Avaya Solution & Interoperability Test Lab

Application Notes for Nuance SpeechAttendant 12.1 with Avaya Aura® Session Manager 7.0 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Nuance SpeechAttendant 12.1 to interoperate with Avaya Aura® Session Manager 7.0 and Avaya Aura® Communication Manager 7.0 using SIP trunks. Nuance SpeechAttendant automates call routing by asking callers to speak the name or dial the extension of a destination.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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

1. Introduction

These Application Notes describe the configuration steps required for Nuance SpeechAttendant 12.1 to interoperate with Avaya Aura® Session Manager 7.0 and Avaya Aura® Communication Manager 7.0 using SIP trunks. Nuance SpeechAttendant automates call routing by asking callers to speak the name or dial the extension of a destination.

In the compliance testing, calls from internal and external callers were routed over SIP trunks to Nuance SpeechAttendant. Nuance SpeechAttendant played different greeting announcements based on ANI and/or DNIS, used speech recognition and/or DTMF digits to determine the route destination, and used SIP REFER to transfer calls to destinations on Avaya Aura® Communication Manager or on the PSTN.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were placed manually from users on the PSTN and on Communication Manager to SpeechAttendant. Speech and DTMF input were used from the callers for requesting transfer to internal user and group destinations on Communication Manager, and to external destinations on the PSTN.

The serviceability test cases were performed manually by disconnecting and reconnecting the Ethernet connection to SpeechAttendant.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included G.711MU, session refresh, ANI, DNIS, speech recognition, DTMF, speaking ahead (barge-in), dialing ahead, call forwarding, find me, voicemail, invalid number, blind transfer, supervised transfer, outgoing call screening, and simultaneous calls.

The serviceability testing focused on verifying the ability of SpeechAttendant to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to SpeechAttendant.

2.2. Test Results

All test cases were executed, and the following were observations on SpeechAttendant:

- The application only supports the G.711MU codec, and does not support codec negotiation and media shuffling.

- When an internal user destination has call forwarding activated for both internal and external calls, all transferred calls from SpeechAttendant will follow the forwarding destination for external calls.

- The difference between SpeechAttendant's implementation of blind versus supervised transfer resides in when the application drops from the call after sending REFER with transfer-to destination as Refer To.

2.3. Support

Technical support on SpeechAttendant can be obtained through the following:

- **Phone:** (866) 434-2564 or (514) 390-3922
- **Email:** SpeechAttendant.Support@nuance.com
- **Web :** www.network.nuance.com

3. Reference Configuration

As shown in **Figure 1**, SIP trunks were used between Session Manager and SpeechAttendant, and the applicable domain name was “dr220.com”.

A five digit Uniform Dial Plan (UDP) was used to facilitate routing with SpeechAttendant. Unique extension ranges were assigned to users on Communication Manager (6xxxx), and to SpeechAttendant (52xxx).

The configuration of Session Manager is performed via the web interface of System Manager. The detailed administration of basic connectivity between Communication Manager, System Manager, and Session Manager is not the focus of these Application Notes and will not be described.

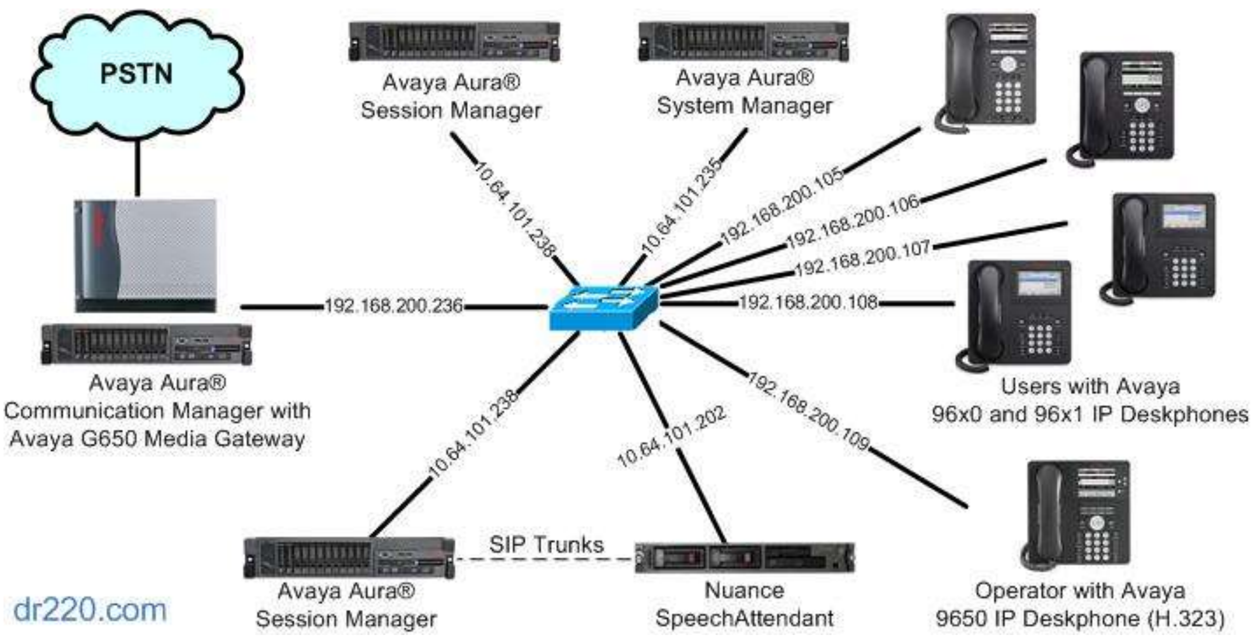


Figure 1: Compliance Testing Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager in Virtual Environment	7.0 SP1 (7.0.0.1.0.441.22477)
Avaya G650 Media Gateway	NA
Avaya Aura® Media Server in Virtual Environment	7.7.0.236
Avaya Aura® Session Manager in Virtual Environment	7.0 (7.0.0.0.0.700007)
Avaya Aura® System Manager in Virtual Environment	7.0 (7.0.0.0.0.4036)
Avaya 9608 IP Deskphone (H.323)	6.6029
Avaya 9620C & 9650 IP Deskphones (H.323)	3.250A
Avaya 9621G & 9641G IP Deskphones (SIP)	7.0.0.39
Nuance SpeechAttendant on Microsoft Windows Server 2012	12.1.0 HotFix 1210HF01, 1210HF02 R2 Standard

5. Configure Avaya Aura® Communication Manager

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

- Verify license
- Administer system parameters features
- Administer SIP trunk group
- Administer SIP signaling group
- Administer SIP trunk group members
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer private numbering
- Administer uniform dial plan
- Administer AAR analysis

In the compliance testing, a separate set of codec set, network region, trunk group, and signaling group were used for integration with SpeechAttendant.

5.1. Verify License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

```
display system-parameters customer-options                      Page   2 of  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 12000 10
           Maximum Concurrently Registered IP Stations: 1800  1
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 41000 1
                   Maximum Video Capable IP Softphones: 24000 20
                       Maximum Administered SIP Trunks: 24000 0
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
```

5.2. Administer System Parameters Features

Use the “change system-parameters features” command to allow for trunk-to-trunk transfers.

For ease of interoperability testing, the **Trunk-to-Trunk Transfer** field was set to “all” to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk, and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented on the Class Of Restriction or Class Of Service levels. Refer to reference [1] for more details.

```
change system-parameters features                              Page   1 of  19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y

              Music (or Silence) on Transferred Trunk Calls? Call-wait
              DID/Tie/ISDN/SIP Intercept Treatment: attendant
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n
```

5.3. Administer SIP Trunk Group

Use the "add trunk-group n" command, where "n" is an available trunk group number, in this case "52". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "sip"
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** "tie"

```
add trunk-group 52                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 52                   Group Type: sip          CDR Reports: y
  Group Name: Nuance                      COR: 1        TN: 1        TAC: 1052
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group:
                                                     Number of Members: 0
```

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

```
add trunk-group 52                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



                     Numbering Format: private
                                              UUI Treatment: service-provider

                                               Replace Restricted Numbers? n
                                              Replace Unavailable Numbers? n

                                                Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no
```

5.4. Administer SIP Signaling Group

Use the "add signaling-group n" command, where "n" is an available signaling group number, in this case "52". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "sip"
- **Transport Method:** "tcp"
- **Near-end Node Name:** An existing C-LAN node name or "procr".
- **Far-end Node Name:** The existing node name for Session Manager.
- **Near-end Listen Port:** An available port for integration with Nuance.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** An existing network region to use with Nuance.
- **Far-end Domain:** The applicable domain name for the network.
- **Direct IP-IP Audio Connections:** "n"

```
add signaling-group 52                                          Page   1 of   2
                                SIGNALING GROUP

 Group Number: 52             Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: Others
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? n
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? y

   Near-end Node Name: procr                  Far-end Node Name: sm7-sig
 Near-end Listen Port: 5052                 Far-end Listen Port: 5052
                                         Far-end Network Region: 2

Far-end Domain: dr220.com
                                              Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                   RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? n
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
```

5.5. Administer SIP Trunk Group Members

Use the "change trunk-group n" command, where "n" is the trunk group number from **Section 5.3**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Signaling Group:** The signaling group number from **Section 5.4**.
- **Number of Members:** The desired number of members, in this case "10".

```
change trunk-group 52                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 52                  Group Type: sip           CDR Reports: y
  Group Name: Nuance                     COR: 1        TN: 1        TAC: 1052
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 52
                                                     Number of Members: 10
```

5.6. Administer IP Network Region

Use the “change ip-network-region n” command, where “n” is the existing far-end network region number used by the SIP signaling group from **Section 5.4**.

For **Authoritative Domain**, enter the applicable domain for the network. Enter a descriptive **Name**. Enter “no” for **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio**, as shown below. For **Codec Set**, enter an available codec set number for integration with SpeechAttendant.

```
change ip-network-region 2                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 2
Location:          Authoritative Domain: dr220.com
    Name: Nuance                Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: no
      Codec Set: 2              Inter-region IP-IP Direct Audio: no
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
```

Navigate to **Page 4**, and specify this codec set to be used for calls with network regions used by Avaya endpoints and by the trunk to the PSTN. In the compliance testing, network region “1” was used by the Avaya endpoints and by the trunk to the PSTN.

```
change 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
 2    2
 3                                                                 all
 4
 5
 6
 7
 8
```

5.7. Administer IP Codec Set

Use the “change ip-codec-set n” command, where “n” is the codec set number from **Section 5.6**. Update the audio codec types in the **Audio Codec** fields as necessary. Note that SpeechAttendant only supports the G.711 codec variant. The codec shown below was used in the compliance testing.

