



## Avaya Solution & Interoperability Test Lab

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# Application Notes for Algo 8180 SIP Audio Alerter with Avaya IP Office - Issue 1.1

### Abstract

These Application Notes describe the configuration steps required for Algo 8180 SIP Audio Alerter to interoperate with Avaya IP Office. Algo 8180 SIP Audio Alerter is a SIP-based device that can register with Avaya IP Office as two separate SIP endpoints, one for loud ringing and one for voice paging.

Readers should pay attention to section 2, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

# 1. Introduction

These Application Notes describe the configuration steps required for Algo 8180 SIP Audio Alerter to interoperate with Avaya IP Office. Algo 8180 SIP Audio Alerter is a SIP-based device that can register with Avaya IP Office as two separate SIP endpoints, one for loud ringing and one for voice paging.

For loud ringing, Algo 8180 SIP Audio Alerter can be configured to ring whenever the associated desk phone receives an incoming call. The loud ringing is useful for users that require louder ringing than what is available from the desk phone. The simultaneous ringing at the desk phone and Algo 8180 SIP Audio Alerter is accomplished via the Avaya IP Office Mobile Twinning feature.

For voice paging, Algo 8180 SIP Audio Alerter can auto-answer an incoming call and allow the caller to broadcast audio over the Algo 8180 SIP Audio Alerter speaker.

## 2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were manually placed to the loud ringing and voice paging extensions, with call controls such as hold/resume, unattended, attended transfer and conference performed from the caller.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

### 2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The loud ringing feature testing included registration, internal and external caller, interactions with the voice paging extension, and interactions with desk phone features such as coverage, call forwarding, and do not disturb. The voice paging feature testing included registration, media shuffling, G.722, internal and external caller, interactions with the loud ringing extension, and interactions with caller actions such as drop, hold/reconnect, blind/attended transfer, and blind/attended conference.

The serviceability testing focused on verifying the ability of Algo 8180 SIP Audio Alerter to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cable to the device.

## 2.2. Test Results

The objectives outlined in **Section 2.1** were verified. All test cases passed, the following observations were made during the compliance testing:

- The call between Algo 8180 Page and Avaya phones (H.323, SIP, and digital) cannot be transferred in attended mode by Avaya phone to SIP phone. This feature is currently not supported on Algo 8180.
- A one way audio issue was encountered when Algo 8180 Page is configured with two-way Talkback. The issue will be fixed in the next release of Algo 8180, contact Algo for a firmware upgrade to correct this issue. With the current release only one-way alerting is supported, which is the primary function of this device.

