



Avaya Solution & Interoperability Test Lab

Application Notes for Avtec Scout VoIP Console 5.3 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager using SIP Trunk 10.1 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Avtec Scout VoIP Console with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Avtec Scout VoIP Console is a SIP-based system that provides a customizable interface and supports telephony features, such as inbound and outbound calls, hold, resume, mute, and transfer. It integrates with Avaya Aura® Session Manager via a SIP trunk.

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 to integrate Avtec Scout VoIP Console with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Avtec Scout VoIP Console is a SIP-based system that provides a customizable interface and supports telephony features, such as inbound and outbound calls, hold, resume, mute, and transfer. It integrates with Avaya Aura® Session Manager via a SIP trunk.

2 General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Avtec Scout VoIP Console, Avaya SIP and H.323 IP Deskphones, and the PSTN, and exercising basic telephony features, such as hold, mute, and transfer.

The serviceability testing focused on verifying that Avtec Scout VoIP Console came back into service after re-connecting the Ethernet cable and rebooting the system. The following sub-section covers the features and functionality that were covered in more detail.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and Avtec Scout VoIP Console did not include use of any specific encryption features as requested by Avtec.

2.1 Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Establishing a SIP trunk between Scout VoIP Console and Session Manager. This includes verifying that Scout VoIP Console successfully responds to SIP OPTIONS messages.
- Calls between Scout VoIP Console and Avaya SIP/H.323 IP Deskphones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between Scout VoIP Console and the PSTN.
- G.711 and G.729 codec support.
- Proper recognition of DTMF tones from Scout VoIP Console.
- Basic telephony features, including hold, mute, redial, multiple calls, and blind/attended transfers.
- Two SIP lines registered to the Avtec Scout SIP Proxy. The SIP lines register to the Scout SIP Proxy and Scout VPGate connects to Session Manager via a SIP trunk.
- Multiple simultaneous calls to the two SIP lines and long duration calls.
- Proper system recovery after a reboot of Scout VoIP Console and loss of IP connectivity.

2.2 Test Results

All test cases passed with the following observations.

- Incoming call notification not heard through headset by design, external speakers are required. However, visual indication of incoming calls is provided by the Scout VoIP Console.
- Each SIP line on Scout VoIP Console supports one call at a time. An incoming call to an active line on Scout VoIP Console results in either busy tone or the call covering to the next coverage point, if configured. However, multiple SIP lines may be configured on Scout VoIP Console.
- Scout VoIP Console does not currently support conferencing.
- SIP TLS transport and SRTP is currently not supported by Scout VoIP Console.
- Although outside the scope of these Application Notes, it is worth noting that if PSTN connectivity provided via Avaya Session Border Controller for Enterprise and shuffling is enabled, *Delayed SDP Handling* may need to be enabled to support attended/supervised transfers over the PSTN.

2.3 Support

Avtec Technical Support for Scout VoIP Console can be obtained via phone, email, or website.

- **Phone:** +1 (800) 545-3034
+1 (803) 358-3601
- **Email:** customersupport@avtecinc.com
- **Web:** <https://www.avtecinc.com/support>

3 Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following products:

- Communication Manager with a G430/G450 Media Gateway and Avaya Aura® Media Server providing media resources.
- Session Manager connected to Communication Manager via a SIP trunk.
- Avaya Aura® System Manager to configure Session Manager.
- Avaya H.323 and SIP Deskphones.
- Scout VoIP Console connected to Session Manager via a SIP trunk.
- Scout VoIP Console was installed on a desktop PC running Microsoft Windows 10 and included the following components: Scout VPGate, Scout SIP Proxy, and Scout Manager. Scout Manager was used to configure the Scout VoIP Console.

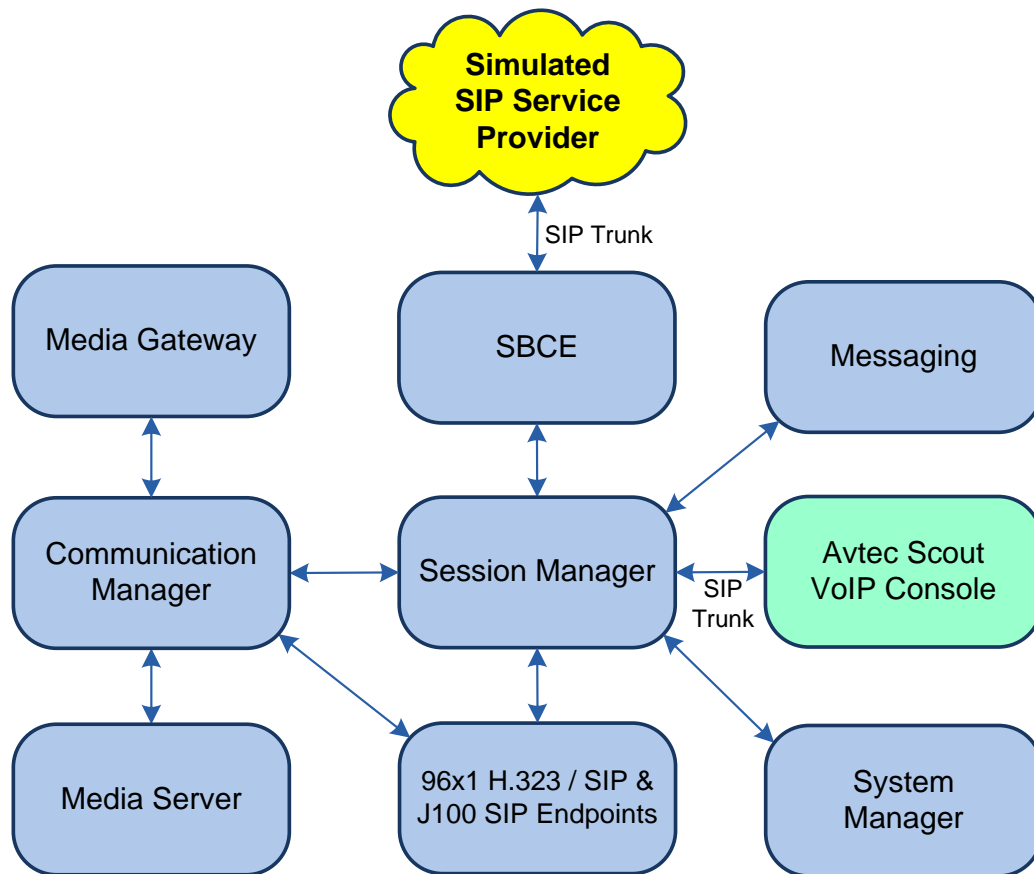


Figure 1: Avaya SIP Telephony Network with Avtec Scout VoIP Console

4 Equipment and Software Validated

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

| Equipment/Software | Release/Version |
|--|---|
| Avaya Aura® Communication Manager | 10.1.0.1.0-SP1 |
| Avaya G430 Media Gateway | FW 42.4.0 |
| Avaya G450 Media Gateway | FW 42.7.0 |
| Avaya Aura® Media Server | v.10.1.0.77 |
| Avaya Aura® System Manager | 10.1.0.1 Build No. – 10.1.0.0.537353 Software Update Revision No: 10.1.0.1.0614394 Service Pack 1 |
| Avaya Aura® Session Manager | 10.1.0.1.1010105 |
| Avaya Messaging | 11.0.0.3204 |
| Avaya 96x1 Series IP Deskphones | 6.8.5.3.2 (H.323) 7.1.13.0.4 (SIP) |
| Avaya J100 SIP Telephones | 4.0.13.0.6 |
| Avtec Scout VoIP Console running on Microsoft Windows 10, including the following components: <ul style="list-style-type: none">Scout ConsoleScout VPGateScout SIP ProxyScout Manager | 5.3.0.13 5.3.0.41 5.3.0.11 5.3.0.13 |

5 Configure Avaya Aura® Communication Manager

This section describes the steps for configuring a SIP trunk to Session Manager and routing calls to Scout. Administration of Communication Manager was performed using the System Access Terminal (SAT). This section covers the following configuration:

- **IP Node Names** to associate names with IP addresses.
- **IP Codec Set** to specify the codec type used for calls to Scout VoIP Console.
- **IP Network Region** to specify the domain name and the IP codec set, to enable IP-IP direct audio (i.e., Shuffling), and to specify the UDP port range.
- **SIP trunk** for calls towards Session Manager and Scout.
- **Private Numbering** to allow the caller's extension to be sent to Scout VoIP Console.
- **Call Routing** to route calls to Scout VoIP Console using AAR.

