Avaya Solution & Interoperability Test Lab

Application Notes for LifeSize Team 220 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the LifeSize Team 220 video system with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using a SIP interface. LifeSize Team 220 supports HD video and consists of the following components: LifeSize camera, phone, codec device, and remote control. It also requires a 3rd party monitor display, preferably one that supports HD video and has an HDMI interface.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the steps required to integrate the LifeSize Team 220 video system with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using a SIP interface. LifeSize Team 220 supports HD video and consists of the following components: LifeSize camera, phone, codec device, and remote control. It also requires a 3rd party monitor display, preferably one that supports HD video and has an HDMI interface.

2. General Test Approach and Test Results

To verify interoperability of the LifeSize Team 220 video system with Communication Manager and Session Manager, voice and video calls were made between LifeSize Team 220, other LifeSize video systems (see **Section 4**), Avaya one-X® Communicator (SIP and H.323 versions), and the Avaya Desktop Video Device. Additional features were exercised on the Team 220, including auto-answer, Do Not Disturb, and audio mute.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1 Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of LifeSize Team 220 video system with Session Manager.
- Video calls between LifeSize Team 220 and other LifeSize video systems, Avaya one-X® Communicator (SIP and H.323 versions), and the Avaya Desktop Video Device.
- Voice calls between LifeSize Team 220 and other LifeSize video systems, Avaya one-X® Communicator (SIP and H.323 versions), and the Avaya Desktop Video Device.
- G.711 codec support.
- Caller ID display on Avaya and LifeSize endpoints.
- Auto-answer and Do Not Disturb on Team 220 for incoming video calls.
- Audio mute on Team 220 and Avaya endpoints for video and voice calls.
- Voice call transfer from an Avaya endpoint to another endpoint while a voice call is active with Team 220.
- Video mute from Avaya endpoints to Team 220. Initiating video mute from Team 220 is currently not supported.
- Video call transfer from Avaya endpoints to Team 220. Initiating a call transfer from Team 220 is currently not supported.
- Proper system recovery after a restart of Team 220 and loss of IP connectivity.

2.2 Test Results

All test cases passed with the following observations:

- If Avaya one-X Communicator places a video call on hold with a LifeSize video system, only the audio portion of the video call is restored after taking the call off hold. Video is no longer available after the hold/resume. In addition to simple hold/resume scenarios, this issue also impacts other call scenarios were a call is placed on hold, such as transfers and conferences. This issue has been fixed in LifeSize firmware version 4.11.6 (2).
- If Avaya one-X Communicator places a video call on hold with a LifeSize video system, a “Call Status” screen appears in the middle of the LifeSize monitor display (blocking the view of the video call behind it). The “Call Status” screen cannot be removed until the call is terminated. This issue has been fixed in LifeSize firmware version 4.11.6 (2).

2.3 Support

For technical support on the Team video system, contact LifeSize Support via phone or website.

- **Phone:** (877) LIFESIZE or (512) 347-9300
- **Web:** http://www.lifesize.com/Support/Get_support.aspx

3. Reference Configuration

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

- Avaya Aura® Communication Manager running on an Avaya S8300D Server with a G450 Media Gateway. Communication Manager was configured as an Evolution Server.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones and video endpoints.
- Avaya Aura® System Manager used to configure Session Manager.

In addition, a LifeSize Team 220 video system, other LifeSize video systems (see **Section 4**), Avaya one-X Communicator (SIP and H.323 versions), and an Avaya Desktop Video Device were used for video calls. All SIP devices registered with Session Manager and were configured as Off-PBX Stations (OPS) on Communication Manager.

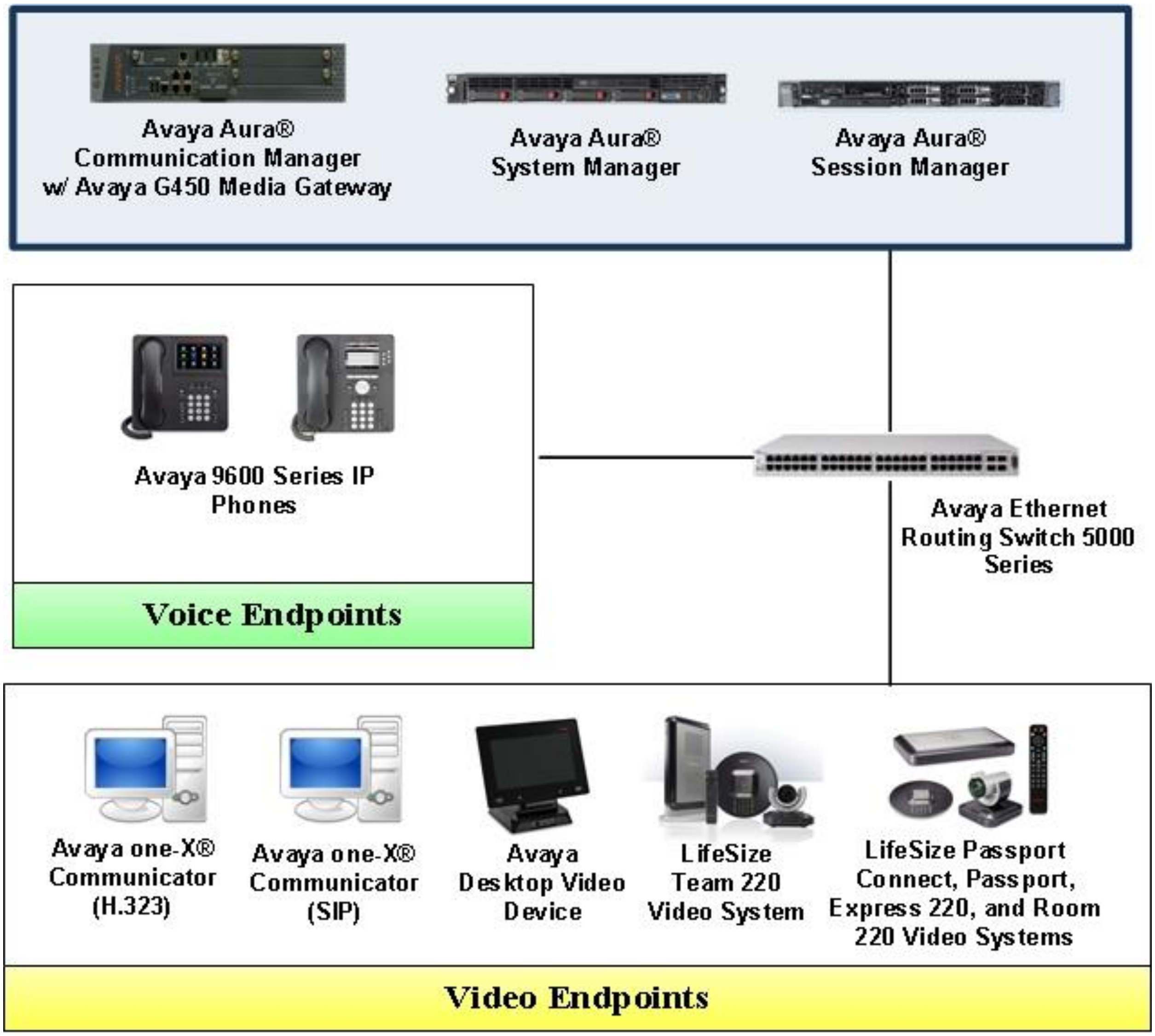


Figure 1: Avaya SIP Network with the LifeSize Team 220 Video System

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
HP ProLiant DL360 G7 Server	Avaya Aura® Session Manager 6.1 SP7
Dell™ PowerEdge™ R610 Server	Avaya Aura® System Manager 6.1 SP8
Avaya S8300D Server with an Avaya G450 Media Gateway	Avaya Aura® Communication Manager 6.0.1 (R016x.00.1.510.1-19736)
Avaya one-X® Communicator	6.1.3.09-SP3-Patch3-35953
Avaya 9600 Series IP Telephones • 96x0 (SIP) • 96x1 (SIP	Avaya one-X® Deskphone Edition SIP 2.6.7 Avaya one-X® Deskphone Edition SIP 6.1
Avaya Desktop Video Device	1.1.1
LifeSize Passport	4.11.1 (16)
LifeSize Passport Connect	4.11.1 (16)
LifeSize Express 220	4.11.1 (16)
LifeSize Team 220	4.11.1 (16)
LifeSize Room 220	4.11.1 (16)

5. Configure Avaya Aura® Communication Manager

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

- Verify Communication Manager license
- Configure Team 220 as an Off-PBX Station (OPS)
- Configure a SIP trunk between Communication Manager and Session Manager

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

5.1 Verify OPS and SIP Trunk Capacity

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

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

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

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

                                                                     USED
                                Platform Maximum Ports: 65000 409
                                      Maximum Stations: 41000 51
                              Maximum XMOBILE Stations: 41000 0
                    Maximum Off-PBX Telephones - EC500: 41000 0
                    Maximum Off-PBX Telephones -   OPS: 41000 19
                    Maximum Off-PBX Telephones - PBFMC: 41000 0
                    Maximum Off-PBX Telephones - PVFMC: 41000 0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   0




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

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

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 77
           Maximum Concurrently Registered IP Stations: 18000 5
             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 8
                   Maximum Video Capable IP Softphones: 18000 3
                       Maximum Administered SIP Trunks: 24000 180
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   1
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

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

5.2 Configure SIP Trunk

In the **IP Node Names** form, assign a host name and IP address for the Session Manager SIP interface. Note the processor host name of Communication Manager. The host names will be used throughout the other configuration screens of Communication Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
SM_21_31            10.64.21.31
default             0.0.0.0
msgserver           10.64.21.41
procr               10.64.21.41
procr6              ::


( 14 of 14   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling group.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to the LifeSize Team endpoint. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. The default settings of the **IP Codec Set** form are shown below.

```
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:
```

Configure **Page 2** of the **IP Codec Set** form as follows (note that other values are possible for the maximum call rates).

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

                          IP Codec Set

                              Allow Direct-IP Multimedia? y
              Maximum Call Rate for Direct-IP Multimedia: 10240:Kbits
     Maximum Call Rate for Priority Direct-IP Multimedia: 10240:Kbits

                    Mode                Redundancy
    FAX             t.38-standard        0
    Modem           off                  0
    TDD/TTY         US                   3
    Clear-channel   n                    0
```

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the Signaling Group form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- Set the **Transport Method** field to *tls*.
- Set the **IP Video** field to *y*. This is an important setting required for video calls.
- Specify the processor of Communication Manager and the Session Manager SIP interface as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values were taken from the **IP Node Names** form.
- Ensure that the TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. Communication Manager supports DTMF transmission using RFC 2833.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- Set the **Initial IP-IP Direct Media** field to *y*.
- The default values for the other fields may be used.

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

 Group Number: 1             Group Type: sip
  IMS Enabled? n       Transport Method: tls
        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: SM_21_31
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region: 1

Far-end Domain:
                                             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): 20
```

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to SIP endpoints. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*. Set the **Member Assignment Method** to *auto*. Specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

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

Group Number: 1                  Group Type: sip           CDR Reports: y
  Group Name: to SM_21_31              COR: 1        TN: 1        TAC: 101
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 50
```

On **Page 3** of the trunk group form, set the **Numbering Format** field to *unk-pvt* (other configurations are possible). This field specifies the format of the calling party number sent to the far-end.

```
add trunk-group 1                                               Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                       Maintenance Tests? y



                     Numbering Format: unk-pvt
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n


                                Modify Tandem Calling Number: no




 Show ANSWERED BY on Display? y
```

Configure the **Private Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '5' whose calls are routed over any trunk group, including SIP trunk group "1", have the extension sent to the far-end for display purposes.

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

Ext Ext              Trk         Private          Total
Len Code             Grp(s)      Prefix           Len
 5  5                                             5     Total Administered: 2
                                                           Maximum Entries: 540
```

5.3 Configure Station for LifeSize Team 220

The **station** and **off-pbx-telephone station-mapping** configuration shown in this section was automatically performed by creating the **User** in Session Manager as described in **Section 6.7**. In this section, simply verify the settings. Note that the **User** has to be added in Session Manager first before it can be viewed on Communication Manager. Alternatively, this configuration could have also been performed manually.

