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 Feature 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 a Feature 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 a Feature 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 Feature 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 require Avaya Aura® Communication Manager operating as a Feature Server. Communication Manager Feature Server only supports IP Multimedia Subsystem (IMS)-SIP users that are registered to Avaya Aura® Session Manager. Communication Manager Feature Server is connected to Session Manager via an IMS-enabled 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 Feature 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 the Communication Manager Feature Server and Session Manager. Detailed administration of Communication Manager Evolution Server will not be described (see the appropriate documentation listed in **Section 8**).

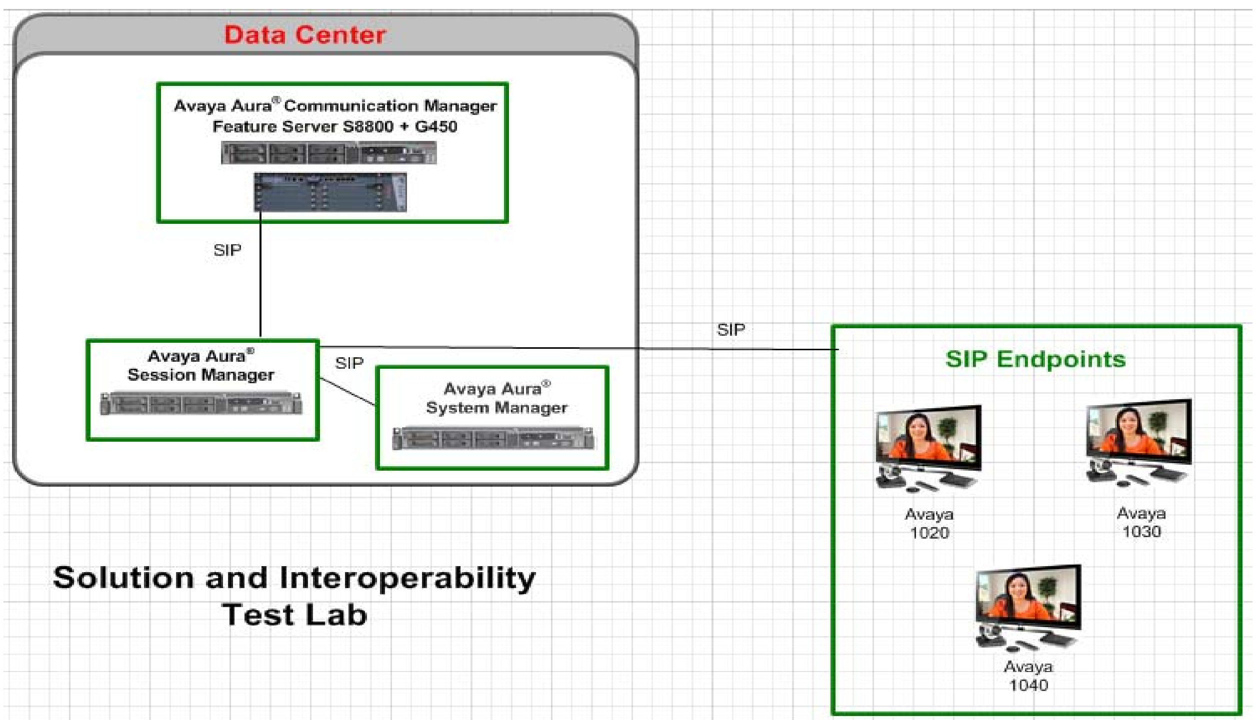


Figure 1 – Sample Configuration

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 Feature 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 Feature Server

This section describes the administration of Communication Manager Feature 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 the Communication Manager Feature 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 0
           Maximum Concurrently Registered IP Stations: 18000 0
             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 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 128
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 50
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   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.)
```

2.1.2. AAR/ARS Routing Check

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

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

 A/D Grp/Sys List Dialing Start at 01? n                              CAS Main? n
Answer Supervision by Call Classifier? n                     Change COR by FAC? n
                                  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?
```

2.1.3. Enable Private Numbering

Use the **change system-parameters customer-options** command to verify that **Private Networking** is enabled as shown below:

```
display system-parameters customer-options                 Page   5 of  11
                                OPTIONAL FEATURES

               Multinational Locations? y          Station and Trunk MSP? y
Multiple Level Precedence & Preemption? n     Station as Virtual Extension? y
                    Multiple Locations? y
                                           System Management Data Transfer? n
         Personal Station Access (PSA)? y              Tenant Partitioning? n
                       PNC Duplication? n      Terminal Trans. Init. (TTI)? y
                  Port Network Support? n              Time of Day Routing? n
                       Posted Messages? n      TN2501 VAL Maximum Capacity? y
                                                      Uniform Dialing Plan? y
                    Private Networking? y    Usage Allocation Enhancements? y
              Processor and System MSP? y
                    Processor Ethernet? y              Wideband Switching? n
                                                                 Wireless? y
```

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               135.9.88.72
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**".
- **Media Encryption**: Enter the value based on the system requirement. For the sample configuration "none" was used.

```
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:

     Media Encryption
 1: none
```

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  19
                               IP NETWORK REGION
  Region: 1
Location: 1        Authoritative Domain: dr.avaya.com
    Name: CMFS-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"
- **IMS Enabled?:** "y"
- **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"
- **Direct IP-IP Early Media?:** "y"

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

Group Number: 1               Group Type: sip
                        Transport Method: tcp
  IMS Enabled? y
     IP Video? y          Priority Video? n
  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:  1
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                    Direct IP-IP Early 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 1                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: SIP Video TG to silasm4     COR: 1        TN: 1       TAC: #001
Direction: two-way        Outgoing Display? y
 Dial Access? n                               Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 1
                                                     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 1                                              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
```

On **Page 3**, set **Numbering Format** to be “private”. Use default values for all other fields.

```
add trunk-group 1                                           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
```

2.7. Administering Numbering Plan

SIP Users registered to Session Manager need to be added to either the private or public numbering table on Communication Manager Feature Server. For the sample configuration, private numbering was used and all extension numbers were unique within the private 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 Feature Server, use the **change private-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 the Communication Manager Feature Server.

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

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

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

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 Feature 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 55002 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 55002                                            Page   1 of   6

                             STATION

Extension: 55002                          Lock Messages? n          BCC: 0
     Type: 9630SIP                        Security Code: 123456      TN: 1
     Port: S00006                       Coverage Path 1: 1          COR: 1
     Name: SIL Video Lab - 1020         Coverage Path 2:            COS: 1
                                        Hunt-to Station:
STATION OPTIONS
                                            Time of Day Lock Table:
              Loss Group: 19
                                                  Message Lamp Ext: 55002
        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 55002                                                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 **pPage 6**, set:

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

```
change station 55002                                                    Page   6 of  6

                             STATION
SIP FEATURE OPTIONS
         Type of 3PCC Enabled: None
                    SIP Trunk: 1
```

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 55002                Page   1 of  3

                STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk      Config  Dual
 Extension                 Prefix                     Selection  Set     Mode
 55002            OPS          -      55002            1         1
                               -
                               -
```

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

- **Mapping Mode**: “both”
- **Calls Allowed:** “all”

```
change off-pbx-telephone station-mapping 55002                Page   2 of  3

                STATIONS WITH OFF-PBX TELEPHONE INTEGRATION


 Station       Appl    Call         Mapping    Calls      Bridged     Location
 Extension     Name    Limit        Mode       Allowed    Calls
 55002         OPS      1           both       all         none
                               -
```

2.10. Save Translations

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

Note: After a change on Communication Manager Feature Server which alters the dial plan, synchronization between Communication Manager Feature 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 Feature 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.

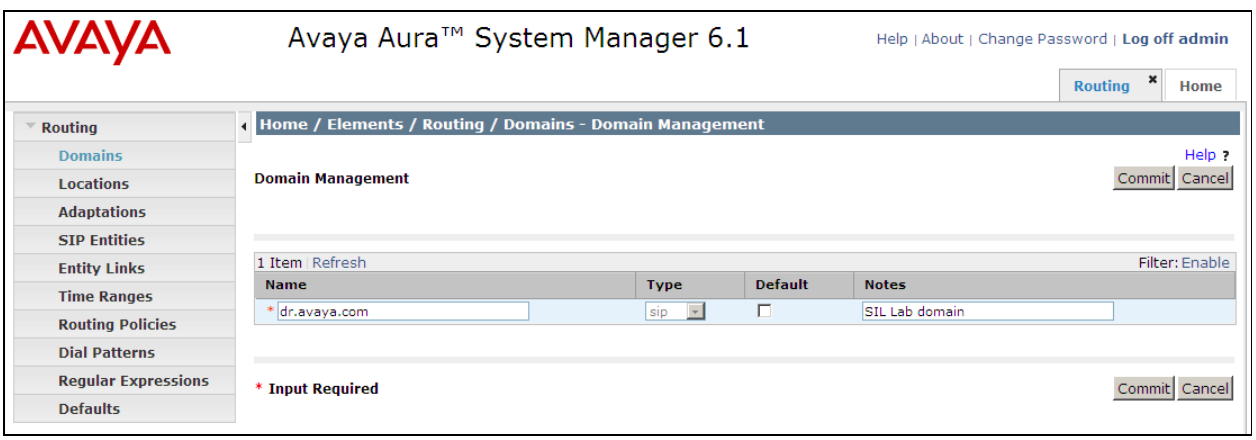
Log in with the appropriate credentials. Once logged in select the **Routing** Link under the **Elements** column. Select a specific item such as **Domains**.

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.

