Avaya Solution & Interoperability Test Lab

Application Notes for Configuration of the Communication Server 1000E Release 7.5 and Nuance Speech Attendant 11.1 - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate Nuance Speech Attendant 11 with Avaya Communication Server 1000E release 7.5 via SIP trunk configured on Avaya Aura® Session Manager release 6.1.

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 provide detail configurations steps required to integrate the Nuance Speech Attendant 11.1 (hereafter referred to as Nuance SA) with Avaya Communication Server 1000E release 7.5 (hereafter referred to as Avaya CS1000) using SIP trunk integration configured on Avaya Aura® Session Manager release 6.1. During the compliance testing, the 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.

2. General Test Approach and Test Results

The general test approach was to have different telephone types of Avaya CS1000 place a call to the Nuance SA and follow its voice instructions to verify core features of the Nuance SA such as: Answering incoming call, receive and handle calling/called party identification, perform speech recognition and transfer calls, handling disconnect event and recognize DTMF digits.

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

2.1 Interoperability Compliance Testing

Interoperability compliance testing covered features and serviceability of the Nuance SA. The feature testing included the following:

- SIP Trunk is established successfully between the Nuance SA Server and Avaya CS1000 via the Session Manager.
- Voice/Speech Recognition and Barge-in.
- Nuance SA can perform Supervised Call Transfer to different telephone types of the Avaya CS1000 (SIP, non-SIP and emulated PSTN telephones).
- Nuance SA can perform Unsupervised Call Transfers to different telephone types of the Avaya CS1000 (SIP, non-SIP and emulated PSTN telephones).
- DTMF Recognition.
- Call forward to voicemail with Message Waiting Indication (MWI) notification.

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

All test cases were tested and passed.

2.3 Support

Technical support for Nuance Speech Attendant can be obtained via email or through their website or phone.

- Web: http://www.nuance.com/support/index.htm
- Email: SpeechAttendant.Support@nuance.com
- Phone: (866) 434-2564 or (514) 390-3922

3. Reference Configuration

Figure 1 illustrates the configuration used to verify the Nuance Speech Attendant (SA) solution with Avaya Aura® Session Manager and Avaya Communication Server 1000E. Nuance SA is deployed on a dedicated server running Windows 2008 R2 Server. Session Manager interfaces to Avaya Communication Sever 1000E 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 Avaya CS1000 to the SA server through the Session Manager. Multiple SIP ports were configured on the Nuance SA server.

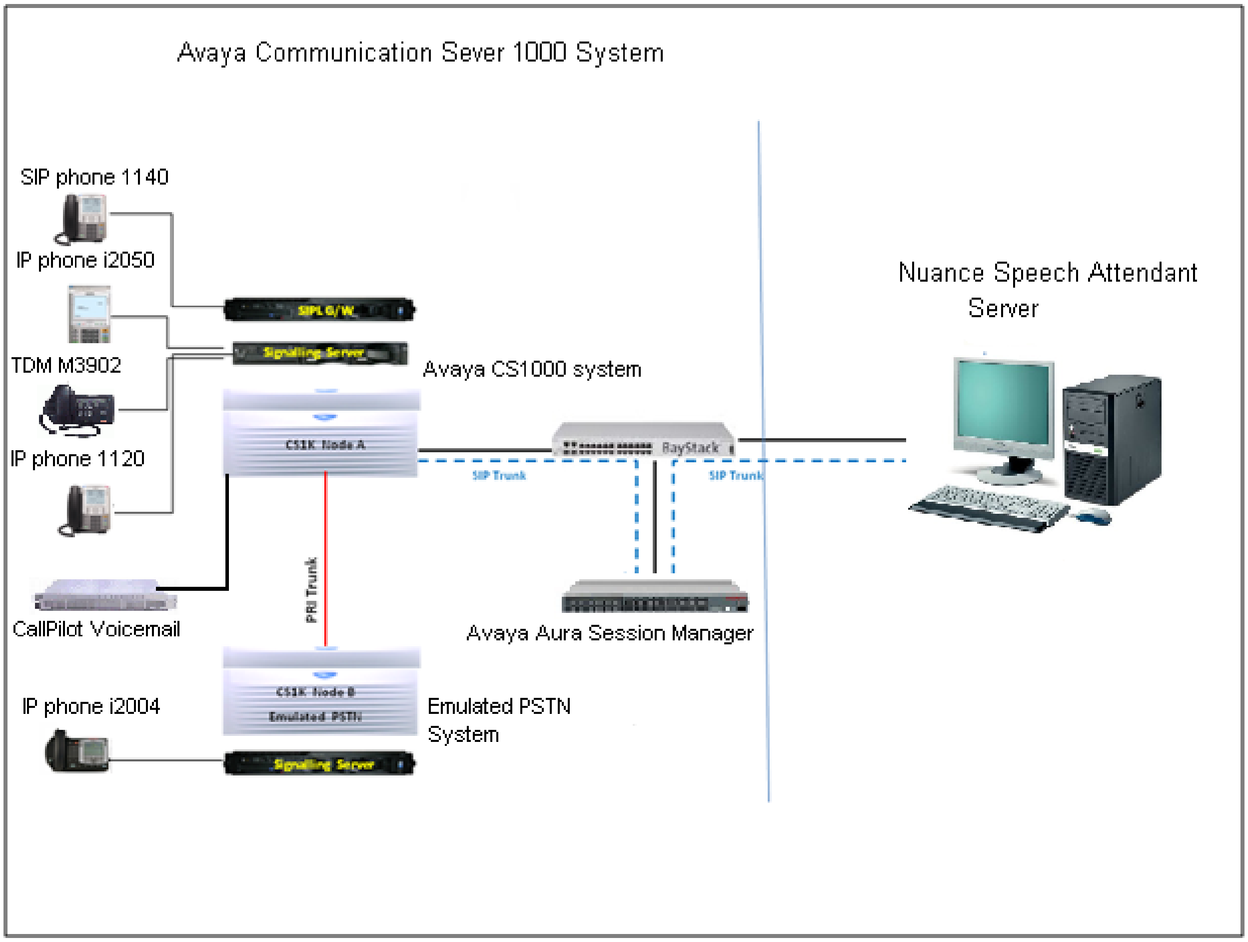


Figure 1: Avaya Communication Server 1000E Network with the Nuance Speech Attendant 11 connecting to Avaya Session Manager via SIP trunk.

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya S8800 server running Avaya Aura® Session Manager	Aura® Session Manager 6.1 (6.1.5.0.615006)
Avaya Communication Server 1000E	Call Server: 7.50Q GA Signaling Server: 7.50.17 GA SIP Line Gateway: 7.50.17.00
Avaya CallPilot® 600i	Version 05.00.41.143
Avaya IP Phone 1165E	Version 0625C8L
Avaya IP Phone 1110	Version 0623C8L
Avaya CS1000 SIP Phones 1140	Version 4.0.3
Avaya TDM phone	3902
Nuance Speech Attendant Server	Window 2008 R2 Operating System

5. Configure Avaya System

This Application Note assumes that Avaya Aura® Session Manager and the Avaya Communication Server 1000E are installed, configured and operational. For detailed information on how to configure and administer the Avaya Systems, please refer to the **Section 9 [1]**.

The following section will describe how to configure the SIP trunk on Avaya Session Manager for Nuance SA and Avaya CS1000.

5.1 Configure Avaya Aura® Session Manager

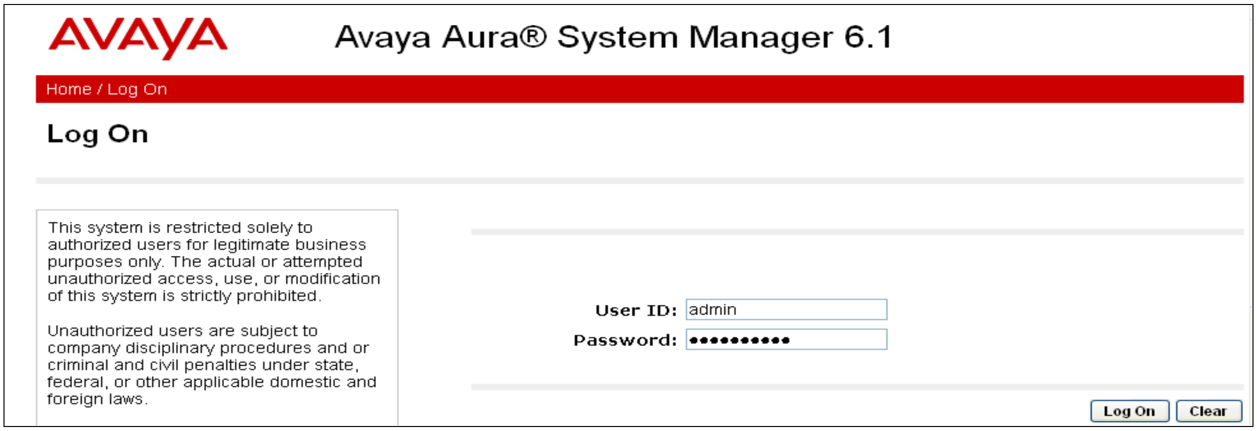
This section provides the procedures for configuring Session Manager. Session Manager is comprised of two functional components: The Session Manager server and the System Manager server. All SIP call provisioning for Session Manager is performed through the System Manager Web interface and is then downloaded into Session Manager.

This section assumes that Session Manager and System Manager have been installed, and network connectivity exists between the two platforms. The following steps describe for configuring Session Manager.

- SIP Domains
- Locations
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policy
- Dial Patterns

5.1.1 Configure SIP Domain

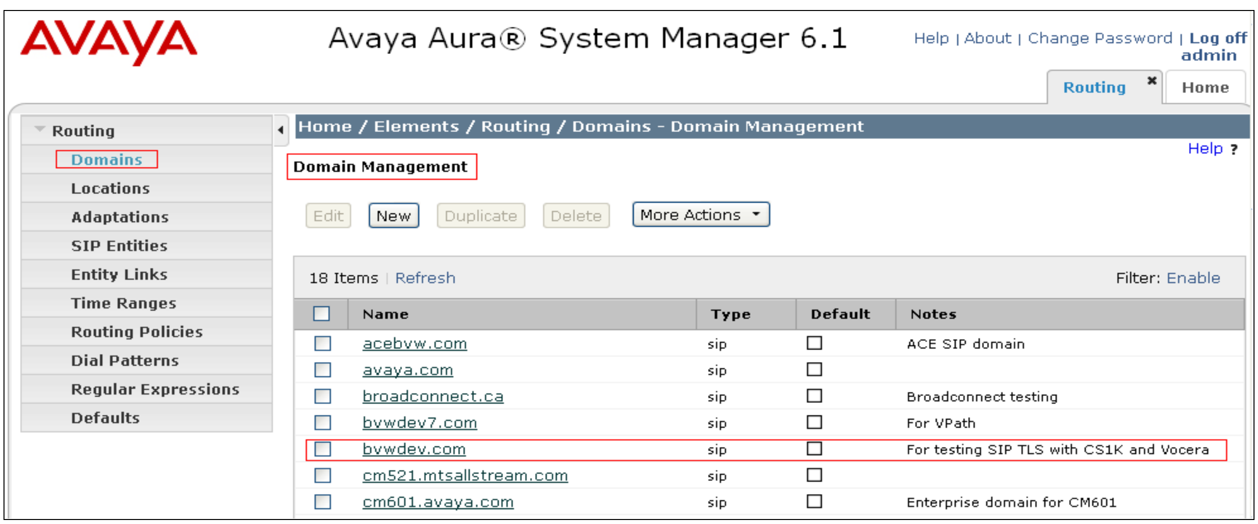
Launch a web browser, enter **https://<IP address of System Manager>** in the URL, and log in with the appropriate credentials.



