DevConnect Program

Application Notes for Avtec Scout VoIP Console 5.6 with Avaya Aura® Communication Manager 10.1 and Avaya Aura® Session Manager 10.1 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 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 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.

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 registers with Avaya Aura® Session Manager as SIP endpoints 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. 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 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:

- 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 reboot of the Scout VoIP Console and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observation:

- 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 registers with Session Manager as a SIP endpoint through the VPGate component installed on Windows 10. Scout Manager was used to configure Scout VoIP Console system.

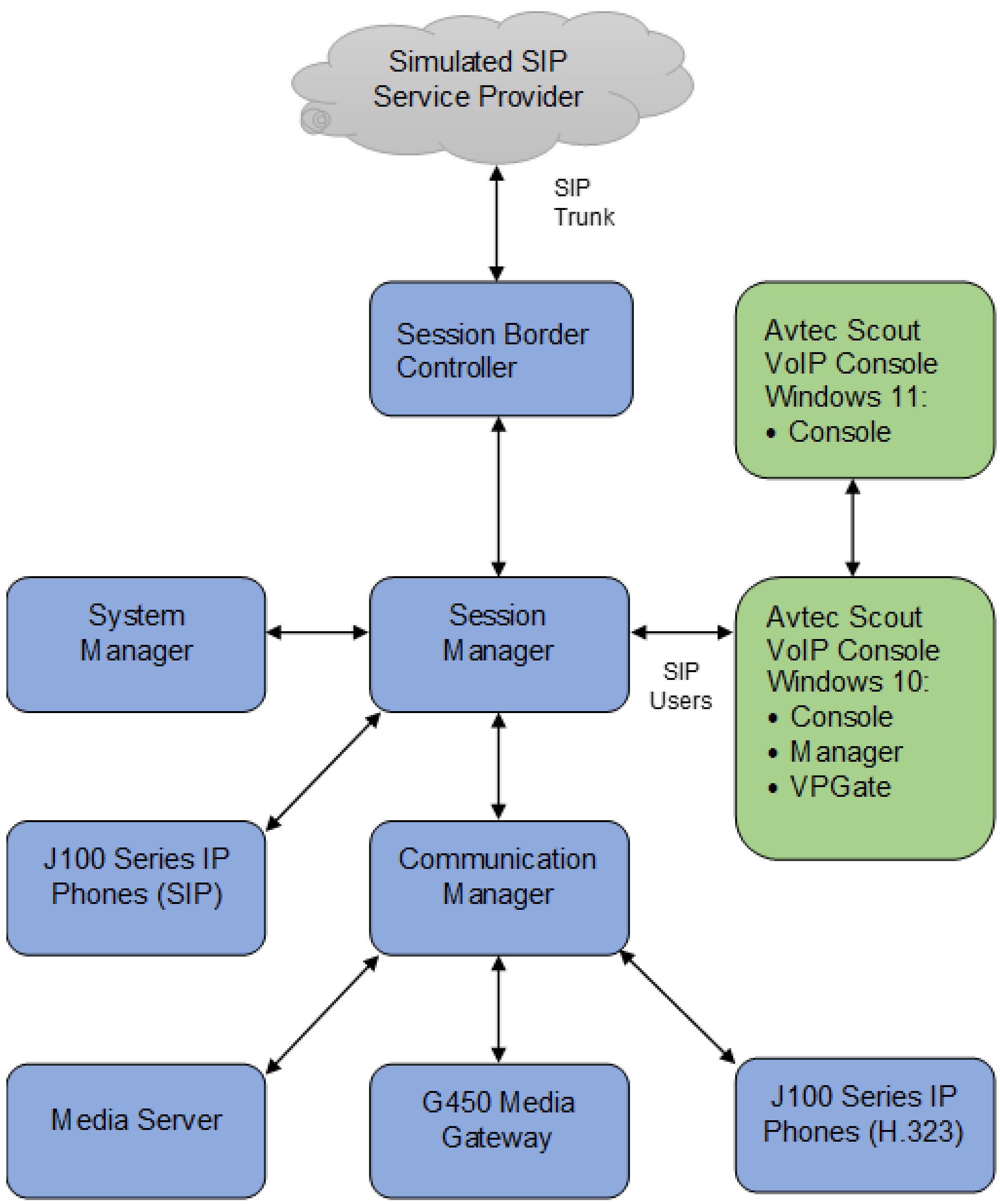


Figure 1: Avaya SIP Network with Avtec Scout VoIP Console

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	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: ▪ Scout Console (Windows 10 and Windows 11) ▪ Scout Central Distributor (Windows 10) ▪ Scout VPGate (Windows 10) ▪ Scout Manager (Windows 10)	 5.6.0.5 5.6.1.6 5.6.1.10 5.6.0.5

5. Configure Avaya Aura® Communication Manager

This section provides the steps for configuring Communication Manager using the System Access Terminal (SAT). The procedure includes the following areas:

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

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: V20                          Software Package: Enterprise
         Location: 2                             System ID (SID): 1
         Platform: 28                            Module ID (MID): 1

                                                          USED
                            Platform Maximum Ports:  48000    106
                                  Maximum Stations:    150     71
                          Maximum XMOBILE Stations:  36000      0
                Maximum Off-PBX Telephones - EC500:    150      0
                Maximum Off-PBX Telephones -   OPS:    150     41
                Maximum Off-PBX Telephones - PBFMC:    150      0
                Maximum Off-PBX Telephones - PVFMC:    150      0
                Maximum Off-PBX Telephones - SCCAN:      0      0
                Maximum Off-PBX Telephones -   EMX:    150      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 (*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
```

5.3. Administer IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed 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                                             Page   1 of   2

                          IP MEDIA PARAMETERS
    Codec Set: 1

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

     Media Encryption                  Encrypted SRTCP: best-effort
 1: 1-srtp-aescm128-hmac80
 2: 10-srtp-aescm256-hmac80
 3: none
 4:
 5:
```

5.4. 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 IP endpoints without using media resources in the Avaya 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: 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
```

5.5. 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*.
- **Enforce SIPS URI for SRTP** was set to *n*.
- Specify Communication Manager (*procr*) 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 5.2.**
- 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
                                             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? y
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. 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 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
```

5.6. Administer 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 "70" (extensions used by Scout VoIP Console are *70111 to 70113*) to route pattern *1* as shown below.

```
change aar analysis 70                                            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
    70                    5    5     1         lev0         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: main
    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: 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
```

6. Configure Avaya Aura® Session Manager

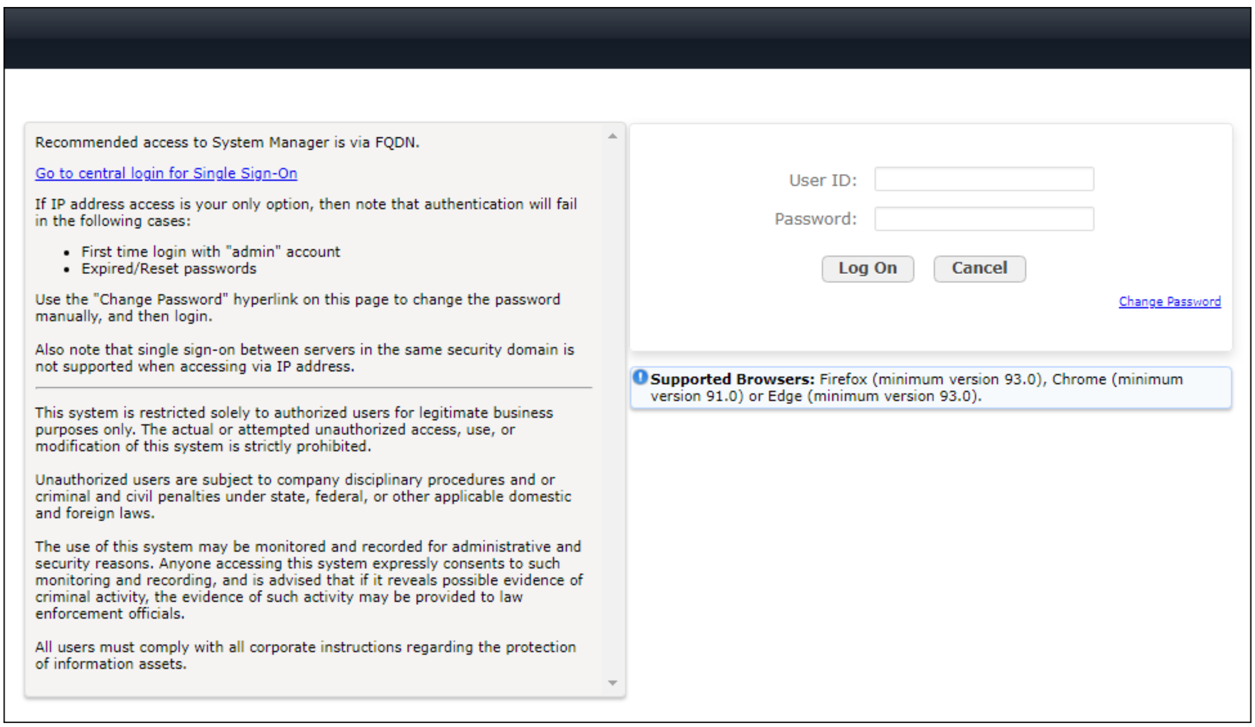
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- 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.

