Avaya Solution & Interoperability Test Lab

Configuring Avaya Modular Messaging with Avaya IP Office 6.0 (81006), Avaya Aura™ Session Manager 5.2 and Avaya Aura™ Communication Manager 5.2.1 as a Feature Server – Issue 1.0

Abstract

These Application Notes present a sample configuration for a network consisting of a Centralized Avaya Modular Messaging Solution supporting Avaya Aura™ Communication Manager as a Feature Server and Avaya IP Office. These three systems are connected via a common Avaya Aura™ Session Manager.

Testing was conducted via the Internal Interoperability Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

The purpose of this interoperability Application Notes is to validate Avaya Modular Messaging for Avaya IP Office with SIP integration via Avaya Aura™ Session Manager. Avaya IP Office users have mailboxes defined in Modular Messaging which they can access via a dedicated pilot number. The sample network is shown in **Figure 1**, where Avaya Aura™ Communication Manager supports the Avaya 9620 IP Telephone (SIP). Avaya IP Office supports the Avaya 4610 IP Telephone (H.323) and the Avaya 2420 Digital Phone. Avaya Modular Messaging consists of Avaya Messaging Application Server and Avaya Message Storage Server. SIP trunks are used to connect these three systems to Avaya Aura™ Session Manager. 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 for multi-vendor systems to interoperate. Avaya Aura™ Session Manager is managed by a separate Avaya Aura™ System Manager, which can manage multiple Avaya Aura™ Session Managers.

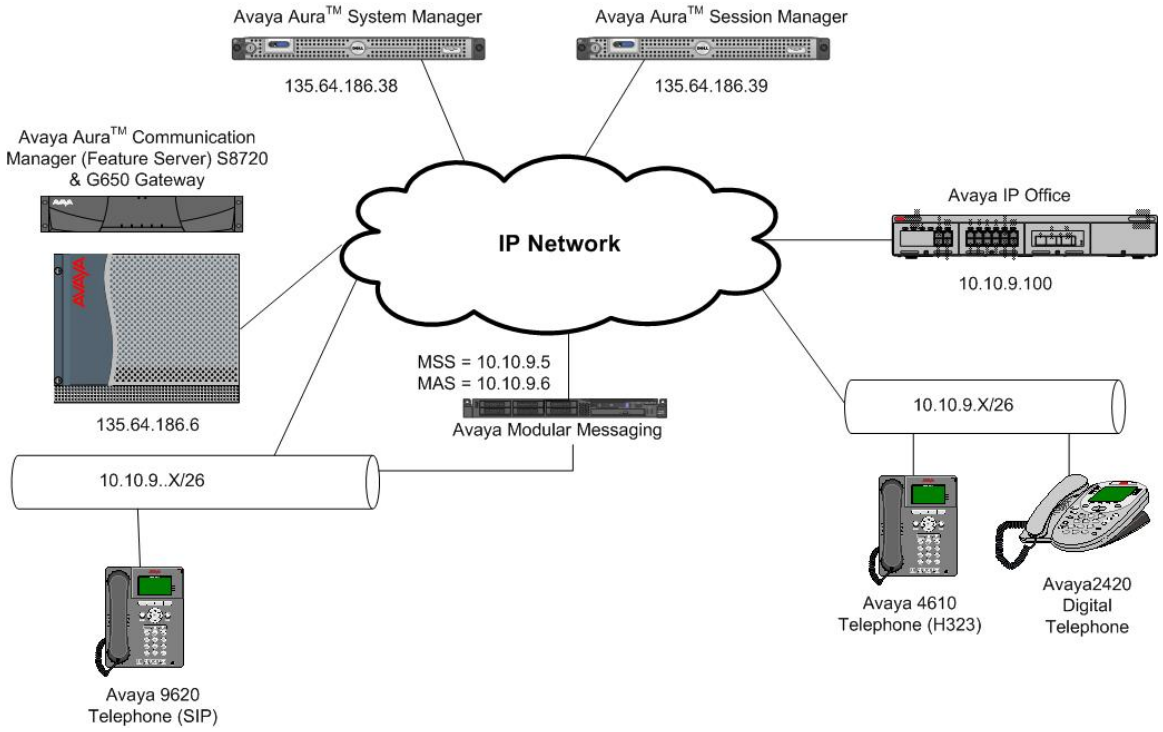


Figure 1: Connection of Avaya Aura™ Communication Manager, Avaya IP Office and Avaya Modular Messaging via Avaya Aura™ Session Manager using SIP Trunks

Avaya telephones are registered to Avaya Aura™ Communication Manager and Avaya IP Office. Avaya Aura™ Communication Manager stations use extensions 320xx. Avaya IP Office registered stations use extensions 701000890xx. One SIP trunk is provisioned from both PBX's to the Avaya Aura™ Session Manager to manage call control for calls between the two systems. One SIP trunk is provisioned to the Avaya Aura™ Session Manager to manage calls to/from Avaya Modular Messaging.

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8720 Media Server	Avaya Aura™ Communication Manager 5.2.1 SP2 (R015x.02.1.016.4)
Avaya G650 Media Gateway • TN799DP C-LAN Circuit Pack • TN2312BP IP Server Interface • TN2602AP IP Media Pro	 HW01 FW035 HW15 FW046 HW08 FW054
Avaya S8510 Server with SM100 Card	Avaya Aura™ Session Manager 5.2 SP2
Avaya S8510 Server	Avaya Aura™ System Manager 5.2 SP2
Avaya S8800 Server	Avaya Modular Messaging 5.2 SP2
Avaya 9620 IP Telephone (SIP)	2.6.2.18
Avaya 2420 Digital Phone	-
Avaya 4610 IP Telephone (H.323)	2.9 SP1
Avaya IP Office (IP500 V2)	Avaya IP Office 6.0 (81006)

3. Configure Avaya Aura™ Communication Manager

This section shows the configuration in Communication Manager. All configurations in this section are administered using the System Access Terminal (SAT). These Application Notes assumed that the basic configuration has already been administered. For further information on Communication Manager, please consult with references **[4]** and **[5]**. The procedures include the following areas:

- Verify Avaya Aura™ Communication Manager License
- Administer System Parameters Features
- Administer IP Node Names
- Administer IP Network Region and Codec Set
- Administer SIP Signaling Group and Trunk Group
- Administer Route Pattern
- Administer Private Numbering
- Administer Dial Plan, Uniform Dial Plan and AAR analysis
- Administer Feature Access Codes
- Administer Hunt Group
- Administer Coverage Path
- Administer SIP Phone
- Save Changes

3.1. Verify Avaya Aura™ Communication Manager License

Use the **display system-parameter customer options** command to compare 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.

Note: The license file installed on the system controls the maximum features 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: 30    0
           Maximum Concurrently Registered IP Stations: 18000 9
             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: 0     0
                       Maximum Video Capable Stations: 10    1
                   Maximum Video Capable IP Softphones: 10    4
                       Maximum Administered SIP Trunks: 100   55
```

3.2. Administer System Parameters Features

Use the **change system-parameters features** command to allow for trunk-to-trunk transfers. This feature is needed to allow for transferring an incoming/outgoing call from/to a remote switch back out to the same or different 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: This feature poses significant security risk and must be used with caution. As an alternative, the trunk-to-trunk feature can be implemented using Class Of Restriction or Class Of Service levels.

```
change system-parameters features                               Page   1 of  18
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? y
                                    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
               Music (or Silence) on Transferred Trunk Calls? no
                       DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n
```

3.3. Administer IP Node Names

Use the **change node-names ip** command to add entries for Communication Manager and Session Manager that will be used for connectivity. In the sample network, **clan1a3** and **135.64.186.6** are entered as **Name** and **IP Address** for the CLAN card in Communication Manager running on the Avaya S8720 Server. In addition, **SM100** and **135.64.186.40** are entered for Session Manager.

```
change node-names ip                                            Page   1 of   2
                                   IP NODE NAMES
    Name                IP Address
Gateway001          135.64.186.1
MBTCM               135.64.186.68
MX6200              135.64.186.15
SM100               135.64.186.40
clan1a3             135.64.186.6
clan1b3             135.64.186.7
default             0.0.0.0
mpro1a2             135.64.186.8
mpro1b2             135.64.186.9
onexmobile          135.64.186.30
procr               135.64.186.10
silstackaes         135.64.186.28
```

3.4. Administer IP Network Region and Codec Set

Use the **change ip-network-region n** command, where **n** is the network region number to configure the network region being used. In the sample network, ip-network-region 3 is used. For the **Authoritative Domain** field, enter the SIP domain name configured for this enterprise and a descriptive **Name** for this ip-network-region. Set **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** to **yes** to allow for direct media between endpoints. Set the **Codec Set** to **3** to use ip-codec-set 3.

```
change ip-network-region 3                                      Page   1 of  19
                              IP NETWORK REGION
  Region: 3
Location:         Authoritative Domain: silstack.com
    Name: To ASM
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 3              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                     IP Audio Hairpinning? n
   UDP Port Max: 3329
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
```

Use the **change ip-codec-set n** command, where **n** is the existing codec set number to configure the desired audio codec.

```
change ip-codec-set 3                                          Page   1 of   2

                           IP Codec Set

    Codec Set: 3

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

3.5. Administer SIP Signaling Group and Trunk Group

3.5.1. SIP Signaling Group

In the test configuration, Communication Manager acts as a Feature Server. An IMS enabled SIP trunk is required. Use signal group 150 along with trunk group 150 to reach the Session Manager. Use the **add signaling-group n** command, where **n** is the signaling-group number being added to the system. Use the values defined in **Section 3.3** and **3.4** for **Near-end Node Name**, **Far-End Node-Name** and **Far-End Network Region**. The **Far-end Domain** is left blank so that the signaling group accepts any authoritative domain. Set **IMS Enabled** to **y**.

```
add signaling-group 150                                         Page   1 of   2

                                     SIGNALING GROUP

 Group Number: 150            Group Type: sip
                        Transport Method: tcp
  IMS Enabled? y
     IP Video? n



 Near-end Node Name: clan1a3               Far-end Node Name: SM100
 Near-end Listen Port: 5063                Far-end Listen Port: 5063
                                        Far-end Network Region: 3
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? y           Alternate Route Timer(sec): 6
```

3.5.2. SIP Trunk Group

Use the **add trunk-group n** command, where **n** is the new trunk group number being added to the system. The following screens show the settings used for trunk group 150. Enter the following:

- **Group Type** **sip**
- **TAC** a numeric value i.e. **150**
- **Service Type** **tie**
- **Signaling Group** the signaling group defined in **Section 3.5.1**, i.e. **150**
- **Number of Members** set to a numeric value, i.e. **10**

```
add trunk-group 150                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 150                 Group Type: sip           CDR Reports: y
  Group Name: Avaya SIP phones            COR: 1        TN: 1        TAC: 150
   Direction: two-way        Outgoing Display? y
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                        Signaling Group: 150
                                                      Number of Members: 10
```

Navigate to **Page 3** and enter **private** for **Numbering Format**.

```
add trunk-group 150                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y

                      Numbering Format: private
                                                   UUI Treatment: service-provider

                                                    Replace Restricted Numbers? y
                                                   Replace Unavailable Numbers? y

 Show ANSWERED BY on Display? Y
```

Navigate to P**age 4** and enter **120** for **Telephone Event Payload Type.**

```
add trunk-group 150                                          Page   4 of  21
                              PROTOCOL VARIATIONS

                       Mark Users as Phone? n
            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: 120
```

3.6. Administer 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 the route pattern number specified in **Section 3.8**. Configure this route pattern to route calls to trunk group number **150** configured in **Section 3.5.2.** Assign the lowest **FRL** (facility restriction level) to allow all callers to use this route pattern.

```
change route-pattern 150                                          Page   1 of   3
                    Pattern Number: 140 Pattern Name: To ASM
                             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: 150  0                                                               n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      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
 2: y y y y y n  n            rest                                          none
 3: y y y y y n  n            rest                                          none
 4: y y y y y n  n            rest                                          none
```

3.7. Administer Private Numbering

Use the **change private-numbering** command to define the calling party number to be sent out through the SIP trunk. In the sample network configuration below, all calls originating from a 5-digit extension beginning with 320 will result in a 5-digit calling number. The calling party number will be in the SIP “From” header.

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

Ext Ext            Trk        Private          Total
Len Code           Grp(s)     Prefix           Len
 5  2                                         5     Total Administered: 4
 5  4                                         5        Maximum Entries: 540
 5  8                                         5
 5  320                                       5
```

3.8. Administer Dial Plan, Uniform Dial Plan and AAR analysis

Configure the dial plan for dialing 11-digit extensions beginning with **701000** to stations registered with IP Office. Use the **change dialplan analysis** command to define **Dialed String 701000** as a **udp Call Type**.

```
change dialplan analysis                                          Page   1 of  12
                              DIAL PLAN ANALYSIS TABLE
                                    Location:  all           Percent Full:    2

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     1            3    dac
     3            5    ext
     701000       11   udp
     9            1    fac
     *            3    fac
     #            3    fac
```

Use the **change uniform-dialplan n** command where **n** is the dial string pattern to configure a **udp** entry for **Dialed String 701000**

```
change uniform-dialplan 7                                         Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                                 Percent Full: 0

  Matching                    Insert              Node
  Pattern       Len Del       Digits     Net Conv Num
 701000          11  0                   aar  n
```

Use the **change aar analysis 0** command to configure an **aar** entry for **Dialed String 701000** to use **Route Pattern 150.** Add another entry to cover calls to the voicemail number **39999.**

```
change aar analysis 0                                             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
    701000                5    5     150      aar          n
    39999                 5    5     150      aar          n
    5                     7    7     254      aar          n
    6                     7    7     254      aar          n
    9                     7    7     254      aar          n
```

3.9. Administer Feature Access Code

Configure a feature access code to use for AAR routing. Use the **change feature access code** command to define an **Auto Alternate Routing (AAR) Access Code**.

```
change feature-access-codes                                       Page   1 of   8
                                FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code: #00
                         Attendant Access Code:
```

```
      Auto Alternate Routing (AAR) Access Code: *8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA: *1     All: *2     Deactivation: *3
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
```

3.10. Administer Hunt Group

Configure a Hunt Group to be used as the call coverage point for the call coverage path assigned to Modular Messaging subscribers. Use the **add hunt-group n** command where **n** is the hunt group number to be assigned. Configure a **Group Name** and **Group Extension** number to be used as the Modular Messaging pilot name and number. Select **ucd-mia** for **Group Type.**

```
add hunt-group 2                                                Page   1 of  60
                                   HUNT GROUP

            Group Number: 2                                  ACD? n
              Group Name: VoiceMail                        Queue? n
         Group Extension: 39999                           Vector? n
              Group Type: ucd-mia                  Coverage Path:
                      TN: 1          Night Service Destination:
                     COR: 1                     MM Early Answer? n
           Security Code:                Local Agent Preference? n
 ISDN/SIP Caller Display:
```

Navigate to **Page 2**. Select **sip-adjunct** for **Message Center**. For **Voice Mail Number** and **Voice Mail Handle** use the Group Extension number and Group Name defined on **Page 1** respectively. For **Routing Digits** use the AAR access code defined in **Section 3.9**.

```
add hunt-group 2                                                Page   2 of  60
                                   HUNT GROUP




                       Message Center: sip-adjunct

     Voice Mail Number        Voice Mail Handle         Routing Digits
                                                  (e.g., AAR/ARS Access Code)
     39999                    VoiceMail                 *8
```

3.11. Administer Coverage Path

Configure a coverage path for the MAS subscribers. Use command **add coverage path n** where **n** is the coverage path number to be assigned. Configure **COVERAGE POINTS**, using value **hn** where **n** is the hunt group number defined in **Section 3.10**.

```
add coverage path 2
                                COVERAGE PATH

                   Coverage Path Number: 2
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?             n              n
               Busy?             y              y
       Don't Answer?             y              y         Number of Rings: 2
                All?             n              n
 DND/SAC/Goto Cover?             y              y
   Holiday Coverage?             n              n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h2             Rng: 2  Point2:
  Point3:                        Point4:
  Point5:                        Point6:
```

3.12. Administer SIP Phone

Configure the SIP Phone discussed in **Section 4.12** to allow call coverage to Modular Messaging. Use the command **change station 32007** and on **Page 1** for **Coverage Path 1** use the coverage path defined in **Section 3.11**.

```
change station 32007                                          Page   1 of   6
                                     STATION

Extension: 32007                         Lock Messages? n               BCC: 0
     Type: 9650SIP                       Security Code:                  TN: 1
     Port: S00004                      Coverage Path 1: 2               COR: 1
     Name: phelan, tom                 Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19
                                                 Message Lamp Ext: 32007

        Display Language: english                  Button Modules: 0

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

                                                         IP Video? n
```

Navigate to **Page 2** and set **MWI Served User Type** to **sip-adjunct** which is the **Message Center** defined in **Section 3.10**. Set **Per Station CPN - Send Calling Number** to **y.**

```
change station 32007                                          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? y
                                                        EC500 State: enabled

    MWI Served User Type: sip-adjunct

                                                    Coverage After Forwarding? s
                                                  Direct IP-IP Audio Connections? y
  Emergency Location Ext: 32007         Always Use? n IP Audio Hairpinning? n
```

3.13. Save Changes

Use the **save translation** command to save all changes.

