Avaya Solution & Interoperability Test Lab

Configuring SIP Trunks between Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager 5.2.1, and Avaya IP Office Release 5.0 – Issue 1.0

Abstract

These Application Notes present a sample configuration for a network that uses Avaya Aura™ Session Manager to connect Avaya Aura™ Communication Manager 5.2.1 and Avaya IP Office using SIP trunks. Session Initiated Protocol (SIP) is a standard based communication protocol capable of supporting voice, video, instant messaging and other multi-media communication. These Application Notes will outline a solution for using SIP as a trunk protocol between Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager 5.2.1 and Avaya IP Office.

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1 Introduction

These Application Notes present a sample configuration for a network that uses Avaya Aura™ Session Manager to connect Avaya Aura™ Communication Manager 5.2.1 and Avaya IP Office using SIP trunks. Session Initiated Protocol (SIP) is a standard based communication protocol capable of supporting voice, video, instant messaging and other multi-media communication. These Application Notes will outline a solution for using SIP as a trunk protocol between Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager 5.2.1 and Avaya IP Office.

As shown in **Figure 1**, the Avaya 96xx IP Telephone (H.323) and 2420 Digital Telephone are supported by Communication Manager which serves as an Access Element within the Avaya Aura™ Session Manager architecture. The Avaya 5610 and 1608 IP Telephones (H.323) and 54xx Digital Telephones are supported by Avaya IP Office 500. SIP trunks are used to connect these two systems to Avaya Aura™ Session Manager, using its SM-100 (Security Module) network interface. All inter-system calls are carried over these SIP trunks. Avaya Aura™ Session Manager can support flexible inter-system call routing based on dialed number, calling number and system location, and can also provide protocol adaptation to allow multi-vendor systems to interoperate. It is managed by a separate Avaya Aura™ System Manager, which can manage multiple Avaya Aura™ Session Managers by communicating with their management network interfaces. Avaya 9620 IP Telephones configured as SIP users utilizes the Avaya Aura™ Session Manager User Registration feature and require Communication Manager Feature Server. Communication Manager as a feature server only supports IMS-SIP users that are registered to Avaya Aura™ Session Manager. The Communication Manager Feature Server is connected to Session Manager via an IMS-enabled SIP signaling group and associated SIP trunk group.

For the sample configuration, Avaya Aura™ Session Manager runs on an Avaya S8510 Server, and Avaya Aura™ Communication Manager 5.2.1 runs on an Avaya S8730 Server with Avaya G650 Media Gateway. The results in these Application Notes should be applicable to other Avaya servers and media gateways that support Avaya Aura™ Communication Manager 5.2.1 and Avaya IP Office on the 500 platform.

These Application Notes will focus on the configuration of the SIP trunks and call routing. Detailed administration of Session Manager, Communication Manager Feature Server, Communication Manager Access Element and the endpoint telephones will not be described (see the appropriate documentation listed in **Section 9**).

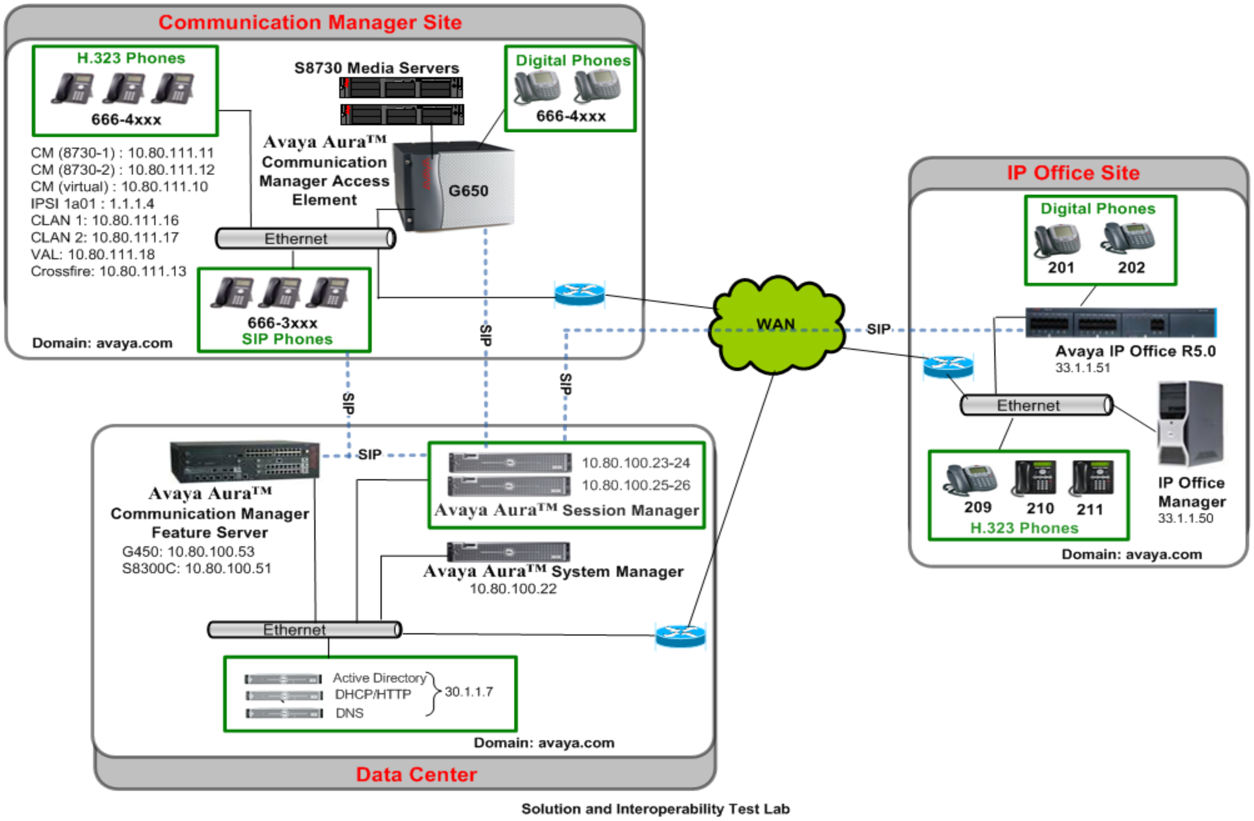


Figure 1 – Sample Configuration

2 Equipment and Software Validated

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

Hardware Component	Software Version
Avaya S8510 Server	Avaya Aura™ Session Manager Release 5.2 (Build 520011)
	Avaya Aura™ System Manager, Release 5.2 (5.2.7.0)
Avaya S8730 Servers with G650 Media Gateway	Avaya Aura™ Communication Manager Release 5.2 (R015x.02.1.016.4)
Avaya S8300C Server with G450 Media Gateway	Avaya Aura™ Communication Manager Release 5.2 (R015x.02.1.016.4)
Avaya 9630 IP Telephone (H.323)	2.0
Avaya 9630 IP Telephone (SIP)	2.5.5.17
Avaya 2420 Digital Telephone	NA
Avaya IP Office Server	Release 5.0 (8)
Avaya 5410 & Avaya 5420 Digital Telephones	NA
Avaya 1608 IP Telephone (H.323)	ha1608ual_2110.bin
Avaya 5610 IP Telephone (H.323)	2.9

3 Configure Avaya IP Office

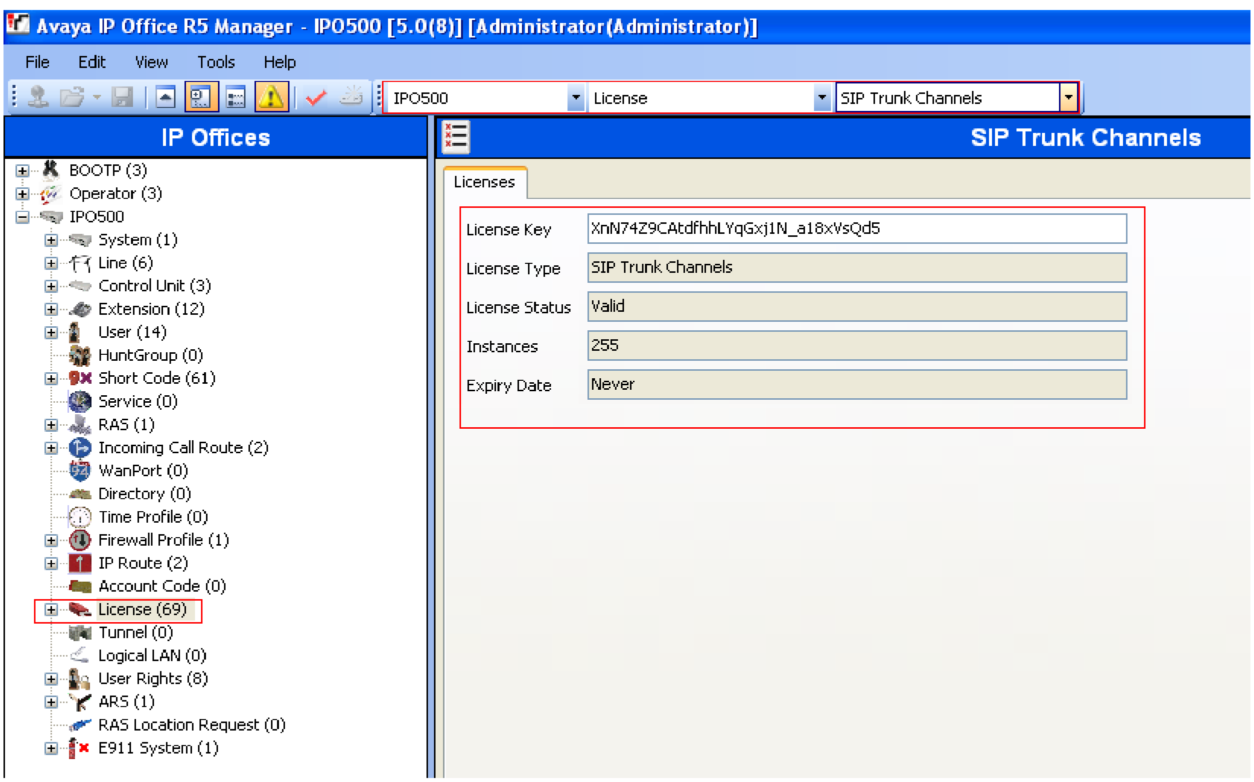
This section provides the procedures for configuring Avaya IP Office. The procedures include the following areas:

- Verify IP Office license
- Obtain LAN IP address
- Configure Network Topology
- Administer SIP Registrar
- Administer Codec Preference
- Administer SIP Trunk
- Administer Short Code
- Configure Incoming Call Route
- Configure Users SIP Names

3.1 Verify IP Office License

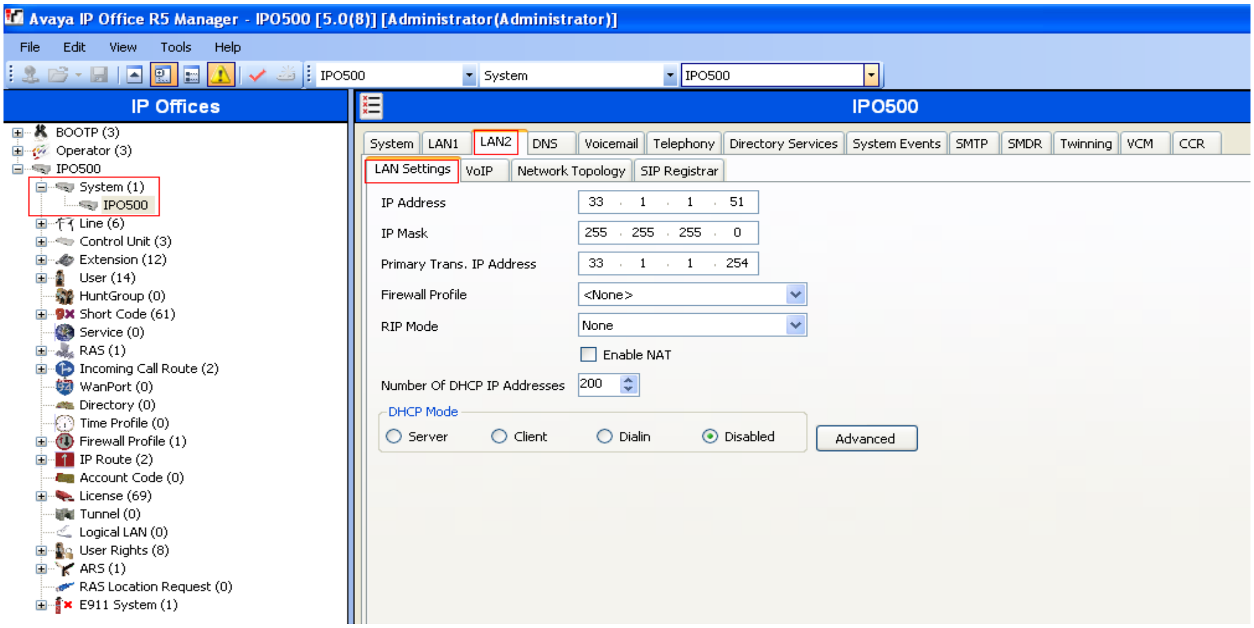
From a PC running the Avaya IP Office Manager application, select **Start > Programs > IP Office > Manager** to launch the Manager application. Select the proper IP Office system, and log in with the appropriate credentials.

The **Avaya IP Office Manager** screen is displayed. From the configuration tree in the left pane, select **License > SIP Trunk Channels** to display the **SIP Trunk Channels** screen in the right pane. Verify that the **License Status** is "Valid".