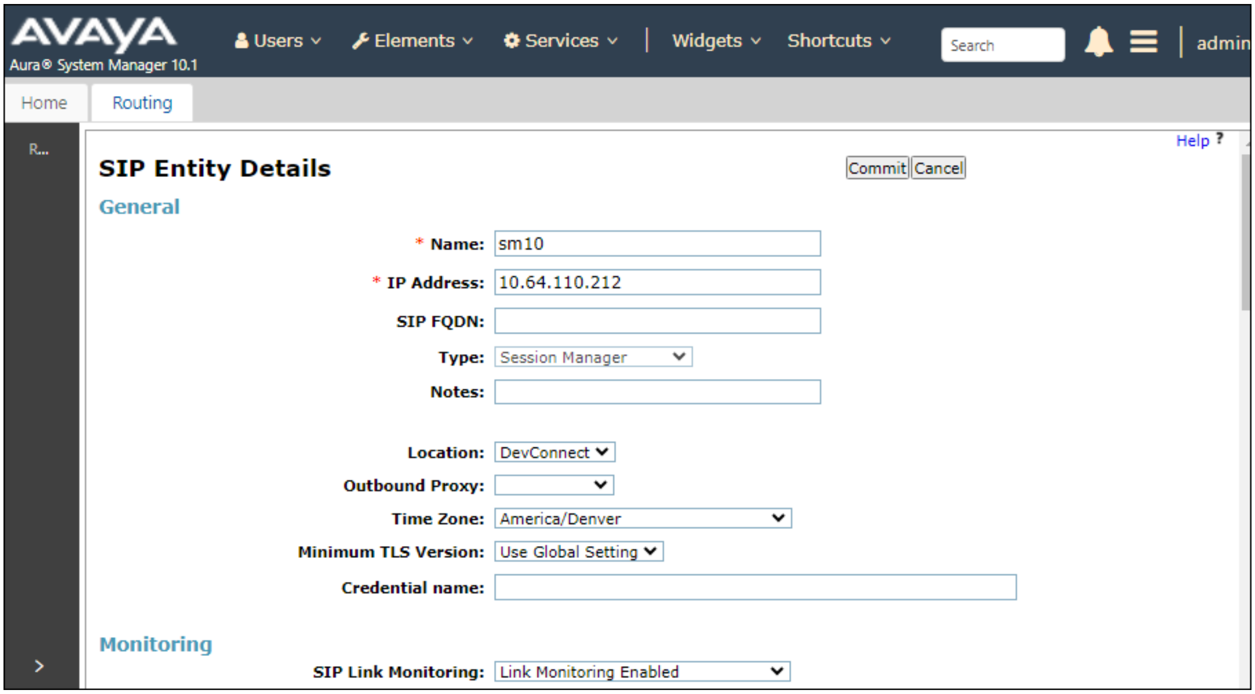
6.1. Launch System Manager

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.

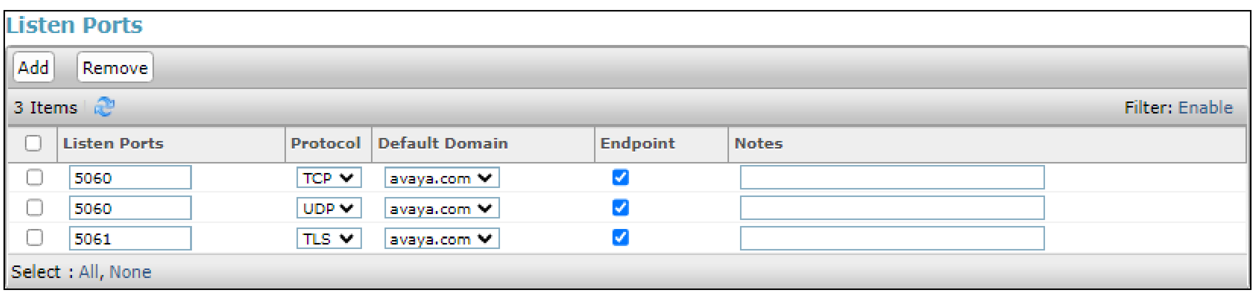


6.2. Set Network Transport Protocol for Scout VoIP Console

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

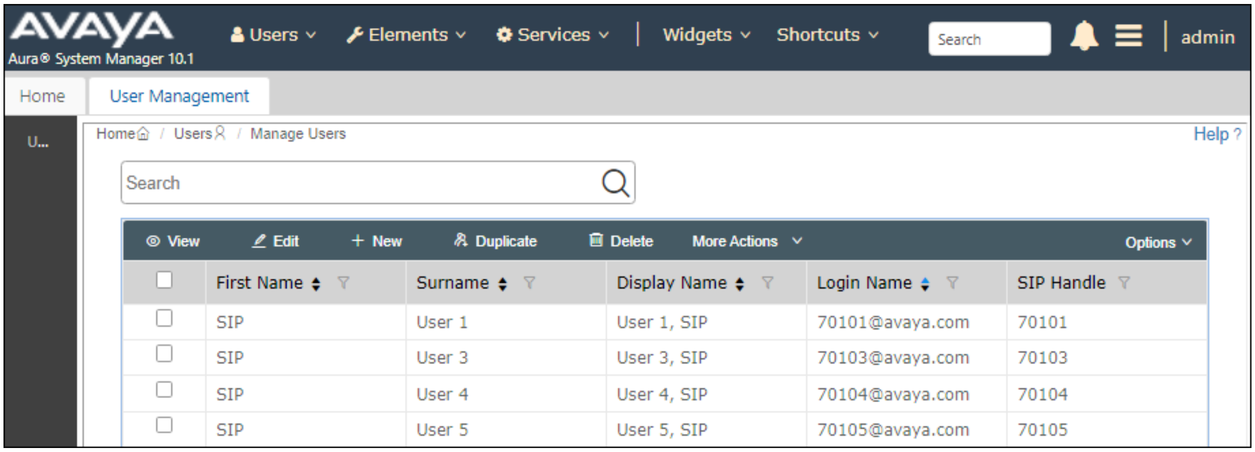


Scroll down to the **Listen Ports** section and verify that the transport network protocol used by Scout VoIP Console is checked in the list below. For the compliance test, the solution used UDP network transport.



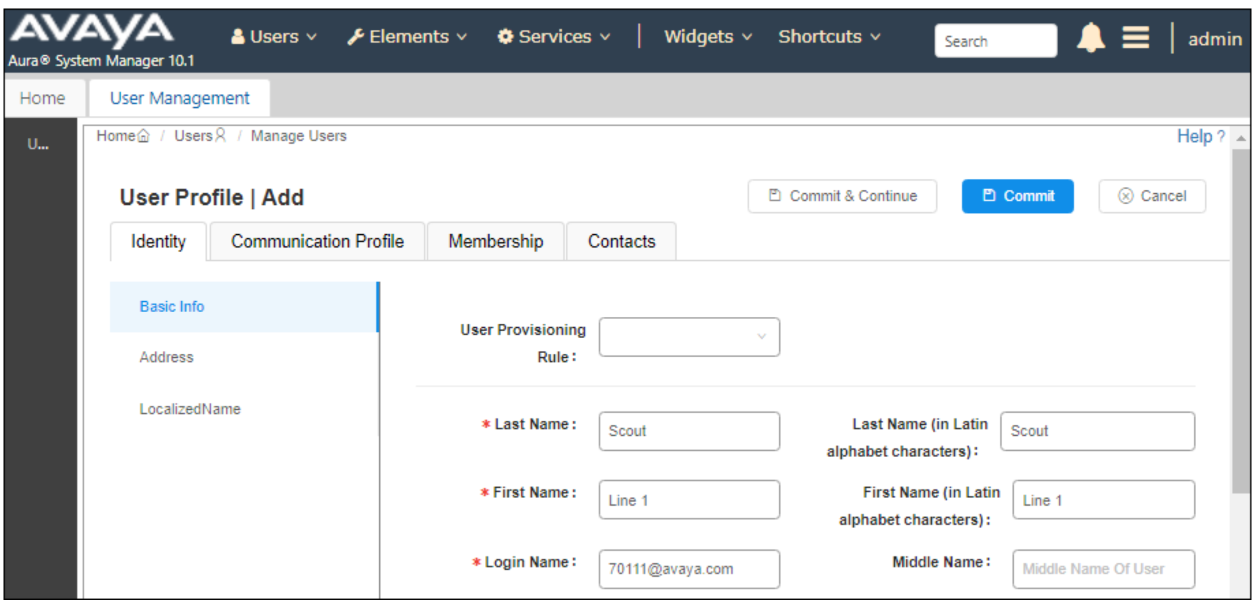
6.3. Administer SIP User

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.



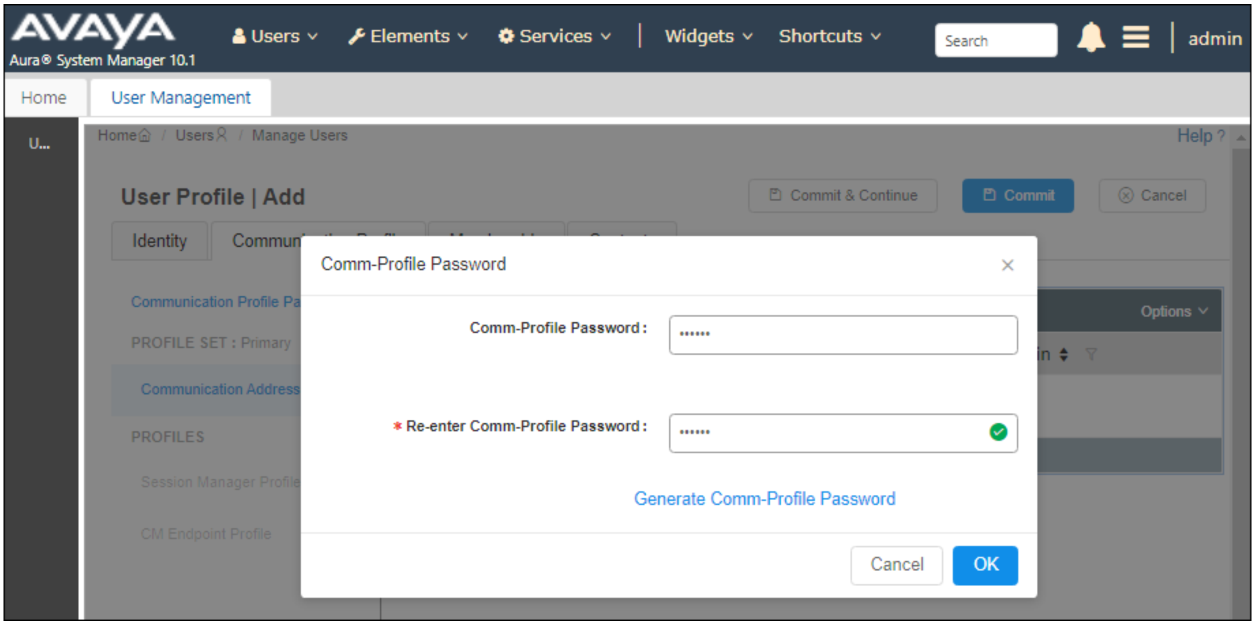
6.3.1. Identity

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 0**. Retain the default values in the remaining fields.