Use the **display station** command to view the station created for the LifeSize Team 220 video system and verify the settings in bold. Note that the **IP Video** field must be set to *y*.

```
display station 53165                                               Page   1 of   6
                                     STATION

Extension: 53165                        Lock Messages? n            BCC: M
     Type: 9630SIP                      Security Code: 123456        TN: 1
     Port: S00006                     Coverage Path 1:              COR: 1
     Name: 53165, LS Team             Coverage Path 2:              COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19
                                                Message Lamp Ext: 53165

        Display Language: english                 Button Modules: 0

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

                                                        IP Video? y
```

Use the **display off-pbx-telephone station-mapping** command to view the mapping of the Communication Manager extensions (e.g., 53165) to the same extension configured in System Manager. Verify the field values shown. For the sample configuration, the **Trunk Selection** field is set to *aar* so that AAR call routing is used to route calls to Session Manager. AAR call routing configuration is not shown in these Application Notes. The **Configuration Set** value can reference a set that has the default settings.

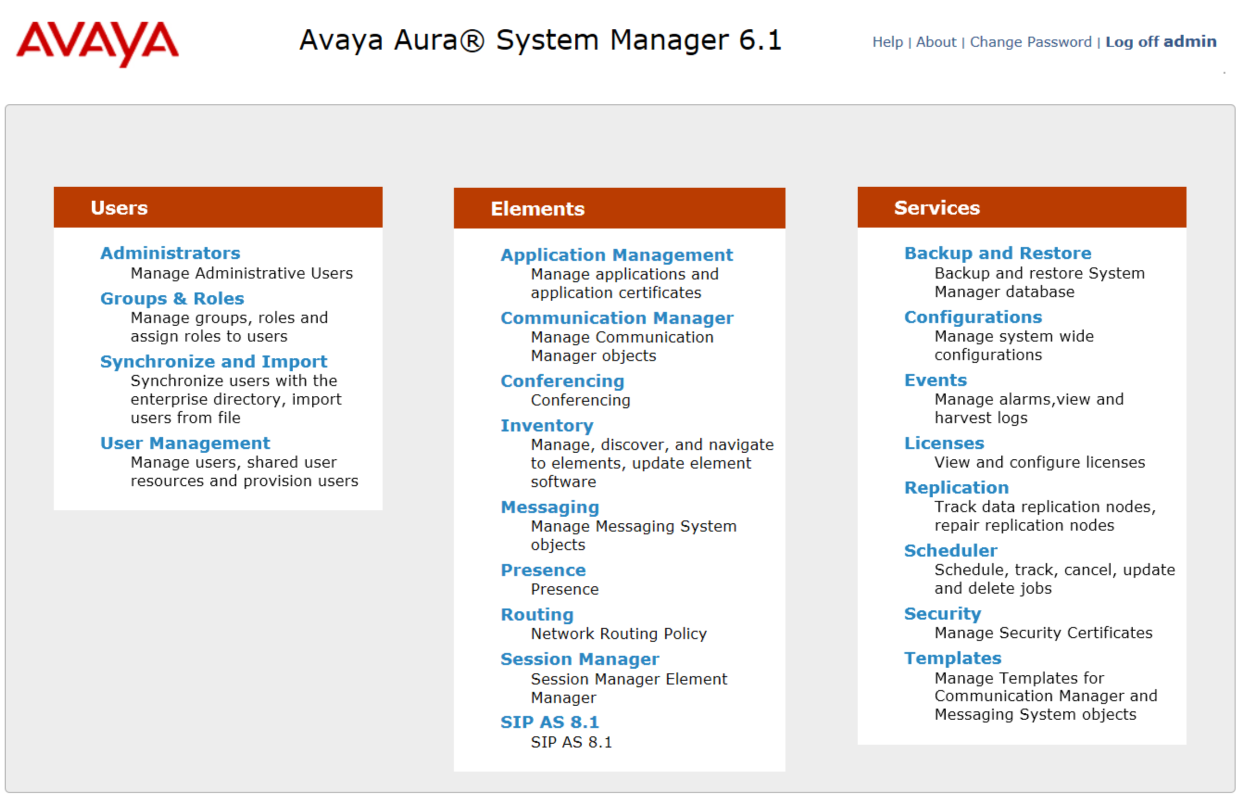
```
change off-pbx-telephone station-mapping 53165                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 53165            OPS          -      53165            aar        1
```

6. Configure Avaya Aura® Session Manager

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

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

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL "https://<*ip-address*>/SMGR", where <*ip-address*> is the IP address of System Manager. Log in with the appropriate credentials. The initial screen is displayed as shown below. The configuration in this section will primarily be performed under **Routing** and **Session Manager** listed within the **Elements** box.



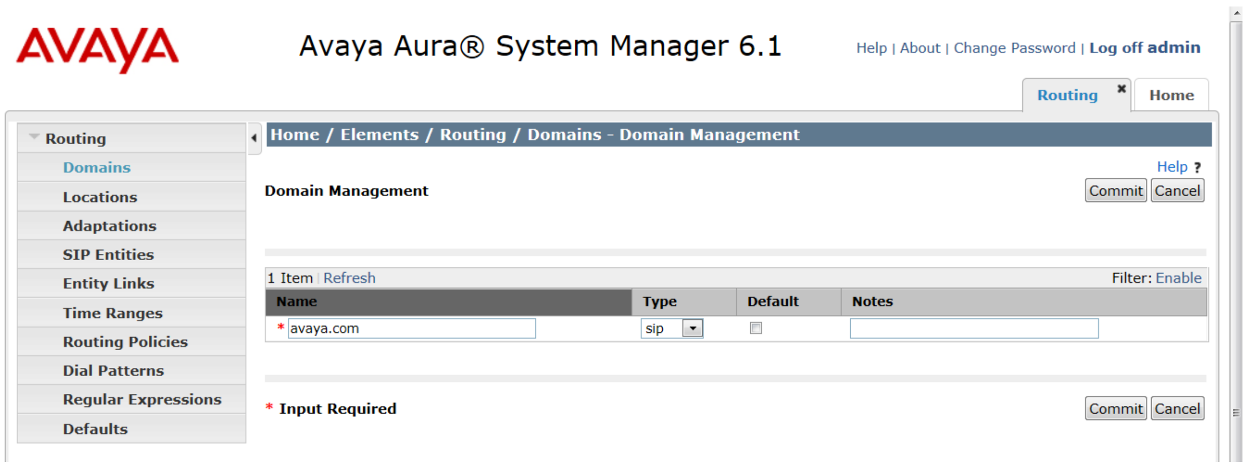
6.1 Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Select **Domains** on the left and clicking the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

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

Click **Commit**.

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



6.2 Add Locations

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

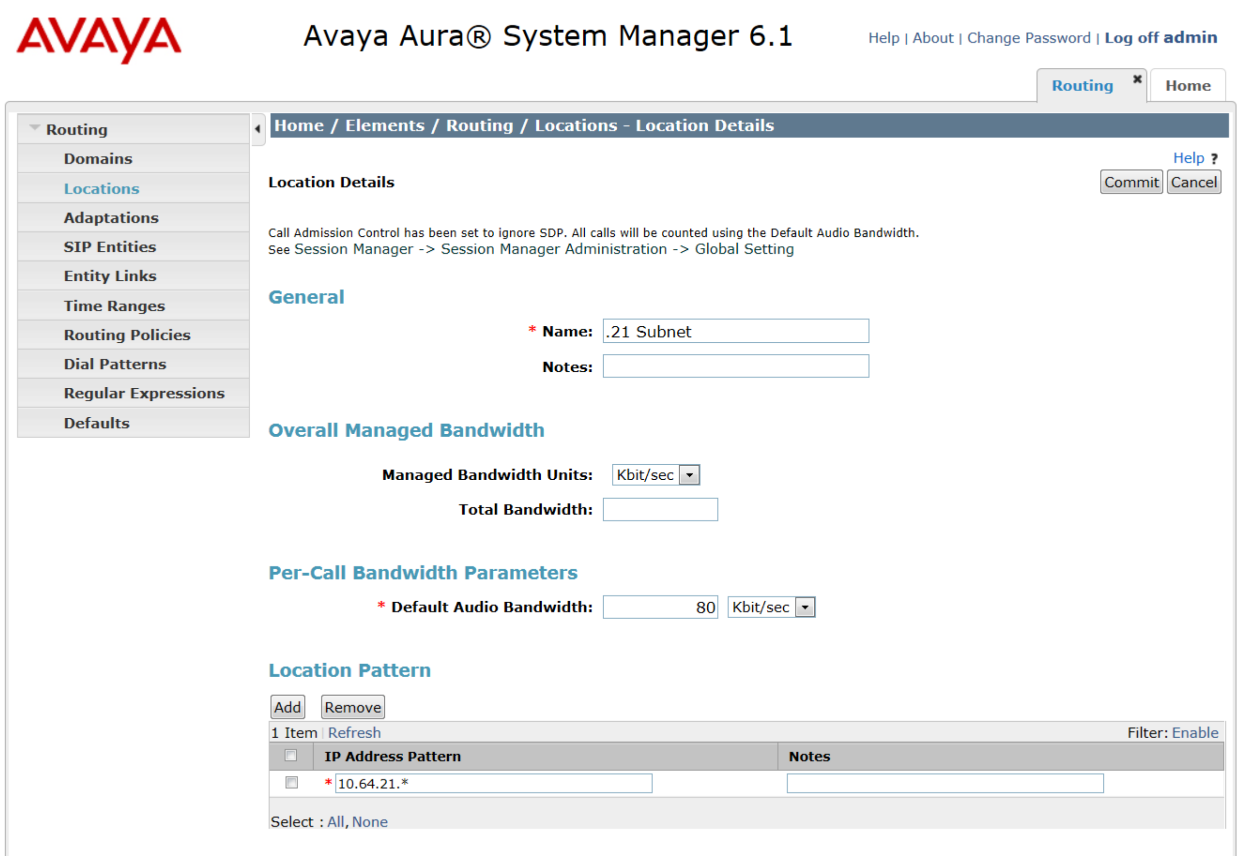
Under *General*:

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

Under *Location Pattern*:

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

The screen below shows addition of the *.21 Subnet* location, which includes Communication Manager and Session Manager. Click **Commit** to save the Location definition.



6.3 Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager and Communication Manager.

6.3.1 Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

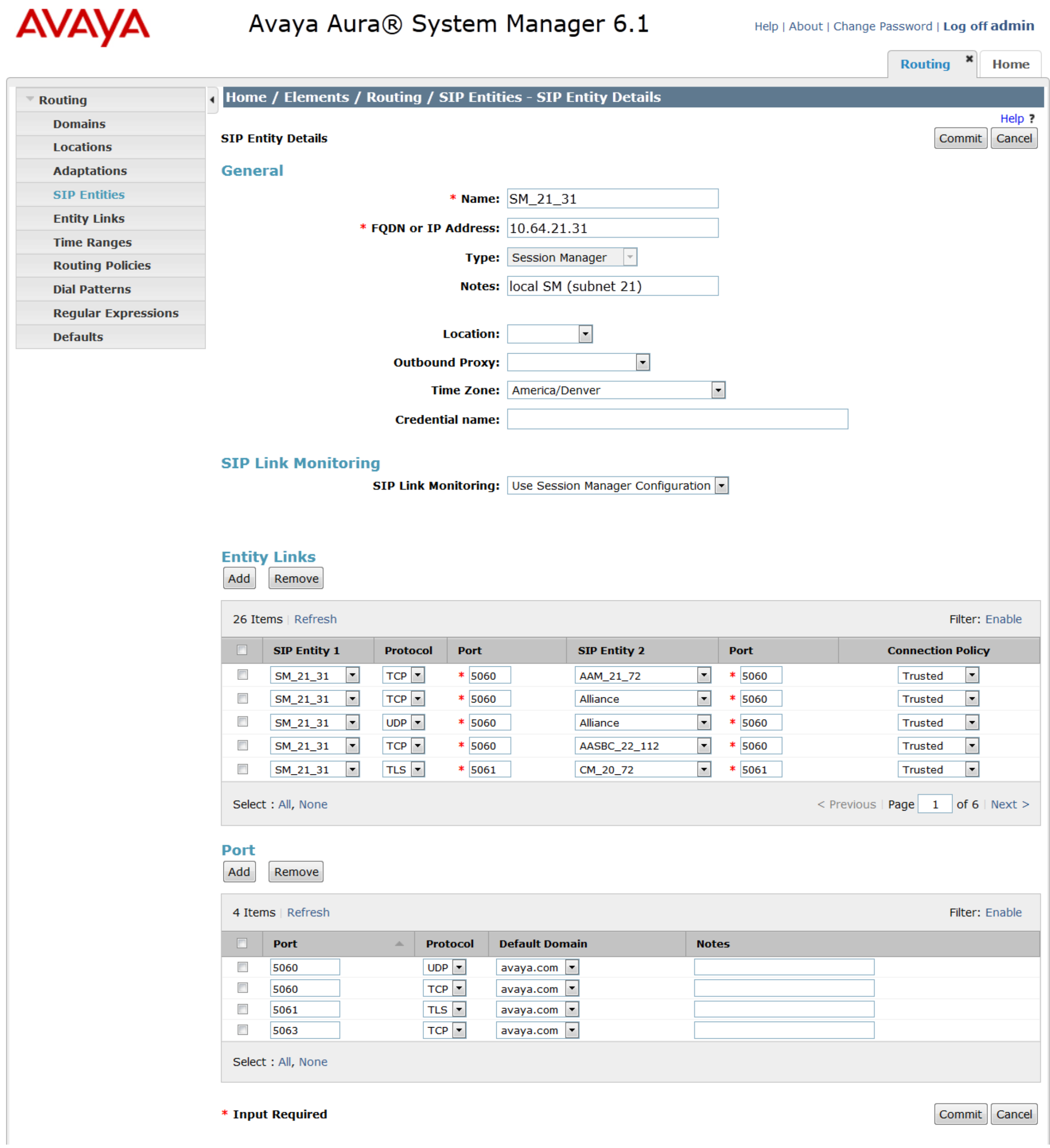
Under *General*:

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

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

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

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


AVAYA
Avaya Aura® System Manager 6.1
Help | About | Change Password | Log off admin
Routing
Home
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Home / Elements / Routing / SIP Entities - SIP Entity Details
Help ?
SIP Entity Details
Commit
Cancel
General
* Name: SM_21_31
* FQDN or IP Address: 10.64.21.31
Type: Session Manager
Notes: local SM (subnet 21)
Location:
Outbound Proxy:
Time Zone: America/Denver
Credential name:
SIP Link Monitoring
SIP Link Monitoring: Use Session Manager Configuration
Entity Links
Add
Remove
26 Items | Refresh
Filter: Enable
SIP Entity 1 | Protocol | Port | SIP Entity 2 | Port | Connection Policy
SM_21_31 | TCP | * 5060 | AAM_21_72 | * 5060 | Trusted
SM_21_31 | TCP | * 5060 | Alliance | * 5060 | Trusted
SM_21_31 | UDP | * 5060 | Alliance | * 5060 | Trusted
SM_21_31 | TCP | * 5060 | AASBC_22_112 | * 5060 | Trusted
SM_21_31 | TLS | * 5061 | CM_20_72 | * 5061 | Trusted
Select : All, None
< Previous | Page 1 of 6 | Next >
Port
Add
Remove
4 Items | Refresh
Filter: Enable
Port | Protocol | Default Domain | Notes
5060 | UDP | avaya.com
5060 | TCP | avaya.com
5061 | TLS | avaya.com
5063 | TCP | avaya.com
Select : All, None
* Input Required
Commit
Cancel

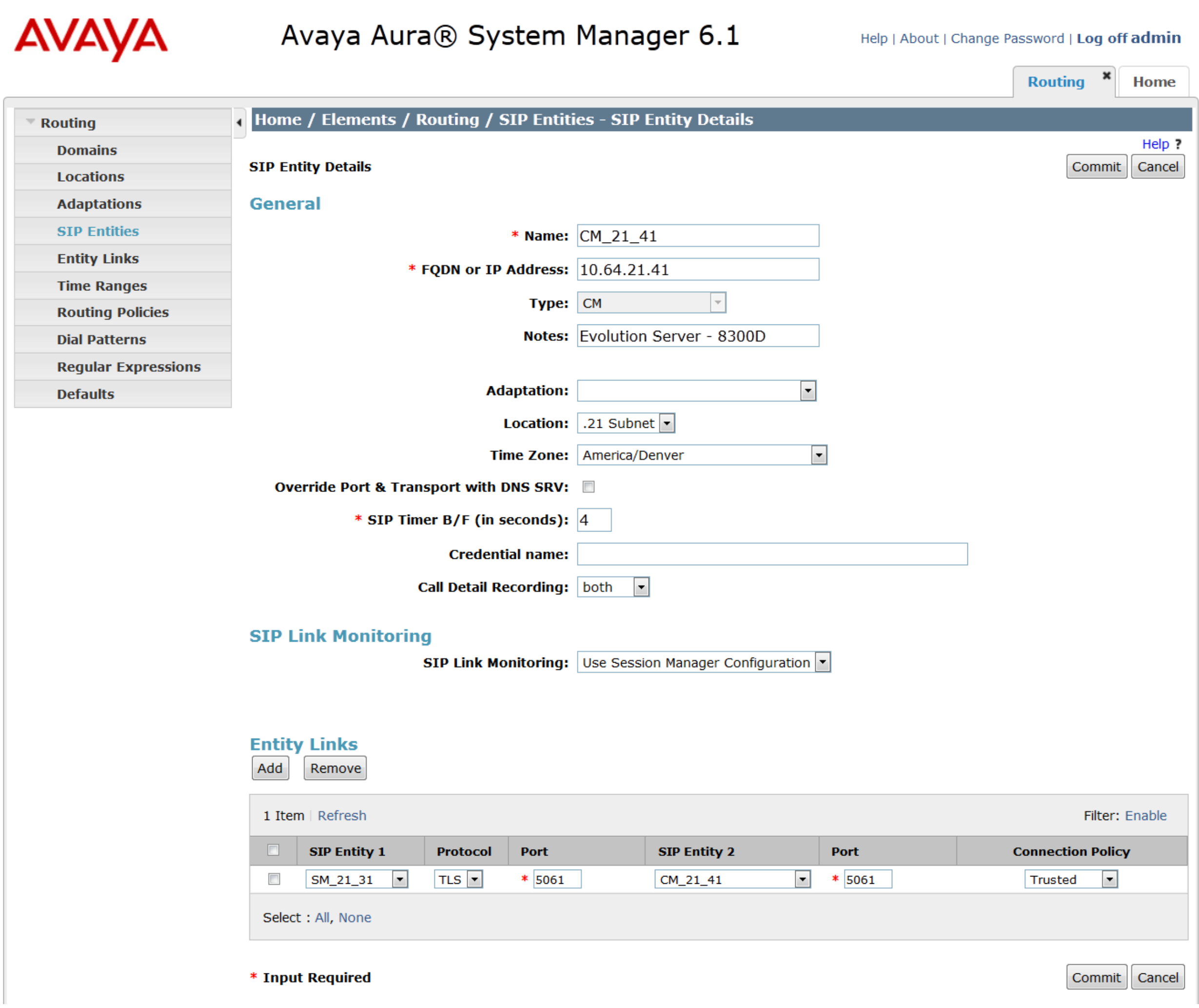
6.3.2 Communication Manager

A SIP Entity must be added for the Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

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

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

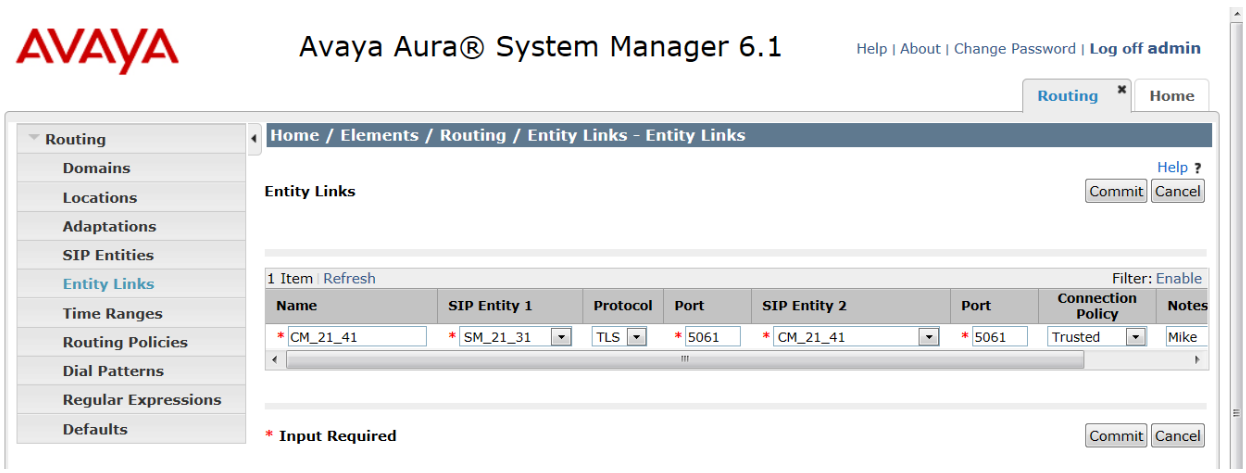


6.4 Add Entity Link

The SIP trunk from Session Manager to Communication Manager is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name.
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Select **Trusted**. *Note: If **Trusted** is not selected, calls from the associated SIP Entity specified in **Section 6.3.2** will be denied.*

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



6.5 Define Communication Manager as Managed Element

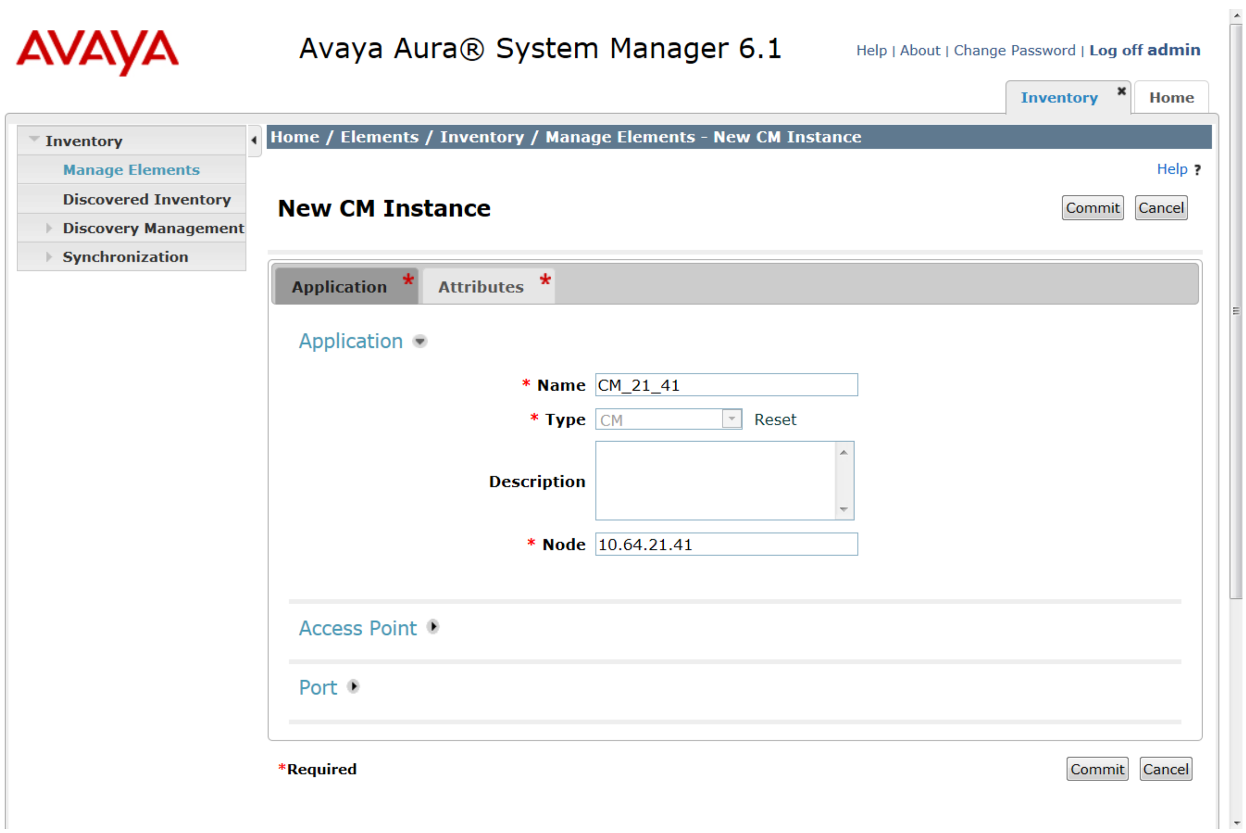
Before adding SIP users, Communication Manager must be added to System Manager as a managed element. This action allows System Manager to access Communication Manager over its administration interface. Using this administration interface, System Manager will notify Communication Manager when new SIP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

To define Communication Manager as a managed element, select **Elements→Inventory→Manage Elements** on the left and click on the **New** button (not shown) on the right. In the **New Entities Instance** screen (not shown), select *CM* in the **Type** field can click **Commit**.

