Avaya Solution & Interoperability Test Lab

Application Notes for AudioCodes 300HD SIP IP Phone Series with Avaya Aura^{TM} Session Manager and Avaya Aura^{TM} Communication Manager Feature Server - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate an AudioCodes 320HD SIP Phone with a SIP infrastructure consisting of Avaya Aura^{TM} Session Manager and Avaya Aura^{TM} Communication Manager Feature Server. The 320HD SIP Phone is part of the AudioCodes 300HD SIP IP Phone Series. During compliance testing, the AudioCodes SIP Phones successfully registered with Session Manager, established calls with other telephones, and executed telephony features such as Hold, Transfer, and Conference.

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

1. Introduction

These Application Notes describe the steps required to integrate an AudioCodes 320HD SIP Phone with a SIP infrastructure consisting of Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager Feature Server. The 320HD SIP Phone is part of the AudioCodes 300HD SIP IP Phone Series. During compliance testing, the AudioCodes SIP Phones successfully registered with Session Manager, established calls with other telephones, and executed telephony features such as Hold, Transfer, and Conference.

These Application Notes assume that Communication Manager and Session Manager are already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult the appropriate document in the reference section at the end of this document.

1.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful registration of the AudioCodes 320HD SIP Phone with Session Manager.
- Calls between 320HD SIP phones and Avaya SIP, H.323, and digital stations.
- G.711, G.729A, and G.722-74K codec support.
- Proper recognition of DTMF tones by navigating voicemail menus.
- Proper operation of voicemail with Message Waiting Indication (MWI).
- Basic telephony features including Hold, Transfer, and Conference.
- Extended telephony features using Communication Manager Feature Name Extensions (FNEs) such as Call Forwarding, Call Pickup, and Send All Calls.
- Proper system recovery after a 320HD SIP phone restart and loss of IP connectivity.

1.2. Support

For technical support on the 300HD SIP IP Phone Series contact AudioCodes Customer Support through their website at http://www.audiocodes.com/support.

2. Reference Configuration

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

- Communication Manager is running on an Avaya S8510 Server with a G450 Media Gateway, serving as a Feature Server for SIP endpoints.
- An Avaya S8720 Server with a G650 Media Gateway serves as an Access Element supporting H.323 and digital telephones.
- Session Manger interconnects the Feature Server and Access Element via SIP trunks and acts as a Registrar/Proxy for SIP telephones.
- Avaya Aura™ System Manager is used to configure Session Manager.
- Avaya Modular Messaging provides voice mail service.

In addition, three AudioCodes 320HD SIP Phones registered with Session Manager and were configured as Off-PBX Stations (OPS) on the Communication Manager Feature Server.

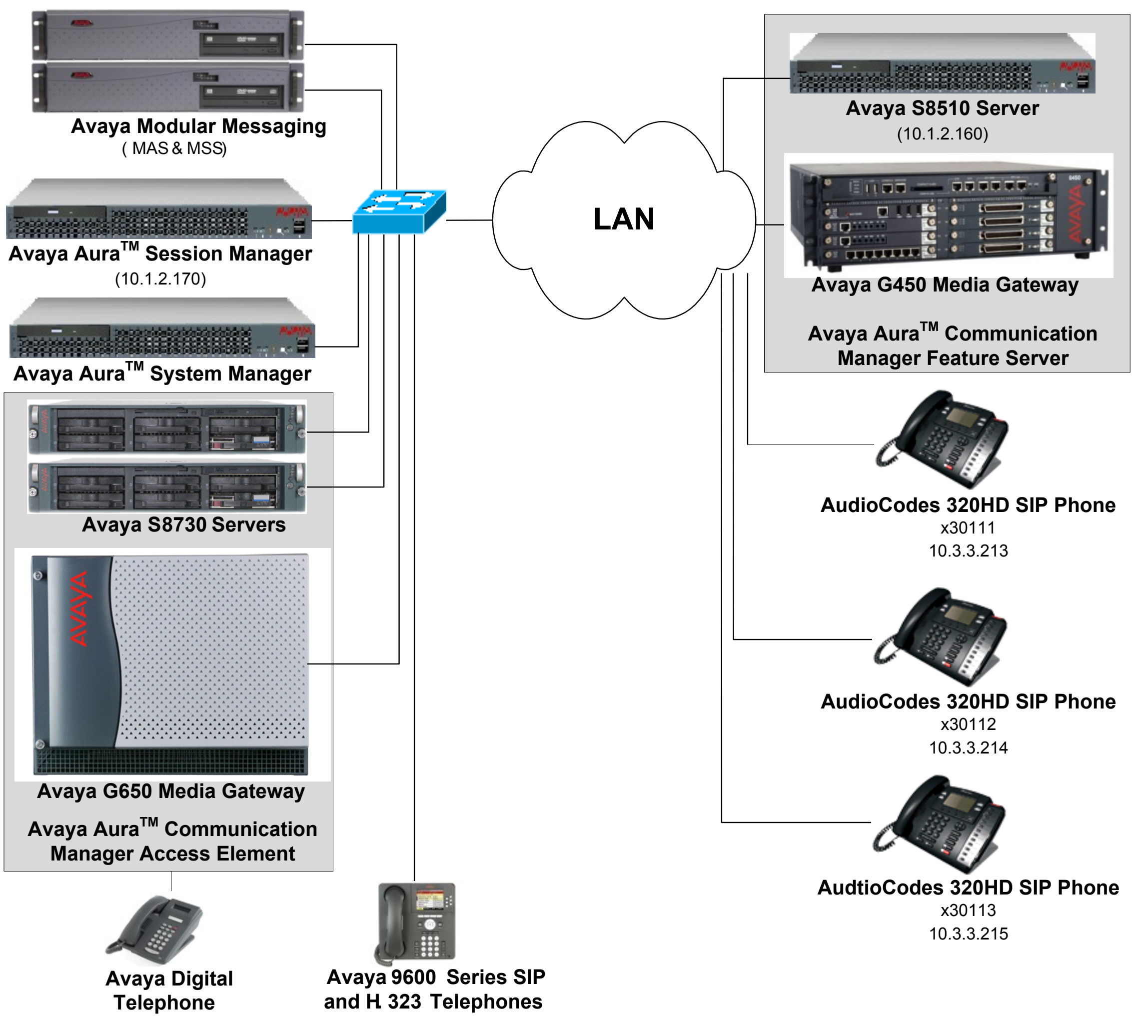


Figure 1: AudioCodes 320HD SIP Phones with Avaya SIP Solution

2.1. SIP Call Flows

The AudioCodes 320HD SIP Phone originates a call by sending a call request (SIP INVITE message) to Session Manager, which then routes the call over a SIP trunk to the Communication Manager Feature Server for origination services. If the call is destined for another local SIP phone, Communication Manager routes the call back over the SIP trunk to Session Manager for delivery to the destination SIP phone. If the call is destined for an H.323 or digital telephone, Communication Manager routes the call back to Session Manager for delivery to the Communication Manager Access Element supporting H.323 and digital endpoints.

For a call arriving at Communication Manager Feature Server that is destined for the AudioCodes 320HD SIP Phones, Communication Manager routes the call over the SIP trunk to Session Manager for delivery to the AudioCodes 320HD SIP Phones.

3. Equipment and Software Validated

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

Hardware Component	Version
Avaya S8730 Servers and G650 Media Gateway (Access Element)	Avaya Aura^{TM} Communication Manager 5.2.1 (R015x.02.1.016.4) with Service Pack 2 (Patch 18111)
Avaya S8510 Servers with G450 Media Gateway (Feature Server)	Aura^{TM} Communication Manager 5.2.1 (R015x.02.1.016.4) with Patch 18151
Avaya Aura^{TM} Session Manager	5.2.2
Avaya Modular Messaging running on Avaya S3500 Servers	5.2
Avaya 9600 Series IP Telephones	3.0 (H.323) 2.0.5 (SIP)
Avaya Digital Telephones	N/A
AudioCodes 320HD SIP Phone	1.2.2_p009_Build_5

4. Configure Avaya Aura^{TM} Communication Manager Feature Server

This section describes the steps for configuring the AudioCodes 320HD SIP Phone as an Off-PBX Station (OPS) and configuring a SIP trunk between the Communication Manager Feature Server and Session Manager. **Section 4.3** covers the station configuration for the 320HD SIP Phones. See [2] for additional information on configuring SIP support on Communication Manager. Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

Note: These Application Notes do not cover call routing to the Communication Manager Access Element.

4.1. Verify OPS and SIP Trunk Capacity

Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks features 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  10
                                OPTIONAL FEATURES

     G3 Version: V15                                 Software Package: Standard
       Location: 1                                RFA System ID (SID): 1
       Platform: 12                               RFA Module ID (MID): 1

                                                               USED
                                Platform Maximum Ports: 44000 316
                                      Maximum Stations: 36000 86
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 10    0
                    Maximum Off-PBX Telephones -   OPS: 200   80
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 0     0

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

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

```
display system-parameters customer-options                      Page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 450   100
           Maximum Concurrently Registered IP Stations: 18000 0
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 5     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 5     0
                       Maximum Administered SIP Trunks: 300   130
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 5     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: 0     0

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

4.2. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for the S8510 Server processor and the Session Manager server. 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
SM1                 10.1.2.170
default             0.0.0.0
procr               10.1.2.160

( 3 of 3   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. *For this solution, shuffling should be disabled to prevent issues with the Hold and Conference features*. 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  19
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: avaya.com
    Name: HQ CM and SIP Phones
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: no
      Codec Set: 1              Inter-region IP-IP Direct Audio: no
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        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 320HD SIP Phones. 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. However, the **IP Codec Set** form may specify multiple codecs, including G.711, G.729A, and G.722-64K, which are supported by the 320HD SIP Phones.

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

                          IP Codec Set

    Codec Set: 1

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

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

- Set the **Group Type** field to *sip*.
- The **Transport Method** field was set to *tcp*.
- Specify the S8510 Server processor and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the recommended TLS port value of *5060* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- Shuffling was disabled in the IP Network Region form which supersedes the setting of the **Direct IP-IP Audio Connections** field on this form. However, Shuffling could have been disabled here instead so that only calls using this signaling group are impacted.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 60                                       Page   1 of   1
                                SIGNALING GROUP

 Group Number: 60            Group Type: sip
                       Transport Method: tcp
  IMS Enabled? y

   Near-end Node Name: procr                  Far-end Node Name: SM1
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 1
Far-end Domain: 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? n                       Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 10
```

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to the 320HD SIP Phones. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

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

Group Number: 60                   Group Type: sip          CDR Reports: y
  Group Name: SM1                         COR: 1        TN: 1        TAC: 160
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                    Signaling Group: 60
                                                  Number of Members: 100
```

On Page 3 of the trunk group form, set the **Numbering Format** field to *public*. This field specifies the format of the calling party number sent to the far-end.

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

                     Numbering Format: private
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n

 Show ANSWERED BY on Display? y
```

Configure the **Public/Unknown 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 '3' and whose calls are routed over SIP trunk group "60" have the number 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  3                  60                           5     Total Administered: 1
                                                             Maximum Entries: 540
```

4.3. Configure Stations

Use the **add station** command to add a station for each 320HD SIP Phone to be supported. Use *9630SIP* for the **Station Type** and include the **Coverage Path** for voice mail, if applicable. The **Name** field is optional. Use the default values for the other fields on Page 1. The SIP station can also be configured automatically by Session Manager as described in **Section 5.8**.

