



Avaya Solution & Interoperability Test Lab

Application Notes for Avtec Scout VoIP Console with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Endpoints – 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 supports inbound and outbound calls, hold, resume, mute, and transfer, and integrates with Avaya Aura® Session Manager via SIP endpoints.

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 supports inbound and outbound calls, hold, resume, mute, and transfer, and integrates with Avaya Aura® Session Manager via SIP endpoints.

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. Additional telephony features, such as call forward, call coverage, call park/unpark, and call pickup were also verified using Communication Manager Feature Access Codes (FACs).

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 subsection 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 this DevConnect Application Note 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 this Application Note, 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:

- SIP registration of Scout VoIP Console with Session Manager.
- Calls between Scout VoIP Console and Avaya SIP and H.323 telephones with Direct IP Media (Shuffling) enabled and disabled.
- Calls between the Scout VoIP Console and the PSTN.
- G.711 and G.729 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, multiple calls, and blind/attended transfers.
- Extended telephony features using Communication Manager FACs for Call Forward, Call Park/Unpark, and Call Pickup.
- Voicemail coverage, MWI support, and logging into voicemail system to retrieve voice messages.
- Use of programmable buttons on the Scout VoIP Console.
- Proper system recovery after a restart of the Scout VoIP Console and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observation(s):

- Incoming call notification is 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.

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:

- Avaya Aura® Communication Manager running in a virtual environment with an Avaya G450 Media Gateway.
- Media resources in the Avaya G450 Media Gateway and Avaya Aura® Media Server.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya Aura® Messaging serving as the voicemail system.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya Aura® Messaging serving as the voicemail system.
- Avaya 96x1 Series H.323 and SIP Deskphones.
- Avaya J100 Series SIP Deskphones.
- Avtec Scout VoIP Console registered to Session Manager via SIP endpoints.
- Avtec Scout VoIP Console was installed on a desktop PC running Microsoft Windows 10 and included the following components: Scout VPGate and Scout Manager. Scout Manager was used to configure the Scout VoIP Console.

Avtec Scout VoIP Console registered with Session Manager as SIP endpoints and configured as Off-PBX Stations (OPS) on Communication Manager.

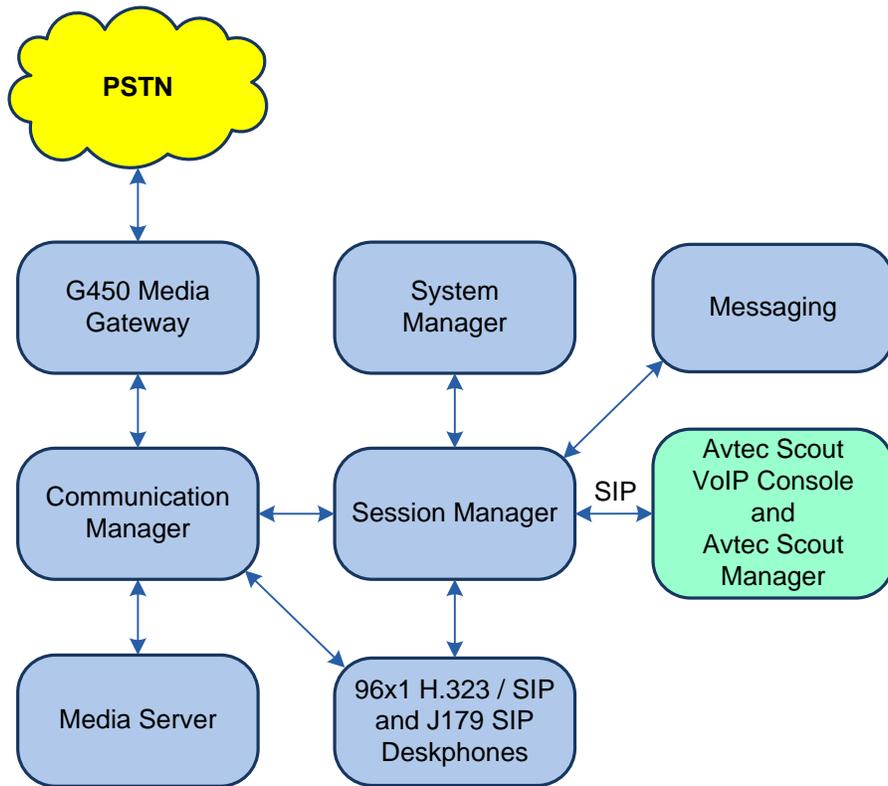


Figure 1: Avaya SIP Network with Avtec Scout VoIP Console

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	8.1.1.0.0-FP1
Avaya G450 Media Gateway	FW 40.25.0
Avaya Aura® Media Server	v.8.0.1.121
Avaya Aura® System Manager	8.1.0.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.0.0.079814
Avaya Aura® Session Manager	8.1.0.0.810007
Avaya Aura® Messaging	7.1.3.1.0-FP3SP1
Avaya 96x1 Series IP Deskphones	6.8304 (H.323) 7.1.7.0.11 (SIP)
Avaya J179 SIP Deskphones	4.0.3.1.4
Avtec Scout VoIP Console running on Microsoft Windows 10, including the following components: <ul style="list-style-type: none">▪ Scout Console▪ Scout VPGate	4.10.0.57 4.10.0.267
Avtec Scout Manager	4.10.0.57

5. Configure Avaya Aura® Communication Manager

This section provides the procedure for configuring Communication Manager. The procedure includes the following areas:

- Verify Communication Manager license
- Administer IP Node Names
- Administer IP Network Region and IP Codec Set
- Administer SIP Trunk Group to Session Manager
- Administer AAR Call Routing

Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

Note: It is assumed that basic configuration of the Communication Manager has already been completed, such as the SIP trunk to Session Manager. The SIP station configuration for Scout VoIP Console is configured through Avaya Aura® System Manager in **Section 6.3**.

5.1. Verify License

Using the SAT, verify that the Off-PBX Telephones (OPS) option is enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

```
display system-parameters customer-options                               Page 1 of 12
                                OPTIONAL FEATURES

G3 Version: V18                                                         Software Package: Enterprise
Location: 2                                                             System ID (SID): 1
Platform: 28                                                            Module ID (MID): 1

                                USED
Platform Maximum Ports: 48000    82
Maximum Stations: 36000          21
Maximum XMOBILE Stations: 36000  0
Maximum Off-PBX Telephones - EC500: 41000  0
Maximum Off-PBX Telephones - OPS: 41000  10
Maximum Off-PBX Telephones - PBFMC: 41000  0
Maximum Off-PBX Telephones - PVFMC: 41000  0
Maximum Off-PBX Telephones - SCCAN: 0      0
Maximum Survivable Processors: 313      0

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

5.2. 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
```

5.3. Administer IP Network Region and IP Codec Set

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 IP endpoints without using media resources in the Avaya G450 Media Gateway or Avaya Aura® 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.

```
change ip-network-region 1                             Page 1 of 20
                                                    IP NETWORK REGION
      Region: 1        NR Group: 1
Location: 1          Authoritative Domain: avaya.com
      Name:                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: 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
```

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 that IP codec set '1' was specified in IP Network Region '1' shown above. Scout VoIP Console didn't use SRTP. The Scout VoIP Console was tested using G.711 and G.729 codecs.

```
change ip-codec-set 2 Page 1 of 2
```

IP MEDIA PARAMETERS

Codec Set: 2

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: none

3:

4:

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:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Set the **Enforce SIPS URI for SRTP** field to *n*.
- Specify Communication Manager (*procr*) and the 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.
- Ensure that the TLS port value of *5061* is configured 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*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to 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? 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: devcon-sm
Near-end Listen Port: 5061                               Far-end Listen Port: 5061
                                                         Far-end Network Region: 1
Far-end Domain: avaya.com
                                                         Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                    RFC 3389 Comfort Noise? n
  DTMF over IP: rtp-payload                             Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
  Enable Layer 3 Test? y                                Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n                 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, Avaya SIP deskphones, and Avaya Aura® Messaging. 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. Configure the other fields in bold and accept the default values for the remaining fields.

```

add trunk-group 10                                     Page 1 of 22
                                     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

```

5.5. AAR Call Routing

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and enter add an entry that routes digits beginning with “78” to route pattern 10 as shown below.

```

change aar analysis 78                               Page 1 of 2
                                     AAR DIGIT ANALYSIS TABLE
                                     Location: all                                     Percent Full: 1

      Dialed      Total      Route      Call      Node      ANI
      String      Min Max      Pattern      Type      Num      Reqd
78              5 5       10         lev0     n

```

Configure a preference in **Route Pattern** 10 to route calls over SIP trunk group 10 as shown below.

```

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
2:
3:
4:
5:
6:
                                     DCS/ IXC
                                     n      user
                                     n      user
                                     n      user
                                     n      user
                                     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

```

6. Configure Avaya Aura® Session Manager

This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Set Network Transport Protocol for Scout VoIP Console
- Administer SIP User

Note: It is assumed that basic configuration of Session Manager has already been performed. This section will focus on the configuration of a SIP user for the Scout VoIP Console.

