



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Configuring Avtec Scout VoIP Console with Avaya Aura® Communication Manager 6.3 and Avaya Aura® Session Manager 6.3 – Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required for Avtec Scout VoIP Console to successfully interoperate with Avaya Aura® Communication Manager 6.3 and Avaya Aura® Session Manager 6.3. The Avtec Scout VoIP Console is a SIP-based system that integrates with Avaya Aura® Communication Manager and Avaya Aura® Session Manager as SIP endpoints.

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 the Avtec Scout VoIP Console to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Avtec Scout VoIP Console is a SIP-based system that integrates with Avaya Aura® Communication Manager and Avaya Aura® Session Manager as SIP endpoints. The Avtec Scout VoIP Console allows for the configuration of multiple SIP extensions that can then Register with Session Manager and become an extension on the enterprise system.

## 2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing.

The Avtec Scout VoIP Console was installed and configured on a desktop computer running Microsoft Windows 7. Multiple extensions were then configured to Register with Session Manager, which was configured to accept the Registrations from these endpoints. These Application Notes cover the setup and configuration described here.

The feature testing covered SIP registration, basic calls, simultaneous calls, display verification, media shuffling, audio codec negotiation, transfers and conferencing. The feature test cases were performed manually.

The serviceability testing focused on verifying the ability of the Avtec Scout VoIP Console device to recover from adverse conditions, such as disconnecting and reconnecting the LAN cable to the Avtec Scout VoIP Console. Additionally, the Communication Manager and Session Manager servers were each individually rebooted to verify the Avtec Scout VoIP Console device was able to properly register and function normally after each server recovered.

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 for 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 plan included feature and serviceability test cases.

The feature testing covered SIP registration, basic calls, simultaneous calls, display verification, media shuffling, audio codec negotiation, transfers and conferencing. Various endpoints were configured for the Avtec Scout VoIP Console system to test connectivity to other extensions and out to the PSTN.

The serviceability testing focused on verifying the ability of the Avtec Scout VoIP Console to recover from adverse conditions, such as disconnecting and reconnecting the Ethernet cable to the device, rebooting Communication Manager, and rebooting Session Manager.

## 2.2. Test Results

All feature test cases were executed and passed with the following observations:

- During the compliance test, it was observed that for Consultative Transfers there was a one-way audio condition during the consult portion of the call. At the time of writing these Application Notes, this issue was being investigated. In the meantime, Avtec can remove the consultative button from their desktop to prevent this issue from occurring. Customers can use blind transfer as a workaround.

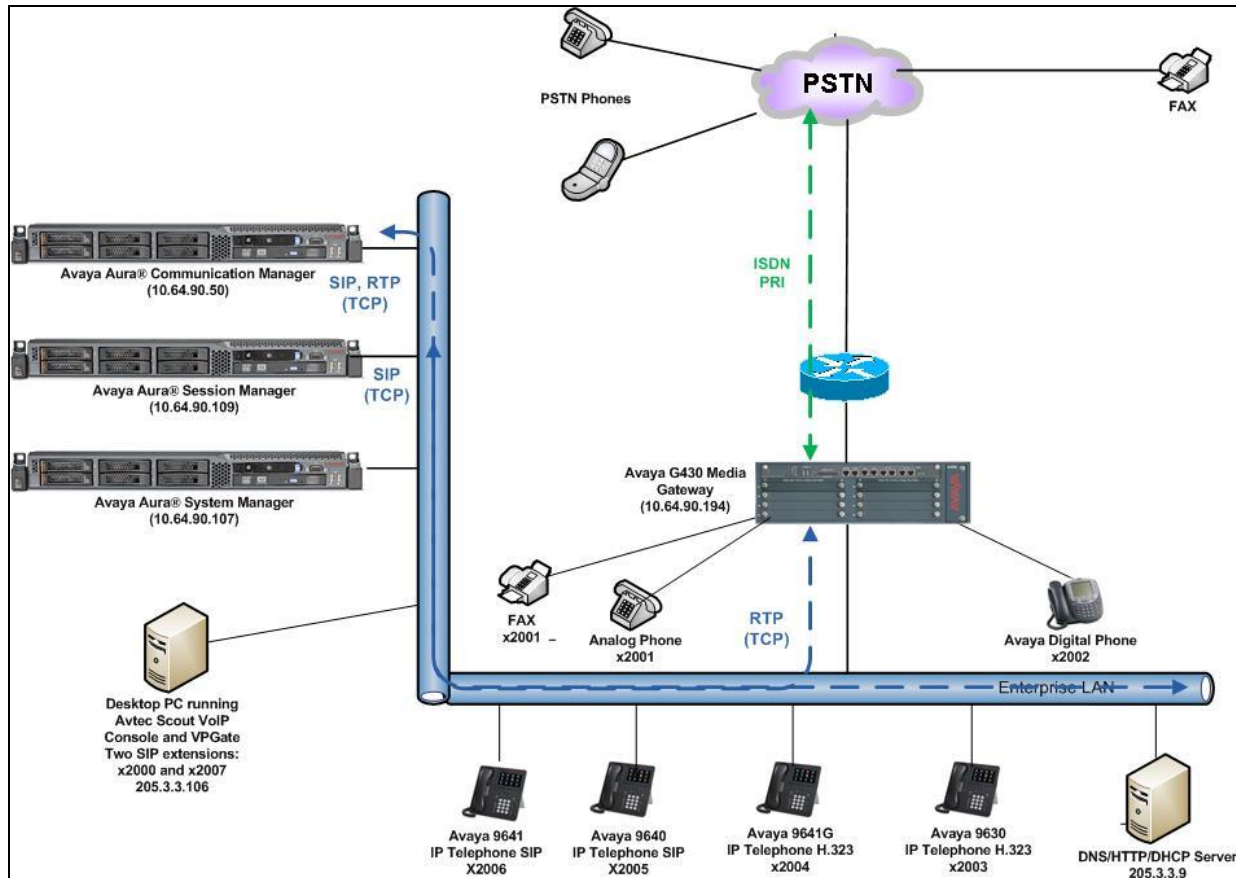
## 2.3. Support

Technical support for Avtec can be obtained through the following:

- **Phone:** 1-800-545-3034
- **Email:** [CustomerSupport@avtecinc.com](mailto:CustomerSupport@avtecinc.com)

### 3. Reference Configuration

The Avtec Scout VoIP Console solution consists of the Avtec Scout VoIP Console and the Avtec VPGate applications running on a Windows PC / Server. The Avtec system can register multiple SIP endpoints with Avaya Aura® Session Manager each of which are displayed on the Console. Each user / extension has their own separate Console from which they can receive and place calls to / from other endpoints. These endpoints can place and receive calls from both Avaya and Avtec devices with various supported features as listed above in **Section 1**. The reference configuration used for the compliance test is shown in **Figure 1** below.



