Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Nuance Speech Attendant with Avaya Aura® Session Manager R6.3 and Avaya Communication Server 1000 R7.6 – 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 Communication Server 1000 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 and be transferred 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 Communication Server 1000 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 be transferred to the requested party.

2. General Test Approach and Test Results

The interoperability compliance testing included feature and serviceability test cases. No performance testing was done and the tests listed in **Section 2.1** were executed and verified.

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 Feature testing focused on Nuance SA successfully recognizing spoken names and extensions entered via DTMF, and then transferring the call to the correct destination. Only supervised transfer was tested and verified. Other features covered included: DNIS and CLID handling, 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.

2.2. Test Results

The objectives outlined in **Section 2.1** were verified and all test cases passed.

- Nuance SA does not support unsupervised/blind transfer call.
- When SA initials a transfer and sends REFER message to CS1000, CS1000 accepts the REFER and sends Re-Invite to hold current call however SA response back with 488 Not Acceptable Here. This does not cause an issue on the call transfer but it is listed here as observation.
- There is no calling name displayed on the transferee's CS1000 phone after Nuance SA completes the transfer, however the calling number is still displayed properly on the transferee phone.

2.3. Support

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

- Web: www.network.nuance.com
- Email: SpeechAttendant.Support@nuance.com.
- Phone: +1 (866) 434-2564 or +1 (514) 390-3922.

3. Reference Configuration

Figure 1 illustrates the setup used to verify the Nuance Speech Attendant (SA) solution with Avaya Aura® Session Manager and Avaya Communication Server 1000. Nuance SA is deployed on a dedicated server running Windows 2008 R2 Enterprise server. Session Manager interfaces to Communication Server 1000 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 Server 1000 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.

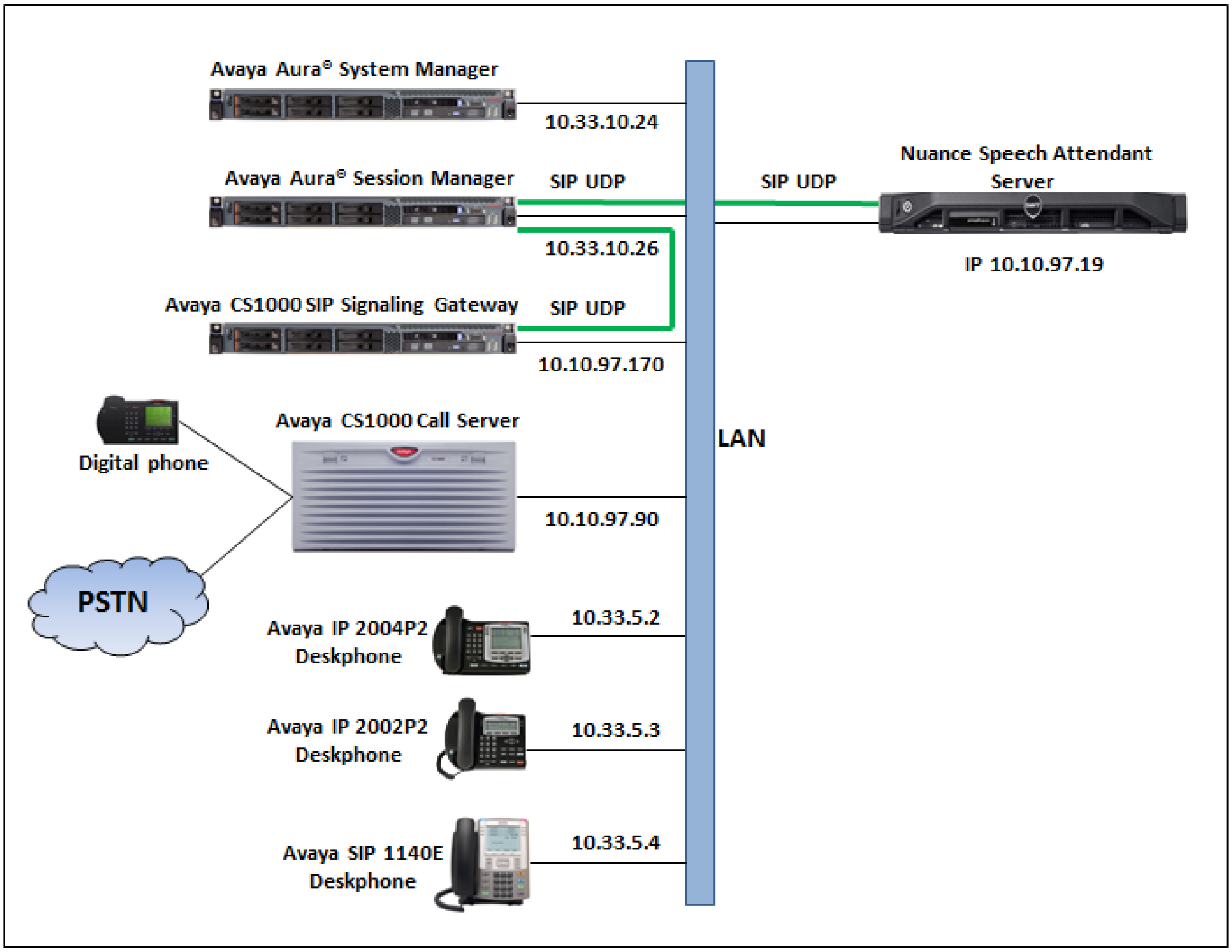


Figure 1: Reference Configuration Diagram

4. Equipment and Software Validated

The following equipment and software was used during the lab testing:

Equipment	Software Version
Avaya Aura® Session Manager running on S8800 Server	Release: 6.3.7 Build No. 6.3.0.0.630002-6.3.7.637004
Avaya Aura® System Manager running on S8800 Server	6.3.7 – FP3 Build No. 6.3.0.8.5682 - 6.3.8.3204
Avaya Communication Server 1000 running on CPPM card	R7.6 Service Pack 4 DepList1 Created 2013-12-17
Avaya Media Gateway Controller CSP Version Boot Version	 MGCC DC03+ MGCB BA18
Avaya 2004P IP Phone	0604DC0
Avaya SIP 1140 IP Phone	4.3
Avaya Digital 3904	N/A
Nuance Speech Attendant	V12

5. Configure Avaya Communication Server 1000

The assumption is that route, trunk and dialing plan of the Avaya CS1000 have been configured. This section only describes the details on how to configure the Avaya CS1000 Call Server to connect to the Session Manager via SIP Signaling Gateway using the Element Manager.

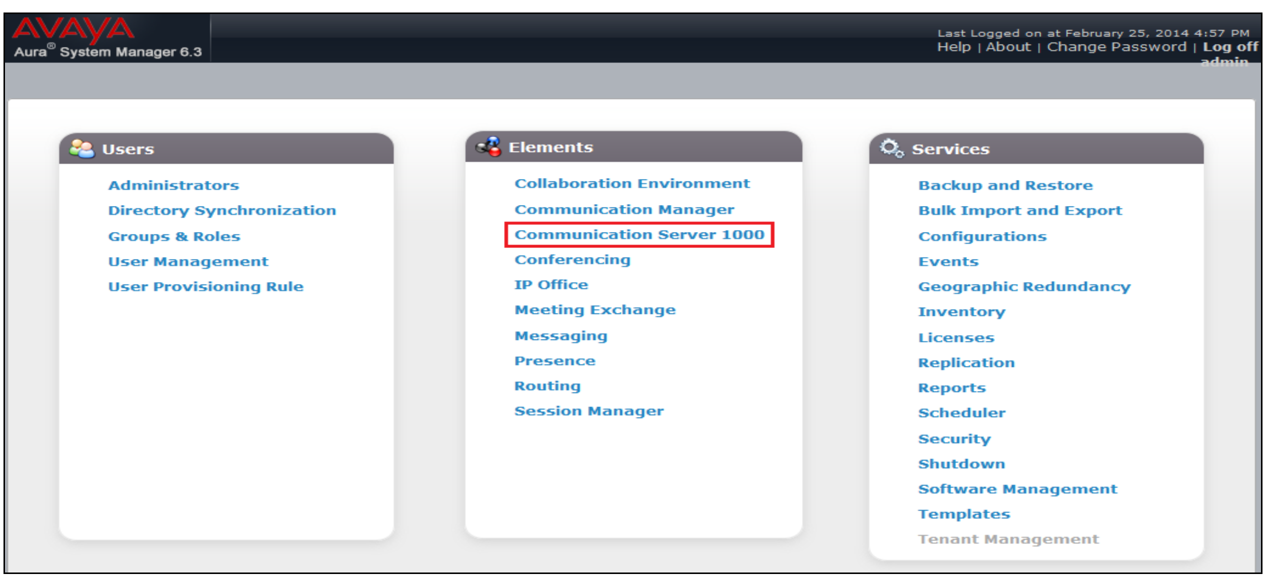
Prerequisites: An Avaya CS1000 server which has been:

- Installed with CS 1000 Release 7.6 Linux Base.
- Joined CS 1000 Release 7.6 Security Domain.
- Deployed with SIP Trunk Application.