```
add station 30111                                           Page   1 of   6
                                     STATION

Extension: 30111                        Lock Messages? n              BCC: 0
     Type: 9630SIP                      Security Code:                 TN: 1
     Port: IP                         Coverage Path 1: 60             COR: 1
     Name: AudioCodes 30111           Coverage Path 2:                COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19
                                                Message Lamp Ext: 30111

        Display Language: english                 Button Modules: 0

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

On Page 2, set the **MWI Served User Type** field to the appropriate value to allow MWI notifications to be sent to the 320 HD SIP Phone.

Note: Shuffling was disabled on the IP Network Region form which supersedes the setting of the **Direct IP-IP Audio Connections** field on Page 2 of the Station form. However, Shuffling could be disabled on a station basis if the customer does not want to disable Shuffling for the entire IP network region. For this solution, Shuffling should be disabled as mentioned earlier.

```
add station 30111                                           Page   2 of   6
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe
          LWC Activation? y                      Coverage Msg Retrieval? y
                                                            Auto Answer: none
            CDR Privacy? n                             Data Restriction? n
                                              Idle Appearance Preference? n
                                            Bridged Idle Line Preference? n
   Bridged Call Alerting? n
  Active Station Ringing: single

        H.320 Conversion? n          Per Station CPN - Send Calling Number?
                                                        EC500 State: enabled
    MWI Served User Type: sip-adjunct
                                                Coverage After Forwarding? s

                                             Direct IP-IP Audio Connections? y
  Emergency Location Ext: 30111         Always Use? n IP Audio Hairpinning? n
     Precedence Call Waiting? y
```

Use the **change off-pbx-telephone station-mapping** command to map the Communication Manager extensions (e.g., 30111) to the same extension configured in Session Manager. Enter the field values shown. For the sample configuration, the **Trunk Selection** value indicates the SIP trunk group between the Communication Manager Feature Server and Session Manager. The SIP trunk group was configured in **Section 4.2**. The **Configuration Set** value can reference a set that has the default settings.

```
change off-pbx-telephone station-mapping 30111                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 30111            OPS          -     30111             60         1
```

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

```
change off-pbx-telephone station-mapping 30111                  Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 30111         OPS      3        both         all         none
```

5. 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 the Communication Manager Feature Server
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Application Sequence
- Define Feature Server as Administrable Entity
- Session Manager, corresponding to the Avaya Aura™ Session Manager Server to be managed by Avaya Aura™ System Manager
- Add SIP Users

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

Note: These Application Notes do not cover the configuration of the Communication Manager Access Element, but the configuration would be similar to that of the Feature Server which is covered.

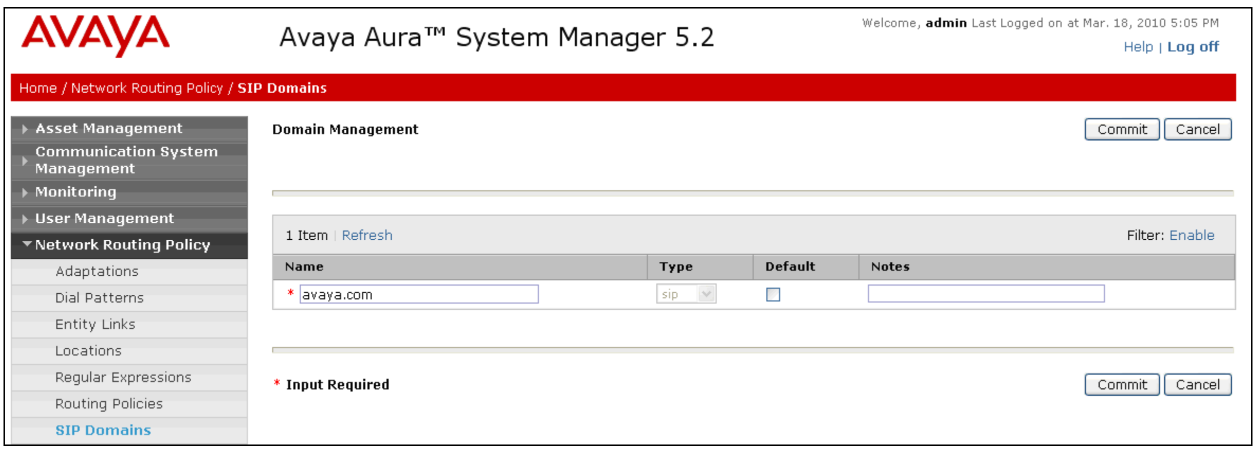
5.1. Specify SIP Domain

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

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

Click **Commit**.

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



5.2. 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 on the right. The following screen will then be shown. Fill in the following:

Under *General*:

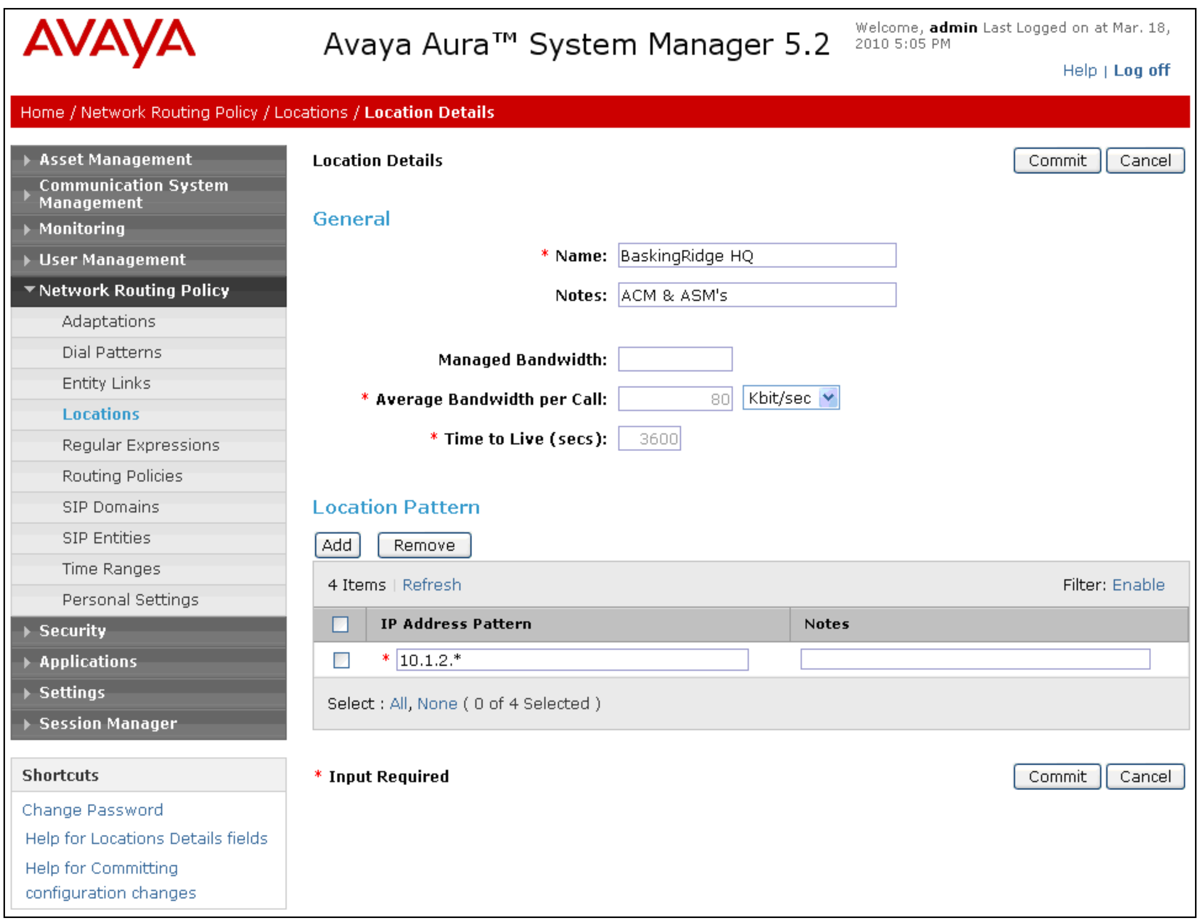
- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

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 *Basking Ridge HQ* location, which includes the Avaya Aura™ Communication Manager Feature Server and Avaya Aura™ Session Manager. Click **Commit** to save the Location definition.

