



DevConnect Program

Application Notes for Avtec Scout VoIP Console 5.6 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.

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. Avtec Scout VoIP Console integrates with Avaya Aura® Session Manager via a SIP trunk in the tested configuration.

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.
- 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 is only heard through external speakers by default. Scout VoIP Console can be configured to receive notification through the headset, but compliance testing used the external speakers.
- 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.

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 the compliance test configuration. Scout VoIP Console connects to Session Manager via a SIP trunk linked to the SIP Proxy component installed on Windows 10. Scout Manager was used to configure Scout VoIP Console System.

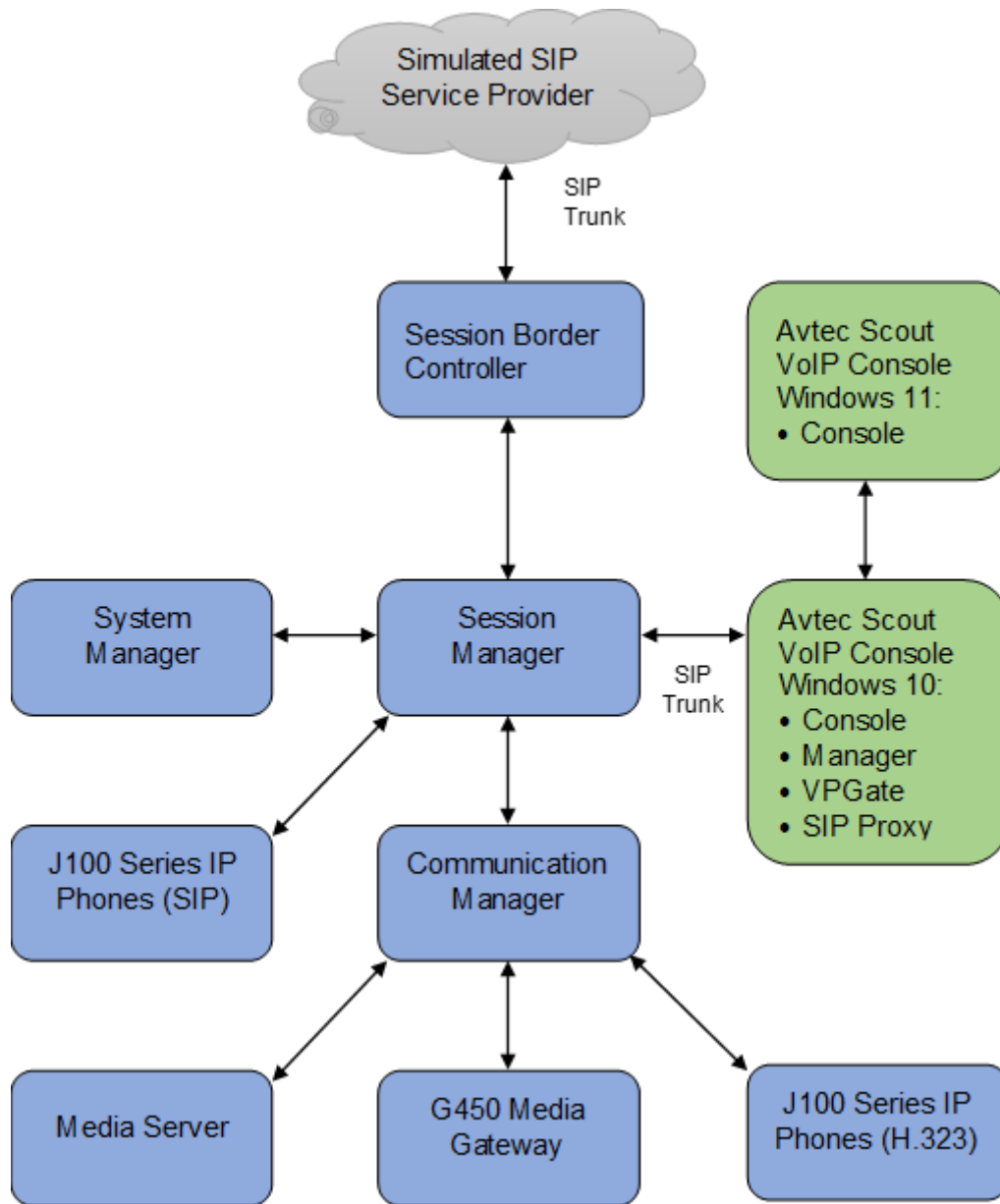


Figure 1: Avaya SIP Telephony Network with Avtec Scout VoIP Console

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	10.1.3.1.0-FP3 SP1 01.0.974.0-27937
Avaya G450 Media Gateway	FW 42.24.0
Avaya Aura® Media Server	v.10.1.0.154
Avaya Aura® System Manager	10.1.3.1 Feature Pack 3 SP1 10.1.3.1.0716418
Avaya Aura® Session Manager	10.1.3.1 Feature Pack 3 SP1 10.1.3.1.1013103
Avaya Session Border Controller	10.1.2.0-64-23285
Avaya J100 SIP Telephones	4.1.2.0.11 (SIP) 6.8.5.4.10 (H.323)
Avtec Scout VoIP Console including the following components: <ul style="list-style-type: none">Scout Console (Windows 10 and Windows 11)Scout Central Distributor (Windows 10)Scout VPGate (Windows 10)Scout SIP Proxy (Windows 10)Scout Manager (Windows 10)	5.6.0.5 5.6.1.6 5.6.1.10 5.6.0.2 5.6.0.5

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 (*sm10*). 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	
aes10	10.64.110.247	
ams10	10.64.110.214	
default	0.0.0.0	
procr	10.64.110.213	
procr6	::	
sm10	10.64.110.212	
(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

1: 1-srtp-aescm128-hmac80

2: 10-srtp-aescm256-hmac80

3: none

4:

5:

Encrypted SRTP: best-effort

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	NR Group: 1	
Location: 1	Authoritative Domain: avaya.com	
Name: Main	Stub Network Region: n	
MEDIA PARAMETERS	Intra-region IP-IP Direct Audio: yes	
Codec Set: 1	Inter-region IP-IP Direct Audio: yes	
UDP Port Min: 2048	IP Audio Hairpinning? n	
UDP Port Max: 65535		
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 the Session Manager (*sm10*) 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 1		Page 1 of 3
SIGNALING GROUP		
Group Number: 1	Group Type: sip	
IMS Enabled? n	Transport Method: tls	
Q-SIP? n		
IP Video? y	Priority Video? n	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: sm10	
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 1		Page 1 of 4	
TRUNK GROUP			
Group Number: 1	Group Type: sip	CDR Reports: y	
Group Name: SM Trunk 1	COR: 1	TN: 1	TAC: 101
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: 1		
	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 1		Page 3 of 4	
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		
Modify Tandem Calling Number: no			
Show ANSWERED BY on Display? y			
DSN Term? n			

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

add trunk-group 1	Page 4 of 4
PROTOCOL VARIATIONS	
Mark Users as Phone? n	
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n	
Send Transferring Party Information? n	
Network Call Redirection? n	
Send Diversion Header? n	
Support Request History? y	
Telephone Event Payload Type:	
Convert 180 to 183 for Early Media? n	
Always Use re-INVITE for Display Updates? n	
Identity for Calling Party Display: P-Asserted-Identity	
Block Sending Calling Party Location in INVITE? n	
Accept Redirect to Blank User Destination? n	
Enable Q-SIP? n	
Interworking of ISDN Clearing with In-Band Tones: keep-channel-active	
Request URI Contents: may-have-extra-digits	