**Figure 1: Avtec Scout VoIP Console with Avaya Aura® Communication Manager and Avaya Aura® Session Manager and the Avaya G430 Gateway**

## 4. Equipment and Software Validated

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

| Equipment   | Software  |
|---|---|
| Avaya S8800 Server  | Avaya Aura® Communication Manager 6.3,<br>R016x.03.0.124.0<br>Avaya Aura® System Platform: 6.3.0.0.18002                    |
| Avaya S8800 Server  | Avaya Aura® System Manager 6.3.0 – FP2<br>Build No. - 6.3.0.8.5682-6.3.8.1627<br>Avaya Aura® System Platform: 6.3.0.0.18002 |
| Avaya S8800 Server  | Avaya Aura® Session Manager 6.3,<br>6.3.2.0.632023  |
| Avaya G430 Gateway with ISDN PRI<br>circuit to the PSTN   |   |
| Avaya 96xx Series IP Telephones <ul style="list-style-type: none"><li>• 96x0 H.323</li><li>• 96x1 H.323</li><li>• 96x0 SIP</li><li>• 96x1 SIP</li></ul> | 3.1<br>6.2.1.19<br>2.6.9.1<br>6.2.1.26  |
| Avtec Project Manager   | 3.1.12.43   |

## 5. Configure Avaya Aura® Communication Manager

The detailed administration of basic connectivity between Avaya Aura® Communication Manager and Avaya Aura® Session Manager is not the focus of these Application Notes and will not be described. For administration of basic connectivity between Avaya Aura® Communication Manager and Avaya Aura® Session Manager, refer to the appropriate documentation listed in **Section 10**. This section provides the procedures for the following:

- Verify Avaya Aura® Communication Manager License

### 5.1. Verify Communication Manager License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the **display system-parameters customer-options** command to verify that there is sufficient capacity for SIP stations by comparing the **Maximum Off-PBX Telephones - OPS** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the number of SIP extensions required for the Avtec Scout VoIP Console system.

```
display system-parameters customer-options                               Page 1 of 11
                                OPTIONAL FEATURES

G3 Version: V16                               Software Package: Enterprise
Location: 2                                   System ID (SID): 1
Platform: 28                                 Module ID (MID): 1

                                USED
                                Platform Maximum Ports: 65000 38
                                Maximum Stations: 41000 8
                                Maximum XMOBILE Stations: 41000 0
Maximum Off-PBX Telephones - EC500: 41000 0
Maximum Off-PBX Telephones - OPS: 41000 4
Maximum Off-PBX Telephones - PBFMC: 41000 0
Maximum Off-PBX Telephones - PVFMC: 41000 0
Maximum Off-PBX Telephones - SCCAN: 0 0
Maximum Survivable Processors: 313 0

(NOTE: You must logoff & login to effect the permission changes.)
```

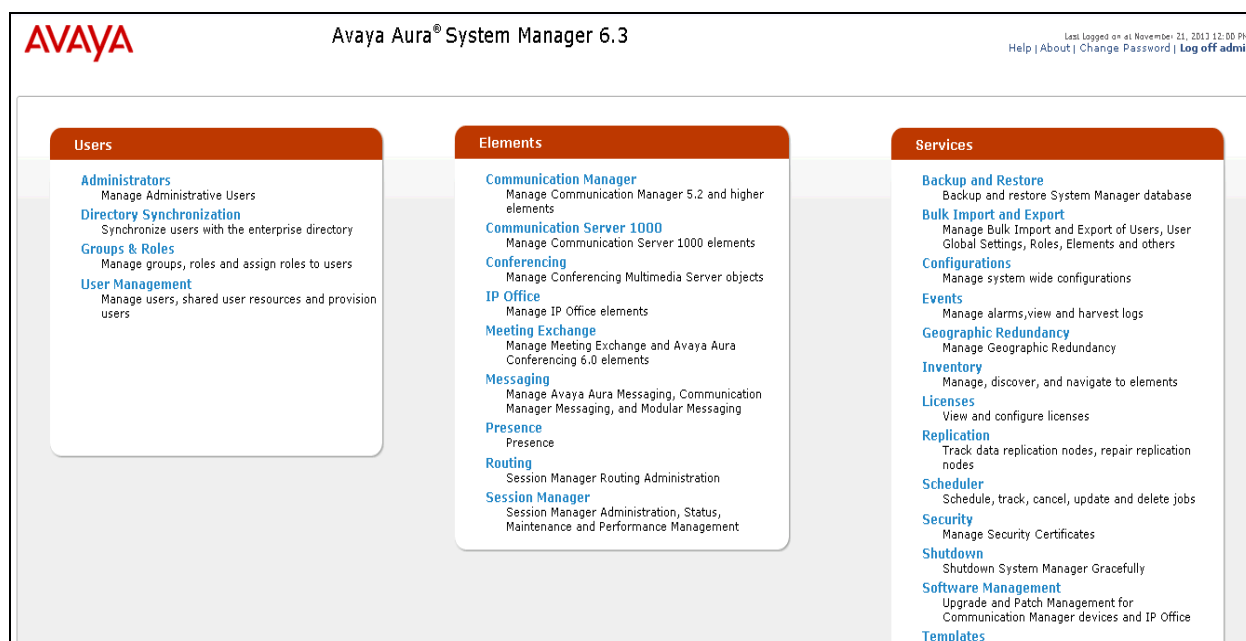
## 6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Avaya Aura® Session Manager. The procedures include the following areas:

- Launch Session Manager administration interface
- Administer endpoints
- Administer users

### 6.1. Launch Avaya Aura® System Manager Administration Interface

Configuration of Session Manager is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager, using the URL “https://<ip-address>/SMGR”, where “<ip-address>” is the IP address of System Manager. Log in using the appropriate credentials. The screen shown below is displayed.