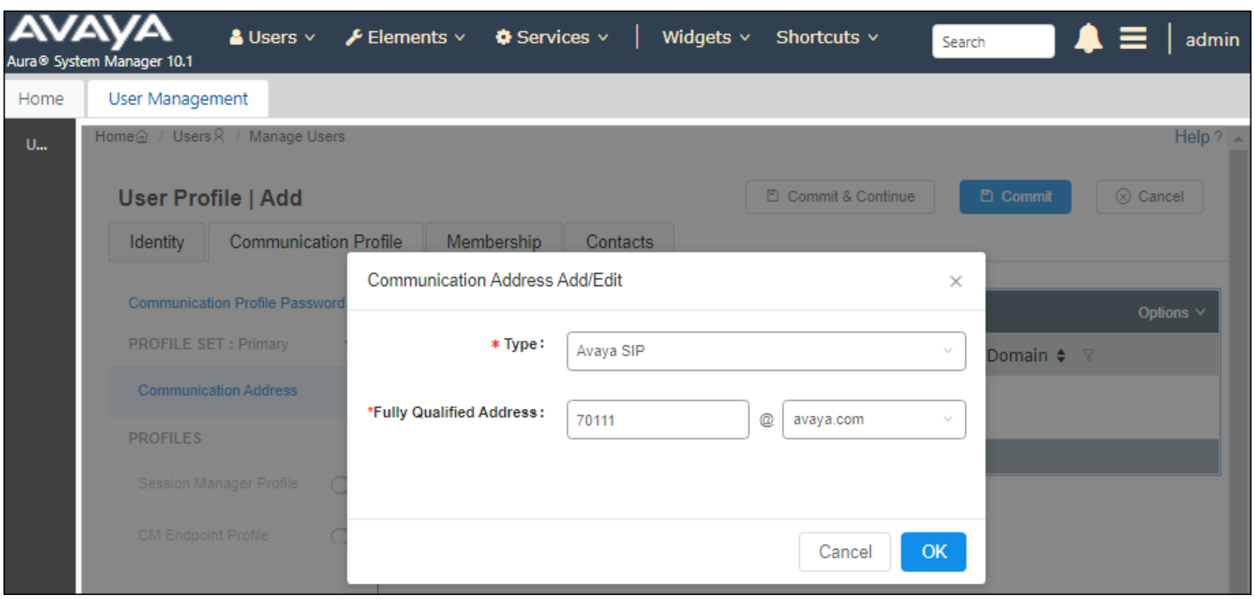
6.3.2. Communication Profile

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**.



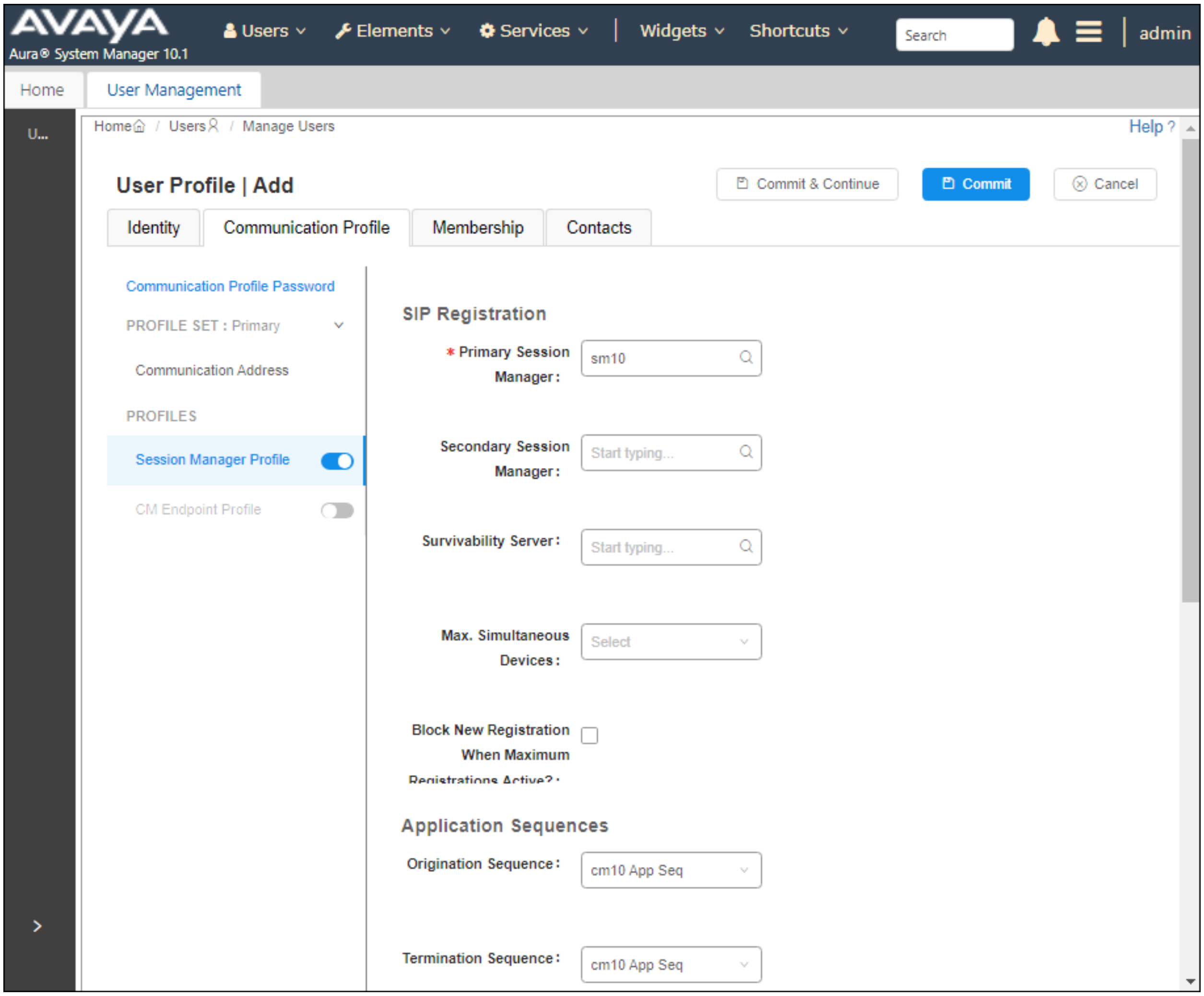
6.3.3. Communication Address

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**.

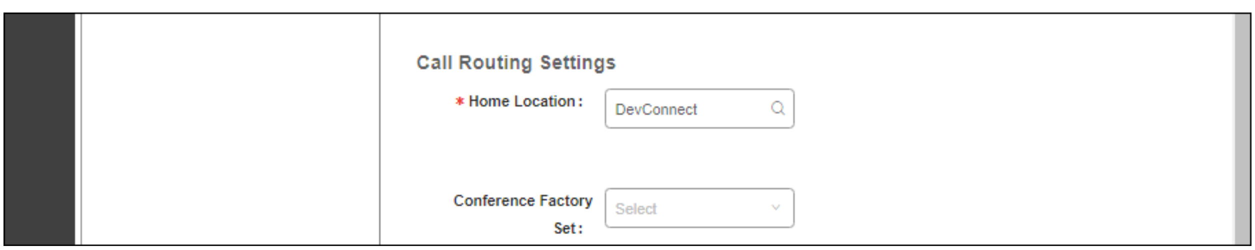


6.3.4. Session Manager Profile

Click on toggle button by **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, and **Termination Application Sequence**, select the values corresponding to the applicable Session Manager and Communication Manager.



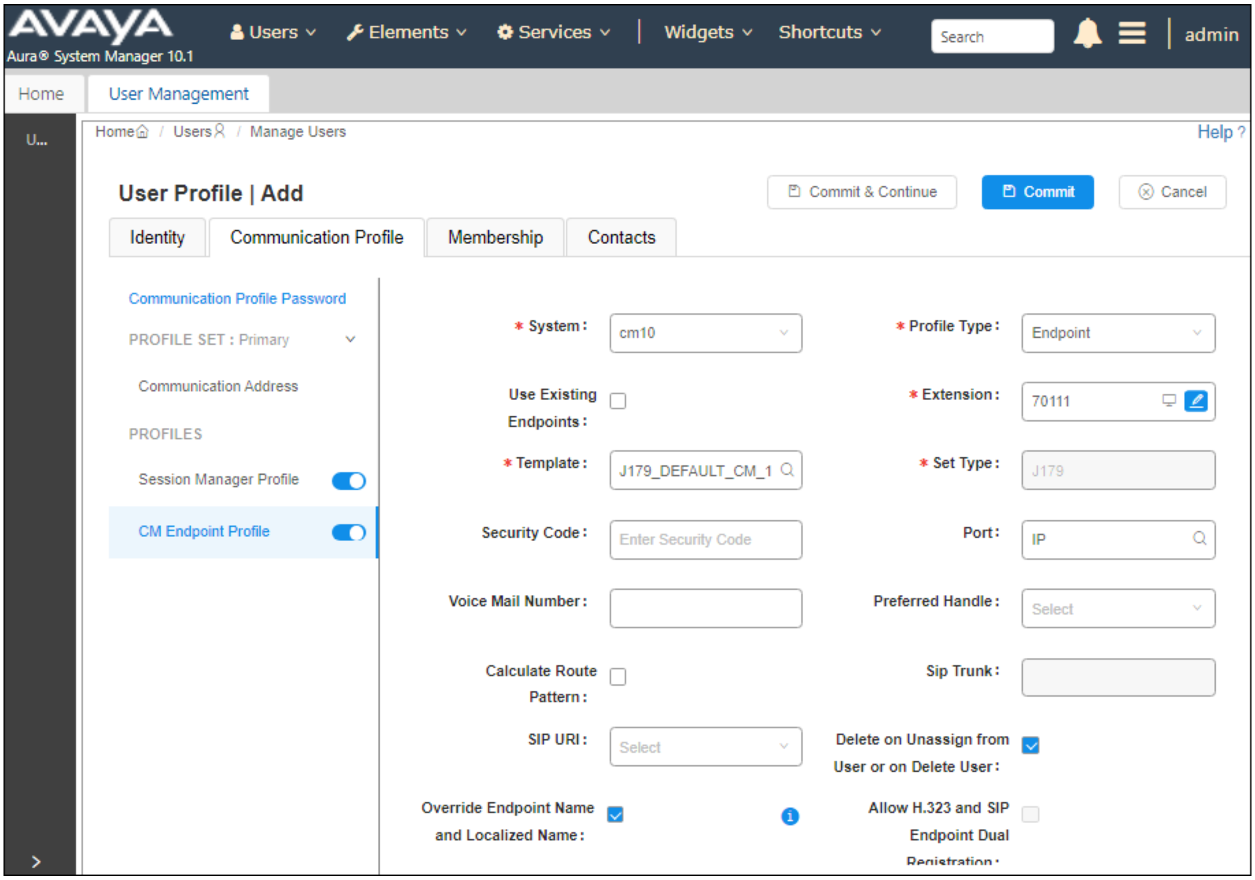
Scroll down to the **Call Routing Settings** section to configure the **Home Location**.