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 1, have the extension sent to Scout VoIP Console.

change private-numbering 0	Page 1 of 2				
NUMBERING - PRIVATE FORMAT					
Ext Len	Ext Code	Trk Grp(s)	Private Prefix	Total Len	
5	7			5	Total Administered: 1 Maximum Entries: 540

5.6 Administer 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 ‘7013’. This would cover call routing to the Scout VoIP Console extensions (i.e., 70131 – 70133). For the compliance test, three Scout VoIP Console lines were configured as shown in **Section 7.3**.

change uniform-dialplan 7	Page 1 of 2				
UNIFORM DIAL PLAN TABLE					
Percent Full: 0					
Matching Pattern	Len	Del	Insert Digits	Net Conv	Node Num
7013	5	0	aar	n	

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and add an entry that routes digits beginning with “7013” to route pattern 1 as shown below. Note that the **Call Type** was set to *lev0*. This routes calls to SIP stations and to Scout VoIP Console.

change aar analysis 7							Page 1 of 2
AAR DIGIT ANALYSIS TABLE							
Location: all							Percent Full: 2
	Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
	7013	5	5	1	lev0		n
	71	5	5	1	aar		n
	8	5	5	1	aar		n
							n
							n

Configure a preference in **Route Pattern 1** to route calls over SIP trunk group 1 as shown below.

change route-pattern 1										Page	1 of	4
Pattern Number: 1										Pattern Name: To main		
SCCAN? n		Secure SIP? n		Used for SIP stations? n								
Grp No	FRL	NPA	Pfx	Hop	Toll	No.	Inserted			DCS/	IXC	
			Mrk	Lmt	List	Del	Digits			QSIG		
							Dgts			Intw		
1:	1	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	Request			Dgts	Format	
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**

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.

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

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 IP address of the Windows system where the SIP Proxy component is running.

The screenshot shows the Avaya Aura System Manager 10.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The 'Routing' tab is selected. The 'Adaptation Details' form is displayed, with the 'General' tab active. The form contains the following fields:

- Adaptation Name:** Avtec Adaptation
- Notes:** (empty)
- Module Name:** DigitConversionAdapter (selected from a dropdown)
- Type:** digit
- State:** enabled (selected from a dropdown)
- Module Parameter Type:** Single Parameter (selected from a dropdown)
- Module Parameter:** 10.64.49.8
- Egress URI Parameters:** (empty)

Below the form are two sections for digit conversion:

- Digit Conversion for Incoming Calls to SM:** Includes an 'Add' button, a 'Remove' button, and a table with columns: Matching Pattern, Min, Max, Phone Context, Delete Digits, Insert Digits, Address to modify, Adaptation Data, and Notes. The table is currently empty.
- Digit Conversion for Outgoing Calls from SM:** Includes an 'Add' button, a 'Remove' button, and a table with the same columns as the incoming section. The table is currently empty.

At the bottom right of the form are 'Commit' and 'Cancel' buttons.

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 lines with different extensions for this compliance test. These SIP extensions registered with the Scout VoIP Console SIP 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*.
- **Location:** Select the location defined previously (not shown).
- **Time Zone:** Time zone for this location.

Under *Adaptations*, select **Add** and under **Name** select the adaptation added in **Section 6.1**. Defaults can be used for the remaining fields. Click **Commit**.

The screenshot displays the Avaya Aura System Manager 10.1 interface. The top navigation bar includes the Avaya logo, 'Aura® System Manager 10.1', and various menu items like Users, Elements, Services, Widgets, and Shortcuts. A search bar and a user profile icon are also present. The main content area is titled 'SIP Entity Details' and has a 'Commit' button. The 'General' section contains the following fields:

- Name:** Avtec Scout
- FQDN or IP Address:** 10.64.49.8
- Type:** SIP Trunk
- Notes:** (empty)
- Location:** (dropdown)
- Time Zone:** America/Denver
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty)
- Securable:** ☐
- Call Detail Recording:** none

The 'Adaptations' section shows a table with one entry:

Order	Name	Module Name	State	Type	Notes
1	Avtec Adaptation	DigitConversionAdapter	enabled	digit	

At the bottom of the 'Adaptations' section, there is a 'Select : All, None' option.

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., *sm10_Avtec Scout_5060_UDP_IPV4*).
- **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 top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. Below this is a search bar and a user profile 'admin'. The main content area is titled 'Entity Links' and contains a table with one item. The table columns are Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, DNS Override, and Connection Policy. The row shows 'sm10_Avtec Scout_5060' as the Name, 'sm10' as SIP Entity 1, 'UDP' as Protocol, '5060' as Port, 'Avtec Scout' as SIP Entity 2, '5060' as Port, 'DNS Override' as checked, and 'trusted' as Connection Policy. There are 'Commit' and 'Cancel' buttons at the top right of the table area.

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.

AVAYA Users Elements Services Widgets Shortcuts Search admin

Aura® System Manager 10.1

Home Routing

R...

Routing Policy Details Commit Cancel Help ?

General

* Name: Avtec Scout

Disabled: ☐

* Retries: 0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Avttec Scout	10.64.49.8	SIP Trunk	

Time of Day

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/> 0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

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 '7013' 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 top navigation bar includes the Avaya logo, version information, and various menu items like Users, Elements, Services, Widgets, and Shortcuts. The main content area is titled 'Dial Pattern Details' and contains two sections: 'General' and 'Originating Locations and Routing Policies'.

General Section:

- Pattern:** 7013
- Min:** 5
- Max:** 5
- Emergency Call:** ☐
- SIP Domain:** -ALL-
- Notes:** (empty text box)

Originating Locations and Routing Policies Section:

This section includes an 'Add' button and a 'Remove' button. Below these is a table with one item selected. The table has columns for Originating Location Name, Originating Location Notes, Routing Policy Name, Rank, Routing Policy Disabled, Routing Policy Destination, and Routing Policy Notes.

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
-ALL-		Avtec Scout	0	<input type="checkbox"/>	Avtec Scout	

At the bottom of the table, there is a 'Select : All, None' option.

7 Configure Avtec Scout VoIP Console

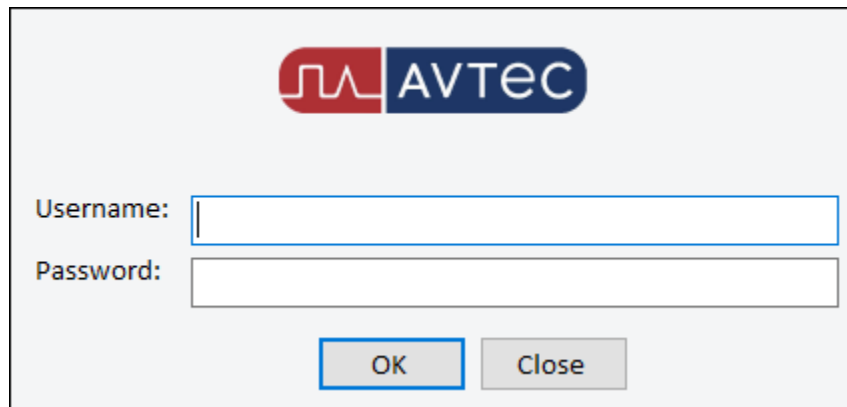
This section covers the configuration of Scout VoIP Console using the **Scout Manager** application. This section assumes that the Scout VoIP Console software has already been installed successfully. In the **Scout Manager** application, the following procedures are performed:

- 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
- Add SIP Transports
- Deploy the Configuration