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

                          IP Codec Set

    Codec Set: 2

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

5.8. Administer Route Pattern

Use the “change route-pattern n” command, where “n” is an existing route pattern number to be used to reach SpeechAttendant, in this case “52”. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern Name:** A descriptive name.
- **Grp No:** The SIP trunk group number from **Section 5.3**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.

```
change route-pattern 52                                         Page   1 of   3
                    Pattern Number: 52  Pattern Name: Nuance
                             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: 52   0                                                               n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

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

5.9. Administer Private Numbering

Use the "change private-numbering 0" command, to define the calling party number to send to SpeechAttendant. Add an entry for the trunk group defined in **Section 5.3**. In the example shown below, all calls originating from a 5-digit extension beginning with 6 and routed to trunk group 52 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  6                52                           5     Total Administered: 1
                                                           Maximum Entries: 540
```

5.10. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 52xxx to SpeechAttendant. Note that other routing methods may be used. Use the "change uniform-dialplan 0" command, and add an entry to specify the use of AAR for routing of digits 52xxx, as shown below.

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

  Matching                   Insert              Node
  Pattern       Len Del      Digits      Net Conv Num

 52              5   0                   aar  n
```

5.11. Administer AAR Analysis

Use the "change aar analysis 0" command, and add an entry to specify how to route calls to 52xxx. In the example shown below, calls with digits 52xxx will be routed as an AAR call using route pattern "52" from **Section 5.8**.

