Avaya Solution & Interoperability Test Lab

Application Notes for Nuance Speech Attendant with Avaya Aura® Session Manager and Avaya Aura® Communication Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Nuance Speech Attendant with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP integration. Nuance Speech Attendant allows callers to speak the name of a person, department, service, or location and be automatically transferred to the requested party without waiting to speak to an operator. In addition, the caller may dial an extension number to transfer to the requested party.

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 to integrate the Nuance Speech Attendant (SA) with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP integration. Nuance Speech Attendant allows callers to speak the name of a person, department, service, or location and be automatically transferred to the requested party without waiting to speak to an operator. In addition, the caller may dial an extension number to transfer to the requested party.

1.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. Feature testing focused on that Nuance SA can successfully recognize spoken names and extensions entered via DTMF and transfer the call to the correct destination. Blind and supervised transfers were verified. Other features covered included: DNIS and CLID handling, adding new transfer entries, recording caller utterances, and accessing Maintenance Mode and Personal Administration Mode to record name and change PIN. In addition, failover support was also tested, which verified the ability to re-route the call to Nuance SA to an alternate destination if it was busy or unavailable.

Serviceability testing focused on verifying the ability of the Nuance SA to recover from adverse conditions, such as server restarts, power failures, and disconnecting cables to the IP network.

1.2. Support

To obtain technical support for Nuance Speech Attendant, contact Nuance via email or through their website.

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

2. Reference Configuration

Figure 1 illustrates the configuration used to verify the Nuance Speech Attendant (SA) solution with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Nuance SA is deployed on a dedicated server running Windows 2003 Server. Session Manager interfaces to Communication Manager using a SIP trunk, and Nuance SA interfaces to Session Manager via SIP. To access the Nuance SA application, a call is simply routed from Communication Manager to the SA server through the Session Manager. Multiple SIP ports were configured on the Nuance SA server. Avaya Aura® System Manager was used to configure Session Manager.

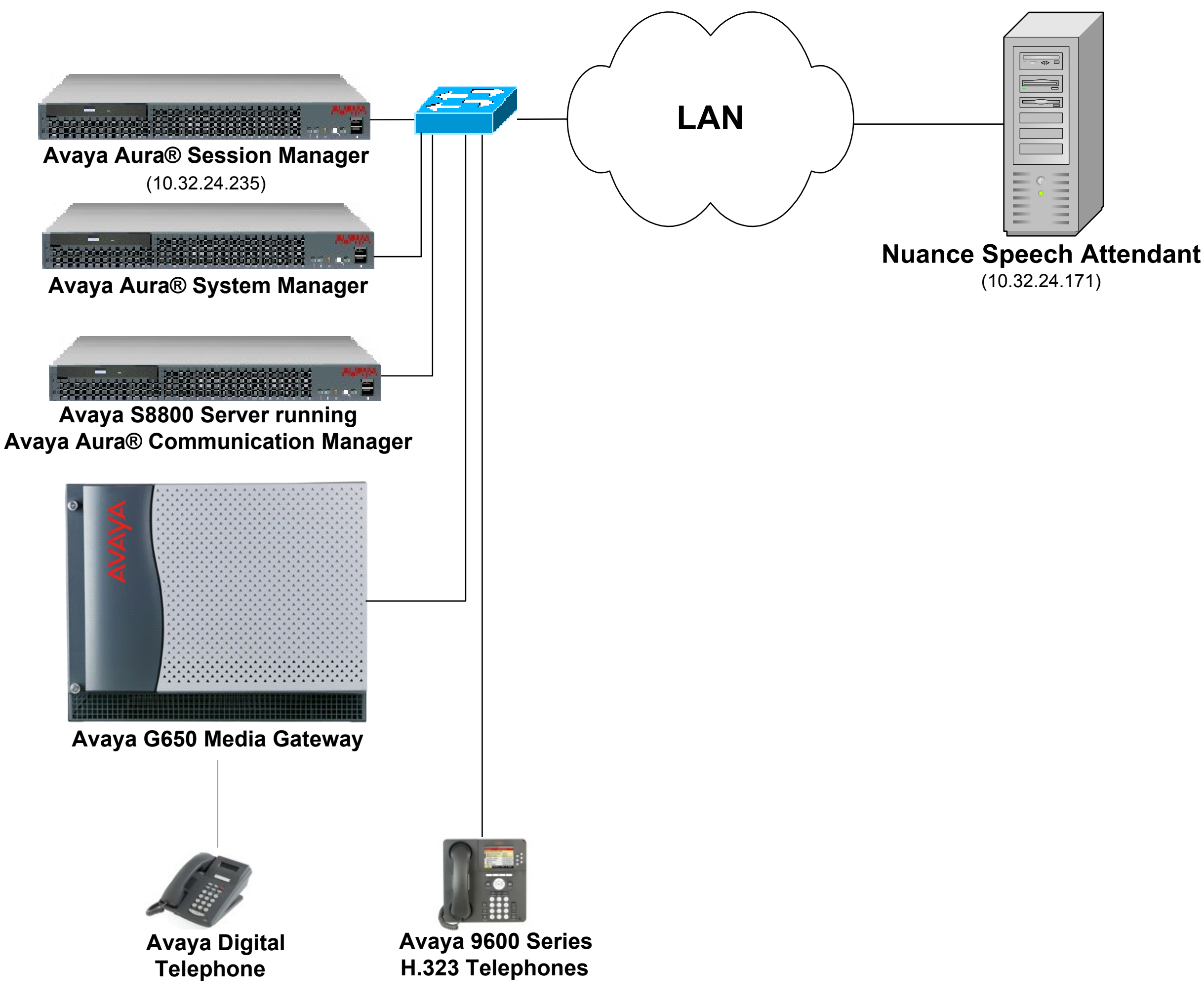


Figure 1: Configuration with Nuance Speech Attendant in an Avaya SIP Network

2.1. Equipment and Software Validated

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

Equipment	Software
Avaya S8800 Server with a G650 Media Gateway	Avaya Aura® Communication Manager 6.0 with Service Pack 1
Avaya Aura® Session Manager	6.0 (6.0.0.0.600020)
Avaya Aura® System Manager	6.0 (6.0.0.0.556-3.0.6.1)
Avaya 9600 Series IP Telephones	3.110b (H.323)
Nuance Speech Attendant (SA)	11.0 with Hotfix 5

3. Configure Avaya Aura® Communication Manager

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

- Verify Communication Manager license
- Configure a SIP trunk between Communication Manager and Session Manager
- Configure AAR Call Routing to Nuance SA

Use the System Access Terminal (SAT) to configure Communication Manager and log in with the appropriate credentials.

3.1. Verify SIP Trunk Capacity

Using the SAT, verify that SIP trunks are enabled in the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

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

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 40
           Maximum Concurrently Registered IP Stations: 18000 17
             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: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 30
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   1
                     Maximum Media Gateway VAL Sources: 250   0
           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.)
```

3.2. Configure SIP Trunk

In the **IP Node Names** form, assign an IP address and host name for the C-LAN board in the G650 Media Gateway, and the Session Manager SIP interface. The host names will be used throughout the other configuration screens of Communication Manager.

```
change node-names ip                                           Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
Gateway001          10.32.24.1
ModMsg              192.50.10.45
clancrm             10.32.24.20
default             0.0.0.0
devcon-asm          10.32.24.235
medprocrm           10.32.24.21
procr               10.32.24.10
procr6              ::
 ( 8 of 8   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G650 Media Gateway. However, Nuance SA does not support shuffling so it should be disabled either in the IP Network Region or in the SIP signaling group configured below. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region '1') is specified in the SIP signaling group.

```
change ip-network-region 1                                     Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: no
      Codec Set: 1              Inter-region IP-IP Direct Audio: no
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 34
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 7
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                  RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to Nuance SA. The form is accessed via the **change ip-codec-set 1** command. Note that IP codec set '1' was specified in IP Network Region '1' shown above. The default settings of the **IP Codec Set** form are shown below. Nuance SA supports G.711.

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

                          IP Codec Set

    Codec Set: 1

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

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

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tcp*.
- Specify the C-LAN board and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TCP port value of *5060* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was disabled on this form since Nuance SA does not support shuffling.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*. Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 50                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 50            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: clancrm               Far-end Node Name: devcon-asm
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1

Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? n
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

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

```
add trunk-group 50                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 50                   Group Type: sip          CDR Reports: y
  Group Name: To devcon-asm               COR: 1        TN: 1        TAC: 1050
   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: 50
                                                    Number of Members: 10
```

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

```
add trunk-group 50                                          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

                                Modify Tandem Calling Number: no

 Show ANSWERED BY on Display? y
```

Configure the **Private Numbering Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with '7' whose calls are routed over any trunk group, including SIP trunk group "50", have the extension sent to the Nuance SA for proper CLID handling.

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

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 5  7                                            5     Total Administered: 1
                                                          Maximum Entries: 540
```

3.3. Configure AAR Call Routing to Nuance Speech Attendant

In the **AAR Digit Analysis Table**, specify a **Dialed String** that would match the Nuance SA extension (22005) and specify the **Route Pattern** that will be used to route the call.

```
change aar analysis 7                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all           Percent Full: 1

          Dialed            Total     Route     Call    Node   ANI
          String          Min  Max  Pattern    Type    Num    Reqd
    22005                 5    5      50        aar            n
```

