

Algo SIP Endpoints and Zoom Phone Interoperability Testing and Configuration Steps

Need Help?

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Table of Contents

Introduction 3

Configuration Steps – Zoom Web Portal 4

Configuration Steps – Algo Endpoint 6

Interoperability Testing..... 9

Troubleshooting..... 11

Introduction

Algo SIP Endpoints can register to Zoom Phone as a third-party SIP Endpoint and provide Paging, Ringing as well as Emergency Alerting capability.

This document provides instructions to add your Algo device to the Zoom web portal. Interoperability testing results are also available.

All testing has been conducted with the Algo 8301 Paging Adapter and Scheduler, 8186 SIP Horn, and 8201 SIP PoE Intercom. These are representative of all Algo SIP speakers, paging adapters and doorphones and similar registration steps would apply. Exceptions are the 8028 SIP Doorphone and 8128 Strobe Light. Please contact Algo Support to inquire.

Please note only one SIP extension may be registered to any given Algo endpoint at a time with Zoom Phone. Multiple Lines feature will be release later in the year. For more information, please contact Zoom support.

Configuration Steps – Zoom Web Portal

To register an Algo SIP Endpoint to Zoom Phone begin by creating an a new common area phone in the Zoom web portal. See the [Zoom support site](#) for more information.

1. Sign in to the Zoom web portal.
2. Click **Phone System Management** > [Users & Rooms](#).
3. Click the [Common Area Phones](#) tab.
4. Click **Add** and enter the following information:

The screenshot shows the 'Add Common Area Phone' dialog box in the Zoom web portal. The dialog box is white with a light gray border and contains the following fields:

- Display Name:** Algo 8180(G2)
- Description (Optional):** (Empty)
- Extension Number:** 804
- MAC Address:** 00:22:ee:12:00:25
- Device Type:** Other (selected from a dropdown menu)

At the bottom right of the dialog box, there are two buttons: 'Cancel' and 'Save'.

- **Site** (only visible if you have [multiple sites](#)): Select the site you want the device to belong to.
- **Display Name:** Enter a display name to identify the device.
- **Description (Optional):** Enter a description to help you identify the location of the device.
- **Extension Number:** Enter an extension number to assign it to the device.
- **MAC Address:** Enter the 12-digit MAC address of the Algo Endpoint. The MAC can be found on the product label or in the Algo Web Interface under **Status**.
- **Device Type:** Select **Other**.
Note: If you don't have the **Other** option, contact your Zoom sales representative. By default, support for a generic SIP profile is not enabled.

- **Emergency Address** (only visible if you don't have multiple sites): Select an [emergency address](#) to assign to the desk phone. If you selected a site for the common area phone, the site's emergency address will be applied to the phone.
5. Click **Save**.
 6. Click **Provision** to view the SIP credentials. You will need this information to complete provisioning using the Algo Web Interface.

Provisioning

MAC Address	00-22-ee-12-00-25
Device Type	Other

You will need to enable TLS1.2 for SIP registration and enable SRTP for secure calling on your IP phone. Please refer to your manufacturer's instructions for these processes.

You'll need following information for manual provisioning.

1. **SIP Domain:** 50744587.zoom.us
2. **Outbound Proxy:** us01sipsj06.zoom.us:5091
3. **User Name:** [\[redacted\]](#)
4. **Authorization ID:** [\[redacted\]](#)
5. **Password:** [\[redacted\]](#)

Also, download CA [certificate](#) and import to trust list on your IP phone.

Note: Please note that Zoom support team will not be able to troubleshoot or configure IP phones that are provisioned in this manner. Some Zoom Phone features may not work on manually provisioned phones. It may vary depending on your desk phone model.

[Close](#)

Configuration Steps – Algo Endpoint

To register an Algo SIP Endpoint navigate to the Web Configuration Interface.

1. Open a web browser.
2. Type the IP Address of the endpoint. If you don't know the address yet, navigate to www.algosolutions.com, find the user guide for your product and go through the Getting Started section.
3. Log in and go to **Basic Settings** -> **SIP** tab.
4. Enter the information provided from Zoom as per below:
 - SIP Domain (Proxy Server) – Zoom SIP Domain
 - Page or Ring Extension – Zoom User Name
 - Authentication ID – Zoom Authorization ID
 - Authentication Password – Zoom Password

Status Basic Settings Additional Features Scheduler Advanced Settings System Logout

SIP Features Multicast

SIP Settings

SIP

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

SIP Domain (Proxy Server) 50744587.zoom.us
i Default port is 5060. To specify a different port, enter PROXY:PORT, e.g. my_proxy.com:5070, or 192.168.1.10:5080.

Ring/Alert Mode Monitor "Ring" event on registered SIP extension
 None

Page Extension 215234142890

Authentication ID 768005571266

Authentication Password ●●●●●●

Display Name (Optional)

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

Save

5. Go to **Advanced Settings** -> **Advanced SIP**.
6. Set the **SIP Transportation** protocol to "TLS".
7. Enter the **Outbound Proxy** provided by Zoom.