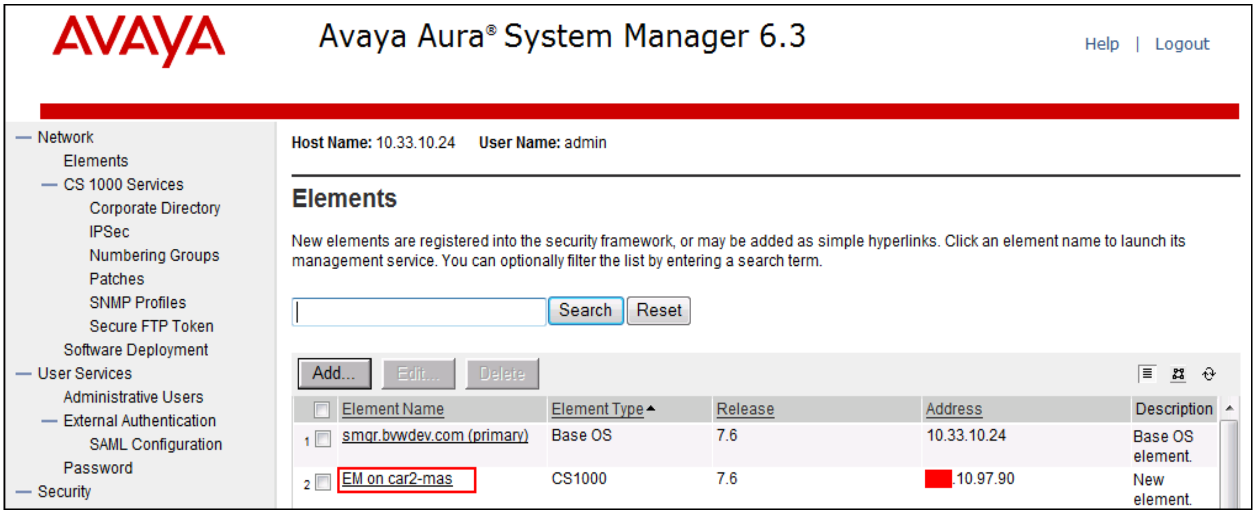
For more information on CS 1000 installation, maintenance, and upgrades, see **Section 10**. The following software packages are enabled in the key code. If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at http://www.avaya.com.

Package Mnemonic	**Package Number**	**Package Description**	**Package Type (New or Existing or Dependency)**	**Applicable Market**
SIP	406	SIP Service package	New package	Global
FFC	139	Flexible Feature Codes	Existing package	Global
SIPL_ Nortel	415	Avaya SIP Line	Existing package	-
SIPL_3ThirdParty	416	Third Party SIP Line	Existing package	-

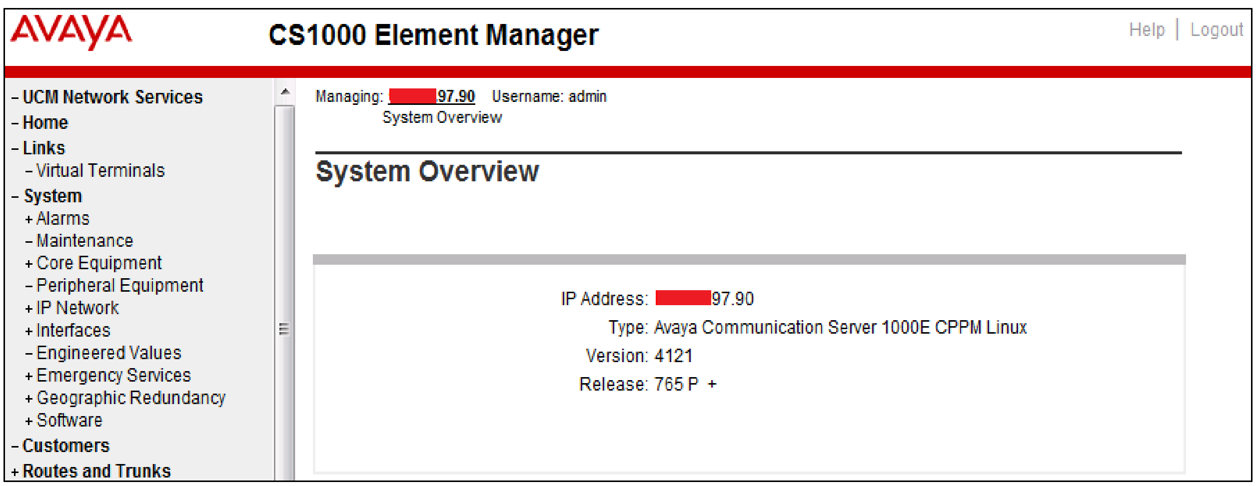
Since release 7.6 Avaya CS1000 Element Manager is integrated to System Manager, to access the Element Manager of CS1000 first log in the System Manager. The screen below shows the System Manager home page with Communication Server 1000 entry in the Elements table. Click on the **Communication Server 1000** to access to CS1000 UCM, the UCM webpage will be opened in the new window.



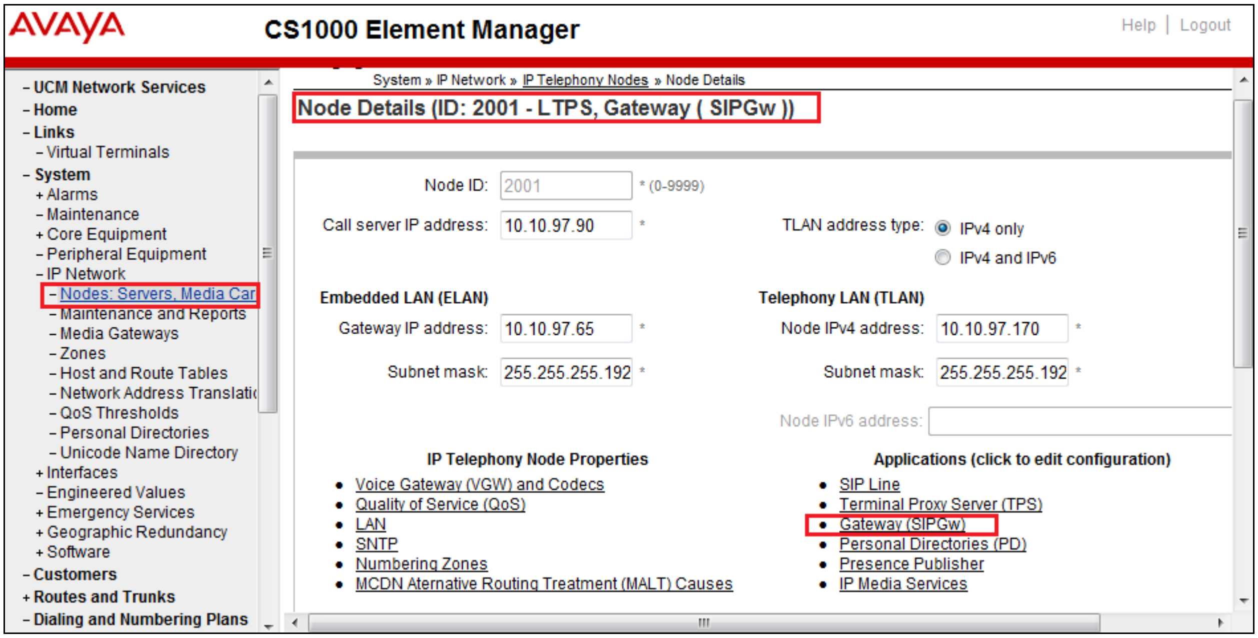
The **Elements** page is shown in the following screenshot. Click Element Name of the CS1000 that needs to be accessed as highlighted in the red box.



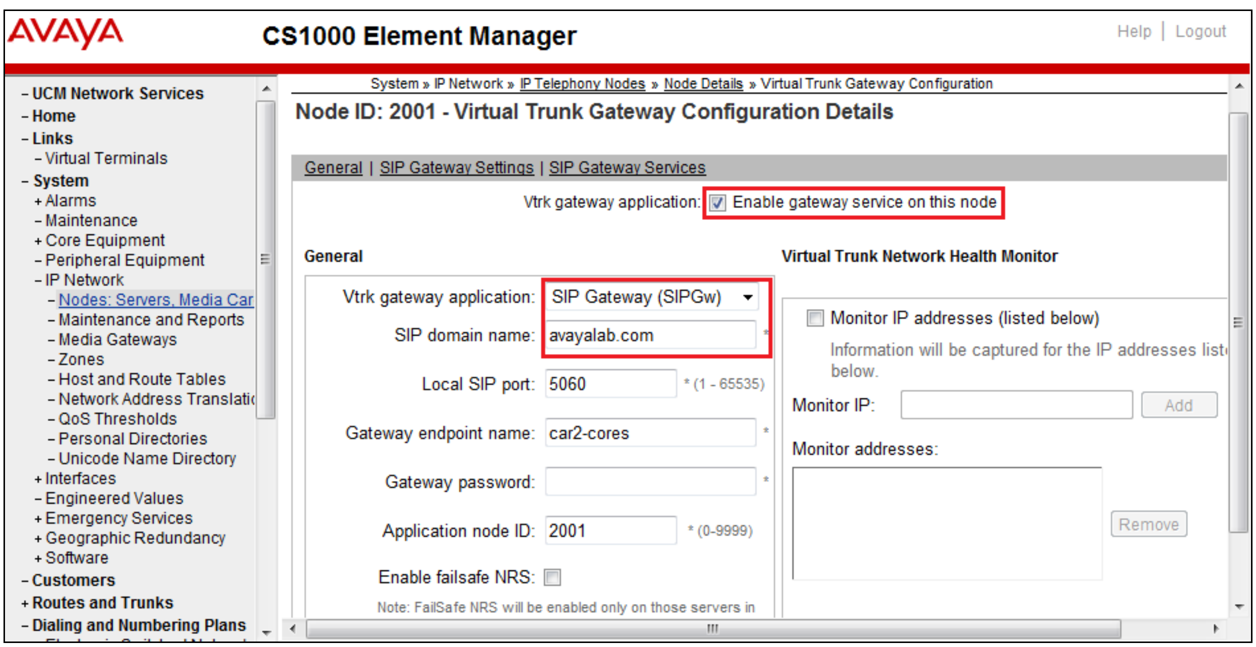
The following screenshot shows the CS1000 Element Manager **System Overview** page.