```
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
    52                    5    5     52        unku         n
```

6. Configure Avaya Aura® Session Manager

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

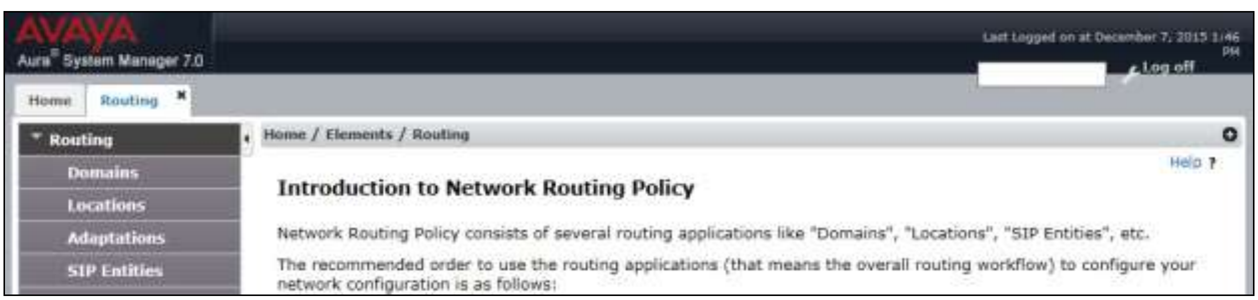
- Launch System Manager
- Administer locations
- Administer SIP entities
- Administer routing policies
- Administer dial patterns

6.1. Launch System Manager

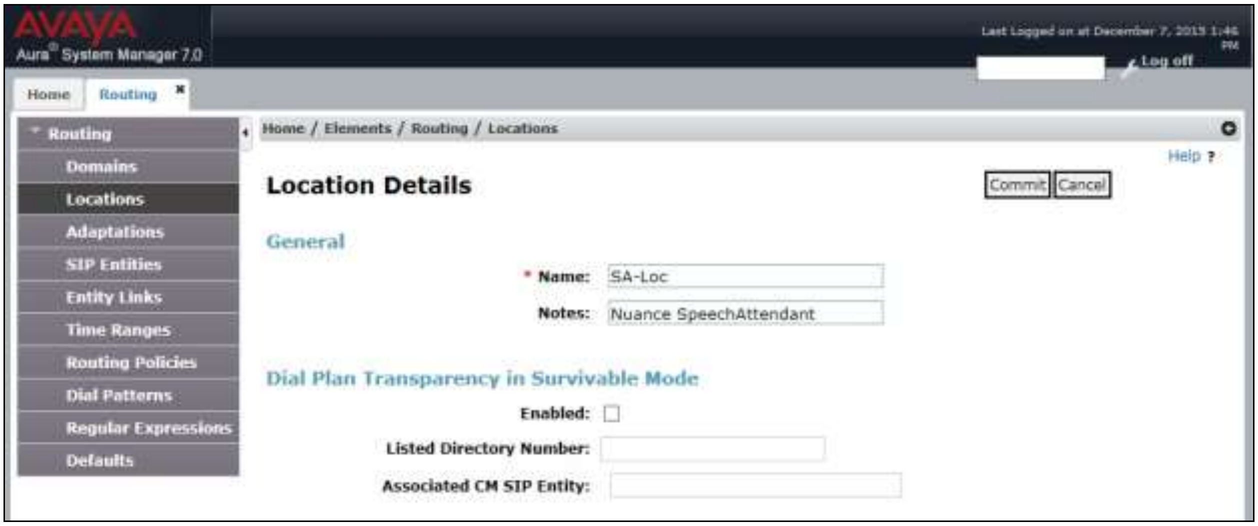
Access the System Manager web interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of System Manager. Log in using the appropriate credentials.

6.2. Administer Locations

In the subsequent screen (not shown), select **Elements → Routing** to display the **Introduction to Network Routing Policy** screen below. Select **Routing → Locations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new location for SpeechAttendant.



The **Location Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**. Retain the default values in the remaining fields.



Scroll down to the **Location Pattern** sub-section, click **Add** and enter the IP address of SpeechAttendant in **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.

6.3. Administer SIP Entities

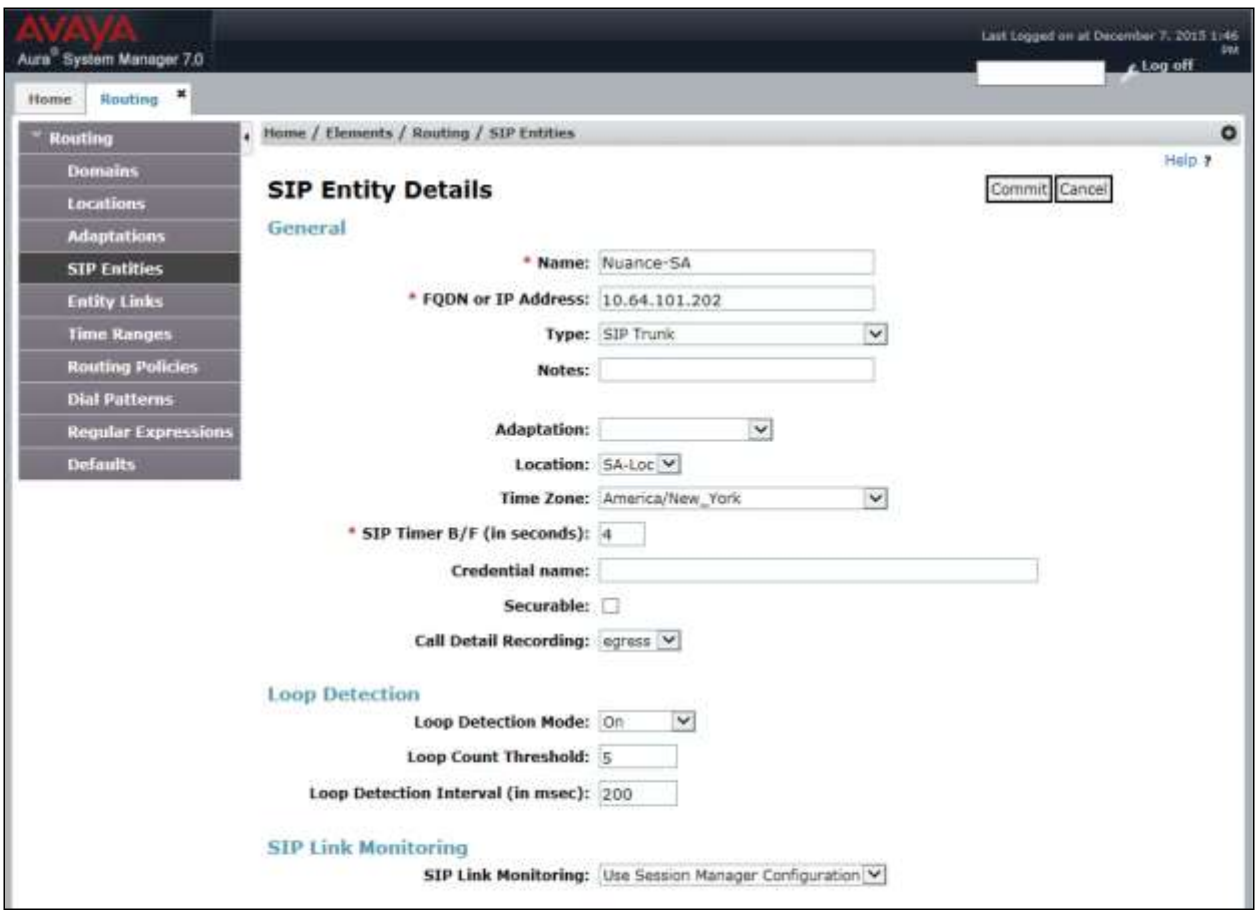
Add two new SIP entities, one for SpeechAttendant and one for the new SIP trunks with Communication Manager.

6.3.1. SIP Entity for SpeechAttendant

Select **Routing → SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for SpeechAttendant.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

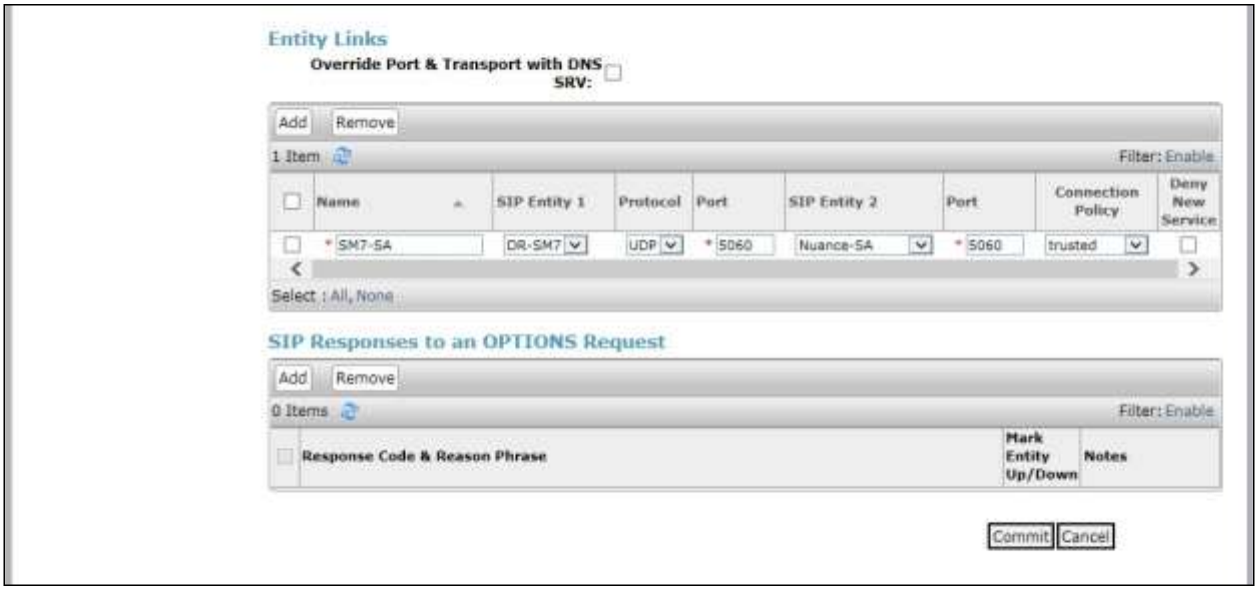
- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the SpeechAttendant server.
- **Type:** "SIP Trunk"
- **Notes:** Any desired notes.
- **Location:** Select the SpeechAttendant location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.



Scroll down to the **Entity Links** sub-section, and click **Add** to add an entity link. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case “DR-SM7”.
- **Protocol:** “UDP”
- **Port:** “5060”
- **SIP Entity 2:** The SpeechAttendant entity name from this section.
- **Port:** “5060”
- **Connection Policy:** “trusted”

Note that SpeechAttendant can support UDP, TCP, and TLS, and the compliance testing used the UDP protocol.

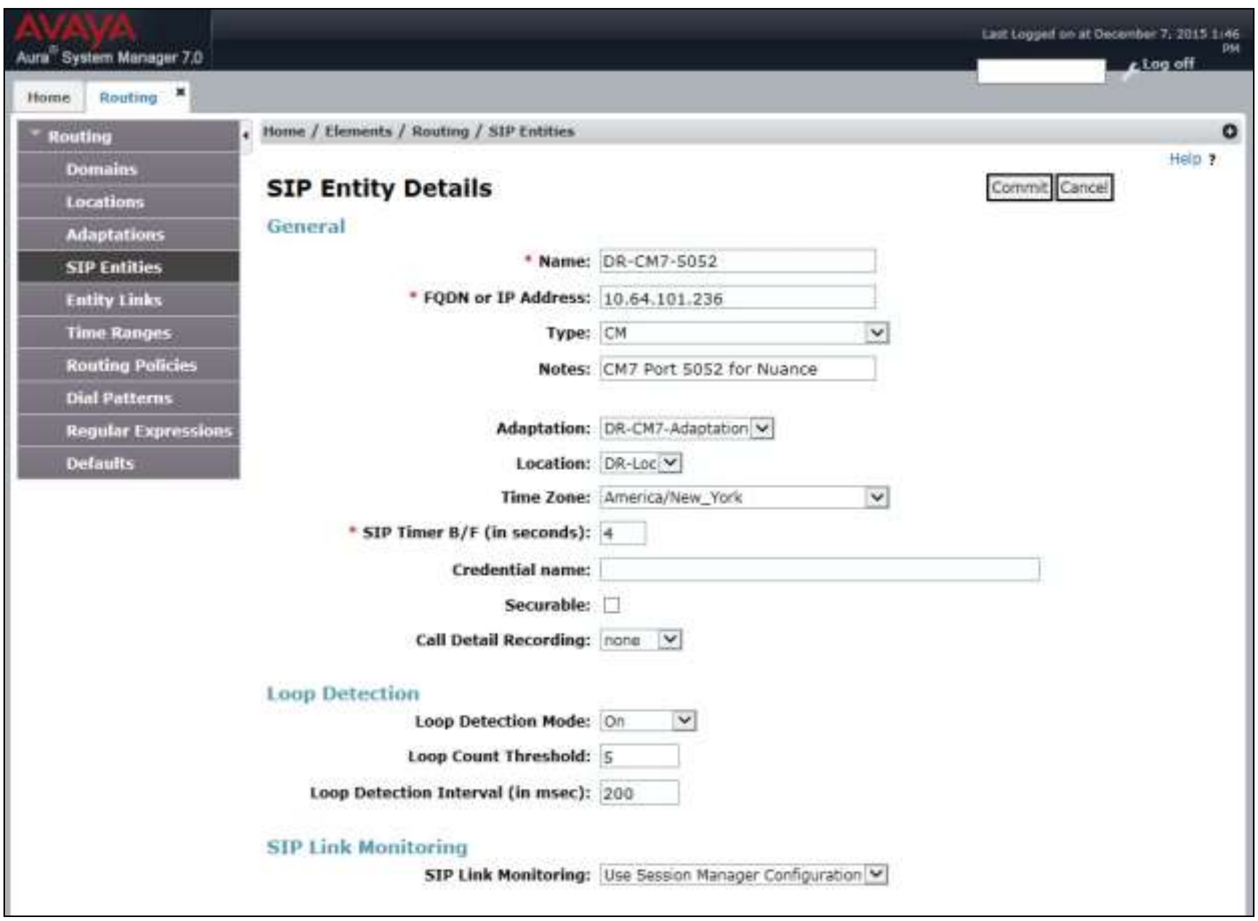


6.3.2. SIP Entity for Communication Manager

Select **Routing → SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Communication Manager. Note that this SIP entity is used for integration with SpeechAttendant.

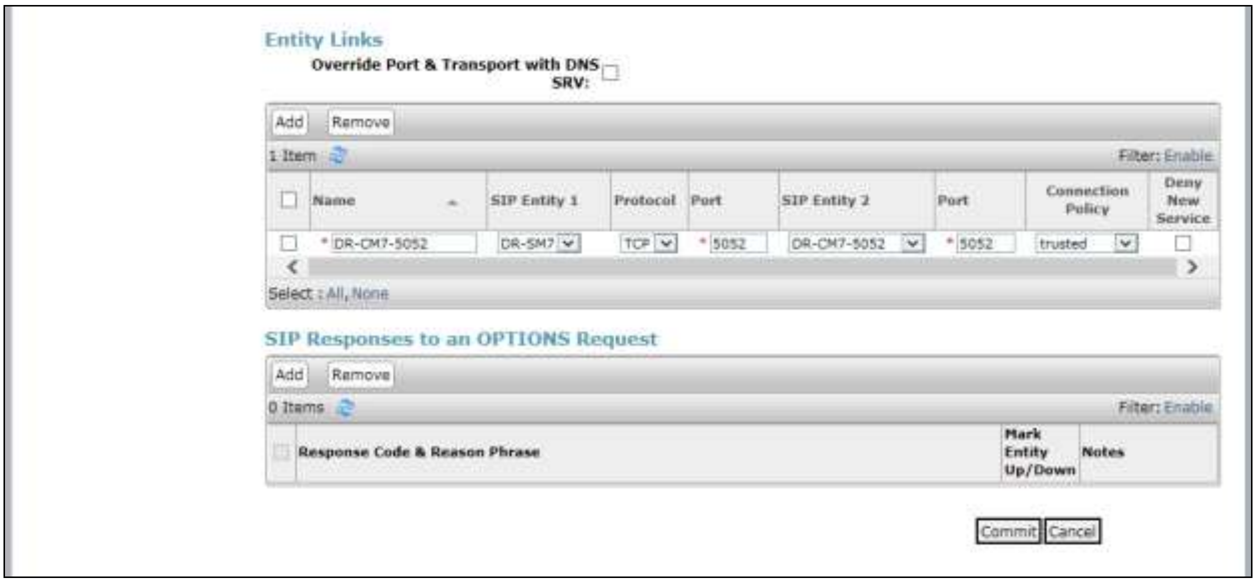
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of an existing CLAN or the processor interface.
- **Type:** "CM"
- **Notes:** Any desired notes.
