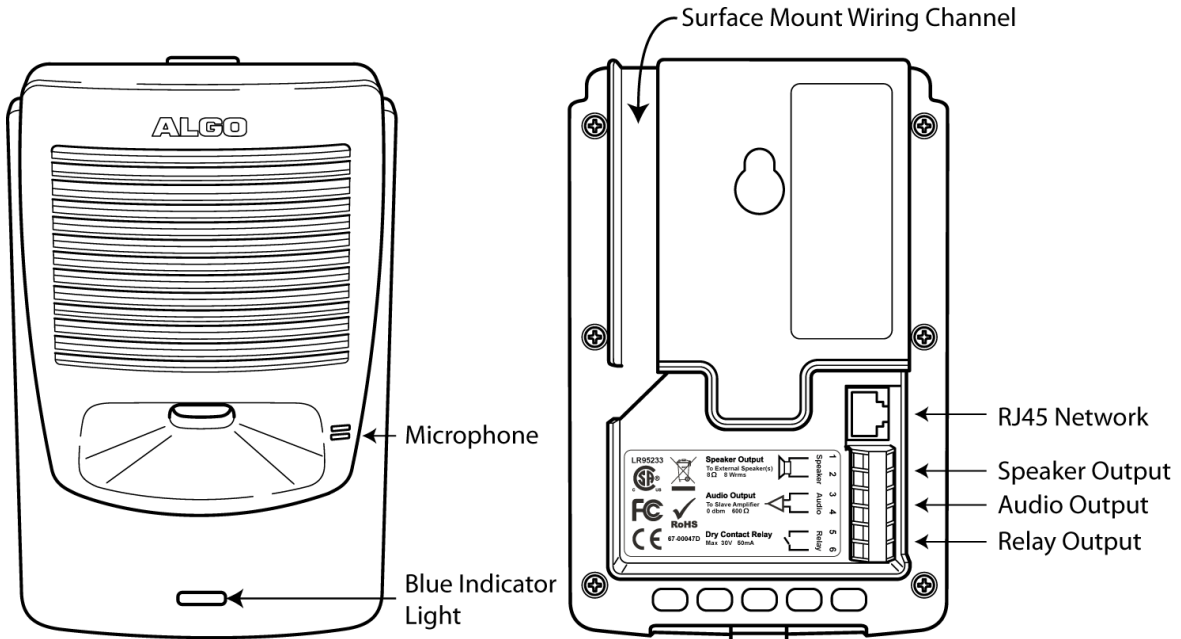




Algo 8180 Pager v1.0

## Algo 8180 Audio Alerter / Pager Firmware Version 2.2



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## Introduction

Thank you for choosing FreedomVoice for your industry-leading cloud based phone system. We are glad to have you on board as part of our team and this document should help you configure the Algo SIP audio pager with FreedomVoice.

The 8180 SIP Audio Alerter is a SIP compliant PoE network audio device for loud ring and voice paging applications using dual endpoints. When registered with a SIP server, the device can either play an audio file from internal memory upon ring detection or auto-answer for voice paging, complete with two-way talkback.

**NOTE: FreedomVoice only provides line activation (basic dial tone) support for this product. Our knowledge of these devices does not extend beyond the scope of a SIP configuration for VoIP. For all other options, please consult Algo for support.**

## Configure the Algo 8180 with a FreedomVoice account

The Algo SIP audio pager only takes a few configuration options to get up and running. This guide assumes that the unit is connected to an Ethernet cable that provides POE and Internet access.

### Start by activating the device

1. Login to Weblink and choose “Settings”, “Activate”.
2. Activate the device under “Linksys” and “PAP2T”.
3. Go to “Support”, “Diagnose Phones”.
4. Click the “Diagnose” button next to the device you just activated.
5. From this screen you will need the “DID Number” along with the “Device Password”.

### Configure the SIP account

1. Access the webUI for the device by typing the local IP address of the device into a web browser (see Algo 8180 documentation for additional details).
2. Select the “Basic Settings” tab and then the “SIP” tab. (**NOTE:** If a field isn’t covered here, leave the default option or leave it blank)
3. **SIP Domain (Proxy Server):** Enter “setup.hostedipbx.com”.

**NOTE:** You can choose to have the unit act as an audio alerter (loud ring) or audio pager (loud speaker). Paging is covered here but if you wanted to use the unit for ringing instead, just fill out the “Ring...” fields instead of the “Page...” fields as outlined in the steps below.

4. **Page Extension:** Enter the DID activated in the activation step.
5. **Page Authentication ID:** Again, enter the DID activated in the activation step.
6. **Page Authentication password:** Enter the device password gathered in step 5 above.
7. Click “Save” in the lower right hand corner.

The screenshot shows the web interface for the ALGO 8180 SIP Audio Alerter. The title bar reads "ALGO 8180 SIP Audio Alerter Control Panel" with "Firmware: 2.2" on the right. Below the title bar are tabs for "Status", "Basic Settings", "Advanced Settings", "System", and "Logout". The "Basic Settings" tab is active, and within it, the "SIP" sub-tab is selected. The "SIP Settings" section contains the following fields and options:

- SIP Domain (Proxy Server):** A text input field containing "sip.west.hostedipbx.com". A tooltip below the field states: "Default port is 5060. To specify a different port, enter PROXY:PORT, e.g. my\_proxy.com:5070, or 192.168.1.10:5080."
- Ring Extension:** An empty text input field.
- Ring Authentication ID:** An empty text input field.
- Ring Authentication Password:** An empty text input field with a blue eye icon to toggle visibility.
- Page Extension:** A text input field containing "7608289222".
- Page Authentication ID:** A text input field containing "7608289222".
- Page Authentication Password:** A text input field with masked characters (dots) and a blue eye icon to toggle visibility.
- Different Ports for Extensions:** A section with two radio buttons: "Enabled" (selected) and "Disabled". A tooltip below states: "Turn this option on for certain proxies, e.g. Cisco Communication Manager to send ring and page SIP requests through different port numbers."

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

### Set QoS markings

1. Select the "Advanced" tab and choose the "Network" tab.
2. **Sip (6-bit DSCP value):** Enter "011000".
3. **RTP (6-bit DSCP value):** Enter "101110".
4. Click "Save" in the lower right hand corner.
5. Choose the "Advanced SIP" tab.
6. **Register Period (seconds):** Enter "300".
7. Click "Save" in the lower right hand corner.

**Setup is complete!** You can check the SIP registration status under the "Status" tab. It should read "Successful" if all settings were entered correctly.

## Technical Support

FreedomVoice Customer Care is available from 3:00 AM PST to 6:00 PM PST, Monday through Friday, Saturday from 6:30am PST to 3:30pm PST and can be reached either by phone or by email. Emergency support is available 24/7.

**Phone:** 888-955-3520 ext. 2

**Support Email:** [support@freedomvoice.com](mailto:support@freedomvoice.com)

Emails are automatically forwarded to our ticketing system. An auto-reply will be sent within a few minutes indicating the case number generated. Emails are generally returned within two hours during normal support hours, but may take longer depending on the current volume of tickets received. All emails should, however, be returned same day. For an issue that requires a faster turn-around time, please use the phone numbers listed above.