```
save translation

                                SAVE TRANSLATION

           Command Completion Status                                  Error Code

           Success                                                    0
```

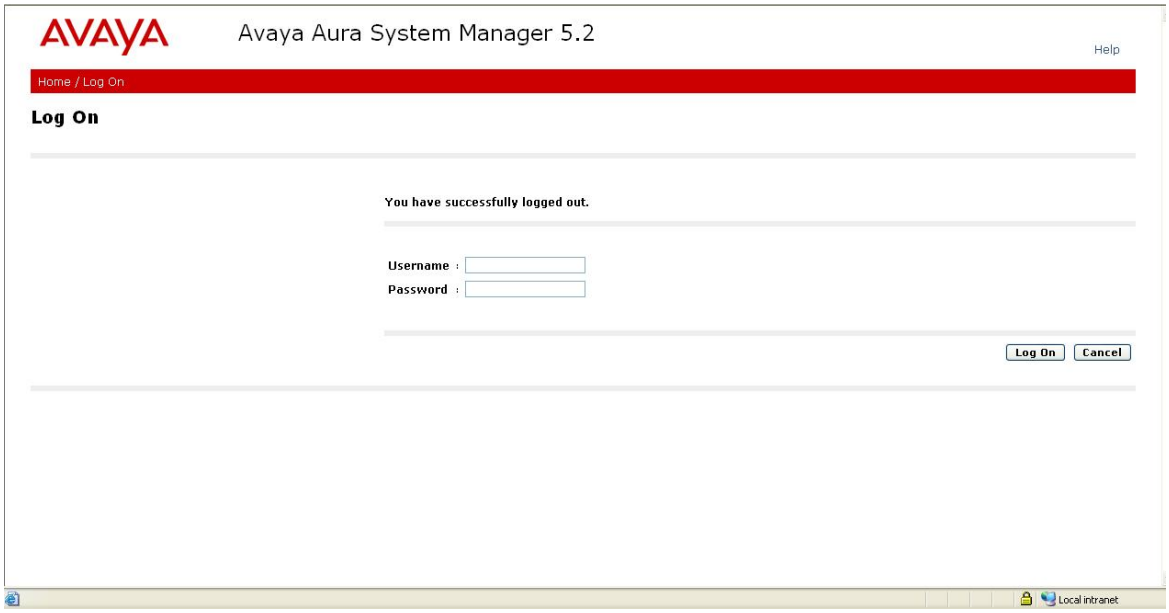
4. Configuring Avaya Aura™ Session Manager

This section provides the procedures for configuring Session Manager. For further information on Session Manager, please consult with references **[1]**, **[2]** and **[3]**. The procedures include the following areas:

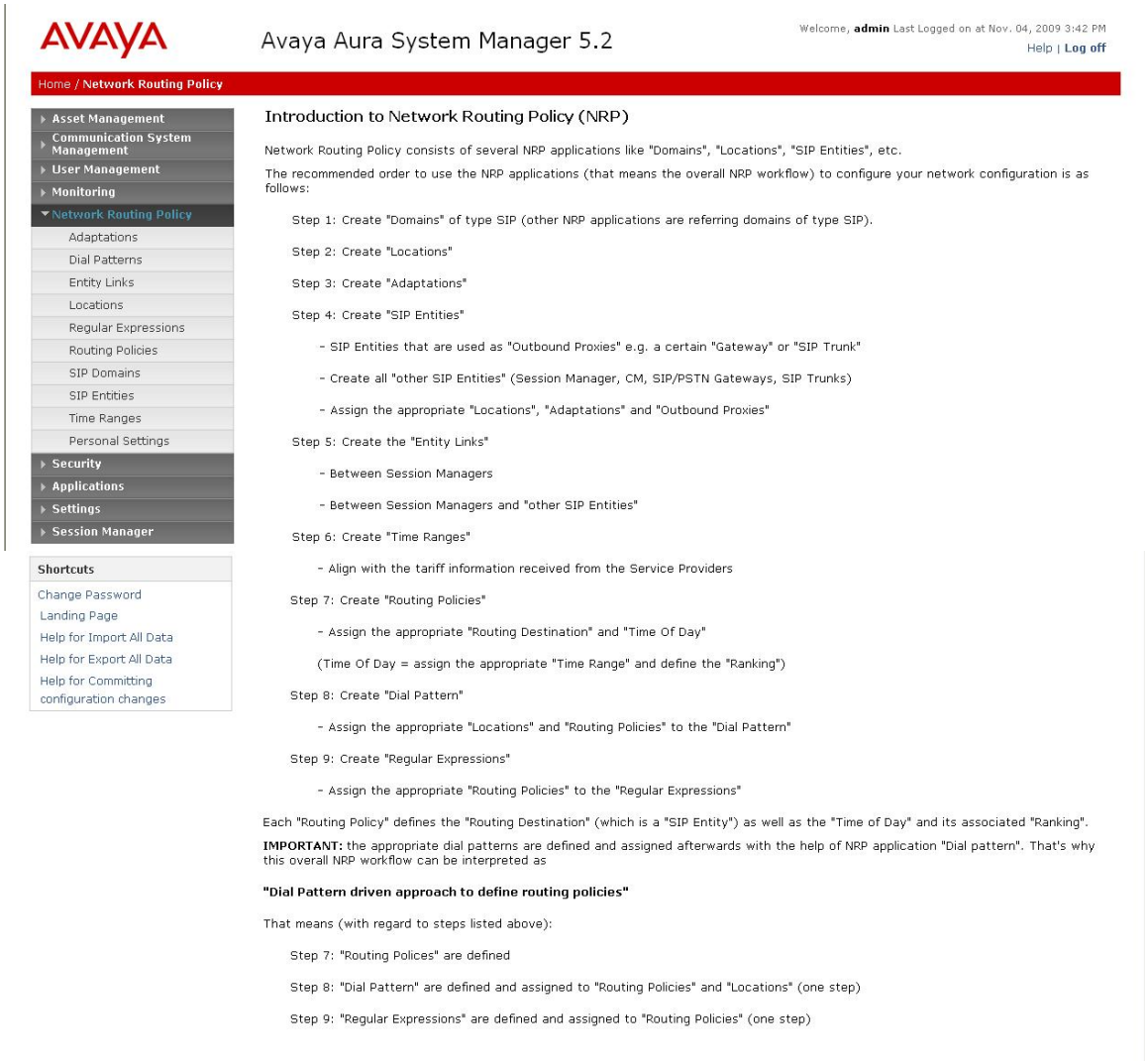
- Log in to Avaya Aura™ Session Manager
- Administer SIP Domain
- Administer Adaptations
- Administer SIP Entities
- Administer Entity Links
- Administer Time Ranges
- Administer Routing Policies
- Administer Dial Patterns
- Administer Regular Expression
- Administer Avaya Aura™ Session Manager
- Add Avaya Aura™ Communication Manager as a Feature Server
- Add Users for SIP Phones

4.1. Log in to Avaya Aura™ Session Manager

Access the Avaya Aura™ System Manager using a Web Browser and entering ***http://<ip-address>/SMGR***, where <ip-address> is the IP address of System Manager. Log in using appropriate credentials and accept the subsequent Copyright Legal Notice.



By selecting **Network Routing Policy** from the left panel menu, a short procedure for configuring Network Routing Policy is shown on the right panel.

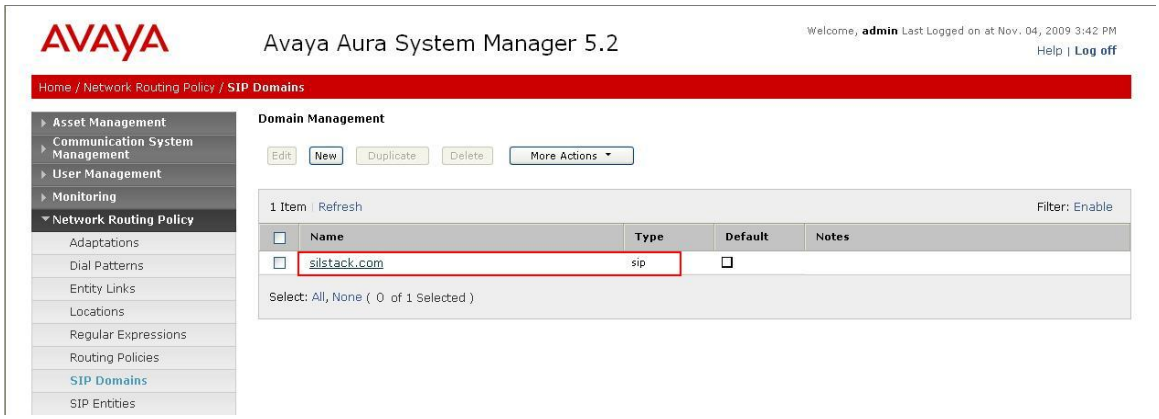


4.2. Administer SIP Domain

Add the SIP domain, for which the communications infrastructure will be authoritative, by selecting **SIP Domains** on the left panel menu and clicking the **New** button (not shown) to create a new SIP domain entry. Complete the following options:

- **Name** The authoritative domain name (e.g., **silstack.com**)
- **Notes** Description for the domain (optional)

Click **Commit** to save changes. Verify the domain is created as in screenshot below.



Note: Since the sample network does not deal with any foreign domains, no additional SIP Domains entry is needed.

4.3. Administer Adaptations

In the sample configuration, multi-site Avaya Modular Messaging represents its subscribers using 11 digit telephone numbers. The 5 digit extension used by Communication Manager is preceded by the 6 digits 120122. DigitConversionAdapter is used in Session Manager to convert between the 5 and 11 digit formats when routing between Modular Messaging and Communication Manager. For the Modular Messaging adaptation (shown on the next page), enter the following information:

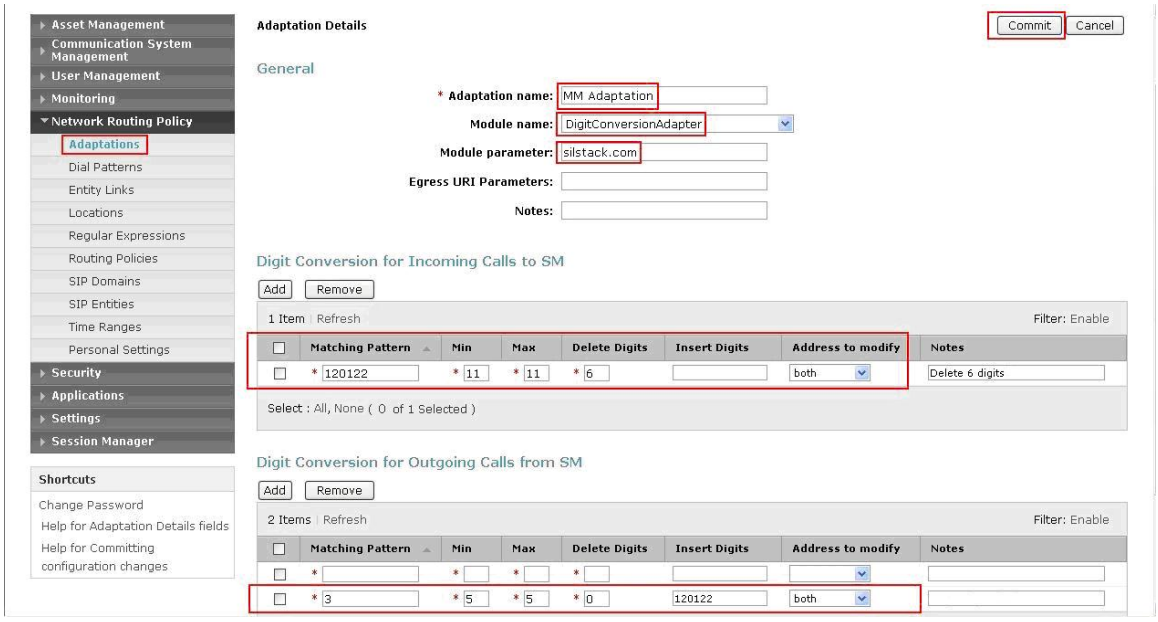
Under **General:**

- **Adaptation Name** An informative name for the adaptation e.g. **Voicemail**
- **Module Name** **DigitConversionAdapter**
- **Module Parameter** The domain name, e.g. **silstack.com**

The remaining fields can be left blank.

Under **Digit Conversion for Incoming Calls to SM** and **Digit Conversion for Outgoing Calls from SM**, click **Add** and then edit the fields in the resulting new row as shown below:

- **Matching Pattern** A Regular expression or partial digit string used to match the incoming dialed number
- **Min** Minimum dialed number length
- **Max** Maximum dialed number length
- **Delete Digits** Number of digits to delete from the beginning
- **Insert Digits** Number of digits to insert at the beginning
- **Address to Modify** Chose between **origination**, **destination** or **both**



Click **Commit** to save.

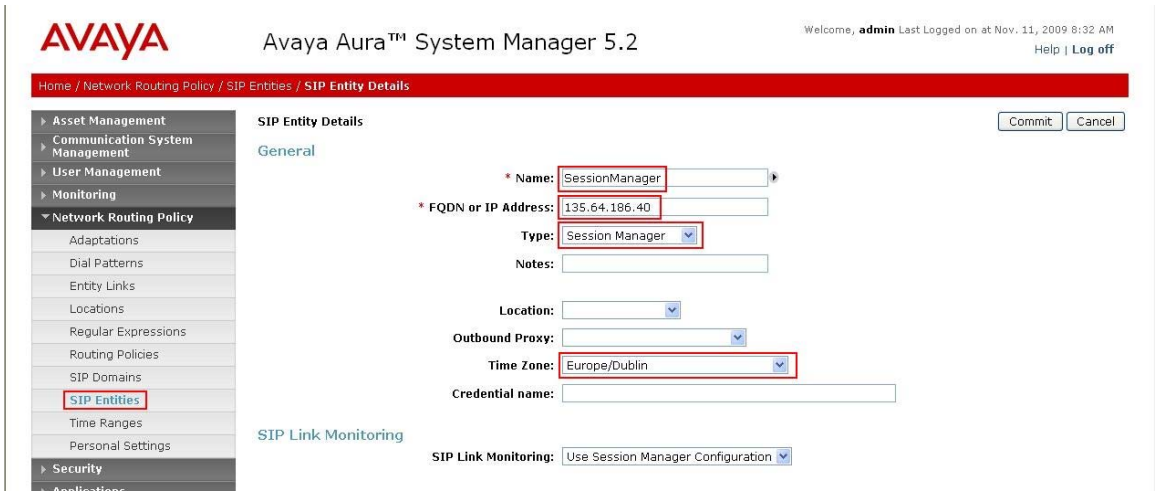
Incoming calls to Communication Manager telephones (SIP INVITE messages) from Modular Messaging that use 11 digit numbers will be converted to the 5 digit form by deleting the first 6 digits. Session Manager will route the call based on the resulting 5 digit extension. Calls routed from Communication Manager to Modular Messaging will have their Request-URI, P-Asserted-Identity, and History-Info headers converted to 11 digit format by insertion of "120122" before being routed to Modular Messaging. See screen below for configuration used.

4.4. Administer SIP Entities

A SIP Entity must be added for Session Manager for each SIP-based telephony system supported by a SIP Trunk. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown). Enter the following for each SIP Entity:

Under **General:**

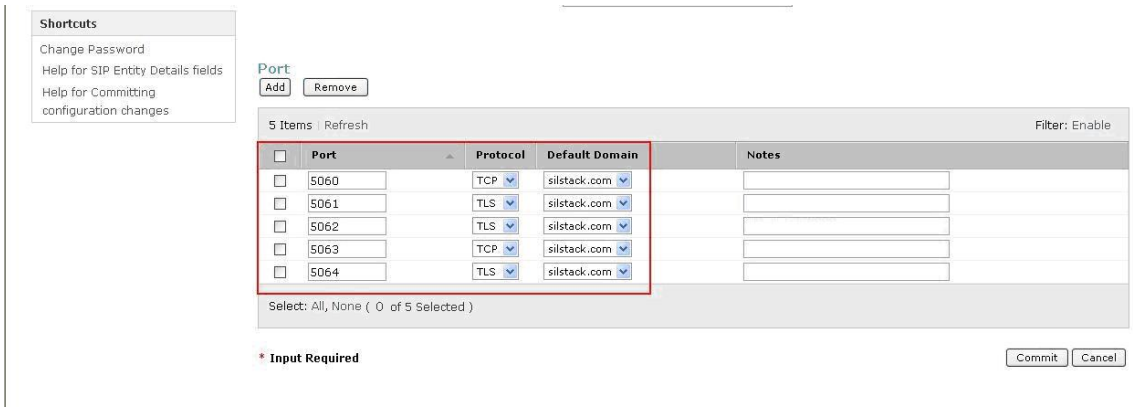
- **Name** — An informative name (e.g., **SessionManager**)
- **FQDN or IP Address** — IP address of the signaling interface on the Session Manager
- **Type** — **Session Manager** for Session Manager, **CM** for Communication Manager, or **Other** for IP Office
- **Time Zone** — Time zone for this location



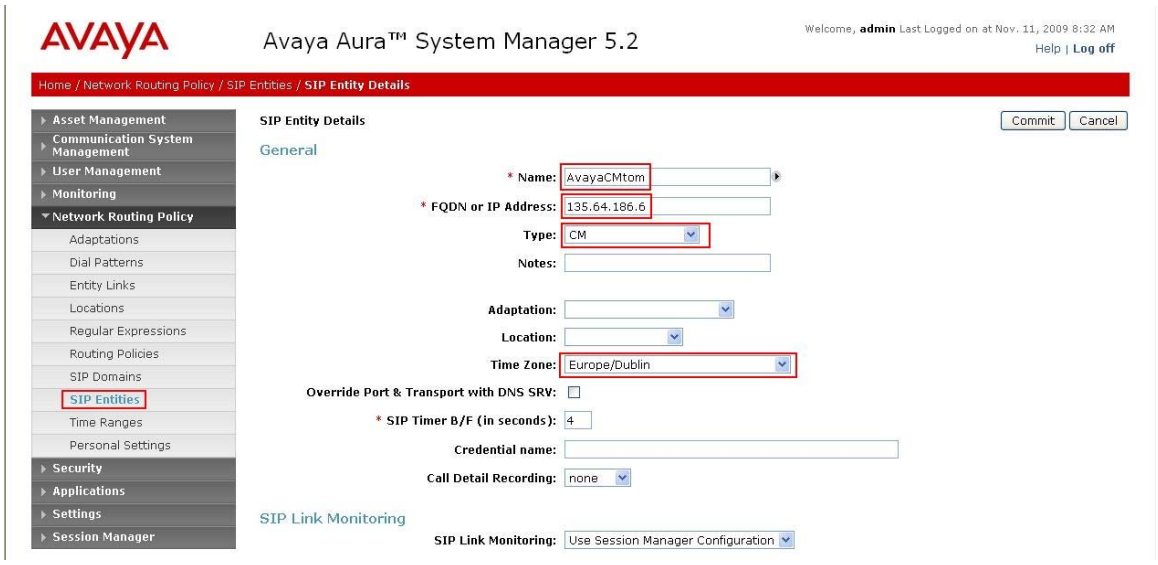
Under **Port**, click **Add,** and then edit the fields in the resulting new row.

- **Port** Port number on which the system listens for SIP requests
- **Protocol** Transport protocol to be used to send SIP requests

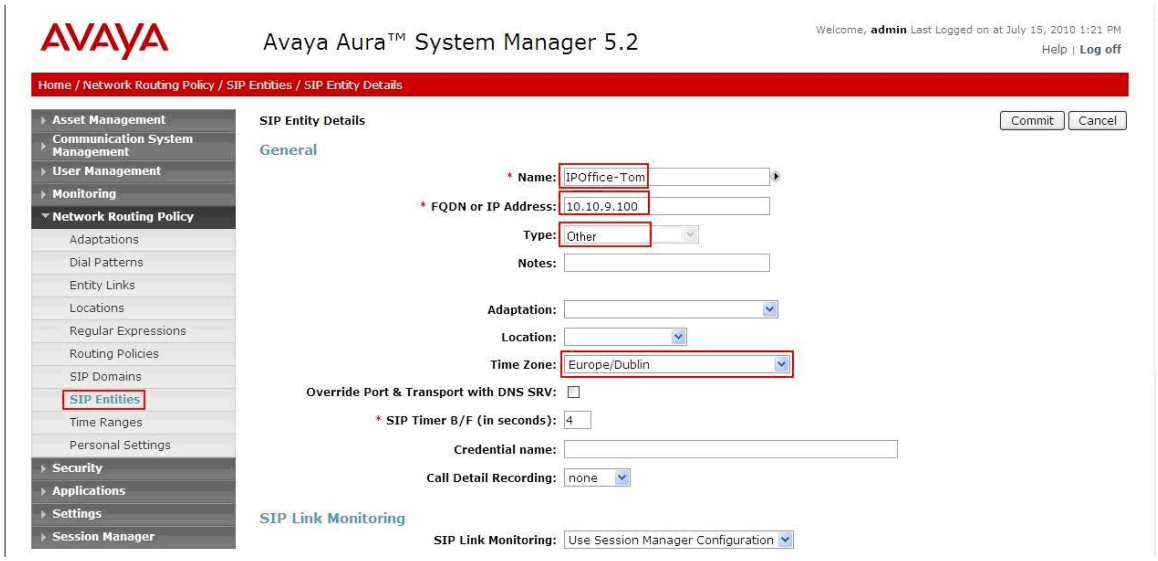
The following screen shows the Port definitions for the Session Manager SIP Entity.



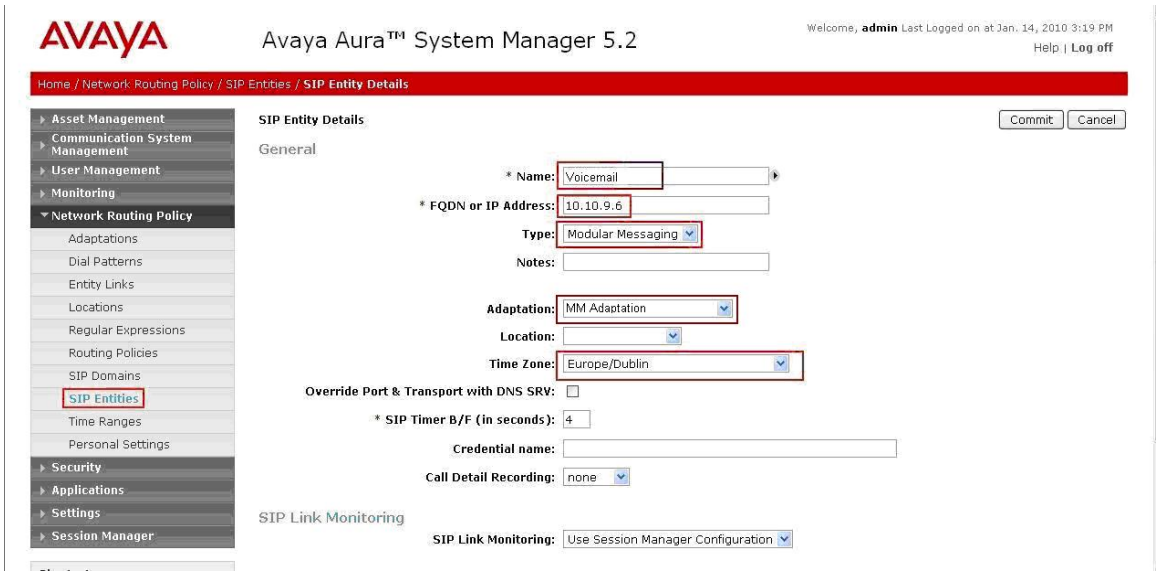
The following screen shows the **SIP Entity Details** for Communication Manager.



The following screen shows the **SIP Entity Details** for IP Office.