In the *Route Pattern* form, specify the SIP trunk group and set the **Numbering Format** for the route preference to *unk-unk*. This would prevent a + sign to be prepended to the dialed digits in conjunction with using private numbering format in the SIP trunk group.

```
change route-pattern 50                                         Page   1 of   3
                    Pattern Number: 50  Pattern Name: To devcon-asm
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 50   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                               unk-unk   none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

4. Configure Avaya Aura® Session Manager

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

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities corresponding to Session Manager and Communication Manager
- Entity Links, which define the SIP trunk parameters used by 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
- Session Manager, corresponding to the Session Manager Server to be managed by System Manager

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

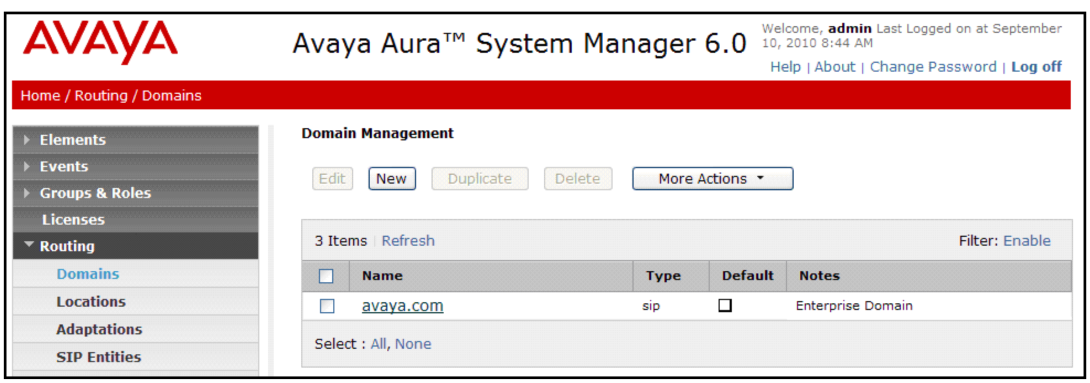
4.1. Specify SIP Domain

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

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

Click **Commit**.

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



4.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

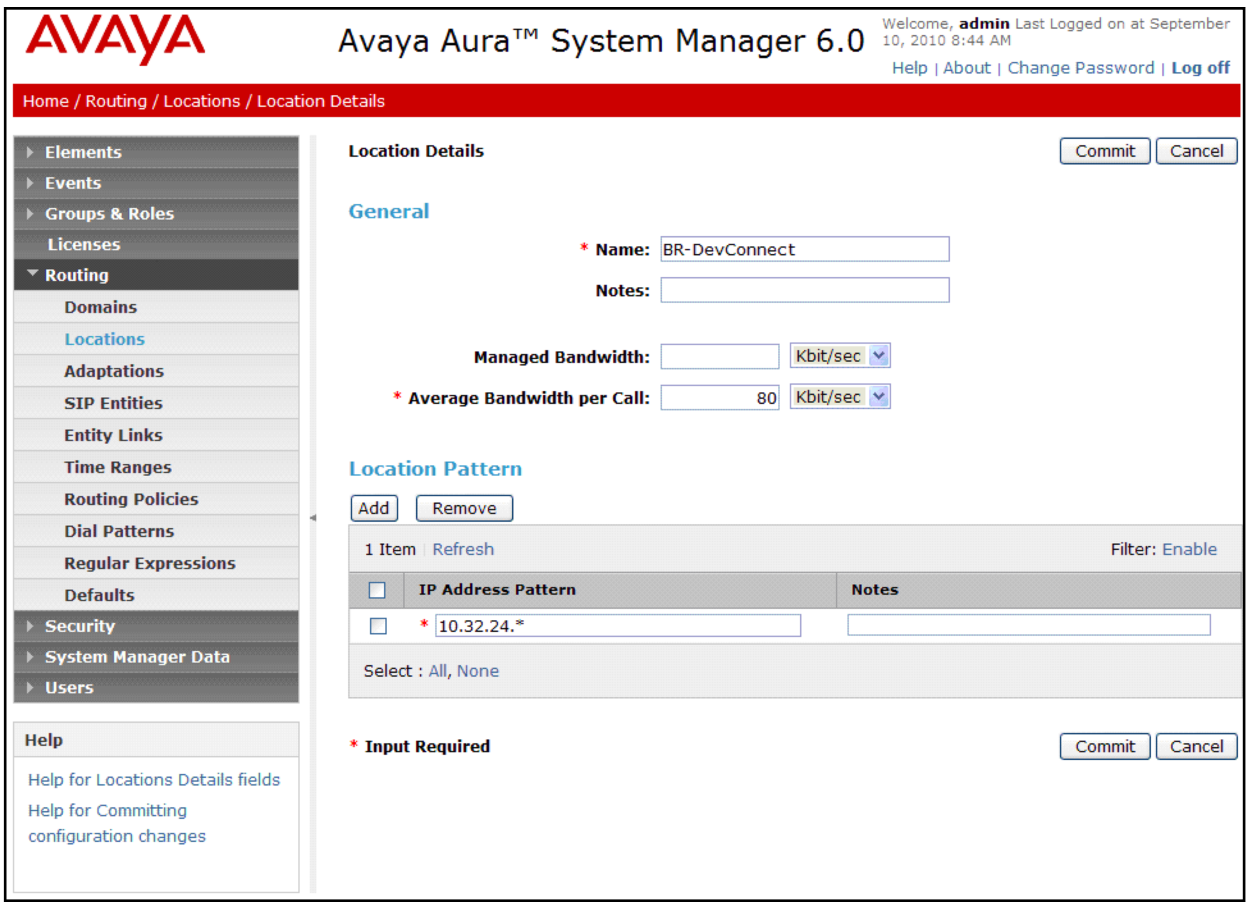
Under *General*:

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

Under *Location Pattern*:

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