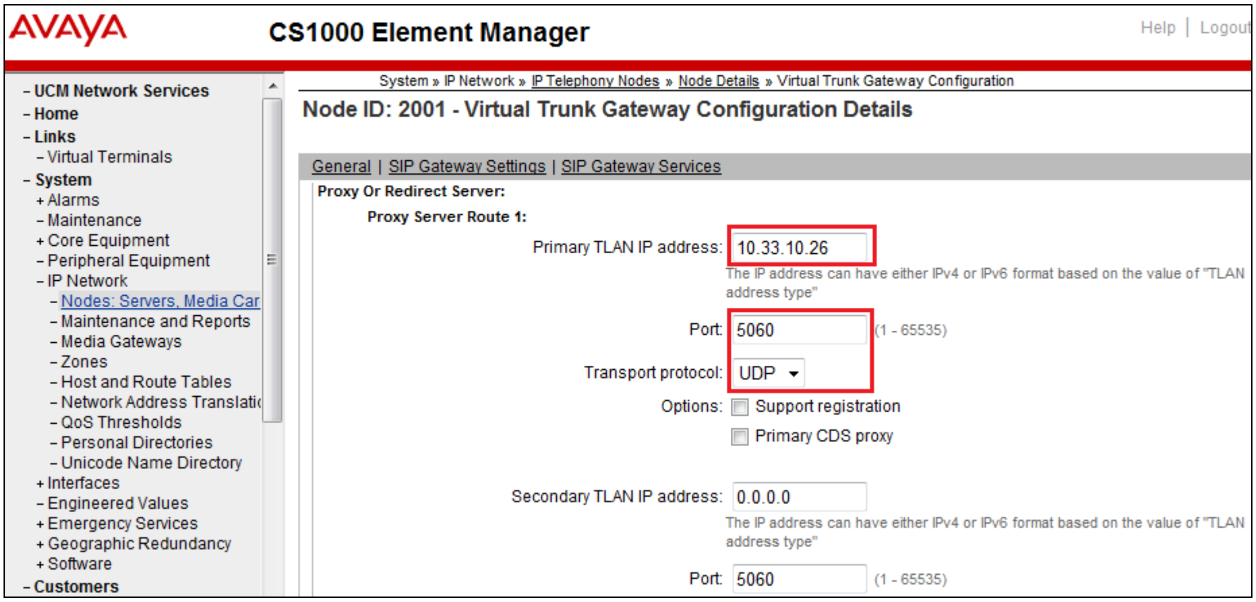
From the left menu column of the EM page, navigate to **System → IP Network → Nodes: Servers, Media Cards**. The **Node ID Telephone** page will appear (not shown). Click on the **Node ID # 2001,** which is **the LTPS, Gateway (SIPGw).** The **Node Details** page will appear. Scroll down under the **Applications**, click on the **Gateway (SIPGw)** link, the **Virtual Trunk Gateway Configuration Details** page will appear in the next 2 screen shots.



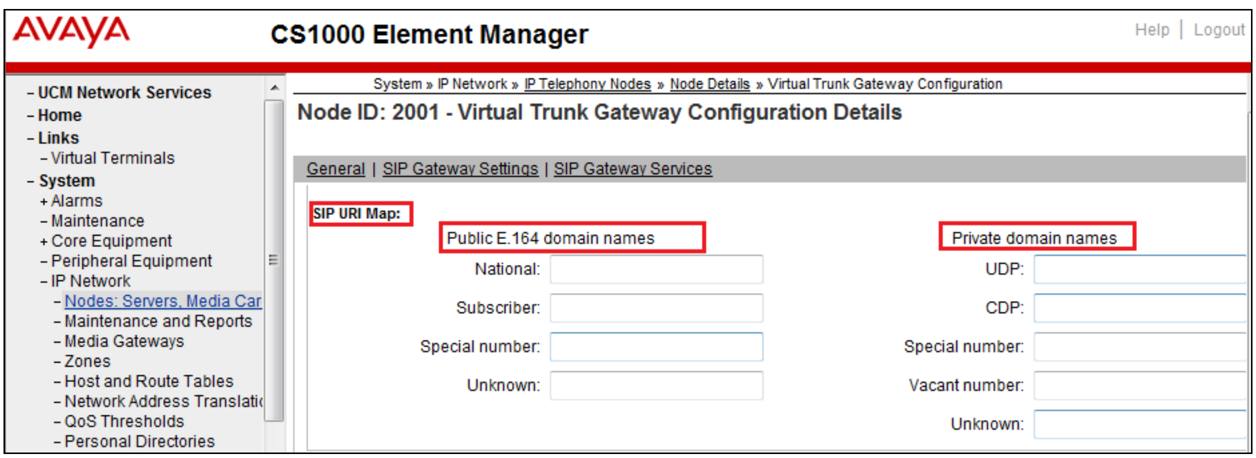
Enter the information highlighted in the red-box for the **General** and **SIP Gateway Settings**. All other fields are left as default. Click **Save**. Note: **SIP domain name avayalab.com** should be matched with what was created in **Section 6.1**.



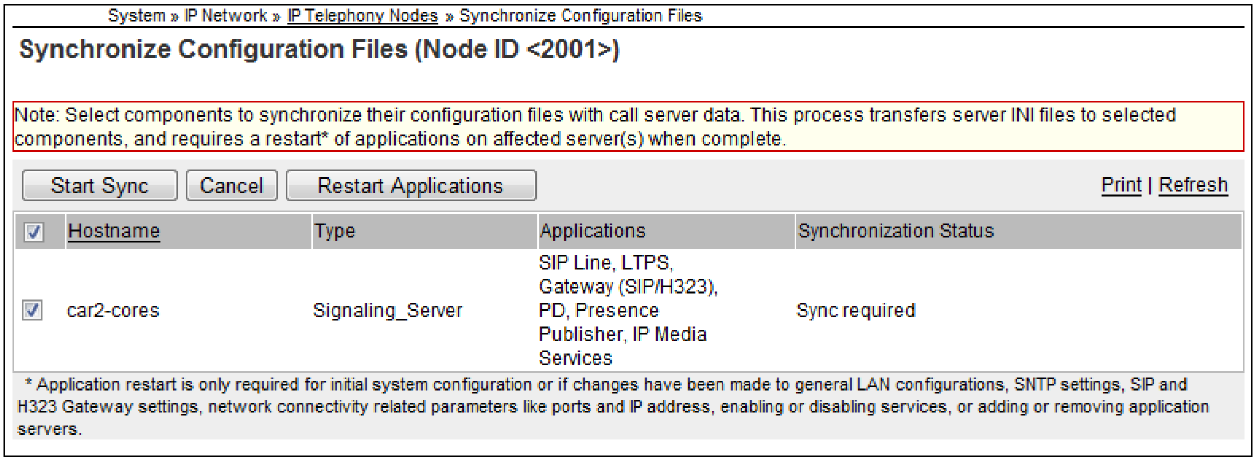
The **Primary TLAN IP** address is the IP address used in **Section 6.3.1** which is the Session Manager IP address 10.33.10.26.



On the same page, scroll-down the parameters box to the **SIP URI Map** section. Under the **Public E.164 domain names** and **Private domain names** subsections leave all fields as blank.



After click **Save**, the system will bring back the **Node ID** page (not shown). Then click **Save** button on the **Node ID** page and that will take the user to the **Node Saved** page (not shown). Click on the **Transfer Now** button, when finished it will bring the user to **Synchronize Configuration Files** page. Then click **Start Sync** button (not shown) to complete the configuration saved process.



6. 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 Server 1000.
- 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.

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL “https://<ip-address>”, where <ip-address> is the IP address of System Manager. Log in with the appropriate credentials (not shown). Select the **Routing** link after logging in.

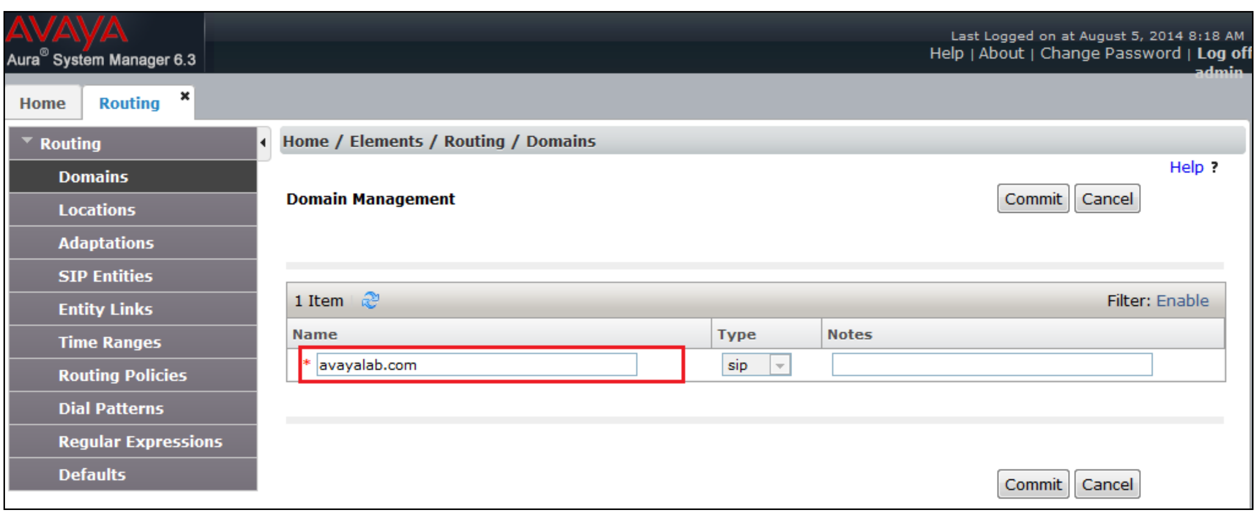
6.1. Specify SIP Domain

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 (not shown). The following screen will then be shown. Fill in the following:

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

Click **Commit**.