- **Adaptation:** Select the applicable adaptation for Communication Manager.
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.



Scroll down to the **Entity Links** sub-section, and click **Add** to add an entity link. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case "DR-SM7".
- **Protocol:** The signaling group transport method from **Section 5.4**.
- **Port:** The signaling group listen port number from **Section 5.4**.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** The signaling group listen port number from **Section 5.4**.
- **Connection Policy:** "trusted"



6.4. Administer Routing Policies

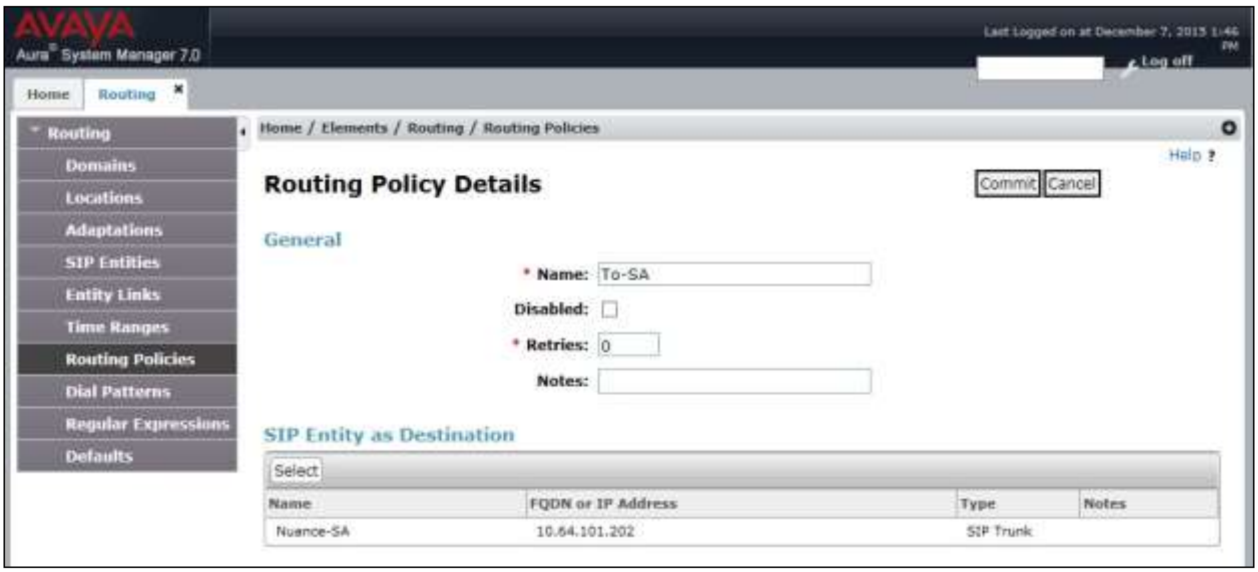
Add two new routing policies, one for SpeechAttendant and one for the new SIP trunks with Communication Manager.

6.4.1. Routing Policy for SpeechAttendant

Select **Routing → Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for SpeechAttendant.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the SpeechAttendant entity name from **Section 6.3.1**. The screen below shows the result of the selection.

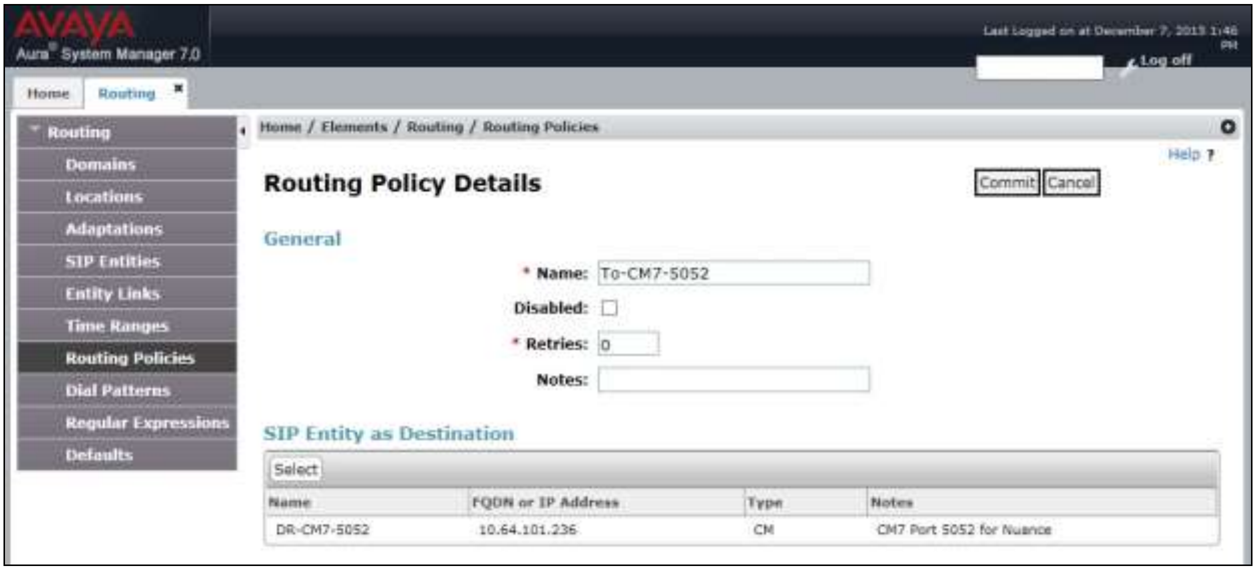


6.4.2. Routing Policy for Communication Manager

Select **Routing → Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for Communication Manager.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communication Manager entity name from **Section 6.3.2**. The screen below shows the result of the selection.



6.5. Administer Dial Patterns

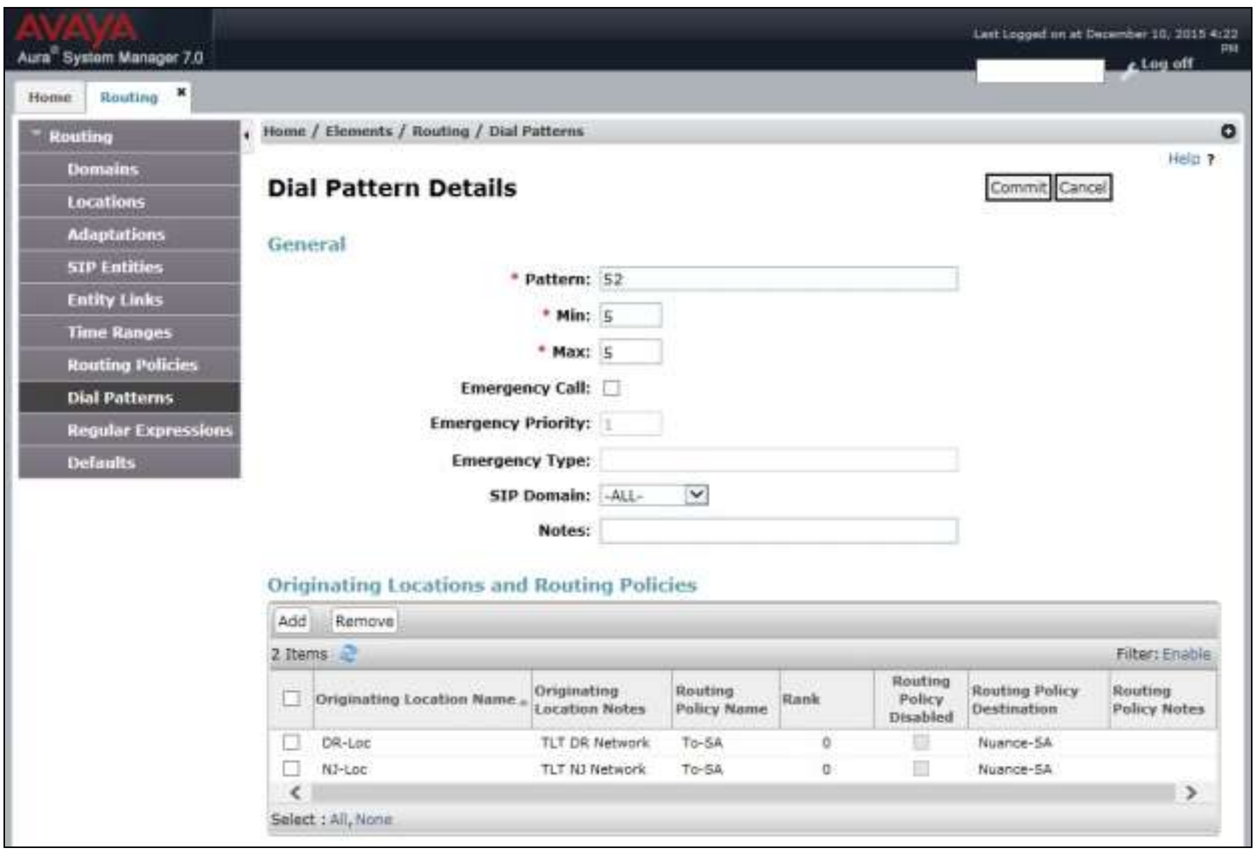
Add a new dial pattern for SpeechAttendant, and update existing dial patterns for Communication Manager.

6.5.1. Dial Pattern for SpeechAttendant

Select **Routing → Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach SpeechAttendant. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match, in this case "52".
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The signaling group domain name from **Section 5.4**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching SpeechAttendant. In the compliance testing, the entry allowed for call originations from all Communication Manager endpoints in locations "DR-Loc" and "NJ-Loc". The SpeechAttendant routing policy from **Section 6.4.1** was selected as shown below.

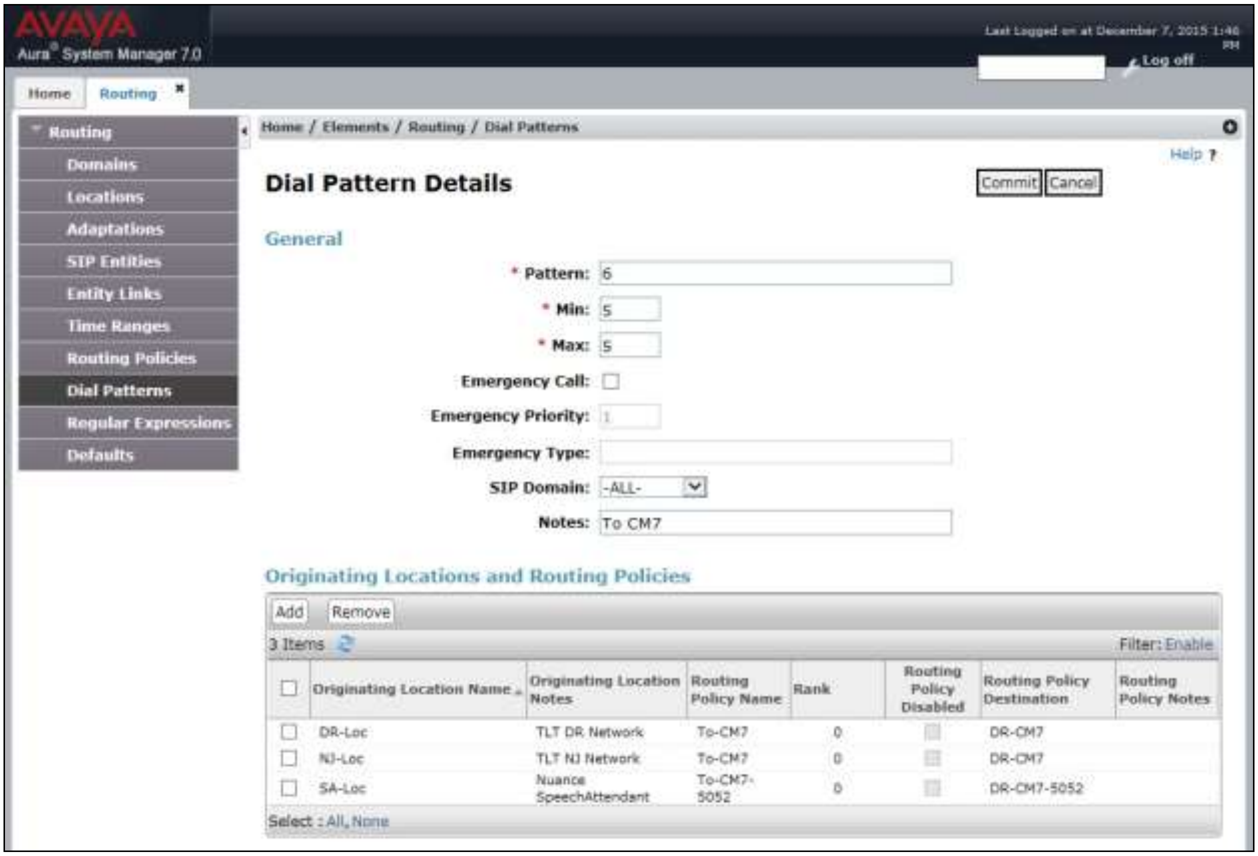


6.5.2. Dial Pattern for Communication Manager

Select **Routing → Dial Patterns** from the left pane, and click on the first existing dial pattern for Communication Manager in the subsequent screen, in this case dial pattern "6" (not shown). The **Dial Pattern Details** screen is displayed.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy as necessary for calls from SpeechAttendant. In the compliance testing, the new policy allowed for call origination from the SpeechAttendant location from **Section 6.2**, and the Communication Manager routing policy from **Section 6.4.2** was selected as shown below. Retain the default values in the remaining fields.

