Configuration Notes

Yeastar S-Series VoIP PBX & Algo 8201

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Contents

Algo 8201 SIP Intercom Test Report

This article is the Interoperability Test Report for Yeastar S-Series VoIP PBX and Algo 8201 SIP Intercom.

Tested Equipment & Software

Equipment	Firmware/Software Version
Algo 8201 SIP Intercom	1.6.2
Yeastar S300	30.10.0.59

Summary of test focus

The following table shows a summary of the validated capabilities.

Feature	Test Result
DUT Services	
SIP Registration	PASS
Inbound Call	PASS
Outbound Call	PASS
Serviceability	PASS
PBX Services	
Paging/Intercom Group	PASS

Definitions

Word definitions in the following test plan table.

- **DUT**: Device Under Test, which in this case is the Algo 8180G2 Audio Alerter.
- **Phone A**: A SIP compatible endpoint used to place and receive calls.
- **Phone B**: A SIP compatible endpoint used to place and receive calls.

Test plan

SIP Registration

Test Case	Expected Result	Test Result
Attempt registering DUT Extension using incorrect password.	Registration failure status is correctly displayed in web interface	PASS
Attempt registering DUT Extension using incorrect username.	Registration failure status is correctly displayed in web interface.	PASS
Correctly register DUT Extension	DUT registers properly and status is correctly displayed in web interface	PASS
Register DUT Extension using UDP protocol.	DUT registers properly and status is correctly displayed in web interface	PASS
Register DUT Extension using TCP protocol.	DUT registers properly and status is correctly displayed in web interface	PASS
Register DUT Extension using TLS protocol.	DUT registers properly and status is correctly displayed in web interface	PASS

Inbound Call

Test Case	Expected Result	Test Result
Call the DUT from Phone A.	A two-way audio call is established.	PASS
Call the DUT from Phone A and mute/un-mute the call.	• Mute: The DUT doesn't plays the audio from Phone A. • Unmute: The DUT plays the audio from Phone A.	PASS
When the DUT is already in a call with Phone A, call the DUT from Phone B.	Phone B receives busy tone while Phone A call continues.	PASS
Call the DUT from Phone A and maintain the call for a period of time.	The call remains up after the Session Refresh (REINVITE) is sent to the DUT.	PASS

Outbound Call

Test Case	Expected Result	Test Result
Press the call button on the DUT to call Phone A.	When the call is answered by Phone A. a two-way audio call is established.	PASS
Call the DUT from Phone A and Phone A doesn't answer the call.	Phone A continues ringing until timeout.	PASS
When an outbound call is established on the DUT and Phone A, call the DUT from Phone B.	Phone B receives busy tone, while Phone A call continues.	PASS

Serviceability

Test Case	Expected Result	Test Result
Disconnect, then reconnect, the ethernet cable from the DUT.	DUT registers with the PBX server after the network is restored.	PASS

PBX Feature: Paging/Intercom Group

The following test cases verify the Paging/Intercom Group of Yeastar S300. The DUT acts as a multicast slaver.

Test Case	Expected Result	Test Result
Verify PBX feature: 1-Way Multicast Paging. **Prerequisite**: • On Yeastar S300, add a 1-Way Multicast Paging group. • On the DUT, set the Multicast mode to Slave/Receiver and configure the same multicast IP address and port as the Yeastar S300.		
Dial the 1-Way Multicast Paging number from Phone A.	DUT answers the call automatically, and the 1-way paging is established.	PASS
Cancel the call by hanging up Phone A.	DUT ends the call and stops playing the paging audio.	PASS

Test Case	Expected Result	Test Result
Verify PBX feature: 1-Way Paging. **Prerequisite**: • On Yeastar S300, add a 1-Way Paging group. • On the DUT, register a Page Extension. The page extension is a member of the 1-Way paging group.		
Dial the 1-Way Paging number from Phone A.	DUT answers the call automatically, and the 1-way paging is established.	PASS
Cancel the call by hanging up Phone A.	DUT ends the call and stops playing the paging audio.	PASS
Verify PBX feature: 2-Way Intercom. **Prerequisite**: • On Yeastar S300, add a 2-Way Intercom group. • On the DUT, register a Page Extension. The page extension is a member of the 2-Way Intercom group.		
Dial the 2-Way Intercom number from Phone A.	DUT answers the call automatically, and the 2-way intercom is established.	PASS
Cancel the call by hanging up Phone A.	DUT ends the call and stops playing the audio.	PASS

