Avaya Solution & Interoperability Test Lab

Application Notes for Komutel SIT PC Attendant Console with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the Komutel SIT (**S**olution for **I**ntegrated **T**elecommunications) PC Attendant Console with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The SIT PC Attendant Console provides a desktop communications center with enhanced control of call handling features. It provides the ability to handle a high volume of calls and offers tools designed to manage telephony functions. In the compliance test, the SIT PC Attendant Console successfully registered with Session Manager, established calls with other telephones, and executed telephony features such as Hold, Transfer, and Conference.

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 steps required to integrate the Komutel SIT (**S**olution for **I**ntegrated **T**elecommunications) PC Attendant Console with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The SIT PC Attendant Console provides a desktop communications center with enhanced control of call handling features. It provides the ability to handle a high volume of calls and offers tools designed to manage telephony functions. In the compliance test, the SIT PC Attendant Console successfully registered with Session Manager, established calls with other telephones, and executed telephony features such as Hold, Transfer, and Conference.

2. General Test Approach and Test Results

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.

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between the SIT PC Attendant Console and Avaya SIP, H.323, and digital stations and exercising common telephony features, such as hold, transfer, and conference.

The serviceability testing focused on verifying that the SIT PC Attendant Console comes back into service after re-connecting the Ethernet connection or rebooting the PC on which the SIT PC Attendant Console is running.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of the SIT PC Attendant Console with Session Manager.
- Calls between SIT PC Attendant Console and Avaya SIP, H.323, and digital stations.
- G.711 codec support.
- Caller ID display on Avaya and Komutel endpoints.
- Proper recognition of DTMF tones.
- Basic telephony features including Hold, Mute, Transfer, and Conference.
- Extended telephony features using Communication Manager Feature Name Extensions (FNEs) such as Call Forwarding and Call Pickup.
- Proper system recovery after a restart of the SIT PC Attendant Console and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations noted:

- When activating a feature on the SIT PC Attendant Console using an FNE, the dial text box should be used to dial the FNE. If the feature being activated requires additional digits (i.e., second dial tone is provided), those digits should be entered using the keypad on the console. If the dial textbox is use, the console will attempt to transfer the call. This is working as designed on the console.

- When an incoming or outgoing call is aborted (i.e., originator hangs up call prior to answer), the SIT PC Attendant Console does not clear its display.

- SIT PC Attendant Console does not update its display after a call transfer.

2.3. Support

For technical support on the SIT PC Attendant Console, contact Komutel Support via phone, email, or website.

- **Phone:** (877) 225-9988
- **Email:** service@komutel.com
- **Web:** http://www.gotoassist.com/ph/komutel

3. Reference Configuration

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

- Avaya Aura® Communication Manager running on an Avaya S8300 Server with an Avaya G450 Media Gateway. Communication Manager was configured as an Evolution 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.

In addition, a Komutel SIT PC Attendant Console registered with Session Manager and was configured as Off-PBX Stations (OPS) on the Communication Manager.

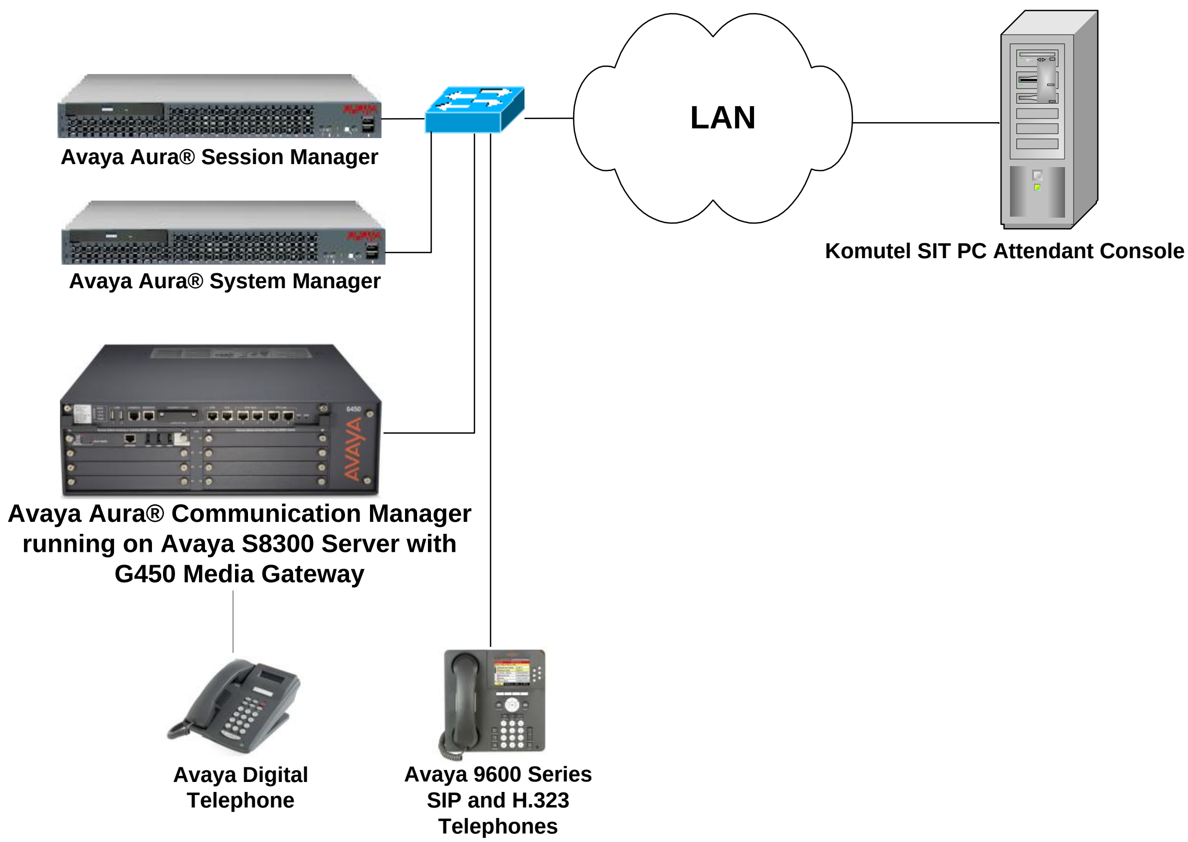


Figure 1: Avaya SIP Network with Komutel SIT PC Attendant Console

3.1. SIP Call Flows

The Komutel SIT PC Attendant Console originates a call by sending a call request (SIP INVITE message) to Session Manager, which then routes the call over a SIP trunk to Communication Manager for origination services. If the call is destined for another local SIP phone, Communication Manager routes the call back over the SIP trunk to Session Manager for delivery to the destination SIP phone. If the call is destined for an H.323 or digital telephone, Communication Manager routes the call directly to the H.323 or digital endpoint.

For a call arriving at Communication Manager that is destined for an SIT PC Attendant Console, Communication Manager routes the call over the SIP trunk to Session Manager for delivery to the SIT PC Attendant 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 running on an Avaya S8300 Server with and G450 Media Gateway	6.2 (R016x.02.0.823.0 with Patch 19926)
Avaya Aura® Session Manager	6.2 (6.2.3.0623006)
Avaya Aura® System Manager	6.2.0 SP 3
Avaya 9600 Series IP Telephones	3.1 SP 5 (H.323) 2.6.7 (SIP)
Avaya Digital Telephones	--
Komutel SIT PC Attendant Console with the modTelephony_SIP.dll plugin.	2.2.0.11058 (SIT) 1.0.4.11109 (plugin)

5. Configure Avaya Aura® Communication Manager

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

- Verify Communication Manager license
- Configure the SIT PC Attendant Console as an Off-PBX Station (OPS)
- Configure a SIP trunk between Communication Manager and Session Manager

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

5.1. Verify OPS and SIP Trunk Capacity

Using the SAT, verify that the Off-PBX Telephones (OPS) and the SIP Trunks options are 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  11
                                OPTIONAL FEATURES

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

                                                              USED
                                Platform Maximum Ports: 6400  223
                                      Maximum Stations: 2400  80
                              Maximum XMOBILE Stations: 2400  0
                    Maximum Off-PBX Telephones - EC500: 9600  0
                    Maximum Off-PBX Telephones -   OPS: 9600  42
                    Maximum Off-PBX Telephones - PBFMC: 9600  0
                    Maximum Off-PBX Telephones - PVFMC: 9600  0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   0

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

On **Page 2** of the **system-parameters customer-options** form, verify that the number of SIP trunks supported by the system is sufficient.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 4000  30
           Maximum Concurrently Registered IP Stations: 2400  5
             Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 2400  0
                       Maximum Administered SIP Trunks: 4000  90
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 50    0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

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

5.2. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for the S8300 Server in the G450 Media Gateway and the Session Manager SIP interface. The host names will be used throughout the 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-asm          10.32.24.235
devcon13            10.32.24.20
lz-asm              192.168.100.235
procr               192.168.100.10
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
```

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 *devcon.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. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling group.