6.1. Launch System Manager

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: Internet Explorer 11.x or Firefox 65.0, 66.0 and 67.0.

6.2. Set Network Transport Protocol for Scout VoIP Console

From the System Manager **Home** screen, select **Elements** → **Routing** → **SIP Entities** and edit the SIP Entity for Session Manager shown below.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The left sidebar is expanded to 'Routing' > 'SIP Entities'. The main content area displays 'SIP Entity Details' for 'devcon-sm'. The 'General' section includes fields for Name (devcon-sm), IP Address (10.64.102.117), SIP FQDN, Type (Session Manager), Notes, Location (Thornton), Outbound Proxy, Time Zone (America/New_York), Minimum TLS Version (Use Global Setting), and Credential name. The 'Monitoring' section includes SIP Link Monitoring and CRLF Keep Alive Monitoring, both set to 'Use Session Manager Configuration'. Buttons for 'Commit' and 'Cancel' are visible.

Scroll down to the **Listen Ports** section and verify that the transport network protocol used by Scout VoIP Console is specified in the list below. For the compliance test, the solution used UDP network transport.

Listen Ports

Add Remove

3 Items Filter: Enable

<input type="checkbox"/>	Listen Ports	Protocol	Default Domain	Endpoint	Notes
<input type="checkbox"/>	5060	TCP	avaya.com	<input type="checkbox"/>	
<input type="checkbox"/>	5060	UDP	avaya.com	<input type="checkbox"/>	
<input type="checkbox"/>	5061	TLS	avaya.com	<input type="checkbox"/>	

Select : All, None

6.3. Administer SIP User

In the subsequent screen (not shown), select **Users** → **User Management** → **Manage Users** to display the **User Management** screen below. Click **New** to add a user.

	First Name	Surname	Display Name	Login Name	SIP Handle
<input type="checkbox"/>	SIP	78000	78000, SIP	78000@avaya.com	78000
<input type="checkbox"/>	SIP	78001	78001, SIP	78001@avaya.com	78001
<input type="checkbox"/>	SIP	78002	78002, SIP	78002@avaya.com	78002

6.3.1. Identity

The **New User Profile** screen is displayed. Enter the desired **Last Name** and **First Name**. For **Login Name**, enter “<ext>@<domain>”, where “<ext>” is the desired Scout VoIP Console SIP extension and “<domain>” is the applicable SIP domain name from **Section 5.3**. Retain the default values in the remaining fields.

User Profile | Add

Commit & Continue | Commit | Cancel

Identity | Communication Profile | Membership | Contacts

Basic Info

Address

LocalizedName

User Provisioning Rule: [v]

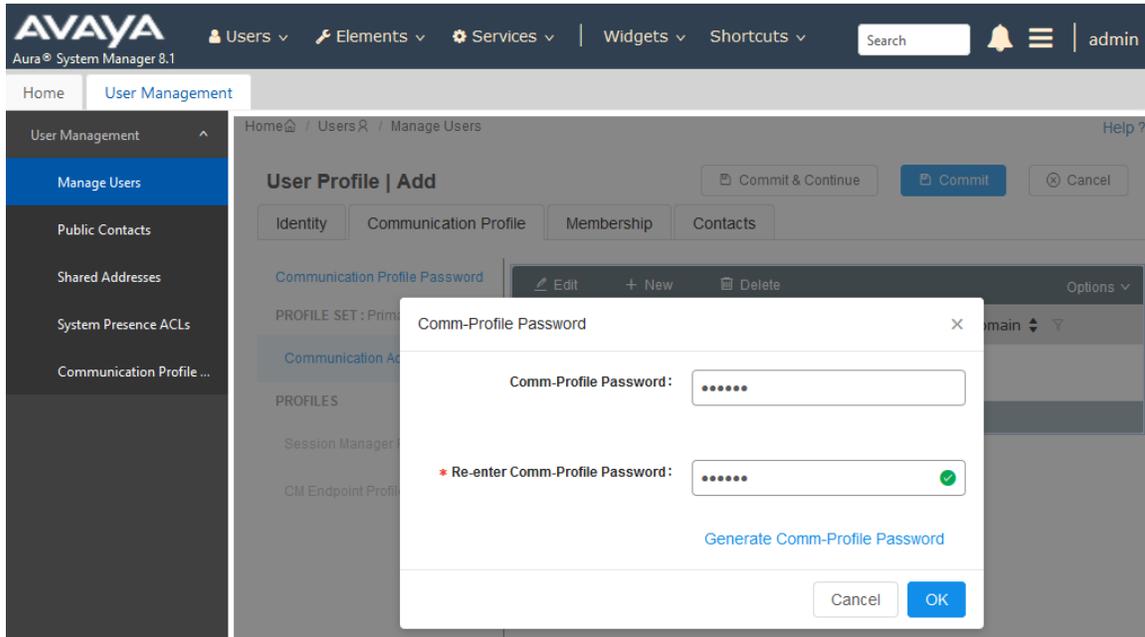
* Last Name: 78020 | Last Name (Latin Translation): 78020

* First Name: Avtec | First Name (Latin Translation): Avtec

* Login Name: 78020@avaya.cor | Middle Name: Middle Name Of U

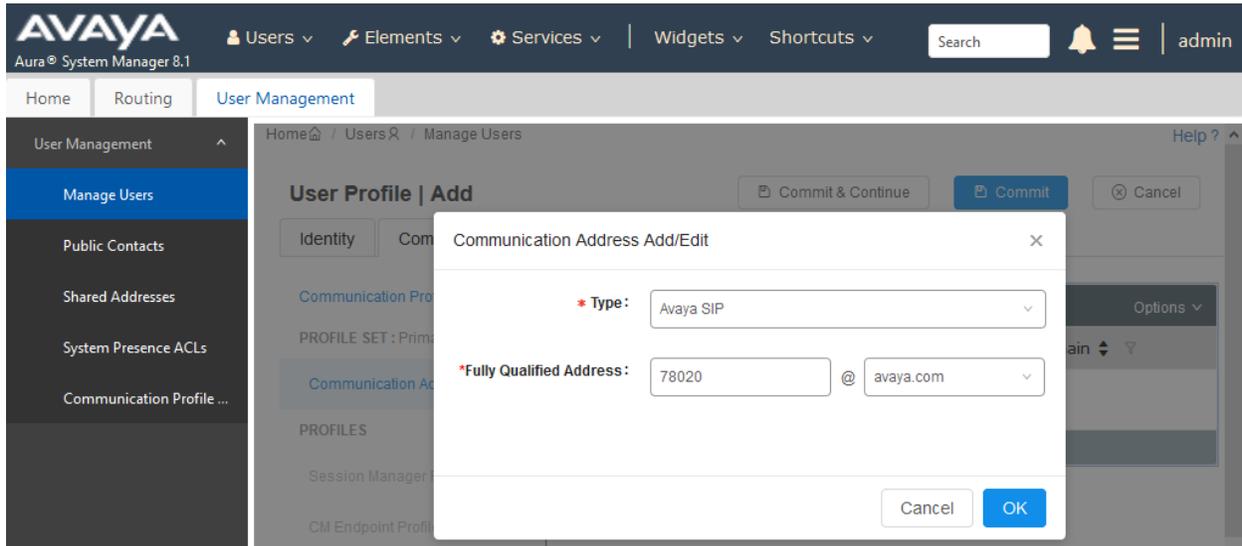
6.3.2. Communication Profile

Select the **Communication Profile** tab. Next, click on **Communication Profile Password**. For **Comm-Profile Password** and **Re-enter Comm-Profile Password**, enter the desired password for the SIP user to use for registration. Click **OK**.



