


# Configuration Notes

## Yeostar S-Series VoIP PBX & Algo 8180G2

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# Algo 8180G2 Test Report

This article is the Interoperability Test Report for Yeastar S-Series VoIP PBX and Algo 8180G2 SIP Audio Alerter.

## Tested equipment & software

Equipment	Firmware/Software Version
Algo 8180G2 Audio Alerter	1.7.2
Yeastar S300	30.10.0.59

## Summary of test focus

The following table shows a summary of the validated capabilities.

Feature	Test Result
<b>DUT Services</b>	
SIP Registration	PASS
Inbound Call: Ring Extension	PASS
Inbound Call: Page Extension	PASS
Inbound Call: Emergency Alert	PASS
Inbound Call: Multicast	PASS
Serviceability	PASS
<b>PBX Services</b>	
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.
- **Ring Extension:** This is the extension that will be called from Phone A or Phone B in order to trigger a “Ring” sound from the DUT. The DUT will expect to play ring tones, but will not answer the call.
- **Page Extension:** This is the extension that will be called from Phone A or Phone B in order to send paging audio to the DUT. The DUT will answer the call automatically.

- **Announcement Extension:** This is the extension that will be called from Phone A or Phone B in order to play selected announcement in the DUT. The DUT will answers the call automatically.
- **Call to Cancel Extension:** When the DUT is playing the emergency announcement, dial this extension from Phone A or Phone B to cancel the announcement.
- **Phone A:** A SIP compatible endpoint used to call the DUT.
- **Phone B:** A SIP compatible endpoint used to call the DUT and Phone A.
- **Phone C:** A SIP compatible endpoint used to register a Ring/Alert Extension.

## Test plan


### SIP Registration

The following test cases verify features related to the registration process with Yeastar S300.

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 multiple extensions.	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 - Ring Extension

The following test cases verify the Ring Extension with different Ring/Alert modes of the DUT.

Test Case	Expected Result	Test Result
<b>Ring/Alert Mode: Monitor "Ring" event on registered SIP extension</b>		
Dial Ring Extension from Phone A.	<ul style="list-style-type: none"> <li>• DUT answers the call automatically and plays the selected ring sound.</li> <li>• DUT continues to ring until the call is canceled by Phone A.</li> </ul>	PASS
<b>Ring/Alert Mode: Use "Subscribe/Notify" dialog event (RFC4235)</b>  <b>Note:</b> Ensure Phone C is registered with the Page Extension.		
Select <b>Alert Event</b> to <b>Ring</b> , and call Phone C (Ring Extension registered) from Phone A.	<ul style="list-style-type: none"> <li>• When Phone C is ringing, DUT plays ring sound.</li> <li>• When Phone C answers the call, DUT stops playing ring sound.</li> </ul>	PASS
Select <b>Alert Event</b> to <b>In-Use</b> , and call Phone C (Ring Extension registered) from Phone A.	<ul style="list-style-type: none"> <li>• When Phone C is ringing, DUT doesn't play ring sound.</li> <li>• When Phone C answers the call, DUT starts playing ring sound.</li> <li>• When Phone C ends the call, DUT stops playing ring sound.</li> </ul>	PASS
Select <b>Alert Event</b> to <b>Ring&amp;In-Use</b> , and call Phone C (Ring Extension registered) from Phone A.	<ul style="list-style-type: none"> <li>• When Phone C is ringing, DUT plays ring sound.</li> <li>• When Phone C answers the call, DUT replays the ring sound.</li> <li>• When Phone C ends the call, DUT stops playing ring sound.</li> </ul>	PASS
<b>Ring/Alert Mode: Use "Subscribe/Notify" presence event (RFC 3856/3863 PIDF)</b>		Not Supported

### Inbound Call - Page Extension

The following test cases verify the inbound paging feature of the DUT.

Test Case	Expected Result	Test Result
Dial Page Extension from Phone A.	<ul style="list-style-type: none"> <li>DUT answers and a one-way audio page is established from Phone A to UUT.</li> <li>The call is terminated by hanging up Phone A.</li> </ul>	PASS
Dial Page Extension from Phone A and mute/unmute the call.	<ul style="list-style-type: none"> <li>Mute: The DUT doesn't plays the audio from Phone A.</li> <li>Unmute: The DUT plays the audio from Phone A.</li> </ul>	PASS
When the Page Extension. is already in a call with Phone A, dial the Page Extension from Phone B.	<ul style="list-style-type: none"> <li>Phone B receives busy tone (DUT configured to allow only one simultaneous Page call).</li> </ul>	PASS
Dial Page Extension from Phone A and maintain the call for a period of time.	<ul style="list-style-type: none"> <li>The call remains up after the Session Refresh (REINVITE) is sent to the DUT.</li> </ul>	PASS

### Inbound Call - Emergency Alert

The following test cases verify the inbound Emergency Alert feature of the DUT.

