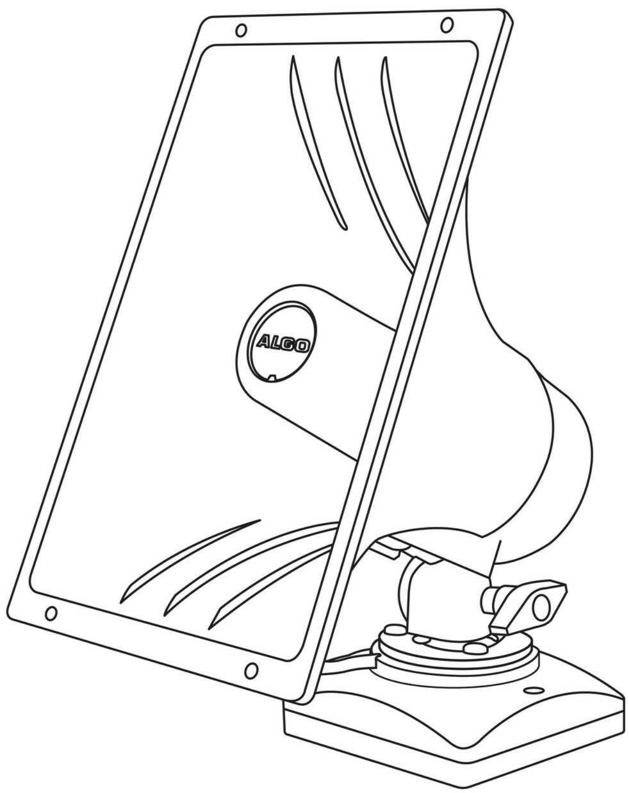
8186 SIP Horn Speaker

FW Version 1.7

User Guide



Order Codes

8186 SIP Horn Speaker

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Important Safety Information

⚠ Important Safety Information

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

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

⚠ Consignes de Sécurité Importantes

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

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

⚠ Información de Seguridad Importante

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

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

⚠ Wichtige Sicherheitsinformationen

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

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

⚠ 安全须知

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

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

⚠ CAUTION

The 8186 Horn Speaker is capable of output levels in excess of 116dB at 1 meter. Ensure nobody is in close proximity to the horn, especially during installation and testing of the product.

⚠ INSTALLATION

The 8186 Horn Speaker should only be installed by a qualified electrician. An improperly installed 8186 could fall from the wall or ceiling and cause serious injury or death.

Local building codes may require one or more additional safety measures, particularly in earthquake prone regions.

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⚠ EMERGENCY COMMUNICATION

If used in an emergency communication application, the 8186 Horn Speaker should be routinely tested. SNMP supervision is recommended for assurance of proper operation. Contact Algo for other methods of operational assurance including the use of the integrated microphone for automated “sound to air” acoustic testing.

⚠ WET OR OUTDOOR ENVIRONMENTS

The 8186 Horn Speaker is intended for indoor or outdoor locations and may be subjected to spray or weather provided the rear wiring cavity is properly sealed to prevent water ingress.

Gaskets included with the 8186 Horn Speaker may be effective against water ingress on some, but not all surfaces in which case additional protective measures must be taken such as a perimeter sealant.

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

Relay input and output connections must not leave the building perimeter without adequate lightning protection.

Overview

Introduction

The 8186 SIP Horn Speaker is a SIP compliant and multicast capable IP speaker suitable for voice paging, loud ringing, and alert/notification applications, particularly wide-area and/or high noise environments (e.g. warehouse, factory). When installed properly, the 8186 can be used for outdoor applications.

An integrated microphone provides talkback capability and ambient noise detection for automatic level control.

Dual SIP extensions provide both voice paging and notification (ring) capability. One or both extensions can be registered with any Communication Server (hosted or enterprise) that supports 3rd party SIP Endpoints. Additional page and ring extensions are also supported, as well as Emergency Alert extensions.

Multiple speakers in a SIP environment require only one speaker to register as a SIP extension. Multicasting capabilities allow the SIP registered speaker to ring/page and simultaneously stream multicast audio to the other speakers. Any number and variety of Algo speakers, paging adapters, and strobes can be configured in a multicast zone.

The 8186 SIP Horn Speaker is configured using central provisioning features or by accessing the web interface using browsers such as Chrome, Firefox, or Edge.

What is Included

- 8186 SIP Horn Speaker
- Mounting bracket
- Gaskets
- Flat head screwdriver
- Getting Started Sheet

What is not Included

- Optional Call Button/Wall Switch (Algo 1202, 1203 or 1204)
- This Installation Guide (www.algosolutions.com/8186/guide)

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Key Features

SIP Extensions

The 8186 connects to an on-premise or hosted communication server in the same way as a SIP telephone. To register the 8186 with the server the following information is required:

1. IP address (e.g. 192.168.1.1) or domain name (e.g. myserver.com) of the SIP Server
2. SIP extension (e.g. 3790)
3. Authentication ID
4. Password

The 8186 supports two SIP extensions which behave differently – **RING** and **PAGE**. One or both may be used depending on the application. If the RING extension is called the 8186 will not answer. Instead, it will play the selected audio file until the ringing stops. Typically the RING extension is programmed as part of a hunt group so that it receives a ring signal simultaneously with one or more phones to function as a loud ringer in noisy or large areas.

If the PAGE extension is called, the 8186 will answer and allow paging over its internal speaker. When the 8186 answers it will play a configurable tone to the caller so they know when they can begin speaking. The same tone is also played over the speaker before the announcement. If Paging to a single 8186, talkback may be enabled using the integrated microphone. The audio direction is determined by the speech activity of the caller.

Loudness

Equipped with a high-efficiency integrated amplifier and tuned high-quality loudspeaker, the 8186 is capable of output levels in excess of 116dB at 1 meter.

Multicasting

Allows multiple units to simultaneously play Ring or Page audio. One 8186 may be configured as a 'Master' device and broadcast an audio stream to any number/combination of Algo IP speaker, paging adapter, or strobe endpoint configured as multicast 'Slaves'. This feature provides scalability without requiring each endpoint Slave to be registered with a SIP extension.

Polycom™ Group Paging

The 8186 support Polycom Group Paging. The 8186 can be added to a Polycom Group Page so that voice paging is heard over Polycom telephone speakers and overhead paging simultaneously.

Ambient Noise Compensation

The 8186's can automatically adjusts loud ring and paging volume to compensate for background ambient noise. If 'Ambient Noise Compensation' is enabled, the alert volume will get louder or quieter by the same dB level as the ambient noise measured just prior to the alert.

Configuration & Provisioning

Configuration can be done through a web interface control panel. Central provisioning may also be used to allow units to be pre-configured for a specific server prior to deployment in the field. Configuration files are automatically downloaded from a server (via TFTP, FTP, HTTP, HTTPS) using DHCP.

Blue Indicator Light

The blue LED by default will be on when the speaker is active. The blue LED will also be on during power up and boot process.

The blue LED can also provide a heartbeat with a flash every 60 seconds to indicate that the speaker is powered and connected to the network.

If the 8186 SIP Horn Speaker is in talkback mode the blue LED will be flashing.

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Setup and Installation

Getting Started - Quick Install & Test

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

1. Connect the 8186 SIP Horn Speaker to an IEEE 802.3af compliant PoE network switch. The blue light will remain on until boot up is completed – about 30 seconds.
2. After the blue LED turns off, connect the reset terminals on the back of the unit to hear the IP address over the speaker. The IP address may also be discovered by downloading the Algo Locator Tool to find Algo devices on your network: www.algosolutions.com/locator
3. Mount the speaker. Summarized instructions are provided in the next section of this sheet.
4. Access the 8186 SIP Horn Speaker web page by entering the IP address into a browser (Chrome, Firefox or Edge) and login using the default password: ***algo***.
5. Enter the IP address or the name for the SIP server into the SIP Domain field under the **Basic Settings > SIP** tab.
6. Enter the Ring and/or Page SIP extension and credentials. Leave the credentials blank for either extension if there is no intended use to have both registered.

   (Note: The speaker supports multiple Ring, Emergency Alert, and Page SIP extensions. The Page extension auto-answers for voice announcements. The Ring and Emergency Alert extensions will play an audio file over the speaker without answering.)
7. Verify the extension is properly registered with the SIP server in the Status tab. Ensure the SIP Registration is “Successful”.
8. Make a test call from a telephone to the speaker for one or all extensions.

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Installation & Mounting

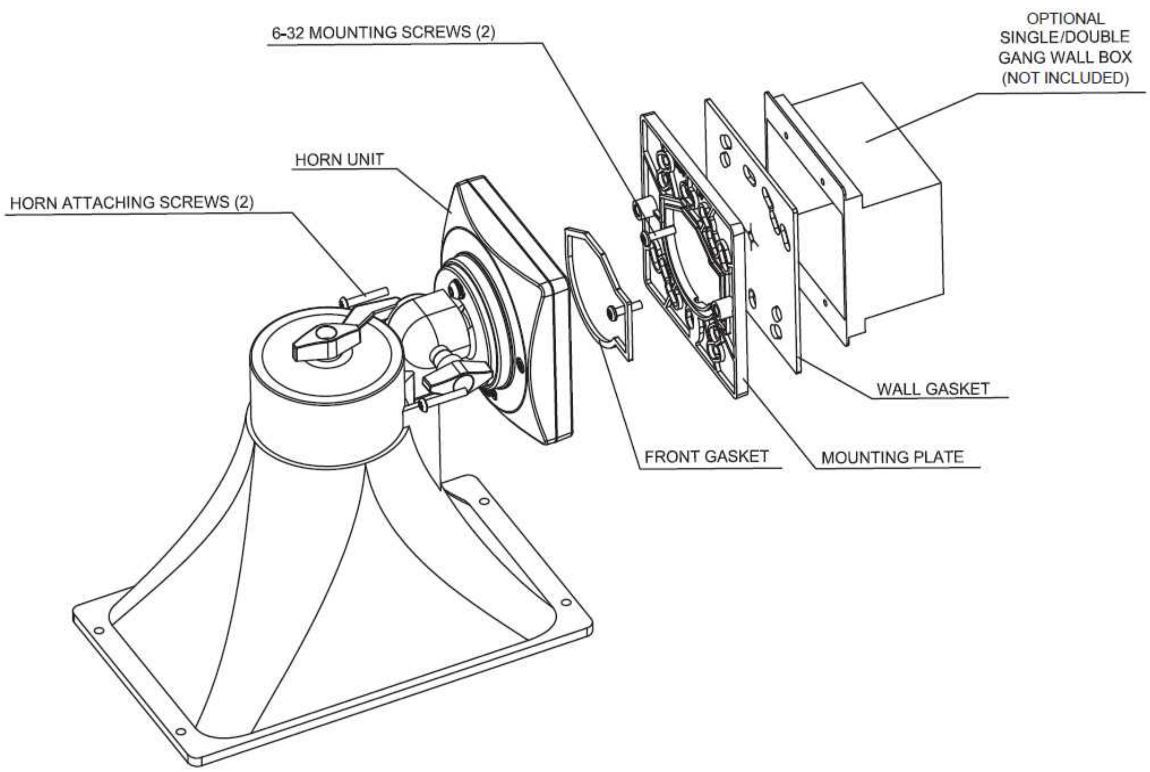
The 8186 SIP Horn Speaker can be wall or ceiling mounted. Concealed wiring may enter from the wall into the wiring cavity. Alternatively, surface wiring may enter through a channel from the bottom edge. The channel is intended for cabling 0.2" or 5mm in diameter and is intentionally snug to protect against moisture ingress.

The 8186 SIP Horn Speaker can be mounted in any orientation but both the bracket and housing must identify TOP. This keeps the bracket wiring channel on the bottom and the RJ45 jack on the top side. Moreover, water needs to be able to drain through the bottom of the mounting plate, so its orientation is extremely important.

The mounting plate may be used to mount over flush or surface mounted electrical boxes or mud rings and fits securely to a 2 gang electrical box (not included) for installation with wiring conduit.

The 8186 SIP Horn Speaker is rated IP65 for wet locations however care must be taken to ensure that water does not enter the wiring cavity. The supplied gaskets or sealant must be used to protect the wiring cavity in wet environments. If sealant is used, ensure the bottom center area of the mounting plate is not obstructed, as water may need to drain out. In dry indoor environments the gaskets are not required. If the wall gasket is used with surface wiring then the gasket should be attached after placing the cable into the wiring channel.

The 8186 SIP Horn Speaker should not be installed beyond a building perimeter without adequately protecting the building wiring from lightning surges.



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Programming and Configuration

After connecting the 8186 to a network PoE, the blue indicator light will turn on during initialization. The 8186 will then attempt to obtain an IP address from the DHCP server. If there is no DHCP server or the attempt was unsuccessful, the 8186 will default to the static IP address **192.168.1.111**.

Note: If you don't have a PoE switch, you'll need a PoE injector that installs between the 8186 and network switch. The PoE injector will supply 48 Vdc to the 8186. Most PoE injectors are capable of providing more power than the 8186 requires (12.95 W). Ensure that the PoE injector is fully compliant to the IEEE 802.3af standard.

After a successful boot up the blue LED will turn off, and the speaker will have obtained an IP address.

Connect the reset terminals on the back of the unit to hear the IP address over the speaker. Connect the reset terminals again to stop playing the IP address over the speaker.

The IP address may also be discovered by downloading the Algo locator tool to find Algo devices on your network: www.algosolutions.com/locator

Enter the IP address (e.g. 192.168.1.111) into a browser such as Chrome, Firefox, or Edge. The web interface should be visible and the default password will be ***algo*** in lower case letters.

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Features

SIP Paging: One 8186

The 8186 SIP 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 speaker with the SIP server, use the **Basic Settings > SIP** tab in the web interface to enter the Communication Server IP address, extension, username, and password. This information will be available from the IT Administrator.

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

Important: once the speaker is using VLAN you will need to be on the same VLAN to access the web interface.

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

There are a number of configurable speaker options:

- 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

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

SIP Ring Event

Set Monitoring Mode to ‘**Monitor Ring**’ and enter credentials. When a call is made to the SIP extension the 8186 will play the selected audio file from memory. Often, the 8186 will be part of a hunt group or ring group to ring in conjunction with a telephone.

Multicast Overview

In addition to the ring and page features, the 8186 is able to send and receive IP audio multicast messages over the network to support larger deployment for both paging and ring/notification. This provides a scalable and efficient method of building large scale notification solutions.

An Algo 8186 can be configured as a Master endpoint. When called from a phone, the SIP registered 8186 auto-answers and plays the page audio over its speaker. Simultaneously, the registered 8186 endpoint broadcasts the audio over the network using RTP multicast to any number/combination of Algo IP speakers, paging adapters, and strobes as required.

The Slave 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. 8180, 8186, 8188, 8128, 8201, 8301, and 8373).

SIP Paging: Multiple 8186s (Using Multicast)

Multicast features in the 8186 SIP Horn Speaker require that only ONE of the speakers be registered as a SIP extension. Additional speakers may be added as multicast Slaves receiving a stream from the SIP registered Master speaker. Please note that any number and combination of Algo speakers, paging adapters and strobes can be part of a multicast.

The Master speaker will page normally while simultaneously streaming audio to the Slave speakers. The Slave speakers do not require SIP extensions and do not need to register with the SIP Communication Server.

To enable multicast streaming from the SIP speaker, go to its web interface and navigate to the **Basic Settings > Multicast** tab**.** Choose multicast mode **'Master/Sender'** and zone **'All Call'**. The multicast addresses pre-populated in the table, under **Advanced Settings > Advanced Multicast** section, will work in most cases and should only be altered for rare cases.

