

Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager 6.0.1, Avaya Aura® Session Manager 6.1 to support DuVoice DV2000 Voice Messaging System – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for the DuVoice DV2000 Voice Messaging System to operate with Avaya SIP enabled enterprise solution. The Avaya SIP enabled enterprise solution consists of Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and various Avaya endpoints. In the compliance testing SIP trunks were used in between the DuVoice DV2000 Voice Messaging System and Avaya Aura® Session Manager.

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

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1. Introduction

These Application Notes describe the configuration steps required for the DuVoice DV2000 Voice Messaging System to operate with Avaya SIP enabled enterprise solution. The Avaya SIP enabled enterprise solution consists of Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and various Avaya endpoints.

DuVoice DV2000 Voice Messaging System is a hospitality application that provides voicemail, automated attendant, and wake-up call features.

In the compliance testing SIP trunks were used in between the DuVoice DV2000 server and Avaya Aura® Session Manager, the DuVoice server used a Dialogic PowerMedia HMP, with a physical connection to the Local Area Network (LAN).

For the voicemail coverage scenarios, voicemail messages were recorded and saved on the DuVoice server. Standard SIP messaging was used to activate/deactivate the MWI, to optionally transfer the call via automated attendant or to schedule wakeup calls when requested manually by the guests.

InnDesk is a Web based wakeup and room management application used by the hotel staff to manage wakeup calls, guest voice mail and room clean/dirty status. From InnDesk the staff can manage guest rooms with web based interface from any staff PC on the networks. InnDesk was used to schedule wakeup call, to view failed wakeup calls, and to manage messages. Not all capabilities of InnDesk were tested, only capabilities related to wake up and voice mail services.

2. General Test Approach and Test Results

The general test approach was to simulate a hospitality environment in the Solution & Interoperability Test Lab by connecting an Avaya Aura® Communication Manager, Avaya Aura® Session Manager and the DuVoice DV2000 Voice Messaging System, as depicted in **Figure 1.**

2.1. Interoperability Compliance Testing

To verify SIP trunk interoperability, the following areas were tested for compliance:

Automated Attendant

- Incoming trunk calls to DuVoice DV2000 Voice Messaging System answered by Auto Attendant
- Incoming trunk calls to DuVoice DV2000 Voice Messaging System answered by Auto Attendant, originated from a simulating PSTN extension
- Transfers to Staff Extensions
- Transfers to Guest Extensions
- Remote Disconnects
- Invalid Options

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Voice Mail

- Incoming trunk calls to DuVoice Voice Messaging System for Voice Mail. Verifying message waiting indicator (light on/off) on different types of end-point (Analog, H.323, Digital and SIP phones).
- Guest to Guest Voice Messaging
- Staff Voice Messaging
- Voice Mail retrieval
- Voice Mail retrieval from a simulated PSTN extension
- Call Blocking

Wake-up call

- Scheduling wake-up calls from guest extensions
- Scheduling wake-up calls from staff extensions
- Scheduling wake-up calls from InnDesk
- Wake-up calls retries
- Wake-up call failed coverage (routes to front desk after expiration of retries)

2.2. Test Results

Interoperability testing of DuVoice DV2000 Voice Messaging System with Avaya SIP-enabled enterprise solution was completed successfully with the following observations/limitations.

- Message Waiting Indicator on Avaya SIP phones failed to illuminate or extinguish when a message was left or when a message was retrieved. When a message is left for the Avaya SIP phone DuVoice sends a SIP Notify message to Avaya Aura® Session Manager to indicate that there is a message for the subscriber and to turn the light on the SIP phone, the Avaya Aura® Session Manager replied with a "400 Bad Request (No subscription-state)". Avaya Session Manager does not support SIP unsolicited Notify message, which DuVoice uses. Message Waiting Indication worked properly on Analog, H.323, and Digital Telephones.

2.3. Support

2.3.1. Avaya

For technical support on the Avaya products described in these Application Notes visit http://support.avaya.com.

2.3.2. DuVoice

For technical support on DuVoice products visit the online support site at http://www.duvoice.com/

3. Reference Configuration

Figure 1 below illustrates the test configuration used. The test configuration simulates an enterprise site with Avaya SIP-enabled enterprise solution connected to the DuVoice DV2000 server via the Local Area Network (LAN).

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The Avaya components used to create the simulated customer site included:

- Avaya S8300 Server running Avaya Aura® Communication Manager.
- Avaya G450 Media Gateway.
- Avaya HP® Proliant DL360 G7 server running Avaya Aura® Session Manager.
- Avaya HP® Proliant DL360 G7 server running Avaya Aura® System Manager.
- Avaya 9620-Series IP Telephones (H.323).
- Avaya 9640 Telephone (SIP).
- Avaya 2420 Digital telephone.
- Analog Telephone.
- Desk top with administration interfaces.

The transport protocol between the Avaya Aura® Session Manager and the DuVoice Server is SIP over UDP. The transport protocol between Avaya Aura® Session Manager and Avaya Aura® Communication Manager across the enterprise IP network is SIP over TLS. Note: For ease of troubleshooting during testing, the compliance test was conducted with the Transport Method set to **tcp** between Avaya Aura® Session Manager and Avaya Aura® Communication Manager.

One SIP trunk group was created between the Avaya Aura® Communication Manager and the Avaya Aura® Session Manager to carry the traffic to and from DuVoice (two-way trunk group). To separate the codec settings required by DuVoice from the codec used by the telephones, two IP network regions were created, each with a dedicated signaling group. For inbound calls, the calls flowed from DuVoice to Avaya Aura® Session Manager. Avaya Aura® Session Manager used the configured dial patterns (or regular expressions) and routing policies to determine the recipient (in this case Avaya Aura® Communication Manager) and on which link to send the call. Once the call arrived at Avaya Aura® Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions could be performed.

Outbound calls to DuVoice were first processed by Avaya Aura® Communication Manager for outbound feature treatment such as automatic route selection and class of service restrictions. Once Avaya Aura® Communication Manager selected the proper SIP trunk; the call is routed to Avaya Aura® Session Manager. The Avaya Aura® Session Manager once again used the configured dial patterns (or regular expressions) and routing policies to determine the route to DuVoice.

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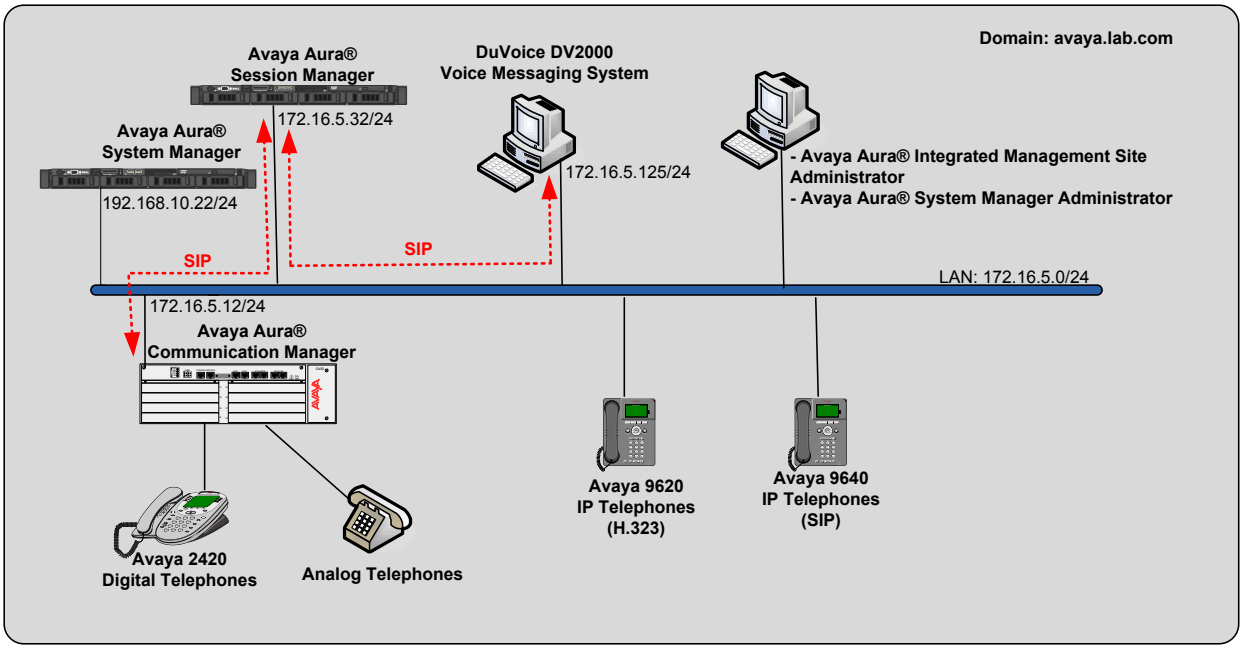


Figure 1: Network Configuration

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3.1. Client types used for testing

The following client types were used for testing:

DID Number	Client Type	Registered with
3040	Avaya 9620 IP Telephone (H323)	Avaya Aura® Communication Manager
3041	Avaya 9620 IP Telephone (H323)	Avaya Aura® Communication Manager
3042	Avaya 9620 IP Telephone (H323)	Avaya Aura® Communication Manager
3045	2500 (Analog)	Avaya Aura® Communication Manager
3046	Avaya 2420 Telephone (Digital)	Avaya Aura® Communication Manager
3047	Avaya 9640 Telephone (SIP)	Avaya Aura® Session Manager

Table 1 – Client Types used for testing

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4. Equipment and Software Validated

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

Component	Version
Avaya	
Avaya Aura® Communication Manager running on an Avaya S8300 Server	6.0.1 SP5.01 (00.1.510.1-19303)
Avaya G450 Media Gateway	31.20.0
Avaya Aura® Session Manager running on a HP® Proliant DL360 G7 Server	6.1 SP5 (ASM 6.1.5.0.615006)
Avaya Aura® System Manager running on a HP® Proliant DL360 G7 Server	6.1 Service Pack 5 Build 6.1.0.0.7345-6.1.5.502
Avaya 9620 IP Deskphones (H.323)	Avaya one-X® Desk phone Edition 3.102
Avaya 9640 IP Deskphone (SIP)	Avaya one-X® Deskphone Edition SIP 2.6.5.1
Avaya 2420 Digital Telephone	--
Avaya Analog Telephone	--
DuVoice	
Duvoice DV2000 Voice Messaging System	5.00.046

Table 2 – Hardware and Software Components Tested

The specific configuration above was used for the compliance testing. Note that this solution will be compatible with other Avaya Server and Media Gateway platforms running similar versions of Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

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5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring Avaya Aura® Communication Manager. A SIP trunk is created between Avaya Aura® Communication Manager and Avaya Aura® Session Manager for use by signaling traffic to and from DuVoice. It is assumed the general installation of Avaya Aura® Communication Manager, Avaya G450 Media Gateway and Avaya Aura® Session Manager has been previously completed.

In configuring Avaya Aura® Communication Manager, various components such as ip-network-regions, signaling groups, trunk groups, etc. need to be selected or created for use with the SIP connection to DuVoice. Unless specifically stated otherwise, any unused ip-network-region, signaling group, trunk group, etc. can be used for this purpose.

The Avaya Aura® Communication Manager configuration was performed using Avaya Integrated Management Site Administrator. Some screens in this section have been abridged and highlighted for brevity and clarity in presentation.

Please note that in the sample screenshots listed below the “display” command was used instead of the “change” or “add” commands, this is because all necessary changes were already in place when the screenshots were taken.

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5.1. Licensing and capacity

Use the **display system-parameters customer-options** command to verify that the **Maximum Administered SIP Trunks** value on **Page 2** is sufficient to support the desired number of simultaneous SIP calls across all SIP trunks at the enterprise including any trunks to DuVoice. The example shows that 4000 licenses are available and 16 are in use. The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity.

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

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 4000   10
           Maximum Concurrently Registered IP Stations: 2400   3
             Maximum Administered Remote Office Trunks: 4000   0
Maximum Concurrently Registered Remote Office Stations: 2400   0
              Maximum Concurrently Registered IP eCons: 68     0
  Max Concur Registered Unauthenticated H.323 Stations: 100    0
                       Maximum Video Capable Stations: 2400   0
                   Maximum Video Capable IP Softphones: 2400   1
                      Maximum Administered SIP Trunks: 4000   16
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000   0
   Maximum Number of DS1 Boards with Echo Cancellation: 80     0
                            Maximum TN2501 VAL Boards: 10     0
                     Maximum Media Gateway VAL Sources: 50     1
           Maximum TN2602 Boards with 80 VoIP Channels: 128    0
          Maximum TN2602 Boards with 320 VoIP Channels: 128    0
   Maximum Number of Expanded Meet-me Conference Ports: 300    0

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


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5.2. IP Node Names

Use the **display node-names ip** command to verify that node names have been previously defined for the IP addresses of the Avaya S8300D Server running Avaya Aura® Communication Manager **(procr**) and for Avaya Aura® Session Manager (**Lab-HG-SM**). These node names will be needed for defining DuVoice signaling group in **Section 5.5**.

```
display node-names ip                                          Page   1 of   2
                                   IP NODE NAMES
    Name                  IP Address
Lab-HG-SM                172.16.5.32
default                  0.0.0.0
msgserver                172.16.5.12
procr                    172.16.5.12
procr6                   ::

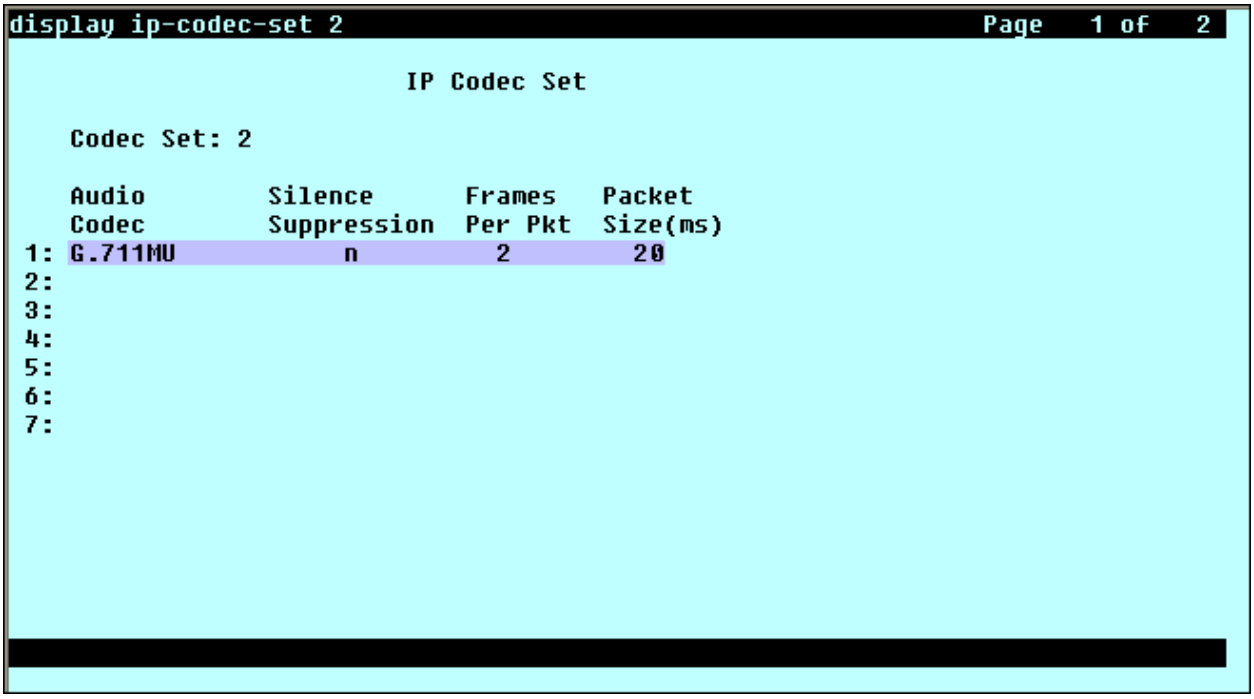



( 5  of 5    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```


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5.3. Codec

Use the **change ip-codec-set** command to define a list of codec's to use for calls between the enterprise and DuVoice. For the compliance test, ip-codec-set 2 was used for this purpose. Currently DuVoice only supports codec G.711MU. Default values can be used for all other fields.



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On **Page 2**, set the **Fax Mode** to **off.**

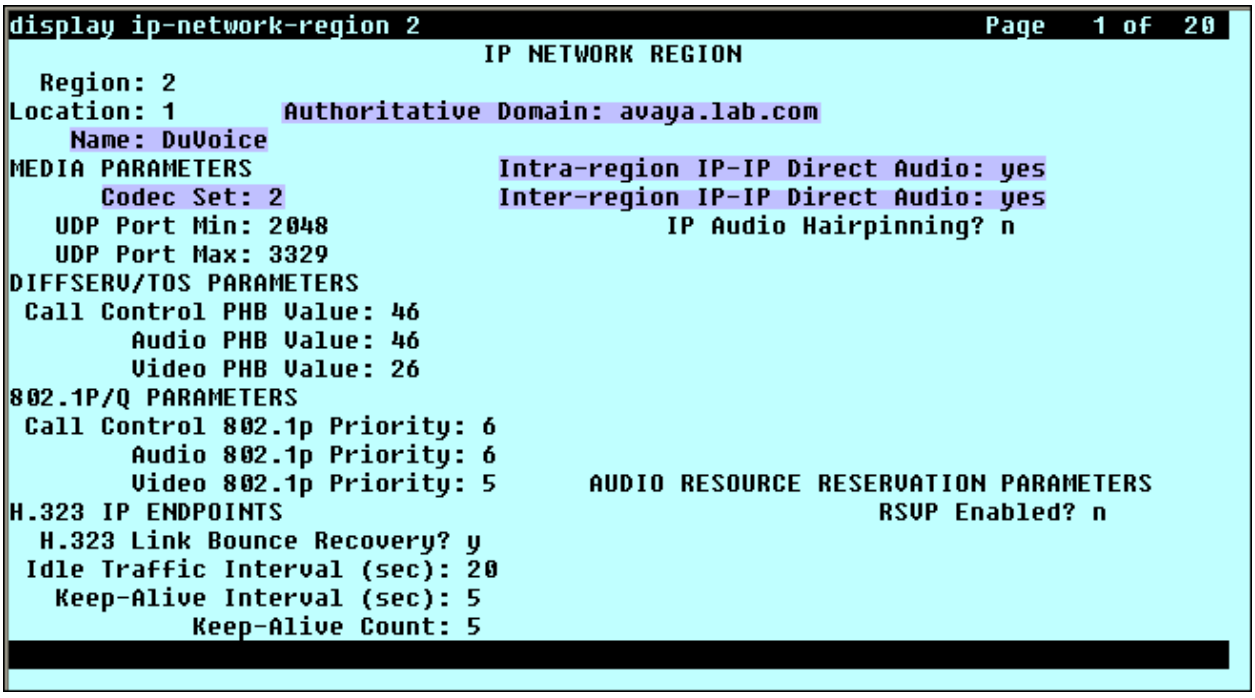


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5.4. IP Network Region

Create a separate IP network region for the DuVoice trunk. This allows for separate codec or quality of service settings to be used (if necessary) for calls between the enterprise and DuVoice versus calls within the enterprise or elsewhere. For the compliance test, IP-network-region 2 was chosen for the DuVoice trunk. Use the **change ip-network-region 2** command to configure region 2 with the following parameters:

- Set the **Authoritative Domain** field to match the SIP domain of the enterprise. In this configuration, the domain name is **avaya.lab.com**. This name appears in the "From" header of SIP messages originating from this IP region.
- Enter a descriptive name in the **Name** field.
- Enable **IP-IP Direct Audio** (shuffling) to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway. Set both **Intra-region** and **Inter-region IP-IP Direct Audio** to **yes.** This is the default setting. Shuffling can be further restricted at the trunk level on the Signaling Group form.
- Set the **Codec Set** field to the IP codec set defined in **Section 5.3**.
- Default values can be used for all other fields.



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On **Page 4**, define the IP codec set to be used for traffic between region 2 and region 1. Enter the desired IP codec set in the **codec set** column of the row with destination region (**dst rgn**) 1. Default values may be used for all other fields. The example below shows the settings used for the compliance test. It indicates that codec set 2 will be used for calls between region 2 (DuVoice) and region 1 (the rest of the enterprise).