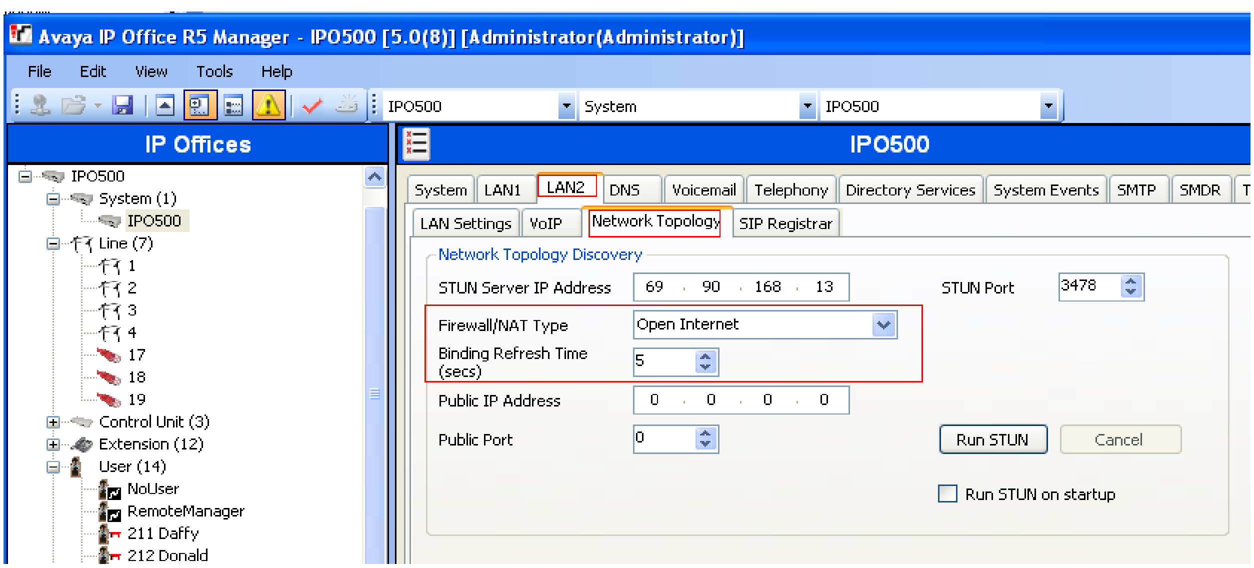
3.2 Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the **IPO500** screen in the right pane. Select the **LAN2** tab, followed by the **LAN Settings** sub-tab in the right pane. Make a note of the **IP Address**, which will be used later to configure SIP trunks. Note that IP Office can support SIP trunks on the LAN1 and/or LAN2 interfaces, and the sample configuration used the LAN2 interface.



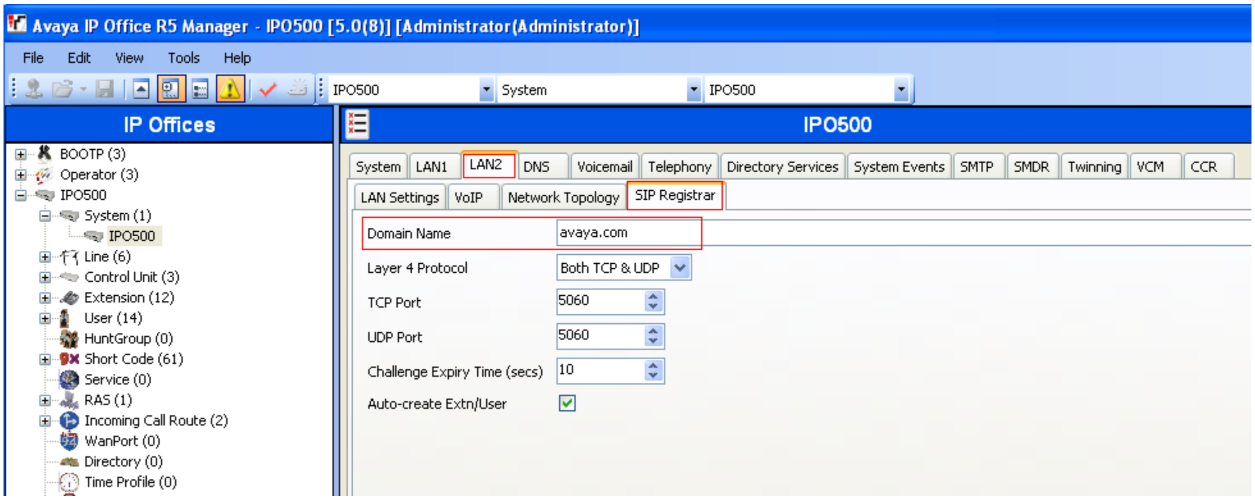
3.3 Configure Network Topology

From the configuration tree in the left pane, select **System** to display the **IPO500** screen in the right pane. Select the **LAN2** tab, followed by the **Network Topology** sub-tab in the right pane. Configure **Firewall/NAT Type** to "Open Internet". Configure **Binding Refresh Time** to "5". Click **OK**.



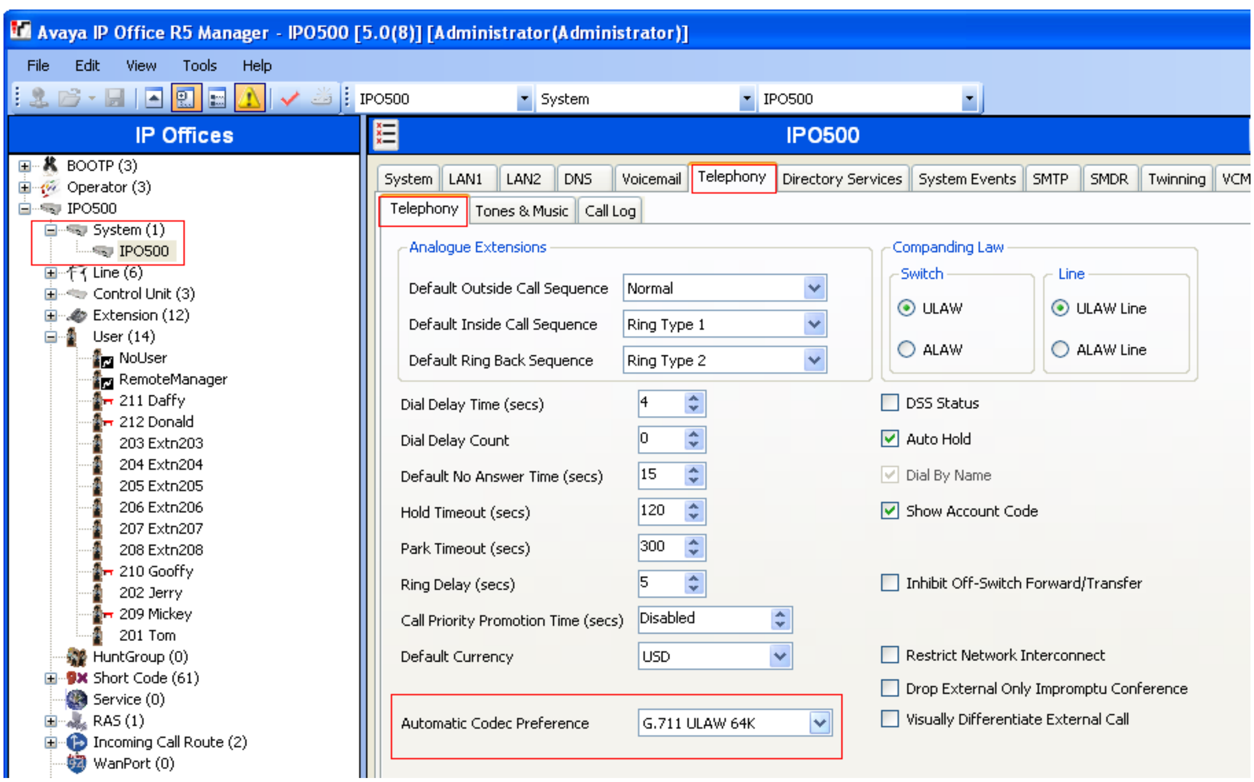
3.4 Administer SIP Registrar

Select **SIP Registrar** sub-tab in the right pane. Enter a valid **Domain Name**. Make a note of the **Layer 4 Protocol** and **TCP Port** and **UDP Port** numbers. These will be used later to configure SIP trunks. Click **OK**.



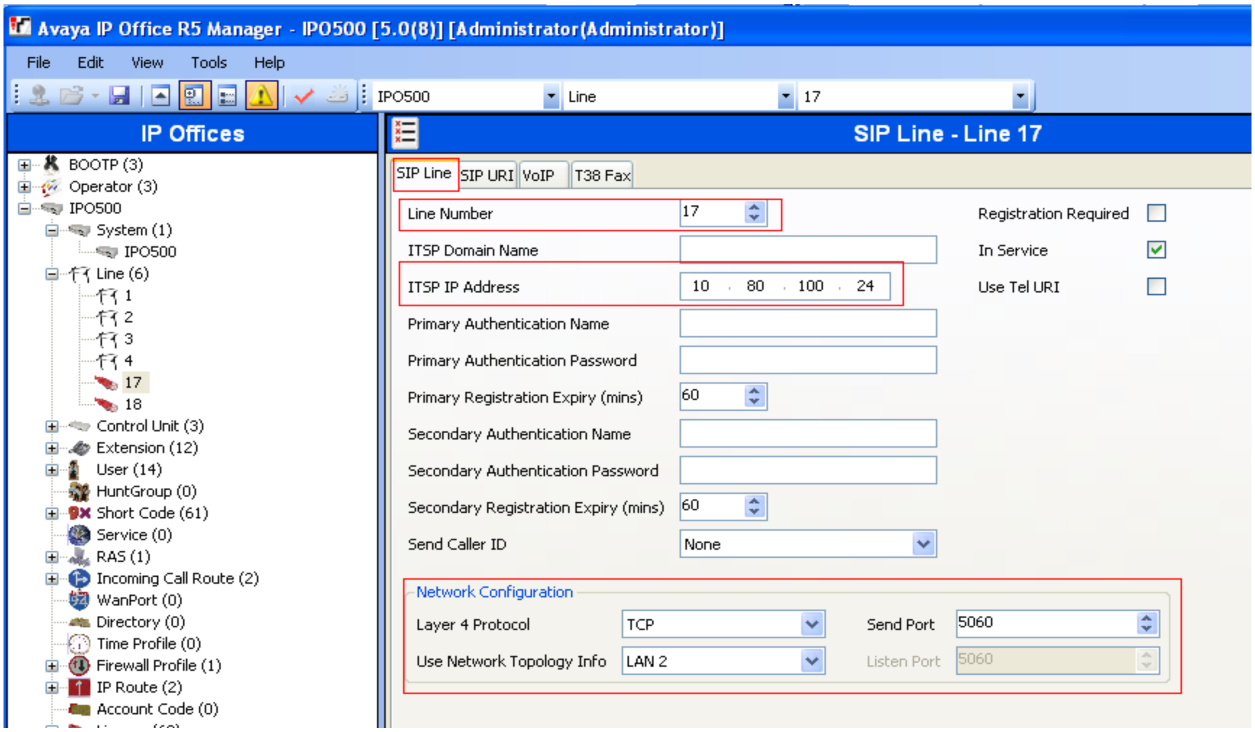
3.5 Administer Codec Preference

From the configuration tree in the left pane, select **System** to display the **IPO500** screen in the right pane. Select the **Telephony** tab. Configure **Automatic Codec Preference** to "G.711 ULAW 64K". Click **OK**.

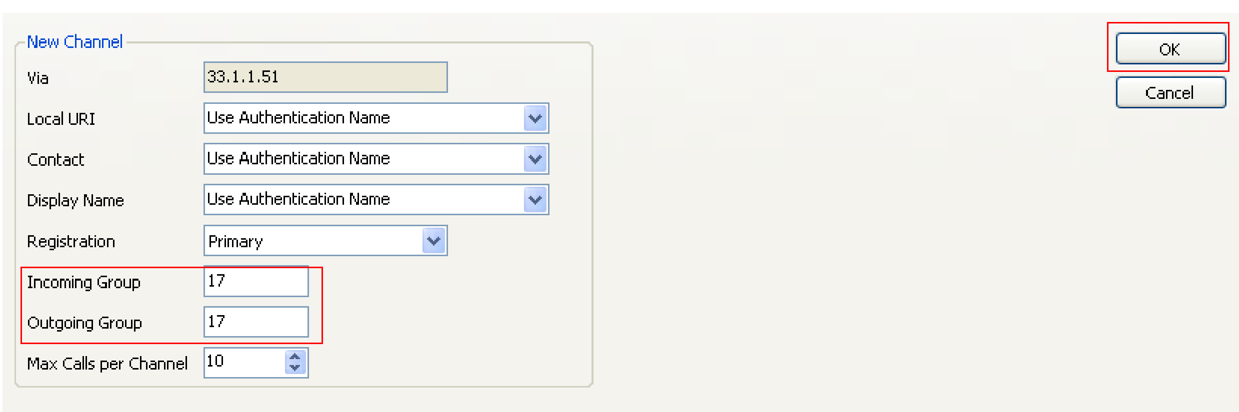


3.6 Administer SIP Trunk

From the configuration tree in the left pane, right-click on **Line** and select **New** > **SIP Line** to add a new SIP Trunk. Enter the “IP address for Session Manager” in **ITSP IP Address** field. Make a note of the **Line Number**. Select **Layer 4 Protocol** as “TCP” and **Send Port** “5060”. Select “LAN2” in the **Use Network Topology Info**. Retain default values for all other fields. Click **OK**.