The screen below shows addition of the *BR-DevConnect* location, which includes the Communication Manager and Session Manager. Click **Commit** to save the Location definition.



4.3. Add SIP Entities

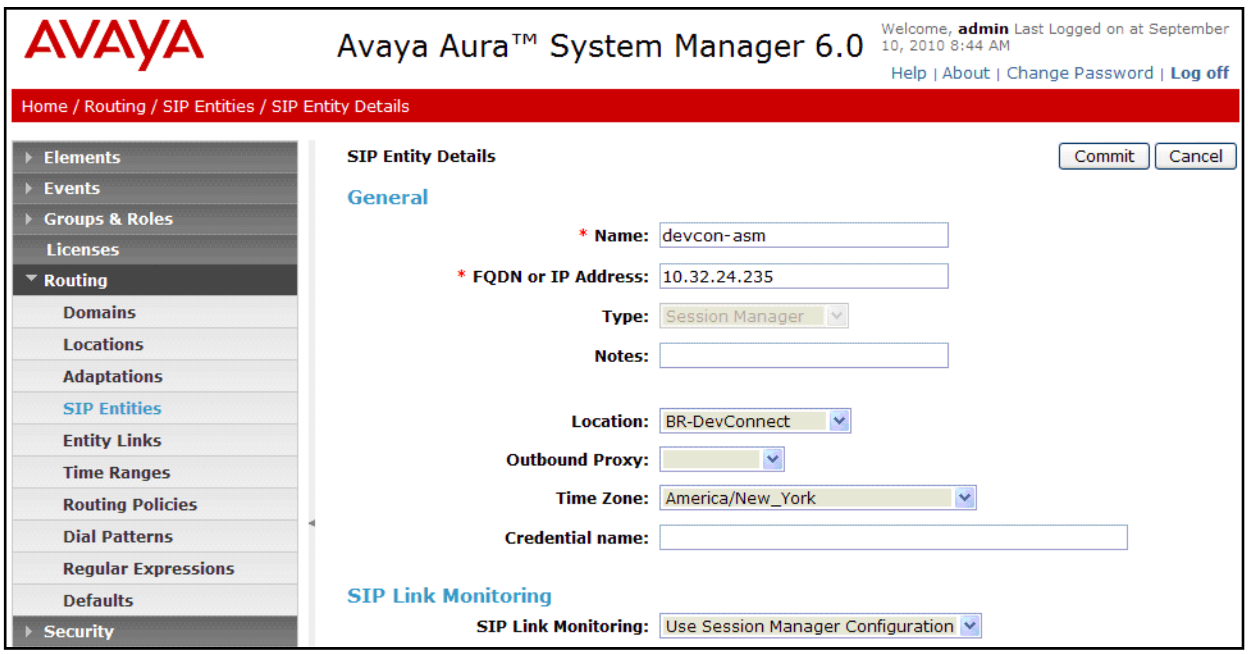
In the sample configuration, a SIP Entity is added for Session Manager, the C-LAN in the G650 Media Gateway, and Nuance Speech Attendant

4.3.1. Avaya Aura® Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

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



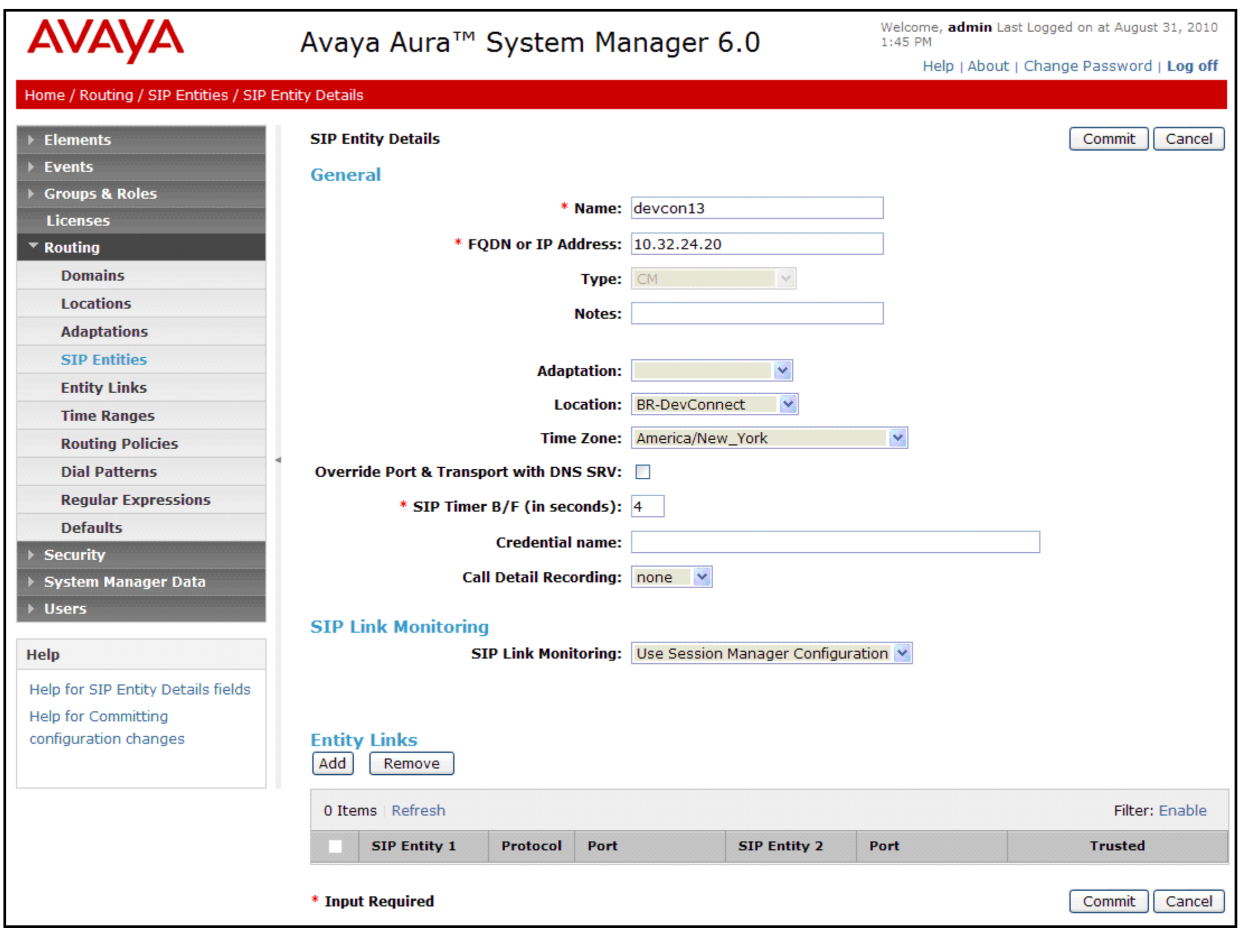
4.3.2. Avaya Aura® Communication Manager

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

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., C-LAN board) on the telephony system.
- **Type:** Select *CM*.
- **Location:** Select one of the locations defined previously.
- **Time Zone:** Time zone for this location.

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



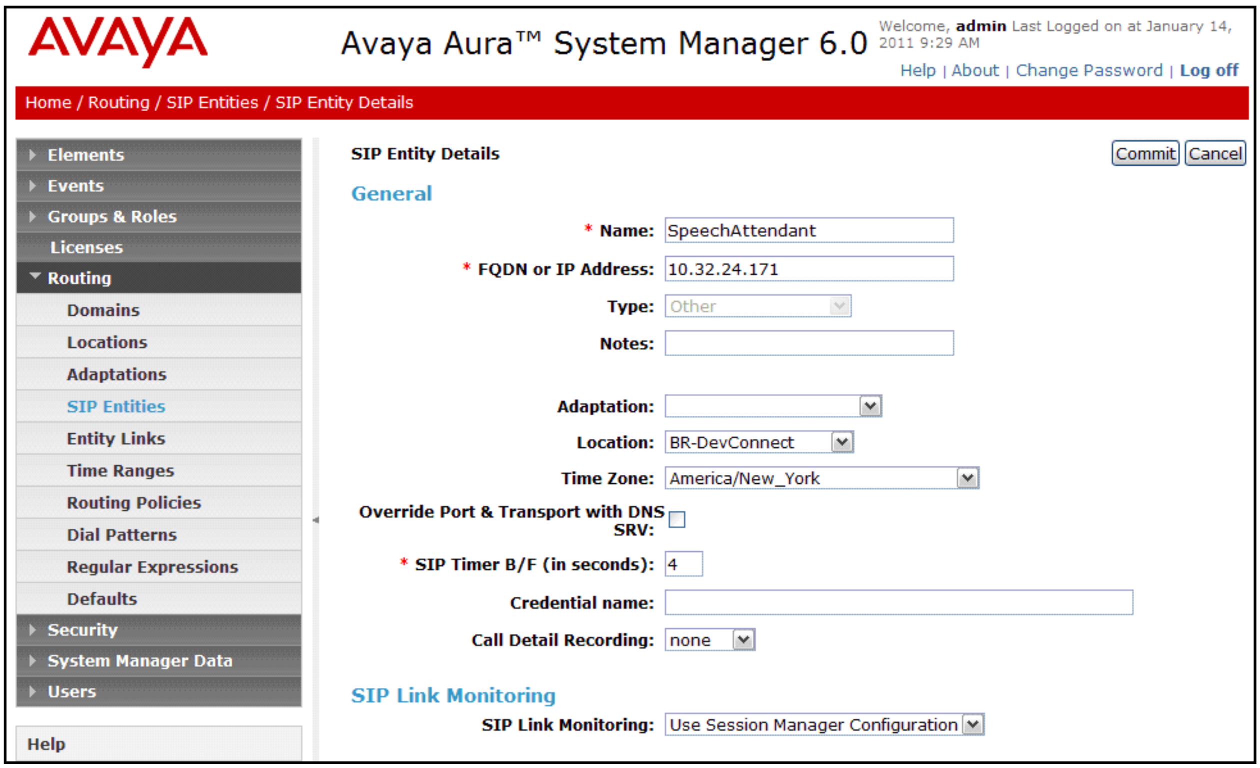
4.3.3. Nuance Speech Attendant

A SIP Entity must be added for Nuance SA. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** Nuance SA IP address.
- **Type:** Select *Other*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

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

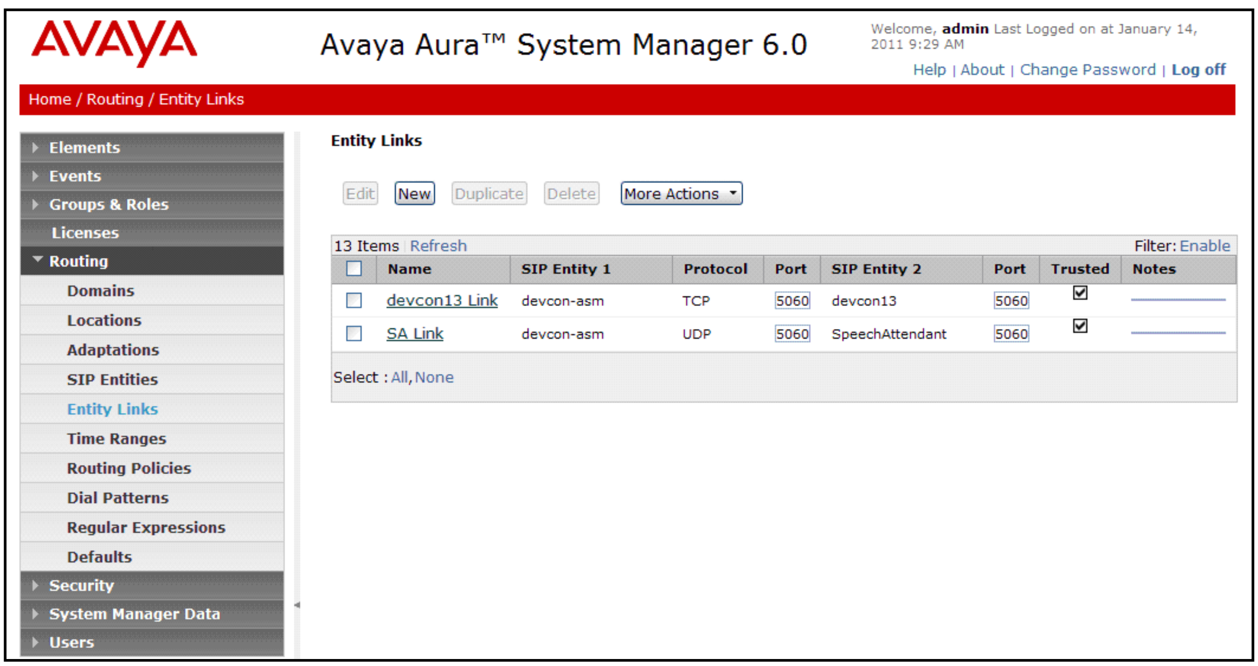


4.4. Add Entity Link

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

- **Name:** A descriptive name (e.g., *devcon13 Link* or *SA Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager or Nuance SA.
- **Port:** Port number on which the other system receives SIP requests.
- **Trusted:** Check this box. *Note: If this box is not checked, calls from the associated SIP Entity specified in **Section 4.4** will be denied.*