## 6.2. Administer Users

Users must be added to Session Manager that correspond to SIP extensions. From the menu, navigate to **User Management**→**Manage Users** as shown below. Select the **New** button from the right pane.

Avaya Aura® System Manager 6.3

Last Logged on at November 21, 2013 12:00 PM  
Help | About | Change Password | Log off admin

User Management

Home / Users / User Management / Manage Users

**User Management**

Users

View Edit **New** Duplicate Delete More Actions

8 Items Refresh Show ALL Filter: Enable

| <input type="checkbox"/> | Last Name      | First Name | Display Name          | Login Name         | SIP Handle | Last Login                          |
|--------------------------|----------------|------------|-----------------------|--------------------|------------|-------------------------------------|
| <input type="checkbox"/> | admin          | admin      | Default Administrator | admin              |            | December 30, 2013 2:22:49 PM -07:00 |
| <input type="checkbox"/> | Scout31        | Avtec      | Scout31, Avtec        | 2000@avayalab2.com | 2000       |                                     |
| <input type="checkbox"/> | Scout31 Test 2 | Avtec      | Scout31 Test 2, Avtec | 2007@avayalab2.com | 2007       |                                     |
| <input type="checkbox"/> | SIP            | 2005       | SIP, 2005             | 2005@avayalab2.com | 2005       |                                     |
| <input type="checkbox"/> | SIP            | 2006       | SIP, 2006             | 2006@avayalab2.com | 2006       |                                     |
| <input type="checkbox"/> | SIP            | 5055       | SIP, 5055             | 5055@avayalab2.com | 5055       |                                     |
| <input type="checkbox"/> | SIP CM62       | 13001      | SIP CM62, 13001       | 13001@avayalab.com | 13001      |                                     |
| <input type="checkbox"/> | SIP CM62       | 13002      | SIP CM62, 13002       | 13002@avayalab.com | 13002      |                                     |

Select : All, None



## 6.2.1. Identity Tab

Enter the following values for the specified fields, all other fields can be left at default values or configured according to customer requirements.

- **Last Name:** Enter a descriptive name
- **First Name:** Enter a descriptive name
- **Login Name:** Enter the extension URI including SIP domain
- **Authentication Type:** Leave this set to the default value of **Basic**
- **Password:** Enter the System Manager password (not shown)
- **Confirm Password:** Enter the System Manager password again to confirm (not shown)

The screenshot shows the 'User Profile Edit' interface for the user '2000@avayalab2.com'. The 'Identity' tab is selected, displaying various user attributes. The 'Last Name' is 'Scout31' and the 'First Name' is 'Avtec'. The 'Authentication Type' is set to 'Basic'. The 'Update Time' is 'November 19, 2013 10:'. The 'Login Name' is '2000@avayalab2.com'. The 'Source' is 'local'. The 'Localized Display Name' and 'Endpoint Display Name' are both 'Scout31, Avtec'. The 'Language Preference' is 'English (United States)' and the 'Time Zone' is '(-7:0)Mountain Time (US & C)'. There are also fields for 'Employee ID', 'Department', and 'Company'. The page includes a left sidebar with navigation links and a top header with the Avaya logo and system version.

Click **Commit & Continue**. Repeat this procedure for each of the Avtec Scout VoIP Console extensions.

## 6.2.2. Communication Profile Tab

Click on the **Communication Profile** tab at the top. Enter the following values for the specified fields below.

- **Communication Profile Password:** Enter a password that will be the security code for this particular SIP extension
- **Confirm Password:** Enter the password again to confirm

Avaya Aura® System Manager 6.3

Last Logged on at November 21, 2013 12:00 PM  
Help | About | Change Password | Log off admin

User Management \* Home

Home / Users / User Management / Manage Users

Status

**New User Profile** [Commit & Continue] [Commit] [Cancel]

Identity \* Communication Profile \* Membership Contacts

Communication Profile

Communication Profile Password: [Text Box]

Confirm Password: [Text Box]

[New] [Delete] [Done] [Cancel]

| Name    |
|---------|
| Primary |

Select : None

Under **Communication Address**, click on **New**. Enter the following values for the specified fields, all other fields can be left at default values or configured according to customer requirements.

- **Type:** Choose **Avaya SIP** from the drop-down menu
- **Fully Qualified Address:** Enter the SIP extension and choose the appropriate SIP domain from the drop-down menu

[New] [Delete] [Done] [Cancel]

| Name    |
|---------|
| Primary |

Select : None

\* Name: Primary [Text Box]

Default : ☒

Communication Address

[New] [Edit] [Delete]

| Type      | Handle | Domain        |
|-----------|--------|---------------|
| Avaya SIP | 2000   | avayalab2.com |

Select : All, None

Type: Avaya SIP [Dropdown]

\* Fully Qualified Address: 2000 [Text Box] @ avayalab2.com [Dropdown]

[Add] [Cancel]

☒ Session Manager Profile

Click the **Add** button when finished.

Expand **Session Manager Profile**. Enter the following values for the specified fields, all other fields can be left at default values or configured according to customer requirements.

- **Primary Session Manager** Select the Session Manager instance that should be used as the home server for the currently displayed Communication Profile.
- **Origination Sequence** Select an Application Sequence that will be invoked when calls are routed *from* this user.
- **Termination Sequence** Select an Application Sequence that will be invoked when calls are routed *to* this user.
- **Home Location** Select the Home Location of this user.