The following screen shows the **SIP Entity Details** for Modular Messaging.

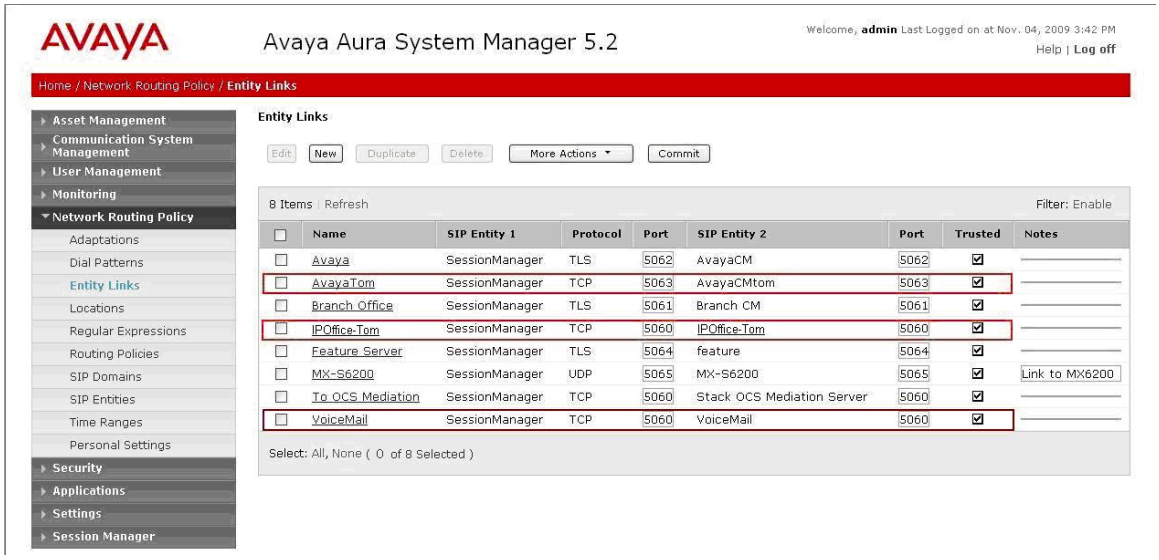


4.5. Administer Entity Links

A SIP trunk between a Session Manager and a telephony system is described by an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button (not shown). Fill in the following fields in the new row that is displayed.

- **Name** An informative name
- **SIP Entity 1** Select **SessionManager**
- **Port** Port number to which the other system sends its SIP requests
- **SIP Entity 2** The other SIP Entity for this link, created in **Section 4.4**
- **Port** Port number to which the other system expects to receive SIP requests
- **Trusted** Whether to trust the other system
- **Protocol** Transport protocol to be used to send SIP requests

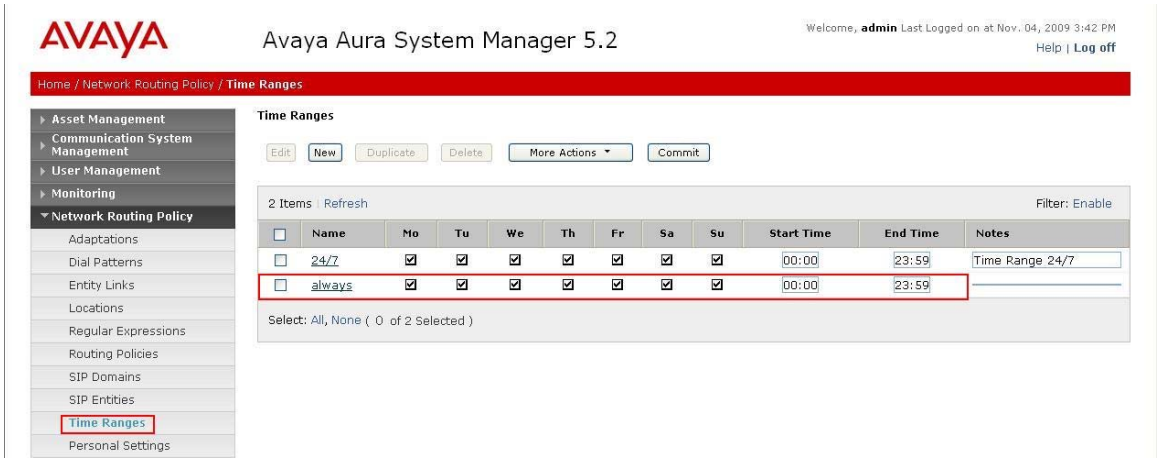
Click **Commit** to save changes. The following screen shows the Entity Links used in the sample network.



4.6. Administer Time Ranges

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

- **Name** An informative name (e.g. **Always**)
- **Mo** through **Su** Check the box under each day of the week for inclusion
- **Start Time** Enter start time (e.g. **00:00** for start of day)
- **End Time** Enter end time (e.g. **23:59** for end of day)



4.7. Administer Routing Policies

Create routing policies to direct how calls will be routed to a system. Two routing policies must be added, one for IP Office and one for Modular Messaging. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown).

Under **General:**

- Enter an informative **Name**

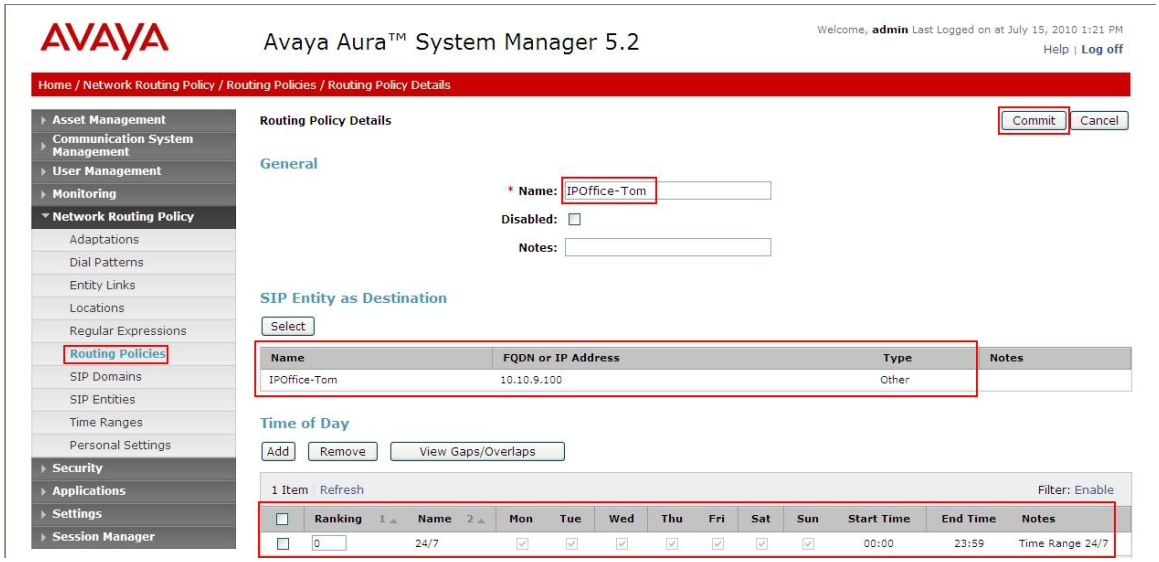
Under **SIP Entity as Destination:**

- Click **Select**, and then select the appropriate SIP entity to which this routing policy applies

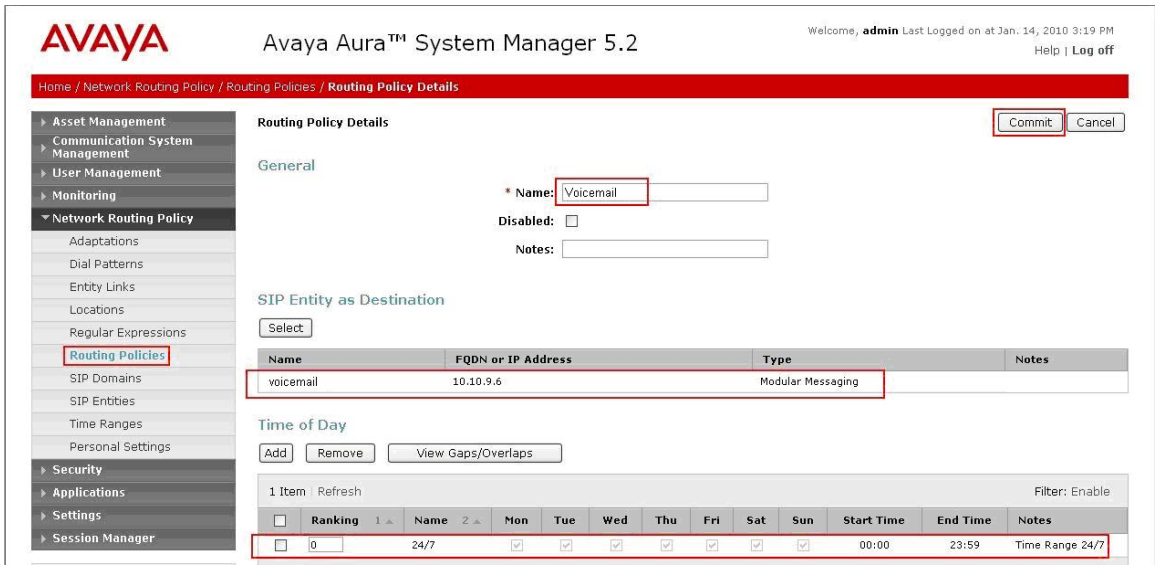
Under **Time of Day:**

- Click **Add**, and then select the time range configured in the previous step.

The following is screen shows the **Routing Policy Details** for IP Office.



The following is screen shows the **Routing Policy Details** for Modular Messaging.

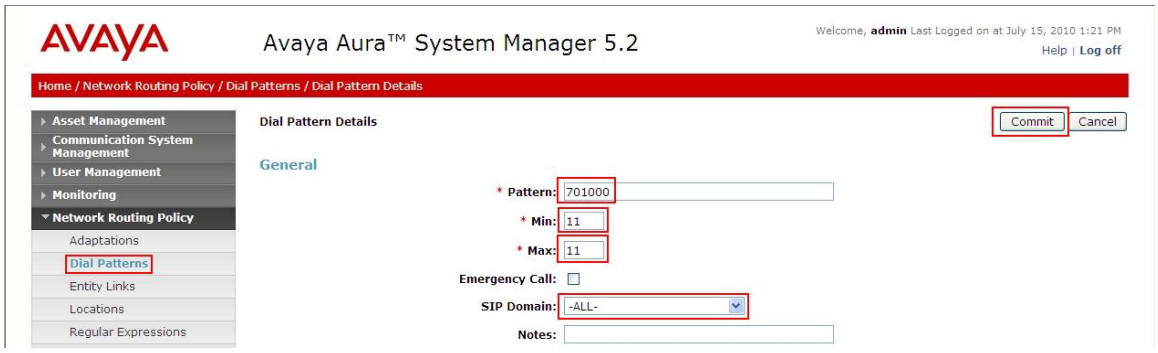


4.8. Administer Dial Patterns

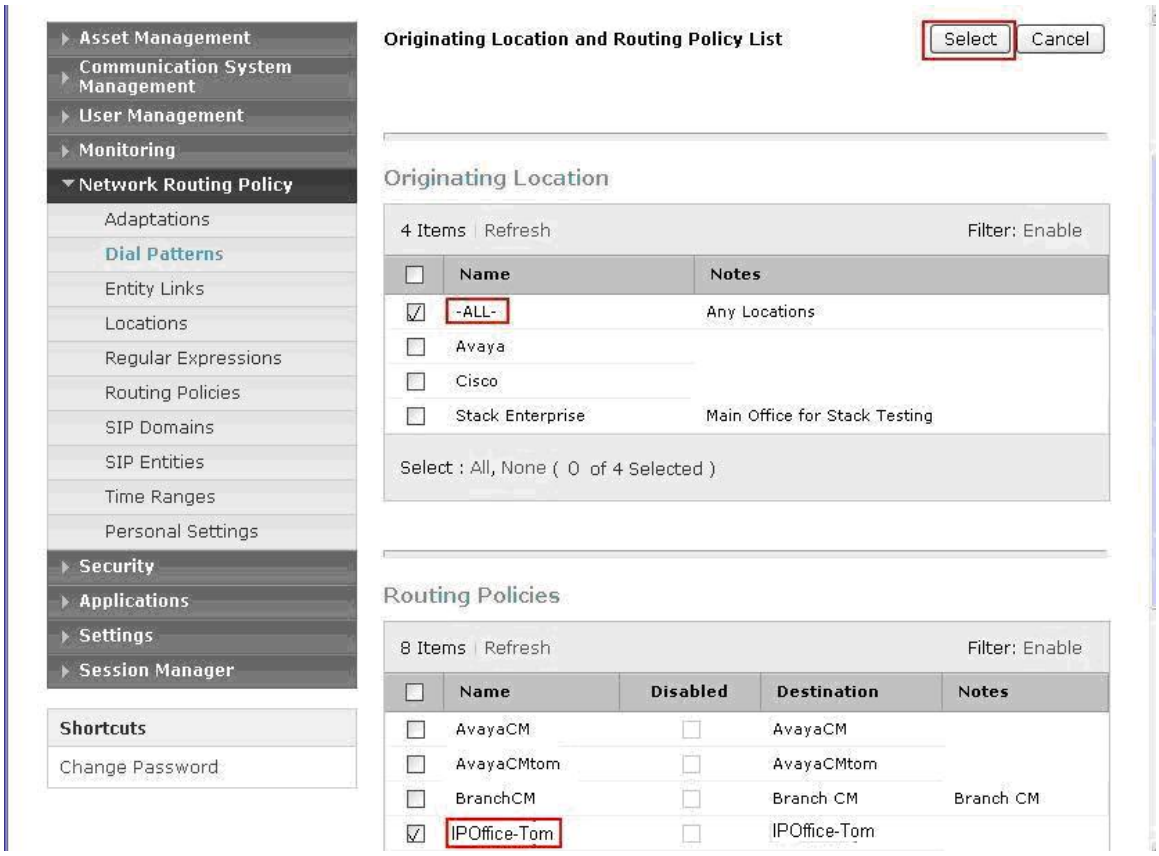
A dial pattern must be defined that will direct calls to the appropriate telephony system. In the sample network, 5-digit extensions beginning with **320** reside on Communication Manager and 11-digit extensions beginning with **701000** reside on IP Office. The 5-digit extension 39999 is for calls to Modular Messaging. To configure IP Office Dial Pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

Under **General:**

- **Pattern** Dialed number or prefix
- **Min** Minimum length of dialed number
- **Max** Maximum length of dialed number
- **Notes** Comment on purpose of dial pattern
- **SIP Domain** Select **ALL**



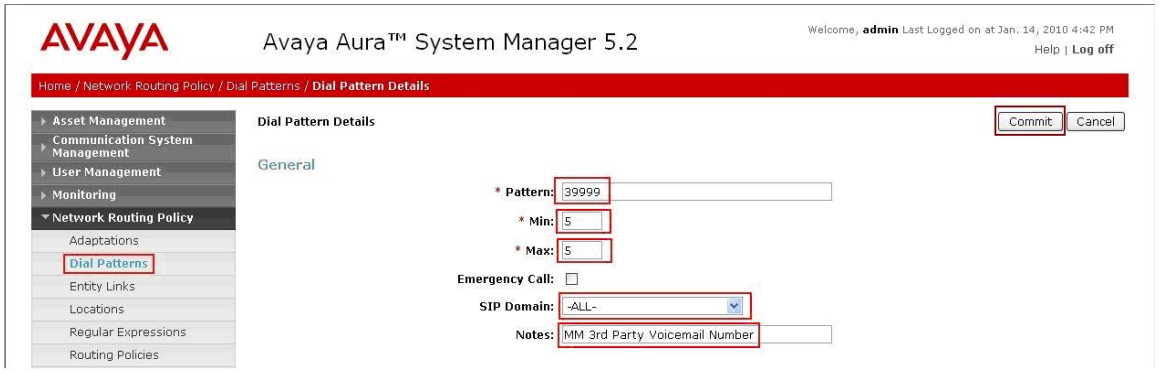
Navigate to **Originating Locations and Routing Policy List** and select **Add** (not shown). Under **Originating Location** select **ALL** and under **Routing Policies** select **IPOffice-Tom.** Click **Select** button to confirm the chosen options and then be returned to the Dial Pattern screen (shown previously), select **Commit** button to save.



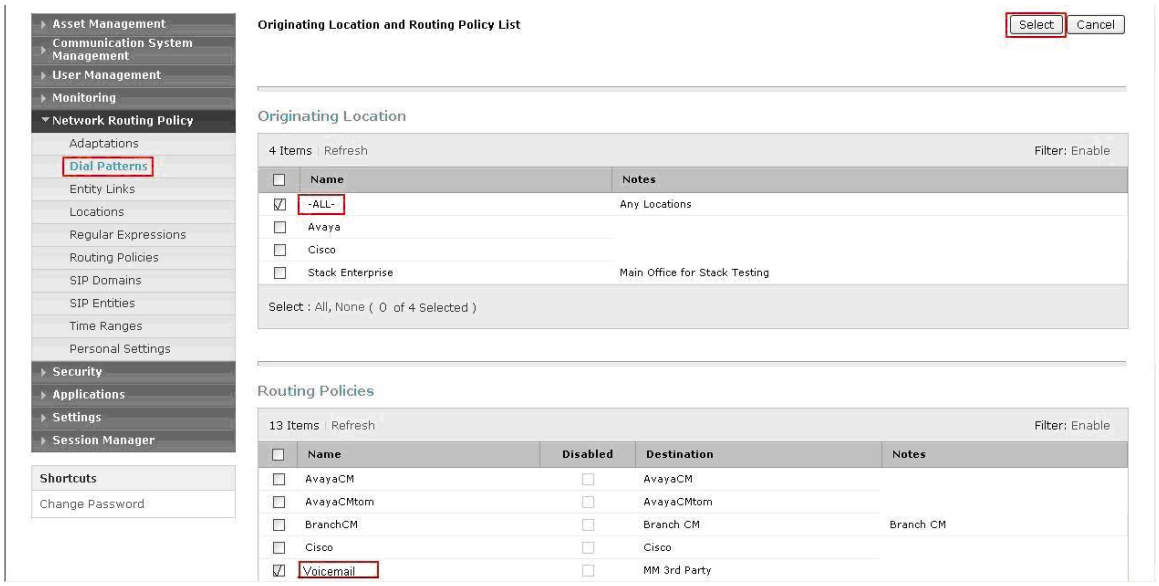
A dial pattern must be defined that will direct calls to the Modular Messaging system. In the sample network, 5-digit extension 39999 will be used as the pilot number for Modular Messaging. For pilot number configuration perform the following. Select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

Under **General:**

- **Pattern** Pilot number
- **Min** Minimum length of pilot number
- **Max** Maximum length of pilot number
- **Notes** Comment on purpose of dial pattern
- **SIP Domain** Select **ALL**



Navigate to **Originating Locations and Routing Policy List** and select **Add** (not shown). Under **Originating Location** select all locations by checking the box next to **ALL** and under **Routing Policies** select the Routing Policy created in **Section 4.7**. Click **Select** button to confirm the chosen options and then be returned to the Dial Pattern screen (shown above), select **Commit** button to save.

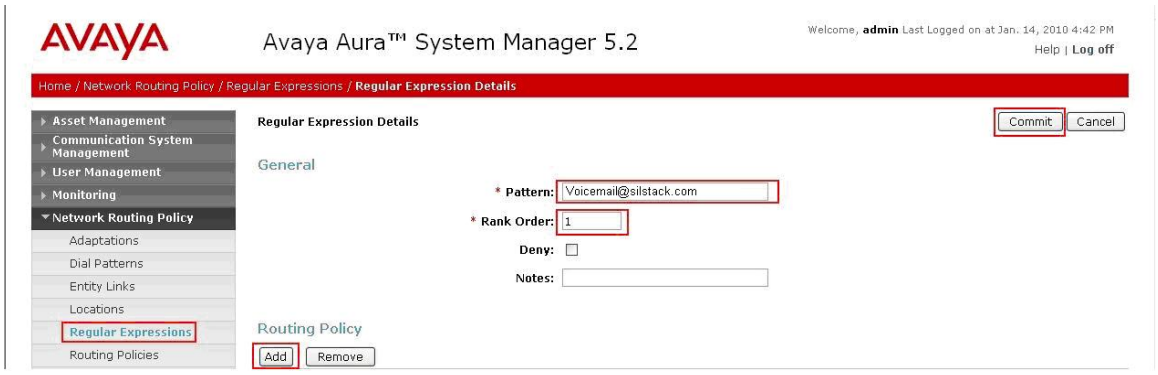


4.9. Administer Regular Expression

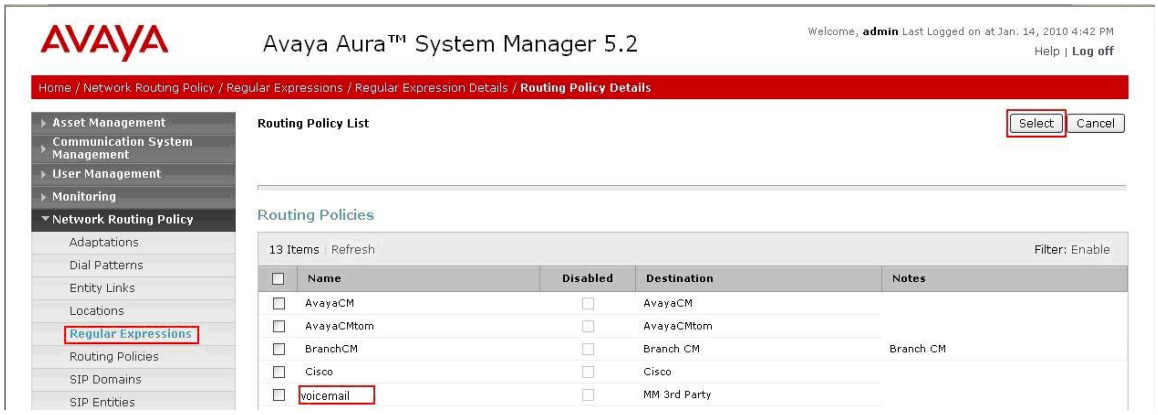
A Regular Expression must be defined for Communication Manager MAS subscribers so that they can communicate with Modular Messaging via Session Manager. The format of the Regular Expression is **voicemail handle**@domain name, where voicemail handle is the handle defined in **Section 3.10**. Select **Regular Expressions** on the left panel menu and then click on the **New** button (not shown).

Under **General:**

- **Pattern** **Voicemail Handle**@Domain Name
- **Rank Order** A unique number
- **Notes** Comment on purpose of Regular Expression



Navigate to **Routing Policies** and select **Add**. Under **Routing Policies** select the routing policy created in **Section 4.7**. Click **Select** button to confirm the chosen option and then be returned to the **Regular Expression Details** screen (shown above), select **Commit** button to save.



4.10. Administer Avaya Aura™ Session Manager

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

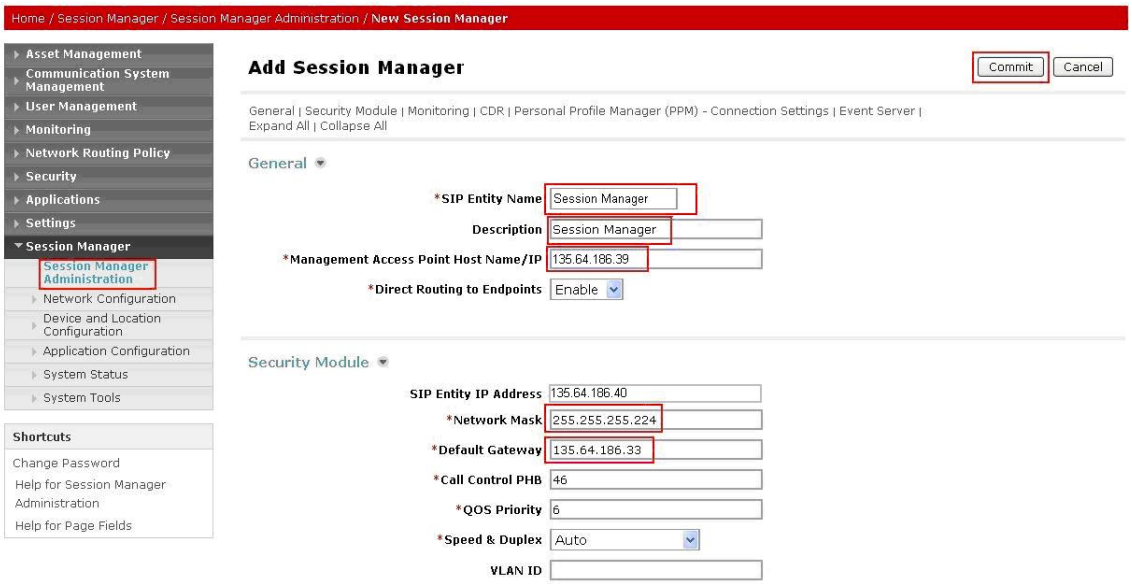