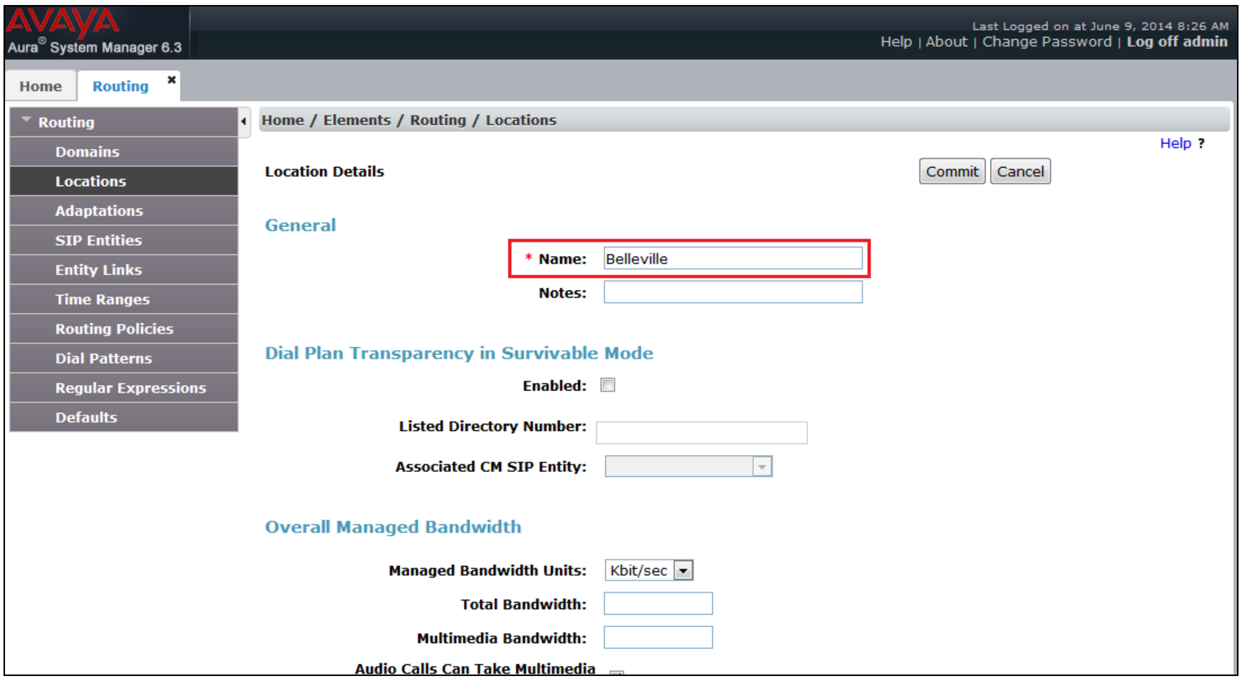
Since the sample configuration does not deal with any other domains, no additional domains need to be added.



6.2. Add Locations

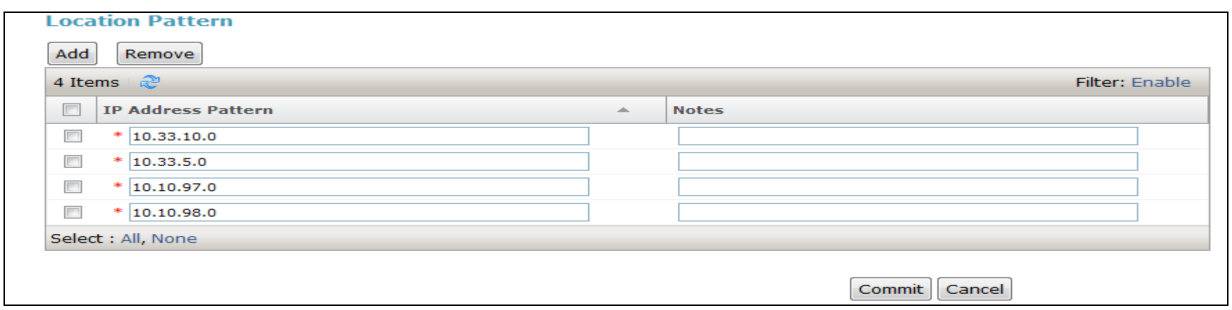
Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and then click the **New** button on the right (not shown). The following screen will then be shown. Fill in the following:

- 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 a location which includes the Communication Server 1000, Session Manager, and Speech Attendant. Click **Commit** to save the Location definition.



6.3. Add SIP Entities

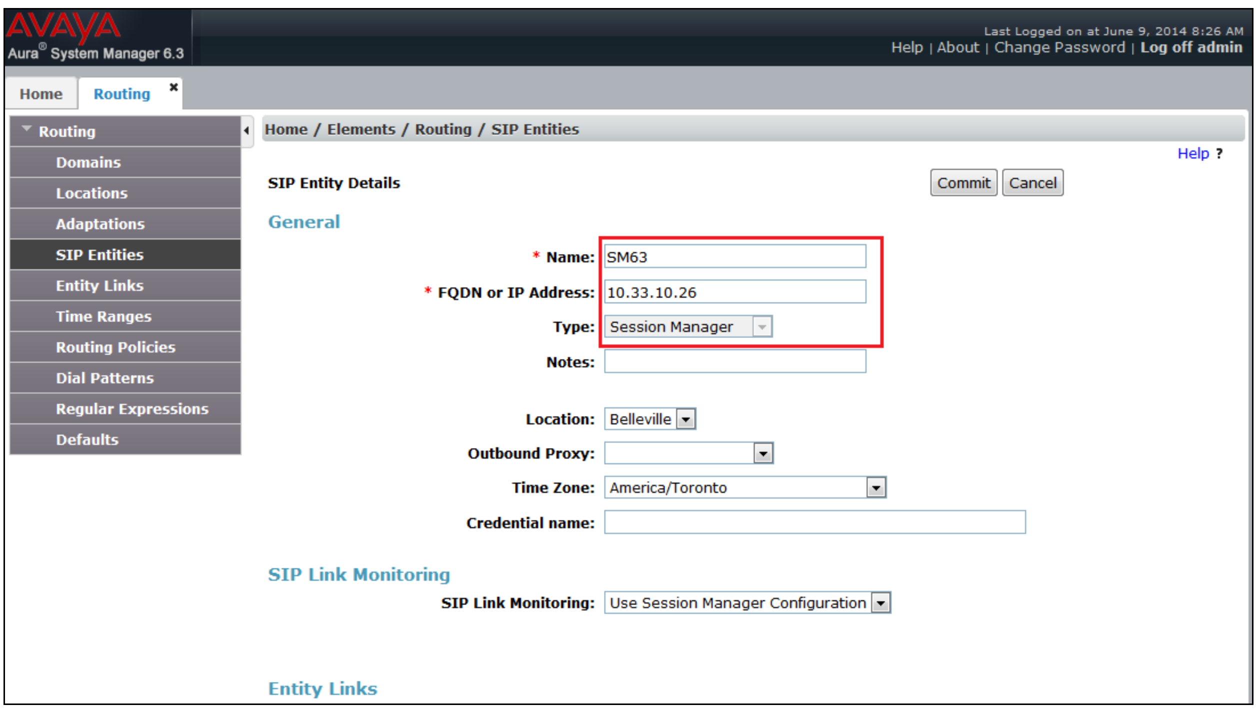
In the sample configuration, a SIP Entity is added for Session Manager, Communication Server 1000, and Nuance Speech Attendant.

6.3.1. Session Manager SIP Entity

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 on the right (not shown). 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 the location 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.
Note: The Ports section of this screen (not shown) is assumed to have been previously configured.



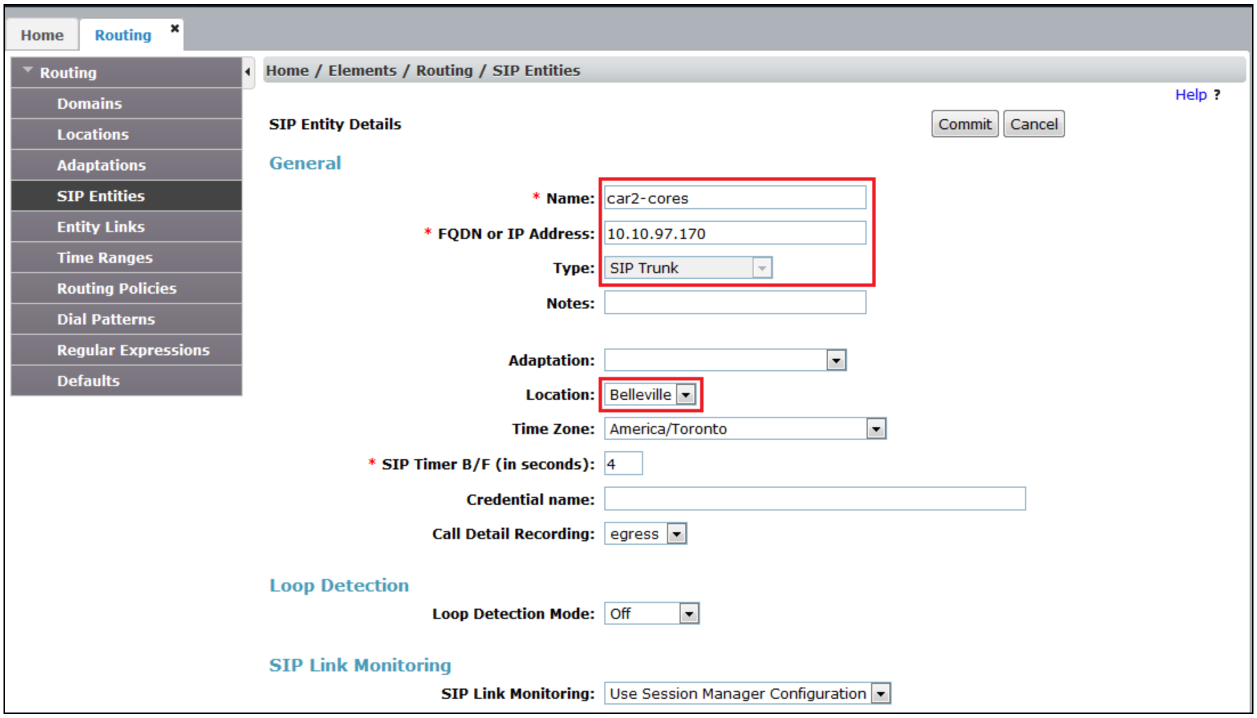
6.3.2. Communication Server 1000 SIP Entity