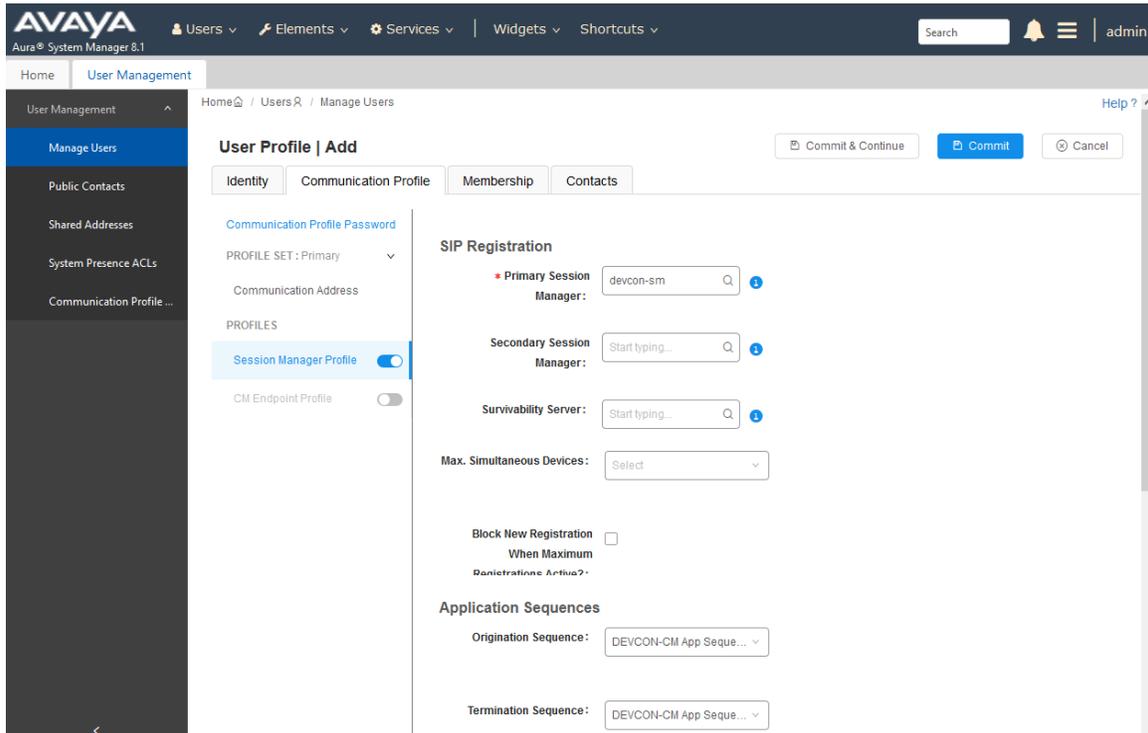
6.3.3. Communication Address

Click on **Communication Address** and then click **New** to add a new entry. The **Communication Address Add/Edit** dialog box is displayed as shown below. For **Type**, retain *Avaya SIP*. For **Fully Qualified Address**, enter and select the SIP user extension and domain name to match the login name from **Section 6.3.1**. Click **OK**.

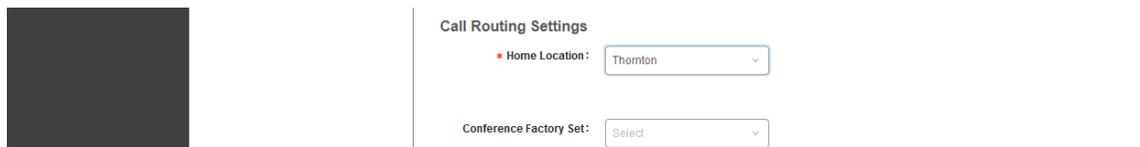


6.3.4. Session Manager Profile

Click on toggle button by **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.



Scroll down to the **Call Routing Settings** section to configure the **Home Location**.



6.3.5. CM Endpoint Profile

Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.3**. For **Template**, select *9641SIP_DEFAULT_CM_8_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click on the Endpoint Editor (i.e, Edit icon in Extension field) to set the **Coverage Path** to voicemail.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The main content area is titled 'User Profile | Add' and features several tabs: 'Identity', 'Communication Profile', 'Membership', and 'Contacts'. The 'Communication Profile' tab is active, showing various configuration fields. On the left, a sidebar menu lists 'Manage Users', 'Public Contacts', 'Shared Addresses', 'System Presence ACLs', and 'Communication Profile...'. The 'CM Endpoint Profile' toggle is turned on. The configuration fields include: 'System' (devcon-cm), 'Profile Type' (Endpoint), 'Extension' (78020), 'Set Type' (9641SIP), 'Port' (IP), 'Template' (9641SIP_DEFAULT_CM), 'Security Code' (Enter Security Code), 'Voice Mail Number', 'Calculate Route Pattern' (checked), 'SIP URI' (Select), 'Delete on Unassign from User or on Delete User' (checked), 'Allow H.323 and SIP Endpoint Dual Penetration' (unchecked), 'Enhanced Call-Info Display for 1-line phones' (unchecked), and 'Override Endpoint Name and Localized Name' (checked). Buttons for 'Commit & Continue', 'Commit', and 'Cancel' are visible at the top right.

Navigate to the **General Options** tab and set the **Coverage Path 1** field to the voicemail coverage path. This provides voicemail coverage for the SIP user. In this example, coverage path 10 was used.

* System	<input type="text" value="devcon-cm"/>	* Extension	<input type="text" value="78020"/>
* Template	<input type="text" value="9641SIP_DEFAULT_CM_8_1"/>	* Set Type	<input type="text" value="9641SIP"/>
* Port	<input type="text" value="IP"/>	* Security Code	<input type="text"/>
Name	<input type="text"/>		

Display Extension Ranges

General Options (G) *		Feature Options (F)	Site Data (S)	Abbreviated Call Dialing (A)
Enhanced Call Fwd (E)		Button Assignment (B)	Profile Settings (P)	Group Membership (M)
* Class of Restriction (COR)	<input type="text" value="1"/>	* Class Of Service (COS)	<input type="text" value="1"/>	
* Emergency Location Ext	<input type="text" value="78020"/>	* Message Lamp Ext.	<input type="text" value="78020"/>	
* Tenant Number	<input type="text" value="1"/>			
* SIP Trunk	<input type="text" value="aar"/>	Type of 3PCC Enabled	<input type="text" value="None"/>	
Coverage Path 1	<input type="text" value="10"/>	Coverage Path 2	<input type="text"/>	
Lock Message	<input type="checkbox"/>	Localized Display Name	<input type="text"/>	

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 Endpoints
- Modify SIP Line Label
- Add Voicemail/MWI Button

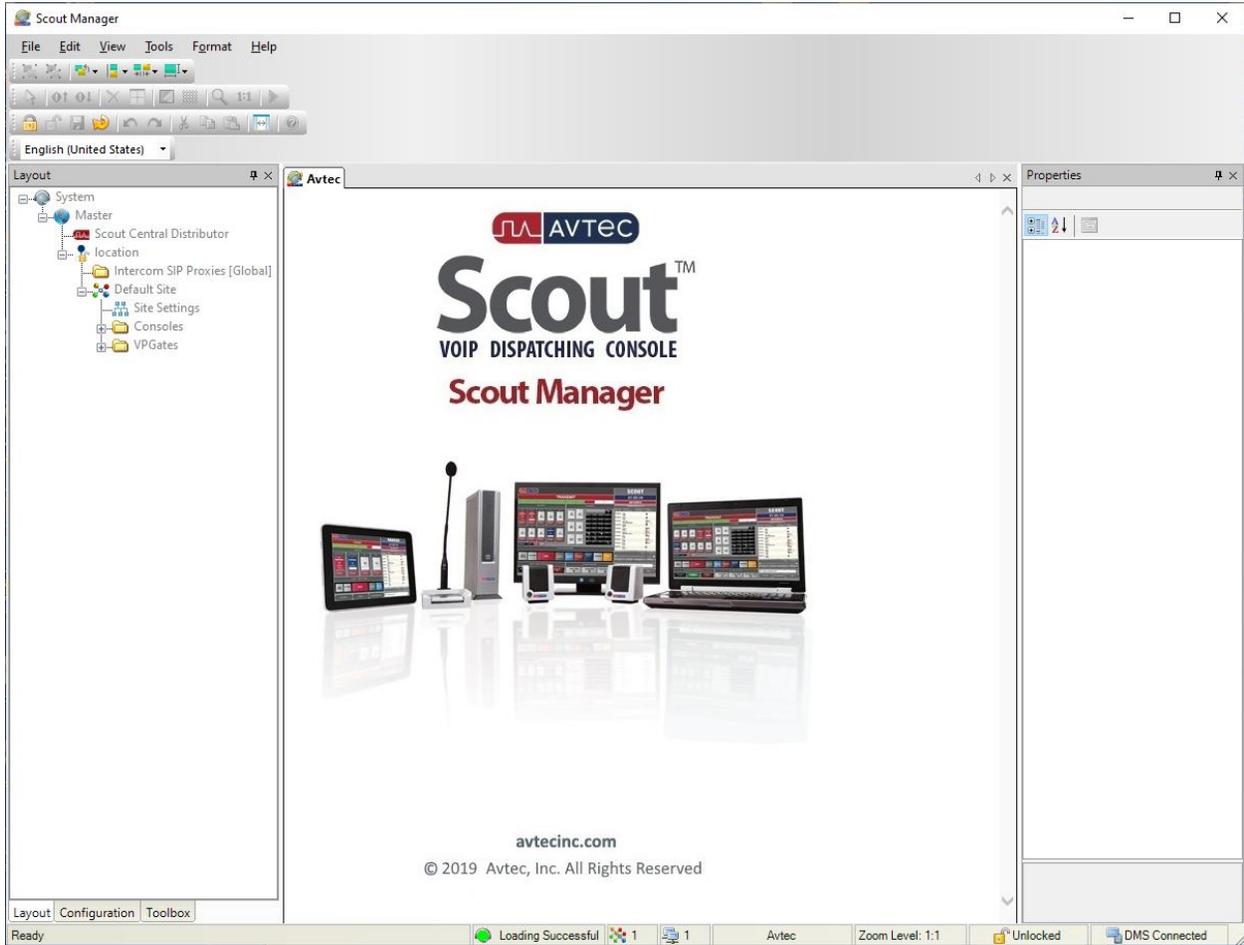
7.1. Launch Scout Manager



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

The image shows a login dialog box for the Scout Manager application. At the top center is the Avtec logo, which consists of a red square with a white waveform icon followed by the word 'AVTEC' in white text on a dark blue rounded rectangle. Below the logo are two input fields: 'Username:' followed by a white text box with a blue border, and 'Password:' followed by a white text box with a grey border. At the bottom of the dialog are two buttons: 'OK' with a blue border and 'Close' with a grey border.