The following screen displays two Entity Links after being configured per the instructions above. The first entity link is for Session Manager and Communication Manager and the second entity link is for Session Manager and Nuance SA.



4.5. Add Routing Policies

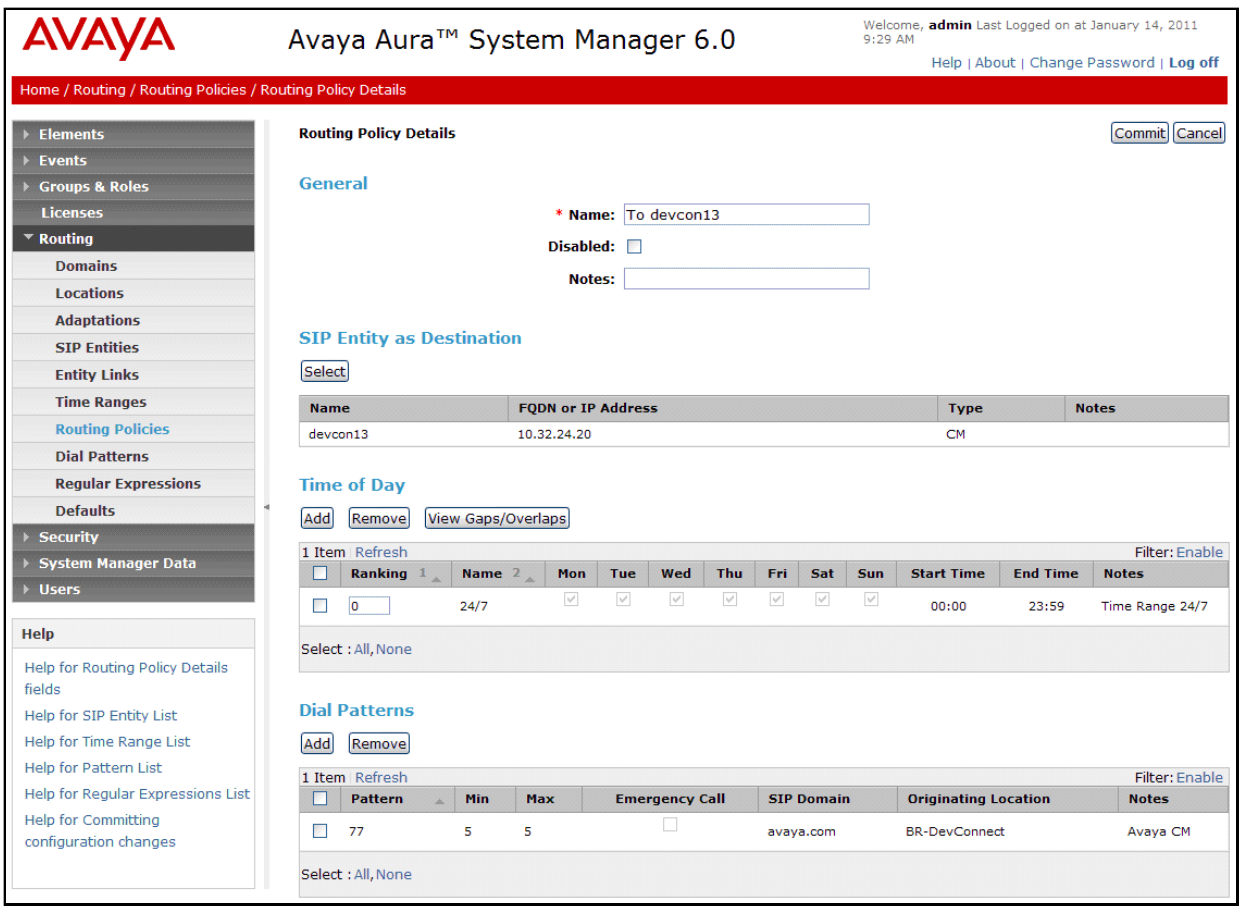
Routing policies describe the conditions under which calls will be routed to the SIP Entities. Two routing policies were added – one for Communication Manager and one for Nuance SA. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

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

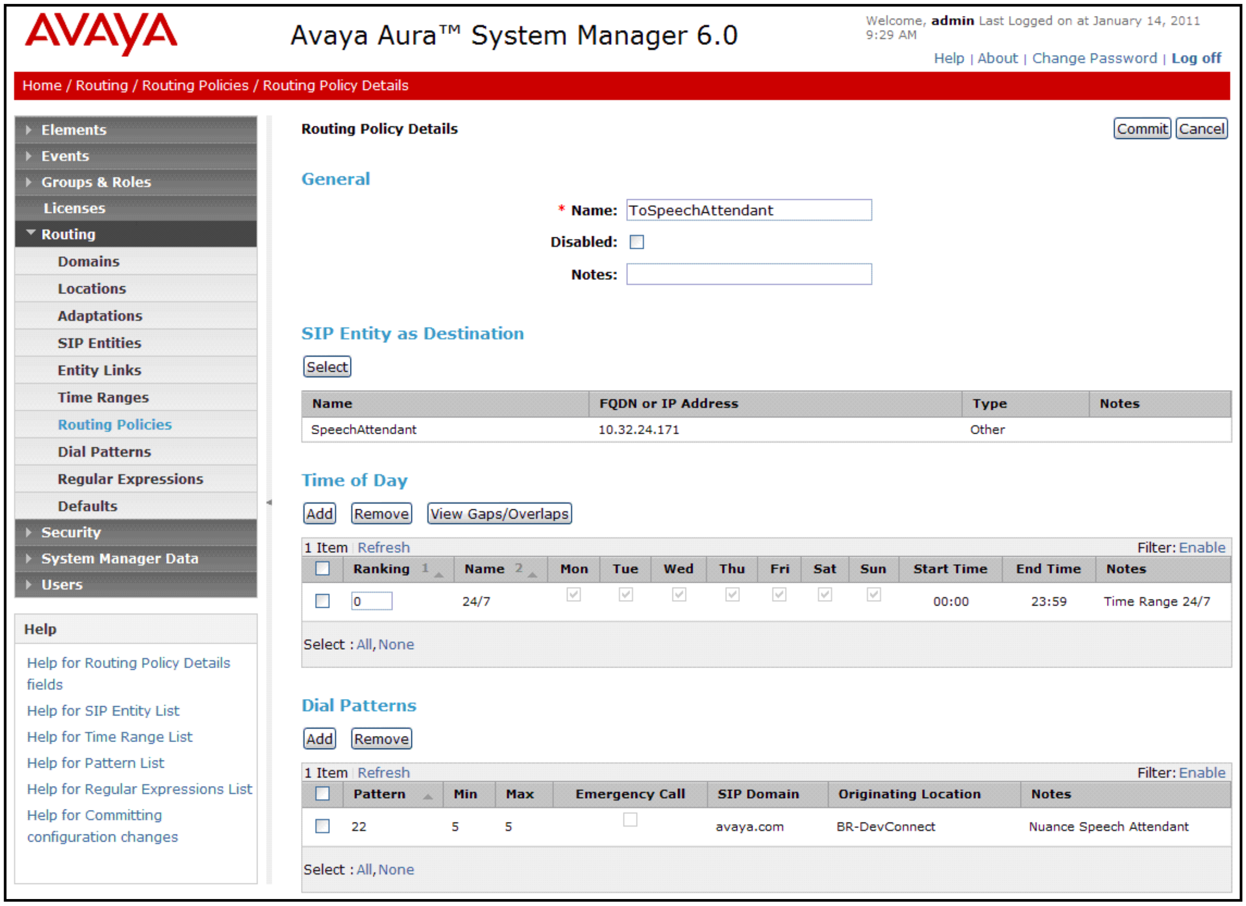
Under *SIP Entity as Destination*:
Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Under *Dial Patterns*:
Enter the dialed digits that would be routed to the specified SIP entity.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screen shows the Routing Policy for Communication Manager.



The following screen shows the Routing Policy for Nuance SA.



4.6. Add Dial Patterns

Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 5-digit extensions beginning with "77" reside on Communication Manager, extension "22005" is the Nuance SA number. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

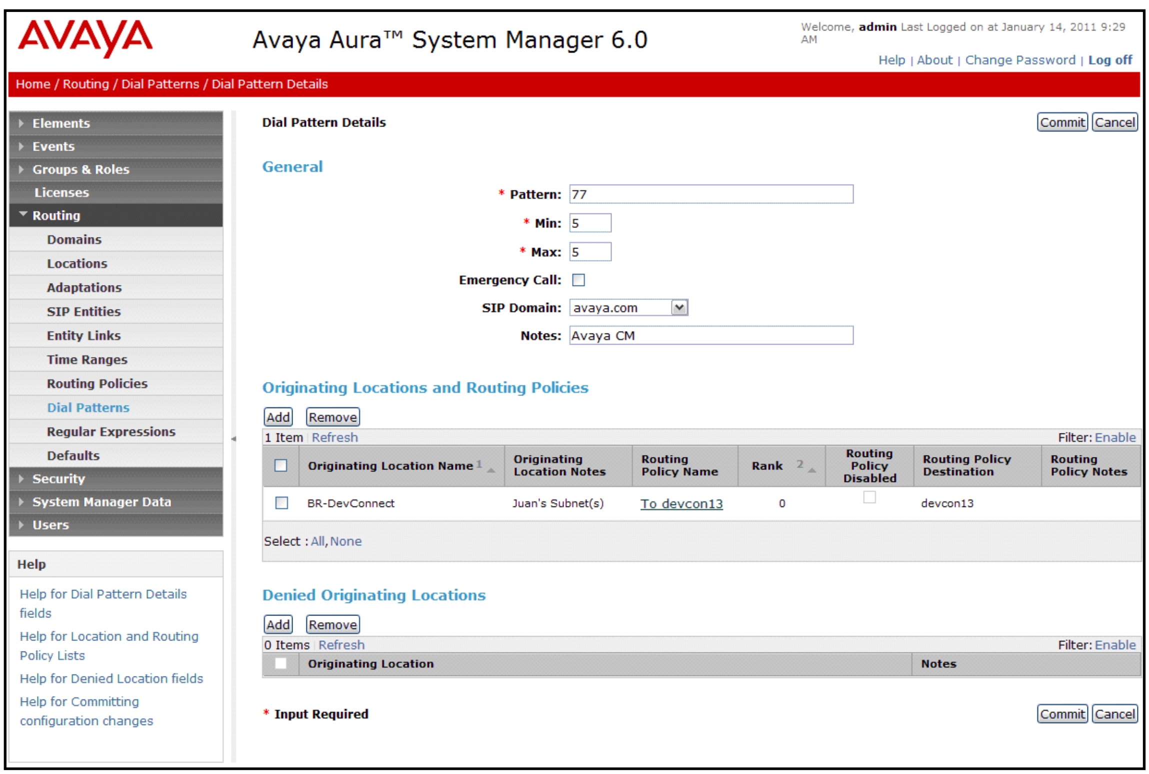
Under *General*:

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

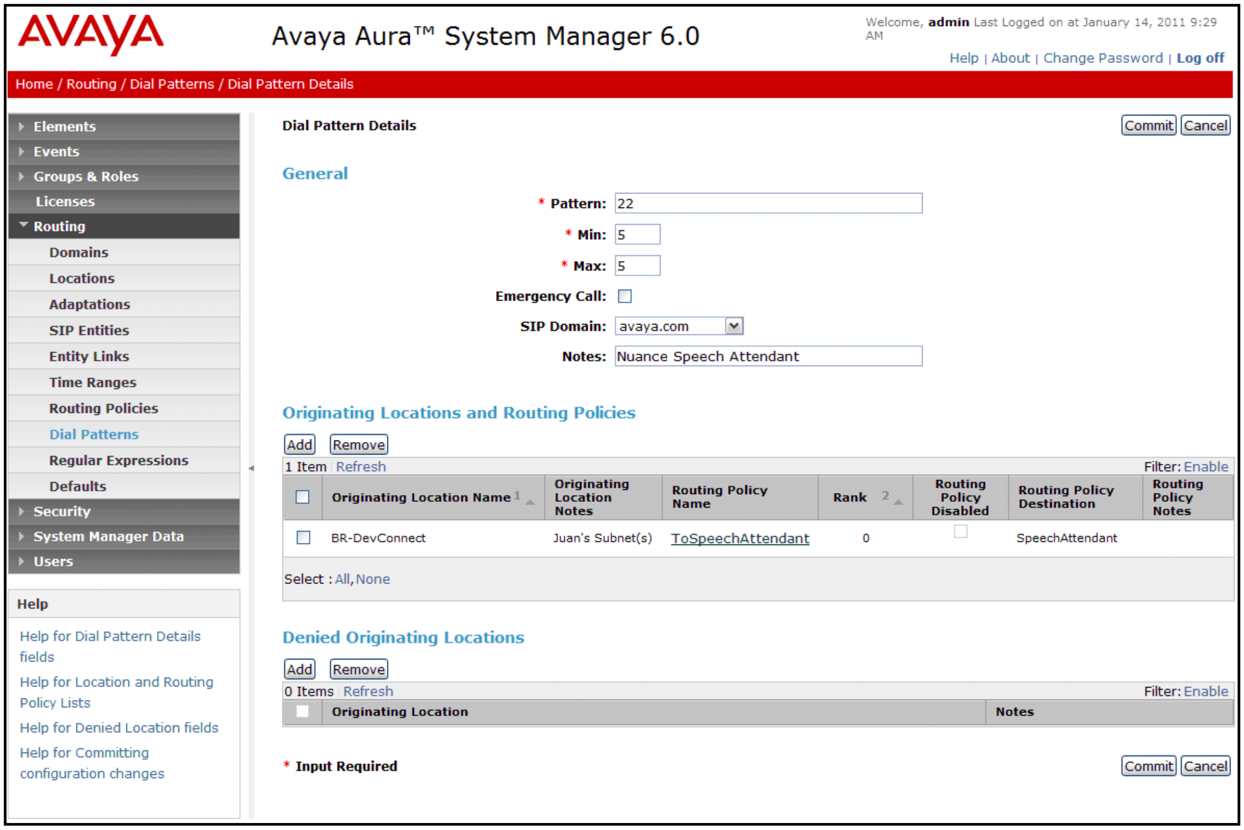
Under *Originating Locations and Routing Policies*:
Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. The following screen shows the dial pattern definitions for local extensions on Communication Manager. Click **Commit** to save this dial pattern.