Follow the procedures in this section to make similar changes to the applicable Communication Manager dial pattern to reach the PSTN. In the compliance testing, SpeechAttendant will add the prefix "9" for outbound calls to the PSTN, and therefore the existing dial pattern for "9" was also changed (not shown below).



7. Configure Nuance SpeechAttendant

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

- Launch Admin Tools
- Administer configuration panel
- Administer phone directory and menu editor
- Administer dialing properties
- Administer ports and entry points

The configuration of SpeechAttendant is typically performed by Nuance Professional Services. The procedural steps are presented in these Application Notes for informational purposes.

7.1. Administer Configuration Panel

From the SpeechAttendant server, double-click the **Admin Tools** icon shown below, which was automatically created as part of installation.

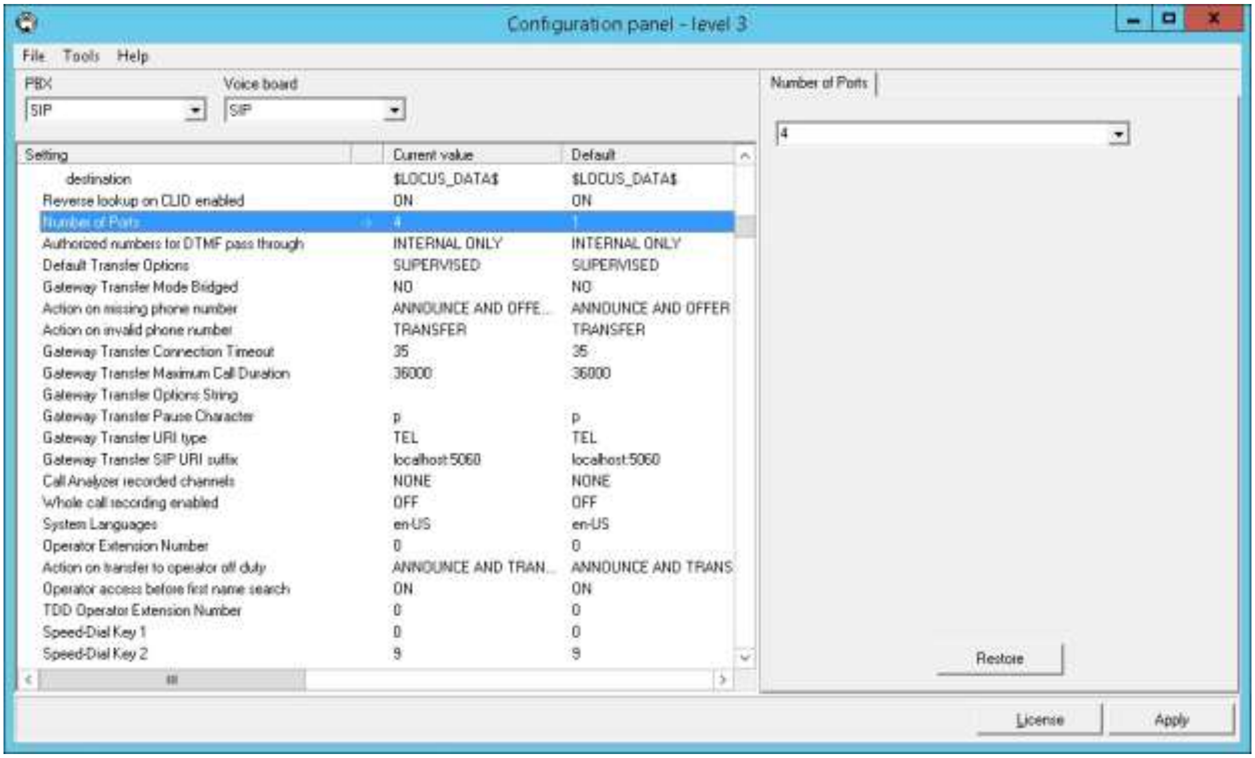
7.2. Administer Configuration Panel

The **Admin Tools** screen is displayed. Select **Configuration Panel**.

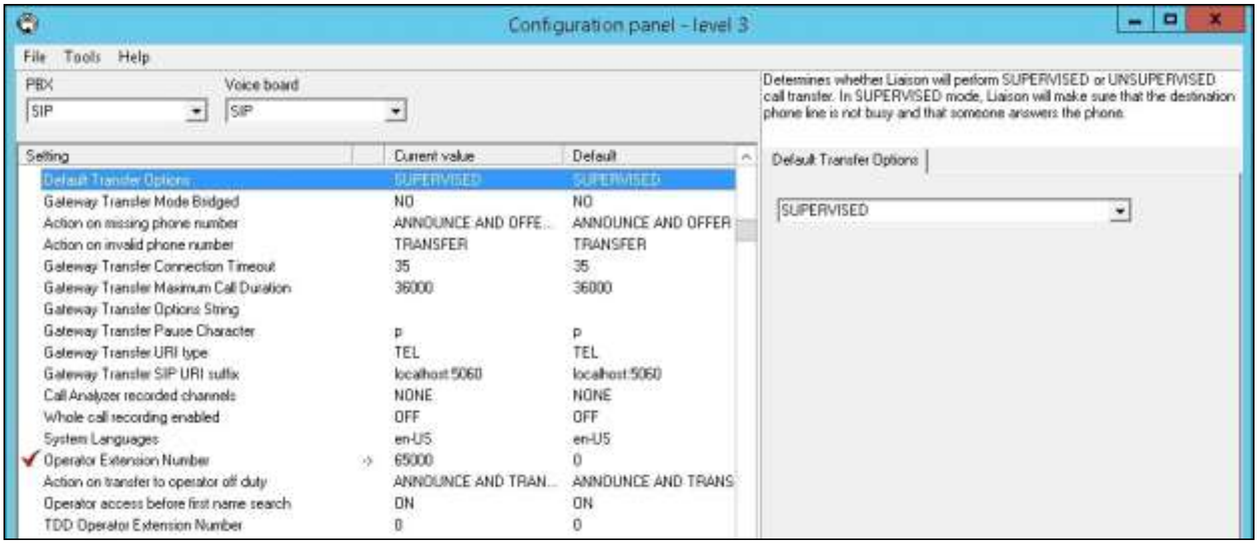
The **Configuration password** screen is displayed. Select "Level 3" and enter the appropriate credential.

The **Configuration panel – level 3** screen is displayed next. In the upper left pane, set **PBX** and **Voice board** to "SIP", as shown below.

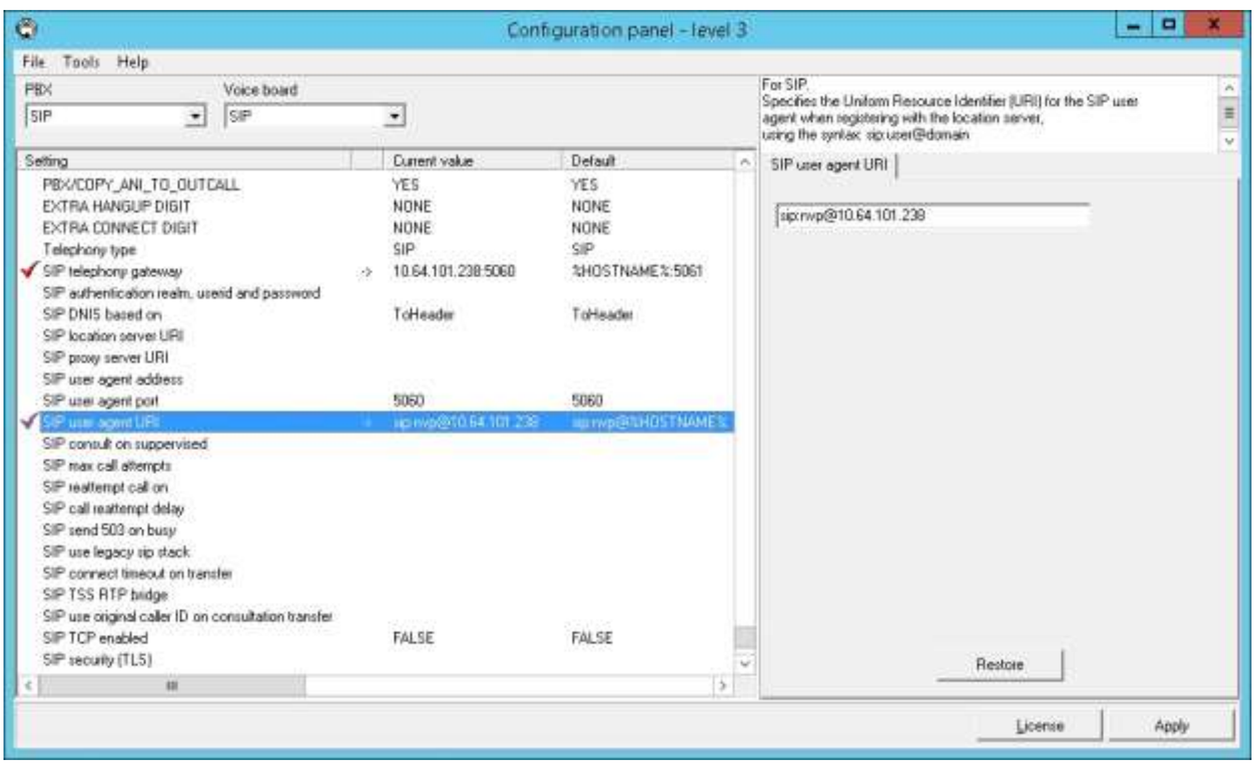
Scroll the screen in the left pane as necessary, and update the **Number of Ports** parameter to the number allowed for by the SpeechAttendant license, in this case "4".



Scroll the screen in the left pane as necessary, and update the **Operator Extension Number** parameter with the extension of an endpoint on Communication Manager for use as the operator. SpeechAttendant will automatically transfer a caller to the operator when all attempts to understand the caller requests have failed. Callers can also ask for the operator directly.



Scroll the screen in the left pane as necessary, to locate the **SIP telephony gateway** and **SIP user agent URI** parameters. Update the two parameters with the IP address of the Session Manager signaling interface and the port number from **Section 6.3.1**, as shown below.

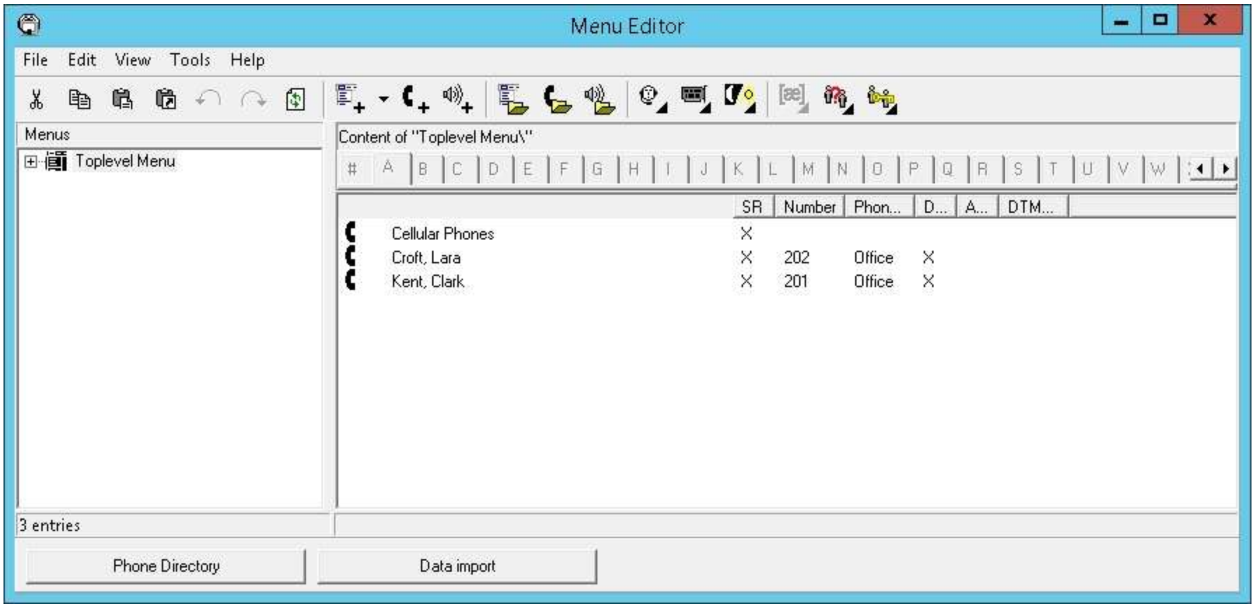


7.3. Administer Phone Directory and Menu Editor

From the **Admin Tools** screen, select **Phone Directory and Menu Editor**, as shown below.

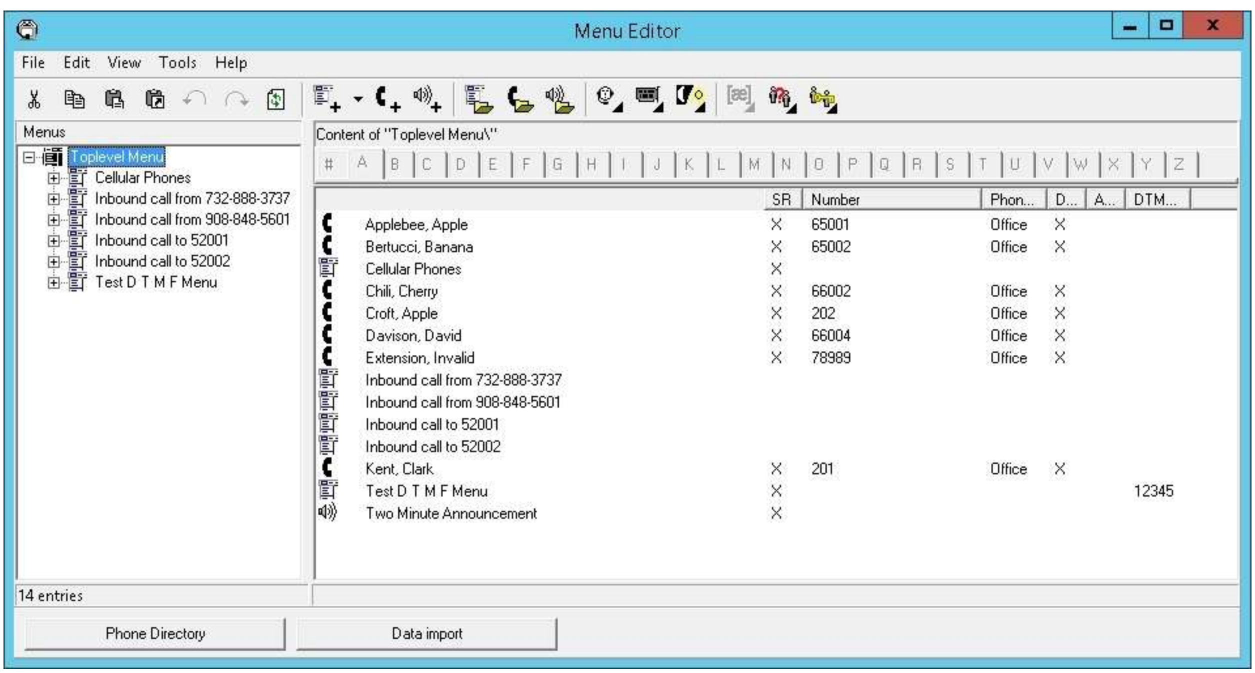