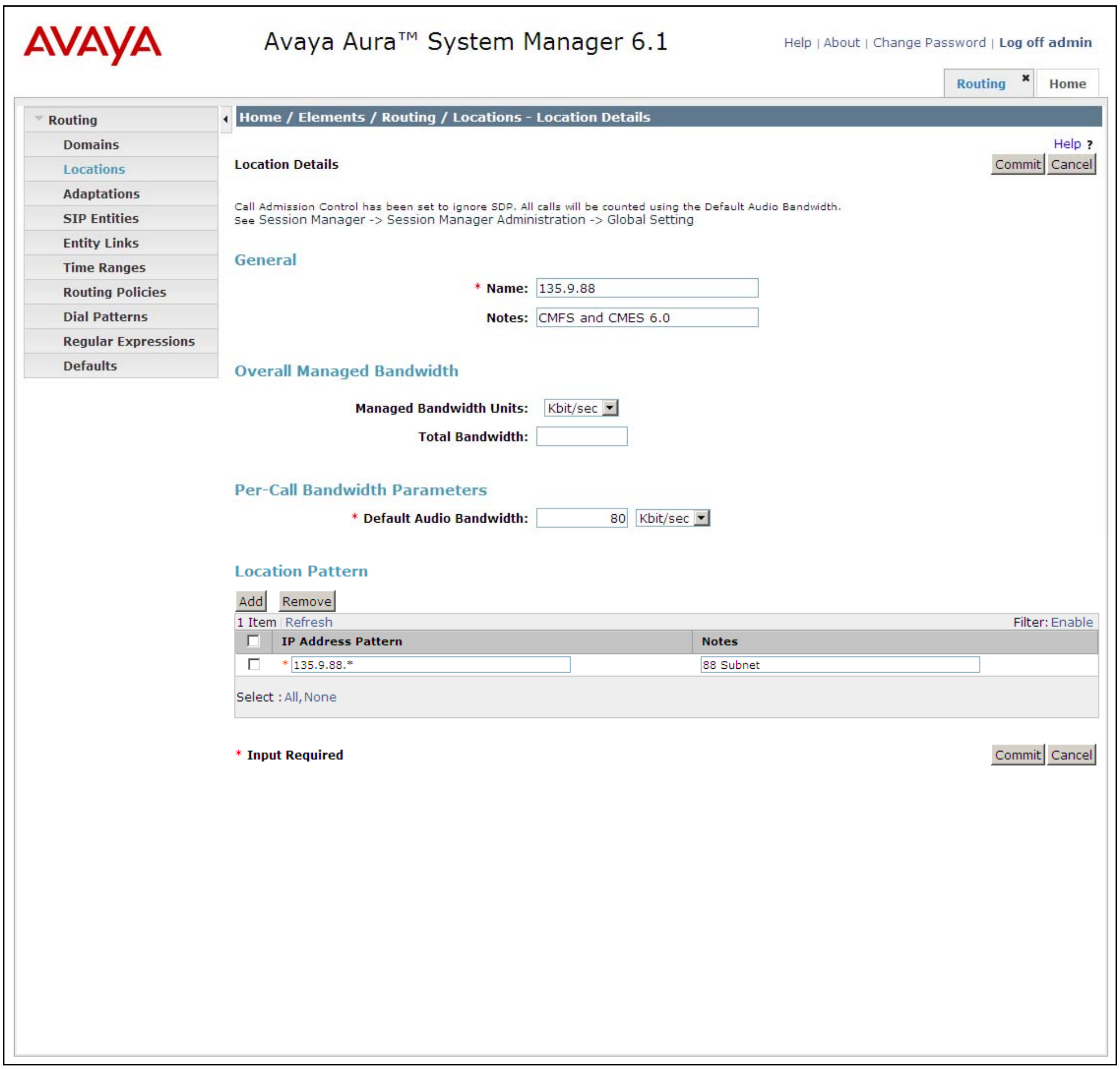


3.2. Define Locations

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 Feature Server in the sample configuration.

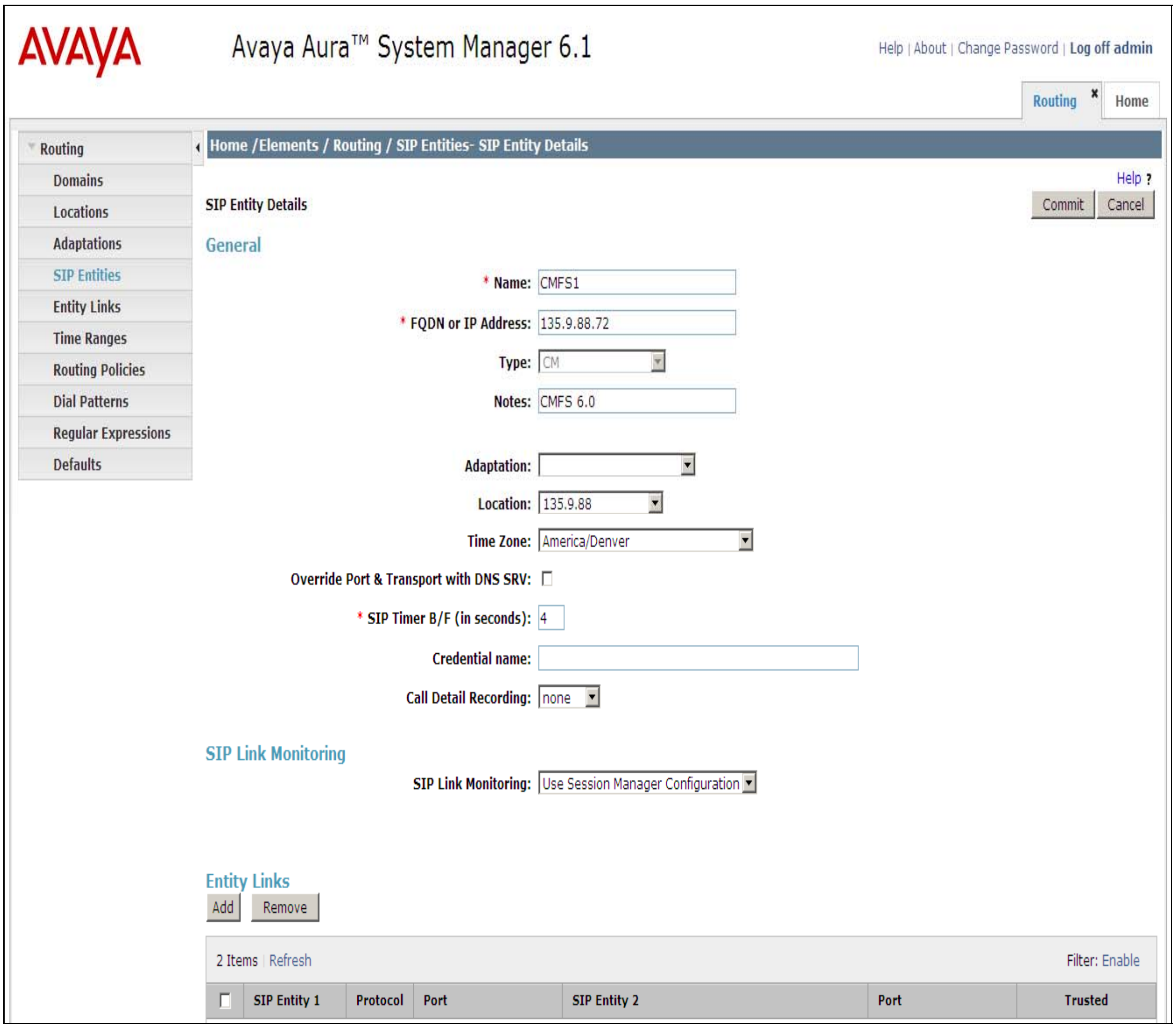


3.3. Add Avaya Aura® Communication Manager Feature Server

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

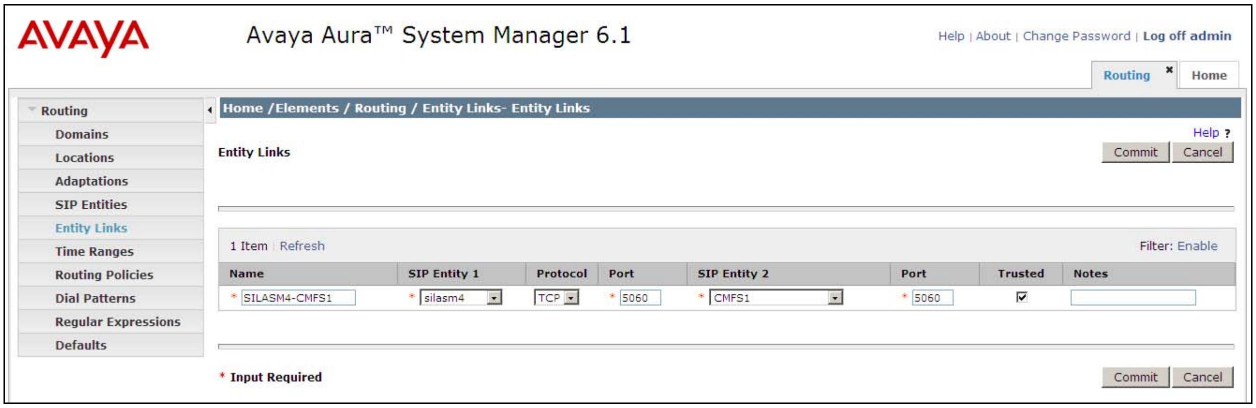
3.3.1. Define SIP Entities for Avaya Aura® Communication Manager Feature Server