```
display ip-network-region 2                                  Page   4 of  20

Source Region: 2     Inter Network Region Connection Management     I       M
                                                                   G  A    t
dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
1    2     y    NoLimit                                           n       t
2    2                                                             all
3
4
5
6
7
8
9
10
11
12
13
14
15
```


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5.5. Signaling Group

Use the **add signaling-group** command to create a signaling group between Avaya Aura® Communication Manager and the Avaya Aura® Session Manager for use by the DuVoice trunk. This signaling group is used for inbound and outbound calls between DuVoice and the enterprise. For the compliance test, signaling group 2 was used for this purpose and was configured using the parameters highlighted below.

- Set the **Group Type** field to **sip**.
- Set the **IMS Enabled** field to **n**. This specifies Avaya Aura® Communication Manager will serve as an Evolution Server for the Avaya Aura® Session Manager.
- Set the **Transport Method** to the recommended default value of **tls** (Transport Layer Security). Note: For ease of troubleshooting during testing, the compliance test was conducted with the **Transport Method** set to **tcp** The transport method specified here is used between Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The transport method used in between the Avaya Aura® Session Manager and DuVoice is UDP.
- Set the **Near-end Listen Port** and **Far-end Listen Port** to a valid unused port instead of the default well-known port value. (For TLS, the well-known port value is 5061). This is necessary so the Avaya Aura® Session Manager can distinguish this trunk from the trunk used for other enterprise SIP traffic. The compliance test was conducted with the **Near-end Listen Port** and **Far-end Listen Port** set to **5070**. (For TCP, the well-known port value is 5060).
- Set the **Peer Detection Enabled** field to **y**. The **Peer-Server** field will initially be set to **others** and cannot be changed via administration. Later, the **Peer-Server** field will automatically change to **SM** once Avaya Aura® Communication Manager detects its peer as an Avaya Aura® Session Manager.
- Set the **Near-end Node Name** to **procr**. This node name maps to the IP address of the Avaya S8300D Server running Avaya Aura® Communication Manager as defined in **Section 5.2**.
- Set the **Far-end Node Name** to **Lab-HG-SM**. This node name maps to the IP address of Avaya Aura® Session Manager as defined in **Section 5.2**.
- Set the **Far-end Network Region** to the IP network region defined for DuVoice in **Section 5.4**.
- Set the **Far-end Domain** to the domain of the enterprise.
- Set **Direct IP-IP Audio Connections** to **y**. This field will enable media shuffling on the SIP trunk allowing Avaya Aura® Communication Manager to redirect media traffic directly between the DuVoice server and the enterprise endpoint. If this value is set to **n**, then the Avaya Media Gateway will remain in the media path of all calls between the SIP trunk and the endpoint. Depending on the number of media resources available in the Avaya Media Gateway, these resources may be depleted during high call volume preventing additional calls from completing.
- Set the **DTMF over IP** field to **in-band**.
- Default values may be used for all other fields.

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```
display signaling-group 2
                                SIGNALING GROUP

 Group Number: 2                Group Type: sip
  IMS Enabled? n          Transport Method: tcp
        Q-SIP? n                                          SIP Enabled LSP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                 Far-end Node Name: Lab-HG-SM
 Near-end Listen Port: 5070                Far-end Listen Port: 5070
                                        Far-end Network Region: 2
                                  Far-end Secondary Node Name:
Far-end Domain: avaya.lab.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: in-band                Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                    IP Audio Hairpinning? n
         Enable Layer 3 Test? y                 Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 6

Command:
```


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5.6. Trunk Group

Use the **add trunk-group** command to create a trunk group for the signaling group created in **Section 5.5**. For the compliance test, trunk group 2 was configured using the parameters highlighted below.

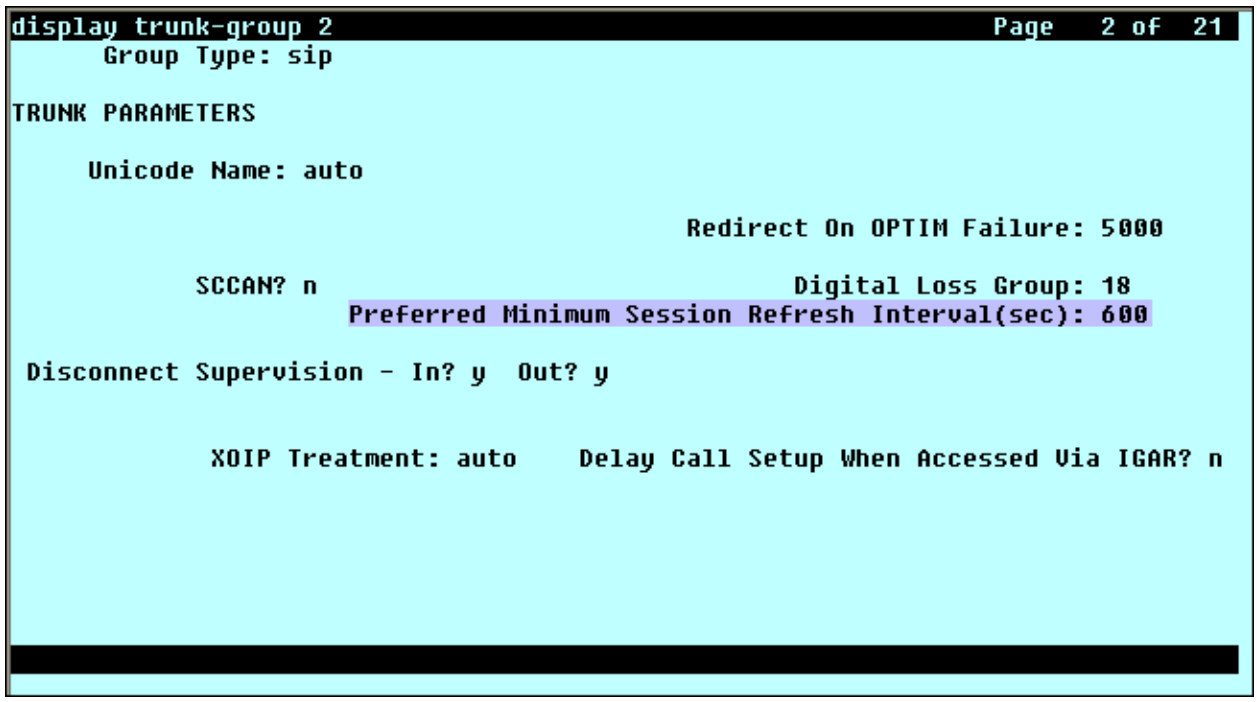
- Set the **Group Type** field to **sip**.
- Enter a descriptive name for the **Group Name**.
- Enter an available trunk access code (TAC) that is consistent with the existing dial plan in the **TAC** field.
- Set the **Direction** to **two-way**.
- Set the **Service Type** field to **public-ntwrk**.
- Set **Member Assignment Method** to **auto**.
- Set the **Signaling Group** to the signaling group shown in the previous step.
- Set the **Number of Members** field to twice the number of ports on the DuVoice server. This value determines how many simultaneous SIP calls can be supported by this trunk. Note: Each SIP call between two SIP end points requires two SIP trunks for the duration of the call.
- Default values were used for all other fields.

```
display trunk-group 2                                        Page   1 of  21
                              TRUNK GROUP

Group Number: 2                    Group Type: sip           CDR Reports: y
  Group Name: DuVoice                       COR: 1        TN: 1        TAC: 602
   Direction: two-way         Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: public-ntwrk           Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 2
                                                     Number of Members: 8
```

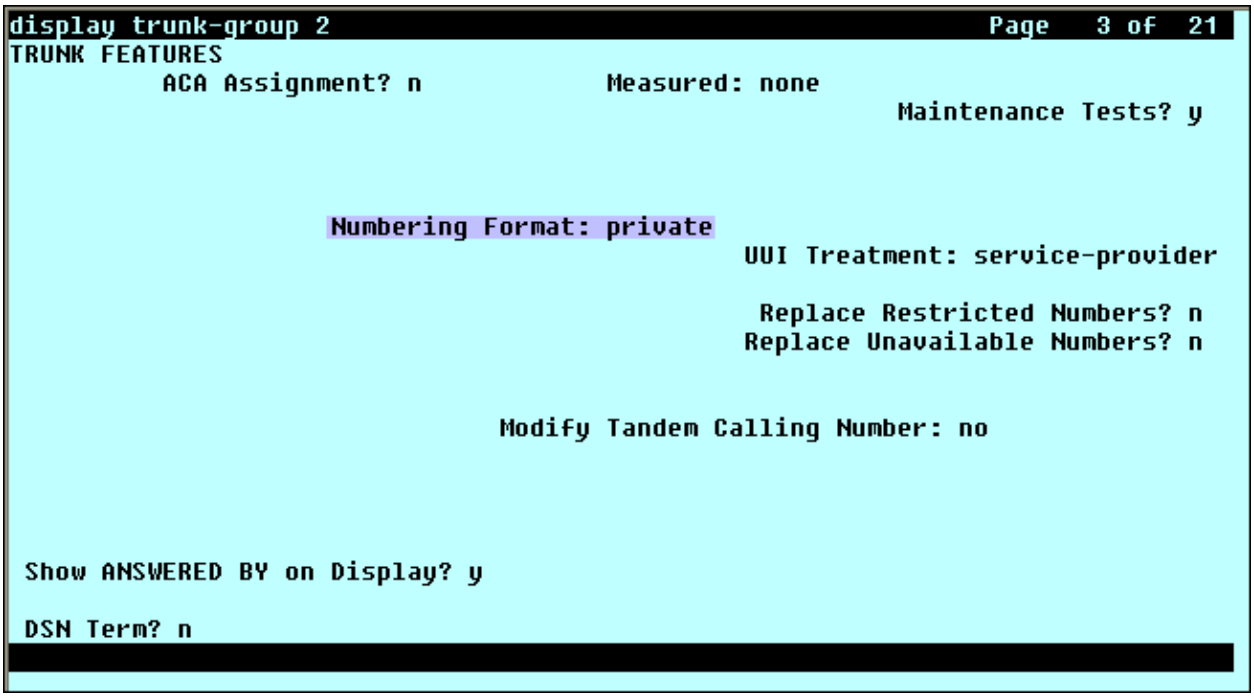

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On **Page 2**, verify that the **Preferred Minimum Session Refresh Interval** is set to a value acceptable to DuVoice. This value defines the interval that re-INVITEs must be sent to keep the active session alive. For the compliance test, the value of **600** seconds was used.



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On **Page 3**, set the **Numbering Format** field to **private**. This field specifies the format of the calling party number (CPN) sent to the far-end. Beginning with Avaya Aura® Communication Manager 6.0, public numbers are automatically preceded with a + sign when passed in the SIP From, Contact and P-Asserted Identity headers.



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On **Page 4**, set the **Network Call Redirection** field to **y**. Set the **Send Diversion Header** field to **y**. This field provides additional information to the network if the call has been re-directed.

Set the **Telephone Event Payload Type** to **blank** (default), DuVoice uses Inband for DTMF over IP. Default values were used for all other fields.

```
display trunk-group 2                                    Page   4 of  21
                            PROTOCOL VARIATIONS

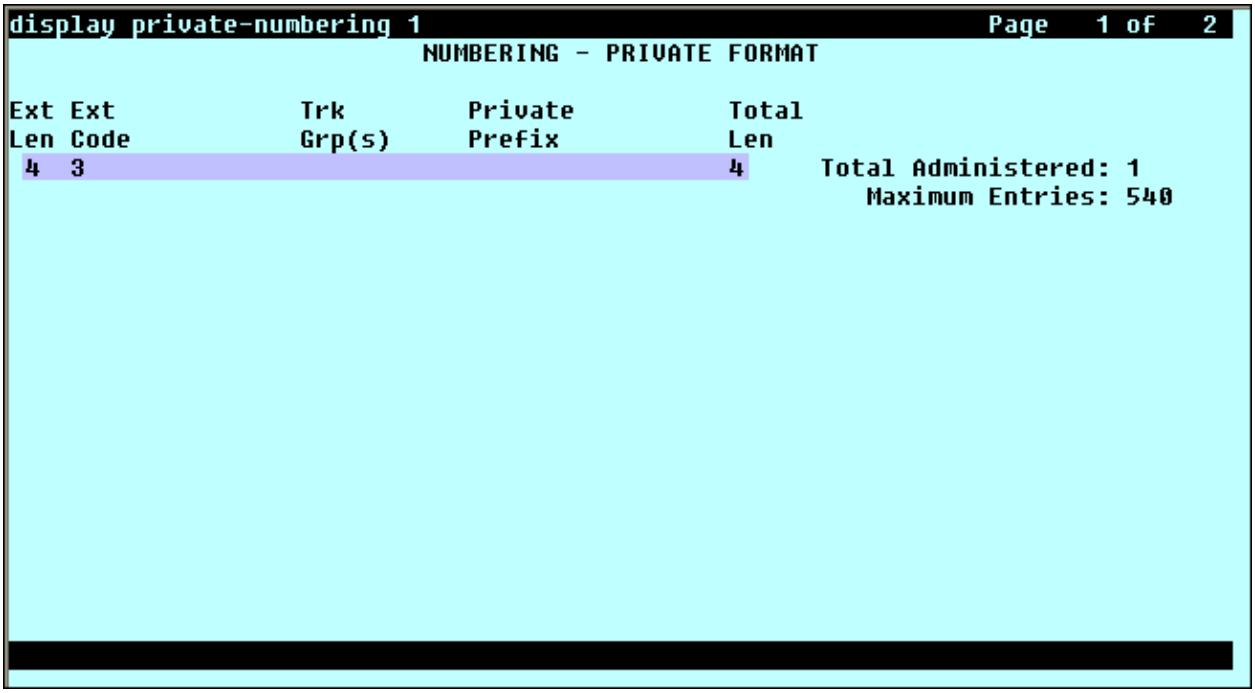
                     Mark Users as Phone? n
           Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                Network Call Redirection? y
                   Send Diversion Header? y
                 Support Request History? y
            Telephone Event Payload Type:


       Convert 180 to 183 for Early Media? n
 Always Use re-INVITE for Display Updates? n
       Identity for Calling Party Display: P-Asserted-Identity
                            Enable Q-SIP? n
```


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5.7. Calling Party Information

In the sample configuration extensions 3040 – 3047 were used on Avaya Aura® Communication Manager. Configure the **Numbering – Private Format** form to send the calling party number to DuVoice. Add an entry so that local stations with a 4-digit extension beginning with 3 are sent to DuVoice. This allows DuVoice to provide the proper greeting on calls that cover to voicemail and to automatically recognize DuVoice subscribers when retrieving messages. Since the **Trk Grp(s)** field is blank, this entry will apply for all outgoing trunk groups.



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5.8. Configure Voicemail Hunt Group

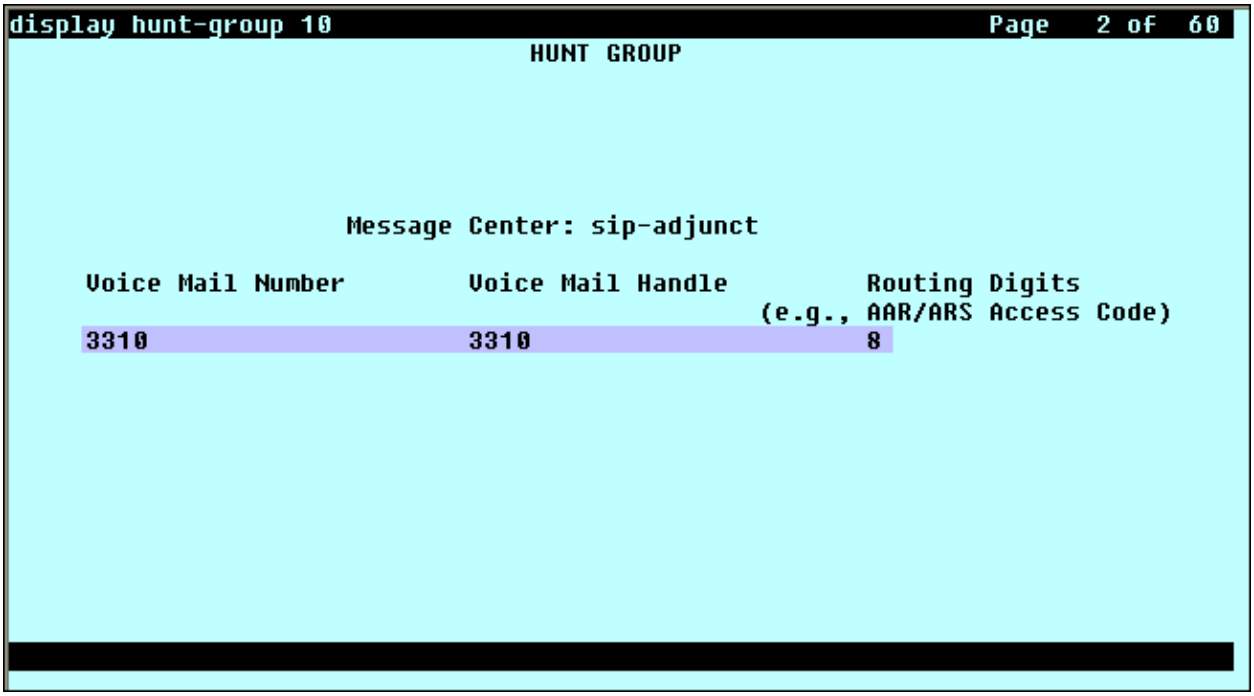
Configure a voicemail hunt group as shown below**.** Specify the voicemail pilot number in the **Group Extension** field. In this example, extension 3310 is dialed by users to access the DuVoice DV2000 Voice Messaging System.

```
display hunt-group 10                                        Page   1 of  60
                                  HUNT GROUP

           Group Number: 10                                 ACD? n
             Group Name: DuVoice                          Queue? n
        Group Extension: 3310                            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:
```


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On **Page 2** of the hunt group, set the **Message Center** field to **sip-adjunct** since DuVoice is accessed via SIP. Set the **Voice Mail Number** and the **Voice Mail Handle** fields to the digits used to route calls to DuVoice (e.g., the same hunt group extension is used here) and set the **Routing Digits** field to the AAR access code. In this example, the AAR feature access code was used to route calls. The voice mail number is used by Avaya Aura® Communication Manager to route calls to DuVoice.