A SIP Entity must be added for the Communication Server 1000. To add a SIP Entity, **select SIP Entities** on the left and click on the **New** button on the right (not shown). The following screen is displayed. Fill in the following:

- Under **General**:

- **Name**: A descriptive name.
- **FQDN or IP Address**: IP address of the SIP gateway node IP.
- **Type**: Select **SIP Trunk**.
- **Location**: Select the location 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.

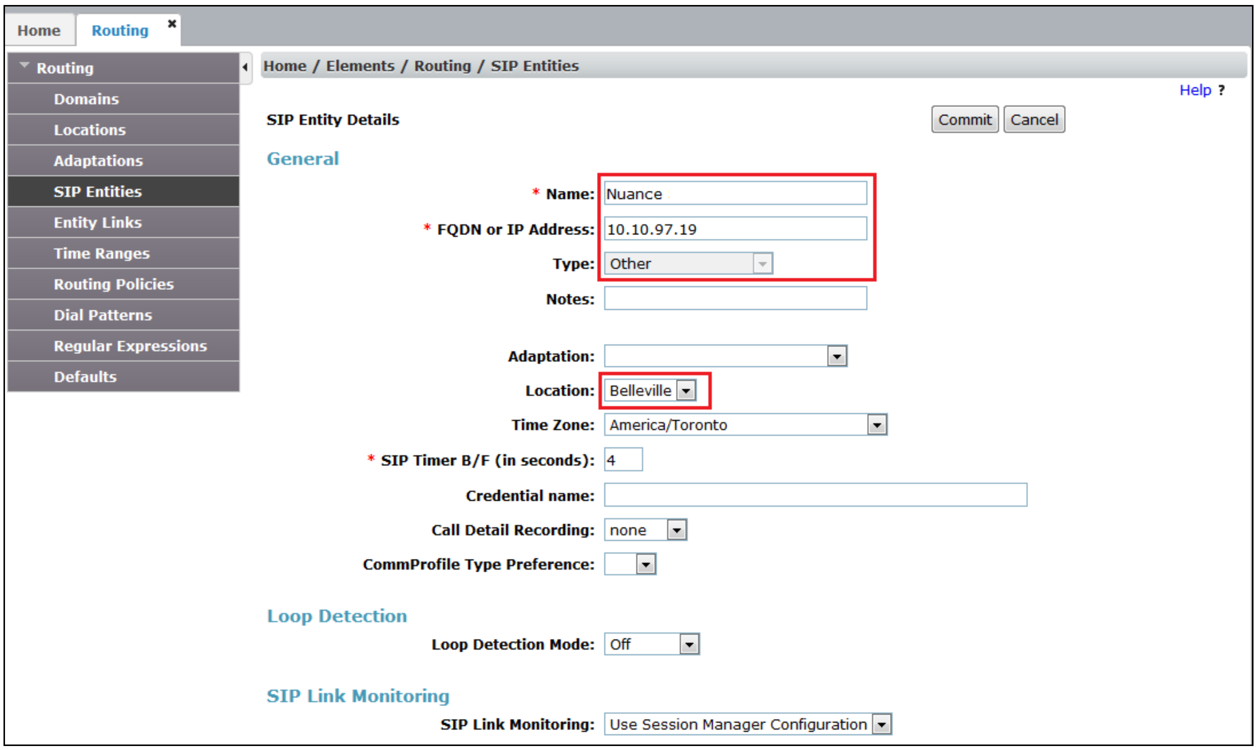


6.3.3. Nuance Speech Attendant SIP Entity

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 on the right (not shown). 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.



6.4. Add Entity Links

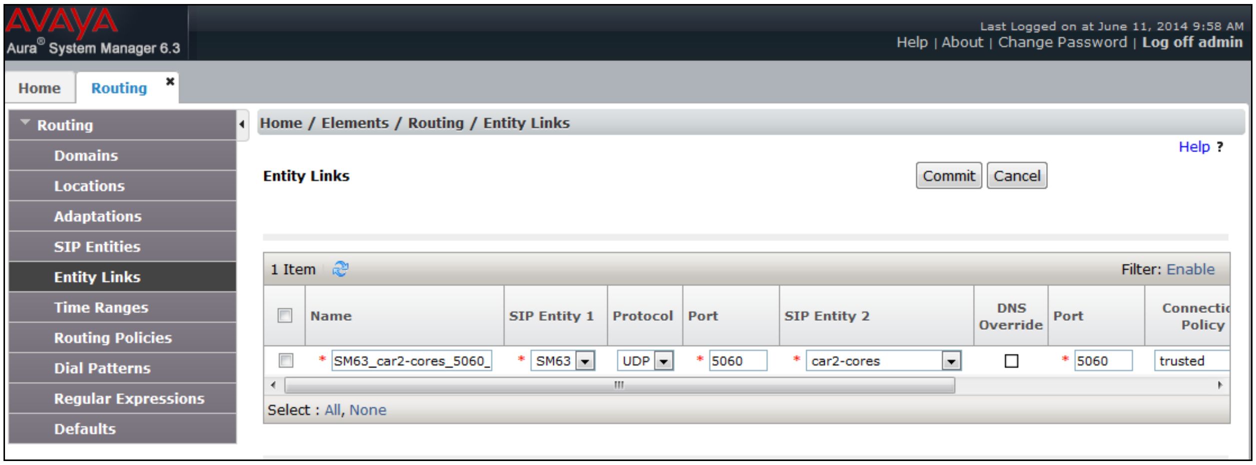
The SIP trunk from Session Manager to Communication Server 1000 and Nuance SA are described by Entity Links. To add an Entity Link, select **Entity Links** on the left and click on the **New** button on the right (not shown). Fill in the following fields in the new row that is displayed:

- **Name**: A descriptive name.
- **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 Server 1000 or

Nuance SA.

- **Port**: Port number on which the other system receives SIP requests.
- **Connection Policy**: Select **Trusted**.

The following screens display the two Entity Links. The first entity link is for Session Manager and Communication Server 1000. The second entity link is for Session Manager and Nuance SA.



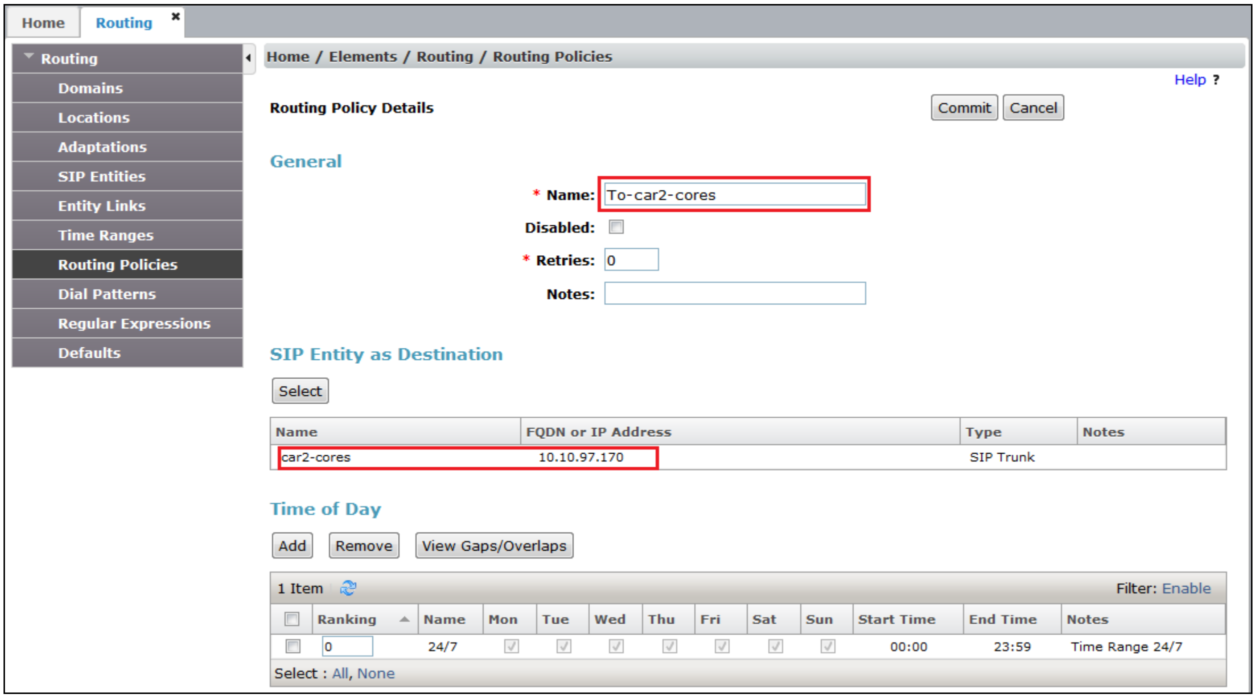
The entity link is below between Session Manager and Nuance Speech Attendant.