```
change ip-network-region 1                                         Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: devcon.com
    Name:
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? y
   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
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to the SIT PC Attendant 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' as shown above. The default settings of the **IP Codec Set** form are shown below. The SIT PC Attendant Console supports G.711.

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

                          IP Codec Set

    Codec Set: 1

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

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 *tcp*.
- Specify the S8300 Server (*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 TCP port value of *5060* 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 60                                          Page   1 of   2
                                SIGNALING GROUP

 Group Number: 60             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM


   Near-end Node Name: procr                  Far-end Node Name: lz-asm
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 1

Far-end Domain: devcon.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 calls to SIP endpoints. 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 fields in bold and accept the default values for the remaining fields.

```
add trunk-group 60                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 60                   Group Type: sip          CDR Reports: y
  Group Name: To lz-asm                  COR: 1        TN: 1        TAC: 1060
   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: 60
                                                     Number of Members: 40
```

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.

```
change trunk-group 60                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     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
```

Configure the **Private Numbering 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 "50", have the extension sent to the far-end for display purposes.

```
change private-numbering 0                                     Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 5  4                                            5     Total Administered: 1
                                                          Maximum Entries: 540
```

5.3. Configure Station for Komutel SIT PC Attendant Console

Use the **add station** command to add a station for the SIT PC Attendant Console to be supported. Use *9630SIP* for the **Station Type** and provide a descriptive **Name**. Use the default values for the other fields on **Page 1**. The SIP station can also be configured automatically by System Manager as described in **Section 6.7**.

```
add station 46100                                           Page   1 of   6
                                     STATION

Extension: 46100                          Lock Messages? n               BCC: 0
     Type: 9630SIP                        Security Code:                  TN: 1
     Port: IP                           Coverage Path 1:                 COR: 1
     Name: Komutel, SIT                 Coverage Path 2:                 COS: 1
                                        Hunt-to Station:
STATION OPTIONS
                                            Time of Day Lock Table:
              Loss Group: 19
                                                  Message Lamp Ext: 46100

        Display Language: english                   Button Modules: 0

          Survivable COR: internal
   Survivable Trunk Dest? y                           IP SoftPhone? n

                                                          IP Video? n
```

Use the **change off-pbx-telephone station-mapping** command to map the Communication Manager extensions (e.g., 46100) to the same extension configured in System Manager. Enter the field values shown. For the sample configuration, the **Trunk Selection** field is set to *aar* so that AAR call routing is used to route calls to Session Manager. AAR call routing configuration is not shown in these Application Notes. The **Configuration Set** value can reference a set that has the default settings.

```
change off-pbx-telephone station-mapping 46100                 Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk        Config  Dual
 Extension                 Prefix                     Selection    Set     Mode
 46100            OPS          -      46100            aar         1
```

On **Page 2**, change the **Call Limit** to match the number of *call-appr* entries in the station form, which should also match the number of lines configured on the SIT PC Conosle. Also, verify that **Mapping Mode** is set to *both* (the default value for a newly added station).

```
change off-pbx-telephone station-mapping 46100                 Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 46100         OPS      3        both         all         none
```

6. Configure Avaya Aura® Session Manager

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

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities corresponding to Session Manager and Communication Manager
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Application Sequence
- Define Communication Manager as Administrable Entity (i.e., Managed Element)
- Session Manager, corresponding to the Session Manager Server to be managed by System Manager
- Add SIP Users

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL "https://<*ip-address*>/SMGR", where <*ip-address*> is the IP address of System Manager. Log in with the appropriate credentials.

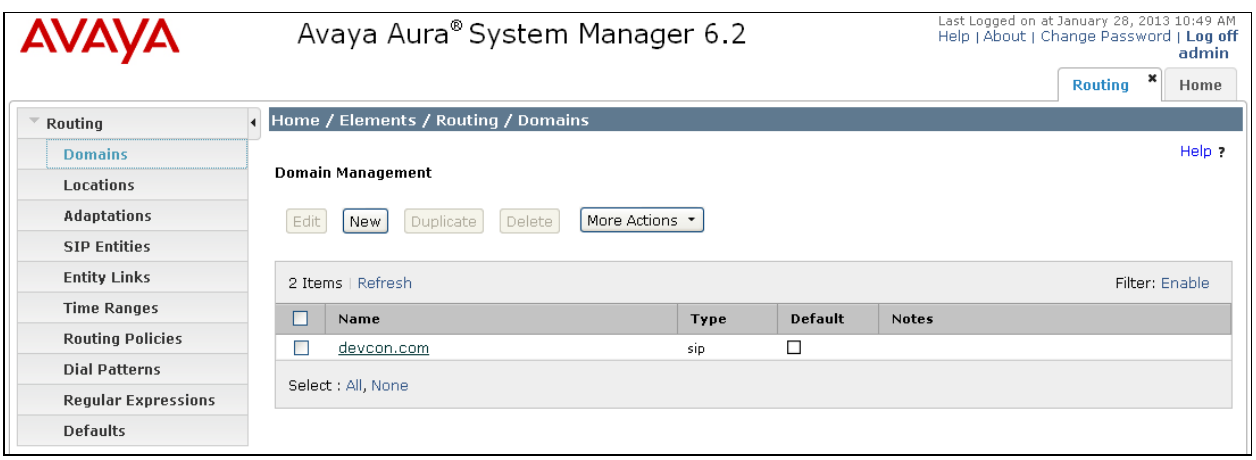
6.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button on the right. The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., *devcon.com*)
- **Notes:** Descriptive text (optional).

Click **Commit** (not shown).

Since the sample configuration does not deal with any other domains, no additional domains need to be added.



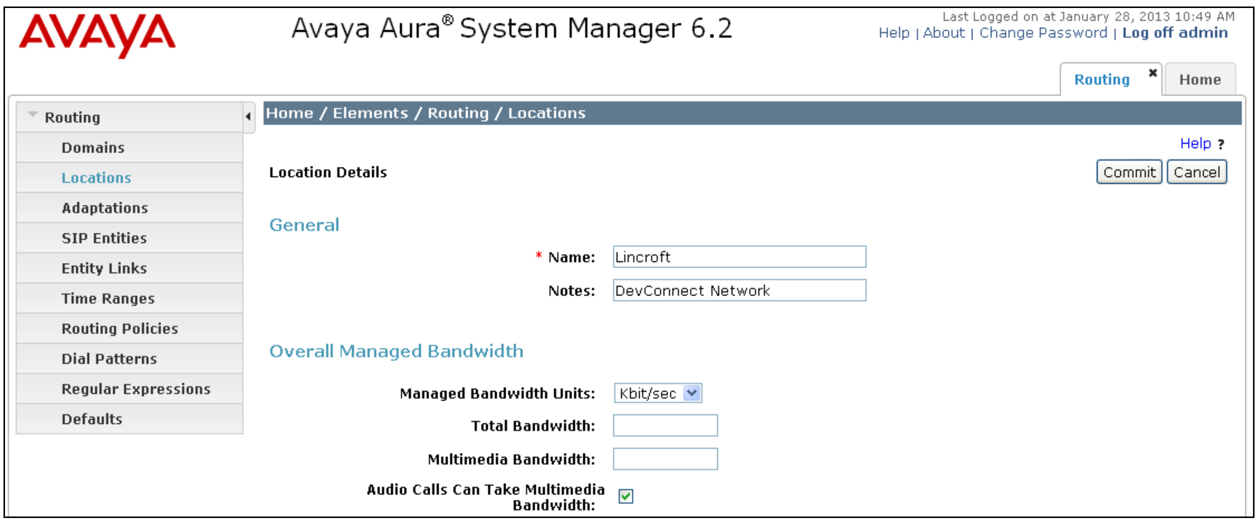
6.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

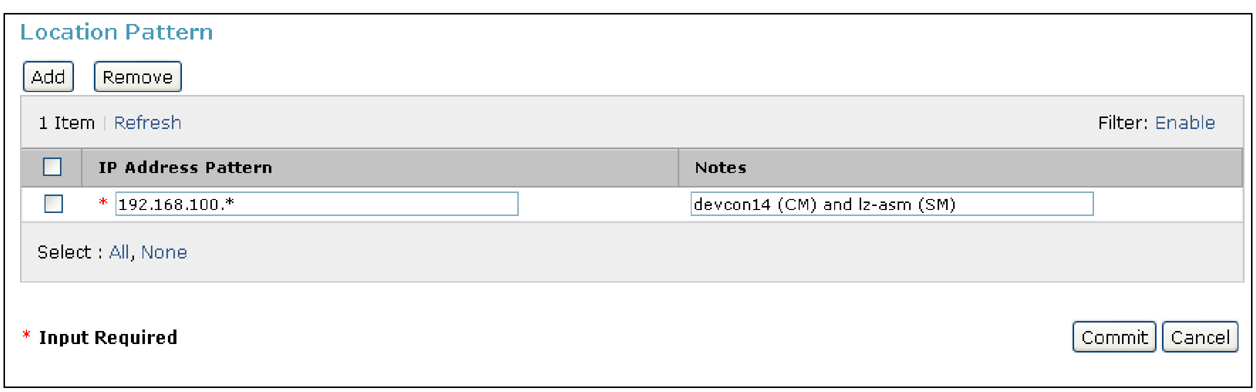
The screen below shows addition of the *Lincroft* location, which includes the Communication Manager and Session Manager.



