Avaya Solution & Interoperability Test Lab

Application Notes for VBrick Distributed Media Engine with Avaya Aura® Session Manager and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the VBrick Distributed Media Engine with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using a SIP 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 VBrick Distributed Media Engine with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using a SIP interface.

The VBrick H.264 Distributed Media Engine (DME) simplifies delivery of high-definition video and other rich media content across multi-site enterprises and campus environments. The Distributed Media Engine, deployed centrally or at the network edge, is a single integrated platform that provides media redistribution, media transformation as well as video-on-demand content serving and storage.

The DME can deliver an optional Video Conference Streaming Gateway Module, enabling the DME to ingest video from popular videoconference systems and convert it for streaming. Each unit can support hundreds or thousands of users, concurrently delivering multiple streams of live and stored video content in a variety of formats. The DME's Video Conferencing Streaming Gateway Module can be invited into a video call by an individual video conference camera & codec or a multipoint control unit (MCU). Conversely, the Gateway Module can initiate a call into a conference. The DME ingests the video from the video conference system, converts the formats for streaming and delivers it across the VBrick ecosystem.

2. General Test Approach and Test Results

This goal of interoperability test plan was to test the ability of the VBrick Distributed Media Engine to interoperate with an Avaya telephony infrastructure. The Avaya components consisted of the following:

- Avaya Aura® System Manager
- Avaya Aura® Session Manager
- Avaya Aura® Communication Manager
- Avaya video endpoints
    - Avaya Desktop Video Device (SIP)
    - Avaya one-X® Communicator (H.323)
    - Avaya one-X® Communicator (SIP)
    - Avaya 1010/1020 Video Conferencing System (SIP)
    - Avaya 1030/1040/1050 Video Conferencing System (SIP)

Compliance testing focused on point-to-point video calls and between the Avaya video endpoints and the VBrick DME, and live streaming of those calls. MCU video call scenarios were not tested.

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 VBrick Distributed Media Engine with Session Manager.
- Video calls between VBrick Distributed Media Engine and Avaya one-X® Communicator (SIP and H.323 versions), the Avaya Desktop Video Device, and Avaya 1020 and 1050 Video Conference Systems.
- Voice calls between VBrick Distributed Media Engine and Avaya one-X® Communicator (SIP and H.323 versions), the Avaya Desktop Video Device, and Avaya 1020 and 1050 Video Conference Systems.
- G.711 and G.729 codec support.
- Caller ID display on.
- Audio mute on VBrick DME and Avaya endpoints for video and voice calls.
- Video mute from Avaya endpoints to VBrick DME.
- Proper system recovery after a restart of the VBrick DME and loss of IP connectivity.

2.2 Test Results

All test cases passed with the following exception/observation:

- Video calls failed between Avaya one-X Communicator (H.323) and the VBrick DME. This issue has been identify within the VBrick DME and the fix will be to issue an upcoming release of the DME.

2.3 Support

For technical support on the VBrick Distributed Media Engine, contact VBrick Support via phone or website.

- **Phone:** (866) 827-4251
- **Web:** http://www.vbrick.com/support/index.asp

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.
- Avaya Aura® Session Manager.
- Avaya Aura® System Manager (used to configure Session Manager).
- Avaya video endpoints
  - Avaya Desktop Video Device (SIP)
  - Avaya one-X® Communicator (H.323)
  - Avaya one-X® Communicator (SIP)
  - Avaya 1010/1020 Video Conferencing System (SIP)
  - Avaya 1030/1040/1050 Video Conferencing System (SIP)

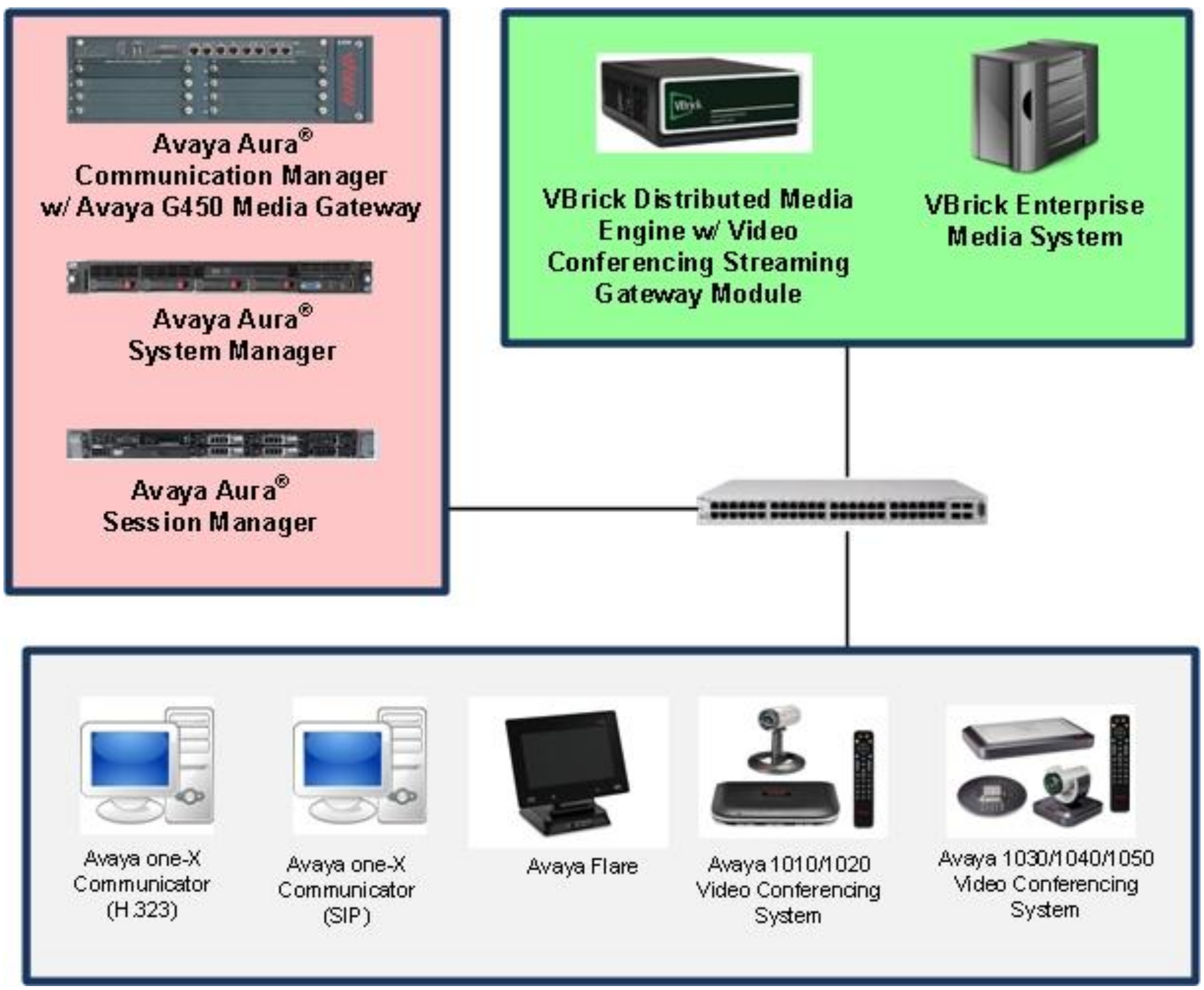


Figure 1: Avaya SIP Network with the VBrick Distributed Media Engine

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 6.2 SP2
Dell™ PowerEdge™ R610 Server	Avaya Aura® System Manager 6.2 SP2
Avaya S8300D Server with an Avaya G450 Media Gateway	Avaya Aura® Communication Manager 6.2 (R016x.02.0.823.0-19926)
Avaya one-X® Communicator	6.1.5.07-SP5-37495
Avaya Desktop Video Device	1.1.1
Avaya 1020 Video Conference System	4.8.3 (26)
Avaya 1050 Video Conference System	4.8.3 (26)
VBrick Distributed Media Engine 7530: • VBrick H.264 Distributed Media Engine • Video Conference Streaming Gateway Module	3.0.2

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 VBrick DME 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 VBrick DME. 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**, enable **Allow Direct-IP Multimedia**. Set **Maximum Call Rate for Direct-IP Multimedia** and **Maximum Call Rate for Priority Direct-IP Multmedia** to desired values or use the default values.

```
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. Type **add signaling-group n** where n is the number of the signaling group. 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 Stations for VBrick Distributed Media Engine

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 6.6**. 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 on Communication Manager. Two users were created, 53124 and 53125. User 53124, shown below, was created for incoming calls to the DME. User 53135, not shown, was created using the same steps for outgoing calls from the DME.

Use the **display station** command to view the station created for the VBrick Distributed Media Engine for incoming calls (i.e. station **53124**) and verify the settings in bold. Note that the **IP Video** field must be set to *y*.

```
display station 53124                                           Page   1 of   6
                                     STATION

Extension: 53124                       Lock Messages? n              BCC: M
     Type: 9630SIP                     Security Code: 123456          TN: 1
     Port: S00006                    Coverage Path 1:                COR: 1
     Name: DME - Incoming, VBrick    Coverage Path 2:                COS: 1
                                     Hunt-to Station:
STATION OPTIONS
                                         Time of Day Lock Table:
              Loss Group: 19
                                               Message Lamp Ext: 53124

        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., 53124) 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.