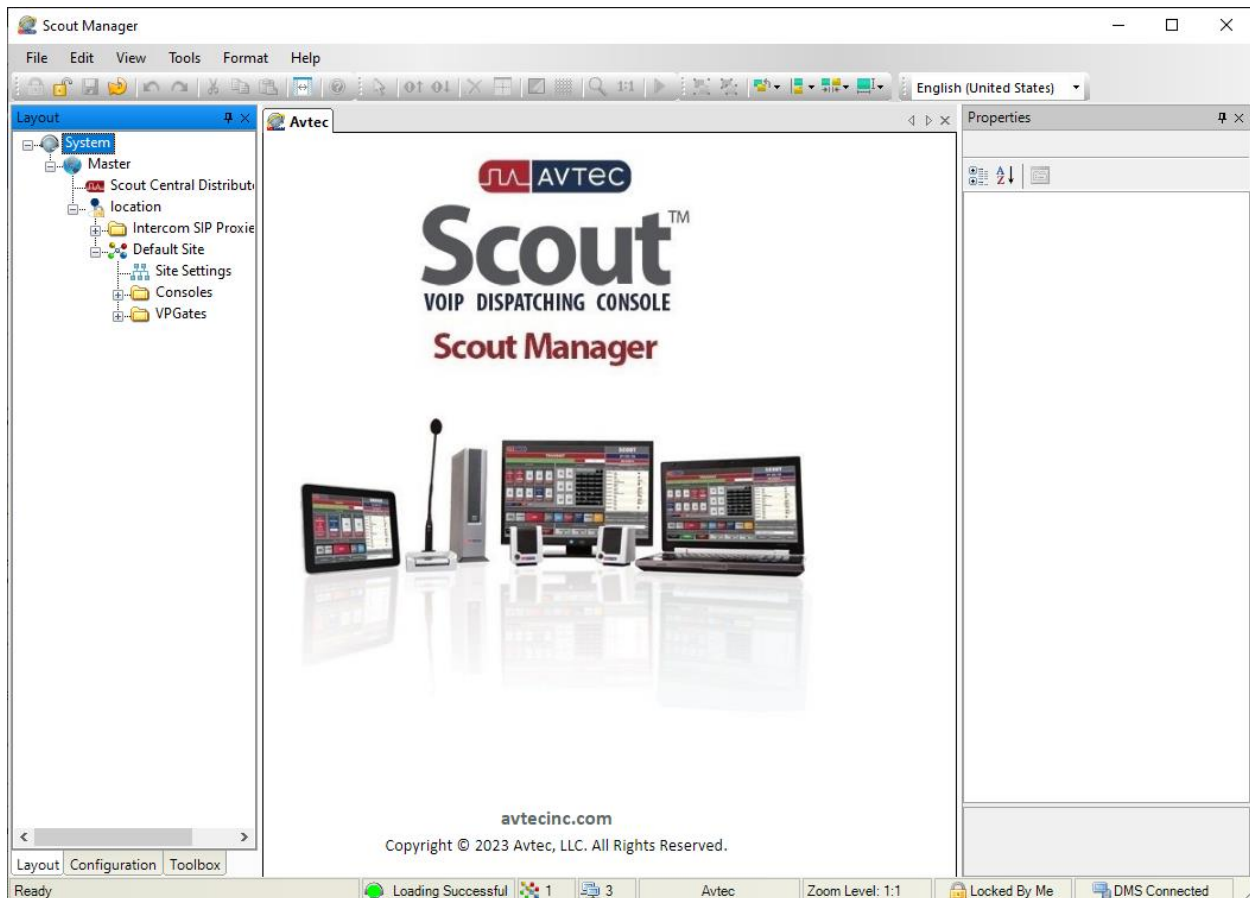
7.1 Launch Scout Manager



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

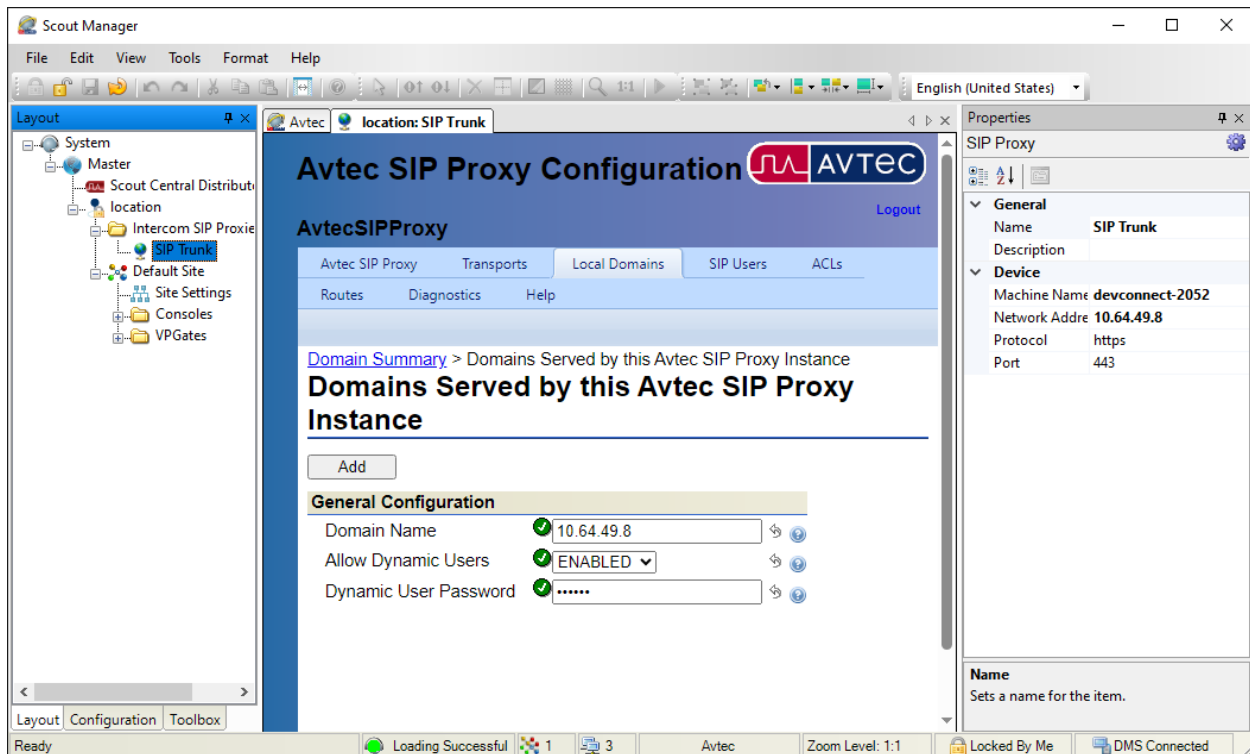
A login dialog box for the Avtec Scout Manager application. It features the Avtec logo at the top, which consists of a red square with a white waveform and the word 'AVTEC' in white on a dark blue background. Below the logo, there are two input fields: 'Username:' and 'Password:'. At the bottom, there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

