Avaya Solution & Interoperability Test Lab

Application Notes for configuring Parlance Operator Assistant with Avaya Aura® Session Manager and Avaya Communication Server 1000 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Parlance Operator Assistant to interoperate with Avaya Aura® Session Manager 7.0 and Avaya Communication Server 1000 7.6 using SIP trunks. Parlance Operator Assistant automates call routing by asking callers to speak the name or dial the extension of a destination.

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

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

1. Introduction

These Application Notes describe the configuration steps required for Parlance Operator Assistant (hereafter referred to as Operator Assistant) to interoperate with Avaya Aura® Session Manager 7.0 (hereafter referred to as Session Manager) and Avaya Communication Server 1000 7.6 (hereafter referred to as Communication Server 1000) using SIP trunks. Parlance Operator Assistant automates call routing by asking callers to speak the name or dial the extension of a destination.

In the compliance testing, calls from internal and external callers were routed over SIP trunks to Parlance Operator Assistant. Parlance Operator Assistant played different greeting announcements based on ANI and/or DNIS, used speech recognition and/or DTMF digits to determine the route destination, and used SIP REFER to transfer calls to destinations on Avaya Communication Server 1000 or on the PSTN.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were placed manually from users on the PSTN and on Communication Server 1000 to Operator Assistant. Speech and DTMF input were used from the callers for requesting transfer to internal user and group destinations on Communication Server 1000, and to external destinations on the PSTN.

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

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

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

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

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

2.2. Test Results

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

- The application only supports the G.711MU codec.

- For Supervised transfer, changes needs to be done in the **PhoneConfig_Overrides.ini** file in the Operator Assistant as shown below, where **10.10.97.228** is the IP address of the Session Manager.

```
[Generic]
;managed_transfer_template = None
basic_transfer_template = sip:%s@10.10.97.228
;sip_2_sip_transfertype = conditional
```

2.3. Support

Technical support on Operator Assistant can be obtained through the following:

- **Phone:** (888) 700-6263
- **Email:** customerservice@parlancecorp.com
- **Web :** www.parlancecorp.com

3. Reference Configuration

As shown in **Figure 1**, SIP trunks were used between Session Manager and Operator Assistant.

A five digit Uniform Dial Plan (UDP) was used to facilitate routing with Operator Assistant. Unique extension ranges were assigned to users on Communication Server 1000 (54xxx), and to Operator Assistant (30xxx).

The configuration of Session Manager is performed via the web interface of System Manager. The detailed administration of basic connectivity between Communication Server 1000, System Manager and Session Manager is not the focus of these Application Notes and will not be described.