The following screen shows the dial pattern definition for Communication Manager extensions beginning with "77".



The following screen shows the dial pattern definition for the Nuance SA number (22005).



4.7. Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add** (not shown), and fill in the fields as described below and shown in the following screen:

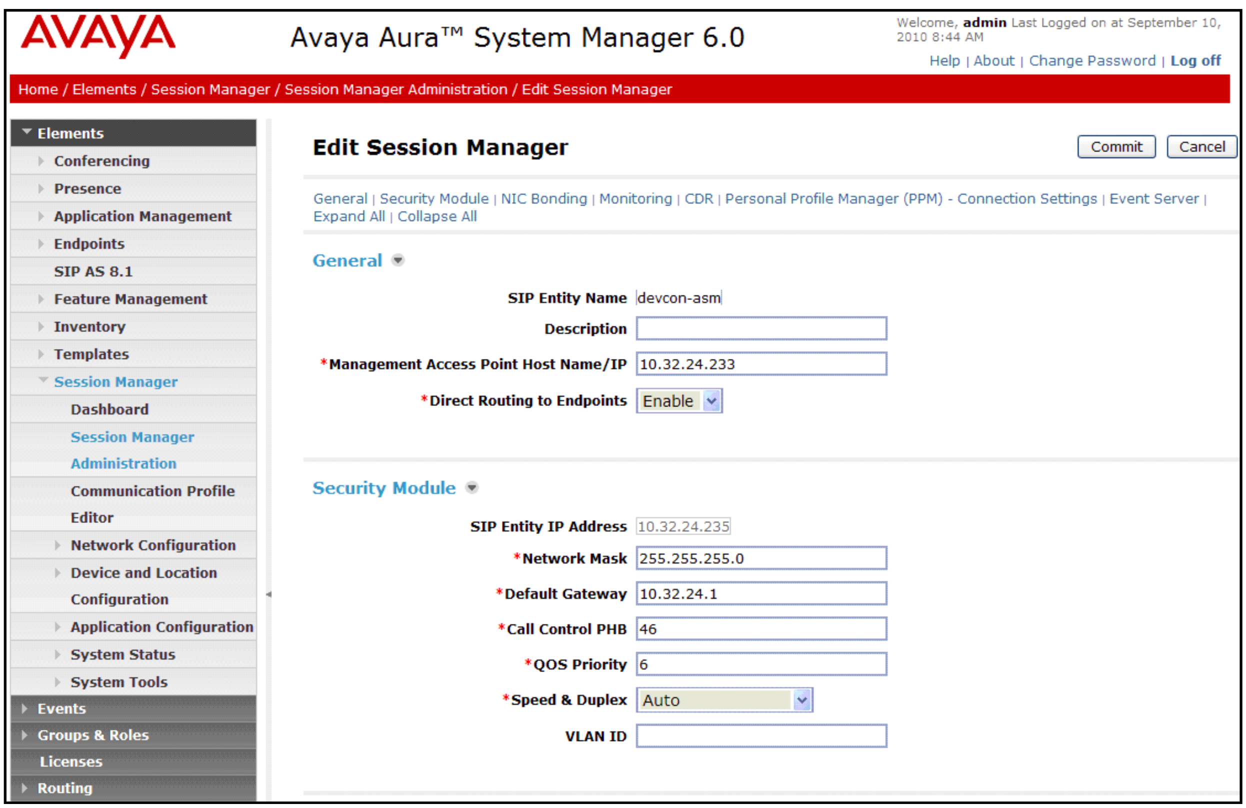
Under *Identity*:

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

Under *Security Module*:

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

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



5. Configure Nuance Speech Attendant

This section covers the procedure for configuring Nuance Speech Attendant (SA). The procedure includes the following areas:

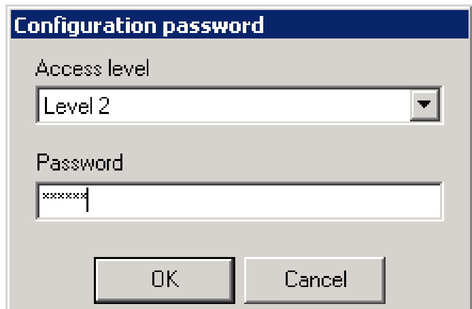
- Configure SIP interface in the Configuration Panel
- Configure the number of SIP ports supported by Nuance SA

Note: Configuration of transfer entries and menus in Nuance SA is outside the scope of these Application Notes and will not be covered.

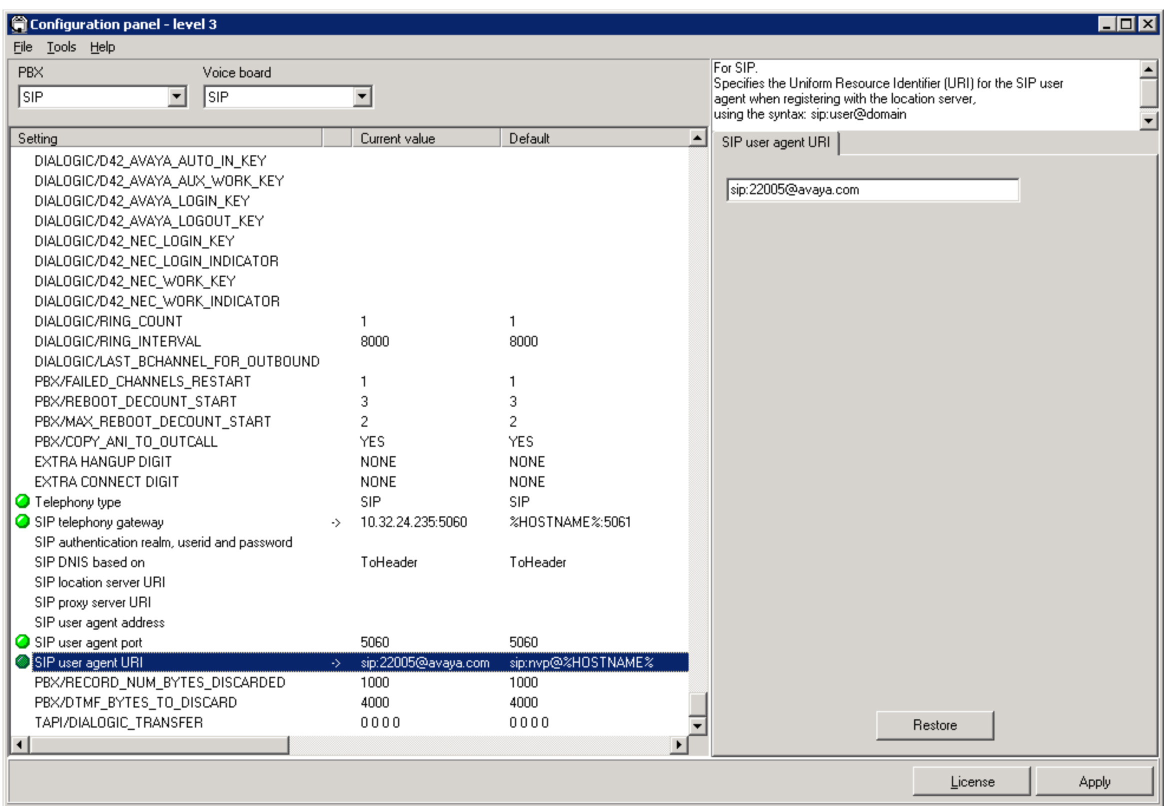
Nuance SA is configured through **Admin Tools** which can be started by navigating to **Start→Programs→Speech Attendant→Admin Tools**. The initial screen is displayed below.

5.1. Configure SIP Interface in Configuration Panel

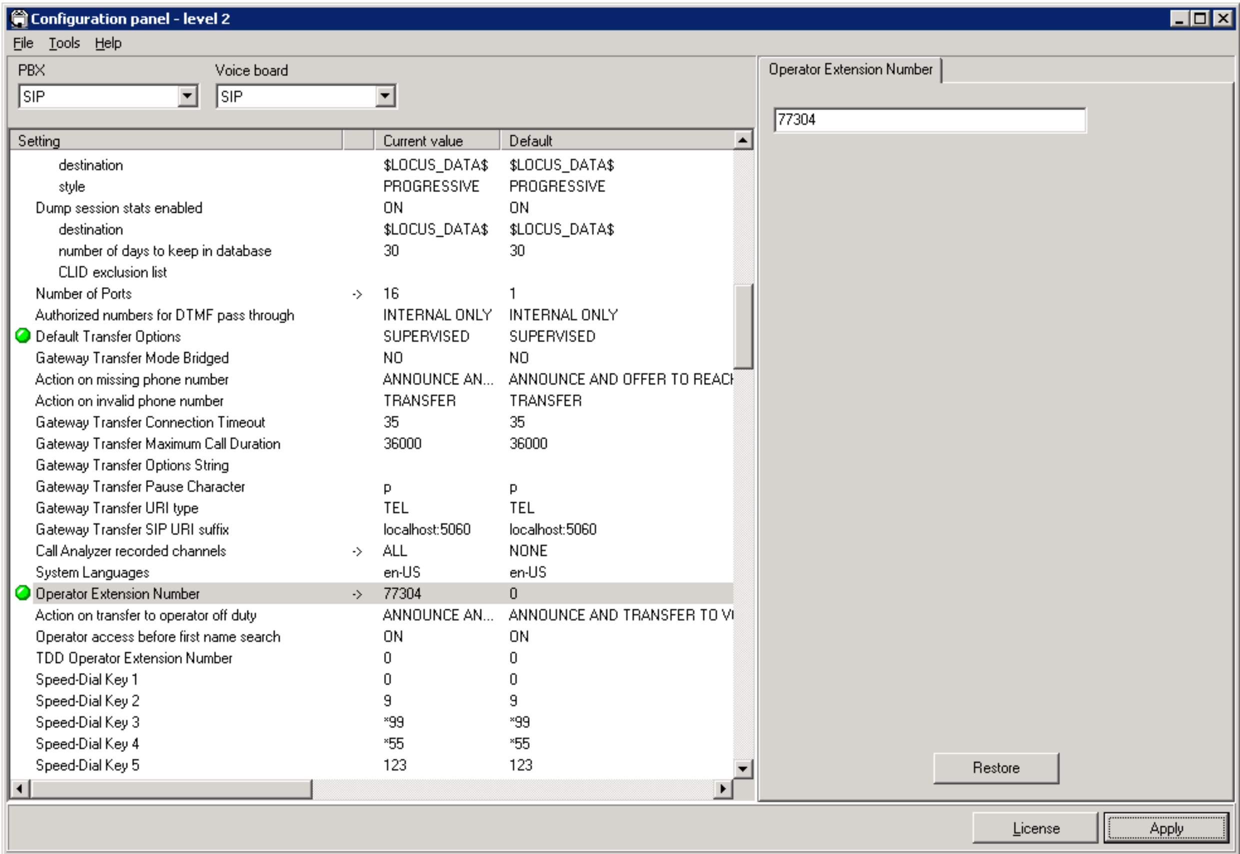
To open the **Configuration Panel**, click on this option in the **Admin Tools** window shown above. The login prompt will be displayed to the user as shown below. Log in with the appropriate credentials using *Level 2* access level.