Navigate to **Elements→Routing→Domains** and click on the **New** button to create a new SIP Domain (screen not shown). Enter the following values and use defaults for the remaining fields:

- **Name** –Enter the Authoritative Domain name specified in CS1000 SIP Gateway in **Section 5.2**, which is **bvwdev.com**.
- **Type** – Select **SIP**

Click **Commit** to save. The following screen shows the Domains page used during the compliance test.



5.1.2 Configure Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside. This is used for bandwidth management or location-based routing.

Navigate to **Routing→Locations**, and click on the **New** button to create a new SIP Entity location (screen not shown).

General section

Enter the following values and use default values for the remaining fields.

- Enter a descriptive Location in the **Name** field (e.g. **Belleville**).
- Enter a description in the **Notes** field if desired.

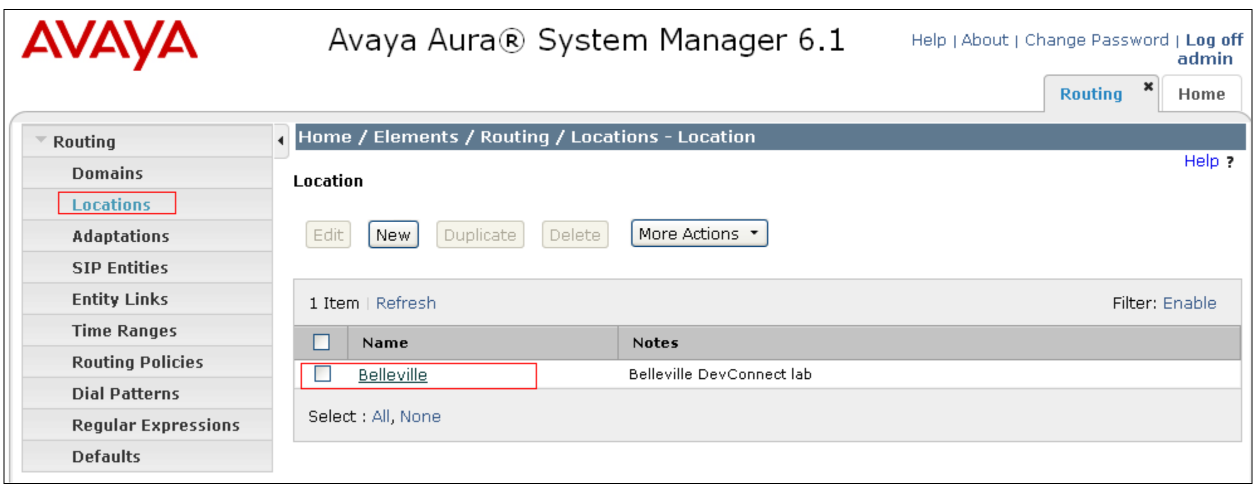
Location Pattern section

Click **Add** and enter the following values:

- The IP address information for the **IP address Pattern** (e.g. **10.10.97.***).
- A description in the **Notes** field if desired.

Repeat these steps in the Location Pattern section if the Location has multiple IP segments. Modify the remaining values on the form, if necessary; otherwise, use all the default values. Click on the **Commit** button.

Repeat all the steps for each new Location. The following screen shows the **Location** page used during the compliance test.



5.1.3 Configure SIP Entities

A SIP Entity must be added for Session Manager and for each network component that has a SIP trunk. During the compliance test the following SIP Entities were configured:

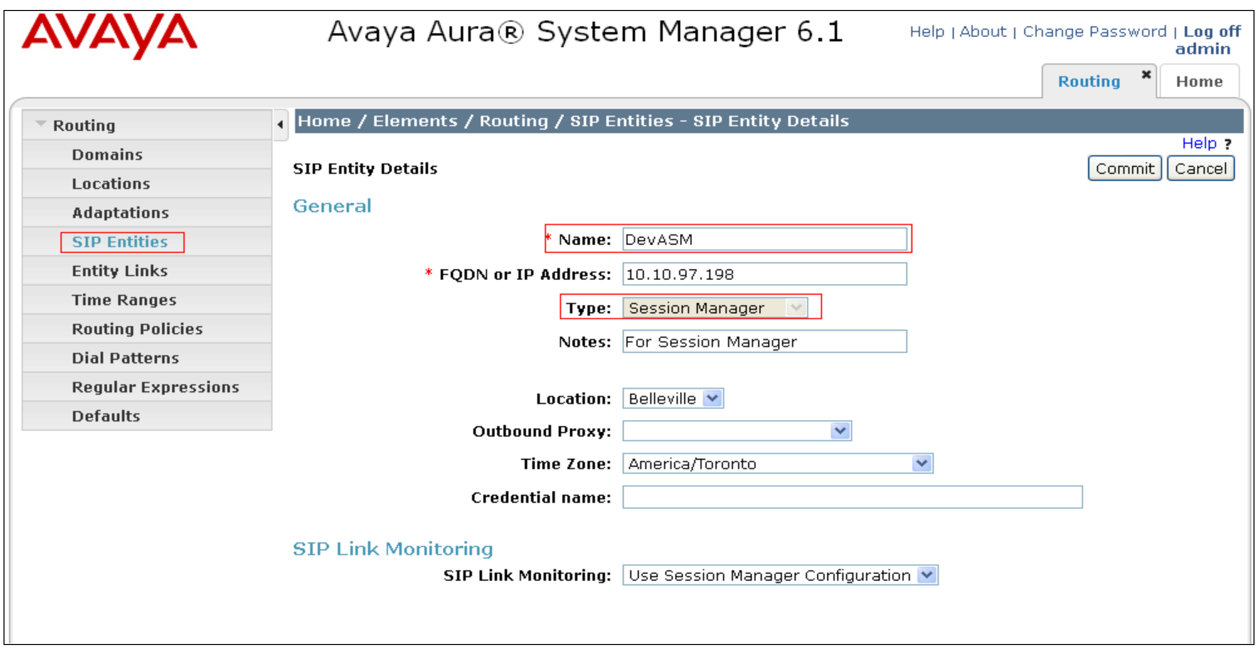
- Session Manager
- Communication Server 1000 SIP Gateway
- Nuance Server

Navigate to **Routing →SIP Entities** and click on the **New** button to create a new SIP entity (screen not shown). Provide the following information:

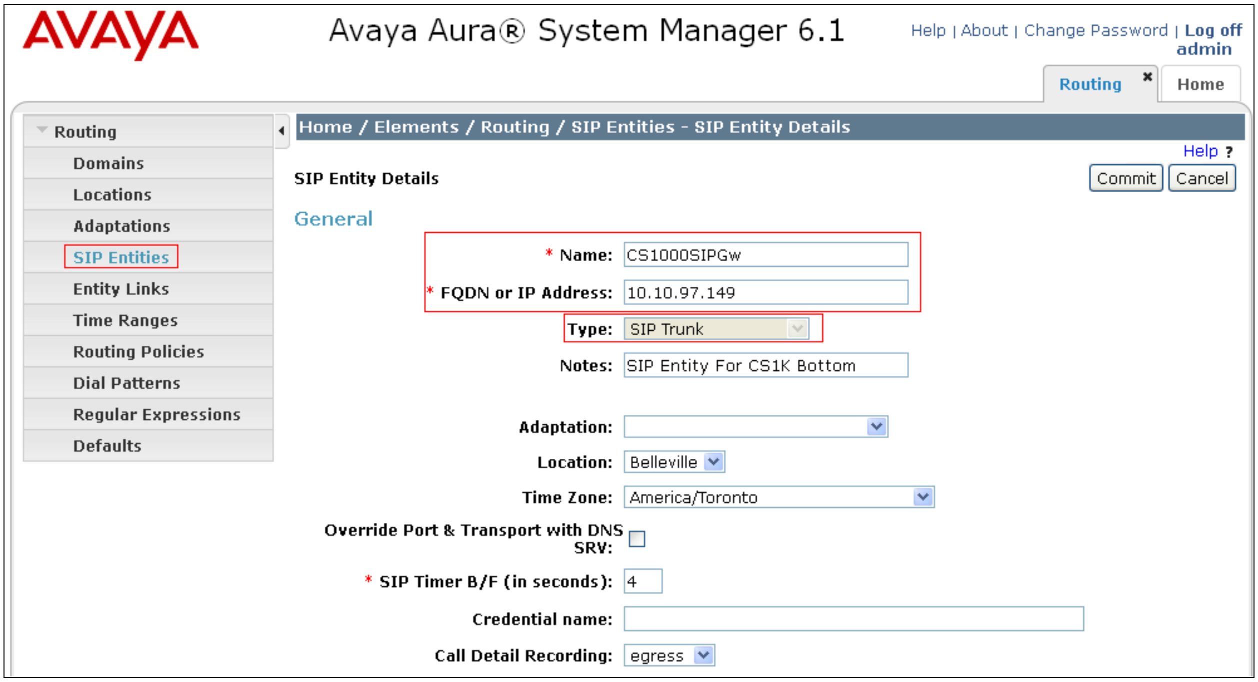
Enter the following and use default values for the remaining fields:

- **Name**: Enter a descriptive name.
- **FQDN or IP Address:** Enter the IP address of the signaling interface on each:
  - CS1000 SIP Gateway: 10.10.97.149
  - Signaling Session Manager: 10.10.97.198
  - Nuance server: 10.10.98.89
- From the **Type** drop down menu, select a type that best matches the SIP Entity:
  - For CS1000 SIP Gateway: select **SIP Trunk**
  - For Session Manager, select **Session Manager**
  - For Nuance Server, select **Other**
- Enter a description in the **Notes** field if desired.
- Select the appropriate time zone.
- Accept the other default values.