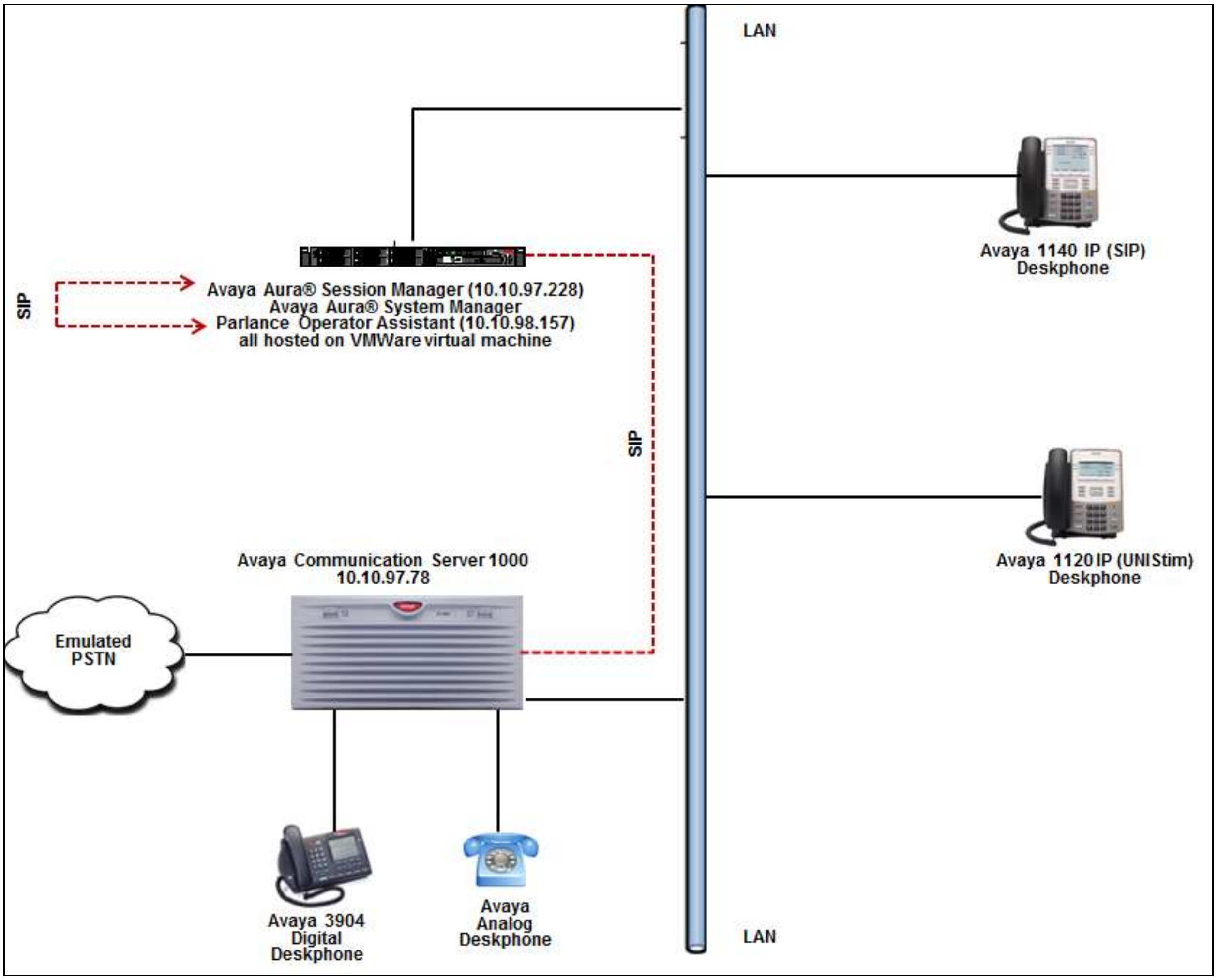


Figure 1: Compliance Testing Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Communication Server 1000	7.65.16 SP7
Avaya Aura® Session Manager in Virtual Environment	7.0.0.2.700201
Avaya Aura® System Manager in Virtual Environment	7.0.0.2
Avaya IP Deskphones: • 1120 (UNIStim) • 1140 (SIP)	 C8Q 4.03.09
Avaya Digital Deskphone	N/A
Avaya Analog Deskphone	N/A
Parlance Operator Assistant running on Microsoft Windows Server 2012 R2	N/A

5. Configure Avaya Communication Server 1000

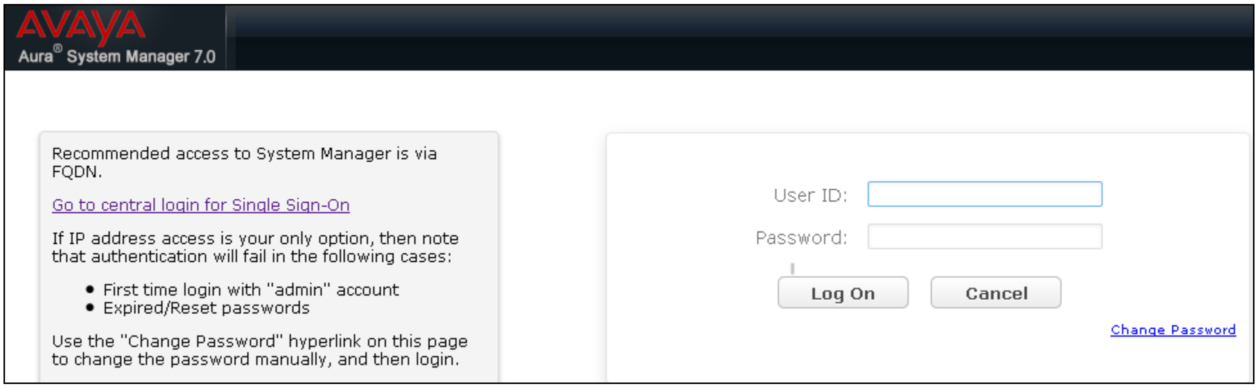
This section describes the Communication Server 1000 configuration necessary to interoperate with Session Manager and Responder. It provides the procedures for configuring Avaya Communication Server 1000 system. The procedures include the following areas:

- Logging into the Element Manager via Unified Communication Manager
- Configuring the SIP Signaling Gateway.
- Configuring a D-Channel.
- Configuring Route and Trunks.
- Configuring Digit Manipulation Block.
- Configuring Route List Block.
- Configuring Distant Steering Code.

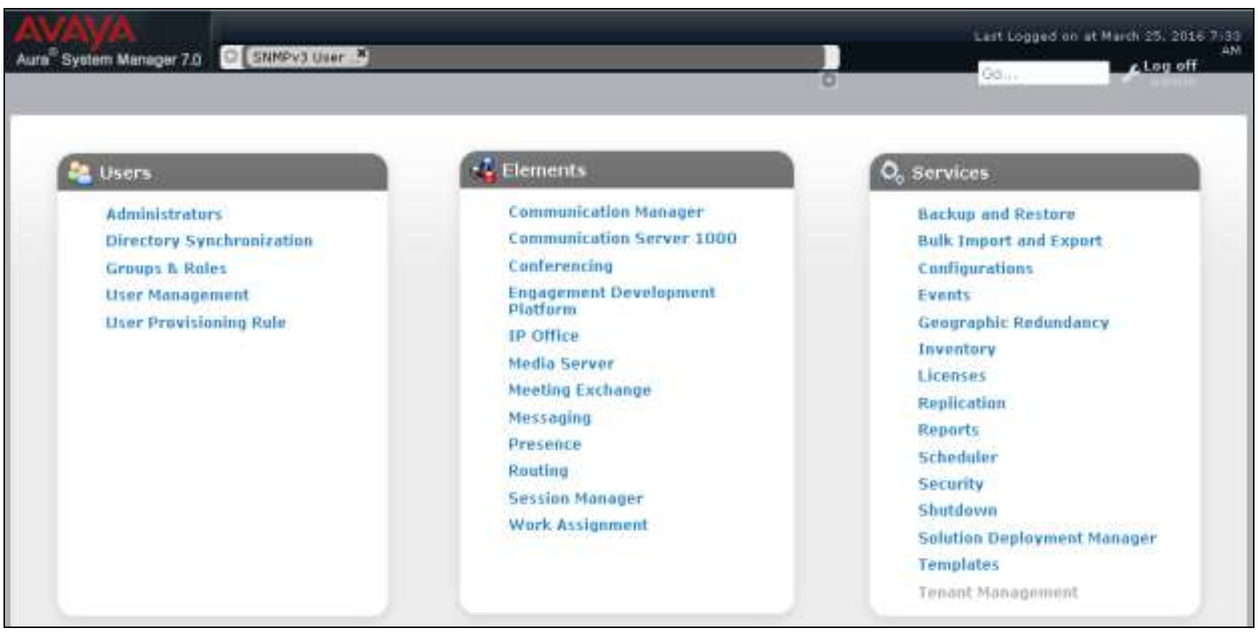
For detail configuration details of the Communication Server 1000 refer to **Section 10**.