Retain the default values in the remaining fields.

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 *J179_DEFAULT_CM_10_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click commit.

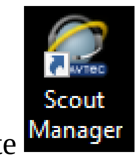


7. Configure Avtec Scout VoIP Console

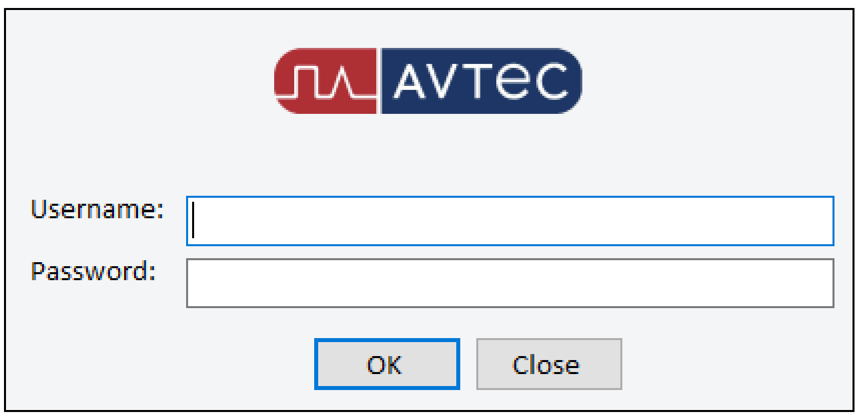
This section covers the configuration of Scout VoIP Console using the **Scout Manager** application. This section assumes that the Scout VoIP Console software has already been installed successfully. In the **Scout Manager** application, the following procedures are performed:

- Launch Scout Manager
- Add Endpoints
- Modify SIP Line Label and Set Endpoint Profile
- Add Voicemail/MWI Button
- Deploy the Configuration

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.



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., *70111*).
- **Service State:** Set to *Available*.

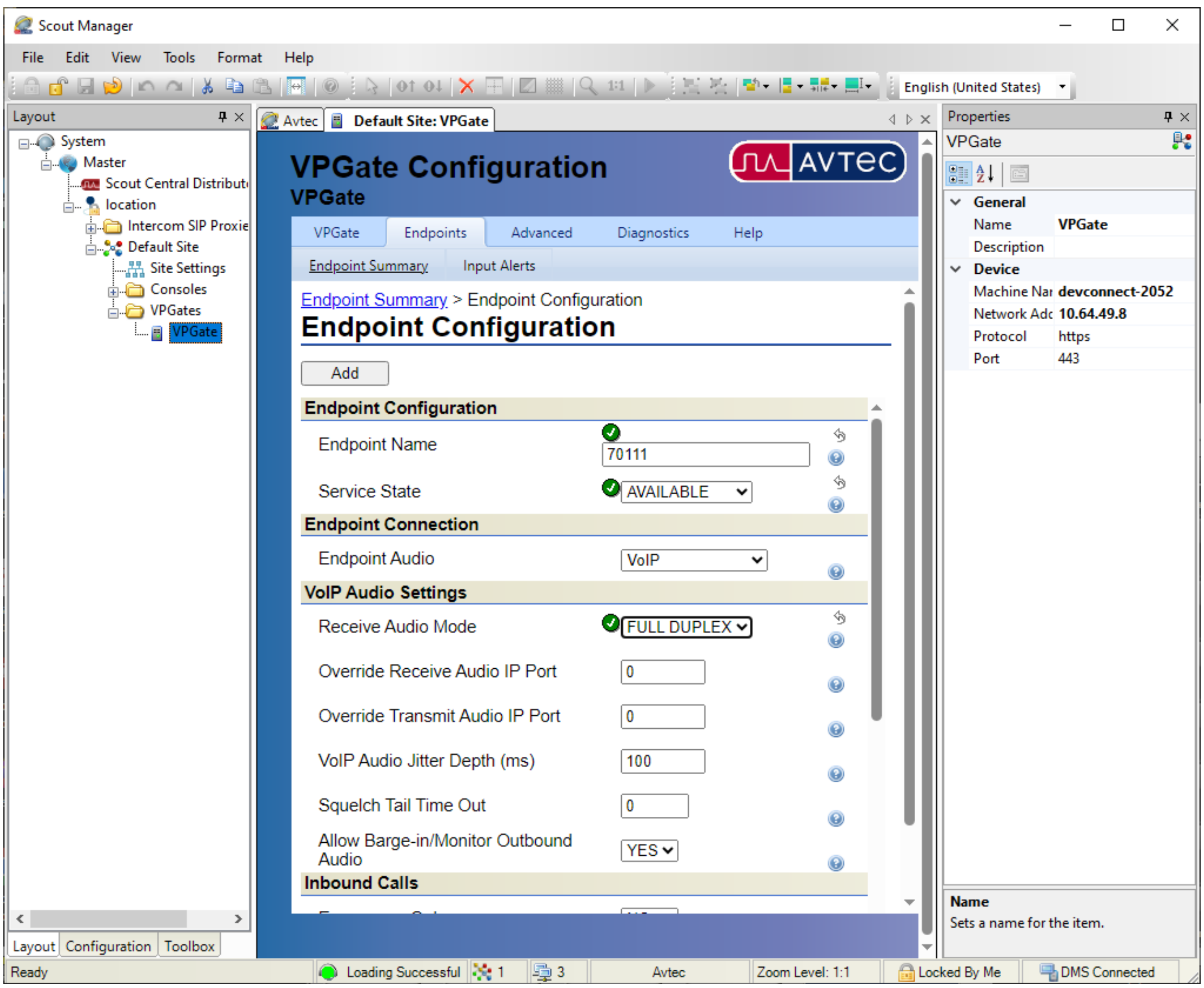
Under **Endpoint Connection**:

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

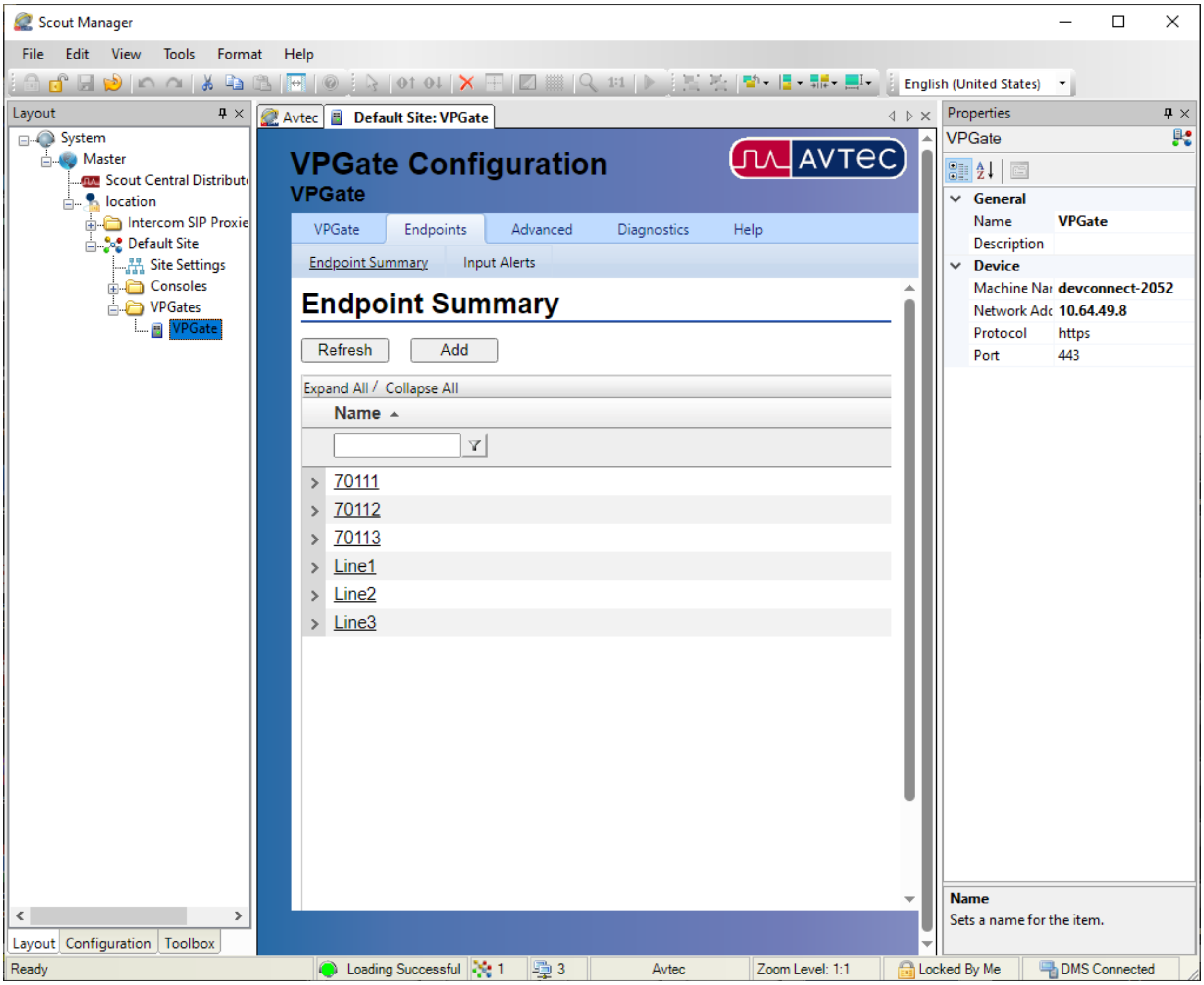
Under **VoIP Audio Settings**:

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

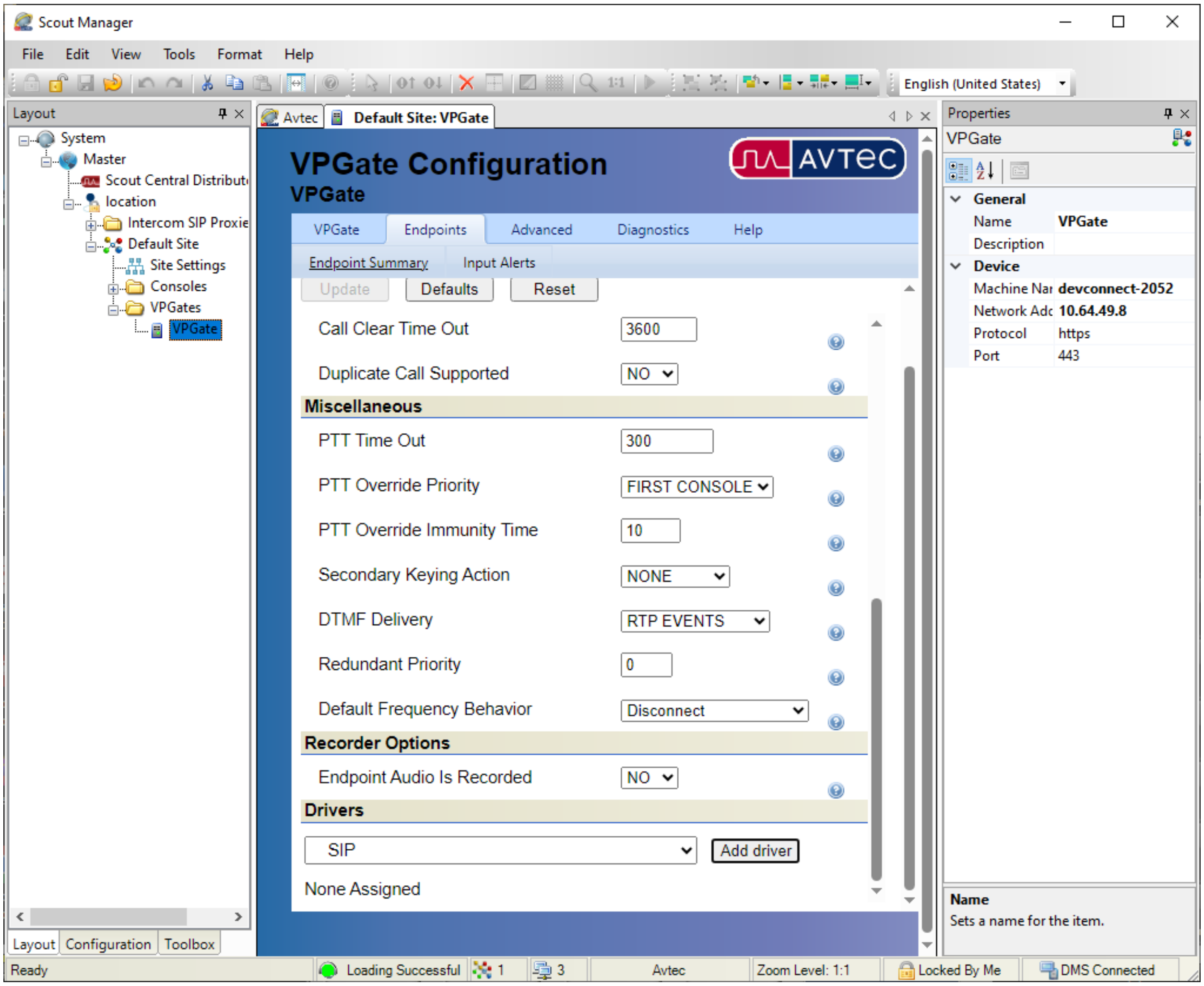
Use the default settings for the remaining fields. Click the **Add** button.



The **Endpoint** previously added is now displayed in the **Endpoint Summary** page shown below. Click on the endpoint that was previously added (i.e., *70111*) to open the configuration again.



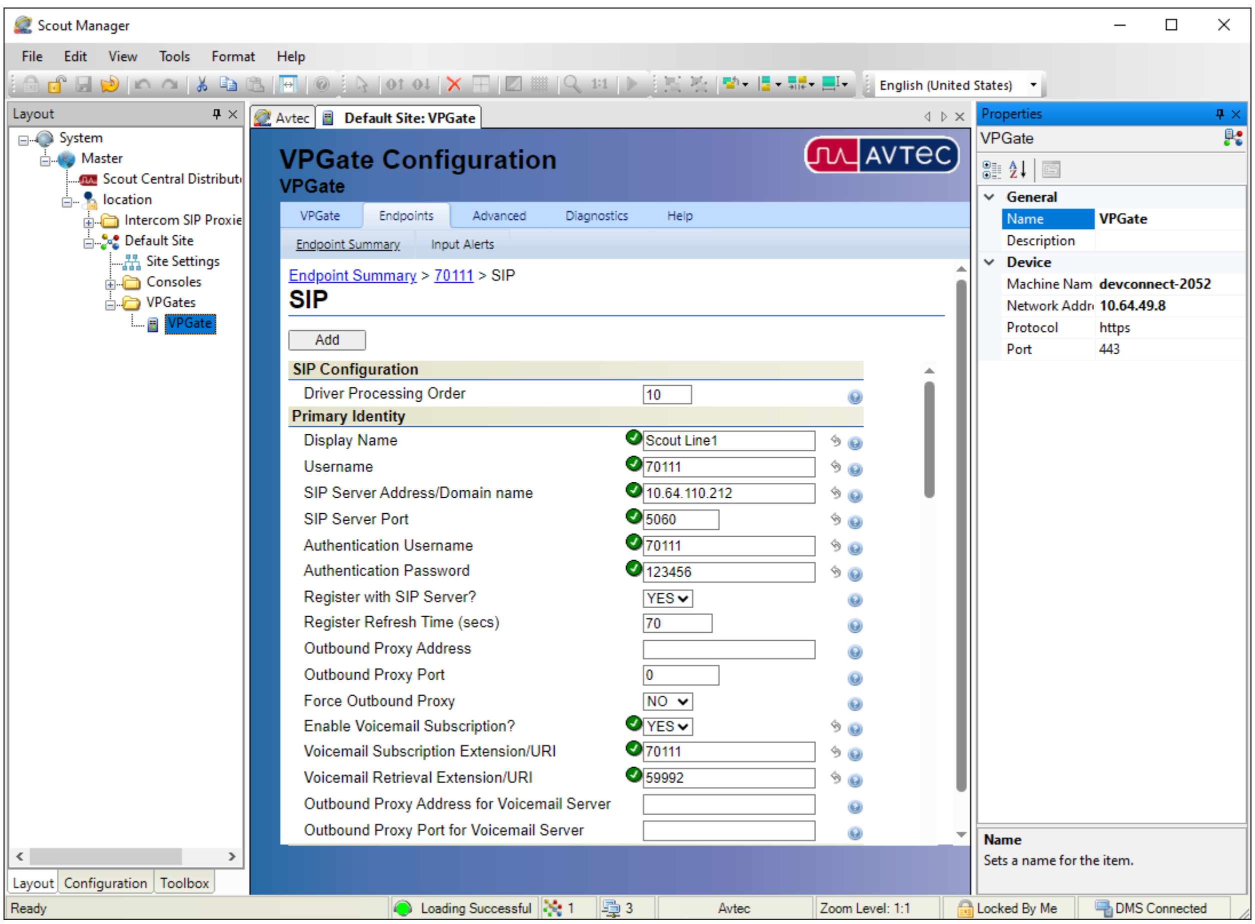
The **Endpoint Configuration** page is displayed. Scroll to the bottom of the page to the **Drivers** section as shown below. Select *SIP* from the drop-down field and click **Add driver**.



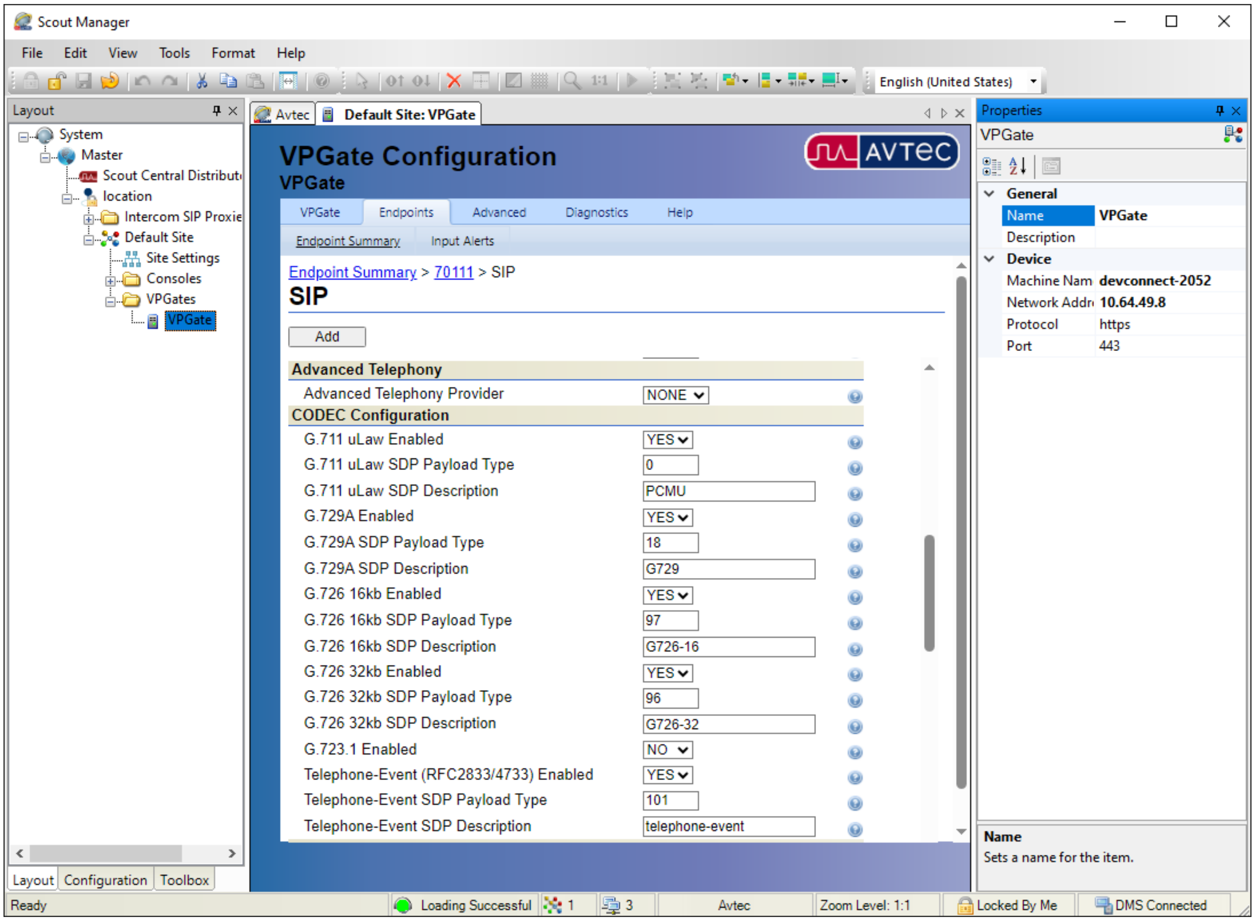
The **SIP** page is displayed as shown below. Under **Primary Identity**, configure the following fields:

▪ **Display Name:**	Specify a descriptive name (e.g., *Scout Line1*).
▪ **Username:**	Specify a descriptive name (e.g., *70111*).
▪ **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., *70111*).
▪ **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., *70111*).
▪ **Voicemail Retrieval Extension/URL:**	Specify the voicemail number.