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

The page pre-announce tone is generated from the Master. The following options are valid for each multicast Slave speaker:

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

Talkback can only be used for the SIP registered Master speaker. When paging with talkback enabled, only the area near the Master speaker is covered for talkback. The microphones in the multicast/Slave speakers are disabled except for ambient noise monitoring.

*Note: See "**Basic Setting Tab – Multicast**" section bellow for more configuration options and instructions.*

SIP Paging: Multiple 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 also page telephone speakers in conjunction with overhead speakers.

SIP Activated Notification Alerts

In addition to voice paging, the 8186 can play audio files for emergency, safety, and security announcements, customer service, shift changes, etc.

Audio files can be stored in speaker memory and played over the speaker in response to an event such as a ring or relay input, and also multicast to other Algo SIP endpoints on the network. See **Additional Features > Emergency Alerts** and **Additional Features > Input/Output** for more details.

Background Music Streaming

The 8301 Paging Adapter & Scheduler (sold separately), set as a Multicast master, can stream background music to other Algo slave devices on the network from a music source connected to the 8301’s AUX Input.

When multicasting music, ensure that Automatic Gain Control (AGC) is ‘Disabled’ in **Basic Settings > Features** tab on all the slave devices. Meanwhile, on the Multicast master device, select ‘G.722’ for the ‘Master Output Codec’ setting in **Advanced Settings > Advanced Multicast** tab.

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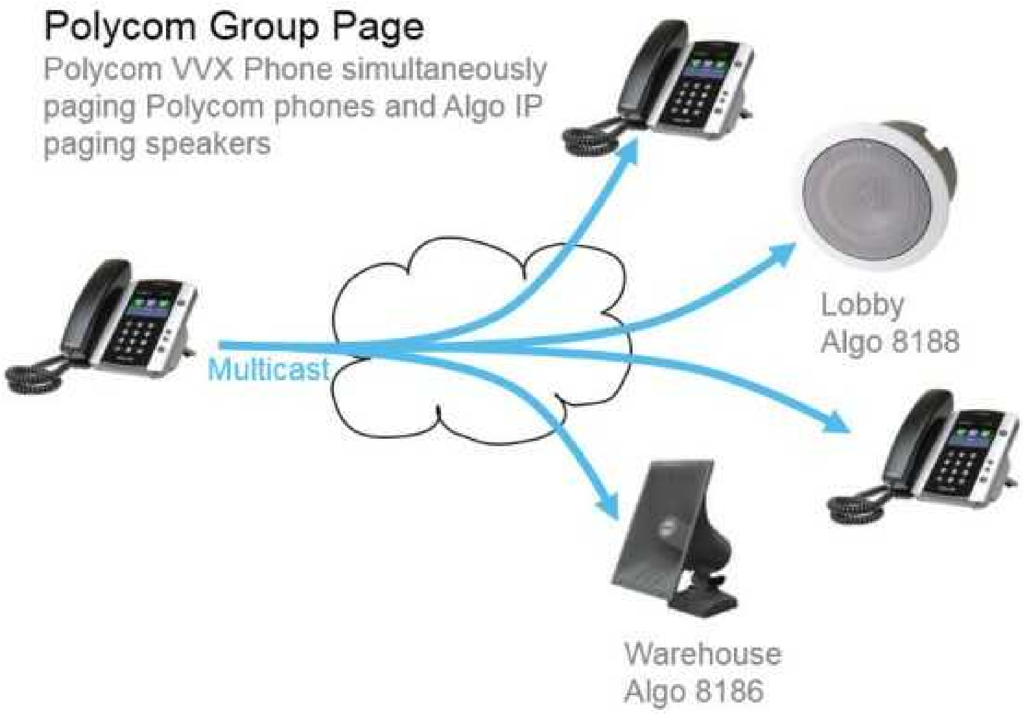
Polycom™ Group Paging



The 8186 SIP Horn Speaker has been designed to support Polycom Group Paging.

The 8186 can be added to a Polycom Group Page so that voice paging is heard over Polycom telephone speakers and overhead paging simultaneously.

The 8186 SIP Horn Speaker may be accessed remotely via SIP and may generate a multicast page within the LAN sending voice page to both Algo paging speakers and Polycom telephones. Audio delay may be added to the 8186 to synchronize with voice page over the Polycom telephone speakers

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TLS for SIP Signaling and Provisioning

Algo devices that support firmware 1.6.4 or later 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-master) secret key.

For further details reference the Algo TLS guide for SIP Signalling and HTTPS Provisioning.

Uploading Public CA Certificates to Algo SIP Endpoints

To install the public CA certificate on the Algo 8186, follow the steps below:

1. Obtain a public certificate from your Certificate Authority.
2. Rename the public certificate 'siptrusted.pem' (only .pem format is supported).
3. In the web interface of the Algo device, navigate to the **Advanced Settings -> File Manager** tab.
4. 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.

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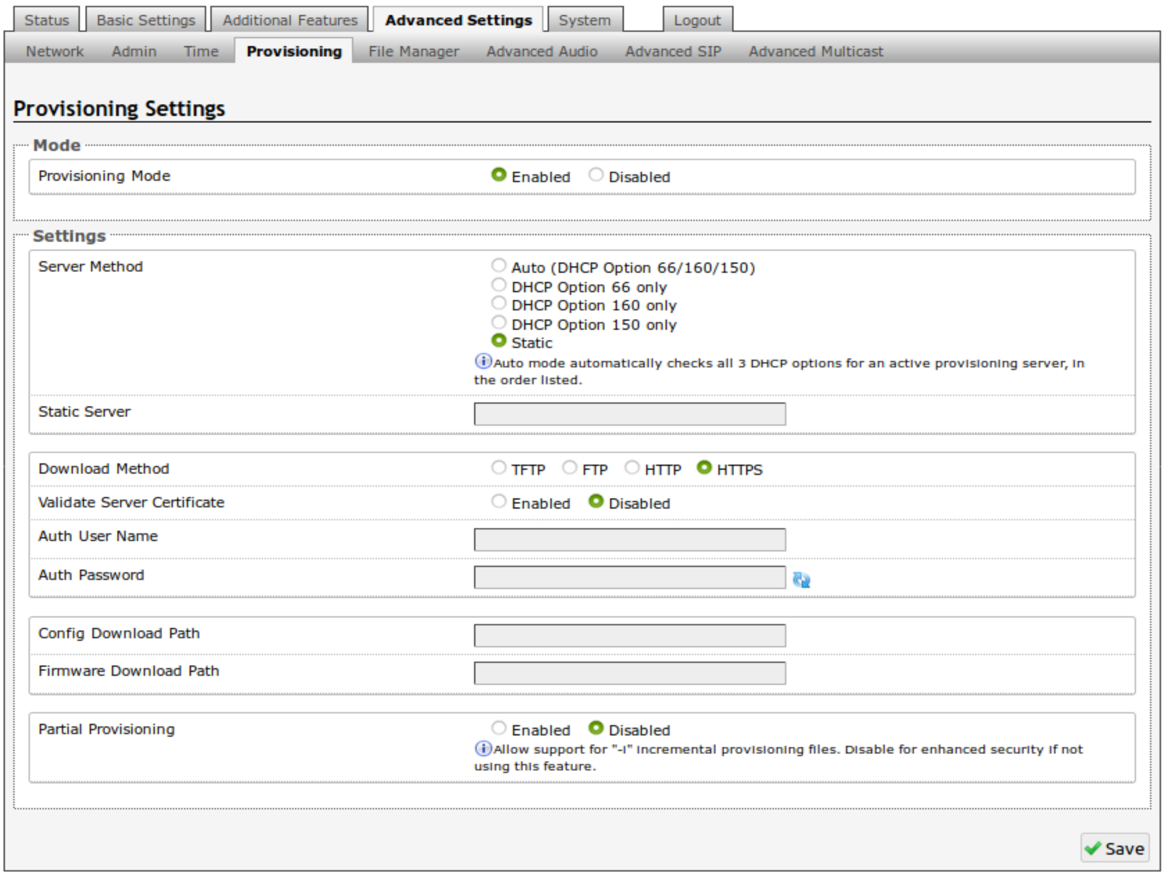
HTTPS Provisioning

Provisioning can be secured by setting the 'Download Method' to 'HTTPS' (under the **Advanced Settings > Provisioning** tab). 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.