Under *Location Pattern*:

- **IP Address Pattern:** A pattern used to logically identify the location.
- **Notes:** Descriptive text (optional).

Click **Commit** to save the **Location** definition.



6.3. Add SIP Entities

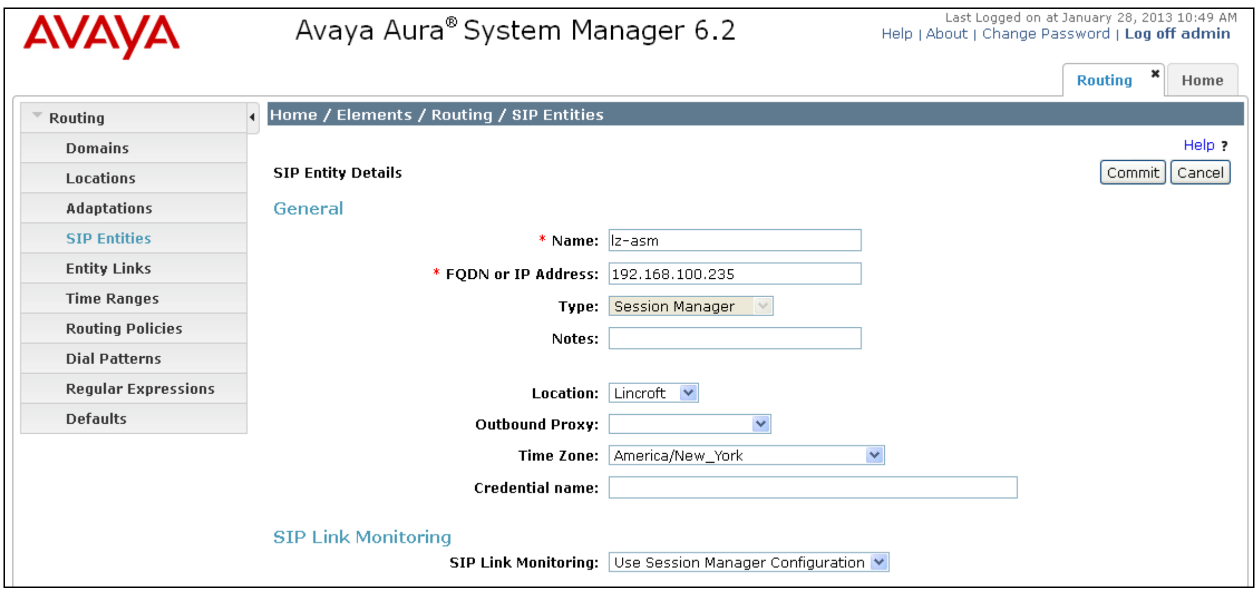
In the sample configuration, a SIP Entity is added for Session Manager and the S8300 Server in the G450 Media Gateway.

6.3.1. Avaya Aura® Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** 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*:

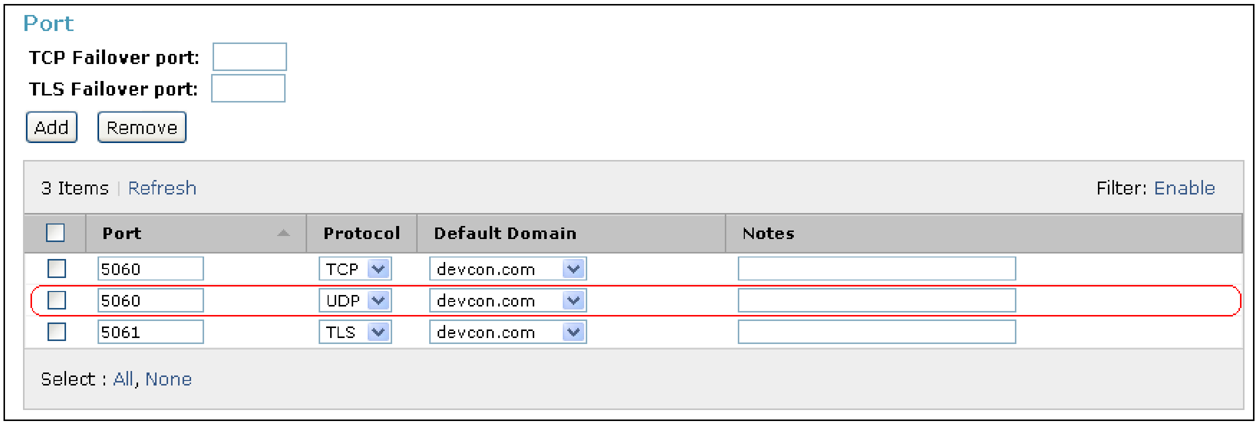
- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.



Under *Port*, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain** The domain used for the enterprise (e.g., *devcon.com*).

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



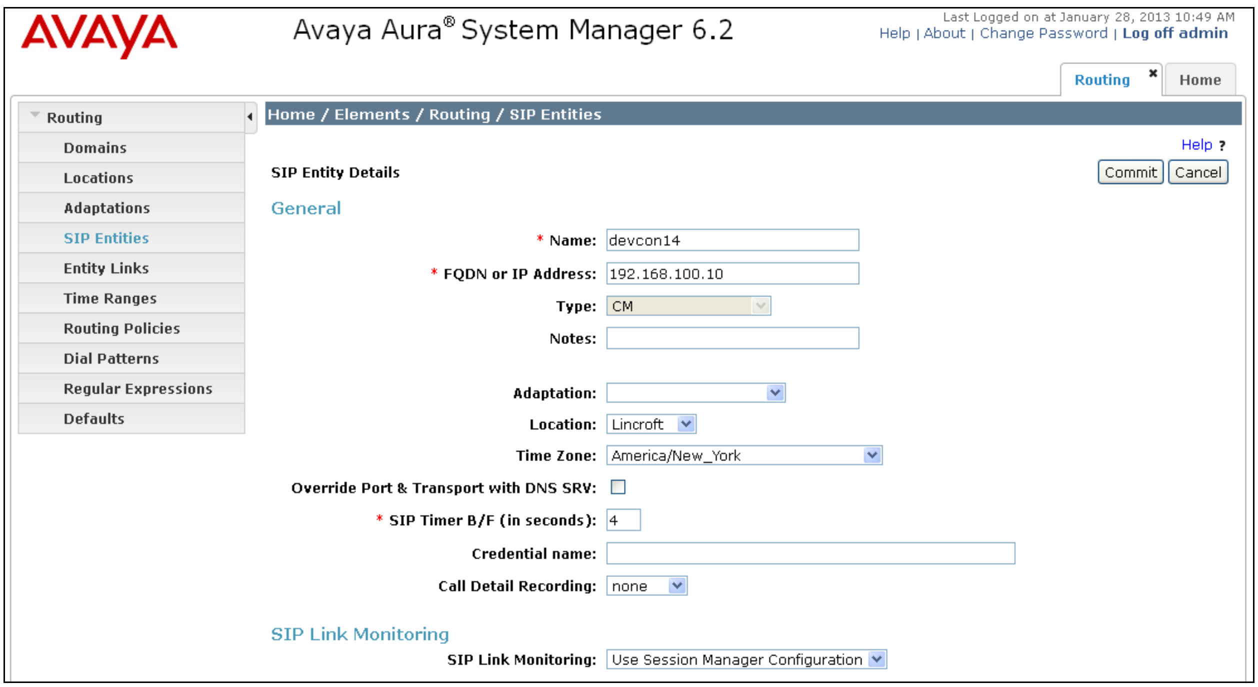
6.3.2. Avaya Aura® Communication Manager

A SIP Entity must be added for the Communication Manager. To add a SIP Entity, select **SIP Entities** 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*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., S8300 Server) on the telephony system.
- **Type:** Select *CM*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