5.1 Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). The host names will be used in other configuration screens of Communication Manager.

| | | |
|---|----------------------|---------------|
| change node-names ip | | Page 1 of 2 |
| | | IP NODE NAMES |
| Name | IP Address | |
| default | 0.0.0.0 | |
| devcon-aes | 10.64.102.119 | |
| devcon-ams | 10.64.102.118 | |
| devcon-sm | 10.64.102.117 | |
| procr | 10.64.102.115 | |
| procr6 | :: | |
| (6 of 6 administered node-names were displayed) | | |
| Use 'list node-names' command to see all the administered node-names | | |
| Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name | | |

5.2 Administer IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to Scout VoIP Console. The form is accessed via the **change ip-codec-set 1** command. Note the codec set number since it will be used in the IP Network Region covered in the next section. Although media encryption was configured, Scout VoIP Console did not use SRTP. Media encryption option of *none* must be specified as shown below. The Scout VoIP Console was tested using G.711 and G.729 codecs.

change ip-codec-set 1

Page1 of 2

IP MEDIA PARAMETERS

Codec Set: 1

| Audio Codec | Silence Suppression | Frames Per Pkt | Packet Size (ms) |
|-------------|---------------------|----------------|------------------|
| 1: G.711MU | n | 2 | 20 |
| 2: G.729 | n | 2 | 20 |
| 3: | | | |
| 4: | | | |
| 5: | | | |
| 6: | | | |
| 7: | | | |

Media Encryption

Encrypted SRTP: best-effort

1: 1-srtp-aescm128-hmac80

2: 2-srtp-aescm128-hmac32

3: none

4:

5:

5.3 Administer IP Network Region

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between Scout VoIP Console and IP endpoints without using media resources in the Avaya Media Gateway or Avaya Media Server. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region 1) is specified in the SIP signaling group.

| | | |
|--|--|--------------|
| change ip-network-region 1 | | Page 1 of 20 |
| IP NETWORK REGION | | |
| Region: 1 | | |
| Location: 1 Authoritative Domain: avaya.com | | |
| Name: SIP Enterprise Stub Network Region: n | | |
| MEDIA PARAMETERS | | |
| Codec Set: 1 | | |
| Intra-region IP-IP Direct Audio: yes | | |
| Inter-region IP-IP Direct Audio: yes | | |
| UDP Port Min: 2048 IP Audio Hairpinning? n | | |
| UDP Port Max: 50999 | | |
| DIFFSERV/TOS PARAMETERS | | |
| Call Control PHB Value: 46 | | |
| Audio PHB Value: 46 | | |
| Video PHB Value: 26 | | |
| 802.1P/Q PARAMETERS | | |
| Call Control 802.1p Priority: 6 | | |
| Audio 802.1p Priority: 6 | | |
| Video 802.1p Priority: 5 | | |
| AUDIO RESOURCE RESERVATION PARAMETERS | | |
| H.323 IP ENDPOINTS RSVP Enabled? n | | |
| H.323 Link Bounce Recovery? y | | |
| Idle Traffic Interval (sec): 20 | | |
| Keep-Alive Interval (sec): 5 | | |
| Keep-Alive Count: 5 | | |

5.4 Administer SIP Trunk to Session Manager

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- **Group Type** was set to *sip*.
- **IMS Enabled** was set to *n*.
- **Transport Method** was set to *tls*.
- Specify the Ethernet processor (*procr*) of Communication Manager and Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form in **Section 5.1**.
- Set the TLS port value (e.g., *5061*) in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- **Direct IP-IP Audio Connections** was enabled on this form.
- **Initial IP-IP Direct Media** was enabled.
- **DTMF over IP** was left at the default value of *rtp-payload*.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

| add signaling-group 10 | | Page 1 of 2 |
|---|------------------------------|------------------------------------|
| SIGNALING GROUP | | |
| Group Number: 10 | Group Type: sip | |
| IMS Enabled? n | Transport Method: tls | |
| Q-SIP? n | | |
| IP Video? y | Enforce SIPS URI for SRTP? n | |
| Peer Detection Enabled? y | Peer Server: SM | Clustered? n |
| Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y | | |
| Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n | | |
| Alert Incoming SIP Crisis Calls? n | | |
| Near-end Node Name: procr | | Far-end Node Name: devcon-sm |
| Near-end Listen Port: 5061 | | Far-end Listen Port: 5061 |
| | | Far-end Network Region: 1 |
| Far-end Domain: avaya.com | | |
| Incoming Dialog Loopbacks: eliminate | | Bypass If IP Threshold Exceeded? n |
| DTMF over IP: rtp-payload | | RFC 3389 Comfort Noise? n |
| Session Establishment Timer(min): 3 | | Direct IP-IP Audio Connections? y |
| Enable Layer 3 Test? y | | IP Audio Hairpinning? n |
| H.323 Station Outgoing Direct Media? n | | Initial IP-IP Direct Media? y |
| | | Alternate Route Timer(sec): 6 |

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to Scout VoIP Console. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Accept the default values for the remaining fields.

| | | | |
|--------------------------|--------------------------------|----------------|-----------|
| add trunk-group 10 | | Page 1 of 5 | |
| TRUNK GROUP | | | |
| Group Number: 10 | Group Type: sip | CDR Reports: y | |
| Group Name: To devcon-sm | COR: 1 | TN: 1 | TAC: 1010 |
| Direction: two-way | Outgoing Display? n | | |
| Dial Access? n | Night Service: | | |
| Queue Length: 0 | | | |
| Service Type: tie | Auth Code? n | | |
| | Member Assignment Method: auto | | |
| | Signaling Group: 10 | | |
| | Number of Members: 10 | | |

On Page 3 of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

| | | | |
|--------------------------------|----------------------------------|---------------------------------|--|
| add trunk-group 10 | | Page 3 of 21 | |
| TRUNK FEATURES | | | |
| ACA Assignment? n | Measured: none | Maintenance Tests? y | |
| Suppress # Outpulsing? n | Numbering Format: private | UUI Treatment: service-provider | |
| | | Replace Restricted Numbers? n | |
| | | Replace Unavailable Numbers? n | |
| | | Hold/Unhold Notifications? y | |
| | Modify Tandem Calling Number: no | | |
| Show ANSWERED BY on Display? y | | | |

On Page 4 of the trunk group form, the default settings were used as shown below.

| | |
|--|--------------|
| add trunk-group 10 | Page 4 of 21 |
| <p>PROTOCOL VARIATIONS</p> <p>Mark Users as Phone? n</p> <p>Prepend '+' to Calling/Alerting/Diverting/Connected Number? n</p> <p>Send Transferring Party Information? n</p> <p>Network Call Redirection? n</p> <p>Send Diversion Header? n</p> <p>Support Request History? y</p> <p>Telephone Event Payload Type: 101</p> <p>Convert 180 to 183 for Early Media? n</p> <p>Always Use re-INVITE for Display Updates? n</p> <p>Identity for Calling Party Display: P-Asserted-Identity</p> <p>Block Sending Calling Party Location in INVITE? n</p> <p>Accept Redirect to Blank User Destination? n</p> <p>Enable Q-SIP? n</p> <p>Interworking of ISDN Clearing with In-Band Tones: keep-channel-active</p> <p>Request URI Contents: may-have-extra-digits</p> | |

5.5 Configure Private Numbering

Configure the **Numbering – Private Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '7' whose calls are routed over any trunk group, including SIP trunk group 10, have the extension sent to Scout VoIP Console.

| change private-numbering 0 | Page 1 of 2 | | | | | | | | | | |
|--|-------------|------------|----------------|------------|----------------|-----------|---|---|--|--|---|
| <p>NUMBERING - PRIVATE FORMAT</p> <table border="1"> <thead> <tr> <th>Ext Len</th> <th>Ext Code</th> <th>Trk Grp(s)</th> <th>Private Prefix</th> <th>Total Len</th> </tr> </thead> <tbody> <tr> <td>5</td> <td>7</td> <td></td> <td></td> <td>5</td> </tr> </tbody> </table> <p>Total Administered: 1 Maximum Entries: 540</p> | | Ext Len | Ext Code | Trk Grp(s) | Private Prefix | Total Len | 5 | 7 | | | 5 |
| Ext Len | Ext Code | Trk Grp(s) | Private Prefix | Total Len | | | | | | | |
| 5 | 7 | | | 5 | | | | | | | |

5.6 AAR Call Routing

Configure the uniform dial plan table to route calls using AAR for dialed digits that are 5-digits long and begin with '78'. This would cover call routing to the Scout VoIP Console extensions (i.e., 78800 – 78801). For the compliance test, four Scout VoIP Console lines were configured as shown in **Section 7.3**.

| change uniform-dialplan 7 | Page 1 of 2 | | | | | | | | |
|--|-------------|------------------|-------------------|---------------|-------------------|----|-----|--|-------|
| <p>UNIFORM DIAL PLAN TABLE</p> <p>Percent Full: 0</p> <table border="1"> <thead> <tr> <th>Matching Pattern</th> <th>Len Del</th> <th>Insert Digits</th> <th>Node Net Conv Num</th> </tr> </thead> <tbody> <tr> <td>78</td> <td>5 0</td> <td></td> <td>aar n</td> </tr> </tbody> </table> | | Matching Pattern | Len Del | Insert Digits | Node Net Conv Num | 78 | 5 0 | | aar n |
| Matching Pattern | Len Del | Insert Digits | Node Net Conv Num | | | | | | |
| 78 | 5 0 | | aar n | | | | | | |

| AAR DIGIT ANALYSIS TABLE | | | | | | Page 1 of 2 |
|--------------------------|-----------|-----------|---------------|-------------|-----------------|-------------|
| Location: all | | | | | Percent Full: 2 | |
| Dialed String | Total Min | Total Max | Route Pattern | Call Type | Node Num | ANI Req'd |
| 7 | 7 | 7 | 254 | aar | | n |
| 78 | 5 | 5 | 10 | lev0 | | n |
| 8 | 7 | 7 | 254 | aar | | n |
| 9 | 7 | 7 | 254 | aar | | n |
| | | | | | | n |

| | | | | | | | | | | | | |
|--|-------|---------------|--------|--------------------------|------|------|-----------------|------|------|-------------|------|------|
| change route-pattern 10 | | | | | | | | | | Page 1 of 3 | | |
| Pattern Number: 10 Pattern Name: To devcon-sm | | | | | | | | | | | | |
| SCCAN? n | | Secure SIP? n | | Used for SIP stations? n | | | | | | | | |
| | | | | | | | | | | | | |
| Grp | FRL | NPA | Pfx | Hop | Toll | No. | Inserted | | | DCS/ | IXC | |
| No | | | Mrk | Lmt | List | Del | Digits | | | QSIG | | |
| | | | | | | | Dgts | | | Intw | | |
| 1: | 10 | 0 | | | | | | | | | n | user |
| 2: | | | | | | | | | | n | user | |
| 3: | | | | | | | | | | n | user | |
| 4: | | | | | | | | | | n | user | |
| 5: | | | | | | | | | | n | user | |
| 6: | | | | | | | | | | n | user | |
| | | | | | | | | | | | | |
| BCC | VALUE | TSC | CA-TSC | ITC | | BCIE | Service/Feature | PARM | Sub | Numbering | LAR | |
| 0 | 1 | 2 | M | 4 | W | | | | Dgts | Format | | |
| Request | | | | | | | | | | | | |
| 1: | y | y | y | y | y | n | n | rest | | unk-unk | none | |
| 2: | y | y | y | y | y | n | n | rest | | | none | |
| 3: | y | y | y | y | y | n | n | rest | | | none | |
| 4: | y | y | y | y | y | n | n | rest | | | none | |
| 5: | y | y | y | y | y | n | n | rest | | | none | |
| 6: | y | y | y | y | y | n | n | rest | | | none | |