5.1. Logging into Element Manager via Avaya Aura® System Manager

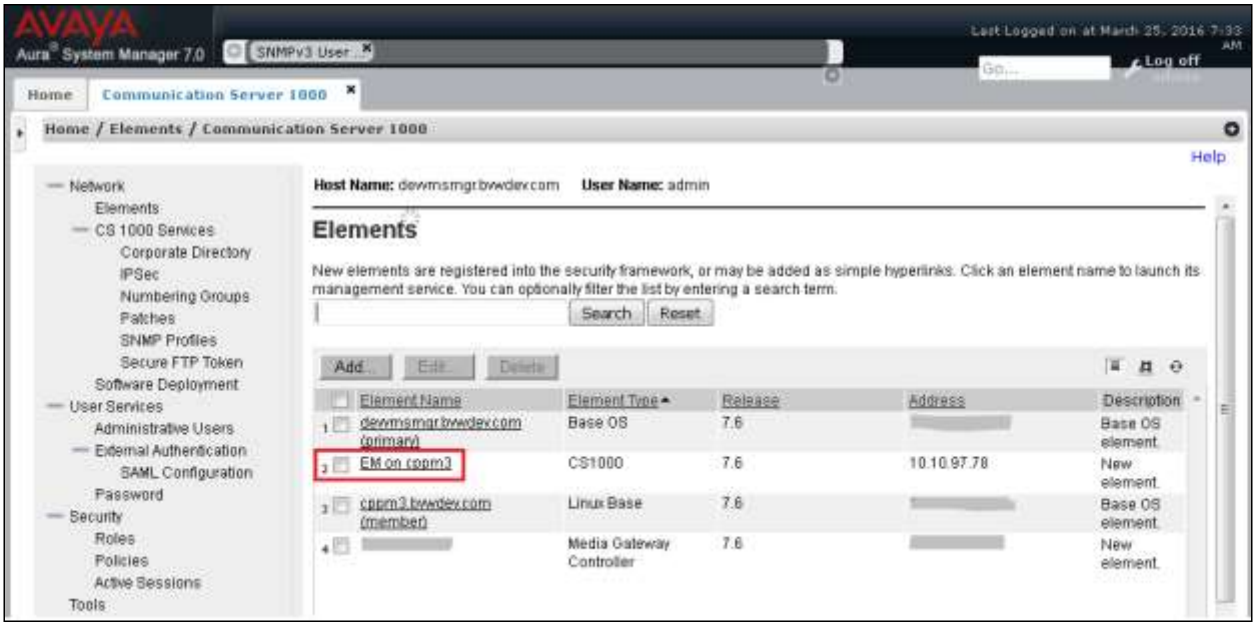
User can login to the Element Manager via System Manager or Unified Communication Manager. During this compliance testing System Manager was used to login to the Element Manager. Access the System Manager web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of System Manager. Log in using the appropriate credentials.



From the main dashboard, select **Communication Server 1000** that is seen under the **Elements** column as shown below.



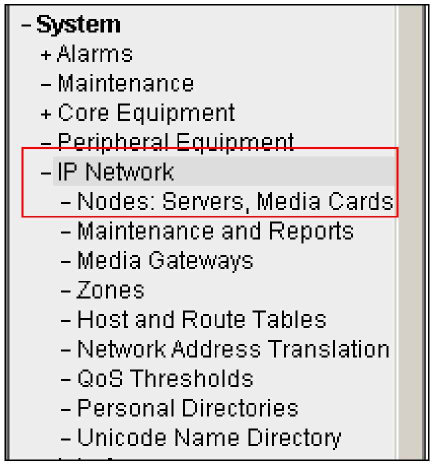
From the **Elements** page of System Manager as shown in screen below, click on the Element **EM on cppm3**. This is the element which is configured to access the Element Manager (EM) for the Communication Server 1000 Call Server.