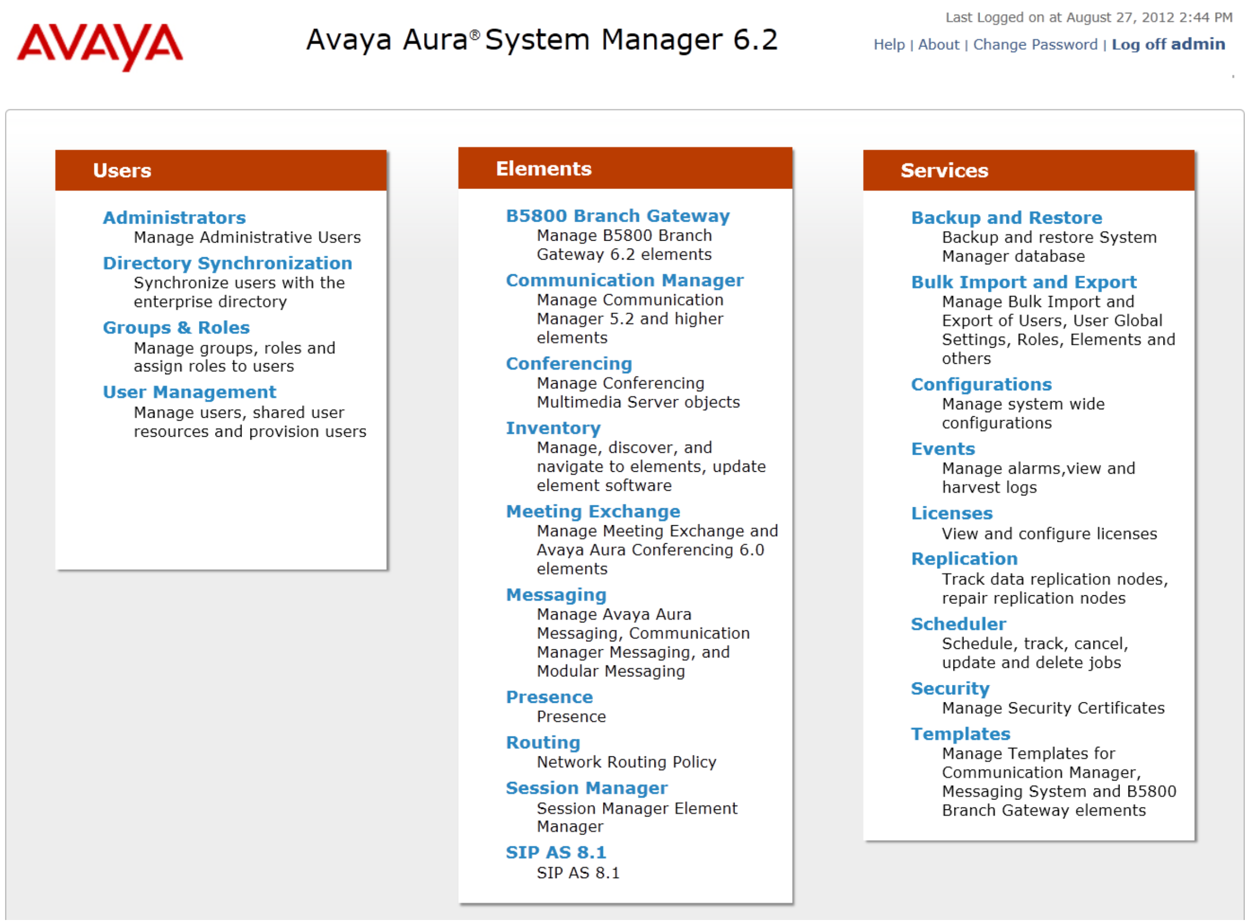
```
display off-pbx-telephone station-mapping 53167                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 53124            OPS          -     53167             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.


Last Logged on at August 27, 2012 2:44 PM
Avaya Aura® System Manager 6.2
Help | About | Change Password | Log off admin
Users
Administrators
Manage Administrative Users
Directory Synchronization
Synchronize users with the enterprise directory
Groups & Roles
Manage groups, roles and assign roles to users
User Management
Manage users, shared user resources and provision users
Elements
B5800 Branch Gateway
Manage B5800 Branch Gateway 6.2 elements
Communication Manager
Manage Communication Manager 5.2 and higher elements
Conferencing
Manage Conferencing Multimedia Server objects
Inventory
Manage, discover, and navigate to elements, update element software
Meeting Exchange
Manage Meeting Exchange and Avaya Aura Conferencing 6.0 elements
Messaging
Manage Avaya Aura Messaging, Communication Manager Messaging, and Modular Messaging
Presence
Presence
Routing
Network Routing Policy
Session Manager
Session Manager Element Manager
SIP AS 8.1
SIP AS 8.1
Services
Backup and Restore
Backup and restore System Manager database
Bulk Import and Export
Manage Bulk Import and Export of Users, User Global Settings, Roles, Elements and others
Configurations
Manage system wide configurations
Events
Manage alarms,view and harvest logs
Licenses
View and configure licenses
Replication
Track data replication nodes, repair replication nodes
Scheduler
Schedule, track, cancel, update and delete jobs
Security
Manage Security Certificates
Templates
Manage Templates for Communication Manager, Messaging System and B5800 Branch Gateway elements

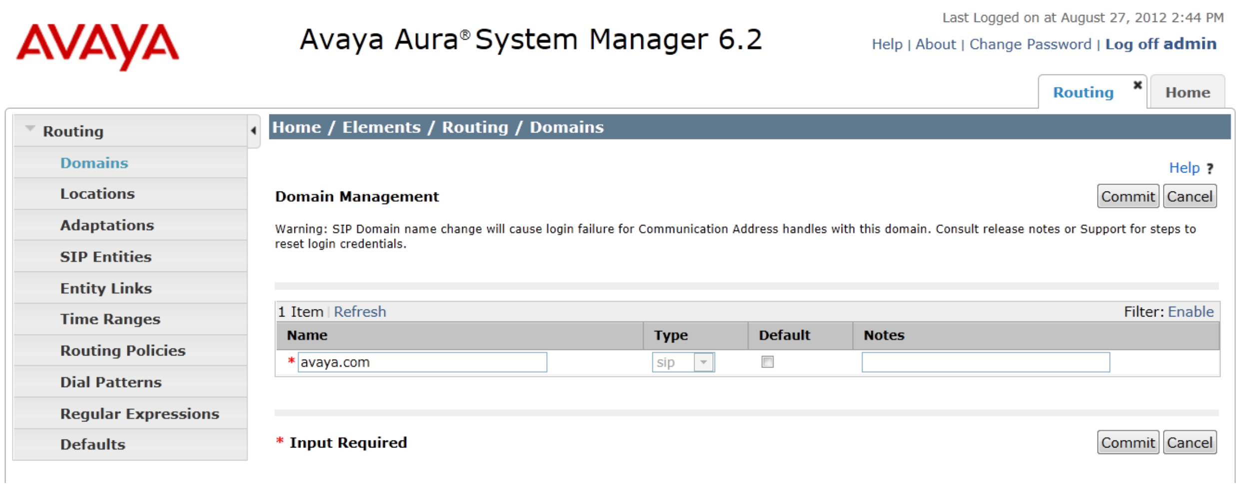
6.1 Specify SIP Domain

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:

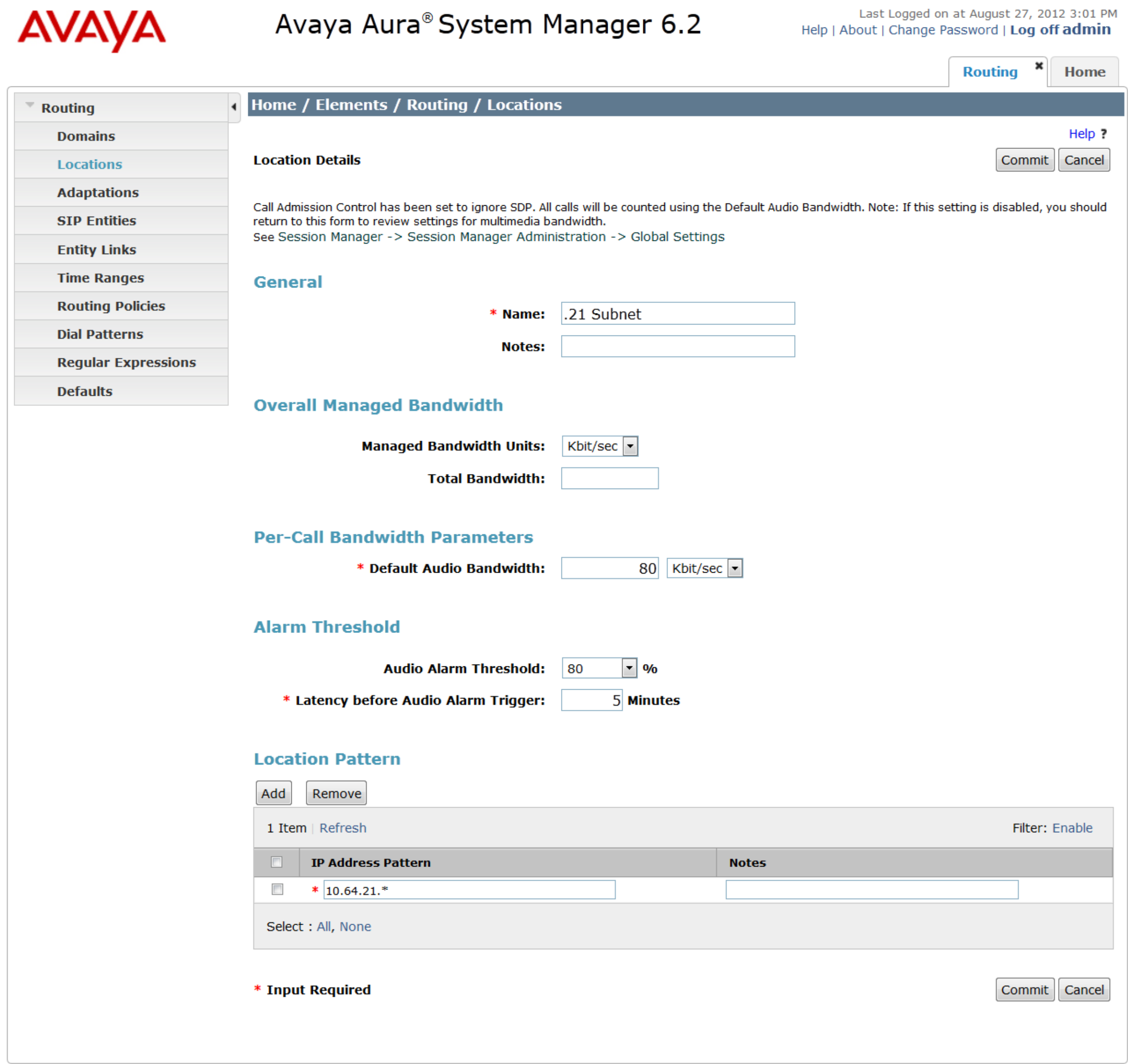