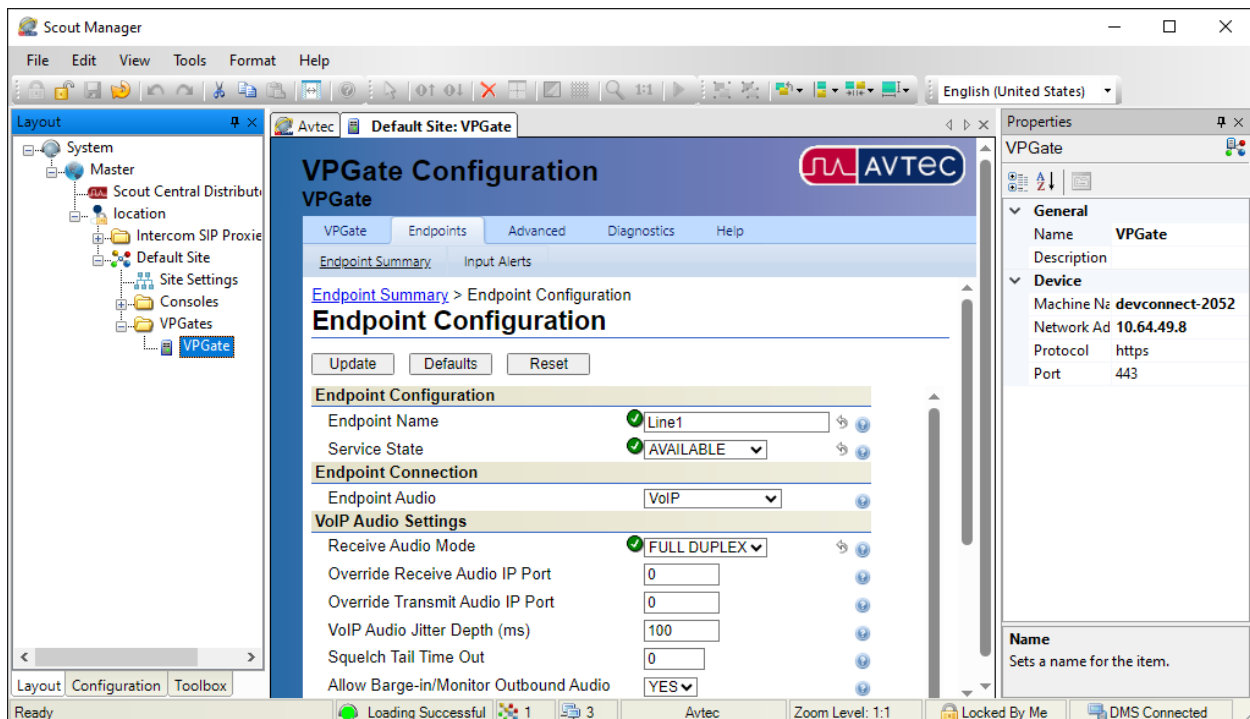
Under **Endpoint Connection**:

- **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 Scout Manager application window. The main pane shows the 'VPGate Configuration' interface with the 'Endpoint Summary' tab selected. The left sidebar shows a tree view with 'VPGate' highlighted. The right sidebar shows the 'Properties' for the selected 'VPGate'.

VPGate Configuration

VPGate

Endpoint Summary

Refresh Add

Expand All / Collapse All

Name
<input type="text"/>
> 70111
> 70112
> 70113
> Line1
> Line2
> Line3

Properties

VPGate

General

Name VPGate

Description

Device

Machine Name devconnect-2052

Network Address 10.64.49.8

Protocol https

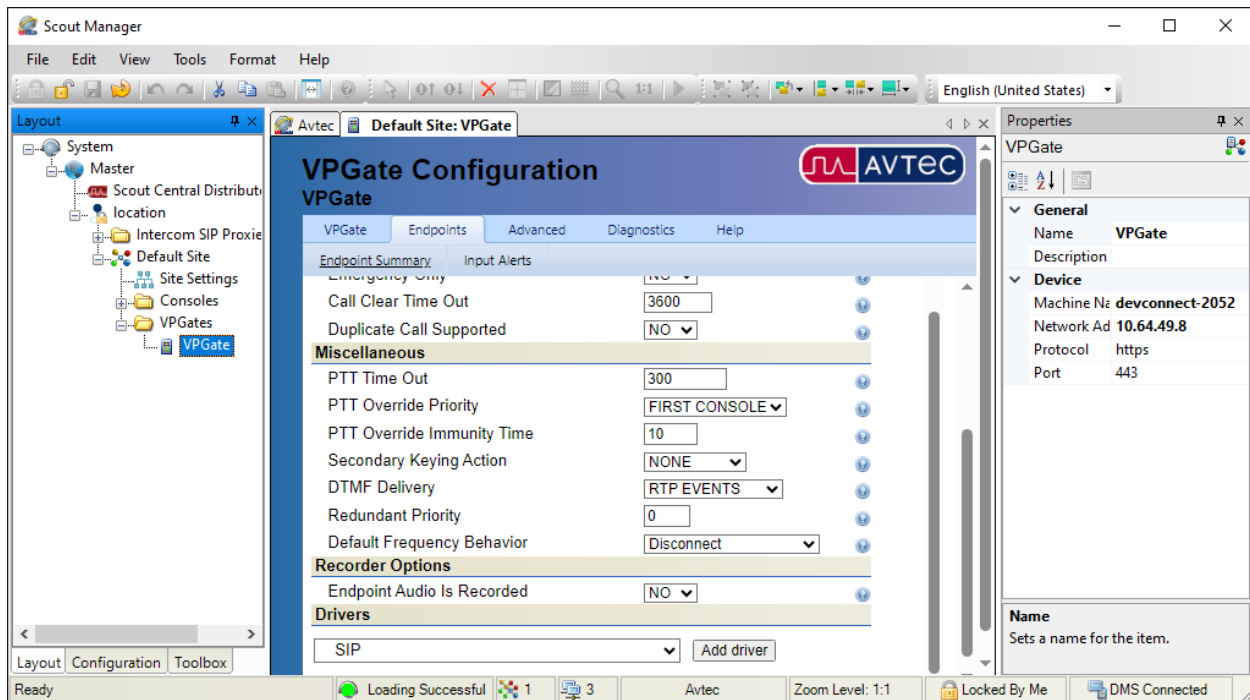
Port 443

Name

Sets a name for the item.

