

## Algo SIP Endpoints and VoIP.MS Interoperability Testing and Configuration Steps

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## Introduction

Algo SIP Endpoints can register to VoIP.MS as third-party SIP Endpoints and provide Paging, Ringing as well as Emergency Alerting capability.

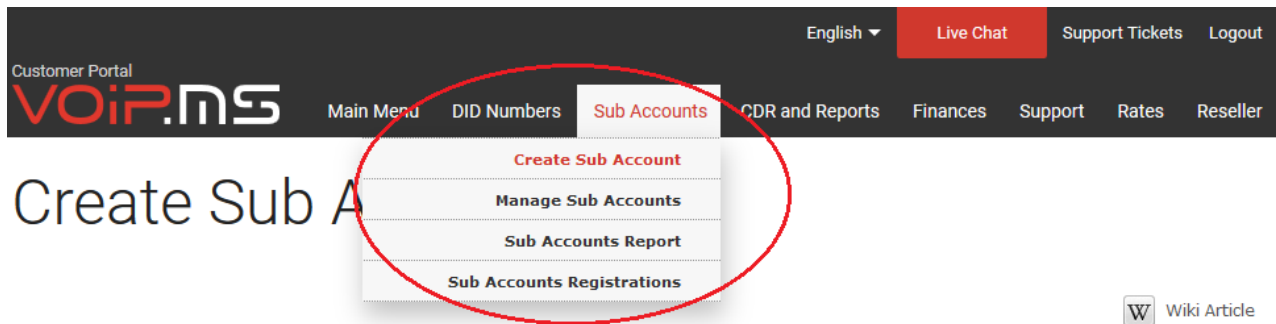
This document provides instructions to register your Algo SIP Endpoint to VoIP.MS. Interoperability testing results are also available towards the end of the document.

Please note all testing has been conducted with the Algo 8301 Paging Adapter and Scheduler, 8180 SIP Audio Alerter (G2), and 8201 SIP PoE Intercom. These are representative of all Algo SIP speakers, paging adapters and doorphones and similar registration steps would apply.

## Configuration Steps

One or multiple SIP accounts may be created and registered to any given Algo SIP Endpoint. To do so, open the VoIP.MS customer portal and navigate to Sub Accounts -> Create Sub Account. Fill out the Username, Password and any other required parameters.

To enable TLS for SIP transportation and SRTP encryption, expand Advanced Options and enable Encrypted SIP Traffic. Please note this is an optional step.



Create new sub account	
Protocol	<input checked="" type="radio"/> SIP (Recommended) <input type="radio"/> IAX2
Authentication type	<input checked="" type="radio"/> User/Password Authentication (Recommended) <input type="radio"/> Static IP Authentication (SIP only) (Advanced Users)
Username	262027_ <input type="text"/> (max 12 chars)
Password (If not using IP Auth)	<input type="text"/>
Confirm your password	<input type="text"/>
Device type	<input type="radio"/> Asterisk, IP PBX, Gateway or VoIP Switch <input checked="" type="radio"/> ATA device, IP Phone or Softphone
CallerID Number	<input checked="" type="radio"/> Use one of my DID's <input type="text"/> <input type="radio"/> Use a Custom CallerID <input type="text"/> <input type="radio"/> I use a system capable of passing its own CallerID
Canada Routing	Premium: Canada \$0.009 <input type="text"/>
International Route	Value <input type="text"/>
Allow International Calls	No - International Calls Disabled <input type="text"/> > <a href="#">Select allowed countries</a>
Allow *225 for Balance	No - *225 for Balance Disabled <input type="text"/>
Music on Hold	No Music <input type="text"/>
Record Calls <b>BETA</b>	No <input type="text"/>
Account name or description (For your own use)	<input type="text"/> (Optional)
<b>Advanced Options</b> <a href="#">Click here to display</a>	

After the extension has been created in the system its credentials must be plugged in the Algo endpoint. In the customer portal navigate to Sub Accounts -> Manage Sub Accounts. The recently created extension will show up in the list.

To register an Algo SIP Endpoint, open a web browser and type the IP address assigned to the unit to access the web configuration panel. Go to Basic Settings -> SIP tab and enter the follow information:

- SIP Domain (Proxy Server) – VoIP.MS server (in the customer portal, hover over the information icon under Authentication Type column to get the SIP domain)
- Page and/or Ring Extension – VoIP.MS Username (available in the sub account management page)
- Authentication ID – VoIP.MS Username
- Authentication Password – Password set in the account creation step

Note: if registering additional extensions for ringing, paging and emergency alerting, enter the unique credentials for the respective extension in the same way. Any combination of page, ring and/or emergency alerts is acceptable, as long the credentials are unique.

The screenshot displays the 'SIP Settings' configuration page. At the top, there are navigation tabs: 'Status', 'Basic Settings' (selected), 'Additional Features', 'Advanced Settings', 'System', and 'Logout'. Below these are sub-tabs: 'SIP' (selected), 'Features', and 'Multicast'. The main content area is titled 'SIP Settings' and contains the following sections:

- SIP**: An information icon and text stating: '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)**: A text input field containing 'houston1.voip.ms'. Below it, a note says: '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**: Radio button options: 'Monitor "Ring" event on registered SIP extension', 'Use "Subscribe/Notify" dialog event (RFC 4235) to monitor event on different extension', 'Use "Subscribe/Notify" presence event (RFC 3856/3863 PIDF) to monitor event on different extension', and 'None' (selected).
- Base/Page Extension**: Text input field containing '262027\_ext2'.
- Authentication ID**: Text input field containing '262027\_ext2'.
- Authentication Password**: Password input field containing '.....'.
- Display Name (Optional)**: Empty text input field.