In the **New CM Instance** screen, fill in the following fields as follows:

In the *Application* tab:

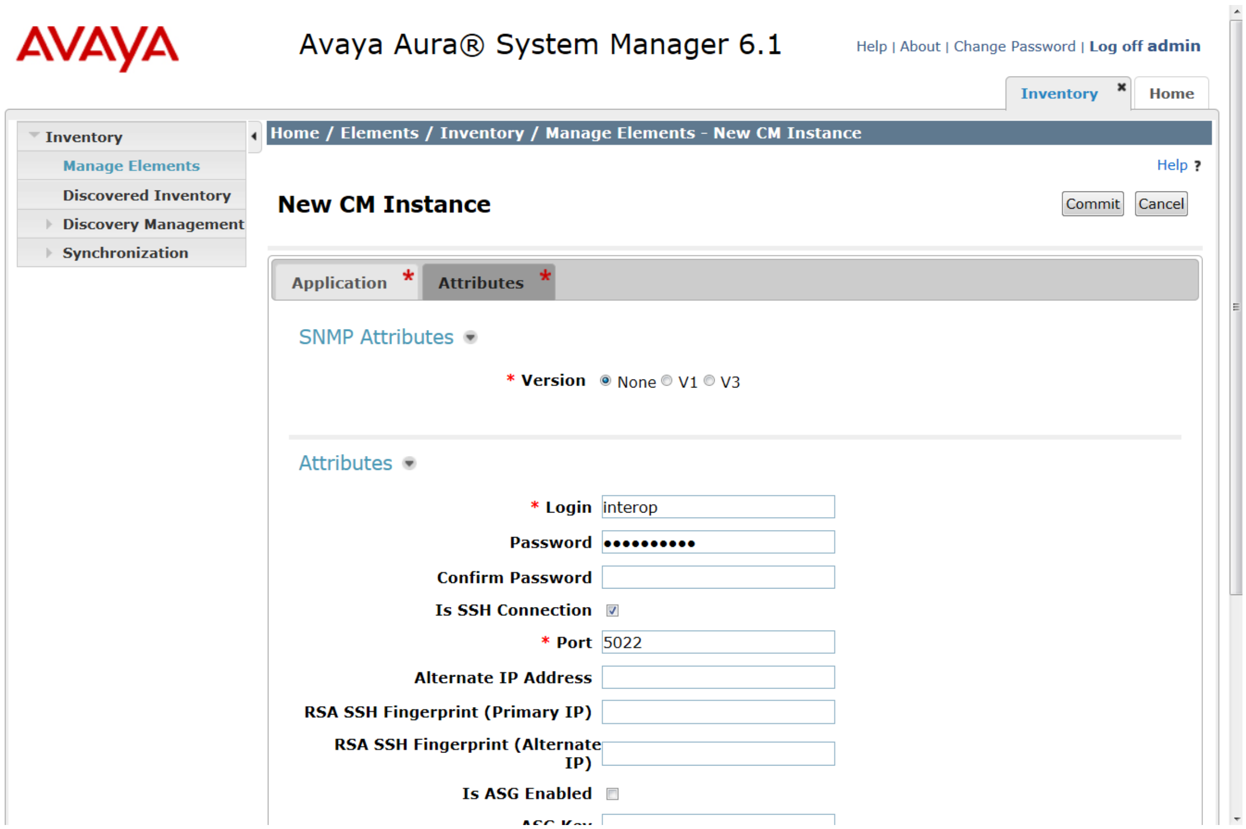
- **Name:** Enter an identifier for Communication Manager.
- **Type:** Select *CM* from the drop-down field.
- **Node:** Enter the IP address of the administration interface for Communication Manager.



In the *Attributes* tab:

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

Defaults can be used for the remaining fields. Click **Commit** to save the settings.

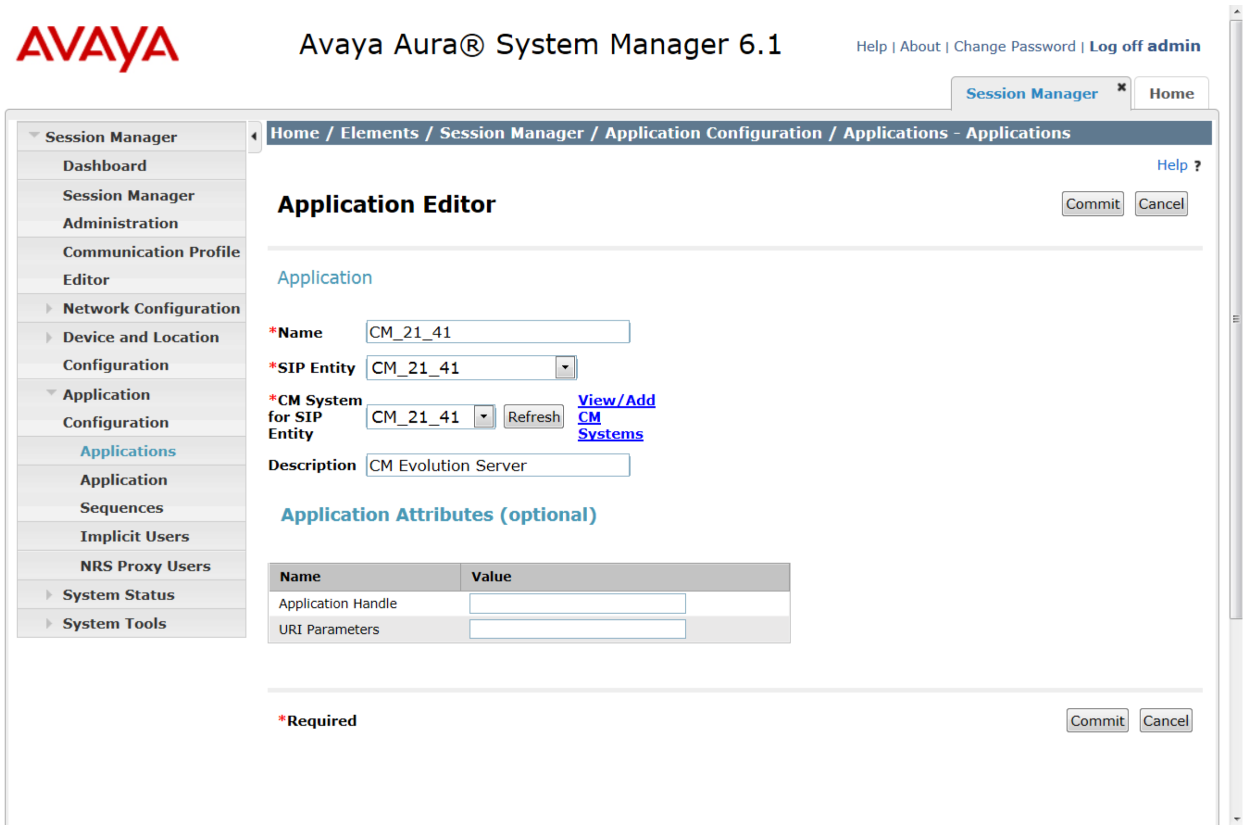


6.6 Add Application Sequence

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

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

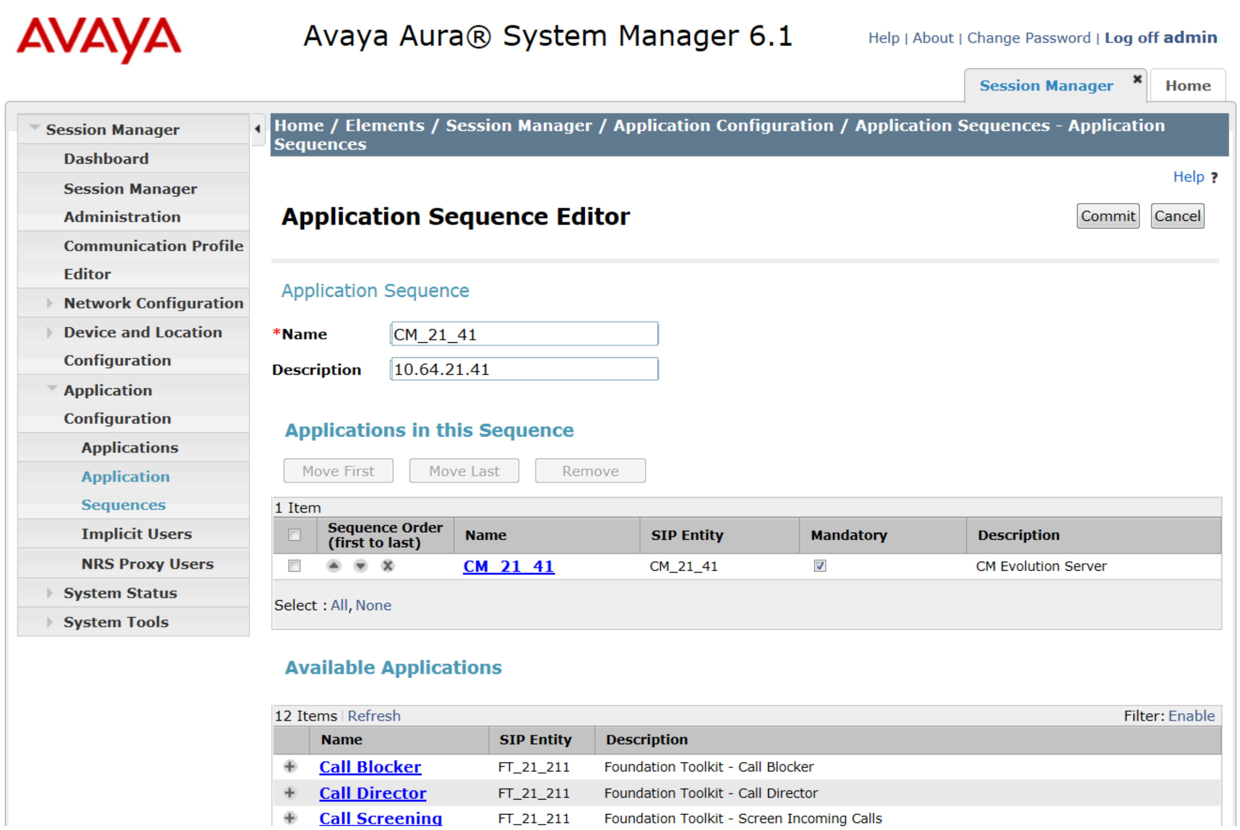
Click **Commit** to save the Application definition.



Next, navigate to **Elements → Session Manager → Application Configuration → Application Sequences** to define the Application Sequence for Communication Manager as shown below. Provide a **Name** for the Application Sequence and under **Available Applications**, click on the plus ( ) sign by *CM_21_41* to add it under the **Application in this sequence** section.