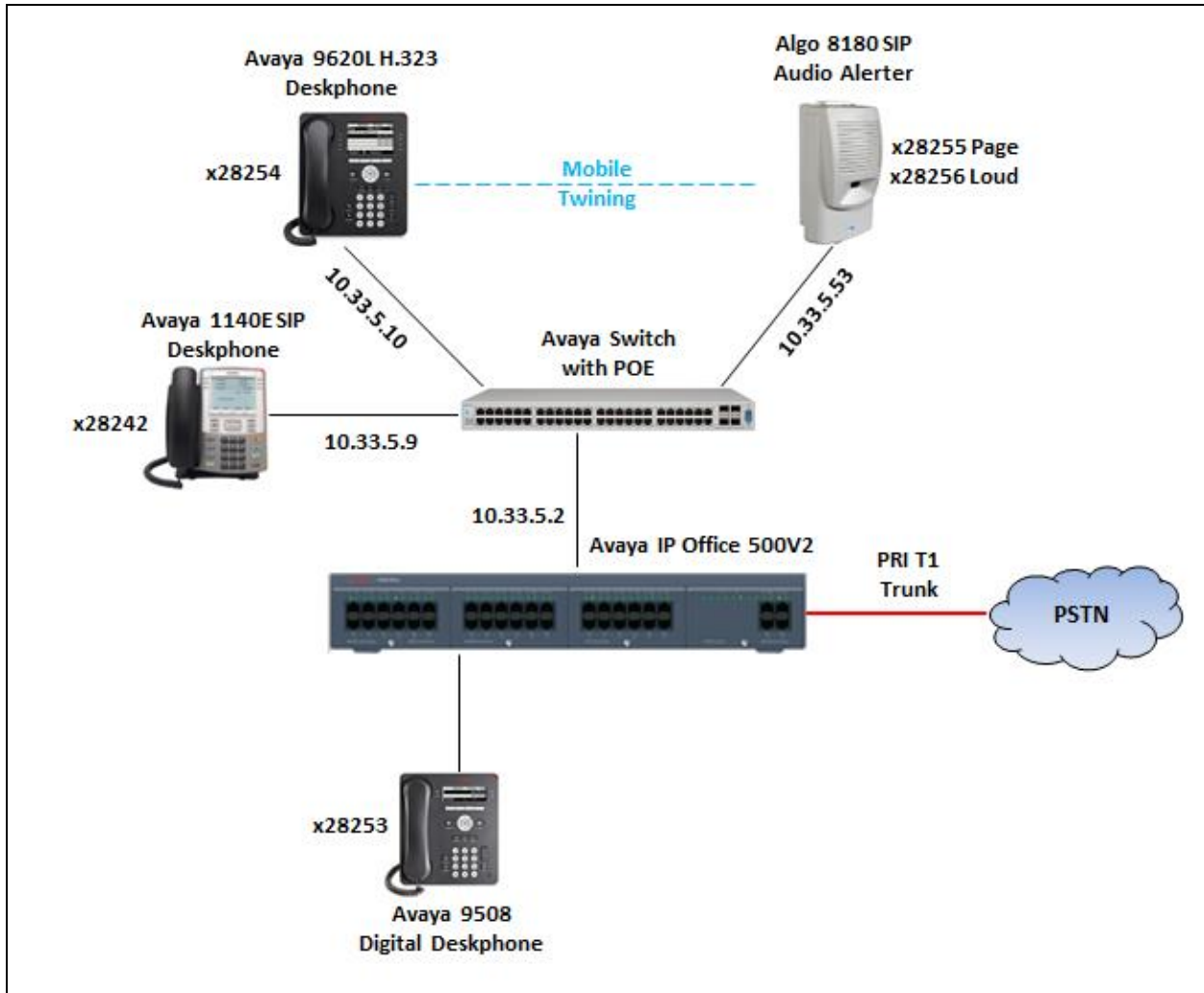
## 2.3. Support

Technical support on Algo 8180 SIP Audio Alerter can be obtained through the following:

- Phone: + 1 604 454 3792
- Web: <http://www.algosolutions.com/support/support.html>
- Email: [support@algosolutions.com](mailto:support@algosolutions.com)

### 3. Reference Configuration

**Figure 1** illustrates the test configuration used during the compliance testing between the Avaya IP Office and Algo 8180 SIP Audio Alerter. Algo 8180 communicated with IP Office through Avaya switch with Power over Ethernet (PoE) and registered with Avaya IP Office as two separate SIP endpoints, and the extensions used for the testing: one for Page and one for Loud Ringer. The PRI T1 trunk was also configured to connect from IP Office to PSTN for test cases off-net via PRI T1 trunk.



**Figure 1: Test Configuration Diagram**

## 4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software
Avaya IP Office 500V2	9.0.3.941
Avaya H.323 9620L IP Deskphone	3.220A
Avaya H.323 9650C IP Deskphone	3.220A
Avaya 1140E SIP Phone	4.3
Avaya 9508 Digital Phone	0.55
Algo 8180 SIP Audio Alerter	2.2

## 5. Configure Avaya IP Office

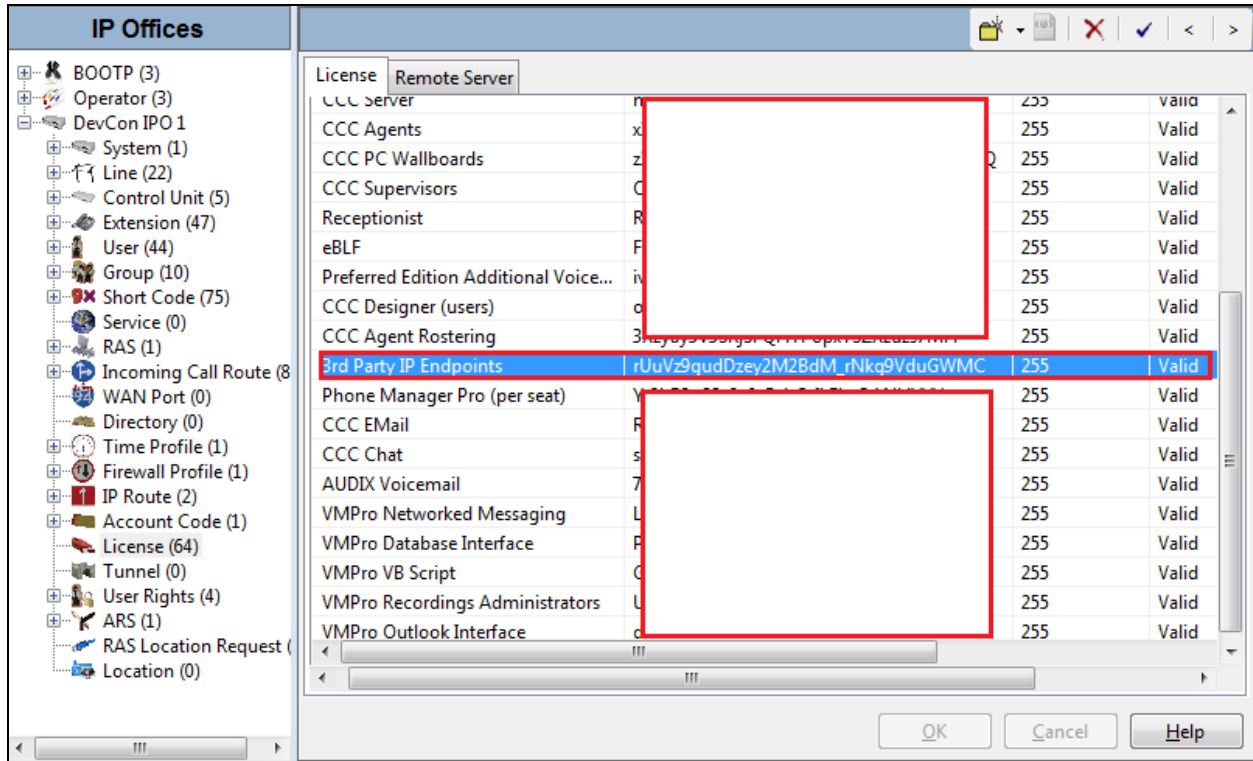
This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

- Verify IP Office license
- Obtain LAN IP address
- Administer SIP registrar
- Administer SIP extensions
- Administer SIP users
- Administer Internal Twinning

## 5.1. Verify IP Office License

From a PC running the Avaya IP Office Manager application, select **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application. Select the proper IP Office system, and log in using the appropriate credentials.