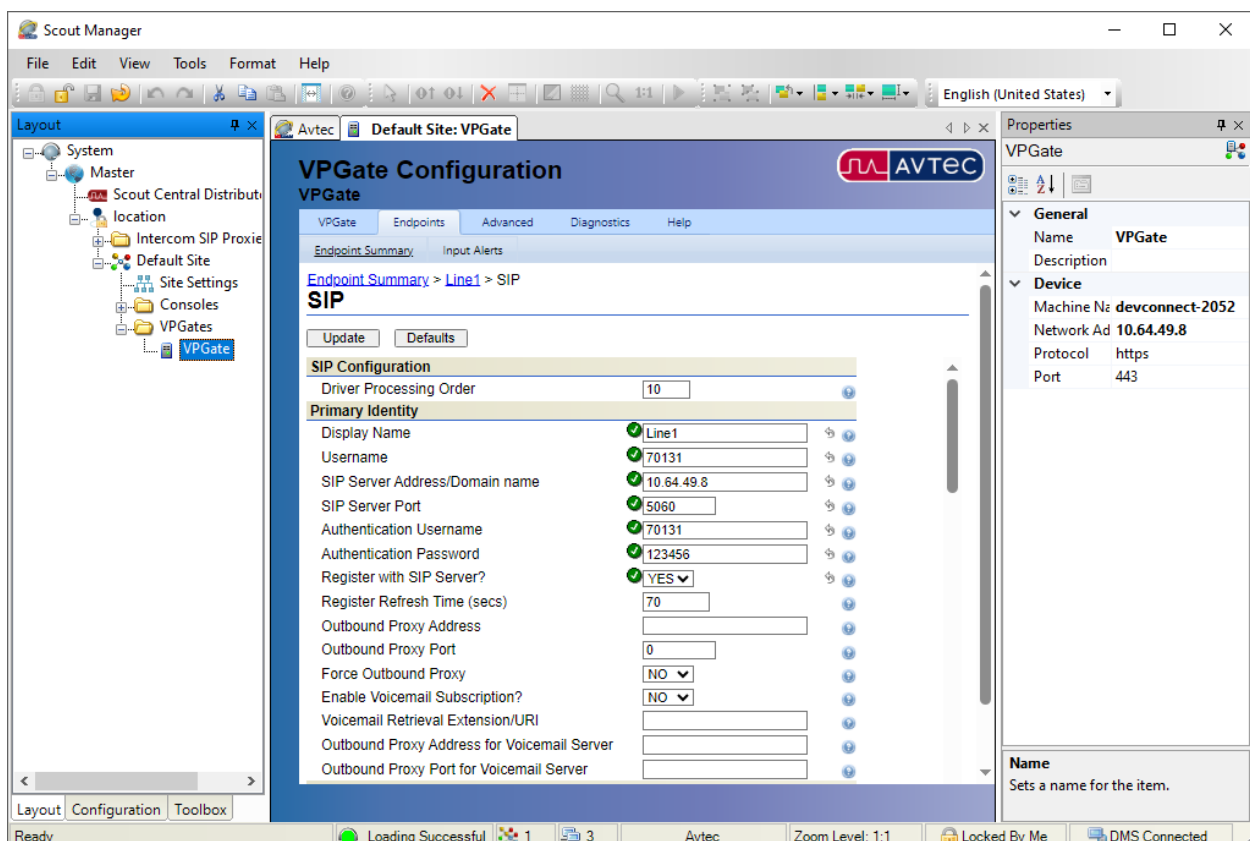
Ready Loading Successful 1 3 Avtec Zoom Level: 1:1 Locked By Me DMS Connected

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**.

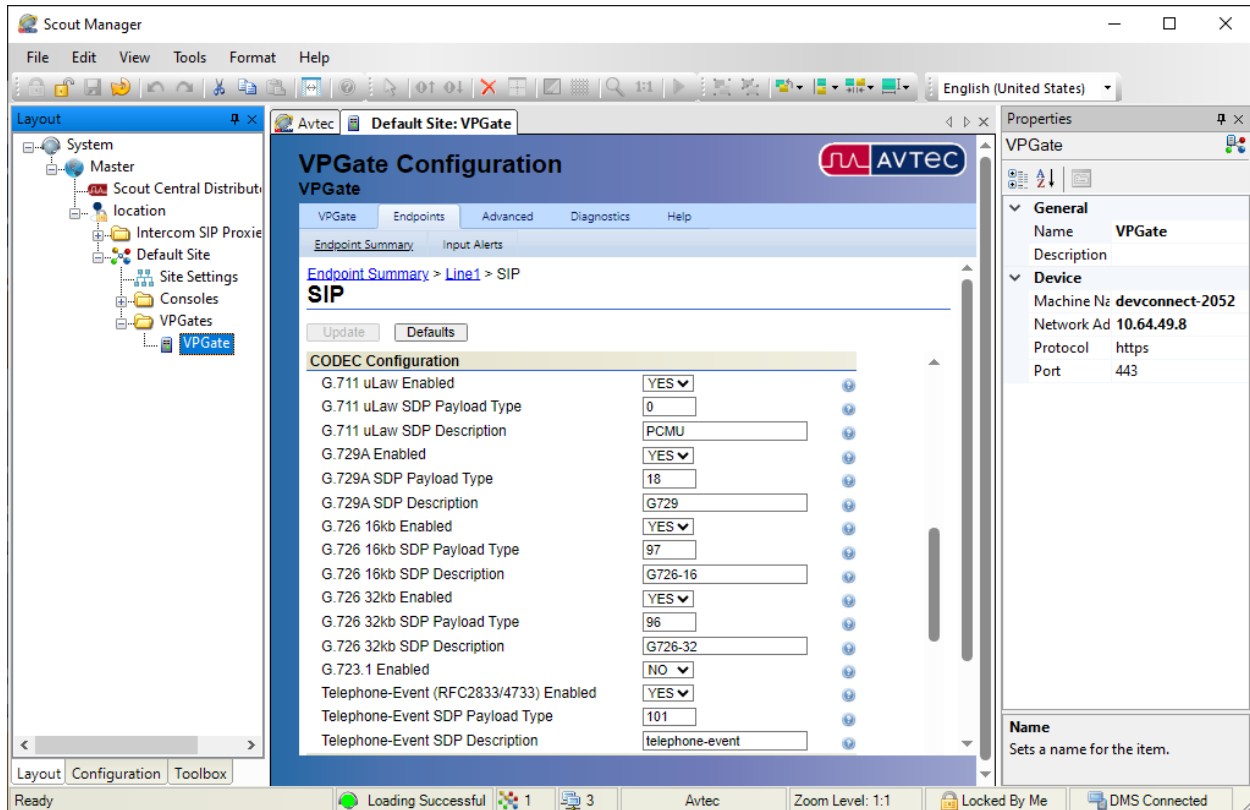


The **SIP** page is displayed as shown below. Under **Primary Identity**, configure the following fields:

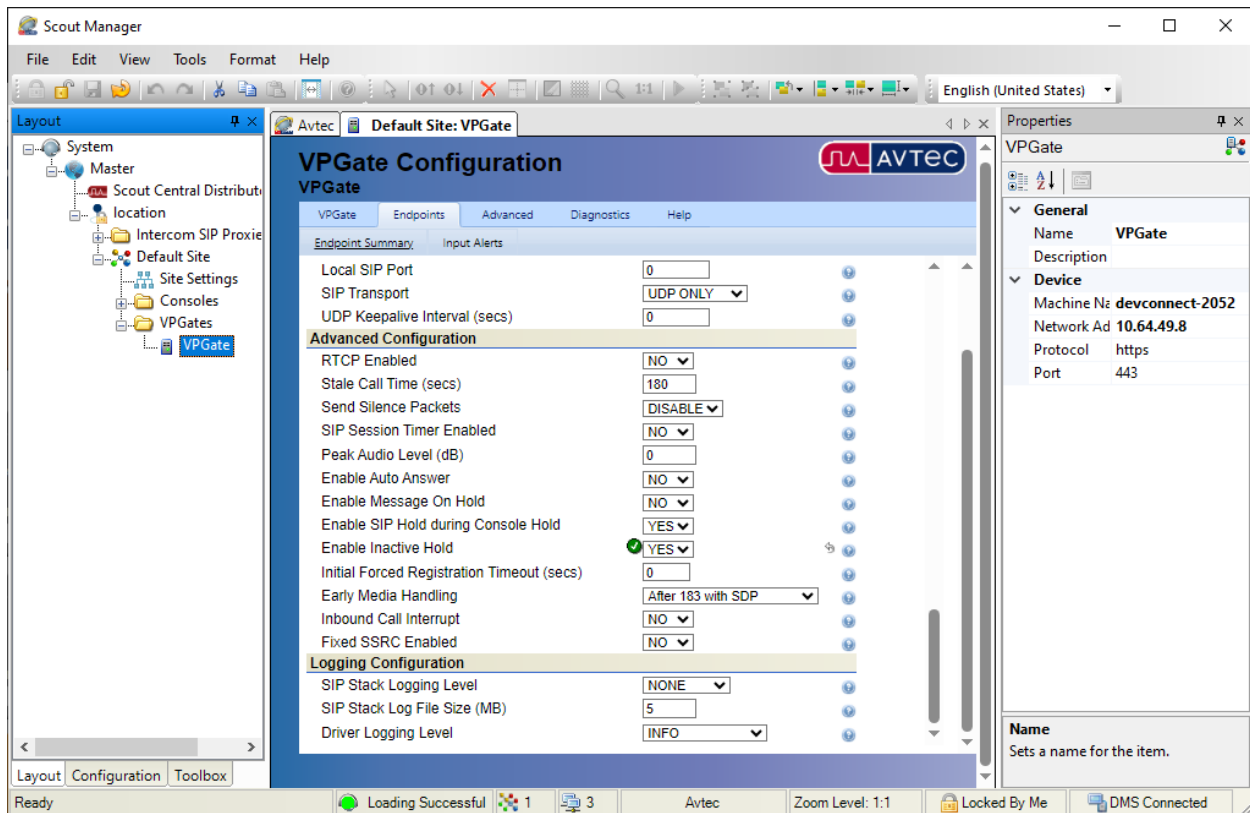
- **Display Name:** Specify a descriptive name (e.g., *Line1*).
- **Username:** Specify a descriptive name (e.g., *70131*).
- **SIP Server Address/Domain name:** Specify the IP address of the Windows system running the SIP Proxy component, which is also the local domain.
- **SIP Server Port:** Specify port *5060*.
- **Authentication Username:** Specify the SIP extension (e.g., *70131*).
- **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.