Under **General:**

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

Under **Security Module:**

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

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



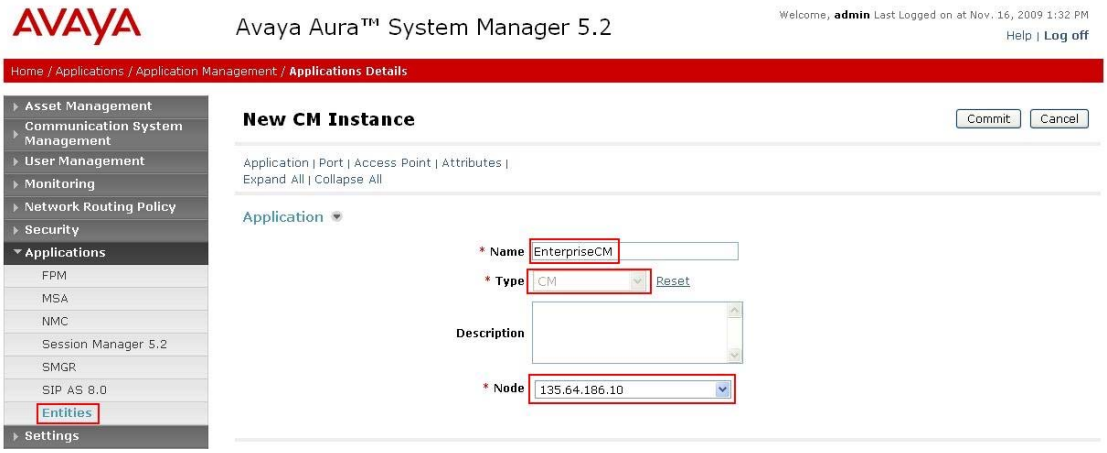
4.11. Add Avaya Aura™ Communication Manager as a Feature Server

In order for Communication Manager to provide configuration and Feature Server support to SIP phones when they register to Session Manager, Communication Manager must be added as an application.

4.11.1. Create an Application Entity

Select **Applications → Entities** on the left. Click on **New** (not shown). Enter the following fields and use defaults for the remaining fields:

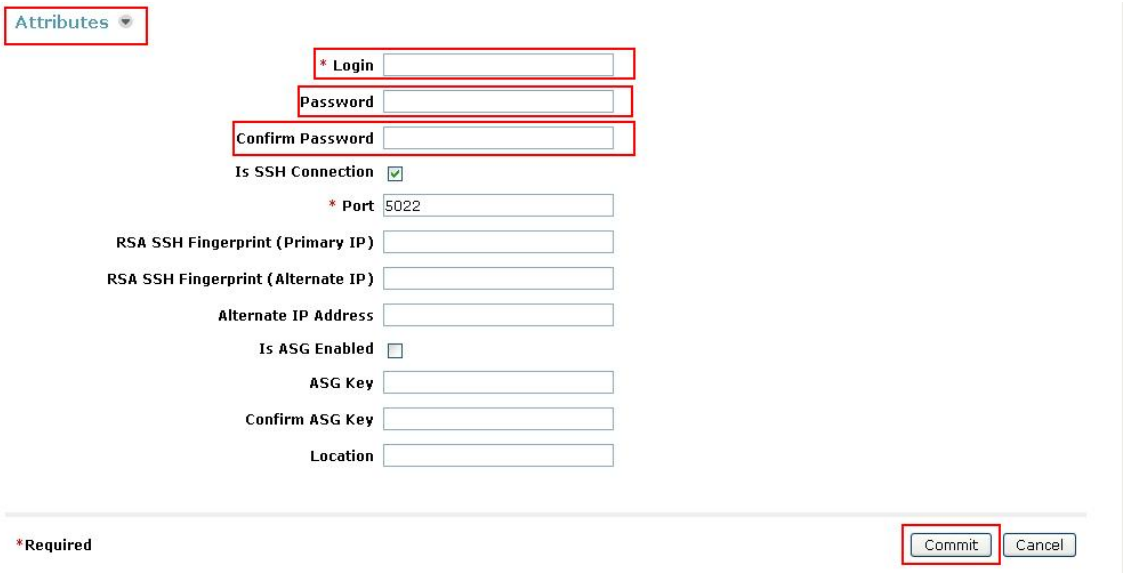
- **Name** A descriptive name
- **Type** Select **CM**
- **Node** Select **Other** and enter the IP address for CM SAT access



Navigate to the **Attributes** section and enter the following:

- **Login** Login used for SAT access
- **Password** Password used for SAT access
- **Confirm Password** Password used for SAT access

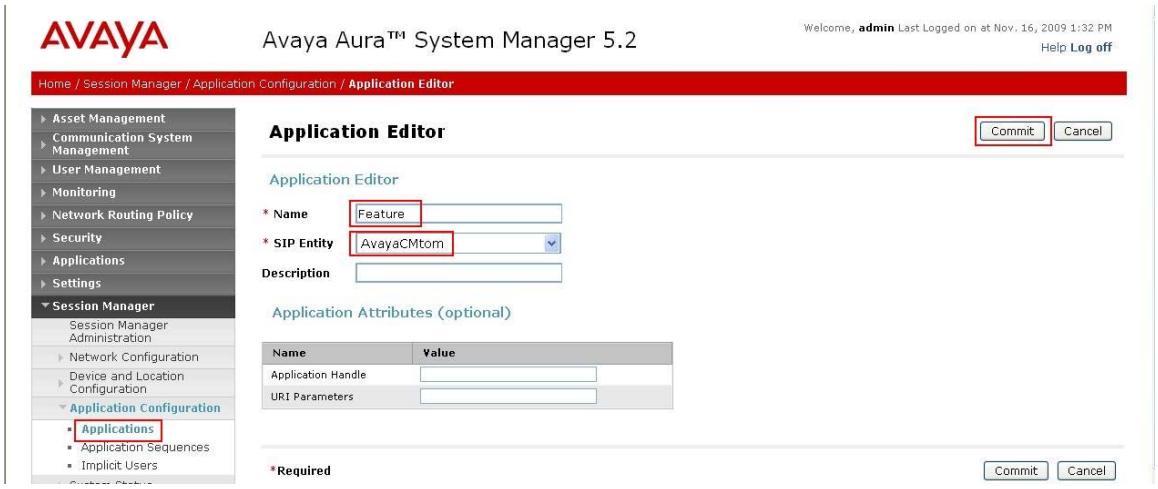
Click on **Commit** to save.



4.11.2. Create a Feature Server Application

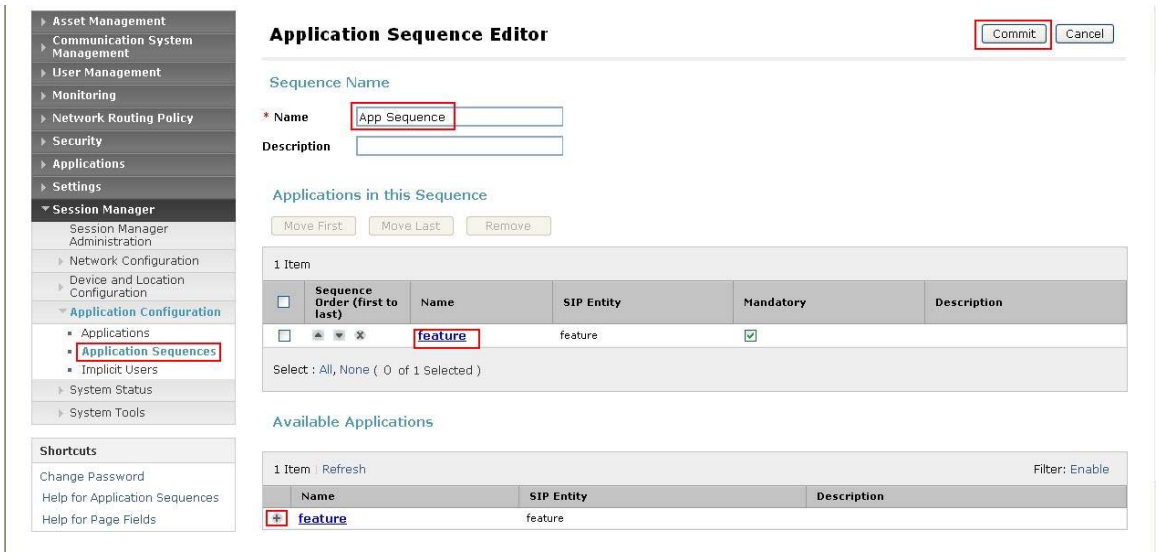
Select **Session Manger → Application Configuration → Applications** on the left. Click on **New** (not shown). Enter following fields and use defaults for the remaining fields and click on **Commit** to save.

- **Name** A descriptive name
- **SIP Entity** Select the CM SIP Entity defined in **Section 4.4**



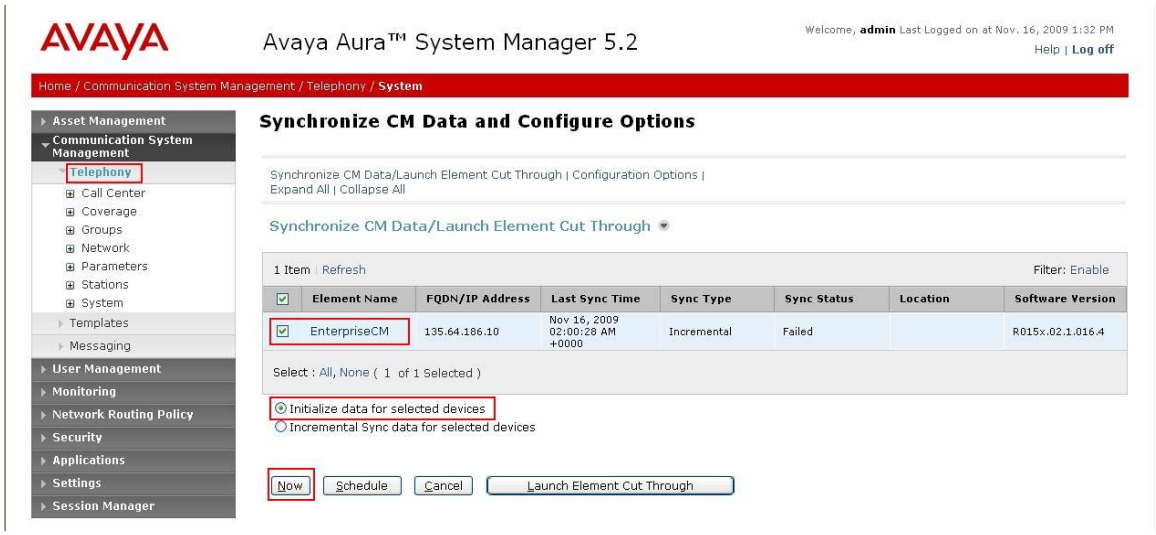
4.11.3. Create a Feature Server Application Sequence

Select **Session Manager → Application Configuration → Application Sequences** on the left. Click on **New** (not shown). Enter a descriptive **Name**. Click on the + sign next to the appropriate **Available Applications** and they will move up to the **Applications in this Sequence** section. Click on **Commit** to save.



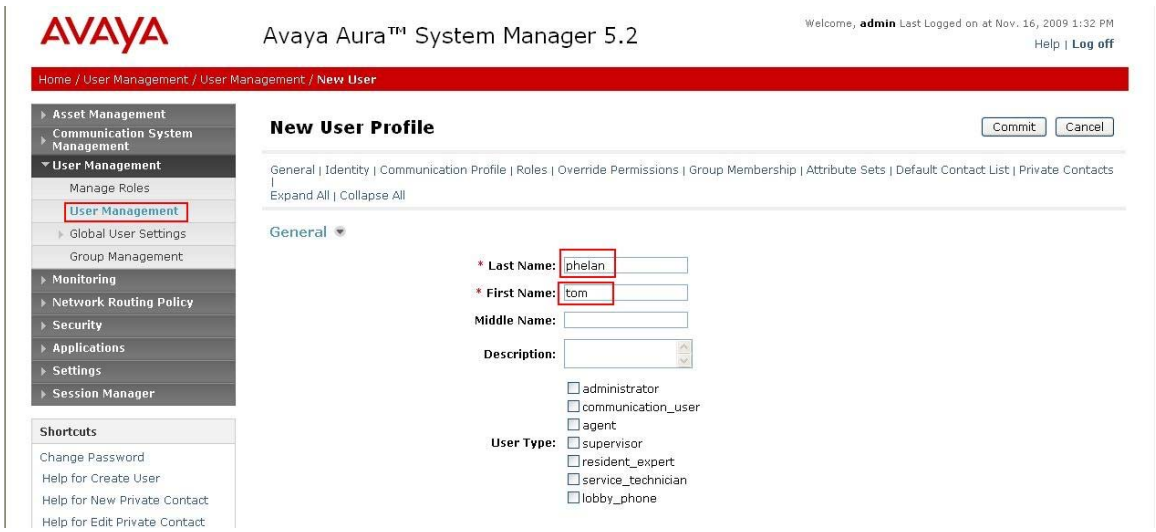
4.11.4. Synchronize Avaya Aura™ Communication Manager Data

Select **Communications System Management → Telephony** on the left. Select the appropriate **Element Name**. Select **Initialize data for selected devices**. Then click on **Now**. This may take some time. Use the menus on the left under **Monitoring → Scheduler** to determine when the task is complete.



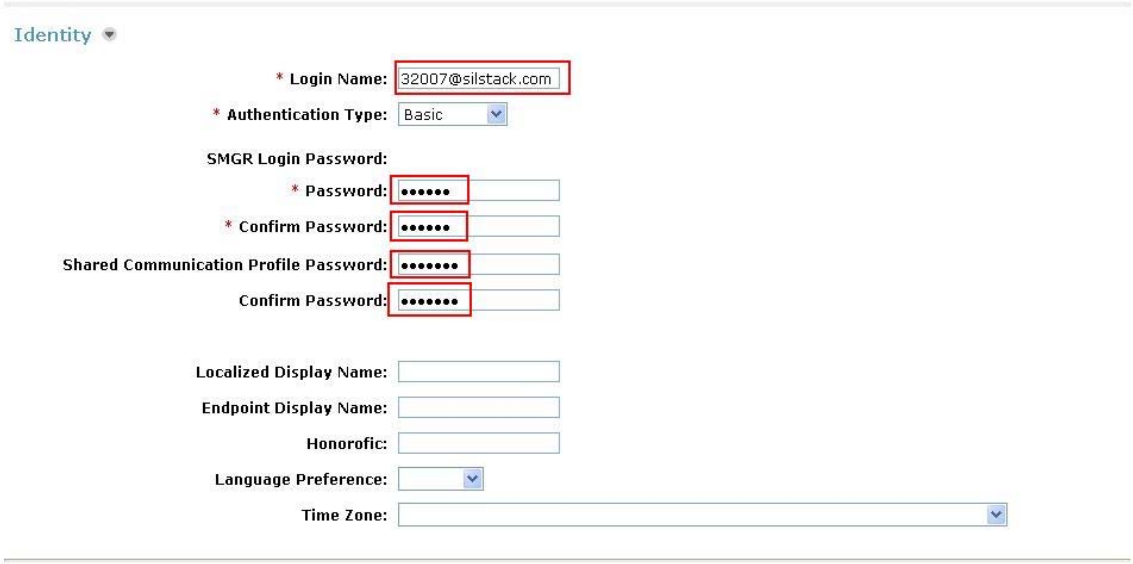
4.12. Add Users for SIP Phones

Users must be added via Session Manager and the details will be updated on the CM. Select **User Management → User Management** on the left. Then click on **New** (not shown). Enter a **First Name** and **Last Name**.



Navigate to the **Identity** section and enter the following and use defaults for other fields:

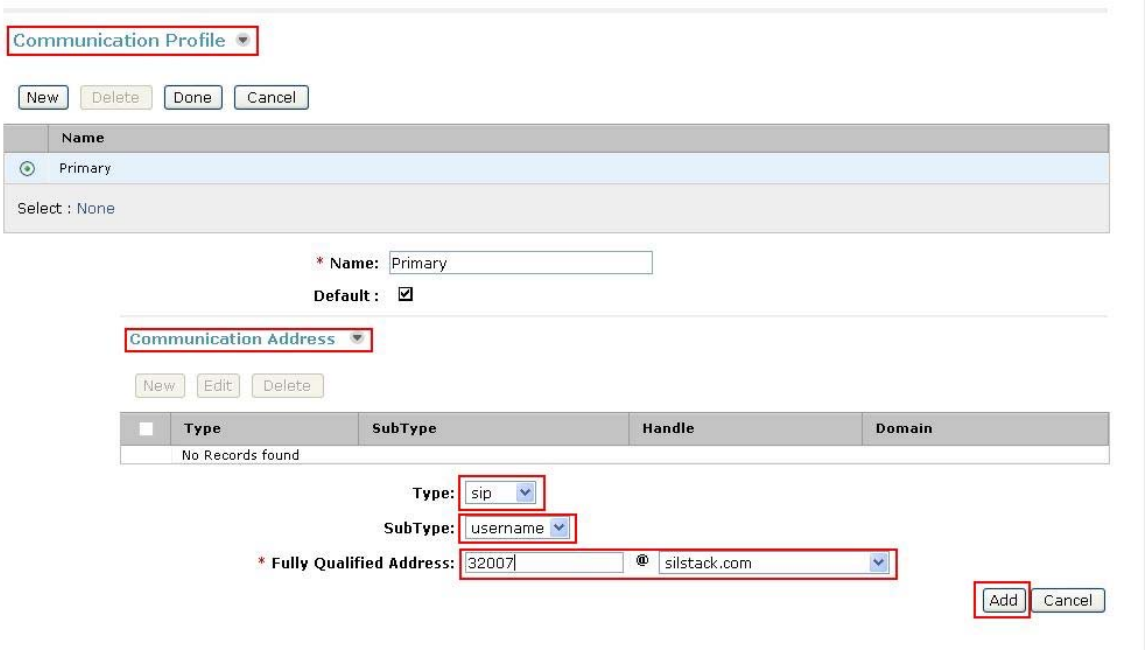
- **Login Name** The desired phone extension number@domain.com where domain was defined in **Section 4.2**
- **Password** Password for user to log into System Manager (SMGR)
- **Shared Communication Profile Password** Password to be entered by the user when logging into the phone.



Navigate to and click on **Communication Profile** section to expand. Then click on **Communication Address** to expand that section. Enter the following and defaults for the remaining fields:

- **Type** Select **SIP**
- **SubType** Select **Username**
- **Fully Qualified Address** Enter the extension number

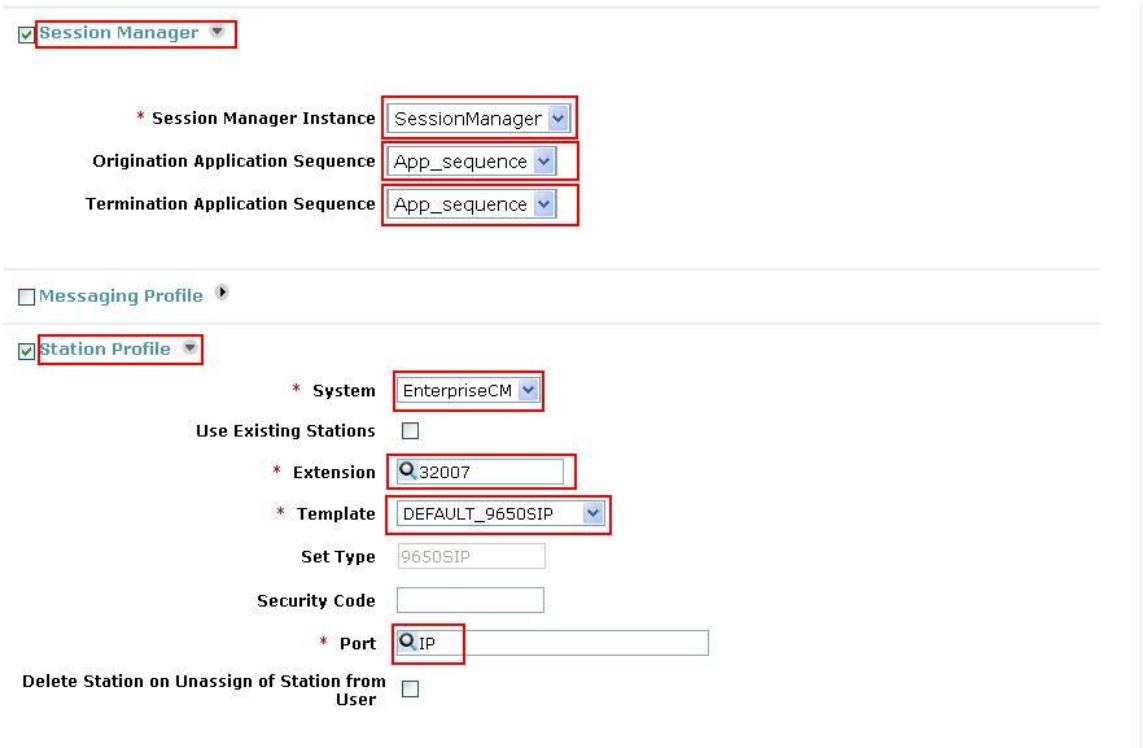
Click on **Add**.



Navigate to and click on **Session Manager** section to expand. Select the appropriate Session Manager server for **Session Manager Instance**. For **Origination Application Sequence** and **Termination Application Sequence** select the application sequence created in **Section 4.11.3**. Click on **Station Profile** to expand that section. Enter the following fields and use defaults for the remaining fields:

- **System** Select the CM Entity
- **Extension** Enter a desired extension number
- **Template** Select a telephone type template
- **Port** Select **IP**

Click on **Commit** to save (not shown).



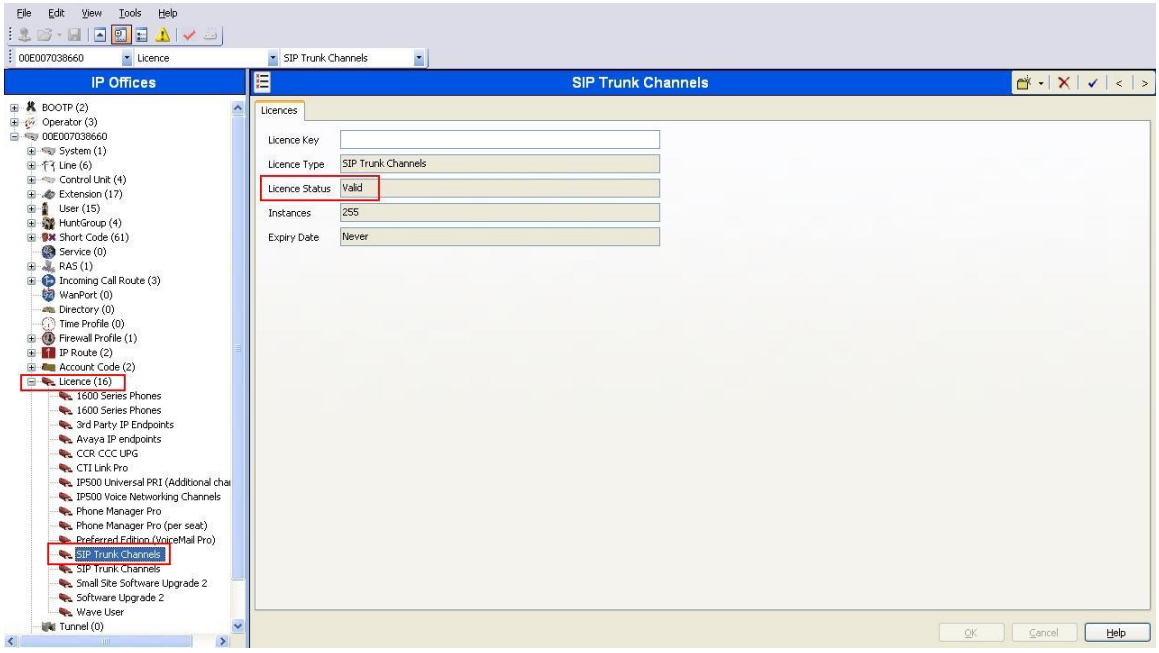
5. Configure Avaya IP Office

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

- Verify Avaya IP Office License
- Obtain LAN IP Address
- Administer Network Topology
- Administer SIP Registrar
- Administer Codec Preference
- Administer SIP Trunk to Avaya Aura™ Session Manager
- Administer Voicemail
- Administer Branch Prefix
- Administer Short Codes
- Administer Voicemail on End Users
- Save Configuration

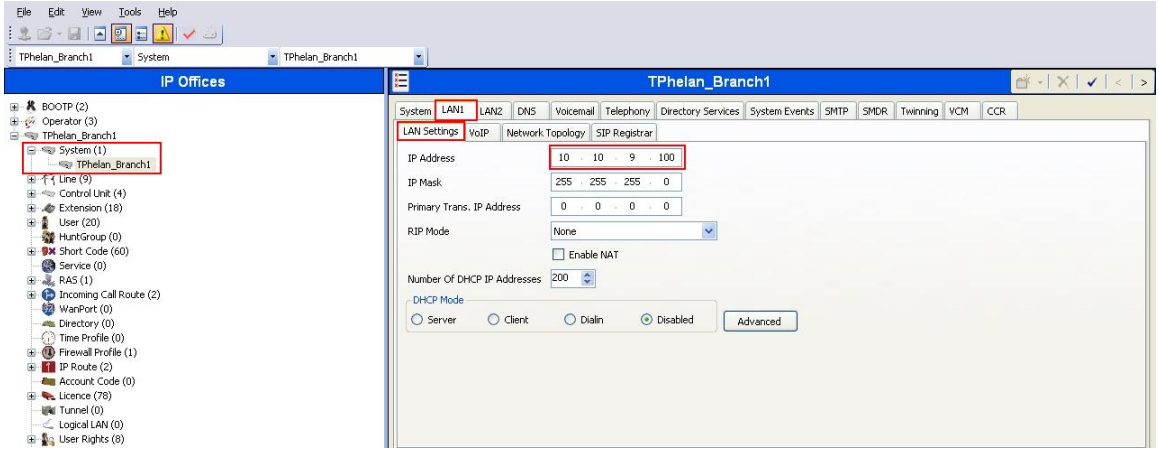
5.1. Verify Avaya IP Office License

From a PC running the Avaya IP Office Manager application, select **Start → Programs → IPOffice → 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** and if not contact your Avaya representative.



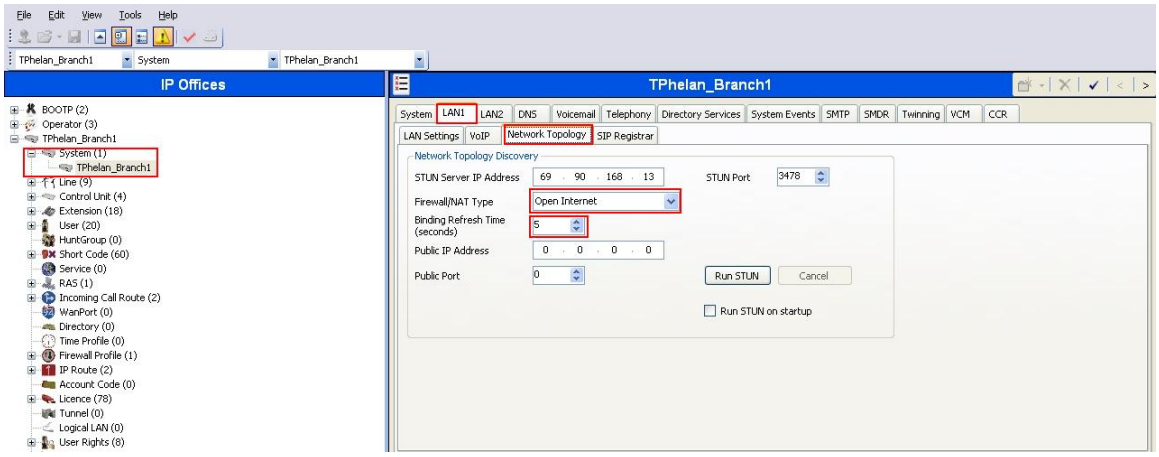
5.2. Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the system screen in the right pane. Select the **LAN1** tab, followed by the **LAN Settings** sub-tab. The **IP address** will be the one defined for the IP Office SIP Entity in **Section 4.4** Note that IP Office can support SIP trunks on the LAN1 and/or LAN2 interfaces, and the sample configuration used the LAN1 interface.