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



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

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



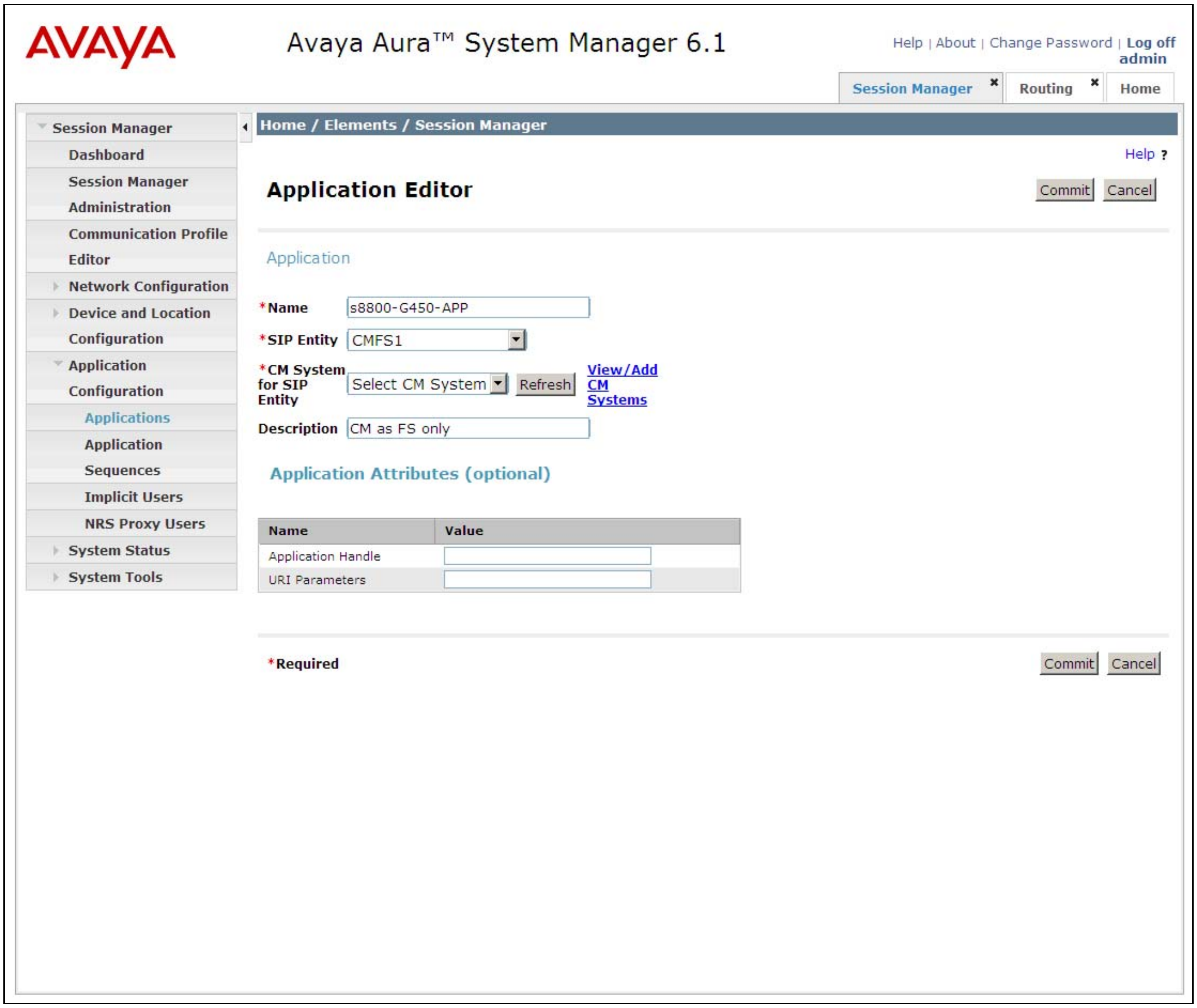
3.3.3. Define Routing Policy for Avaya Aura® Communication Manager Feature Server

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

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

To define the Avaya Aura® Communication Manager Feature 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.

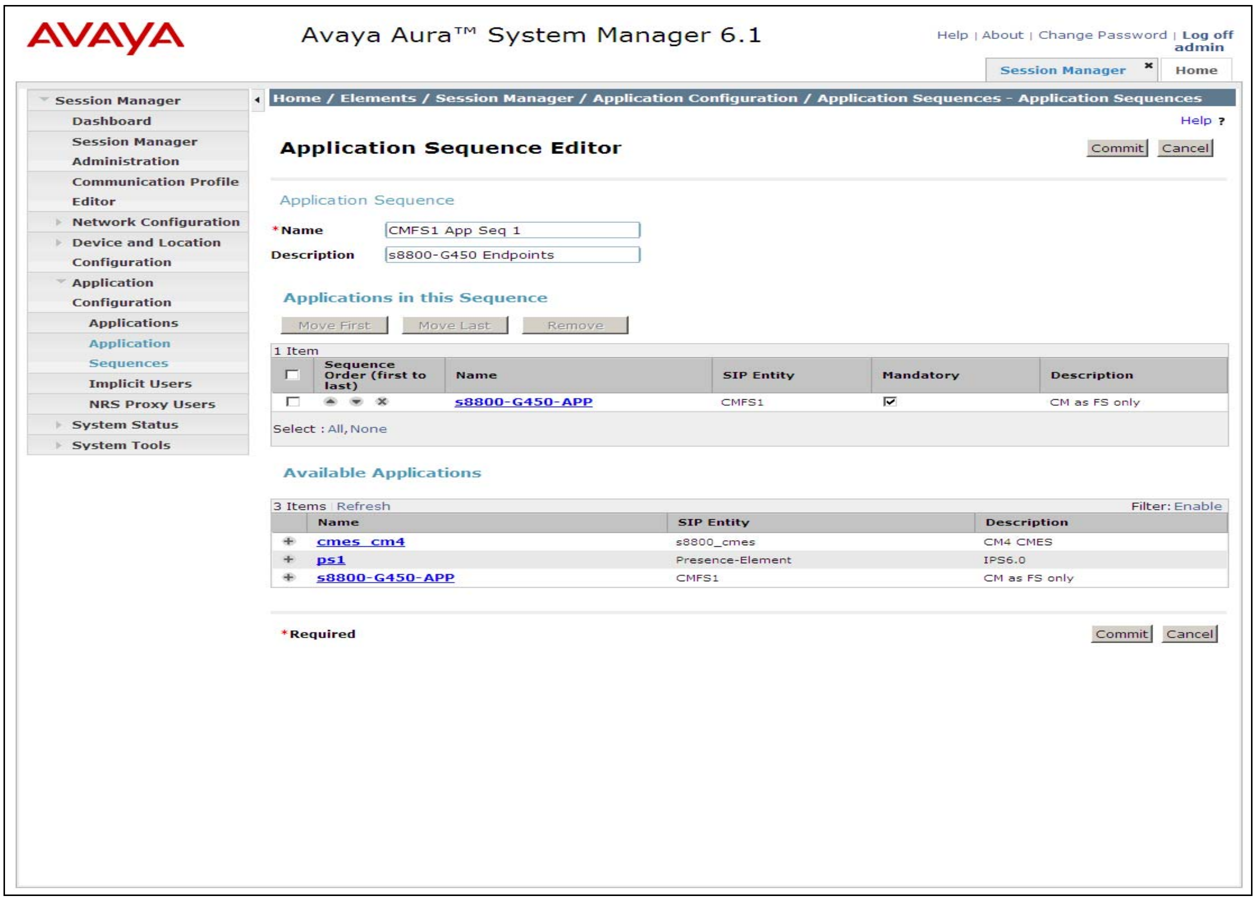


3.3.5. Define Application Sequences for Avaya Aura® Communication Manager Feature Server

To define the Avaya Aura® Communication Manager Feature Server Application Sequences,

- **Elements -> Session Manager → Application Configuration → Application Sequences**
  - o Click **New** (Not shown)
  - o Under *Name*, enter a name of the Application Sequence.
  - o Under *Description*, enter a description if desired.
  - o 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.
  - o Click **Commit** to save.

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

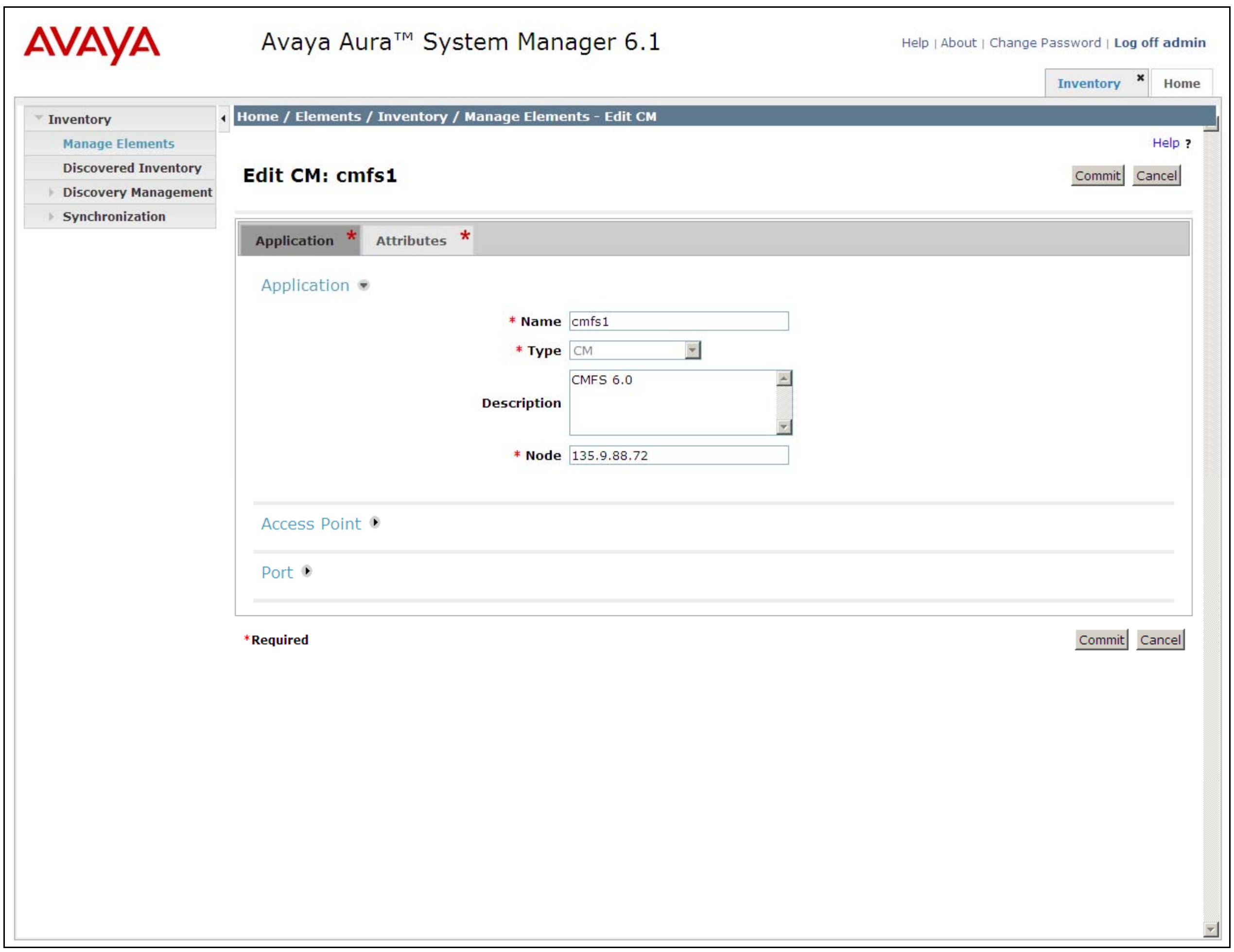


3.3.6. Define Avaya Aura® Communication Manager Feature as an Administrable Entity

Before adding SIP users, the Avaya Aura® Communication Manager Feature 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 Feature Server when new SIP users are added.