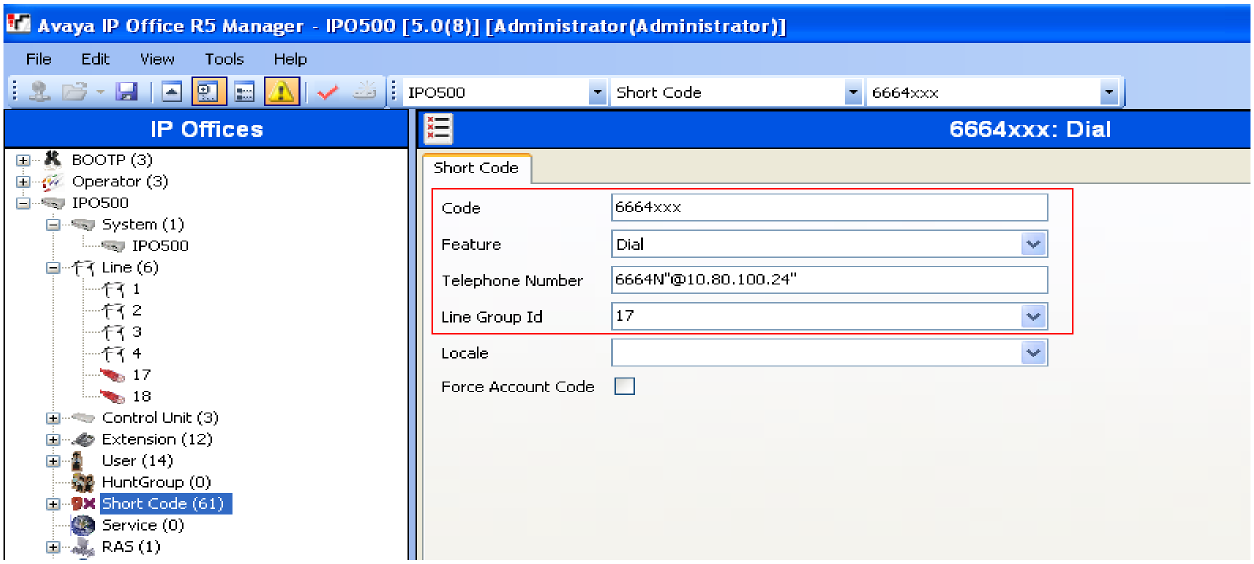


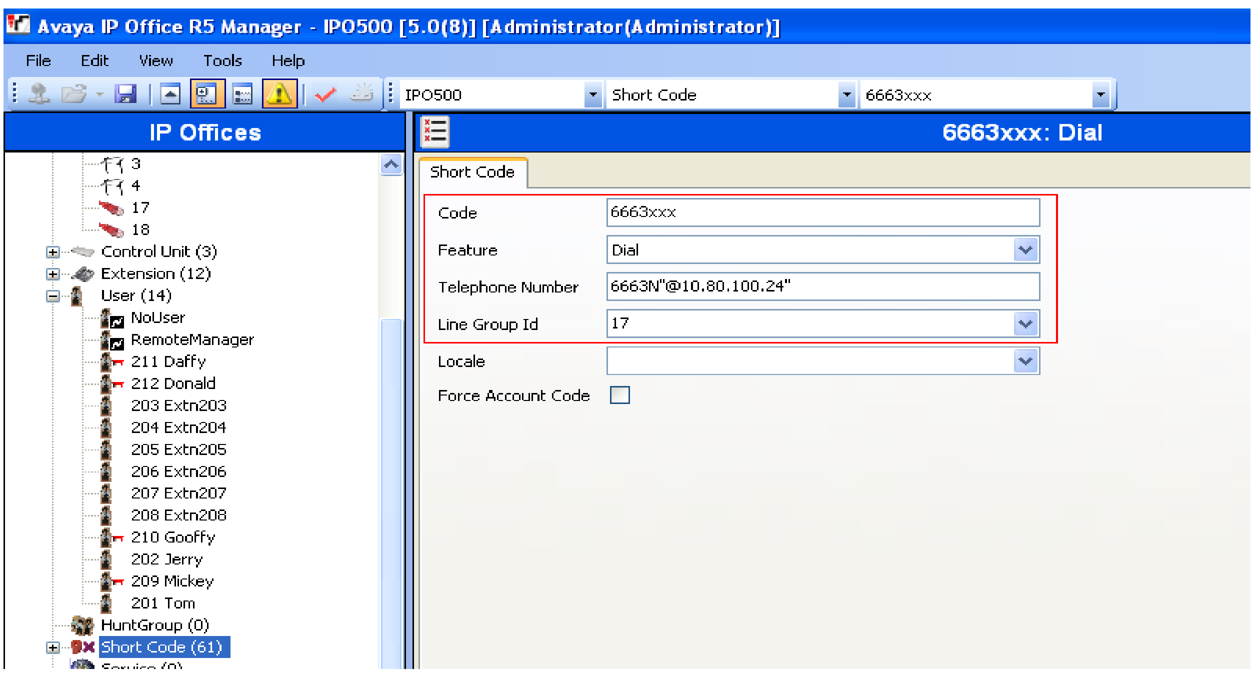
Select the **SIP URI** tab, and click on **Add…** radio button. In the **Incoming Gr**oup and **Outgoing Group** enter the “Line Number” from the above step. Retain default values for all other fields. Click **OK**.



3.7 Administer Short Code

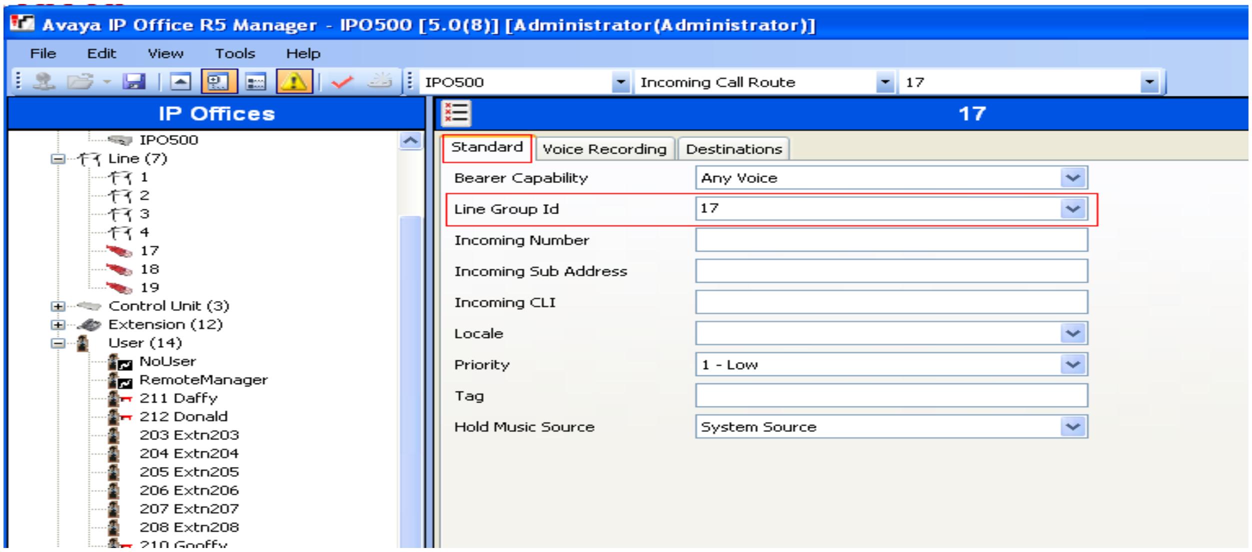
From the configuration tree in the left pane, right-click on **Short Code**, and select **New**. Enter the dialing string that will be used to call the users on Communication Manager in the **Code** field. Select "Dial" from the drop down menu for **Feature** and enter the phone number appended with "@<ip-address of Session Manager>" in the **Telephone Number**. Select SIP trunk administered in **Section 3.6** in the **Line Group Id**. Shown below are two short code which were added for the sample configuration.



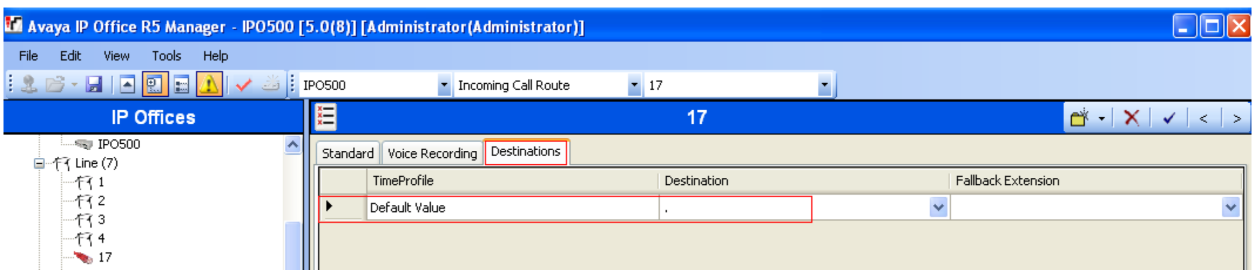


3.8 Configure Incoming Call Route

From the configuration tree in the left pane, right-click on **Incoming Call Route**, and select **New**. Under the **Standard** tab, enter the SIP trunk administered in **Section 3.6** in the **Line Group Id**.

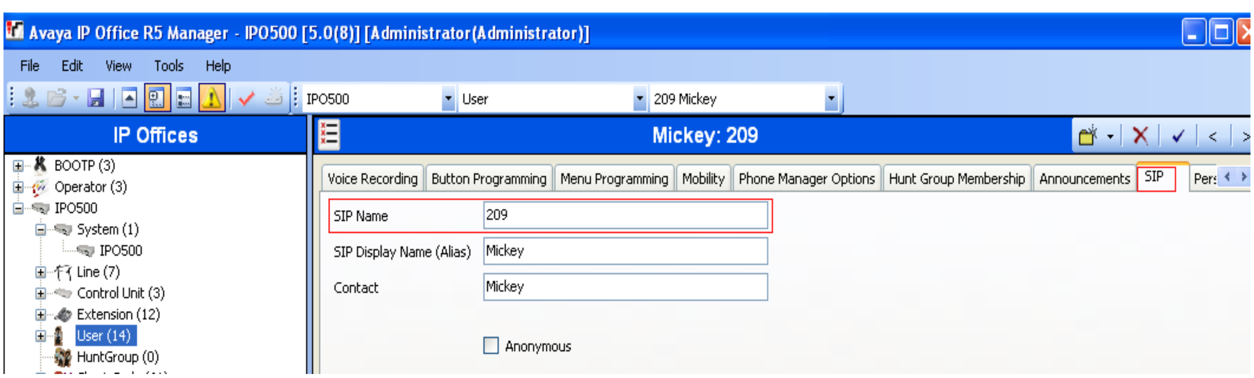


Under the **Destination** tab, enter "." as the Default Value. This will enable all incoming calls to be routed to any extension.



3.9 Configure SIP User Names

From the configuration tree in the left pane, right-click on **User** and select **SIP** tab. Modify the **SIP Name** to be the same as the user's extension number. The other fields can be left as default. Repeat this for all users.



3.10 Save Configuration

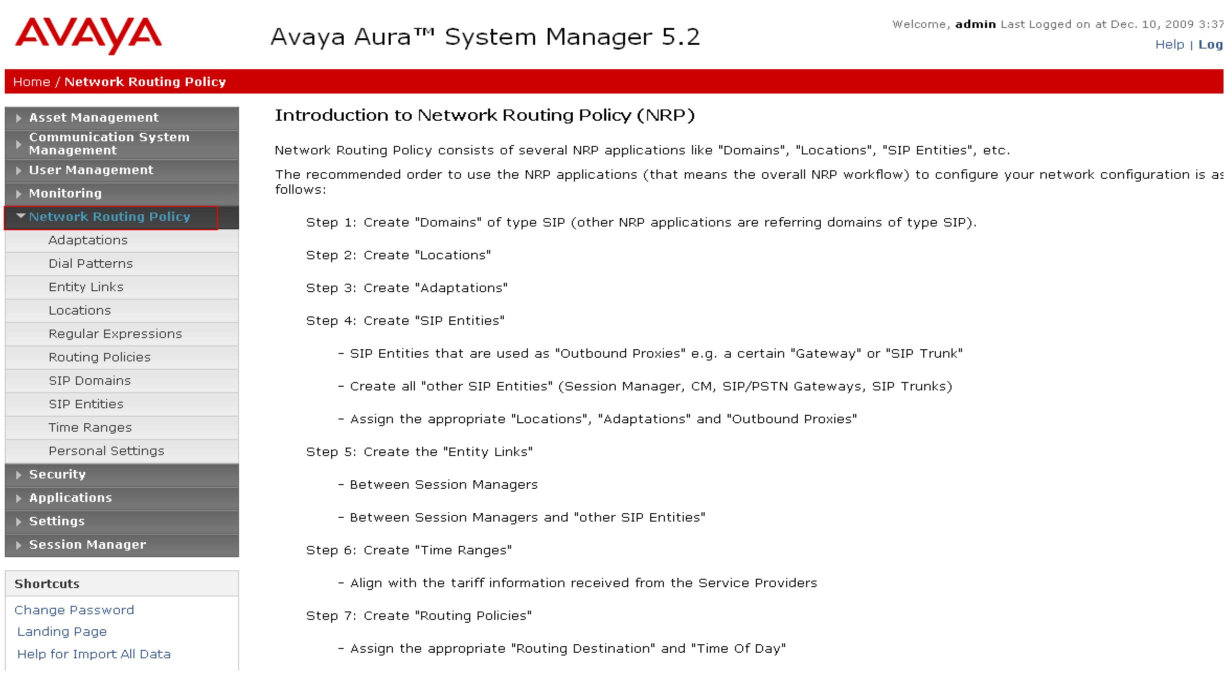
Select **File > Save Configuration** to save and send the configuration to the IP Office server.

4 Configure Avaya Aura™ Session Manager

This section provides the procedures for configuring Avaya Aura™ 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 the SIP telephony systems and Avaya Aura™ Session Manager
- Entity Links, which define the SIP trunk parameters used by Avaya Aura ™ Session Manager when routing calls to/from SIP Entities
- Time Ranges during which routing policies are active
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Session Manager, corresponding to the Session Manager Server to be managed by Avaya Aura™ System Manager.

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura™ System Manager, using the URL "http://<ip-address>/IMSM", where "<ip-address>" is the IP address of Avaya Aura™ System Manager. Log in with the appropriate credentials and accept the Copyright Notice. The menu shown below is displayed. Expand the **Network Routing Policy** Link on the left side as shown. The sub-menus displayed in the left column below will be used to configure all but the last of the above items (**Sections 4.1** through **4.7**).



4.1 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**.

4.2 Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management. For the sample configuration, Locations are added for the Communication Manager Feature Server, Communication Manager Access Element and IP Office.