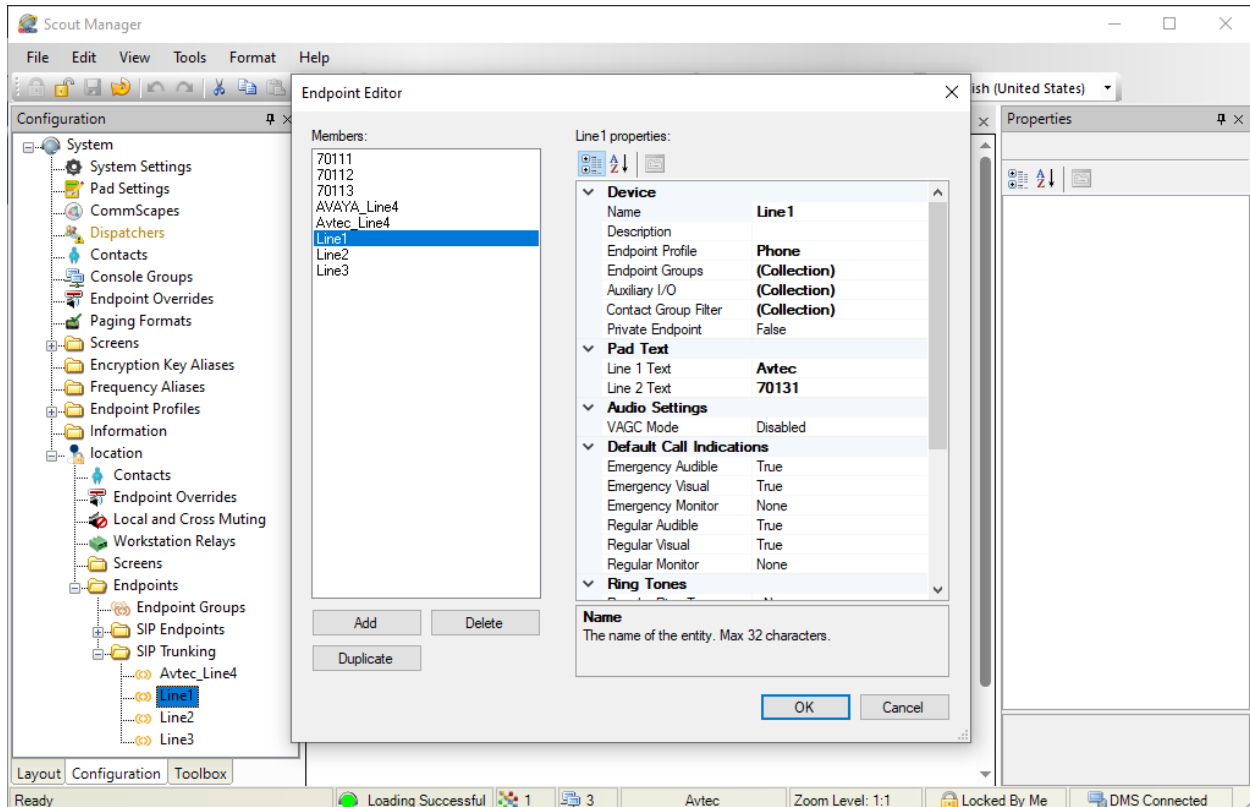


Lastly, scroll down to the **Advanced Configuration** section and **Enable Inactive Hold**. This is required for attended/supervised transfers to work properly. Click the **Add** button.



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 and expand the left view to expose **SIP Trunking** where the added endpoint displays. Double-click on the SIP endpoint added above. In the **Endpoint Editor**, change **Name**, **Line 1 Text**, and **Line 2 Text** to relevant values (e.g., *Line1*, *Avtec*, and *70131* respectively). 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.



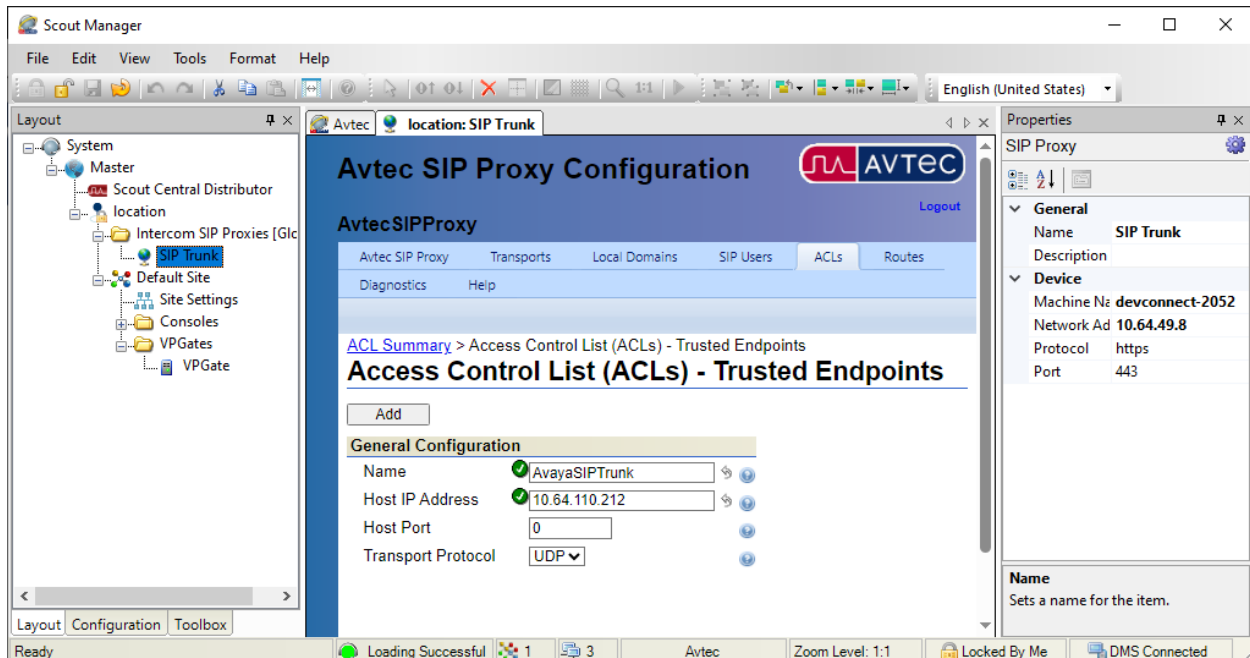
7.5 Add Access Control List (ACLs) – Trusted Endpoint

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., *10.64.110.212*).
- **Host Port:** Enter '0' to trust traffic coming from any port.
- **Transport Protocol:** Specify the transport protocol (i.e., *UDP*).