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



5.3. Add SIP Entities

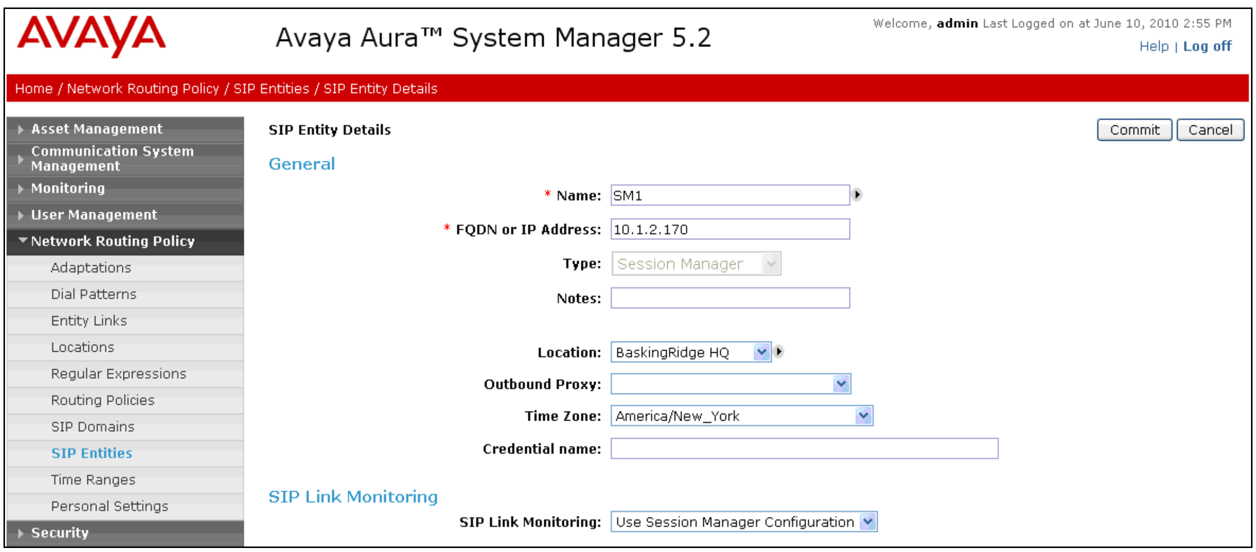
In the sample configuration, a SIP Entity is added for Session Manager and the S8510 Server associated with the Feature Server.

5.3.1. Avaya Aura™ Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button on the right. The following screen is displayed. Fill in the following:

Under *General*:

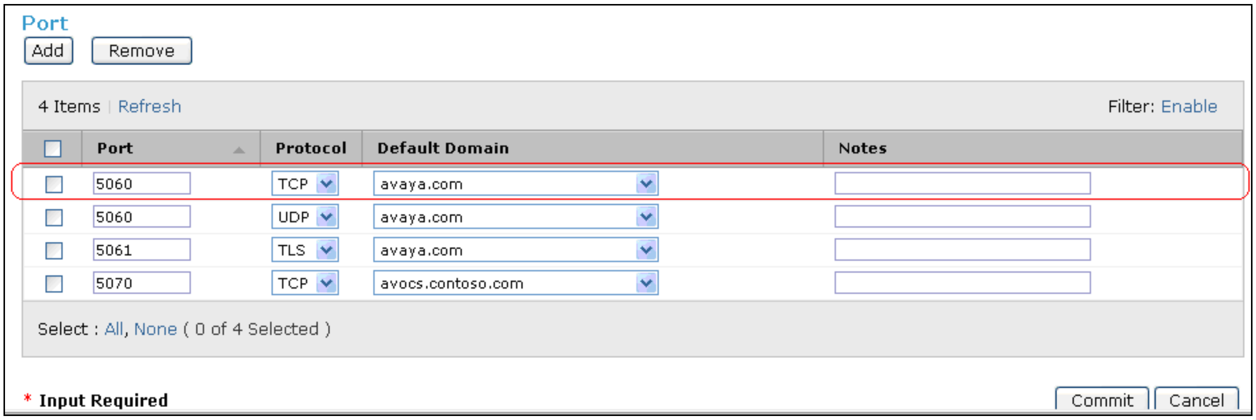
- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Specify *Session Manager*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.



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

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

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



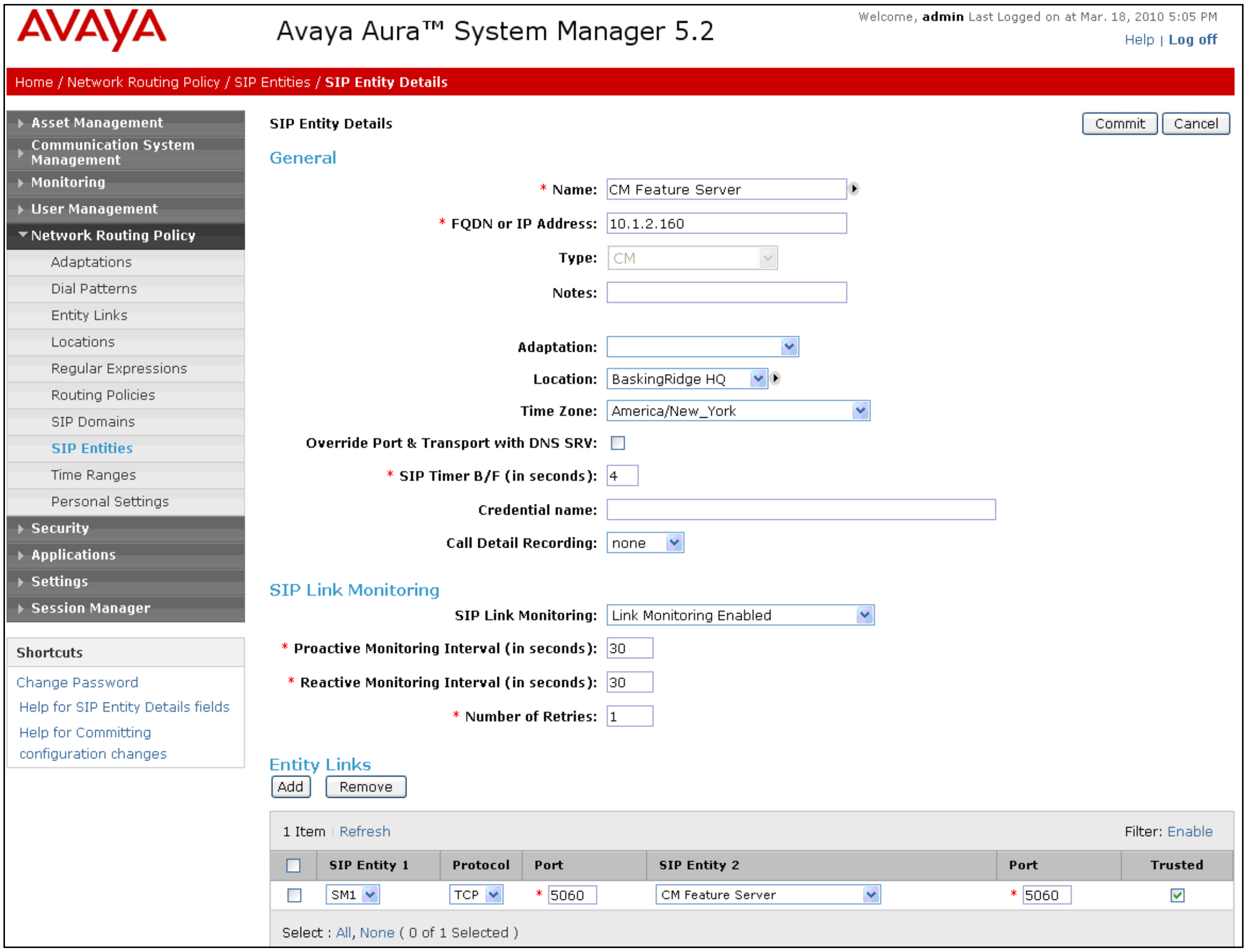
5.3.2. Avaya Aura™ Communication Manager Feature Server

A SIP Entity must be added for the Communication Manager Feature Server. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button 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 ASM or the signaling interface on the telephony system.
- **Type:** Specify *CM*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

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

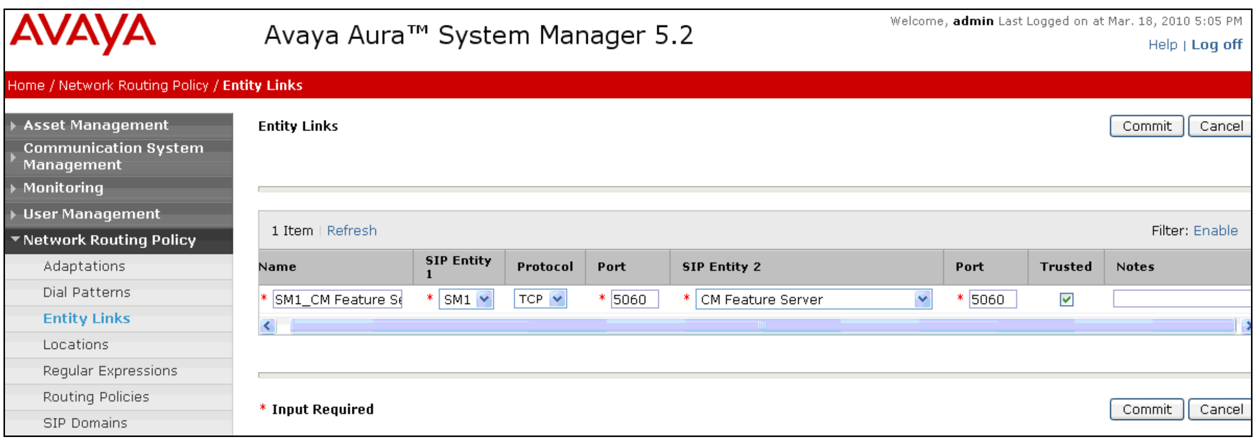

AVAYA
Avaya Aura™ System Manager 5.2
Welcome, admin Last Logged on at Mar. 18, 2010 5:05 PM
Help | Log off
Home / Network Routing Policy / SIP Entities / SIP Entity Details
Asset Management
Communication System Management
Monitoring
User Management
Network Routing Policy
Adaptations
Dial Patterns
Entity Links
Locations
Regular Expressions
Routing Policies
SIP Domains
SIP Entities
Time Ranges
Personal Settings
Security
Applications
Settings
Session Manager
Shortcuts
Change Password
Help for SIP Entity Details fields
Help for Committing configuration changes
SIP Entity Details
Commit
Cancel
General
* Name: CM Feature Server
* FQDN or IP Address: 10.1.2.160
Type: CM
Notes:
Adaptation:
Location: BaskingRidge HQ
Time Zone: America/New_York
Override Port & Transport with DNS SRV:
* SIP Timer B/F (in seconds): 4
Credential name:
Call Detail Recording: none
SIP Link Monitoring
SIP Link Monitoring: Link Monitoring Enabled
* Proactive Monitoring Interval (in seconds): 30
* Reactive Monitoring Interval (in seconds): 30
* Number of Retries: 1
Entity Links
Add
Remove
1 Item | Refresh
Filter: Enable
SIP Entity 1
Protocol
Port
SIP Entity 2
Port
Trusted
SM1
TCP
* 5060
CM Feature Server
* 5060
Select : All, None (0 of 1 Selected)