5.2. Configuring the SIP Signaling Gateway

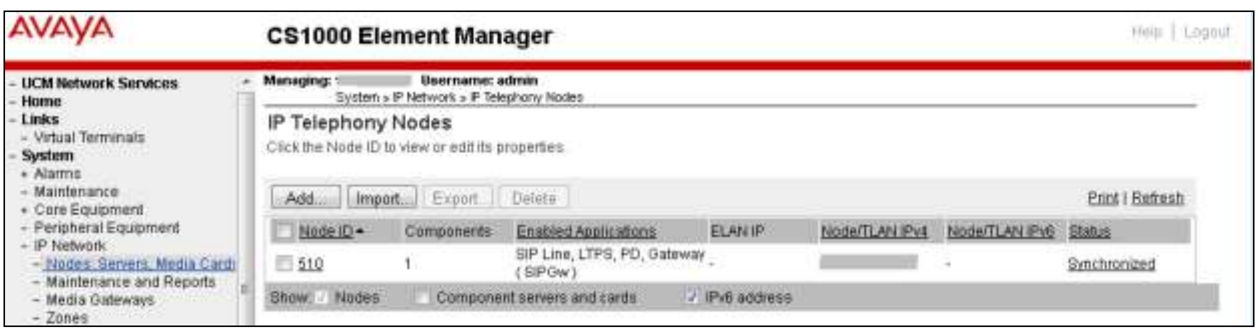
This section describes the configuration required on the SIP Signaling Gateway so that the Communication Server 1000 can communicate with the Session Manager via SIP Trunks.

To add a Node, from the EM left navigator screen, navigate to **System →IP Network → Nodes: Servers, Media Cards** as shown below.

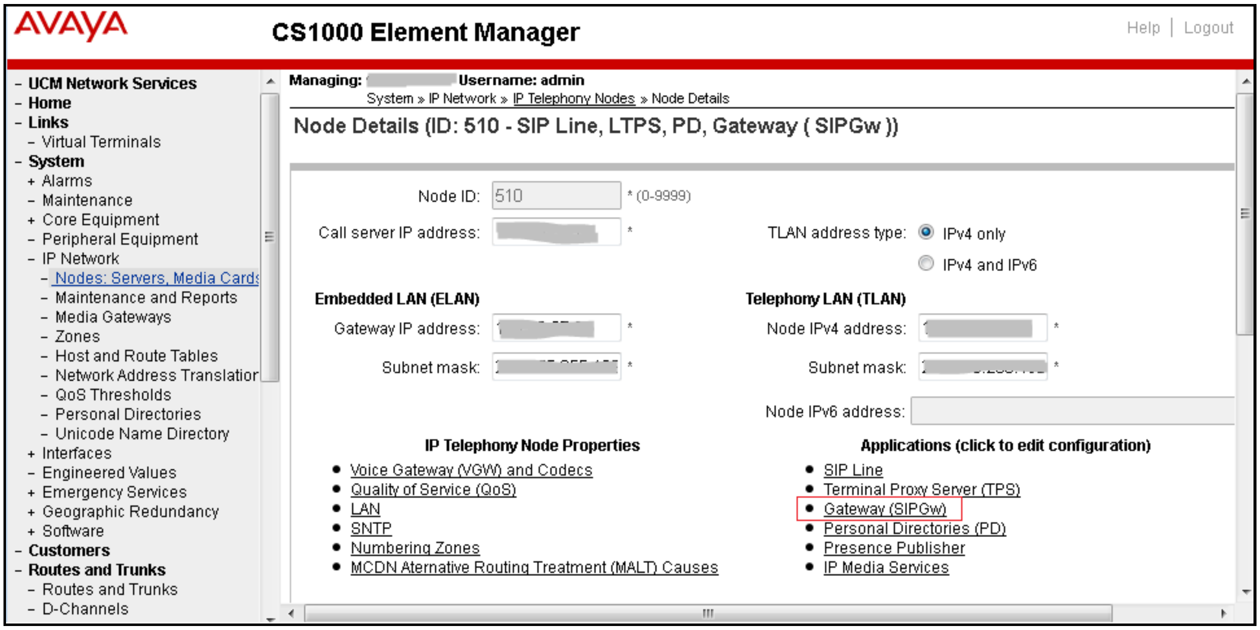


Assumption is made here that the IP Telephony node is already added.

