Avaya Solution Interoperability Test Lab

Configuring Avaya 10x0 Series SIP Video Endpoints with Avaya Aura® Session Manager Release 6.1 and Avaya Aura® Communication Manager Evolution Server Release 6.0.1 – Issue 1.0

Abstract

These Application Notes describe the configuration of the Avaya 10x0 Series SIP Video Endpoints with Avaya Aura® Session Manager and Avaya Aura® Communication Manager as an Evolution Server.

- Avaya Aura® Session Manager provides SIP proxy/routing functionality, routing SIP sessions across a TCP/IP network with centralized routing policies and registrations for SIP endpoints.
- Avaya Aura® Communication Manager operates as an Evolution Server for the SIP endpoints which communicate with Avaya Aura® Session Manager over SIP trunks.

These Application Notes provide information for the setup, configuration, and verification of the call flows tested on this solution.

Table of Contents:

1. Introduction

These Application Notes present a sample configuration for a network that uses Avaya Aura® Session Manager to support registration of Avaya 10x0 (1010, 1020, 1030, 1040, and 1050) SIP Video endpoints and enables connectivity to Avaya Aura® Communication Manager Evolution Server 6.0.1 using SIP trunks.

As shown in **Figure 1,** Avaya Aura® Session Manager is managed by Avaya Aura® System Manager. Avaya 10x0 Video Endpoints configured as SIP endpoints utilize the Avaya Aura® Session Manager User Registration feature and Avaya Aura® Communication Manager operating as an Evolution Server. Communication Manager Evolution Server is connected to Session Manager via a SIP signaling group and associated SIP trunk group.

For the sample configuration, Avaya Aura® Session Manager runs on an Avaya S8510 Server. Avaya Aura® Communication Manager 6.0.1 Evolution Server runs on a S8800 server with an Avaya 450 Gateway. The results in these Application Notes should be applicable to other Avaya servers and media gateways that support Avaya Aura® Communication Manager 6.0.1.

These Application Notes will focus on the configuration of Communication Manager Evolution Server and Session Manager. Detailed administration of Communication Manager Feature Server will not be described (see the appropriate documentation listed in **Section 8**).

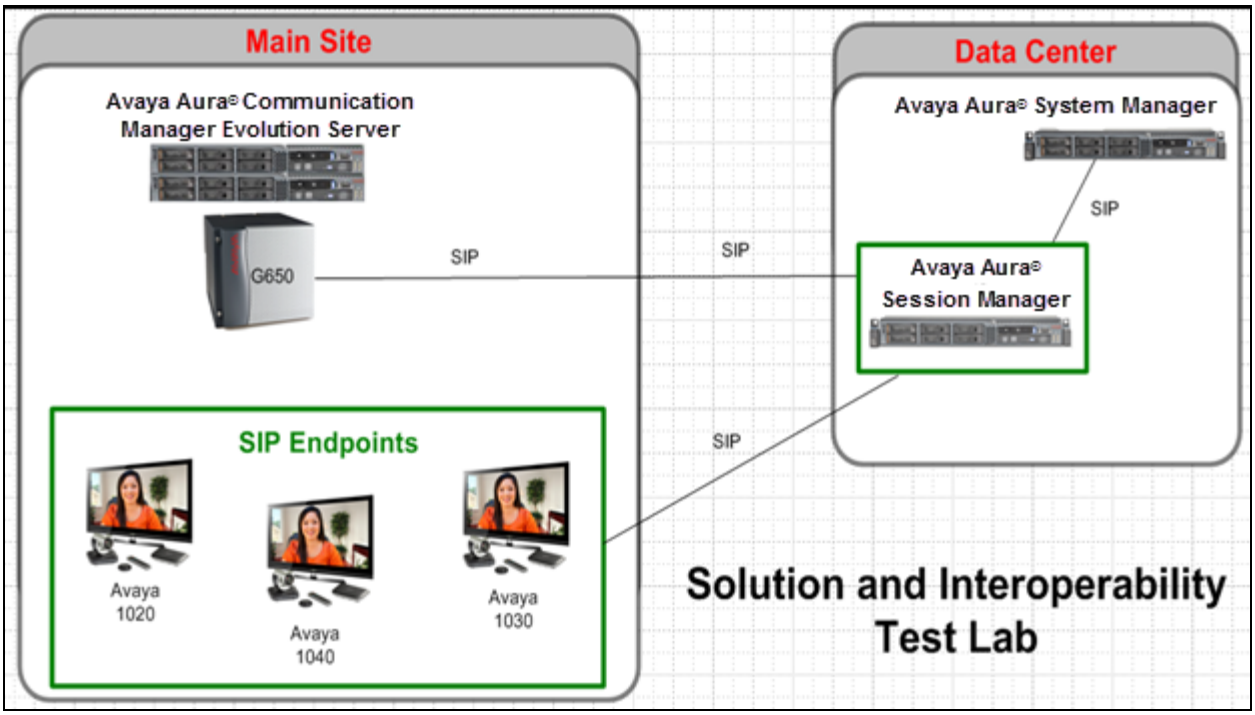


Figure 1 – Sample Configuration

1.1. Equipment and Software Validated

The following equipment and software were used for the sample configuration.

Equipment	Software
Avaya Aura® Session Manager	Release 6.1.0.0.610023
Avaya Aura® System Manager	Release 6.1 Load: 6.0.2.0.5
Avaya Aura® Communication Manager • Avaya S8800 Server Evolution Server	Release R016x.00.1.510.1-18599
Avaya IP Telephones10x0 Video Endpoints (SIP): • 1020 • 1030 • 1040	 FW: AV_PP1_4.7.3(14) FW: AV_XX2_4.7.3(14) FW: AV_XX2_4.7.3(14)

2. Configuring Avaya Aura® Communication Manager Evolution Server

This section describes the administration of Communication Manager Evolution Server using a System Access Terminal (SAT). Alternatively, some of the station administration could be performed using the Communication System Management application on System Manager. These instructions assume the G450 Media Gateway is already configured on Communication Manager Evolution Server. Some administration screens have been abbreviated for clarity.

- Verify System Capabilities and Communication Manager Licensing
- Administer IP node names
- Administer codec type
- Administer IP network region
- Administer SIP signaling group
- Administer SIP trunk group
- Administer numbering plan
- Administer station endpoints
- Administer off-pbx-telephone station-mapping
- Save translations

After completing these steps, the **save translation** command should be performed.

2.1. Verify System Capabilities and Licensing

This section describes the procedures to verify the correct system capabilities and licensing have been configured. If there is insufficient capacity or a required feature is not available, contact an authorized Avaya sales representative to make the appropriate changes.

2.1.1. SIP Trunk Capacity Check

Issue the **display system-parameters customer-options** command to verify that an adequate number of SIP trunk members are licensed for the system as shown below:

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

IP PORT CAPACITIES                                               USED
                     Maximum Administered H.323 Trunks: 12000 328
           Maximum Concurrently Registered IP Stations: 18000 17
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                       Maximum Video Capable Stations: 18000 12
                   Maximum Video Capable IP Softphones: 18000 109
                       Maximum Administered SIP Trunks: 24000 15493
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 80
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   1
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   15
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

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

2.1.2. AAR/ARS Routing Check

Verify that **ARS** is enabled (on **Page 3** of **system-parameters customer options** form

```
display system-parameters customer-options                    Page   3 of  11
                                OPTIONAL FEATURES

 A/D Grp/Sys List Dialing Start at 01? y                          CAS Main? y
Answer Supervision by Call Classifier? y                  Change COR by FAC? y
                                      ARS? y  Computer Telephony Adjunct Links? y
                  ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? y                        DCS (Basic)? y
          ASAI Link Core Capabilities? y                  DCS Call Coverage?
```

2.2. Add Node Name of Avaya Aura® Session Manager

Using the **change node-names ip** command, add the node-name and IP for the Session Manager's software asset, if not previously added.

```
change node-names ip                                          Page   1 of   2
                                   IP NODE NAMES
    Name                IP Address
default               0.0.0.0
procr                 135.9.88.13
procr6                ::
silasm4               135.9.88.62
```

2.3. Configure Codec Type

Issue the **change ip-codec-set n** command where "n" is the next available number. Enter the following values:

- Enter "**G.711MU**" and **"G.729"** as supported types of **Audio Codecs**
- **Silence Suppression:** Retain the default value "**n**".
- **Frames Per Pkt**: Enter "**2**".
- **Packet Size (ms):** Enter "**20**".

```
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: G.729             n         2        20
 3:
```

2.4. Configure IP Network Region

Using the **change ip-network-region 1** command set the **Authoritative Domain**. For the sample configuration "dr.avaya.com" was used. Verify the **Intra-region IP-IP Direct Audio**, and **Inter-region IP-IP Direct Audio** fields are set to **yes**.

```
change ip-network-region 1                                   Page   1 of  20
                                IP NETWORK REGION
  Region: 1
Location: 1        Authoritative Domain: dr.avaya.com
    Name: CMES-Video
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: 16585
```

2.5. Add SIP Signaling Group

Issue the **add signaling-group n** command, where "n" is an available signaling group number, for one of the SIP trunks to the Session Manager, and fill in the indicated fields. In the sample configuration, trunk group "1" and signaling group "1" were used to connect to Avaya Aura® Session Manager.

- **Group Type:** "sip"
- **Transport Method:** "tcp"
- **IP Video?**: "y"
- **Peer Detection Enabled?**: "y"
- **Peer Server:** Use default value. **Note:** default value is replaced with "**SM**" after SIP trunk to Session Manager is established
- **Near-end Node Name:** procr from **Section 2.2**
- **Far-end Node Name:** Session Manager node name from **Section 2.2**
- **Near-end Listen Port:** "5060"
- **Far-end Listen Port:** "5060"
- **Far-end Domain:** Authoritative Domain from **Section 2.4**
- **Enable Layer 3 Test:** "y"
- **Initial IP-IP Direct Media?:** "y"

```
display signaling-group 10                                      Page   1 of   1
                                SIGNALING GROUP

Group Number: 10              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                          SIP Enabled LSP? n
     IP Video? y          Priority Video? n      Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM


    Near-end Node Name: procr                Far-end Node Name: silasm4
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region:  2
Far-end Domain: dr.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
```

2.6. Add SIP Trunk Group

Add the corresponding trunk group controlled by this signaling group via the **add trunk-group n** command, where "n" is an available trunk group number and fill in the indicated fields.

- **Group Type:** "sip"
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** "tie"
- **Signaling Group:** The number of the signaling group added in **Section 2.5**
- **Number of Members:** The number of SIP trunks to be allocated to calls routed to Session Manager (must be within the limits of the total number of trunks configured in **Section 2.1.1**).

```
add trunk-group 10                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 10                   Group Type: sip          CDR Reports: y
  Group Name: SIP Video TG to silasm4     COR: 1        TN: 1       TAC: #010
Direction: two-way        Outgoing Display? y
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                      Signaling Group: 10
                                                    Number of Members: 64
```

Once the add command is completed, trunk members will be automatically generated based on the value in the **Number of Members** field.

On **Page 2**, set the **Preferred Minimum Session Refresh Interval** to 1200. **Note**: to avoid extra SIP messages, all SIP trunks connected to Session Manager should be configured with a minimum value of 1200.

```
add trunk-group 10                                          Page   2 of  21
                                  Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                          Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                     Preferred Minimum Session Refresh Interval(sec): 1200
```

2.7. Administering Numbering Plan

SIP Users registered to Session Manager needs to be added to either the private or public numbering table on Communication Manager Evolution Server. For the sample configuration, public numbering was used and all extension numbers were unique within the public network. However, in many customer networks, it may not be possible to define unique extension numbers for all users within the private network. For these types of networks, additional administration may be required as described in References [3] and [8] in **Section 8**.

To enable SIP endpoints to dial extensions defined in Communication Manager Evolution Server, use the **change public-unknown-numbering x** command, where "x" is the number used to identify the private number plan. For the sample configuration, extension numbers starting with 5-XXXX are used on Communication Manager Evolution Server.

- **Ext Len:** Enter the extension length allowed by the dial plan
- **Ext Code:** Enter leading digit (s) from extension number
- **Trunk Grp(s):** Enter the SIP Trunk Group number for the SIP trunk between the Evolution Server and Session Manager
- **CPN Prefix:** Leave blank unless an enterprise canonical numbering scheme is defined in Session Manager. If so, enter the appropriate prefix.
- **Total CPN Len:** Enter the total CPN length.

```
change public-unknown-numbering 5                                Page   1 of  2
                        NUMBERING – PUBLIC/UNKNOWN FORMAT
                                                  Total
Ext Ext            Trk        CPN                 CPN
Len Code           Grp(s)     Prefix              Len
 5  5              10                             5     Total Administered: 1
                                                            Maximum Entries: 9999
```

2.8. Configure Stations

The method is the same for administering all of the Avaya 1000 series video endpoints with the exception of the 1040 and 1050's. The only difference is that the 1040 can be administered to have up to 3 call appearances and the 1050 can have up to 7 call appearances for conferencing via their internal MCU's. The 1010, 1020, and 1030 have to be administered with only one call-appearance since they are a single-line endpoint with no conferencing or transferring capabilities.

For each SIP user to be defined in Session Manager, add a corresponding station on Communication Manager Evolution Server. **Note**: instead of manually defining each station using the Communication Manager SAT interface, the preferred option is to automatically generate the SIP station when adding a new SIP user. See **Section 3.3.7** for more information on adding SIP users.