The **Phone Directory and Menu Editor** screen below is displayed next. Log in using the appropriate credentials.

The **Menu Editor** screen below is displayed, with default directory entries in the right pane.



Follow reference [3] to create additional entry points in the left pane and additional directory entries in the right pane pertinent to customer needs. The screenshot below shows the entry points and directory entries used in the compliance testing. A mixture of blind and supervised transfer methods were configured for the directory entries.

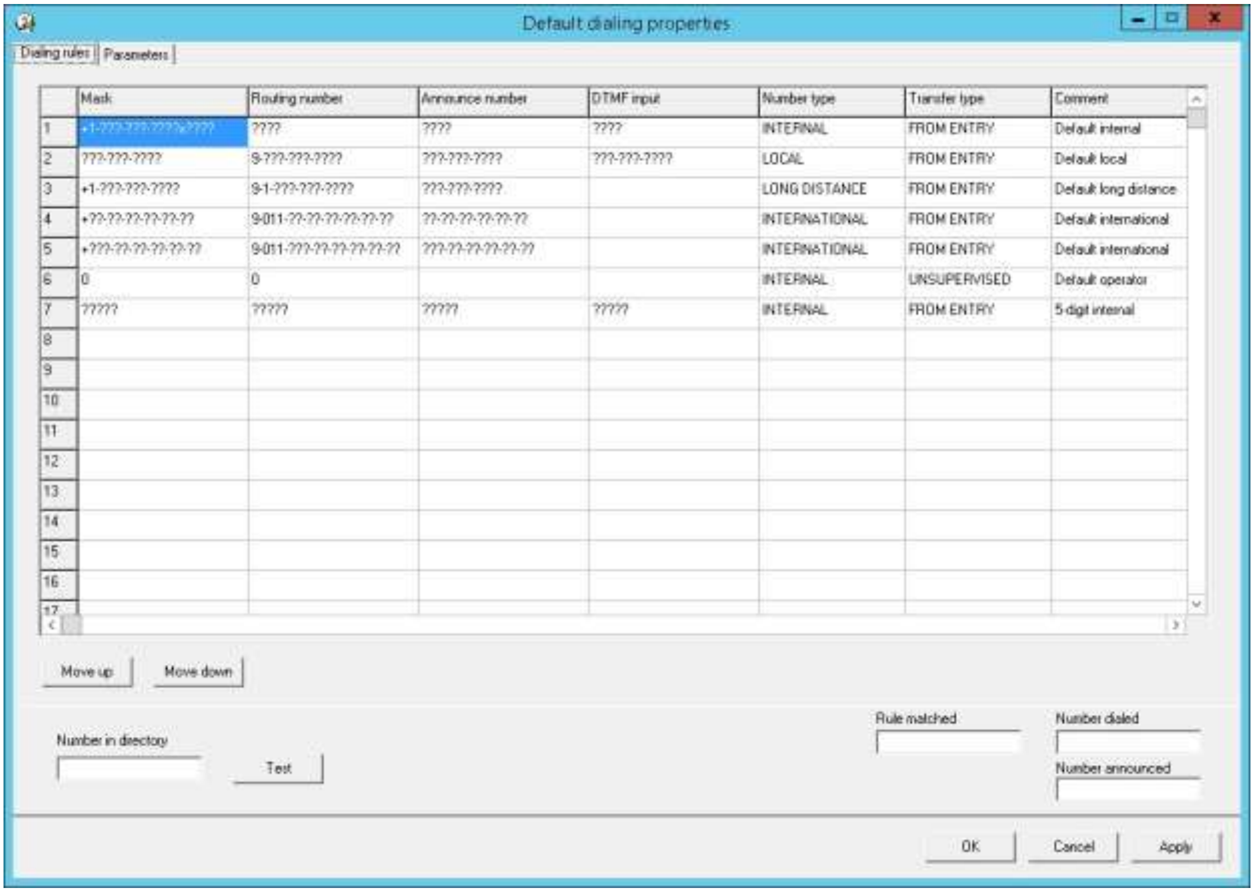
Select **Tools → Dialing Properties** from the top menu.



7.4. Administer Dialing Properties

The **Default dialing properties** screen is displayed. Follow reference [3] to update and add dialing properties entries as necessary for routing of calls pertinent to the customer network.

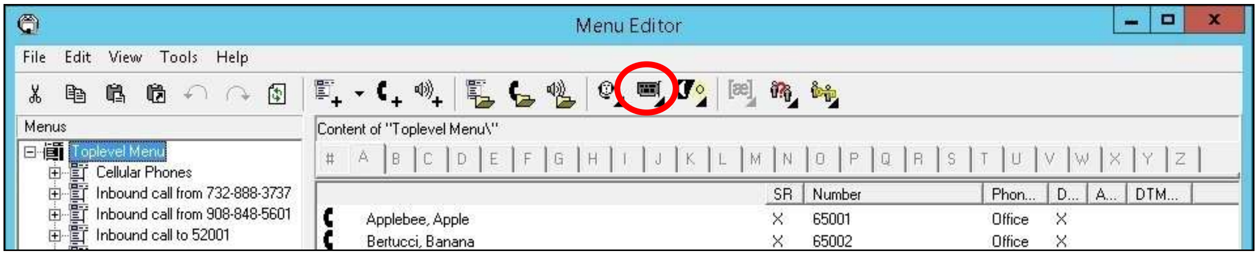
In the compliance testing, the last entry in the screenshot below was added for routing of calls to internal destinations consisting of 5-digit extensions.



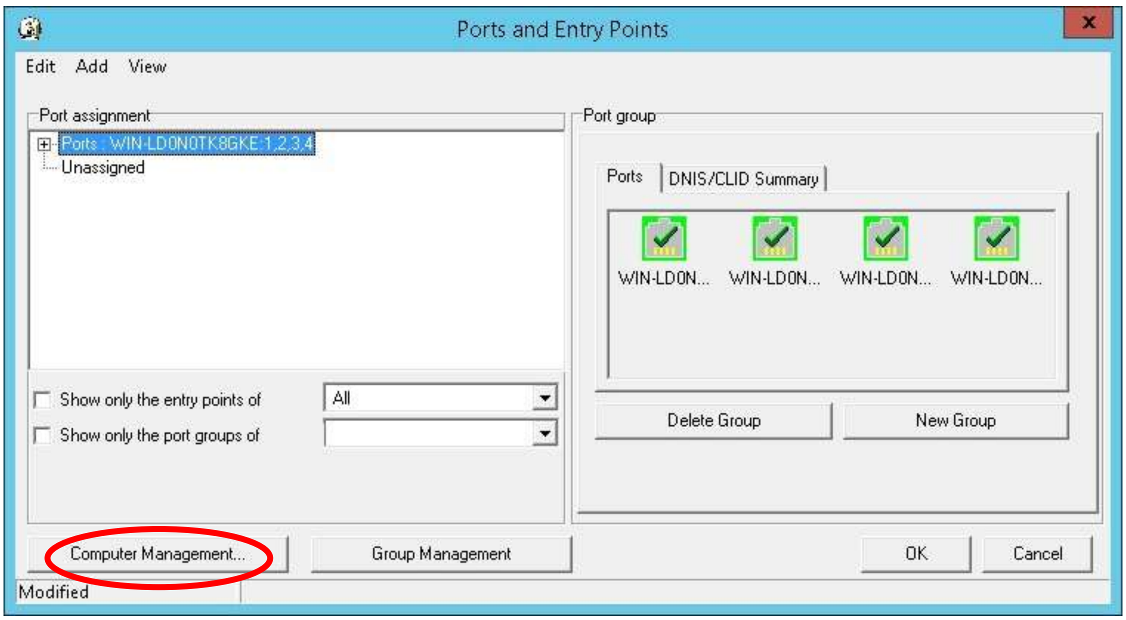
	Mask	Routing number	Announce number	DTMF input	Number type	Transfer type	Comment
1	+1-???-???-????x????	????	????	????	INTERNAL	FROM ENTRY	Default internal
2	???-???-????	9-???-???-????	???-???-????	???-???-????	LOCAL	FROM ENTRY	Default local
3	+1-???-???-????	9-1-???-???-????	???-???-????		LONG DISTANCE	FROM ENTRY	Default long distance
4	+??-??-??-??-??-??	9-011-??-??-??-??-??-??	??-??-??-??-??-??		INTERNATIONAL	FROM ENTRY	Default international
5	+???-??-??-??-??-??	9-011-???-??-??-??-??-??	???-??-??-??-??-??		INTERNATIONAL	FROM ENTRY	Default international
6	0	0			INTERNAL	UNSUPERVISED	Default operator
7	?????	?????	?????	?????	INTERNAL	FROM ENTRY	5 digit internal
8							
9							
10							
11							
12							
13							
14							
15							
16							
17							

7.5. Administer Ports and Entry Points

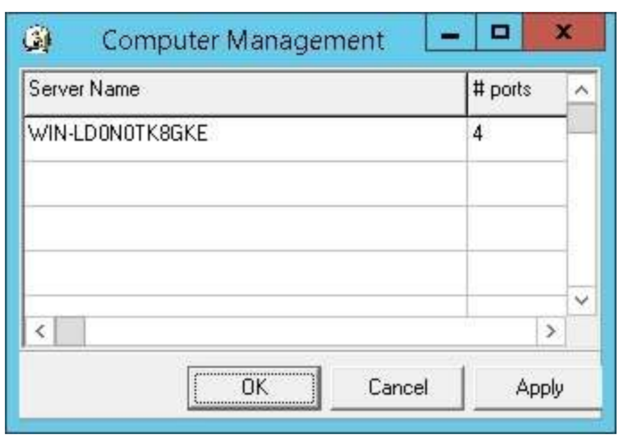
From the **Menu Editor** screen, click the **Ports and entry points** icon shown below.



The **Ports and Entry Points** screen below is displayed. Select **Computer Management**.



The **Computer Management** screen is displayed. Set the **# ports** value to the value allowed for by the license, in this case “4”.



8. Verification Steps

This section provides tests that can be performed to verify proper configuration of Communication Manager, Session Manager, and SpeechAttendant.

8.1. Verify Avaya Aura® Communication Manager

From the SAT interface, verify the status of the SIP trunk groups by using the "status trunk n" command, where "n" is the trunk group number administered in **Section 5.3**. Verify that all trunks are in the "in-service/idle" state as shown below.