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5.9. Configure Coverage Path

Configure the coverage path for the voice mail hunt group, which is group **h10** in this sample configuration. The default values shown for **Busy**, **Don't Answer**, and **DND/SAC/Goto Cover** can be used for the coverage criteria.

```
display coverage path 10
                                COVERAGE PATH

                 Coverage Path Number: 10
    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: h10             Rng:    Point2:
  Point3:                         Point4:
  Point5:                         Point6:

Command:
```


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5.10. Configure Station with Voicemail Coverage

When adding a station with voicemail coverage, configure the appropriate coverage path that points to the voicemail hunt group. The coverage path configured in **Section 5.9** was specified as shown below.

```
display station 3042                                          Page   1 of   5
                                    STATION

Extension: 3042                       Lock Messages? n                BCC: 0
     Type: 9620                       Security Code: *                 TN: 1
     Port: S00004                   Coverage Path 1: 10               COR: 1
     Name: 3042-H323                Coverage Path 2:                  COS: 1
                                    Hunt-to Station:
STATION OPTIONS
                                         Time of Day Lock Table:
             Loss Group: 19        Personalized Ringing Pattern: 1
                                             Message Lamp Ext: 3042
           Speakerphone: 2-way             Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal            Media Complex Ext:
   Survivable Trunk Dest? y                        IP SoftPhone? n

                                                     IP Video? n
                          Short/Prefixed Registration Allowed: default

                                                Customizable Labels? y
```

On **Page 2** of the station form, set the **MWI Served User Type** field to **sip-adjunct**.

```
display station 3042                                          Page   2 of   5
                                    STATION
FEATURE OPTIONS
           LWC Reception: spe              Auto Select Any Idle Appearance? n
          LWC Activation? y                      Coverage Msg Retrieval? y
  LWC Log External Calls? n                              Auto Answer: none
             CDR Privacy? n                        Data Restriction? n
   Redirect Notification? y                 Idle Appearance Preference? n
 Per Button Ring Control? n               Bridged Idle Line Preference? n
   Bridged Call Alerting? n                   Restrict Last Appearance? y
  Active Station Ringing: single
                                                    EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
      Service Link Mode: as-needed                    EC500 State: disabled
        Multimedia Mode: enhanced                Audible Message Waiting? n
    MWI Served User Type: sip-adjunct          Display Client Redirection? n
                                              Select Last Used Appearance? n
                                                Coverage After Forwarding? s
                                                  Multimedia Early Answer? n
                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 3042          Always Use? n IP Audio Hairpinning? n
     Precedence Call Waiting? y
```


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5.11. Configuring Call Routing

In this configuration, AAR was used to route calls to DuVoice as specified on **Page 2** of the hunt group configured in **Section 5.8**. The DuVoice pilot number is 3310 and those digits were used to route calls to DuVoice whenever a call covered to voicemail or when a user dials the DuVoice Messaging System directly. For information in configuring AAR or ARS, refer to [4].

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6. Configure Avaya Aura® Session Manager

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

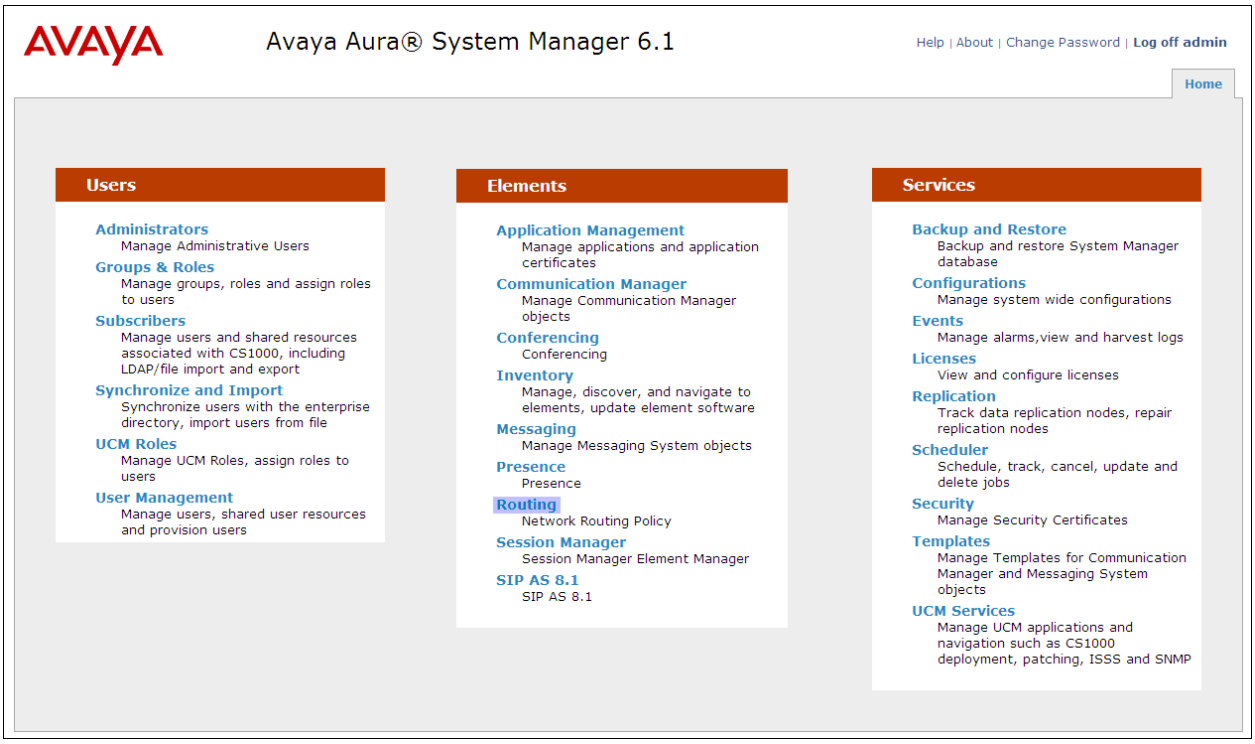
- SIP domain
- Logical/physical Location that can be occupied by SIP Entities
- Adaptation module to perform dial plan manipulation
- SIP Entities corresponding to Avaya Aura® Communication Manager and Avaya Aura® Session Manager
- Entity Links, which define the SIP trunk parameters used by Avaya Aura® Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Avaya Aura® Session Manager, corresponding to the Avaya Aura® Session Manager Server to be managed by Avaya Aura® System Manager.

It may not be necessary to create all the items above when creating a connection to DuVoice since some of these items would have already been defined as part of the initial Avaya Aura® Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities, and Avaya Aura® Session Manager itself. However, each item should be reviewed to verify the configuration.

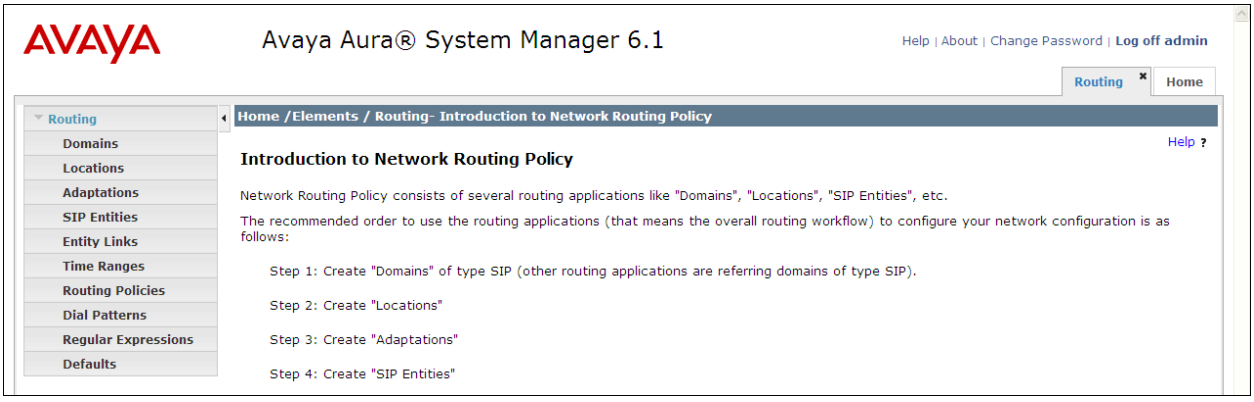
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6.1. System Manager Login and Navigation

Avaya Aura® Session Manager Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager, using the URL "https://<ip-address>/SMGR", where "<ip-address>" is the IP address of Avaya Aura® System Manager. Log in with the appropriate credentials and click on **Login** (not shown). The screen shown below is then displayed, click on **Routing**.



The navigation tree displayed in the left pane below will be referenced in subsequent sections to navigate to items requiring configuration. Most items will be located under the **Routing** link shown below.



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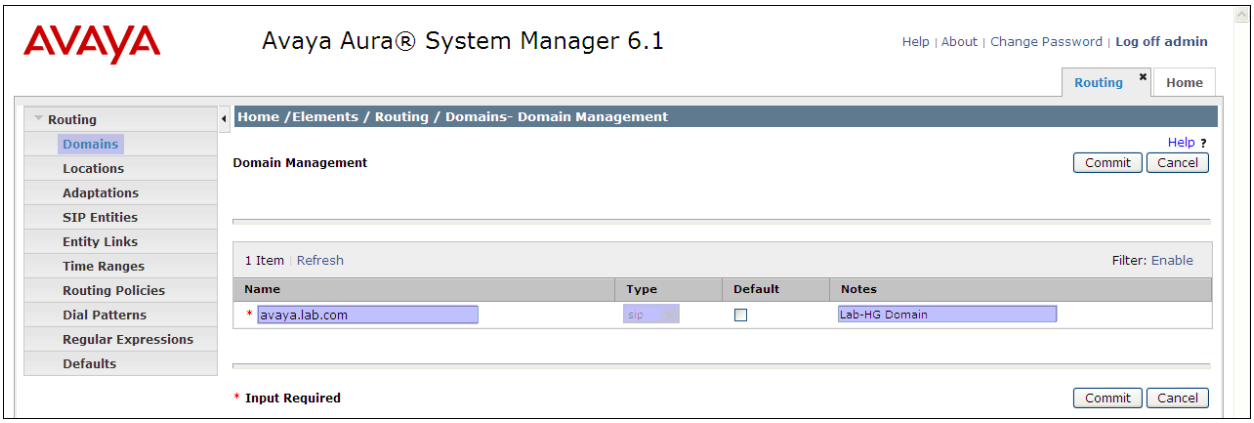
6.2. Specify a SIP Domain

Create a SIP domain for each domain for which Avaya Aura® Session Manager will need to be aware in order to route calls. For the compliance test, this includes the enterprise domain (**avaya.lab.com**).

To add a domain Navigate to **Routing → Domains** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:** Enter the domain name.
- **Type:** Select **sip** from the pull-down menu.
- **Notes:** Add a brief description (optional).

Click **Commit**. The screen below shows the entry for the enterprise domain.



6.3. Add Location

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. To add a location, navigate to **Routing →Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown).

In the **General** section, enter the following values. Use default values for all remaining fields:

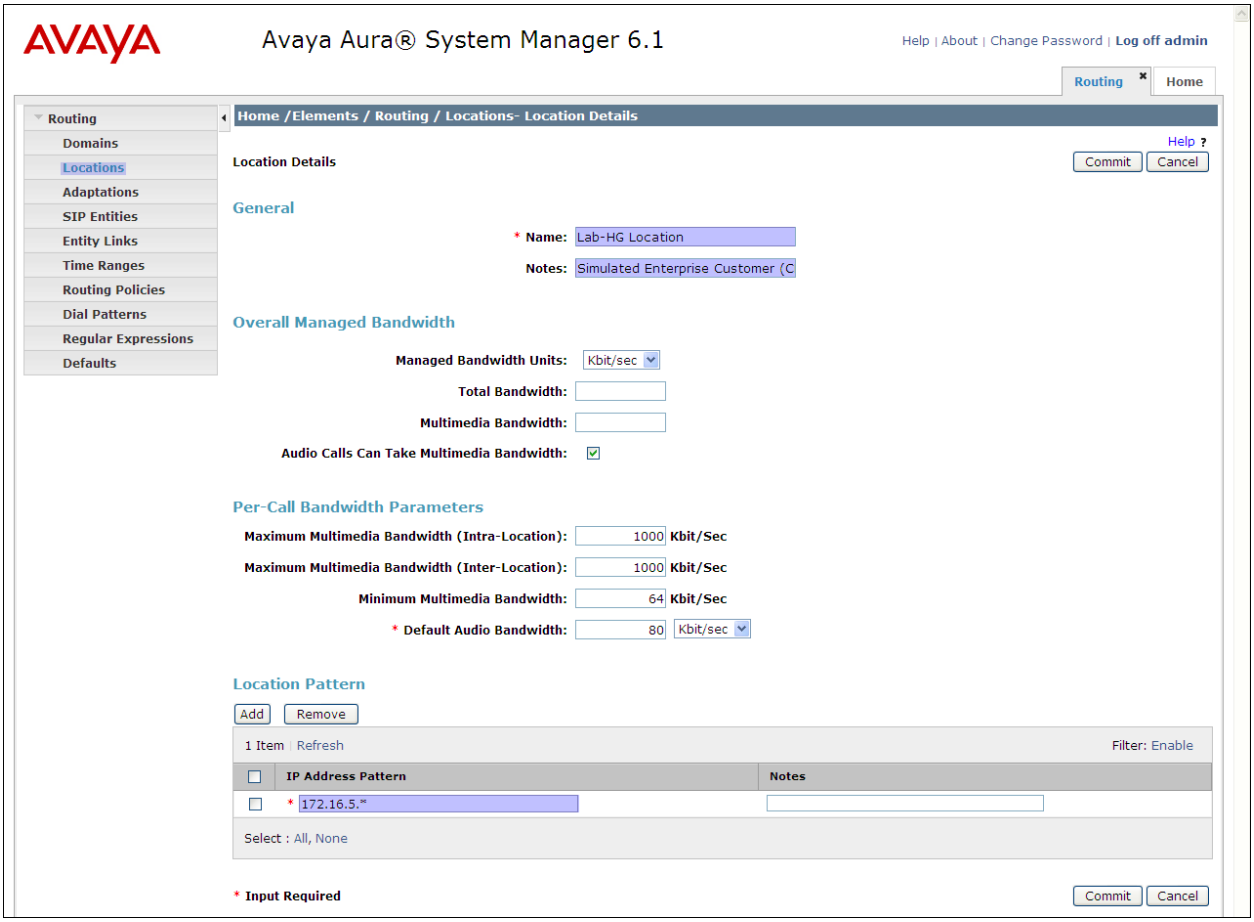
- **Name:** Enter a descriptive name for the location.
- **Notes:** Add a brief description (optional).

In the **Location Pattern** section, click **Add** and enter the following values. Use default values for all remaining fields:

- **IP Address Pattern:** An IP address pattern used to identify the location.
- **Notes:** Add a brief description (optional).

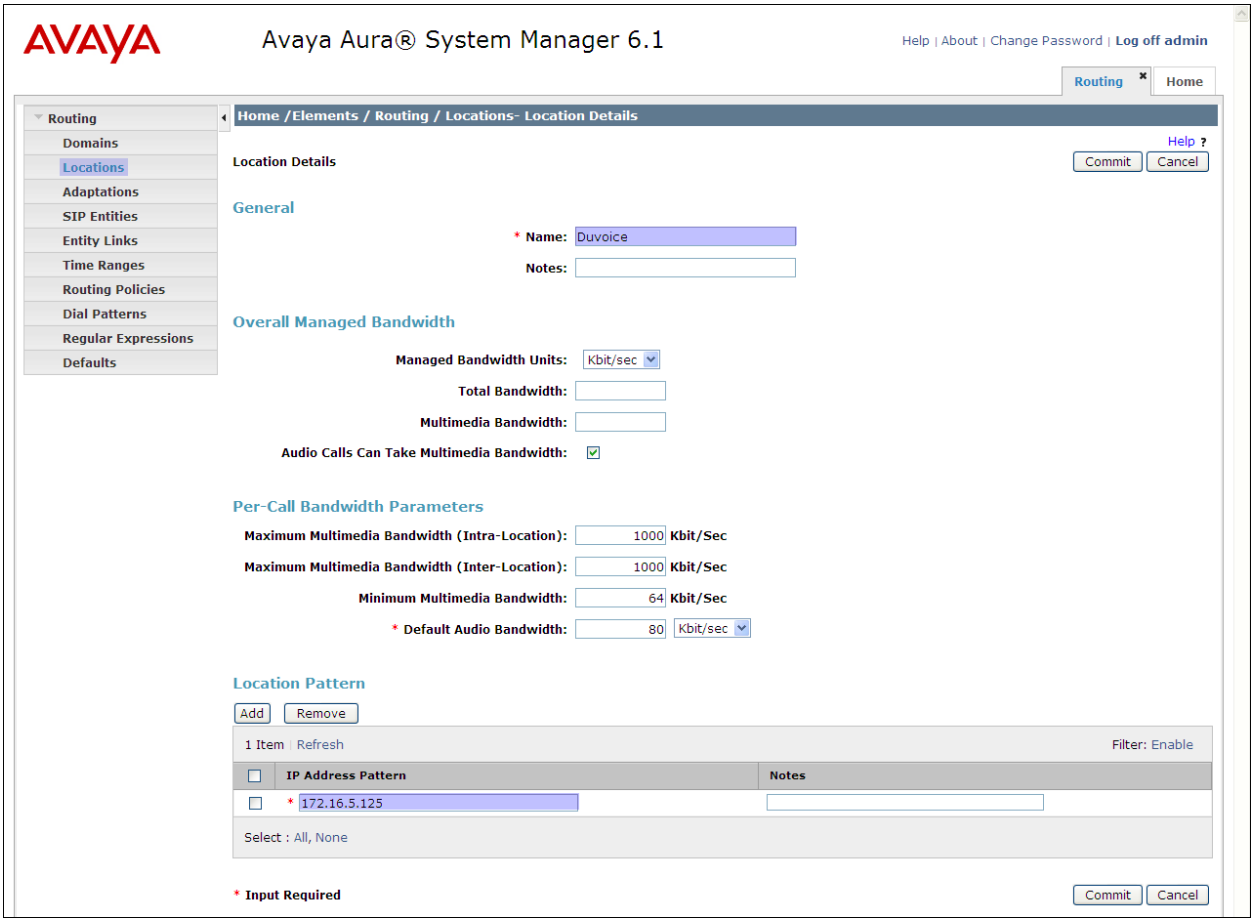
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The screen below shows the addition of the **Lab-HG Location**, which includes all equipment on the **172.16.5.x** subnet including Avaya Aura® Communication Manager, and Avaya Aura® Session Manager. Click **Commit** to save.



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Repeat the preceding procedure to create a separate Location for the DuVoice Server. The screen below shows the addition of the **DuVoice** location, which specifies the specific inside IP address for the DuVoice Server. Click **Commit** to save.