⚠	*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
```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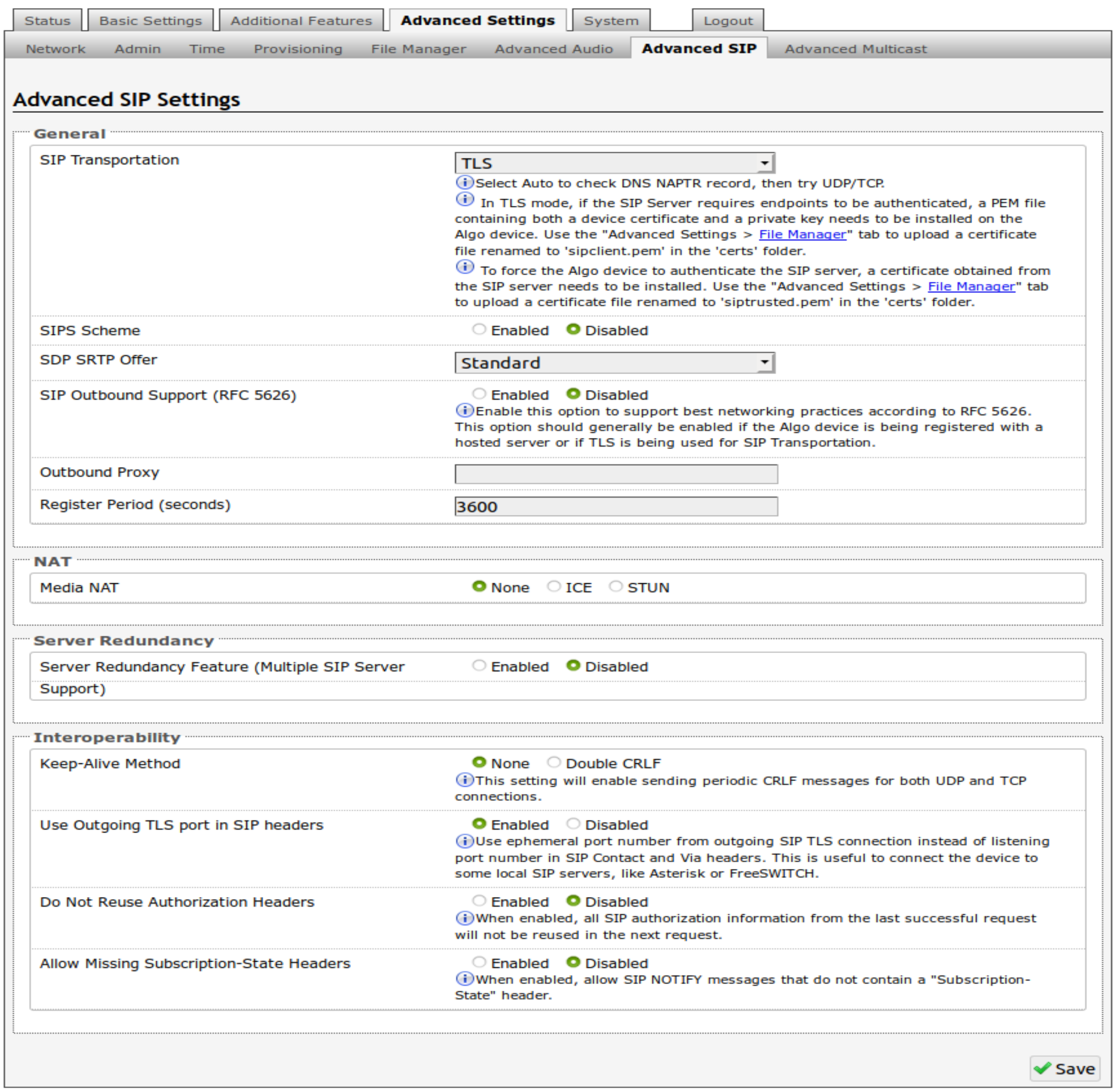
SIP Signaling (and RTP Audio)

SIP signalling is secured by setting ‘SIP Transportation’ to ‘TLS’ (under the **Advanced Settings > 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. In order 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.



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Important: In order for a SIP server to validate the Algo device, an additional certificate has to be manually installed on the 8186. 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'.

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Wiring Connections

Network Connection

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

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

There are two lights on the Ethernet jack:

Green light: On when Ethernet is working, flickers off to indicate activity on the port.

Amber light: Off when successful 100Mbps link is established. Typically on only briefly at power up.

Under normal conditions, the Amber light will turn on immediately after the Ethernet cable is first connected. This indicates that PoE power has been successfully applied. Once the device connects to the network, it will switch to the Green light instead, which will typically flicker indicating traffic on the network.

Connecting Input Devices

The dry contact relay on the 8186 SIP Horn Speaker can be prompted by any normally open, normally closed switch, Algo 1202 Call Button, Algo 1203 Call Switch and Algo 1204 Volume Control Switch. The input switches can be connected to the back of the 8186 via the “IN” terminal.

Connection options can be configured from normally open switch, to normally closed switch, to Algo 1202 Call Button with large blue button, to Algo 1203 single gang backlit Call Switch, to Algo 1204 Volume Control Switch or as an EOL resistor termination. The connection options can then be configured to complete an ‘Action’ when Relay Input is triggered.

Note: See “Additional Features Tab – Input/Output” section of this user guide for additional information on input device configuration

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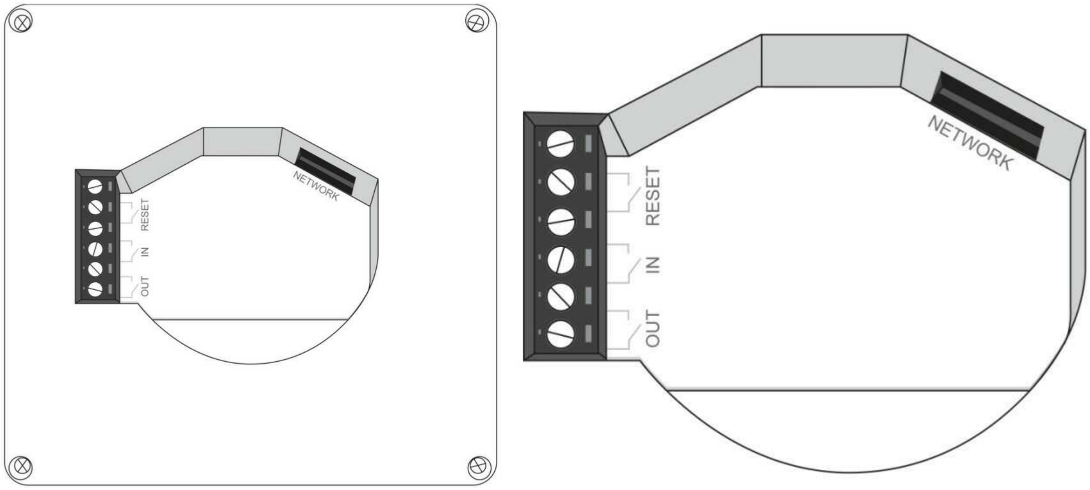
Inputs/Outputs

On the back, the 8186 SIP Horn Speaker has a relay output, relay input and terminal block reset.



Terminal Block Relay In

By default, these terminals are inactive. Connection options are a normally closed switch, normally open switch, 1202 Call Button, 1203 Call Switch, 1204 Volume Control Switch or EOL resistor termination.

Terminal Block Relay Out

By default these terminals provide a normally open contact closure when the 8186 SIP Horn is active.

Reset

Terminal Block Reset

The reset relay terminal on the back can be used to reset the 8186 SIP Horn Speaker only at time of power up. To return all the settings to the factory default for the 8186, reboot or power cycle the 8186. Wait until the LED flashes, then connect the reset terminals and hold until the 8186 LED begins a double flash pattern. Release the reset connection and allow the unit to complete its boot process.

Do not short the reset terminals until the LED begins flashing.

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

Once booting has completed, shorting the reset terminals will cause the device to speak its IP address.

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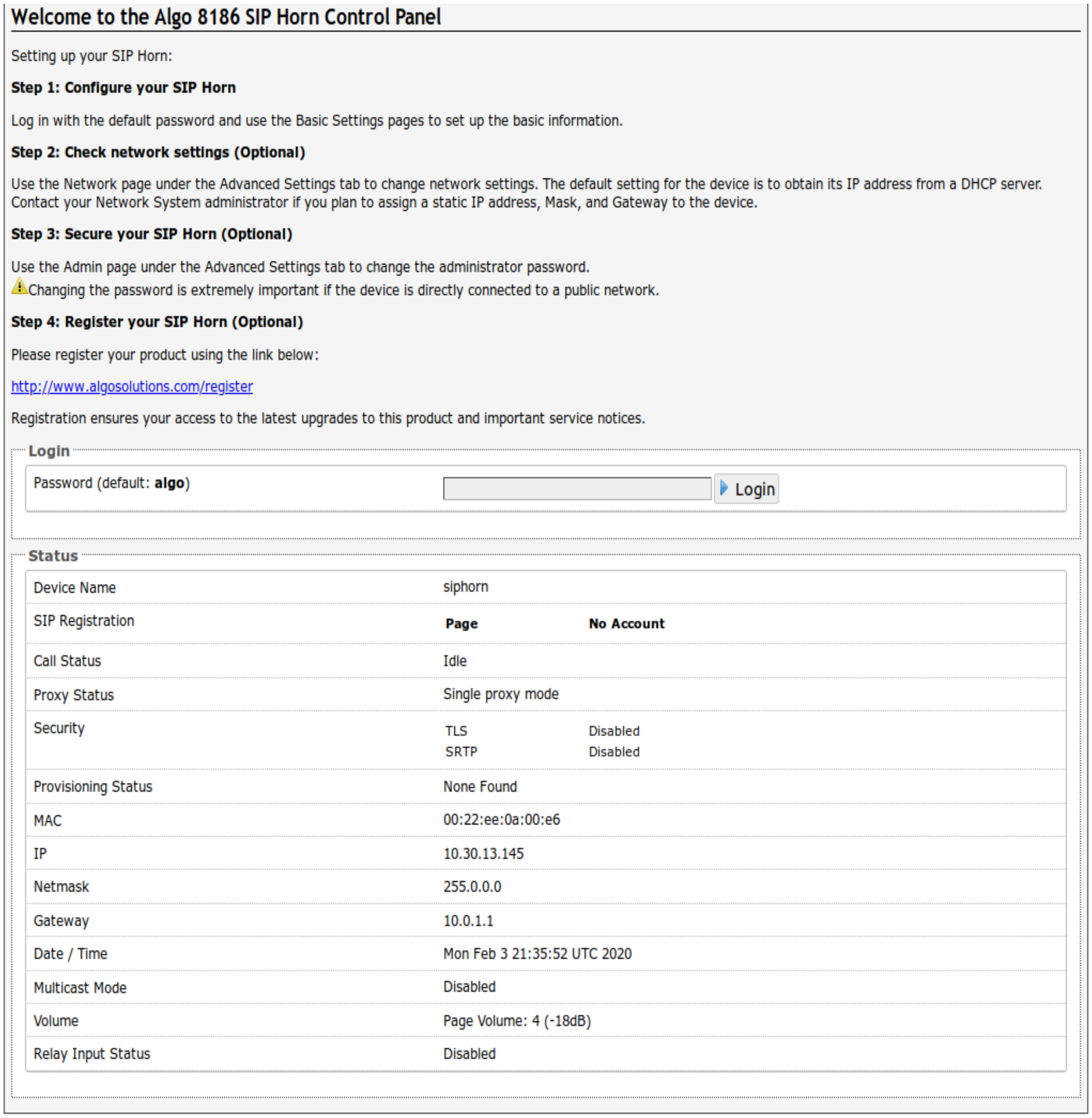
Web Interface Status and Login

Web Interface Login

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



Web Interface is accessed by entering 8186's IP Address into the web browse.

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

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Status

The device's Status page will be available before and after log on. The section can be used to check 8186's SIP Registration status of the Ring/Page extensions, Call Status, Multicast Mode (Slave/Master), Relay Input Status, Proxy Status, and general MAC, IP, Netmask, Date/Time, and Timezone information.

> *The Status page can be hidden when logged out for security purposes under the* ***Advanced Settings > Admin*** *tab.*

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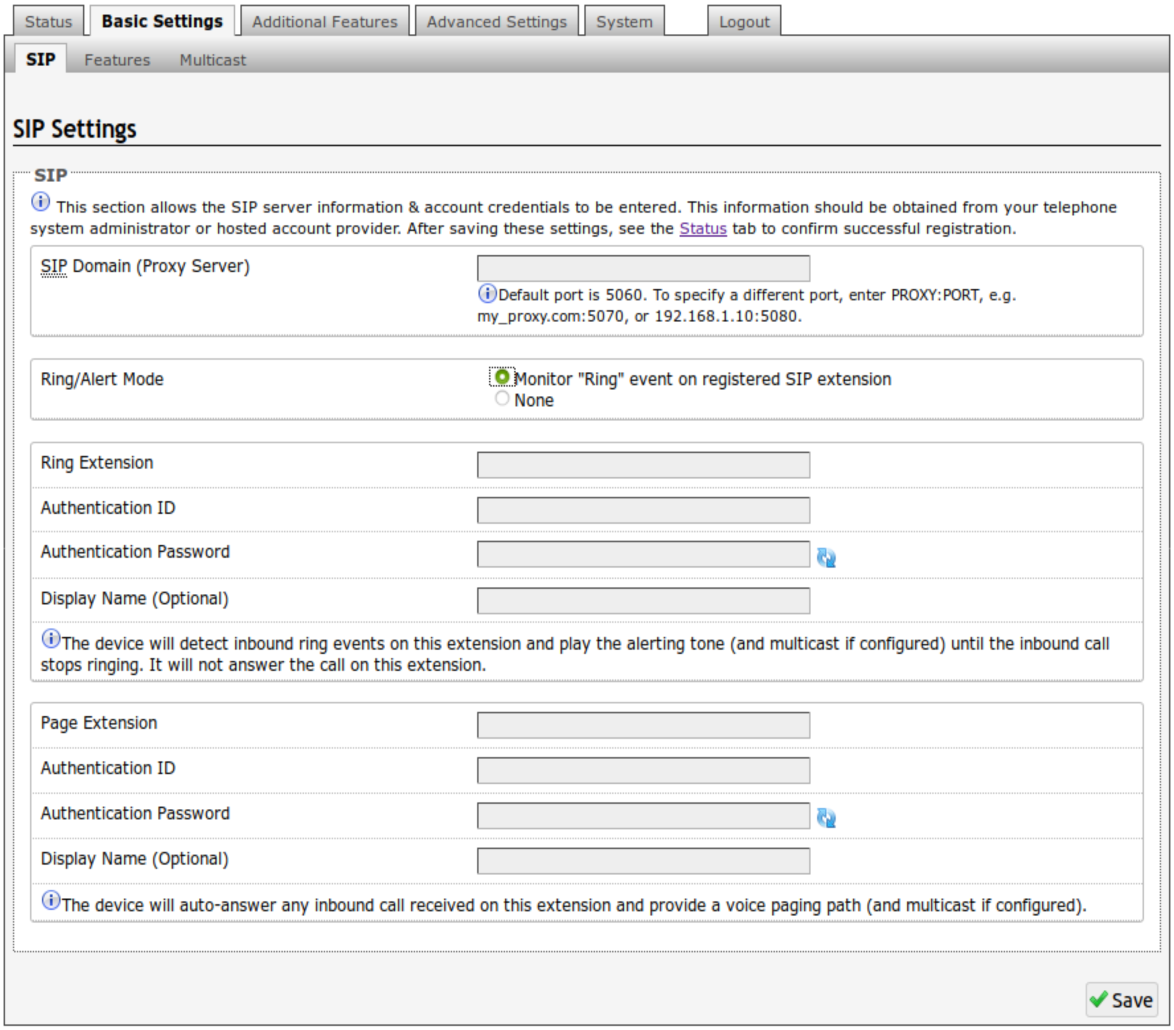
Web Interface Basic Settings

Basic Settings Tab – SIP



SIP Server information and Credentials should be obtained from your telephone system administrator or hosted account provider. After saving the settings, see the Status page to confirm that the registration was successful.

*Important: Any time 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. 192.168.1.1) or domain name (e.g. myserver.com) of the SIP Server.

Ring/Alert Mode

Option for adding a second SIP extension for Ring event. If activated, screen expands to enter SIP extension parameters for a Ring/Alert Extension.

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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 8186 speaker's Ring parameter. The device will detect inbound ring events on this extension and play the alerting tone (and multicast if required) until the inbound call stops ringing. It will not answer the call on this extension.

Page Extension

This is the SIP extension for the 8186 speaker. The device will auto-answer any inbound call received on this extension and provide a voice paging path (and multicast if configured).

Authentication ID

May also be called Username for some SIP servers and in some cases may be the same as the SIP extension used for the associated Ring and/or Page parameter.