☒ Session Manager Profile

SIP Registration

\* Primary Session Manager
 

ASM63

| Primary | Secondary | Maximum |
|---------|-----------|---------|
| 7       | 0         | 7       |

Secondary Session Manager
 

(None)

Survivability Server
 

(None)

Max. Simultaneous Devices
 

1

Block New Registration When Maximum Registrations Active?
 ☐

Application Sequences

Origination Sequence
 

App\_to\_CM63

Termination Sequence
 

App\_to\_CM63

Call Routing Settings

\* Home Location
 

SessionManager

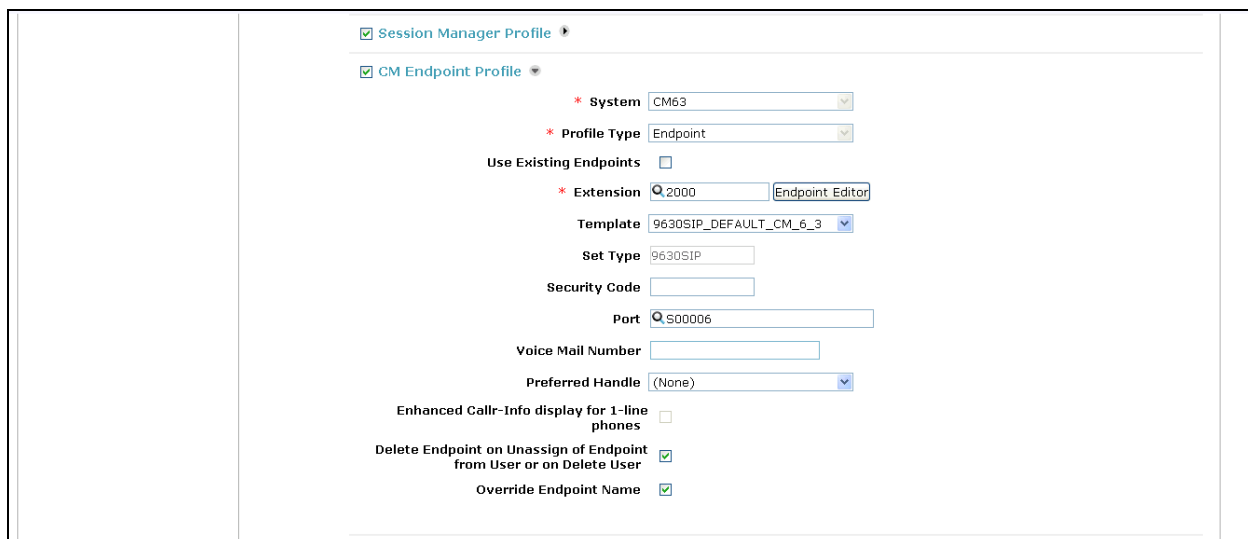
Conference Factory Set
 

(None)

☒ CM Endpoint Profile

Expand **CM Endpoint Profile**. The Avtec endpoints were defined using the template for the Avaya 9630 SIP phone during compliance testing. Enter the following values for the specified fields, all other fields can be left at default values or configured according to customer requirements.

- **System** Select the Communication Manager on which the endpoint exists or will be created.
- **Profile Type** Set to **Endpoint**.
- **Use Existing Endpoints** Only check this box to use an endpoint that was previously administered on Communication Manager. When left unchecked, a corresponding endpoint will be created on Communication Manager.
- **Extension** Enter the extension of the endpoint that you want to associate with this user.
- **Template** Select an appropriate template. For the compliance test, the Avaya 9630 SIP phone template was used.
- **Security Code** Enter the security code to be used by the Avtec endpoint when registering to Session Manager.
- **Port** The Port field is automatically filled in.



Session Manager Profile

CM Endpoint Profile

\* System CM63

\* Profile Type Endpoint

Use Existing Endpoints ☐

\* Extension 2000 Endpoint Editor

Template 9630SIP\_DEFAULT\_CM\_6\_3

Set Type 9630SIP

Security Code

Port S00006

Voice Mail Number

Preferred Handle (None)

Enhanced Callr-Info display for 1-line phones ☐

Delete Endpoint on Unassign of Endpoint from User or on Delete User ☒

Override Endpoint Name ☒

Click the **Commit & Continue** button. Repeat the procedures in this section to add a user for each required endpoint.

## 7. Configure Avtec Scout VoIP Console

The Avtec Scout VoIP Console solution consists of two main components -- the actual Scout console and the VPGate (VoIP Gateway) back-end piece. Each is a separate piece of software that must be installed and configured. This section describes how to create endpoints in the Avtec Scout VoIP Console to connect to Avaya's system. Complete installation and configuration details for the Avtec Scout VoIP Console and VPGate may be found in the Avtec documentation listed in **Section 10**. The procedures include the following areas:

- Configure Endpoints

### 7.1. Configure Avtec Scout VoIP Console Endpoints

To log in to VPGate, bring up a web browser and navigate to <http://localhost:3082/>, or if you want to connect to it from a different computer, browse to [http://\[computer name or IP\]:3082/](http://[computer name or IP]:3082/), using the name or IP address of the computer that is running the VPGate software (not shown). On the tabs at the top of the page, select **Endpoints**. This will bring up the **Endpoint Summary** page. Press the **Add** button to create the new endpoint as shown below.

The screenshot displays the VPGate Configuration web interface. The main content area is titled "Endpoint Summary" and features a table of endpoints. The table has columns for Name, Service State, Connection, Status, Priority, and Primary VPGate. The endpoints listed are Avtec\_1, Avtec\_2, Avtec\_3, Outpost\_1a, Outpost\_1b, Outpost\_2a, and Outpost\_2b. Avtec\_1 and Avtec\_2 are in an "Available" state, while the others are "Unavailable". The interface also includes a "Project Explorer" on the left and a "Properties" pane on the right.

| Name       | Service State | Connection | Status  | Priority | Primary VPGate |
|------------|---------------|------------|---------|----------|----------------|
| Avtec_1    | Available     | ✓          | Primary | 8        | 205.3.3.106    |
| Avtec_2    | Available     | ✓          | Primary | 8        | 205.3.3.106    |
| Avtec_3    | Unavailable   | ✗          | N/A     | 8        |                |
| Outpost_1a | Unavailable   | ✗          | N/A     | 8        |                |
| Outpost_1b | Unavailable   | ✗          | N/A     | 0        |                |
| Outpost_2a | Unavailable   | ✗          | N/A     | 0        |                |
| Outpost_2b | Unavailable   | ✗          | N/A     | 8        |                |

This brings up the **Endpoint Configuration** page. Under **Endpoint Name**, give it a unique identifier (This is the name VPGate and Scout use to identify the endpoint. It doesn't have to

correspond to anything, just name it something meaningful.). Change the next field, **Service State**, to **AVAILABLE**. Under **VoIP Audio Settings**, change the **Receive Audio Mode** to **FULL DUPLEX**, unless for some reason a simplex or half-duplex line is being used. Other fields can be left at their default values for standard use. When you are finished, press **Add**. It will create the endpoint and return you to the **Endpoint Summary** page.