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

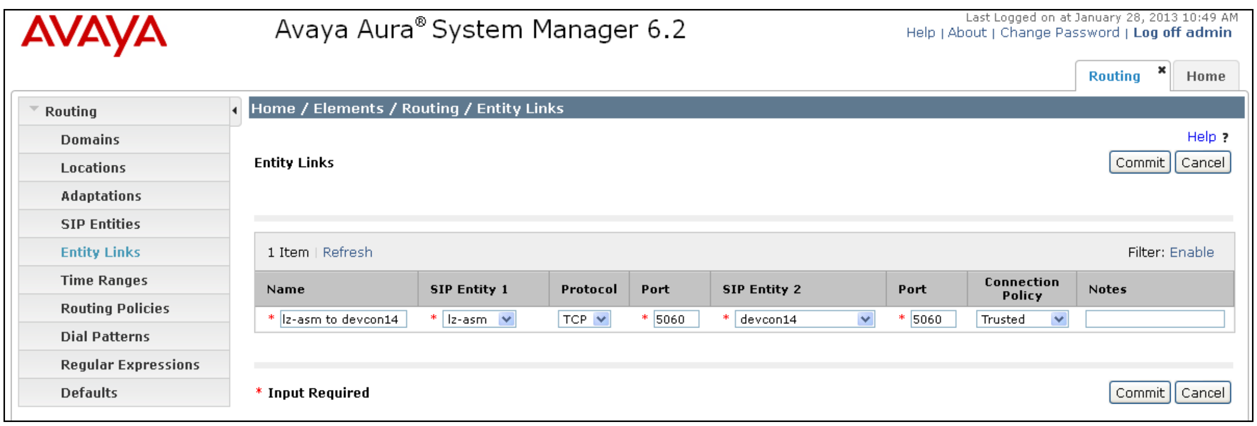


6.4. Add Entity Link

The SIP trunk from Session Manager to Communication Manager 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., *lz-asm to devcon14*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager.
- **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 6.3.2*** *will be denied.*

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



6.5. Define Communication Manager as Managed Element

Before adding SIP users, Communication Manager must be added to System Manager as a managed element. This action allows System Manager to access Communication Manager over its administration interface. Using this administration interface, System Manager will notify Communication Manager when new SIP users are added.

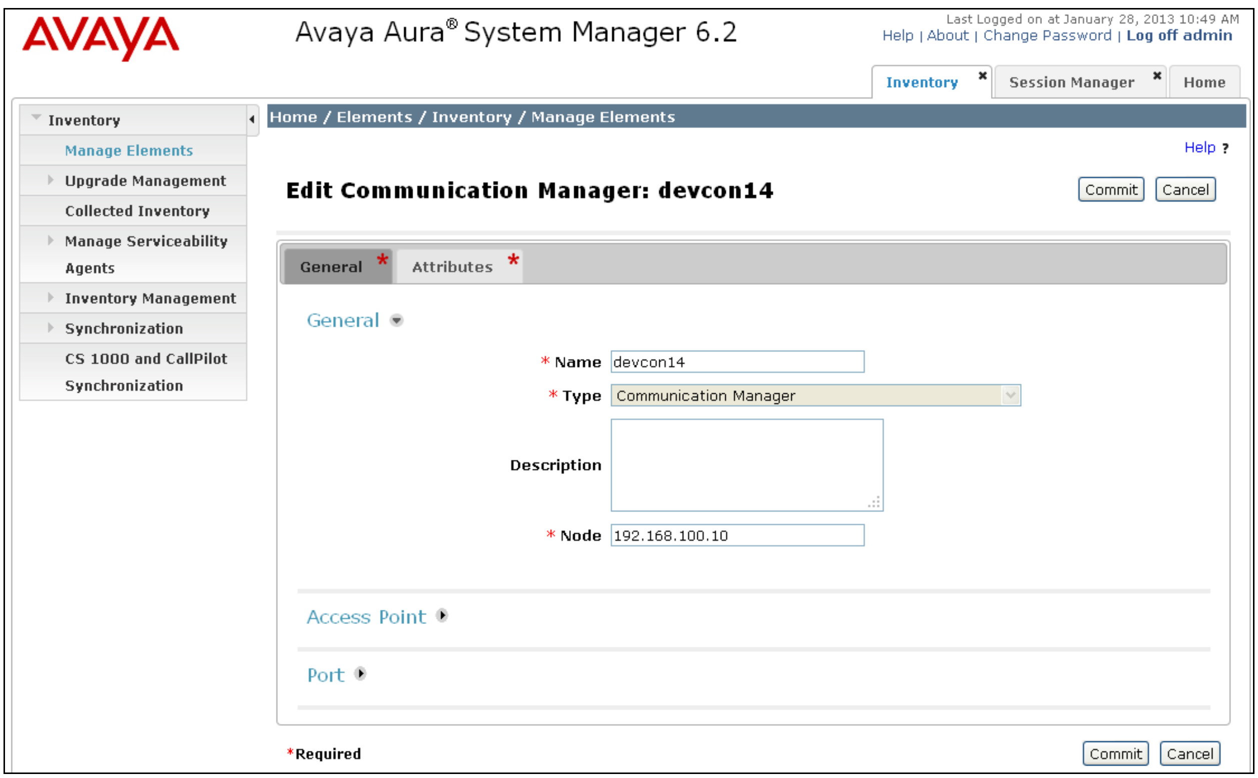
To define Communication Manager as a managed element, select **Elements→Inventory→Manage Elements** on the left and click on the **New** button (not shown) on the right. In the **Type** field that is displayed, select *Communication Manager* from the drop-down menu.

In the **New Communication Manager** screen, fill in the following fields as follows:

Under *General*:

- **Name:** Enter an identifier for Communication Manager.
- **Node:** Enter the IP address of the administration interface for Communication Manager.

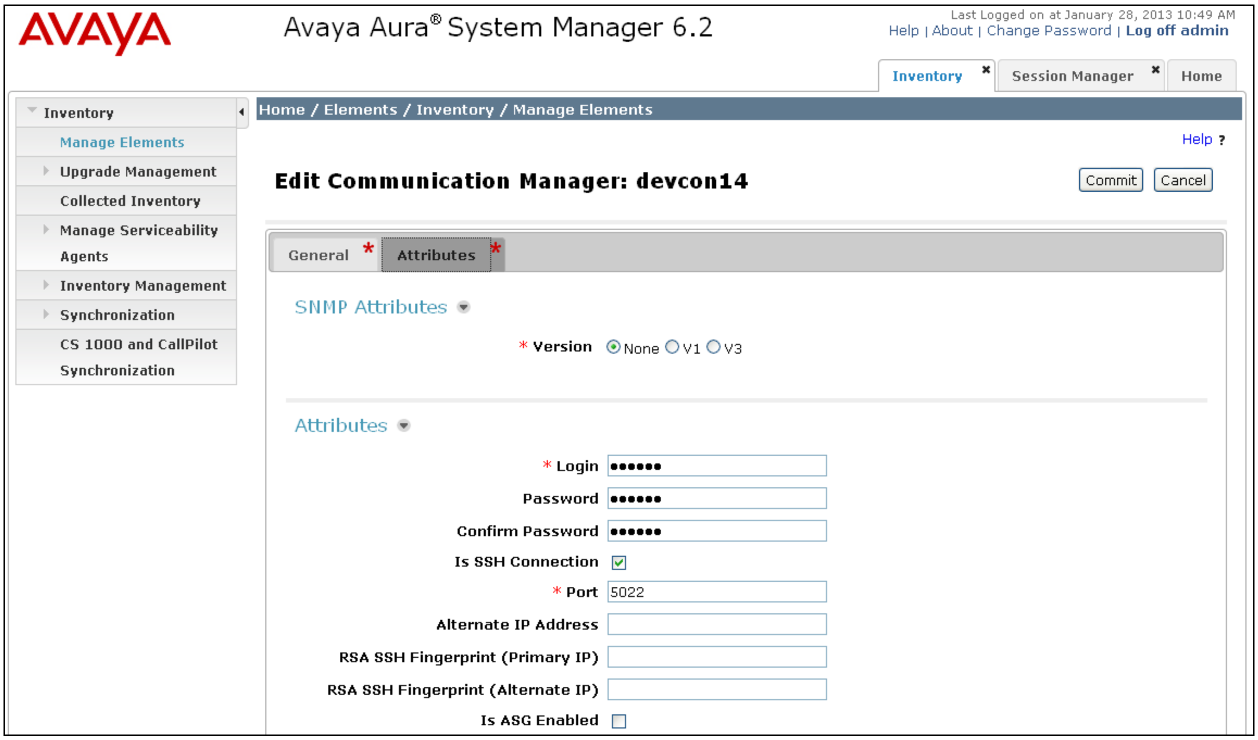
Defaults can be used for the remaining fields.



Under *Attributes*:

- **Login / Password:** Enter the login and password used for administration access.
- **Is SSH Connection:** Enable SSH access.
- **Port:** Enter the port number for SSH administration access (5022).

Click **Commit** to save the settings.