The phone number defined for the station will be the number the SIP user enters to register to Session Manager. Use the **add station x** command where "x" is a valid extension number defined in the system. In this example extension 50095 is an Avaya 1020 video endpoint. On **Page 1** of the **add station** form:

- **Phone Type:** Set to 9630SIP
- **Name:** Display name for user
- **Security Code:** Number used when user logs into station. **Note:** this code should match the "**Shared Communication Profile Password**" field defined when adding this user in Session Manager. See **Section 3.3.7**.
- **IP Video?** Enable endpoint for video

```
add station 50095                                        Page   1 of  6

                              STATION

Extension: 50095                        Lock Messages? n             BCC: 0
     Type: 9630SIP                      Security Code: 123456         TN: 1
     Port: S00006                     Coverage Path 1: 1             COR: 1
     Name: SIL Video Lab - 1030       Coverage Path 2:               COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19
                                                Message Lamp Ext: 50095
        Display Language: english                 Button Modules: 0
          Survivable COR: internal
   Survivable Trunk Dest? y                         IP SoftPhone? n

                                                       IP Video? y
```

Note: It is important to assign only one call-appearance for the 1010, 1020, and 1030's.

```
add station 50095                                        Page   4 of  6

                              STATION
SITE DATA
       Room:                                                Headset? n
       Jack:                                                Speaker? n
      Cable:                                               Mounting: d
      Floor:                                           Cord Length: 0
   Building:                                             Set Color:

ABBREVIATED DIALING
     List1:                     List2:                     List3:




BUTTON ASSIGNMENTS
 1: call-appr                             5:
 2:                                       6:
 3:                                       7:
 4:                                       8:
```

On **Page 6**, set:

- **SIP Trunk option:** Enter SIP Trunk Group defined in **Section 2.6**

```
change station 50095                                         Page   6 of  6

                              STATION
SIP FEATURE OPTIONS
         Type of 3PCC Enabled: None
                    SIP Trunk: 10
```

2.9. Configure Off-PBX-Telephone Station-Mapping

Use the **change off-pbx-telephone station-mapping** command for each extension associated with SIP users defined in Session Manager. On **Page 1**, enter the SIP Trunk Group defined in **Section 2.6** and use default values for other fields.

```
change off-pbx-telephone station-mapping 50095                Page   1 of  3

                STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk      Config  Dual
 Extension                 Prefix                    Selection  Set     Mode
 50095            OPS          -     50095            10        1
                               -
                               -
```

On **Page 2**, enter the following values:

- **Mapping Mode**: "both"
- **Calls Allowed:** "all"

```
change off-pbx-telephone station-mapping 50095                Page   2 of  3

                STATIONS WITH OFF-PBX TELEPHONE INTEGRATION


 Station       Appl    Call        Mapping    Calls      Bridged     Location
 Extension     Name    Limit       Mode       Allowed    Calls
 50095         OPS      1          both       all         none
                               -
```

2.10. Save Translations

Configuration of Communication Manager Evolution Server is complete. Use the **save translations** command to save these changes

Note: After a change on Communication Manager Evolution Server which alters the dial plan, synchronization between Communication Manager Evolution Server and Session Manager needs to be completed and SIP phones must be re-registered. To request an on demand synchronization, log into the System Manager console and use the **Synchronize CM Data** feature under the Communication System Management menu.

3. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring the Session Manager and includes the following items:

- Administer SIP domain
- Define Logical/Physical Locations that can be occupied by SIP Entities

- For each SIP entity in the sample configuration:
    - Define SIP Entity
    - Define Entity Links, which define the SIP trunk parameters used by Avaya Aura® Session Manager when routing calls to/from SIP Entities
    - Define Routing Policies, which control call routing between the SIP Entities
    - Define Dial Patterns, which govern to which SIP Entity a call is routed
- Define Communication Manager Evolution Server as an Managed Element
- Adding SIP Endpoints/SIP URE users

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager, using the URL "http://<fqdn>/SMGR" or "http://<ip-address>/SMGR", where "<fqdn> is the fully qualified domain name of Avaya Aura® System Manager or the "<ip-address>" is the IP address of Avaya Aura® System Manager.

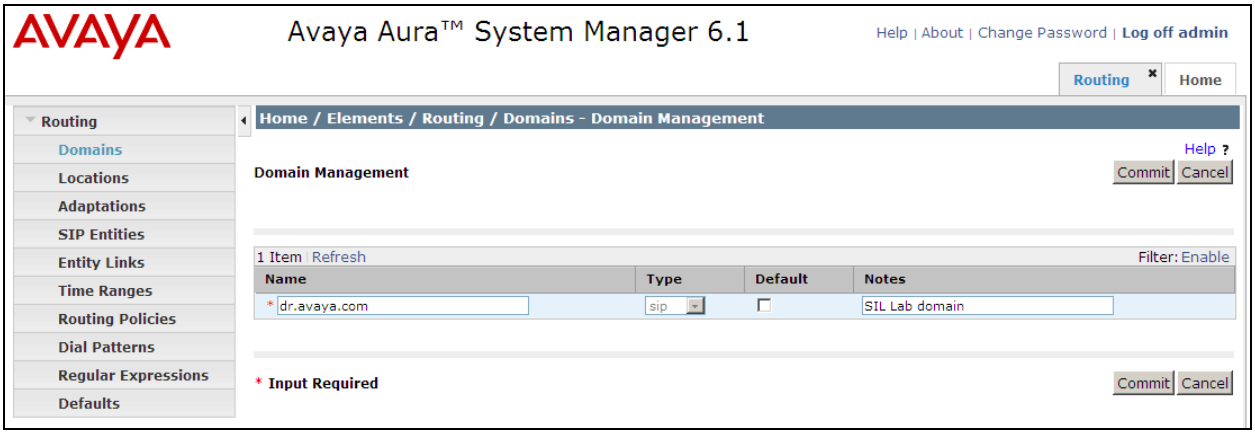
Log in with the appropriate credentials.
Once logged in select the **Routing** Link under the **Elements** column. Select a specific item such as **Domains**.

3.1. Administer SIP Domains

Select **Domains**.

- Click **New** (Not shown)
- Under *Name* add the same name given in **Section 2.4** for the **Authoritative Domain**
- Under *Notes* add a brief description.
- Click **Commit** to save.

The screen below shows the information for the sample configuration.

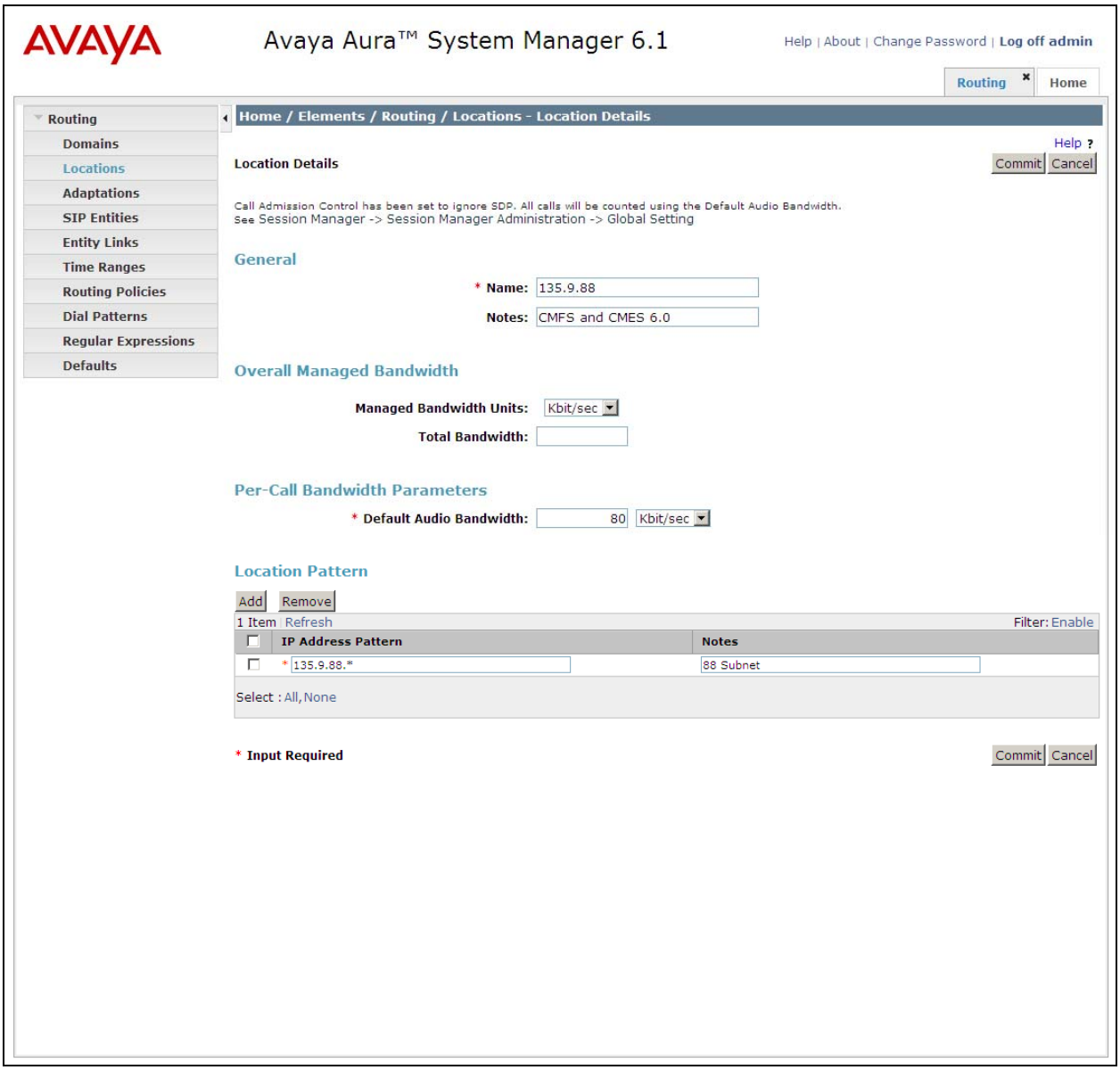


3.2. Define Locations

Select **Locations.** Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management or location-based routing.

- Click **New** (Not shown)
- In the *General* Section, under *Name* add a descriptive name.
- Under *Notes* add a brief description.
- In the *Location Pattern* Section, click **Add**. Under IP Address Pattern section, enter pattern used to logically identify the location. Under *Notes* add a brief description.
- Click **Commit** to save.

The screen below shows the information for Communication Manager Evolution Server in the sample configuration.

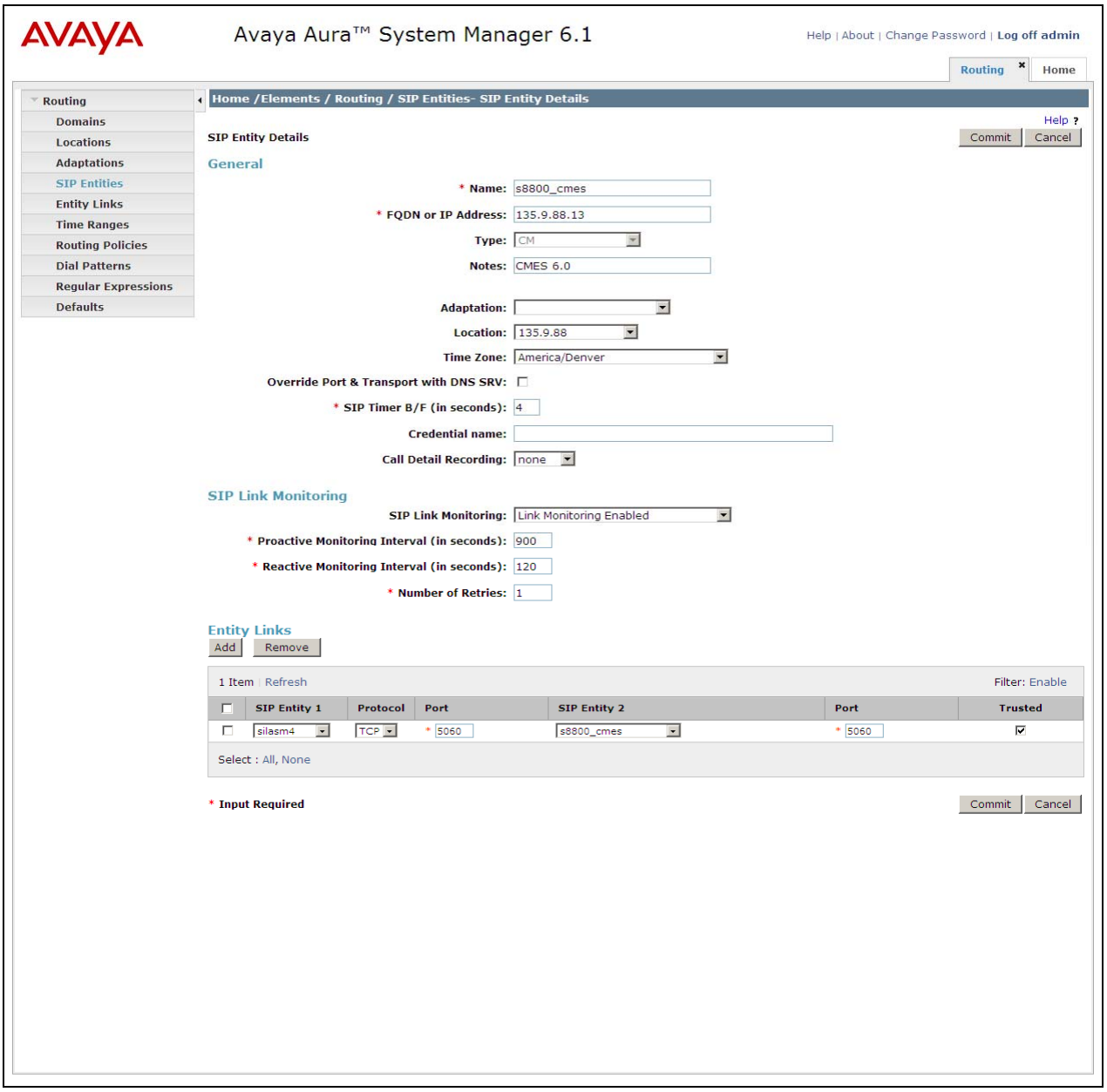


3.3. Add Avaya Aura® Communication Manager Evolution Server

The following section captures relevant screens for defining Avaya Aura® Communication Manager Evolution Server applicable for the sample configuration.