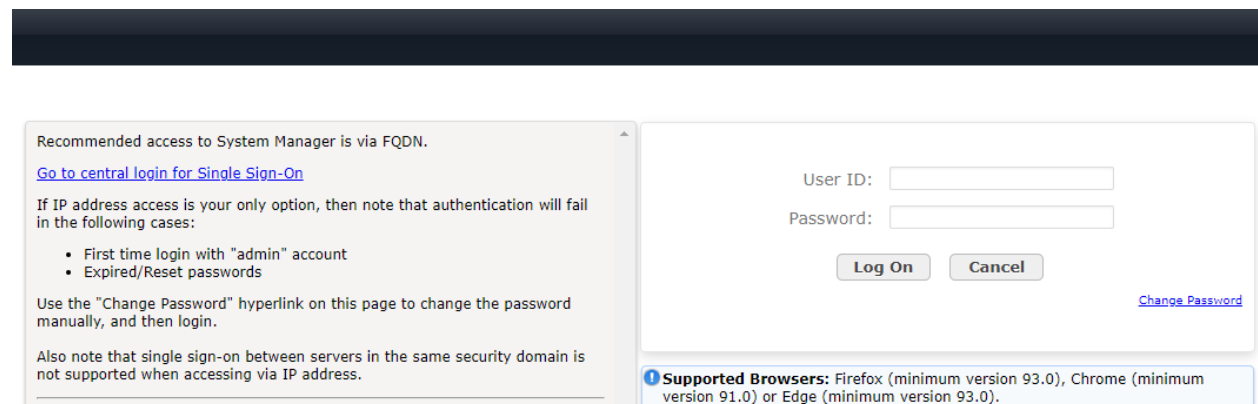
6 Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- Adaptation
- SIP Entity for Scout VoIP Console
- Entity Link, which defines the SIP trunk parameters used by Session Manager when routing calls to/from Scout VoIP Console
- Routing Policies and Dial Patterns
- Session Manager, which is managed by System Manager, and specifies the frequency of SIP Options

Note: It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of the adaptation, SIP entity, entity link, and call routing for Avtec Scout VoIP Console only.

Access the System Manager Web interface by using the URL `https://<ip-address>` in an Internet browser window, where `<ip-address>` is the IP address of the System Manager server. Log in using the appropriate credentials.



Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

User ID:

Password:

[Change Password](#)

Supported Browsers: Firefox (minimum version 93.0), Chrome (minimum version 91.0) or Edge (minimum version 93.0).

6.1 Add Adaptation

Session Manager can be configured with Adaptations that can modify SIP messages before or after routing decisions have been made; for example, replacing a domain name with an IP address as shown in this section. To create an **Adaptation** that will be applied to the Scout VoIP Console SIP entity in **Section 6.2**, navigate to **Elements → Routing → Adaptations** and click on the **New** button (not shown). In the **General** section, enter the following values. Use default values for all remaining fields.

- **Adaptation Name:** Enter a descriptive name for the Adaptation (e.g., *Avtec Adaptation*).
- **Module Name:** Select **DigitConversionAdapter**.
- **Module Parameter Type:** Select **Single Parameter**.
- **Module Parameter:** Enter the Scout VoIP Console IP address.

The screenshot shows the Avaya Aura System Manager 10.1 interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 10.1', and a menu with 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and a user profile 'admin' are also present. The left sidebar shows a tree view with 'Routing' selected, and 'Adaptations' highlighted. The main content area is titled 'Adaptation Details' and contains a 'General' section with the following fields:

- * Adaptation Name:** Text input field containing 'Avtec Adaptation'.
- Notes:** Text input field.
- * Module Name:** Dropdown menu showing 'DigitConversionAdapter'.
- Type:** Text input field containing 'digit'.
- State:** Dropdown menu showing 'enabled'.
- Module Parameter Type:** Dropdown menu showing 'Single Parameter'.
- Module Parameter:** Text input field containing '192.168.100.251'.
- Egress URI Parameters:** Text input field.

At the top right of the form, there are 'Commit' and 'Cancel' buttons, and a 'Help ?' link.

6.2 Add SIP Entity for Avtec Scout VoIP Console

In the sample configuration, one SIP trunk was configured for Scout VoIP Console, which provided four lines with four different extensions for this compliance test. These SIP extensions registered with the internal Scout VoIP Console SIP registrar/proxy, not with Session Manager.

A SIP Entity must be added for Scout VoIP Console. This SIP entity will have an adaptation rule to convert the domain name in the SIP URI of INVITE message to the Scout VoIP Console IP address and vice versa. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name (e.g., *Avtec Scout*).
- **FQDN or IP Address:** IP address of the Scout VoIP Console.
- **Type:** Select *SIP trunk*.
- **Adaptation:** Specify the adaptation configured in **Section 6.1**.
- **Location:** Select the location defined previously (not shown).
- **Time Zone:** Time zone for this location.

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.

The screenshot displays the Avaya Aura System Manager 10.1 web interface. The top navigation bar includes the Avaya logo, user information, and various menu items like Users, Elements, Services, Widgets, and Shortcuts. The left sidebar shows a tree view with 'Routing' selected, and 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' and contains a 'General' tab. The form fields are as follows:

- Name:** Avtec Scout
- FQDN or IP Address:** 192.168.100.251
- Type:** SIP Trunk
- Notes:** (empty text area)
- Adaptation:** Avtec Adaptation
- Location:** (empty dropdown)
- Time Zone:** America/New_York
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty text area)
- Securable:** ☐
- Call Detail Recording:** egress
- Loop Detection Mode:** On
- Loop Count Threshold:** 5
- Loop Detection Interval (in msec):** 200

Buttons for 'Commit' and 'Cancel' are located at the top right of the form. A 'Help' link is also present.

6.3 Add Entity Link for Avtec Scout VoIP Console

This section covers the configuration of an Entity Link for Scout VoIP Console. This entity link will specify that SIP entity configured in **Section 6.2**.

The SIP trunk from Session Manager to Scout VoIP Console is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *Avtec Scout Link*).
- **SIP Entity 1:** Select Session Manager.
- **Protocol:** Select the appropriate protocol (e.g., *UDP*).
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the Scout SIP entity configure in **Section 6.2**.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Select *Trusted*. *Note: If Trusted is not selected, calls from the associated SIP Entity specified in Section 6.2 will be denied.*

Click **Commit** to save the Entity Link definition.

The screenshot shows the Avaya Aura System Manager 10.1 interface. The left sidebar contains a navigation menu with the following items: Routing, Domains, Locations, Conditions, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, and Routing Policies. The main content area is titled 'Entity Links' and features a table with the following columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, and Port. The table contains one row with the following data: Name: Avtec Scout Link, SIP Entity 1: devcon-sm, Protocol: UDP, Port: 5060, SIP Entity 2: Avtec Scout, Port: 5060. Above the table, there are 'Commit' and 'Cancel' buttons. Below the table, there are 'Select : All, None' and another set of 'Commit' and 'Cancel' buttons. The top of the interface shows the Avaya logo, navigation tabs (Users, Elements, Services, Widgets, Shortcuts), a search bar, and a user profile (admin).

6.4 Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the Scout VoIP Console SIP Entity specified in **Section 6.2**. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:

Click **Select** and then select the appropriate SIP entity to which this routing policy applies. In this case, the Scout VoIP Console SIP entity is selected.

Defaults can be used for the remaining fields. Click **Commit** to save the Routing Policy definition. The following screen shows the Routing Policy for Scout VoIP Console.

The screenshot displays the Avaya Aura System Manager 10.1 interface. The left sidebar shows the navigation menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes 'Commit' and 'Cancel' buttons. The 'General' section contains fields for 'Name' (Avtec Scout Policy), 'Disabled' (checkbox), 'Retries' (0), and 'Notes'. The 'SIP Entity as Destination' section has a 'Select' button and a table listing the selected entity 'Avtec Scout' with its FQDN or IP Address (192.168.100.251) and Type (SIP Trunk). The 'Time of Day' section includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons, a table with 1 item, and a 'Filter: Enable' option. The table shows a policy for '24/7' with a ranking of 0, active from 00:00 to 23:59, and a note 'Time Range 24/7'.

| Name | FQDN or IP Address | Type | Notes |
|-------------|--------------------|-----------|-------|
| Avtec Scout | 192.168.100.251 | SIP Trunk | |

| Ranking | Name | Mon | Tue | Wed | Thu | Fri | Sat | Sun | Start Time | End Time | Notes |
|---------|------|-----|-----|-----|-----|-----|-----|-----|------------|----------|-----------------|
| 0 | 24/7 | ✓ | ✓ | ✓ | ✓ | ✓ | ✓ | ✓ | 00:00 | 23:59 | Time Range 24/7 |

6.5 Add Dial Patterns

Dial patterns must be defined to direct calls to the appropriate SIP Entity. In the sample configuration, a 5-digit number beginning with '788' will be routed to lines on Scout VoIP Console.

To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under *General*:

- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern (optional).

Under *Originating Locations and Routing Policies*:

Click **Add** and then select the appropriate location and routing policy from the list. In this case, the Scout VoIP Console routing policy is selected.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definitions for extensions on Scout VoIP Console.