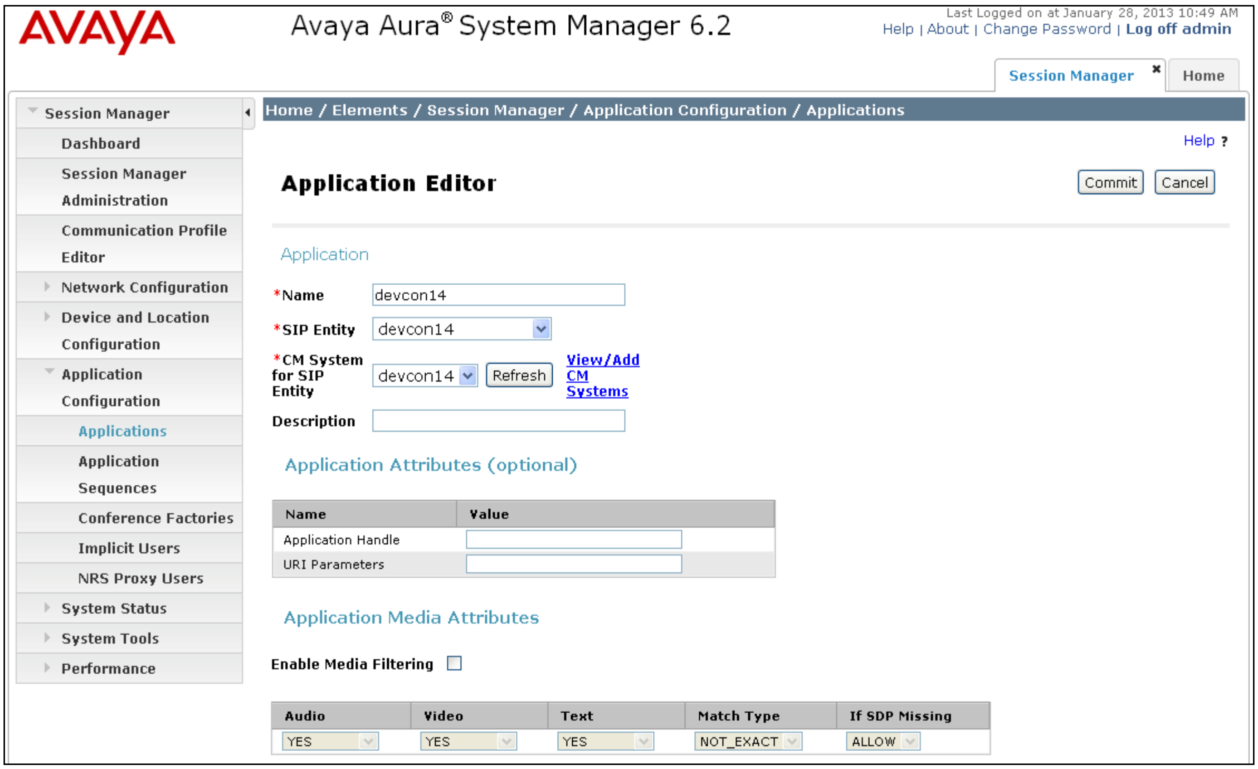


6.6. Add Application Sequence

To define an application for Communication Manager, navigate to **Elements → Session Manager → Application Configuration → Applications** on the left and select **New** button (not shown) on the right. Fill in the following fields:

- **Name:** Enter name for application.
- **SIP Entity:** Select the Communication Manager SIP entity.
- **CM System for SIP Entity** Select the Communication Manager managed element.

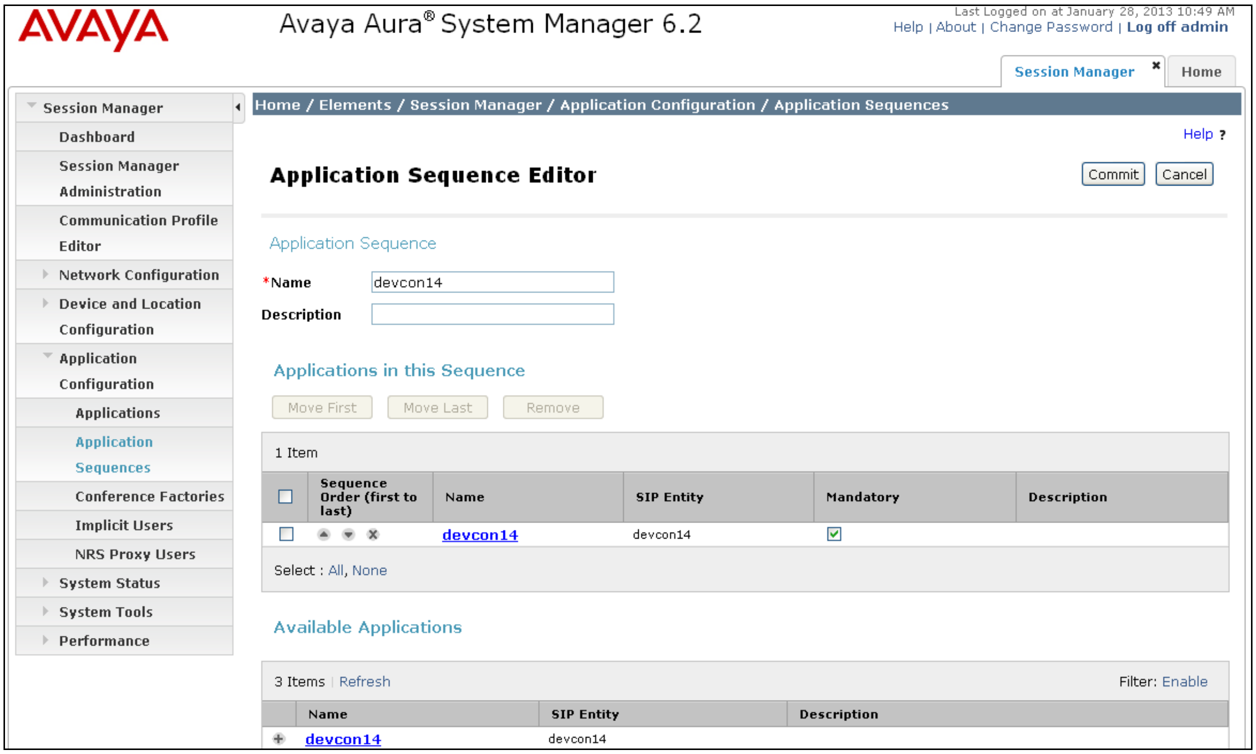
Click **Commit** to save the Application definition.



Next, define the Application Sequence for Communication Manager by selecting **Application Sequences**.

Verify a new entry is added to the **Applications in this Sequence** table and the **Mandatory** column is ☑ as shown below.

Note: The Application Sequence defined for Communication Manager Evolution Server can only contain a single Application.



6.7. Add SIP Users

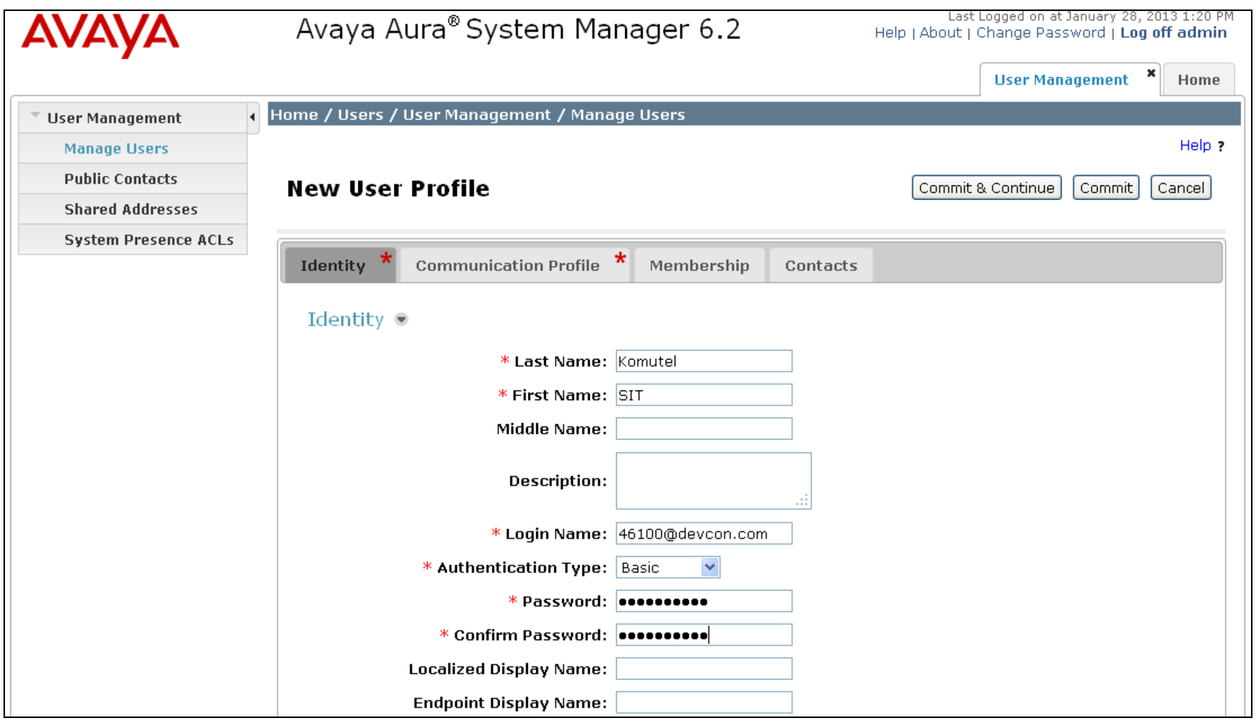
Add a SIP user for the SIT PC Attendant Console as defined in **Section 5.3.** Alternatively, use the option to automatically generate the SIP stations on Communication Manager Evolution Server when adding a new SIP user. For this compliance test, the SIP user was created through System Manager as shown in this section.