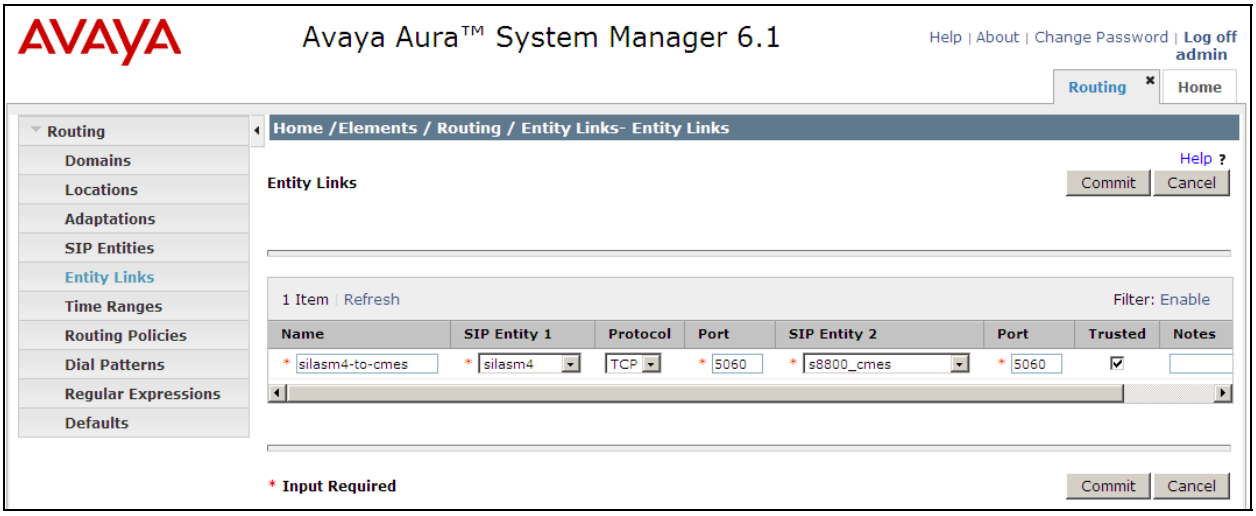
3.3.1. Define SIP Entities for Avaya Aura® Communication Manager Evolution Server

The following screen shows addition of Communication Manager Evolution Server. The IP address used is that of the Processor Ethernet (procr) of Avaya Communication Manager Evolution Server.



3.3.2. Define Entity Links for Avaya Aura® Communication Manager Evolution Server

The following screen shows the Entity Link defined for Avaya Aura® Communication Manager Evolution Server.



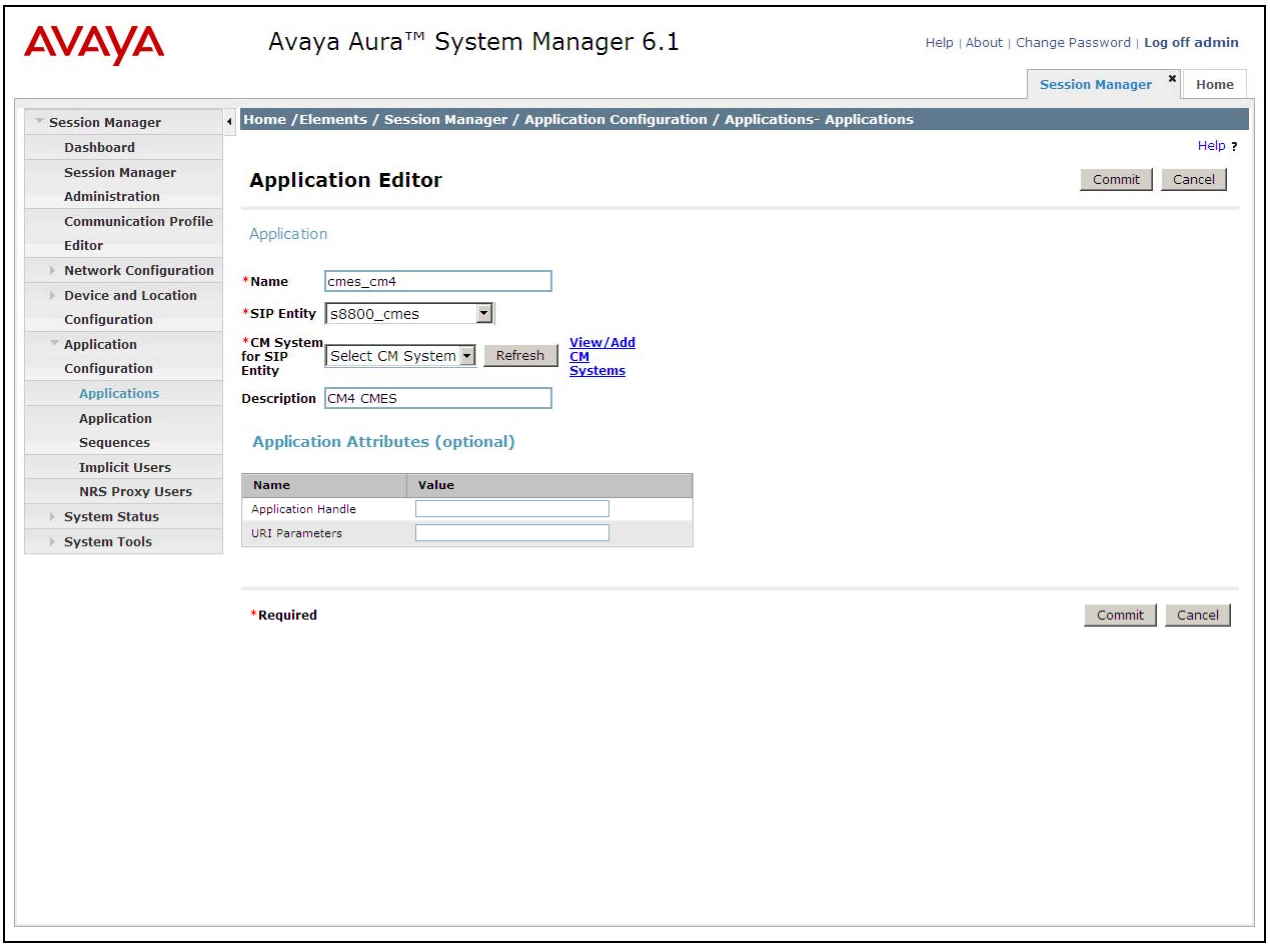
3.3.3. Define Routing Policy for Avaya Aura® Communication Manager Evolution Server

Since the SIP users are registered on Session Manager, a routing policy does not need to be defined for Communication Manager Evolution Server.

3.3.4. Define Applications for Avaya Aura® Communication Manager Evolution Server

To define Avaya Aura® Communication Manager Evolution Server Applications,

- **Elements -> Session Manager->Application Configuration → Applications**
  - Click **New** (Not shown)
  - Under *Name*, enter a name for the Application entry
  - Under *SIP Entity* drop-down menu, select the appropriate SIP Entity.
  - Under *CM System for SIP Entity*, this field can be left as the default of Select CM System.
  - Under *Description*, enter a description if desired.
  - Click **Commit** to save.

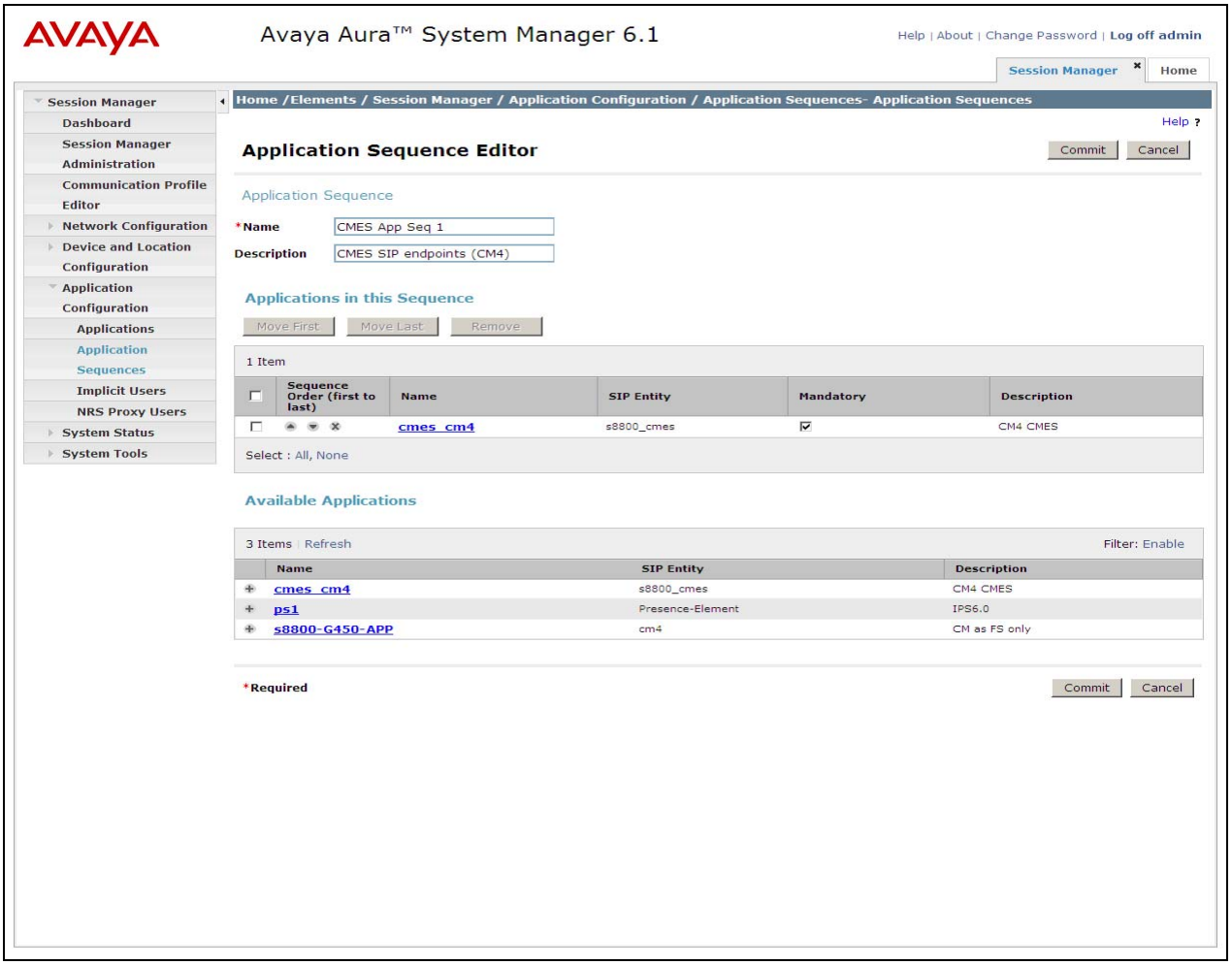


3.3.5. Define Application Sequences for Avaya Aura® Communication Manager Evolution Server

To define Avaya Aura® Communication Manager Evolution Server Application Sequences,

- **Elements -> Session Manager->Application Configuration → Application Sequences**
  - Click **New** (Not shown)
  - Under *Name*, enter a name of the Application Sequence.
  - Under *Description*, enter a description if desired.
  - Under *Available Applications*, select the Application that was created in **Section 3.3.4**. The way to select the Application of choice is to click on the "+" symbol next to the Application desired. This will be added to the **Applications in this Sequence** list.
  - Click **Commit** to save.