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



7.2. Add Endpoints

Endpoints are created under VPGate configuration. 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., 78020).
- **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 screenshot displays the Avtec Scout Manager VPGate Configuration interface. The main window is titled "VPGate Configuration" and shows the "Endpoint Configuration" page. The interface includes a navigation tree on the left, a menu bar at the top, and a properties panel on the right. The "Endpoint Configuration" page has an "Add" button and several sections:

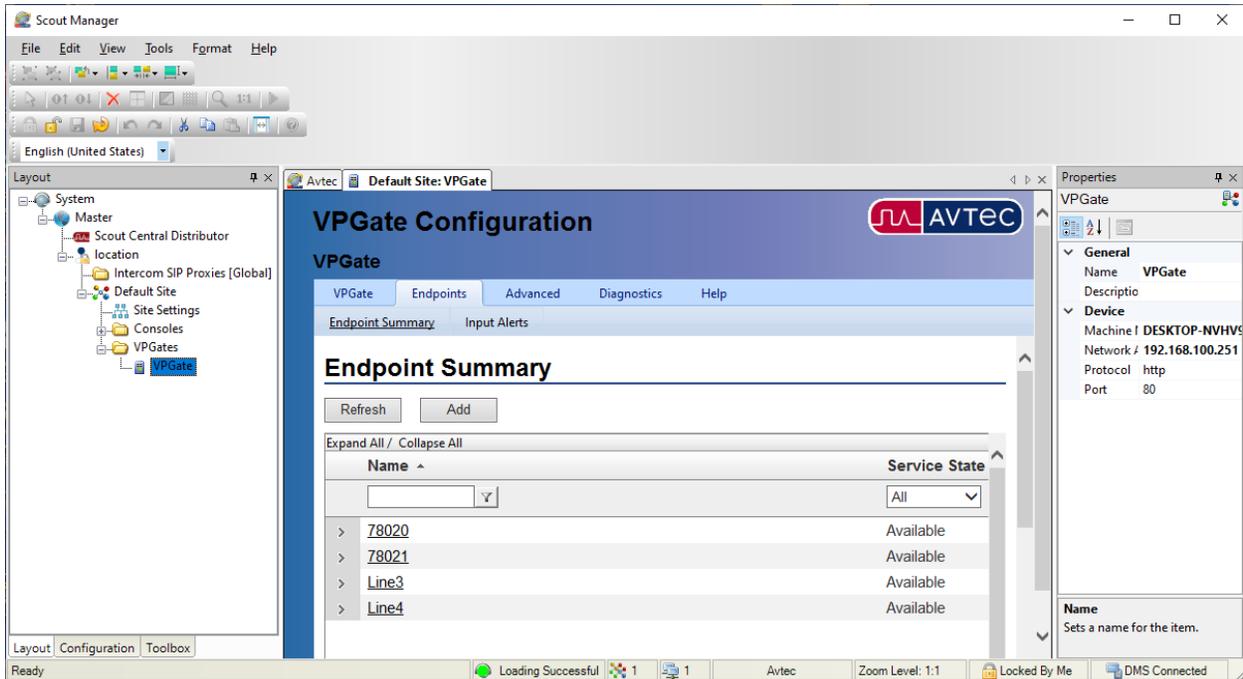
- Endpoint Configuration:**
 - Endpoint Name: 78020
 - Service State: AVAILABLE
- Endpoint Connection:**
 - Endpoint Audio: VoIP
- VoIP Audio Settings:**
 - Receive Audio Mode: FULL DUPLEX
 - Override Receive Audio IP Port: 0
 - Override Transmit Audio IP Port: 0
 - VoIP Audio Jitter Depth (ms): 100
 - Squelch Tail Time Out: 0
 - Allow Barge-in/Monitor Outbound Audio: YES
- Inbound Calls:**
 - Emergency Only: NO
 - Call Clear Time Out: 3600
 - Duplicate Call Supported: NO
- Miscellaneous:**

The properties panel on the right shows the following information:

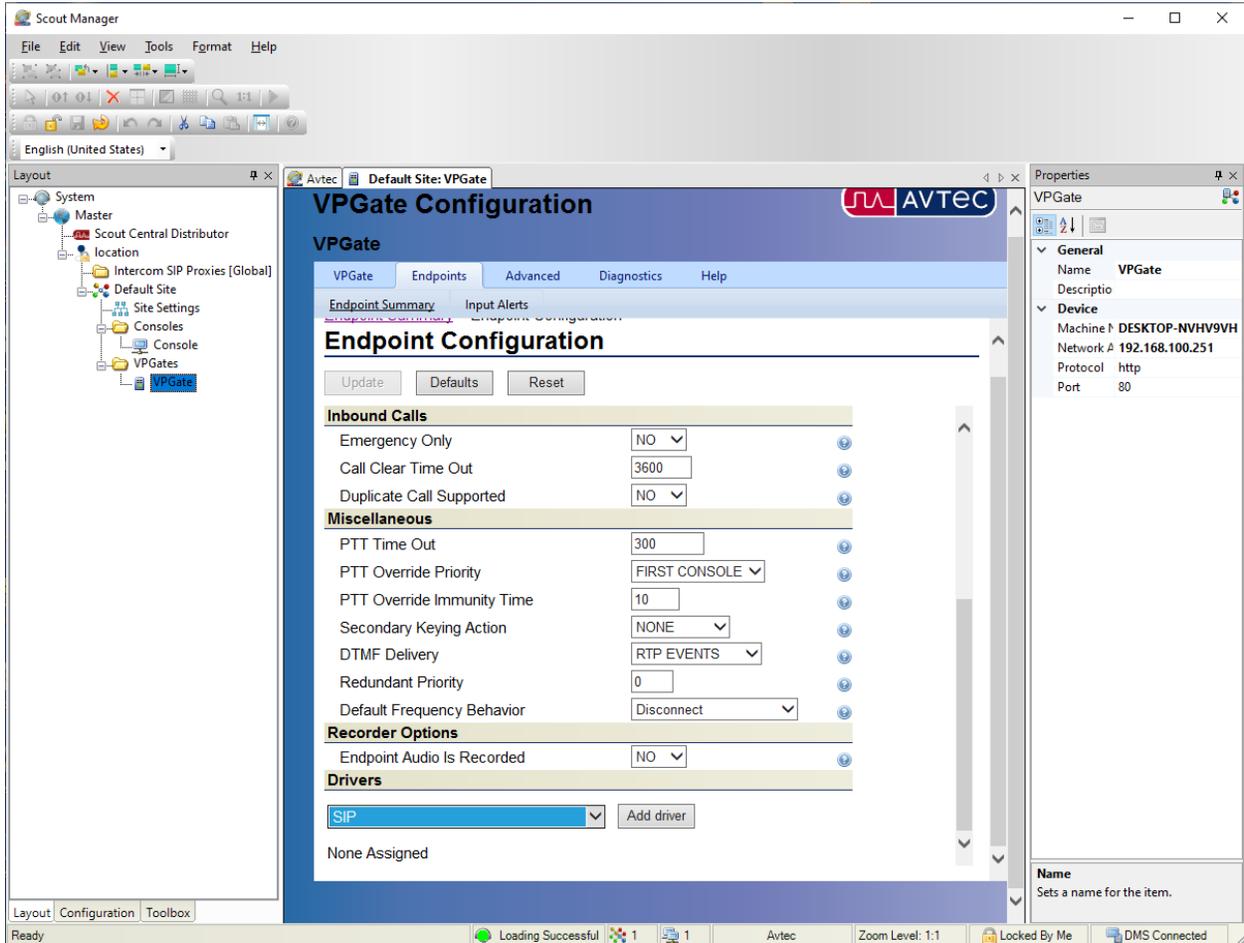
- General:**
 - Name: VPGate
 - Description:
- Device:**
 - Machine: DESKTOP-NVHV5
 - Network: 192.168.100.251
 - Protocol: http
 - Port: 80

The status bar at the bottom shows "Ready", "Loading Successful", "1", "1", "Avtec", "Zoom Level: 1:1", "Locked By Me", and "DMS Connected".