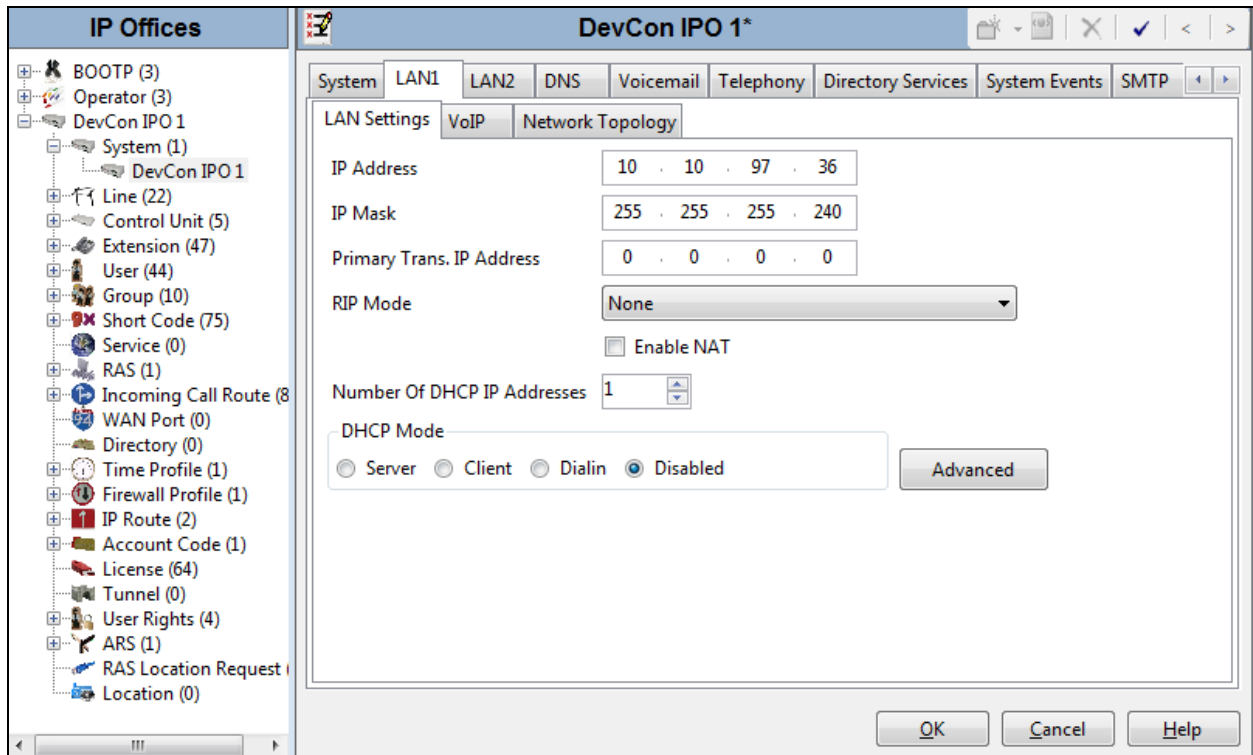
The **Avaya IP Office Manager** screen is displayed. From the configuration tree in the left pane, select **License**, the list of license displayed in the right panel. Verify that the **3rd Party IP Endpoints** status is “**Valid**”.



License	Remote Server	Seats	Status
CCC Server		255	valid
CCC Agents		255	Valid
CCC PC Wallboards		255	Valid
CCC Supervisors		255	Valid
Receptionist		255	Valid
eBLF		255	Valid
Preferred Edition Additional Voice...		255	Valid
CCC Designer (users)		255	Valid
CCC Agent Rostering		255	Valid
<b>3rd Party IP Endpoints</b>	rUuVz9qudDzey2M2BdM_rNkg9VduGWMC	255	<b>Valid</b>
Phone Manager Pro (per seat)		255	Valid
CCC EMail		255	Valid
CCC Chat		255	Valid
AUDIX Voicemail		255	Valid
VMPPro Networked Messaging		255	Valid
VMPPro Database Interface		255	Valid
VMPPro VB Script		255	Valid
VMPPro Recordings Administrators		255	Valid
VMPPro Outlook Interface		255	Valid