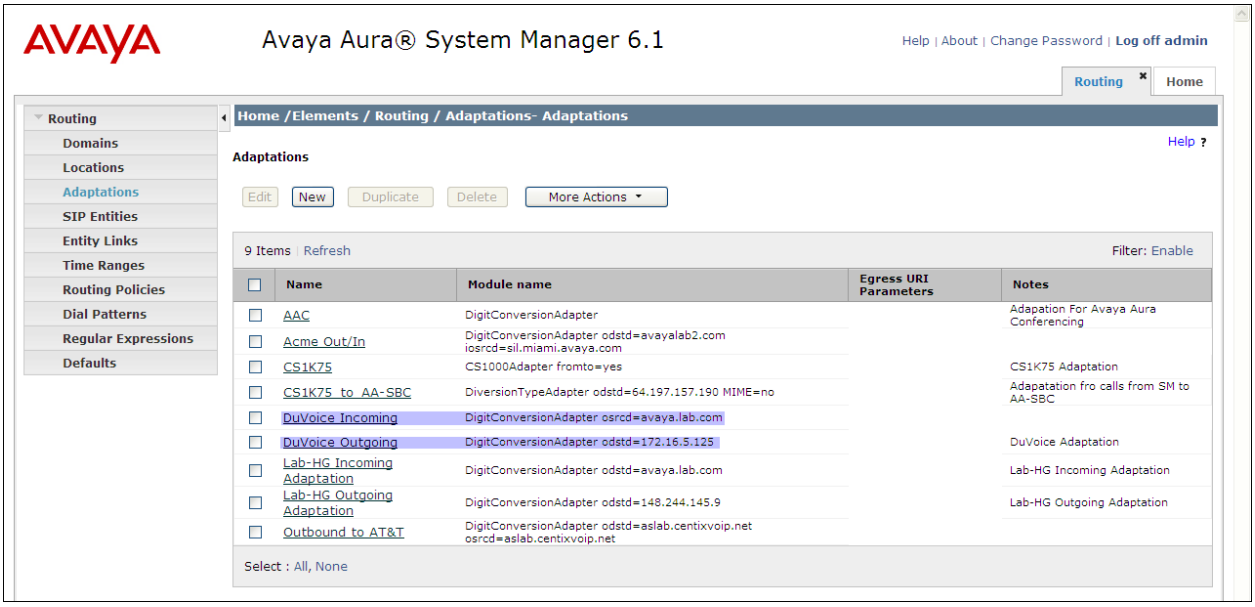


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6.4. Add Adaptation Module

Avaya Aura® Session Manager can be configured with adaptation modules that can modify SIP messages before or after routing decisions have been made. A generic adaptation module **DigitConversionAdapter** supports digit conversion of telephone numbers in specific headers of SIP messages. Other adaptation modules are built on this generic, and can modify other headers to permit interoperability with third party SIP products.

To view or change adaptations, select **Routing → Adaptations**. Click on the checkbox corresponding to the name of an adaptation and **Edit** to edit an existing adaptation, or the **New** button to add an adaptation. Click the **Commit** button after changes are completed. The following screen shows a portion of the list of adaptations in the sample configuration. Two adaptations were used for DuVoice, an incoming adaptation (for incoming calls from DuVoice) and an outgoing adaptation (for outgoing calls to DuVoice).



The adaptations named **DuVoice Incoming** and **DuVoice Outgoing** Adaptations were configured and used in the compliance test.

Settings for **DuVoice Outgoing** Adaptation**:**

In the **General** section, enter the following values. Use default values for all remaining fields:

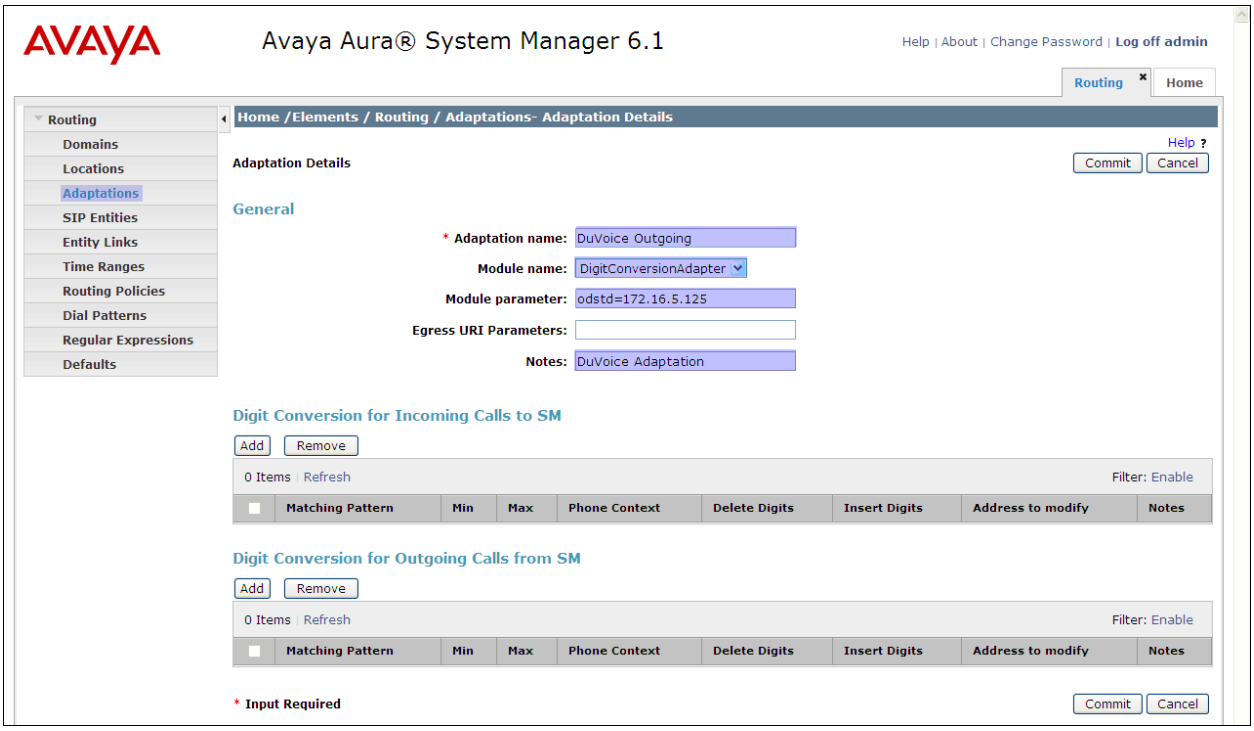
- **Adaptation Name:** Enter a descriptive name for the adaptation.
- **Module Name:** Enter **DigitConversionAdapter**.
- **Module parameter** Enter **odstd=x** where **x** is the IP address of the DuVoice server.

The **odstd=172.16.5.125** module parameter enables the outbound destination domain to be overwritten with the IP address of the DuVoice server. For example, for outbound calls from Avaya to DuVoice, the Request-URI will contain IP address **172.16.5.125** as expected by DuVoice.

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Click **Commit** to save.

The **DuVoice Outgoing** adaptation shown below will later be assigned to the **DuVoice** SIP Entity. This adaptation uses the **DigitConversionAdapter**.



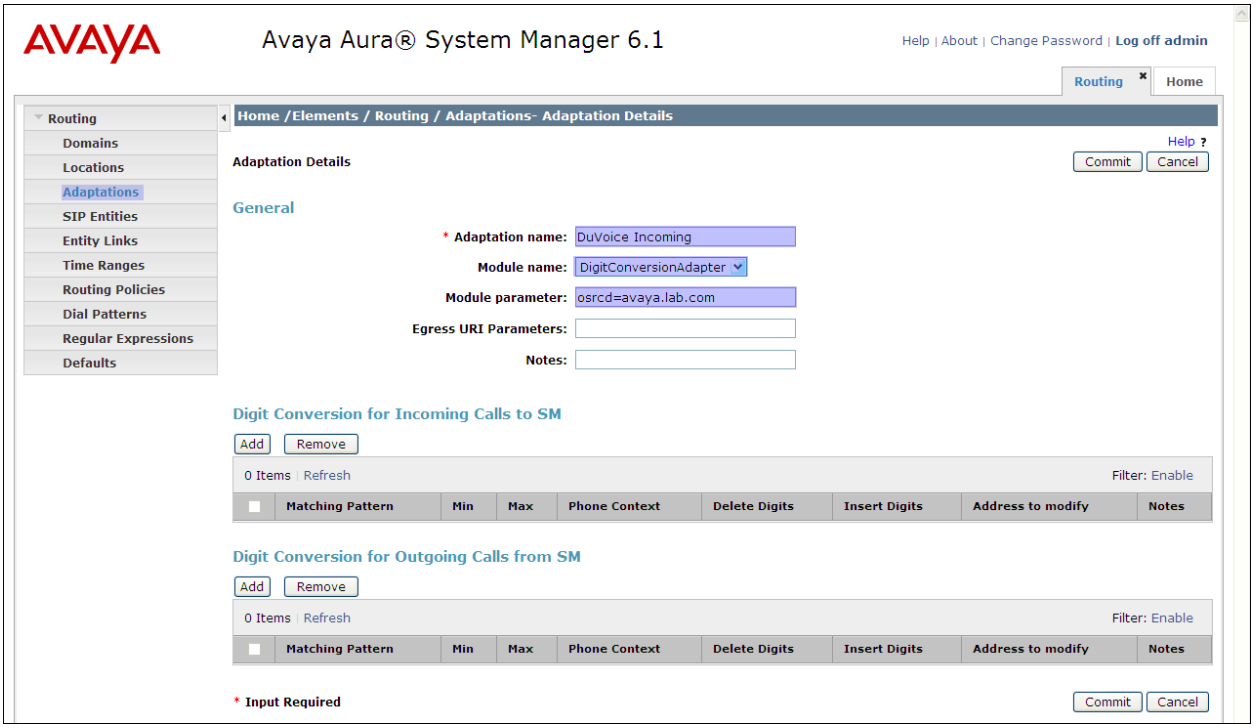
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The adaptation named **DuVoice Incoming** shown below will later be assigned to the SIP Entity for calls destined to Avaya Aura® Communication Manager. This adaptation uses the **DigitConversionAdapter** and specifies the **osrcd=avaya.lab.com** parameter to adapt the domain to the domain name expected by Avaya Aura® Communication Manager. More specifically, this configuration replaces the domain in the From header and P-Asserted-Identity header with the given value. For example, for inbound calls from DuVoice to Avaya, the From header sent to Avaya Aura® Communication Manager will contain the domain **avaya.lab.com** as expected by Avaya Aura® Communication Manager.

Settings for **DuVoice Incoming** Adaptation**:**

In the **General** section, enter the following values. Use default values for all remaining fields:

- **Adaptation Name:** Enter a descriptive name for the adaptation.
- **Module Name:** Enter **DigitConversionAdapter**.
- **Module parameter** Enter **osrcd= x** where **x** is the domain name used.



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6.5. Add SIP Entities

A SIP Entity must be added for Avaya Aura® Session Manager and for each SIP telephony system connected to it which includes Avaya Aura® Communication Manager and the DuVoice server. Navigate to **Routing → SIP Entities** in the left-hand navigation pane and click on the **New** button in the right pane (not shown).

In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling.
- **Type:** Select **Session Manager** for Session Manager, **CM** for Communication Manager or **Other**. Use **Other** for DuVoice.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager**. If applicable, select the **Adaptation Name** created in **Section 6.4** that will be applied to this entity.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Select the time zone for the location above.

To define the ports used by Avaya Aura® Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for **Session Manager** SIP entities.

In the **Port** section, click **Add** and enter the following values. Use default values for all remaining fields:

- **Port:** Port number on which the Session Manager can listen for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain:** The domain used for the enterprise.

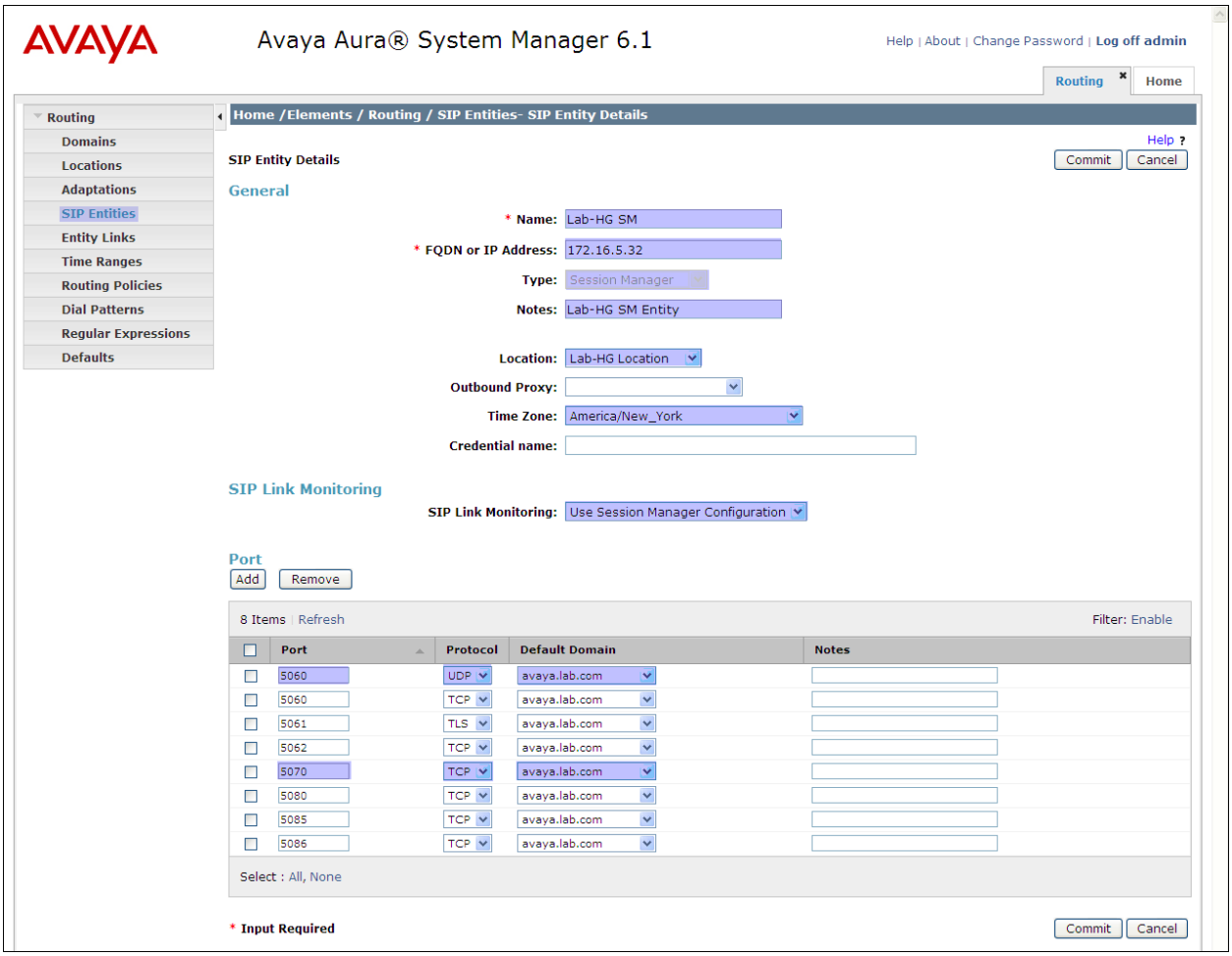
Defaults can be used for the remaining fields. Click **Commit** to save.

For the compliance test, only two Ports were used:

- **5060** with **UDP** for connecting to DuVoice.
- **5070** with **TCP** for connecting to Avaya Aura® Communication Manager.
  Note: In a customer environment the entity link to Avaya Aura® Communication Manager would normally use TLS. For the compliance test, TCP was used to aid in troubleshooting since the signaling traffic would not be encrypted.

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The following screen shows the addition of Session Manager. The IP address of the virtual SM-100 Security Module is entered for **FQDN or IP Address**.

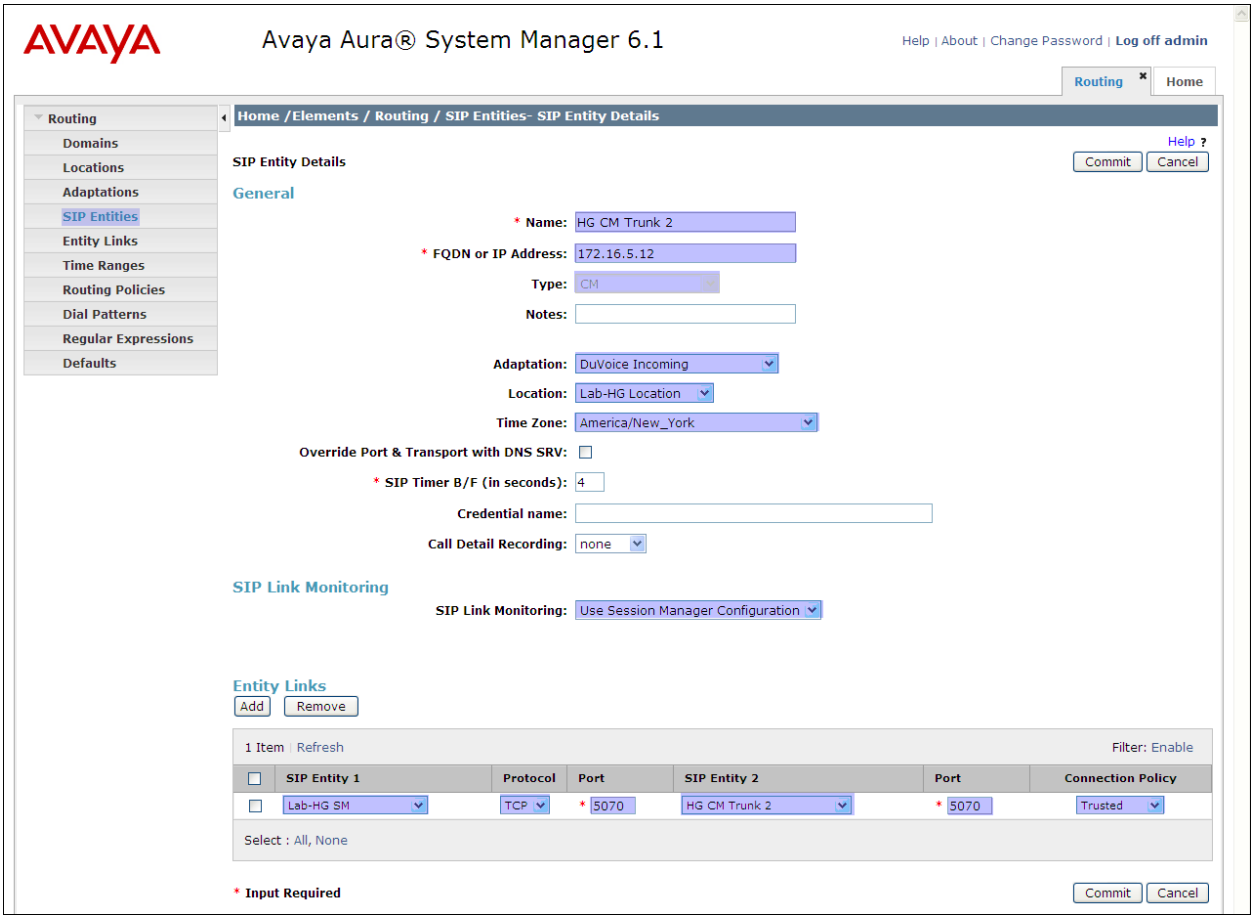


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The following screen shows the addition of Avaya Aura® Communication Manager.