The **Endpoint** previously added is now displayed in the **Endpoint Summary** page shown below. Click on the endpoint that was previously added (i.e., 78020) to open the configuration again.

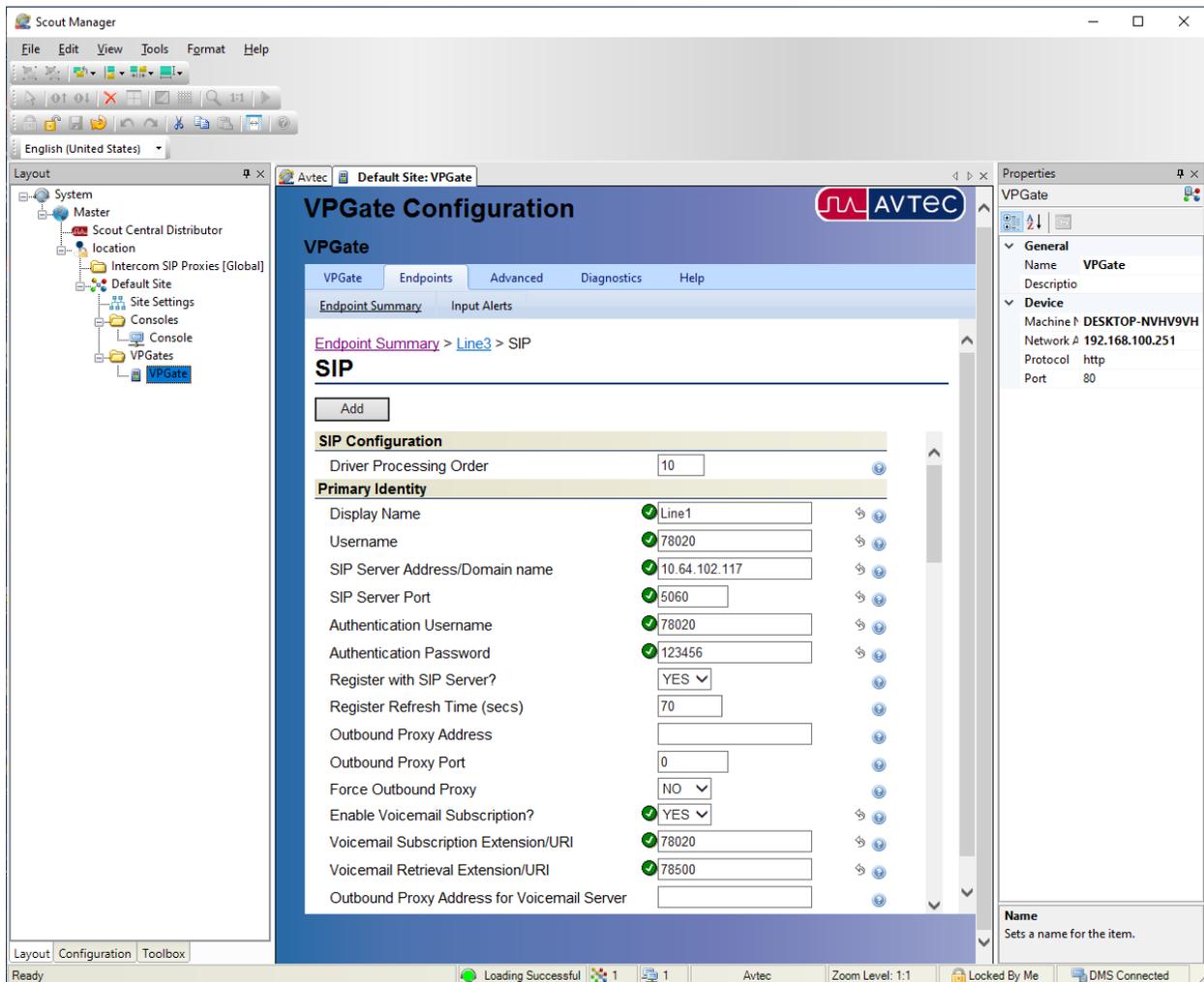


The **Endpoint Configuration** page is displayed. Scroll to the bottom of the page to the **Drivers** section as shown below. Select **SIP** from the drop-down field and click **Add driver**.



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., *78020*).
- **SIP Server Address/Domain name:** Specify the signaling IP address of Session Manager.
- **SIP Server Port:** Specify port *5060*.
- **Authentication Username:** Specify the SIP extension (e.g., *78020*).
- **Authentication Password:** Specify the password used for SIP registration as configured in **Section 6.3.2**.
- **Register with SIP Server:** Enable this option.
- **Enable Voicemail Subscription:** Enable this option.
- **Voicemail Subscription Extension/URL:** Specify the SIP extension (e.g., *78020*).
- **Voicemail Retrieval Extension/URL:** Specify the voicemail number.



Scroll down to the **Codec Configuration** section and specify the codecs to be supported. In this example, G.711, G.729, and G.726 were enabled.

The screenshot displays the Avtec Scout Manager interface for VPGate configuration. The main window is titled "VPGate Configuration" and shows the "CODEC Configuration" section. The "Endpoint Summary" is set to "Line3 > SIP". The "CODEC Configuration" table lists various codecs and their settings:

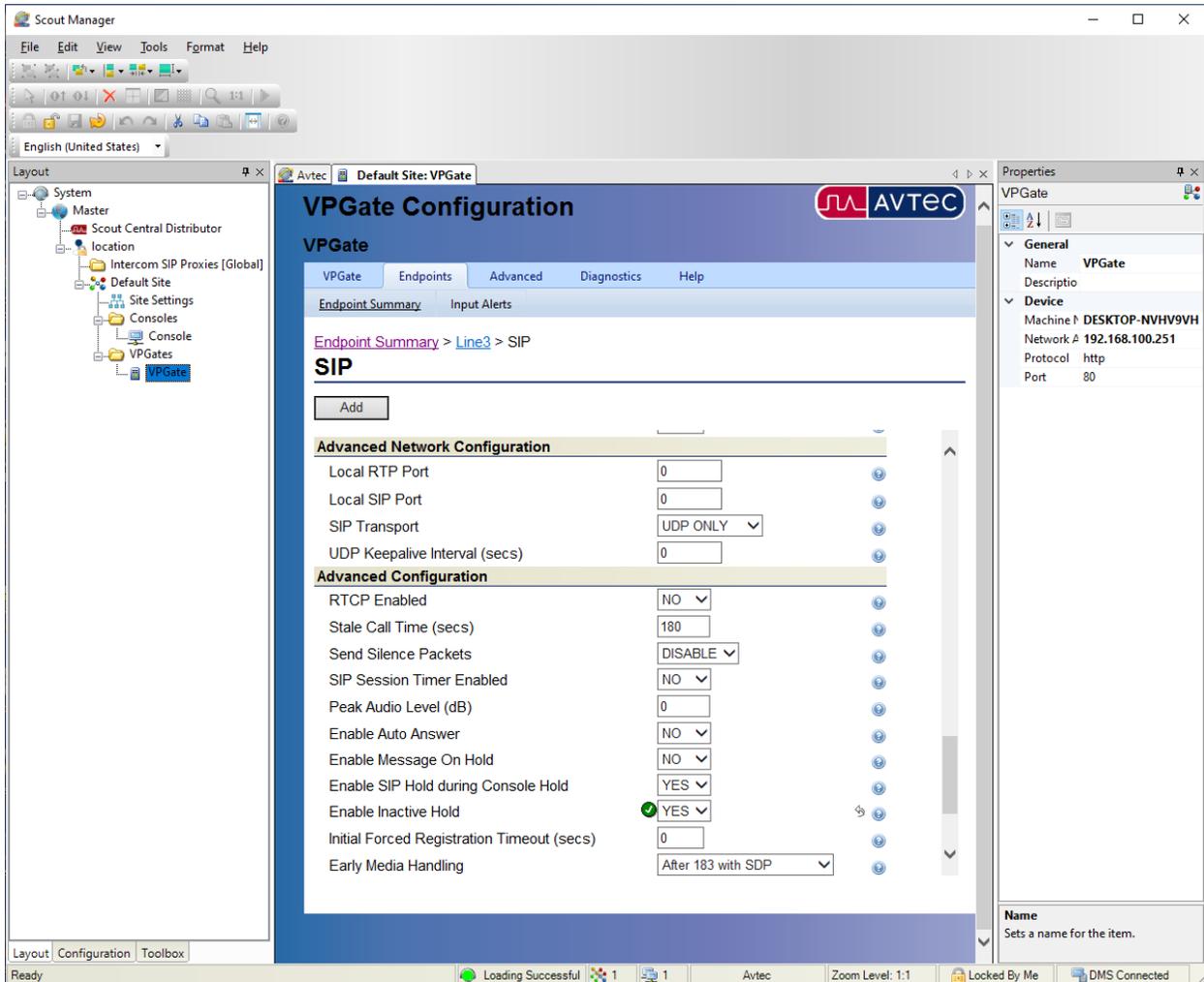
Codec Name	Enabled	SDP Payload Type	SDP Description
G.711 uLaw	YES	0	PCMU
G.729A	YES	18	G729
G.726 16kb	YES	97	G726-16
G.726 32kb	YES	96	G726-32
G.723.1	NO		
Telephone-Event (RFC2833/4733)	YES	101	telephone-event