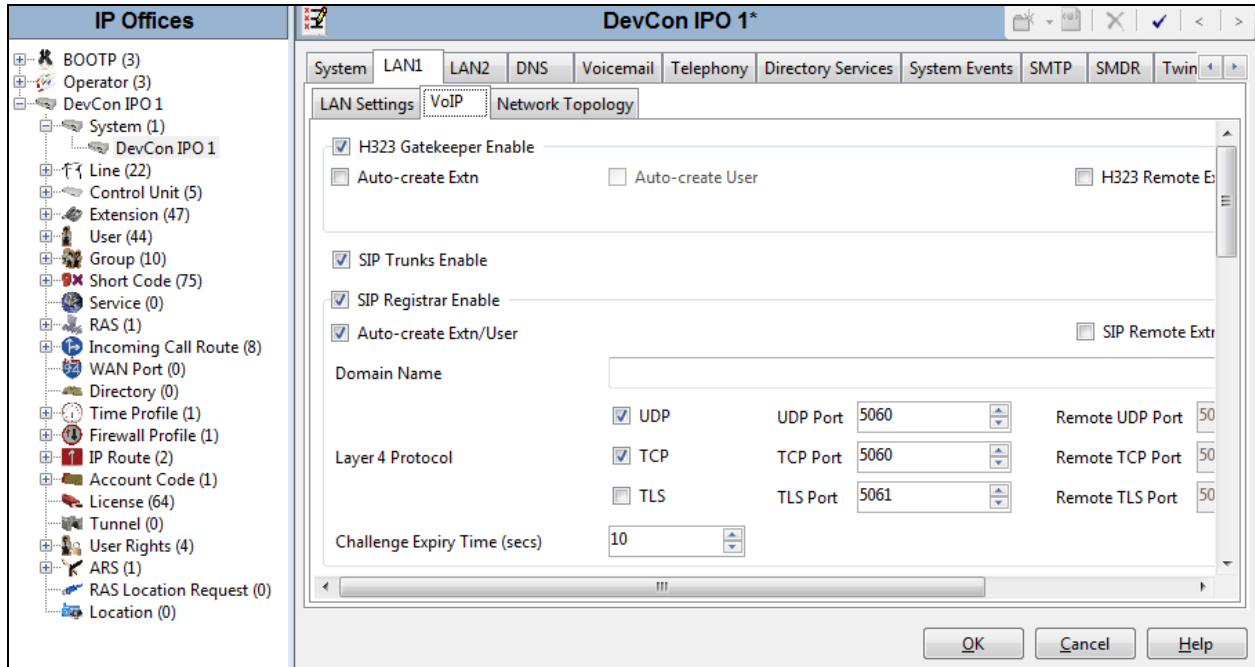
## 5.2. Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the DevCon IPO 1 screen in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure Algo. Note that IP Office can support SIP extensions on the **LAN1** and/or **LAN2** interfaces, and the compliance testing used the **LAN1** interface.



### 5.3. Administer SIP Registrar

Select the **VoIP** sub-tab. Make certain that **SIP Registrar Enable** is checked, as shown below. Enter a valid **Domain Name** for SIP endpoints to use for registration with IP Office. In the compliance testing, the **Domain Name** was left blank, so the SIP endpoints used the LAN IP address for registration.





## 5.4. Administer SIP Extensions

From the configuration tree in the left pane, right-click on **Extension**, and select **New** → **SIP Extension** (not shown) from the pop-up list to add a new SIP extension. For **Base Extension**, enter the voice paging extension “**28255**”. Retain the default values in the remaining fields.

The screenshot shows the 'SIP Extension: 8010 28255' configuration window. The left pane displays a tree view of system components, with 'Extension (47)' highlighted. The main configuration area includes the following fields:

- Extn: VoIP, T38 Fax
- Extension Id: 8010
- Base Extension: 28255
- Caller Display Type: On
- Reset Volume After Calls:
- Device Type: Unknown SIP device
- Location: Automatic
- Module: 0
- Port: 0
- Force Authorization:

Buttons at the bottom: OK, Cancel, Help.

Select the **VoIP** tab, and retain the default values in all fields. Repeat this section to add additional SIP extensions as desired. In the compliance testing, one SIP extension “**28255**” was created for voice paging, and one SIP extension “**28256**” was created for loud ringing.

The screenshot shows the 'SIP Extension: 8010 28255' configuration window with the 'VoIP' tab selected. The left pane is the same as the previous screenshot. The main configuration area includes the following fields:

- Extn: VoIP, T38 Fax
- IP Address: 0 . 0 . 0 . 0
- Codec Selection: System Default
- Reserve License: None
- Fax Transport Support: None

Codec Selection List:

- Unused: (empty)
- Selected: G.711 ULAW 64K, G.711 ALAW 64K, G.722 64K, G.729(a) 8K CS-ACELP, G.723.1 6K3 MP-MLQ

Checkboxes on the right:

- VoIP Silence Suppression
- Local Hold Music
- Allow Direct Media Path
- Re-invite Supported
- Codec Lockdown

Buttons at the bottom: OK, Cancel, Help.

## 5.5. Administer SIP User

From the configuration tree in the left pane, right-click on **User**, and select **New** (not shown) from the pop-up list. Enter desired values for Name and Full Name. For Extension, enter the first SIP base extension from **Section 5.4**.