During compliance testing Node **510** was added. Click on this Node as shown in screen below to view the configured values.



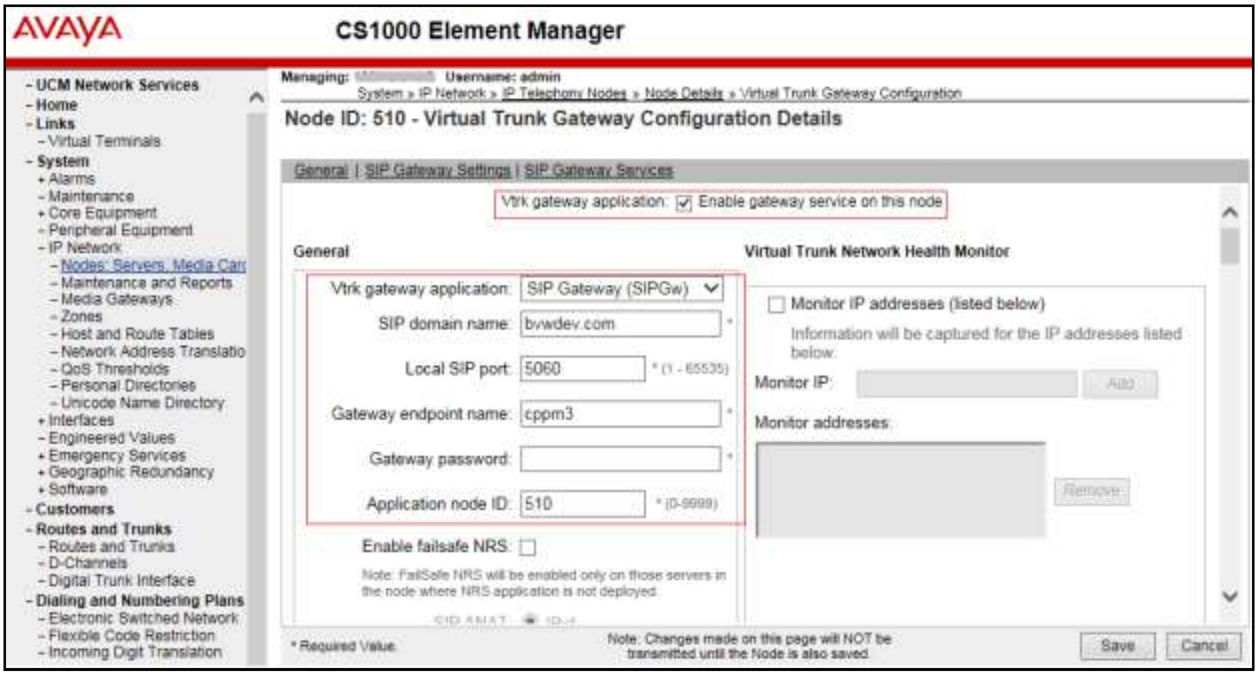
Open the SIP Signaling Gateway configuration by clicking on **Gateway (SIPGw)** as shown below from the Node Details page.



The following values were configured during compliance testing as shown in the screen below.

- **Vtrk gateway application**: Check the *Enable gateway service on this node* box.
- **Vtrk gateway application**: Select *SIP Gateway (SIPGw)* from the drop down menu.
- **SIP domain name**: *bvwdev.com*. This will be the same domain name that will be configured on the Session Manager.
- **Local SIP port**: *5060*.
- **Gateway endpoint name**: *cppm3*.
- **Application node ID**: *510*.