Second, define an Application Sequence for call application sequencing in Avaya Aura® Communication Manager Evolution Server as shown below:

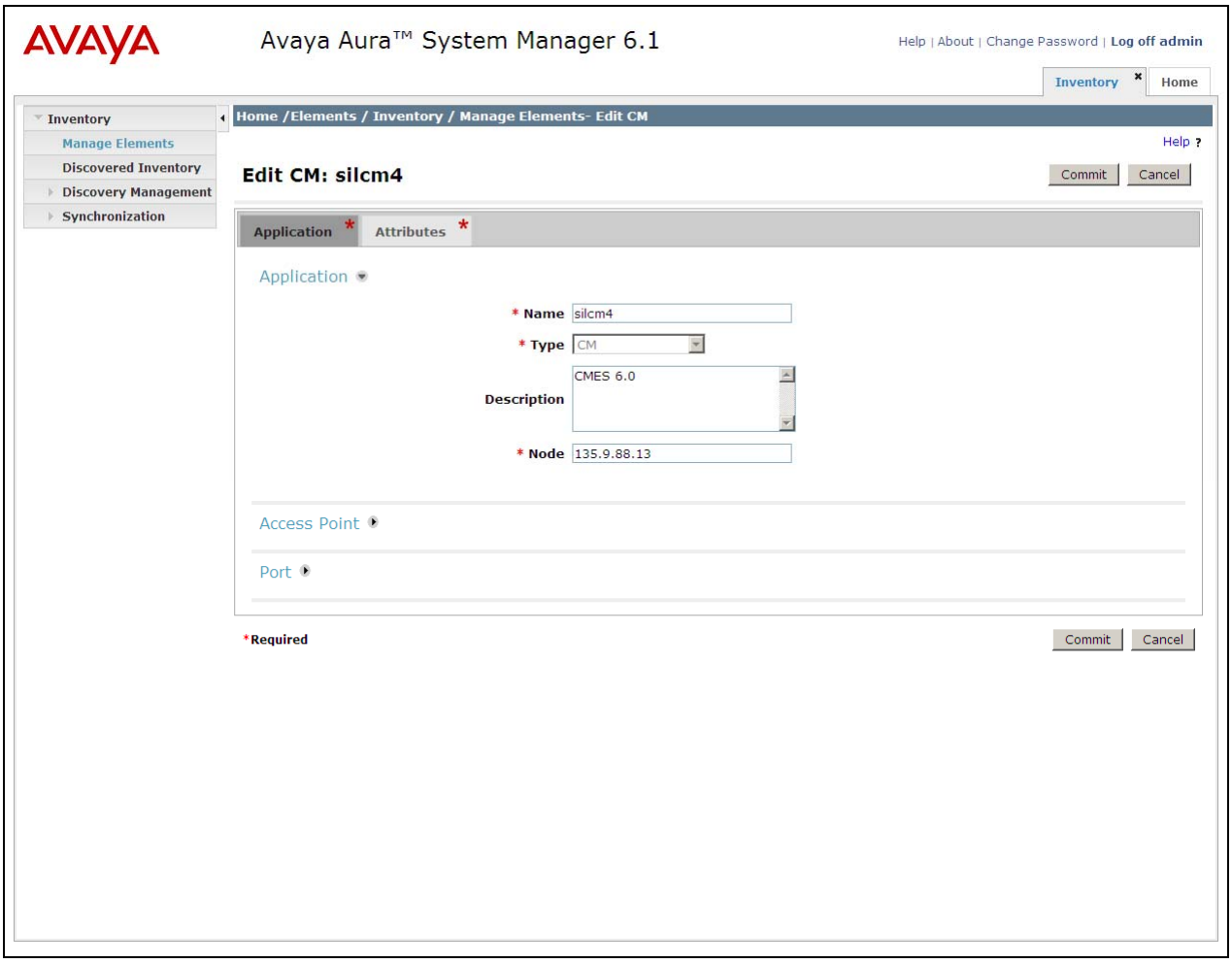


3.3.6. Define Avaya Aura® Communication Manager Evolution as an Administrable Entity

Before adding SIP users, Avaya Aura® Communication Manager Evolution Server must also be added to System Manager as an administrable entity. This action allows System Manager to access Communication Manager over its administration interface similar to how other administration tools such as Avaya Site Administrator access Communication Manager. Using this administration interface, System Manager will notify Communication Manager Evolution Server when new SIP users are added.

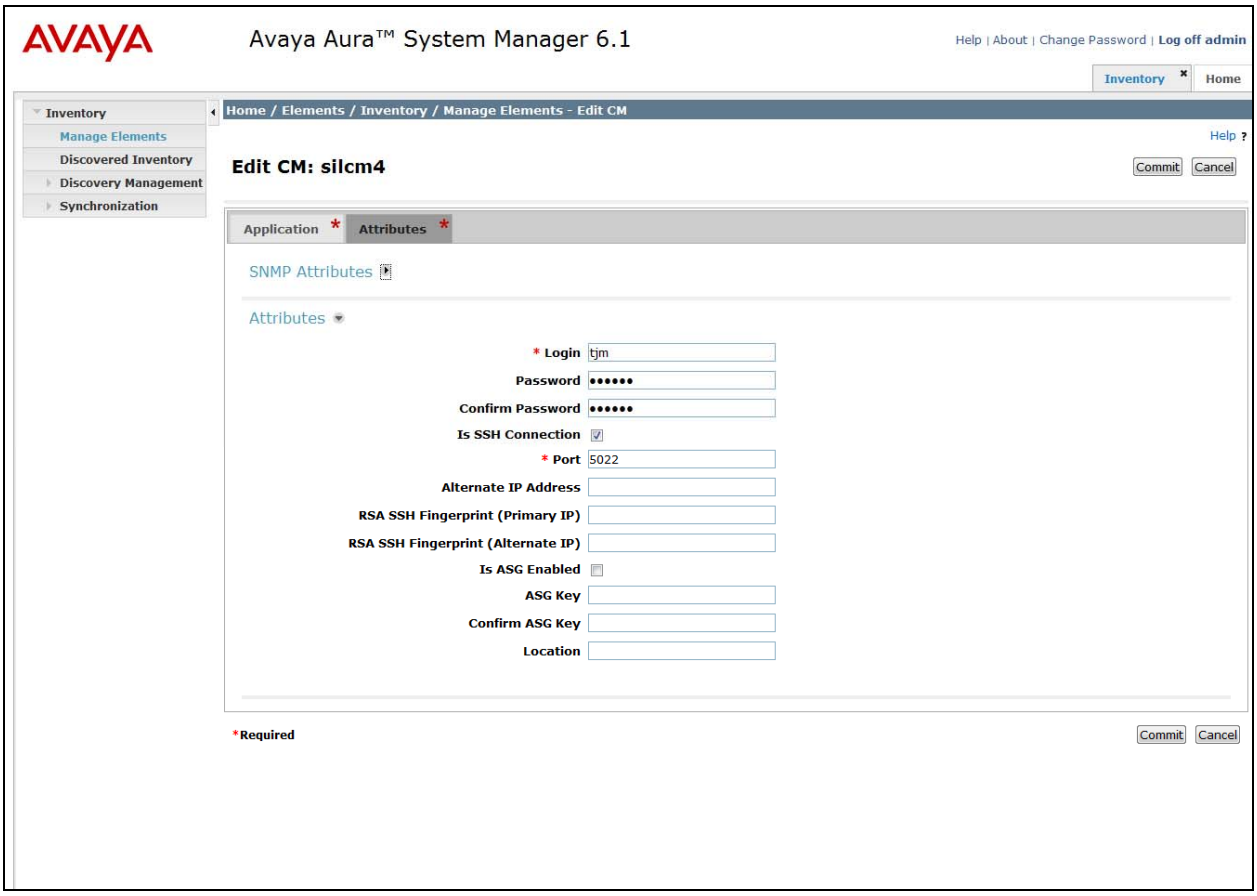
To define Avaya Aura® Communication Manager Evolution Server as an administrable entity, expand **Elements -> Inventory -> Manage Elements** (**Application** tab) and click **New** (Not shown).

- Under *Name*, enter an identifier for Communication Manager Evolution Server.
- Under *Type*, select CM from drop-down menu.
- Under *Node*, enter the IP address of the administration interface for the Evolution Server as shown below:



Defining Avaya Aura® Communication Manager Evolution Server as an administrable entity (continued):

- **Manage Elements – Attributes** tab
  - Under *Login and Password*, enter the login and password used for administration access to the Evolution Server.
  - Select SSH access.
  - Under *Port*, enter the port number for the administration interface of 5022 as shown below:



3.3.7. Add SIP Users

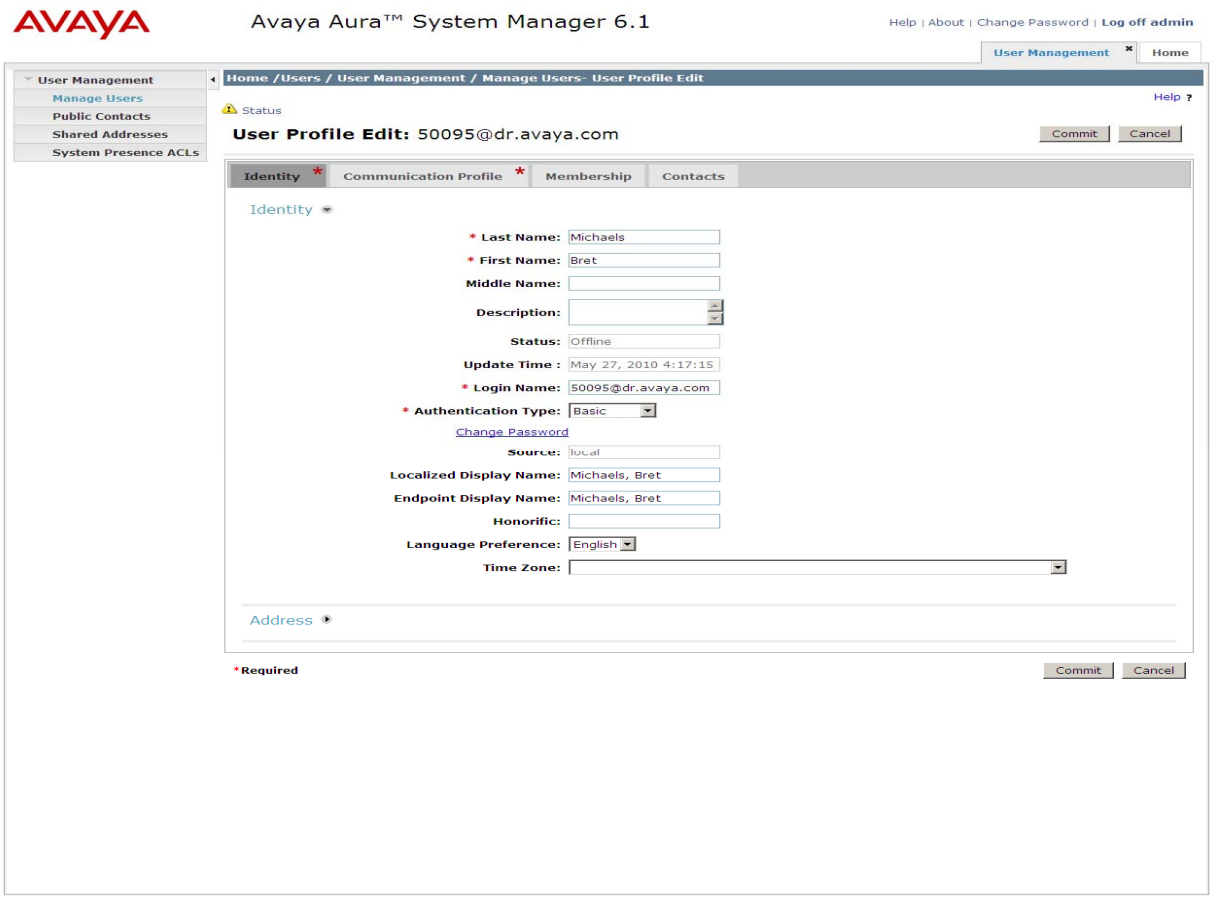
Add SIP users corresponding to the 96XX SIP stations defined in **Section 2.8.** Alternatively, use the option to automatically create station on Communication Manager when new user is added.

Under **Users** column, select **User Management → Manage Users** and click **New** (not shown).

Step 1 (Identity tab): Enter values for the following required attributes for a new SIP user in the **Identity** section of the new user form.

- **Last Name:** enter last name of user
- **First Name:** enter first name of user
- **Login Name:** enter extension no.@sip domain from **Section 3.1**
  Note: This field is primary handle of user.
- **Authentication Type:** select **Basic**
- **Password:** enter password which will be used to log into System Manager application.
  Note: This field is displayed when adding new user.
- **Confirm Password:** repeat value entered above.

The screen below shows results from **Step 1** when adding a new SIP user.



Step 2 (Communication Profile tab): Select the Communication Profile tab and Select **New** to define a **Communication Profile** for the new SIP user.

Enter values for the following required attributes:

- **Communication Profile Password:** enter a numeric value which will be used to logon to SIP phone.
  Note: this field must match the Security Code field on the station form defined in **Section 2.8.**
- **Confirm Password:** repeat numeric password
- **Name:** enter name of communication profile
- **Default:** enter checkmark to indicate profile is default profile

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

- **Type:** select Avaya SIP
- **Fully Qualified Address:** enter extension number
  Note: value is shown in **Handle** field after address is added.
- **@:** select SIP domain defined in **Section 3.1**
  Note: value is shown in **Domain** field.

Click **Add** to save the **Communication Address** for the new SIP user.

Step 3 (Communication Profile tab): Assign the **Application Sequence** defined in **Section 3.3.4** to the new SIP user as part of defining the **Communication Profile**. The **Application Sequence** can be used for both the originating and terminating sequence.