Under *General*:

- **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 the Communication Manager and Session Manager. Click **Commit** to save the Location definition.


AVAYA
Avaya Aura® System Manager 6.2
Last Logged on at August 27, 2012 3:01 PM
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 / Locations
Help ?
Location Details
Commit
Cancel
Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth. Note: If this setting is disabled, you should return to this form to review settings for multimedia bandwidth.
See Session Manager -> Session Manager Administration -> Global Settings
General
* Name: .21 Subnet
Notes:
Overall Managed Bandwidth
Managed Bandwidth Units: Kbit/sec
Total Bandwidth:
Per-Call Bandwidth Parameters
* Default Audio Bandwidth: 80 Kbit/sec
Alarm Threshold
Audio Alarm Threshold: 80 %
* Latency before Audio Alarm Trigger: 5 Minutes
Location Pattern
Add
Remove
1 Item | Refresh
Filter: Enable
IP Address Pattern
Notes
* 10.64.21.*
Select : All, None
* Input Required
Commit
Cancel

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:

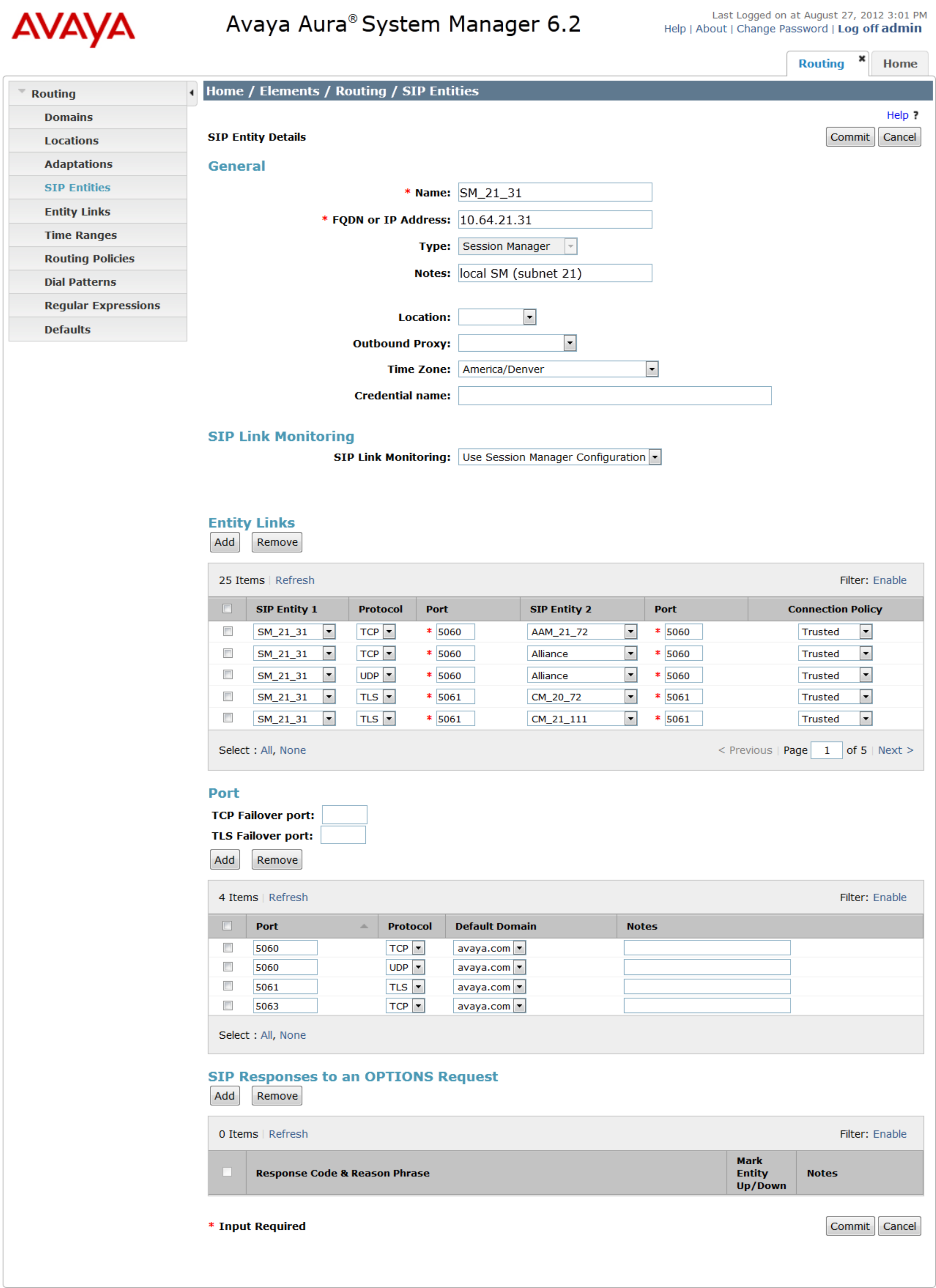
Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Notes :** Optional text.
- **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 6.2
Last Logged on at August 27, 2012 3:01 PM
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
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
25 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 TLS * 5061 CM_20_72 * 5061 Trusted
SM_21_31 TLS * 5061 CM_21_111 * 5061 Trusted
Select : All, None
< Previous | Page 1 of 5 | Next >
Port
TCP Failover port:
TLS Failover port:
Add
Remove
4 Items | Refresh
Filter: Enable
Port
Protocol
Default Domain
Notes
5060 TCP avaya.com
5060 UDP avaya.com
5061 TLS avaya.com
5063 TCP avaya.com
Select : All, None
SIP Responses to an OPTIONS Request
Add
Remove
0 Items | Refresh
Filter: Enable
Response Code & Reason Phrase
Mark Entity Up/Down
Notes