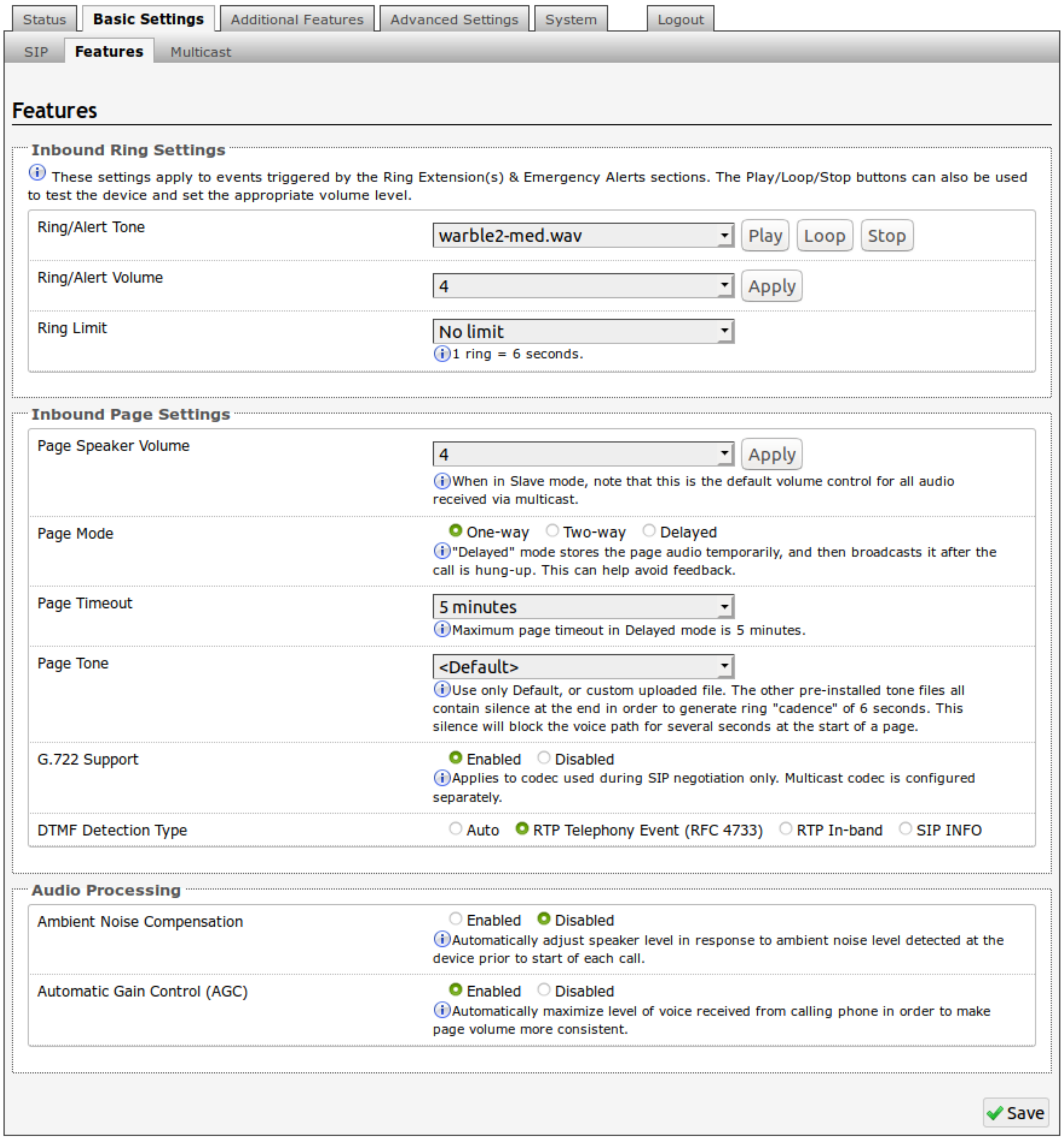
Authentication Password

SIP password provided by the system administrator for the SIP account used for the associated Ring and/or Page parameter.

Display Name

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.

Basic Settings Tab – Features



Ring/Alert Tone

Select an audio file to play when a ring event is detected on the SIP Ring extension. The file may be played immediately to the 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 Master's selected ring tone.

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

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Ring/Alert Volume

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

> *Caution: The 8186 SIP Horn Speaker is capable of output levels in excess of 116dB at 1 meter. Ensure nobody is in close proximity to the horn, especially during installation and testing of the product.*

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.

Page Speaker Volume

Speaker page volume control for SIP or multicast paging. This setting is an amplifier gain control and output level will depend on streaming level. This setting will apply to all inbound multicast (slave mode), regardless of content.

Page Mode

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

In delay mode, press "*" to cancel a page while the recording state is in process to prevent it from being sent after hanging up.

Page Timeout

A time limit may be set for an active page.

Page Tone

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

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

G.722 Support

Enable or disable the G.722 codec.

DTMF Detection Type

Select the preferred DTMF detection method.

Ambient Noise Compensation

To configure, set the volume to an appropriate level for a quiet environment and enable the Ambient Noise Compensation. The integrated microphone will measure the ambient noise during idle periods and automatically increment the speaker volume, if any increase in background noise is detected. Ambient noise level is averaged over 10 seconds. The noise compensation will not be functional when playing background music.

Automatic Gain Control (AGC)

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

Basic Settings Tab – Multicast

Multicast IP Addresses

Each 8186 SIP Horn Speaker has its own IP address, and shares a common multicast IP and port number (multicast zone) for multicast packets. The master speaker transmits to a configurable multicast zone, and the slave units listen to all the multicast zones assigned to them.

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

1. Multicast IP addresses range: 224.0.0.0/4 (from 224.0.0.0 to 239.255.255.255)
2. Port numbers range: 1 to 65535
3. By default, the 8186 SIP Horn Speaker 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 8186 SIP Horn Speaker supports nine “basic” multicast zones. These zones are defined by the multicast IP addresses.

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

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

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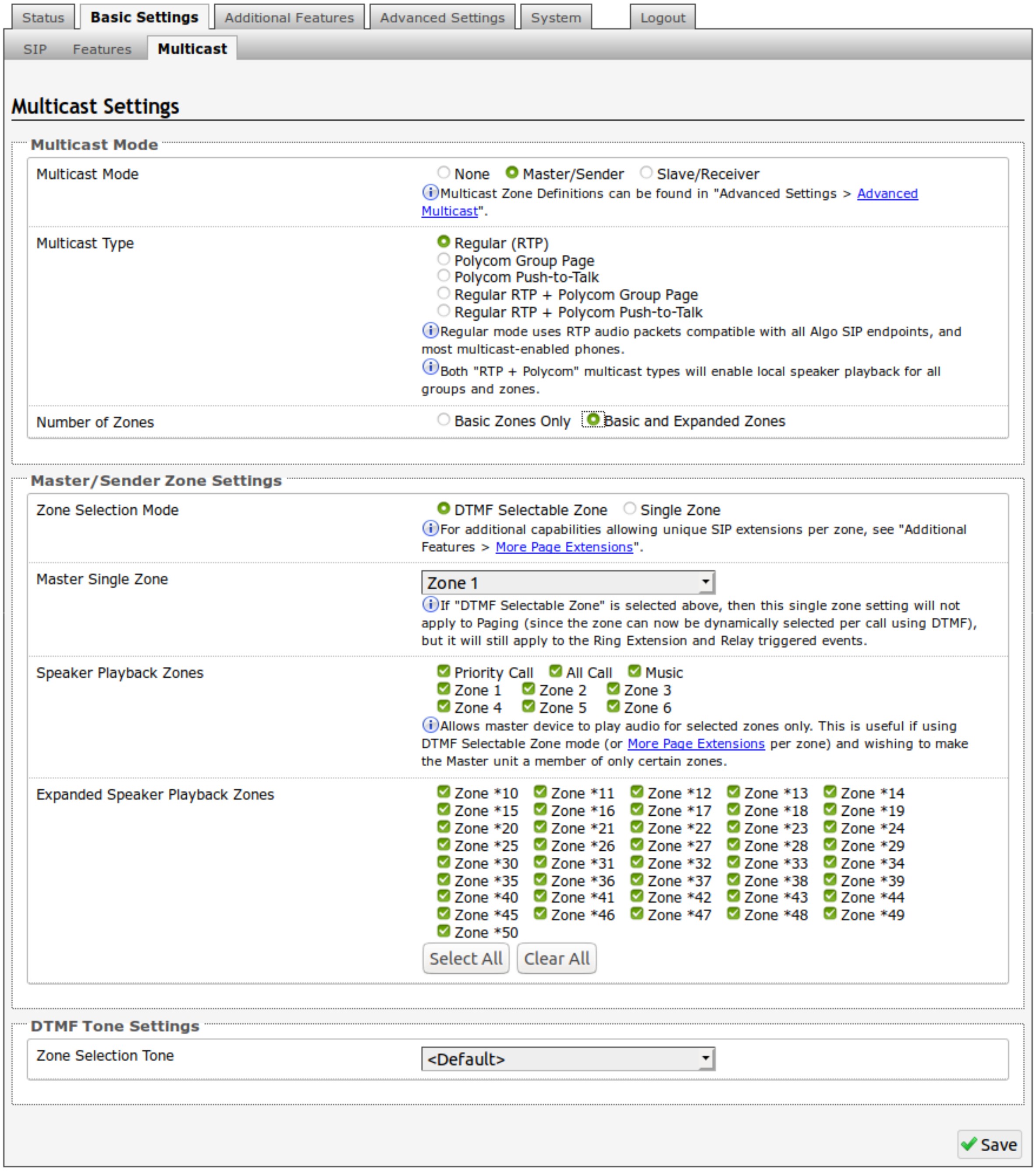
"Expanded" zones can also be enabled, in the **Basic Settings > Multicast tab**, allowing up to 50 zones in total. These have the same behaviors as the basic zones, but are hidden by default to simplify the interface.

Basic Settings Tab – Multicast (Master Settings)



Note: See (Advanced Settings > Advanced Multicast) section for more information on populated IP values below:

Multicast Mode (Master/Sender Selected)

If master is enabled the 8186 will broadcast an IP stream when activated in addition to playing the audio over its own speaker. (Note that the 8186 cannot be both a multicast Master and Slave simultaneously).

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 Slave zones, but are hidden by default to simplify the interface.

Multicast Type

The 8186 SIP Horn Speaker 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 8186 with “Polycom Zone” (IP Address and Port) and “Polycom Default Channel”. *Always ensure that the multicast settings on all Slaves match those of the Master.*

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

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.

> *Note: DTMF Codes for groups 10 and higher start with an “*”.*

Zone Selection Mode

‘Single Zone’ always broadcasts on one IP address. ‘DTMF Selectable Zone’ mode, offers dynamic zone selection and requires only the master device to have a registered SIP Extension. The zone definitions can be found in the **Advanced Settings > Advanced Multicast** tab.

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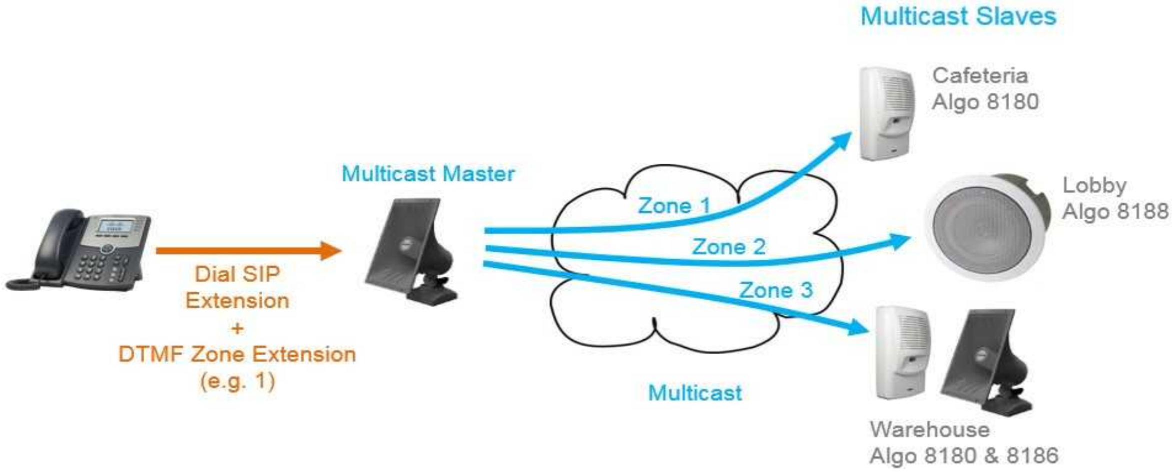
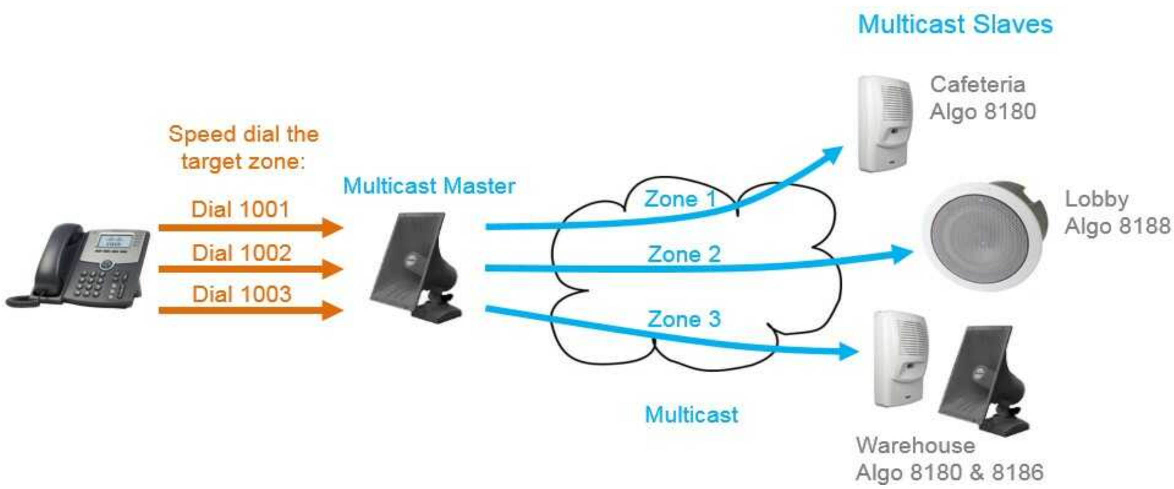


In 'DTMF Selectable Mode', to page, dial the SIP extension of the master device: ####, then dial the desired DTMF page zone (e.g. 1, 2, etc.) on the keypad when prompted.

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 ""*

Alternatively, multiple SIP extensions can be registered on the Master device. Each extension is mapped to a unique zone, allowing zones to be called directly (for instance from speed-dial keys) without the use of DTMF. See **Additional Features > More Page Extensions** tab.



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.

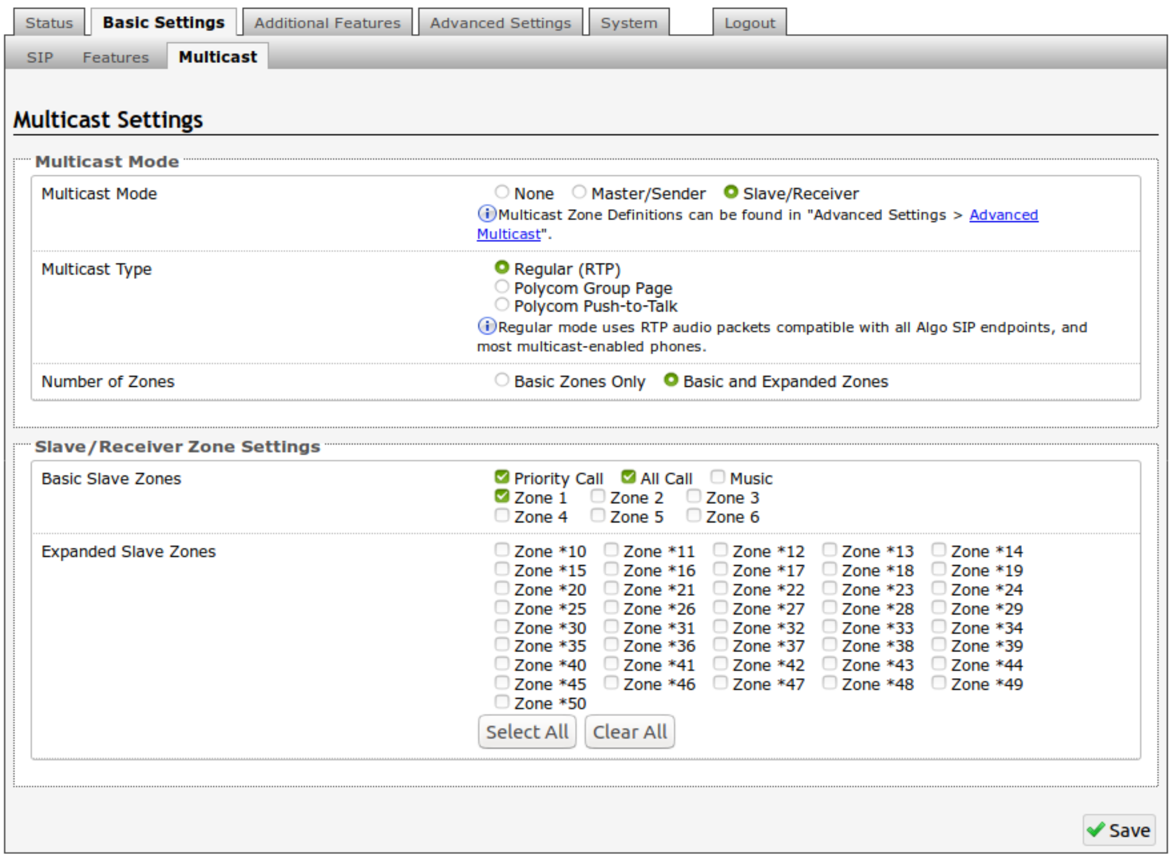
Master Single Zone

The zone that multicast stream will be sent to. 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 ‘Master Single Zone’ setting is still used for any multicast events triggered by the Ring, analog input, or the relay input.