Select the **Session Manager Profile** box and enter the appropriate values for the following attributes:

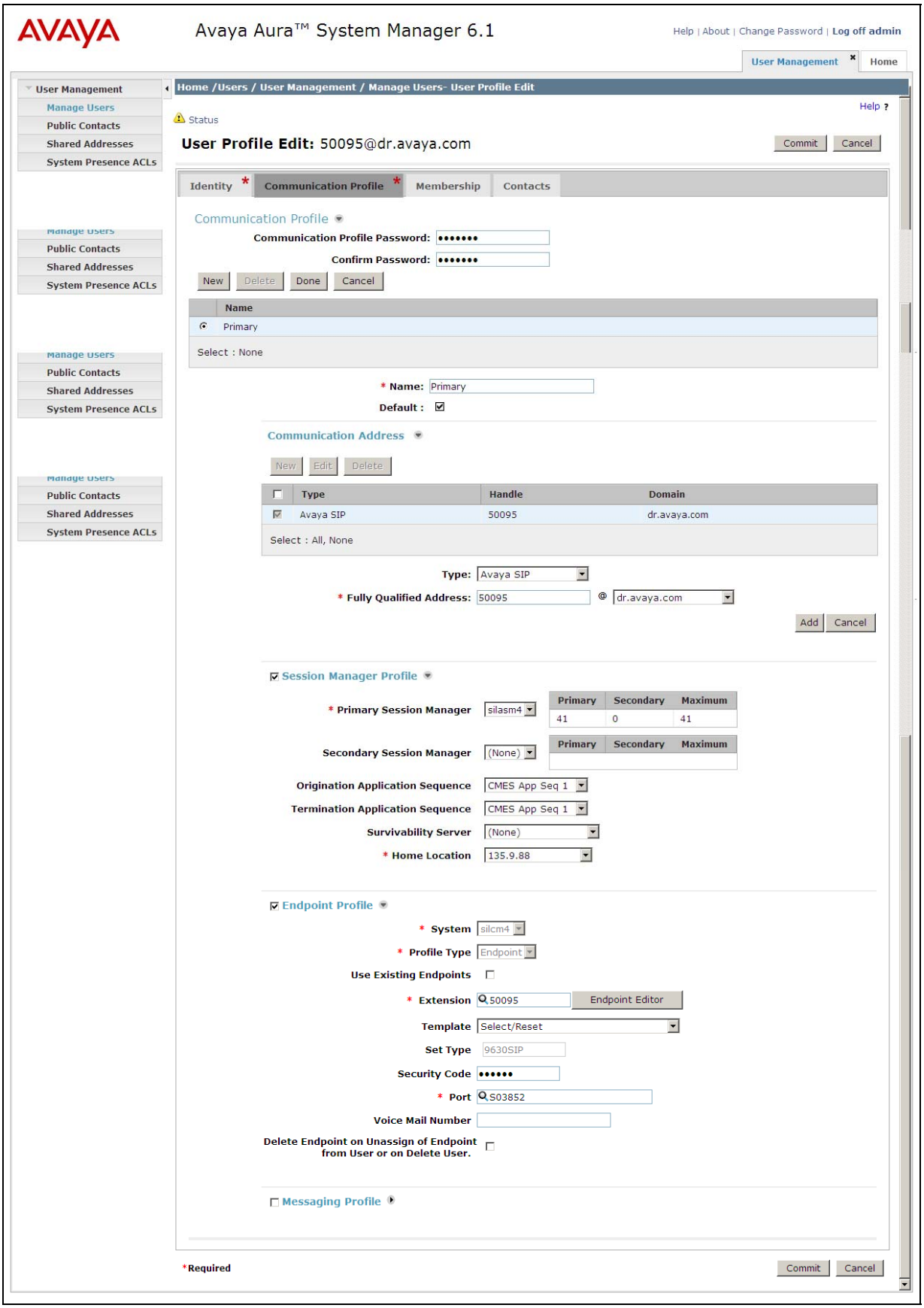
- **Primary Session Manager:** select the appropriate Session Manager instance
- **Origination Application Sequence:** enter the appropriate sequence
- **Termination Application Sequence:** enter the appropriate sequence
- **Home Location:** select the appropriate location that was created in **Section 3.2**

Enter values for the following required attributes of the **Endpoint Profile** section:

- **System:** select the SIP Entity of the Communication Manager Evolution Server defined in **Section 3.3.6** from menu
- **Profile Type:** enter Endpoint
- **Use Existing Stations:** enter checkmark if station was already defined. Else, station will automatically be created.
- **Extension**: enter extension number
- **Template:** select the template (system defined or user defined) associated with the type of endpoint to be added.
- **Set Type:** select "9630SIP" for this video endpoint
- **Security Code:** enter numeric value used to logon to SIP phone. **Note:** this field must match the value entered for the **Shared Communication Profile Password** field
- **Port**: select port number from the list for the selected template

Click **Commit** to save new user profile.

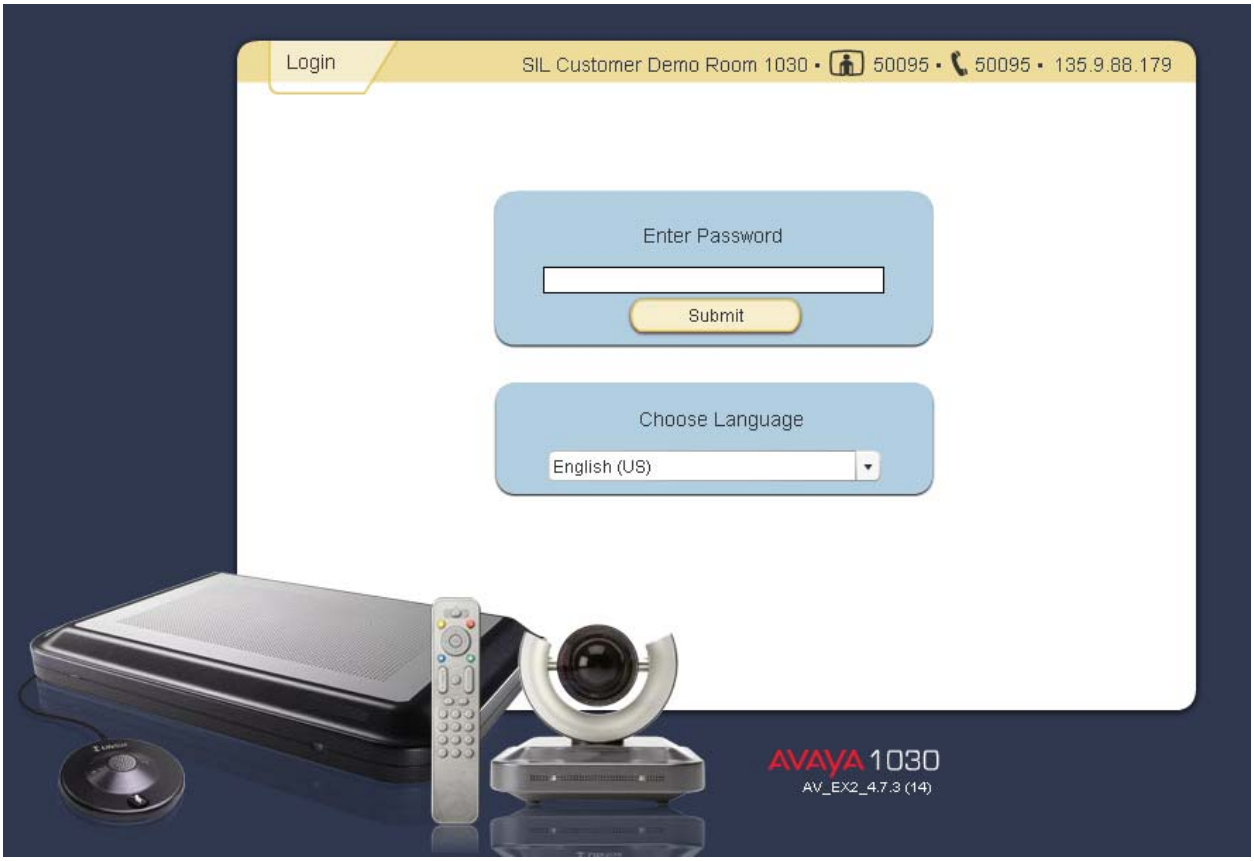
The screen shown on the next page displays the Communication Profile information when adding a new SIP user to the sample configuration.



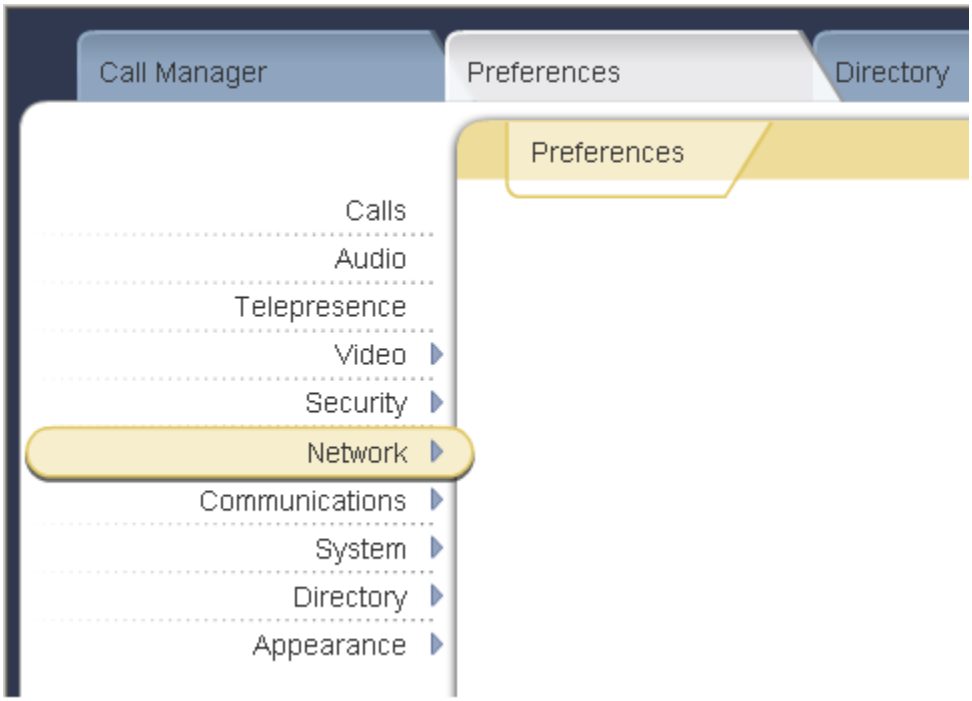
4. Configure Avaya 10x0 Video Endpoint

To administer the 10x0 video endpoints log in to the web interface using the IP address of the video endpoint. You will be redirected to a screen that looks similar to the one below. This is a sample configuration on how to administer a 10x0 video endpoint.

Step 1: Enter the proper login credentials and press **Submit**. Most of the Preferences can be customized to meet your needs. Mentioned below are the absolute necessary items that need to be administered to get the 10x0 up and running on the network.



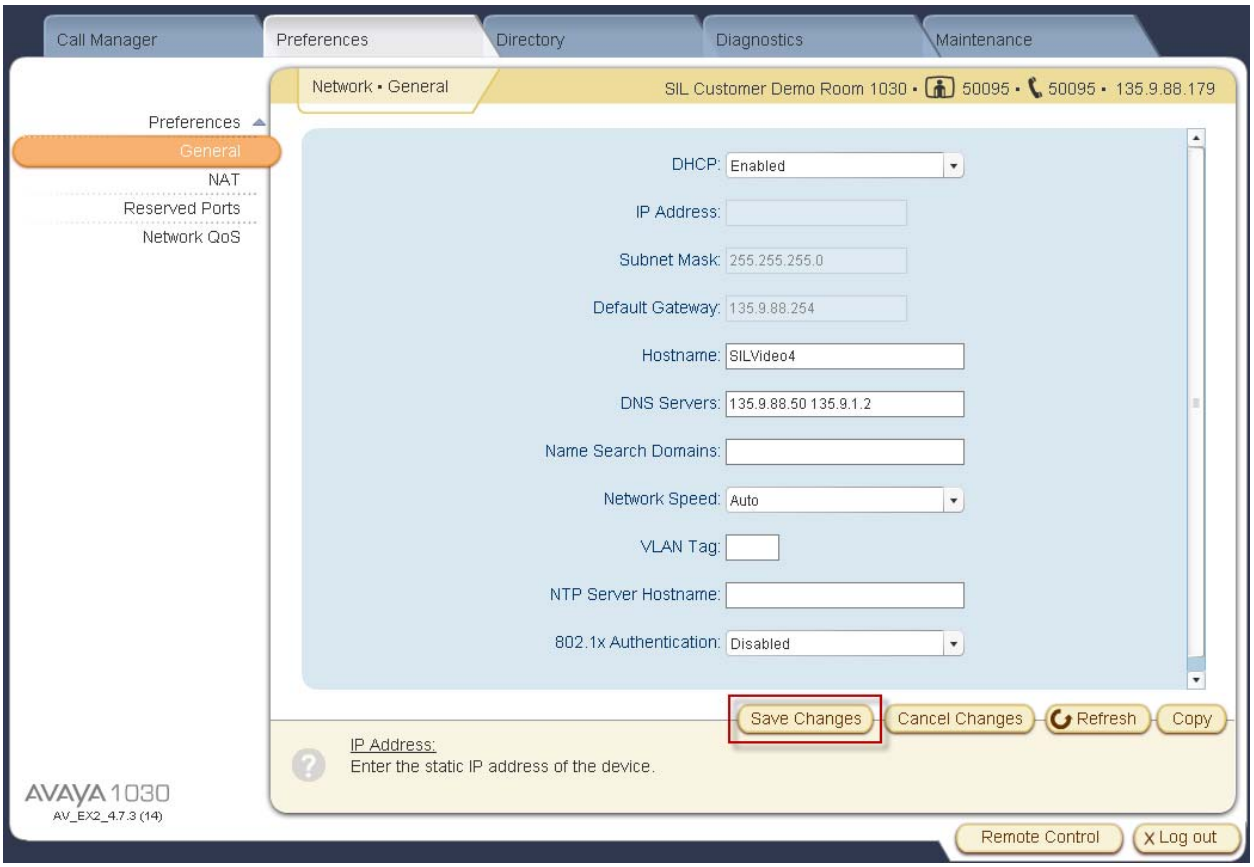
Once logged in select the **Preferences** tab and then the **Network** option.