* Input Required
Commit
Cancel

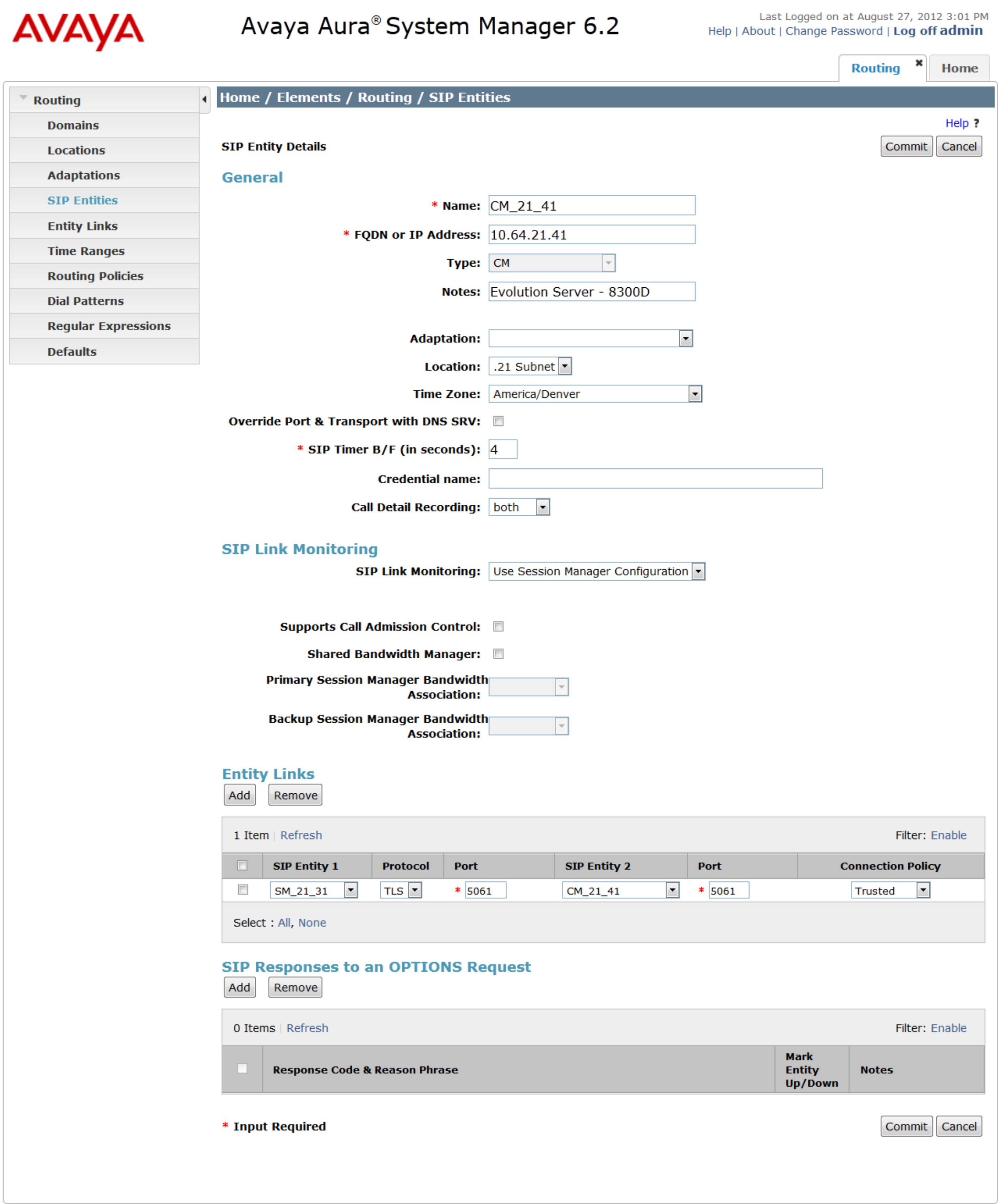
6.3.2 Communication Manager

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*.
- **Notes :** Optional text.
- **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.



AVAYA Avaya Aura® System Manager 6.2

Last Logged on at August 27, 2012 3:01 PM
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

Help ?

SIP Entity Details

Commit Cancel

General

* Name: CM_21_41
* FQDN or IP Address: 10.64.21.41
Type: CM
Notes: Evolution Server - 8300D

Adaptation:
Location: .21 Subnet
Time Zone: America/Denver
Override Port & Transport with DNS SRV:
* SIP Timer B/F (in seconds): 4
Credential name:
Call Detail Recording: both

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configuration

Supports Call Admission Control:
Shared Bandwidth Manager:
Primary Session Manager Bandwidth Association:
Backup Session Manager Bandwidth Association:

Entity Links

Add Remove

1 Item | Refresh Filter: Enable

	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
	SM_21_31	TLS	* 5061	CM_21_41	* 5061	Trusted

Select : All, None

SIP Responses to an OPTIONS Request

Add Remove

0 Items | Refresh Filter: Enable

	Response Code & Reason Phrase	Mark Entity Up/Down	Notes

* Input Required

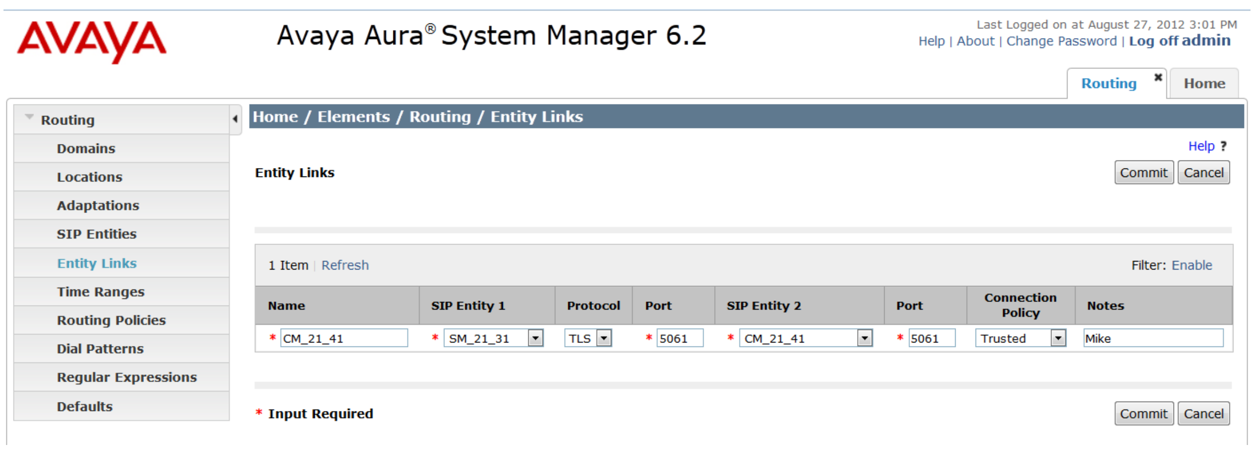
Commit Cancel

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.
- **Trusted:** Check this box. *Note: If this box is not checked, 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

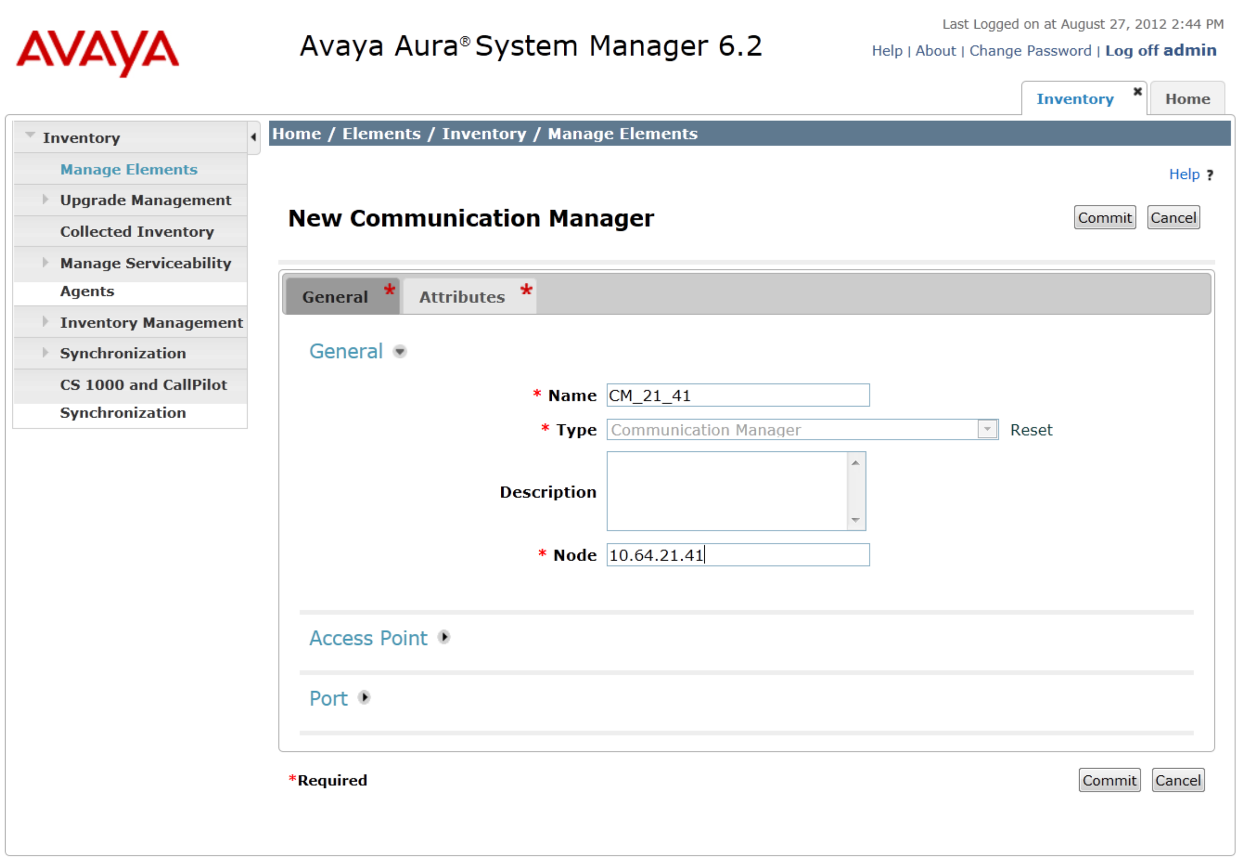
Before adding SIP users, Communication Manager must be added to System Manager as a managed element. This action allows System Manager to access Communication Manager over its administration interface. Using this administration interface, System Manager will notify Communication Manager when new SIP 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 Elements** screen (not shown), select *Communication Manager* in the **Type** field.