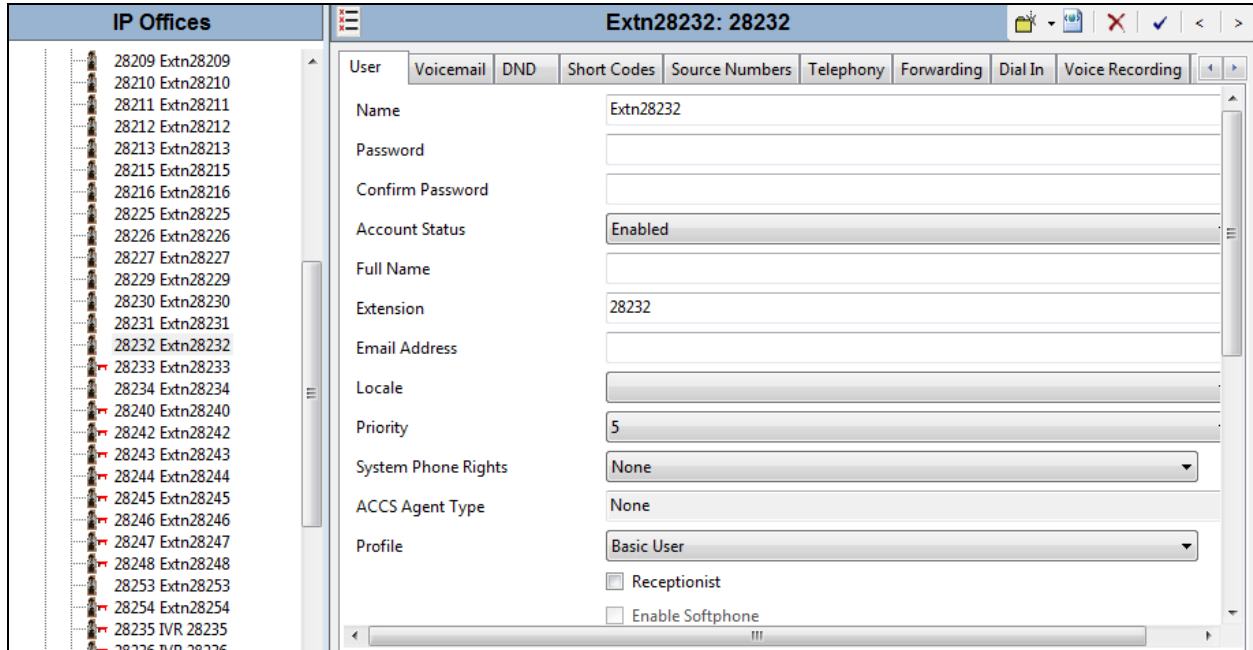
The screenshot shows the 'User' configuration page for extension 28255. The left pane shows the configuration tree with 'User (44)' selected. The main pane has tabs for 'User', 'Voicemail', 'DND', 'Short Codes', 'Source Numbers', 'Telephony', 'Forwarding', 'Dial In', and 'Voice Recording'. The 'User' tab is active, showing fields for Name (28255), Password, Confirm Password, Account Status (Enabled), Full Name (Algo 8180 Page), Extension (28255), Email Address, Locale, Priority (5), System Phone Rights (None), ACCS Agent Type (None), and Profile (Basic User). There are checkboxes for 'Receptionist' and 'Enable Softphone'.

Select the **Telephony** tab, followed by the **Supervisor Settings** sub-tab, and enter a desired **Login Code**. Repeat this section to add a new user for each SIP extension from **Section 5.4**. In the compliance testing, user “28256” was created for loud ringing, and user “28255” was created for voice paging.

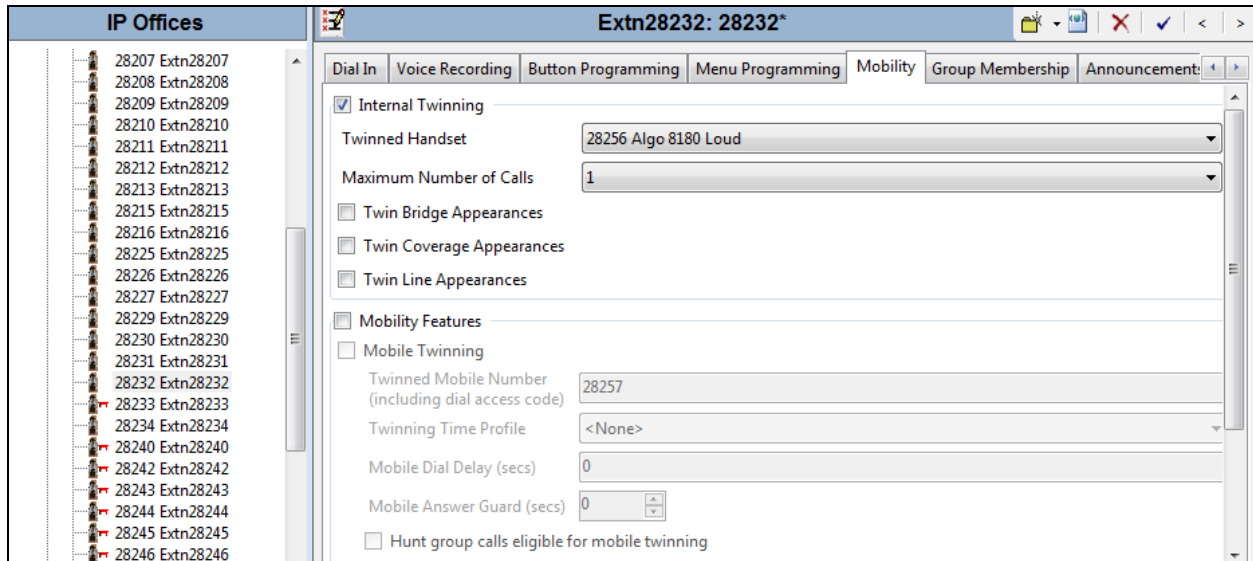
The screenshot shows the 'Supervisor Settings' sub-tab for extension 28255. The left pane shows the configuration tree with 'User (44)' selected. The main pane has tabs for 'Call Settings', 'Supervisor Settings', 'Multi-line Options', 'Call Log', and 'TUI'. The 'Supervisor Settings' tab is active, showing fields for Login Code (masked with dots), Login Idle Period (secs), Monitor Group (<None>), Coverage Group (<None>), Status on No-Answer (Logged On (No change)), and After Call Work Time (secs) (System Default (10)). There are checkboxes for Force Login, Force Account Code, Incoming Call Bar, Outgoing Call Bar, Inhibit Off-Switch Forward/Tran, Can Intrude, Cannot be Intruded, Can Trace Calls, CCR Agent, Automatic After Call Work, and Deny Auto Intercom Calls. There are also radio buttons for 'All Calls' and 'External Incoming'.