At the bottom right, there is a 'Save' button with a green checkmark icon. A red circle highlights the 'SIP Domain' field, and another red circle highlights the 'Base/Page Extension', 'Authentication ID', and 'Authentication Password' fields.

If Encrypted SIP Traffic has been enabled, make sure to set the SIP Transportation protocol to “TLS”, SDP SRTP Offer to Standard and Register Period to 300, under Advanced Settings -> Advanced SIP. Please note this step is only applicable if Encrypted SIP Traffic has been enabled in the Client Portal.

**Advanced SIP Settings**

**General**

SIP Transportation: TLS

SIPS Scheme:  Enabled  Disabled

SDP SRTP Offer: Standard

SIP Outbound Support (RFC 5626):  Enabled  Disabled

Outbound Proxy: [Empty]

Register Period (seconds): 300

Ensure the SIP Registration Status shows “Successful” for each extension registered.

**Device Status**

Welcome to the Algo 8180G2 SIP Audio Alerter Control Panel

Setting up your SIP Audio Alerter:

**Step 1: Configure your SIP Audio Alerter**

Log in with the default password and use the Basic Settings pages to set up the basic information.

**Step 2: Check network settings (Optional)**

Use the Network page under the Advanced Settings tab to change network settings. The default setting for the device is to obtain its IP address from a DHCP server. Contact your Network System administrator if you plan to assign a static IP address, Mask, and Gateway to the device.

**Step 3: Secure your SIP Audio Alerter (Optional)**

Use the Admin page under the Advanced Settings tab to change the administrator password.

⚠ Changing the password is extremely important if the device is directly connected to a public network.

**Step 4: Register your SIP Audio Alerter (Optional)**

Please register your product using the link below:

<http://www.algosolutions.com/register>

Registration ensures your access to the latest upgrades to this product and important service notices.

Status	
Device Name	sipalerter
SIP Registration	Successful (Extension 262027_ext2)
Call Status	Idle
Proxy Status	Single proxy mode
Security	TLS Enabled SRTP Enabled

## Interoperability Testing

Feature	Endpoints Tested	Firmware	Description	Result
<b>Register to VoIP.MS</b>	8301 Paging Adapter and Scheduler, 8186 SIP Horn, 8201 SIP PoE Intercom	1.7.9	Verify 3 <sup>rd</sup> Party SIP Endpoints can register successfully.	<b>Successful</b>
<b>Register Multiple SIP Extensions Simultaneously</b>	8301 Paging Adapter and Scheduler, 8186 SIP Horn	1.7.9	Verify the server will sustain multiple simultaneous extensions registered to the same endpoint (e.g. page, ring, and emergency alert).	<b>Successful</b>
<b>One-Way Paging</b>	8301 Paging Adapter and Scheduler, 8186 SIP Horn	1.7.9	Verify one-way page mode functionality, by calling the registered page extension.	<b>Successful</b>
<b>Two-Way Communication</b>	8301 Paging Adapter and Scheduler, 8186 SIP Horn, 8201 SIP PoE Intercom	1.7.9	Verify two-way page mode functionality, by calling the registered page extension.	<b>Successful</b>
<b>Ringling</b>	8301 Paging Adapter and Scheduler, 8186 SIP Horn	1.7.9	Verify ringing mode functionality by calling the registered ring extension.	<b>Successful</b>
<b>Emergency Alerting</b>	8301 Paging Adapter and Scheduler, 8186 SIP Horn	1.7.9	Verify emergency alerting functionality by calling the registered extension	<b>Successful</b>
<b>Outbound Calling</b>	8301 Paging Adapter and Scheduler, 8186 SIP Horn, 8201 SIP PoE Intercom	1.7.9	Verify emergency alerting functionality by calling the registered extension.	<b>Successful</b>
<b>UDP for SIP Signaling</b>	8301 Paging Adapter and Scheduler, 8186 SIP Horn, 8201 SIP PoE Intercom	1.7.9	Verify UDP for SIP Signaling is supported.	<b>Successful</b>
<b>TCP for SIP Signaling</b>	8301 Paging Adapter and Scheduler, 8186 SIP Horn, 8201 SIP PoE Intercom	1.7.9	Verify TCP for SIP Signaling is supported.	<b>Successful</b>

<b>TLS for SIP Signaling</b>	8301 Paging Adapter and Scheduler, 8186 SIP Horn, 8201 SIP PoE Intercom	1.7.9	Verify TLS for SIP Signaling is supported.	<b>Successful</b>
<b>SDP SRTP Offer</b>	8301 Paging Adapter and Scheduler, 8186 SIP Horn, 8201 SIP PoE Intercom	1.7.9	Verify support for SRTP calling using Standard (RTP/SAVP) or optional (RTP/AVP) profile.	<b>Successful</b>



## Troubleshooting

### **SIP Registration Status = “Rejected by Server”**

Meaning: The server receives register requests 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.
- Ensure the SIP Transportation method (Advanced Settings -> Advanced SIP) is configured correctly, according to the configuration applied in the VoIP.MS Client Portal.
- Check the System Log (System -> System Log tab). If you see “500 Server Internal Error”, it often this means that this is not actually the correct address/port for the SIP server (although the server does know enough to reject the request).

### **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 an internet connection is available for the endpoint to reach the hosted server.
- Ensure the firewall (if present) is not blocking the incoming packets from the server.