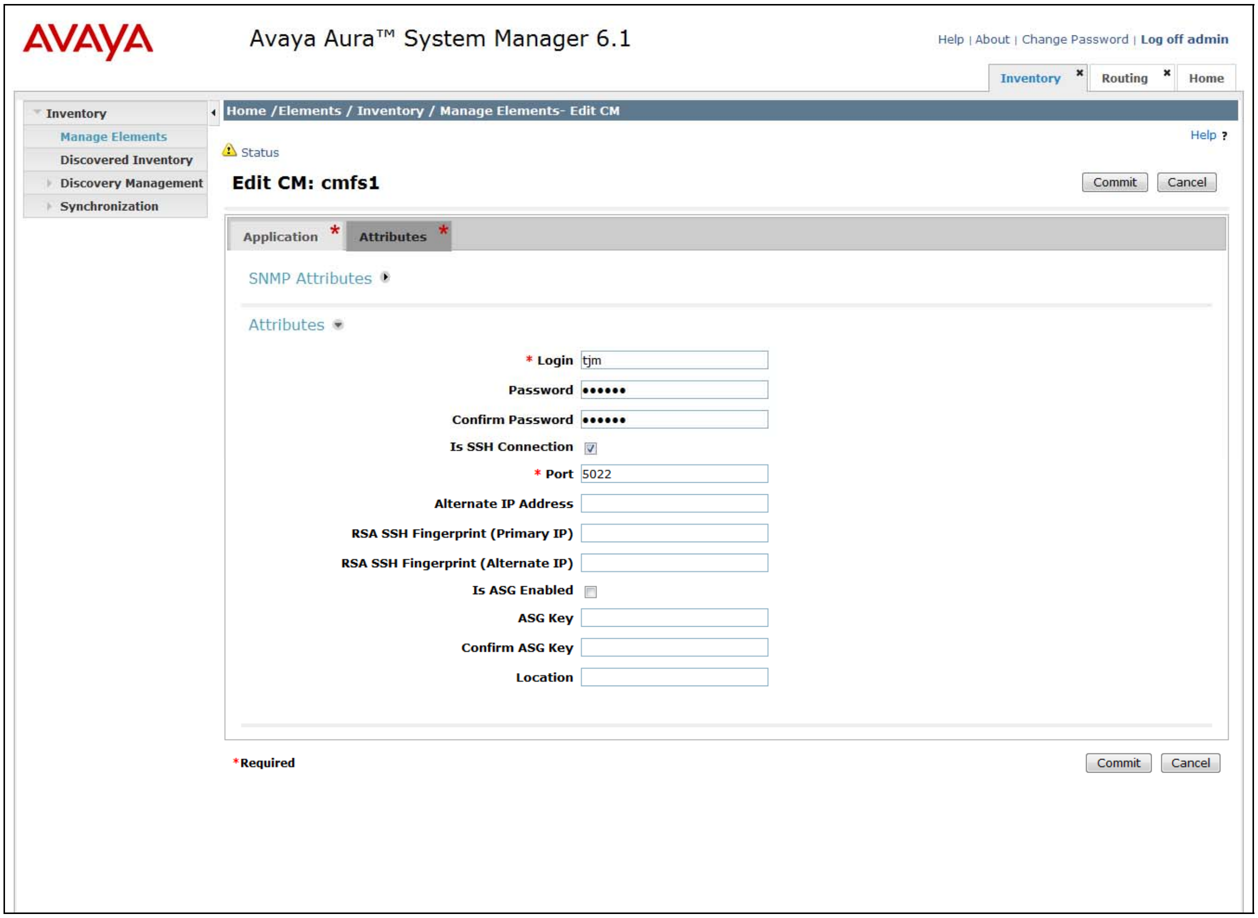
To define Avaya Aura® Communication Manager Feature Server as an administrable entity,

- **Elements -> Inventory -> Manage Elements** (**Application** tab)
  - Click **New** (Not shown)
  - Under *Name*, enter an identifier for Communication Manager Feature Server.
  - Under *Type* drop-down menu, select CM.
  - Under *Node*, enter the IP address of the administration interface for the Feature Server as shown below:



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

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



3.3.7. Add SIP Users

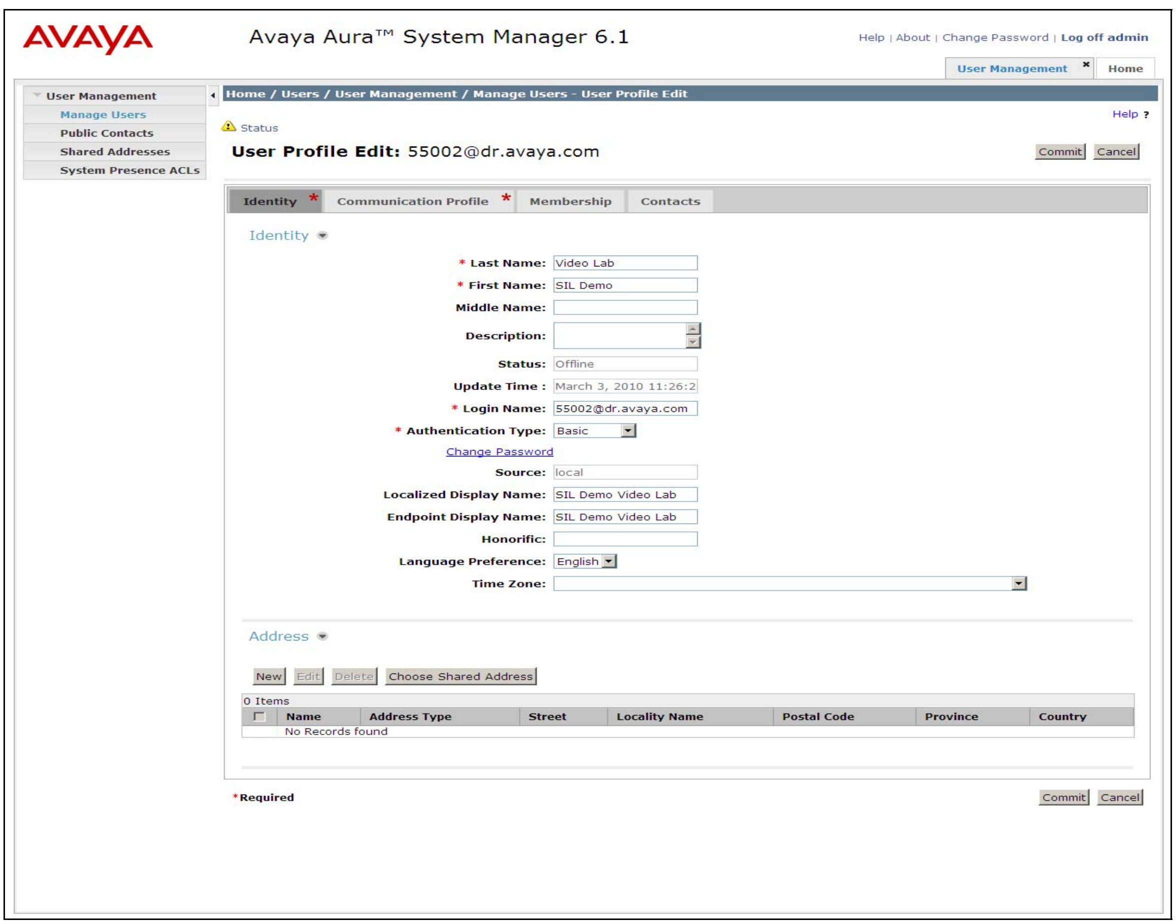
Add SIP users corresponding to the 96XX SIP stations defined in **Section 2.8.** Alternatively, use the option to automatically create station on Communication Manager when new user is added.

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:

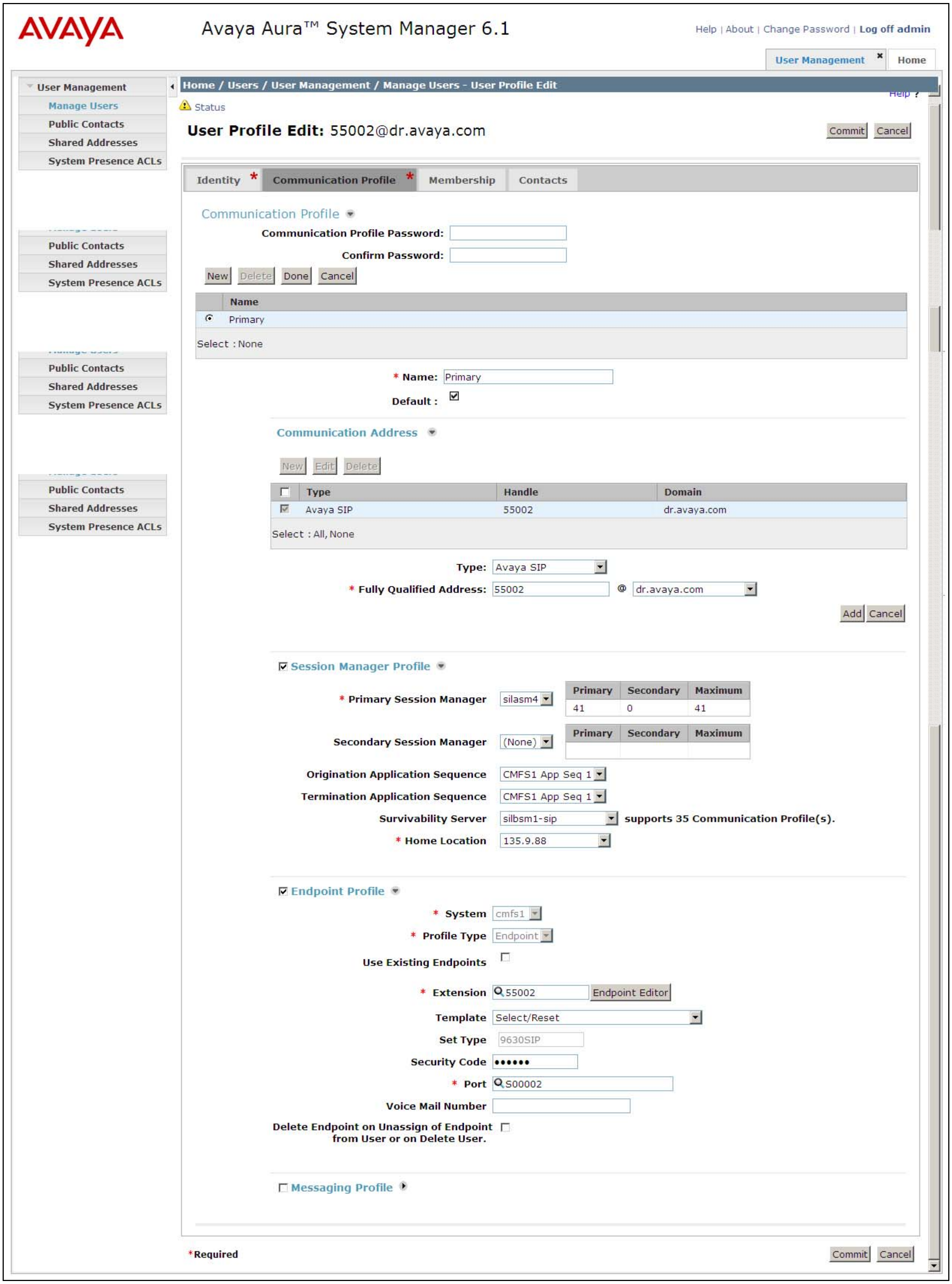
- **Primary Session Manager:** select the appropriate Session Manager instance
- **Origination Application Sequence:** enter the appropriate sequence
- **Termination Application Sequence:** enter the appropriate sequence
- **Home Location:** select the appropriate location that was created in **Section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