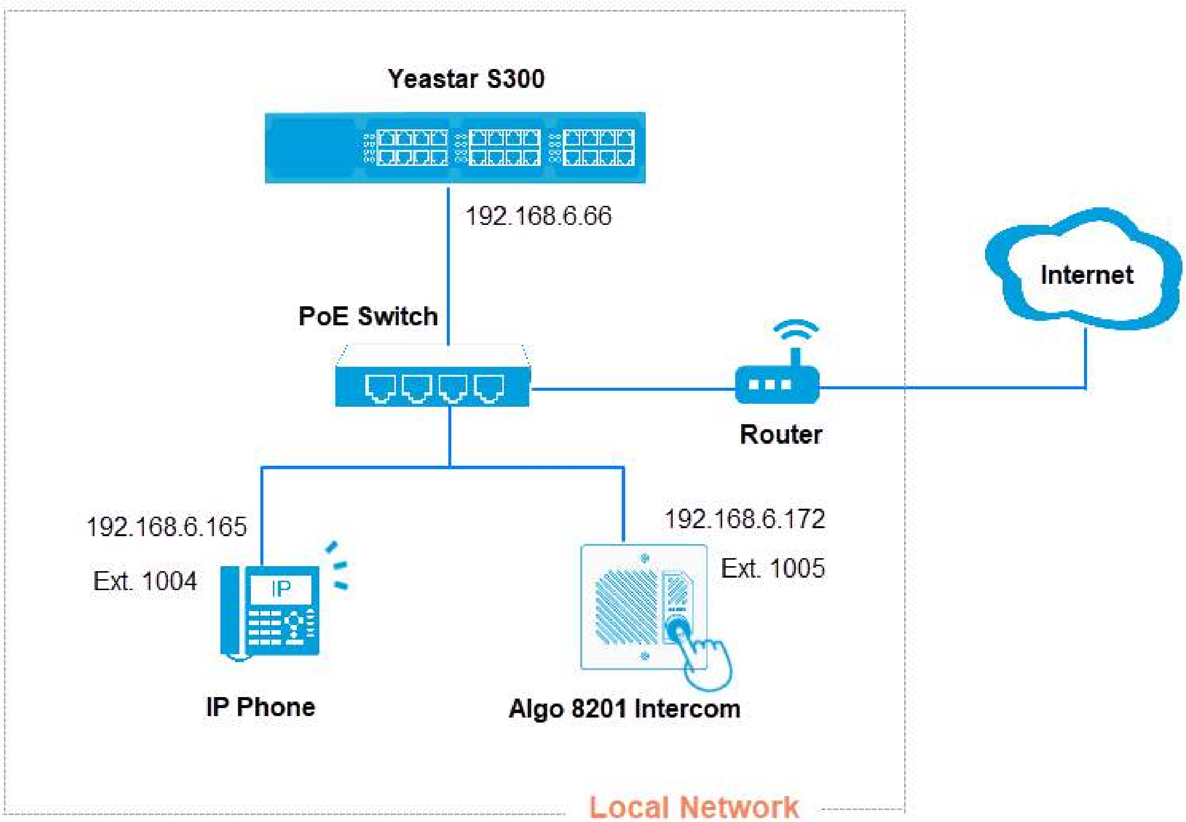
Register Algo 8201 SIP Intercom with Yeastar S-Series VoIP PBX

This guide describes the configuration steps required for Algo 8201 SIP Intercom to interoperate with Yeastar S-Series VoIP PBX.

Below is a guideline of how to register an extension on Algo 8201 SIP Intercom. You may need to configure the other settings of the Algo 8201 SIP Intercom depending on your VoIP solution.

Network Topology

The following diagram shows how the testing network is configured for reference.