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:

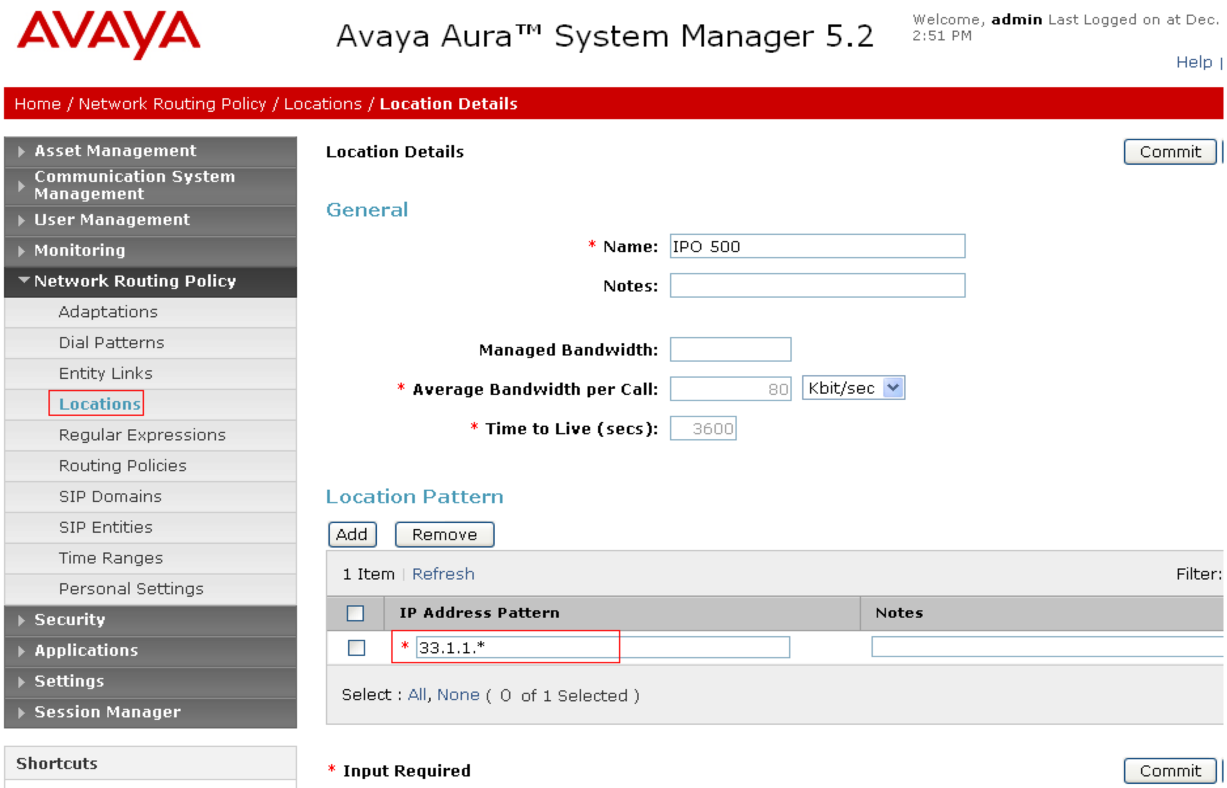
Under *General*:

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

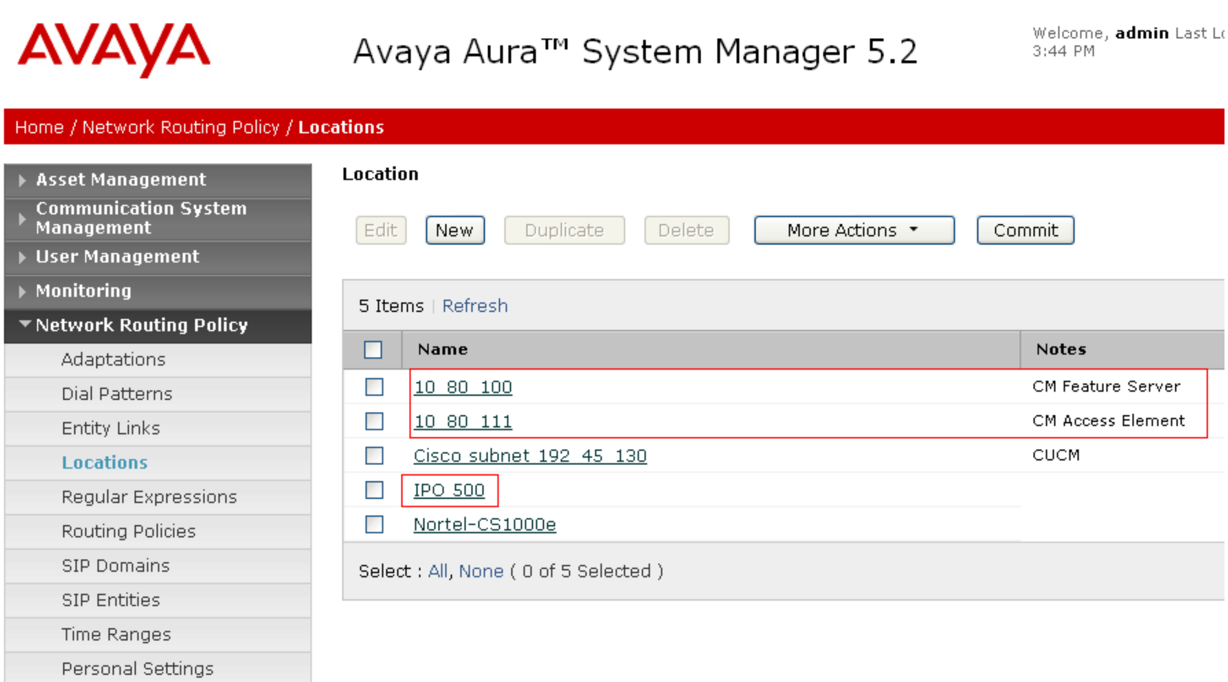
Under *Location Pattern*:

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

The screen below shows the information for IP Office. Click **Commit** to save.



The following screen shows the updated Locations after all the three locations are added.



4.3 Add SIP Entities

A SIP Entity must be added for Avaya Aura™ Session Manager and for each SIP-based telephony system supported by it using SIP trunks. In the sample configuration a SIP Entity is added for the ASM, the C-LAN board in the Avaya G650 Media Gateway, and Avaya IP Office. 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:** "Session Manager" for Avaya Aura™ Session Manager, "CM" for Communication Manager Access Element, "CM" for Communication Manager Feature Server, and "SIP Trunk" for Avaya IP Office.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

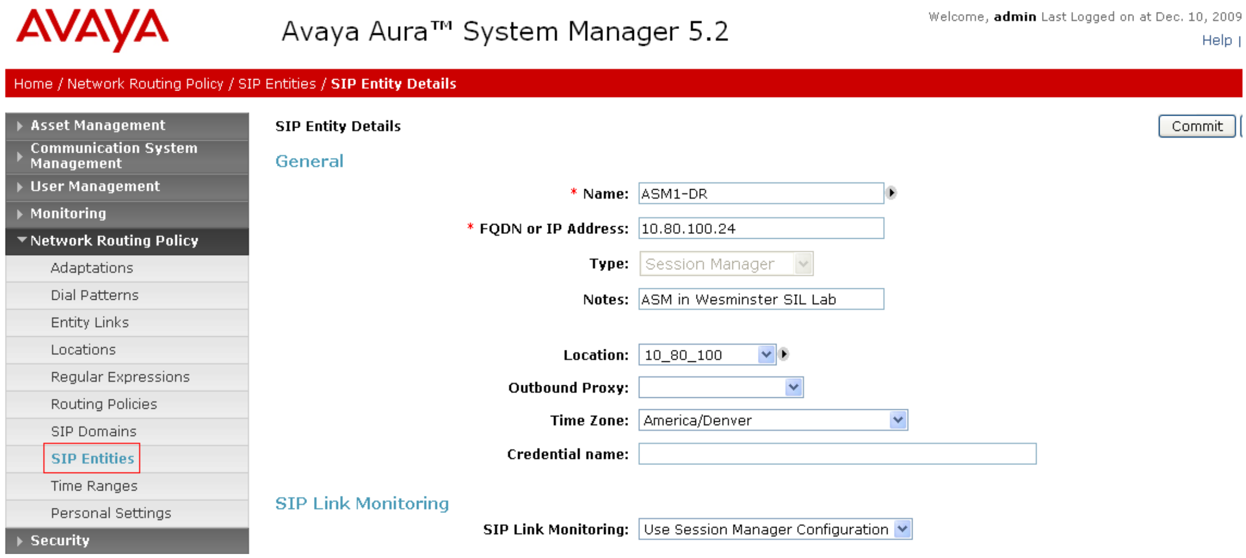
Under *SIP Link Monitoring*:

- **SIP Link Monitoring:** Select "Use Session Manager Configuration" for Communication Manager Access Element, Session Manager and Avaya IP Office.

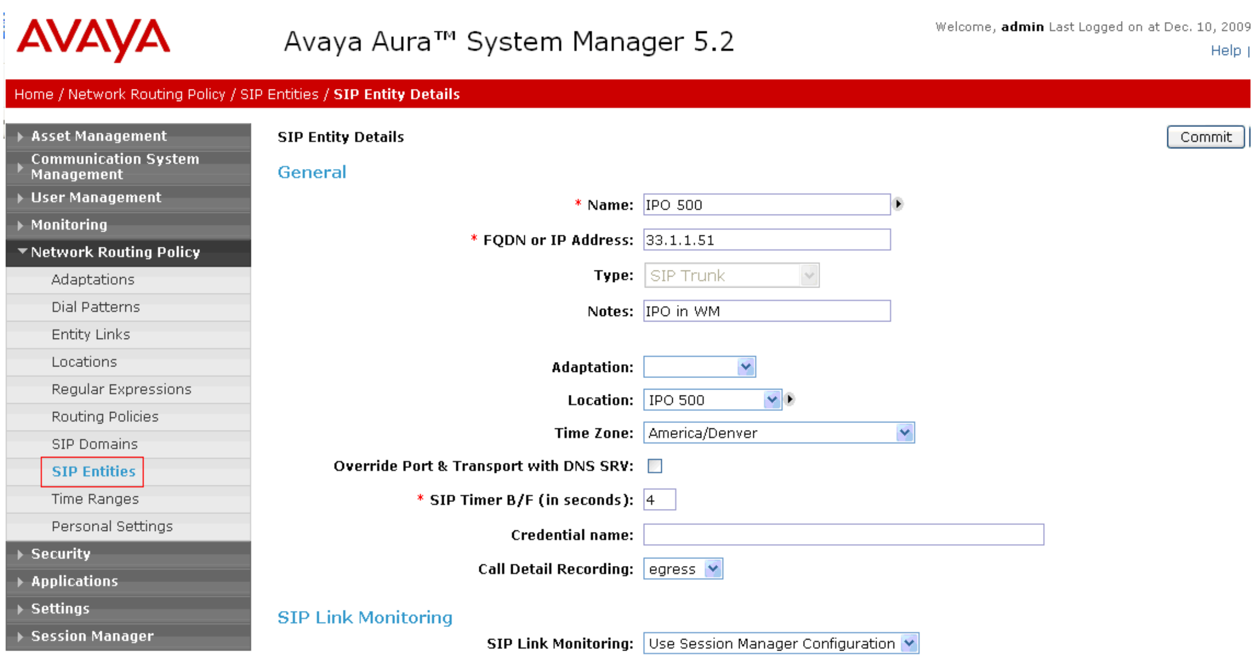
  Select "Link Monitoring Enabled" for Communication Manager Feature Server,

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

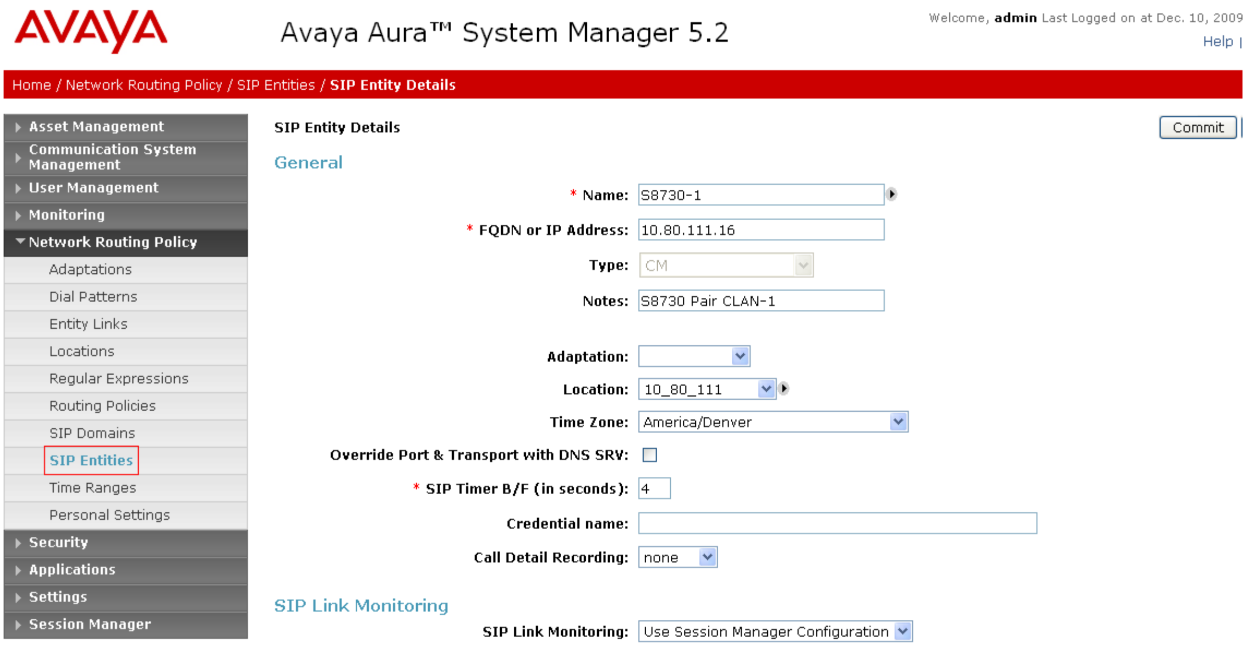
The following screen shows addition of Avaya Aura ™ Session Manager. The IP address used is that of the SM-100 Security Module.



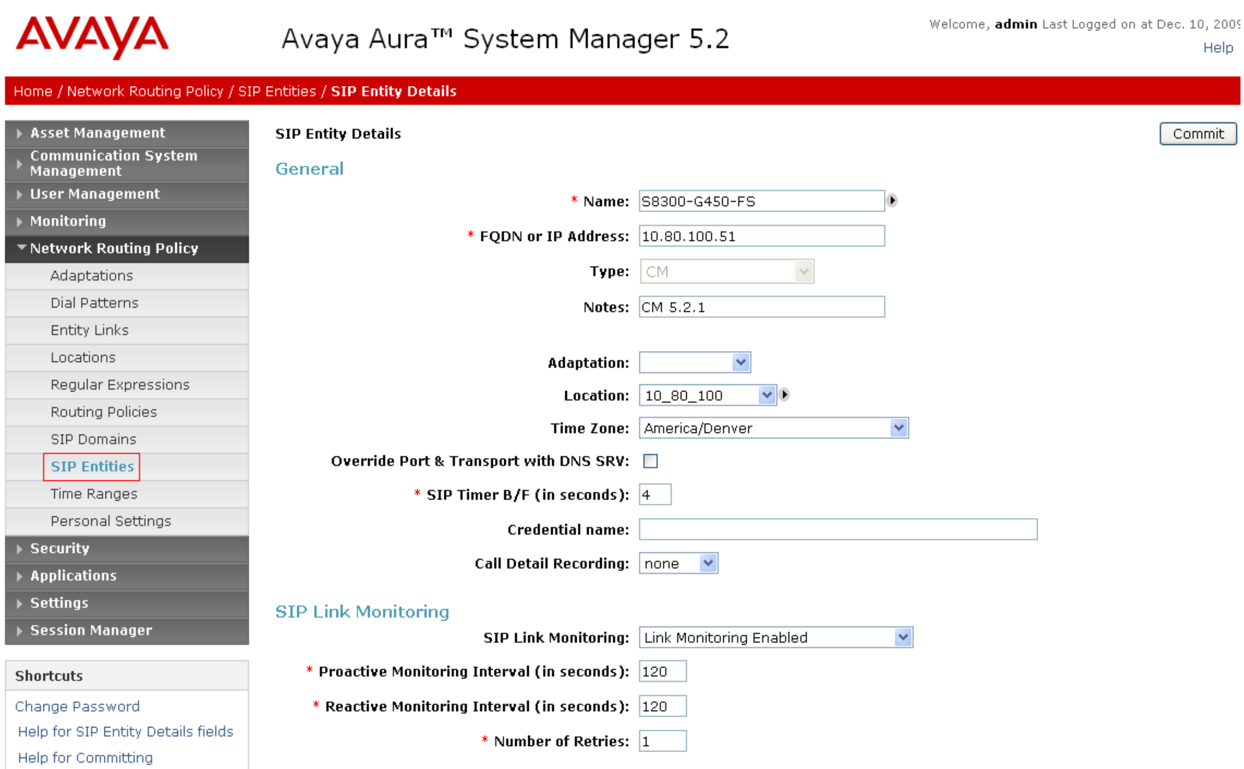
The following screen shows addition of Avaya IP Office.