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

In the *Application* tab:

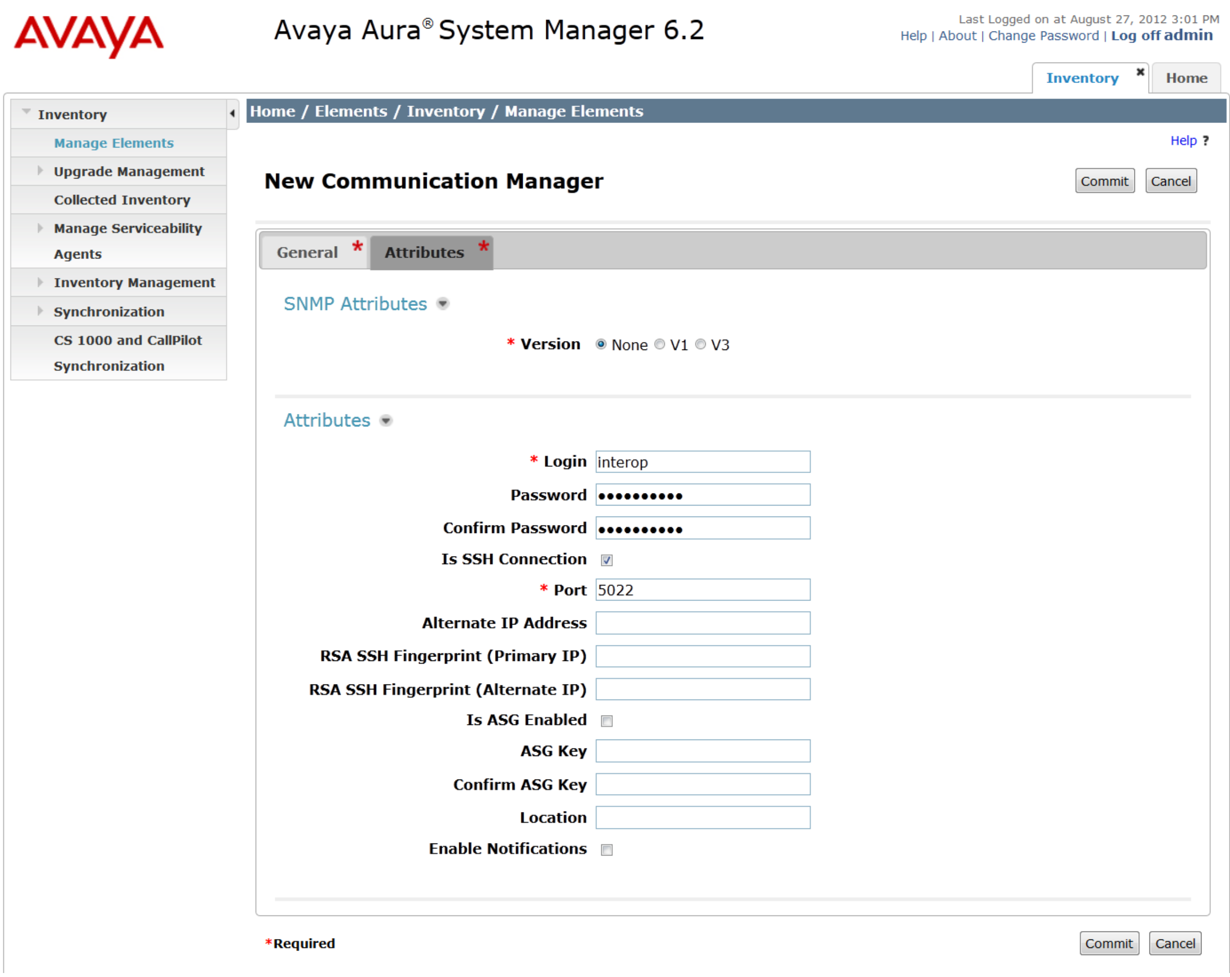
- **Name:** Enter an identifier for Communication Manager.
- **Type:** *Communication Manager* was previously selected.
- **Node:** Enter the IP address of the administration interface for Communication Manager.



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 SIP User

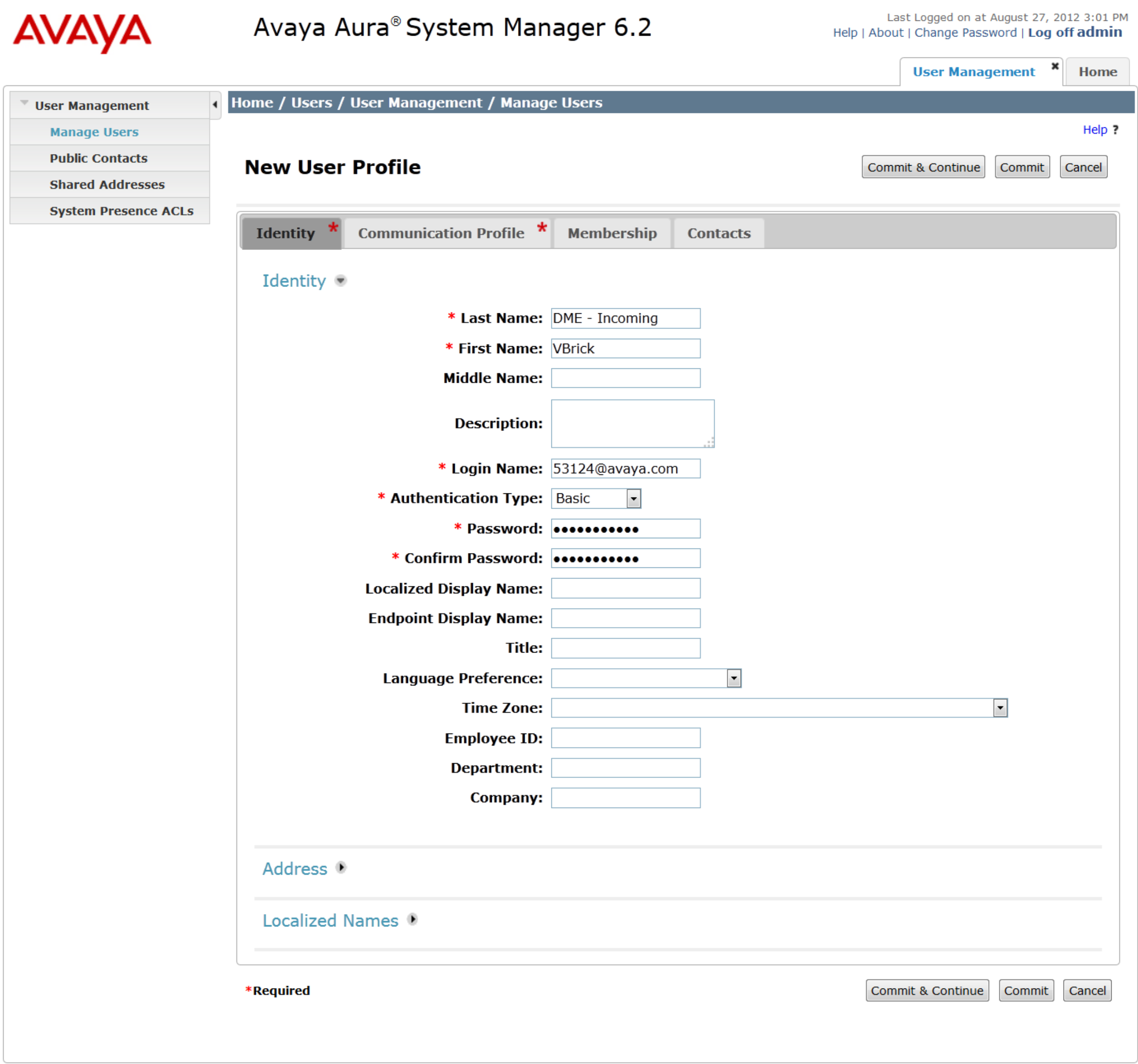
Add two SIP users for the VBrick Distributed Media Engine; one for incoming calls to the DME, and one for outgoing calls from the DME**.** The example screens below show the user created for incoming calls. The following configuration will automatically create the SIP station on Communication Manager.

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., 53167@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.


AVAYA
Avaya Aura® System Manager 6.2
Last Logged on at August 27, 2012 3:01 PM
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
Help ?
New User Profile
Commit & Continue
Commit
Cancel
Identity *
Communication Profile *
Membership
Contacts
Identity
* Last Name: DME - Incoming
* First Name: VBrick
Middle Name:
Description:
* Login Name: 53124@avaya.com
* Authentication Type: Basic
* Password: ••••••••••••
* Confirm Password: ••••••••••••
Localized Display Name:
Endpoint Display Name:
Title:
Language Preference:
Time Zone:
Employee ID:
Department:
Company:
Address
Localized Names
*Required
Commit & Continue
Commit
Cancel

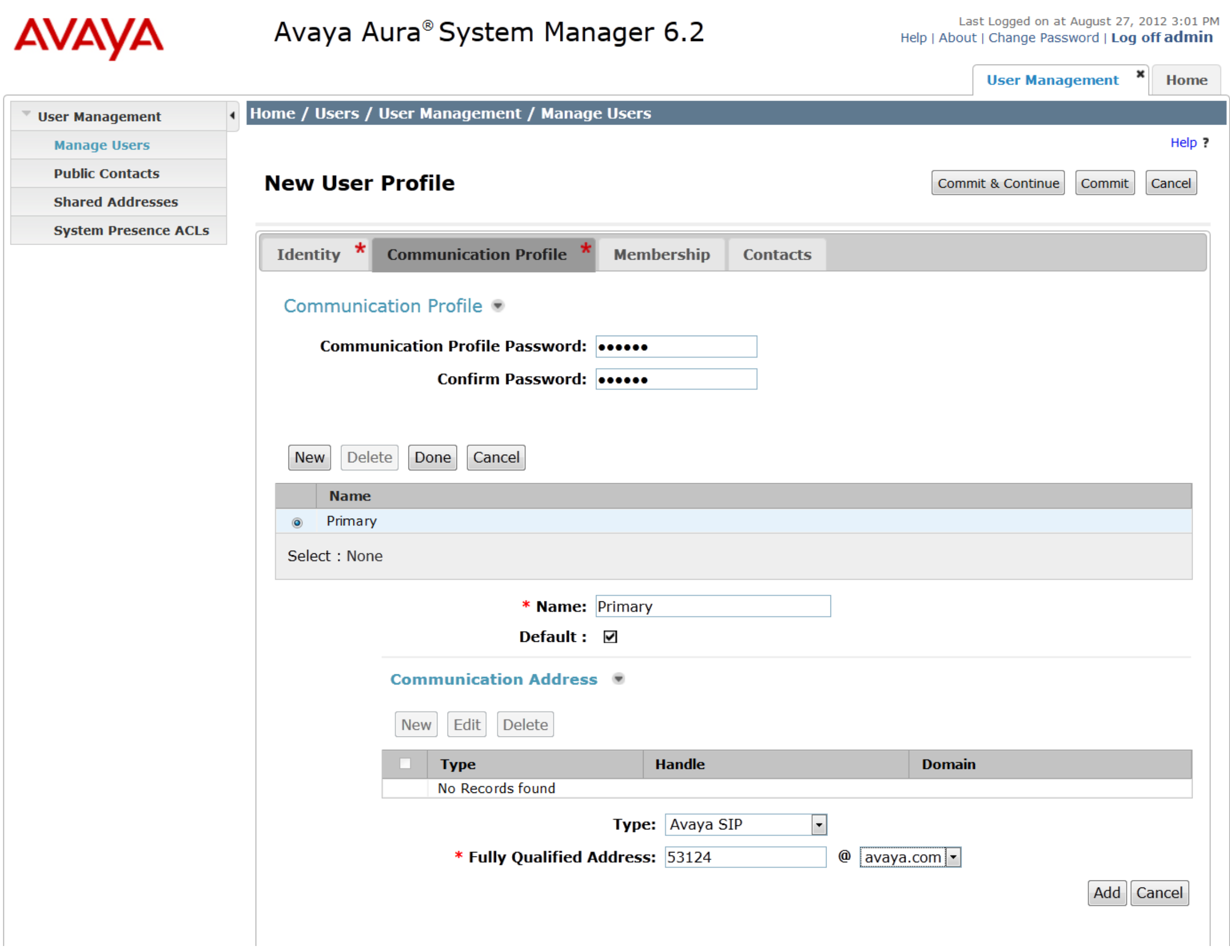
Enter values for the following required attributes for a new SIP user in the **Communication Profile** tab of the new user form.

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

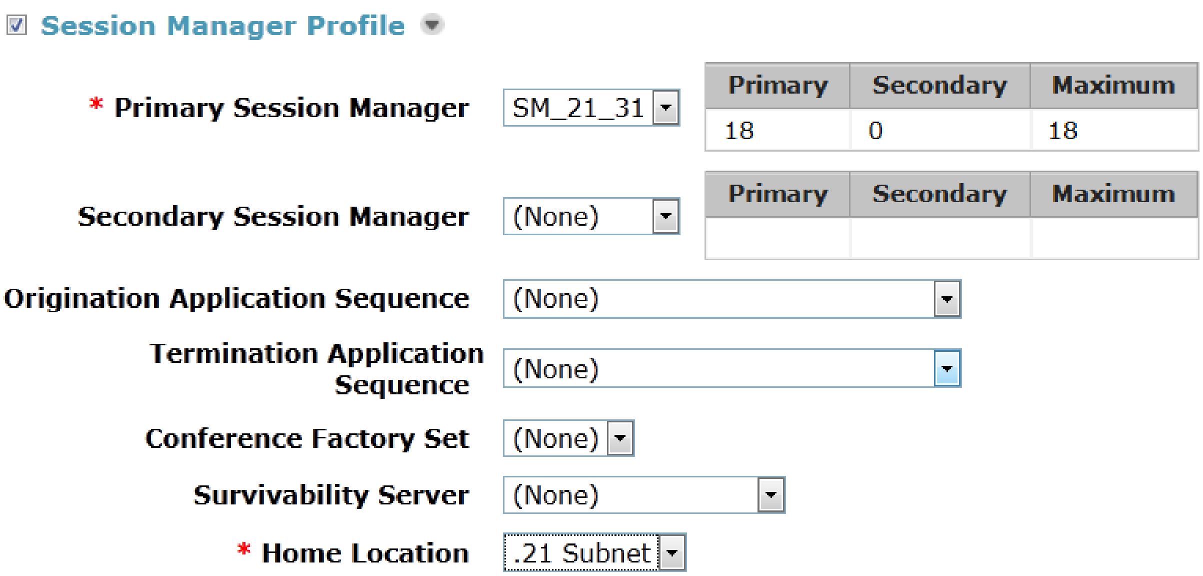
Scroll down to the **Communication Address** section and select **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **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**.