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

- **System:** select the SIP Entity of the Communication Manager Feature 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.


AVAYA
Avaya Aura™ System Manager 6.1
Help | About | Change Password | Log off admin
User Management
Home
User Management
Manage Users
Public Contacts
Shared Addresses
System Presence ACLs
Home / Users / User Management / Manage Users - User Profile Edit
Status
User Profile Edit: 55002@dr.avaya.com
Commit
Cancel
Identity
Communication Profile
Membership
Contacts
Communication Profile
Communication Profile Password:
Confirm Password:
New
Delete
Done
Cancel
Name
Primary
Select : None
* Name: Primary
Default :
Communication Address
New
Edit
Delete
Type
Handle
Domain
Avaya SIP
55002
dr.avaya.com
Select : All, None
Type: Avaya SIP
* Fully Qualified Address: 55002 @ dr.avaya.com
Add
Cancel
Session Manager Profile
* Primary Session Manager silasm4
Primary Secondary Maximum
41 0 41
Secondary Session Manager (None)
Primary Secondary Maximum
Origination Application Sequence CMFS1 App Seq 1
Termination Application Sequence CMFS1 App Seq 1
Survivability Server silbsm1-sip supports 35 Communication Profile(s).
* Home Location 135.9.88
Endpoint Profile
* System cmfs1
* Profile Type Endpoint
Use Existing Endpoints
* Extension 55002
Endpoint Editor
Template Select/Reset
Set Type 9630SIP
Security Code ••••••
* Port S00002
Voice Mail Number
Delete Endpoint on Unassign of Endpoint from User or on Delete User.
Messaging Profile
*Required
Commit
Cancel

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.

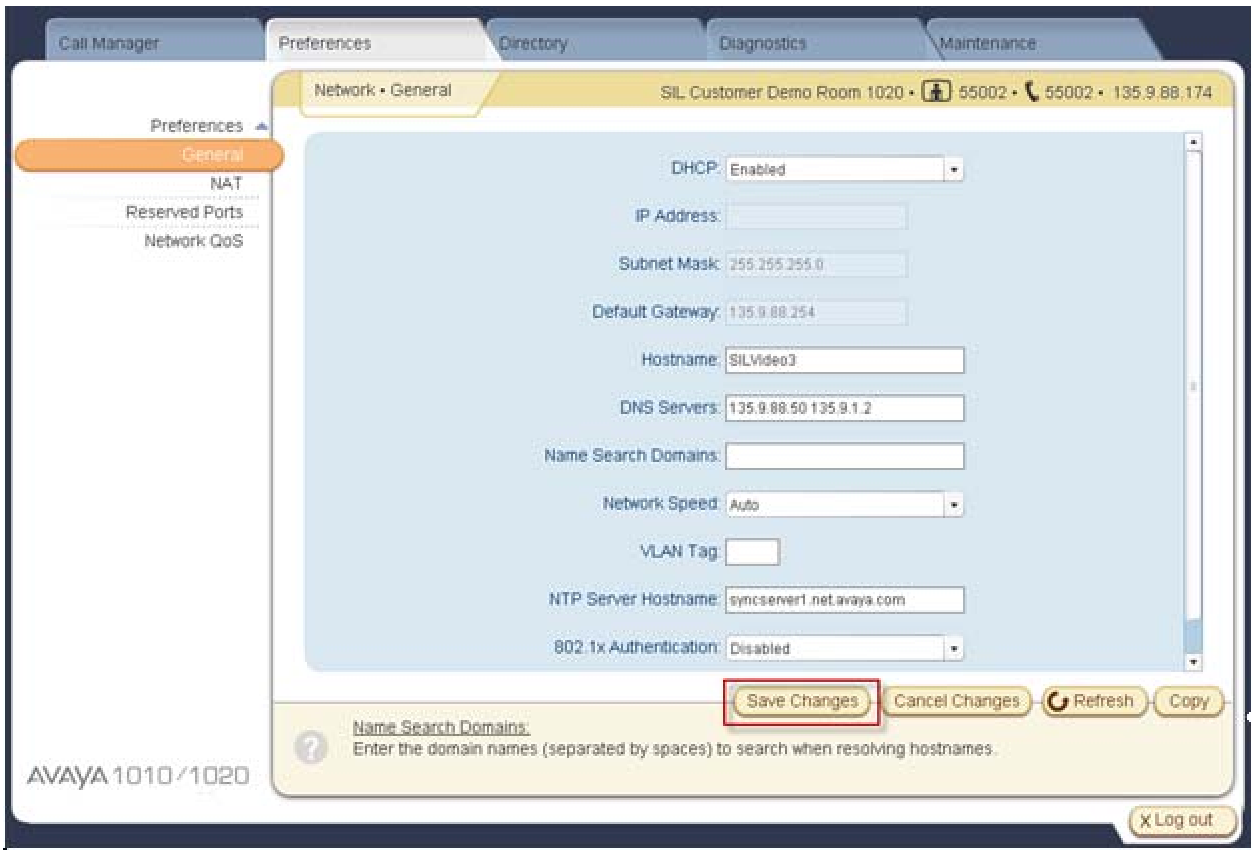
Step 1: Enter the proper login credentials and press **Submit**. Most of the Preferences can be customized to meet your needs. Mentioned below are the absolute necessary items that need to be administered to get the 10x0 up and running on the network.

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

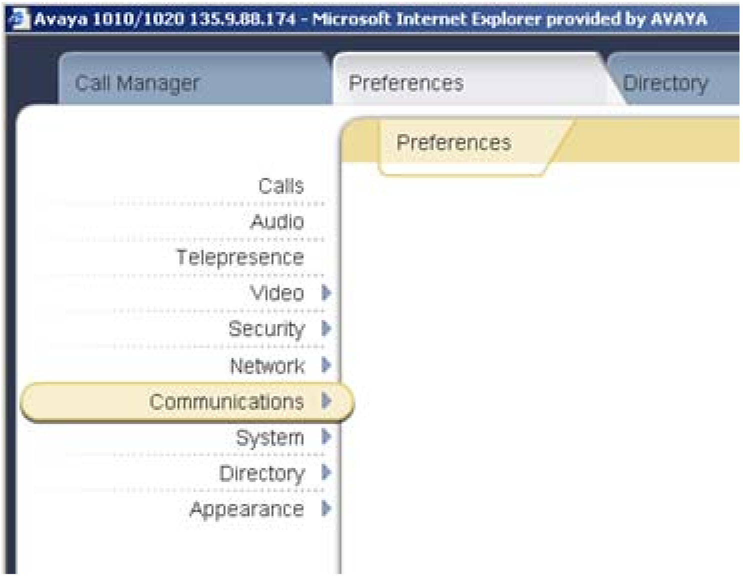
Select **General** option and enter values for the following required attributes.

- **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.

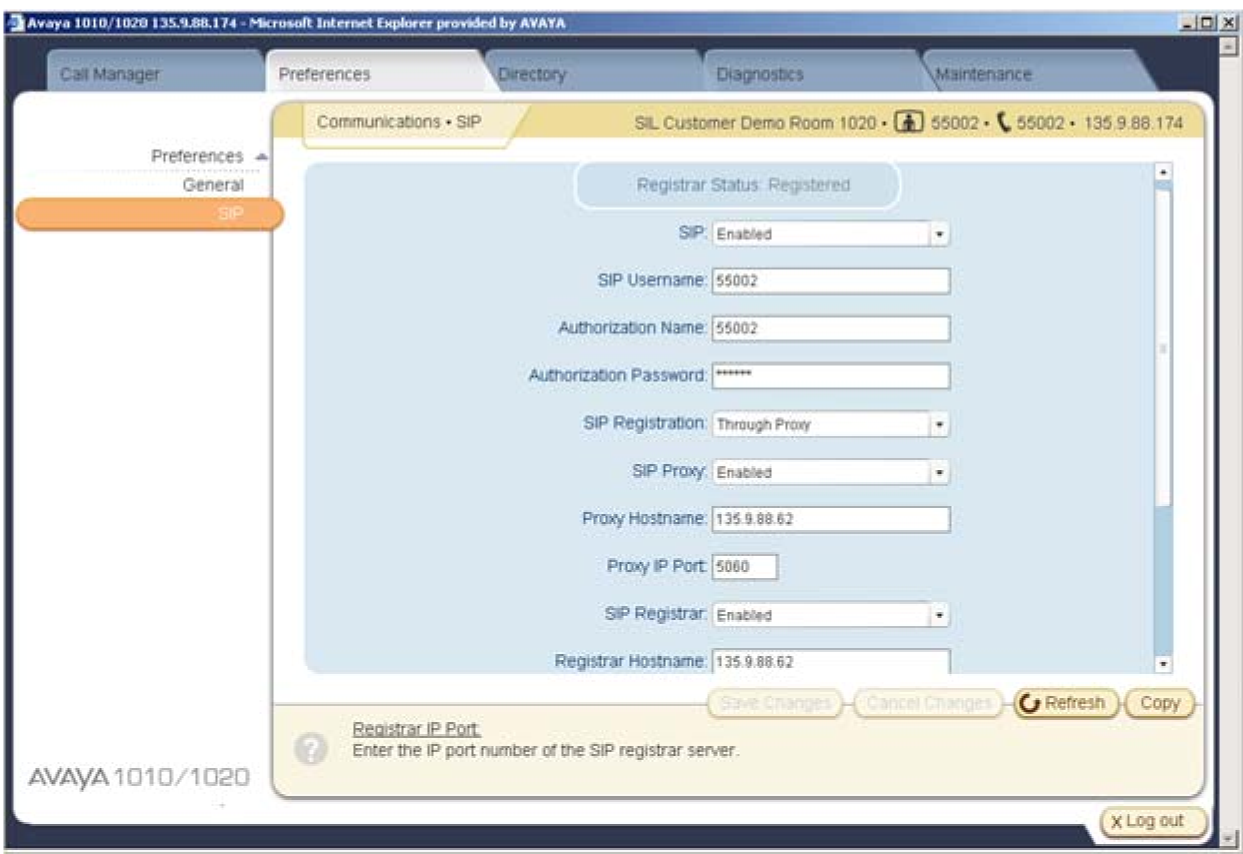


Select the **Preferences** option and select the **Communications** option.