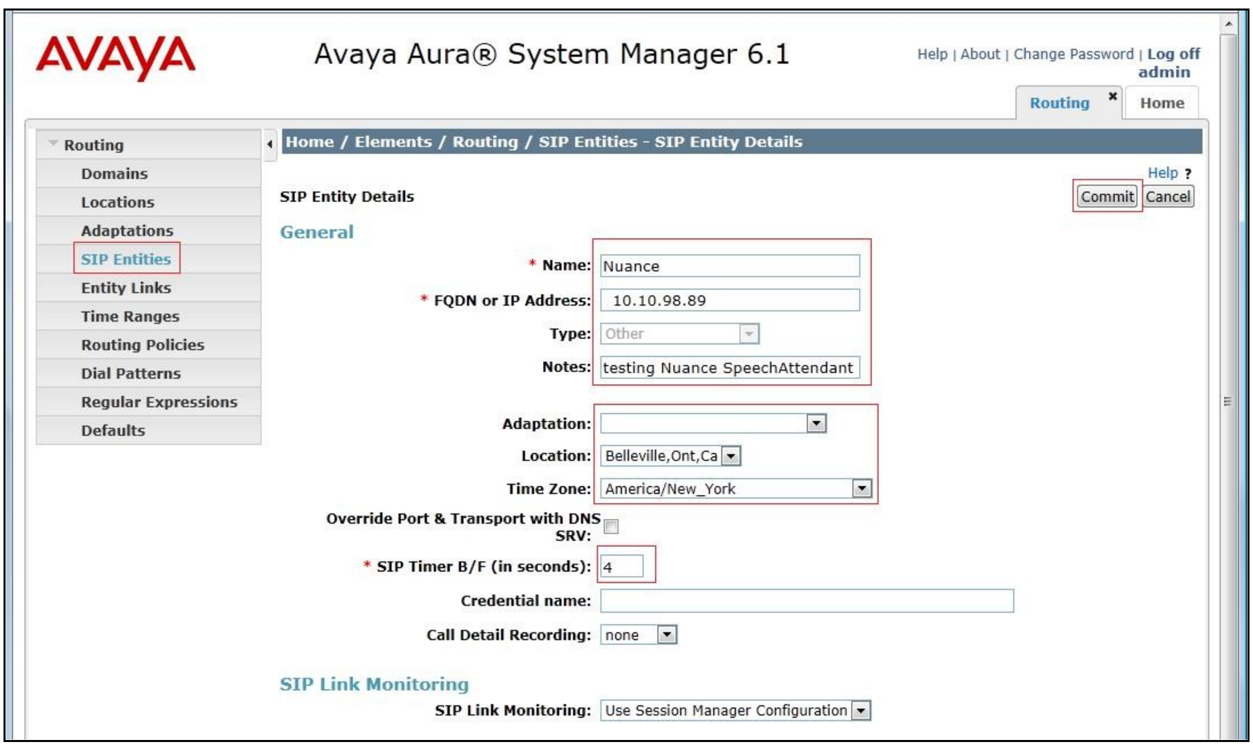
Click on the **Commit** button to save each SIP entity. Repeat all the steps for each new entity. The screen below shows the detail of **Session Manger SIP Entity**.



The screen below shows the details of CS1000 SIP Entity.



The screen below shows the detail of **Nuance SIP Entity**.



5.1.4 Configure Entity Links

Entity Links define the connections between the SIP Entities (in this case, CS1000 SIP gateway and Nuance server) and Session Manager. In the compliance test, the following entity links are defined from Session Manager.

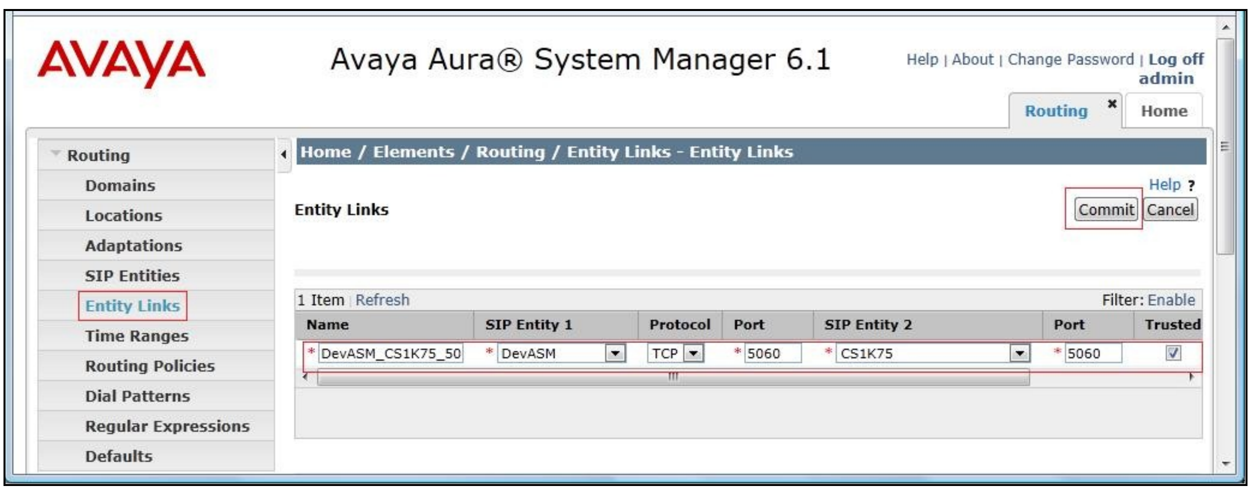
- Session Manager ⇔ Communication Server 1000 SIP Gateway
- Session Manager ⇔ Nuance Server

Navigate to **Routing →Entity Links** and click on the **New** button to create a new entity link (screen not shown). Provide the following information:

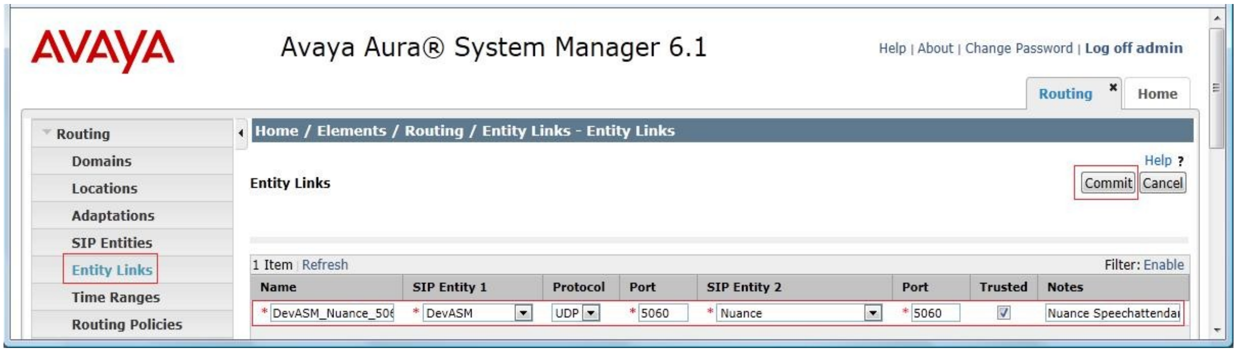
- **Name**: Enter a descriptive name.
- In the **SIP Entity 1** drop down menu, select the Session Manager SIP Entity created in **Section** Error! Reference source not found. (e.g. **DevASM**).
- In the **Protocol** drop down menu, select the UDP protocol.
- In the **Port** field, enter the port to be used (e.g. **5060**).
- In the **SIP Entity 2** drop down menu, select **CS1000SIPGw** if entity link between Session Manager and CS1000 SIP gateway is created, and select **Nuance** for the Nuance entity.
- In the **Port** field, enter the port to be used (e.g. **5060**).
- Check the **Trusted** box.
- Enter a description in the **Notes** field if desired.

Click on the **Commit** button to save each Entity Link definition. Repeat all the steps for each new SIP Entity Link.

The following screen shows Entity Links page during creating new entity link between Session Manager and CS1000 SIP Gateway.



The following screen shows Entity Link page in creating new entity link between Session Manager and Nuance server.



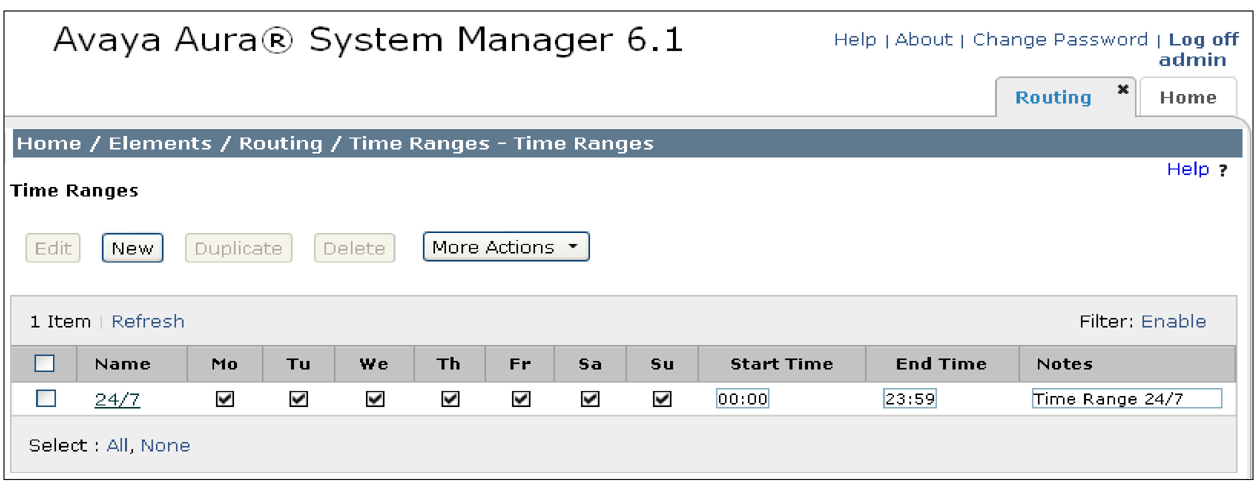
5.1.5 Time Ranges

Time Ranges define admission control criteria to be specified for Routing Policies (**Section** Error! Reference source not found.). In the reference configuration, no restrictions were used.

To add a Time Range, navigate to **Routing→Time Ranges**, and click on the **New** button (screen not shown). Provide the following information:

- Enter a descriptive name in the **Name** field (e.g. **24/7**).
- Check each day of the week.
- In the **Start Time** field, enter **00:00**.
- In the **End Time** field, enter **23:59**.
- Enter a description in the **Notes** field if desired.

Click the **Commit** button. The following screen shows the Time Range page used during the compliance test.



5.1.6 Configure Routing Policy

Routing Policies associate destination SIP Entities (**Section** Error! Reference source not found.) with Time of Day admission control parameters (**Section** Error! Reference source not found.) and Dial Patterns (**Section** Error! Reference source not found.). In the reference configuration, Routing Policies are defined for:

- Inbound calls to CS1000 SIP gateway.
- Inbound calls to Nuance server.

To add a Routing Policy, navigate to **Routing →Routing Policies** and click on the **New** button on the right (screen not shown). Provide the following information:

General section

- Enter a descriptive name in the **Name** field (e.g. **To_CS1K75_Bottom**, **To_Nuance**).
- Enter a description in the **Notes** field if desired.