The **Configuration Panel** shown below is displayed. The **Configuration Panel** allows the SIP interface, transfer mode, and operator extension number to be configured. SA supports blind and supervised transfers. To configure the SIP interface, set the **PBX** and **Voice Board** fields to *SIP*. Next, set the **Telephony type** field to *SIP* and specify the Session Manager IP address and port in the **SIP telephony gateway** field. Lastly, configure the SIP port that SA listens on in the **SIP user agent port** field and specify the SIP URI for SA, which contains its extension and domain name (e.g., *sip:22005@avaya.com*).

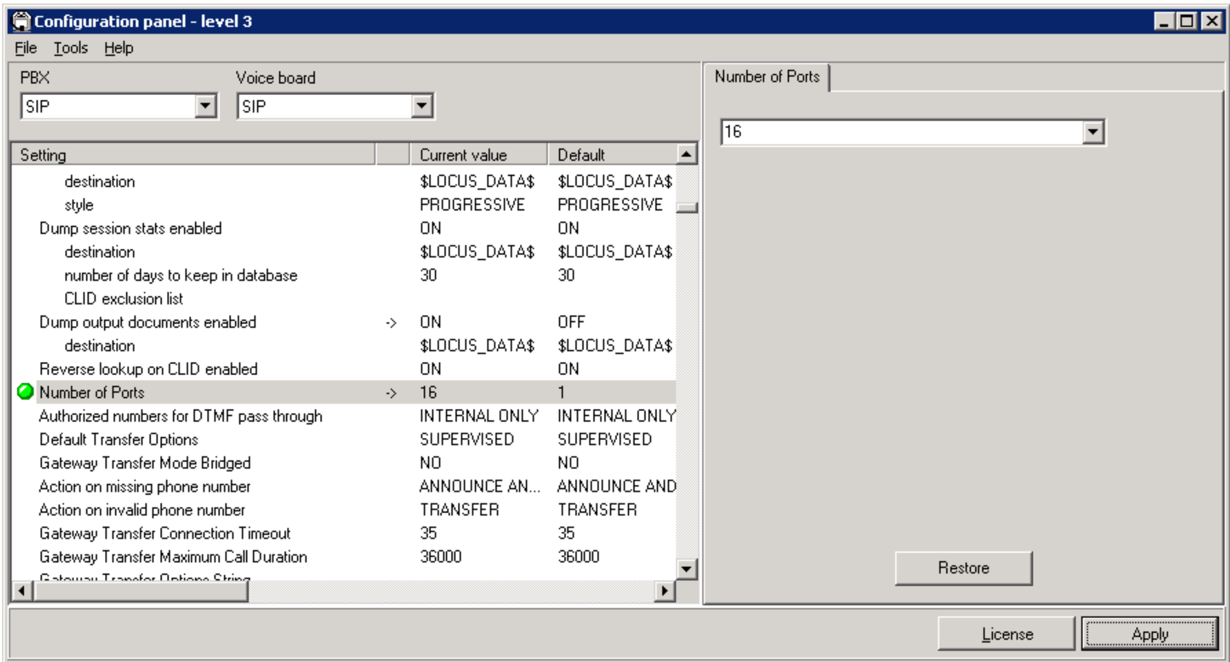


In the **Configuration Panel**, specify the transfer mode (supervised or unsupervised) in the **Default Transfer Options** field, and specify the **Operator Extension Number**, which should be set to a valid extension on Communication Manager.

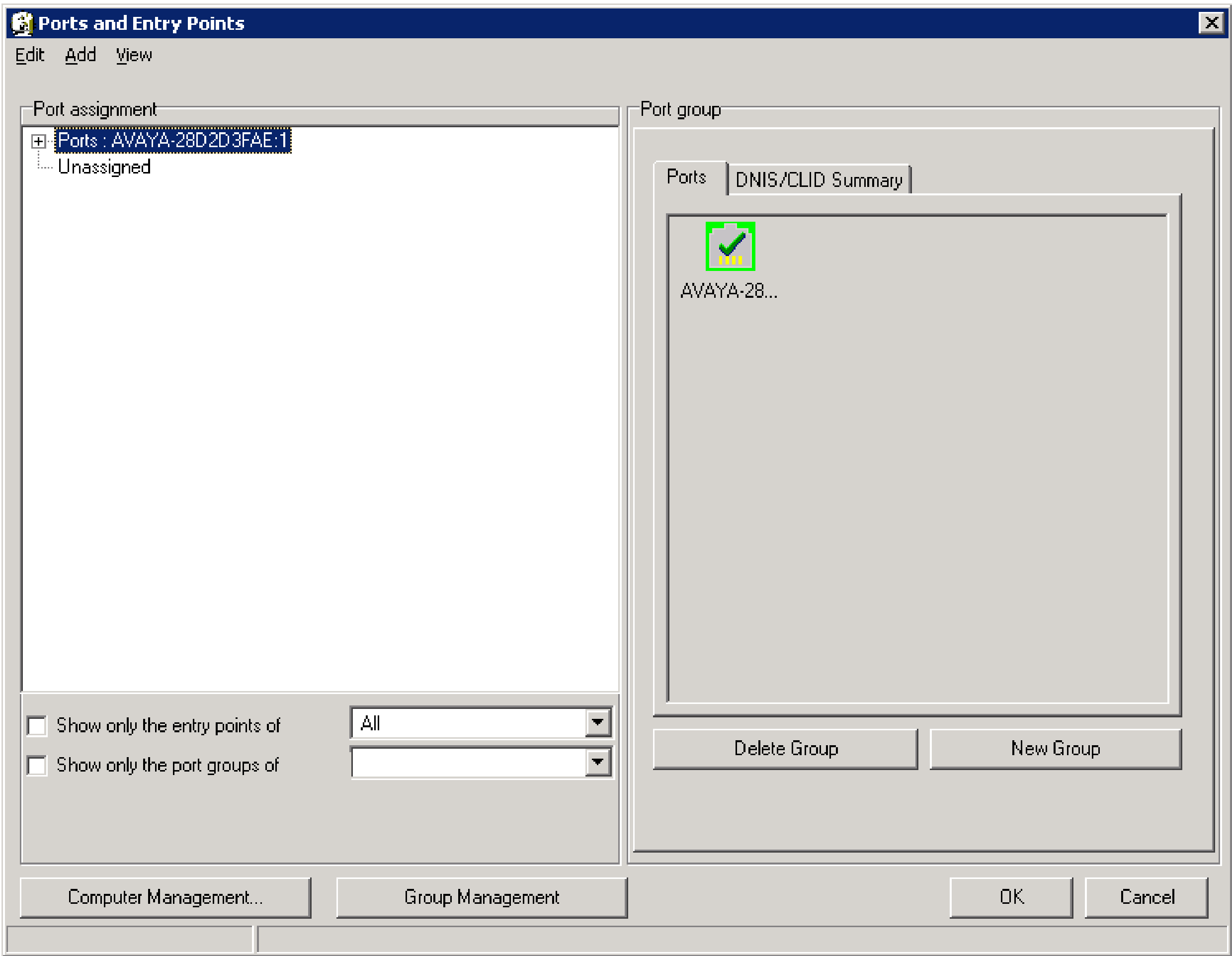


5.2. Configure Number of Ports Supported

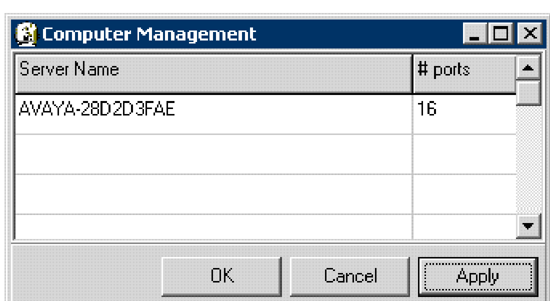
In the **Configuration Panel**, set the **Number of Ports** field to the desired value. In this example, 16 ports were configured. Click **Apply** and then close the **Configuration Panel**. Allow the SA application to be restarted when prompted.



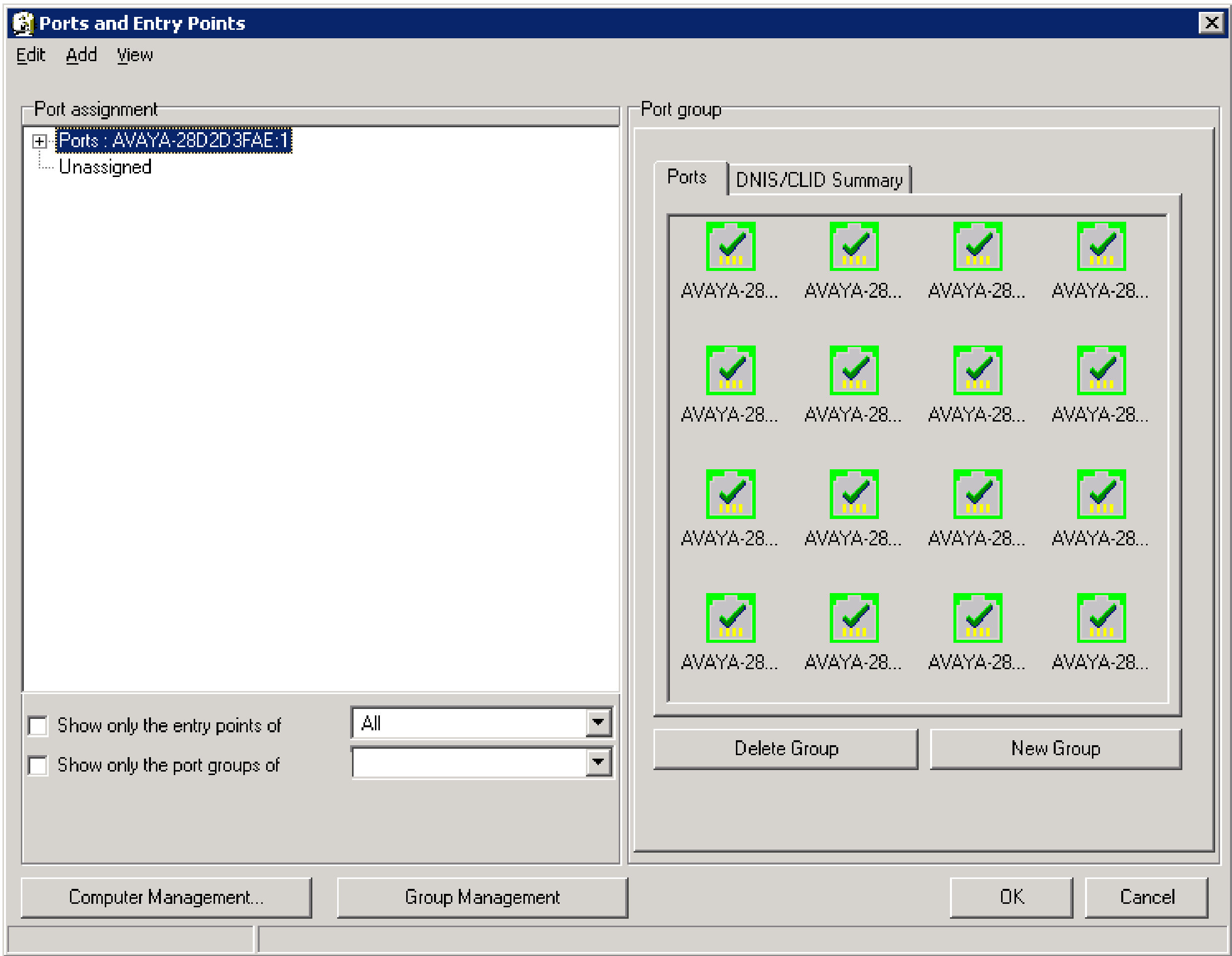
From **Admin Tools**, click on the **Phone Directory and Menu Editor** option and login with the appropriate credentials. In the **Menu Editor** (not shown), navigate to **Edit→Ports and Entry Points** to display the window below. Once the **Ports and Entry Points** window is opened, click on the **Computer Management** button at the bottom of the screen.



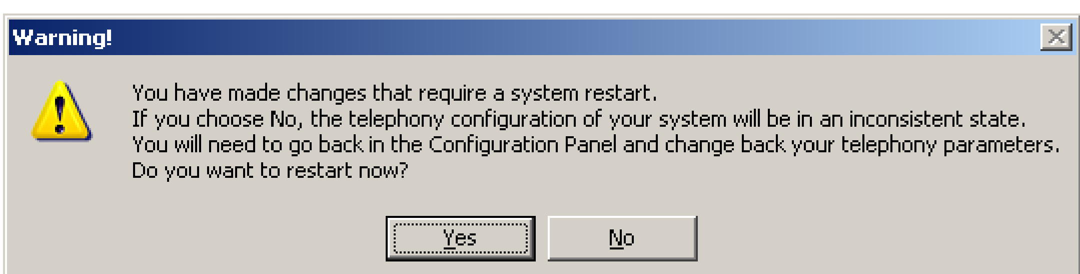