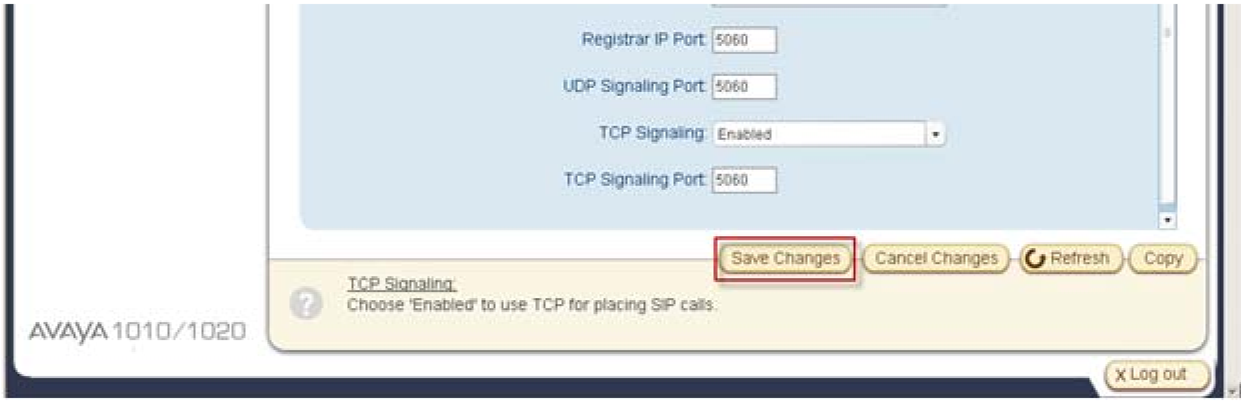
Select the **SIP** option.

- **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** unique and meaningful to the endpoint.
- **Authorization Password:** enter the SIP Server authorization password which should match the Shared Comm 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

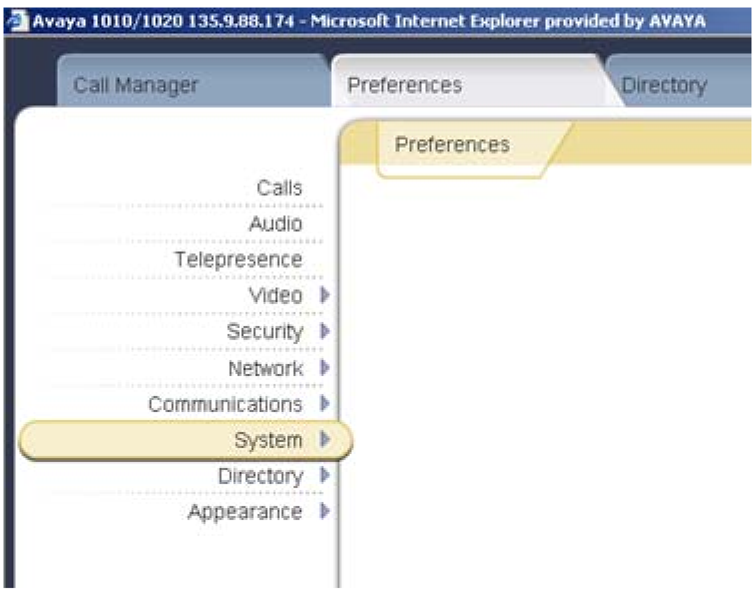


- **Registrar IP Port:** enter the IP port number of the SIP registrar server
- **UDP Signaling Port:** enter the UDP port number of the SIP configuration
- **TCP Signaling:** choose 'Enable' to use TCP for placing SIP call
- **TCP signaling Port:** enter the TCP port number of the SIP configuration

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



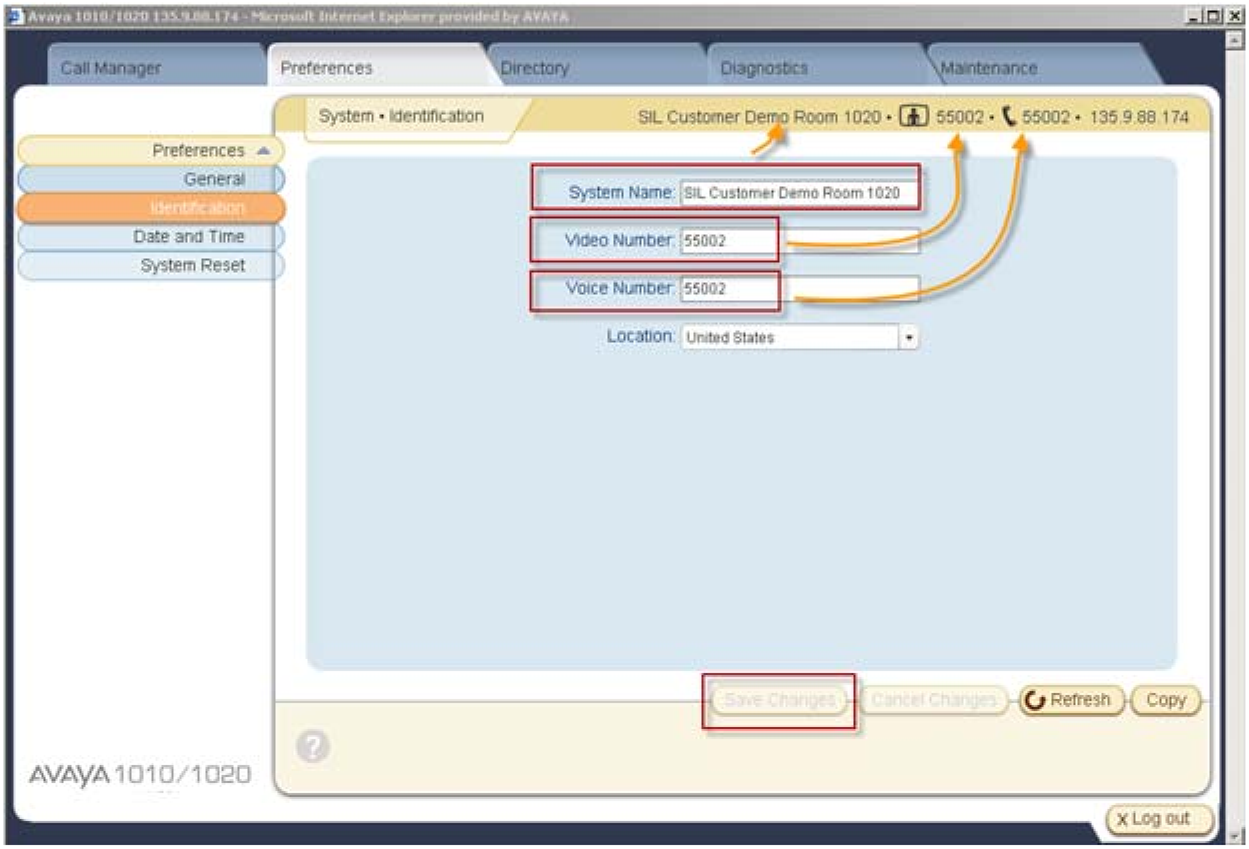
Select the **Preferences** option again and select **System**.



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.