**VPGate Configuration**  
Avtec\_VPGate

VPGate Endpoints Advanced Diagnostics Help

Endpoint Summary Input Alerts

[Endpoint Summary](#) > Endpoint Configuration

### Endpoint Configuration

[Add](#)

**Endpoint Configuration**

Endpoint Name

Service State

**Endpoint Connection**

Endpoint Audio

**VoIP Audio Settings**

Receive Audio Mode

Override Receive Audio IP Port

Override Transmit Audio IP Port

VoIP Audio Jitter Depth (ms)

Squelch Tail Time Out

Allow Barge-in/Monitor Outbound Audio

**Inbound Calls**

Emergency Only

Unanswered Time Out

Call Clear Time Out

Duplicate Call Supported

**Miscellaneous**

PTT Time Out

PTT Override Priority

PTT Override Immunity Time

Secondary Keying Action

Redundant Priority

Click on the endpoint name again to go back to the **Endpoint Configuration** page. At the bottom of the page is a section labeled **Drivers**. From the drop-down menu, select **SIP** and press **Add driver**. This brings up the SIP configuration page as shown below.

Avtec System Settings VPGate1

# VPGate Configuration

Avtec\_VPGate

VPGate Endpoints Advanced Diagnostics Help

Endpoint Summary Input Alerts

Endpoint Summary > Avtec\_1 > SIP

## SIP

Update Defaults

### SIP Configuration

Driver Processing Order 2

### Primary Identity

Display Name Avtec\_1

Username 2000

SIP Server Address/Domain name 10.64.90.109

SIP Server Port 5060

Authentication Username 2000

Authentication Password 1234

Register with SIP Server? YES

Register Refresh Time 3600

Outbound Proxy Address

Outbound Proxy Port 0

Force Outbound Proxy NO

### Secondary Identity

Enable Secondary Identity? NO

### CODEC Configuration

G.711 uLaw Enabled YES

G.711 uLaw SDP Payload Type 0

G.711 uLaw SDP Description PCMU

G.729A Enabled YES

G.729A SDP Payload Type 18

G.729A SDP Description G729

Most of the fields under **Primary Identity** are used to tell the endpoint how to connect to the SIP server that it will be registering against. Under **Display Name**, enter what you would like the Caller ID to display for this endpoint when you call another endpoint. Under **Username**, enter the username this endpoint will use to connect to the SIP server. For **SIP Server Address/Domain name**, enter the IP address of the SIP server itself. For **SIP Server Port**, enter the port number for the SIP server. Set **Authentication Username** and **Authentication Password** to the username and password this endpoint will use to register with the SIP server. **Register Refresh Time** sets how frequently the endpoint will send registrations to the SIP server. This field defaults to 70 seconds. Avaya recommends setting this field to 3600 (1 hour).

The **Codec Configuration** section lists the various types of codecs the endpoint supports. If you want the endpoint to advertise that it supports a specific codec, set the respective codec field to **YES**. If you do not wish to advertise that the endpoint supports a specific codec, set the field to **NO**. (Note: special licensing is required to use the G.723.1 codec.) Also under the **Codec Configuration** section is a field labeled **Telephone-Event (RFC2833/4733) Enabled**. If you are using RTP Events, you should set this field to **YES**.

The screenshot shows the VPGate Configuration web interface. The browser tabs include 'Avtec', 'System Settings', and 'VPGate1'. The page title is 'VPGate Configuration' with the subtitle 'Avtec\_VPGate'. The navigation menu has tabs for 'VPGate', 'Endpoints', 'Advanced', 'Diagnostics', and 'Help'. Below this, there are sub-tabs for 'Endpoint Summary' and 'Input Alerts'. The breadcrumb trail is 'Endpoint Summary > Avtec\_1 > SIP'. The main heading is 'SIP'. There are 'Update' and 'Defaults' buttons. The 'CODEC Configuration' section contains the following fields:

| Field                                  | Value           |
|--|-----------------|
| G.711 uLaw Enabled                     | YES             |
| G.711 uLaw SDP Payload Type            | 0               |
| G.711 uLaw SDP Description             | PCMU            |
| G.729A Enabled                         | YES             |
| G.729A SDP Payload Type                | 18              |
| G.729A SDP Description                 | G729            |
| G.726 16kb Enabled                     | NO              |
| G.726 32kb Enabled                     | NO              |
| G.723.1 Enabled                        | NO              |
| Telephone-Event (RFC2833/4733) Enabled | YES             |
| Telephone-Event SDP Payload Type       | 101             |
| Telephone-Event SDP Description        | telephone-event |

The 'Dial Configuration' section contains the following fields:

| Field                    | Value |
|--------------------------|-------|
| Hook Flash Enabled       | NO    |
| Digit Time               | 100   |
| Interdigit Time          | 100   |
| Digit Level              | -10   |
| Dial Address/Domain Name |       |
| Dial Port                | 0     |
| Enable Call Transfer     | YES   |

The 'Ring Down' section contains the following field:

| Field          | Value |
|----------------|-------|
| Ring Down Type | NONE  |

Under **Advanced Configuration**, there is a field labeled **Enable Inactive Hold for Transfers**. If a system sends back an "Inactive" for the "m" line in the SDP when placing the call on hold during a transfer, it will be necessary to set this field to **YES**. It is set to **NO** by default. Other fields can be left at their default values unless there is some specific reason to change them. (For



example, to configure the endpoint to automatically ring down to another extension when it is selected, or to set up an Auto-Answer or Message on Hold wav file.)

