



David,  
I downloaded this guide and inserted XBLUE info...

I got this guide at:  
[http://www.algosolutions.com/pdf/user\\_guides/SIP%20Registration%20Guide.pdf](http://www.algosolutions.com/pdf/user_guides/SIP%20Registration%20Guide.pdf)

# Getting Started with Algo IP Endpoints: SIP Registration Guide

Need Help?

(604) 454-3792 or [support@algosolutions.com](mailto:support@algosolutions.com)

## Table of Contents

Introduction..... 3

Instructions..... 3

Troubleshooting..... 7

Appendix..... 8



Cisco Call Manager..... 8

Nextiva..... 8

Vonage..... 9

RingCentral ..... 9

Digium..... 9

## Introduction

Algo IP products register with most hosted / cloud or premise-based telephone systems supporting 3<sup>rd</sup> party SIP endpoints. This guide provides instructions to get an Algo SIP endpoint registered as well as to troubleshoot a failed registration.

For a list of known phone systems which support Algo SIP devices and specific instructions, please visit the URL below:

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

Please check the [Appendix](#) for instructions for a select number of the well-known platforms. For a complete list, please check the URL above.

Note: Additional phone systems may not be listed, however, as long as 3<sup>rd</sup> party SIP endpoints are supported by the phone system then Algo endpoints will register without issue.

## Instructions

1. Log into the web interface by typing the device's IP address in the web browser. For devicespecific instructions to discover the IP address, check its User Guide, or use the [Network Device Locator](#).
2. Get in touch with your service provider or network administrator to request the following:
  - a. SIP Server Address / Domain Name
  - b. Create a new SIP Extension, Authentication ID and Password  
Note: On some phone systems if no Authentication ID is available use the extension # in this field.
  - c. (Not mandatory – dependent upon service provider settings) IP address / Domain Name for Outbound Proxy
3. Enter the SIP server's IP address / Domain Name into the **SIP Domain (Proxy Server)** field under the **Basic Settings** -> **SIP** tab.
4. Enter the **Page** and/or **Ring Extension, Authentication ID** and **Password**.

David, you mentioned ringing as well as paging so you'll need two extensions on the XBLUE phone system, one for paging the other for ringing.

**SIP Settings**

**SIP**

This section allows the SIP server information & account credentials to be entered. This information should be obtained from your telephone service provider.

SIP Domain (Proxy Server):  (Default port is 5060. To specify a different port, enter the port number.)

Monitor "Ring" event on registered: ☒ Monitor "Ring" event on registered ☐ None

**Ring Extension**

Authentication ID:  Password for 15:

Authentication Password:

The device will detect inbound ring events on this extension and play the ring tone (and multicast if configured) until the user answers the call.

**Page Extension**

Authentication ID:  Password for 16:

Authentication Password:

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

192.168.254.19:8850/#

IP3022(V2) Configurati... https://184.179.8.72:65... Research

## X-25 Smart Configuration

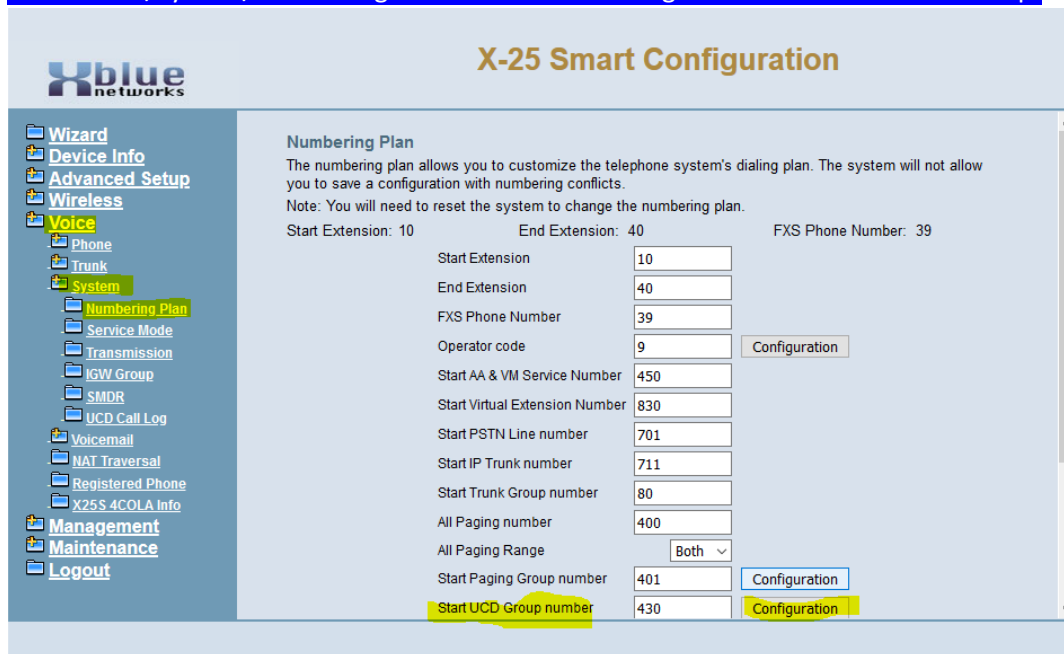
removed from the system.