To add new SIP users, expand **Users** and select **Manage Users** from left and select **New** button (not shown) on the right.

Enter values for the following required attributes for a new SIP user in the **Identity** section of the new user form.

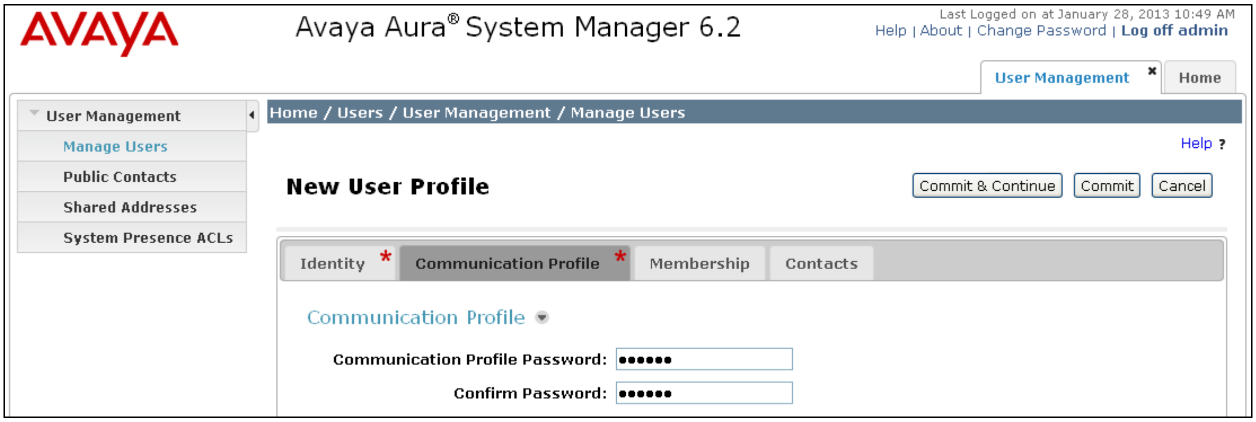
- **Last Name:** Enter the last name of the user.
- **First Name:** Enter the first name of the user.
- **Login Name:** Enter *<extension>@<sip domain>* of the user (e.g., *46100@devcon.com*).
- **Authentication Type:** Select *Basic*.
- **Password:** Enter the password which will be used to log into System Manager
- **Confirm Password:** Re-enter the password from above.

The screen below shows the information when adding a new SIP user to the sample configuration.



Select the **Communication Profile** tab and configure the following fields:

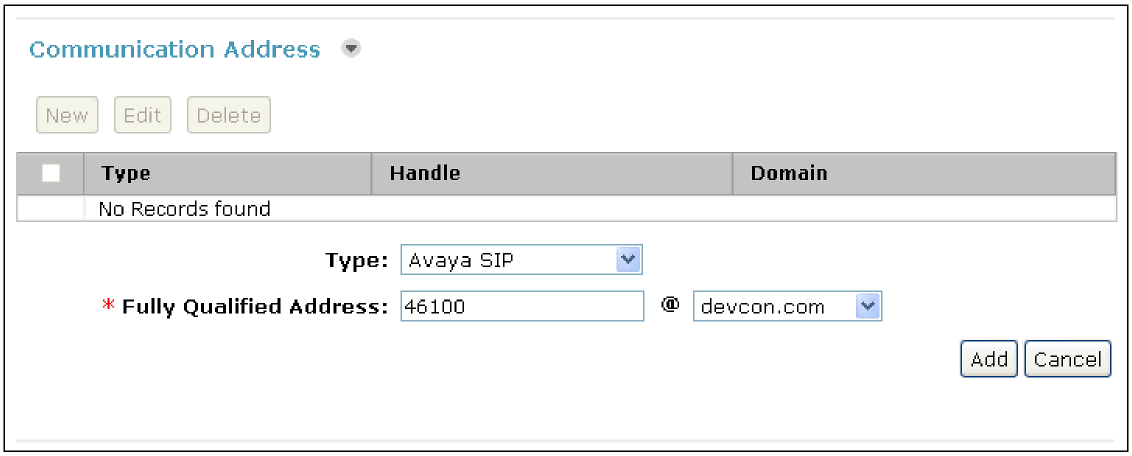
- **Communication Profile Password:** Enter the password which will by the Komutel SIT to log into Session Manager.
- **Confirm Password:** Re-enter the password from above.



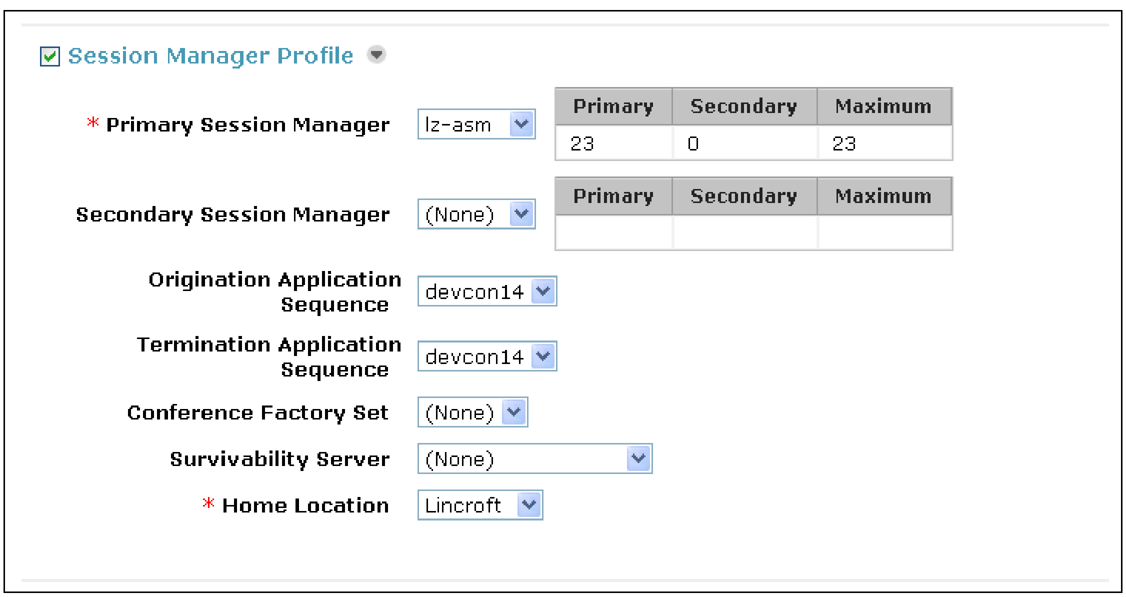
Click **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** Select *Avaya SIP*.
- **Fully Qualified Address:** Enter extension number and select SIP domain.

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Add**.