SIP Entity as Destination section

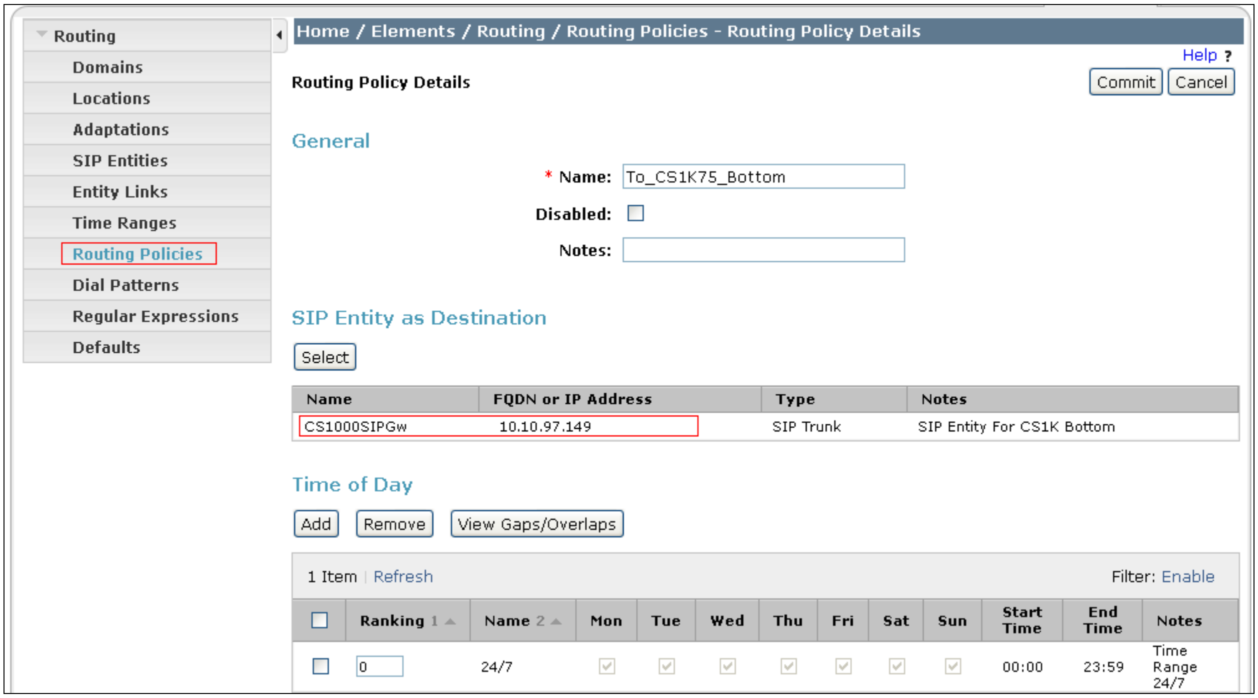
- Click the **Select** button.
- Select a SIP Entity that will be the destination for this call.
- Click the **Select** button and return to the Routing Policy Details form.

Time of Day section

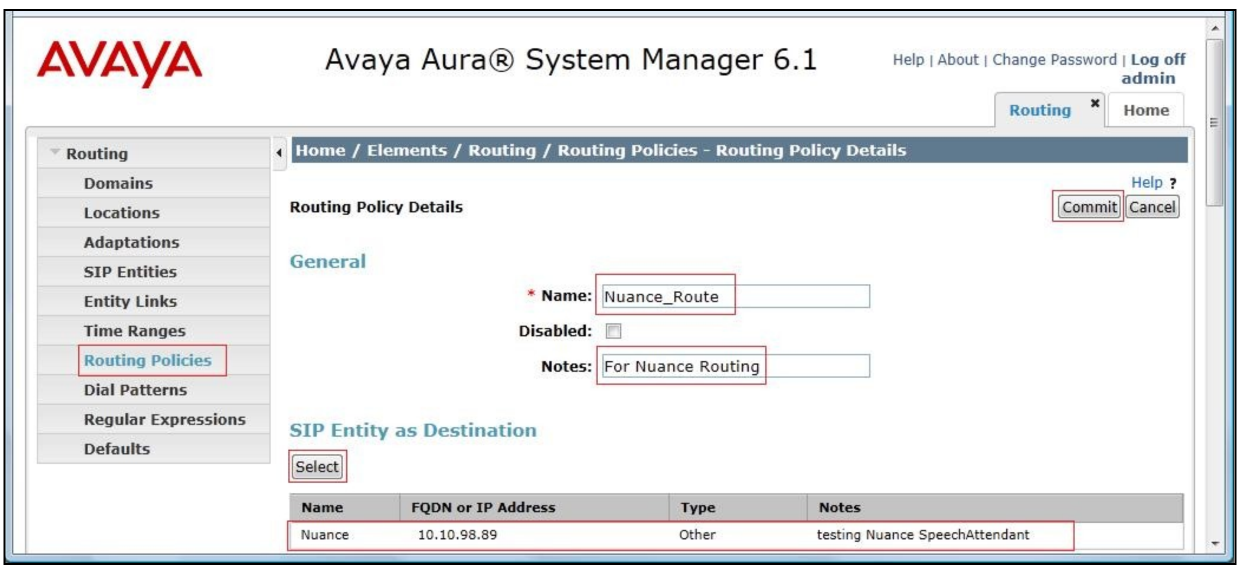
- Leave default values.

Click **Commit** to save Routing Policy definition. Repeat the steps for each new Routing Policy.

The following screen shows the Routing Policy used for CS1000 during the compliance test.



The following screen shows the Routing Policy used for Nuance during the compliance test



5.1.7 Configure Dial Patterns

Dial Patterns define digit strings to be matched for inbound and outbound calls. In the compliance test, the following dial patterns are defined from Session Manager.

- 70xxx – dial pattern used to route calls to CS1000.

To add a Dial Pattern, select **Routing →Dial Patterns** and click on the **New** button (screen not shown) on the right pane. Provide the following information:

General section

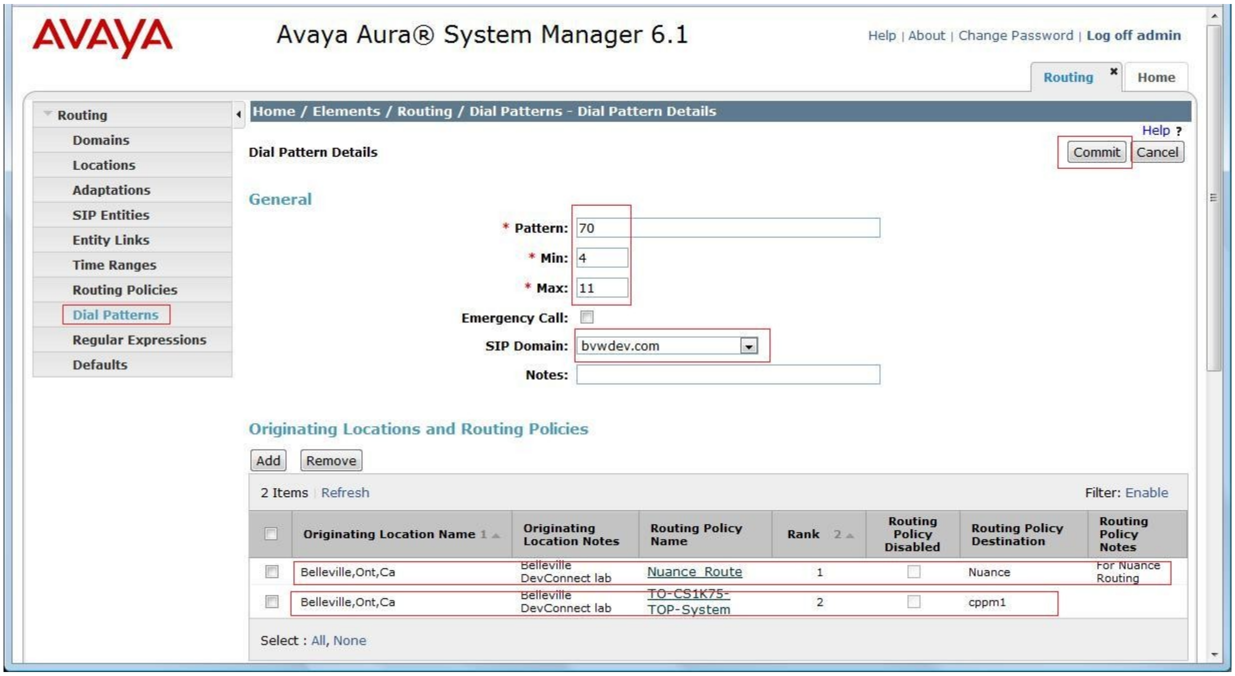
- Enter a unique pattern in the **Pattern** field (e.g. **70**).
- In the **Min** field enter the minimum number of digits (e.g. **4**).
- In the **Max** field enter the maximum number of digits (e.g. **11**).
- In the **SIP Domain** drop down menu select the domain **bvwdev.com** defined in **Section 5.1.1**.

Originating Locations and Routing Policies section

- Click on the **Add** button and a window will open (screen not shown).
- Click on the box for the appropriate Originating Locations, and Routing Policies (see **Section 5.1.6**) that pertain to this Dial Pattern.
    - Select the Originating Location to apply the selected routing policies to **All**.
    - Select appropriate Routing Policies.
    - Click on the **Select** button and return to the **Dial Pattern** page.

Click the **Commit** button to save the new definition. Repeat steps for the remaining Dial Patterns. The following screen shows the dial pattern **70xxx** used to route calls to CS1000 system during the compliance test.

Note: There are 2 routes in the **Originating Locations and Routing Policies**; **Nuance** and **To-CS1K75-TOP-System** routes. **Nuance** route is set to **Rank 1** for which the Session Manager will always route the call to Nuance Speech Attendant first. The **To-CS1K75-TOP-System** route (**Rank 2**) is for the condition when all SIP ports on Nuance SA are busy, the Session Manager will re-route the call to this alternate destination, **To-CS1K75-TOP_System.**


AVAYA
Avaya Aura® System Manager 6.1
Help | About | Change Password | Log off admin
Routing
Home
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Home / Elements / Routing / Dial Patterns - Dial Pattern Details
Help ?
Dial Pattern Details
Commit
Cancel
General
* Pattern: 70
* Min: 4
* Max: 11
Emergency Call:
SIP Domain: bvwdev.com
Notes:
Originating Locations and Routing Policies
Add
Remove
2 Items | Refresh
Filter: Enable
Originating Location Name
Originating Location Notes
Routing Policy Name
Rank
Routing Policy Disabled
Routing Policy Destination
Routing Policy Notes
Belleville,Ont,Ca
Belleville DevConnect lab
Nuance_Route
1
Nuance
For Nuance Routing
Belleville,Ont,Ca
Belleville DevConnect lab
TO-CS1K75-TOP-System
2
cppm1
Select : All, None

5.2 Configure Avaya Communication Server 1000E

The assumption is that the 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 Avaya Session Manager via SIP Signaling Gateway using the Element Manager.