6.5. Add Routing Policy

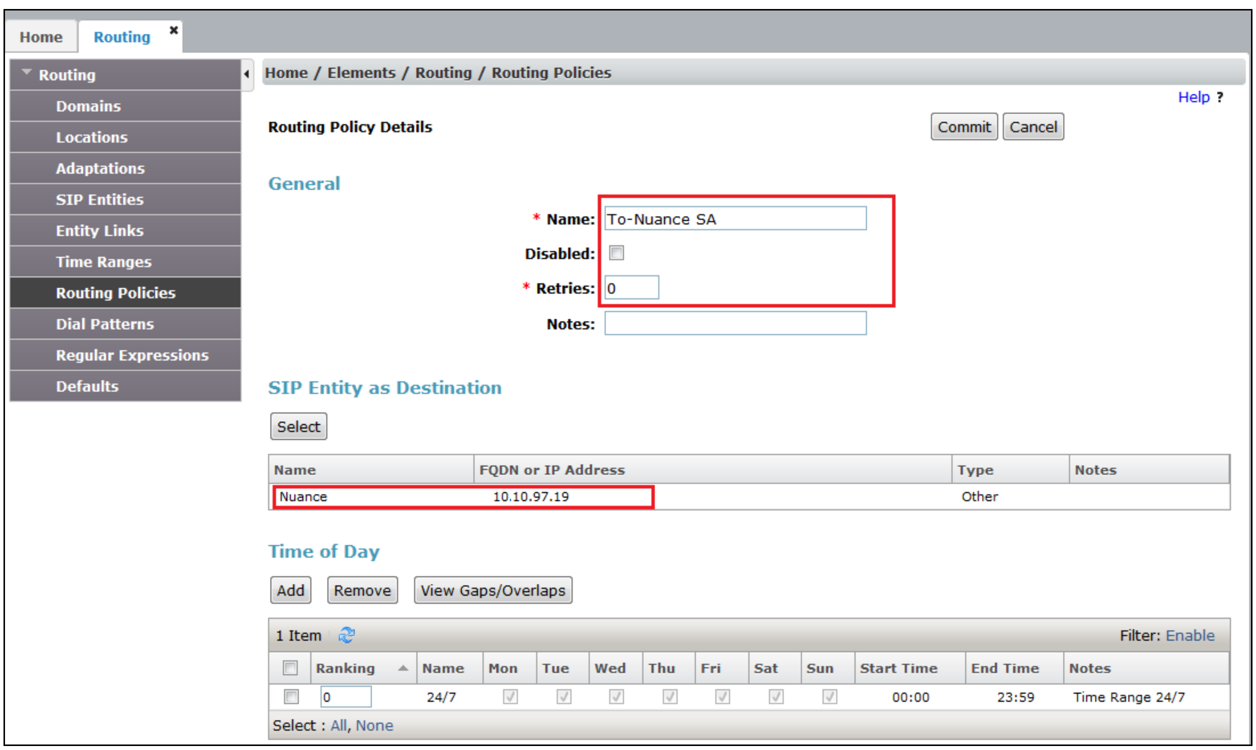
Routing policies describe the conditions under which calls will be routed to the SIP Entities. Two routing policies were added – one for Communication Server 1000 and one for Nuance SA. To add a routing policy, select Routing Policies on the left and click on the **New** button on the right (not shown). 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.

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 Server 1000.



The following screen shows the Routing Policy for Nuance SA. Note in the example below, this routing policy was given a Ranking of 0. The ranking will be discussed in the next section.



6.6. Add Dial Pattern

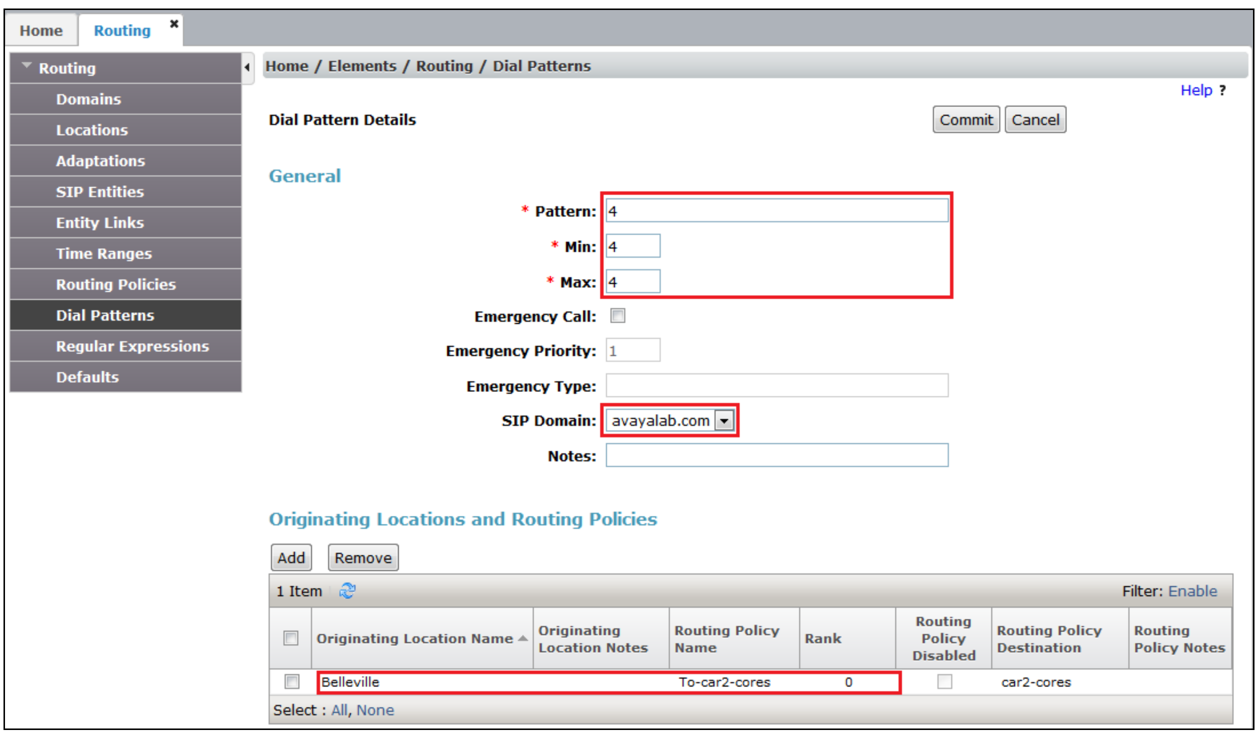
Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 4-digit extensions beginning with “4” reside on Communication Server 1000, extension “70000” is the Nuance SA number. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button on the right (not shown). Fill in the following:

Under General:

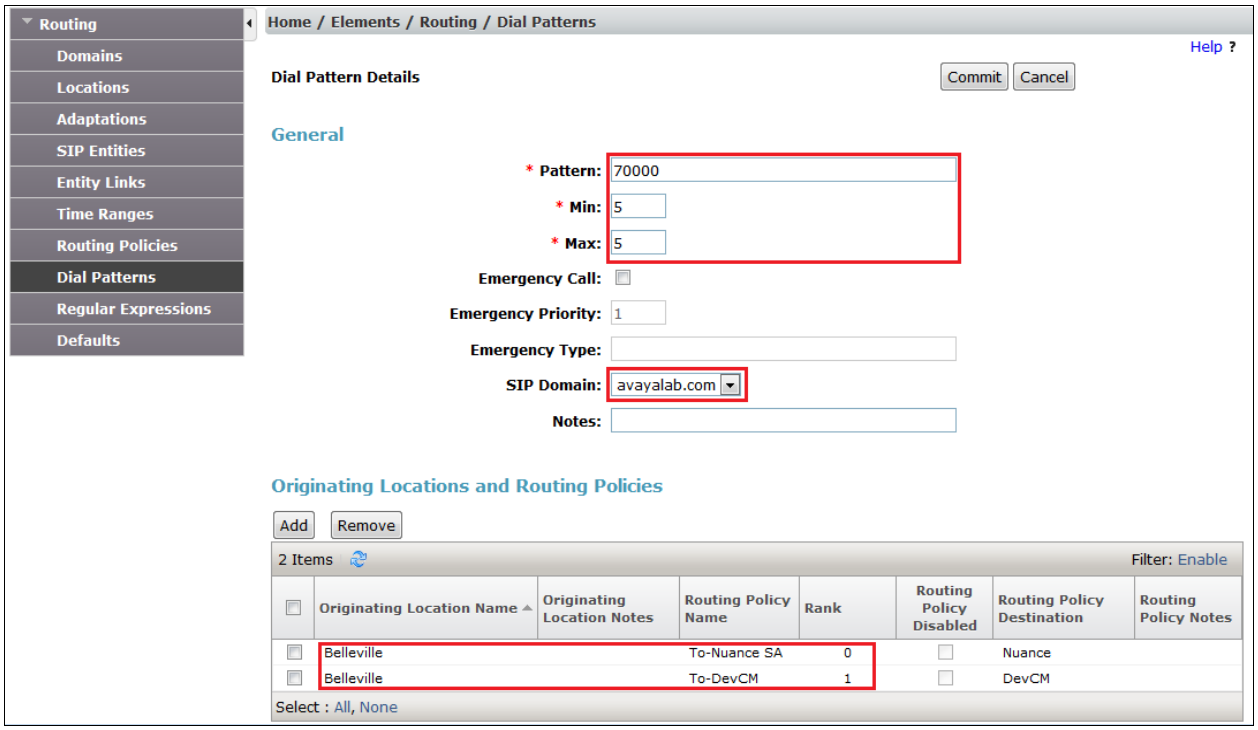
- **Pattern**: Dialed number or prefix.
- **Min**: Minimum length of dialed number.
- **Max**: Maximum length of dialed number.
- **SIP Domain**: Select avayalab.com SIP domain as defined in **Section 6.1**.
- Under Originating Locations and Routing Policies: Click Add, and then select the appropriate 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 Server 1000 . Click **Commit** to save this dial pattern.