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

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



6.7 Add SIP User

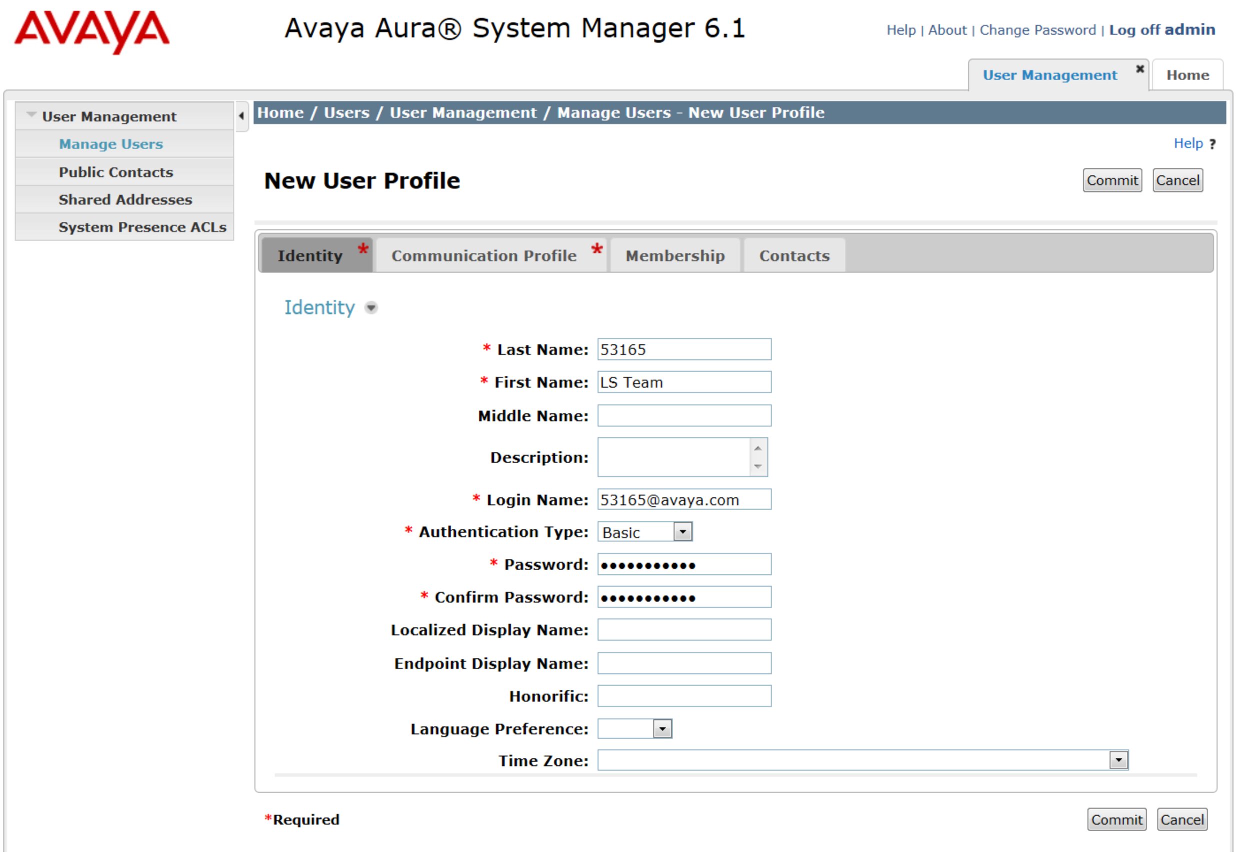
Add a SIP user for LifeSize Team 220**.** The following configuration will automatically create the SIP station on Communication Manager Evolution Server.

To add new SIP users, navigate to **Users → User Management → Manage Users** from the left and select **New** button (not shown) on the right.

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

- **Last Name:** Enter the last name of the user.
- **First Name:** Enter the first name of the user.
- **Login Name:** Enter *<extension>@<sip domain>* of the user (e.g., 53165@avaya.com).
- **Authentication Type:** Select *Basic***.**
- **Password:** Enter the password which will be used to log into System Manager
- **Confirm Password:** Re-enter the password from above.

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



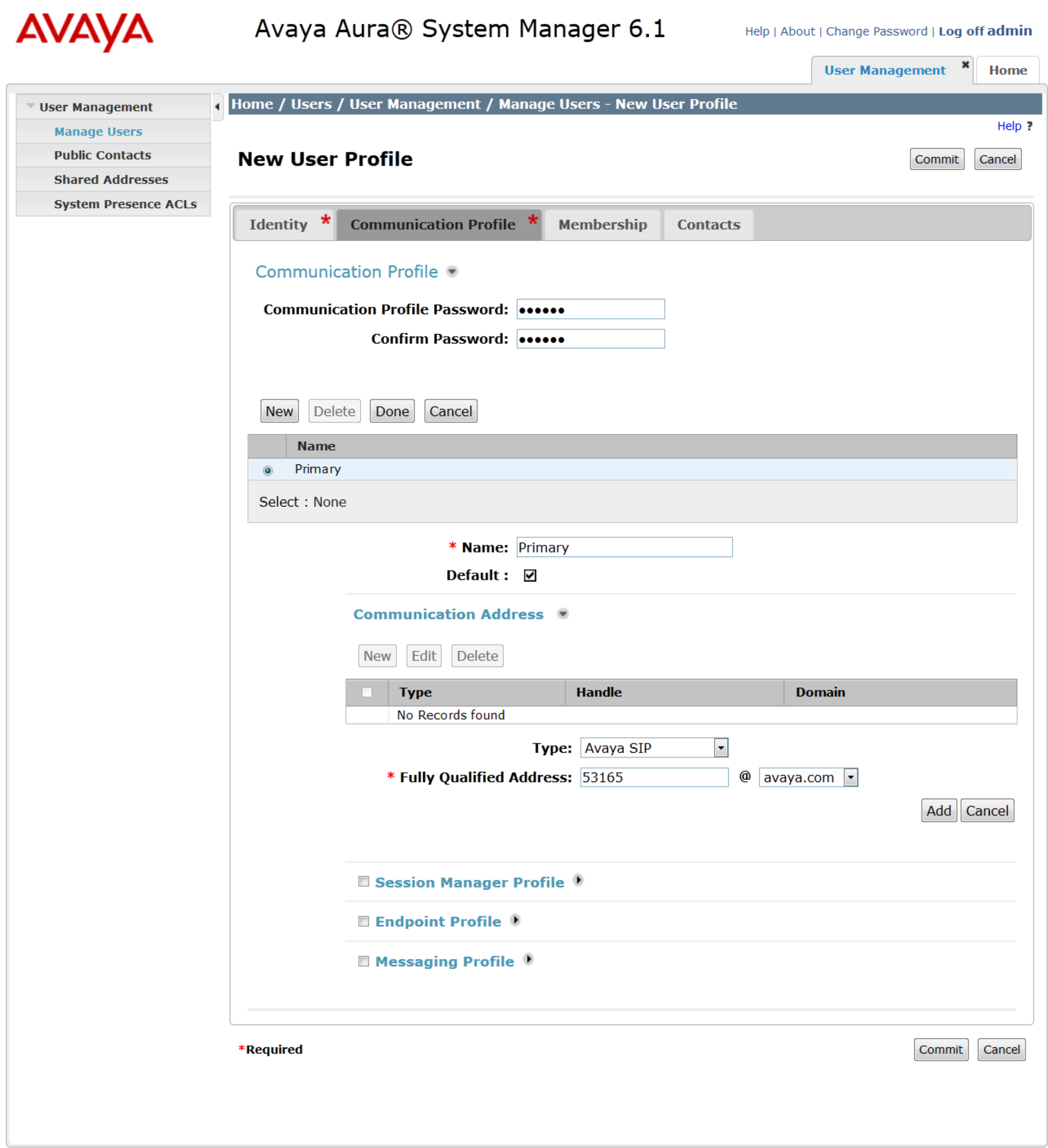
Enter values for the following required attributes for a new SIP user in the **Communication Profile tab** of the new user form.

- **Communication Profile Password:** Enter the password which will be used by Team to register with Session Manager.
- **Confirm Password:** Re-enter the password from above.

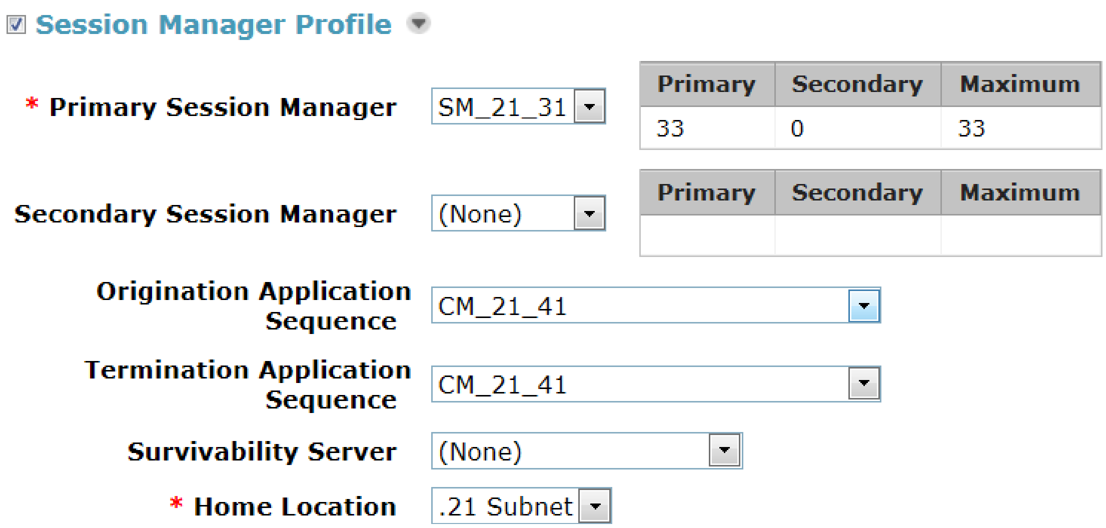
Scroll down to the **Communication Address** section and select **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

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

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

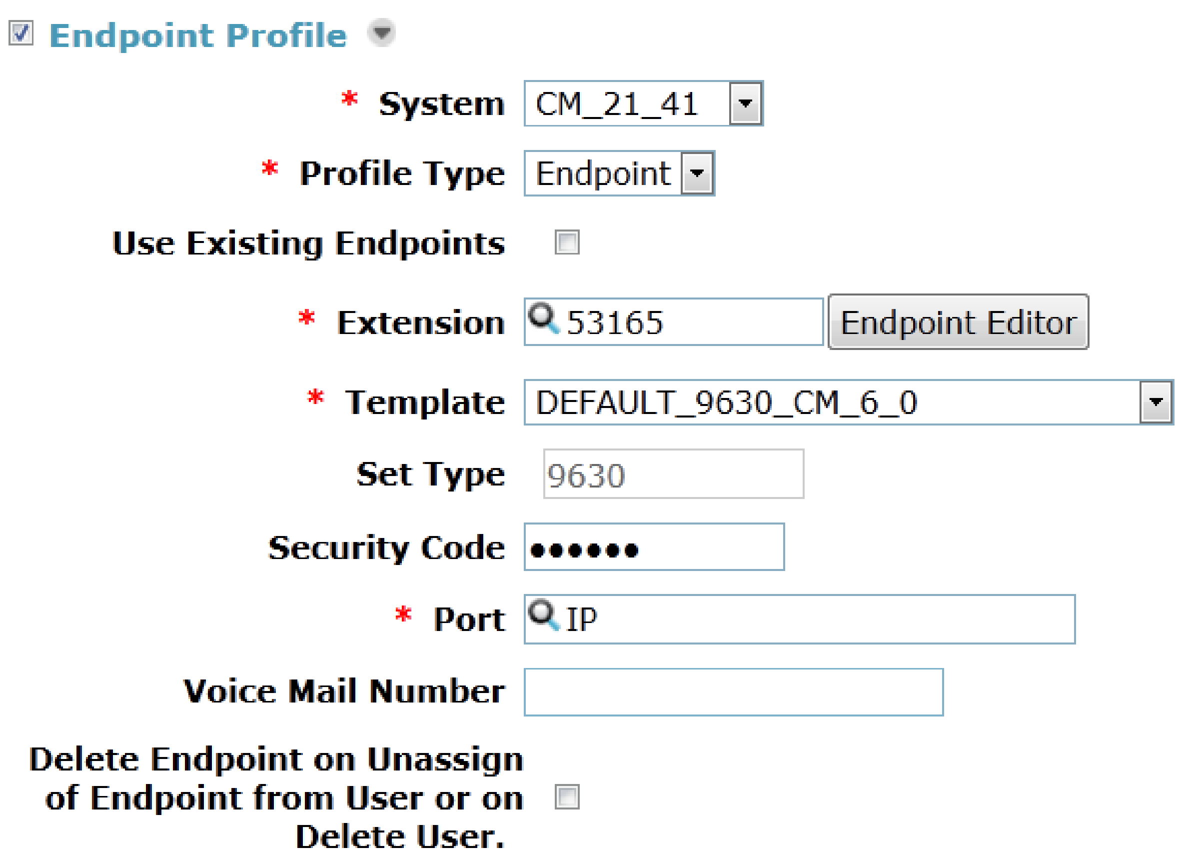

AVAYA
Avaya Aura® System Manager 6.1
Help | About | Change Password | Log off admin
User Management
Home
User Management
Manage Users
Public Contacts
Shared Addresses
System Presence ACLs
Home / Users / User Management / Manage Users - New User Profile
Help ?
New User Profile
Commit
Cancel
Identity *
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
No Records found
Type: Avaya SIP
* Fully Qualified Address: 53165 @ avaya.com
Add
Cancel
Session Manager Profile
Endpoint Profile
Messaging Profile
*Required
Commit
Cancel

In the *Session Manager Profile* section, specify the Session Manager entity from **Section 6.3.1** for **Primary Session Manager** and assign the **Application Sequence** defined in **Section 6.6** to the new SIP user as part of defining the **SIP Communication Profile**.   The **Application Sequence** can be used for both the originating and terminating sequence. Set the **Home Location** field to the **Location** configured in **Section 6.2**.