The following screen shows addition of Communication Manager Access Element. The IP address used is that of the C-LAN board in the Avaya G650 Media gateway.



The following screen shows addition of Communication Manager Feature Server. The IP address used is that of the S8300C server.



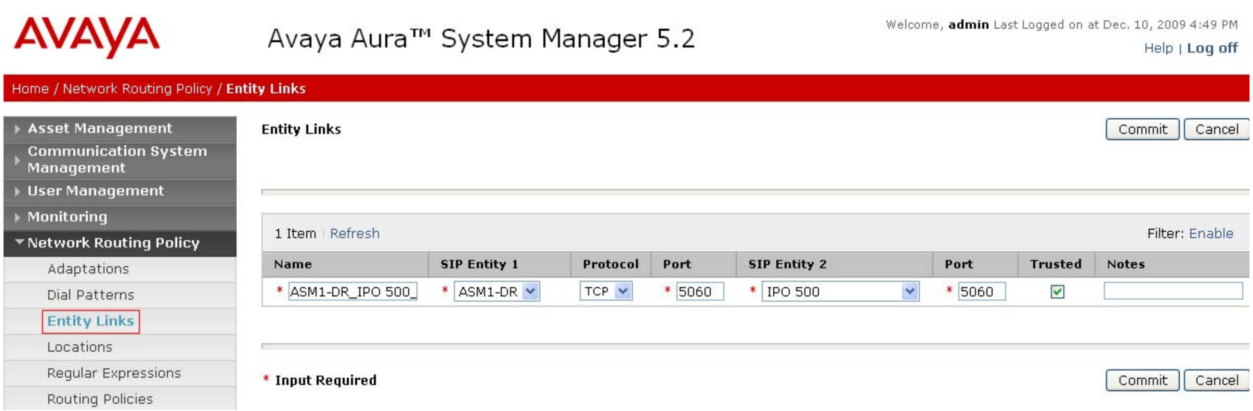
4.4 Add Entity Links

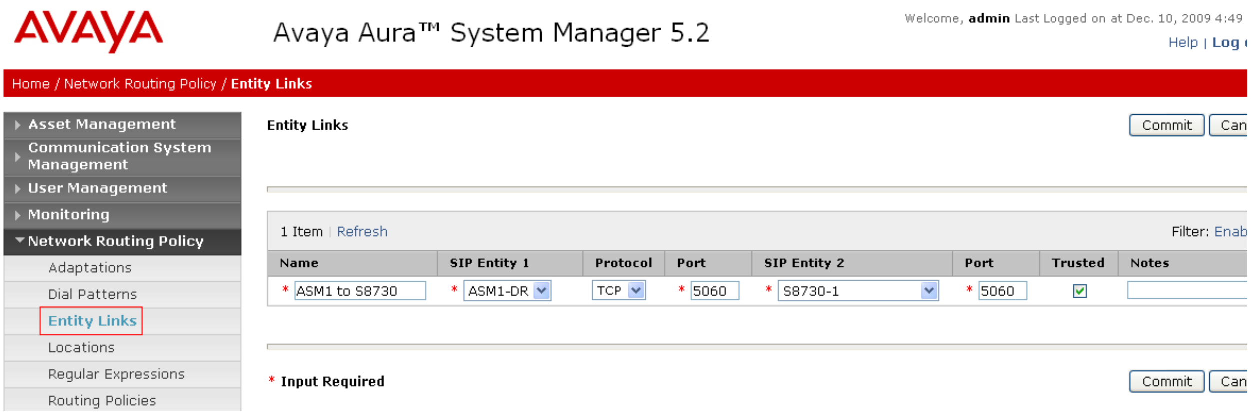
A SIP trunk between Avaya Aura™ Session Manager and a telephony system is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button on the right. Fill in the following fields in the new row that is displayed:

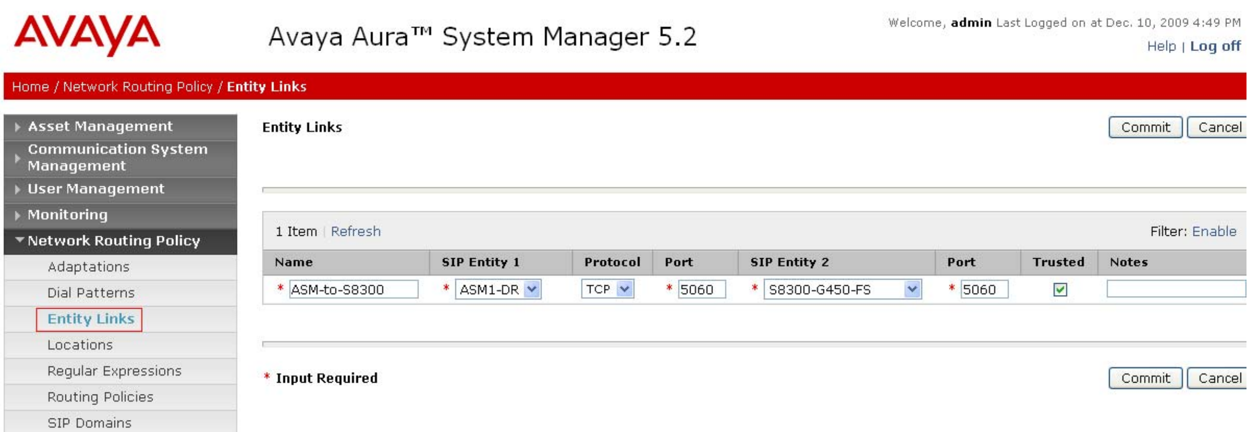
- **Name:** A descriptive name.
- **SIP Entity 1:** Select the Avaya Aura™ Session Manager.
- **Port:** Port number to which the other system sends SIP requests In the sample configuration, TCP Protocol was used.
- **SIP Entity 2:** Select the name of the other system.
- **Port:** Port number on which 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 4.3** will be denied.

Click **Commit** to save each Entity Link definition. The following screens illustrate adding the three Entity Links for:

1. Avaya IP Office
2. Communication Manager Access Element
3. Communication Manager Feature Server





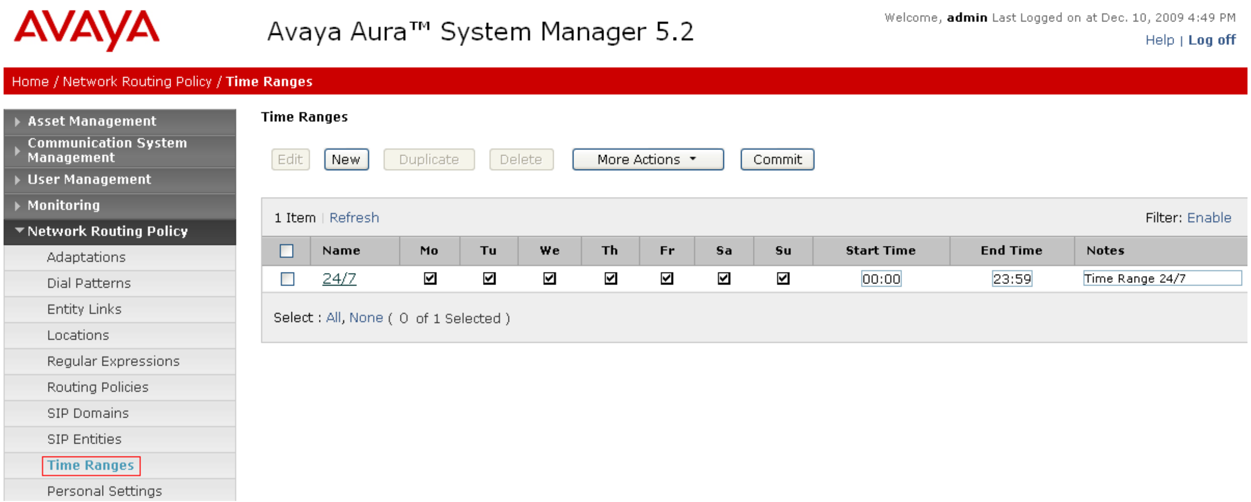


4.5 Add Time Ranges

Before adding routing policies (see next section), time ranges must be defined during which the policies will be active. In the sample configuration, one policy was defined that would allow routing to occur at anytime. To add this time range, select **Time** Ranges, and click on the left and click on the **New** button on the right. Fill in the following:

- **Name:** A descriptive name (e.g., "Anytime").
- **Mo** through **Su** Check the box under each of these headings
- **Start Time** Enter 00:00.
- **End Time** Enter 23:59

Click **Commit** to save this time range.


AVAYA
Avaya Aura™ System Manager 5.2
Welcome, admin Last Logged on at Dec. 10, 2009 4:49 PM
Help | Log off
Home / Network Routing Policy / Time Ranges
Asset Management
Communication System Management
User Management
Monitoring
Network Routing Policy
Adaptations
Dial Patterns
Entity Links
Locations
Regular Expressions
Routing Policies
SIP Domains
SIP Entities
Time Ranges
Personal Settings
Time Ranges
Edit
New
Duplicate
Delete
More Actions
Commit
1 Item | Refresh
Filter: Enable
Name
Mo
Tu
We
Th
Fr
Sa
Su
Start Time
End Time
Notes
24/7
00:00
23:59
Time Range 24/7
Select : All, None (0 of 1 Selected)

4.6 Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 4.3**. Two routing policies must be added – one for IP Office, one for Communication Manager Access Element. To add a routing policy, select **Routing Policies** on the left and click on the **New** button on the right. The following screen is displayed. Fill in the following:

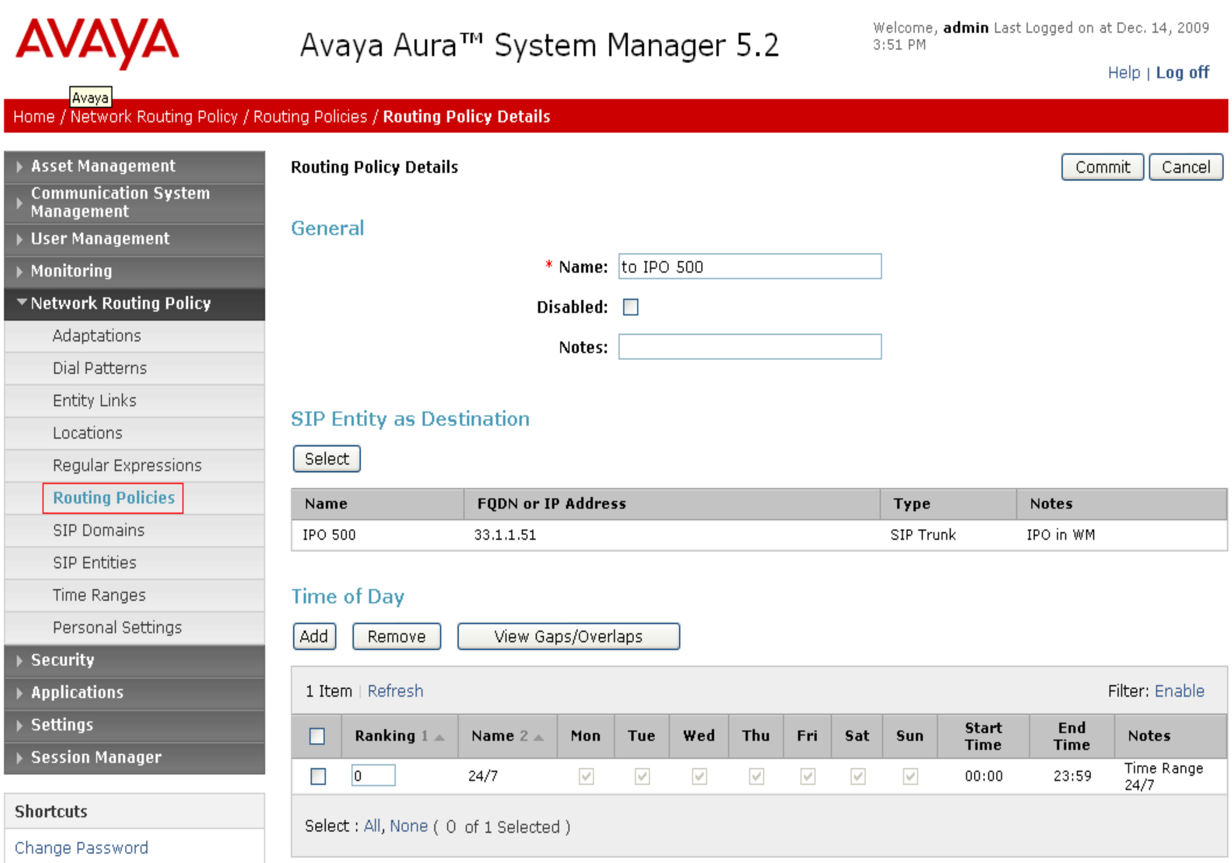
Under *General*:
Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:
Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