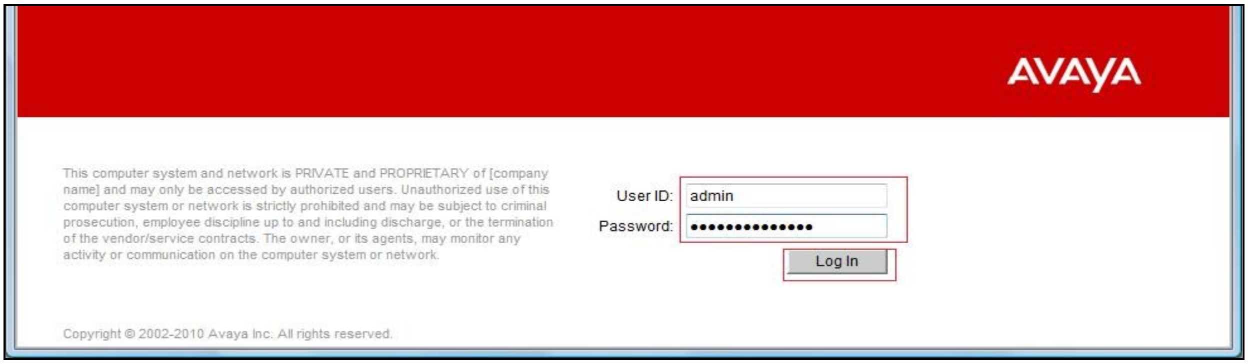
Prerequisites:

- An Avaya CS1000 server which has been:
  - Installed with CS 1000 Release 7.5 Linux Base.
  - Joined CS 1000 Release 7.5 Security Domain.
  - Deployed with SIP Trunk Application.
  - For more information on CS 1000 installation, maintenance, and upgrades, see **Section 9**.

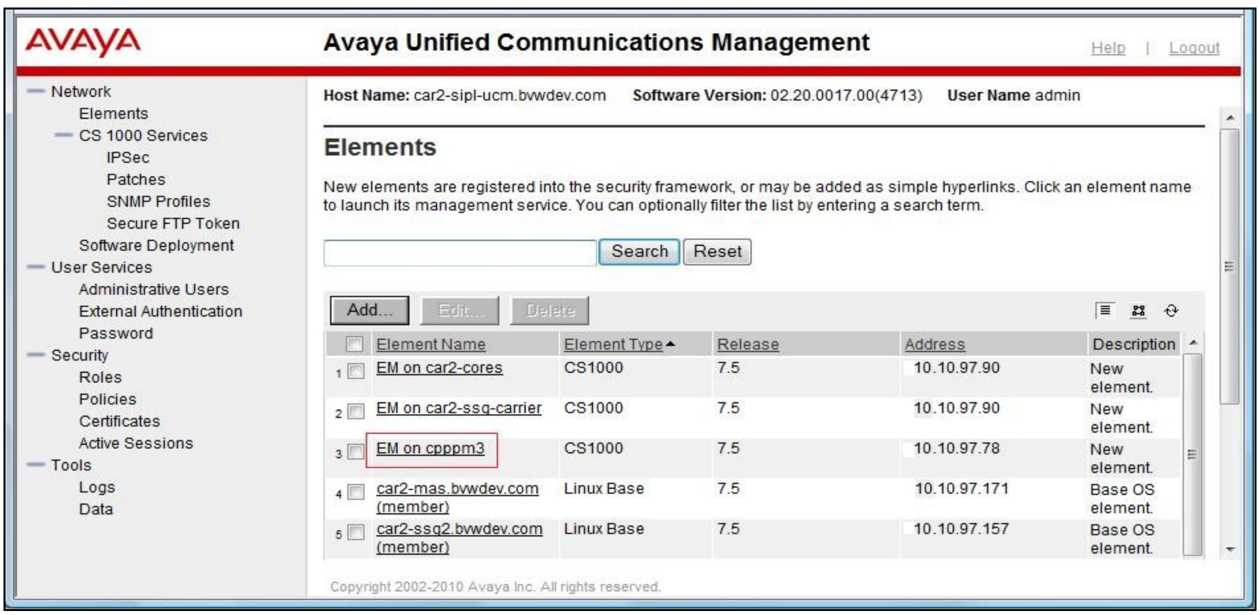
- The following software packages are enabled in the key-code.
- If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at http://www.avaya.com.

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	Nortel SIP Line package	Existing package	--
SIPL_3RDPARTY	416	Third-Party SIP Line Package	Existing package	--

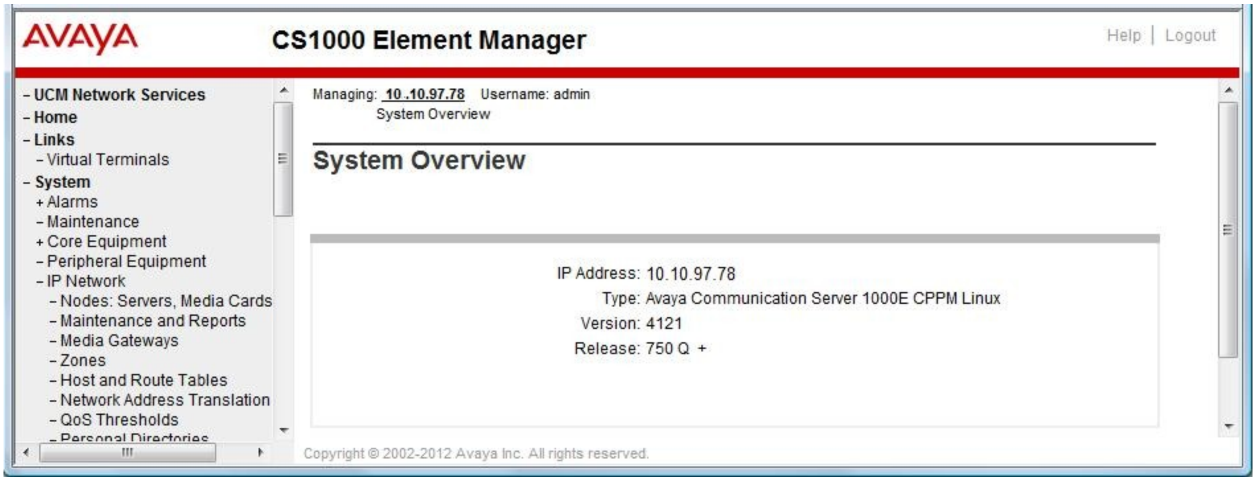
Log on to the Unified Communication Manager (hereafter referred to as UCM) Common Services of the Avaya CS1000, using Microsoft Internet Explorer 6.0260 or later to access the UCM by addressing the IP address or FQDN (Full Qualified Domain Name) of the UCM and then input the username/password which was defined during the primary security server setup.



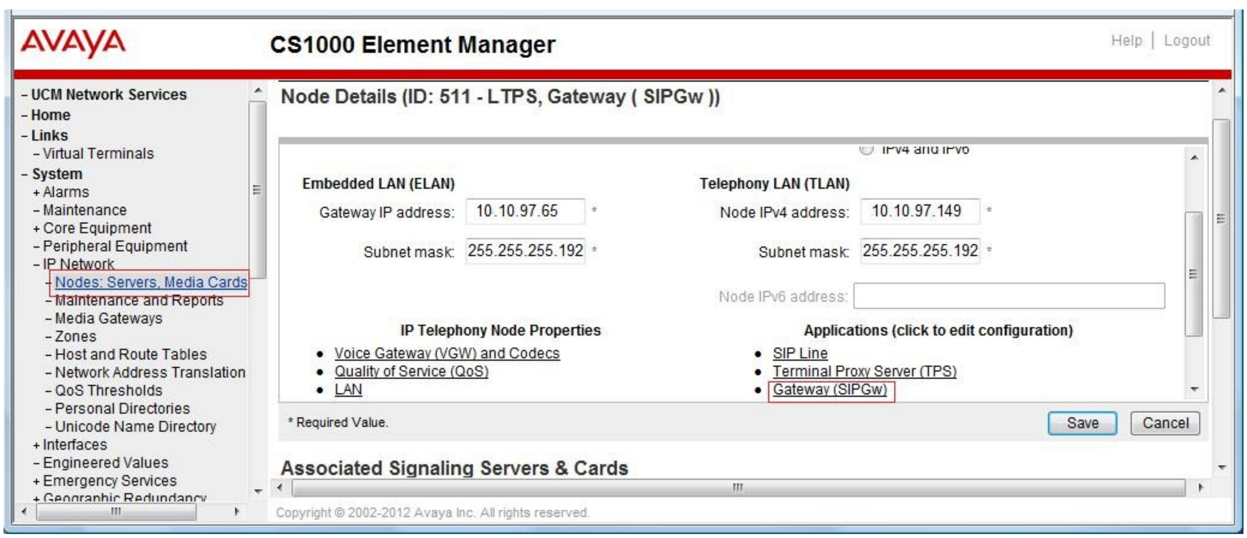
After logging in on the UCM, the **Avaya Unified Communications Management**



Click on the CS1000 CS element highlighted in red-box as shown above. The **System Overview** (EM) home page will appear.

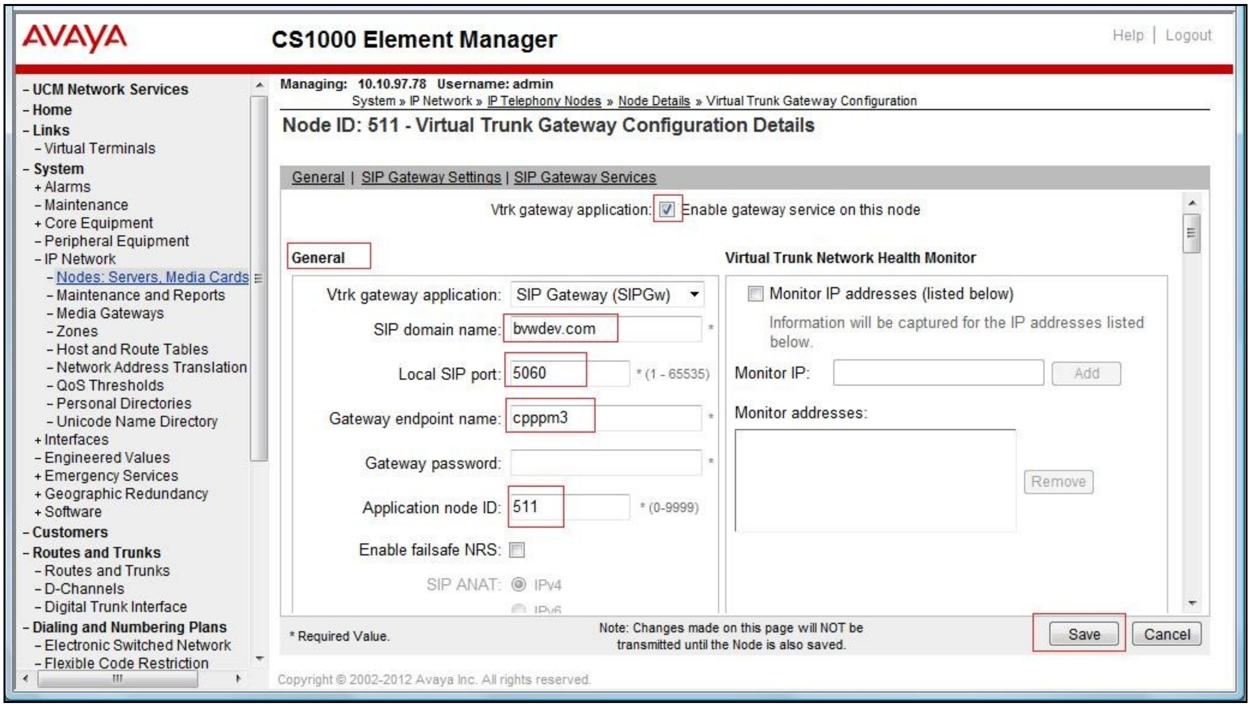


From the left menu column of the EM page, navigate to **System → IP Network → Nodes: Servers, Media Cards**. The **Node ID Telephone** page will appear (not shown).
Click on the **Node ID # 511,** which is **the LTPS, Gateway (SIPGw).** The **Node Details** page will appear.