The screenshot displays the Avaya Aura System Manager 10.1 interface. The left sidebar shows the navigation menu with 'Routing' selected. The main content area is titled 'Dial Pattern Details' and includes 'Commit' and 'Cancel' buttons. The 'General' section contains the following fields:

- Pattern:** 788
- Min:** 5
- Max:** 5
- Emergency Call:** ☐
- SIP Domain:** -ALL- (dropdown menu)
- Notes:** Avtec Scout

The 'Originating Locations and Routing Policies' section features an 'Add' button and a table with 1 item. The table has columns for Originating Location Name, Originating Location Notes, Routing Policy Name, Rank, Routing Policy, Routing Policy Destination, and Routing Policy Notes. The selected item is '-ALL-' with a Rank of 0 and Routing Policy of Avtec Scout. Below the table is a 'Select : All, None' option.

The 'Denied Originating Locations' section includes a 'Remove' button and a table with 0 items. The table has columns for Originating Location and Notes.

6.6 Add Session Manager

Adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add** (not shown), and fill in the fields as described below and shown in the following screen:

Under *General*:

- **SIP Entity Name:** Select the name of the SIP Entity added for Session Manager
- **Description:** Descriptive comment (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager

Use default values for the remaining fields. Click **Commit** to add this Session Manager.

The screenshot displays the Avaya Aura System Manager 10.1 web interface. The top navigation bar includes the Avaya logo, user information (Users, Elements, Services, Widgets, Shortcuts), a search bar, and a user profile (admin). The left sidebar shows the navigation menu with 'Session Manager' selected. The main content area is titled 'Edit Session Manager' and contains two sections: 'General' and 'Security Module'. The 'General' section includes fields for 'SIP Entity Name' (devcon-sm), 'Description', '*Management Access Point Host Name/IP' (10.64.102.116), '*Direct Routing to Endpoints' (Enable), 'Avaya Aura Device Services Server Pairing', and 'Maintenance Mode'. The 'Security Module' section includes fields for 'SIP Entity IP Address' (10.64.102.117), '*Network Mask' (255.255.255.0), '*Default Gateway' (10.64.102.1), '*Call Control PHB' (46), and '*SIP Firewall Configuration' (SM 6.3.8.0). The interface also shows a sidebar with navigation options like 'Session Manager', 'Dashboard', and 'Session Manager Administration'. The top bar includes the Avaya logo, user information, and search functionality.

The following screen shows the **Monitoring** section, which determines how frequently Session Manager sends SIP Options messages to Scout VoIP Console. Use default values for the remaining fields. Click **Commit** to add this Session Manager. In the following configuration, Session Manager sends a SIP Options message every *900* secs. If there is no response, Session Manager will send a SIP Options message every *120* secs.

Monitoring ▼

Enable SIP Monitoring ☒

*Proactive cycle time (secs)

900

*Reactive cycle time (secs)

120

*Number of Tries

1

*Number of Successes

1

Enable CRLF Keep Alive Monitoring ☐

*CRLF Ping Interval (secs)

0

7 Configure Avtec Scout VoIP Console

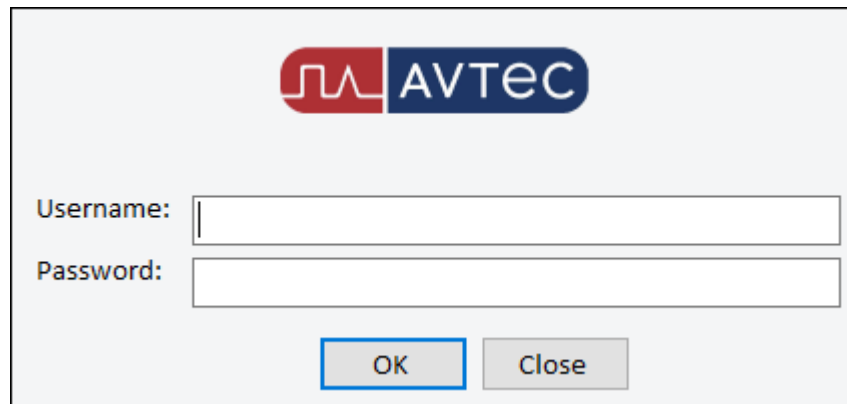
This section covers the configuration of Scout VoIP Console using the **Scout Manager** application. This section assumes that the Scout VoIP Console software has already been installed successfully. In the **Scout Manager** application, the following procedures are performed:

- Launch Scout Manager
- Add Local Domain
- Add Endpoints
- Modify SIP Line Label and Set Endpoint Profile
- Add Access Control List (ACLs) – Trusted Endpoints
- Add Routes
- Deploy the Configuration

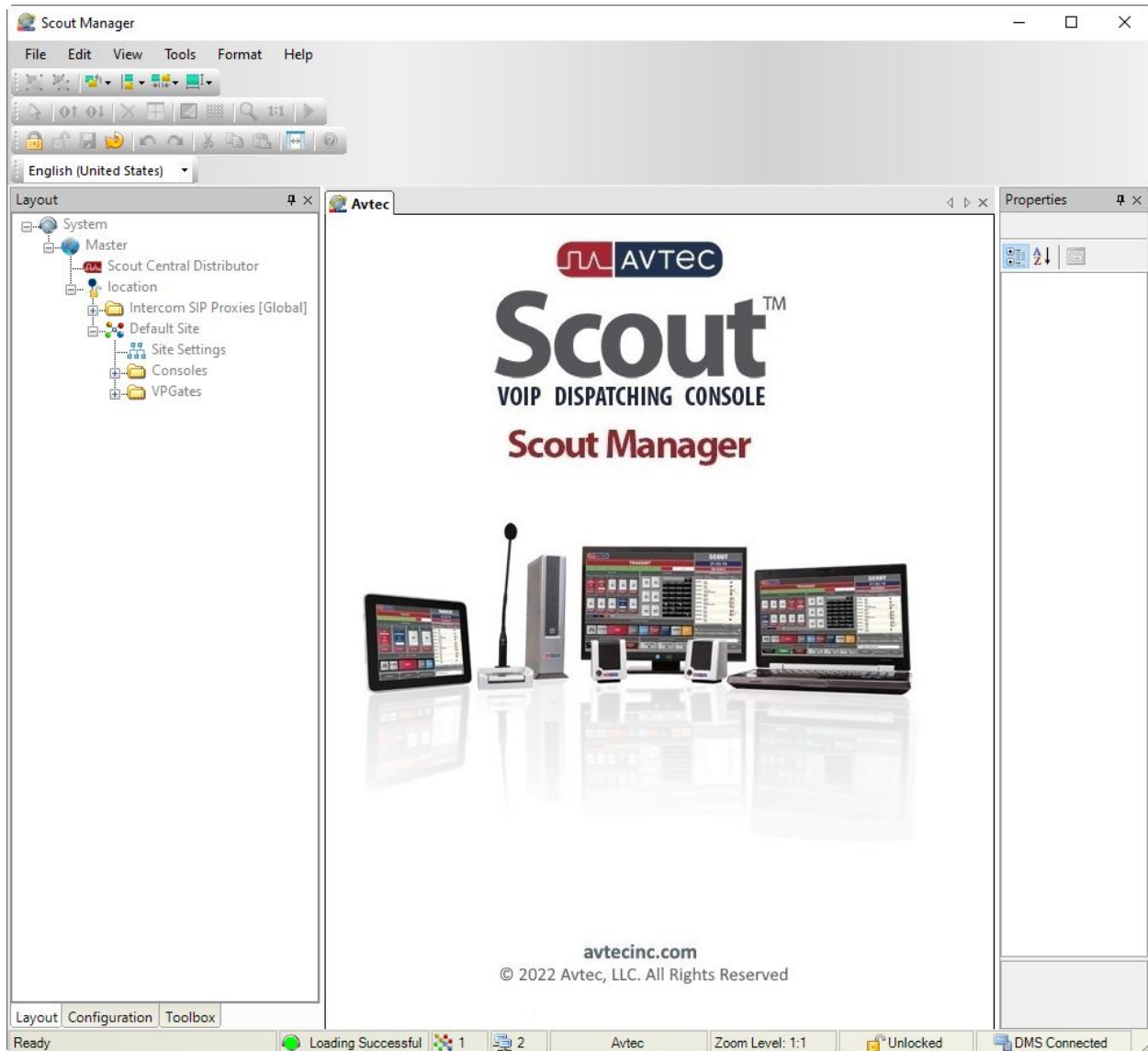
7.1 Launch Scout Manager



Log into the **Scout Manager** application by clicking on the appropriate icon. The following screen is displayed. Log in with the appropriate credentials.