Click **Update**.

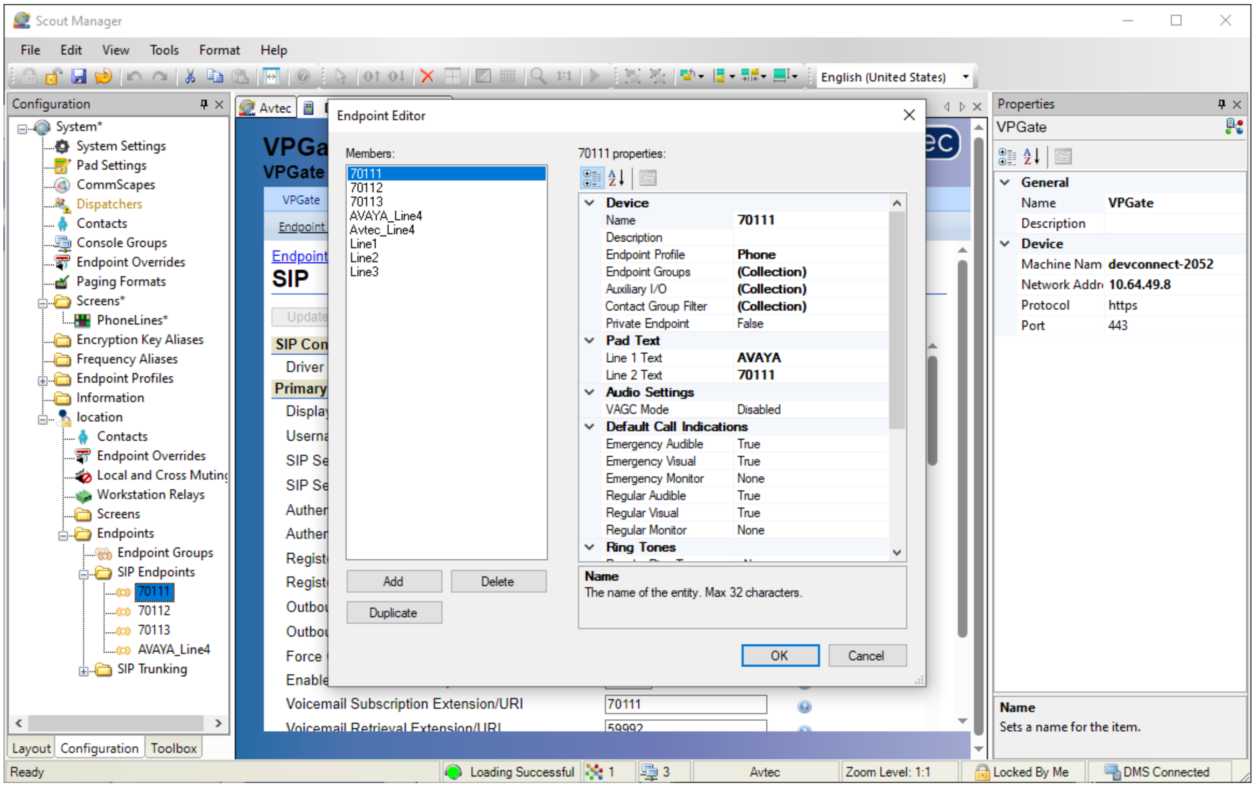


Scroll down to the **Codec Configuration** section and specify the codecs to be supported. In this example, G.711 and G.729 were enabled.



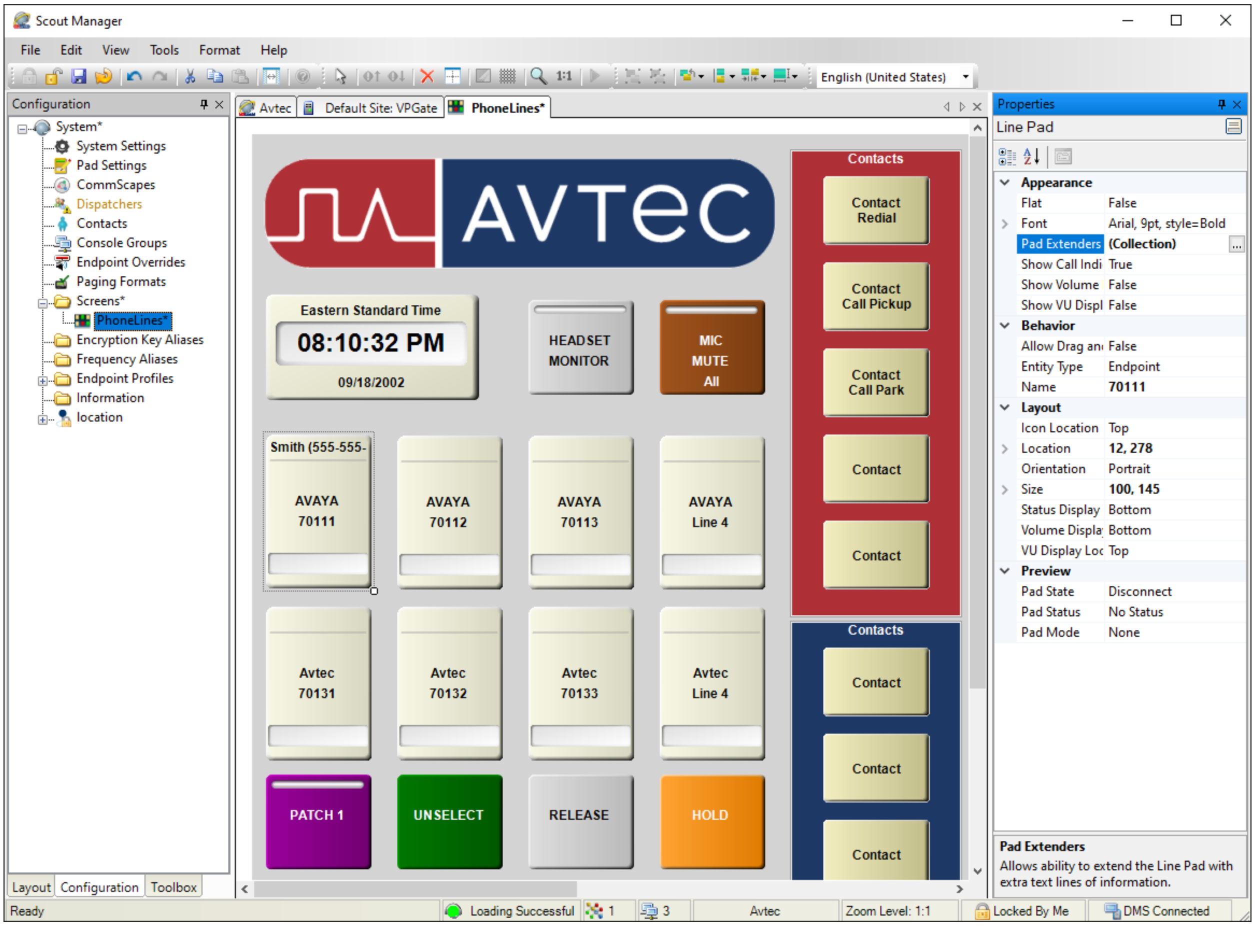
7.3. Modify SIP Line Label and Set Endpoint Profile

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 Endpoints** where the added endpoint displays. Double-click on the SIP endpoint added above. In the **Endpoint Editor**, change **Name** and **Line 2 Text** to the SIP extension (e.g., *70111*). 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.