5.4. Add Entity Link

The SIP trunk from Session Manager to the Communication Manager Feature Server is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *SM1_CM Feature Server*).
- **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 the Communication Manager Feature Server.
- **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 0*** *will be denied.*

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



5.5. Add Routing Policies

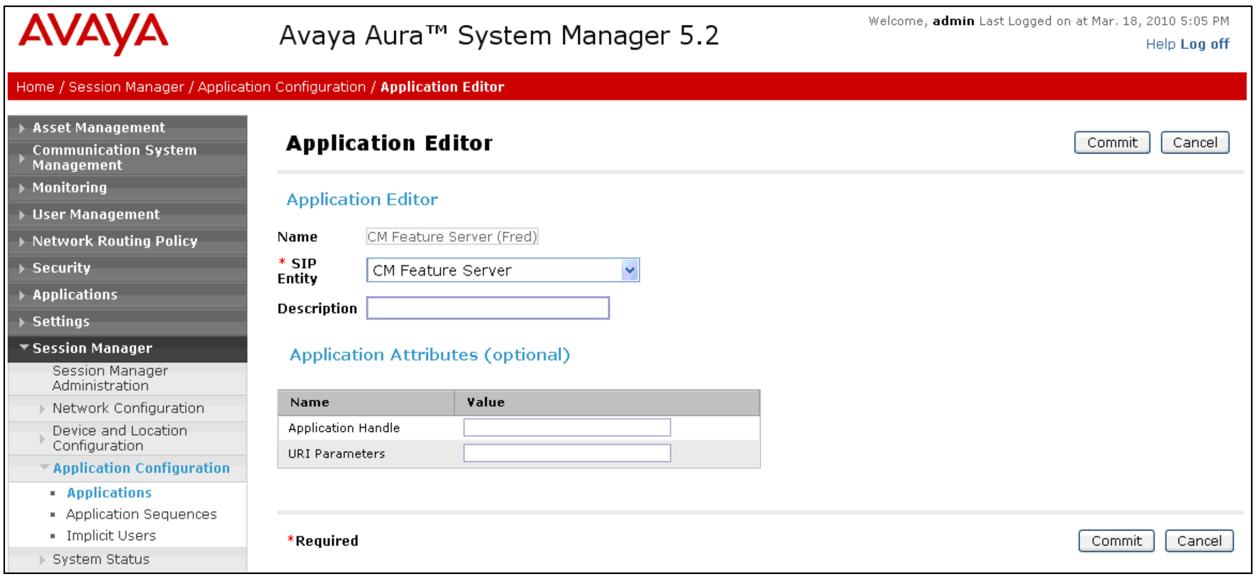
Since the SIP users are register with Session Manager, a routing policy is not required to be defined for the Communication Manager Feature Server.

5.6. Add Application Sequence

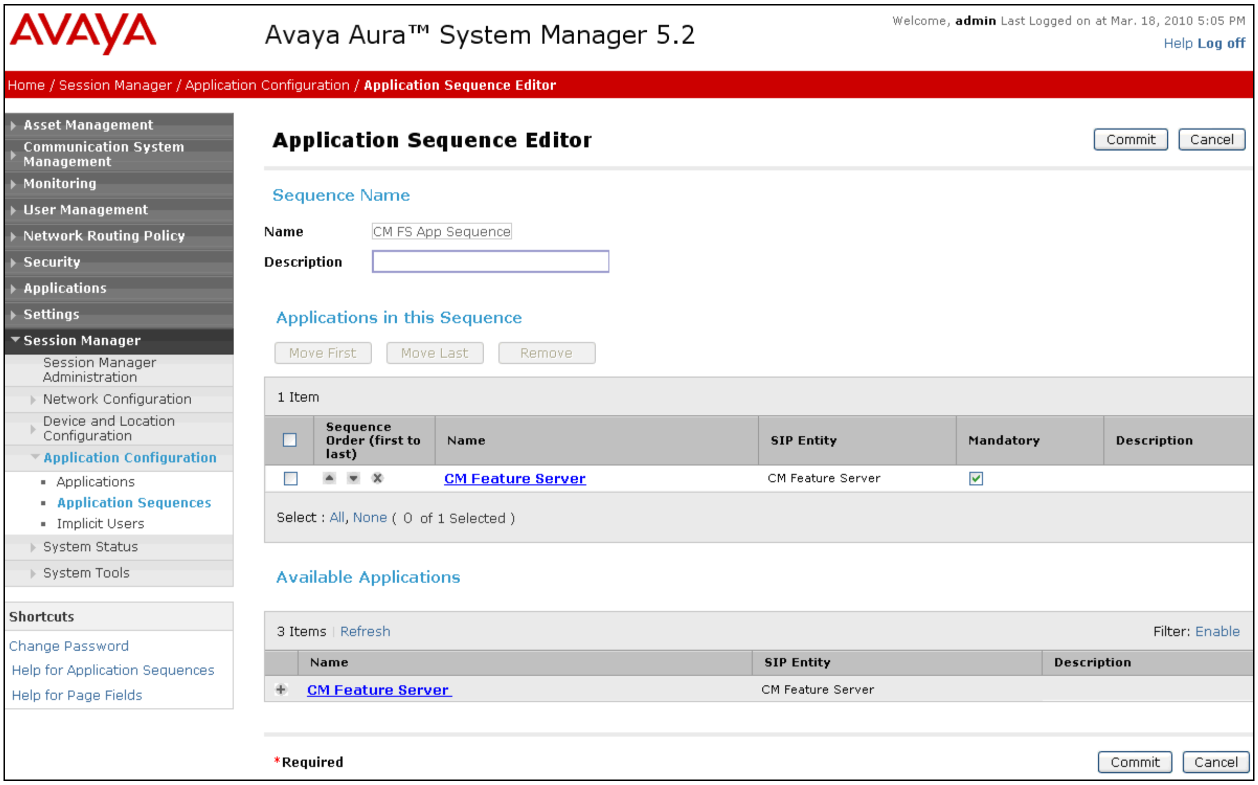
Define an application for the Communication Manager Feature Server. Fill in the following fields:

- **SIP Entity:** Select the Communication Manager Feature Server.

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



Next, define the Application Sequence for Communication Manager Feature Server as shown below.



5.7. Define Feature Server as Administrable Entity

Before adding SIP users, the Communication Manager Feature Server must be added to System Manager as an administrable entity. This action allows System Manager to access Communication Manager over its administration interface. Using this administration interface, System Manager will notify the Communication Manager Feature Server when new SIP users are added.

To define the Feature Server as an administrable entity, select **Entities** on the left and click on the **New** button on the right. Fill in the following fields as follows:

Under *Application*:

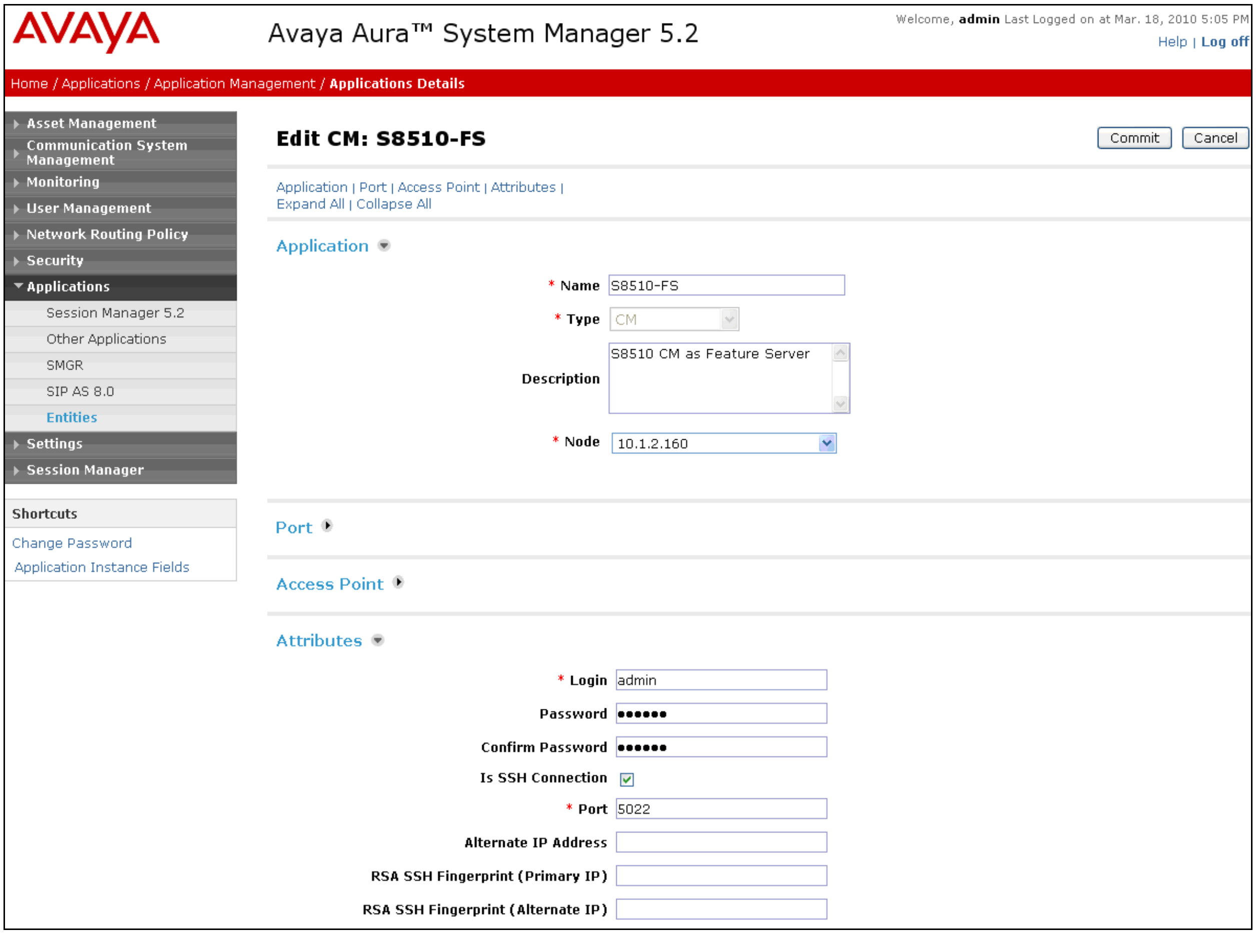
- **Name:** Enter an identifier for the Communication Manager Feature Server.
- **Type:** Select *CM* from the drop-down field.
- **Node:** Enter the IP address of the administration interface for the Feature Server.