The right-hand "Properties" pane shows the following details for the VPGate:

- General:** Name: VPGate
- Device:** Machine: DESKTOP-NVHV9VH, Network A: 192.168.100.251, Protocol: http, Port: 80

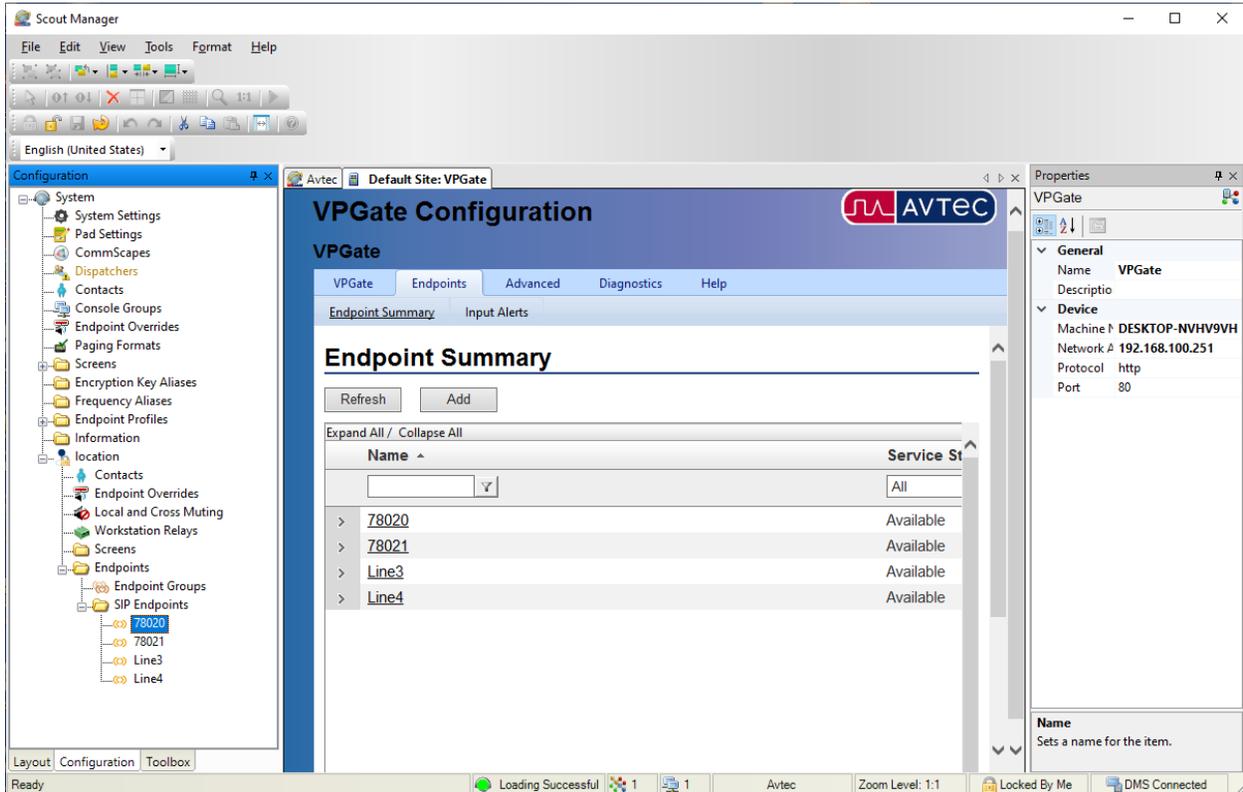
The bottom status bar indicates "Loading Successful" and "DMS Connected".

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.



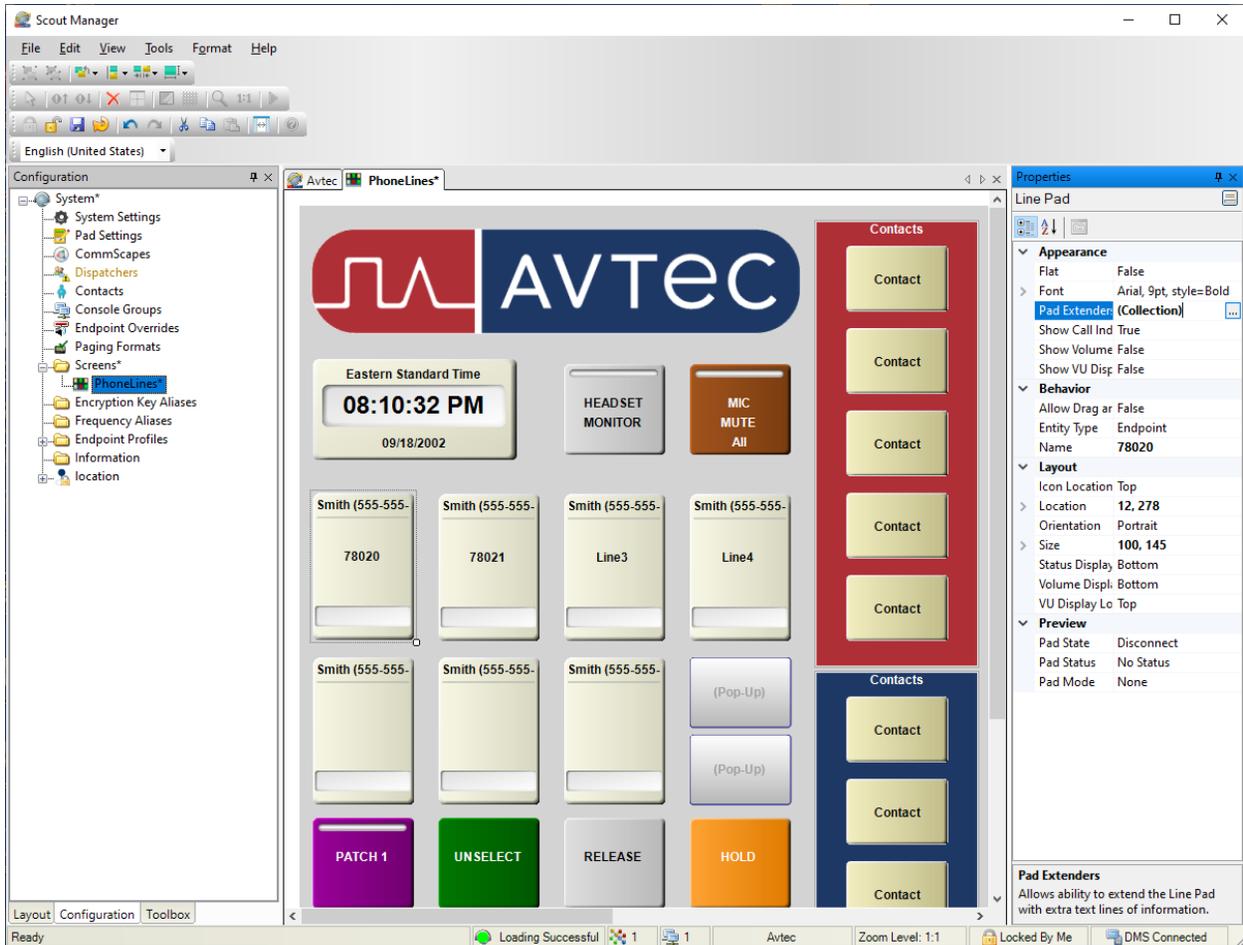
7.3. Modify SIP Line Label

To modify the label of the SIP line button on Scout VoIP Console, select the **Configuration** tab in the left pane and change the label for the SIP endpoint, as desired, by highlighting the current SIP line (e.g., *Line1*) and changing it to the SIP extension (e.g., *78020*) as shown below.