This is where all that came from

No.	Phone	Display	MAC	Day	Night	PickUp	blue Phones
No.	Number	Password	Name	Binding	COS	COS	Group
1	10	.....		00:19:15:6f:eb:ee	0	0	1
2	11	.....		00:19:15:e4:5a:18	0	0	1
3	12	.....		00:19:15:d9:a9:ae	0	0	1
4	13	.....		00:19:15:e2:2c:ea	0	0	1
5	14	.....		00:19:15:dd:18:30	0	0	1
6	15	.....			0	0	1
7	16	.....			0	0	1
8					0	0	1
9					0	0	1
10					0	0	1
11					0	0	1

Set ringing for 15:

Go to Voice/System/Numbering Plan and click on Configuration button for UCD Group:



**X-25 Smart Configuration**

**Numbering Plan**

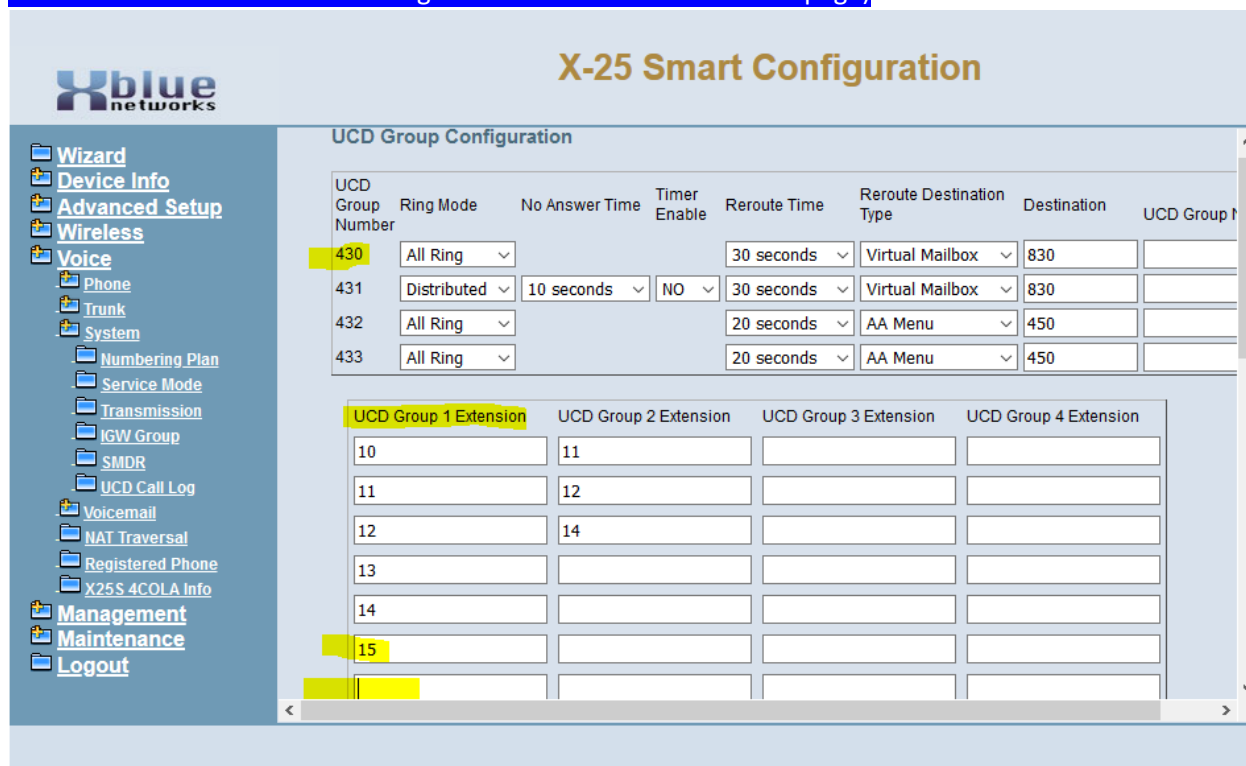
The numbering plan allows you to customize the telephone system's dialing plan. The system will not allow you to save a configuration with numbering conflicts.  
Note: You will need to reset the system to change the numbering plan.