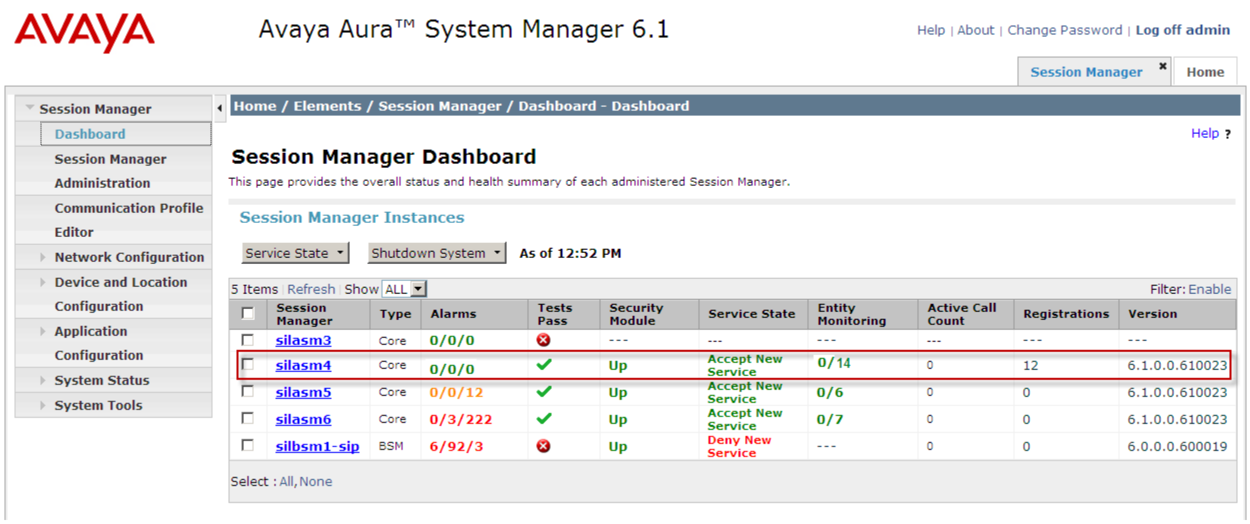
5. Verification Steps

5.1. Verify Avaya Aura® Session Manager Configuration

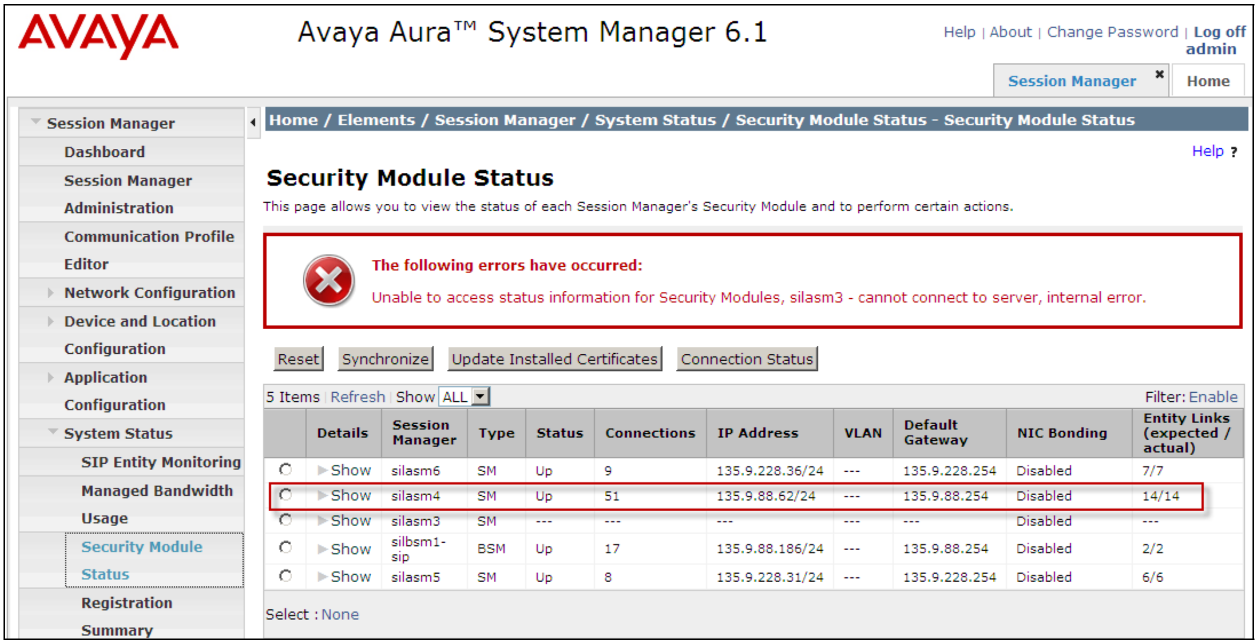
5.1.1. Verify Avaya Aura® Session Manager is Operational

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

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

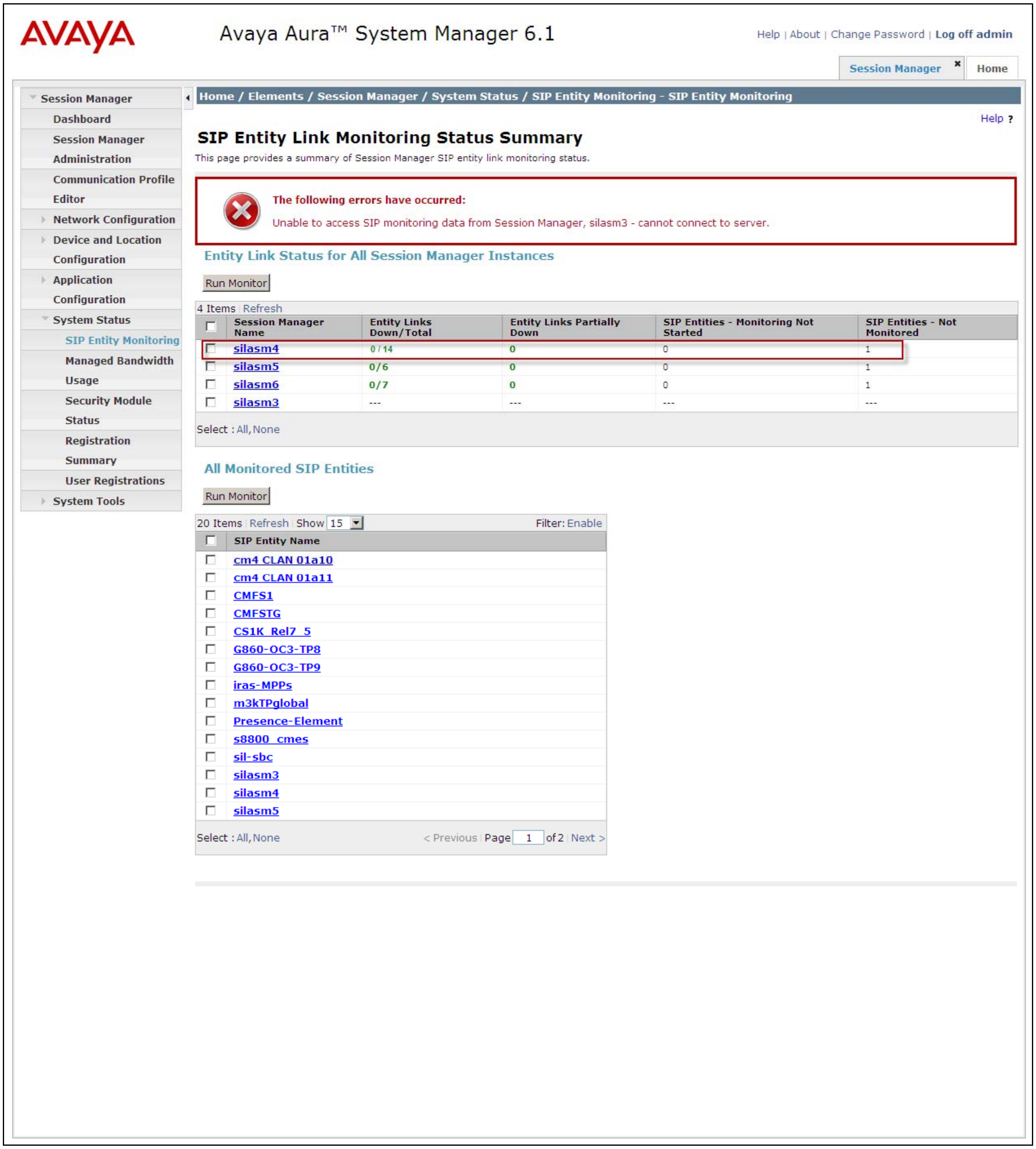


Navigate to **Elements → Session Manager → System Status → Security Module Status** to view more detailed status information on the status of Security Module for Session Manager. Verify the **Status** column displays "**Up**" as shown below.