Speaker Playback Zones

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

Basic Settings Tab – Multicast (Slave Settings)



Multicast Mode (Slave Selected)

If Slave is enabled the 8186 will activate when receiving a multicast message. Will mimic audio stream, but use local volume settings (‘Page Speaker Volume’ in Basic Settings > Features).

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 behavior as the basic Slave zones, but are hidden by default to simplify the interface.

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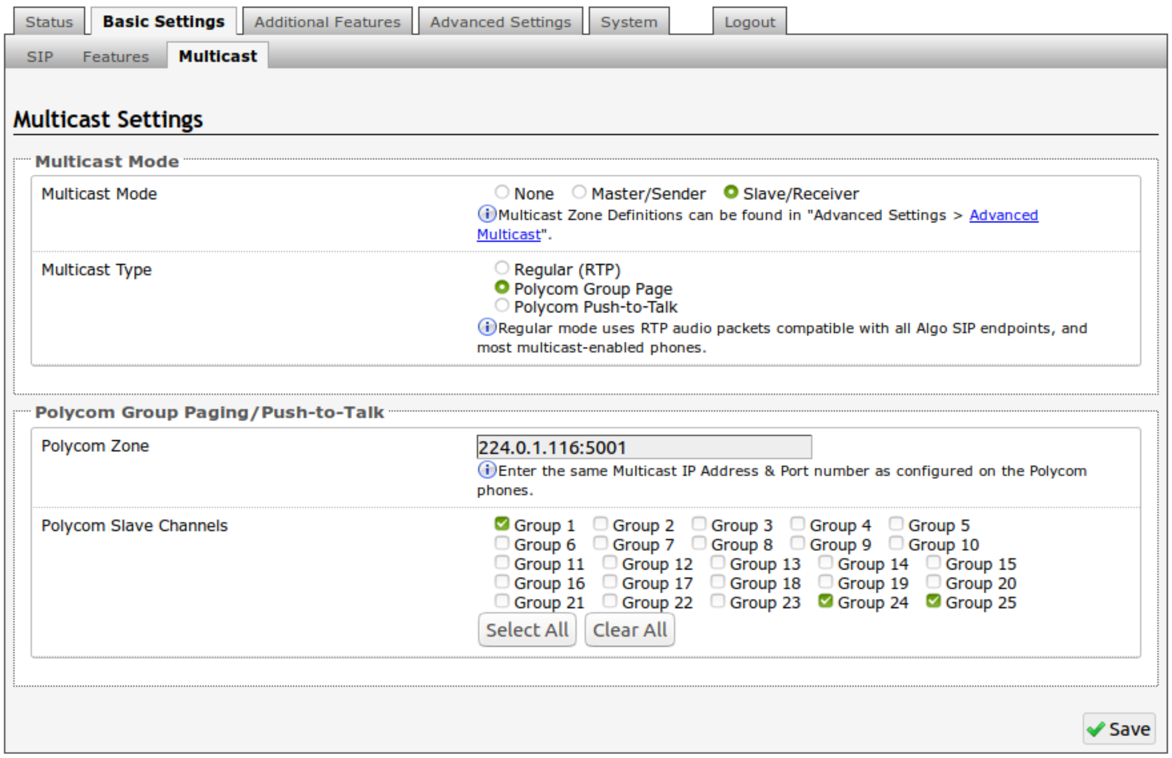
Multicast Type - Regular

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

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

The 8186 SIP Horn Speaker may receive multicast paging compatible with Polycom ***“on premise group paging”*** protocol.

To configure the 8186 as a slave 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.



The Polycom phone used as page audio source for the 8186(s), 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 master, 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 master, a tone does not have to be set as the Algo master will provide its own tone. Polycom Group Tones can be set in Advanced Settings > Advanced Multicast tab.

Slave Zones

Select one or more multicast zones for the 8186 to monitor. Note that multicast zone priority is based on the zone definition list order (top to bottom).

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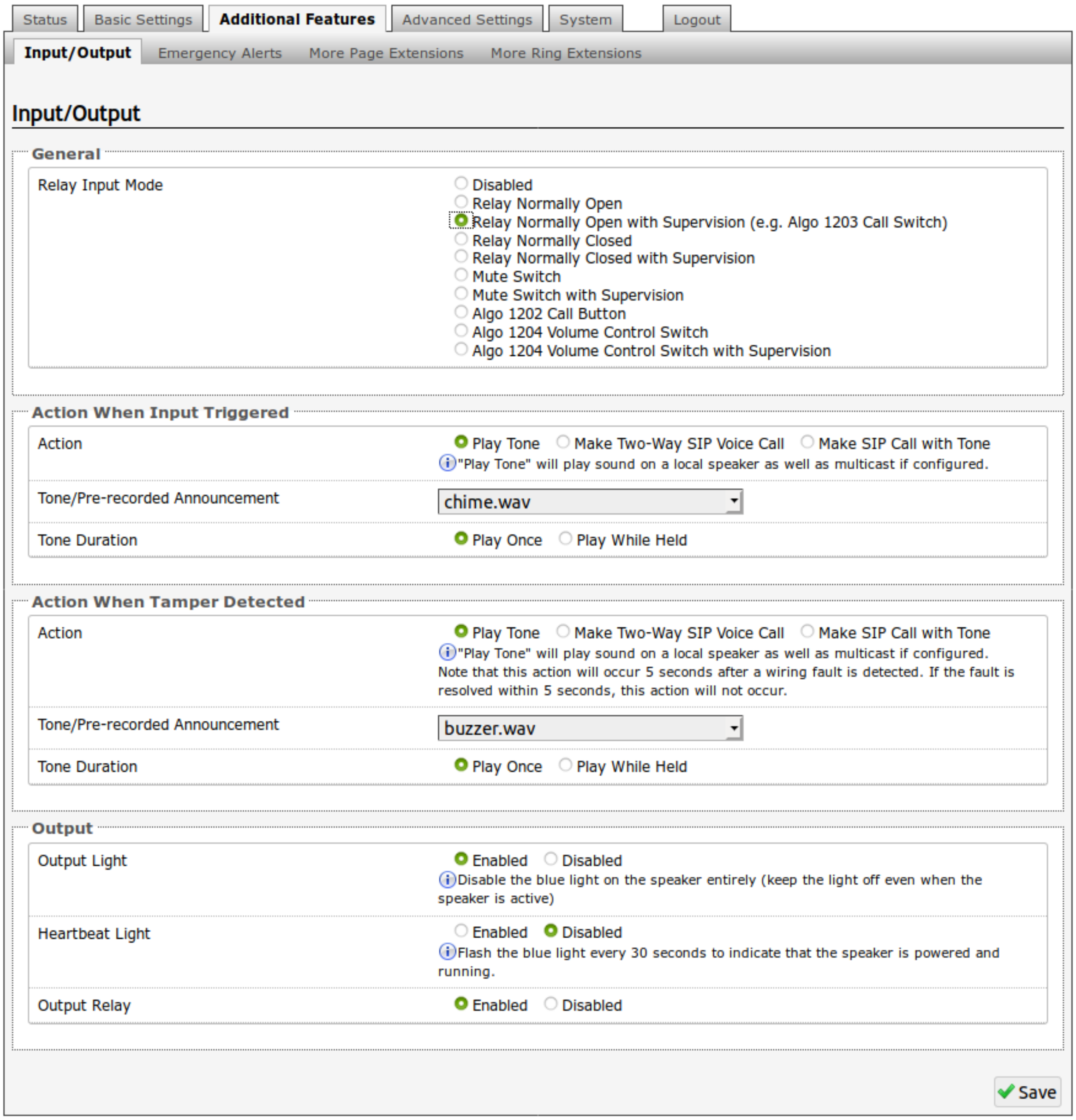
Web Interface Additional Features

Additional Features Tab – Input/Output



When triggered by an input relay, the 8186 SIP Horn Speaker can perform actions such as playing a pre-recorded announcement over the speaker(s), sending the announcement as a private message to a phone, or initiating a two-way conversation between the speaker and a phone.

Relay Input Mode

The input relay to the 8186(s) can be prompted by any normally open or normally closed switch. Algo offers the 1202 Call Button, the 1203 Call Switch, or the 1204 Volume Control

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Switch with supervision. 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 8186, the 1203 can prompt a single action with one-touch, or a continuous action if the button is held.

Mute Switch

Apply an external switch (short-circuit) across the Relay Input terminals 5 & 6 in order to mute the speaker. 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.

1202 Call Button

The 1202 Call Button is a one-touch button for event notification and response. It can be used with the 8186 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.

While the 8186 can be configured to play the audio file only once, it can also be enabled to play it continuously with just one touch on 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 Switch

The 1204 Volume Control Switch is a simple 2 terminal potentiometer that will allow attenuation below the max volume level (configured under ‘Basic Settings > Features’)

Algo’s 1204 can be used for variable volume control. The maximum volume should still be set in the Basic Settings > Features tab as usual, and then the Volume Control Switch will allow attenuation below this level. Enabling Priority Multicast Override allows priority multicast to override the volume set by the Volume Control Switch. Enabling ‘Mute On Lowest Setting’ allows audio to be completely muted when volume control switch is turned all the way down.

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Action – Play Tone

When the 8186 input is triggered, a tone or a pre-recorded audio file will play over the local speaker, or multicast if enabled. This function can be used to call 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

Action – Make Two-Way SIP Voice Call

When the 8186 input is triggered, a voice path will open for an intercom-like call via the 8186 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
    - Allow 2nd Button Press
- Outbound SIP Call Settings:
    - Outbound Ring Limit
    - Ringback Tone
    - Maximum Call Duration

Action – Make SIP Call with Tone

When the 8186 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 Tone (seconds)
    - Maximum Tone Duration
- Outbound SIP Call Settings:
    - Outbound Ring Limit

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 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

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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.

Extension to Dial

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

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

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.

Allow 2nd Button Press

If enabled, 2nd button press will either simply End Call or End and Restart Call. Therefore, if an input is triggered for the second time (since the first input trigger enables one of the four actions listed above) the SIP call will either simply be terminated or terminated and immediately called again.

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.

Ring back 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.

Maximum Call Duration

Select the maximum call length. The call will be terminated once the maximum time is reached. In the event that a call inadvertently reaches voicemail or gets accidentally left on hold, this setting ensures that the 8186 returns on-hook.

Output Light

Enable/Disable the blue light on the speaker entirely (keep the light off even when the speaker is active).