In the **Computer Management** window, set the # ports column to the desired value. In this example, 16 ports were configured. Click **OK**.



Under the Port group section of the **Ports and Entry Points** window, a list of ports in the system will be displayed. Make sure they are all green (enabled). If the ports are grey (disabled), click on each icon to enable them as shown below. Click **OK**.



Next, close the **Configuration Panel**. Allow the system to restart when prompted by clicking **Yes** in the dialog box below.



6. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. Feature testing focused on Nuance SA successfully recognizing spoken names and extensions entered via DTMF, and then transferring the call to the correct destination. Blind and supervised transfers were verified. Other features covered included barge-in / no barge-in, adding new transfer entries, recording caller utterances, and accessing Maintenance Mode and Personal Administration Mode to record name and change PIN.

Serviceability testing focused on verifying the ability of the Nuance SA to recover from adverse conditions, such as server restarts, power failures, and disconnecting cables to the IP network.

All test cases passed with the following observations noted:

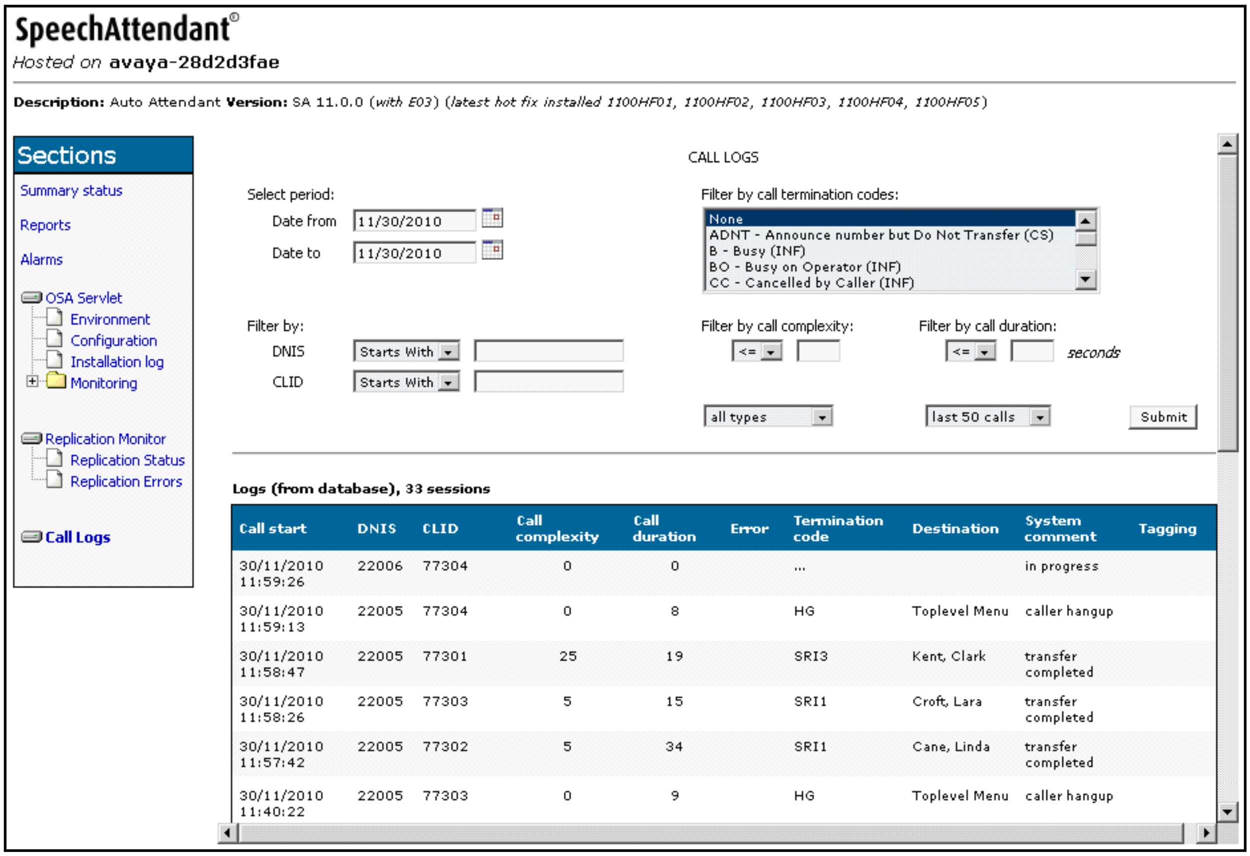
- If all SIP ports on Nuance SA are busy, the call is not re-routed by Session Manager to another destination when an alternate route is specified in its Route Policies and Dial Patterns. Session Manager does not re-route calls based on a busy condition (i.e., a "486 Busy Here" message is received). However, if a "503 Service Not Available" message is returned to Session Manager, it will re-route the call to an alternate destination, if configured properly. In a future SA hotfix, Nuance will provide the ability to return a 503 message to Session Manager when it is busy. If SA is down or not running, Session Manager will re-route the call.

- Nuance SA does not support shuffling (i.e., direct IP-IP media) and should be disabled either in the IP Network Region or SIP signaling group.

7. Verification Steps

This section provides the verification steps that may be performed to verify that Nuance SA is operating properly with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP integration.

1. Place a call to Nuance SA. From the Nuance SA server, open the SA Monitor from **Admin Tools** and login with the appropriate credentials. Verify that SA detects an active call as shown below. For example, the first entry indicates that the call is *in progress*.



2. Verify that the Nuance SA greeting is heard and SA transfers the call to the proper destination specified in a spoken name or extension entered via DTMF.

8. Conclusion

These Application Notes describe the configuration steps required to integrate Nuance Speech Attendant with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP integration. All feature and serviceability test cases were completed successfully.

9. Additional References

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

[1] *Administering Avaya Aura® Communication Manager*, June 2010, Release 6.0, Issue 6.0, Document Number 03-300509.

[2] *Administering Avaya Aura® Session Manager*, August 2010, Issue 3, Release 6.0, Document Number 03-603324.

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.