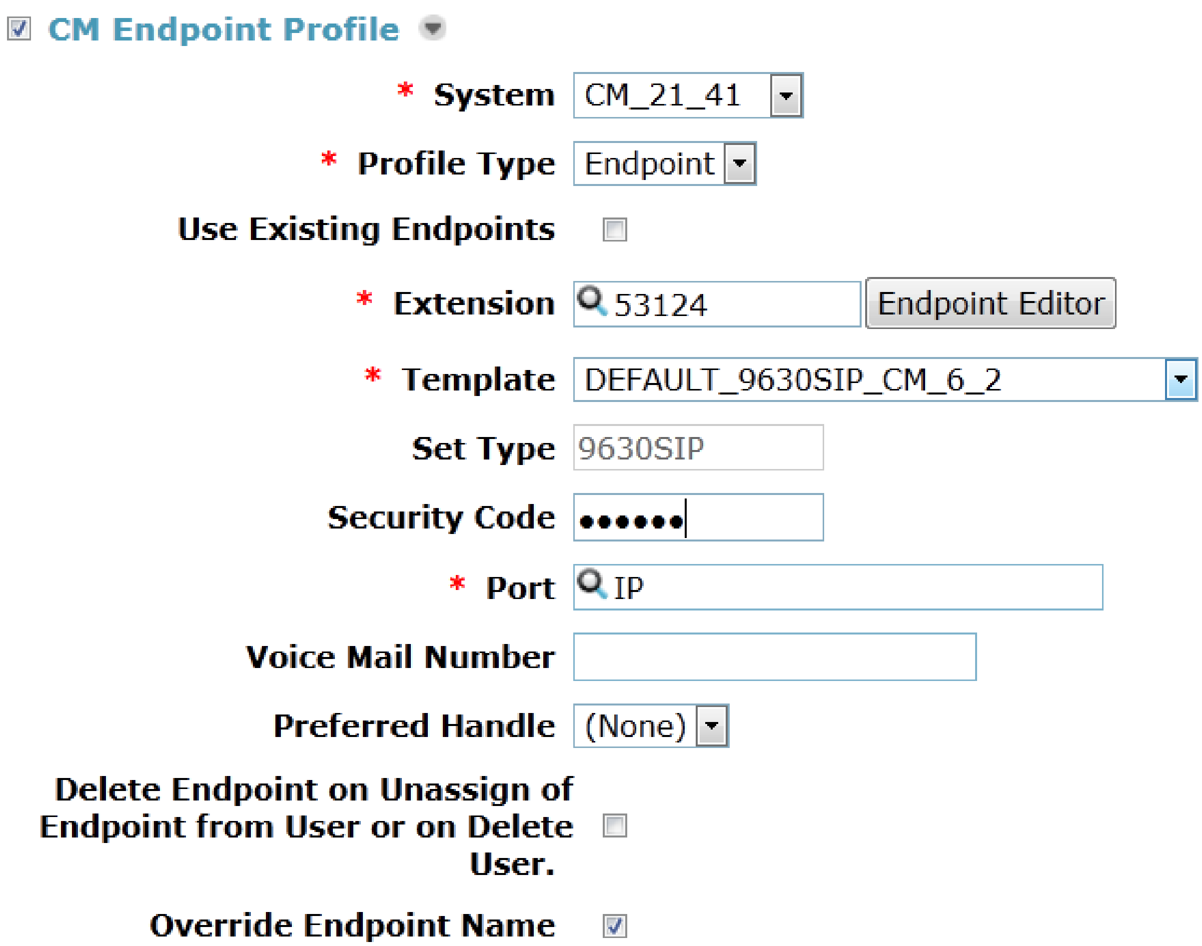


In the *Session Manager Profile* section, specify the Session Manager entity from **Section 6.3.1** for **Primary Session Manager**. Set the **Home Location** field to the **Location** configured in **Section 6.2**.

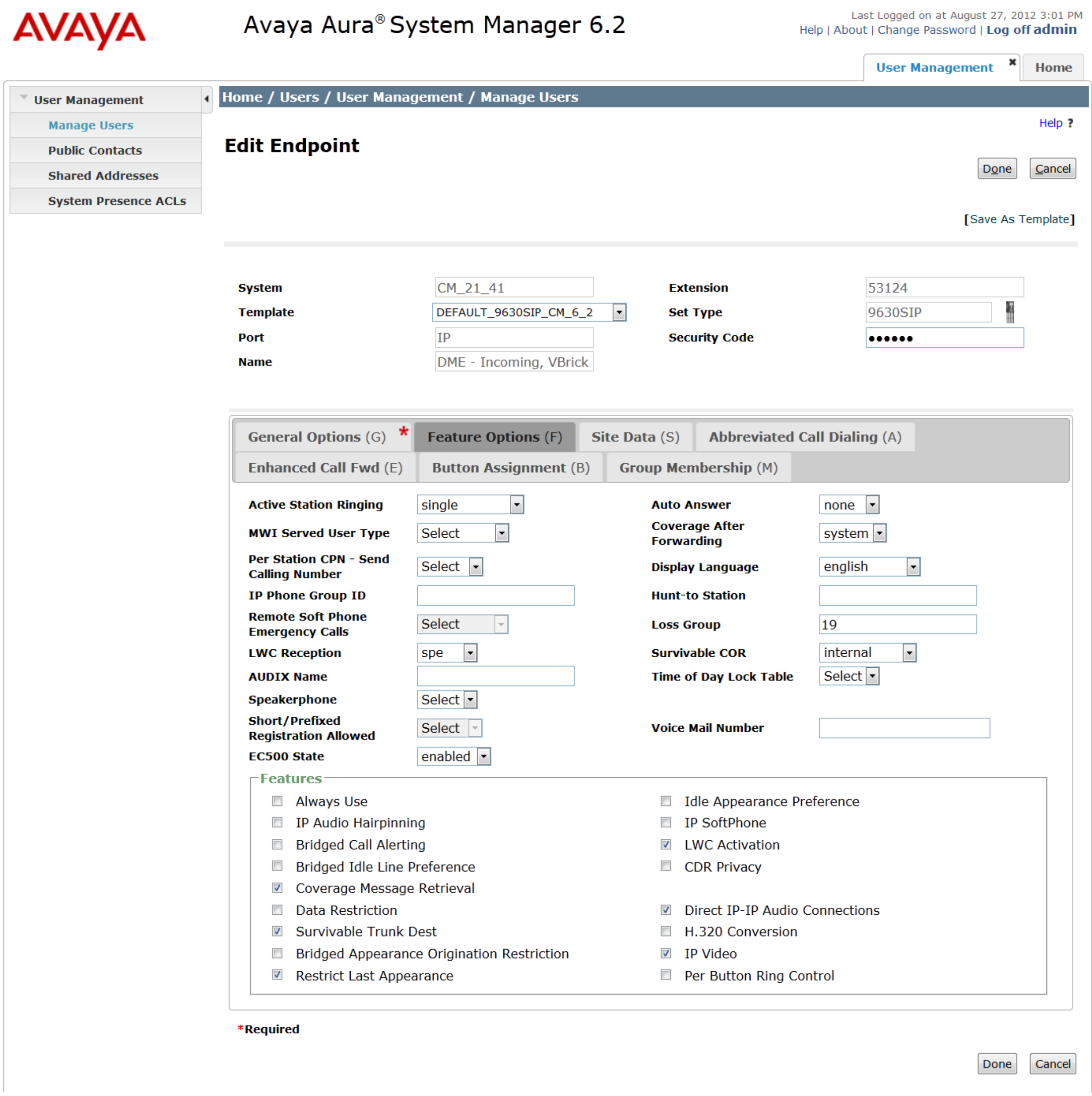


In the **Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager.
- **Profile Type** Select *Endpoint*.
- **Use Existing Stations:** If this field is not selected, the station will automatically be added in Communication Manager.
- **Extension:** Enter extension number of SIP user.
- **Template:** Select the *DEFAULT_9630SIP_CM_6_2* template.
- **Port:** Enter *IP*.



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



6.7 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:

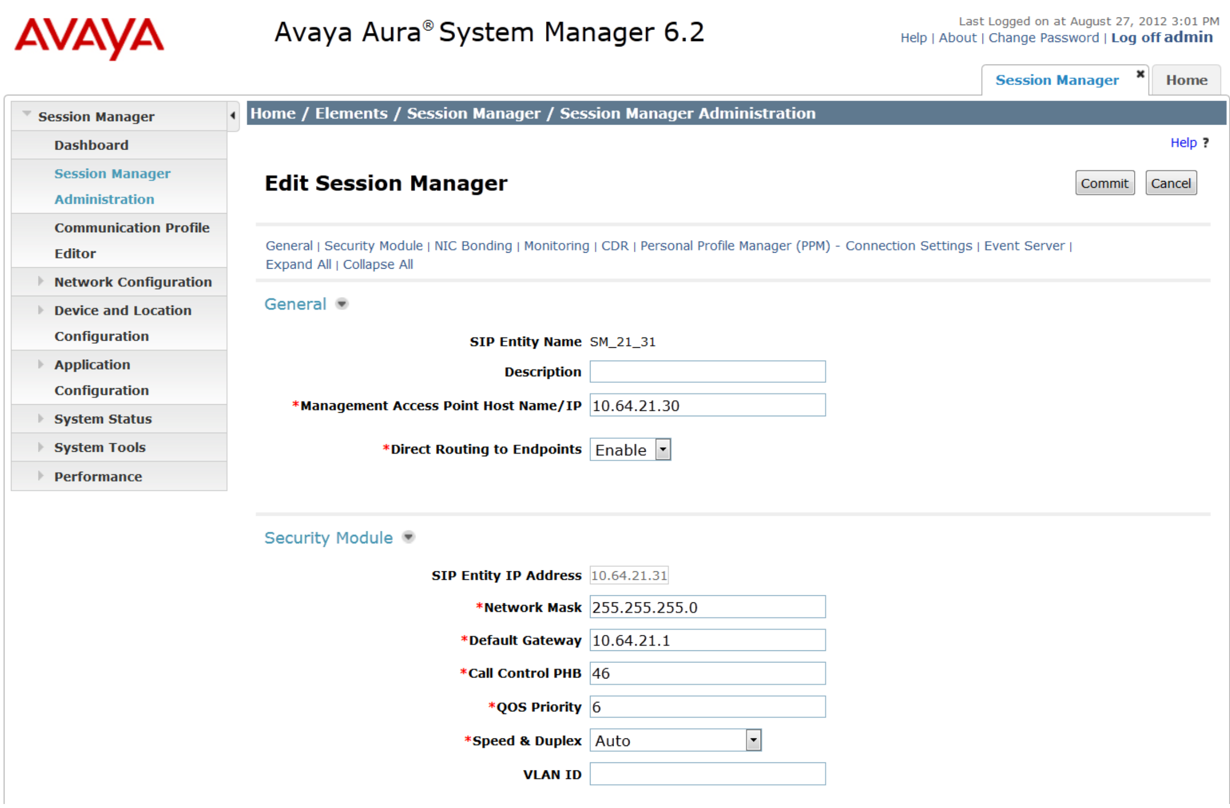
Under *Identity*:

- **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

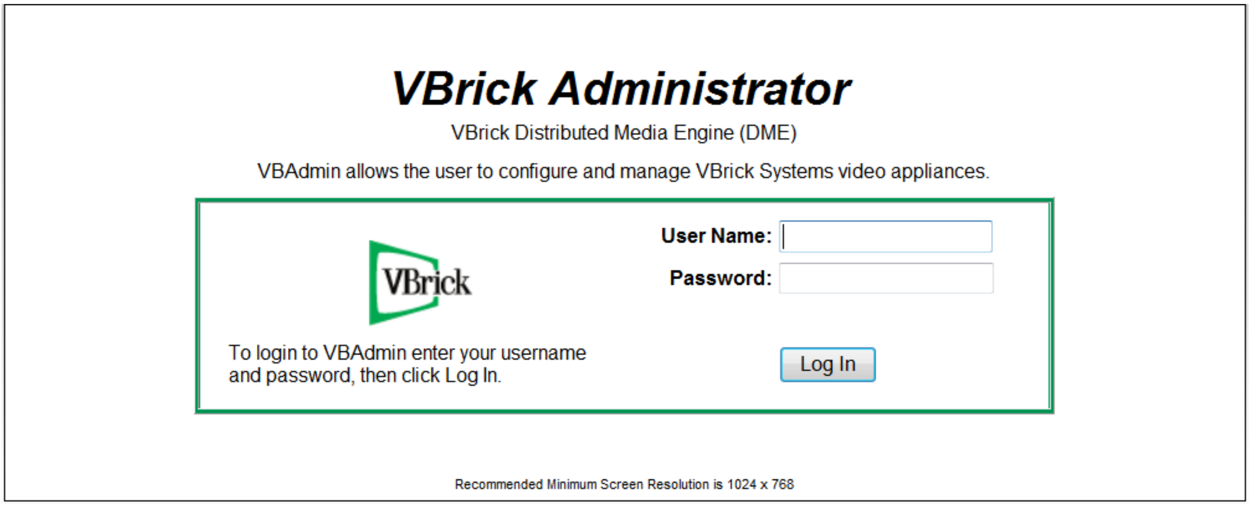
Use default values for the remaining fields. Click **Commit** to add this Session Manager.



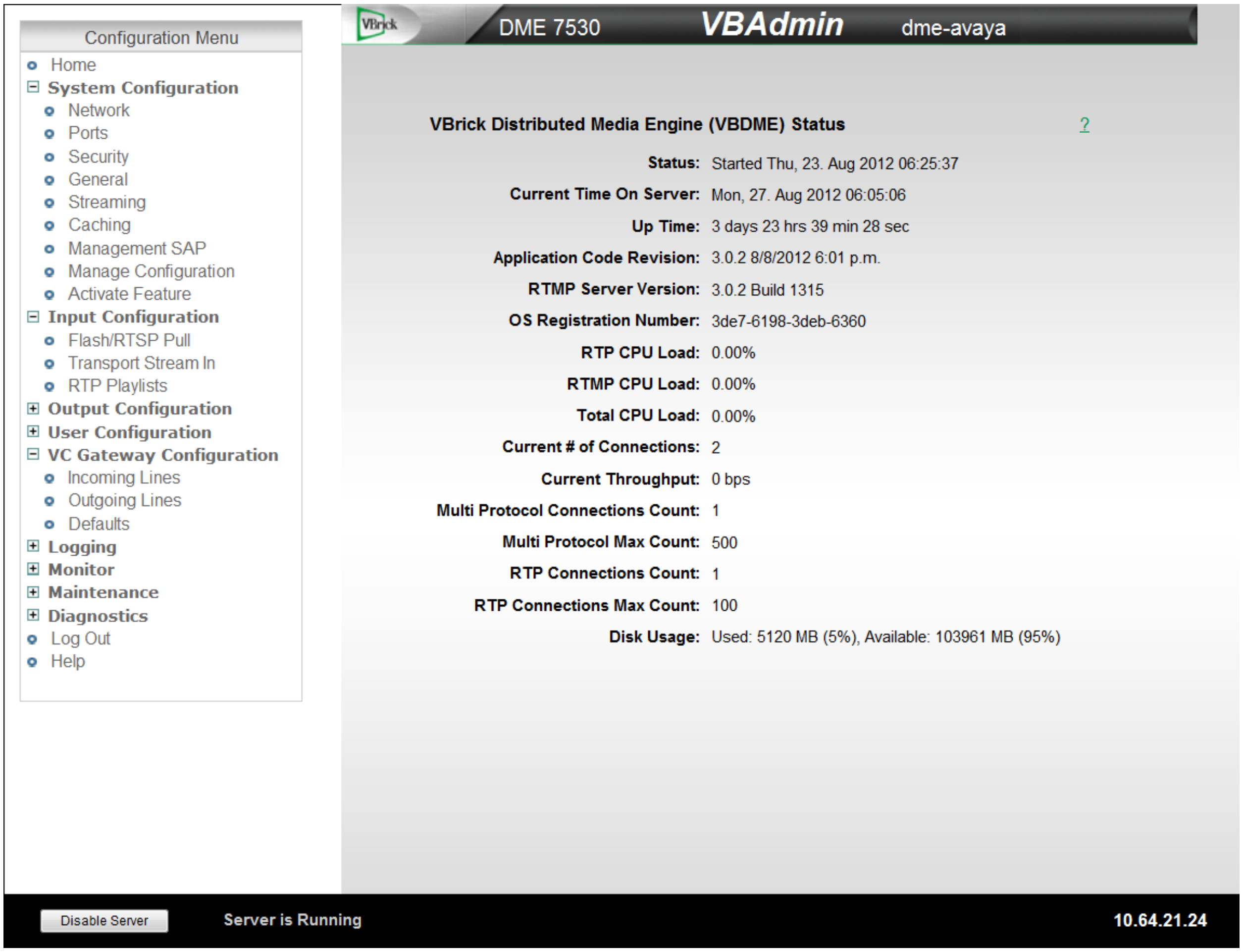
7. Configure VBrick Distributed Media Engine

The configuration of the VBrick Distributed Media Engine was performed via the VBrick DME's embedded web interface. Refer to reference **[3]** for additional information on configuring the DME.

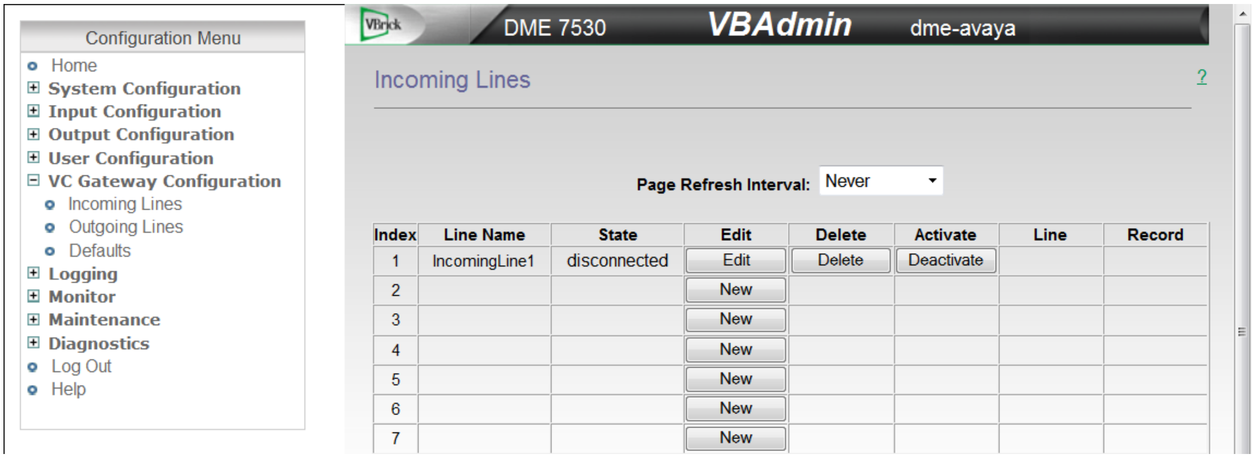
From an internet browser, enter https://*<ip-addr>* in the URL field, where *<ip-addr>* is the VBrick DME's IP address. The following **Login** screen is displayed. Log into the system with the appropriate user name and password.



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



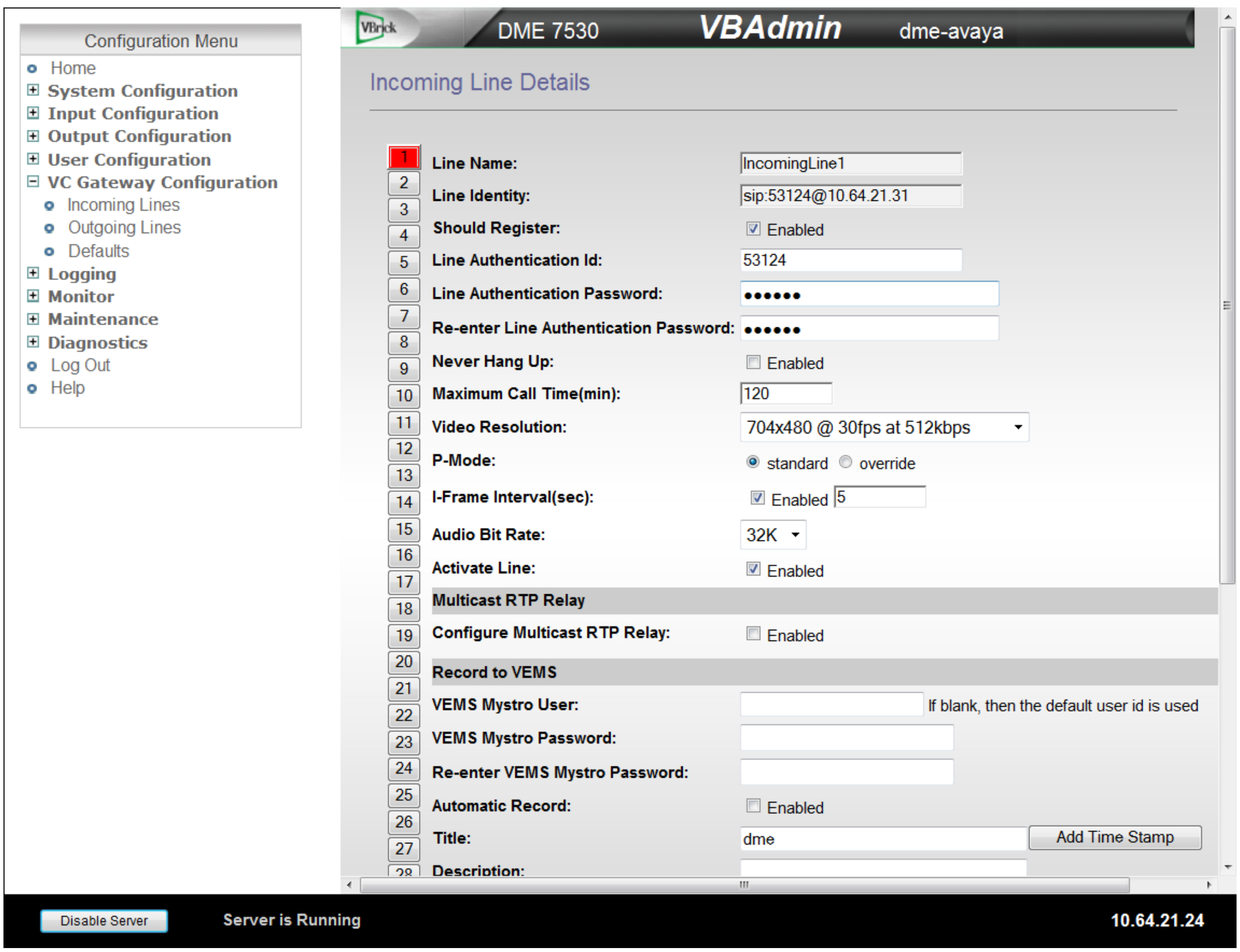
To set up the DME to receive incoming calls, navigate to **VC Gateway Configuration → Incoming Lines**. The following screen is displayed. Click the appropriate **Edit** button to edit an existing incoming line, or a **New** button to create a new incoming line.