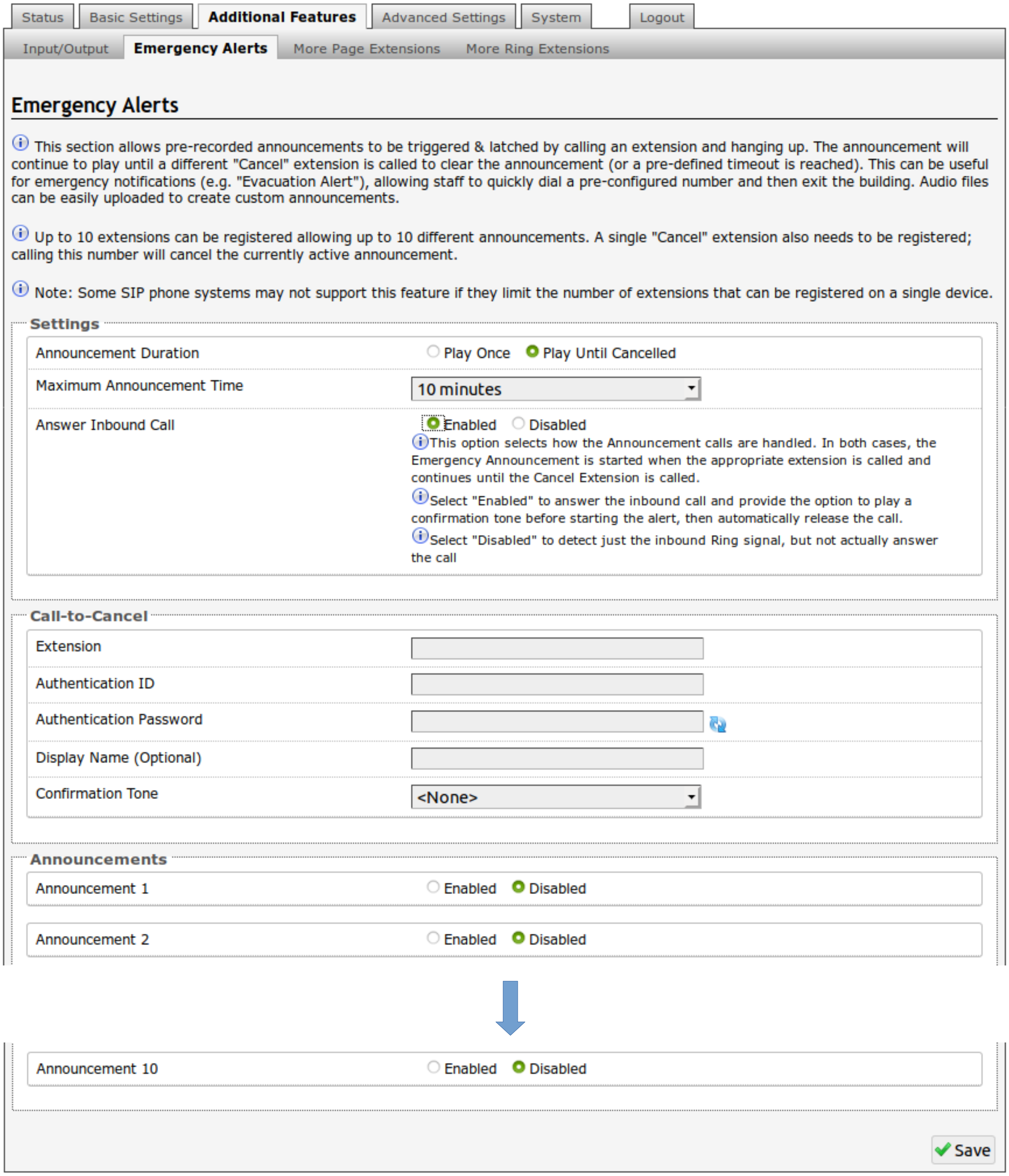
Heartbeat Light

If enabled, the small blue indicator will flash every 30 seconds as visual confirmation that the 8186 is powered and running.

Output Relay

Enable or disable the output relay. Please note this is a normally open relay only.

Additional Features Tab – Emergency Alerts



Emergency Alerts allow for an announcement to be triggered & latched by calling a pre-configured Emergency extension and hanging up. The announcement can be chosen to play once or to play until cancel. If “Play Until Cancelled” is selected, announcement will continue to play until the "Call-to-Cancel" extension is called to clear the announcement (or a defined timeout is reached). The 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.

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If the “Answer Inbound Call” option is “Enabled” the call is auto-answered and a 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 extensions (including this device), the auto-answer may intercept that call and prevent it from ringing on other devices.

Up to 10 extensions can be registered allowing up to 10 different announcements. Audio files can also be easily uploaded to create custom announcements.

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.

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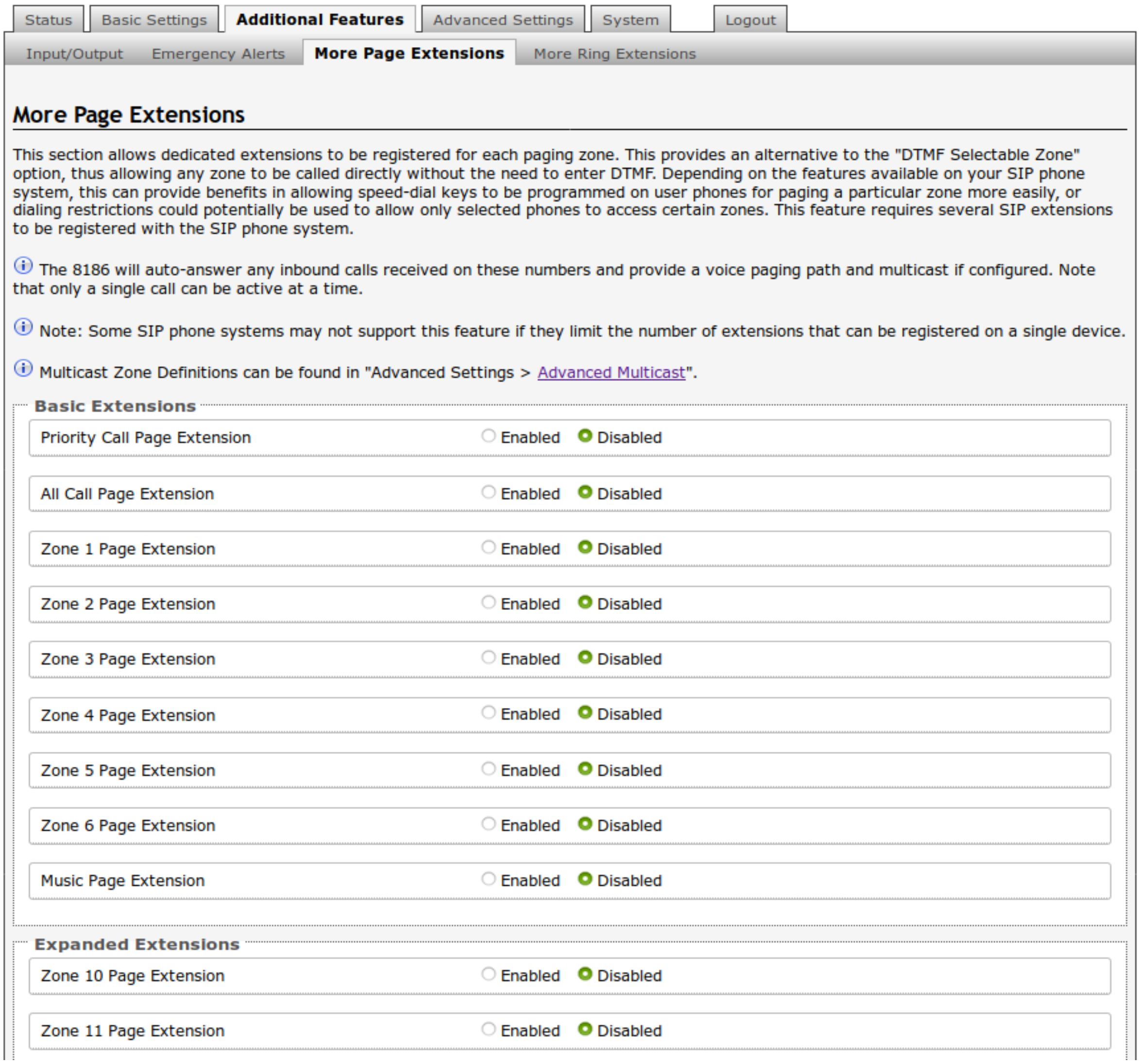
Additional Features Tab – More Page Extensions



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) click “Enable” beside the target extension and enter the Extension, Authentication ID, and Authentication password.

The 8186 will auto-answer any inbound calls received on these numbers and provide a voice paging path and multicast if configured. Note that only a single call can be active at a time.

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.

Multicast Zone Definitions can be found in Advanced Settings > Advanced Multicast.

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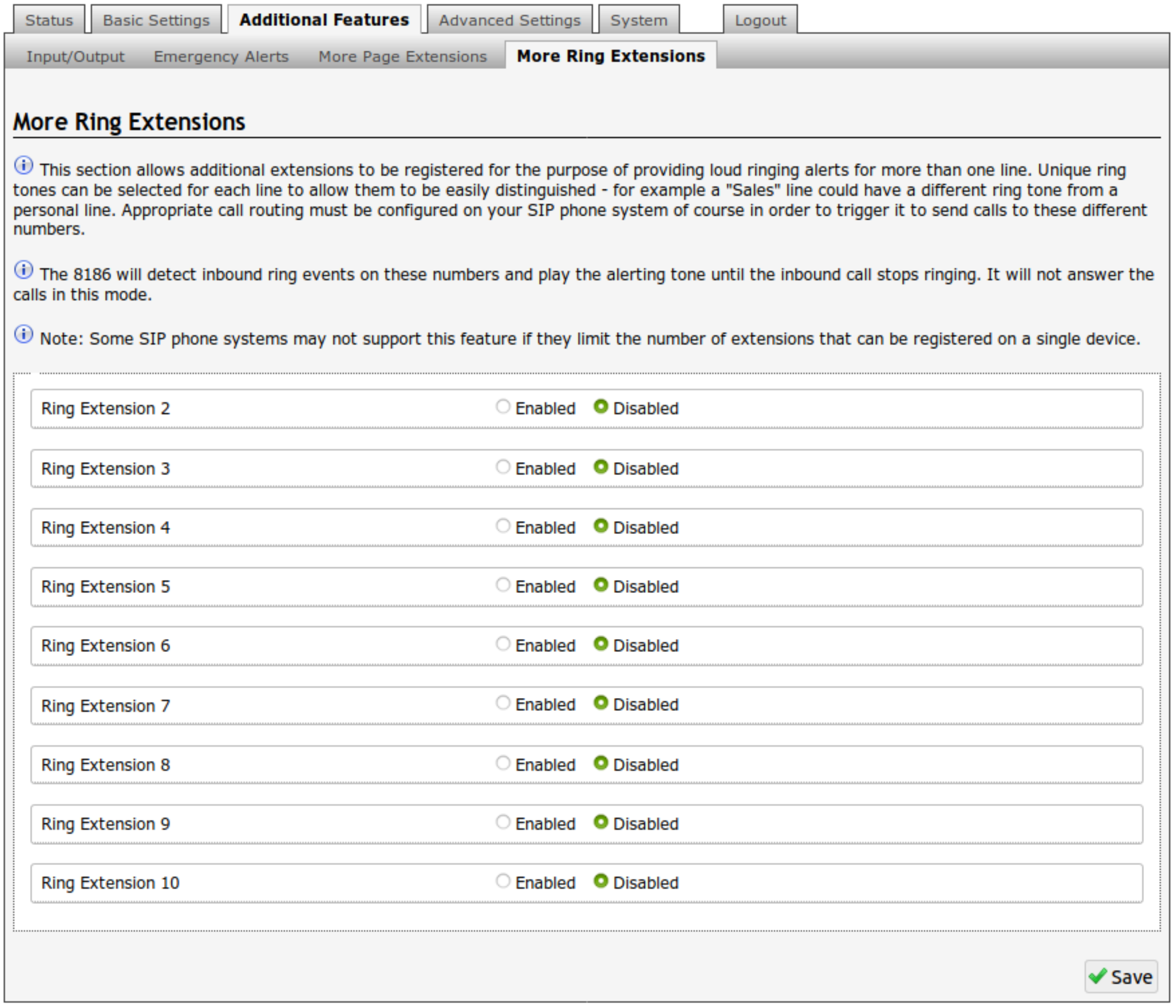
Additional Features Tab – More Ring Extensions



Up to 10 SIP Ring extensions can be registered. To configure additional ring extensions, click “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.

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.

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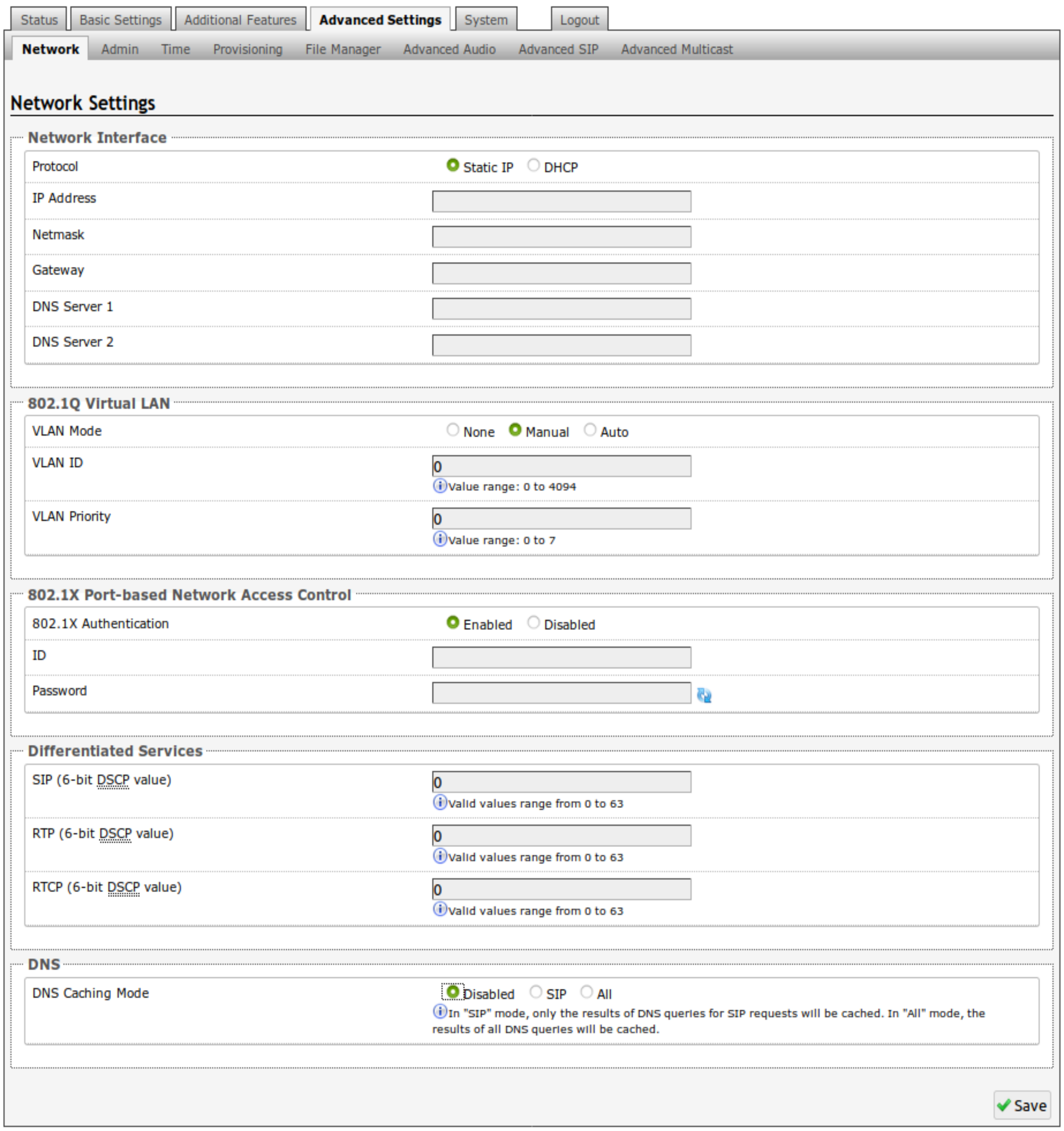
Web Interface Advanced Settings

Advanced Settings Tab – Network



Protocol

DHCP is an IP standard designed to make administration of IP addresses simpler. When selected, DHCP will automatically configure IP addresses for each 8186 on the network. Alternatively, the 8186 can be set to a static IP address.

VLAN Mode

Enables or disables VLAN Tagging. VLAN Tagging is the networking standard that supports Virtual LANs (VLANs) on an Ethernet network. The standard defines a system of

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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 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.

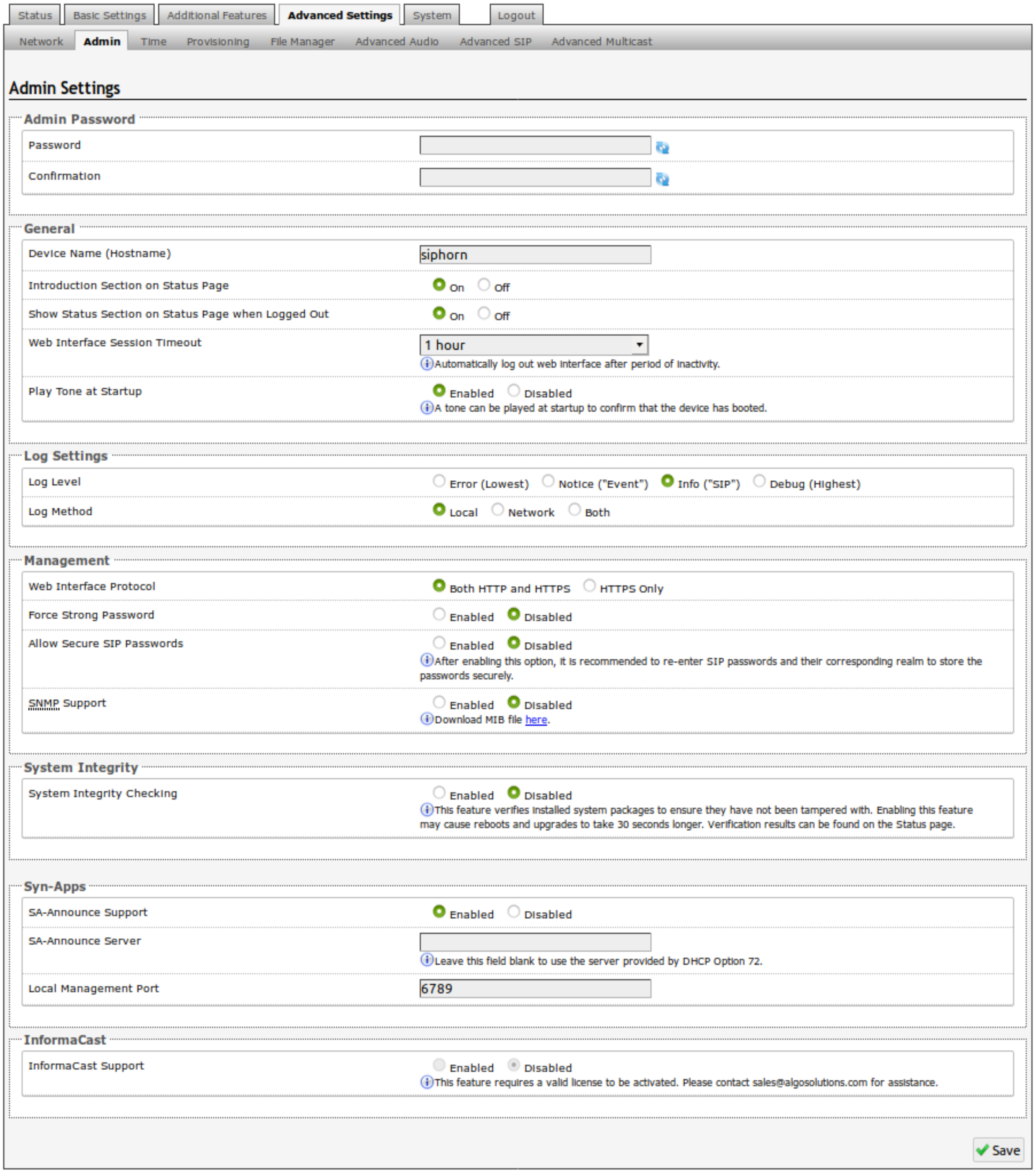
Differentiated Services (6-bit DSCP value)

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

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.

Advanced Settings Tab – Admin



Password

Password to log into the 8186 SIP Horn Speaker web interface. You should change the default password ***algo*** in order to secure the device on the network. If you have forgotten your password, you will need to perform a reset using the Reset Button in order to restore the password (as well as all other settings) back to the original factory default conditions.

For additional password security see “Force Strong Password” below.

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Confirmation

Re-enter network admin password.

Device Name (Hostname)

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

Introduction Section on Status Page

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

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.

Web Interface Session Timeout

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

Play Tone at Startup

A tone can be played at startup to confirm that the device has booted.

Log Level

Use on the advice of Algo technical support only.

Log Method

Allows the 8186 SIP Horn Speaker to write to external Syslog server if the option for external (or both) is selected.

Log Server

If external (or both) is selected this is the address of the Syslog server on the network.

Web Interface Protocol

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

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 re-entered in the Basic Settings > SIP tab, and any other locations where SIP extension 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.

SNMP Support

Additional SNMP support is anticipated for future, but the 8186 SIP Horn Speaker will respond to a simple status query for automated supervision. Contact Algo technical support for more information.

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 page.

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 8186 when Multicast Master Mode is disabled (set to ‘None’) in the Basic Settings > Multicast tab.

SA-Announce Server

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

Local Management Port

Enter the local management port for the SA-Announce Server.

InformaCast Support

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

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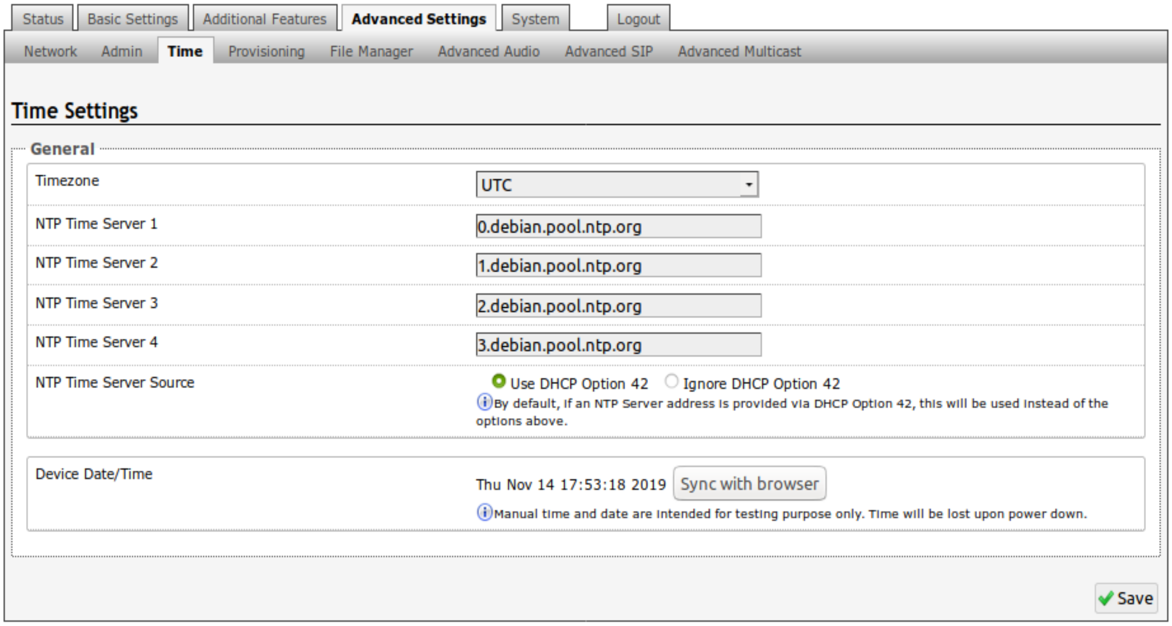
Advanced Settings Tab – Time



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