5.3. Administer Network Topology

From the configuration tree in the left pane, select **System** to display the system screen in the right pane. Select the **LAN1** tab, followed by the **Network Topology** sub-tab. Configure **Firewall/NAT Type** to **Open Internet** and **Binding Refresh Time** to **5**. Click **OK** (not shown).

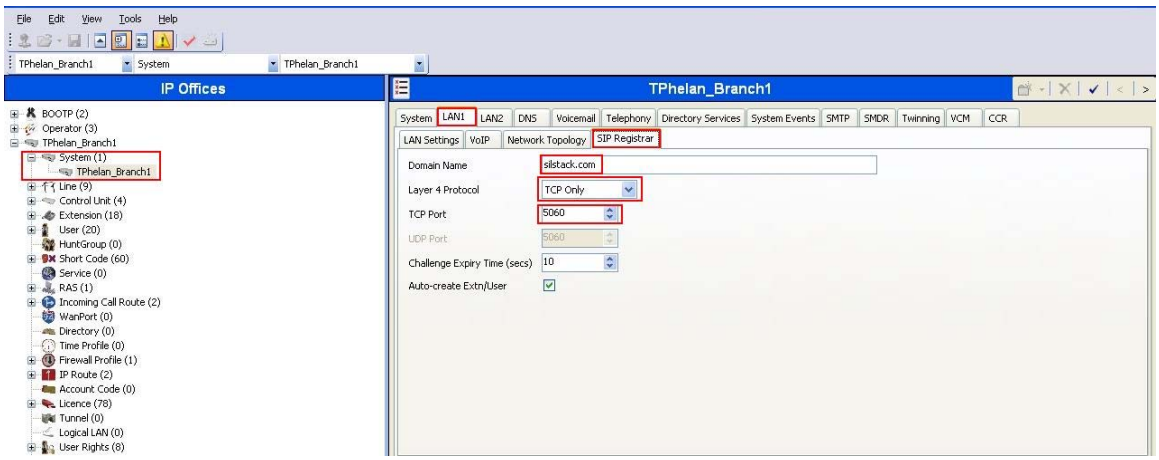


5.4. Administer SIP Registrar

From the configuration tree in the left pane, select **System** to display the system screen in the right pane. Select the **LAN1** tab, followed by the **SIP Registrar** sub-tab in the right pane and enter following values:

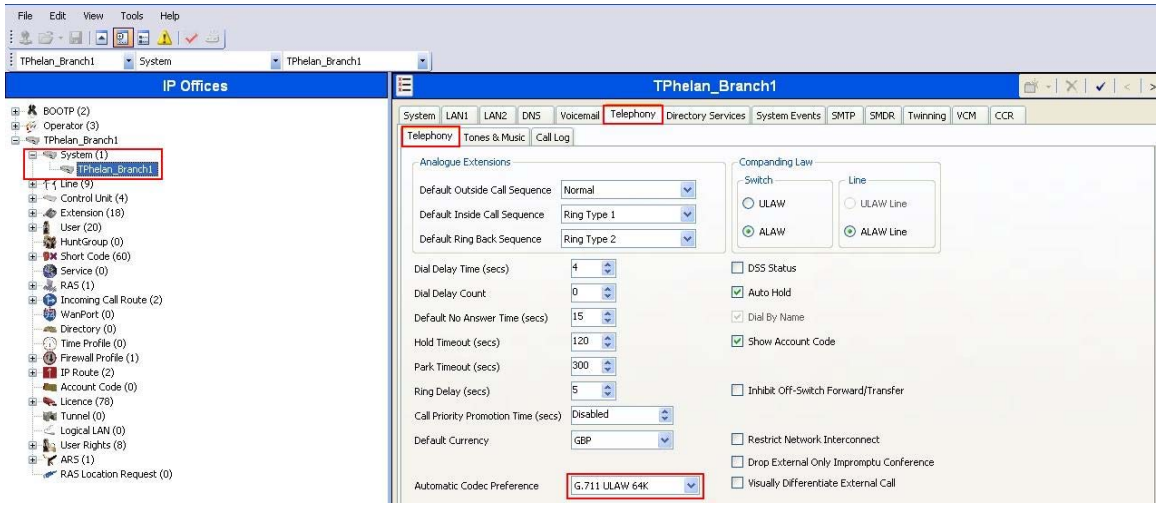
- **Domain Name** Enter a valid Domain Name
- **Layer 4 Protocol** Select **TCP only**
- **TCP Port** Select **5060**

Click **OK** (not shown).



5.5. Administer Codec Preference

From the configuration tree in the left pane, select **System** to display the system screen in the right pane. Select the **Telephony** tab followed by the **Telephony** sub-tab in the right pane. Configure **Automatic Codec Preference** to **G.711 ULAW 64K**. Click **OK** (not shown).

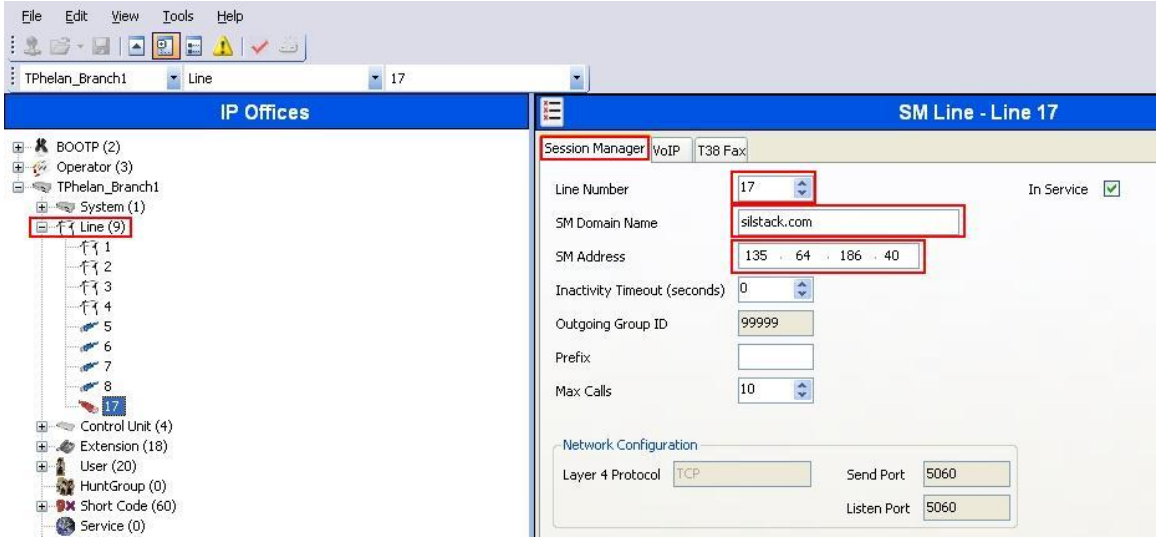


5.6. Administer SIP Trunk to Avaya Aura™ Session Manager

From the configuration tree in the left pane, right-click on **Line** and select **New → SM Line** to add a new SIP Trunk towards Session Manager. Select the **Session Manager** tab and enter the following values:

- **Line Number** Select a unique Line Number
- **SM Domain** Enter a Domain Name
- **SM Address** Enter the IP address for SM-100 card

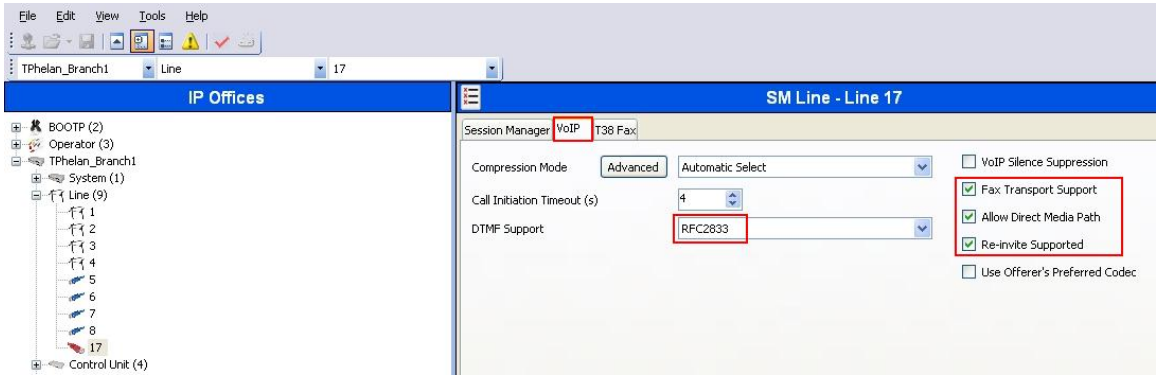
Retain default values for all other fields. Click **OK** (not shown).



Select the **VoIP** tab and enter the following values:

- **DTMF Support** **RFC2833**
- **Fax Transport Support** Tick the box
- **Allow Direct Media Path** Tick the box
- **Re-invite Supported** Tick the box

Retain default values for all other fields. Click **OK** (not shown).

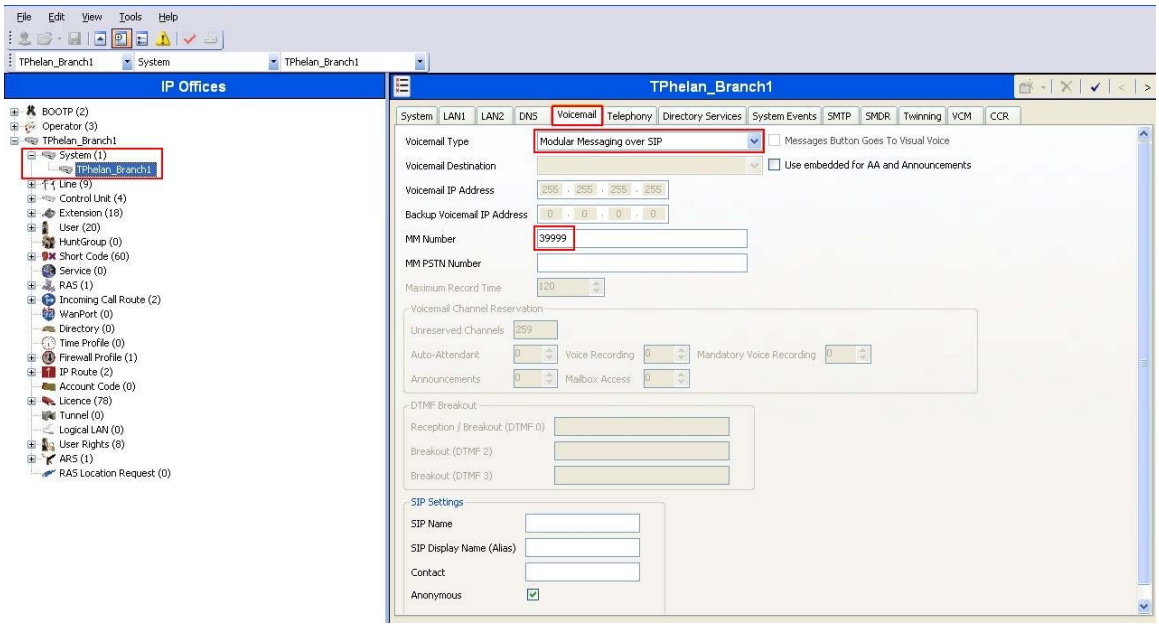


5.7. Administer Voicemail

From the configuration tree in the left pane, select **System** to display the system screen in the right pane. Select the **Voicemail** tab and enter the following values:

- **Voicemail Type** **Modular Messaging over SIP**
- **MM Number** Enter the MM Pilot number

Retain default values for all other fields. Click **OK** (not shown).

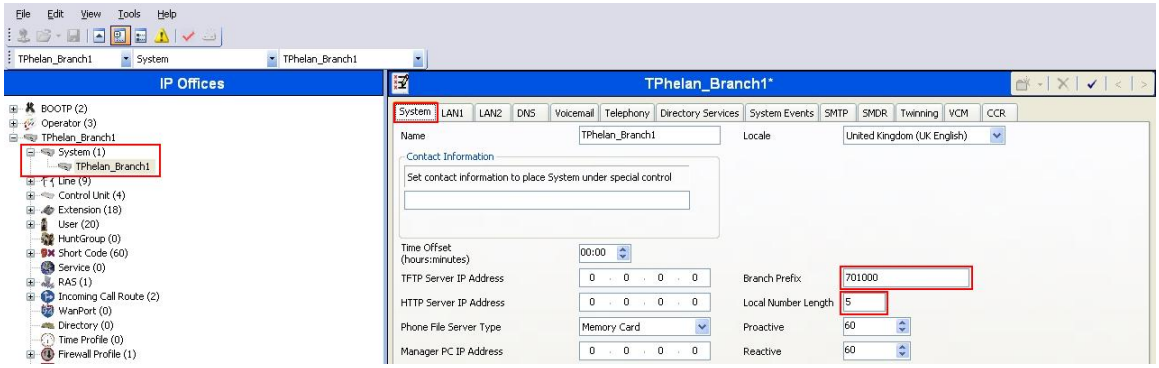


5.8. Administer Branch Prefix

From the configuration tree in the left pane, select **System** to display the system screen in the right pane. Select the **System** tab and enter the following values:

- **Branch Prefix** Enter a desired Branch Prefix
- **Local Number Length** Enter a desired Local Number Length

Retain default values for all other fields. Click **OK** (not shown).

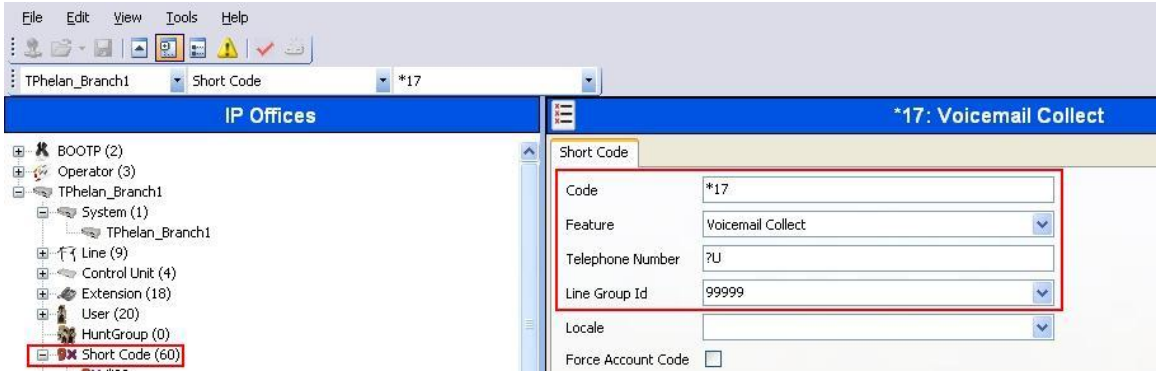


5.9. Administer Short Codes

From the configuration tree in the left pane, right-click on **Short Code**, and select **New**. Enter the following details to define a short code to access Voicemail:

- **Code** Enter a dialing string that will be used to dial Voicemail
- **Feature** Select **Voicemail Collect**
- **Telephone Number** Enter **?U**
- **Line Group ID** Select **Outgoing Group ID** from **Section 5.6**

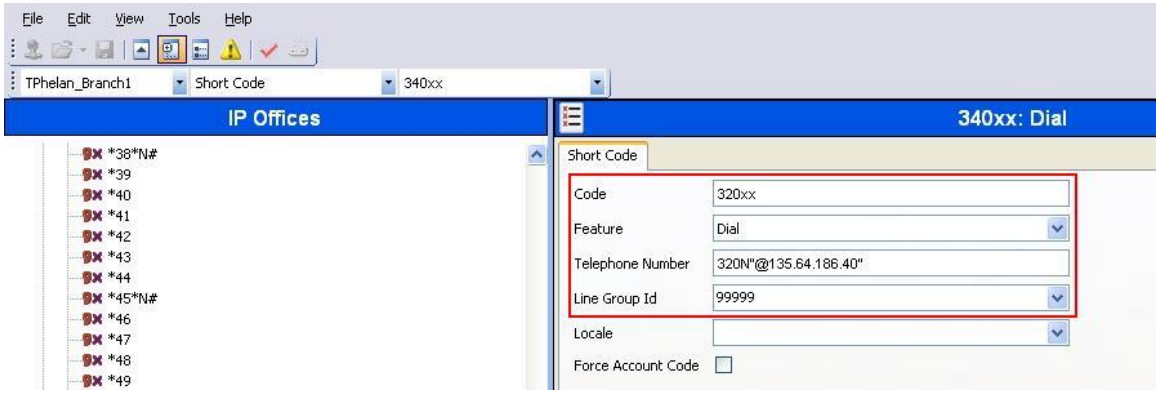
Retain default values for all other fields. Click **OK** (not shown).



Add another Short Code to dial Communication Manager extensions using the following details:

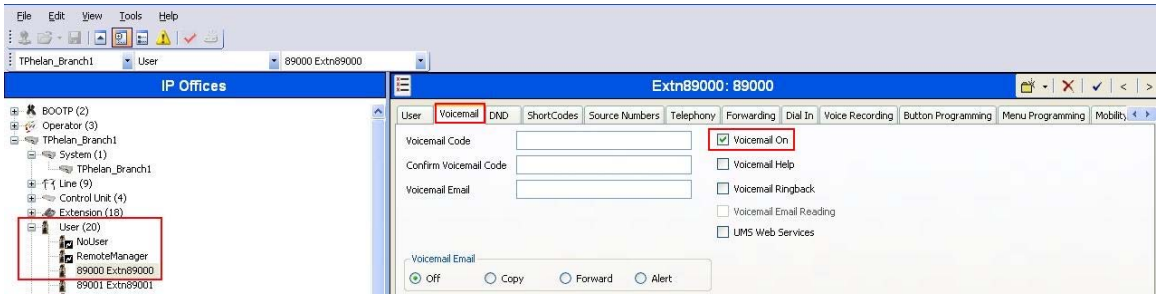
- **Code** Enter a dialing string used to call CM phones
- **Feature** Select **Dial**
- **Telephone Number** Enter the phone number appended with "**@<ip-address of SM-100 card>**"
- **Line Group ID** Select **Outgoing Group ID** from **Section 5.6**

Retain default values for all other fields. Click **OK** (not shown).



5.10. Administer Voicemail on End Users

From the configuration tree in the left pane, select **User**. Select a user and in the right-pane under the **Voicemail** tab, tick the box next to **Voicemail On**. Click **OK** to save (not shown).



5.11. Save Configuration

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

6. Configure Avaya Modular Messaging

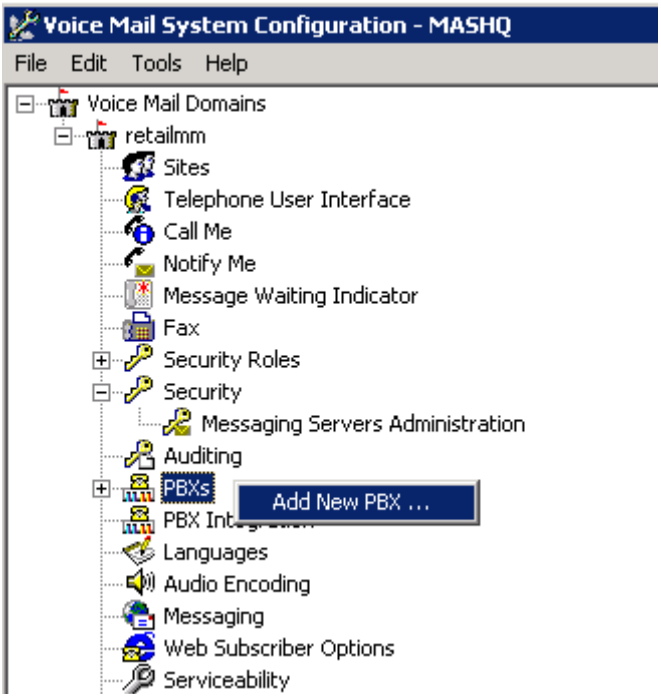
In sample configuration, the Communication Manager and IP Office telephone systems were added as sites to a multi-site Modular Messaging system, which was modified to support their subscribers and communication with Session Manager. The procedures include the following areas:

- Administer PBXs
- Administer Sites
- Administer Subscribers

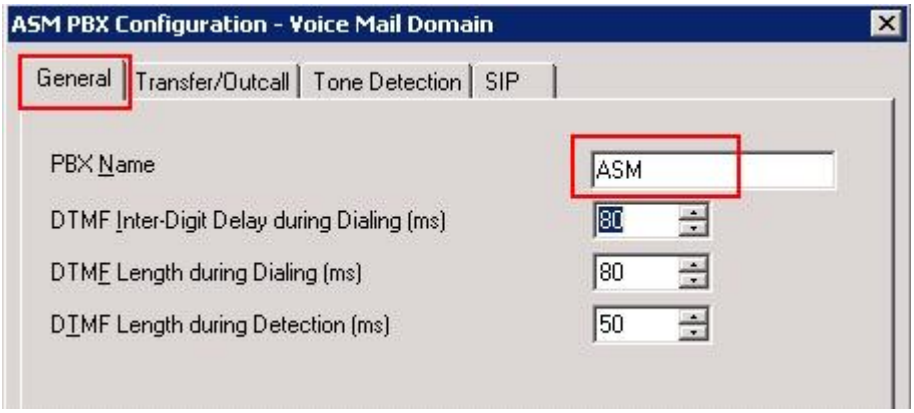
See references **[6]** to [**8]** in **Section 9** for standard installation and configuration information. General knowledge of the configuration tools and interfaces is assumed.

6.1. Administer PBXs

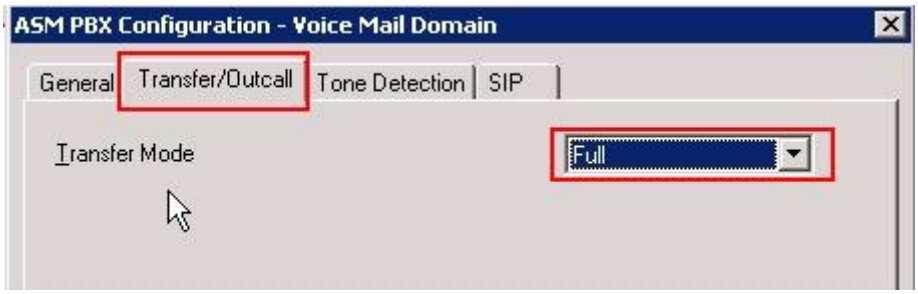
The aggregate Session Manager, Communication Manager and IP Office systems are defined to Modular Messaging as a PBX**.** In the MAS, open the **Voice Mail System Configuration** program, select **PBXs**, and right-click on the mouse to select **Add New PBX**, as shown below.



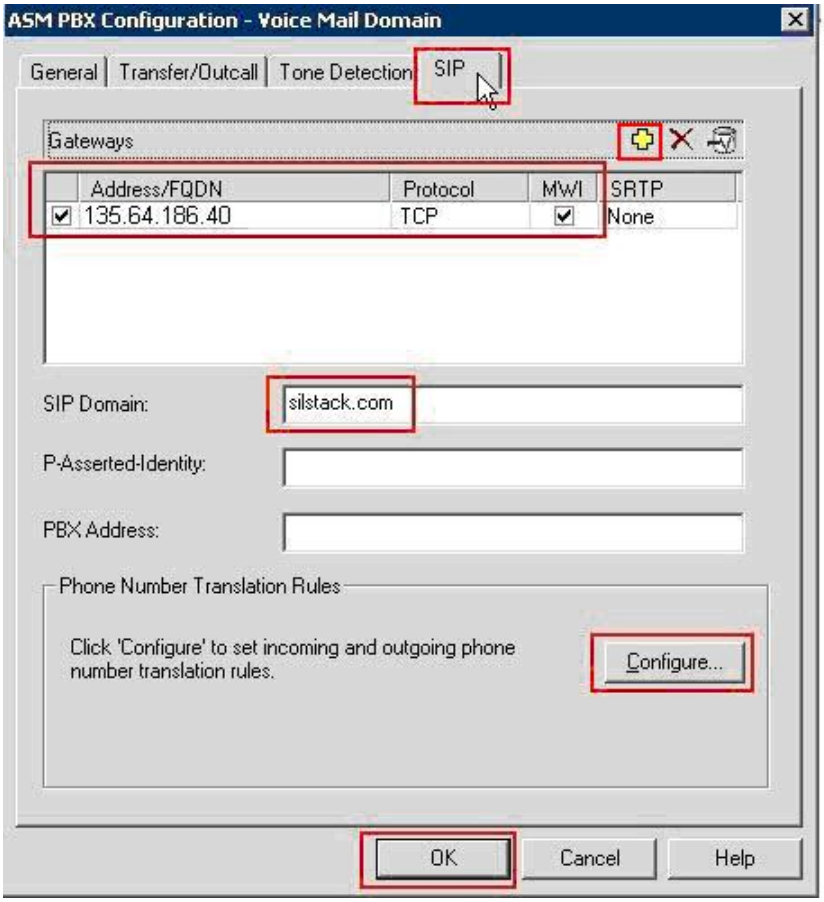