The following screen shows the dial pattern definition for Communication Server 1000 extensions beginning with “4”.



The following screen shows the dial pattern definition for the Nuance SA number (70000). As mentioned in **Section 6.5**, the "To-Nuance SA" routing policy was assigned a Rank of 0. If desired, additional routing policies may be created with the desired ranks (the lower the rank number, the higher the priority). In the example below, under Originating Locations and Routing Policies, a second routing policy (To-DevCM) is assigned to this dial pattern. The To-DevCM routing policy was assigned a rank of 1. When a call to extension70000 is dialed, Session Manager will first use the routing policy with the lowest rank and route the call to Nuance SA. By default, Nuance SA is configured to send a 503 SIP message when all the SA ports are busy. As a result, when Session Manager receives a 503 SIP message response, Session Manager will then attempt to reroute the call using the next lowest ranked routing policy.



7. Configure Nuance Speech Attendant

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

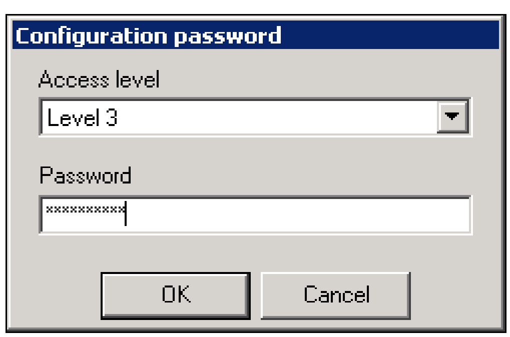
- Configure SIP interface in the Configuration Panel.
- Configure the number of SIP ports supported by Nuance SA.

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

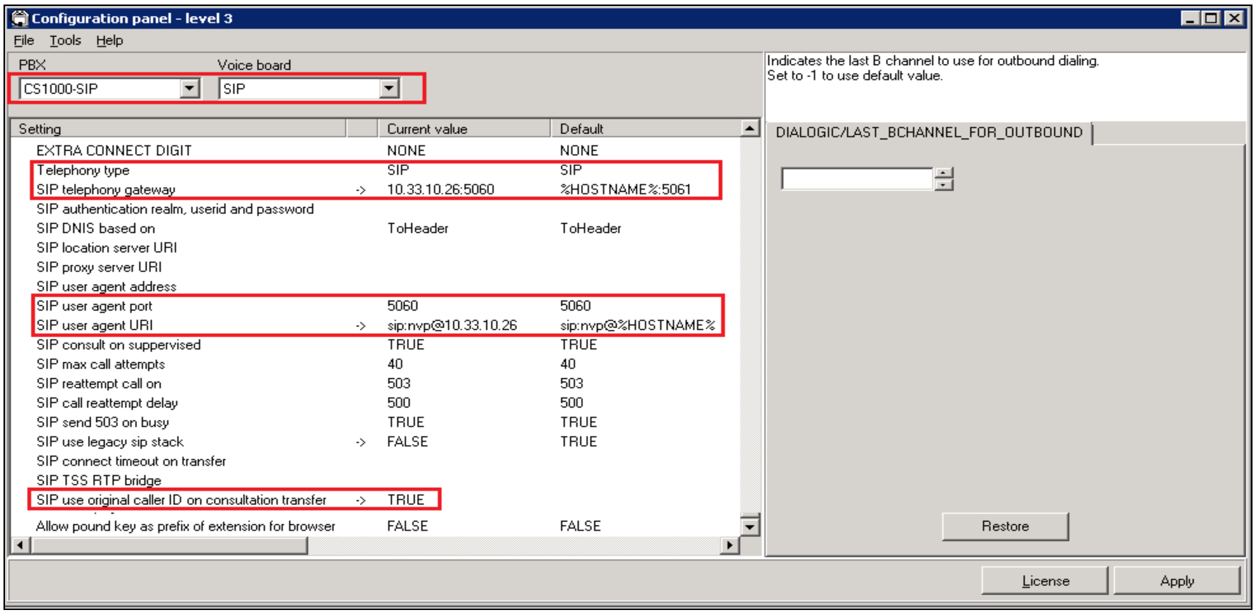
Nuance SA is configured through Admin Tools which can be started by navigating to Start →Programs→Speech Attendant→Admin Tools (not shown). The initial screen is displayed below.

7.1. Configure SIP Interface in Configuration Panel

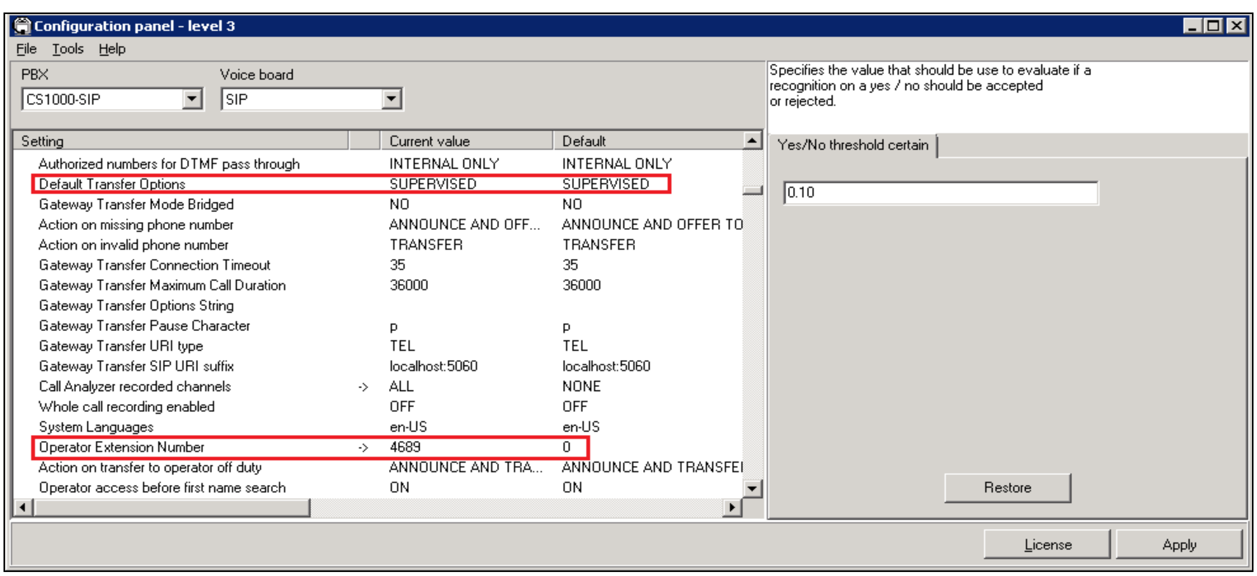
To open the Configuration Panel, click on this option in the Admin Tools window shown above. The login prompt will be displayed to the user as shown below. Log in with the appropriate credentials using Level 3 access level.



The Configuration Panel shown below is displayed. The Configuration Panel allows the SIP interface, transfer mode, and operator extension number to be configured. SA only supports supervised transfer with CS1000. To configure the SIP interface, set the PBX to **CS1000-SIP** 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. Configure the SIP port that SA listens on in the SIP user agent port field, specify the SIP user agent URI and set SIP use original caller ID on consultation transfer to TRUE for SA.

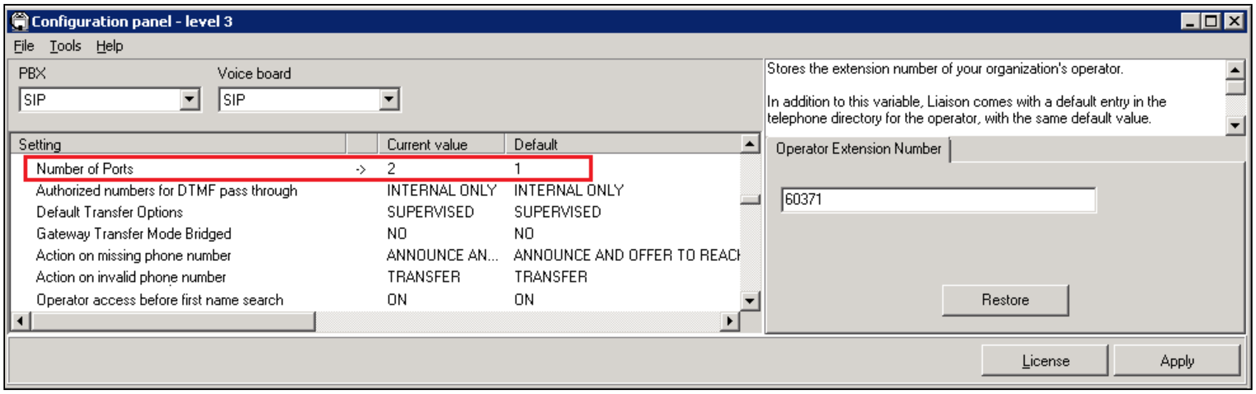


In the Configuration Panel, specify the transfer mode as supervised in the Default Transfer Options field, and specify the Operator Extension Number, which should be set to a valid extension on Communication Server 1000.