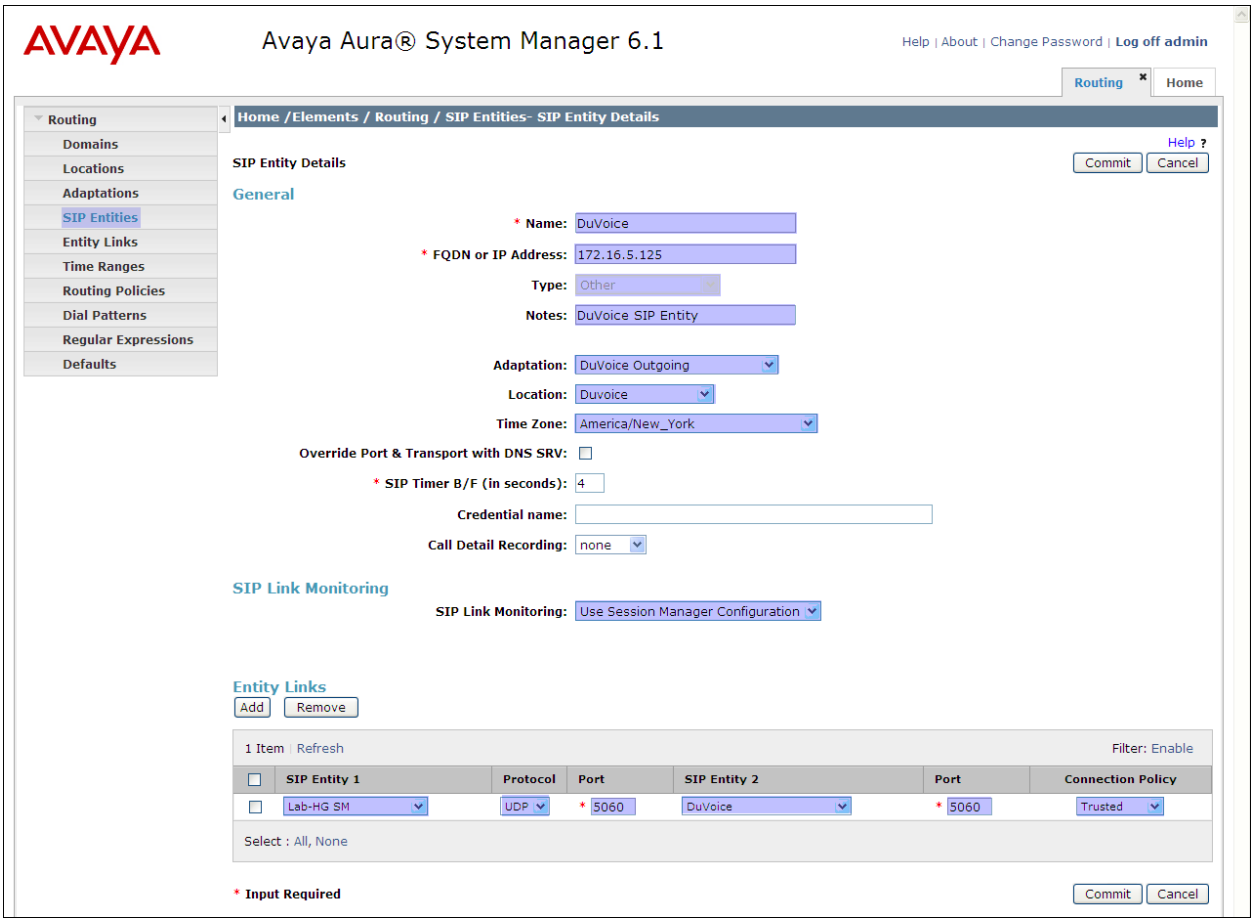
A separate SIP entity for Avaya Aura® Communication Manager, other than the one created for Avaya Aura® Session Manager during installation, is required in other to send traffic to and from Avaya Aura® Communication Manager.

The **FQDN or IP Address** field is set to the IP address of the Avaya S8300D Server running Avaya Aura® Communication Manager. For the **Adaptation** field, select the adaptation module previously defined for dial plan digit manipulation in **Section 6.4**. For the **Location** field, select the location for Avaya Aura® Communication Manager and Avaya Aura® Session Manager previously defined in **section 6.3**.



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The following screen shows the addition of the DuVoice SIP entity. The **FQDN or IP Address** field is set to the IP address of the DuVoice server (see **Figure 1**). Set the **Type** to **Other** for DuVoice. For the **Adaptation** field, select the adaptation module previously defined for dial plan digit manipulation created in **Section 6.4**. For the **Location** field, select the location for DuVoice previously defined in **section 6.3**.



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6.6. Add Entity Links

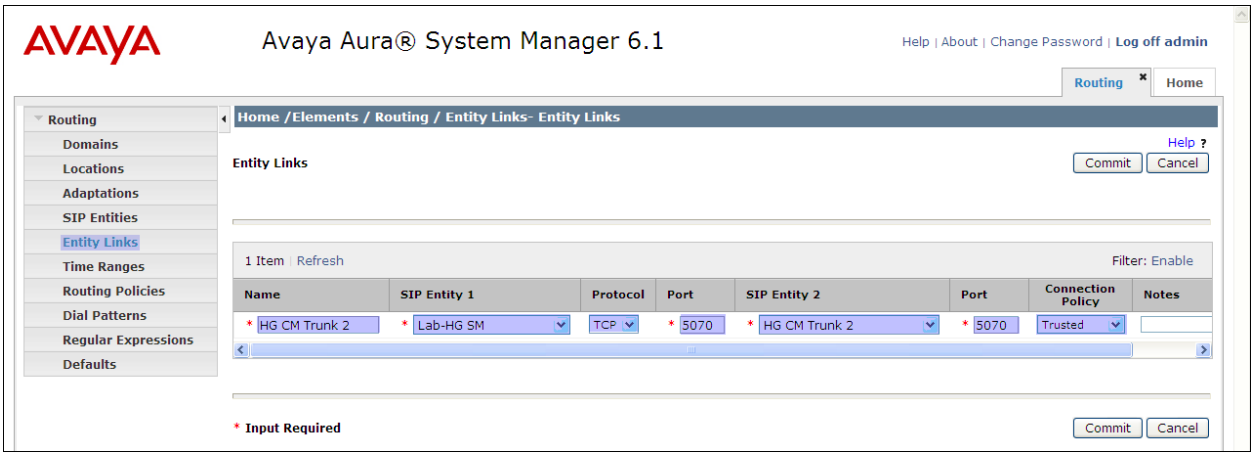
A SIP trunk between Avaya Aura® Session Manager and a telephony system is described by an entity link. Two entity links were created; one to Avaya Aura® Communication Manager to be used only for DuVoice traffic to and from Avaya Aura® Communication Manager and one to the DuVoice server. To add an entity link, navigate to **Routing → Entity Links** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). Fill in the following fields in the new row that is displayed:

- **Name:** Enter a descriptive name.
- **SIP Entity 1:** Select the Avaya Aura® Session Manager.
- **Protocol:** Select the transport protocol used for this link.
- **Port:** Port number on which Session Manager will receive SIP requests from the far-end. For Communication Manager, this must match the **Far-end Listen Port** defined on the Communication Manager signaling group in **Section 5.5**.
- **SIP Entity 2:** Select the name of the other system. For Communication Manager, select the Communication Manager SIP entity defined in **Section 6.5**.
- **Port:** Port number on which the other system receives SIP requests from the Session Manager. For Communication Manager, this must match the **Near-end Listen Port** defined on the Communication Manager signaling group in **Section 5.5**.
- **Trusted:** Select **Trusted**. Note: If this box is not checked, calls from the associated SIP Entity specified in **Section 6.5** will be denied.

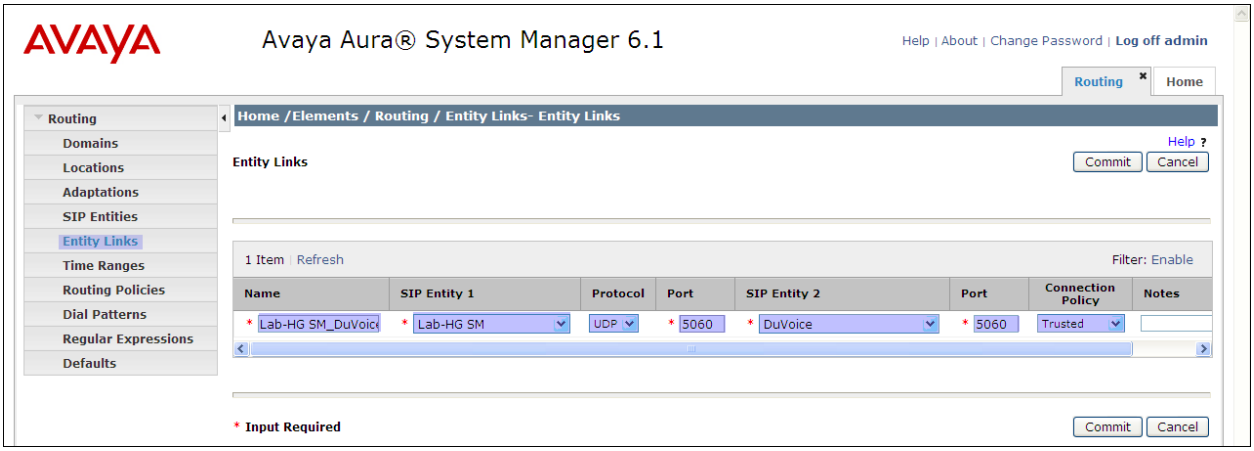
Click **Commit** to save. The following screens illustrate the entity links to Avaya Aura® Communication Manager. It should be noted that in a customer environment the entity link to Avaya Aura® Communication Manager would normally use TLS. For the compliance test, TCP was used to aid in troubleshooting since the signaling traffic would not be encrypted. The protocol and ports defined here must match the values used on the Avaya Aura® Communication Manager signaling group form in **Section 5.5**.

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Entity Link to Avaya Aura® Communication Manager:



Entity Link to the DuVoice server:



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6.7. Add Routing Policies

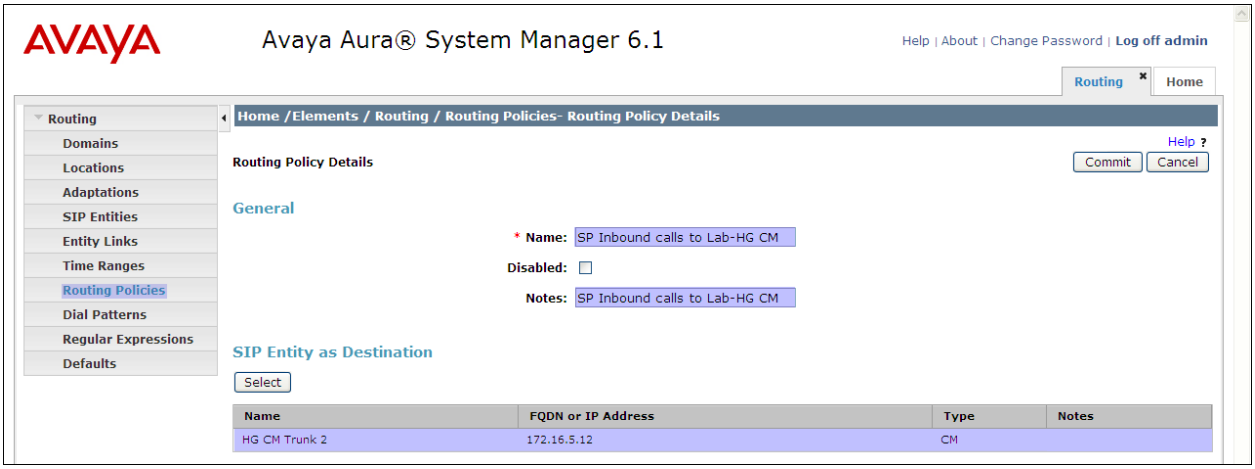
Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.5**. Two routing policies must be added: one for Avaya Aura® Communication Manager and one for DuVoice. To add a routing policy, navigate to **Routing** → **Routing Policies** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). The following screen is displayed. Fill in the following:

In the **General** section, enter the following values. Use default values for all remaining fields:

- **Name:** Enter a descriptive name.
- **Notes:** Add a brief description (optional).

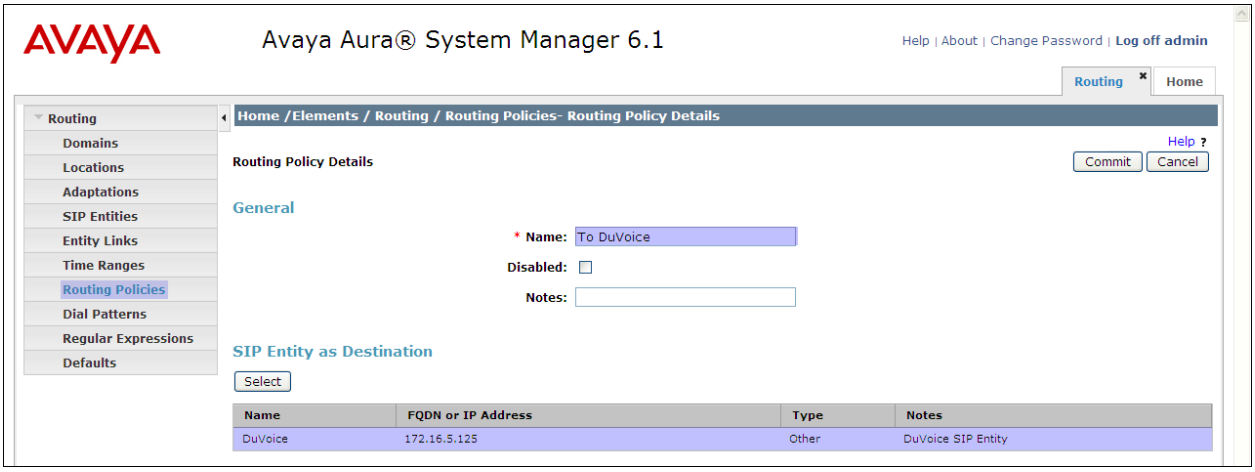
In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Select the appropriate SIP entity to which this routing policy applies and click **Select**. The selected SIP entity displays on the **Routing Policy Details** page as shown below. Use default values for the remaining fields. Click **Commit** to save.

The following screens show the routing policy for Avaya Aura® Communication Manager.



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The following screens show the routing policy for DuVoice.



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6.8. Add Dial Patterns

Dial Patterns are needed to route calls through Avaya Aura® Session Manager. For the compliance test, dial patterns were needed to route calls from Avaya Aura® Communication Manager to DuVoice and vice versa. Dial Patterns define which route policy will be selected for a particular call based on the dialed digits, destination domain and originating location. To add a dial pattern, navigate to **Routing → Dial Patterns** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). Fill in the following, as shown in the screens below:

In the **General** section, enter the following values. Use default values for all remaining fields:

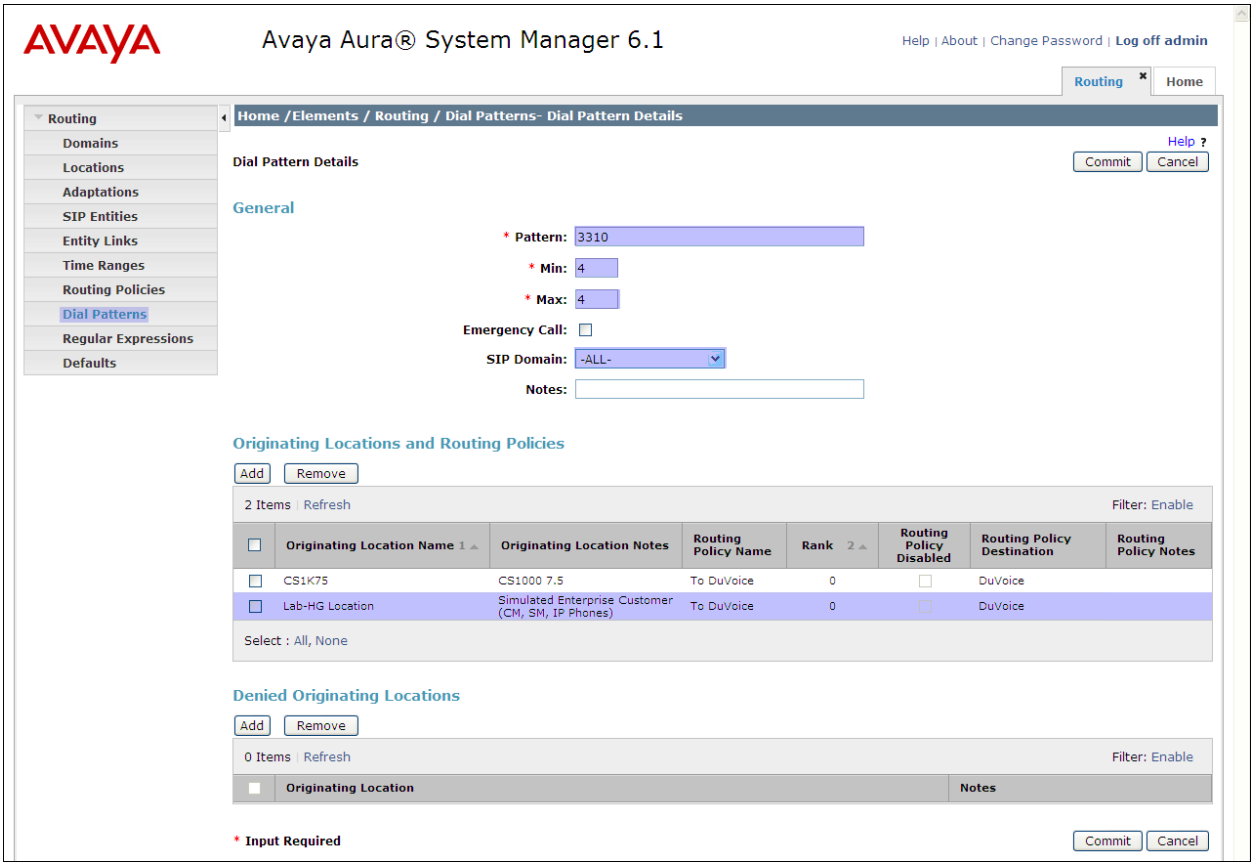
- **Pattern:** Enter a dial string that will be matched against the Request-URI of the call.
- **Min:** Enter a minimum length used in the match criteria.
- **Max:** Enter a maximum length used in the match criteria.
- **SIP Domain:** Enter the destination domain used in the match criteria.
- **Notes:** Add a brief description (optional).

In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria. Lastly, select the routing policy from the list that will be used to route calls that match the specified criteria. Click **Select**.

Default values can be used for the remaining fields. Click **Commit** to save.