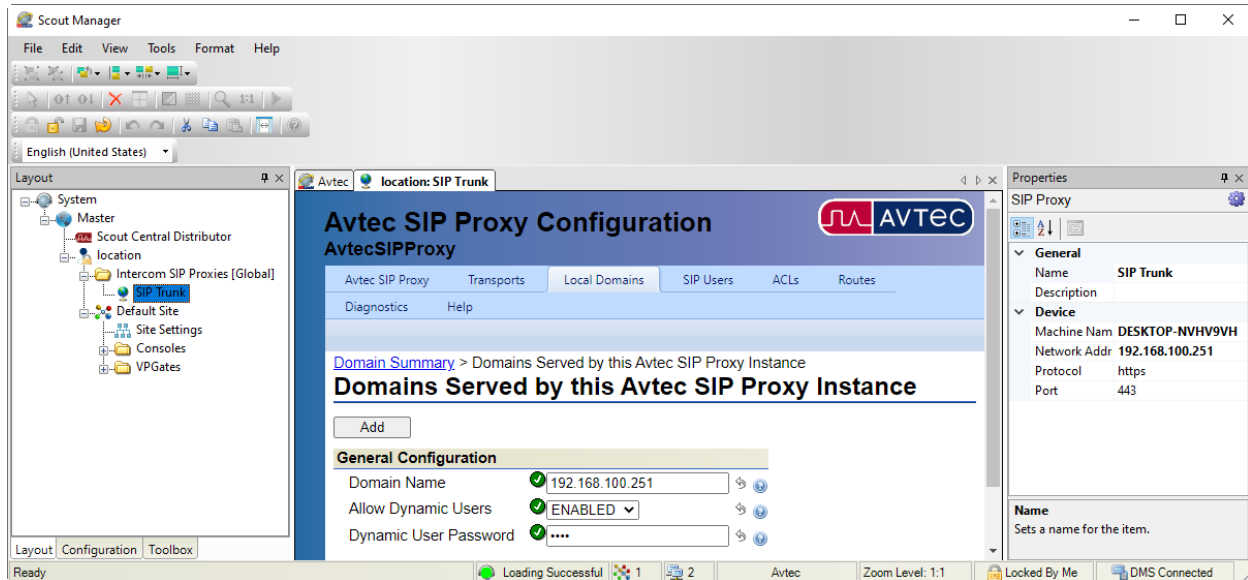
A login dialog box for the Scout Manager application. It features the Avtec logo at the top, which consists of a red square with a white waveform and the word 'AVTEC' in white on a blue background. Below the logo, there are two input fields: 'Username:' and 'Password:'. At the bottom, there are two buttons: 'OK' and 'Close'.

Once logged in, the **Scout Manager** screen appears as shown below. Click on the **Lock** icon to allow configuration.



7.2 Add Local Domain

Create an Avtec SIP Proxy domain. Navigate to **SIP Trunk** → **Local Domains** and click the **Add** button (not shown). The **Domains Served by this Avtec SIP Proxy Instance** page is displayed as shown below. For the **Domain Name**, enter the IP address of the Scout VoIP Console. Enable the **Allow Dynamic Users** option and specify a **Dynamic User Password** to be used by SIP extensions during registration with the Scout SIP Proxy. Click the **Add** button.



7.3 Add Endpoints

Endpoints are created under VPGate configuration. One endpoint was created per SIP line. Navigate to **VPGate → Endpoints** and click the **Add** button in the Endpoint Summary page (not shown). The **Endpoint Configuration** page is displayed as shown below.

Under **Endpoint Configuration**:

- **Endpoint Name:** Specify a descriptive name (e.g., *Line1*).
- **Service State:** Set to *AVAILABLE*.

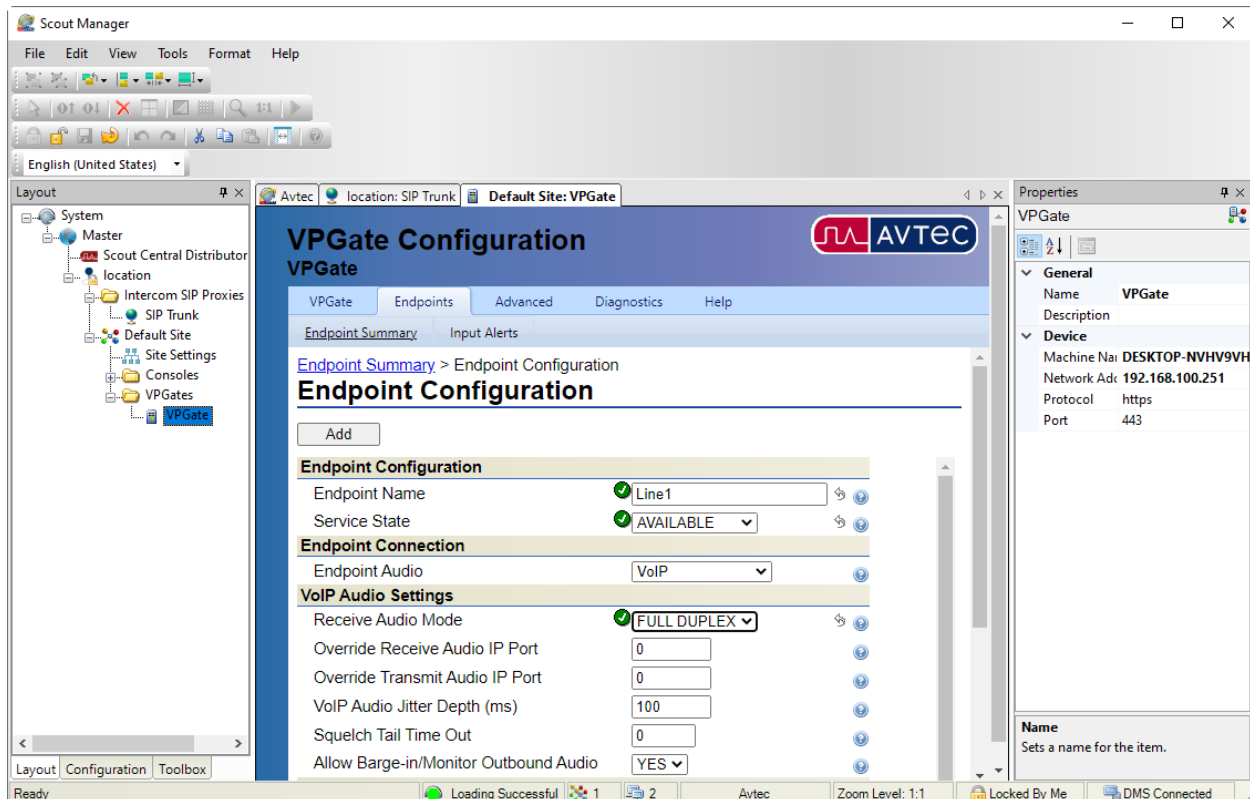
Under **Endpoint Connection**:

- **Endpoint Audio:** Set to *VoIP*.

Under **VoIP Audio Settings**:

- **Receive Audio Mode:** Set to *FULL DUPLEX*.

Use the default settings for the remaining fields. Click the **Add** button.

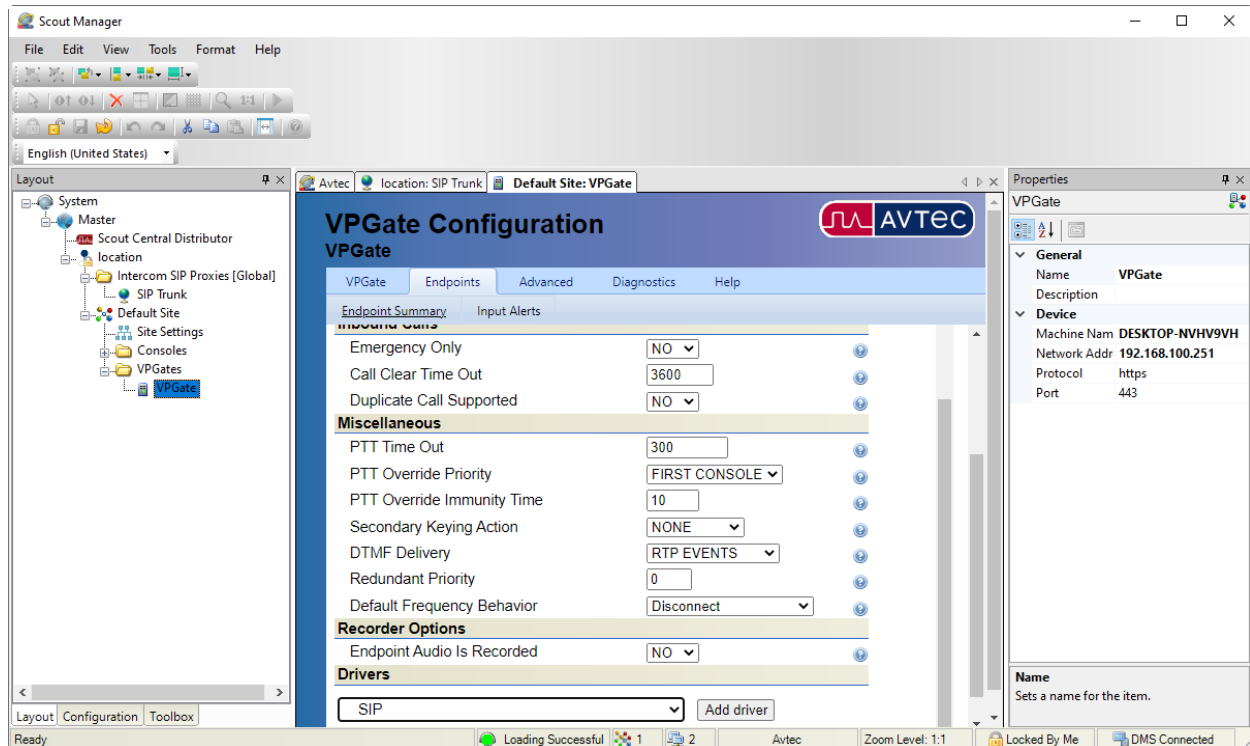


The endpoint previously added is now displayed in the **Endpoint Summary** page shown below. Click on the endpoint that was previously added (i.e., *Line1*) to open the configuration again.

The screenshot displays the Avtec Scout Manager VPGate Configuration application. The main window is titled "VPGate Configuration" and features a sidebar on the left with a tree view showing the system hierarchy: System > Master > Scout Central Distributor > location > Intercom SIP Proxies [Global] > SIP Trunk > Default Site > Site Settings > VPGates > VPGate. The main content area is divided into tabs: VPGate, Endpoints, Advanced, Diagnostics, and Help. The "Endpoints" tab is active, showing the "Endpoint Summary" page. This page includes a "Refresh" button, an "Add" button, and a table of endpoints. The table has columns for "Name" and "Service Stat". The endpoints listed are 78010, 78011, Line1, and Line2, all with a status of "Available". The "Line1" row is highlighted with a red border. On the right side of the window, there is a "Properties" pane for the selected "VPGate" endpoint, showing details such as Name (VPGate), Description, Machine Name (DESKTOP-NVHV9VH), Network Address (192.168.100.251), Protocol (https), and Port (443). The status bar at the bottom indicates "Ready", "Loading Successful", and "Zoom Level: 1:1".