On the **General** tab of the resulting displayed window, enter an appropriate **PBX Name**. Defaults can be used for the remaining fields.



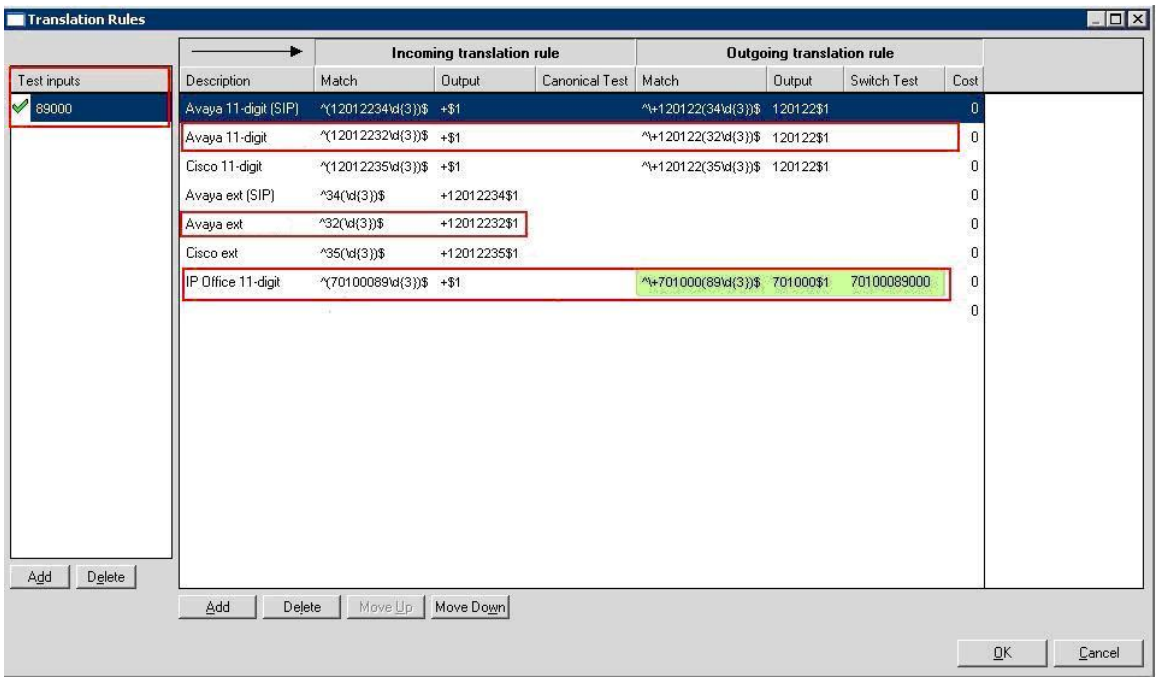
On the **Transfer/Outcall** tab, select **Full** for **Transfer Mode**.



Default values can be used for the **Tone Detection** tab. On the **SIP** tab navigate to the **Gateways** section, click on the + icon and add the Session Manager's Asset Card IP address under **Address/FQDN**, **TCP** for **Protocol**, and click the MWI box so message waiting notifications will be sent. Fill in **SIP Domain** with the domain from **Section 4.2**. Click on **Configure** to specify number translation rules for translating between the local dial plans of the Communication Manager and IP Office telephone systems and the canonical 11 digit form used by Modular Messaging.

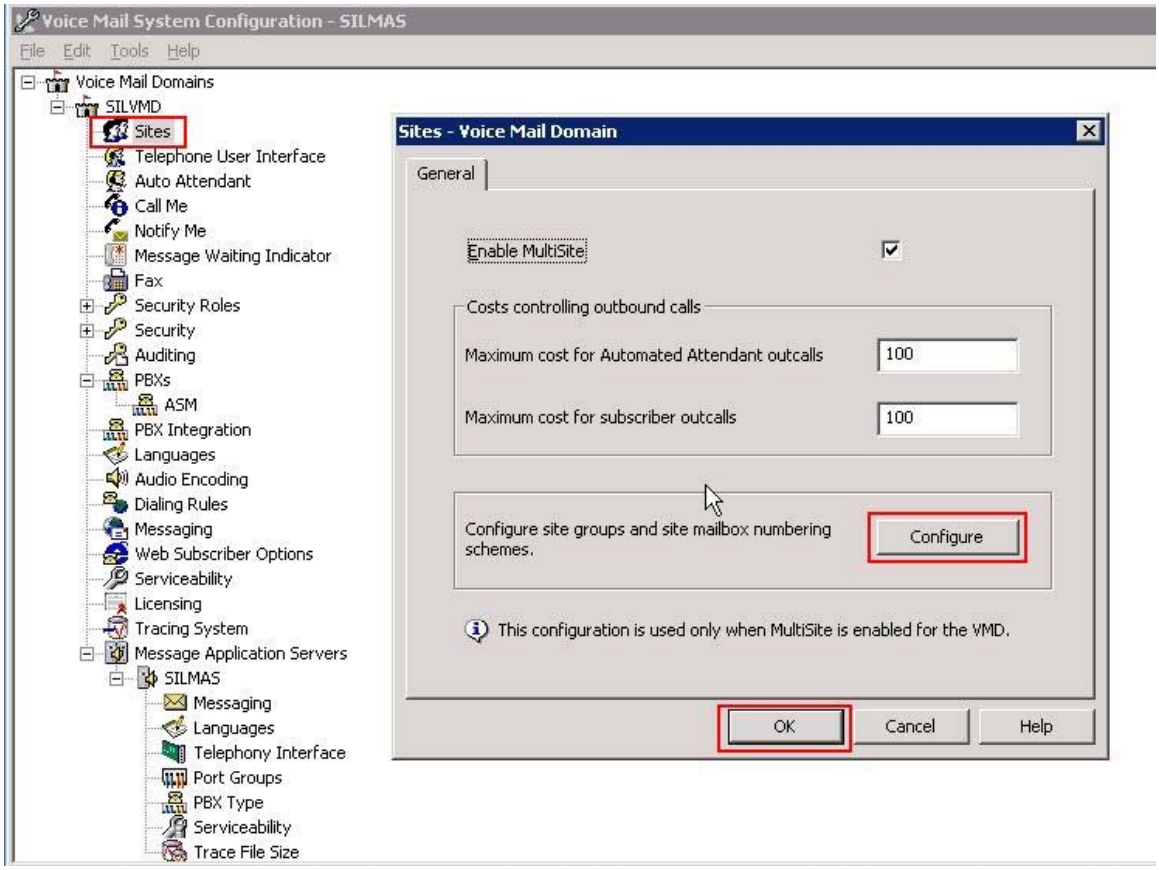


In the right pane, create the appropriate rules to translate between the 5-digit (Communication Manager) and 11-digit (IP Office) extensions dialed and the canonical 11 digit numbers used by Modular Messaging. For the sample configuration, three rules are required and were added by selecting **Add**. As described in **Section 4.3**, Session Manager will translate between 11-digit numbers used by Modular Messaging and 5-digit numbering used by the Communication Manager telephone system. The **Avaya CM 11-digit** and **IP Office 11-digit** Incoming and Outgoing translation rules specify that Modular Messaging will change the number to/from the canonical form. Modular Messaging, when configured for Multi-Site, requires the number to be in canonical form (a number with a + prefix). The **Avaya ext** rule supports features such as extension dialing by subscribers while accessing Modular Messaging and translate the 5-digit extension format into canonical 11-digit format. Proper operation of the rules can be verified by adding **Test inputs** in the left pane and viewing the resulting output in the corresponding rule in the right pane. Click on **OK** when finished, then again on **OK** in the original **Add new PBX** window (see previous screen).



6.2. Administer Sites

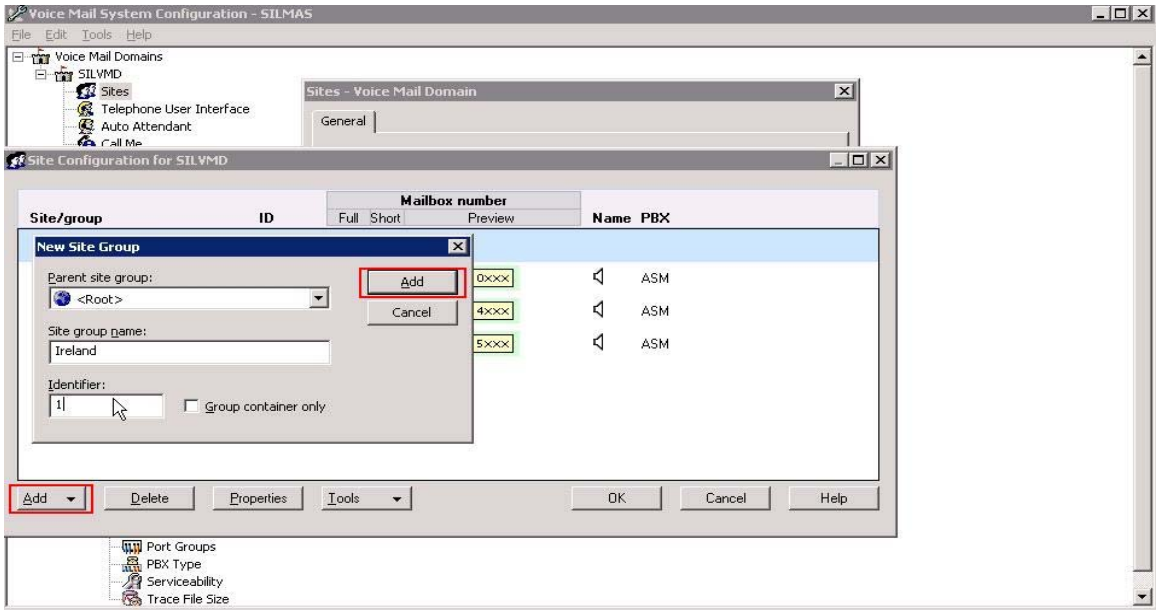
The Communication Manager and IP Office telephone systems must be added as sites in Modular Messaging. This is done by double-clicking **Sites** in the Voice Mail System Configuration tool, as shown below. In the **Sites** window that is displayed, click on **Configure**.



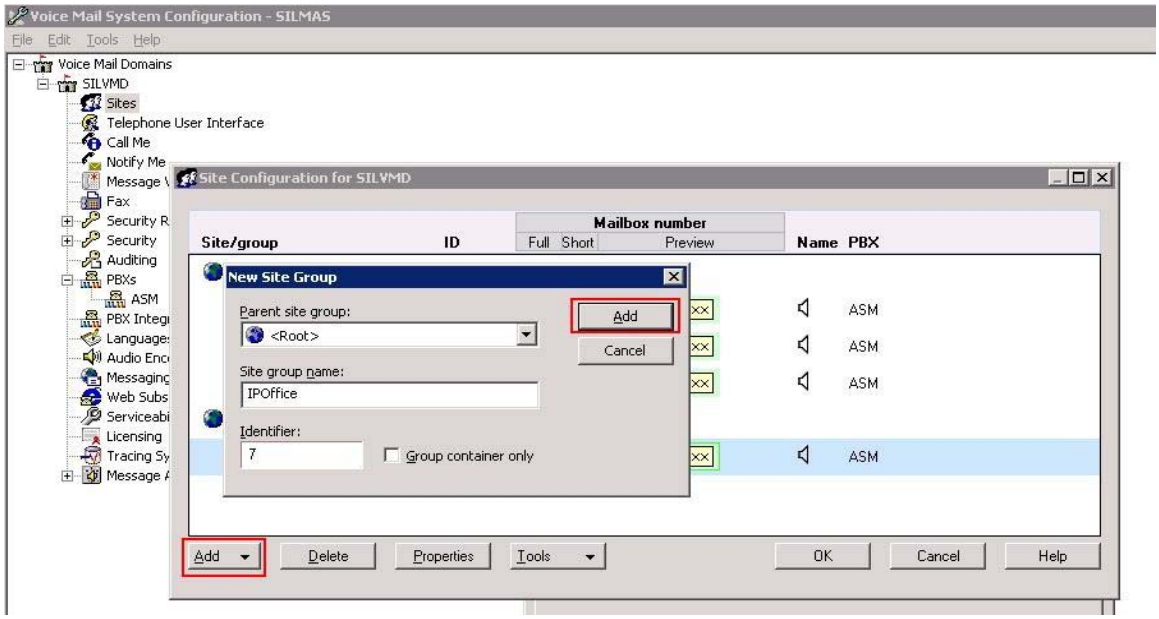
The Site Configuration window is displayed. First create a site group which will be referenced when adding a new site for Communication Manager. Click on **Add** button and select **Group** and enter the following in the **New Site Group** window:

- **Parent site group** Parent site name (default is **Root**)
- **Site group name** Site group name (e.g. **Ireland**)
- **Identifier** A unique number identifying the site group

Click on **Add** when finished.



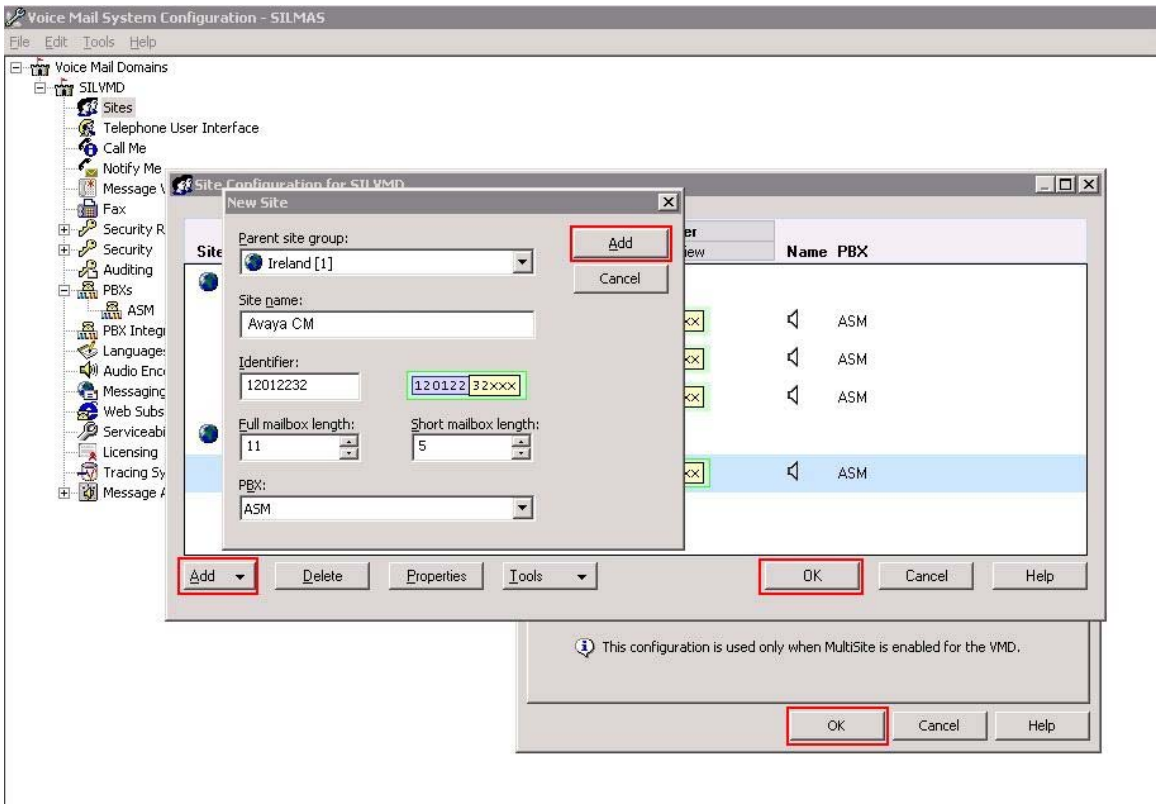
Repeat the above procedure to create a site group which will be referenced when adding a new site for IP Office.

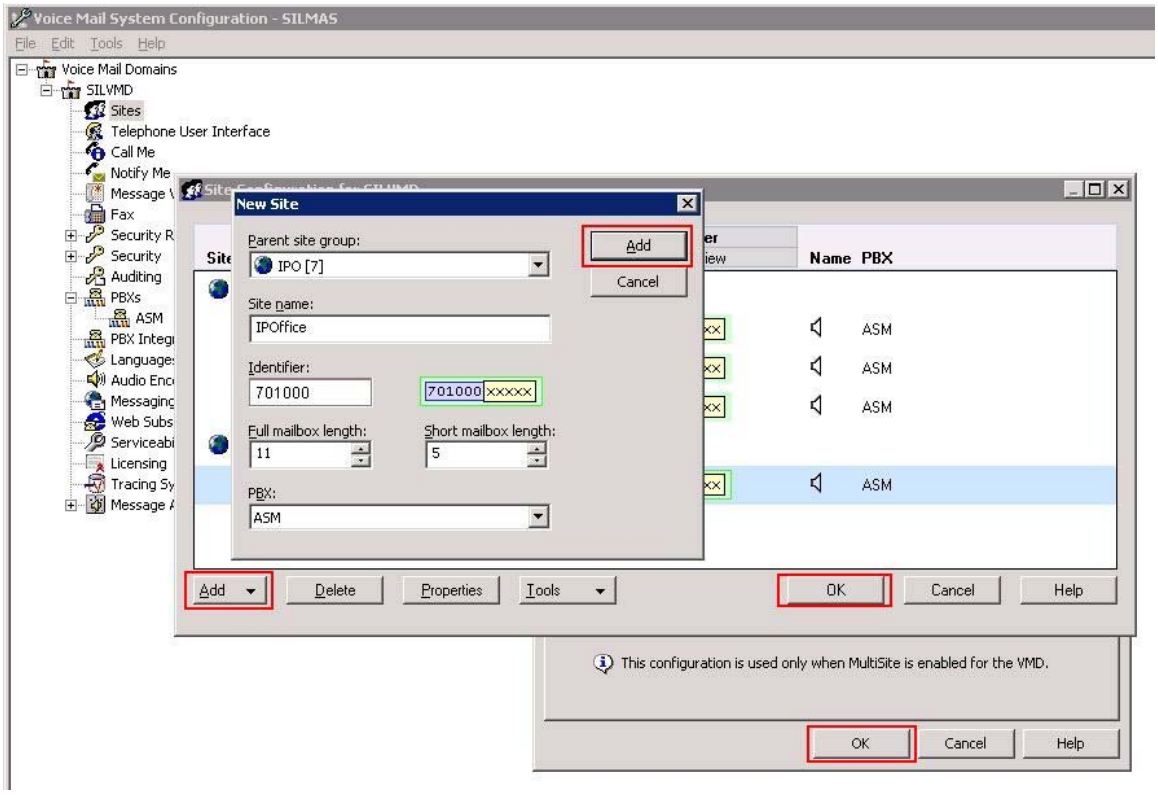


Click on **Add** and select **Site** to add the Communication Manager site, and enter the following in the **New Site** window:

- **Parent site group** Parent site name (e.g., **Ireland**)
- **Site name** Site name
- **Identifier** The unique initial digits of the 11-digit mailbox number, identifying the site
- **Full mailbox length** Enter **11** for the full mailbox number length
- **Short mailbox length** Enter **5** for the extension length
- **PBX** Enter name of the PBX added in the previous **Section 6.1**.

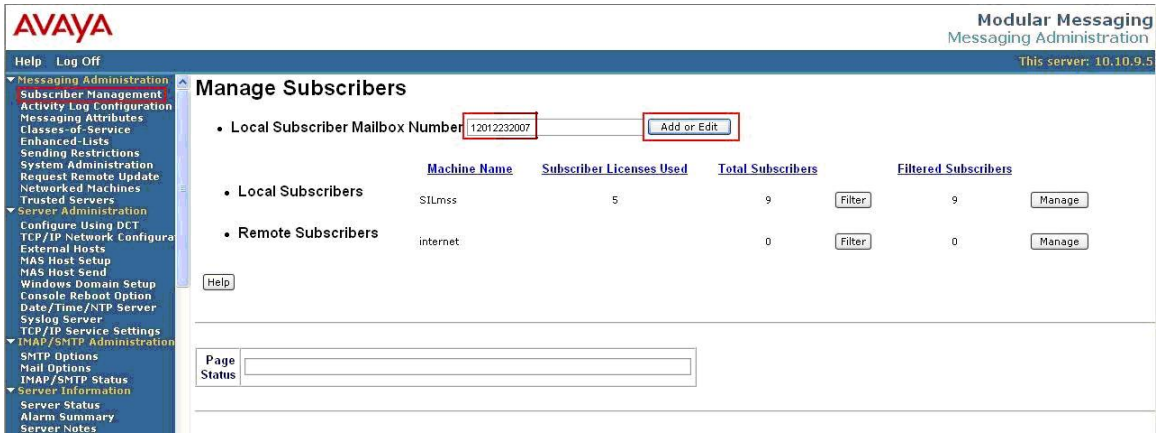
Click on **Add** when finished. The following two screenshots show the adding of the Communication Manager and IP Office sites. When all sites are added, click **OK** in the **Site Configuration** window, and then click on **OK** in the original **Sites** window.



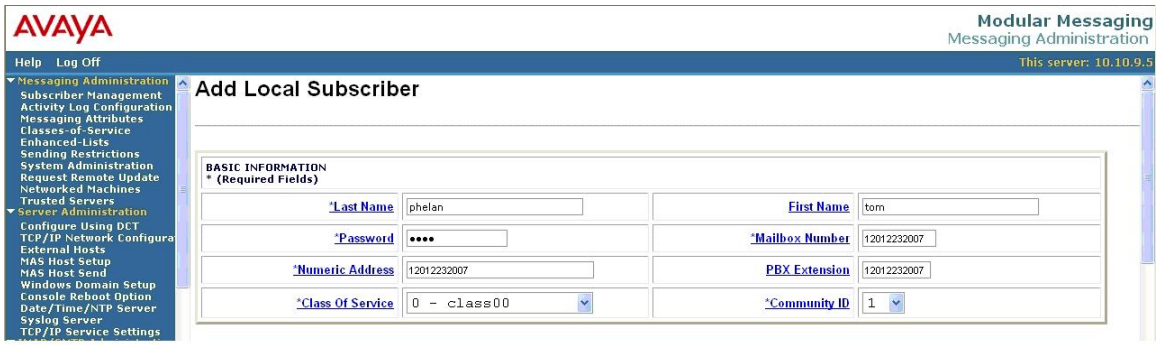


6.3. Administer Subscribers

Log in to the MSS. Select **Messaging Administration → Subscriber Management** from the left pane, to display the **Manage Subscribers** screen. For the **Local Subscriber Mailbox Number** field toward the top of the screen, enter a mailbox number of the subscriber. Click **Add or Edit** box to define more information for the mailbox subscriber.



The **Add Local Subscriber** screen is displayed next. Enter the desired string into the **Last Name, First Name and Password**. In the interoperability testing, the same telephone extensions for the Communication Manager and IP Office subscribers were used for the **Mailbox Number**, **Numeric Address**, and **PBX Extension** fields. Scroll down to the bottom of the screen and click **Save** (not shown). Repeat this section to add all subscribers.



7. Verification

This section provides the tests that can be performed on Communication Manager, Session Manager, Modular Messaging and IP Office to verify their proper configuration.

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 being investigated. Verify that all trunks are in the **in-service/idle** state as shown below.

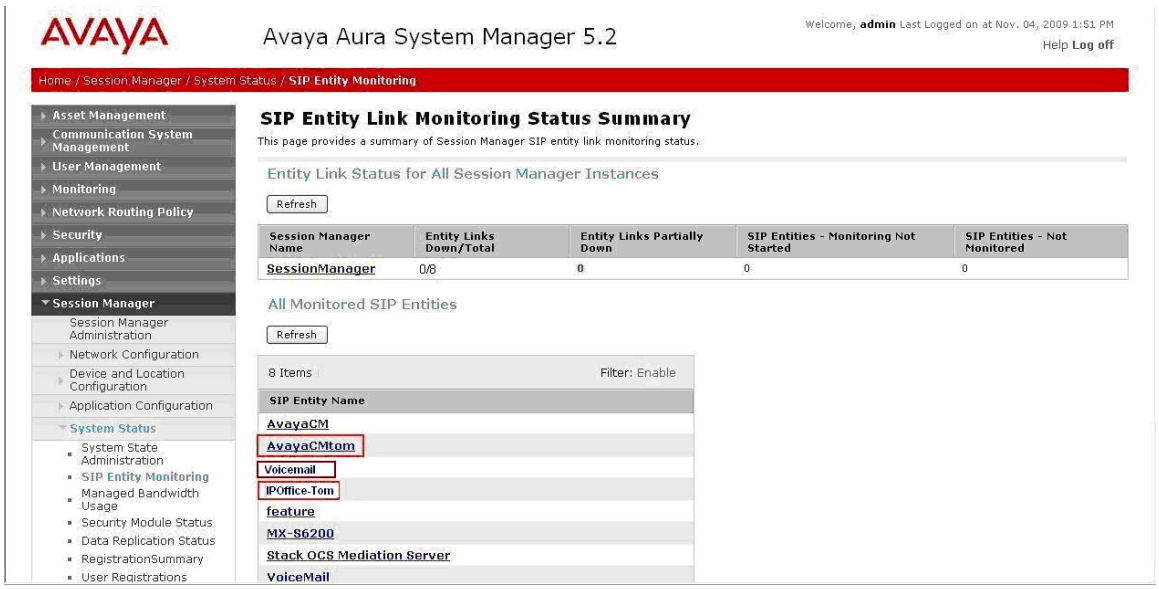
```
status trunk 150