The screenshot shows the VPGate Configuration web interface. The browser window has tabs for Avtec, System Settings, and VPGate1. The page title is "VPGate Configuration" with the Avtec logo. The navigation bar includes "VPGate", "Endpoints", "Advanced", "Diagnostics", and "Help". Below this is a sub-navigation bar with "Endpoint Summary" and "Input Alerts". The main content area shows the breadcrumb "Endpoint Summary > Avtec 1 > SIP" and the title "SIP". There are "Update" and "Defaults" buttons. The configuration is organized into sections: "Ring Down" (Dial Port: 0, Enable Call Transfer: YES), "Quality of Service Priorities" (RTP/RTCP Packets: B8, SIP Packets: 68), "Advanced Network Configuration" (Local RTP Port: 0, Local SIP Port: 0, SIP Transport: UDP ONLY, UDP Keepalive Interval: 0), and "Advanced Configuration" (RTCP Enabled: NO, Stale Call Time: 180, SIP Session Timer Enabled: NO, Peak Audio Level: 0, Enable Auto Answer: NO, Enable Message On Hold: NO, Enable Inactive Hold for Transfers: YES, SIP Stack Logging Level: STACK, SIP Stack Log File Size: 5, Driver Logging Level: DEBUGHIGH). The "Enable Inactive Hold for Transfers" option is highlighted with a red rectangle.

| Field                                 | Value     |
|---------------------------------------|-----------|
| Dial Port                             | 0         |
| Enable Call Transfer                  | YES       |
| <b>Ring Down</b>                      |           |
| Ring Down Type                        | NONE      |
| <b>Quality of Service Priorities</b>  |           |
| RTP/RTCP Packets                      | B8        |
| SIP Packets                           | 68        |
| <b>Advanced Network Configuration</b> |           |
| Local RTP Port                        | 0         |
| Local SIP Port                        | 0         |
| SIP Transport                         | UDP ONLY  |
| UDP Keepalive Interval                | 0         |
| <b>Advanced Configuration</b>         |           |
| RTCP Enabled                          | NO        |
| Stale Call Time                       | 180       |
| SIP Session Timer Enabled             | NO        |
| Peak Audio Level                      | 0         |
| Enable Auto Answer                    | NO        |
| Enable Message On Hold                | NO        |
| Enable Inactive Hold for Transfers    | YES       |
| SIP Stack Logging Level               | STACK     |
| SIP Stack Log File Size               | 5         |
| Driver Logging Level                  | DEBUGHIGH |

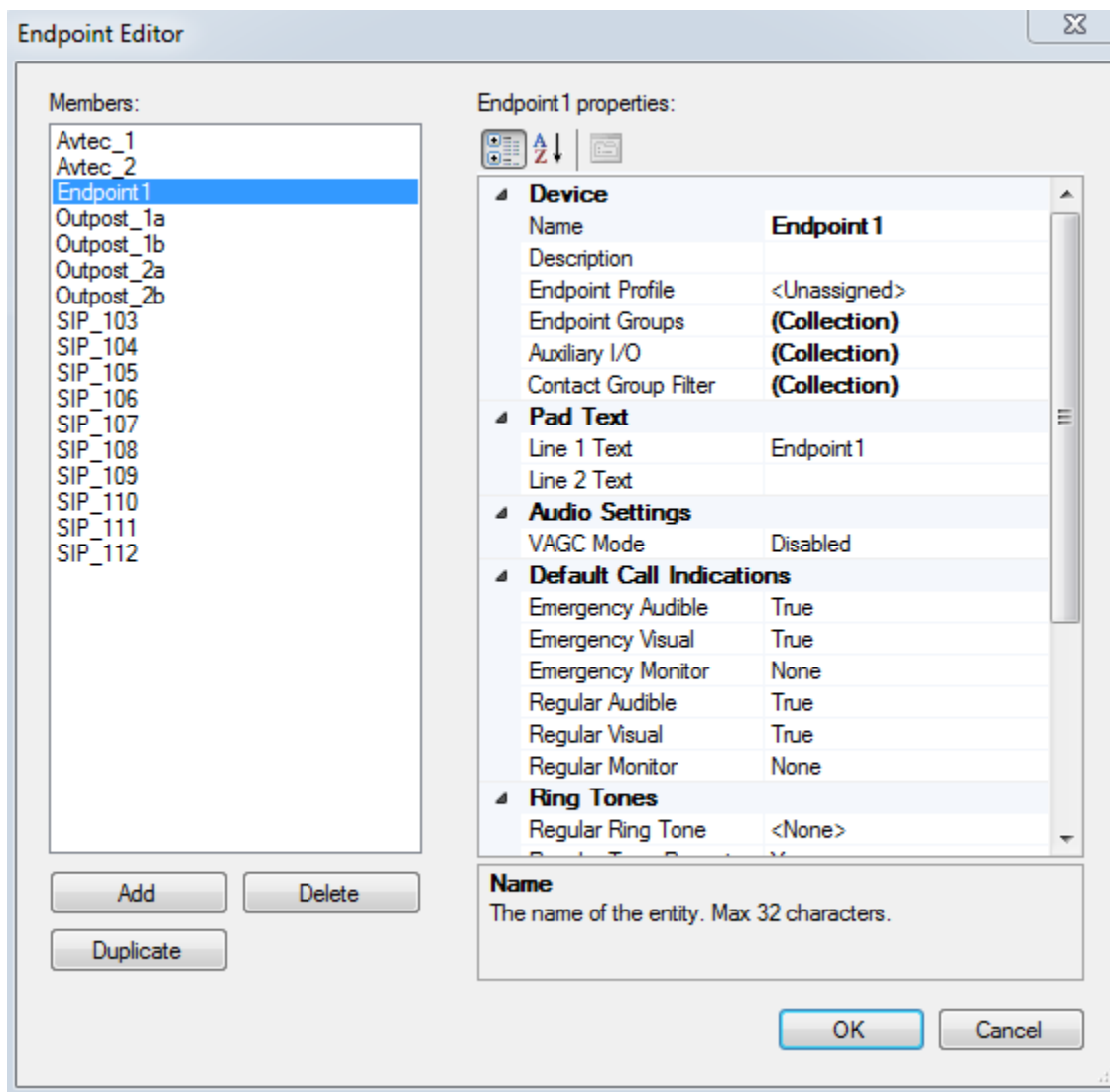
When you have finished configuring the endpoint, press **Add** (as was shown above) or **Update** when editing an existing endpoint.

Follow the same procedure above, making sure you use a unique identifier in the **Endpoint Name** field, for every endpoint that needs to be configured.