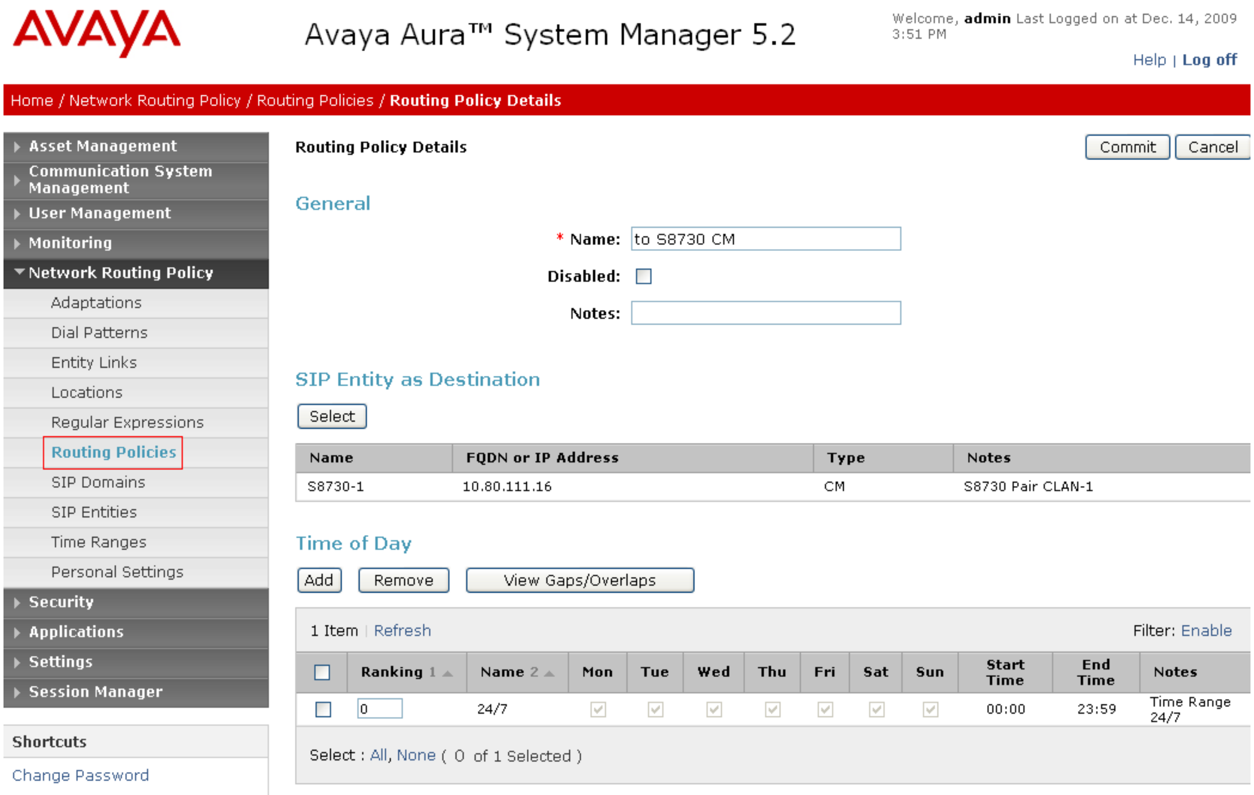
Under *Time of Day*:
Click **Add**, and select the time range configured in the previous section.

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

The following screens show the Routing Policy for IP Office.



The following screens show the Routing Policy for Communication Manager Access Element.



No Routing Policy is required for Communication Manager Feature Server, as these phones are registered directly to Session Manager.

4.7 Add Dial Patterns

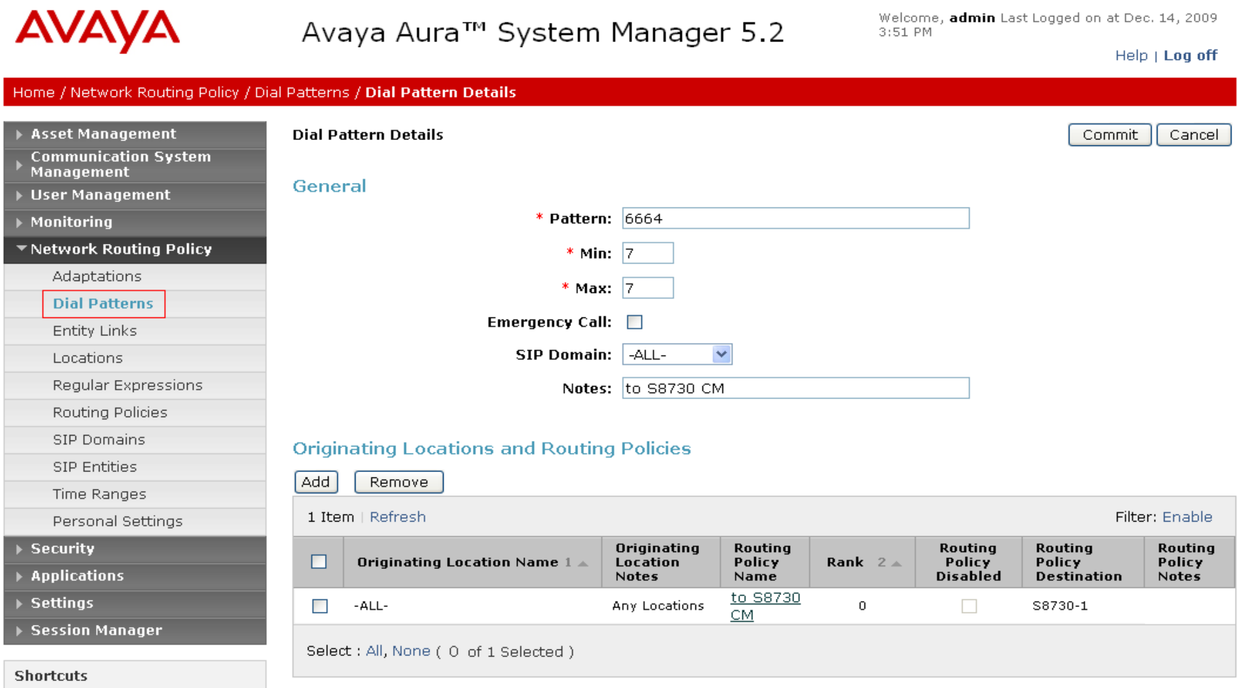
Define dial patterns to direct calls to the appropriate SIP Entity. 7-digit extensions beginning with “6664” reside on Communication Manager Access Element, and 3-digit extensions beginning with “2” reside on Avaya IP Office. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button on the right. Fill in the following, as shown in the screen below, which corresponds to the dial pattern for routing calls to Avaya Aura™ Communication Manager Access Element:

Under *General*:

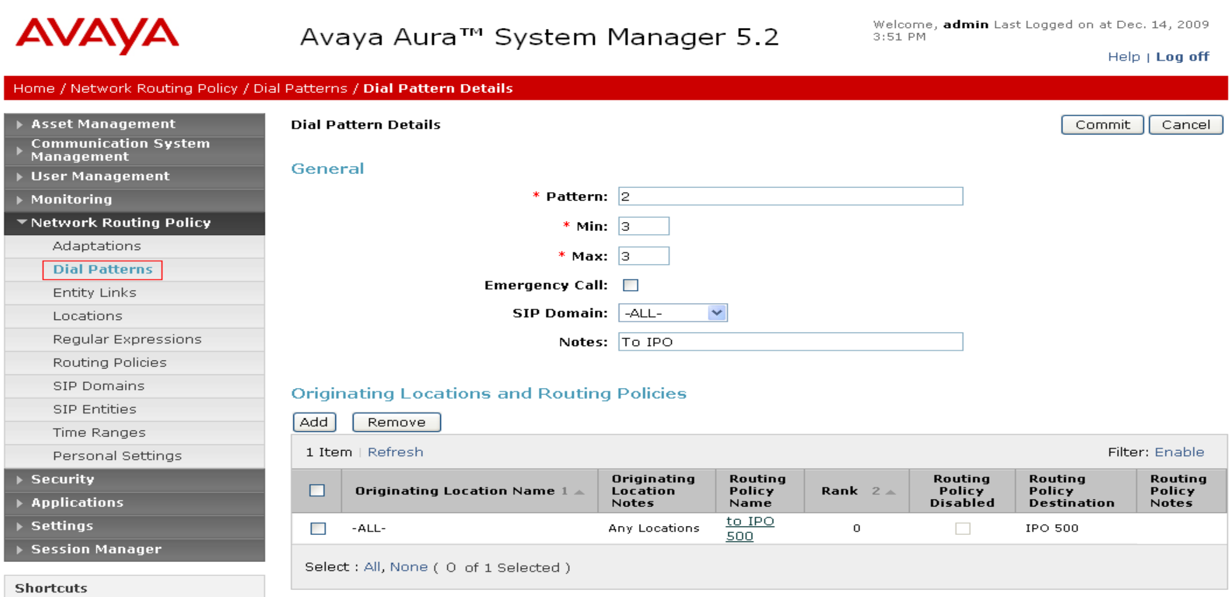
- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain specified in **Section 4.1**
- **Notes** Comment on purpose of dial pattern.

Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the appropriate location and routing policy from the list.

In the sample configuration, all calls originating from endpoints connected to Avaya IP Office dial “666-xxxx” where “4xxx” is the 4-digit extension on Communication Manager Access Element.



In the sample configuration, all calls originating from endpoints connected to Communication Manager Access Element or Feature server dial “2xx” where “2xx” is the 3-digit extension on Avaya IP Office.



5 Configure Avaya Aura™ Communication Manager Access Element

This section describes configuring Avaya Aura™ Communication Manager Access Element in the following areas. Some administration screens have been abbreviated for clarity.

- Verify Communication Manager license
- Administer system parameters features
- Administer IP node names
- Administer IP interface
- Administer IP codec set and network region
- Administer SIP trunk group and signaling group
- Administer SIP trunk group members and route patterns
- Administer private numbering
- Administer dial plan and AAR analysis

5.1 Verify Communication Manager License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity or a required feature is not enabled, contact an authorized Avaya sales representative to make the appropriate changes.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 500   0
           Maximum Concurrently Registered IP Stations: 18000 4
             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: 100   0
                        Maximum Video Capable Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 50    20
```

5.2 Configure System Parameters Features

Use the "change system-parameters features" command to allow for trunk-to-trunk transfers. Submit the change.

This feature is needed to be able to transfer an incoming/outgoing call from/to the remote switch back out to the same or another switch For simplicity, the **Trunk-to-Trunk Transfer** field was set to "all" to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk, and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented using Class Of Restriction or Class Of Service levels. Refer to the appropriate documentation in **Section 9** for more details.

```
change system-parameters features                               Page   1 of  18
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y
                             Music/Tone on Hold: none
```

5.3 Configure IP Node Names

Use the "change node-names ip" command to add entries for the C-LAN that will be used for connectivity, and Avaya Aura ™ Session Manager and Avaya IP Office. The actual node names and IP addresses may vary. Submit these changes.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
8730-1              10.80.111.11
8730-2              10.80.111.12
ASM1                10.80.100.24
CLAN-1              10.80.111.16
CLAN-2              10.80.111.17
IPO                 33.1.1.51
VAL                 10.80.111.18
XFire               10.80.111.13
default             0.0.0.0
gateway1            10.80.111.1
procr               0.0.0.0
```

5.4 Configure IP Interface for C-LAN

Add the C-LAN to the system configuration using the “add ip-interface 1a03” command. The actual slot number may vary. In this case, “1a03” is used as the slot number. Enter the C-LAN node name assigned from **Section 5.3** into the **Node Name** field.

Enter proper values for the **Subnet Mask** and **Gateway Node Name** fields. In this case, “24” and “Gateway001” are used to correspond to the network configuration in these Application Notes. Set the **Enable Interface** and **Allow H.323 Endpoints** fields to “y”. Default values may be used in the remaining fields. Submit these changes.

```
add ip-interface 1a03                                           Page   1 of   3
                                  IP INTERFACES


                  Type: C-LAN
                  Slot: 01A03       Target socket load and Warning level: 400
           Code/Suffix: TN799  D          Receive Buffer TCP Window Size: 8320
     Enable Interface? y                           Allow H.323 Endpoints? y
                  VLAN: n                           Allow H.248 Gateways? y
        Network Region: 1                            Gatekeeper Priority: 5


                                 IPV4 PARAMETERS
             Node Name: CLAN-1
           Subnet Mask: /24
     Gateway Node Name: gateway1
```

5.5 Configure IP Codec Sets and Network Regions

Configure the IP codec set to use for calls to the Avaya IP Office. Use the “change ip-codec-set n” command, where “n” is an existing codec set number to be used for interoperability. Enter the desired audio codec type in the **Audio Codec** field. Retain the default values for the remaining fields and submit these changes.

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

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2: G.729             n         2        20
 3:

     Media Encryption
 1: none
```

In the test configuration, network region "1" was used for calls to the Avaya IP Office via Avaya Aura ™ Session Manager. Use the "change ip-network-region 1" command to configure this network region. For the **Authoritative Domain** field, enter the SIP domain name configured for this enterprise network (See **Section 3.4**). This value is used to populate the SIP domain in the From header of SIP INVITE messages for outbound calls. It is also must match the SIP domain in the request URI of incoming INVITEs from other systems. For the **Codec Set** field, enter the corresponding audio codec set configured above in this section. Enable the **Intra-region IP-IP Direct Audio**, and **Inter-region IP-IP Direct Audio**. These settings will enable direct media between Avaya IP telephones and the far end. Retain the default values for the remaining fields, and submit these changes.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       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: 16585
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: 26change
```

5.6 Configure SIP Signaling Group and Trunk Group

5.6.1 SIP Signaling Group

In the test configuration, trunk group "10" and signaling group "10" were used to reach Avaya Aura™ Session Manager. Use the "add signaling-group n" command, where "n" is an available signaling group number. Enter the following values for the specified fields, and retain the default values for all remaining fields. Submit these changes.

- **Group Type:** "sip"
- **Transport Method:** "tcp"
- **Near-end Node Name:** C-LAN node name from **Section 5.3**.
- **Far-end Node Name:** Avaya Aura™ Session Manager node name from **Section 5.3**.
- **Near-end Listen Port:** "5060"
- **Far-end Listen Port:** "5060"
- **Far-end Network Region:** Avaya network region number "1" from **Section 5.5**.
- **DTMF over IP:** "rtp-payload"

Note: Leave the Far End Domain as blank.

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

 Group Number: 10             Group Type: sip
                        Transport Method: tcp
  IMS Enabled? n
     IP Video? n




   Near-end Node Name: CLAN-1                Far-end Node Name: ASM1
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        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? n                       Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 10
```

5.6.2 SIP Trunk Group

Use the "add trunk-group n" command, where "n" is an available trunk group number. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "sip"
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** "tie"
- **Number of Members:** The number of SIP trunks to be allocated to calls routed to Session Manager (must be within the limits of the total trunks configured in **Section 5.1**).

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

Group Number: 10                     Group Type: sip          CDR Reports: y
  Group Name: SIP trunk to ASM1             COR: 1        TN: 1        TAC: #10
   Direction: two-way       Outgoing Display? y
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

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

Navigate to **Page 3**, and enter "private" for the **Numbering Format** field as shown below. Use default values for all other fields.