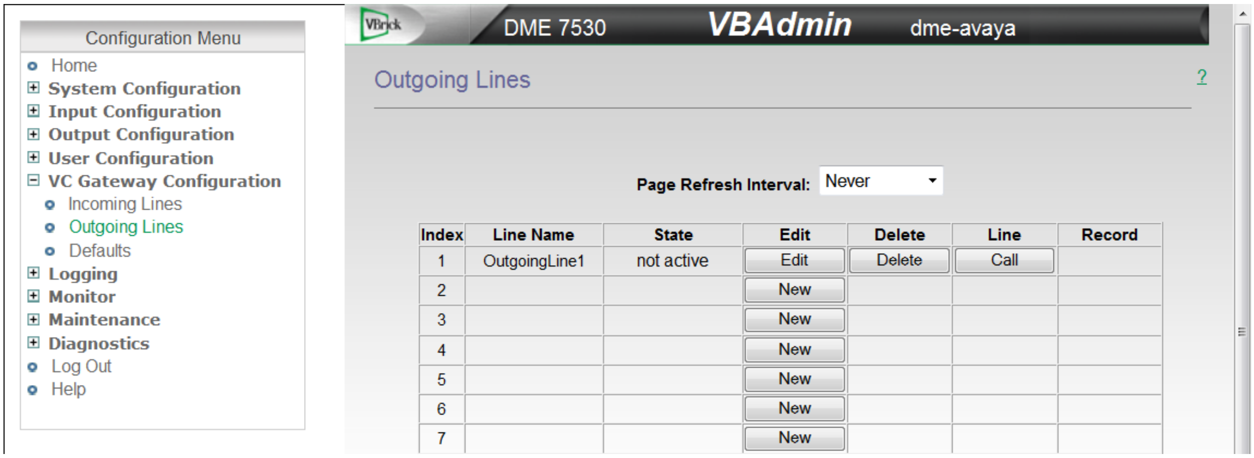
Enter the following information on the **Incoming Line Details** screen:

- **Line Name** Enter desired name.
- **Line Identity** Specify *sip:user@<Session Manager IP address>*
- **Should Register** Check *Enabled.*
- **Line Authentication ID** Specify extension used by VBrick DME to register with Session Manager
- **Line Authentication Password** Specify the password used by VBrick DME to register with Session Manager
- **Re-enter Line Authentication Password** Specify the password used by VBrick DME to register with Session Manager

Default values may be used for the remaining fields. Click the **OK** button (not shown) at the bottom of the screen to save the changes.



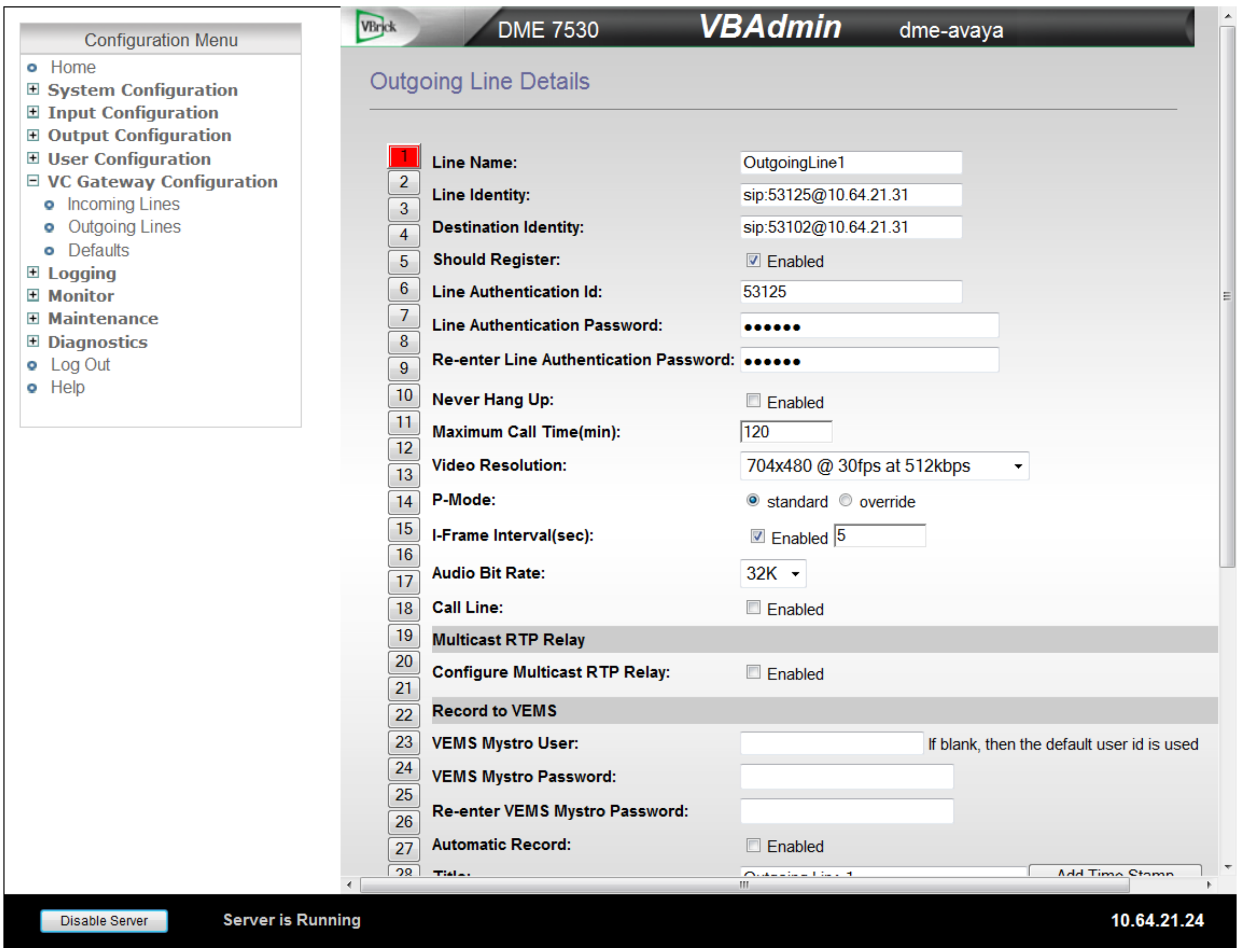
To set up the DME to initiate outgoing calls, navigate to **VC Gateway Configuration → Outgoing Lines**. The following screen is displayed. Click the appropriate **Edit** button to edit an existing outgoing line, or a **New** button to create a new outgoing line.



Enter the following information on the **Outgoing Line Details** screen:

- **Line Name** Enter desired name.
- **Line Identity** Specify *sip:user@<Session Manager IP address>*
- **Destination Identity** Specify *sip:user@<Session Manager IP address>* (this is the user to be called)
- **Should Register** Check *Enabled.*
- **Line Authentication ID** Specify extension used by VBrick DME to register with Session Manager
- **Line Authentication Password** Specify the password used by VBrick DME to register with Session Manager
- **Re-enter Line Authentication Password** Specify the password used by VBrick DME to register with Session Manager

Default values may be used for the remaining fields. Click the **OK** button (not shown) at the bottom of the screen to save the changes.



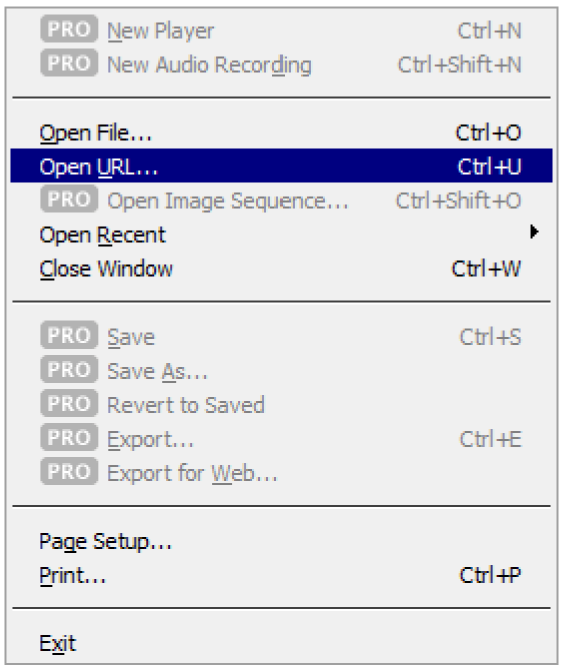
8. Verification Steps