Start Extension: 10      End Extension: 40      FXS Phone Number: 39

Start Extension	10	
End Extension	40	
FXS Phone Number	39	
Operator code	9	Configuration
Start AA & VM Service Number	450	
Start Virtual Extension Number	830	
Start PSTN Line number	701	
Start IP Trunk number	711	
Start Trunk Group number	80	
All Paging number	400	
All Paging Range	Both	
Start Paging Group number	401	Configuration
Start UCD Group number	430	Configuration

XBLUE set all ringing phones in UCD Group 1

Make sure that 15 is in the Group and 16 is not. (I'm not sure how the Algo will behave but this makes sure that 16 won't be detected as a Ring and 15 won't be detected as a page)



**X-25 Smart Configuration**

**UCD Group Configuration**

UCD Group Number	Ring Mode	No Answer Time	Timer Enable	Reroute Time	Reroute Destination Type	Destination	UCD Group
430	All Ring			30 seconds	Virtual Mailbox	830	
431	Distributed	10 seconds	NO	30 seconds	Virtual Mailbox	830	
432	All Ring			20 seconds	AA Menu	450	
433	All Ring			20 seconds	AA Menu	450	

UCD Group 1 Extension	UCD Group 2 Extension	UCD Group 3 Extension	UCD Group 4 Extension
10	11		
11	12		
12	14		
13			
14			
15			

You should be able to call Ext 16 and do a page announcement, and the loudspeaker should ring

Note: Algo SIP speakers and the 8301 paging adapter support two types of SIP extensions. A Page Extension auto-answers for voice page announcements. A Ring Extension plays an audio WAV file (e.g. ring tone, alert announcement, etc.). It is not required that you register both extensions for a device to register. Leave blank the appropriate extension if only registering a single extension. For converting and uploading a custom WAV files, reference the [Tone Conversion Guide](#).

5. (Not mandatory) If the service provider uses an **Outbound Proxy**, enter its address under **Advanced Settings -> Advanced SIP**.

The screenshot displays the 'Advanced SIP Settings' page. The 'Advanced Settings' tab is selected, and the 'Advanced SIP' sub-tab is active. The 'General' section contains the following settings: 'SIP Transportation' is set to 'Auto' (with a note: 'Select Auto to check DNS NAPTR record, then try UDP/TCP.'), 'SDP SRTP Offer' is set to 'Disabled', 'SIP Outbound Support (RFC 5626)' is set to 'Disabled' (with a note: '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' is set to 'sip.myproxy.com', and 'Register Period (seconds)' is set to '3600'. The 'NAT' section shows 'Media NAT' set to 'None'. The 'Server Redundancy' section shows 'Server Redundancy Feature (Multiple SIP Server Support)' set to 'Disabled'. The 'Interoperability' section shows 'Keep-Alive Method' set to 'None' (with a note: 'This setting will enable sending periodic CRLF messages for both UDP and TCP connections.'). A 'Save' button is visible in the bottom right corner.