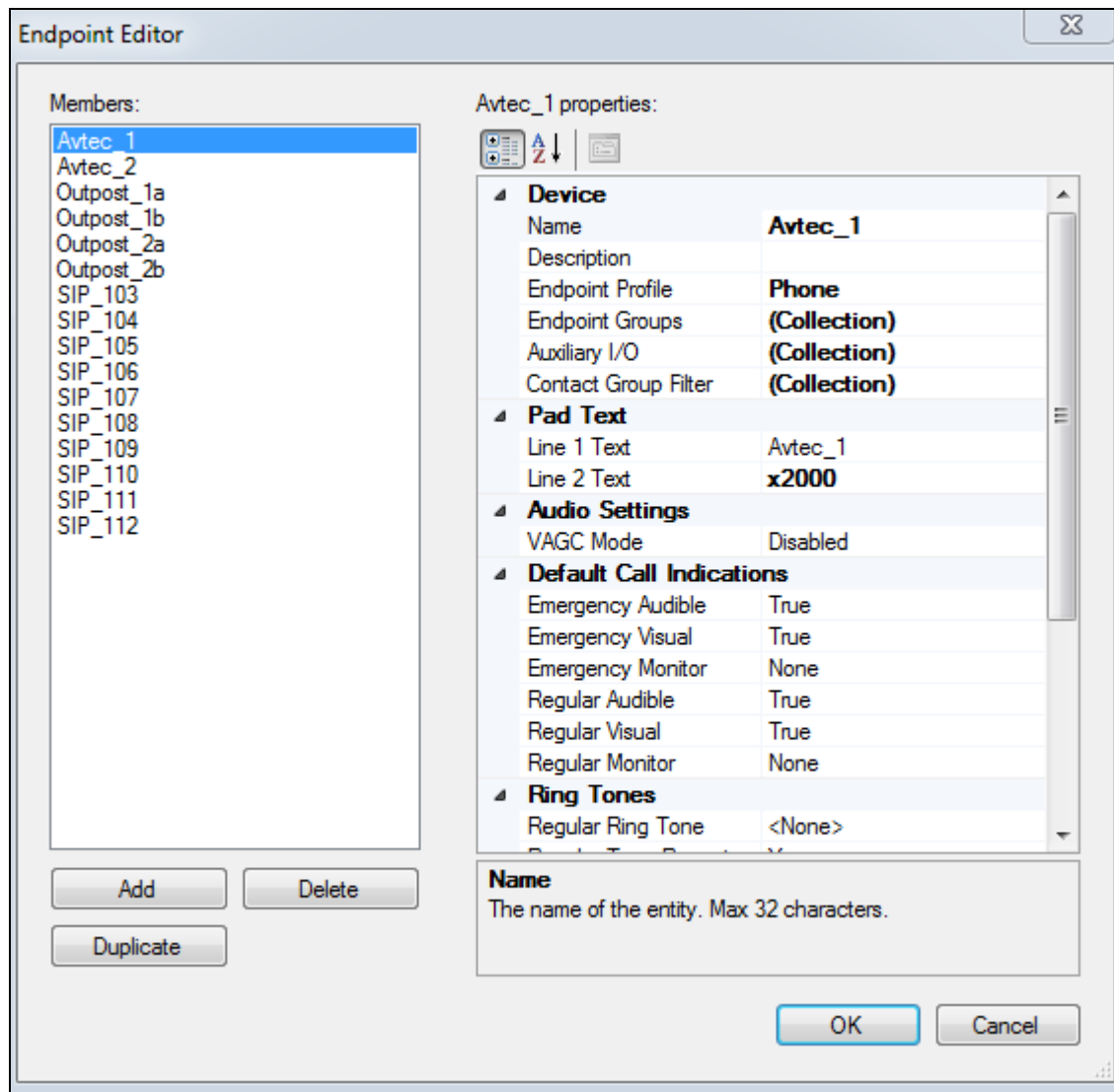
Now that the endpoint is configured, it will be necessary to add a "line pad" to the Avtec Scout VoIP Console so you can use it. First, start up the **Project Manager** software. (This was installed alongside the Avtec Scout VoIP Console software.) The Avtec Scout VoIP Console

installation and help files describe in detail how to create a new project and new screens. These next steps assume there is a basic project defined and that new endpoints simply need to be added.

In **Project Manager**, in the **Project Explorer** tree on the left-hand side, right-click the **Endpoints** folder and choose **Add New→Endpoint** (not shown). This creates a new endpoint with the generic name, **Endpoint1**, under the **Endpoints** folder. Double-click that endpoint to bring up the **Endpoint Editor** window as shown below.



In the **Properties** window on the right side, change the **Name** field to match the name entered for the endpoint in VPGate. Under **Endpoint Profile**, select the preferred profile to use with this endpoint. (The preinstalled **Phone** profile should work fine. Profiles define basic things like default ring and call-progress tones, whether the endpoint can be put on hold, etc.) Other fields can be customized, but these are the required fields that are needed to make the endpoint function properly.



Next, it will be necessary to add the endpoint to a screen (assuming a screen for the endpoint has already been created.) There are several different ways to do that. The simplest way is to click on the endpoint name in the **Project Explorer** window and drag it over to the screen wherever you would like to place it. This can also be done through the **Toolbox** window. In the left-hand window of **Project Manager**, click on the **Toolbox** tab. Find the **Line Pad** icon. Click and drag the icon from the Toolbox to the screen wherever you want to place the endpoint. Then in the **Properties** window on the right-hand side, under **Behavior**, select the drop-down next to the **Name** field and choose the endpoint you created. If you already have a line pad defined on a screen and want to reassign it, select the line pad and change the **Name** field in the **Properties** window in this same way.



Save the project and send it. To do this, press the **Send Project** button on the toolbar at the top, assuming you have **Project Manager** configured to send projects either locally or to a centralized project server.