                            TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0150/001 T00036   in-service/idle    no
0150/002 T00037   in-service/idle    no
0150/003 T00038   in-service/idle    no
0150/004 T00039   in-service/idle    no
0150/005 T00040   in-service/idle    no
0150/006 T00041   in-service/idle    no
0150/007 T00042   in-service/idle    no
0150/008 T00043   in-service/idle    no
0150/009 T00044   in-service/idle    no
0150/010 T00045   in-service/idle    no
```

Verify the status of the SIP signaling-group by using the **status signaling-group n** command, where **n** is the signaling group number being investigated. Verify that the signaling group is in the **in-service** state as shown below.

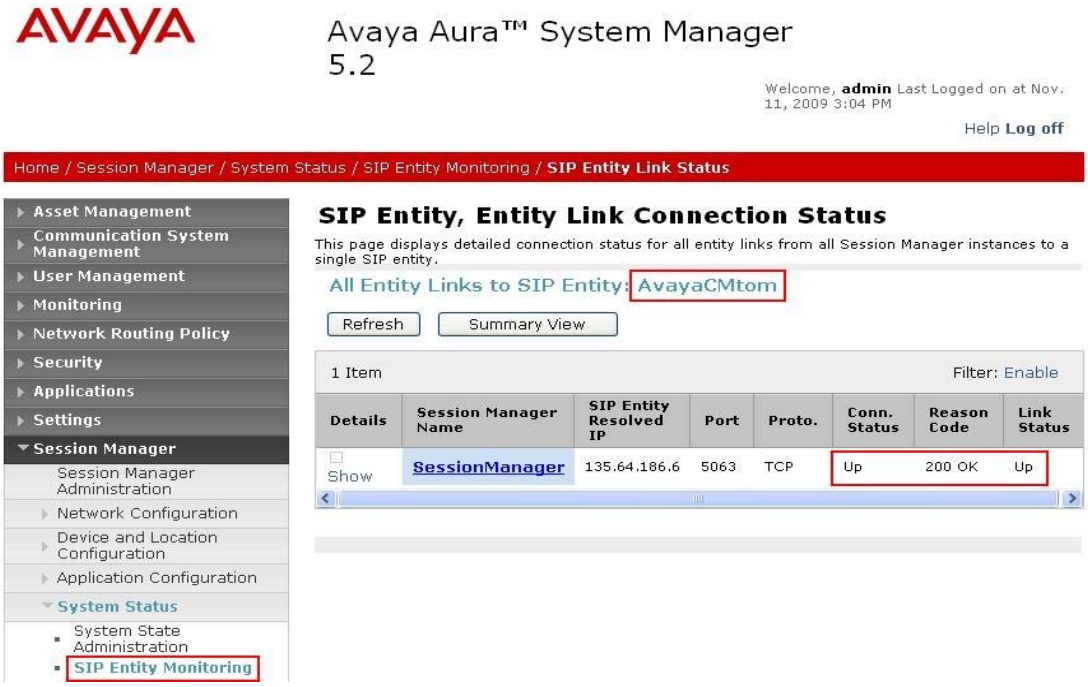
```
status signaling-group 150
                         STATUS SIGNALING GROUP