## 5.6. Administer Internal Twinning

From the configuration tree in the left pane, select the desk phone user that will be associated with the loud ringing user. In this case, desk phone user “**Extn28232**”.



Select the **Mobility** tab, and check **Internal Twinning**. For the Twinned Handset, select the strobe extension user from Section 5.5. Retain the default values in the remaining fields. Note that with the Internal Twinning configuration, the Algo extension 28257 will be acting like a secondary extension of the extension 28232 which is configured as primary and a direct call to the secondary will always get busy. This is the design intent of the Internal Twinning feature in IP Office. In order to place a direct call to the Algo strobe light extension, do not configure it twinned with a deskphone.



## 6. Configure Algo 8180 SIP Audio Alerter

This section provides the procedures for configuring Algo 8180 SIP Audio Alerter. The procedures include the following areas:

- Launch web interface
- Administer configuration

### 6.1. Launch Web Interface

Access the SIP Audio Alerter web-based interface by using the URL “http://ip-address” in an Internet browser window, where “ip-address” is the IP address of the SIP Audio Alerter. Note that the default IP address of the SIP Audio Alerter is 192.168.1.111. The **Welcome to the Algo 8180 SIP Audio Alerter Control Panel** screen is displayed, as shown below. Log in using the appropriate credentials.

**Welcome to the Algo 8180 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/8180reg>

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

**Login**

Password (default: **algo**)

## 6.2. Administer Algo 8180

Select **Basic Settings** → **SIP** from the top menu, to display the screen below. Configure the **SIP Settings** section toward the bottom of the screen as desired to match the configuration. Enter the following values for the specified fields, and retain the default values in the remaining fields.

- **Sip Domain (Proxy Server)** : The LAN IP address from **Section 5.2**
- **Ring/Alert Mode:** Select **Monitor “Ring” event on the registered SIP extension**
- **Page Function:** Select **Enabled**
- **Ring Extension:** Enter the loud ringing SIP base extension from **Section 5.4**
- **Authentication ID:** Enter the loud ringing SIP user name from **Section 5.5**
- **Ring Password:** Enter the loud ringing SIP user login code from **Section 5.5**
- **Page Extension:** Enter the voice paging SIP base extension from **Section 5.4**
- **Page Auth ID:** Enter the voice paging SIP user name from **Section 5.5**
- **Page Password:** Enter the voice paging SIP user login code from **Section 5.5**

The screenshot shows the 'SIP Settings' configuration page. At the top, there are navigation tabs: 'Status', 'Basic Settings' (selected), 'Advanced Settings', 'System', and 'Logout'. Below these are sub-tabs: 'SIP', 'Ring', 'Page', 'Features', and 'Multicast'. The main heading is 'SIP Settings', followed by the text 'Here you can configure the basic SIP settings.' The configuration area is divided into several sections:

- SIP**:
  - SIP Domain (Proxy Server)**: Text input field containing '10.10.97.36'. A help icon and text below state: '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
  - NoneA help icon and text below state: 'Server support required on "Subscribe/Notify" events'
- Page Function**: Radio button options:
  - Enabled
  - Disabled
- Allow SIP REGISTER**: Radio button options:
  - Enabled
  - DisabledA help icon and text below state: 'Disable it when connected in trunk mode'

Below these are two groups of fields, each with a help icon:

- Ring Extension**: Text input field containing '28256'
- Authentication ID**: Text input field containing '28256'
- Authentication Password**: Password input field with masked characters '•••••'
- Page Extension**: Text input field containing '28255'
- Authentication ID**: Text input field containing '28255'
- Authentication Password**: Password input field with masked characters '•••••'

Select **Basic Settings** → **Page** from the top menu, to display the **Page Settings** below. In the **Page Mode**, the **One-way** option is selected by default and only this mode is currently supported in the 8180 during the compliance test. The Talkback mode should not be selected because the one-way audio issue and it will be supported in the next release.