Test Case	Expected Result	Test Result
Dial <b>Announcement</b> Extension from Phone A.	<ul style="list-style-type: none"> <li>DUT answers the call automatically and plays the selected announcement.</li> <li>DUT keeps playing the selected announcement even the call is canceled by Phone A.</li> </ul>	PASS
When DUT is playing an announcement, dial <b>Call to Cancel</b> Extension from Phone A.	<ul style="list-style-type: none"> <li>DUT stops playing the selected announcement.</li> </ul>	PASS

### Inbound Call: Multicast

The following test cases verify the Multicast Master/Sender feature on the DUT. The DUT acts as a multicast master.

Test Case	Expected Result	Test Result
<b>Prerequisite:</b> <ul style="list-style-type: none"> <li>• On the DUT, set the Multicast mode to Master/Sender and configure the multicast IP address and port.</li> <li>• On the DUT, register a Zone 1 Page Extension.</li> <li>• On the other phones, configure the same multicast IP address and port as the DUT to receive multicast.</li> </ul>		
Dial the Zone 1 Page Extension from Phone A.	<ul style="list-style-type: none"> <li>• DUT plays the selected Page Tone and plays the audio from Phone A.</li> <li>• The other phones plays the DUT selected Page Tone and plays the audio from Phone A.</li> </ul>	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
<b>Verify PBX feature: 1-Way Multicast Paging.</b> <b>Prerequisite:</b> <ul style="list-style-type: none"> <li>• On Yeastar S300, add a 1-Way Multicast Paging group.</li> <li>• On the DUT, set the Multicast mode to Slave/Receiver and configure the same multicast IP address and port as the Yeastar S300.</li> </ul>		
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
<b>Verify PBX feature: 1-Way Paging.</b> <b>Prerequisite:</b> <ul style="list-style-type: none"> <li>On Yeastar S300, add a 1-Way Paging group.</li> <li>On the DUT, register a Page Extension. The page extension is a member of the 1-Way paging group.</li> </ul>		
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
<b>Verify PBX feature: 2-Way Intercom.</b> <b>Prerequisite:</b> <ul style="list-style-type: none"> <li>On Yeastar S300, add a 2-Way Intercom group.</li> <li>On the DUT, register a Page Extension. The page extension is a member of the 2-Way Intercom group.</li> </ul>		
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

### Serviceability

The following test cases verify the serviceability of the DUT.

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



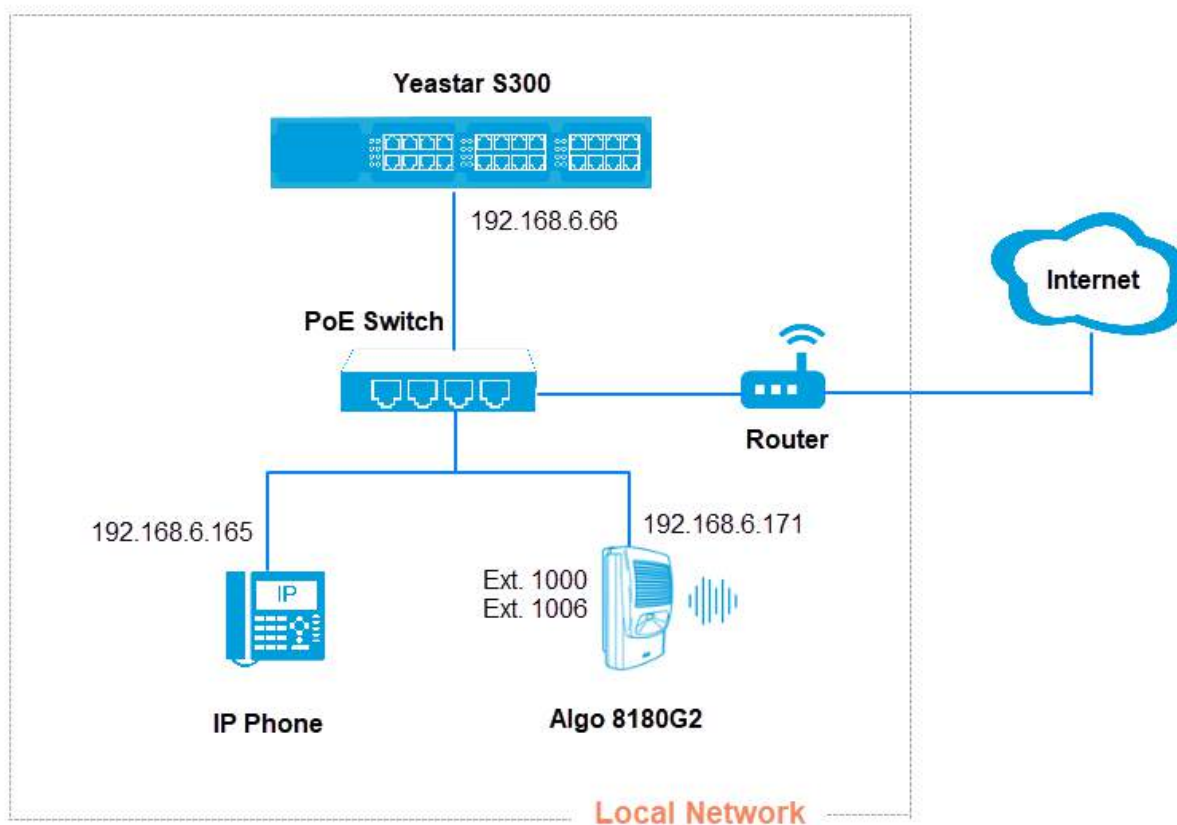
# Register Algo 8180G2 Audio Alerter with Yeastar S-Series VoIP PBX