```
status trunk 52

                            TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0052/001 T00146   in-service/idle    no
0052/002 T00147   in-service/idle    no
0052/003 T00148   in-service/idle    no
0052/004 T00149   in-service/idle    no
0052/005 T00150   in-service/idle    no
0052/006 T00151   in-service/idle    no
0052/007 T00152   in-service/idle    no
0052/008 T00153   in-service/idle    no
0052/009 T00154   in-service/idle    no
0052/010 T00155   in-service/idle    no
```

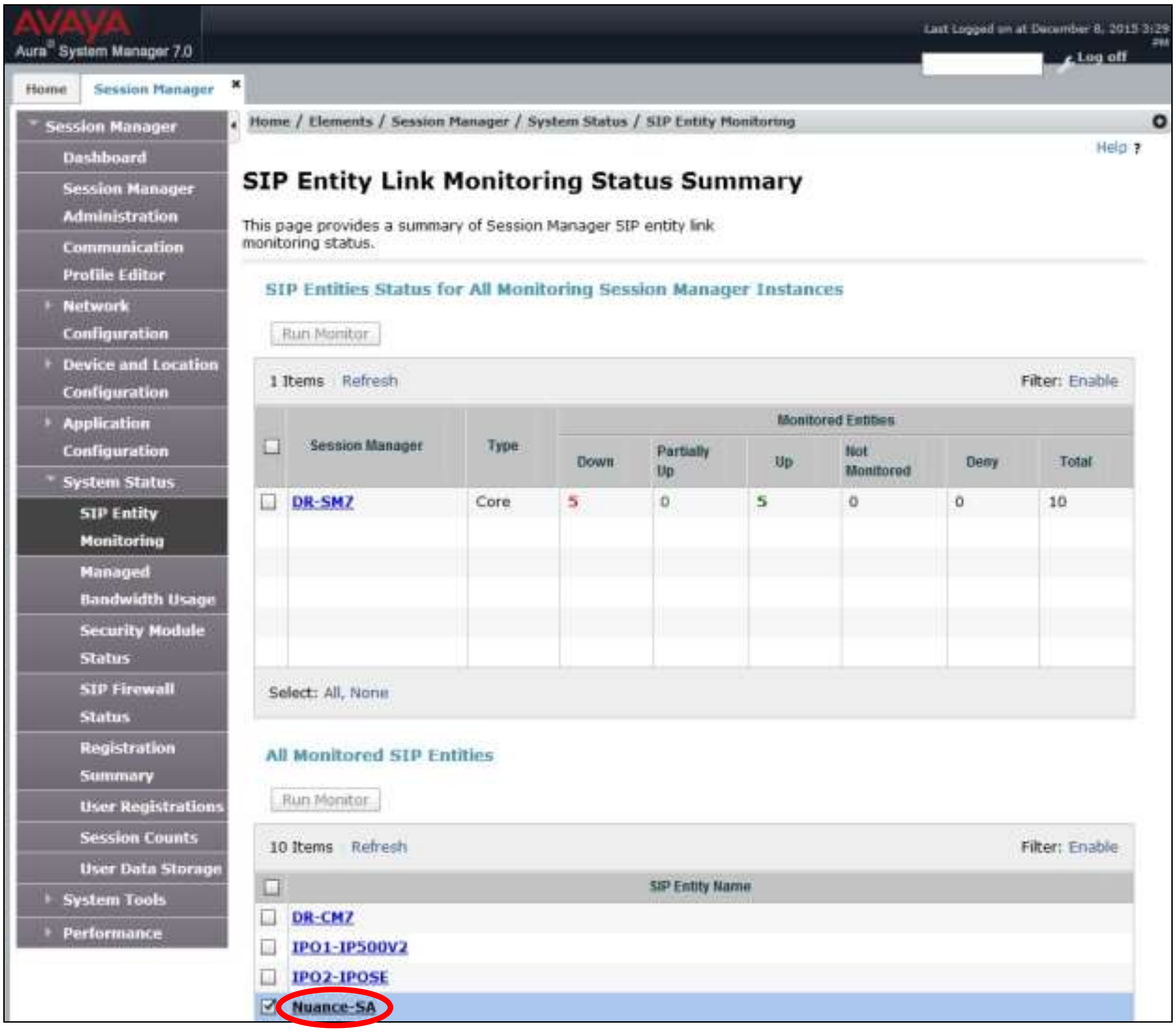
Verify the status of the SIP signaling groups by using the "status signaling-group n" command, where "n" is the signaling group number administered in **Section 5.4**. Verify that the **Group State** is "in-service", as shown below.