This section provides the steps that may be performed to verify proper configuration of the VBrick Distributed Media Engine with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

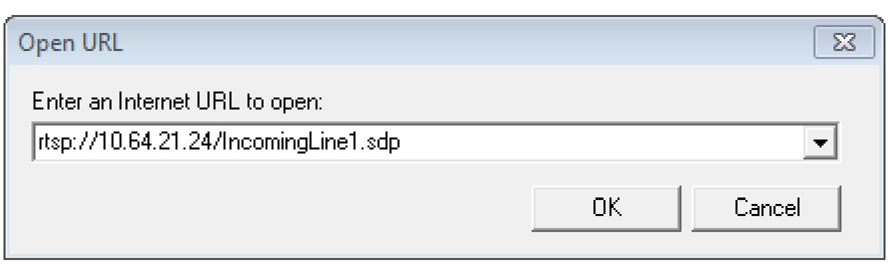
1. Place a video call from an Avaya endpoint to the VBrick DME. Verify that the call is successfully established with 2-way audio and video.

2. Place a video call from the VBrick DME to an Avaya endpoint. Verify that the call is successfully established with 2-way audio and video.

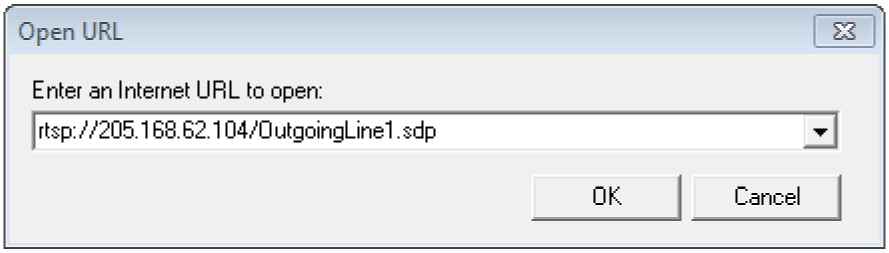
3. Repeat steps 1 and 2 above and while the each call is established, use QuickTime Player to view the streaming video call. Open QuickTime Player and select **File → Open URL…**.



To view the stream of the call incoming to the DME, enter the following URL: *rstp://<DME_IP>/<Line_Name>.sdp*, where *<DME_IP>* is the IP address of the DME and *<Line_Name>*is the incoming line name.



To view the stream of the call outoging from the DME, enter the following URL: *rstp://<DME_IP>/<Line_Name>.sdp*, where *<DME_IP>* is the IP address of the DME and *<Line_Name>*is the outgoing line name.



9. Conclusion

These Application Notes have described the administration steps required to integrate the VBrick Distributed Media Engine with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. VBrick Distributed Media Engine successfully registered with Session Manager and voice and video calls were established with Avaya one-X Communicator (SIP), Avaya Desktop Video Devices, and Avaya 1020 and 1050 video conference systems. All test cases passed with exceptions/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*, July 2012, Document Number 03-300509.
[2] *Administering Avaya Aura® Session Manager*, July 2012, Document Number 03-603324.

The following VBrick product documentation is available at http://www.vbrick.com.

[3] VBrick Distributed Media Engine, VBrick H.264 v3.0 DME Admin Guide, May 2012, Document Number 4410-0294-0003.

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.