This guide describes the configuration steps required for Algo 8180G2 SIP Audio Alerter to interoperate with Yeastar S-Series VoIP PBX.

Below is a guideline of how to register a Ring extension and a Page extension on Algo 8180G2. You may need to configure the other settings of the Algo 8180G2 Audio Alerter depending on your VoIP solution.

## Network Topology

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



## Yeastar S300 configuration

Add two SIP extensions on Yeastar S300, and provide the extension details in Algo 8101G2 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**.

The screenshot shows the 'Add Extension' configuration window with the 'Basic' tab selected. Under the 'General' section, the 'Type' is set to 'SIP' (highlighted with a red box), and 'IAX' and 'FXS' are unselected. The 'Extension' is set to '1000', 'Caller ID' is '1000', 'Registration Name' is '1000', and 'Caller ID name' is 'Algo-Ring'. 'Concurrent Registrations' is set to '1' and 'Registration Password' is masked with dots.

3. Add an extension, this extension will be registered as the Algo Page extension.
  - a. Click **Add**.
  - b. Leave the default settings or change the General settings according to your needs.
  - c. Click **Save** and **Apply**.

The screenshot shows the 'Add Extension' configuration window with the 'Basic' tab selected. Under the 'General' section, the 'Type' is set to 'SIP' (highlighted with a red box), and 'IAX' and 'FXS' are unselected. The 'Extension' is set to '1006', 'Caller ID' is '1006', 'Registration Name' is '1006', and 'Caller ID name' is 'Algo-Page'. 'Concurrent Registrations' is set to '1' and 'Registration Password' is masked with dots.

## Algo 8180G2 configuration

1. Access the Algo 8180G2 web interface, enter the password, and click **Login**.  
The default password is *algo*.
2. Go to **Basic Settings**→**SIP**, enter the following settings.

### SIP Settings

**SIP**

*(i)* This section allows the SIP server information & account credentials to be entered. This information should be obtained from your telephony provider. For more information, see the [Status](#) tab to confirm successful registration.

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

Ring/Alert Mode  Monitor "Ring" event on registered SIP extension  Use "Subscribe/Notify" dialog event (RFC 4235)  Use "Subscribe/Notify" presence event (RFC 4235)  None

Ring/Alert Extension

Authentication ID

Authentication Password

Display Name (Optional)

*(i)* The device will detect inbound ring events on this extension and play the alerting tone (and multicast if configured) until the extension is answered.

Base/Page Extension

Authentication ID

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).

- **SIP Domain (Proxy Server):** Enter the IP address of Yeastar S-Series VoIP PBX.
- **Ring/Alert Mode:** Select **Monitor "Ring" event on registered SIP extension.**
- **Ring/Alert Extension**

Enter the extension details of Ring/Alert extension.

- **Ring/Alert Extension:** Enter the extension number.
- **Authentication ID:** Enter the extension's **Registration Name.**
- **Authentication Password:** Enter the extension's **Registration Password.**

- **Base/Page Extension**

Enter the extension details of Base/Page extension.

- **Ring/Alert Extension:** Enter the extension number.
- **Authentication ID:** Enter the extension's **Registration Name.**
- **Authentication Password:** Enter the extension's **Registration Password.**

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	sipalerter		
SIP Registration	<b>Page Ring #1</b>	<b>Successful Successful</b>	(Extension 1006) (Extension 1000)

**Result:**

- When you dial the Ring/Alert extension 1000, the Algo 8180G will play ring tones until the you hang up the call.

 **Note:** The call is not answered.

- When you dial the Base/Page extension 1006, the Algo 8180G will answer the call automatically.