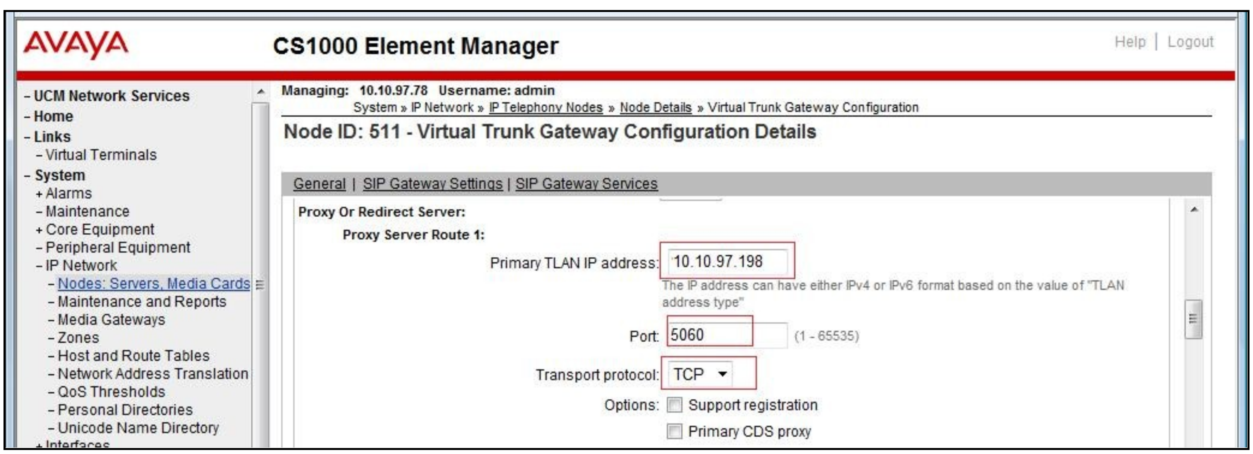


Scroll down under the **Applications**, click on the **Gateway (SIPGw)** link, the **Virtual Trunk Gateway Configuration Details** page will appear in the next 2 screen shots. Enter the information highlighted in the red-box for the **General** and **SIP Gateway Settings**. All other fields are left as default. Click **Save**.

Note: **SIP domain name** should be matched with what was created in **Section 5.1.1**.



The **Primary TLAN IP** address is the IP address used in **Section 5.1.3** the Session Manager IP address.



On the same page, scroll-down the parameters box to the **SIP URI Map** section. Under the **Public E.164 domain names**

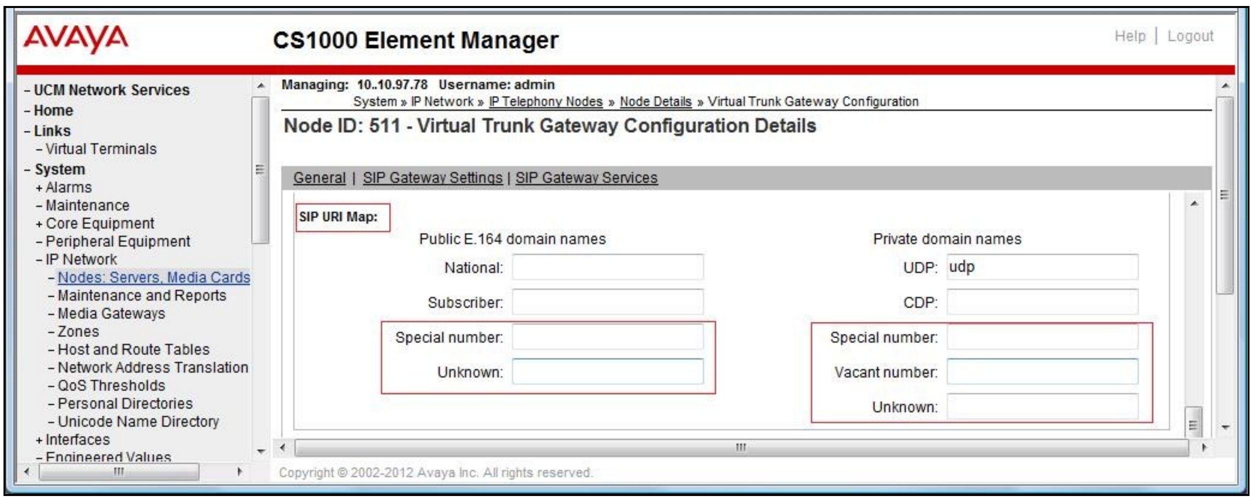
- **Special Number**: leave this SIP URI field as blank
- **Unknown**: leave this SIP URI field as blank

Under the **Private domain names**

- **Special Number**: leave this SIP URI field as blank
- **Vacant number**: leave this SIP URI field as blank
- **National**: leave this SIP URI field as blank
- **UDP**: **udp**
- **CDP**: leave this blank

Click **Save** (not shown).

Note: This will remove the phone context information in the SIP invite URL.



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 (not shown). Then click **Start Sync** button (not shown) to complete the configuration saved process.

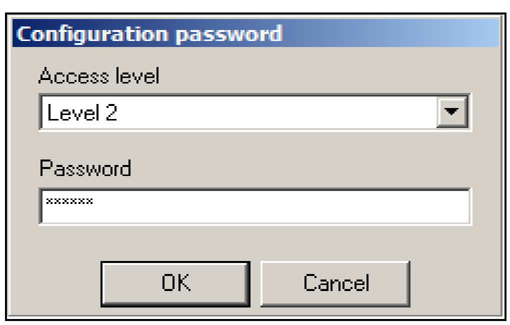
6. Configure Nuance Speech Attendant Server

The Nuance Speech Attendant is a turnkey solution that comprises the Speech Attendant application software pre-installed on a server running Windows Server 2008 R2. This section describes how to configure the Nuance SA application, once the server is received at the customer premise, to interoperate with the Avaya CS1000.

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.

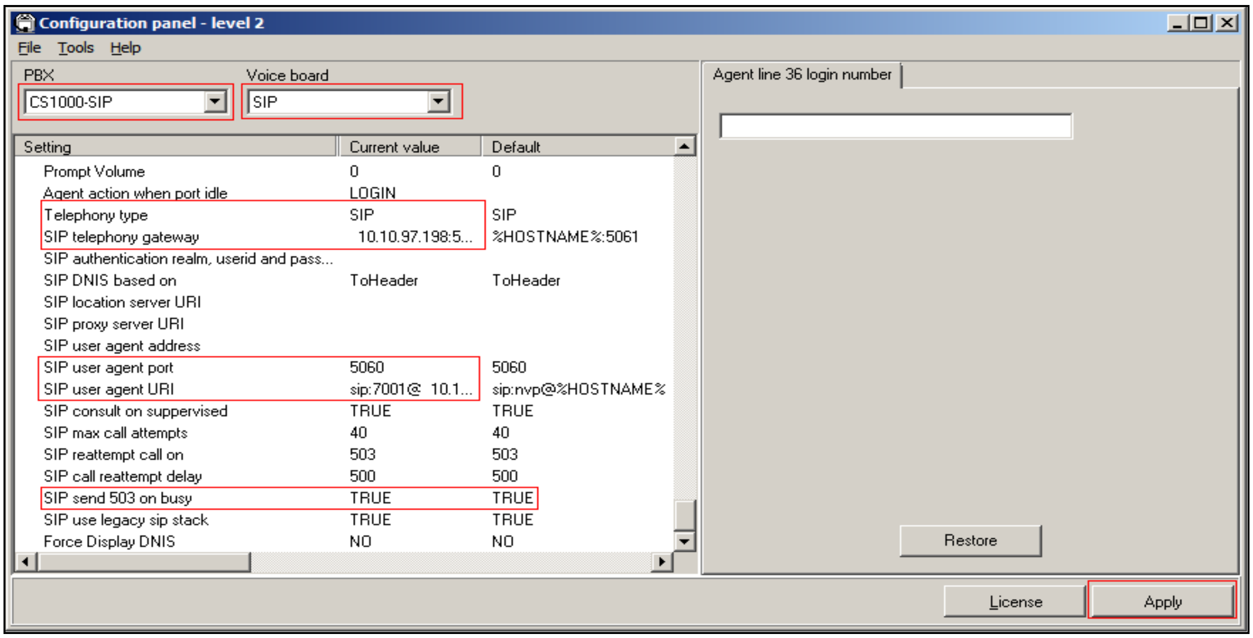
6.1 Configuring Nuance Speech Attendant SIP Interface to Connect to Avaya Communication Server 1000E

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 credential using **Level 2** access level.