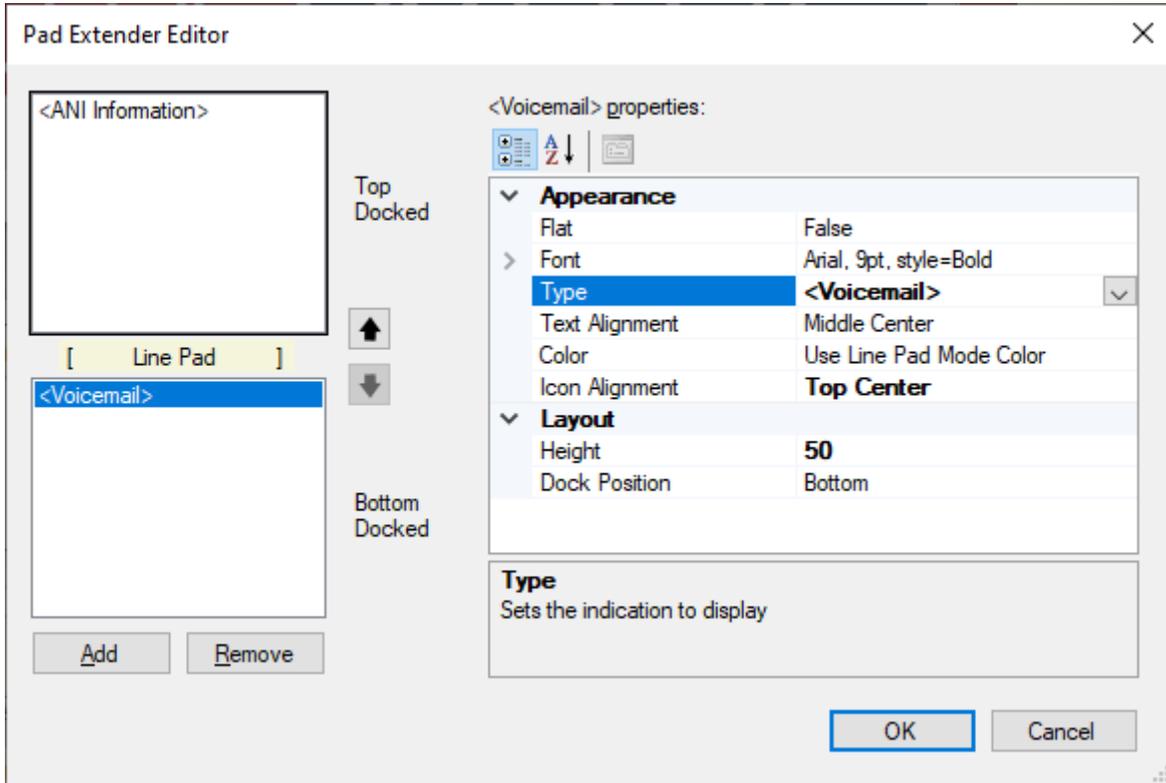


7.4. Add Voicemail/MWI Button

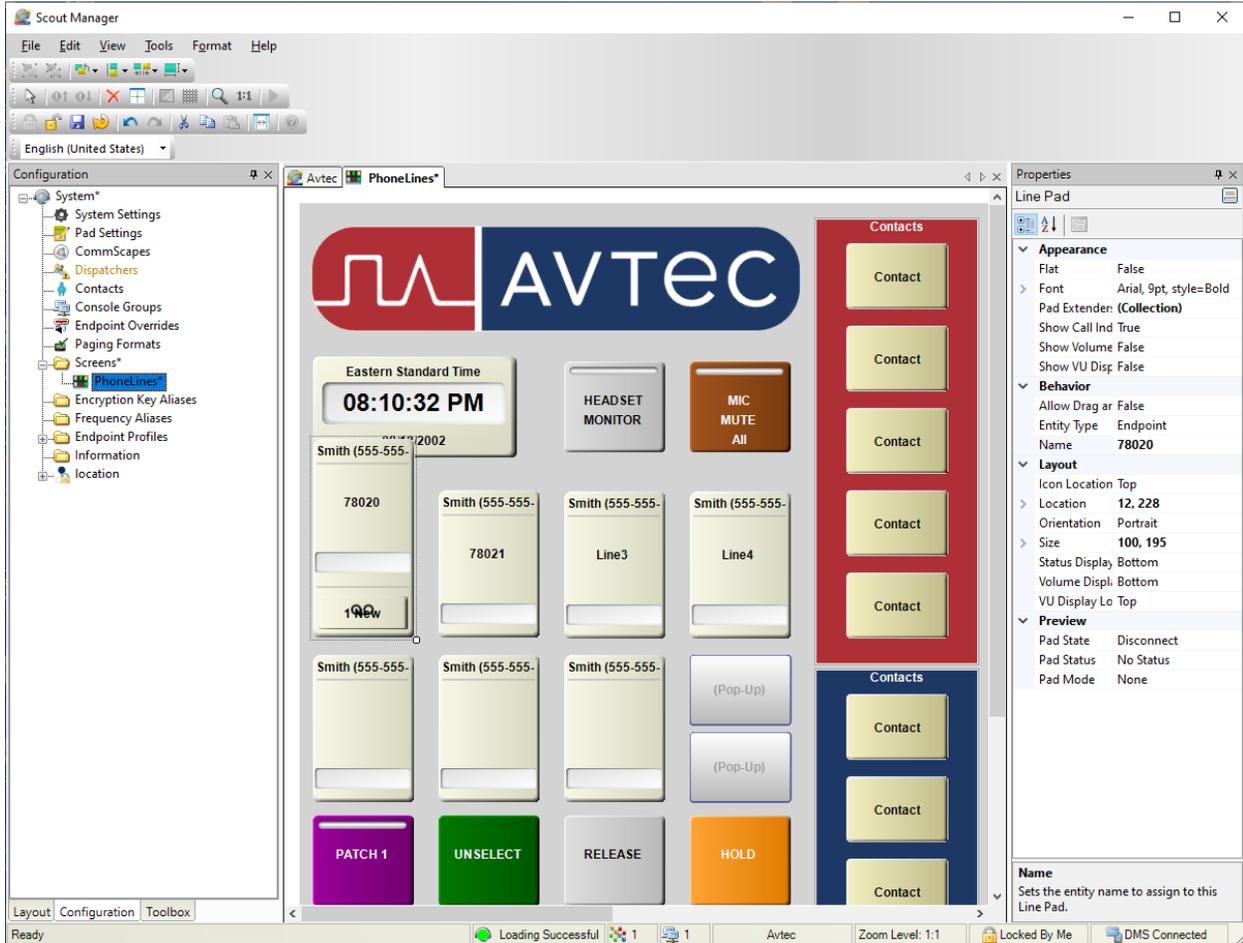
A voicemail/MWI button calls the voicemail system when pressed and provides an indication of any new voicemail messages. To add a voicemail/MWI button associated with a SIP line, select the **Configuration** tab in the left pane and open the **PhoneLines** screen shown below. Next, click on the SIP line button (e.g., 78020) that will be configured with a voicemail/MWI button. In the **Line Pad** section, click on the ellipses (...) button by **Pad Extender** to display the **Pad Extender Editor** window.