Time Zone

Select a time zone.

NTP Time Servers 1/2/3/4

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

NTP Time Server Source

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 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 only for logging purposes and are not typically required.

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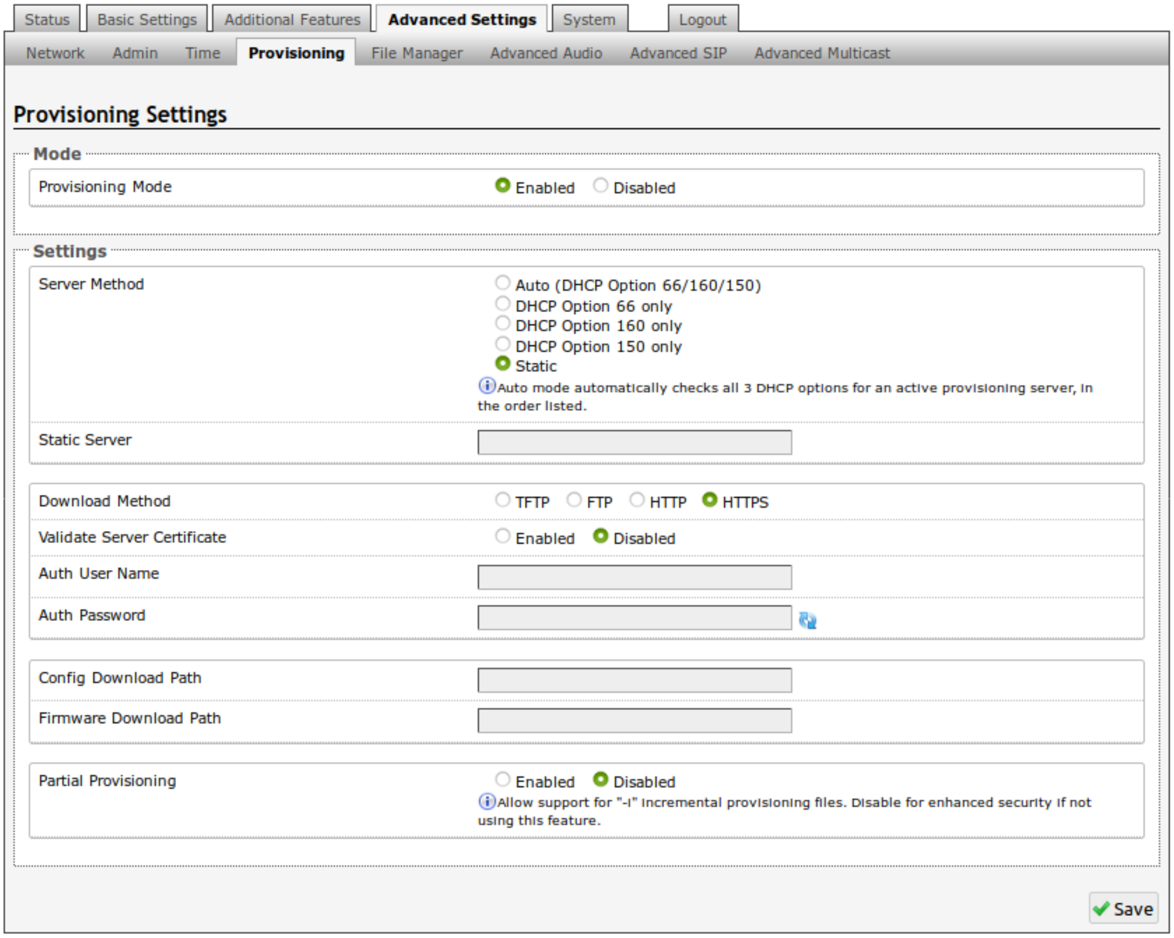
Advanced Settings Tab – Provisioning



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.

Provisioning allows installers to pre-configure the 8186 SIP Horn Speaker units prior to installation on a network. It is typically used for large deployments to save time and ensure consistent setups.

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.

For example, the 8186 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.

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

Generic (for all Algo 8186 Speakers)	**algop8186.conf**
Specific (for a specific MAC address)	**algom[MAC].conf**

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

MD5 Checksum

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

A tool as such 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

1. Connect the 8186 to the network
2. Access the 8186 Web Interface Control Panel
3. Configure the 8186 with desired options
4. Click on the System tab and then Maintenance.
5. Click "Download" to download the current configuration file
6. Save the file settings.txt
7. Rename file settings.txt to algop8186.conf
8. File algop8186.conf can now be uploaded onto the Provisioning server

If using a generic configuration file, extensions and credentials have to be entered manually once the 8186 has automatically downloaded the configuration file.

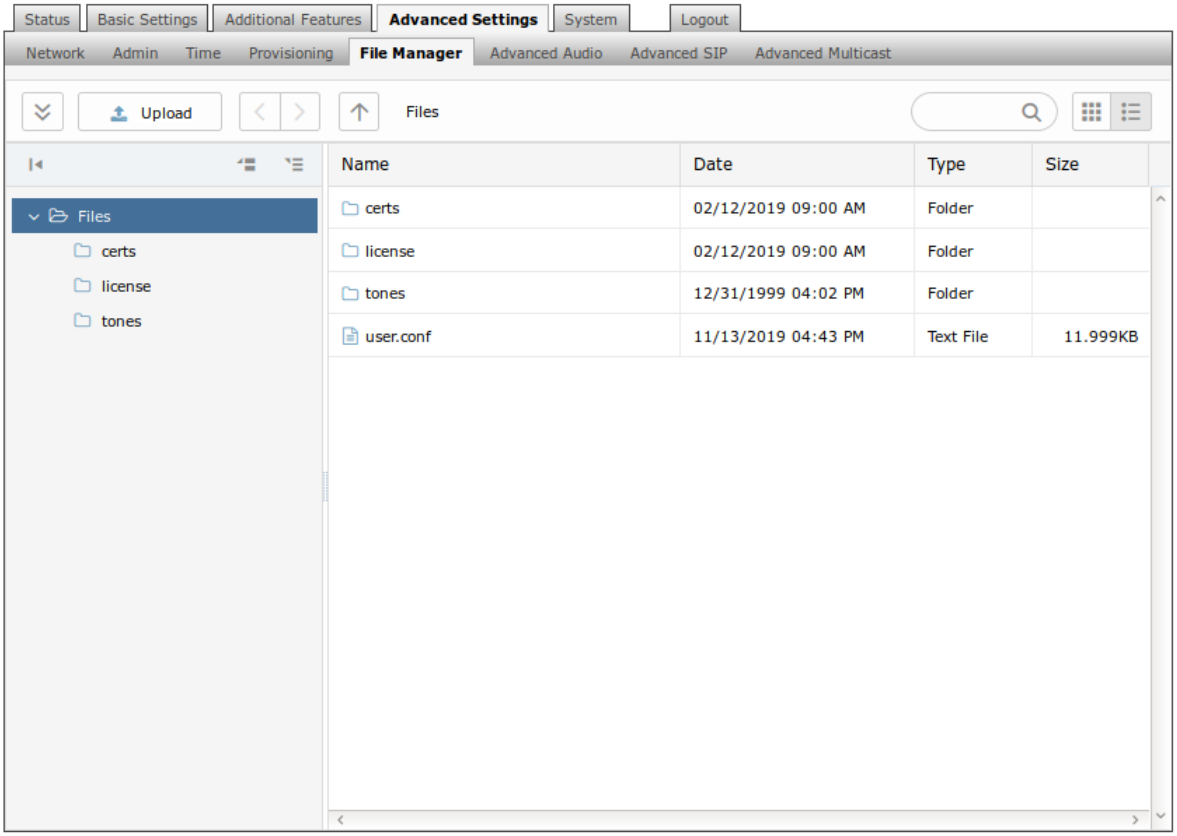
Generating a specific configuration file

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

The specific configuration file will only be downloaded by the 8186 with the MAC address specified in the configuration file name. Since all the necessary settings can be included in this file, the 8186 will be ready to work immediately after the configuration file is downloaded. The MAC address of each 8186 speaker can be found on the back label of the unit.

For more Algo SIP endpoint provisioning information, see:
www.algosolutions.com/provision

Advanced Settings Tab – File Manager



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 by downloading the original by 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
- 8kHz or 16kHz sampling rate
- 16-bit PCM, or u-law
- Mono
- Smaller than 200MB

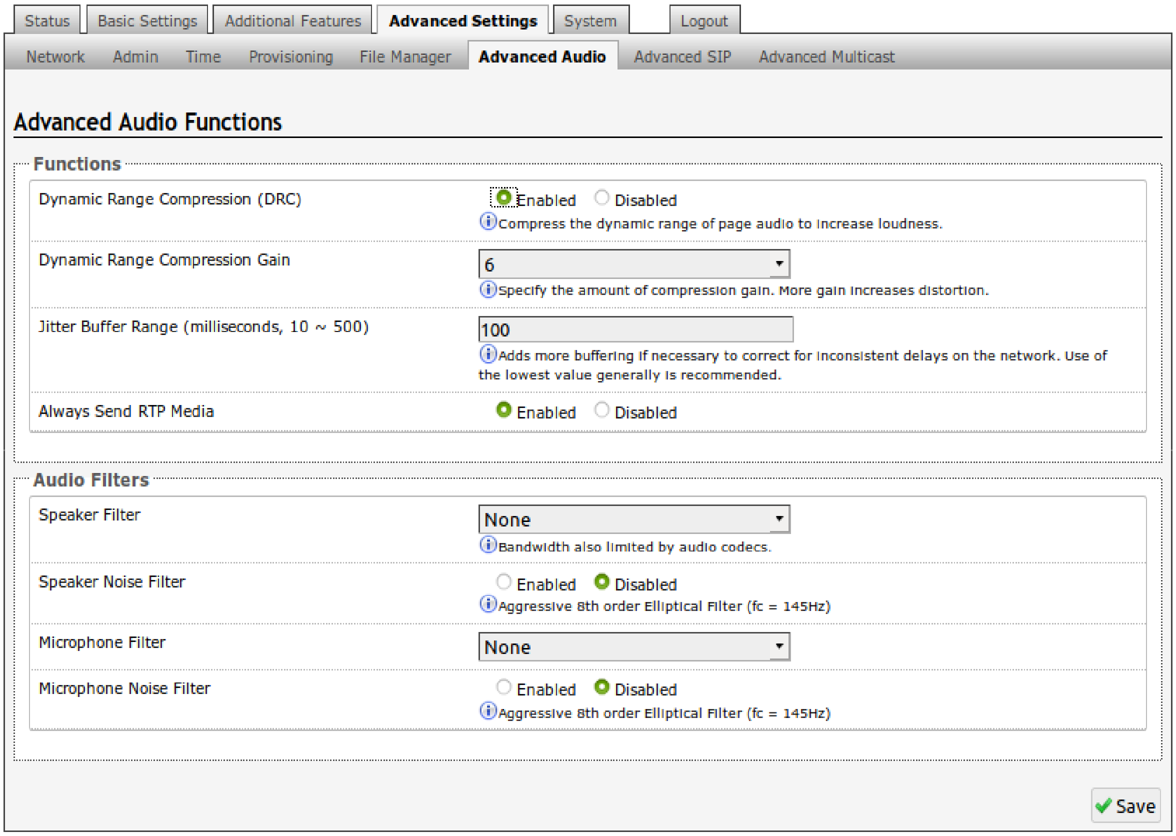
File names must be limited to 32 characters, with no spaces.

For further instructions reference the Custom Tone Conversion and Upload Guide.

Tone Files Included in Memory

The 8186 SIP Horn Speaker includes several pre-loaded audio files that can be selected to play for various events. The web interface allows selection of the audio file and also the ability to play the file immediately over the speaker for testing. Files may also be deleted or renamed.

Advanced Settings Tab – Advanced Audio



Dynamic Range Compression (DRC)

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

Dynamic Range Compression Gain

'Dynamic Range Compression' must be enabled to display this setting. 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.

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 145Hz 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 145Hz to reduce mains induced noise (fans).

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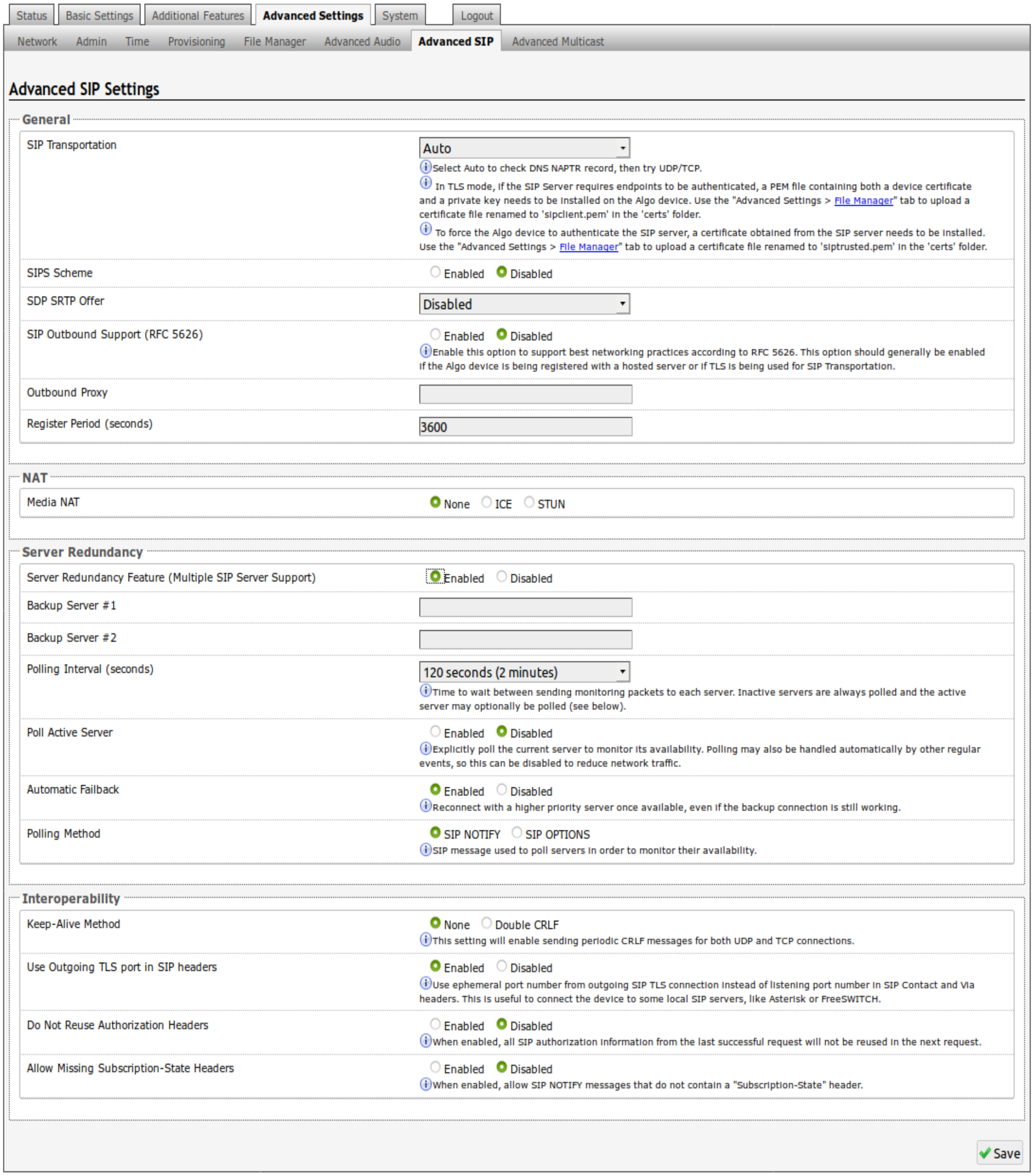
Advanced Settings Tab – Advanced SIP



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.

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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’, makes RTP voice data encryption mandatory, 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.

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 8186 SIP Horn Speaker will re-register with the SIP server. Default setting is 3600 seconds (1 hour). Only change if instructed otherwise.

Media NAT

IP address for STUN server if present or IP address/credentials for a TURN server.

Server Redundancy Feature

Two secondary SIP servers may be configured. The 8186 SIP Horn Speaker will attempt to register with the primary server but switch to a secondary server if necessary. The configuration allows re-registration to the primary server upon availability or to stay with a server until unresponsive.

Backup Server #1

Only visible if ‘Server Redundancy Feature’ is enabled. If primary server is unreachable the 8186 SIP Horn Speaker will attempt to register with the backup servers. If enabled, the 8186 will always attempt to register with the highest priority server.

Backup Server #2

Only visible if ‘Server Redundancy Feature’ is enabled. If backup server #1 is unreachable the 8186 SIP Horn Speaker will attempt to register with the 2nd backup server. If enabled, the 8186 will always attempt to register with the highest priority server.

Polling Intervals (seconds)

Only visible if ‘Server Redundancy Feature’ is enabled. Time period between sending monitoring packets to each server. Non-active servers are always polled, and active server may optionally be polled (see below).

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Poll Active Server

Only visible if ‘Server Redundancy Feature’ is enabled. Explicitly poll current server to monitor availability. May also be handled automatically by other regular events, so it can be disabled to reduce network traffic.