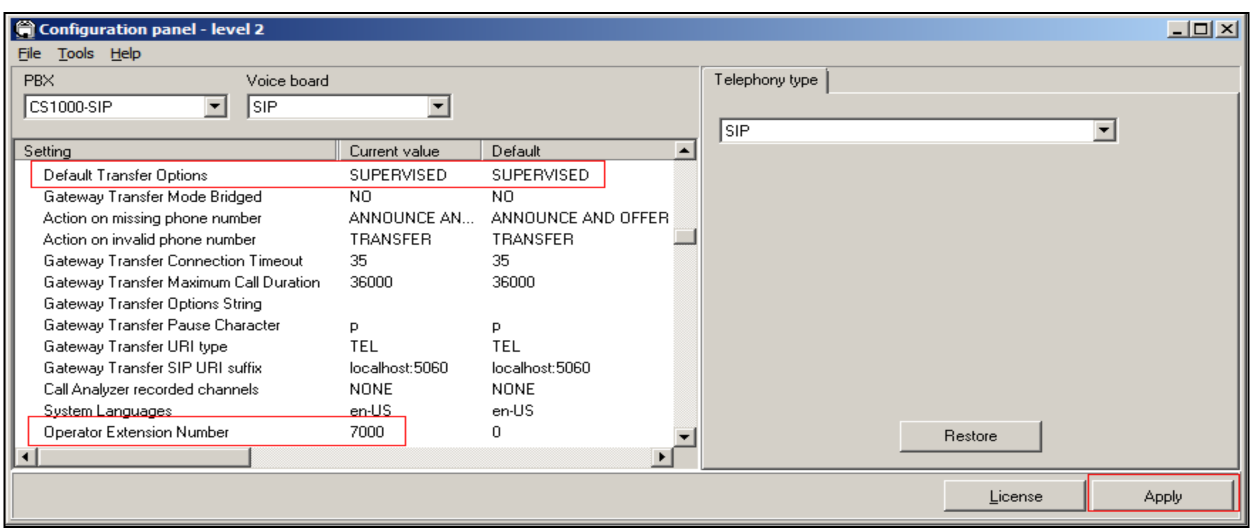
At the Configuration Panel, configure the following parameters as follow:

- **PBX** **CS1000-SIP**
- **Voice board** **SIP**
- **Telephone type** **SIP**
- **SIP Telephony Gateway** **10.10.97.198**
- **SIP user agent port** **5060**
- **SIP user agent URI** **sip:7001@10.10.97.198**
- **SIP send 503 on busy** **TRUE**



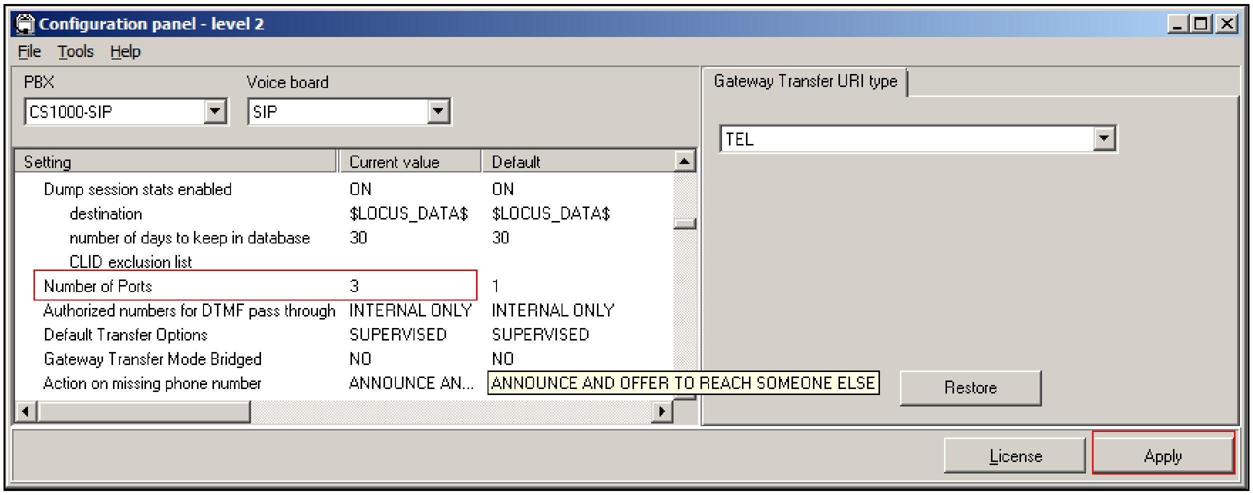
Continue in the **Configuration Panel**, specify the following:

- **Default Transfer Options** **SUPERVISED**
- **Operator Extension Number** **7000**

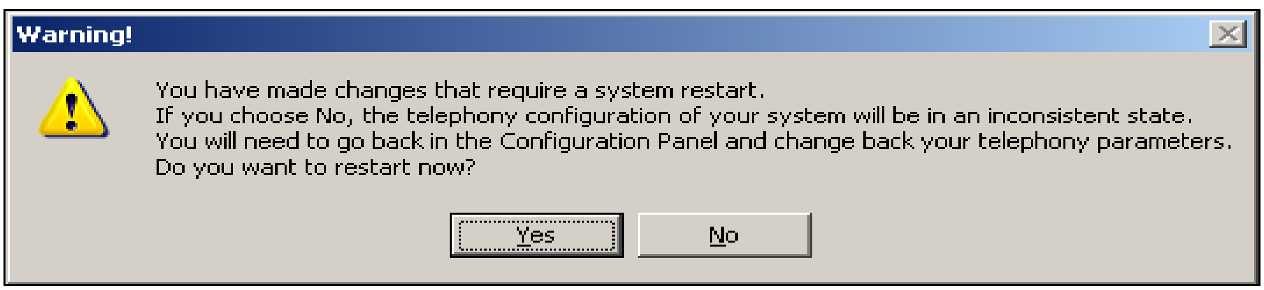


6.2 Configure Number of Ports Supported

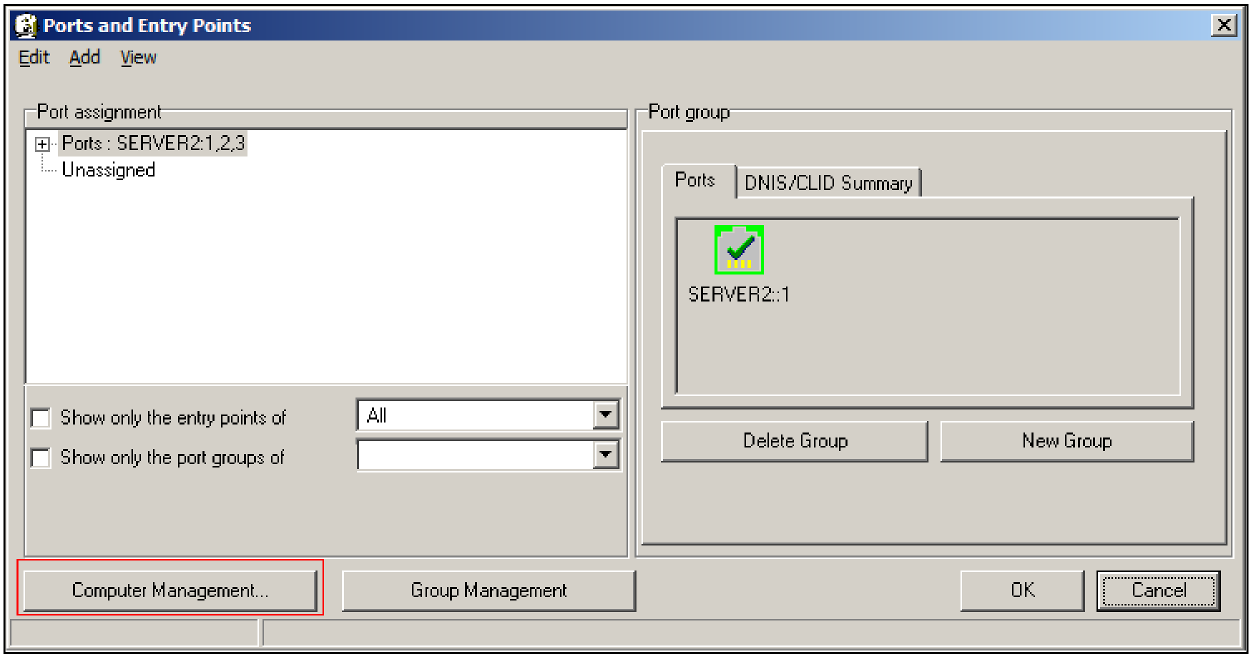
In the **Configuration Panel**, set the **Number of Ports** field to the desired value. In this example, 3 ports were configured. Click **Apply**.



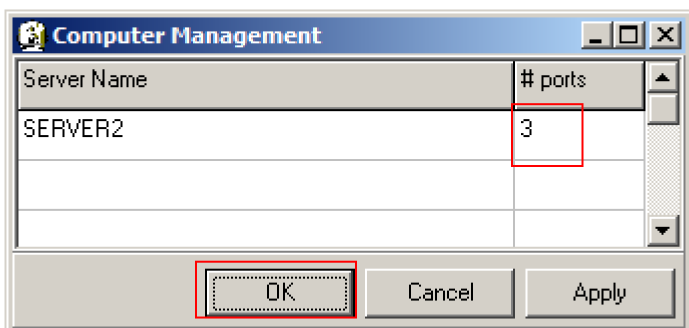
Next, close the **Configuration Panel**. Allow the system to restart when prompted by clicking **Yes** in the dialog box below.



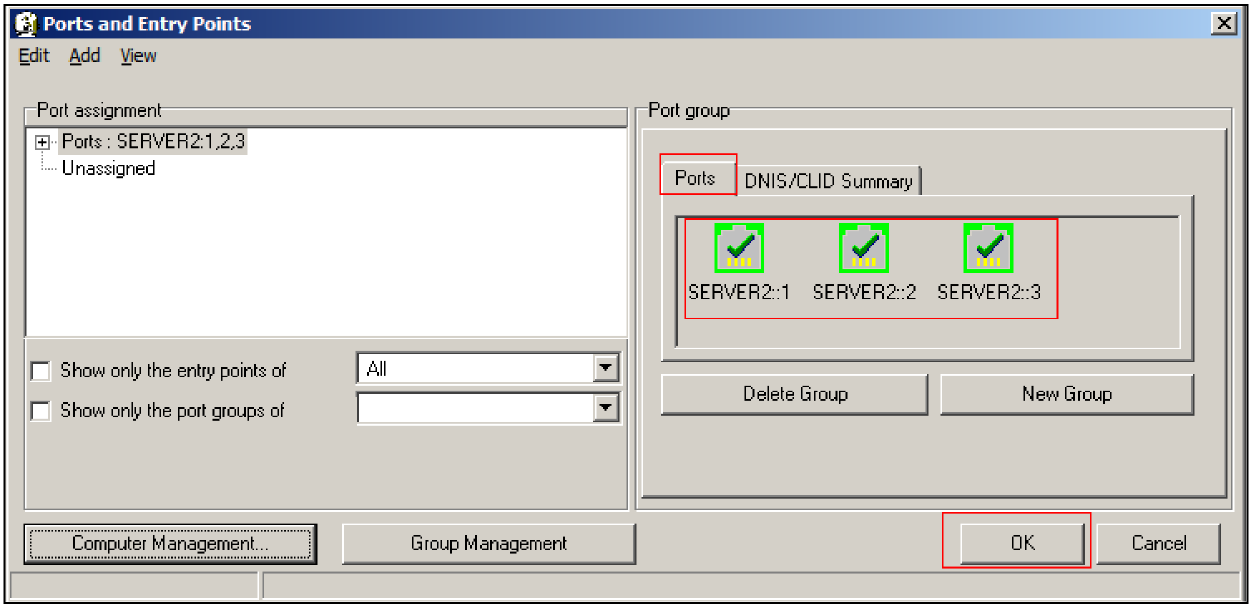
From **Admin Tools**, click on the **Phone Directory and Menu Editor** option and login with the appropriate credential. In the **Menu Editor** (not shown), navigate to **Edit → Ports and Entry Points** to display the window below. Once the **Ports and Entry Points** window is opened, click on the **Computer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



In the **Computer Management** window, set the **# ports** column to the desired value. In this example, 3 ports were configured. Click **OK**.