Examples of the dial patterns used for the compliance testing are shown below. The first example shows dial pattern **3310** for the voice mail hunt group which routes the calls to DuVoice, have a **SIP Domain** of **all,** an originating location name of **Lab-HG Location,** and uses routing policy name **To DuVoice.**

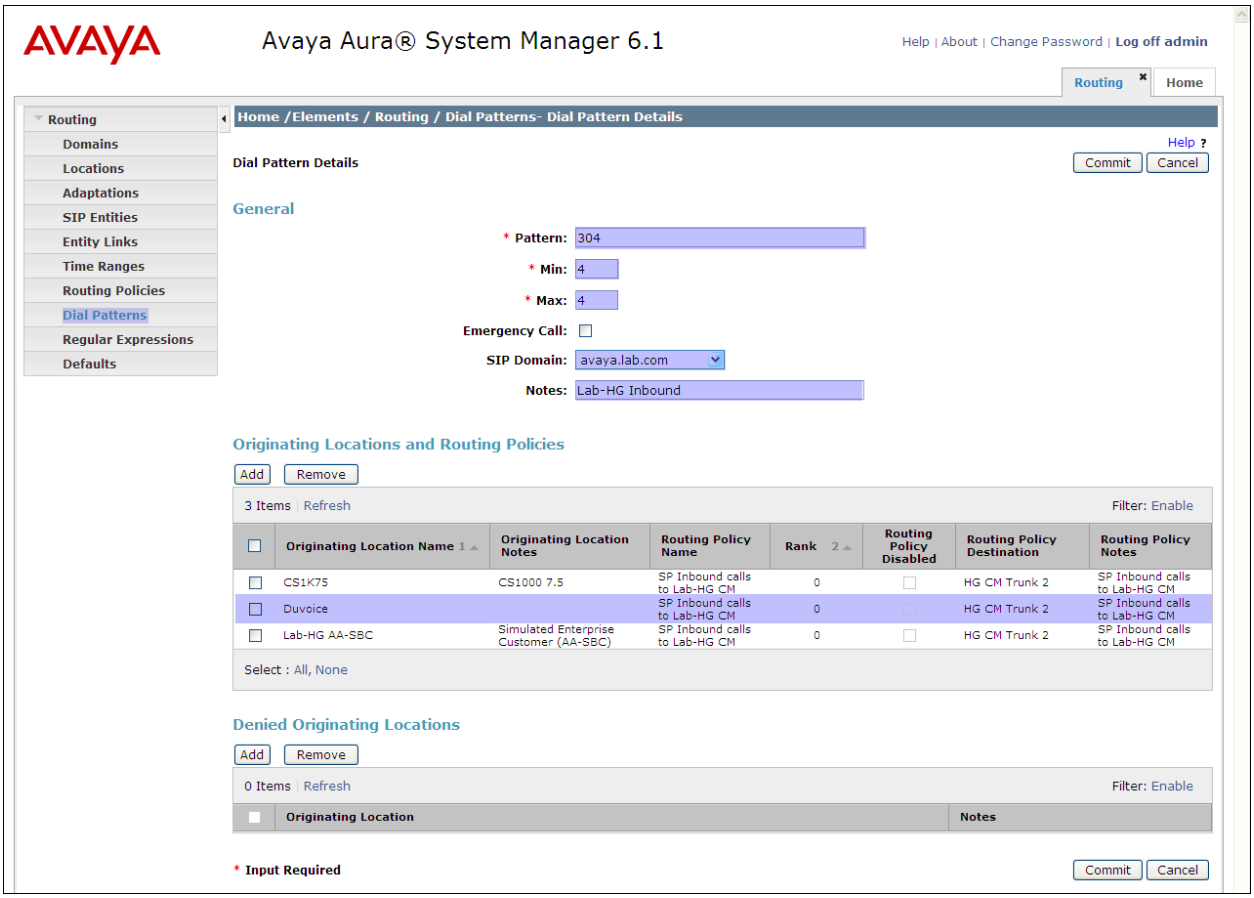
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The following dial pattern example used for the compliance testing uses dial pattern **304**, these are the first 3 digits of the four digit extensions numbers assigned to the stations. Have a **SIP Domain** of **avaya.lab.com**, an originating Location name of **DuVoice**, and a routing policy name of **SIP inbound calls to Lab-HG CM**.

Note: Only 4 digits extension numbers assigned to stations were used for the compliance testing (e.g., 3040, 3041, etc.)



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6.9. Add/View Avaya Aura® Session Manager

The creation of an Avaya Aura® Session Manager element provides the linkage between Avaya Aura® System Manager and Avaya Aura® Session Manager. This was most likely done as part of the initial Avaya Aura® Session Manager installation. To add an Avaya Aura® Session Manager, navigate to **Elements → Session Manager → Session Manager Administration** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). If the Avaya Aura® Session Manager already exists, click **View** (not shown) to view the configuration. Enter/verify the data as described below and shown in the following screen:

In the **General** section, enter the following values:

- **SIP Entity Name:** Select the SIP Entity created for Session Manager.
- **Description**: Add a brief description (optional).
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

In the **Security Module** section, enter the following values:

- **SIP Entity IP Address:** Should be filled in automatically based on the SIP Entity Name. Otherwise, enter IP address of Session Manager signaling interface.
- **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 **Save** (not shown) to add this Avaya Aura® Session Manager. The screen below shows the Avaya Aura® Session Manager values used for the compliance test.

The screen below shows the Avaya Aura® Session Manager values used for the compliance test.

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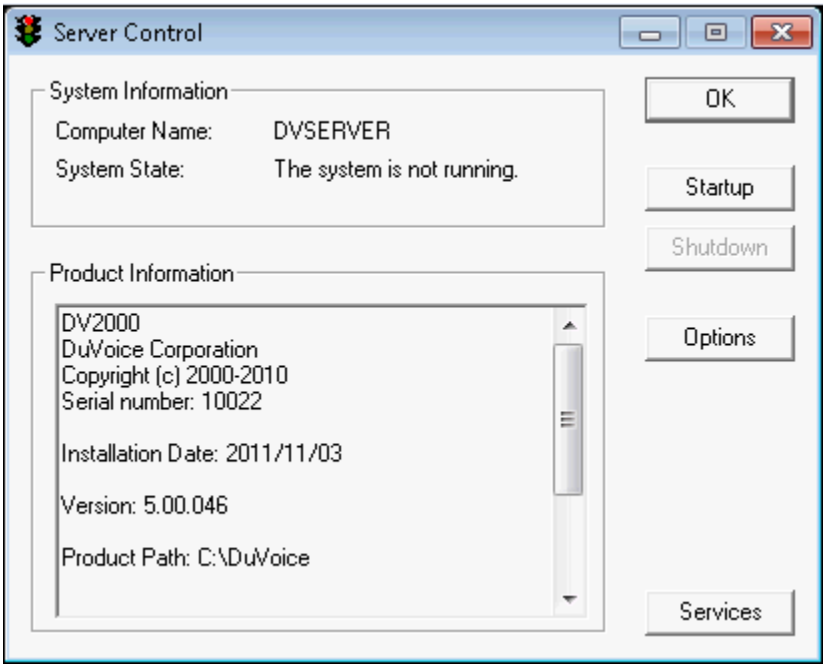
7. Configure DuVoice

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

- Shutdown server
- Administer setup wizard
- Administer connectors
- Verify System Configuration Setting
- Startup the server
- Administer mailboxes
- Verify Port Activity

7.1. Shutdown Server

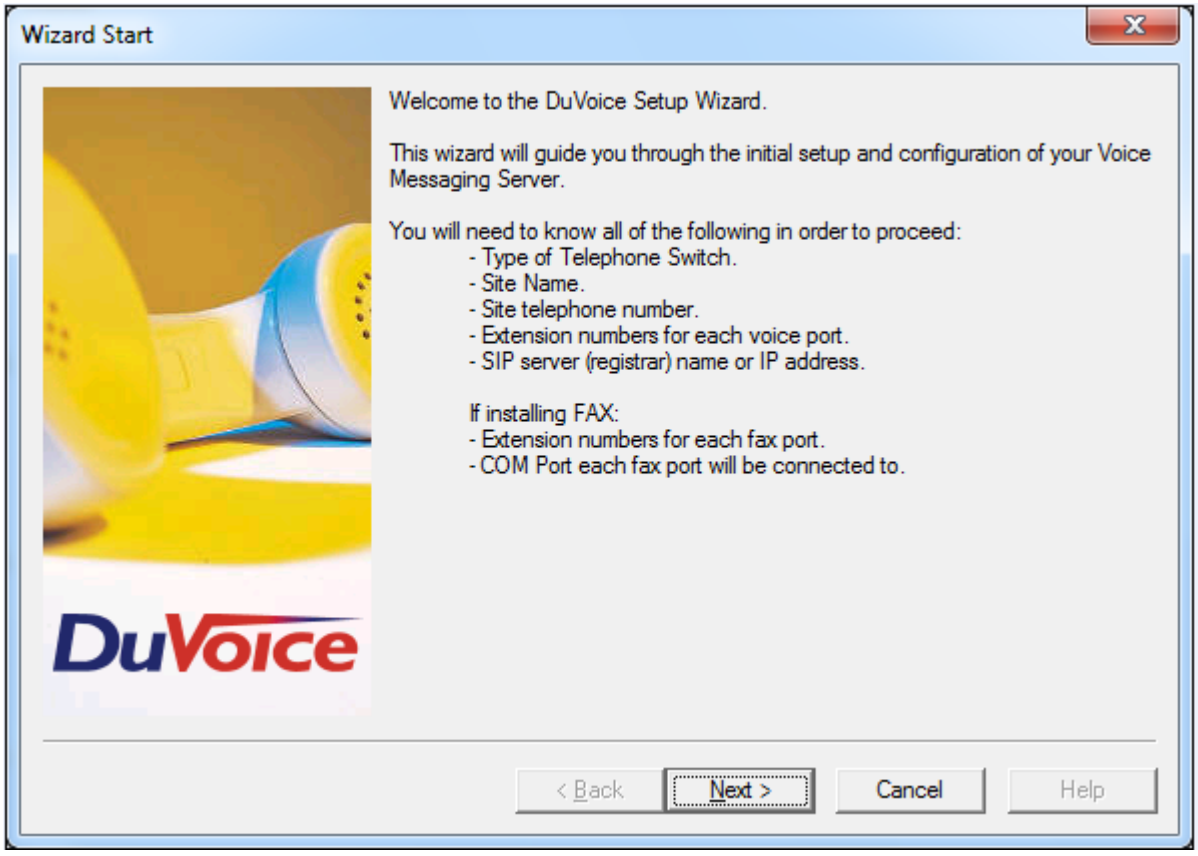
From the DuVoice server, select **Start > All Programs > DuVoice > Server Control**. The **Server Control** screen is displayed. Select **Shutdown** to stop the server, in order to make subsequent configuration changes.



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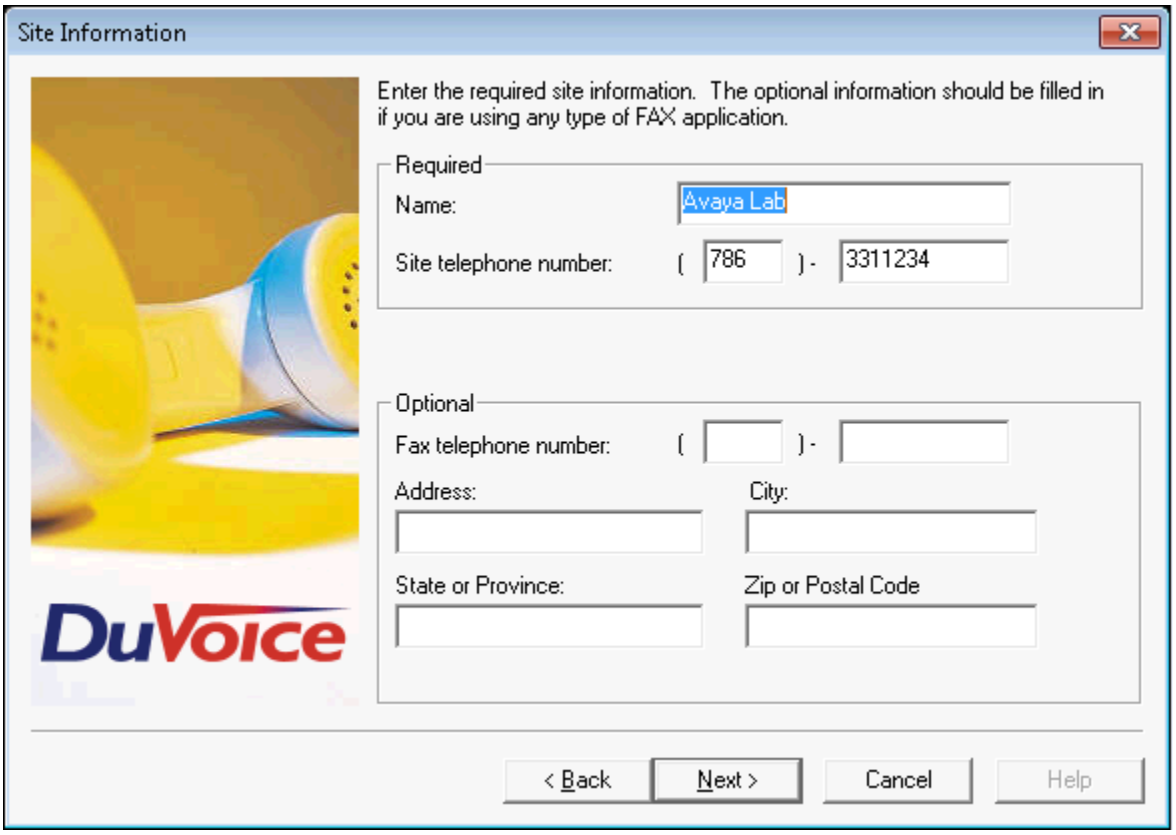
7.2. Administer Setup Wizard

From the DuVoice server, select **Start > All Programs > DuVoice > System Configuration**. The **Wizard Start** screen is displayed upon initial access. Click **Next**.



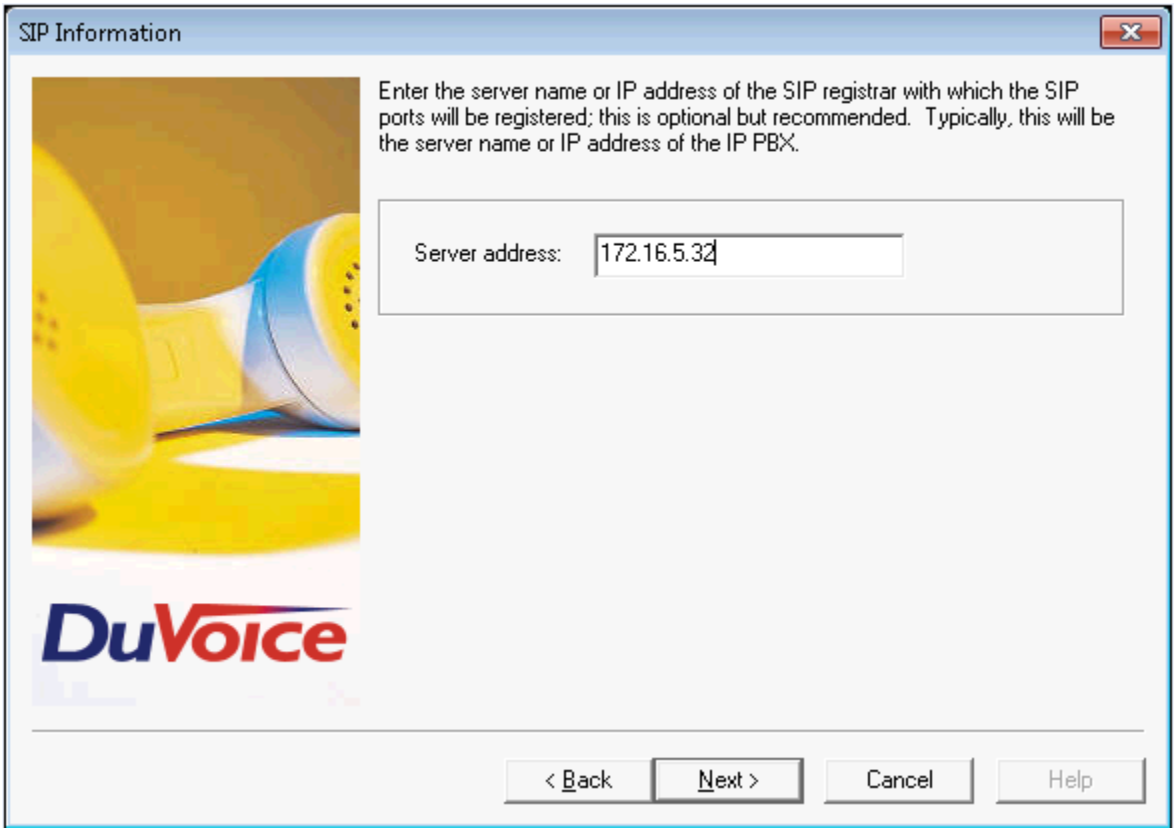
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The **Site Information** screen is displayed next. Enter desired values for **Name** and **Site telephone number**, and click **Next**.



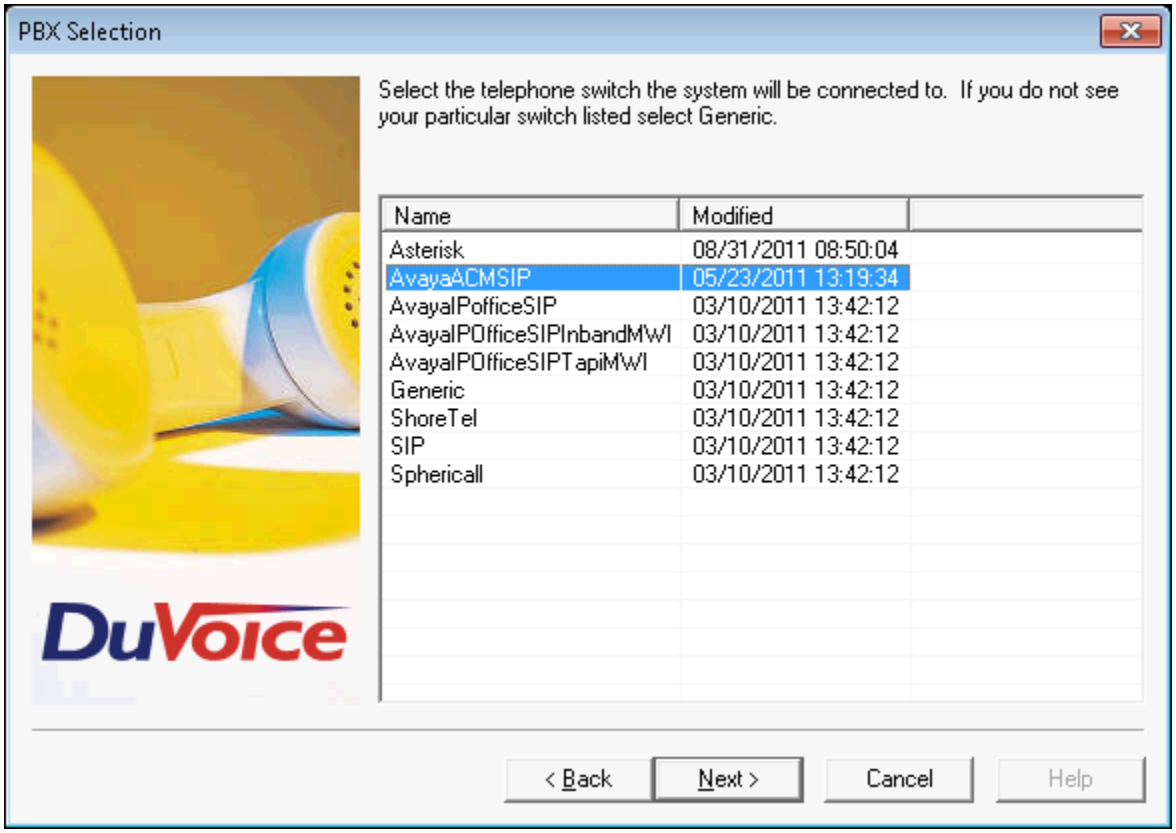
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The **SIP Information** screen is displayed next. Enter the IP address of **Session Manager**, and click **Next**.