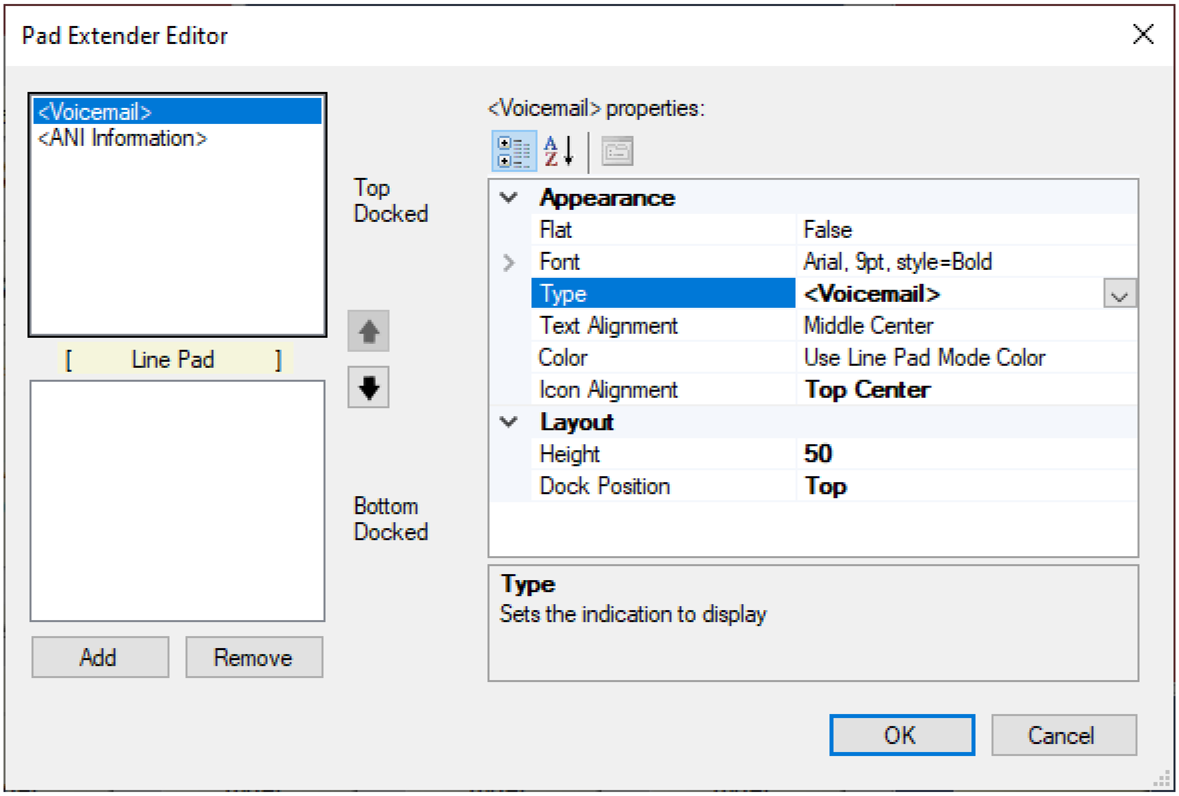


7.4. Add Voicemail/MWI Button

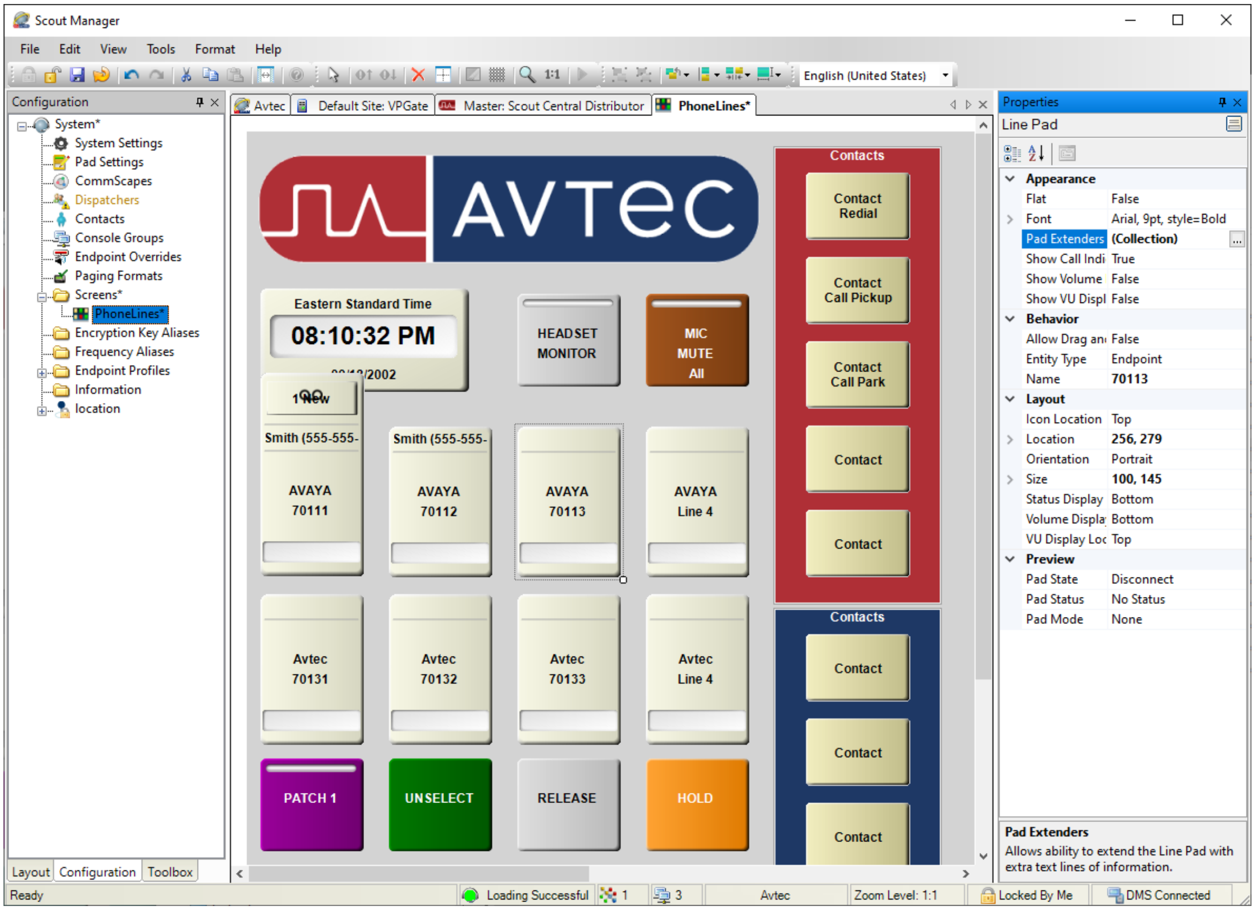
A voicemail/MWI button calls the voicemail system when pressed and provides an indication of any new voicemail messages. To add a voicemail/MWI button associated with a SIP line, select the **Configuration** tab in the left pane and open the **PhoneLines** screen shown below. Next, click on the SIP line button (e.g., *70111*) that will be configured with a voicemail/MWI button. In the **Line Pad** section, click on the ellipses (…) button by **Pad Extenders** 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 *Top*. Click **OK**.

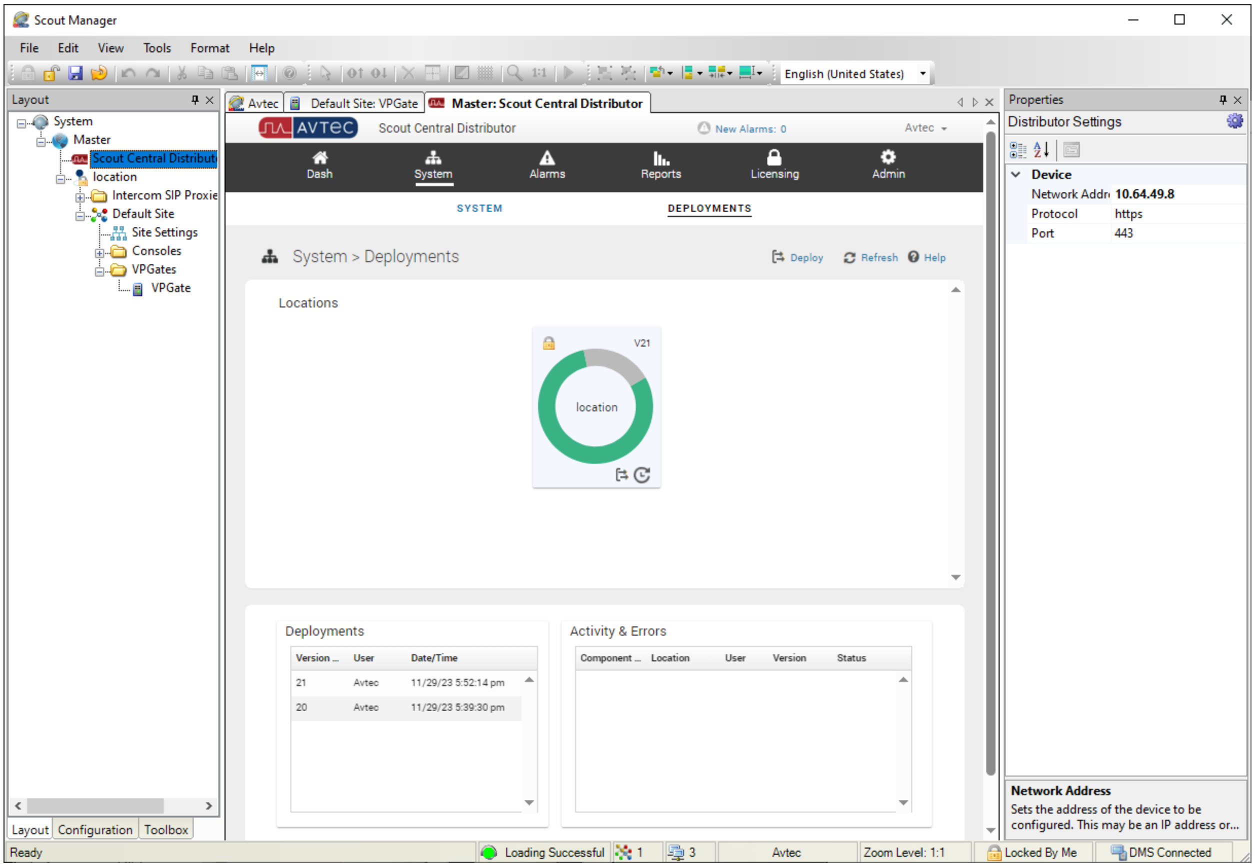


The voicemail/MWI button appears with the SIP line button as shown below. Click **Save**.