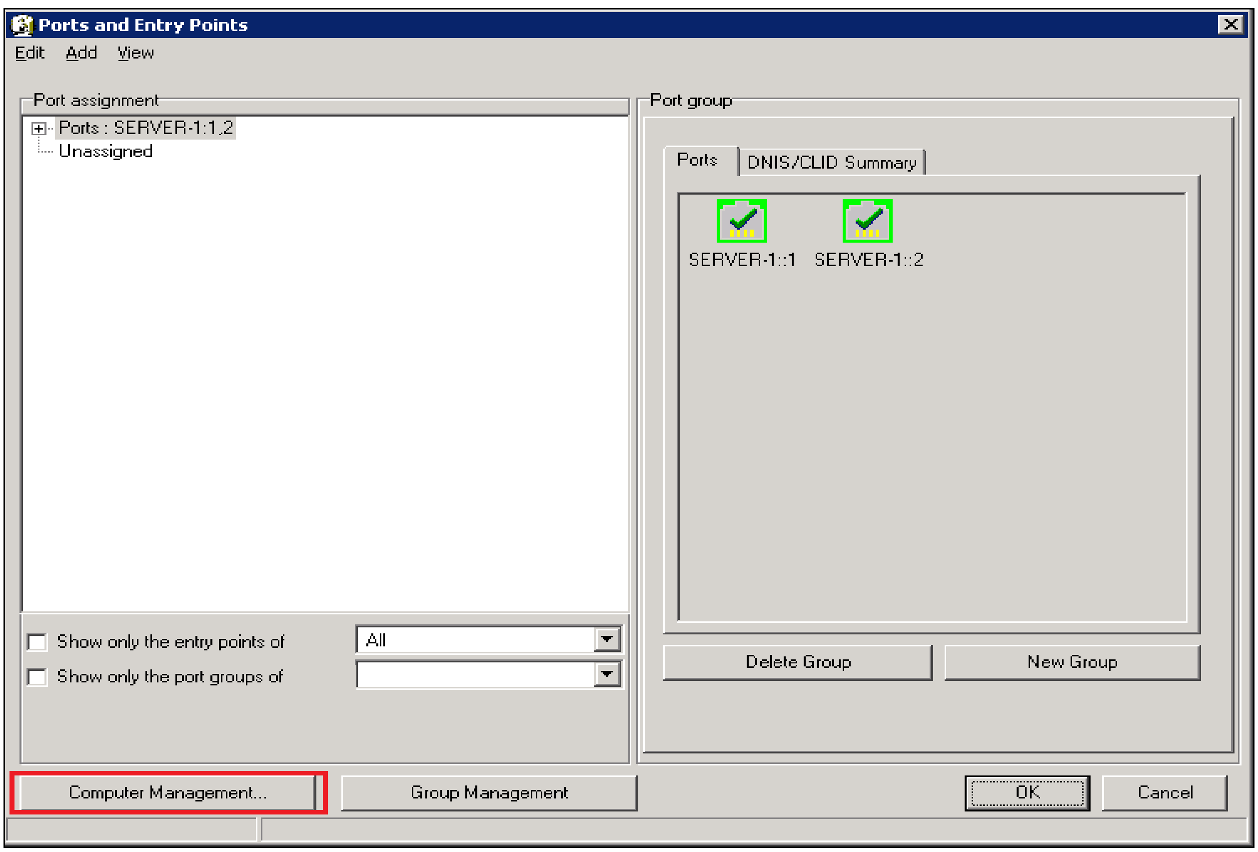


7.2. Configure Number of Ports

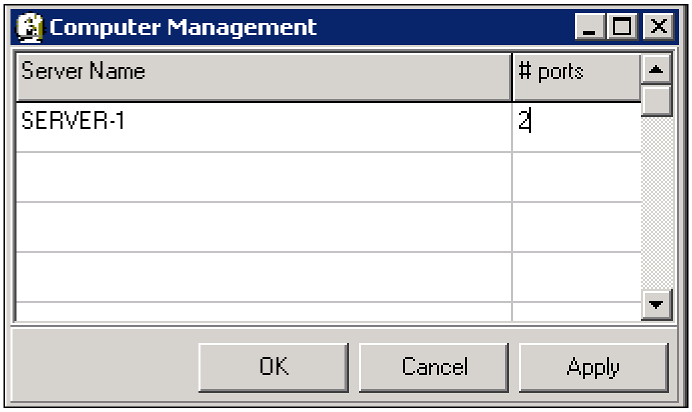
In the **Configuration Panel**, set the **Number of Ports** field to the desired value. In this example, 2 ports were configured. Click **Apply** and then close the Configuration Panel. Allow the SA application to be restarted when prompted.



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, 2 ports were configured. Click **OK**.

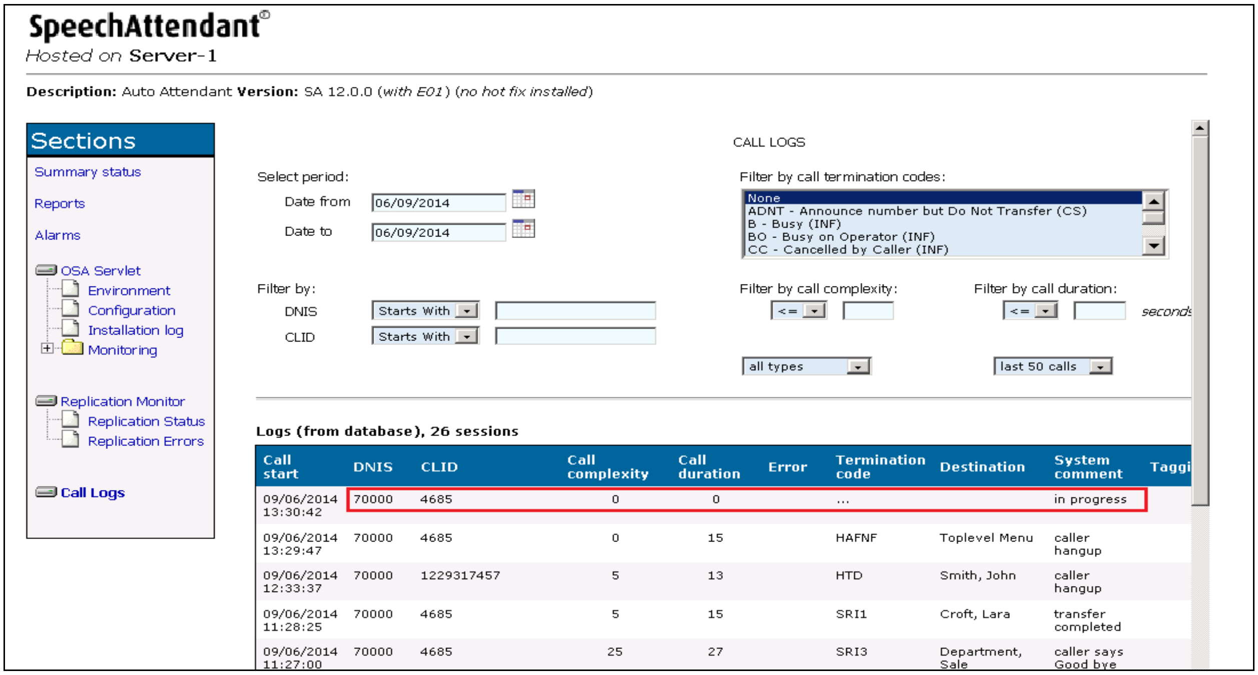


Next, close the **Configuration Panel**. Allow the system to restart when prompted.

8. 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 Communication Server 1000 using SIP integration.

- Place a call to Nuance SA. From the Nuance SA server, open the SA Monitor from Admin Tools (refer to **Section 7** for accessing 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**.



Call start	DNIS	CLID	Call complexity	Call duration	Error	Termination code	Destination	System comment	Taggi
09/06/2014 13:30:42	70000	4685	0	0		...		in progress	
09/06/2014 13:29:47	70000	4685	0	15		HAFNF	Toplevel Menu	caller hangup	
09/06/2014 12:33:37	70000	1229317457	5	13		HTD	Smith, John	caller hangup	
09/06/2014 11:28:25	70000	4685	5	15		SRI1	Croft, Lara	transfer completed	
09/06/2014 11:27:00	70000	4685	25	27		SRI3	Department, Sale	caller says Good bye	

- 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.

9. Conclusion

These Application Notes describe the configuration steps required to integrate Nuance Speech Attendant with Avaya Aura® Session Manager and Avaya Communication Server 1000 using SIP integration. All feature and serviceability test cases were completed successfully. Refer to **Section 2.2** for test results and any observations.

10. Additional References

This section references product documentation relevant to these Application Notes.

Documentation for Avaya products can be found at http://support.avaya.com.

[1] *Administering Avaya Aura® Session Manager*, Release 6.3, Document Number 03-300509, Issue 9, October 2013

[2] *IP Peer Networking Installation and Commissioning, Avaya Communication Server 1000,* Release 7.6, Document Number NN43001-313, Revision: 05.02, Jun 2013.

[3] *Communication Server 1000E Overview*, *Avaya Communication Server 1000*, Release 7.6, Document Number NN43041-110, Revision: 05.02, Jun 2013.

Nuance product documentation is 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.