```
add trunk-group 10                                             Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y

                     Numbering Format: private
                                                UUI Treatment: service-provider
```

Navigate to **Page 4**, and enter “101” for the **Telephone Event Payload Type** field as shown below. Use default values for all other fields. Submit these changes.

```
add trunk-group 10                                             Page   4 of  21
                              PROTOCOL VARIATIONS

                      Mark Users as Phone? y
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? n
                    Send Diversion Header? n
                  Support Request History? y
             Telephone Event Payload Type: 101
```

5.7 Configure Route Pattern

Configure a route pattern to correspond to the newly added SIP trunk group. Use the “change route-pattern n” command, where “n” is an available route pattern. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Pattern Name:** A descriptive name.
- **Grp No:** The trunk group number from **Section 5.6.2**.
- **FRL:** Enter a level that allows access to this trunk, with 0 being least restrictive.
- **No. Del Dgts**: Enter “3”. For the sample configuration, the user dials “233-2xx”, however “233” will be deleted and only “2xx” will be sent to Session Manager via the SIP trunk.

```
change route-pattern 15                                         Page   1 of   3
                    Pattern Number: 15  Pattern Name:
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 10   0                    3                                          n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                Dgts Format
                                                        Subaddress
 1: y y y y y n  n            rest                                          none
```

5.8 Configure Private Numbering

Use the "change private-numbering 3" command, to define the calling party number to be sent to Avaya IP Office. Add an entry for the trunk group defined in **Section 5.6.2** to reach Avaya IP Office endpoints. In the sample configuration, all calls originating from endpoints connected to Communication Manager Access Element dial "233-2xx" where "2xx" is the 3-digit extension on Avaya IP Office. The call will be routed over the SIP trunk defined in **Section 5.6.2**. Submit these changes.

```
change private-numbering 3                                          Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 6  233              10         233              6     Total Administered: 5
 7  666              10         303              10       Maximum Entries: 540
 7  6664             2                           7
 7  6665             10                          7
 7  6664003          10                          7
```

5.9 Administer Dial Plan and AAR Analysis

This section provides sample Automatic Alternate Routing (AAR) used for routing calls with dialed digits 233-2xx to Avaya IP Office. Note that other methods of routing may be used. Use the "change dialplan analysis" command, and add an entry to specify use of AAR for routing of digits 233-2xx. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Dialed String:** Dialed prefix digits to match on, in this case "2".
- **Total Length:** Length of the full dialed number, in this case "6"
- **Call Type:** "aar"

```
change dialplan analysis                                            Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all          Percent Full:    1

       Dialed    Total  Call    Dialed    Total  Call     Dialed    Total  Call
       String    Length Type    String    Length Type     String    Length Type
     0             1    attd
     1             2    dac
     2             6    aar
     400           7    ext
     500           5    ext
     522           7    ext
     666           7    ext
     71            5    aar
     777           7    ext
     8             1    fac
     9             1    fac
     *             3    fac
     #             3    dac
```

Use the "change aar analysis 233" command, and add an entry to specify how to route the calls to Avaya IP Office endpoints. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Dialed String:** Dialed prefix digits to match on, in this case "233".
- **Total Min:** Minimum number of digts.
- **Total Max:** Maximum number of digits.
- **Route Pattern:** The route pattern number from **Section 5.7**.
- **Call Type:** "aar"

```
change aar analysis 233                                         Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    1

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    233                   6    6     15       aar          n
    3                     7    7     999      aar          n
    4                     7    7     999      aar          n
    5                     7    7     999      aar          n
    522                   7    7     20       aar          n
    6                     7    7     10       aar          n
    6663                  7    7     20       aar          n
    6665000               7    7     20       aar          n
    7                     7    7     2        lev0         n
    8                     7    7     999      aar          n
    9                     7    7     999      aar          n
```

5.10 Save Translations

Configuration of Communication Manager Access Element is complete. Use the "save Translations command to save these changes.

6 Configure Avaya Aura™ Communication Manager Feature Server

This section covers the administrative steps to route calls between SIP endpoints registered to Session Manager and Avaya IP Office via the SIP trunk. Avaya 9620 IP Telephones configured as SIP users utilizes the Avaya Aura™ Session Manager User Registration feature and require Communication Manager Feature Server. Communication Manager as a feature server only supports IMS-SIP users that are registered to Avaya Aura™ Session Manager. The Communication Manager Feature Server is connected to Session Manager via an IMS-enabled SIP signaling group and associated SIP trunk group. Actual administration for SIP endpoints is not covered in this document.

This section describes configuring Avaya Aura™ Communication Manager Feature Server in the following areas. Some administrative screens are not shown in this section, as they might be similar to **Section 5.**

- Verify Communication Manager license
- Administer system parameters features
- Administer IP node names
- Administer IP interface
- Administer IP codec set and network region
- Administer SIP trunk group and signaling group
- Administer SIP trunk group members and route patterns
- Administer private numbering
- Administer dial plan and AAR analysis

6.1 Verify Communication Manager License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections. The license file installed on the system controls the maximum permitted. If there is insufficient capacity or a required feature is not enabled, contact an authorized Avaya sales representative to make the appropriate changes.

6.2 Configure System Parameters Features

Use the "change system-parameters features" command to allow for **trunk-to-trunk transfers** as shown in **Section 5.2**.

6.3 Configure IP Node Names

Use the “change node-names ip” command to add entries for Avaya Aura ™ Session Manager and Avaya IP Office. The actual node names and IP addresses may vary. Submit these changes.

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
ASM1                    10.80.100.24
default                 0.0.0.0
procr                   10.80.100.51
IPO                     33.1.1.51
```

6.4 Configure SIP Signaling Group and Trunk Group

6.4.1 SIP Signaling Group

In the test configuration, trunk group “10” and signaling group “10” were used to reach Avaya Aura™ Session Manager. Use the “add signaling-group n” command, where “n” is an available signaling group number. Enter the following values for the specified fields, and retain the default values for all remaining fields. Submit these changes.

- **Group Type:** “sip”
- **Transport Method:** “tcp”
- **IMS Enabled:** “y”
- **Near-end Node Name:** procr
- **Far-end Node Name:** Avaya Aura™ Session Manager node name from **Section 6.3**.
- **Near-end Listen Port:** “5060”
- **Far-end Listen Port:** “5060”
- **DTMF over IP:** “rtp-payload”
- **Enable Layer 3 Tests:** “y”

Note: Leave the Far End Domain as blank.

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

 Group Number: 10             Group Type: sip
                        Transport Method: tcp
  IMS Enabled? y
     IP Video? n


   Near-end Node Name: procr                  Far-end Node Name: ASM1
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         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                     Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n           Alternate Route Timer(sec): 10
```

6.4.2 SIP Trunk Group

Use the “add trunk-group n” command, where “n” is an available trunk group number. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** “sip”
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** “tie”
- **Number of Members:** The number of SIP trunks to be allocated to calls routed to Session Manager

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

Group Number: 10                   Group Type: sip          CDR Reports: y
  Group Name: SIP trunk to ASM1           COR: 1        TN: 1        TAC: #10
   Direction: two-way       Outgoing Display? y
 Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                     Auth Code? n

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

Navigate to **Page 3**, and enter “private” for the **Numbering Format** field as shown below. Use default values for all other fields. Submit these changes.

```
add trunk-group 10                                               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
```

6.5 Configure Route Pattern

Configure a route pattern to correspond to the newly added SIP trunk group. Use the "change route-pattern n" command, where "n" is an available route pattern. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Pattern Name:** A descriptive name.
- **Grp No:** The trunk group number from **Section 6.4.2**.
- **FRL:** Enter a level that allows access to this trunk, with 0 being least restrictive.
- **No. Del Dgts**: Enter "3". For the sample configuration, the user dails "233-2xx", however "233" will be deleted and only "2xx" will be sent to Avaya IP Office via the SIP trunk.

```
change route-pattern 15                                           Page   1 of   3
                    Pattern Number: 15  Pattern Name:
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                                DCS/ IXC
    No          Mrk Lmt List Del  Digits                                  QSIG
                             Dgts                                         Intw
 1: 10   0                    3                                            n   user
 2:                                                                        n   user
 3:                                                                        n   user
 4:                                                                        n   user
 5:                                                                        n   user
 6:                                                                        n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                            none
```

6.6 Configure Private Numbering

Use the "change private-numbering 3" command, to define the calling party number to be sent to Avaya IP Office. Add an entry for the trunk group defined in **Section 6.4.2** to reach Avaya IP Office endpoints. In the sample configuration, all calls originating from endpoints connected to Communication Manager Access Element dial "233-2xx" where "2xx" is the 3-digit extension on Avaya IP Office. The call will be routed over the SIP trunk defined in **Section 6.4.2**. Submit these changes.

```
change private-numbering 3                                        Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 7  5                10                          7     Total Administered: 3
 7  6                10                          7        Maximum Entries: 540
 6  233              10         233              6
```

6.7 Administer Dial Plan and AAR Analysis

This section provides sample Automatic Alternate Routing (AAR) used for routing calls with dialed digits 233-2xx to Avaya IP Office. Note that other methods of routing may be used. Use the "change dialplan analysis" command, and add an entry to specify use of AAR for routing of digits 233-2xx. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Dialed String:** Dialed prefix digits to match on, in this case "2".
- **Total Length:** Length of the full dialed number, in this case "6"
- **Call Type:** "aar"

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all          Percent Full:    1

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     0            1    attd
     1            2    dac
     2            6    aar
     #            3    dac
```

Use the "change aar analysis 233" command, and add an entry to specify how to route the calls to Avaya IP Office endpoints. Enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Dialed String:** Dialed prefix digits to match on, in this case "233".
- **Total Min:** Minimum number of digts.
- **Total Max:** Maximum number of digits.
- **Route Pattern:** The route pattern number from **Section 6.5**.
- **Call Type:** "aar"

```
change aar analysis 233                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    2

          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    233                   6    6     15        aar          n
    522                   7    7     10        aar          n
    666                   7    7     10        aar          n
    7                     7    7     10        aar          n
```

6.8 Save Translations

Configuration of Communication Manager Feature Server is complete. Use the "save Translations command to save these changes.

Note: After a change on Communication Manager Feature Server which alters the dial plan, synchronization between Communication Manager Feature Server and Session Manager needs to be completed and SIP phones must be rebooted. To force synchronization, execute "stop -s sm-mgmt" followed by "start -s sm-mgmt" on Session Manager command line interface.

7 Verification Steps

This section provides the tests that can be performed on Avaya IP Office, Communication Manager and Session Manager to verify proper configuration of these systems.

7.1 Verify Avaya Aura™ Communication Manager

Verify the status of the SIP trunk group by using the "status trunk n" command, where "n" is the trunk group number administered in **Section 5.6** and **Section 6.4**. Verify that all trunks are in the "in-service/idle" state as shown below. Perform this on both Communication Manager Access Element and Feature Server.

```
status trunk 10

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0010/001 T00024   in-service/idle    no
0010/002 T00025   in-service/idle    no
0010/003 T00026   in-service/idle    no
0010/004 T00027   in-service/idle    no
0010/005 T00028   in-service/idle    no
0010/006 T00029   in-service/idle    no
0010/007 T00030   in-service/idle    no
0010/008 T00031   in-service/idle    no
0010/009 T00032   in-service/idle    no
0010/010 T00033   in-service/idle    no
```

Verify the status of the SIP signaling groups by using the "status signaling-group n" command, where "n" is the signaling group number administered in **Section 5.6** and **Section 6.4**. Verify the signaling group is "in-service" as indicated in the **Group State** field shown below. Perform this on both Communication Manager Access Element and Feature Server.

```
status signaling-group 10
                         STATUS SIGNALING GROUP

        Group ID: 10                            Active NCA-TSC Count: 0
      Group Type: sip                            Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

Make a call between the Avaya 9600 Series IP Telephone and the Avaya IP Office 500 IP Telephone. Verify the status of connected SIP trunks on Communication Manager Access Element SAT terminal by using the "status trunk x/y", where "x" is the number of the SIP trunk group from **Section 5.6.2** to reach Avaya Aura™ Session Manager, and "y" is the member number of a connected trunk. Verify on Page 1 that the **Service State** is "in-service/active". On Page 2, verify that the IP addresses of the C-LAN and Avaya Aura™ Session Manager are shown in the **Signaling** section. The Audio Connection will be "ip-direct". The Near-end IP address will be the IP address of the 9620 IP Telephone and the Far end IP address will be the IP address of the Avaya IP Office.

```
  status trunk 10/7                                              Page   1 of   3
                              TRUNK STATUS

   Trunk Group/Member: 0010/007             Service State: in-service/active
                 Port: T00030            Maintenance Busy? no
   Signaling Group ID: 10

     IGAR Connection? no

      Connected Ports: S00009
```

```
status trunk 10/7                                              Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: 01A0317
  Signaling   IP Address                                  Port
   Near-end:  10.80.111.16                              : 5060
    Far-end:  10.80.100.24                              : 5060
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:           H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                        Codec Type: G.711MU
   Audio      IP Address                                  Port
   Near-end:  10.80.50.38                               : 10106
    Far-end:  33.1.1.51                                 : 49156

 Video Near:
  Video Far:
 Video Port:
  Video Near-end Codec:               Video Far-end Codec:
```

Make a call between the Avaya 9600 Series IP Telephone registered to Session Manager and the Avaya IP Office 500 IP Telephone. Verify the status of connected SIP trunks on Communication Manager Feature Server SAT terminal by using the "status trunk x", where "x" is the number of the SIP trunk group from **Section 6.4.2**.

Note: Two ports on the trunk will be used for this call.