The screenshot shows the 'Advanced SIP Settings' page. The 'SIP Transportation' dropdown menu is set to 'TLS'. Below it, there are two informational icons: a blue 'i' icon and a red 'x' icon. The 'Outbound Proxy' text field contains the value 'us01sipsj03.zoom.us:5091'. Other settings include 'SIPS Scheme' (Disabled), 'SDP SRTP Offer' (Disabled), and 'SIP Outbound Support (RFC 5626)' (Disabled).

8. Ensure the SIP Registration Status shows "Successful".

The screenshot shows the 'Device Status' page. It includes a welcome message and four steps for configuration. At the bottom, there is a 'Status' table with the following data:

Status	
Device Name	pagingadapter
SIP Registration	Page Successful (Extension 215234142890)
Call Status	Idle
Proxy Status	Single proxy mode
Security	TLS Enabled SRTP Disabled
Provisioning Status	None Found



Note: if registering additional extensions for ringing, paging or emergency alerting, enter the unique credentials for the respective extension in the same way.

Only one SIP extension may be registered to any given Algo endpoint at a time with Zoom Phone. Multiple Lines feature will be release later in the year. For more information, please contact Zoom support.

Interoperability Testing

Register to Zoom Phone

- Endpoints: 8301 Paging Adapter and Scheduler, 8186 SIP Horn, 8201 SIP PoE Intercom
- Firmware: 1.7.6
- Description: Verify 3rd Party SIP Endpoints are registered successfully.
- Result: **Successful**

Register Multiple SIP Extensions Simultaneously

- Endpoints: 8301 Paging Adapter and Scheduler, 8186 SIP Horn
- Firmware: 1.7.6
- Description: Verify the server will sustain multiple simultaneous extensions registered to the same endpoint (e.g. page, ring, and emergency alert).
- Result: **Not supported at this time. Please see note below.**

Please note only one SIP extension may be registered to any given Algo endpoint at a time with Zoom Phone. Multiple Lines feature will be release later in the year. For more information, please contact Zoom support.

One-Way Page

- Endpoints: 8301 Paging Adapter and Scheduler, 8186 SIP Horn
- Firmware: 1.7.6
- Description: Verify one-way page mode functionality, by calling the registered page extension.
- Result: **Successful**

Two-Way Page

- Endpoints: 8301 Paging Adapter and Scheduler, 8186 SIP Horn, 8201 SIP PoE Intercom
- Firmware: 1.7.6
- Description: Verify two-way page mode functionality, by calling the registered page extension.
- Result: **Successful**

Ringling

- Endpoints: 8301 Paging Adapter and Scheduler, 8186 SIP Horn
- Firmware: 1.7.6
- Description: Verify ringing mode functionality by calling the registered ring extension.
- Result: **Successful**

Emergency Alerts

- Endpoints: 8301 Paging Adapter and Scheduler, 8186 SIP Horn
- Firmware: 1.7.6
- Description: Verify emergency alerting functionality by calling the registered extension.
- Result: **Successful**

Outbound Calls

- Endpoints: 8301 Paging Adapter and Scheduler, 8186 SIP Horn, 8201 SIP PoE Intercom
- Firmware: 1.7.6
- Description: Verify emergency alerting functionality by calling the registered extension.
- Result: **Successful**

TLS for SIP Signaling

- Endpoints: 8301 Paging Adapter and Scheduler, 8186 SIP Horn, 8201 SIP PoE Intercom
- Firmware: 1.7.6
- Description: Verify TLS for SIP Signaling is supported.
- Result: **Successful**

SDP SRTP Offer

- Endpoints: 8301 Paging Adapter and Scheduler, 8186 SIP Horn, 8201 SIP PoE Intercom
- Firmware: 1.7.6
- Description: Verify support for SRTP calling using Standard (RTP/SAVP) or optional (RTP/AVP) profile.
- Result: **Successful**

Troubleshooting

SIP Registration Status = “Rejected by Server”

Meaning: The server received register request from the endpoint and responds with an unauthorized message.

- Ensure the SIP credentials (extension, authentication ID, password) are correct.
- Under Basic Settings -> SIP, click on the blue circular arrows to the right of the Password field. If the Password is not what it should be, the web browser is probably auto filling the password field. If so, any change on a page containing a password could be filled in with an undesired string.

SIP Registration Status = “No reply from server”

Meaning: the device is not able to communicate across the network to the phone server.

- Double check the "SIP Domain (Proxy Server)", under Basic Settings -> SIP tab field is filled out correctly with the address of your server and port number.
- Ensure the firewall (if present) is not blocking the incoming packets from the server.
- Ensure TLS is configured for SIP Transportation Method (Advanced Settings -> Advanced SIP).