The screenshot shows the ALGO 8180 SIP Audio Alerter Control Panel. The top navigation bar includes 'Status', 'Basic Settings', 'Advanced Settings', 'System', and 'Logout'. Below this, a sub-menu shows 'SIP', 'Ring', 'Page', 'Features', and 'Multicast'. The main content area is titled 'Page Settings' and contains the following configuration options:

Page Volume	7	Apply
Page Mode	<input checked="" type="radio"/> One-way <input type="radio"/> Talkback <small><b>i</b> Talkback mode allows bidirectional half-duplex communication. G.722 support should be disabled if using talkback mode.</small>	
Page Timeout	None	
Page Tone	page-notif.wav	
G.722 Support	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled <small><b>i</b> Only for playback.</small>	
Automatic Gain Control (AGC)	<input type="radio"/> Enabled <input checked="" type="radio"/> Disabled	
Audio Delay (milliseconds)	0	

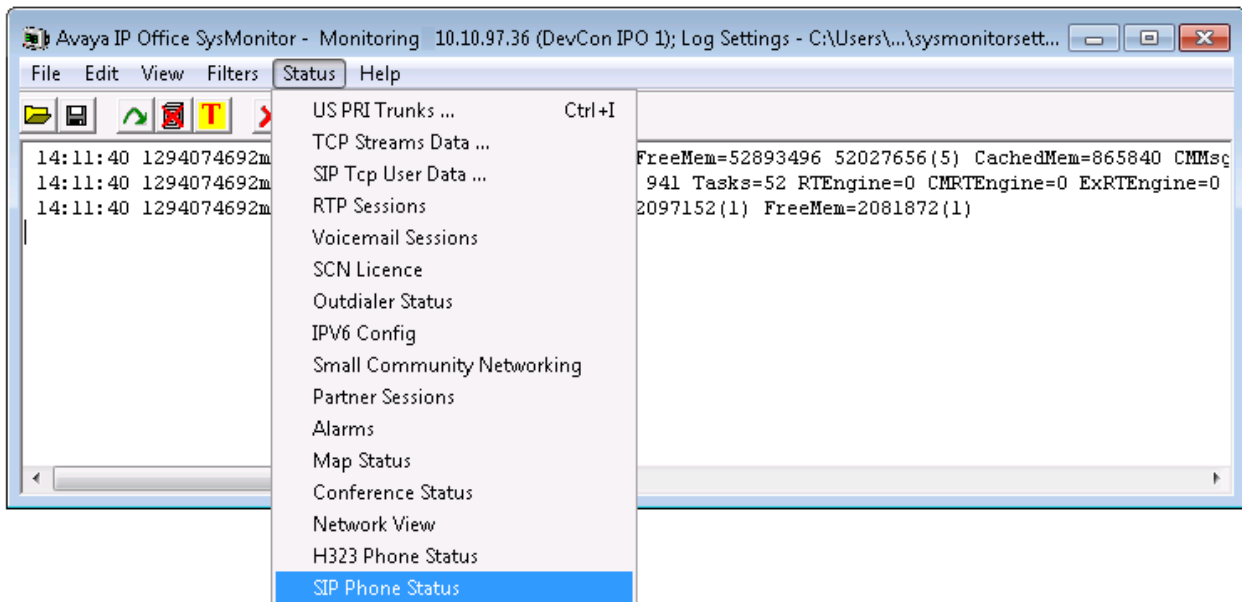
At the bottom right of the settings area is a 'Save' button with a green checkmark icon. The footer of the page reads '© Copyright 2014 Algo Communication Products Ltd.'

## 7. Verification Steps

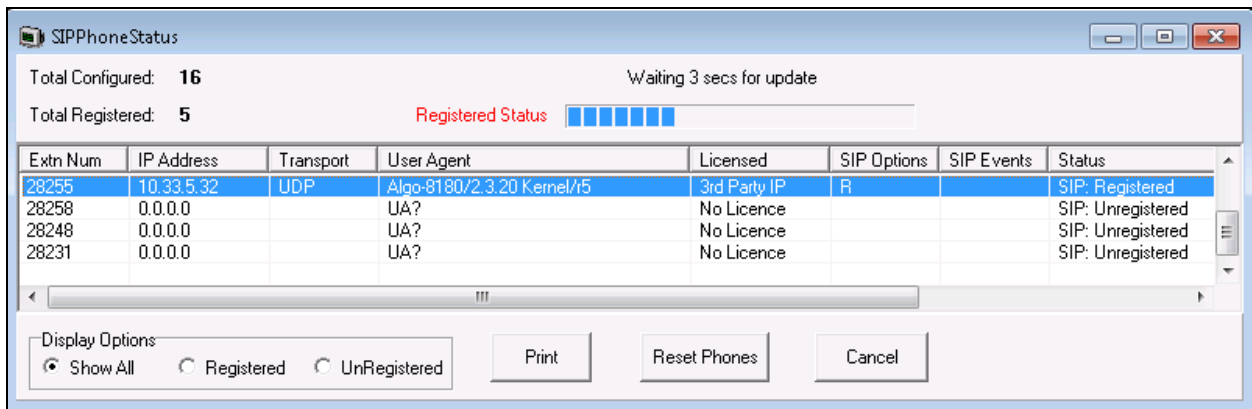
This section provides the tests that can be performed to verify proper configuration of Avaya IP Office and Algo 8180 SIP Audio Alerter.

### 7.1. Verify Avaya IP Office

From a PC running the Avaya IP Office Monitor application, select **Start → Programs → IP Office → System Monitor** to launch the application. The **Avaya IP Office SysMonitor** screen is displayed, as shown below. Select **Status → SIP Phone Status** from the top menu.