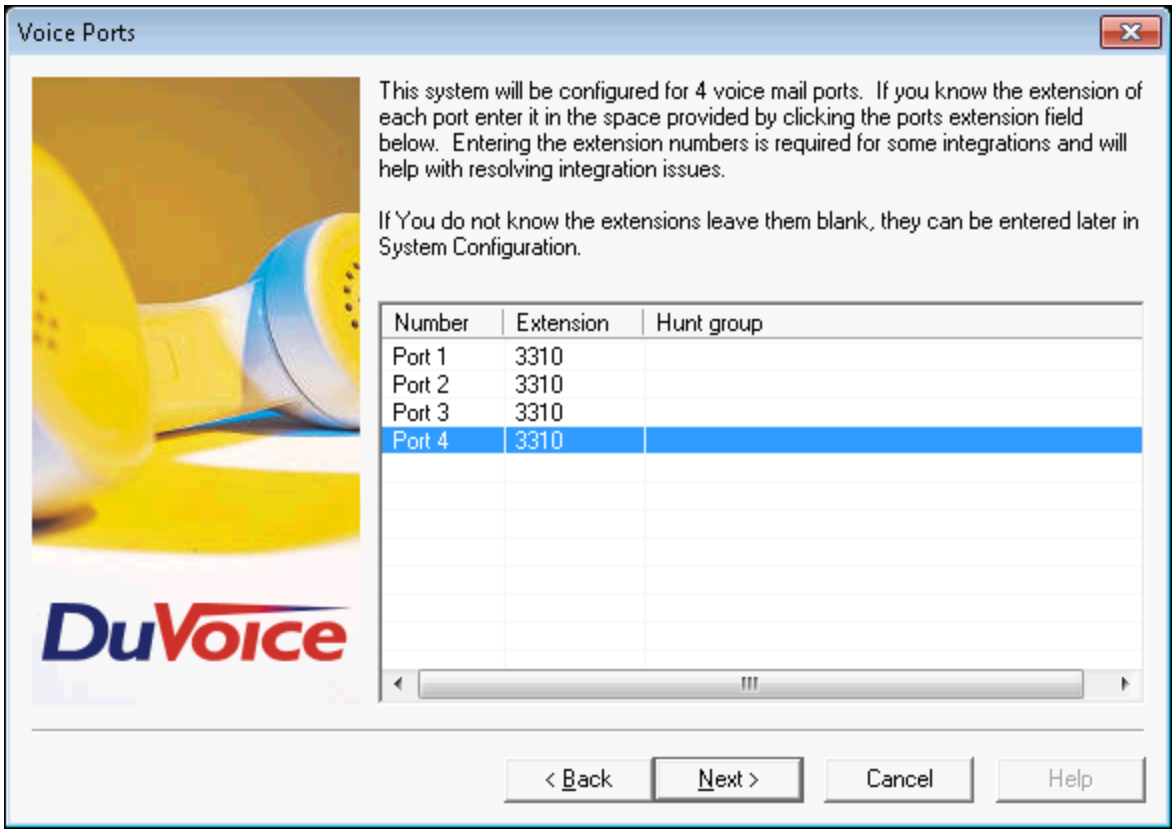
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The **PBX Selection** screen is displayed next. Select the **AvayaACMSIP** entry, and click **Next**.



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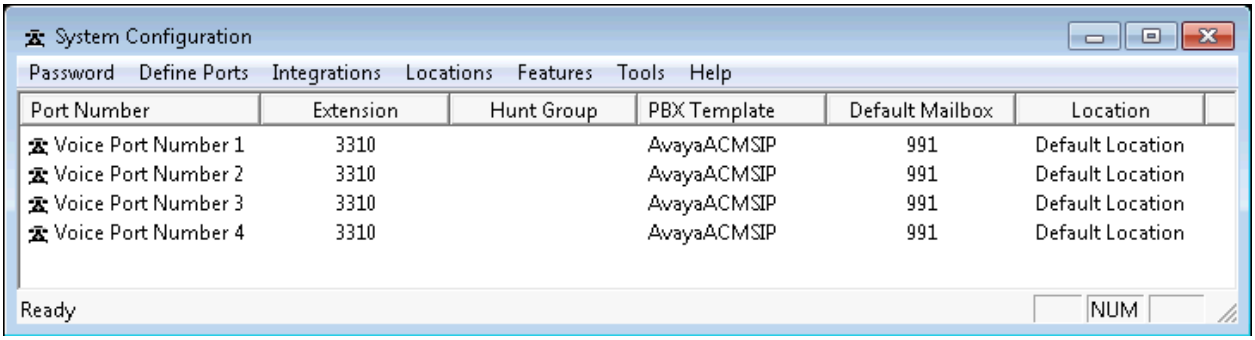
The **Voice Ports** screen is displayed next. Under the **Extension** column for each port number, enter the hunt group pilot number configured in **Section 5.8**. The same hunt group pilot number is assigned to all of the ports, **3310** was used for the sample configuration. Click **Next**, followed by **Finish** in the subsequent screen to complete the wizard.



7.3. Administer Connectors

From the DuVoice server, select **Start > All Programs > DuVoice > System Configuration**.

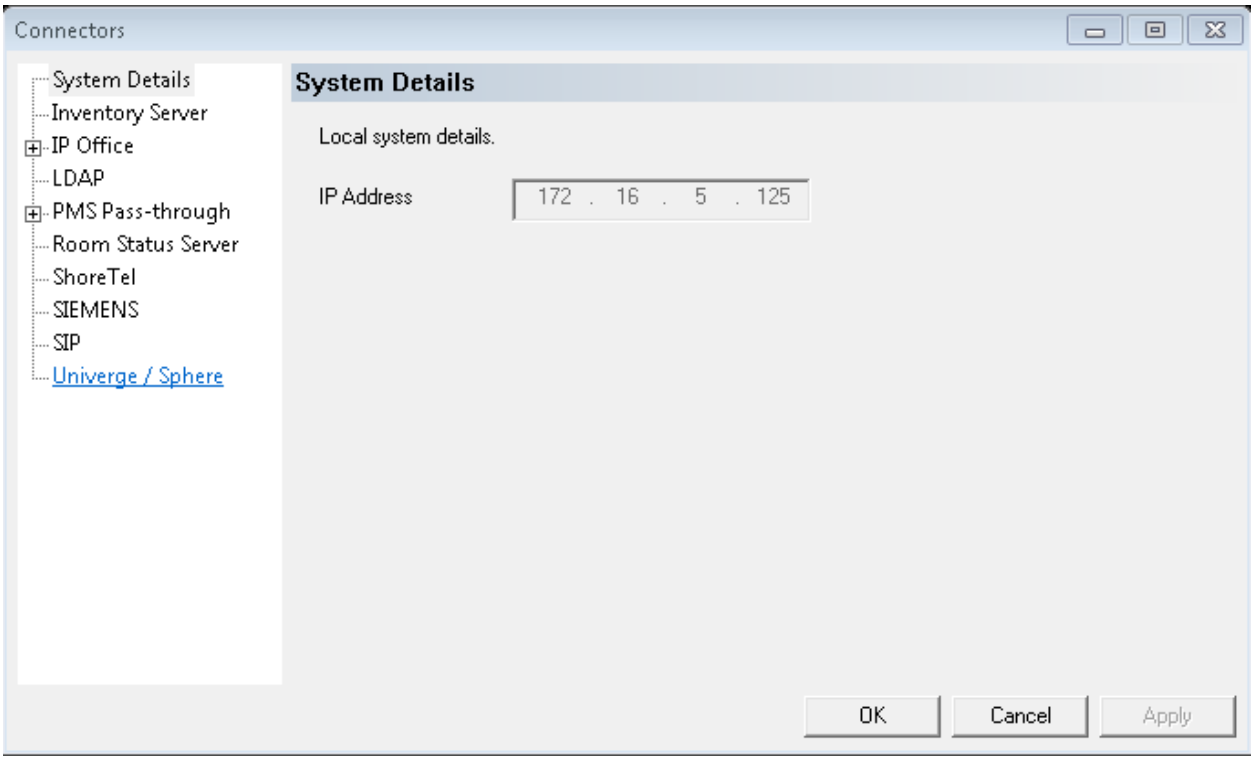
The **System Configuration** screen is displayed:



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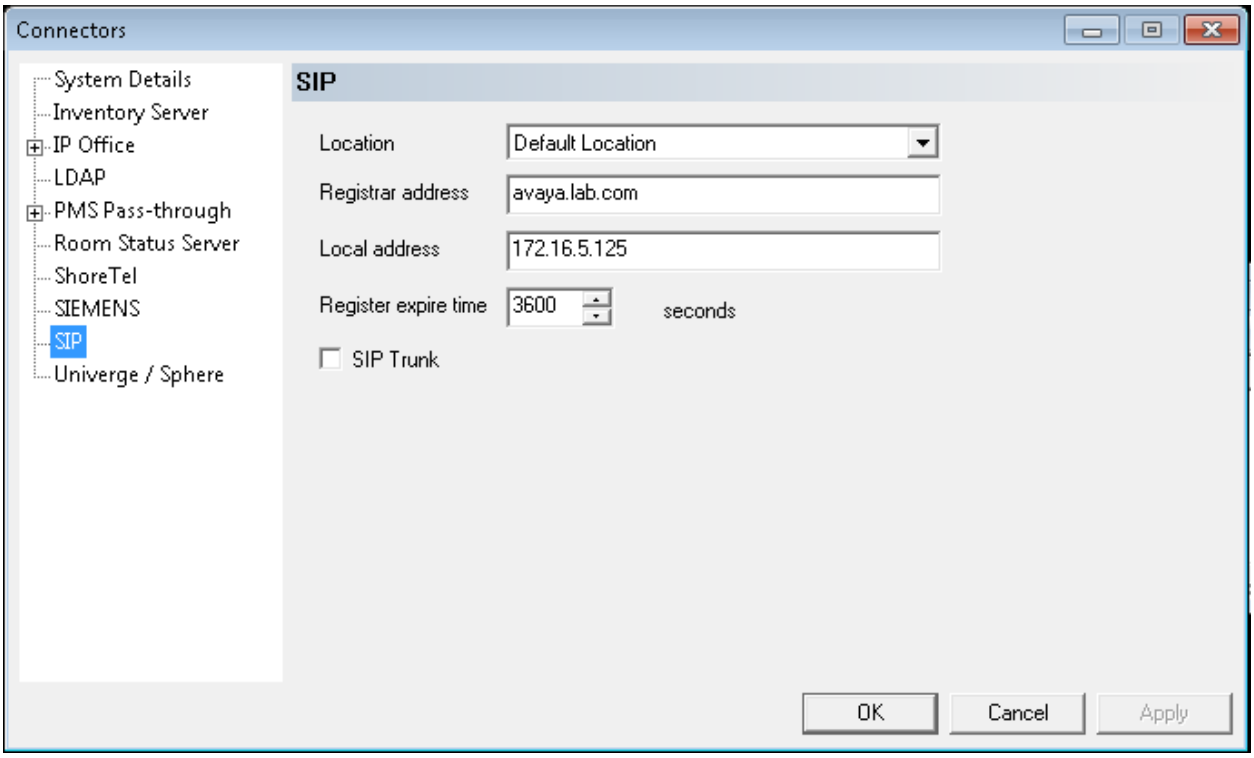
Select **Features > Connectors** from the top menu.

The **Connectors** screen is displayed:



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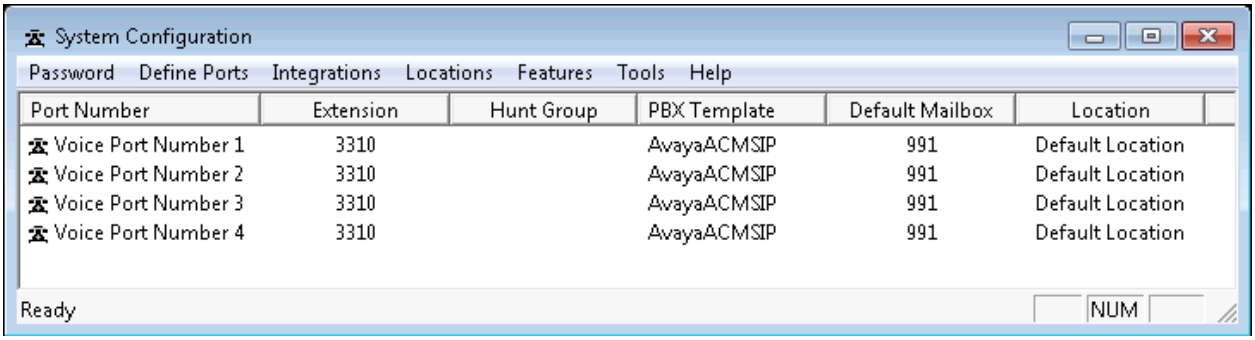
Select **SIP** from the left pane, the **SIP** screen for **Connectors** is displayed next. For the **Registrar address** enter the SIP domain name defined under **Section 6.2**. **Note**: The IP address of Session Manager may be required under the **Registrar address** instead of the domain name; it depends on the enterprise capabilities to handle DNS queries for the DuVoice server. For **Local address**, enter the IP address of the DuVoice server. Set the **Register expire time** to the DuVoice recommended value of **3600**. Verify that SIP Trunk is not checked.



7.4. Verify the System Configuration Setting

Select **Start> All Programs> DuVoice> System Configuration**.

The **System Configuration** screen is displayed:

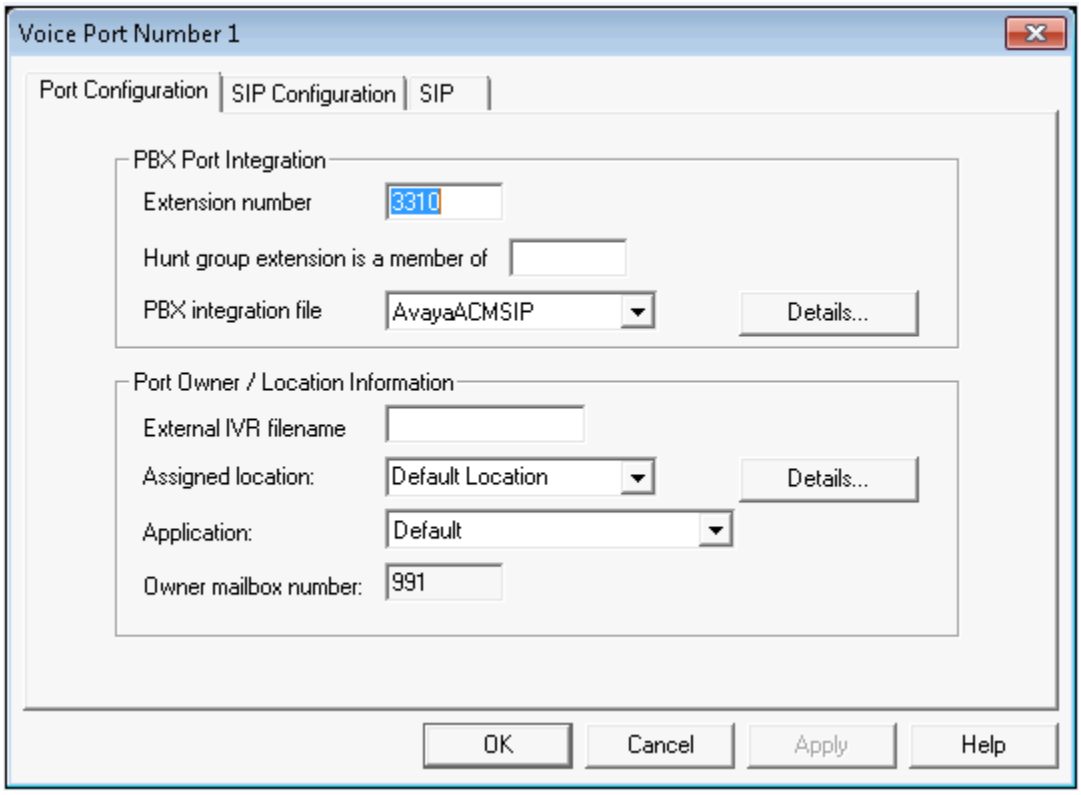


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Double click on **Voice Port Number 1**.

Under the **Port Configuration** tab, verify the following values. Use default values for all remaining fields:

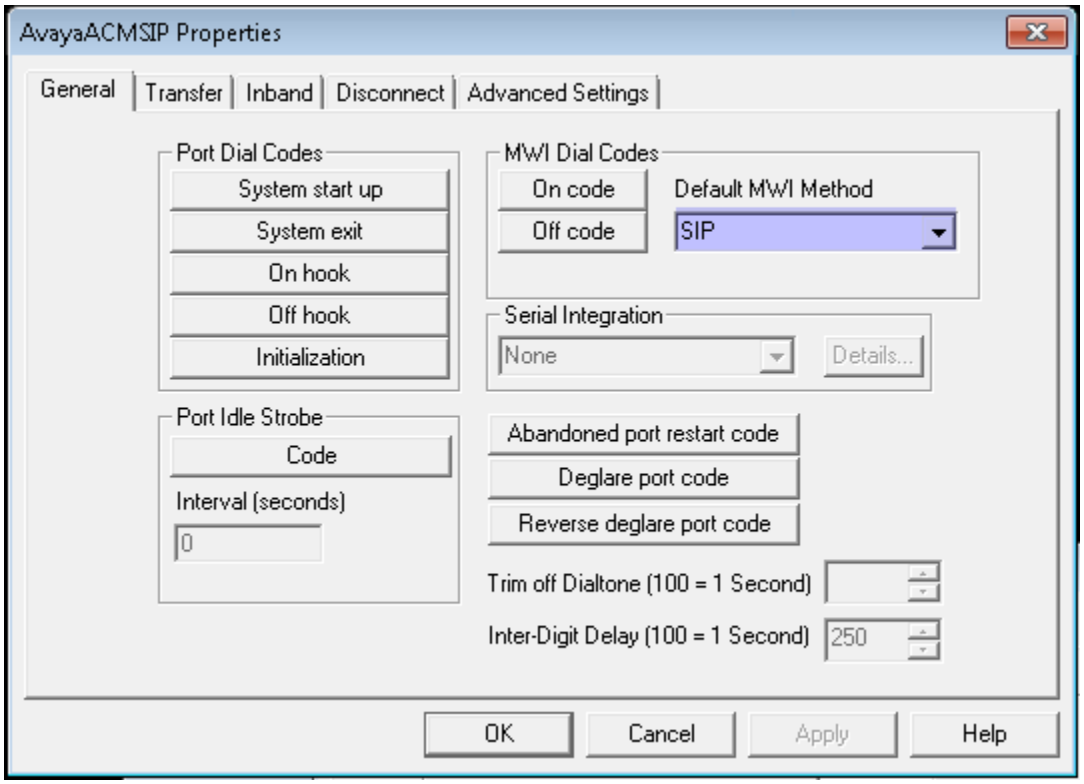
- **Extension number:** Verify that the extension number is set to the hunt group pilot number configured in **Section 5.8**
- **PBX integration file:** Verify that the PBX integration file is set to **AvayaACMSIP**.



Click on **Details**, shown on the above screen under **PBX Port Integration**.