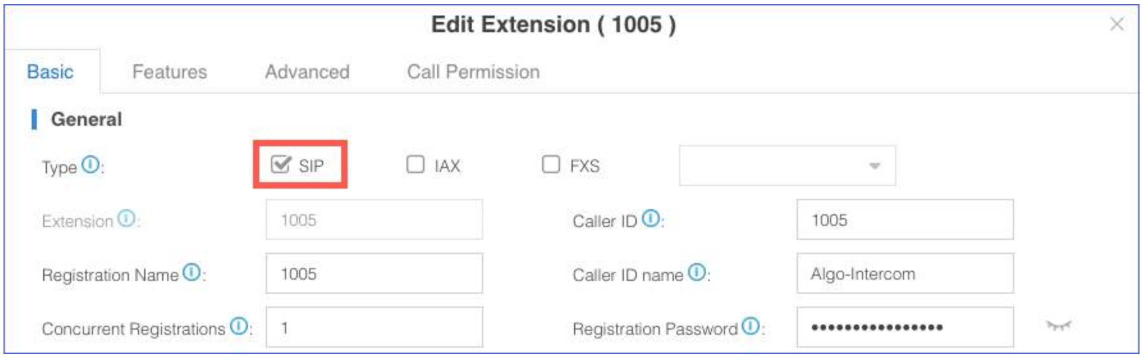


Yeastar S300 configuration

Add a SIP extension on Yeastar S300, and provide the extension details in Algo 8201 web page.

1. Log in Yeastar S300 web interface, go to **Settings→PBX→Extensions**.
2. Add an extension, this extension will be registered as the Algo Ring extension.

a. Click **Add**.
b. Leave the default settings or change the General settings according to your needs.
c. Click **Save** and **Apply**.

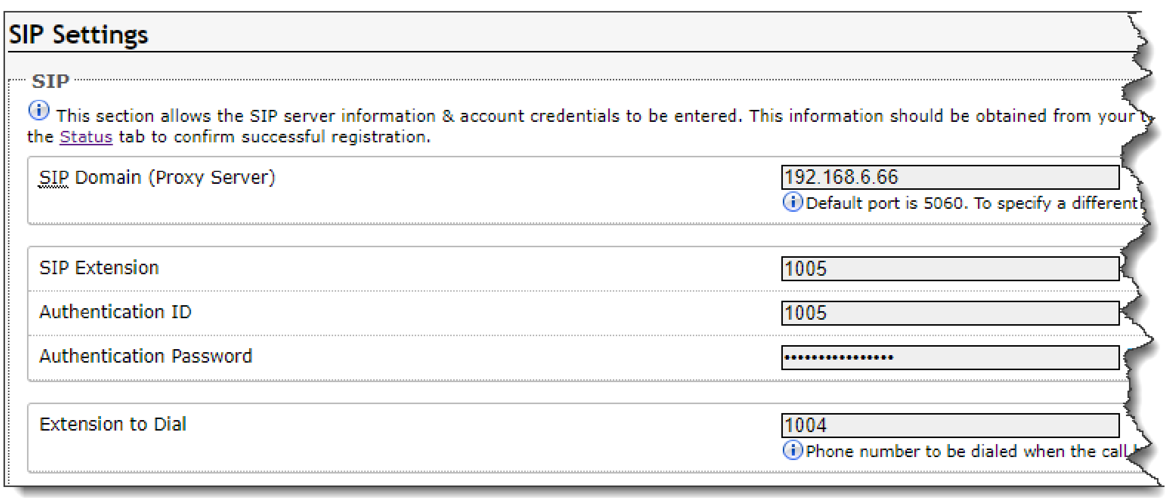


Algo 8201 SIP Intercom configuration

1. Access the Algo 8201 web interface, enter the password, and click **Login**.

   The default password is *algo*.

2. Go to **Basic Settings**, enter the following settings:



- **SIP Domain (Proxy Server)**: Enter the IP address of Yeastar S-Series VoIP PBX.
- **SIP Extension**: Enter the extension number.
- **Authentication ID**: Enter the extension's **Registration Name**.
- **Authentication Password**: Enter the extension's **Registration Password**.

- **Extension to Dial**: Enter an extension of Yeastar S-Series VoIP PBX. When a visitor presses the blue call button on Algo 8201, the extension will be dialed.

3. Click **Save**.
4. Go to **Status** to check the registration status.

   If the extension is registered successfully, the status will display "Successful".

Status

Device Name	sipintercom	
SIP Registration	**Successful**	(Extension 1005)
Call Status	Idle	

Result:

- When a visitor presses the blue call button on the Algo 8201 SIP Intercom, the extension 1004 will ring.