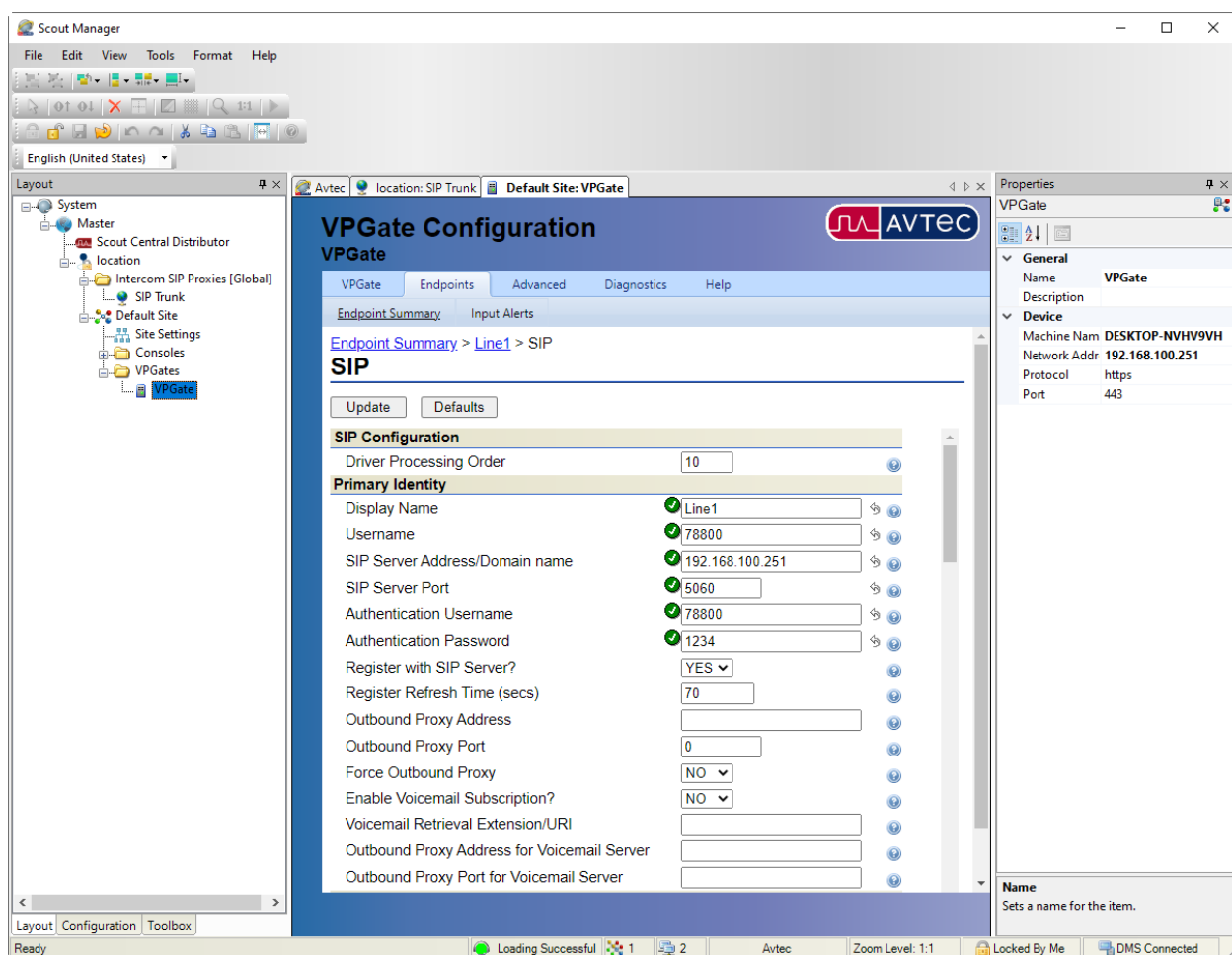
| Name | Service Stat |
|---------|--------------|
| > 78010 | Available |
| > 78011 | Available |
| > Line1 | Available |
| > Line2 | Available |

The **Endpoint Configuration** page is displayed. Scroll to the bottom of the page to the **Drivers** section as shown below. Select *SIP* from the drop-down field and click **Add driver**.

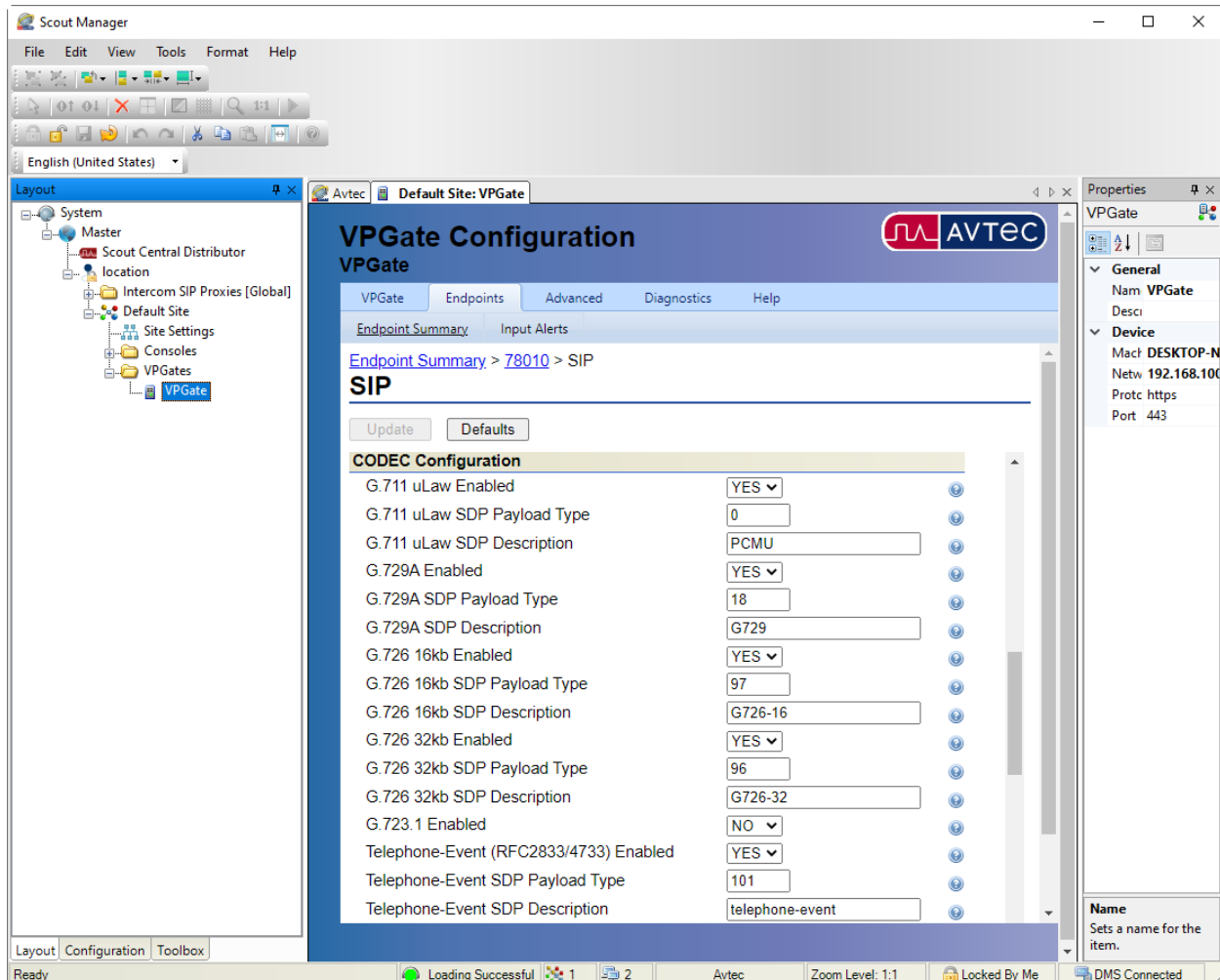


The **SIP** page is displayed as shown below. Under **Primary Identity**, configure the following fields:

- **Display Name:** Specify a descriptive name (e.g., *Line1*).
- **Username:** Specify a descriptive name (e.g., *78800*).
- **SIP Server Address/Domain name:** Specify the IP address of Scout VoIP Console, which is also the local domain.
- **SIP Server Port:** Specify port *5060*.
- **Authentication Username:** Specify the SIP extension (e.g., *78800*).
- **Authentication Password:** Specify the password used for SIP registration as configured in **Section 7.2**.
- **Register with SIP Server:** Enable this option.