Click the **Add** button.

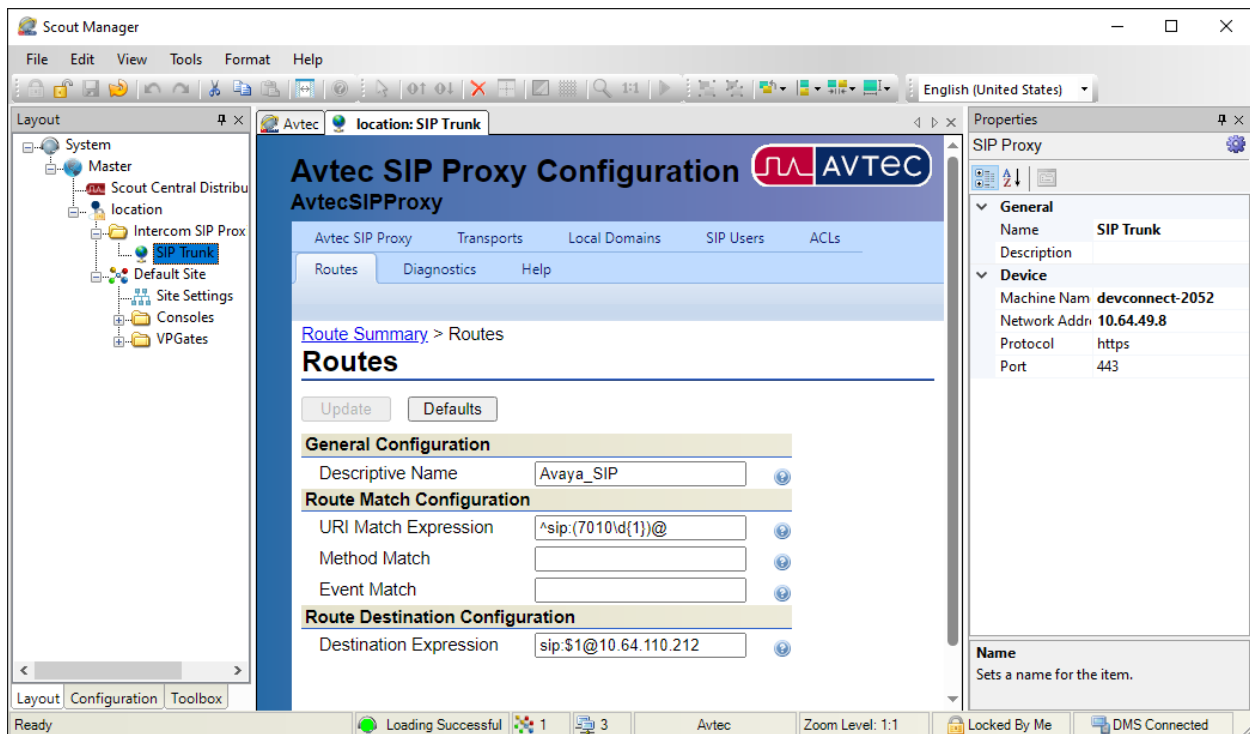


7.6 Add Routes

Configure routes for calls to Session Manager destined for local SIP and H.323 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 SIP 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 '7010', such as *^sip:(7010\d{1})@*
- **Destination Expression:** Enter the destination URI that routes calls to Session Manager. In this case, the expression used was *sip:\$1@10.64.110.212*.

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 window within the Scout Manager application. The main window is titled 'Avtec SIP Proxy Configuration' and features a sidebar with a tree view showing the system hierarchy: System > Master > Scout Central Distributor > location > Intercom SIP Proxy > SIP Trunk. The main content area is divided into tabs: 'Routes' (selected), 'Diagnostics', and 'Help'. The 'Routes' tab shows a 'Route Summary' table with the following data:

Name	Match URI	Method	Event
Avaya_H323	^sip:(700\d{2})@		
Avaya_SIP	^sip:(7010\d{1})@		
Messaging	^sip:(59992)@		
PSTN	^sip:(91\d{10})@		

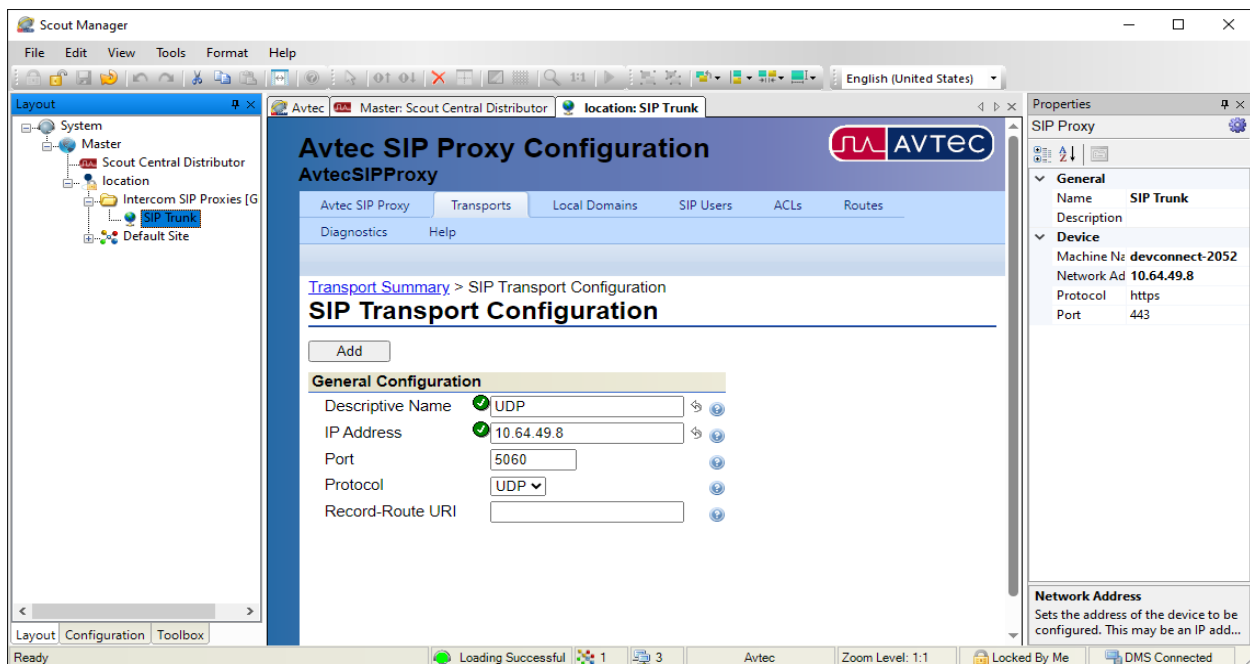
Below the table is an 'Add' button. On the right side, the 'Properties' panel is open, showing the configuration for the 'SIP Trunk'. The 'General' section includes the Name 'SIP Trunk' and a Description. The 'Device' section includes the Machine Name 'devconnect-2052', Network Address '10.64.49.8', Protocol 'https', and Port '443'. At the bottom of the Properties panel, there is a 'Name' field with the instruction 'Sets a name for the item.'