        Group ID: 150                             Active NCA-TSC Count: 0
      Group Type: sip                              Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

7.2. Verify Avaya Aura™ Session Manager

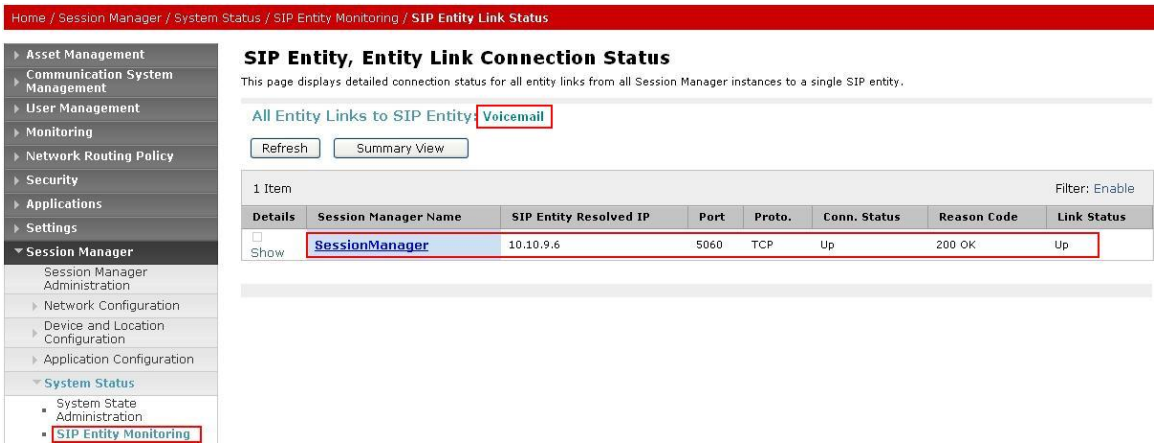
Select **Session Manager → System Status → SIP Entity Monitoring**. Verify as shown below that none of the SIP entities for Communication Manager, Modular Messaging or Avaya IP Office links are down, indicating that they are all reachable for routing.



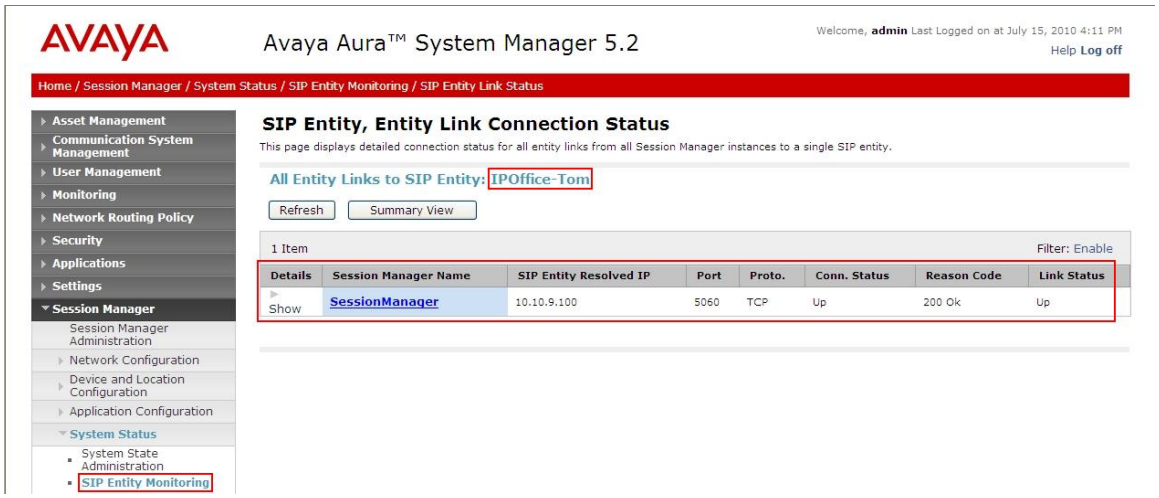
Click on the SIP Entity Names AvayaCMtom, IPOffice-Tom and Voicemail shown in the previous screen and verify that the connection status is **Up**, as shown in screenshots below.



Voicemail SIP Entity:

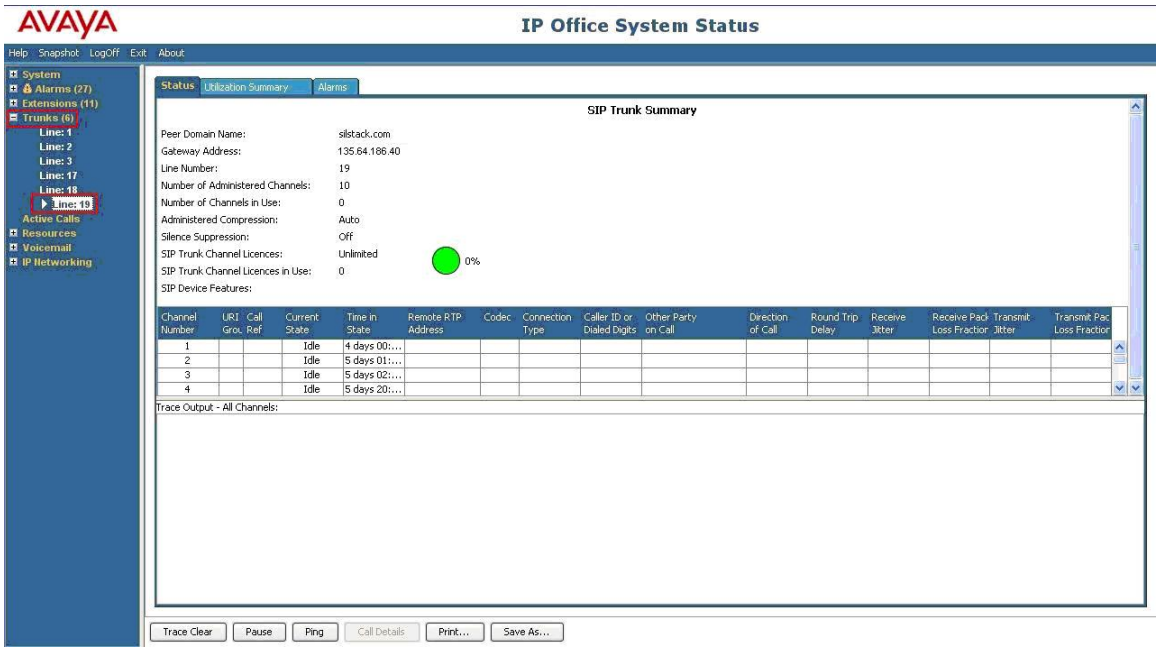


IPOffice-Tom SIP Entity:



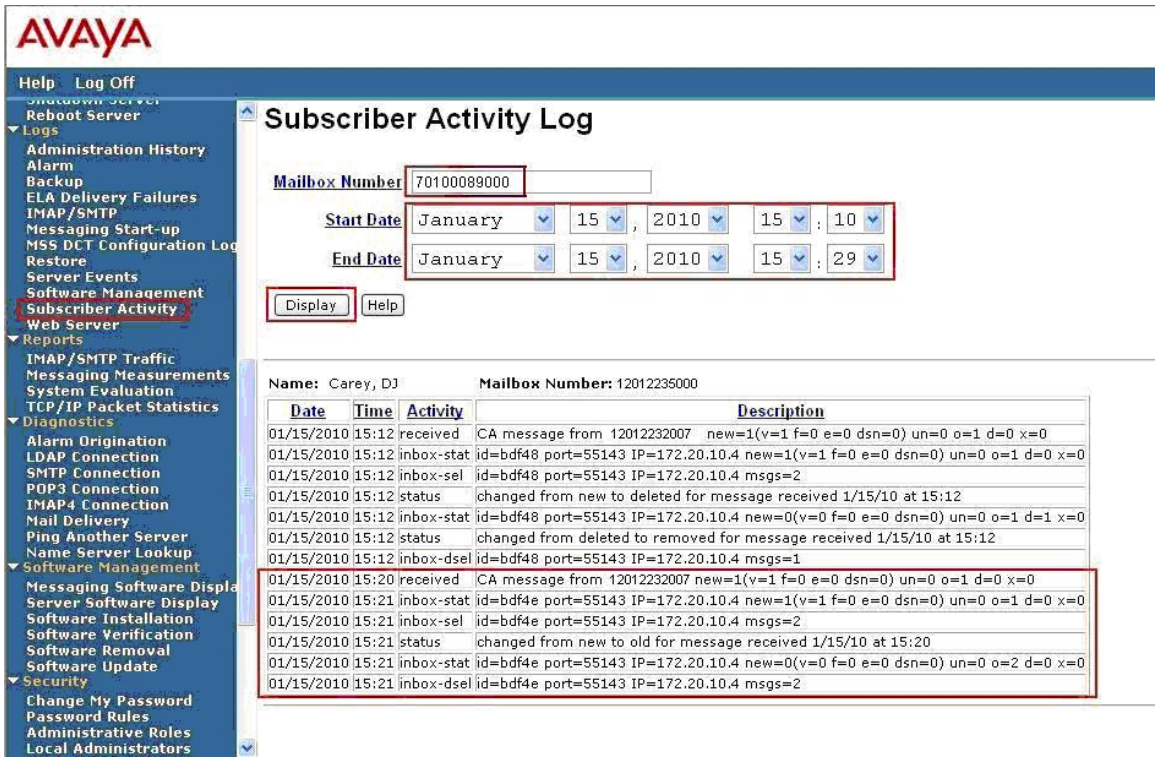
7.3. Verify Avaya IP Office

IP Office can be debugged with the System Status Application. Log in to the IP Office Manager PC and select **Start → Programs → IP Office → System St**atus to launch the application. Log into the application using the appropriate credentials. In the left panel, click on the **Trunks** entry and select the SIP trunk created in **Section 5.6**. Press the **Trace All** button (not shown). The messages on the line are displayed.



7.4. Verify Avaya Modular Messaging

Make a call from a Communication Manager subscriber to an IP Office subscriber and verify that the call covers to Modular Messaging upon no answer. Leave a voice message for the IP Office subscriber. From the IP Office subscriber, dial the Modular Messaging pilot number to retrieve the message. Verify that the Modular Messaging system identifies the IP Office subscriber as a local subscriber, and that the voice message can be retrieved. Log in to the MSS web interface and **select Logs → Subscriber Activity** from the left pane. Enter the **mailbox number** of the IP Office subscriber (70100089000), enter the appropriate **start date** and **end date** for the above activities, and click **Display**. Verify that a listing of the detailed activities is displayed into the bottom portion of the right hand pane. Verify that there is an entry showing the message left by the Communication Manager subscriber (in this case 12012232007). Also verify that there is an entry showing the message being retrieved.



7.5. Verified Scenarios

The verification scenarios for the configuration described in these Application Notes included the following:

- The IP Office subscribers were properly recognized by Modular Messaging upon dialing the Modular Messaging pilot number, and that the IP Office subscribers can log in without entering the mailbox number.
- The IP Office subscribers were properly identified by Modular Messaging as the calling party for voice messages left for other subscribers.
- Modular Messaging turns the message waiting indicator ON and OFF appropriately for voice messages left and retrieved for the IP Office subscribers.
- Modular Messaging appropriately identifies the original dialed endpoint as the called party for scenarios with Multiple Call Forwarding, where a called party has calls forwarded to another party that covers to Modular Messaging upon no answer.

8. Conclusion

As illustrated in these Application Notes, Avaya Aura™ Communication Manager can interoperate with Avaya IP Office using SIP trunks via Avaya Aura™ Session Manager. The following is a list of interoperability items observed:

- Find Me feature does not work properly if the Find Me subscriber and the Find Me destination subscriber are on different PBX's
- Fax sending to Avaya IP Office subscribers does not work properly

9. Additional References

Product documentation for Avaya products may be found at http://support.avaya.com

[1] *Avaya Aura™ Session Manager Overview*, Doc # 03-603323, Issue 2

[2] *Administering Avaya Aura™ Session Manager*, Doc # 03-603324, Issue 2

[3] *Maintaining and Troubleshooting Avaya Aura™ Session Manager*, Doc # 03-603325, Issue 2

[4] *SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers*, Doc # 555-245-206, Issue 9

[5] *Administering Avaya Aura™ Communication Manager*, Doc # 03-300509, Issue 5.0

[6] *Modular Messaging Admin Guide Release 5.2 with Avaya MSS*

[7] *Modular Messaging for the Avaya Message Storage Server (MSS) Configuration* Release 5.2 Installation and Upgrades

[8] *Avaya S8300/S85x0/S84x0/S87x0 SIP Integration using Avaya Session Manager*

[9] *Avaya IP Office Manager, Doc # 15-601011, Issue 24k*

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