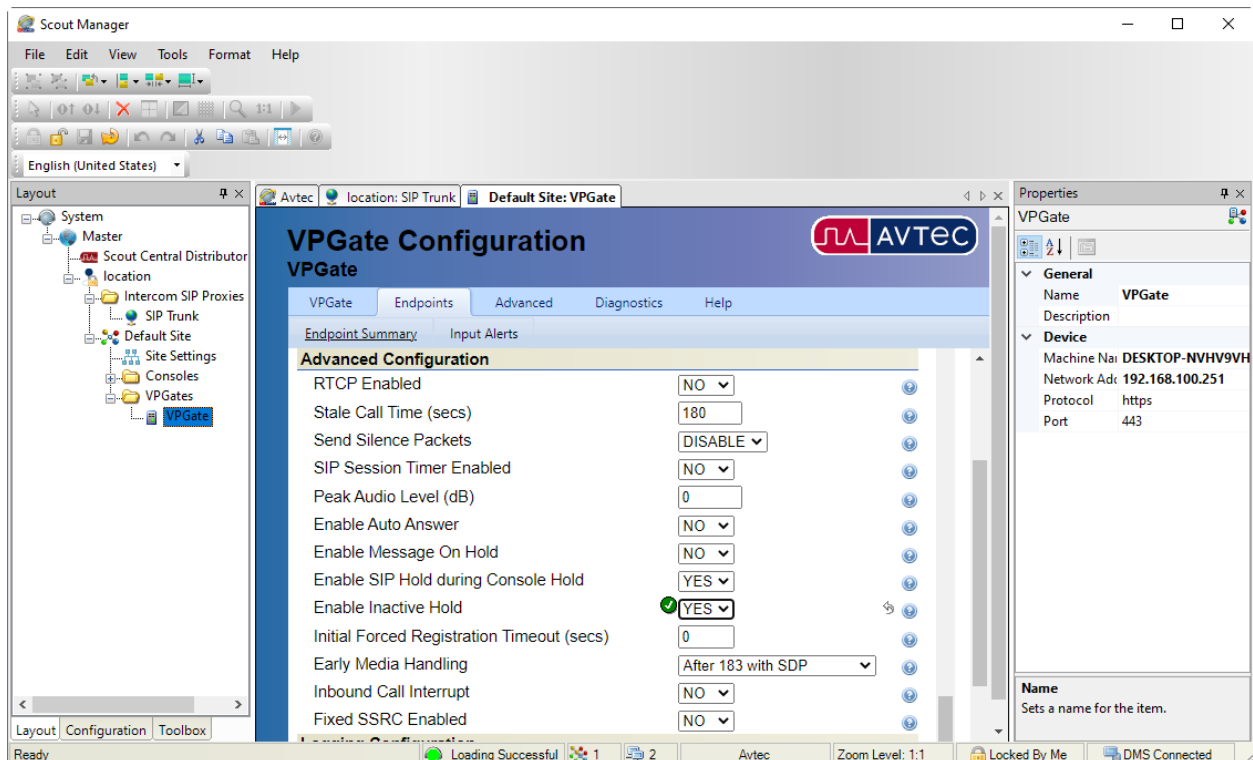


Scroll down to the **Codec Configuration** section and specify the codecs to be supported. In this example, G.711 and G.729 were enabled.



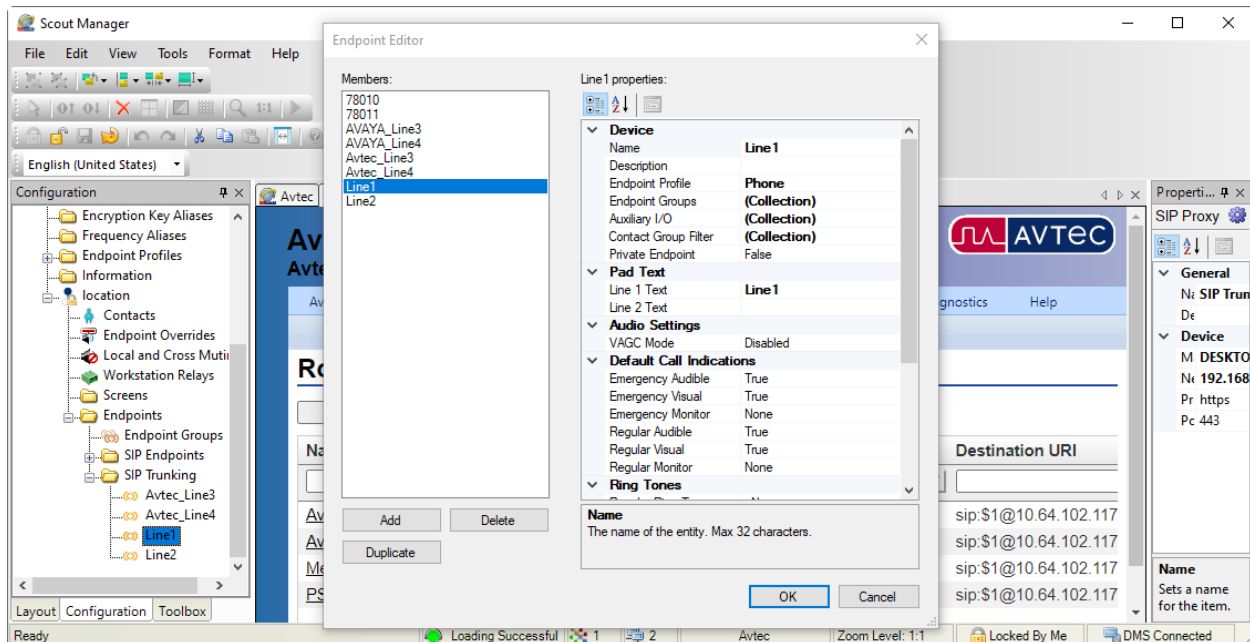
Lastly, scroll down to the **Advanced Configuration** section and **Enable Inactive Hold**. This is required for attended/supervised transfers to work properly. Click the **Add** button.

Note: Although outside the scope of these Application Notes, it is worth noting that if PSTN connectivity provided via Avaya Session Border Controller for Enterprise and shuffling is enabled, *Delayed SDP Handling* may need to be enabled to support attended/supervised transfers over the PSTN.



7.4 Modify SIP Line Label and Set Endpoint Profile

To modify the label of the SIP line button on Scout VoIP Console and set the endpoint profile, select the **Configuration** tab in the left pane, double-click on the SIP endpoint added above. In the **Endpoint Editor**, change **Name** and **Line 1 Text** to the SIP extension (e.g., *Line1*). Set **Endpoint Profile** to *Phone* so that dial tone and ringback (or other audio feedback, such as busy tone) is heard when Scout Console goes off-hook and places an outbound call.



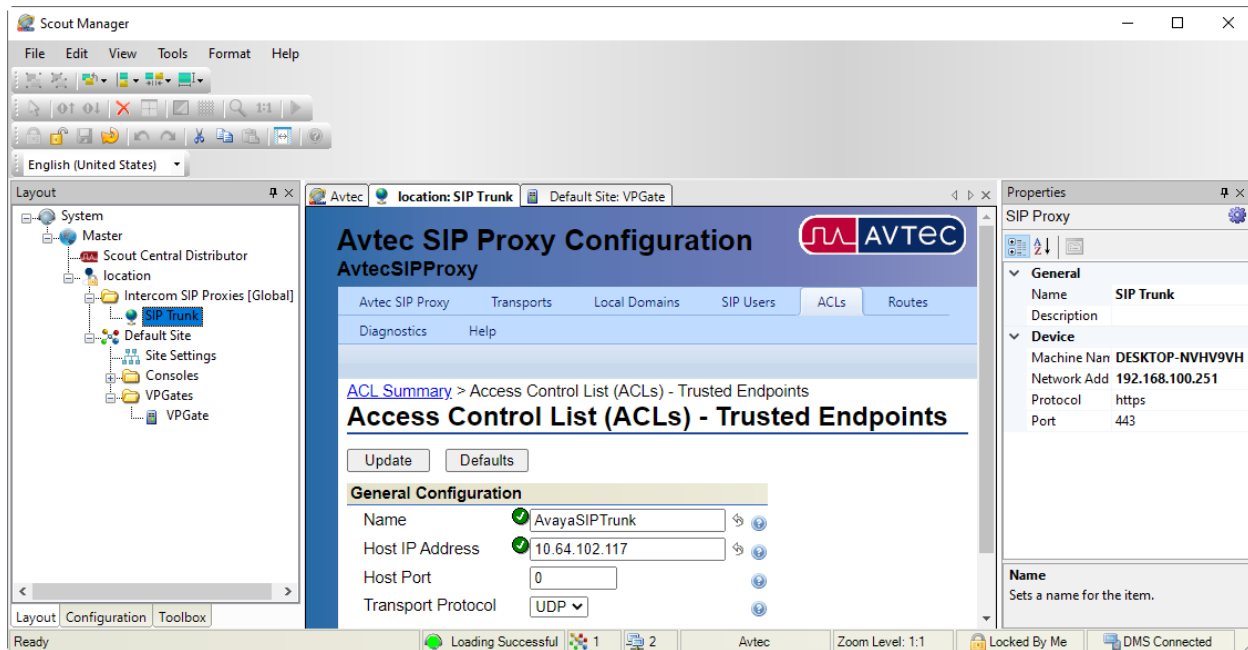
7.5 Add Access Control List (ACLs) – Trusted Endpoint

Configure an access control list (ACL) for SIP trunking. An ACL is added to trust calls from Session Manager. Navigate to **SIP Trunk** → **ACLs** and click the **Add** button as shown below (not shown).

The **Access Control List (ACL) – Trusted Endpoints** page is displayed as shown below. Configure the following parameters as follows:

- **Name:** Provide a descriptive name (e.g., *AvayaSIPTrunk*).
- **Host IP Address:** Specify the SIP signaling interface of Session Manager (i.e., *10.64.102.117*).
- **Host Port:** Enter '0' to trust traffic coming from any port.
- **Transport Protocol:** Specify the transport protocol (i.e., *UDP*).

Click the **Add** button.