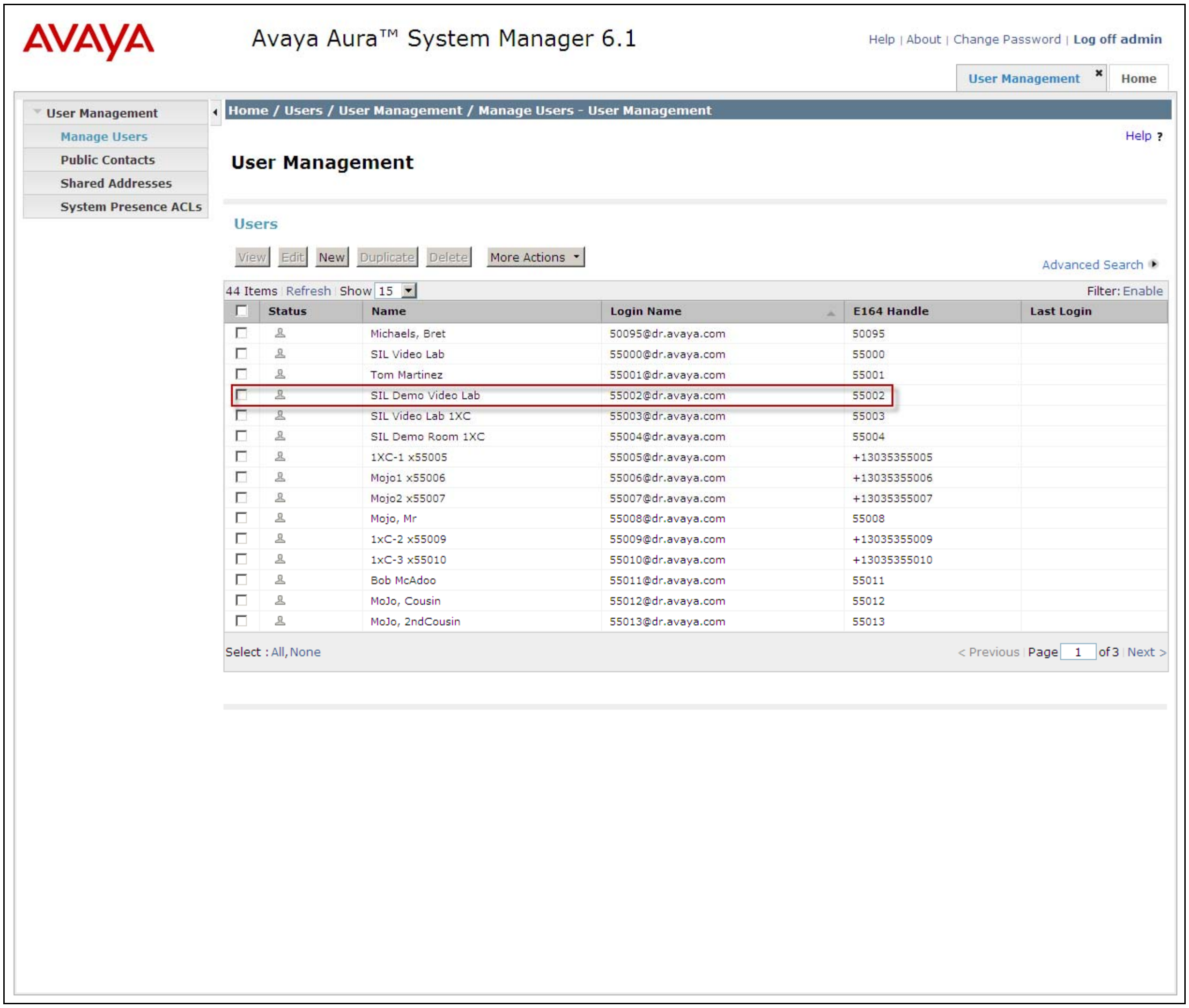
5.1.2. Verify SIP Link Status

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

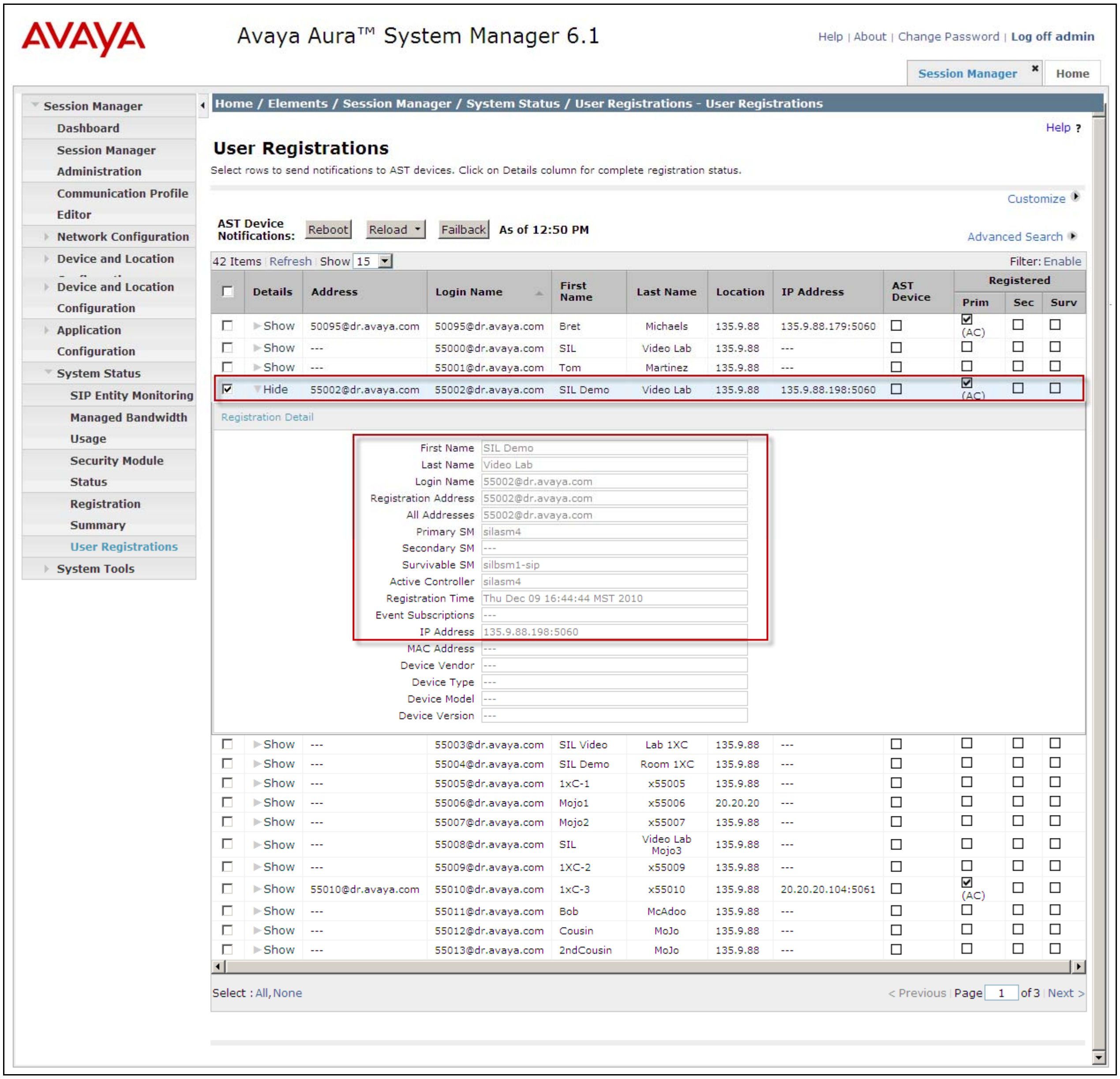


5.1.3. Verify Registrations of SIP Endpoints

Verify SIP users have been created in the Session Manager. In the sample configuration, Extension 55002 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 Feature 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 1
                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy
0001/001 T00001   in-service/idle    no
0001/002 T00002   in-service/idle    no
0001/003 T00003   in-service/idle    no
0001/004 T00004   in-service/idle    no
0001/005 T00005   in-service/idle    no
0001/006 T00006   in-service/idle    no
0001/007 T00007   in-service/idle    no
0001/008 T00008   in-service/idle    no
0001/009 T00009   in-service/idle    no
0001/010 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 1
                         STATUS SIGNALING GROUP

        Group ID: 1
      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 Feature Server as shown below:

```
list trace tac #001                                         Page   1
                                LIST TRACE
time            data

10:53:34 TRACE STARTED 12/02/2010 CM Release String cold-00.1.510.1-18599
10:54:29 SIP<INVITE sip:55002@dr.avaya.com;transport=tcp;user=ph
10:54:29 SIP<one SIP/2.0
10:54:29     dial
10:54:29     term trunk-group 1    cid 0xb9
10:54:29     dial
10:54:29     seize trunk-group 1 member 21  cid 0xb9
10:54:29     Calling Number & Name NO-CPNumber NO-CPName
10:54:29     Proceed trunk-group 1 member 21  cid 0xb911:01:07     Setup
10:54:29 SIP>SIP/2.0 180 Ringing
10:54:29     Alert trunk-group 1 member 21  cid 0xb9
10:54:31     active trunk-group 1 member 21  cid 0xb9
10:54:31     G711MU ss:off ps:20
             rgn:1 [135.9.88.216]:60656
             rgn:1 [135.9.88.174]:60142
10:54:31     G711MU ss:off ps:20
             rgn:1 [135.9.88.174]:60142
             rgn:1 [135.9.88.216]:60656
10:54:31 SIP>SIP/2.0 200 OK
10:54:31     Video: H264 [135.9.88.216]:60658
10:54:31     Video: H264 [135.9.88.174]:60144
                      logChl:110 sessId:2 bw:21760 tx/rx:11520
10:54:31     Video: H264 [135.9.88.174]:60144
10:54:31     Video: H264 [135.9.88.216]:60658
                      logChl:110 sessId:2 bw:21760 tx/rx:11520
10:54:31 SIP<ACK sip:55002@135.9.88.72;transport=tcp SIP/2.0
10:54:37 SIP<BYE sip:55002@135.9.88.72;transport=tcp SIP/2.0
10:54:37 SIP>SIP/2.0 200 OK
10:54:37     idle station    55002 cid 0xb9
```

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 6663000                                            Page   1
                                LIST TRACE
time            data

10:58:38 TRACE STARTED 08/21/2010 CM Release String cold-00.1.510.1-18599
10:59:07     active station    55002 cid 0xbc
10:59:07 SIP>INVITE sip:55002@dr.avaya.com SIP/2.0
10:59:07     dial
10:59:07     term trunk-group 1    cid 0xbc
10:59:07     dial
10:59:07     seize trunk-group 1 member 23  cid 0xbc
10:59:07     Setup digits 55002
10:59:07     Calling Number & Name 55002 SIL Demo Vide
10:59:07 SIP<SIP/2.0 100 Trying
10:59:07     Proceed trunk-group 1 member 23  cid 0xbc
10:59:07 SIP<INVITE sip:55002@dr.avaya.com SIP/2.0
10:59:07 SIP>SIP/2.0 100 Trying
10:59:07 SIP>SIP/2.0 180 Ringing
10:59:07 SIP<SIP/2.0 180 Ringing
10:59:07     Alert trunk-group 1 member 23  cid 0xbc
10:59:09 SIP>SIP/2.0 200 OK
10:59:09 SIP<SIP/2.0 200 OK
10:59:09     active trunk-group 1 member 23  cid 0xbc
10:59:09     G711MU ss:off ps:20
             rgn:1 [135.9.88.216]:60664
             rgn:1 [135.9.88.174]:60150
10:59:09     G711MU ss:off ps:20
             rgn:1 [135.9.88.174]:60150
             rgn:1 [135.9.88.216]:60664
10:59:09     Video: H264 [135.9.88.216]:60666
10:59:09     Video: H264 [135.9.88.174]:60152
                      logChl:110 sessId:2 bw:21760 tx/rx:11520
10:59:09     Video: H264 [135.9.88.174]:60152
10:59:09     Video: H264 [135.9.88.216]:60666
                      logChl:110 sessId:2 bw:21760 tx/rx:11520
10:59:09 SIP>ACK sip:55002@135.9.88.72;transport=tcp SIP/2.0
10:59:10 SIP<ACK sip:55002@135.9.88.72;transport=tcp SIP/2.0
11:00:43 SIP>BYE sip:55002@135.9.88.72;transport=tcp SIP/2.0
11:00:43 SIP<BYE sip:55002@135.9.88.72;transport=tcp SIP/2.0
11:00:43 SIP>SIP/2.0 200 OK
11:00:43     idle trunk-group 1 member 23  cid 0xbc
```

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 (CMFS) to another 1020/1030/1040 video endpoint registered on SM (CMFS). 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 (CMFS) to another 1020/1030/1040 video endpoint registered on SM (CMFS). 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 (CMFS) to another 1020/1030/1040 video endpoint registered on SM (CMFS). 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 (CMFS) to another 1020/1030/1040 video endpoint registered on SM (CMFS). 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 (CMFS) to another 1020/1030/1040 video endpoint registered on SM (CMFS). 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 (CMFS) to another 1020/1030/1040 video endpoint registered on SM (CMFS). 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 (CMFS) to another 1020/1030/1040 video endpoint registered on SM (CMFS). 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 (CMFS) to another 1020/1030/1040 video endpoint registered on SM (CMFS). 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)
CMFS	Communication Manager Feature 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 Feature 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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