Under *Attributes*:

- **Login / Password:** Enter the login and password used for administration access.

- **Is SSH Connection:** Enable SSH access.
- **Port:** Enter the port number for SSH administration access (5022).

Click **Commit** to save the settings.



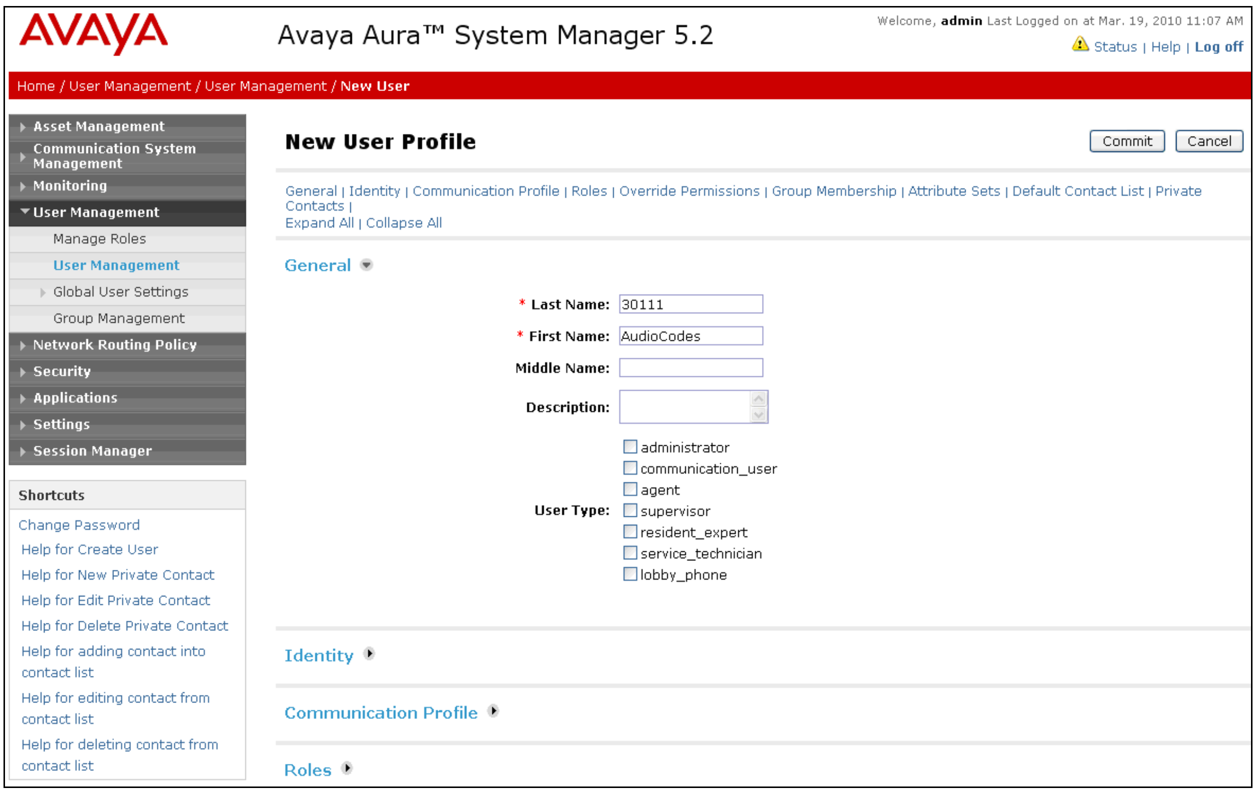
5.8. Add SIP Users

Add SIP users corresponding to the AudioCodes 320HD SIP Phone defined in **Section 4.3.** Alternatively, use the option to automatically generate the SIP stations on Communication Manager Feature Server when adding a new SIP user.

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

- **Last Name:** Enter the last name of the user.
- **First Name:** Enter the first name of the user.

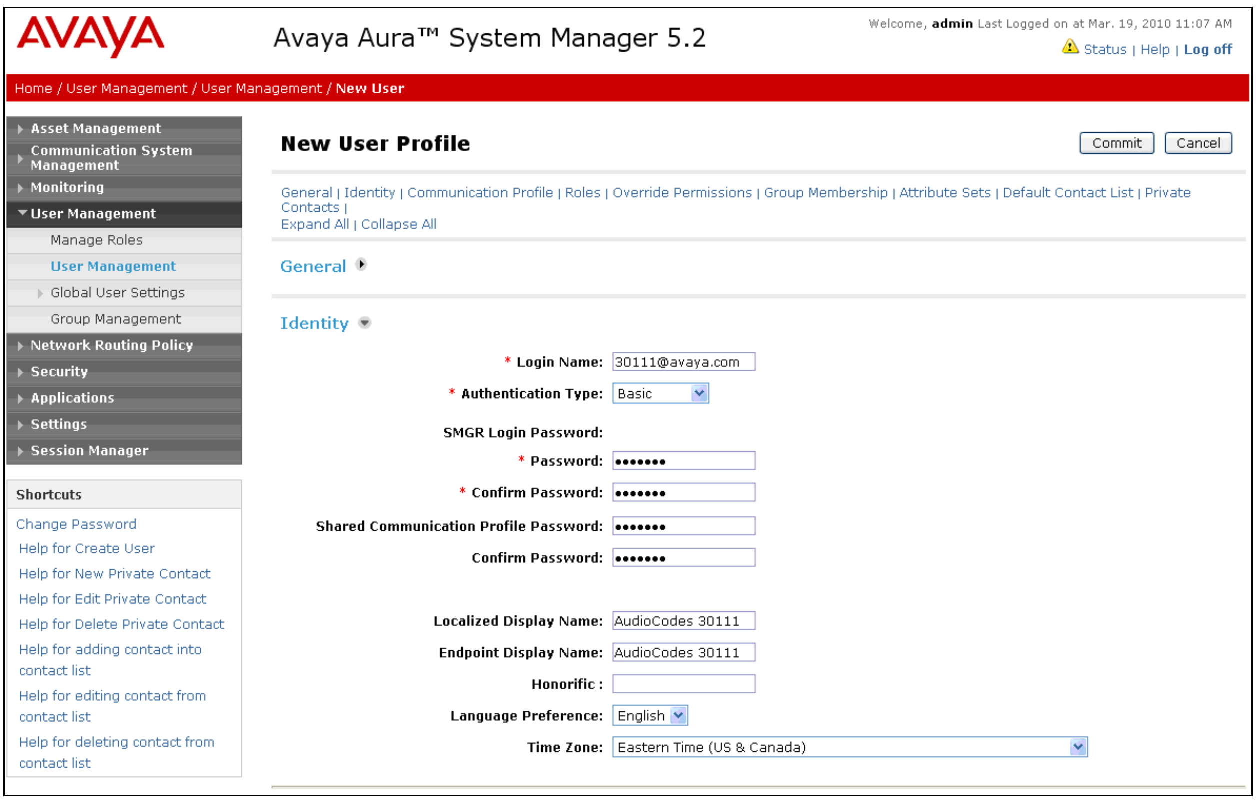
The screen below shows the information when adding a new SIP user to the sample configuration.



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

- **Login Name:** Enter *<extension>@<sip domain>* of the user (e.g., 30111@avaya.com).
- **Authentication Type:** Select *Basic*.
- **SMGR Login Password:** Enter the password which will be used to log into System Manager
- **Confirm Password:** Re-enter the password from above.
- **Shared Communication Profile Password:** Enter the password which will by the SIP phone to log into Session 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.