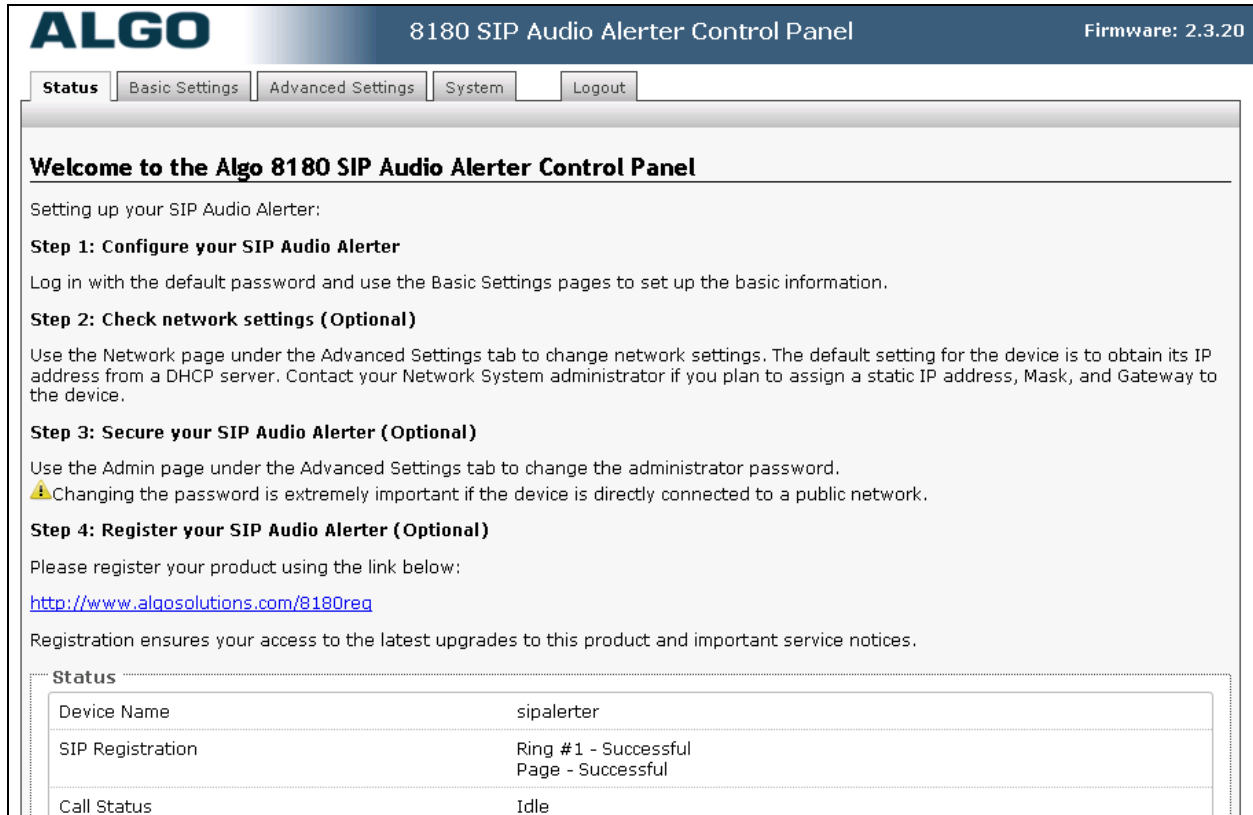
The **SIPPhoneStatus** screen is displayed. Verify that there is an entry for each SIP extension from **Section 5.4**, that the **User Agent** is “Algo-8180”, and that the **Status** is “SIP: Registered”, as shown below.





## 7.2. Verify Algo 8180 SIP Audio Alerter

From the Algo 8180 SIP Audio Alerter web-based interface, select **Status** from the top menu. Verify that **SIP Registration** displays “Ring – Successful” and “Page – Successful”, as shown below.



**ALGO** 8180 SIP Audio Alerter Control Panel Firmware: 2.3.20

Status Basic Settings Advanced Settings System Logout

### Welcome to the Algo 8180 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/8180reg>  
Registration ensures your access to the latest upgrades to this product and important service notices.

Status	
Device Name	sipalerter
SIP Registration	Ring #1 - Successful Page - Successful
Call Status	Idle

The following tests were conducted to verify the solution between the Algo 8180 SIP Audio Alerter and Avaya IPO.

- Verify that the incoming call to the twinning extension on the IPO rings the 8180 and the 8180 stops ringing if the twinning extension answers the call
- Verify that the incoming call to the 8180 Page is automatically answered with clear audio path
- Verify that the telephone that places the incoming call to the 8180 can do conference, transfer, mute, un-mute and provide busy tone if it is on another call
- Verify that the solution works with different Avaya clients (e.g. digital, analog, IP etc).
- Verify that 8180 goes into an idle state when the call is completed
- Verify that the 8180 re-registers without issues if the Ethernet cable is unplugged and plugged back in

## 8. Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 2.1**, with some exceptions outlined in **Section 2.2**. Algo 8180 SIP Audio Alerter version 2.3.0 is considered to be in compliance with Avaya IP Office System Release 9.0.

## 9. Additional References

Product documentation for the Avaya IP Office products may be found at:

<https://support.avaya.com/css/Products/>

Product documentation for the Algo 8180 SIP Audio Alerter products may be found at:

<http://partneraccess.spectralink.com/products/wi-fi/spectralink-8000-portfolio/pivot-87-series>

Avaya IP Office Documents:

[1] IP Office 9.0 Installation, Document number 15-601042 Issue 28, 11 October 2013

[2] IP Office 9.0 Manager 9.0, Document number 15-601011 Issue 9.01, 09 September 2013

[3] IP Office 9.0 Administering Voicemail Pro, Document number 15-601063 Issue 9.0 Release 1.0, September 2013

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