6. Check SIP Registration Status. If the status is not “Successful”, read the [Troubleshooting](#) section below.

The screenshot shows the 'Status' tab selected in the top navigation bar. Below the navigation bar, the 'Device Status' section is visible. It includes a welcome message and four steps for setting up the SIP Horn. The 'SIP Registration' status is highlighted with a red bar, showing 'Page' as 'Successful' for 'Extension 2065'. Below this, a table lists various device settings.

Status	
Device Name	siphorn
SIP Registration	Page Successful (Extension 2065)
Call Status	Idle
Proxy Status	Single proxy mode
Security	TLS Disabled SRTP Disabled
Provisioning Status	None Found
MAC	00:22:ee:0a:31:89
IP	10.30.18.38
Netmask	255.0.0.0
Gateway	10.0.0.1
Date / Time	Tue Jun 5 21:57:56 UTC 2018
Multicast Mode	Slave Mode, Idle
Volume	Page Volume: -2 (-36dB), Ring Volume: -2 (-36dB)
Relay Input Status	Disabled

7. Call to the endpoint by dialling its Page / Ring extension

## Troubleshooting

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

**Meaning:** The server receives Register packets from the endpoint and responds with an unauthorized message.

- Ensure the credentials (extension, authentication ID, password) on the device match on the Server.
- 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 autofilling the password field. If so, any change on a page containing a password could be filled in with an undesired string.
- Many VoIP Phone Systems don't accept more than 1 device registered to an extension. Make sure that the endpoint is registering with an extension that is not being used by any other device.
- 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).

- Lastly, if the Extension and Authentication ID are not the same, copy & paste the Extension to Authentication ID.

## **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.
- Check if the service provider uses an Outbound Proxy. If so, enter it under Advanced Settings -> Advanced SIP
- Ensure the firewall (if present) is not blocking the incoming packets from the server.

## **Registration Drops Constantly**

- Enable the Keep-alive method. Navigate to Advanced Settings -> Advanced SIP, set Keep-alive to "Double CRLF" and set the period to 30 seconds.

## Appendix

### Cisco Call Manager

A step-by-step guide can be found at the address below:

<http://www.algosolutions.com/pdf/Cisco%20CUCM%2012.0%20Config%20Guide%20for%20Algo%208186%20SIP%20Horn%20Speaker.pdf>.

When registering two or more extensions to a single 3rd party SIP endpoint in CUCM, it must be configured using as "Third Party Advanced" as opposed to "Third Party Basic". This will allow more than one extension per MAC address. Configuration as Basic is only designed for one extension.

It will be necessary to delete the basic extension and reconfigure, because it cannot be changed into Advanced after it's created.

### Nextiva

Extension field is the SIP Username and Authentication ID is Authentication Name.



## Vonage

- Alerter/Speaker/Horn - <https://businesssupport.vonage.com/articles/answer/Algo-Alerter-1002>
- Doorphone - <https://businesssupport.vonage.com/articles/answer/Algo-Doorphone-1003>

## RingCentral

To provision an Algo unit on RingCentral platform, log in to the online administration portal, go to "Phone & Devices" -> "User Phones". Click "+ Add Device". On step 3 "Buy Phones" choose "Other phones" and "Existing Phone". Keep following the steps until the end.

Once this is completed, click on "Setup & Provision" -> "Other Phones". Copy and paste the credentials on the Algo Endpoint:

- Ring Central User Name is entered into the Extension field
- Ring Central Authorization ID is entered into the Authentication ID field • Ring Central Password is entered into the Password field

The SIP Domain must be: sip.ringcentral.com:5060

Outbound Proxy must be: sip20.ringcentral.com:5090 or sip10.ringcentral.com:5090

If RingCentral provides a different port number than what is referenced above, try their values first.

Note: a SIP endpoint may also be configured as Paging Device and assigned to a Group on the RingCentral platform.

## Digium

On the Digium platform follow the path below:

Setup -> Manage -> Modify extension -> Phone Settings -> Common Settings -> Phone Password

Extension field on the Algo endpoint = extension number

Authentication ID field on the Algo endpoint = extension number