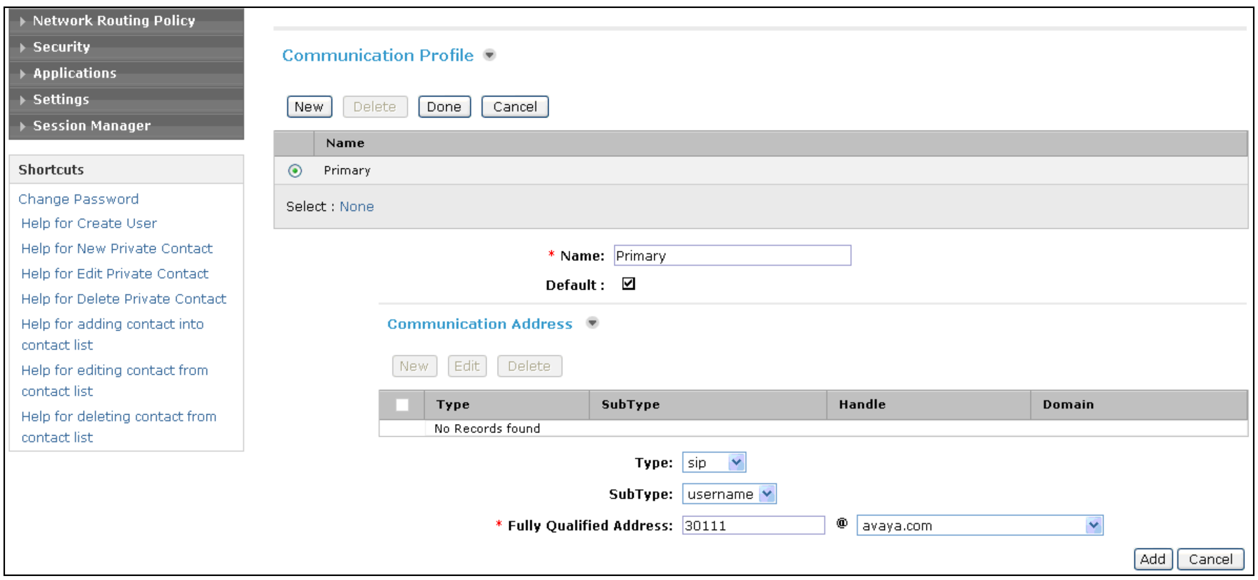
Scroll down to the *Communication Profile* section and select **New** to define a **Communication Profile** for the new SIP user. Enter values for the following required fields:

- **Name:** Enter name of communication profile.
- **Default:** Select field to indicate that this is the default profile.

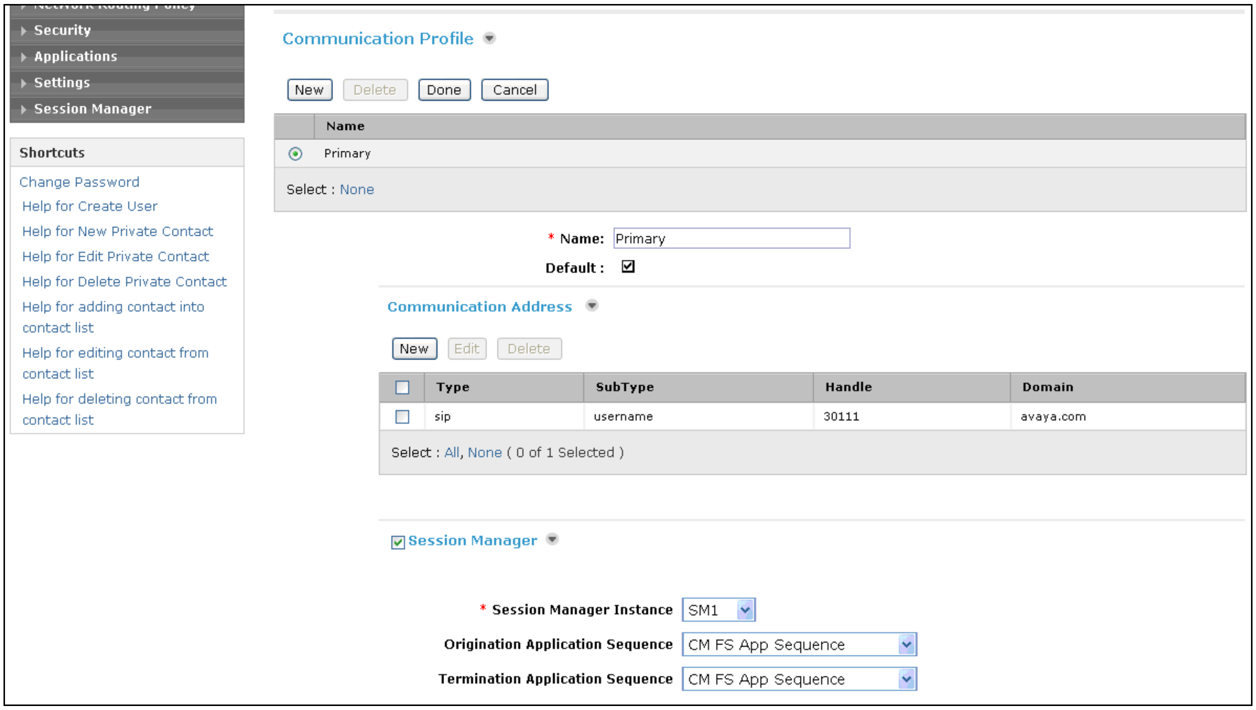
Click **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** Select *sip*.
- **SubType:** Select *username*.
- **Fully Qualified Address:** Enter extension number and SIP domain.

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



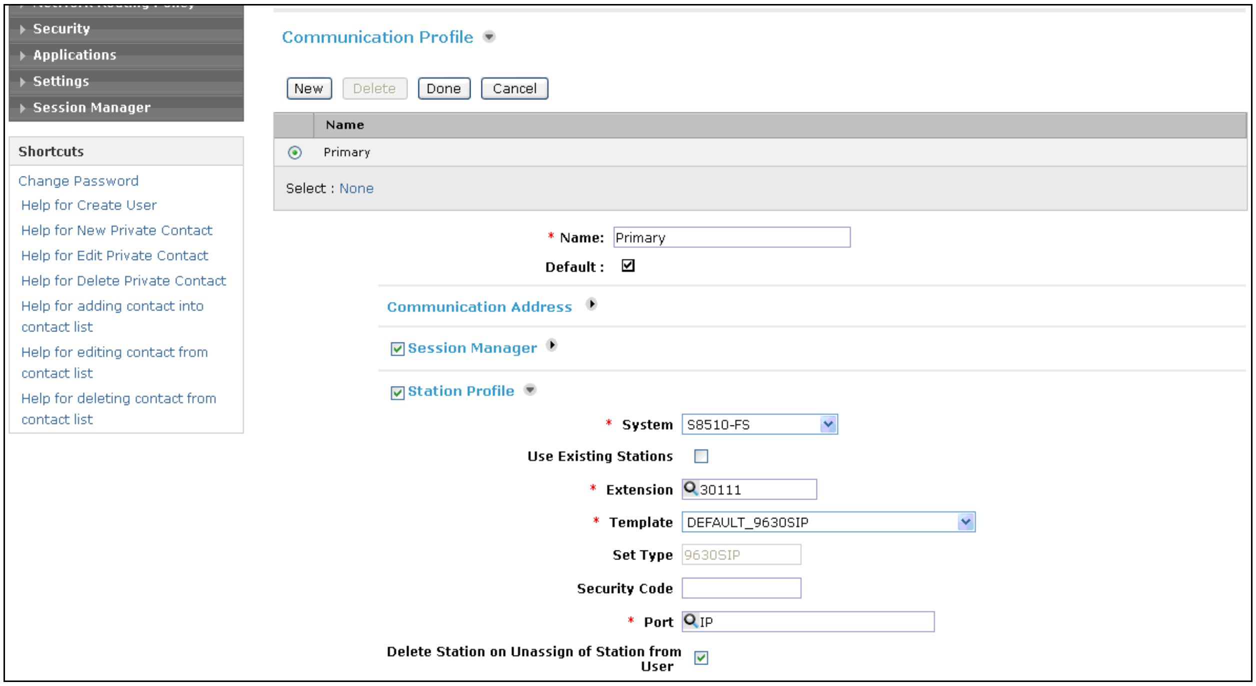
In the *Session Manager* section, specify the Session Manager entity and assign the **Application Sequence** defined in **Section 5.6** to the new SIP user as part of defining the **SIP Communication Profile**. The **Application Sequence** can be used for both the originating and terminating sequence.



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

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

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



5.9. Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between Avaya Aura™ System Manager and Avaya Aura™ Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add**, and fill in the fields as described below and shown in the following screen:

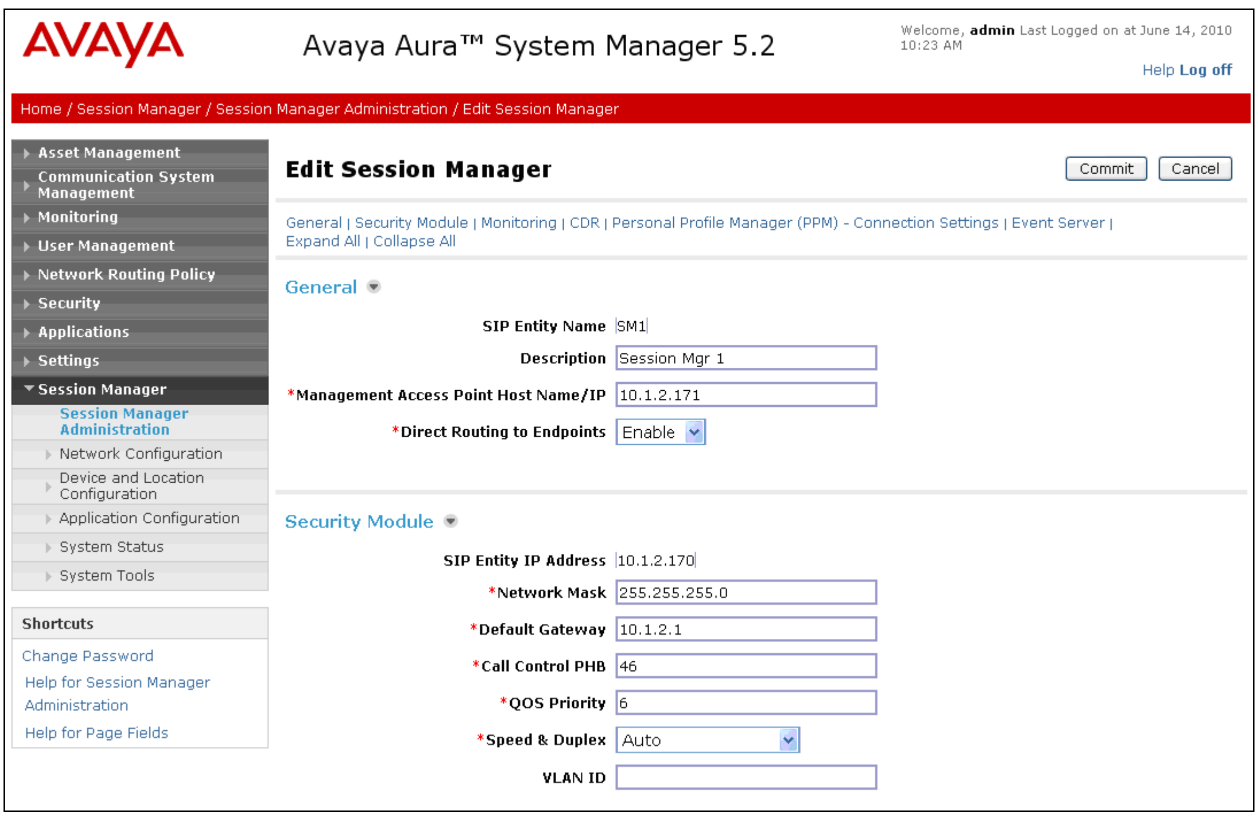
Under *Identity*:

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

Under *Security Module*:

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

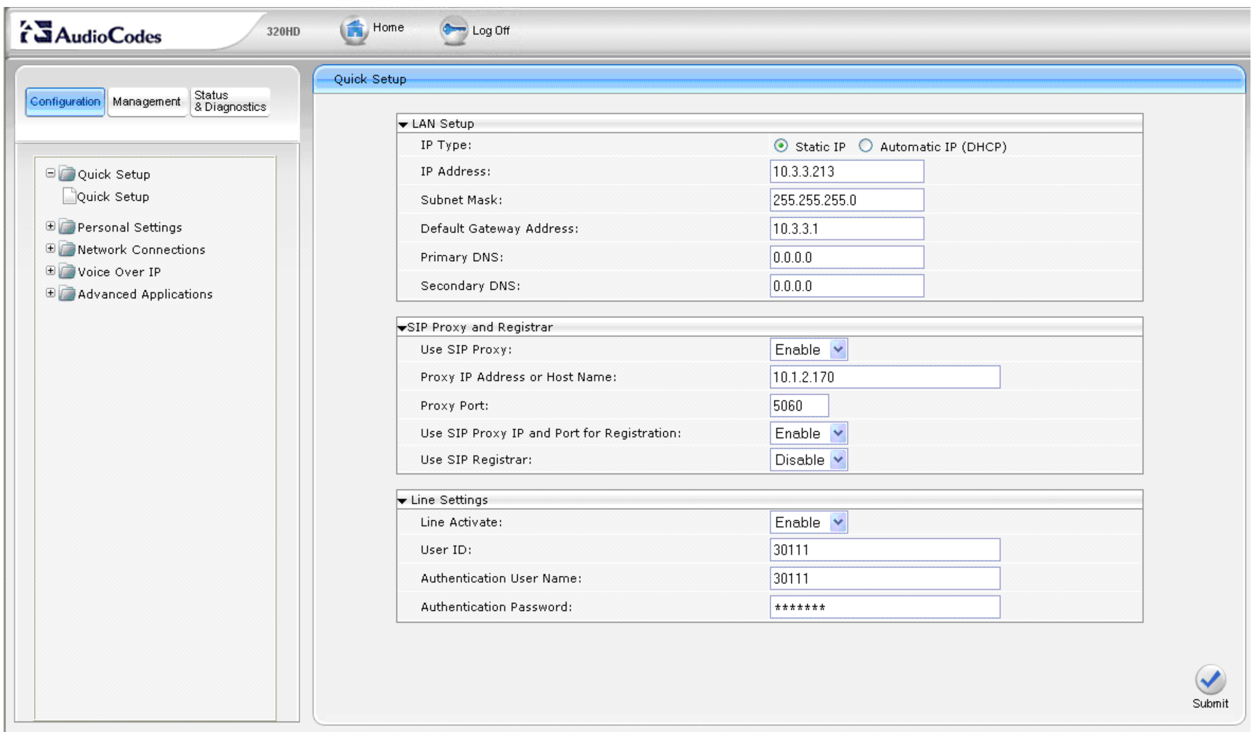
Use default values for the remaining fields. Click **Save** to add this Session Manager.



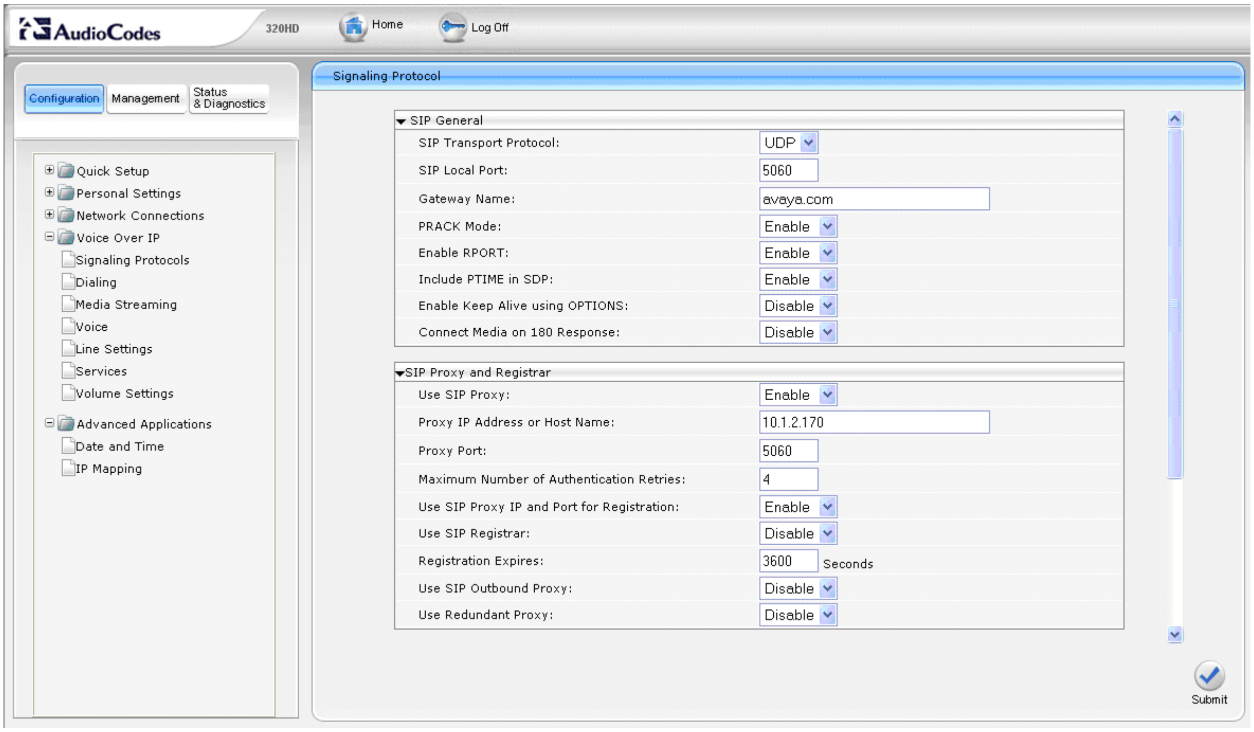
6. Configure AudioCodes 320HD SIP Phone

The configuration of the 320HD SIP Phone was performed via the phone's menu-driven LCD user interface and its embedded Web interface. The phone's LAN connection interface was initially configured via the phone's LCD screen. To access the **Network Settings** menu, click the MENU key on the phone and navigate to Administration→Network Settings to configure the IP parameters for the phone. A valid password will be required. The rest of the configuration was performed through the phone's embedded Web interface. Refer to [5] for additional information on configuring the 320HD SIP Phone.

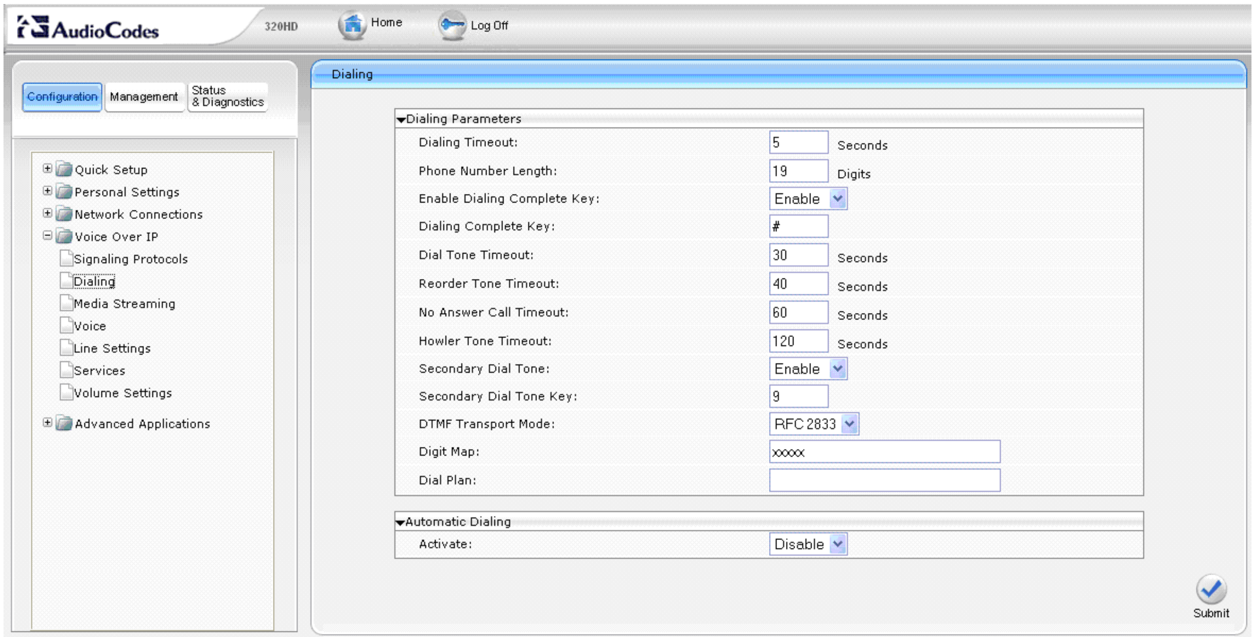
From an internet browser, enter http://<*ip-addr*> in the URL field, where <*ip-addr*> is the phone's IP address. Navigate to the **Quick Setup** screen shown below. The **LAN Setup** section was previously configured through phone's LCD screen. In the **SIP Proxy and Registrar** section, set the **Use SIP Proxy** field to *Enable* and set the **Proxy IP Address or Host Name** field to the IP address of Session Manager. Configure the **Proxy Port** field with the appropriate port value (i.e., 5060). The **Line Settings** section should be configured with the phone's extension and password that were configured for the user on Session Manager. This section provides the authentication information required to register with Session Manager. Click **Submit**.