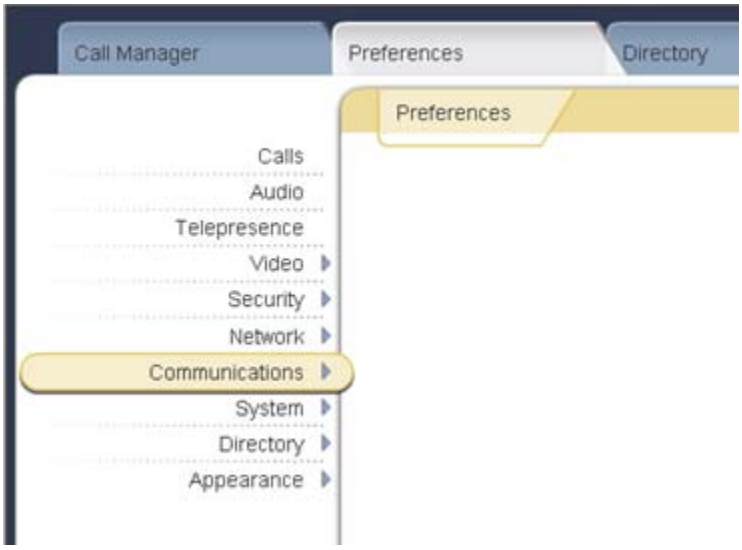
Select **General** option and enter values for the following required attributes.

- **DHCP:** select either Enabled/Disabled
- **IP Address:** enter IP Address if DHCP is **disabled**
- **Subnet Mask:** enter Subnet Mask if DHCP is **disabled**
- **Default Gateway:** enter Default Gateway if DHCP is **disabled**
- **Hostname:** enter the appropriate Hostname
- **DNS Servers:** enter the appropriate DNS Servers
- **NTP Server Hostname:** enter NTP Server Hostname

Select the **Save Changes** button to save the administration just added.

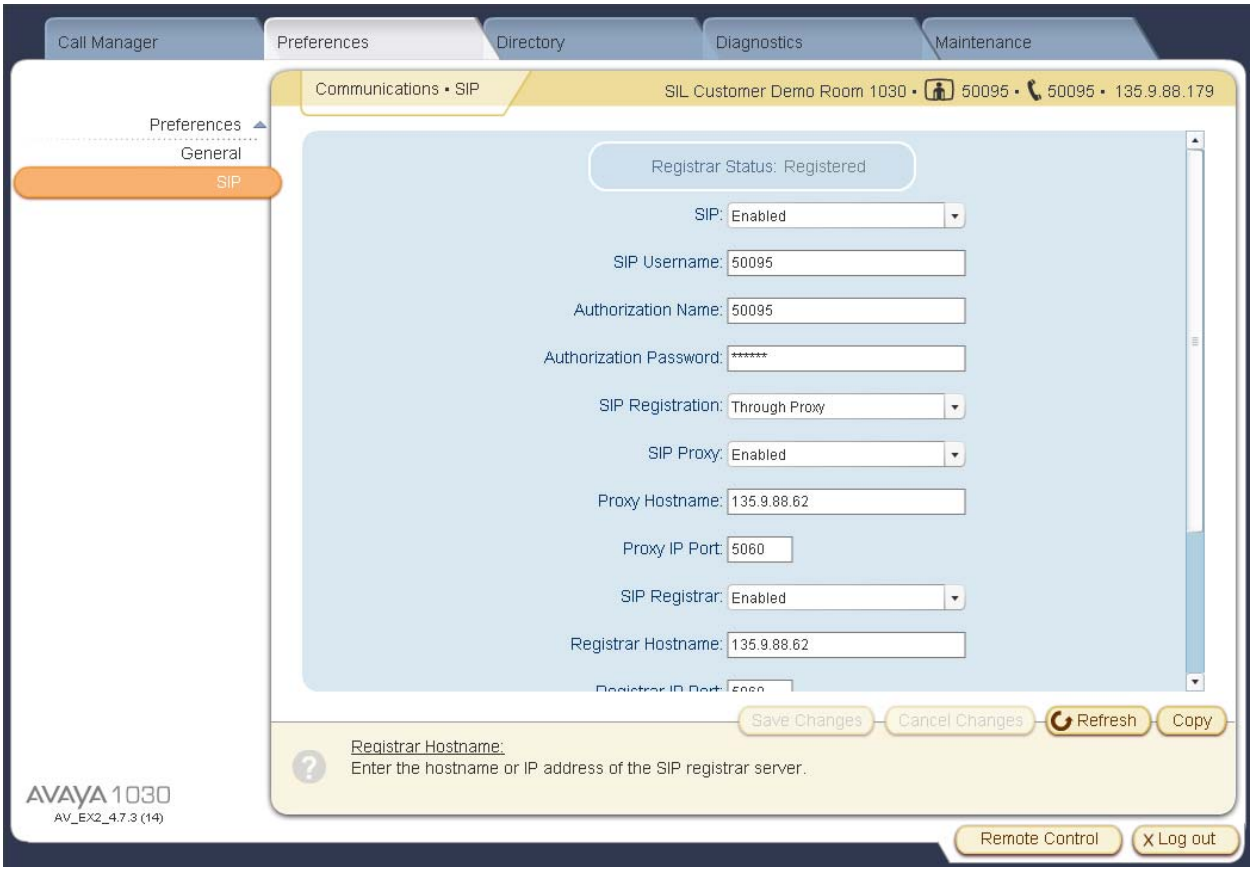


Select the **Preferences** option and select the **Communications** option.



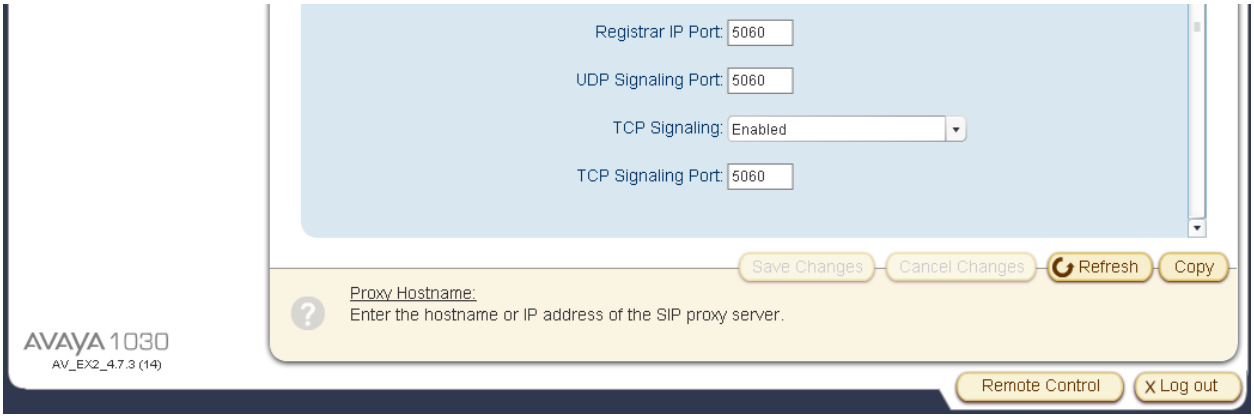
Select the **SIP** option.

- **SIP:** select Enabled
- **SIP Username:** enter the SIP Username for the device. **NOTE**: The SIP Username should be unique and meaningful to the endpoint.
- **Authorization Name:** enter the SIP Server authorization username. **NOTE**: The Authorization Name should be the extension number that is used to register the SIP endpoint defined in **Section 3.3.7**
- **Authorization Password:** enter the SIP Server authorization password which should match the Shared Communication Profile Password defined in **Section 3.3.7**
- **SIP Registration:** select the communication path to use when registering with a SIP Registrar
- **SIP Proxy:** choose 'Enabled' to use the SIP proxy
- **Proxy Hostname:** enter the hostname or IP address of the SIP proxy server. **NOTE:** This is the Session Manager software asset card IP address
- **Proxy IP Port:** enter the IP port number of SIP proxy server
- **SIP Registrar:** choose 'Enabled' to use the SIP registrar
- **Registrar Hostname:** enter the hostname or IP address of the SIP registrar server

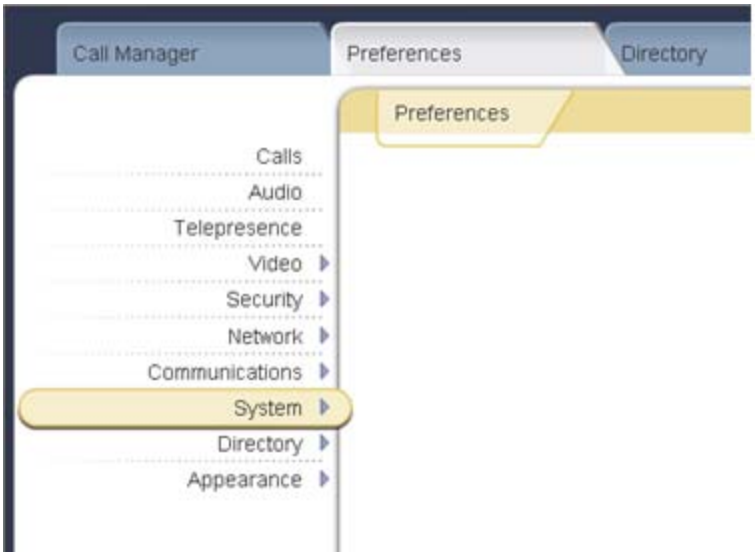


- **Registrar IP Port:** enter the IP port number of the SIP registrar server
- **UDP Signaling Port:** enter the UDP port number of the SIP configuration
- **TCP Signaling:** choose 'Enable' to use TCP for placing SIP call
- **TCP signaling Port:** enter the TCP port number of the SIP configuration

Select the **Save Changes** button to save the administration just added.



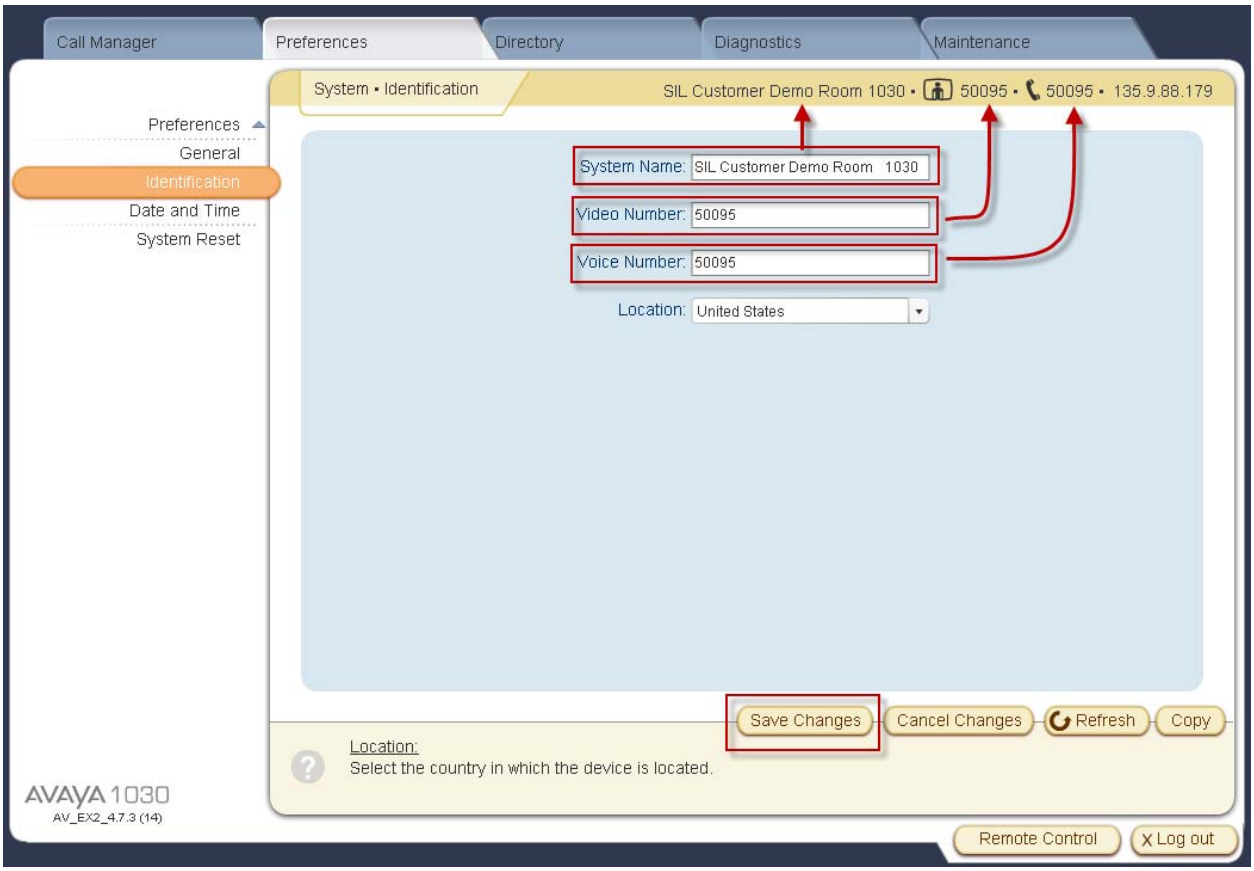
Select the **Preferences** option again and select **System**.



Select the **Identification** option. This option will allow the user to display the name and video/voice numbers on the menu bar.

- **System Name:** enter a descriptive name for the system
- **Video Number:** enter the video number of the endpoint
- **Voice Number:** enter the voice number of the endpoint

Select the **Save Changes** button to save the administration just added.