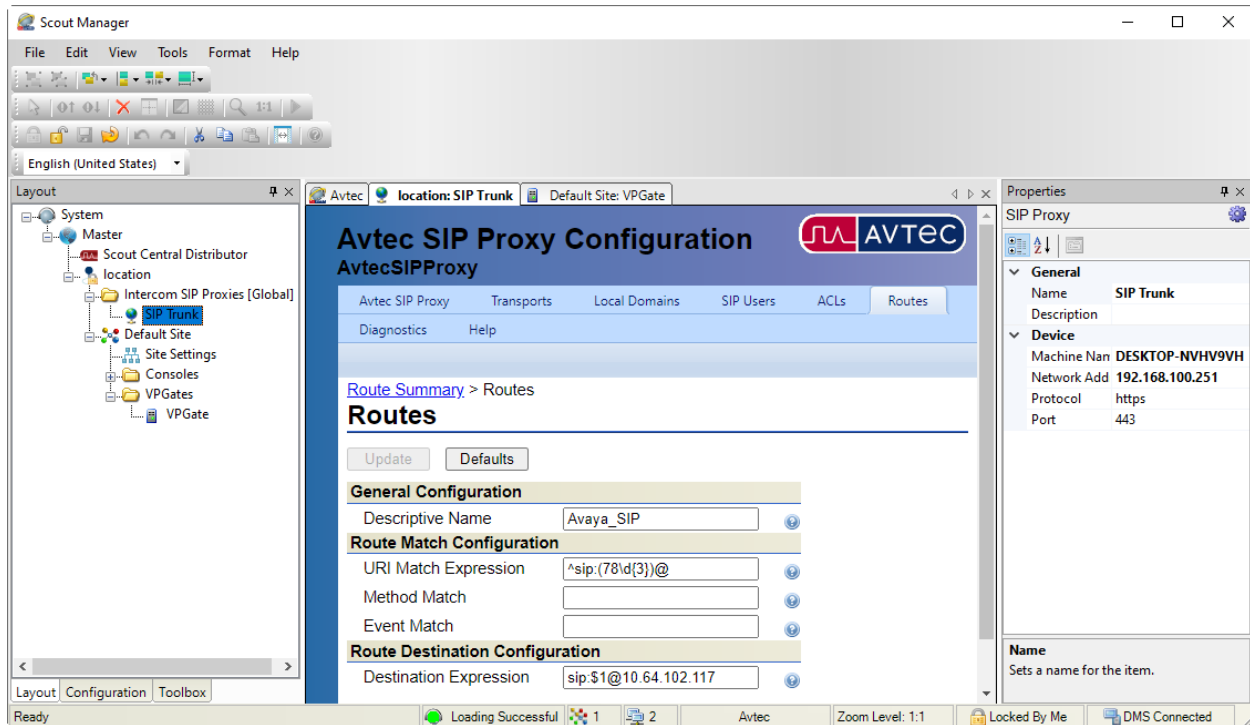


7.6 Add Routes

Configure routes for calls to Session Manager destined for local H.323 and SIP stations, voicemail system, and PSTN, as necessary. Navigate to **SIP Trunk** → **Routes** and click the **Add** button (not shown). The **Routes** page is displayed. For example, to route calls to local H.323 stations, the following parameters were configured:

- **Descriptive Name:** Provide a descriptive route name (e.g., *Avaya_SIP*).
- **URI Match Expression:** Enter an expression that identifies a route for calls with a 5-digit dial number starting with '78', such as *^sip:(78\d{3})@*
- **Destination Expression:** Enter the destination URI that routes calls to Session Manager. In this case, the expression used was *sip:\$1@10.64.102.117*.

Click the **Add** button.



Add additional Routes as necessary. For the compliance test, the following Routes were configured.

The screenshot displays the Avtec SIP Proxy Configuration web interface. The left sidebar shows a tree view of the system configuration, including Master, Scout Central Distributor, location, Intercom SIP Proxies, SIP Trunk, Default Site, Site Settings, Consoles, VP Gates, and VPGate. The main content area is titled 'Avtec SIP Proxy Configuration' and 'AvtecSIPProxy'. It features a navigation bar with tabs: Avtec SIP Proxy, Transports, Local Domains, SIP Users, ACLs, Routes, Diagnostics, and Help. The 'Routes' tab is selected, showing a 'Route Summary' table. The table has columns for Name, Match URI, Method, Event, and Destination URI. It lists four routes: Avaya_H323, Avaya_SIP, Messaging, and PSTN, all with a Match URI of ^sip:(77\d{3})@, ^sip:(78\d{3})@, ^sip:(878600)@, and ^sip:(91\d{10})@ respectively, and a Destination URI of sip:\$1@10.64.102.117. An 'Add' button is located above the table. The right sidebar shows the 'Properties' panel for the selected route, with sections for General, Device, and Name. The status bar at the bottom indicates 'Ready', 'Loading Successful', and 'Zoom Level: 1:1'.

| Name | Match URI | Method | Event | Destination URI |
|------------|------------------|--------|-------|-----------------------|
| Avaya_H323 | ^sip:(77\d{3})@ | | | sip:\$1@10.64.102.117 |
| Avaya_SIP | ^sip:(78\d{3})@ | | | sip:\$1@10.64.102.117 |
| Messaging | ^sip:(878600)@ | | | sip:\$1@10.64.102.117 |
| PSTN | ^sip:(91\d{10})@ | | | sip:\$1@10.64.102.117 |

7.7 Deploy the Configuration

Lastly, deploy the changes to the **Scout VoIP Console**. Select the **Layout** tab in the left pane and then open the **Scout Central Distributor** screen as shown below. Navigate to **System** → **Deployment** to display the following screen. Click on the **Deploy** hyperlink.

The screenshot displays the Scout Manager application window. The title bar reads 'Scout Manager'. The menu bar includes 'File', 'Edit', 'View', 'Tools', 'Format', and 'Help'. The toolbar contains various icons for file operations and navigation. The left sidebar shows a tree view with 'System' expanded, containing 'Master', 'Scout Central Distributor', 'Location', 'Intercom SIP Proxy', 'Default Site', 'Site Settings', 'Consoles', 'VP Gates', and 'VP Gate'. The main content area is titled 'Scout Central Distributor' and features a navigation bar with 'Dash', 'System', 'Alarms', 'Reports', and 'Licensing'. Below this, there are tabs for 'SYSTEM' and 'DEPLOYMENTS'. The 'DEPLOYMENTS' tab is active, showing 'System > Deployments' with a 'Deploy' button. The 'Locations' section displays a circular progress indicator for 'location' with a 'V22' label. Below this, there is a 'Deployments' table with columns 'Version', 'User', and 'Date/Time'. The table contains one row: '22', 'Avtec', '11/17/22 5:20:32 pm'. To the right of the table is an 'Activity & Errors' section with columns 'Component', 'Location', 'User', and 'Version'. The right sidebar shows 'Properties' for 'Distributor Settings' with a 'Device' section containing 'Network Addr: 192.168.100.251', 'Protocol: https', and 'Port: 443'. At the bottom, there is a 'Network Address' section with a description: 'Sets the address of the device to be configured. This may be an IP address o...'. The status bar at the bottom shows 'Ready', 'Loading Successful', '1', '2', 'Avtec', 'Zoom Level: 1:1', 'Locked By Me', and 'DMS Connected'.

In the **Select Deployment Locations** screen shown below, click the **Deploy** button to deploy the changes to the **Scout VoIP Console**.

Select Deployment Locations

Select deployment location(s) and then click **Deploy**.

☒ [Select All](#) [Missing Locations?](#)

| Location Name | |
|-------------------------------------|----------|
| <input checked="" type="checkbox"/> | location |

The **Scout VoIP Console** below displays the SIP line configured above with *Line1* as the button label.



8 Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Avtec Scout VoIP Console with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The following steps can be used to verify installations in the field.

1. Launch the Avtec Scout VoIP Console. The Scout VoIP Console will be displayed as shown below. If the SIP trunk is down, the line buttons will display *Unavailable*. The line buttons (i.e., *Line1* and *Line2*) shown below indicate that the SIP trunk is in-service.



- Verify that the SIP trunk between Session Manager and Scout VoIP Console is up by navigating to **Home → Elements → Session Manager → System Status → SIP Entity Monitoring** on System Manager. Below is the status of the SIP trunk to Scout VoIP Console.

AVAYA Aura® System Manager 10.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾

Search 🔍 🔔 ☰ | admin

Home Session Manager

Network Configur... ▾
Device and Locati... ▾
Application Conf... ▾
System Status ▾
Load Factor
SIP Entity Monit...
Managed Band...
Security Module...
SIP Firewall Status

SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

Status Details for the selected Session Manager:

All Entity Links to SIP Entity: Avtec Scout

Summary View

1 Item Filter: Enable

| | Session Manager Name | Session Manager IP Address Family | SIP Entity Resolved IP | Port | Proto. | Deny | Conn. Status | Reason Code | Link Status |
|-----------------------|---------------------------|-----------------------------------|------------------------|------|--------|-------|--------------|-------------|-------------|
| <input type="radio"/> | devcon-sm | IPv4 | 192.168.100.251 | 5060 | UDP | FALSE | UP | 200 OK | UP |

Select : None

- Verify that the SIP trunk between Communication Manager and Session Manager is in-service using the **status trunk** command on Communication Manager.
- Place an incoming call to Scout VoIP Console and answer the call. Verify two-way audio is provided.
- Place an outgoing call from Scout VoIP Console to an Avaya local station or PSTN and answer the call. Verify two-way audio is provided.

9 Conclusion

These Application Notes describe the configuration steps required to integrate Avtec Scout VoIP Console with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. A SIP trunk was established between Avtec Scout VoIP Console and Avaya Aura® Session Manager and basic telephony features were verified. All feature and serviceability test cases were completed successfully with observations noted in **Section 2.2**.

10 References

This section references the product documentation relevant to these Application Notes. Avtec Scout VoIP Console documentation is available through the application via online help.

- [1] *Administering Avaya Aura® Communication Manager*, Release 10.1.x, Issue 1, December 2021, available at <http://support.avaya.com>.
- [2] *Administering Avaya Aura® System Manager*, Release 10.1.x, Issue 6, June 2022, available at <http://support.avaya.com>.
- [3] *Administering Avaya Aura® Session Manager*, Release 10.1.x, Issue 3, April 2022, available at <http://support.avaya.com>.
- [4] *Administering Avaya Session Border Controller for Enterprise*, Release 10.1.x, Issue 1, December 2021, available at <http://support.avaya.com>.

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