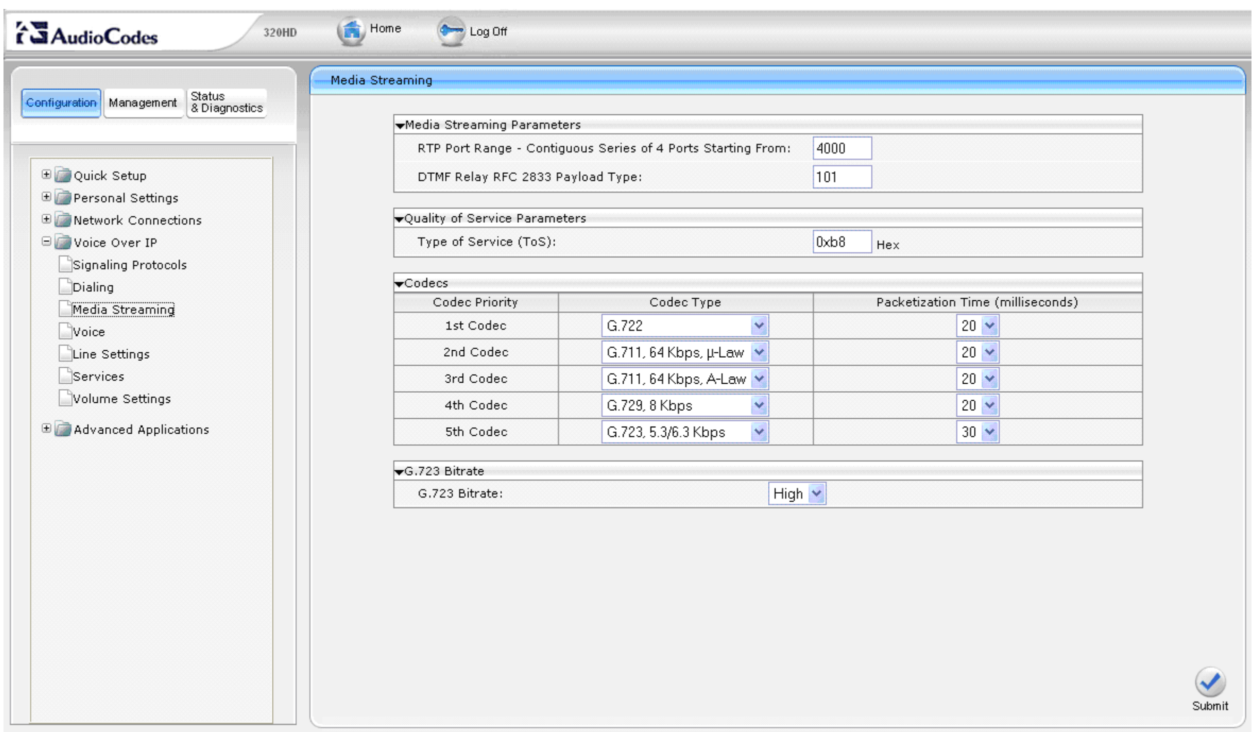
Navigate to **Voice Over IP→Signaling Protocols** and set the **Gateway Name** field to the SIP domain (e.g., *avaya.com*). All of the other fields should be properly set based on the **Quick Setup** configuration above. Click **Submit**.



Navigate to **Voice Over IP→Dialing** and set the **Digit Map** field to *xxxxx* to allow 5-digit extensions to be dialed without waiting for an inter-digit timeout to expire. This corresponds to the dial plan used during the compliance test. Click **Submit**.



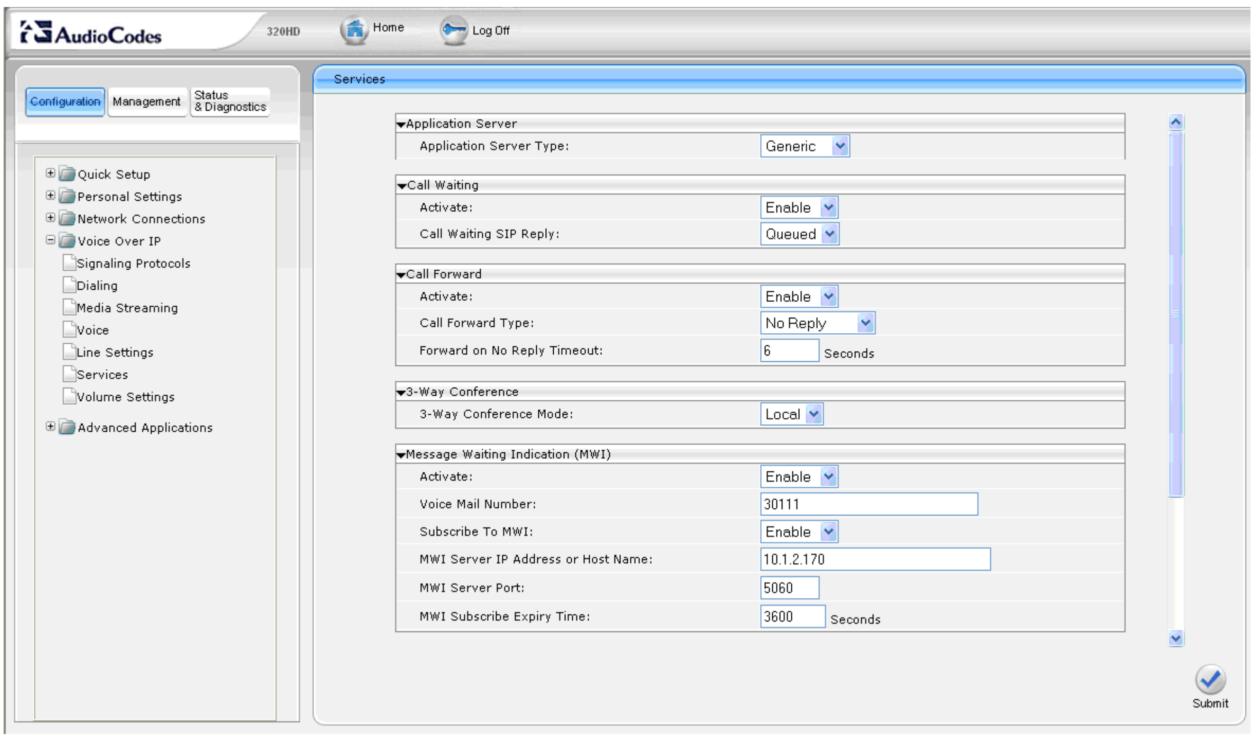
The following screen simply shows the codecs supported by the endpoint. No additional configuration is required here.



Navigate to **Voice Over IP→Services** to enable MWI. Under **Message Waiting Indication (MWI)**, set the **Activate** field to *Enable* and set the **Voice Mail Number** field to the phone's extension.

Note: The **Voice Mail Number** field is usually set to the voicemail pilot number so that when the Voicemail button on the phone is pressed, the voicemail system is called for the user to check or send voicemail messages. However, for MWI to work, this field should be configured as specified above. The voicemail system can still be dialed manually by the user.

Next, set the **Subscribe To MWI** field to *Enable*, set the **MWI Server IP Address or Host Name** field to the IP address of Session Manager, and increase the **MWI Subscribe Expiry Time** field to a minimum of 3600. Click **Submit**.



7. General Test Approach and Test Results

To verify interoperability of the AudioCodes 300HD SIP IP Phone Series with Communication Manager Feature Server and Session Manager, calls were made between 320HD SIP Phones and Avaya SIP, H.323, and digital stations using various codec settings and exercising common PBX features. The telephony features were activated and deactivated using phone buttons and FNEs. The PBX features listed in **Section 1.1** were covered.

The following observations were noted during the compliance test:

- Shuffling should be disabled for calls to the 300HD SIP IP Phone Series to prevent issues with Hold and Conference.

- The Voice Mail Number on the 300HD SIP IP Phone Series should be set to the phone's extension for MWI to work. As a result, the user will have to manually dial the voicemail pilot number instead of simply pressing the Voicemail button on the phone.

- When a 320HD SIP Phone calls another phone with Call Forwarding enabled, there is no indication on the 320HD SIP Phone that the call was forwarded.

8. Verification Steps

The following steps can be used to verify and/or troubleshoot installations in the field.

1. Verify that the 320HD SIP Phones have successfully registered with Session Manager.
2. Verify basic telephony features by establishing calls between a 320HD SIP Phone and another phone.
3. Call a 320HDSIP phone that currently has no voice messages, and leave a message. Verify that the message waiting indicator (i.e., Voicemail button) illuminates. Call the voicemail system and retrieve voice messages. Verify that after hearing all messages, that the message waiting indicator is extinguished.

9. Conclusion

These Application Notes have described the administration steps required to integrate the AudioCodes 300HD SIP IP Phone Series with Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager. The 320HD SIP Phone successfully registered with Session Manager and basic telephony features were verified. Noted observations during the compliance test were also covered.

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^{TM} Communication Manager*, Release 5.2, May 2009, Issue 5.0, Document Number 03-300509.

[2] *SIP Support in Avaya Aura^{TM} Communication Manager Running on the Avaya S8xxx Servers*, May 2009, Issue 9, Document Number 555-245-206.

[3] *Administering Avaya Aura^{TM} Communication Manager as a Feature Server*, May 2010, Issue 1.3, Release 5.2, Document Number 03-603479.

[4] *Administering Avaya Aura^{TM} Session Manager*, November 2009, Issue 2, Release 5.2, Document Number 03-603324.

[5] *AudioCodes Administrator Manual 320HD IP Phone*, Version 1.2.2, April 2010, Document Number LTRT-08105.

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.