```
status signaling-group 52
                        STATUS SIGNALING GROUP

        Group ID: 52
      Group Type: sip

     Group State: in-service
```

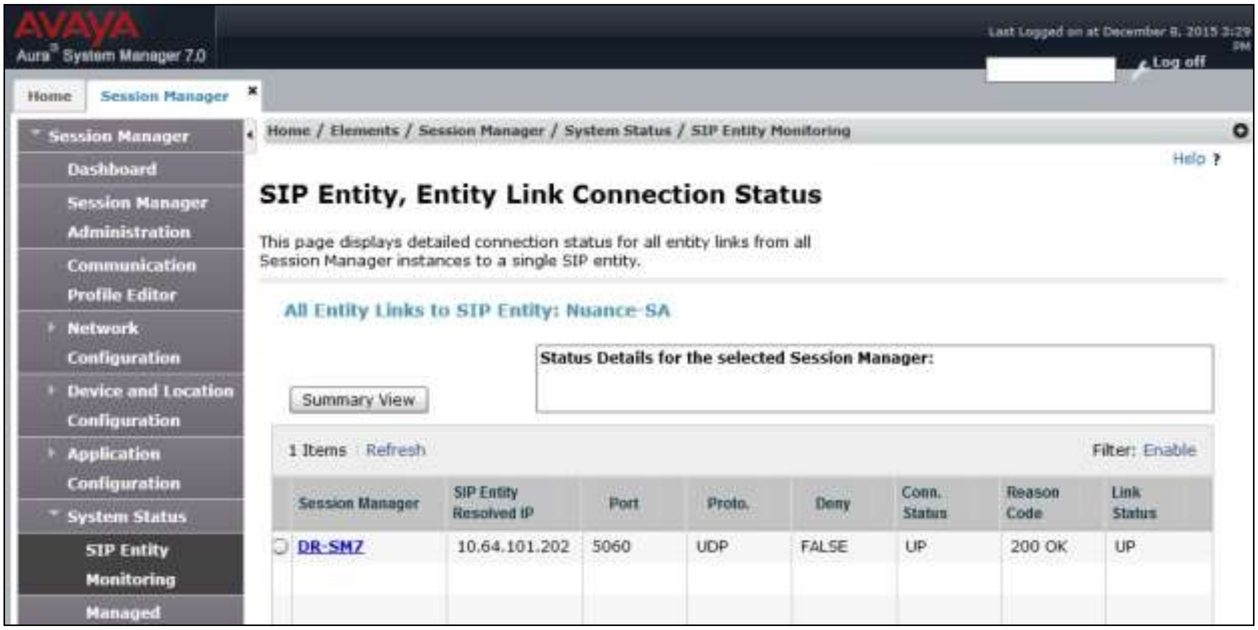
8.2. Verify Avaya Aura® Session Manager

From the System Manager home page (not shown), select **Elements → Session Manager** to display the **Session Manager Dashboard** screen (not shown).

Select **Session Manager → System Status → SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click the SpeechAttendant entity name from **Section 6.3.1**.



The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn Status** and **Link Status** are "UP", as shown below.

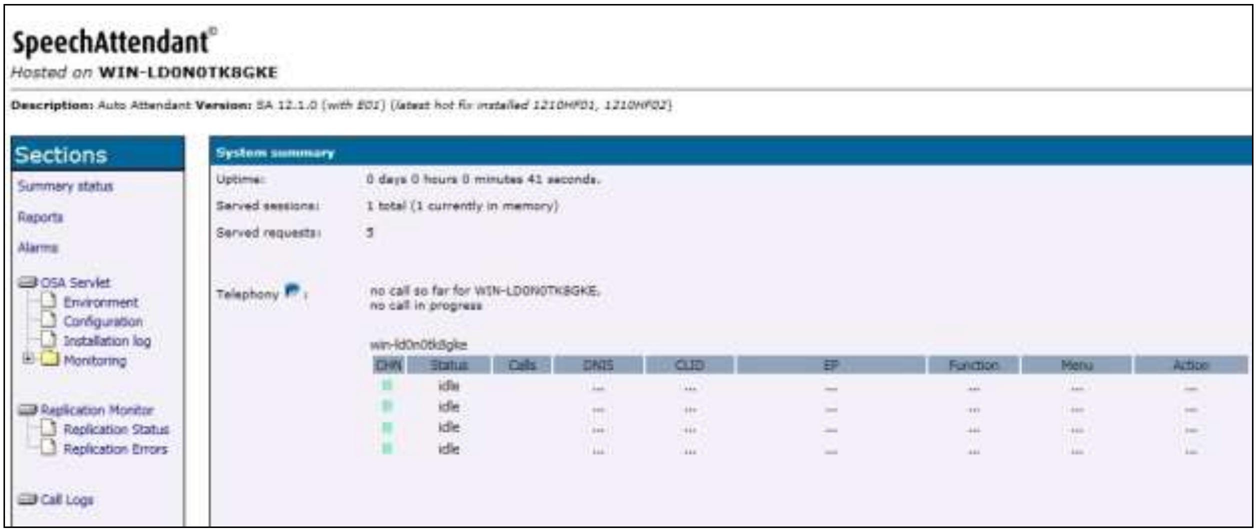


8.3. Verify Nuance SpeechAttendant

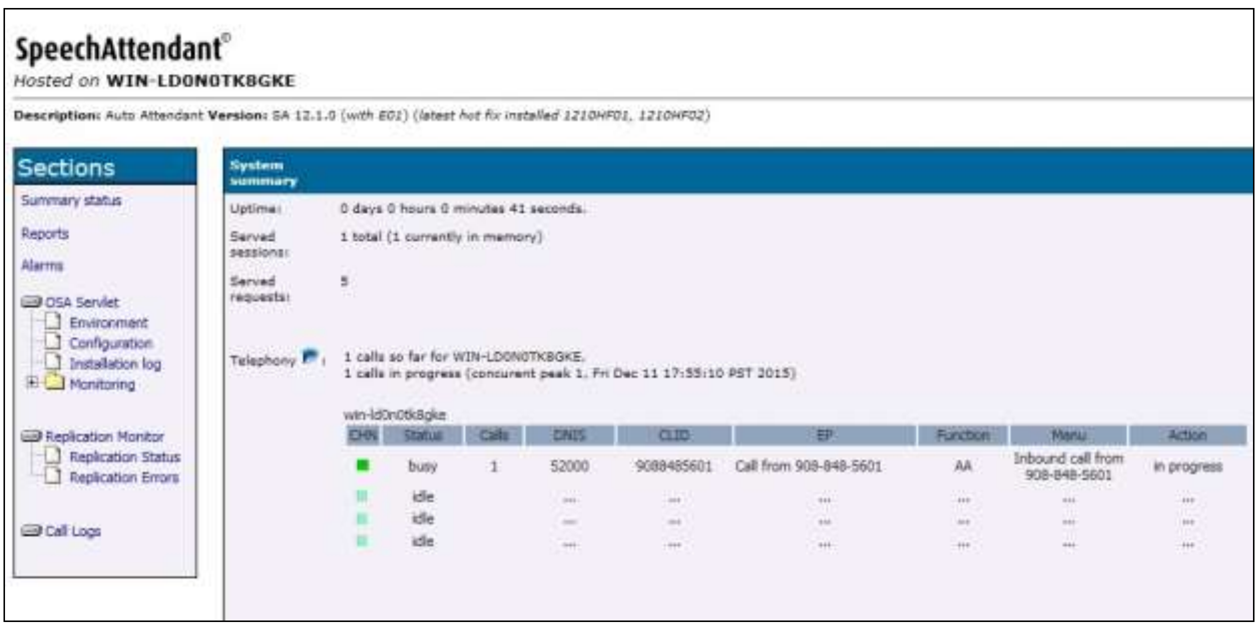
From a PC, launch an Internet browser window and access the SpeechAttendant web-based status interface by using the URL "http://<ip-address>/OpenSpeech/Attendant/servlet/aa?action=status", where "ip-address" is the IP address of the SpeechAttendant server.

The **Speech Attendant Login** screen is displayed. Log in using the appropriate credentials.

The screen below is displayed next. Verify that the **Status** for all channels are "Idle", as shown below.



Establish an incoming trunk call from PSTN with SpeechAttendant. Verify that the calling party hears the appropriate greeting, and that the status screen reflects the active call with pertinent call information, as shown below.



9. Conclusion

These Application Notes describe the configuration steps required for Nuance SpeechAttendant 12 to successfully interoperate with Avaya Aura® Session Manager 7.0 and Avaya Aura® Communication Manager 7.0 using SIP trunks. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

10. Additional References

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

1. *Administering Avaya Aura® Communication Manager*, Release 7.0, Issue 1, August 2015, available at http://support.avaya.com.

2. *Administering Avaya Aura® Session Manager*, Release 7.0, Issue 1, August 2015, available at http://support.avaya.com.

3. *Nuance SpeechAttendant Nuance OpenSpeech Attendant Administration Guide*, April 2014, available at https://network.nuance.com/portal/server.pt.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.