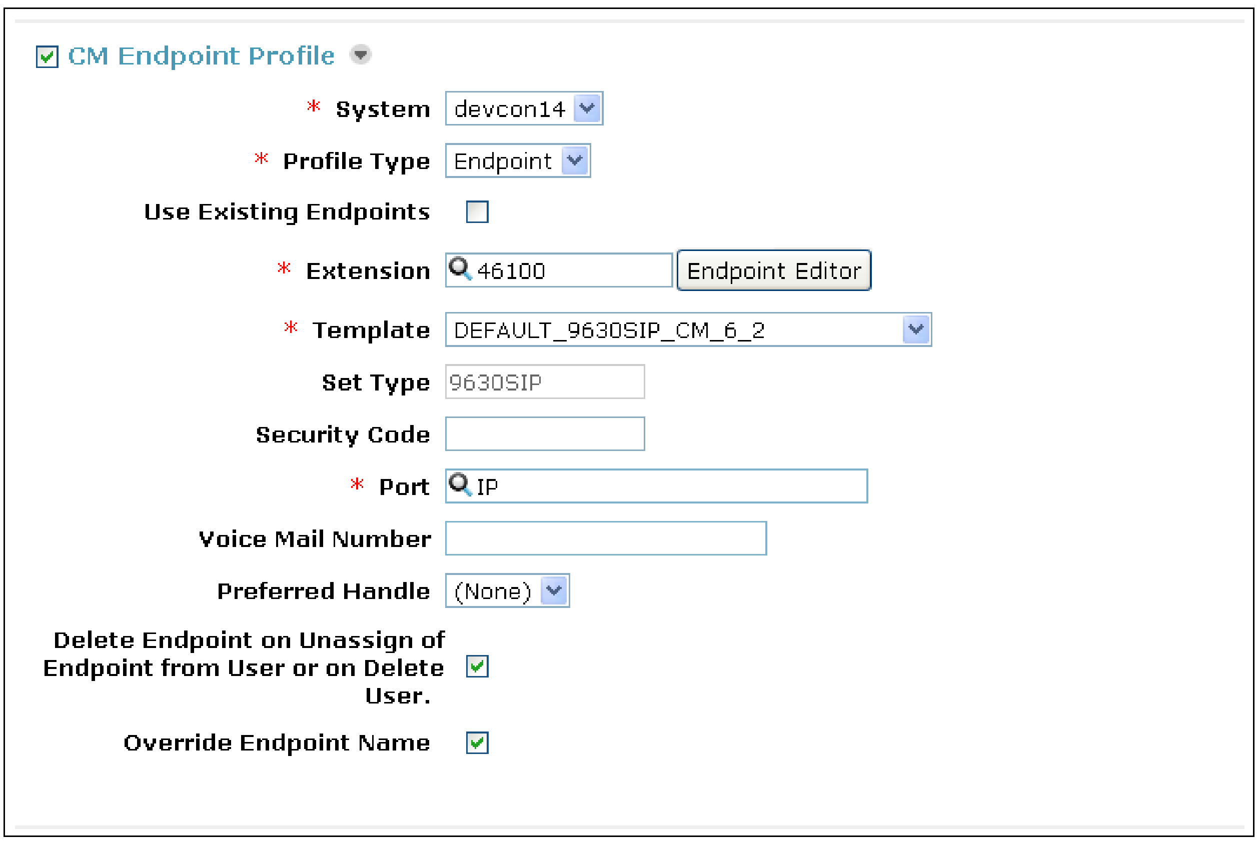
In the *Session Manager Profile* section, specify the Session Manager entity from **Section 6.3.1** for **Primary Session Manager** and assign the **Application Sequence** defined in **Section 6.6** to both the originating and terminating sequence fields. Set the **Home Location** field to the **Location** configured in **Section 6.2**.



In the **CM Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager.
- **Profile Type:** Select *Endpoint*.
- **Use Existing Stations:** If field is not selected, the station will automatically be added in Communication Manager.
- **Extension:** Enter extension number of SIP user.
- **Template:** Select template for type of SIP phone.
- **Port:** Enter *IP*.
- **Delete Ednpoint on Unassign of Endpoint From User or on Delete User:** Enable field to automatically delete station when **Station Profile** is un-assigned from user.

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Commit** to add the SIP user.



6.8. Add Session Manager

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

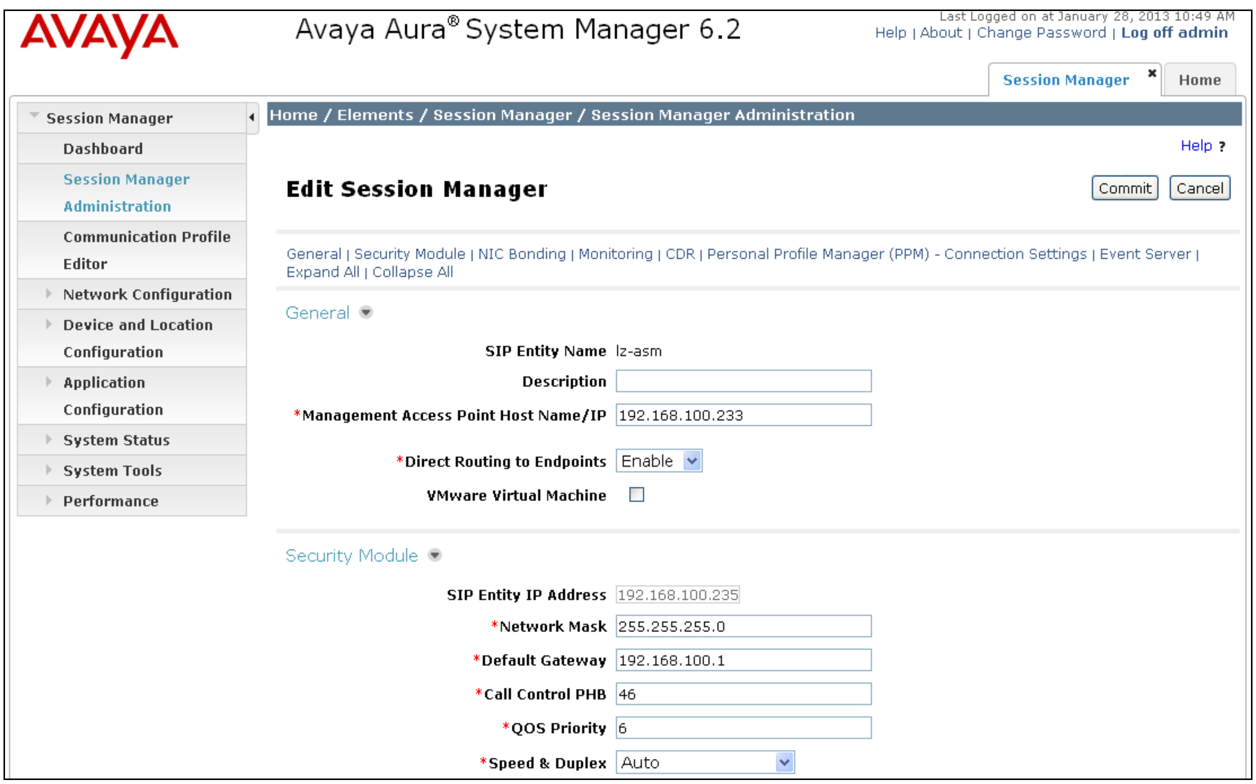
Under *General*:

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

Under *Security Module*:

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

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



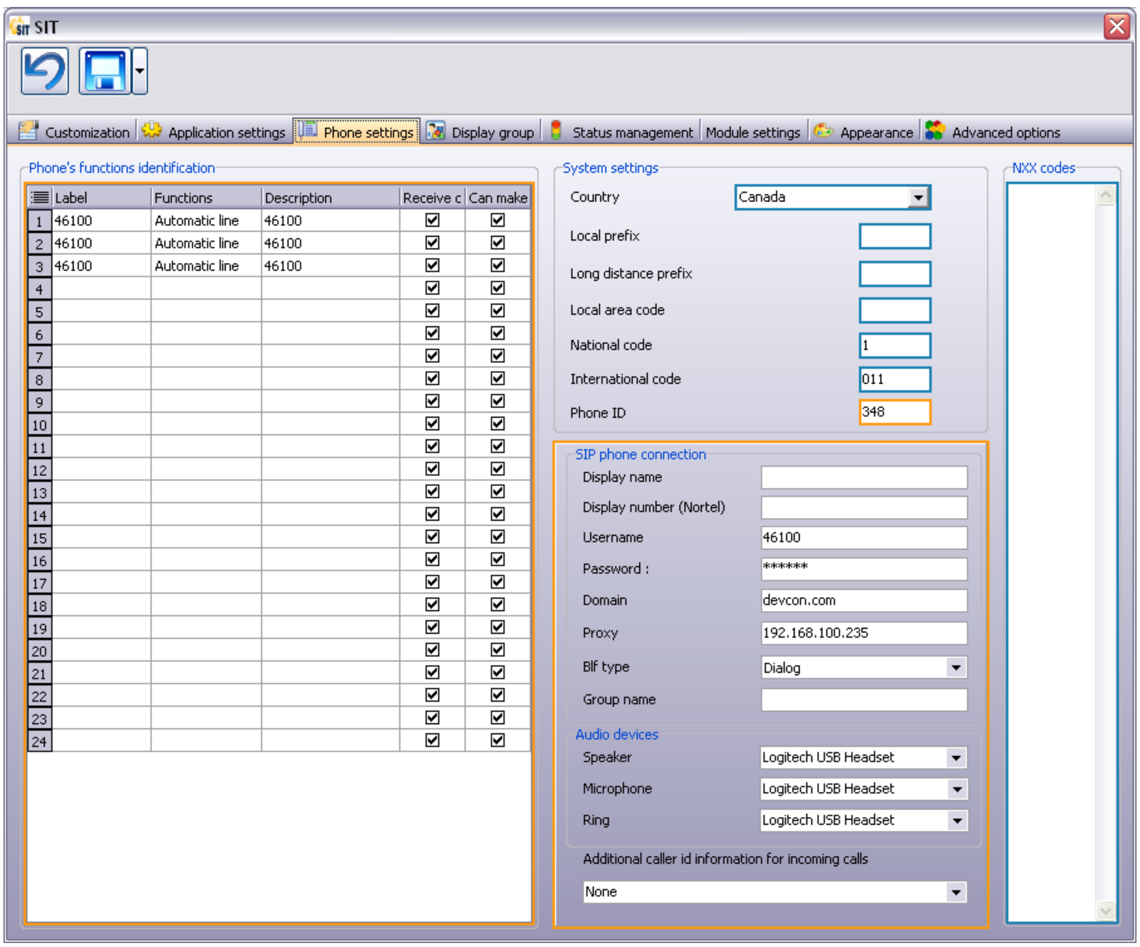
7. Configure Komutel SIT PC Attendant Console

Launch the SIT application and login in with the appropriate credentials. To configure the console's lines, navigate to the **Tools** menu option(not shown), and then select the **Phone Settings** tab. Depending on the number of lines that are available, choose *Automatic Line* in the **Functions** column, enter the **DN** in the **Description** column and type the text that will be displayed on the line's button in the **Label** column. As shown below, the console was configured with three line appearances with extension 46100.