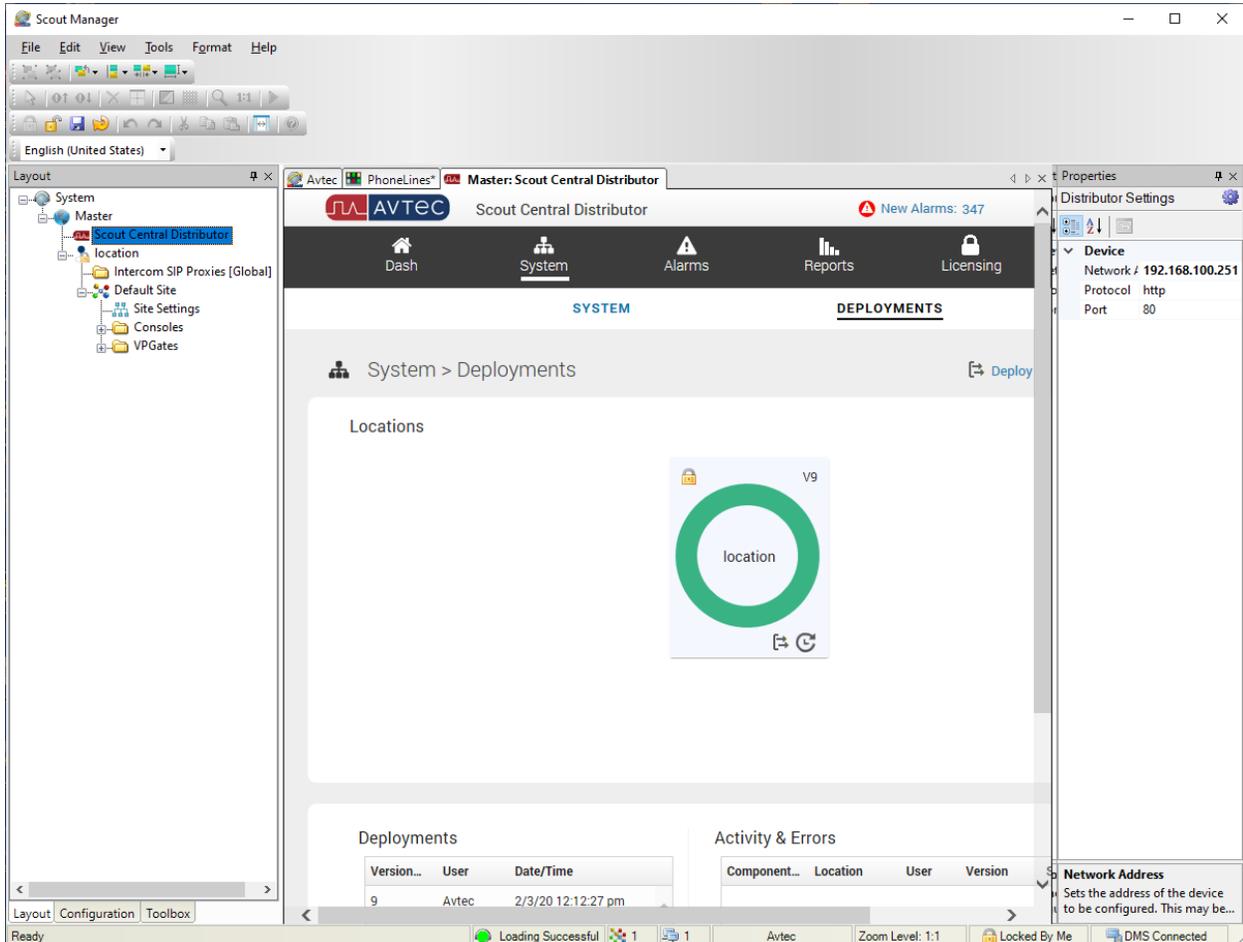
The **Pad Extender Editor** window is displayed below. Click the **Add** button. In the properties section to the right, set the **Type** field to *Voicemail*. Select the **Dock Position** for the button. In this example, the button was docked at the *Bottom*. Click **OK**.



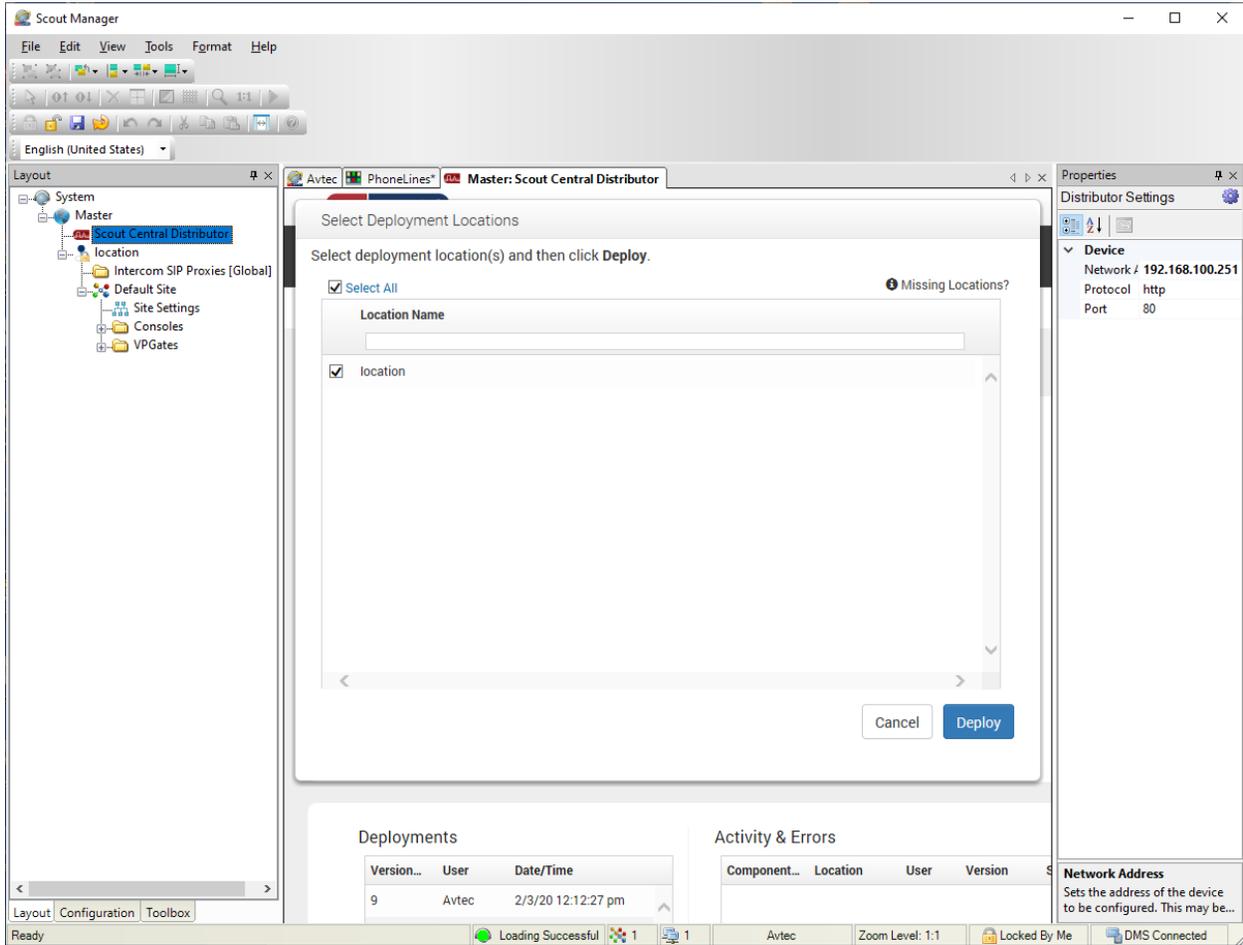
The voicemail/MWI button appears with the SIP line button as shown below. Click **Save**.



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.



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



The **Scout VoIP Console** below displays the voicemail/MWI button associated with the appropriate SIP line (e.g., 78020). Note that the label on the SIP line is the SIP extension (78020) , which was changed in **Section 7.3**.



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. Verify that the Scout VoIP Console has successfully registered with Session Manager. In System Manager, navigate to **Elements** → **Session Manager** → **System Status** → **User Registrations** to check the registration status as shown below.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The top navigation bar includes the Avaya logo, user profile, and various menu items like Users, Elements, Services, Widgets, and Shortcuts. The main content area is titled "User Registrations" and contains a table of 10 items. The table has columns for Address, First Name, Last Name, Actual Location, IP Address, Remote Office, Shared Control, Simult. Devices, AST Device, and Registered. The Registered column has sub-columns for Prim, Sec, and Surv. The table shows various user registrations with their respective details and registration status.

	Details	Address	First Name	Last Name	Actual Location	IP Address	Remote Office	Shared Control	Simult. Devices	AST Device	Registered		
											Prim	Sec	Surv
<input type="checkbox"/>	Show	78000@avaya.com	SIP	78000	---	192.168.100.54	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	78020@avaya.com	Avtec	78020	---	192.168.100.251	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	---	Equinox	78040	---	---	<input type="checkbox"/>	<input type="checkbox"/>	0/1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	78001@avaya.com	SIP	78001	---	192.168.100.58	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	78002@avaya.com	SIP	78002	---	192.168.100.59	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	78030@avaya.com	Agent	78030	---	192.168.100.49	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/> (AC)	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	Show	78021@avaya.com	Avtec	78021	---	192.168.100.251	<input type="checkbox"/>	<input type="checkbox"/>	1/1	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

2. Launch the Avtec Scout VoIP Console. The Scout VoIP Console will be displayed as shown below. If the SIP line is down, the line buttons will display *Unavailable*. The line buttons shown below indicate that the SIP lines for extensions 78020 and 78021 are in-service.



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.
6. Verify basic telephony features by establishing calls between Scout VoIP Console and another phone.

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. Avtec Scout VoIP Console was successfully registered with Session Manager via SIP endpoints and basic and supplementary telephony features were verified. All test cases passed with observations noted in **Section 2.2**.

10. Additional References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at support.avaya.com. Avtec Scout VoIP Console documentation is available through the application via online help.

- [1] *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 2, July 2019.
- [2] *Administering Avaya Aura® Session Manager*, Release 8.1, Issue 1, June 2019.

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