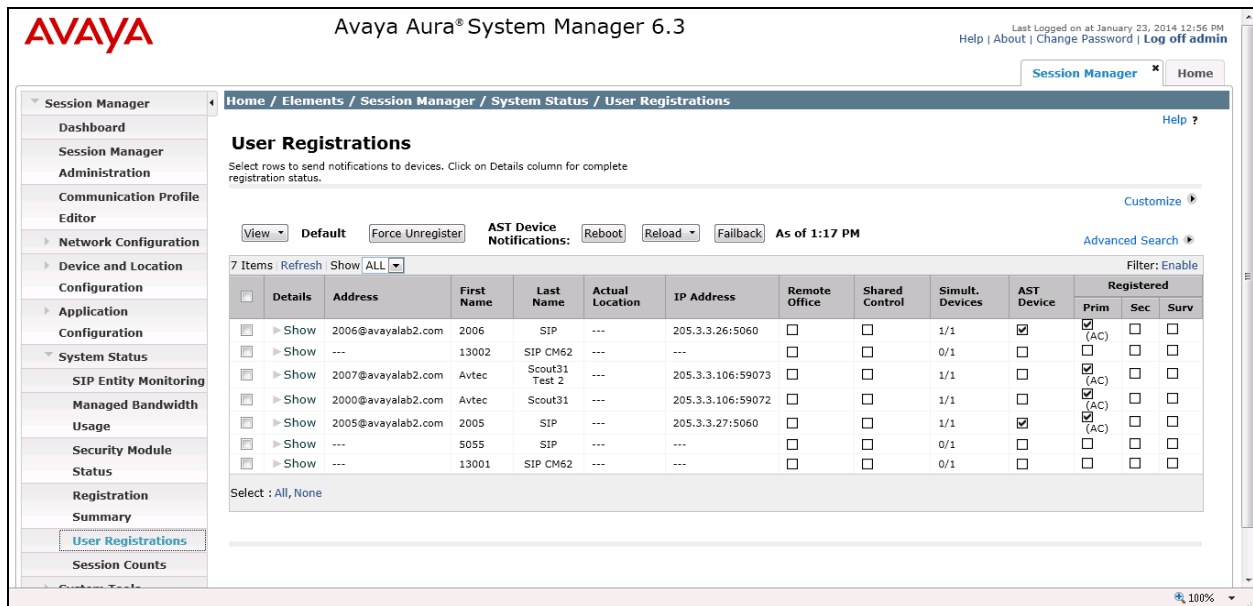
Now your endpoint(s) should be usable.

## 8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and the Avtec Scout VoIP Console.

### 8.1. Verify User Registrations

On Session Manager, verify the registration status of the Avtec Scout VoIP Console device by navigating to **Session Manager → System Status → User Registrations**. Verify that all the users administered in **Section 6.1** are listed as registered users.



The screenshot displays the Avaya Aura System Manager 6.3 interface. The left sidebar contains navigation links such as Session Manager, Dashboard, Session Manager Administration, Communication Profile Editor, Network Configuration, Device and Location Configuration, Application Configuration, System Status, SIP Entity Monitoring, Managed Bandwidth Usage, Security Module Status, Registration Summary, User Registrations (highlighted), and Session Counts. The main content area is titled 'User Registrations' and includes a breadcrumb trail: Home / Elements / Session Manager / System Status / User Registrations. Below the title, there are instructions: 'Select rows to send notifications to devices. Click on Details column for complete registration status.' and a 'Customize' link. A toolbar contains buttons for View (Default), Force Unregister, AST Device Notifications, Reboot, Reload, and Failback, along with a timestamp 'As of 1:17 PM' and an 'Advanced Search' link. A table lists 7 items with columns: Details, Address, First Name, Last Name, Actual Location, IP Address, Remote Office, Shared Control, Simult. Devices, AST Device, and Registered (Prim, Sec, Surv). The table data is as follows:

| Details | Address            | First Name | Last Name      | Actual Location | IP Address        | Remote Office            | Shared Control           | Simult. Devices | AST Device                          | Registered (Prim, Sec, Surv)  |
|---------|--------------------|------------|----------------|-----------------|-------------------|--------------------------|--------------------------|-----------------|-------------------------------------|---|
| ▶ Show  | 2006@avayalab2.com | 2006       | SIP            | ---             | 205.3.3.26:5060   | <input type="checkbox"/> | <input type="checkbox"/> | 1/1             | <input checked="" type="checkbox"/> | (AC) <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> |
| ▶ Show  | ---                | 13002      | SIP CM62       | ---             | ---               | <input type="checkbox"/> | <input type="checkbox"/> | 0/1             | <input type="checkbox"/>            | <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>      |
| ▶ Show  | 2007@avayalab2.com | Avtec      | Scout31 Test 2 | ---             | 205.3.3.106:59073 | <input type="checkbox"/> | <input type="checkbox"/> | 1/1             | <input checked="" type="checkbox"/> | (AC) <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> |
| ▶ Show  | 2000@avayalab2.com | Avtec      | Scout31        | ---             | 205.3.3.106:59072 | <input type="checkbox"/> | <input type="checkbox"/> | 1/1             | <input checked="" type="checkbox"/> | (AC) <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> |
| ▶ Show  | 2005@avayalab2.com | 2005       | SIP            | ---             | 205.3.3.27:5060   | <input type="checkbox"/> | <input type="checkbox"/> | 1/1             | <input checked="" type="checkbox"/> | (AC) <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/> |
| ▶ Show  | ---                | 5055       | SIP            | ---             | ---               | <input type="checkbox"/> | <input type="checkbox"/> | 0/1             | <input type="checkbox"/>            | <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>      |
| ▶ Show  | ---                | 13001      | SIP CM62       | ---             | ---               | <input type="checkbox"/> | <input type="checkbox"/> | 0/1             | <input type="checkbox"/>            | <input type="checkbox"/> <input type="checkbox"/> <input type="checkbox"/>      |

Below the table, there is a 'Select : All, None' option.

### 8.2. Verify Avtec Scout VoIP Console

Make a call to one of the extensions or DID's for the Avtec Scout VoIP Console endpoint(s). Verify that the call has ringback tone, two-way audio for at least 30 seconds, and that the call properly disconnects.

## 9. Conclusion

These Application Notes describe the configuration steps required for Avtec Scout VoIP Console to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

All feature and serviceability test cases were completed with observations listed in **Section 7**.

## 10. Additional References

This section references the product documentation relevant to these Application Notes.

Avaya documentation referenced below can be found at <http://support.avaya.com>

1. *Implementing Avaya Aura® Communication Manager*, Document 03-603558, Issue 5.0, Release 6.3, October 2013
2. *Administering Avaya Aura® Communication Manager*, Document 03-300509, Issue 9.0, Release 6.3, October 2013
3. *Implementing Avaya Aura® Session Manager*, Document 03-603473, Release 6.2, March 2013
4. *Administering Avaya Aura® Session Manager*, Document 03-603324, Issue 3, Release 6.3, October 2013

Avtec documentation can be found at <https://portal.avtecinc.com>

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