Automatic Failback

Only visible if ‘Server Redundancy Feature’ is enabled. Reconnect with higher priority server once available, even if backup connection is still fine.

Polling Method

Only visible if ‘Server Redundancy Feature’ is enabled. SIP message used to poll servers to monitor availability.

Keep-alive Method

If Double CRLF is selected the 8186 will periodically send a CRLF message for both UDP and TCP connections to maintain connection with the SIP Server.

Keep-alive Interval

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

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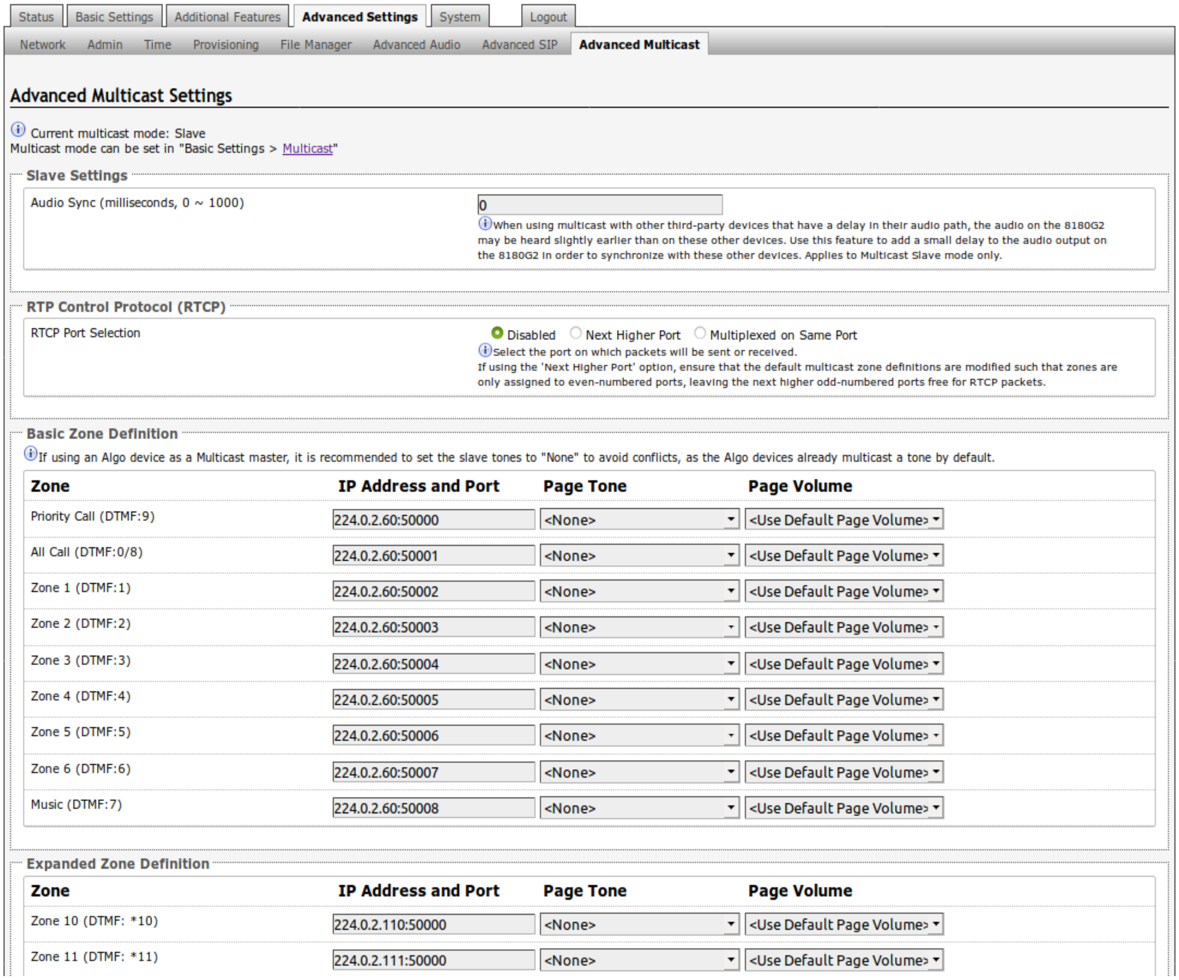
Advanced Settings Tab – Advanced Multicast



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

Audio Sync (Slave Mode)

When using multicast with other third-party devices that have a delay in their audio path, the audio on the 8186 may be heard slightly earlier than on these other devices. By adding audio delay up to one second, the 8186 may be synchronized with other speakers or telephones that have greater latency. This feature applies to Multicast Slave mode only.

Master Output Codec (Master Mode)

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

Master Output Packetization Time (Master Mode)

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

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.

Zone Definition

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

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

Page Tone and Page Volume

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

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

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

Polycom Slave Tones

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

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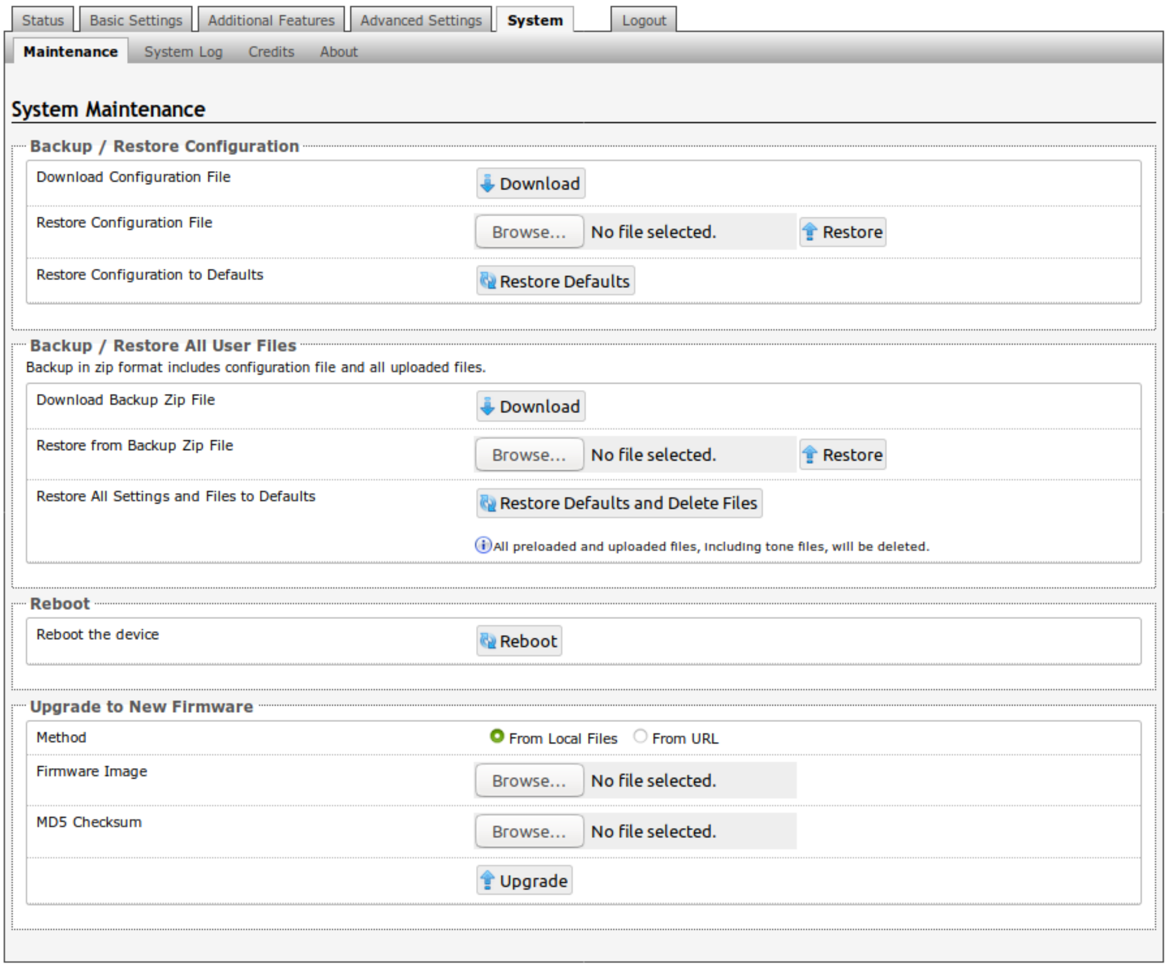
Web Interface System

System Tab – Maintenance



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 8186 SIP Horn Speaker device settings to factory default values.

Download Backup 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 the Device

Reboots the device.

Method

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

Firmware Image

Point to the firmware image provided by Algo.

MD5 Checksum

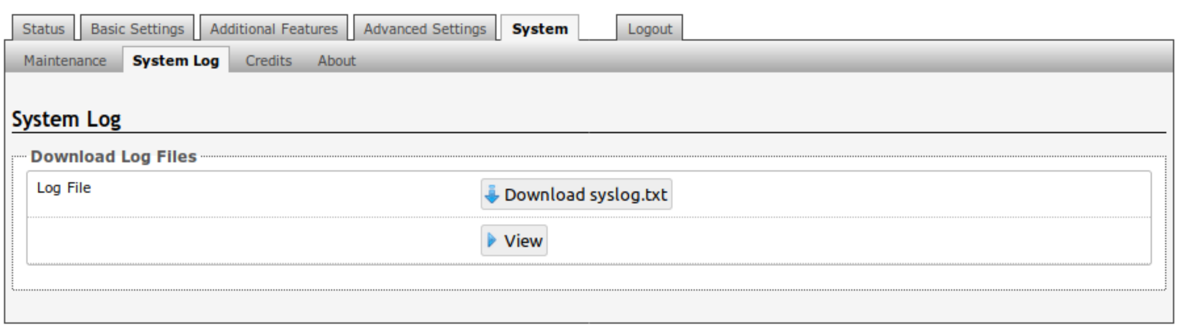
Point to the checksum file provided by Algo.

How To Upgrade 8186 SIP Horn Speaker Firmware

1. From the top menu, click on System, then Maintenance.
2. In the Upgrade section, click on Choose File and select the 8186 speaker firmware file to upload. Note that both the FW firmware and MD5 checksum files must be loaded.
3. Click Upgrade
4. After the upgrade is complete, confirm that the firmware version has changed (refer to top right of the Control Panel).

System Tab – System Log

System log files are automatically created and assist with troubleshooting in the event the 8186 SIP Horn Speaker does not behave as expected.



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Specifications

Power Input:	PoE (IEEE 802.3af Class 0) 48V, 12.95W (Max 12.95W - Idle nominal 2W)
SIP:	SIP Extensions: • 50 Page (Capable of hands-free talkback) • 10 Emergency Alert • 10 Ring SIP Signalling/Transport Protocols: UDP, TCP, TLS, RTP, SRTP
Multicast & Third-Party Compatibility:	RTP Multicast (Send and Receive 50 Zones) Polycom Group Page Singlewire InformaCast (additional license required) Syn-Apps Revolution
Configuration & Provisioning:	Configuration: Web interface or provisioning server Web Interface: HTTP, HTTPS Provisioning: TFTP, FTP, HTTP, HTTPS DHCP Options 66, 150, 160 Reboot via SIP 'check-sync' Supervision: Compatible with any third party SNMP monitoring software or the Algo 8300 Controller
Networking:	Networking: IPv4, DHCP, VLAN Link Layer: LLDP, CDP QoS: DSCP NAT: STUN, TURN, CRLF Keep Alive, SIP Outbound Address Resolution: DNS, SRV Record Redundancy: Secondary and tertiary SIP server Time: NTP Server (up to four)
Audio:	Speaker: Double re-entrant horn speaker, 11" x 6 5/8" (30cm x 16.8cm) rectangular SPL: 116 dBA at 1m (1 kHz sine wave) Frequency Response: 350 – 9,000 Hz (- 10 dB) Dispersion Angle: 76H x 51V (2 kHz -6dB); Oriented Vertically 11” tall x 6 5/8” Wide Microphone: Omnidirectional - talkback and ambient noise monitoring Audio Codecs: G.711 u-law, G.711 A-law, G.722 Wideband Audio Memory: 1 GByte audio storage
Input/Output:	Relay Input: Normally open or normally closed dry contact supervision using Algo 1202 Call Button, Algo 1203 Call Switch, 1204 Volume Control or EOL resistor termination Relay Output: Max 30 V 50 mA (normally open)

Relay Input Current Draw Detection Thresholds:

	Active	Idle	Tamper
Normally Open	>4mA	<4mA	N/A
Normally Open with Supervision	>20mA	4-20mA	<4mA
Normally Closed	<4mA	>4mA	N/A
Normally Closed with supervision	4-20mA	>20mA	<4mA

Nominal 12V source, current limited to 40mA
Typical supervision resistor value = 1k ohm

Environmental & Mechanical:

Environmental: -40 to +50° C (-40 to +122° F);
Suitable for outdoor and wet environments when properly installed.
IPX6 rating
Dimensions (Product): 11” x 6.6” x 10.5” (28cm x 16.8cm x 26.7cm)
Weight (Product): 5.0 lbs (2.3 kg)
Weight (Shipping): 6.5 lbs (2.9 kg)

Compliance:

RoHS, CE, FCC Class A, CISPR 22 Class A, CISPR 24, CSA/UL (USA & Canada), EN60950

FCC Compliance Statement

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy, and if it is not installed and used in accordance with the instruction manual, it may cause harmful interference to radio communications. Operation 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 own expense.