The status bar at the bottom of the window shows 'Ready', 'Loading Successful', '1' (with a small icon), '3' (with a small icon), 'Avtec', 'Zoom Level: 1:1', 'Locked By Me', and 'DMS Connected'.

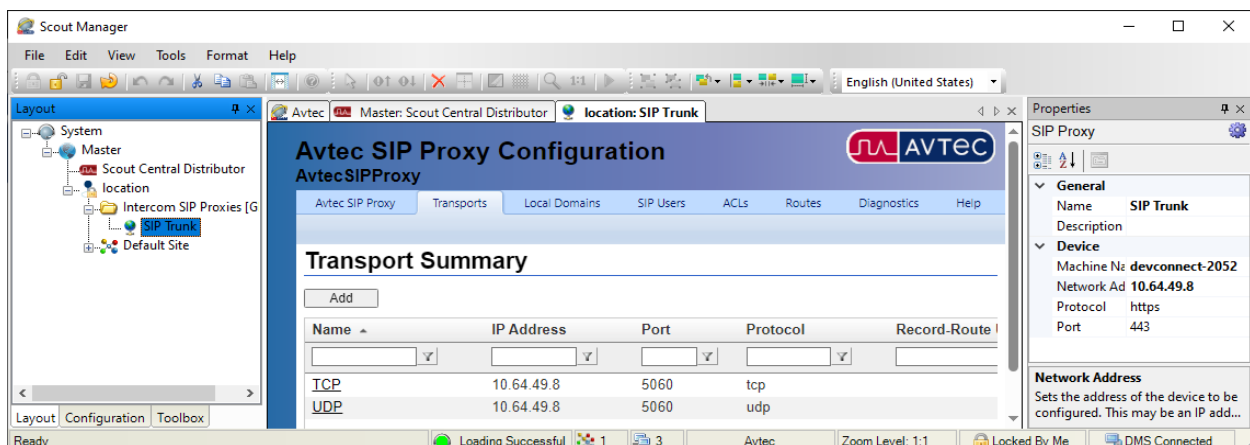
7.7 Add SIP Transports

Configure the TCP and UDP listen ports for the Avtec SIP proxy component. Navigate to **SIP Trunk** → **Transports** and click the **Add** button (not shown). The **SIP Transport Configuration** page is displayed. Input the following values:

- **Descriptive Name:** Provide a descriptive route name (e.g., *UDP*).
- **IP Address:** Enter the IP address of the Windows system running the SIP Proxy component.
- **Port:** Enter the port used to listen for UDP traffic
- **Protocol:** Select the protocol, (e.g., *UDP*)



Click **Add**. Add a similar transport for the TCP protocol. The **Transport Summary** will display the added transports:



7.8 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 left sidebar shows a tree view with "System" expanded, containing "Master", "Scout Central Distributor" (highlighted), "location", "Intercom SIP Proxies [G]", and "Default Site". The main content area is titled "Master: Scout Central Distributor" and features a navigation bar with "Dash", "System", "Alarms", "Reports", "Licensing", and "Admin". The "System" tab is active, showing a "DEPLOYMENTS" section. This section includes a "Locations" area with a circular progress indicator labeled "location" and "V21". Below this is a "Deployments" table with columns "Version", "User", and "Date/Time". The table contains two rows: Version 21, User Avtec, Date/Time 11/29/23 5:52:14 pm; and Version 20, User Avtec, Date/Time 11/29/23 5:39:30 pm. To the right of the table is an "Activity & Errors" section with a table with columns "Component", "Location", "User", "Version", and "Status". The right sidebar contains "Distributor Settings" with a "Device" section showing "Network Ad 10.64.49.8", "Protocol https", and "Port 443". Below this is a "Network Address" section with a text input field. The status bar at the bottom shows "Ready", "Loading Successful", "1", "3", "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**.

The screenshot shows a dialog box titled "Select Deployment Locations". Inside the dialog, there is a header bar with the title. Below the header, there is a instruction: "Select deployment location(s) and then click **Deploy**." To the left of this instruction is a checkbox labeled "Select All" which is checked. To the right is a link "Missing Locations?". Below this is a section titled "Location Name" with a text input field. Underneath the input field is a list box containing one item, "location", which is also checked. At the bottom right of the dialog are two buttons: "Cancel" and "Deploy".

The **Scout VoIP Console** below displays the SIP line configured above with *Avtec 70131*, *Avtec 70132*, and *Avtec 70133* as the button labels.



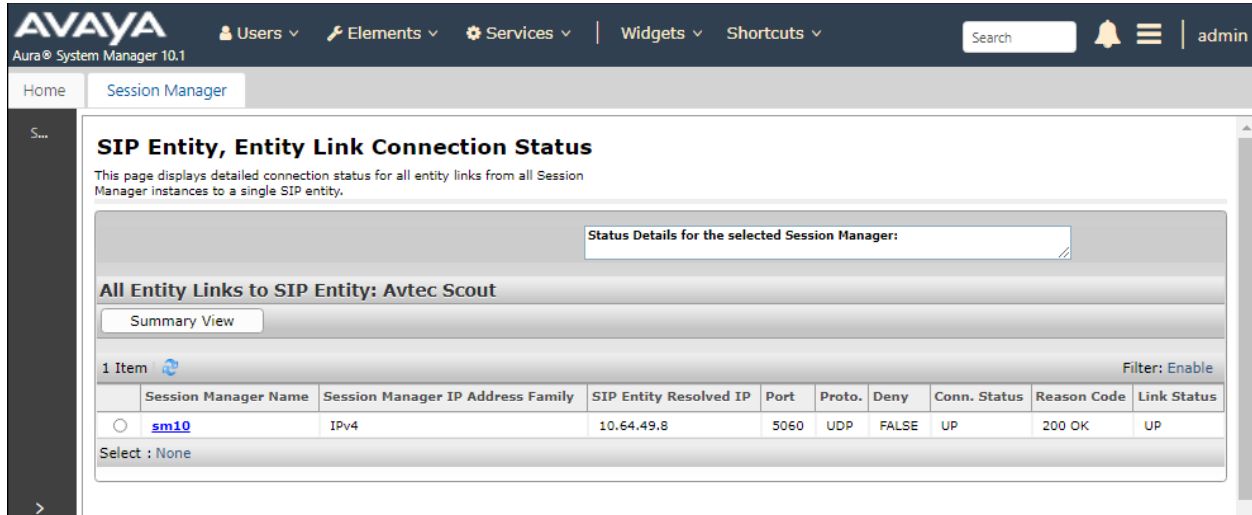
8 Verification Steps

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., *Avtec 70131*, *Avtec 70132*, and *Avtec 70133*) shown below indicate that the SIP trunk is in-service.



2. 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

S...


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"/>	sm10	IPv4	10.64.49.8	5060	UDP	FALSE	UP	200 OK	UP

Select : None

3. Verify that the SIP trunk between Communication Manager and Session Manager is in-service using the **status trunk** command on Communication Manager.
4. Place an incoming call to Scout VoIP Console and answer the call. Verify two-way audio is provided.
5. 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 6, June 2023, available at <http://support.avaya.com>.
- [2] *Administering Avaya Aura® System Manager*, Release 10.1.x, Issue 12, September 2023, available at <http://support.avaya.com>.
- [3] *Administering Avaya Aura® Session Manager*, Release 10.1.x, Issue 6, May 2023, available at <http://support.avaya.com>.
- [4] *Administering Avaya Session Border Controller*, Release 10.1.x, Issue 5, October 2023, available at <http://support.avaya.com>.

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