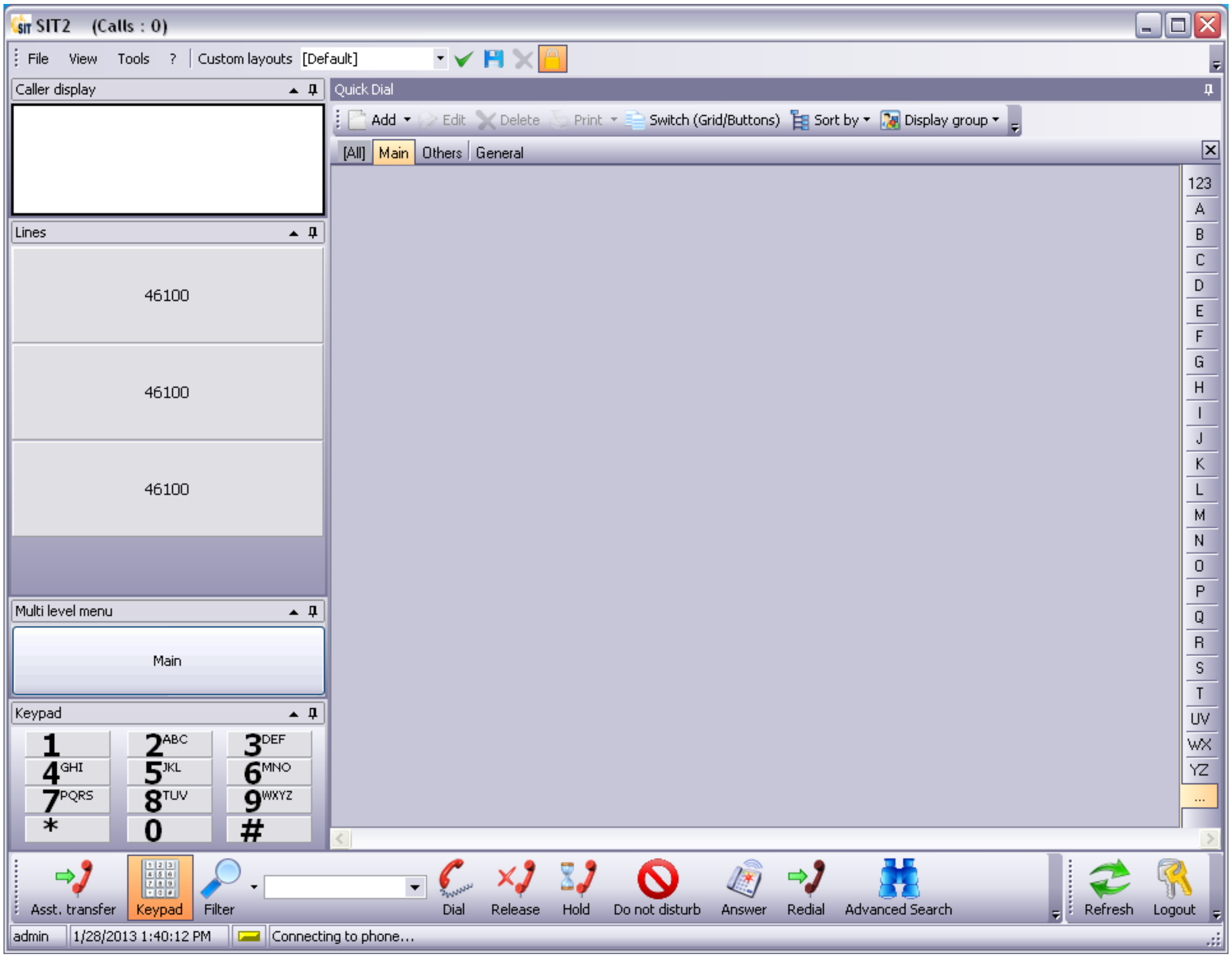
In the **SIP phone connection** section, configure the SIP parameters, including:

- The **Username** and **Password** used to register with Session Manager.
- The SIP **Domain** (e.g., *devcon.com*).
- The **Proxy**, which specifies the IP address of the SIP interface of Session Manager.

In the **Audio devices** section, specify the audio device or headset that will be used with the console. Click the **Save** button at the top of screen.



After the configuration is completed, the SIT PC Attendant Console appears as follows.

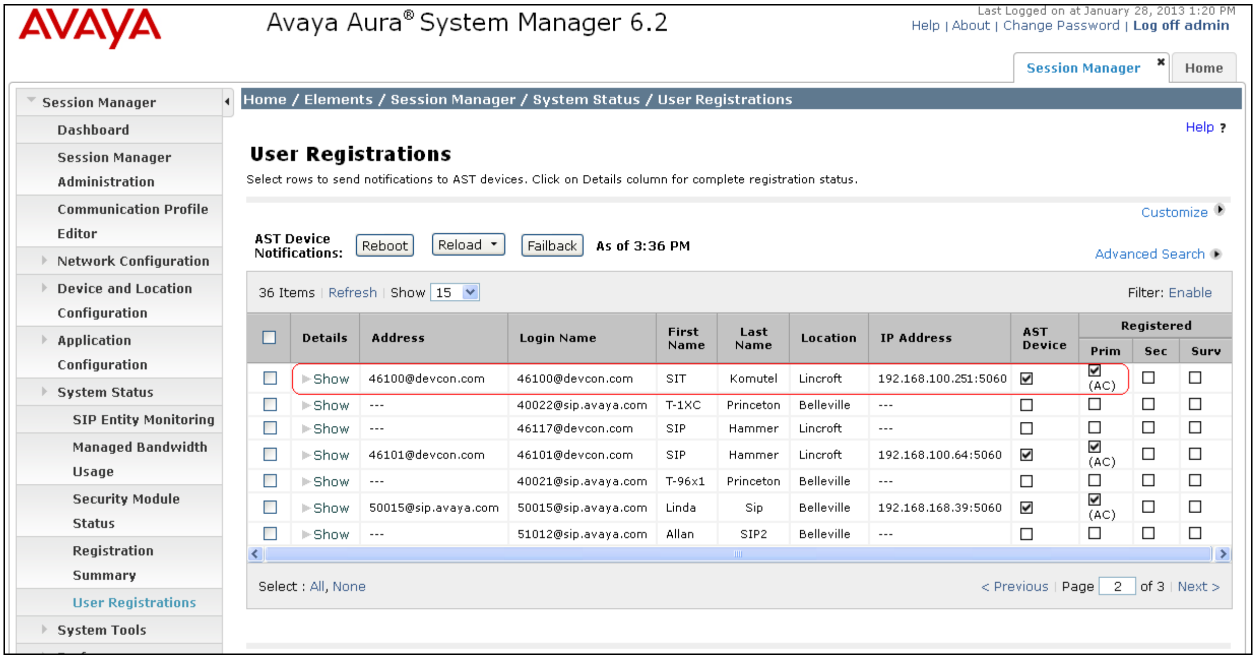


Note: Although the Komutel SIT installation is outside the scope of these Application Notes, note that during the installation, the options corresponding to *Communication Manager* and *Install SITSIP (softphone)* must be selected. In addition, make sure that the appropriate sound files for various ring notifications and alerts have been copied to the directory where the SIT executable resides.

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Komutel SIT PC Attendant Console with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the SIT PC Attendant Console has successfully registered with Session Manager.



2. Verify basic telephony features by establishing calls between an SIT PC Attendant Console and another phone.

9. Conclusion

These Application Notes have described the administration steps required to integrate the Komutel SIT PC Attendant Console with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The SIT PC Attendant Console successfully registered with Session Manager and basic telephony features were verified. All test cases passed with observations noted in **Section 2.2**.

10. References

This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at http://support.avaya.com.

[1] *Administering Avaya Aura® Communication Manager*, Release 6.2, Issue 7.0, December 2012, Document Number 03-300509.

[2] *Administering Avaya Aura® Session Manager*, July 2012, Issue 3, Release 6.2, Document Number 03-603324.

[3] *Komutel User Guide for SIT PC Attendant Console*, Revised 2012-07-16.

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.