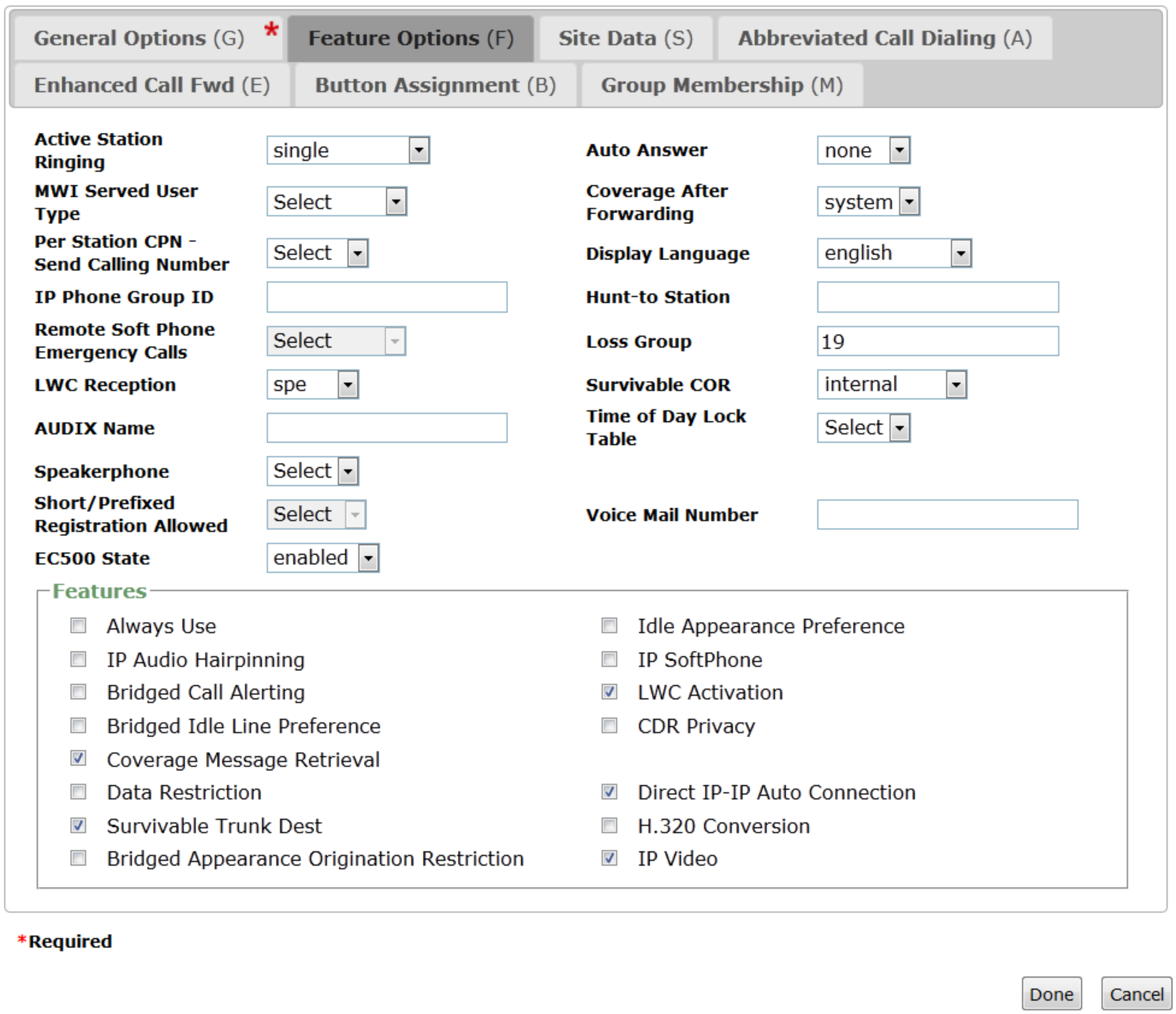


In the **Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager.
- **Profile Type** Select *Endpoint*.
- **Use Existing Stations:** If field is not selected, the station will automatically be added in Communication Manager.
- **Extension:** Enter extension number of SIP user.
- **Template:** Select template for type of SIP phone.
- **Port:** Enter *IP*.



Next, click on the **Endpoint Editor** button by the **Extension** field. The following screen is displayed. In the **Feature Options** section, select **IP Video Softphone** and click **Done**. The user will be returned to the previous screen. Click the **Commit** button to save the new SIP user profile.



6.8 Add Session Manager

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

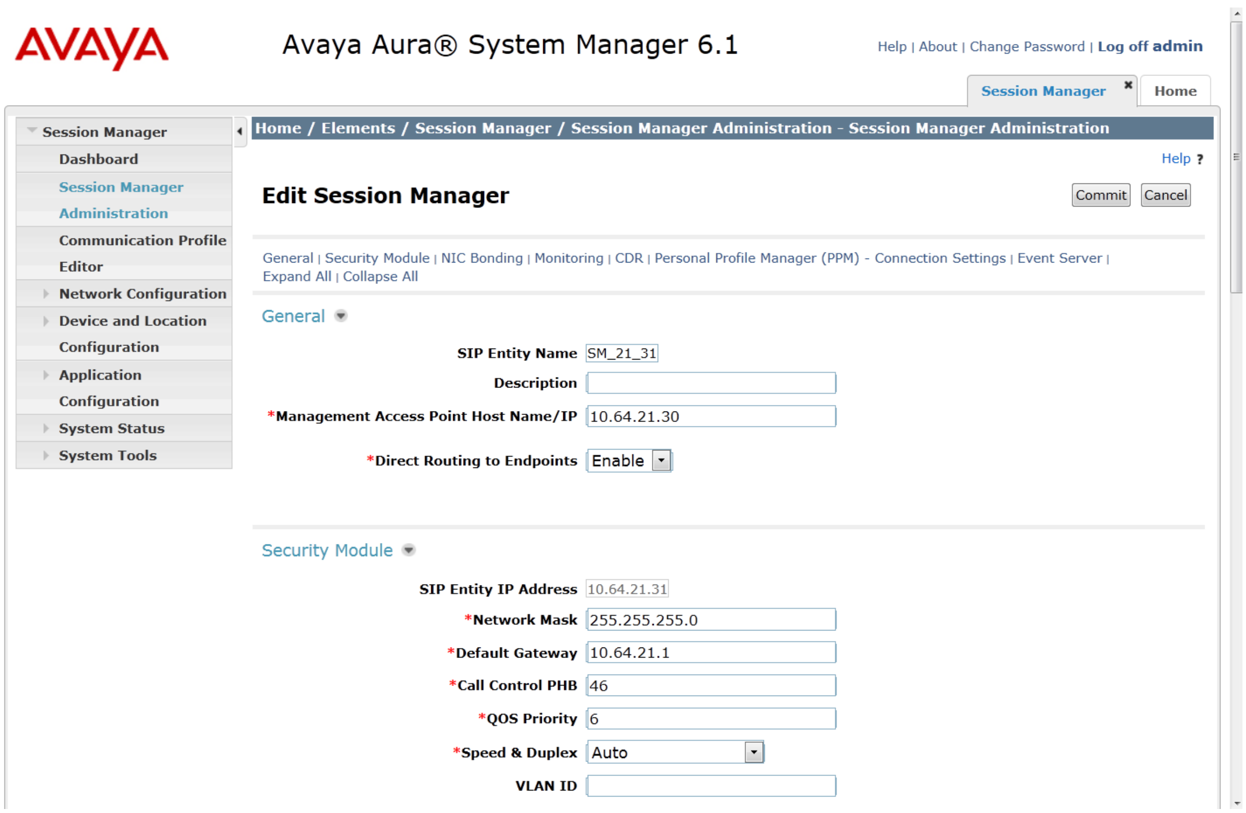
Under *Identity*:

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

Under *Security Module*:

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

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

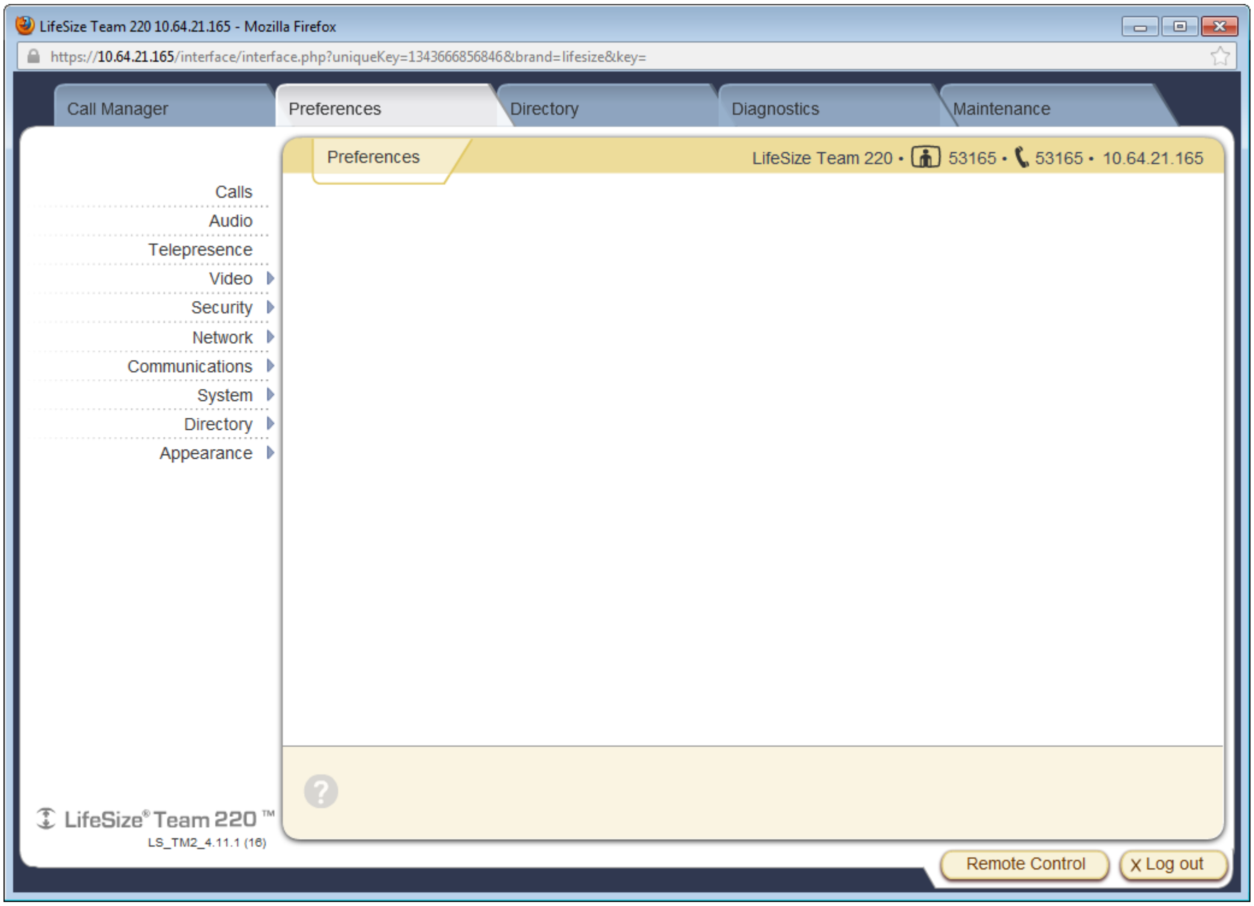


7. Configure LifeSize Team 220

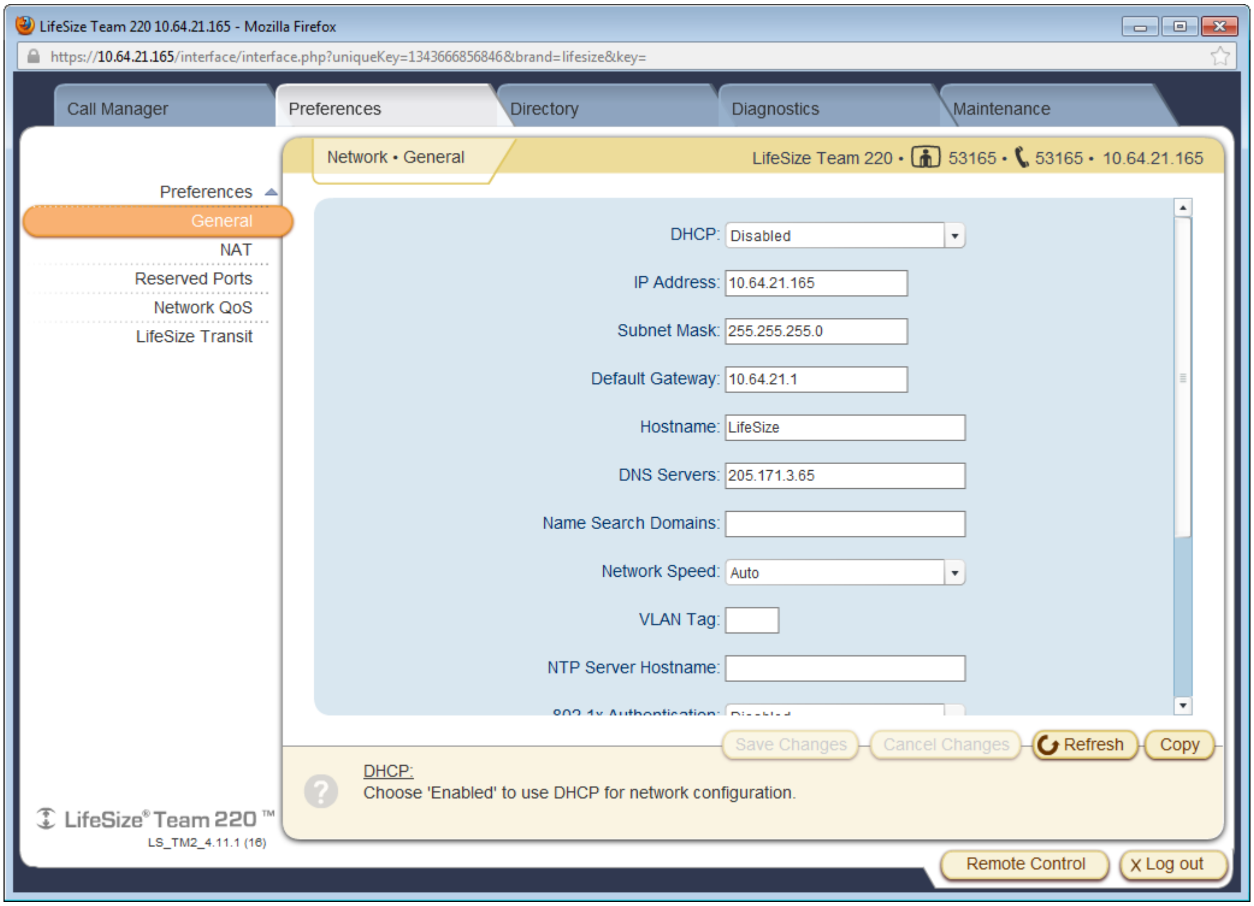
The configuration of the LifeSize Team 220 video system was performed via the Team 220's embedded Web interface or user interface on the monitor display using the remote control. However, the Team 220's LAN connection interface was initially configured via its monitor using the remote control. To configure the IP parameters for Team 220, navigate to the **System Menu → Administrator Preferences** and then log in with the appropriate credentials. Next, select **Network** and then select **General** to configure the LAN interface. The LAN configuration will be shown later in this section. The rest of the configuration was performed via the Team 220's embedded Web interface as shown in this section. Refer to reference **[4]** for additional information on configuring the Team video system.

From an internet browser, enter https://<*ip-addr*> in the URL field, where <*ip-addr*> is the Team 220's IP address. The following **Login** screen is displayed. Log in with the appropriate password.

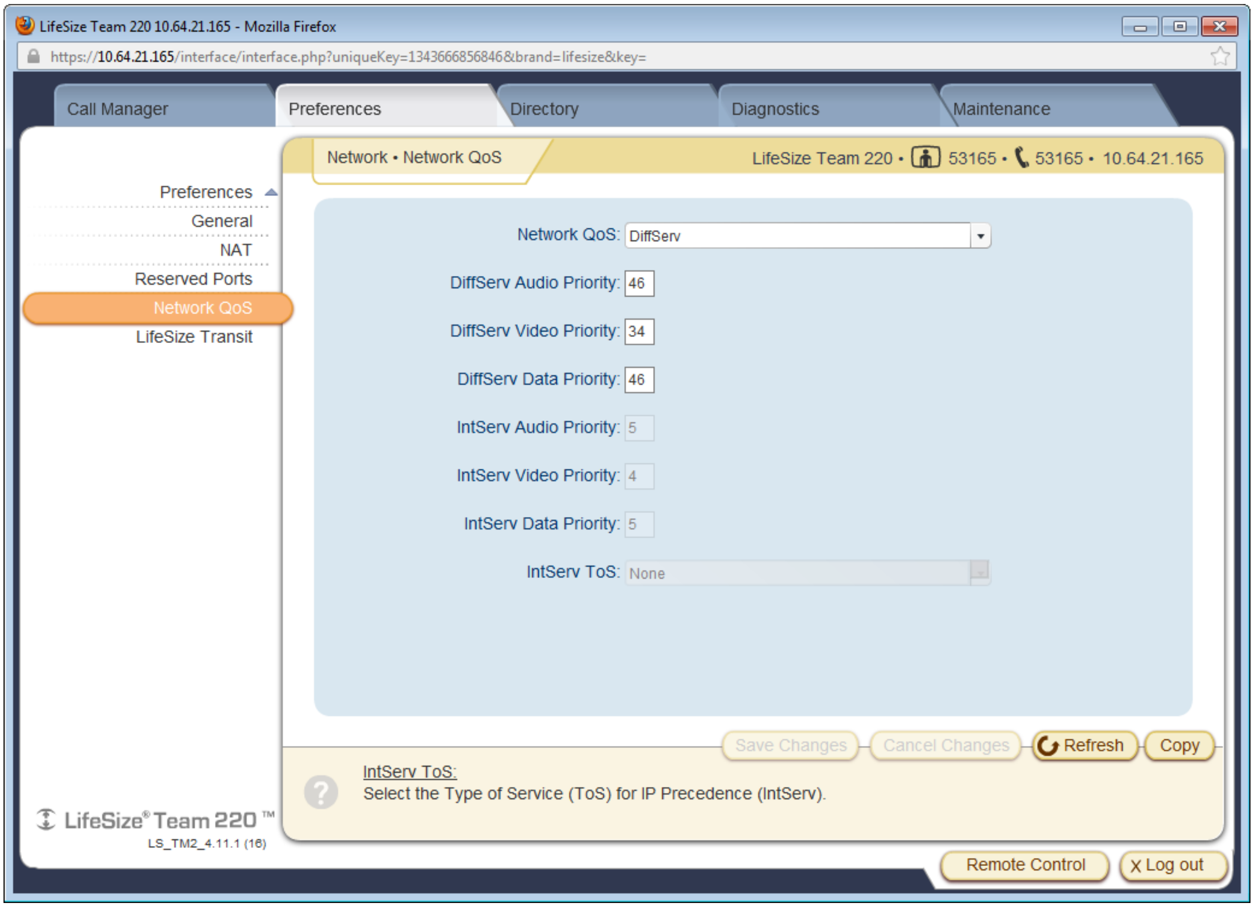
After logging in, the main screen is displayed as shown below.



To view the LAN configuration, navigate to **Network → General**. The following screen is displayed. In this configuration, a static IP address was assigned. As mentioned earlier, the initial IP configuration was performed via the monitor using the remote control.

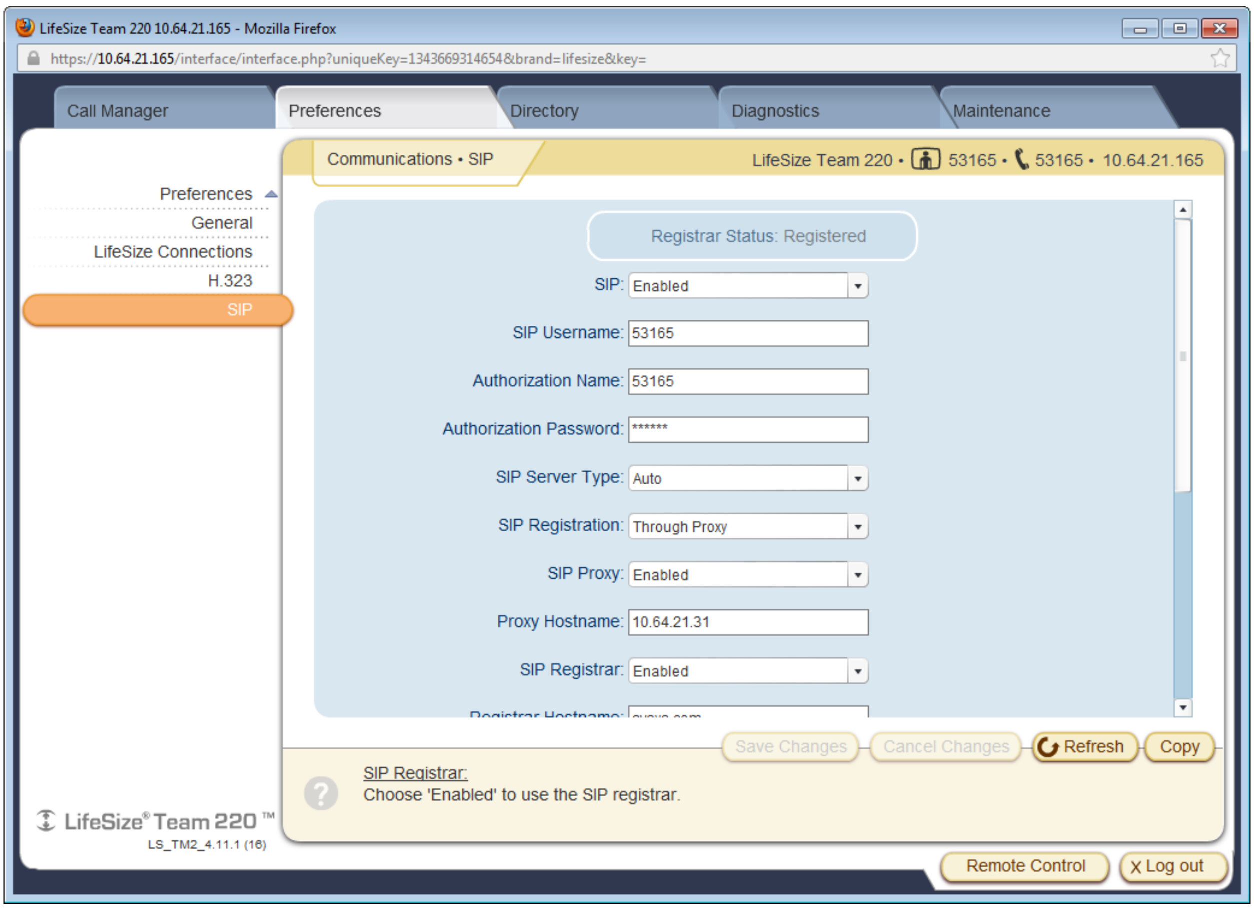


If network QoS is implemented using DiffServ, the **DiffServ Video Priority** may be configured on Team 220 so that it tags its video RTP packets with the appropriate DiffServ value. To configure DiffServ on Team 220, navigate to **Network → Network QoS** to display the screen below. Set the **Network QoS** field to *DiffServ* and set the **DiffServ Video Priority** field to the appropriate value as specified by your network administrator. Click the **Save Changes** button.