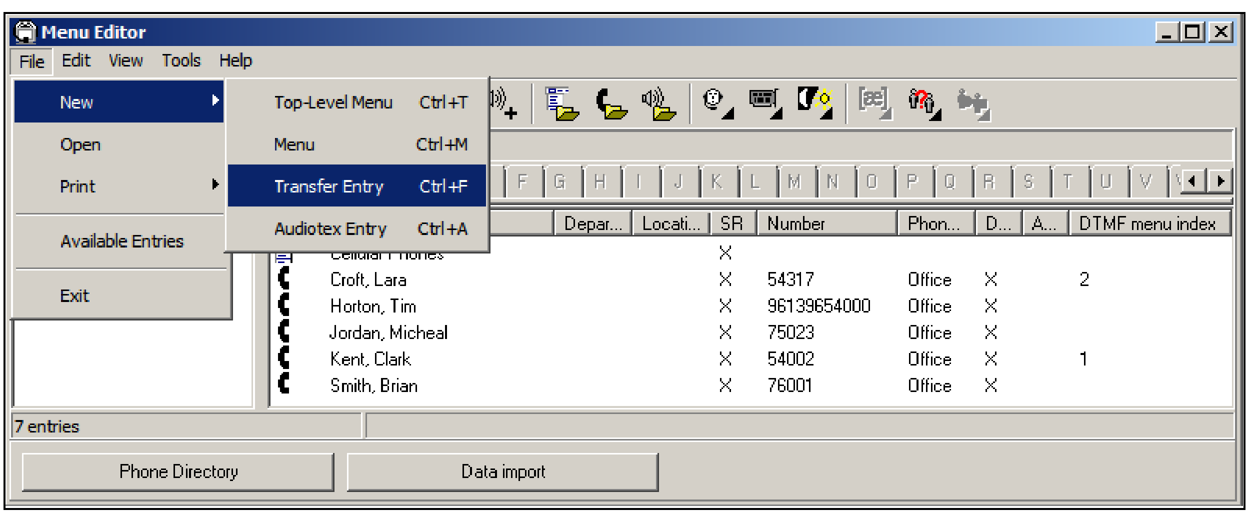


Under the **Port group** section of the **Ports and Entry Points** window, a list of **Ports** in the system will be displayed. Make sure they are all green (enabled). If the ports are grey (disabled), click on each icon to enable them (green) as shown below. Then click **OK**.

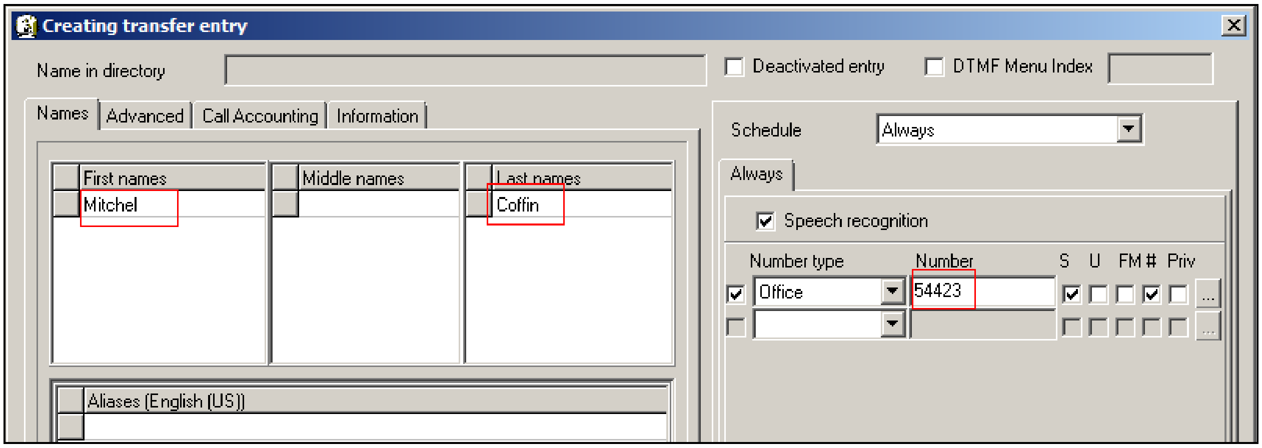


6.3 Configure Transfer Entry In Phone Directory

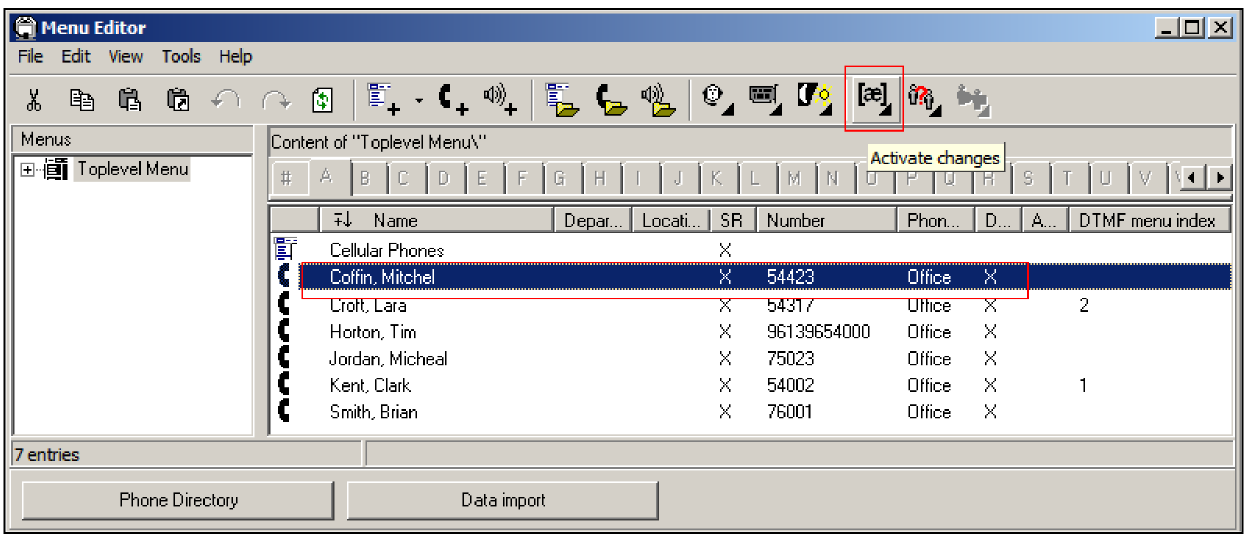
This section shows how to create a **Transfer Entry** in the **In Phone Directory** so the attendant can transfer the call to. From **Admin Tools**, click on the **Phone Directory and Menu Editor** option and login with the appropriate credential. In the **Menu Editor** (not shown), navigate to **File → New → Transfer Entry** as shown bellow.



At the **Creating transfer entry** window, enter the information of the new entry as shown in the red-boxes. Other fields are left as default. Then click **OK** (not shown).



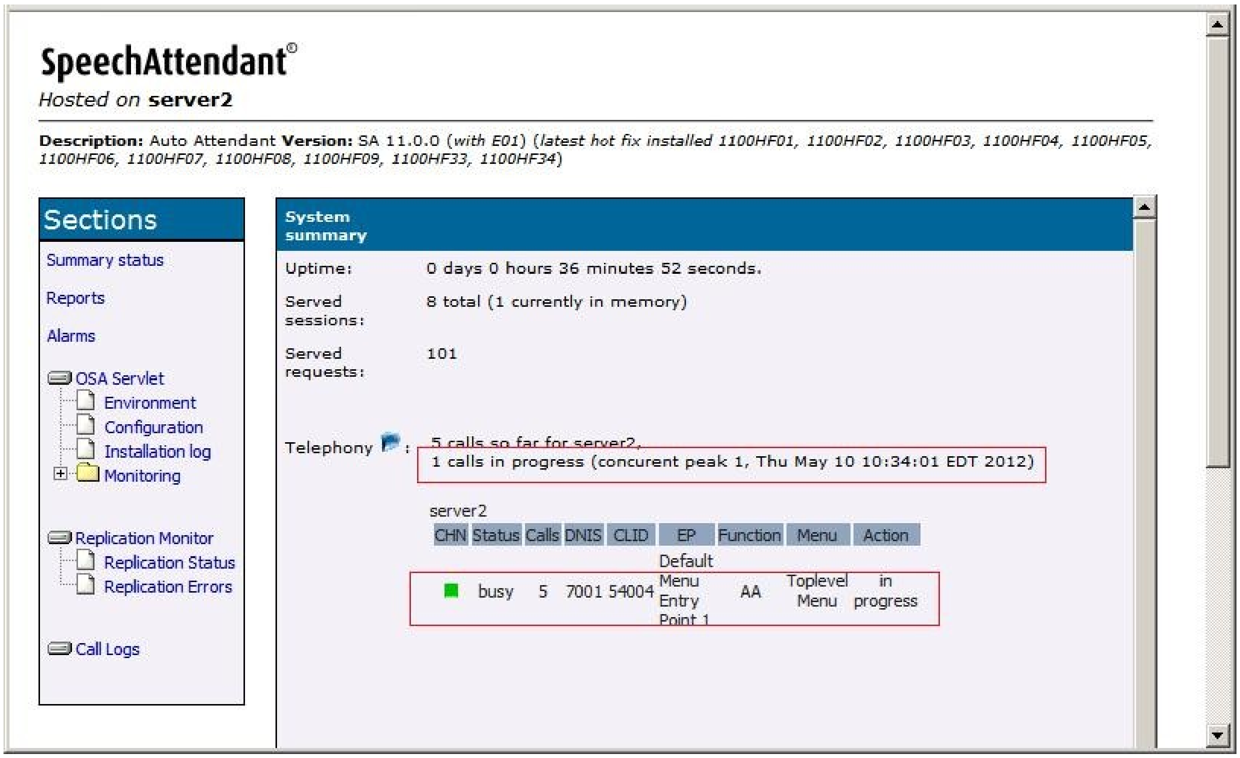
At the **Menu Editor** window, the new entry **Coffin, Mitchel** is presented. To activate the entry, click on the icon highlighted in red-box as shown bellow.



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 Communication Server 1000E 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.



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 have described the administration steps required to integrate the Nuance Speech Attendant 11.1 with the Avaya Communication Server 1000E via SIP trunk configured on the Avaya Aura® Session Manager. All feature and serviceability test cases were completed successfully.

9. 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® Session Manager*, August 2010, Issue 3, Release 6.0, Document Number 03-603324.
[2] *Communication Server 1000 Installation and Commissioning*, April 2012, Release 7.5, Issue 05.08, Document Number NN43041-310.
[3] *Signaling Server IP Line Applications Fundamentals for Avaya Communication Server 1000 (Avaya CS 1000)*, April 2012, Release 7.5, Issue 03.11, Document Number NN43001-125.

Product information for Nuance Speech Attendant can be found at http://www.nuance.com/for-business/by-solution/employee-productivity-solutions/speech-attendant/index.htm

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.