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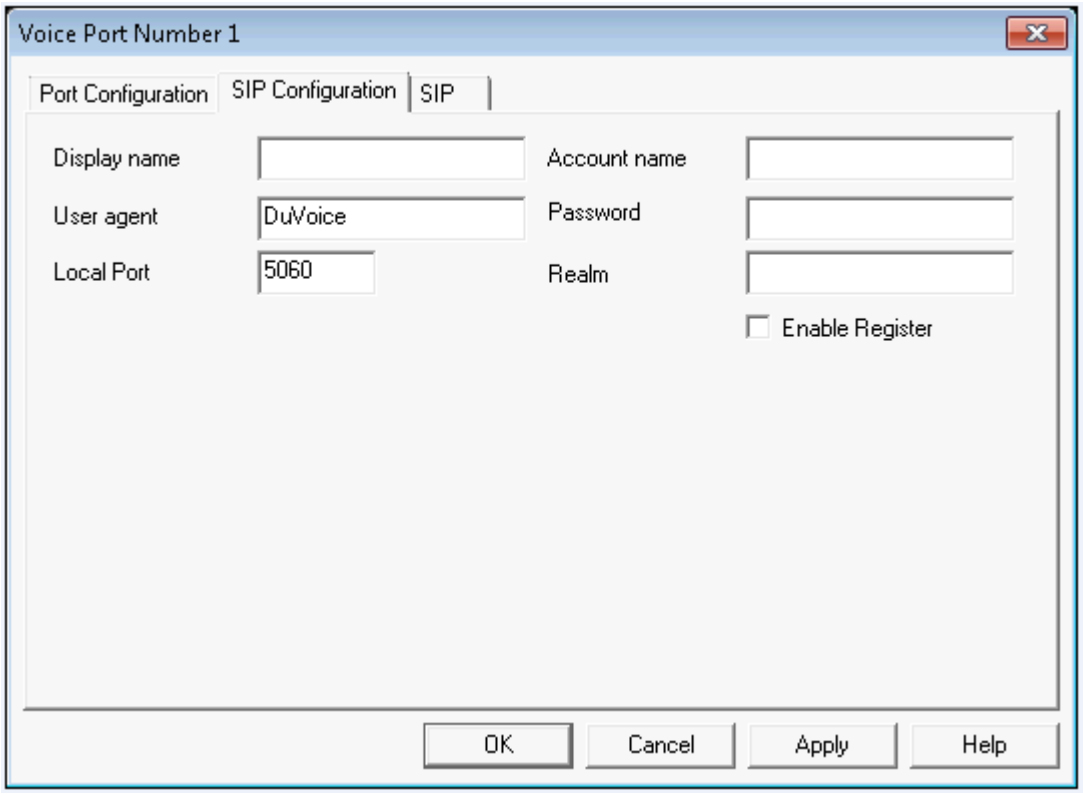
Verify that the **Default MWI Method** is set to **SIP**. Click on **Apply** and **OK** if changes were made.



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Under the **SIP Configuration** tab verify the **SIP** Configuration settings.

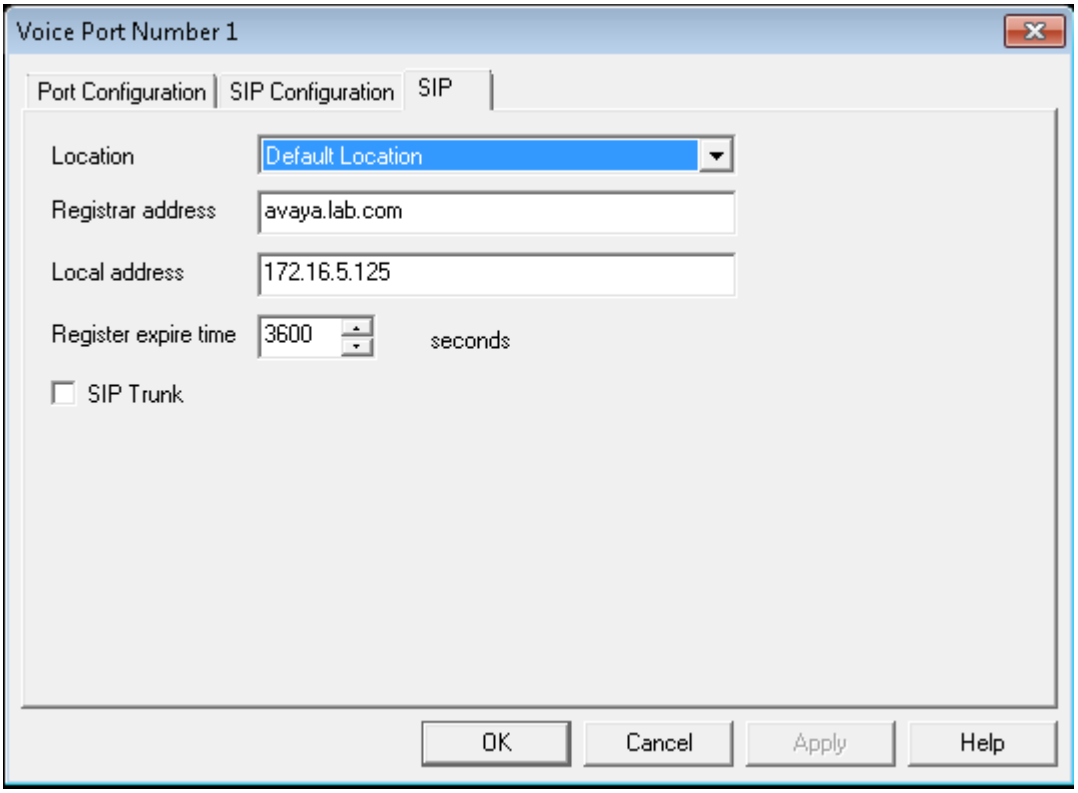
- **Local Port:** Verify that the local port is set to 5060 configured in **Section 6.5.**
- **Enable Register:** Verify that **Enable Register** is not checked.



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Under the **SIP** tab verify additional SIP setting.

- **Registrar address:** Verify that the Registrar address is set to the SIP domain name configured in **Section 6.2. Note**: The IP address of Session Manager may be required under the **Registrar address** instead of the domain name; it depends on the enterprise capabilities to handle DNS queries for the DuVoice server.
- **Local address:** This field should contain the IP address of the DuVoice server.
- **Register expire time:** Verify that the Register expire time is set to **3600**
- **SIP trunk:** Verify that SIP Trunk is not checked.

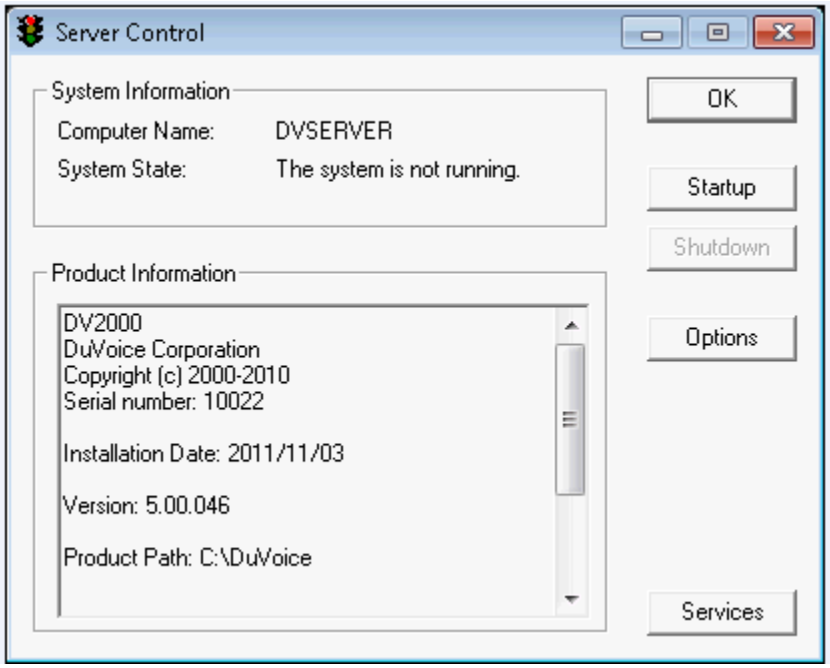


Repeat this section for each remaining Voice Port Number **2-4** for the sample configuration.

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7.5. Startup the Server

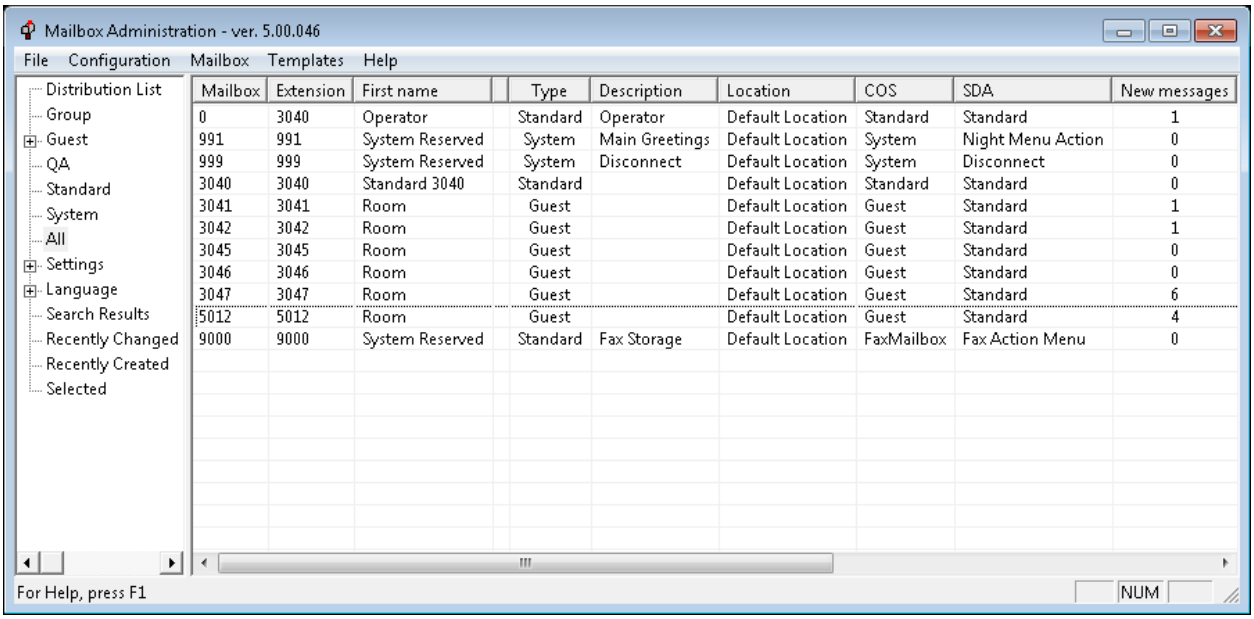
From the DuVoice server, select **Start > All Programs > DuVoice > Server Control**. The **Server Control** screen is displayed. Select **Startup** to start the server.



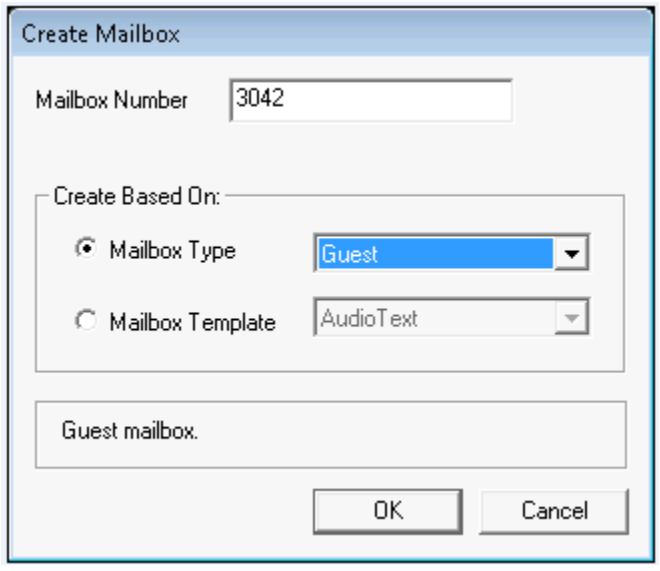
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7.6. Administer Mailboxes

From the DuVoice server, select **Start > All Programs > Mailbox Administration**.
The **Mailbox Administration** screen is displayed. Select **Mailbox > Create** from the top menu.

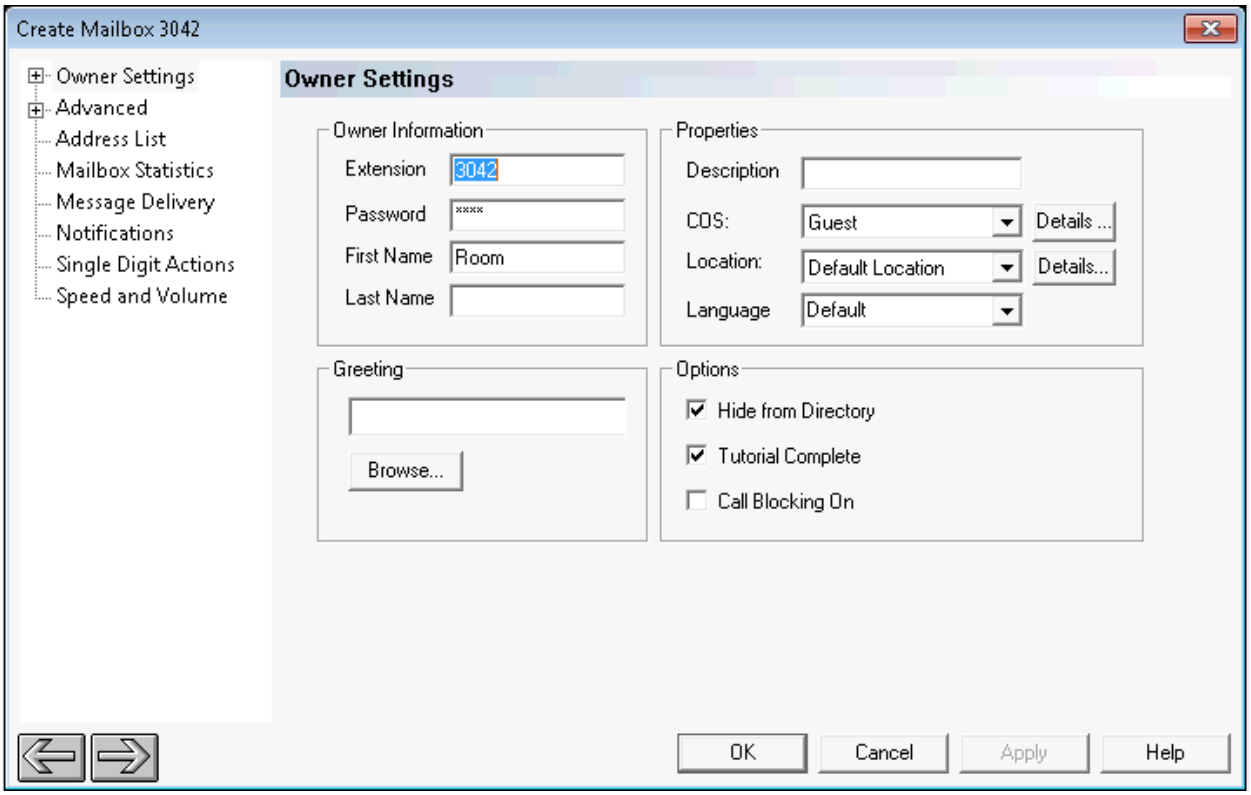


The **Create Mailbox** screen is displayed next. For **Mailbox Number**, enter the first voicemail user extension from **Section 5.10**, in this case "3042" was created. For **Mailbox Type**, select "Guest" for guest users, and "Standard" for front desk and staff users.



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The **Create Mailbox 3042** screen is displayed next. Enter desired values for **Password**, **First Name**, and **Last Name**, and retain the default values in the remaining fields.



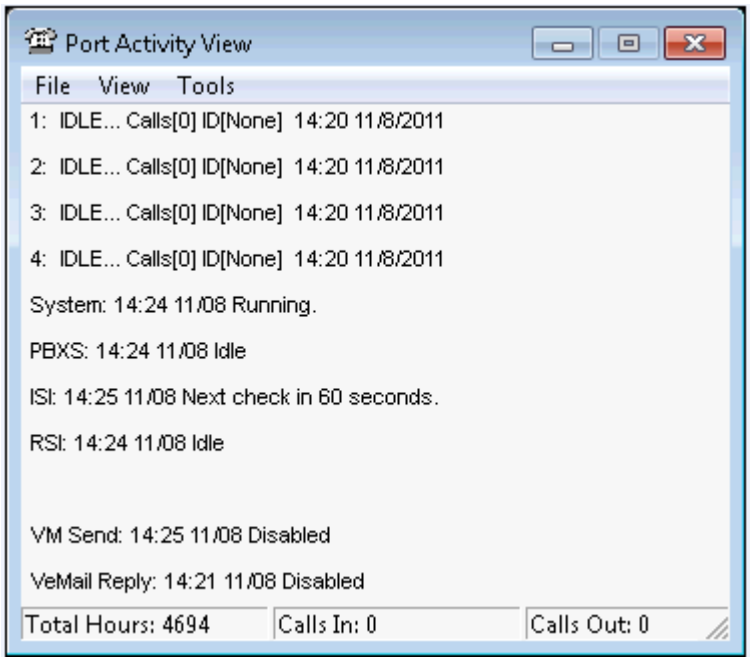
Repeat this section for all voicemail users.

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7.7. Verify Port Activity

From the DuVoice server, select **Start > All Programs > Port Activity**.

The **Port Activity View** screen is displayed. Verify that all configured ports are **IDLE** and ready to accept calls.



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8. Verification and Troubleshooting

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful troubleshooting commands that can be used to troubleshoot the solution.

Verification Steps:

- **Automated Attendant:**
  Place a call to the hunt group pilot number from a guest extensions defined in the DuVoice server, when asked enter a valid guest extension (defined in the DuVoice server) to be transferred to. Verify that the transfer takes place, ring back and speech path in both directions.

- **Voice Mail:**
  Place a call to reach a guest, do not answer the call. Verify the caller hears the system greeting, leave a voice message. Verify the MWI is turned on at the guest telephone. Make a call from the guest extension to the hunt group pilot number, Verify the greeting is played and that the message can be retrieved. Verify the MWI is turned off.

- **Wakeup call:**
  From a guest extension call the hunt group pilot number to schedule a wakeup call. Verify that the wakeup call takes place at the scheduled time.

Troubleshooting Steps:

Avaya Aura® Communication Manager:

- **list trace station** <extension number> - Traces calls to and from a specific station.
- **list trace tac** <trunk access code number> - Trace calls over a specific trunk group.
- **status station** <extension number> - Displays signaling and media information for an active call on a specific station.
- **status trunk** <trunk group number> - Displays trunk group information.
- **status trunk** <trunk group number/channel number> - Displays signaling and media information for an active trunk channel.

Avaya Aura® Session Manager:

- **Call Routing Test** - The Call Routing Test verifies the routing for a particular source and destination. To run the routing test, navigate to **Elements → Session Manager → System Tools → Call Routing Test**. Enter the requested data to run the test.
- **traceSM -x** – Avaya Aura® Session Manager command line tool for traffic analysis. Login to the Avaya Aura® Session Manager management interface to run this command.

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9. Conclusion

These Application Notes describe the configuration steps required for the DuVoice DV2000 Voice Messaging System to operate with Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and various Avaya endpoints. DuVoice passed compliance testing.

10. References

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

[1] *Installing and Configuring Avaya Aura® System Platform*, Release 6.0.3, February 2011.
[2] *Administering Avaya Aura® System Platform*, Release 6.0.3, February 2011.
[3] *Installing and Configuring Avaya Aura® Communication Manager, Release 6.0.1, December 2010.*
[4] *Administering Avaya Aura® Communication Manager*, Release 6.0, June 2010, Document Number 03-300509.
[5] *Avaya Aura® Communication Manager Feature Description and Implementation, Release 6.0* June 2010, *D*ocument Number 555-245-205.
[6] *Installing and Upgrading Avaya Aura® System Manager 6.1*, November 2010.
[7] *Installing and Configuring Avaya Aura® Session Manager*, Release 6.1, April 2011. Number 03-603473.
[8] *Administering Avaya Aura® Session Manager,* Release 6.1, November 2010, Document Number 03-603324.
[9] *DuVoice DV2000 Messaging Products System Reference Guide Version 5.00.044 – Available on the DuVoice DV2000 server as part of the installation.*

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