Next, configure the Team's SIP parameters. From the main screen, navigate to **Communications → SIP** to display the screen below. Configure the fields as follows:

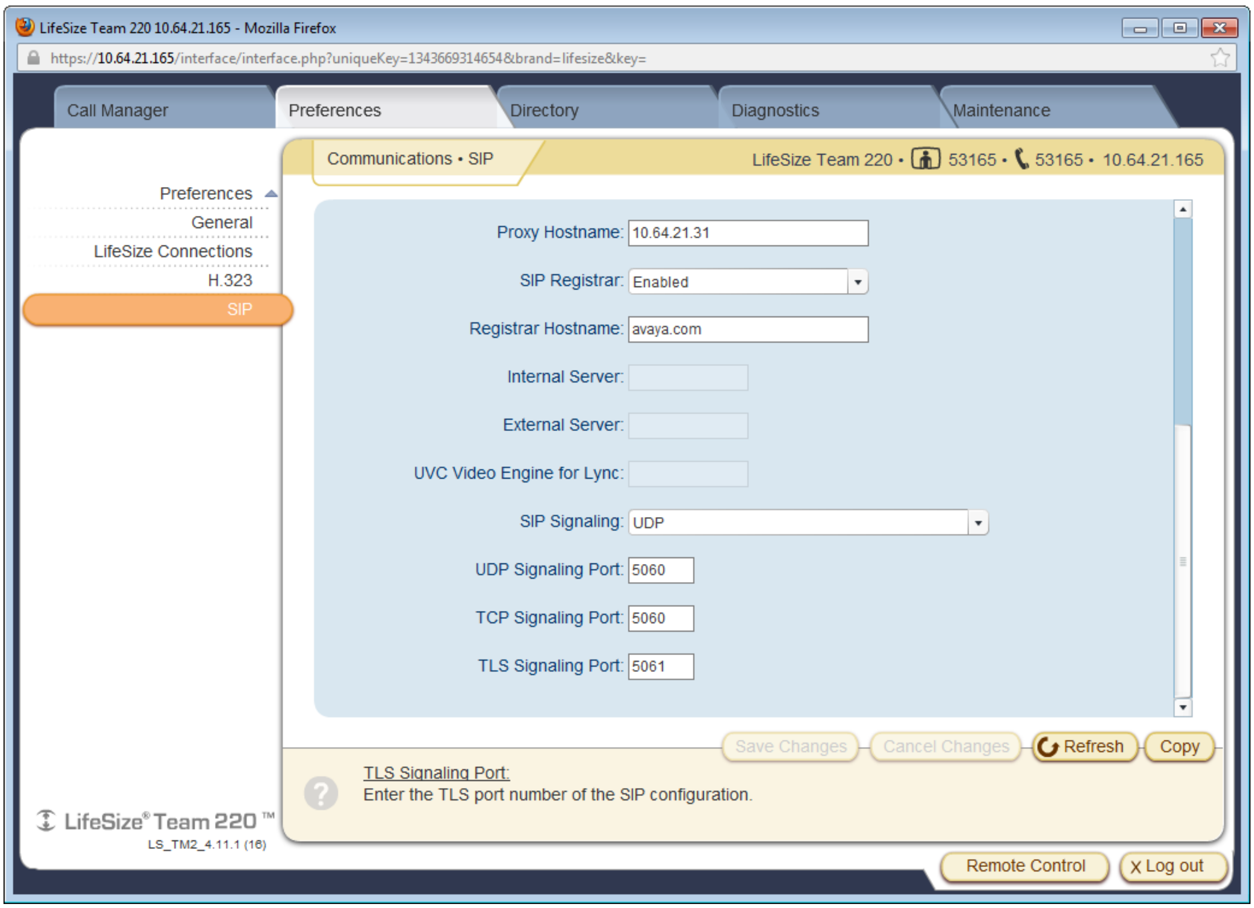
- **SIP** Set to *Enabled*.
- **SIP Username** Specify the Team 220's extension (e.g., 53165).
- **Authorization Name** Specify the Team 220's extension, which will be used to register with Session Manager.
- **Authorization Password** Specify the password used by Team 220 to register with Session Manager
- **SIP Server Type** Set to *Auto*.
- **SIP Registration** Set to *Through Proxy*.
- **SIP Proxy** Set to *Enabled*.
- **Proxy Hostname** Specify the IP address of Session Manager's SIP interface (e.g., 10.64.21.31).
- **SIP Registrar** Set to *Enabled*.



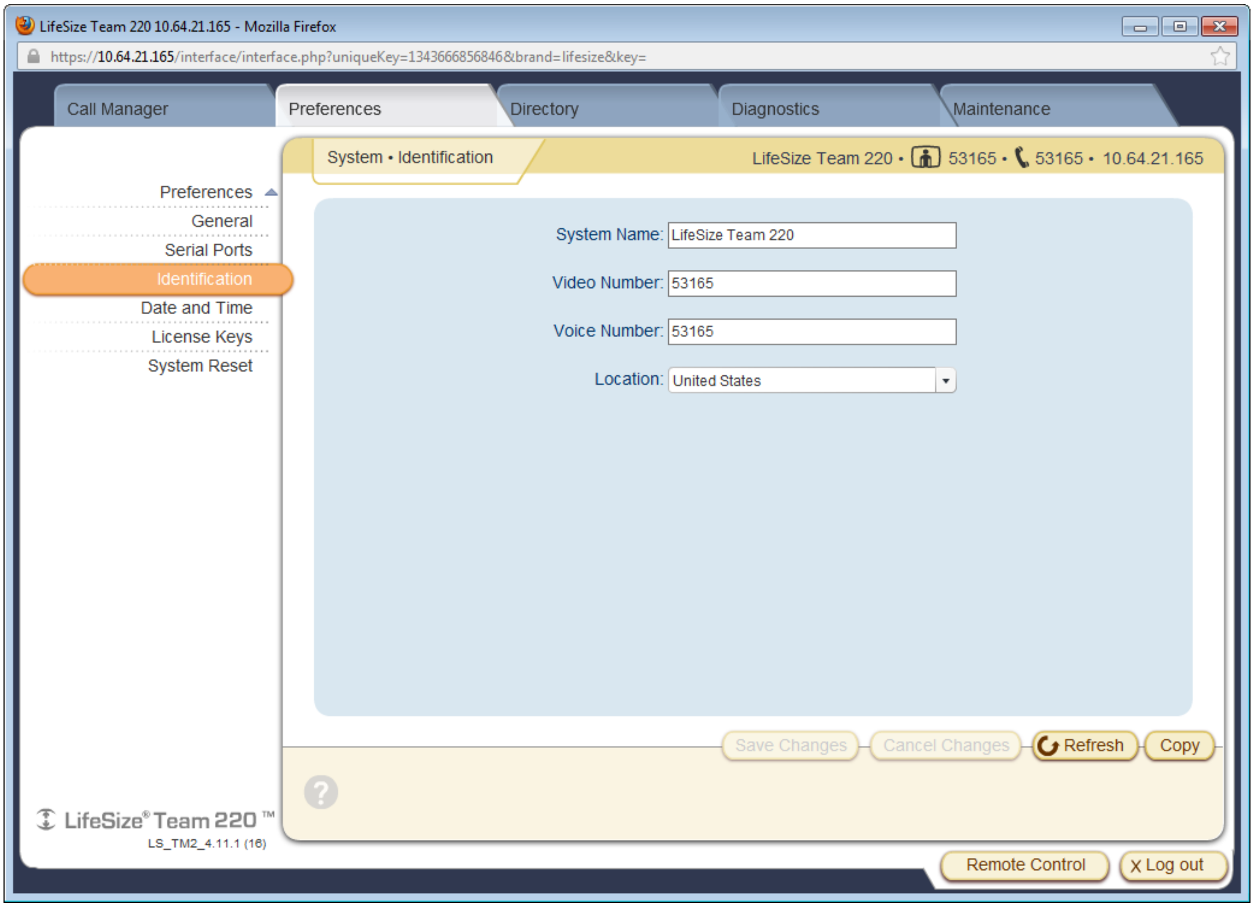
On the same **Communications → SIP** screen, scroll down to configure the rest of the SIP parameters as follows:

- **Registrar Hostname** Specify the IP address of Session Manager's SIP interface.
- **SIP Registrar** Set to *Enabled*.
- **Registrar Hostname** Set to *avaya.com*.
- **SIP Signaling** Set to *UDP*.
- **UDP Signaling Port** Specify the port used to communicate with Session Manager via UDP.

When the configuration is completed, click the **Save Changes** button.



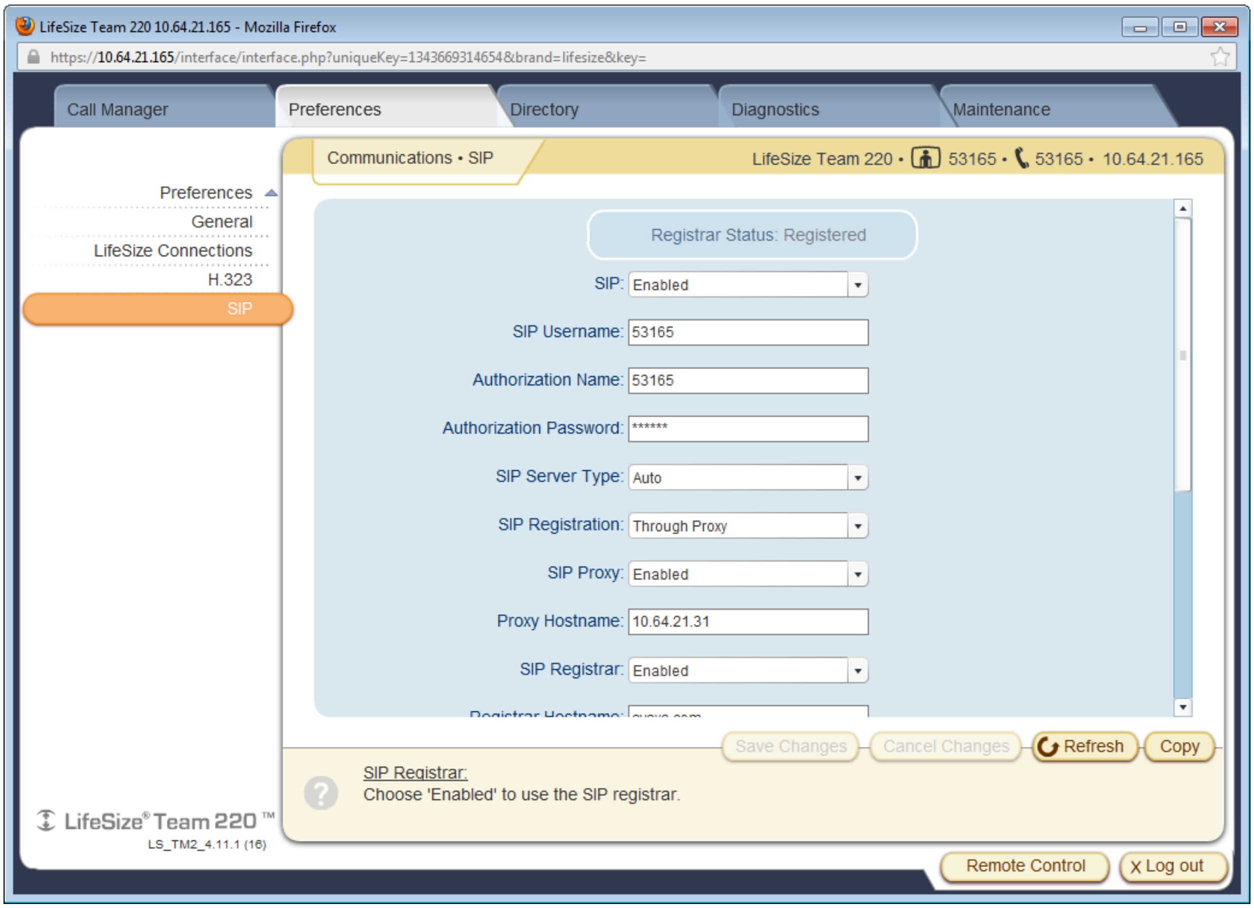
Lastly, to display the Team 220's extension on the top of the monitor, configure the **Identification** screen. From the main screen, navigate to **System → Identification** and set the **Video Number** and **Voice Number** fields to the Team's extension as shown below. Click **Save Changes** when done.



8. Verification Steps

This section provides the steps that may be performed to verify proper configuration of the LifeSize Team 220 video system with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

1. Verify that the LifeSize system has successfully registered with Session Manager. Navigate to **Communications → SIP** and verify that the **Registrar Status** indicates *Registered* as shown below.



2. Place an outgoing video call from Team 220 to another video system registered with Session Manager and verify that the video completes with 2-way audio and video.

3. Place an outgoing voice call from Team 220 to an Avaya IP telephone and verify that the voice call completes with 2-way audio.

9. Conclusion

These Application Notes have described the administration steps required to integrate the LifeSize Team 220 video system with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. LifeSize Team 220 successfully registered with Session Manager and voice and video calls were established with LifeSize Team, Avaya one-X Communicator and Avaya IP telephones. All test cases passed with observations noted in **Section 2.2**.

10. References

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

[1] *Administering Avaya Aura® Communication Manager*, March 2012, Document Number 03-300509.

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

The following LifeSize product documentation is available at http://www.lifesize.com.

[3] *LifeSize® Video Communication Systems Installation Guide*, Februay 2011.

[4] *LifeSize® Video Communication Systems User and Administrator Guide*, February 2011.

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