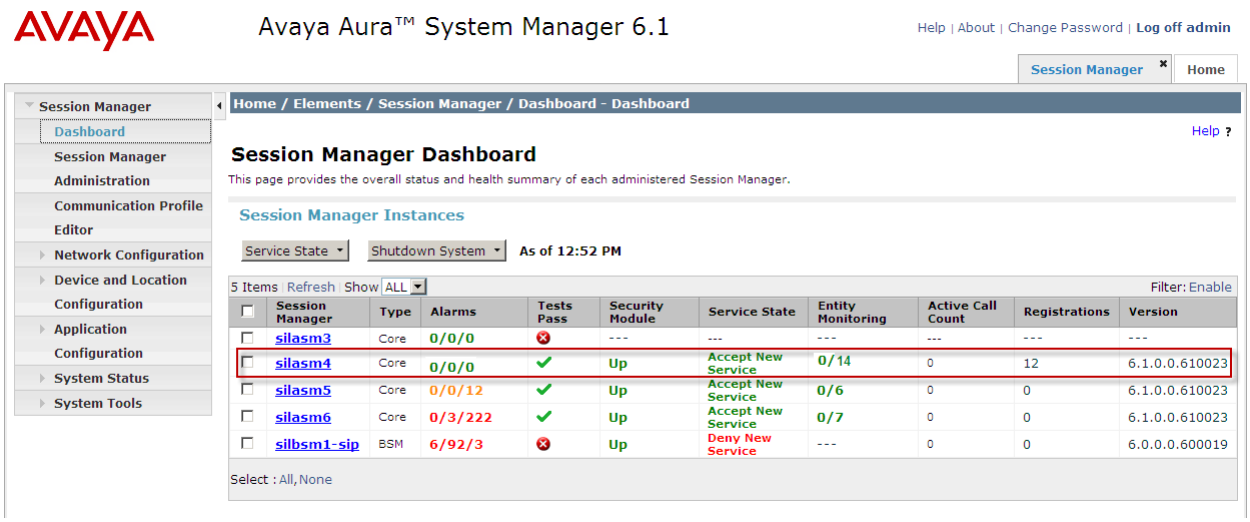
5. Verification Steps

5.1. Verify Avaya Aura® Session Manager Configuration

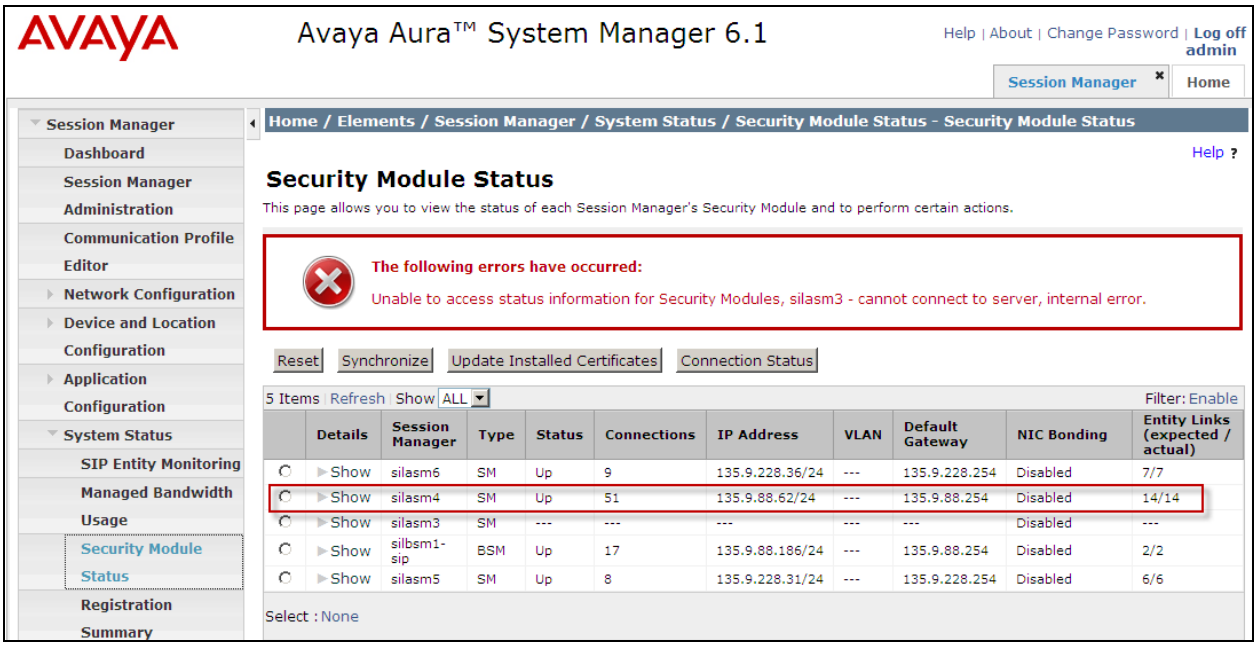
5.1.1. Verify Avaya Aura® Session Manager is Operational

Navigate to **Elements → Session Manager → Dashboard**

Verify the overall system status for Session Manager as shown below:

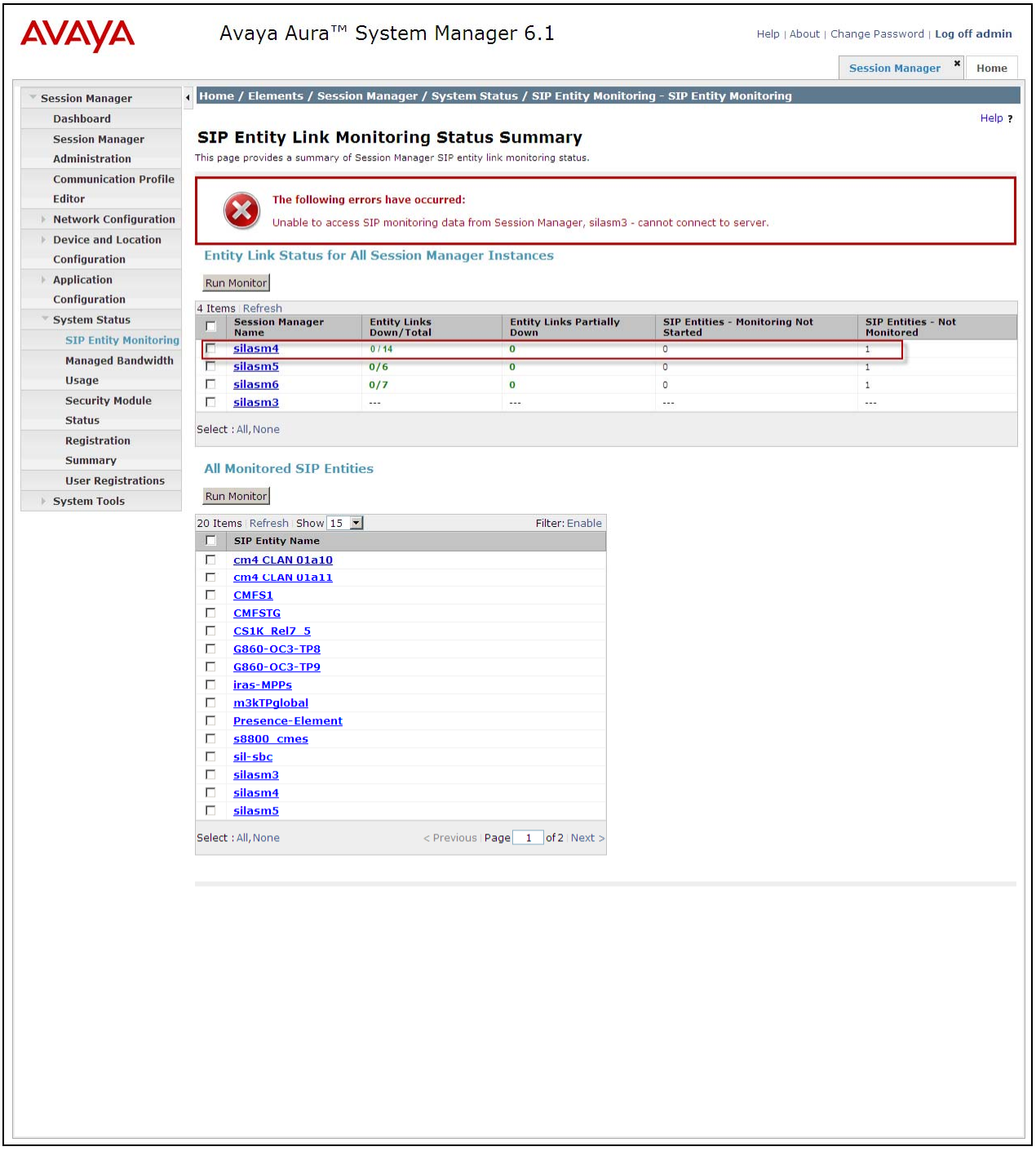


Navigate to **Elements → Session Manager → System Status → Security Module Status** to view more detailed status information on the status of Security Module for Session Manager. Verify the **Status** column displays "**Up**" as shown below.



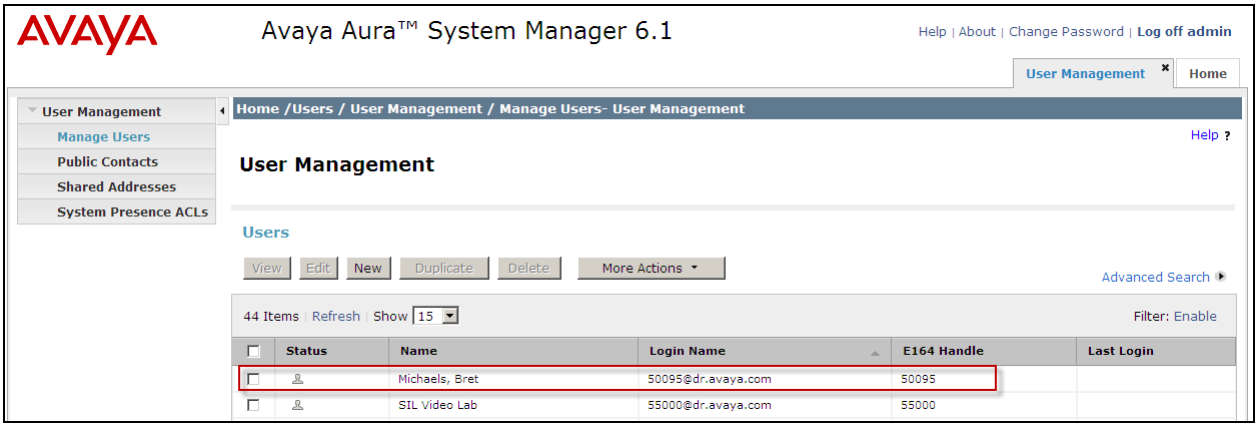
5.1.2. Verify SIP Link Status

Expand the Session Manager menu on the left and click **SIP Entity Monitoring**. Verify all SIP Entity Links are operational as shown below:

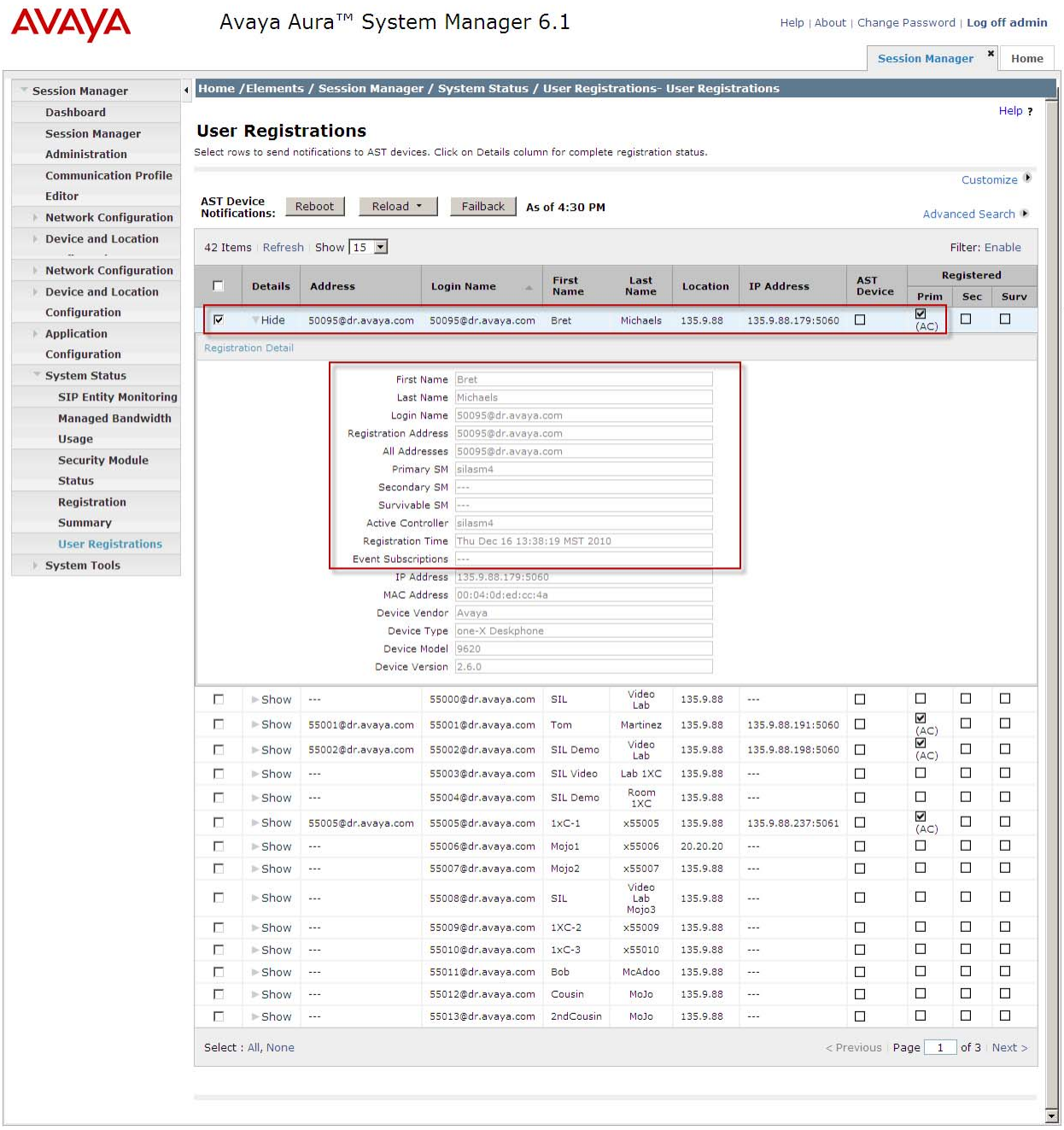


5.1.3. Verify Registrations of SIP Endpoints

Verify SIP users have been created in the Session Manager. In the sample configuration, Extension 50095 SIP user was created as shown in the highlighted area below:



Navigate to **Elements → Session Manager → System Status → User Registrations** to verify the SIP endpoints have successfully registered with the Session Manager as shown below:



5.2. Verify Avaya Aura® Communication Manager Evolution Server Configuration

Verify the status of the SIP trunk group by using the**status trunk n** command, where "**n**" is the trunk group number administered in **Section 2.6.** Verify that all trunks are in the "in-service/idle" state as shown below:

```
status trunk 10
                              TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy
0010/001 T00001   in-service/idle    no
0010/002 T00002   in-service/idle    no
0010/003 T00003   in-service/idle    no
0010/004 T00004   in-service/idle    no
0010/005 T00005   in-service/idle    no
0010/006 T00006   in-service/idle    no
0010/007 T00007   in-service/idle    no
0010/008 T00008   in-service/idle    no
0010/009 T00009   in-service/idle    no
0010/010 T00010   in-service/idle    no
```

Verify the status of the SIP signaling groups by using the **status signaling-group n** command, where "**n**" is the signaling group number administered in **Section 2.5** Verify the signaling group is "in-service" as indicated in the **Group State** field shown below:

```
status signaling-group 10
                          STATUS SIGNALING GROUP

        Group ID: 10
      Group Type: sip

     Group State: in-service
```

Use the Communication Manager SAT command, **list trace tac #**, where "**tac #"** is the trunk access code defined in **Section 2.6** to trace trunk group activity for the SIP trunk between the Session Manager and Communication Manager Evolution Server as shown below:

```
list trace tac #010                                          Page   1
                                LIST TRACE
time            data

18:32:04 TRACE STARTED 12/20/2010 CM Release String cold-00.1.510.1-18599
18:32:41 SIP<INVITE sip:55001@dr.avaya.com;transport=tcp SIP/2.0
18:32:41     dial 55001# route:UDP|AAR
18:32:41     term trunk-group 10    cid 0x13fe
18:32:41     dial 55001# route:UDP|AAR
18:32:41     route-pattern  10 preference 1  cid 0x13fe
18:32:41     seize trunk-group 10 member 19  cid 0x13fe
18:32:41     Calling Number & Name NO-CPNumber NO-CPName
18:32:41     Proceed trunk-group 10 member 19  cid 0x13fe
18:32:42 SIP>SIP/2.0 180 Ringing
18:32:42     Alert trunk-group 10 member 19  cid 0x13fe
18:32:44     active trunk-group 10 member 19  cid 0x13fe
18:32:44     G711MU ss:off ps:20
             rgn:2 [135.9.88.191]:60304
             rgn:2 [135.9.88.179]:60040
18:32:44     G711MU ss:off ps:20
             rgn:2 [135.9.88.179]:60040
             rgn:2 [135.9.88.191]:60304
18:32:44 SIP>SIP/2.0 200 OK
18:32:44     Video: H264 [135.9.88.191]:60306
18:32:44     Video: H264 [135.9.88.179]:60042
                      logChl:110 sessId:2 bw:21760 tx/rx:11520
18:32:44     Video: H264 [135.9.88.179]:60042
18:32:44     Video: H264 [135.9.88.191]:60306
                      logChl:110 sessId:2 bw:21760 tx/rx:11520
18:32:44 SIP>INFO sip:50095@135.9.88.179;transport=tcp SIP/2.0
18:32:44 SIP<ACK sip:55001@135.9.88.13;transport=tcp SIP/2.0
18:32:44 SIP<SIP/2.0 200 OK
18:32:51 SIP<BYE sip:55001@135.9.88.13;transport=tcp SIP/2.0
18:32:51 SIP>SIP/2.0 200 OK
18:32:51     idle station    50095 cid 0x13fe
```

Use the Communication Manager SAT command, **list trace station xxx**, where "**xxx"** is the extension number of the 96XX SIP telephone as shown below:

```
list trace station 50095                                          Page   1
                                LIST TRACE
time            data

18:35:36 TRACE STARTED 12/20/2010 CM Release String cold-00.1.510.1-18599
18:36:13     active station    50095 cid 0x13ff
18:36:13 SIP>INVITE sip:55001@dr.avaya.com SIP/2.0
18:36:13     dial 55001# route:UDP|AAR
18:36:13     term trunk-group 10    cid 0x13ff
18:36:13     dial 55001# route:UDP|AAR
18:36:13     route-pattern  10 preference 1  cid 0x13ff
18:36:13     seize trunk-group 10 member 20  cid 0x13ff
18:36:13     Setup digits 55001
18:36:13     Calling Number & Name *50095 Michaels, Bre
18:36:13 SIP<SIP/2.0 100 Trying
18:36:13     Proceed trunk-group 10 member 20  cid 0x13ff
18:36:13 SIP<SIP/2.0 422 Session Interval Too Small
18:36:13 SIP>ACK sip:55001@dr.avaya.com SIP/2.0
18:36:13 SIP>INVITE sip:55001@dr.avaya.com SIP/2.0
18:36:13 SIP<SIP/2.0 100 Trying
18:36:13 SIP<SIP/2.0 180 Ringing
18:36:13     Alert trunk-group 10 member 20  cid 0x13ff
18:36:15 SIP<SIP/2.0 200 OK
18:36:15     active trunk-group 10 member 20  cid 0x13ff
18:36:15     G711MU ss:off ps:20
             rgn:2 [135.9.88.191]:60312
             rgn:2 [135.9.88.179]:60048
18:36:15     G711MU ss:off ps:20
             rgn:2 [135.9.88.179]:60048
             rgn:2 [135.9.88.191]:60312
18:36:15     Video: H264 [135.9.88.191]:60314
18:36:15     Video: H264 [135.9.88.179]:60050
                      logChl:110 sessId:2 bw:21760 tx/rx:11520
18:36:15     Video: H264 [135.9.88.179]:60050
18:36:15     Video: H264 [135.9.88.191]:60314
                      logChl:110 sessId:2 bw:21760 tx/rx:11520
18:36:16 SIP<INFO sip:+50095@135.9.88.13;transport=tcp SIP/2.0
18:36:16 SIP>SIP/2.0 200 OK
18:36:16 SIP>ACK sip:55001@135.9.88.72;transport=tcp SIP/2.0
18:36:22 SIP>BYE sip:55001@135.9.88.72;transport=tcp SIP/2.0
18:36:22     idle station    50095 cid 0x13ff
```

5.3. Call Scenarios Verified

Verification scenarios for the configuration described in these Application Notes included the following call scenarios:

Calls initiated from the GUI of the respective endpoint

- Place a point-to-point video call from a 1020/1030/1040 video endpoint registered to SM (CMES) to another 1020/1030/1040 video endpoint registered on SM (CMES). Answer the call and verify two-way video and two-way talk path for all combinations of calls between10x0 video endpoints. Verify Call statistics on the endpoint GUI.
- Place a point-to-point video call from a 1040 video endpoint registered to SM (CMES) to another 1020/1030/1040 video endpoint registered on SM (CMES). Answer the call and verify two-way video and talk path. Place a video conference call from 1040 to a 1020. Answer the call and verify three-way video and audio conference call. Add a fourth video endpoint to the call and verity video and audio. Verify Call statistics on the endpoint GUI.
- Place a point-to-point audio call from a 1020/1030/1040 video endpoint registered to SM (CMES) to another 1020/1030/1040 video endpoint registered on SM (CMES). Answer the call and verify two-way talk path for all combinations of calls between10X0 video endpoints. Verify Call statistics on the endpoint GUI.
- Place a point-to-point audio call from a 1040 video endpoint registered to SM (CMES) to another 1020/1030/1040 video endpoint registered on SM (CMES). Answer the call and verify two-way talk path. Place an audio conference call from 1040 to a 1020/1030/1040. Answer the call and verify talk path on conference call. Add a fourth video endpoint to the call and verify talk path. Verify Call statistics on the endpoint GUI.

Calls initiated from the Web interface of the respective endpoint

- Place a point-to-point video call from a 1020/1030/1040 video endpoint registered to SM (CMES) to another 1020/1030/1040 video endpoint registered on SM (CMES). Answer the call and verify two-way video and two-way talk path for all combinations of calls between10x0 video endpoints. Verify Call statistics on the endpoint GUI.
- Place a point-to-point video call from a 1040 video endpoint registered to SM (CMES) to another 1020/1030/1040 video endpoint registered on SM (CMES). Answer the call and verify two-way video and talk path. Place a video conference call from 1040 to a 1020. Answer the call and verify three-way video and audio conference call. Add a fourth video endpoint to the call and verity video and audio. Verify Call statistics on the endpoint GUI.
- Place a point-to-point audio call from a 1020/1030/1040 video endpoint registered to SM (CMES) to another 1020/1030/1040 video endpoint registered on SM (CMES). Answer the call and verify two-way talk path for all combinations of calls between10X0 video endpoints. Verify Call statistics on the endpoint GUI.
- Place a point-to-point audio call from a 1040 video endpoint registered to SM (CMES) to another 1020/1030/1040 video endpoint registered on SM (CMES). Answer the call and verify two-way talk path. Place an audio conference call from 1040 to a 1020/1030/1040. Answer the call and verify talk path on conference call. Add a fourth video endpoint to the call and verify talk path. Verify Call statistics on the endpoint GUI.

6. Acronyms

AAR	Automatic Alternative Routing (Routing on Communication Manager)
ARS	Alternative Routing Service (Routing on Communication Manager)
CMES	Communication Manager Evolution Server
IMS	IP Multimedia Subsystem
IP	Internet Protocol
RTP	Real Time Protocol
SAT	System Access Terminal (Communication Administration Interface)
SIL	Solution Interoperability Lab
SIP	Session Initiation Protocol
SM	Avaya Aura® Session Manager
SMGR	System Manager (used to configure Session Manager)
TAC	Trunk Access Code (Communication Manager Trunk Access)
TCP	Transmission Control Protocol
TCP/IP	Transmission Control Protocol/Internet Protocol
TLS	Transport Layer Security
URE	User Relation Element

7. Conclusion

These Application Notes describe how to configure Avaya Aura® Session Manager and Avaya Aura® Communication Manager operating as a Evolution Server to support the Avaya 10x0 Series SIP video endpoints. Interoperability testing included successfully making bi-directional calls between several different types of video endpoints and the use of the conferencing feature of the internal MCU of the 1040. These successful calls were generated via the GUI of each respective video endpoint as well as each video endpoints respective Web interface.

8. Additional References

This section references the product documentation relevant to these Application Notes.

Session Manager

1) Avaya Aura® Session Manager Overview, Doc ID 03-603323, available at http://support.avaya.com.
2) Installing and Administering Avaya Aura® Session Manager, Doc ID 03-603324, available at http://support.avaya.com.
3) Avaya Aura® Session Manager Case Studies, dated January 2, 2010, available at http://support.avaya.com
4) Maintaining and Troubleshooting Avaya Aura® Session Manager, Doc ID 03-603325, available at http://support.avaya.com.

Communication Manager

5) Hardware Description and Reference for Avaya Aura® Communication Manager (COMCODE 555-245-207) http://support.avaya.com/elmodocs2/comm_mgr/r4_0/avayadoc/03_300151_6/245207_6/245207_6.pdf
6) SIP Support in Avaya Aura® Communication Manager Running on Avaya S8xxx Servers, Doc ID 555-245-206, May 2009, available at http://support.avaya.com.
7) Administering Avaya Aura® Communication Manager, Doc ID 03-300509, May 2009, available at http://support.avaya.com.
8) Administering Avaya Aura® Communication Manager as a Feature Server, Doc ID 03-603479, November 2009, available at http://support.avaya.com

Avaya 1000 Series Video Endpoints

9) Avaya 1010/1020 Installation Guide, Issue 1, June 2010, available at http://support.avaya.com
10) Avaya 1010/1020 User Guide, Issue 1, June 2010, available at http://support.avaya.com
11) Avaya 1030 Installation Guide, Issue 1, June 2010, available at http://support.avaya.com
12) Avaya 1040 Installation Guide, Issue 1, June 2010, available at http://support.avaya.com
13) Avaya 1050 Installation Guide, Issue 1, June 2010, available at http://support.avaya.com
14) Avaya Video Communications System Administrator Guide (1050/1040/1030) , Issue 1, June 2010, available at http://support.avaya.com
15) Avaya Video Communications System User Guide (1050/1040/1030) , Issue 1, June 2010, available at http://support.avaya.com
16) Avaya Video Camera 100 Installation Guide, Issue 1, June 2010, available at http://support.avaya.com
17) Avaya Video Conferencing Manager Deployment Guide, Issue 1, June 2010, available at http://support.avaya.com

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