```
  status trunk 10

                             TRUNK GROUP STATUS

  Member   Port     Service State      Mtce Connected Ports
                                       Busy

  0010/001 T00006   in-service/active  no   T00008
  0010/002 T00007   in-service/idle    no
  0010/003 T00008   in-service/active  no   T00006
  0010/004 T00009   in-service/idle    no
  0010/005 T00014   in-service/idle    no
  0010/006 T00015   in-service/idle    no
  0010/007 T00043   in-service/idle    no
  0010/008 T00044   in-service/idle    no
  0010/009 T00045   in-service/idle    no
  0010/010 T00046   in-service/idle    no
```

Issue "status trunk x/y", where "x" is the number of the SIP trunk group to reach Avaya Aura™ Session Manager, and "y" is the member number of a connected trunk. Verify on Page 1 that the **Service State** is "in-service/active". On Page 2, verify that the IP addresses of the S8300C Media Server and Avaya Aura™ Session Manager are shown in the **Signaling** section. The Audio Connection will be "ip-direct". The IP address will be the IP address of the 9620 IP Telephone and the IP address of Avaya IP Office in the **Audio** section. In the screen below, 10.80.50.41 is the IP address of the 9620 IP Telephone registered to Session Manager.

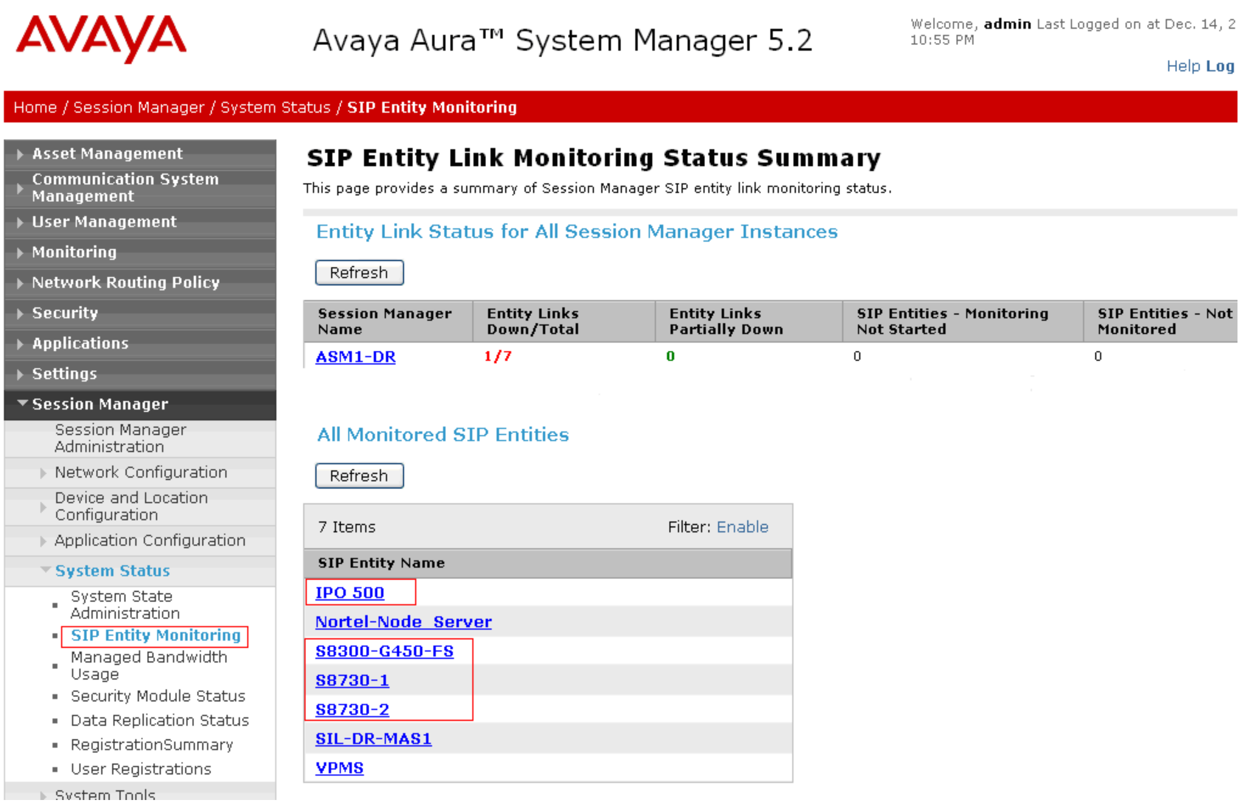
```
status trunk 10/1                                             Page   1 of   3
                                  TRUNK STATUS

 Trunk Group/Member: 0010/001             Service State: in-service/active
               Port: T00006            Maintenance Busy? no
 Signaling Group ID: 10

   IGAR Connection? no

    Connected Ports: T00008

status trunk 10/01                                            Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: 01A0017
  Signaling   IP Address                             Port
   Near-end:  10.80.100.51                         : 5060
    Far-end:  10.80.100.24                         : 5060
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:           H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                        Codec Type: G.711MU
   Audio      IP Address                             Port
   Near-end:  33.1.1.51                            : 49156
    Far-end:  10.80.50.41                          : 5004
```

Issue “status trunk x/y”, where “x” is the number of the SIP trunk group to reach Avaya Aura™ Session Manager, and “y” is the member number of a connected trunk. Verify on Page 1 that the **Service State** is “in-service/active”. On Page 2, verify that the IP addresses of the S8300C Media Server and Avaya Aura™ Session Manager are shown in the **Signaling** section. The IP address will be the IP address of the 9620 IP Telephone and the IP address of Avaya IP Office in the **Audio** section. In the screen below, 10.80.50.41 is the IP address of the 9620 IP Telephone registered to Session Manager.

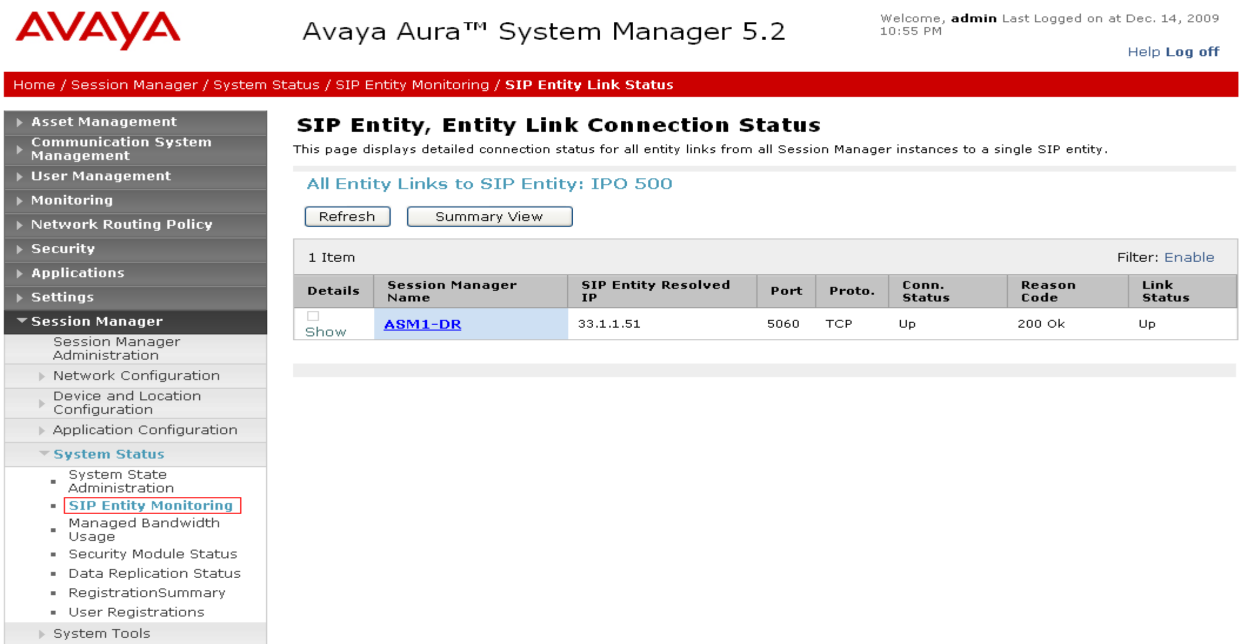
```
status trunk 10/3                                              Page   1 of   3
                              TRUNK STATUS

 Trunk Group/Member: 0010/003             Service State: in-service/active
               Port: T00008            Maintenance Busy? no
 Signaling Group ID: 10

   IGAR Connection? no

    Connected Ports: T00006

status trunk 10/3                                              Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: 01A0017
  Signaling   IP Address                                Port
   Near-end:  10.80.100.51                            : 5060
    Far-end:  10.80.100.24                            : 5060
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:           H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                         Codec Type: G.711MU
   Audio      IP Address                                Port
   Near-end:  10.80.50.41                             : 5004
    Far-end:  33.1.1.51                               : 49156
```

7.2 Verify Avaya Aura™ Session Manager

Expand the Session Manager menu on the left and click SIP Entity Monitoring.



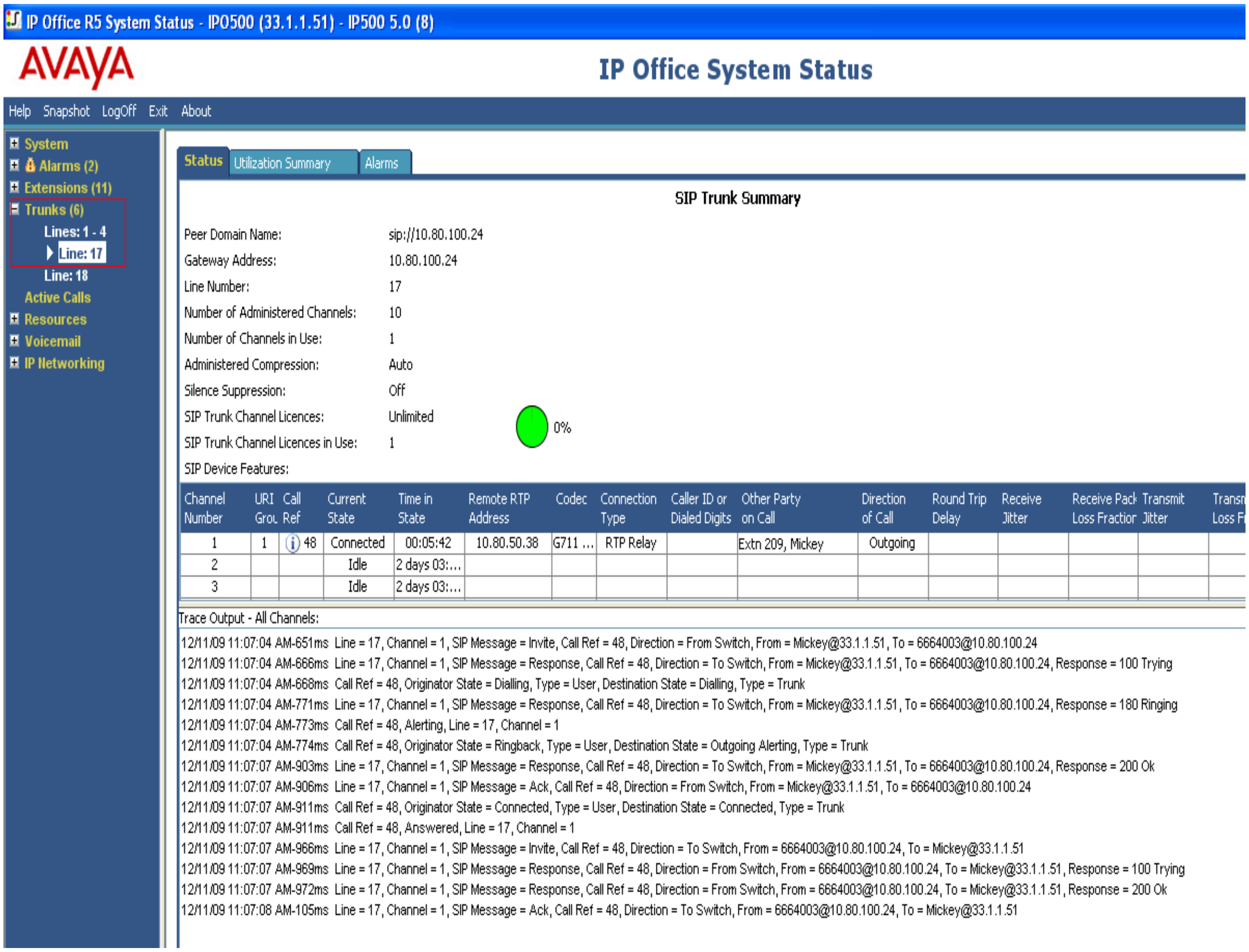
Select the corresponding SIP Entity and verify that the links are up as shown below for Avaya IP Office.



7.3 Verify Avaya IP Office

IP Office can be debugged with the System Status Application. Log into the IP Office Manager PC and select Start > Programs > IP Office > System Status to launch the application. Log into the application using the appropriate credentials.

In the left panel, double-click on the Trunks entry and select SIP trunk created in **Section 3.6**. Press the **Trace All** button. The messages on the line are displayed.



7.4 Verification Scenarios

Verification scenarios for the configuration described in these Application Notes included the following. Proper display of the calling and called party name and number information was verified for all calls.

- Place a call from an extension on the Avaya IP Office to an extension on Communication Manger Access Element. Answer the call and verify talkpath.
- Repeat previous case in the opposite direction.
- Place a call from an extension on the Avaya IP Office to an extension on Communication Manger Feature Server. Answer the call and verify talkpath.
- Repeat previous case in the opposite direction.
- Verify that calls can be transferred from an extension on Avaya IP Office to an extension on Communication Manager.
- Verify that calls can be transferred from an extension on Communication Manager to an extension on Avaya IP Office.
- Verify that extensions on Avaya IP Office can conference in extensions on Communication Manager.
- Verify that extensions on Communication Manager can conference in extensions on Avaya IP Office.

8 Conclusion

These Application Notes describe how to configure a sample configuration for a network that uses Avaya Aura™ Session Manager to connect Avaya Aura™ Communication Manager 5.2.1 and Avaya IP Office using SIP trunks. Interoperability testing included verification of successful bi-directional calls among several types of endpoints with various features including transfer, and conference.

9 Additional References

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

Session Manager:

[1] Avaya Aura™ Session Manager Overview, Doc ID 03-603323, available at http://support.avaya.com.

[2] Installing and Administering Avaya Aura™ Session Manager, Doc ID 03-603324, available at http://support.avaya.com.

[3] Maintaining and Troubleshooting Avaya Aura™ Session Manager, Doc ID 03-603325, available at http://support.avaya.com.

Communication Manager:

[4] *SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers*, Doc ID 555-245-206, May 2009, available at http://support.avaya.com.

[5] *Administering Avaya Aura™ Communication Manager*, Doc ID 03-300509, May 2009, available at http://support.avaya.com.

[6] *Administering Avaya Aura™ Communication Manager as a Feature Server*, Doc ID 03-603479, November 2009, available at http://support.avaya.com

IP Office:

[7] Avaya IP Office Manager, Doc ID 15-601011, available at http://support.avaya.com.

Avaya Application Notes:

[8] *Configuring 96xx SIP Phones on Avaya Aura™ Session Manager Release 5.2*, available at http://www.avaya.com.

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 Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com