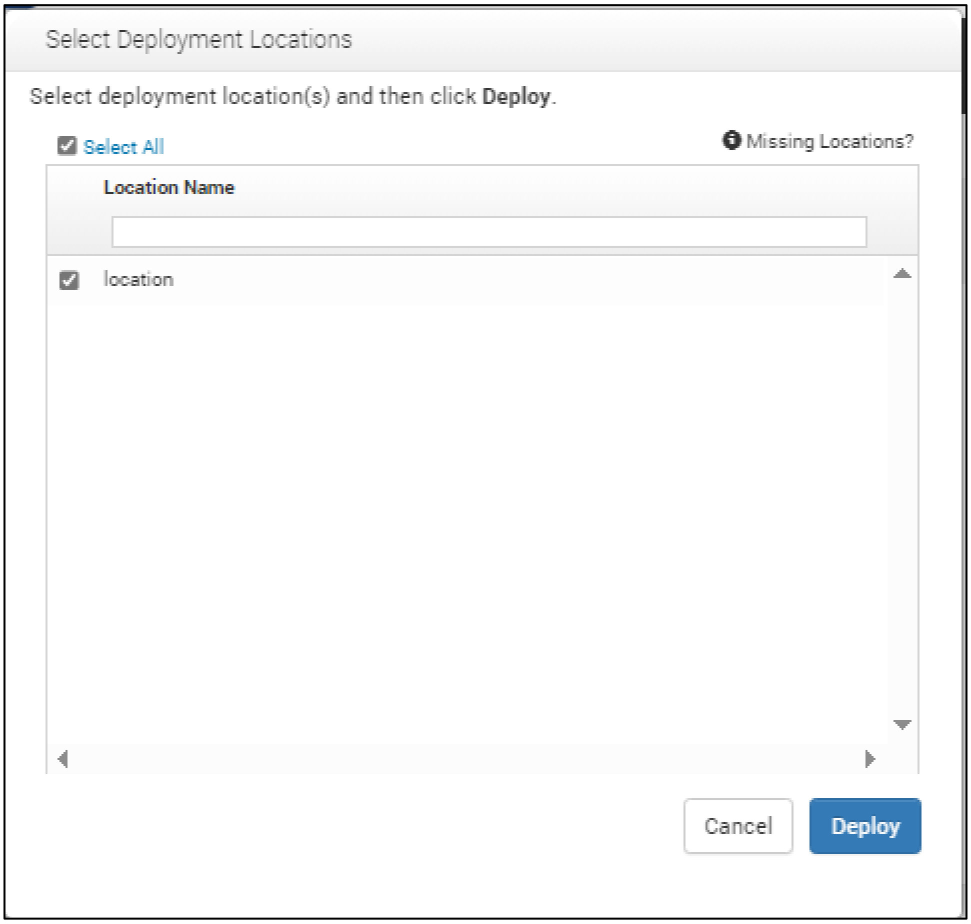


7.5. Deploy the Configuration

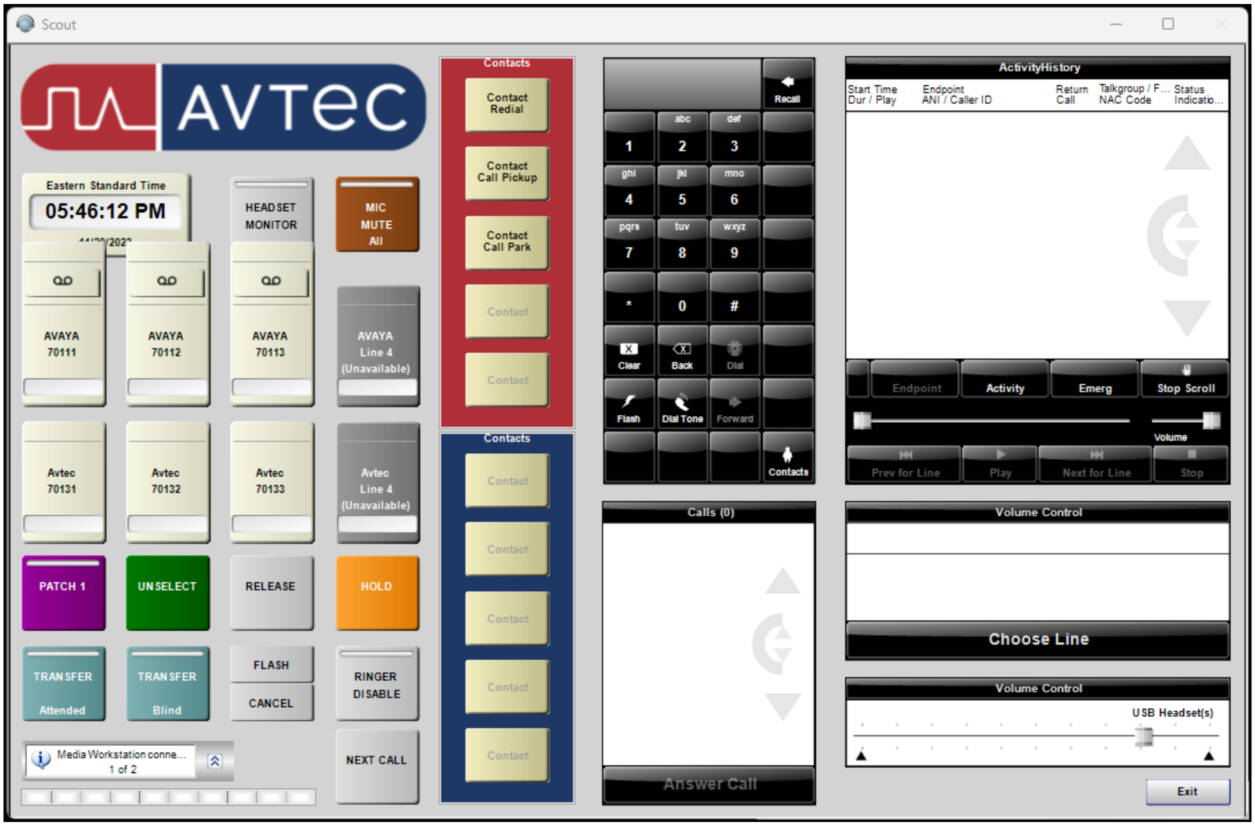
Lastly, deploy the changes to the **Scout VoIP Console**. Select the **Layout** tab in the left pane and then open the **Scout Central Distributor** screen as shown below. Navigate to **System → Deployment** to display the following screen. Click on the **Deploy** hyperlink.



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



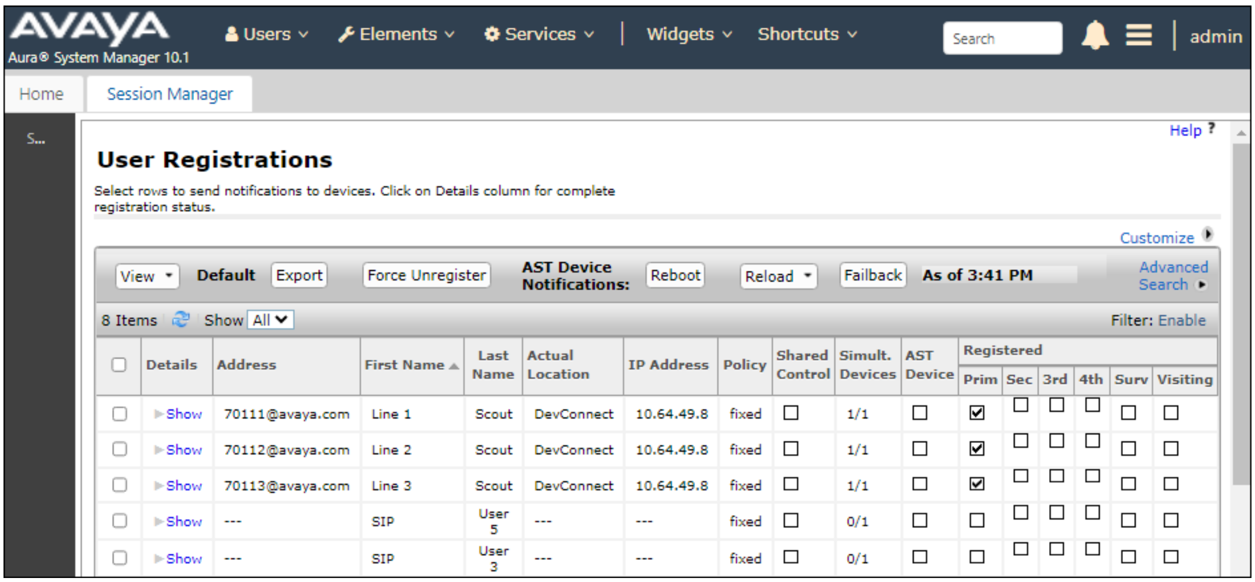
The **Scout VoIP Console** below displays the voicemail/MWI button associated with the appropriate SIP line (e.g., *70111*). Note that the label on the SIP line has the SIP extension *70111*, which was changed in **Section 7.3**.



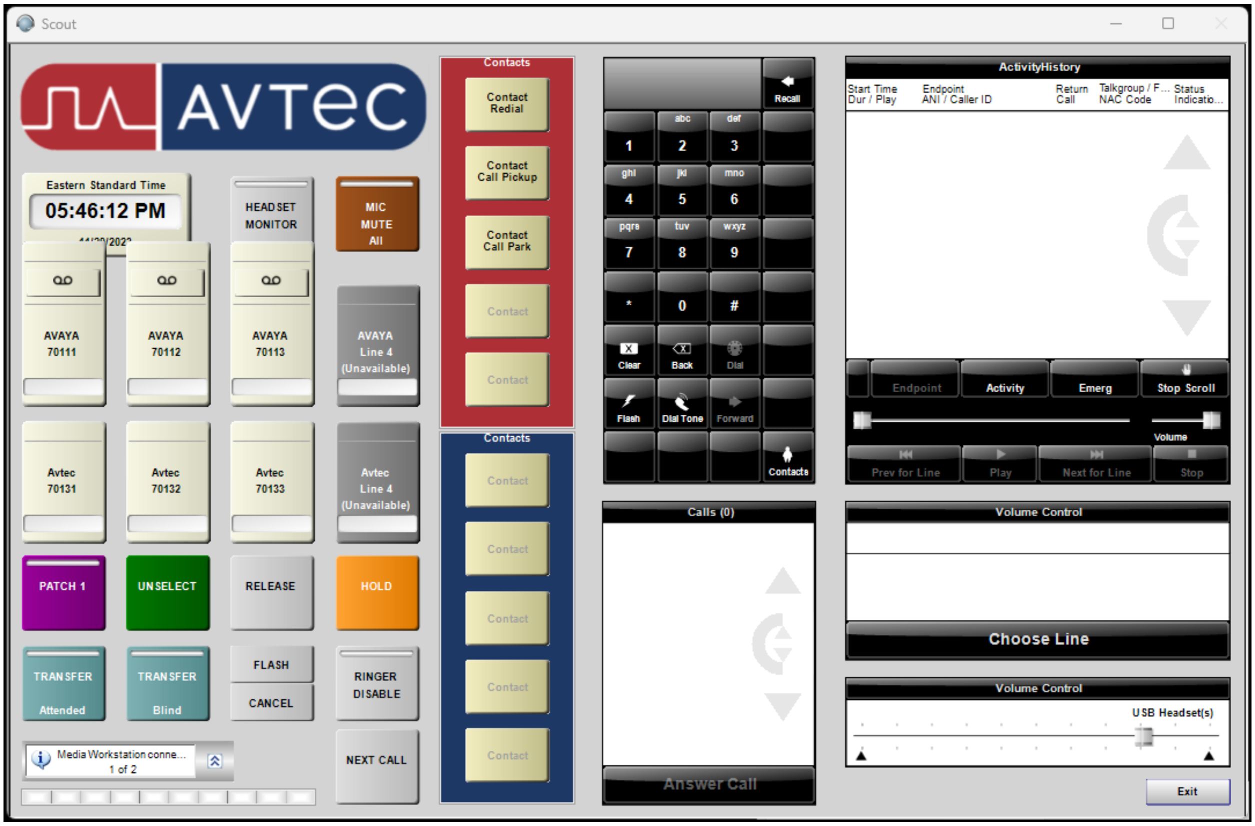
8. Verification Steps

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. In this example, 3 extensions, *70111*, *70112*, and *70113* are registered showing the IP address of the Windows system that has VPGate component installed.



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 (i.e., *Avaya 70111*, *Avaya* 70112, and *Avaya* 70113) shown below indicate that the SIP lines 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 as a SIP endpoint 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 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.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.