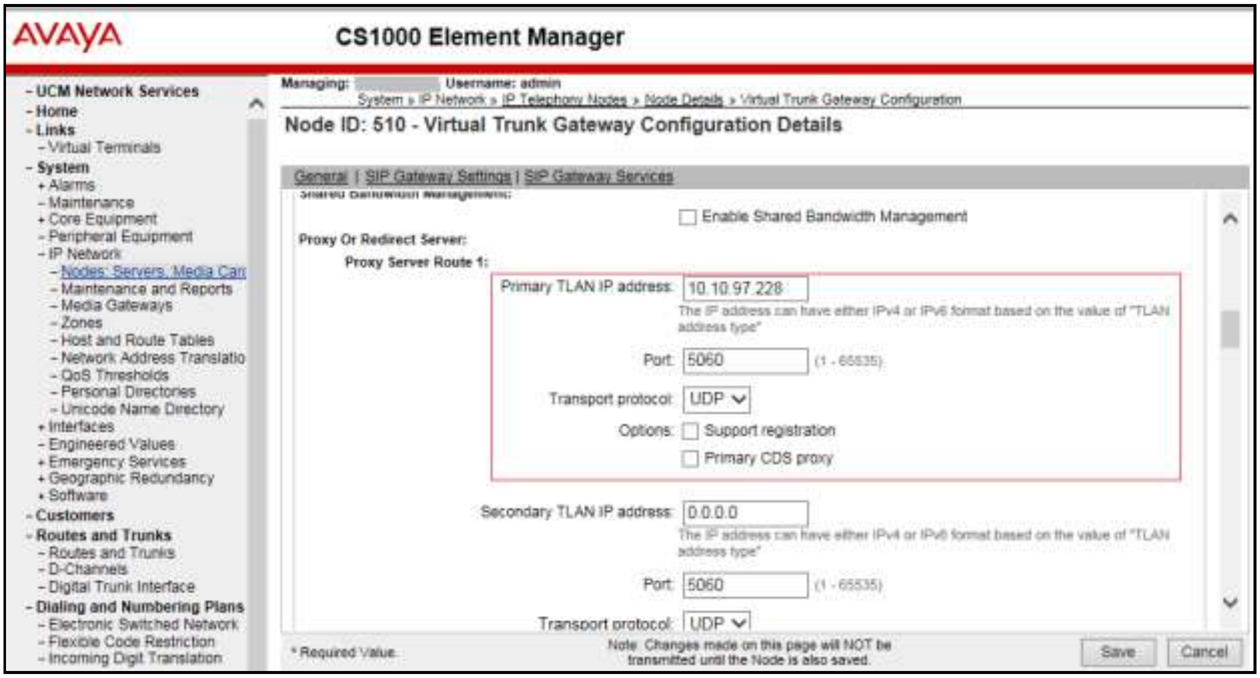
Retain default values for other fields.



Scroll down to the **Proxy or Redirect Server** section. The following values were configured during compliance testing.

- **Primary TLAN IP address**: *10.10.97.228*. This is the IP address of the Session Manager.
- **Port**: 5060
- **Transport protocol**: Select *UDP* from the drop down menu.

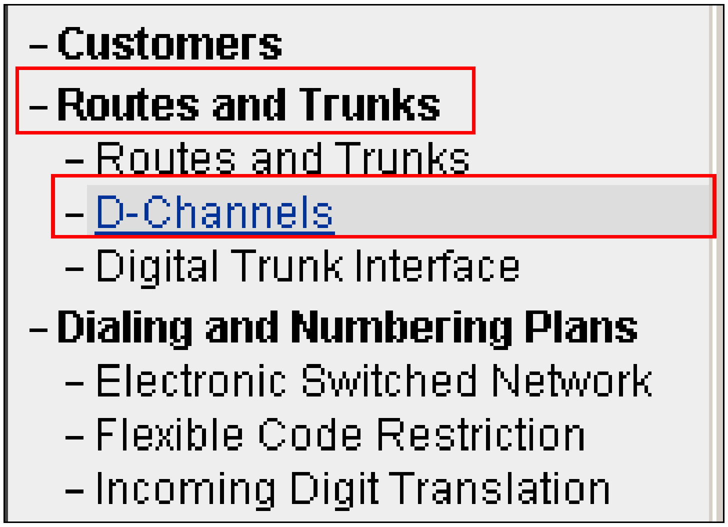
Retain default values for other fields.



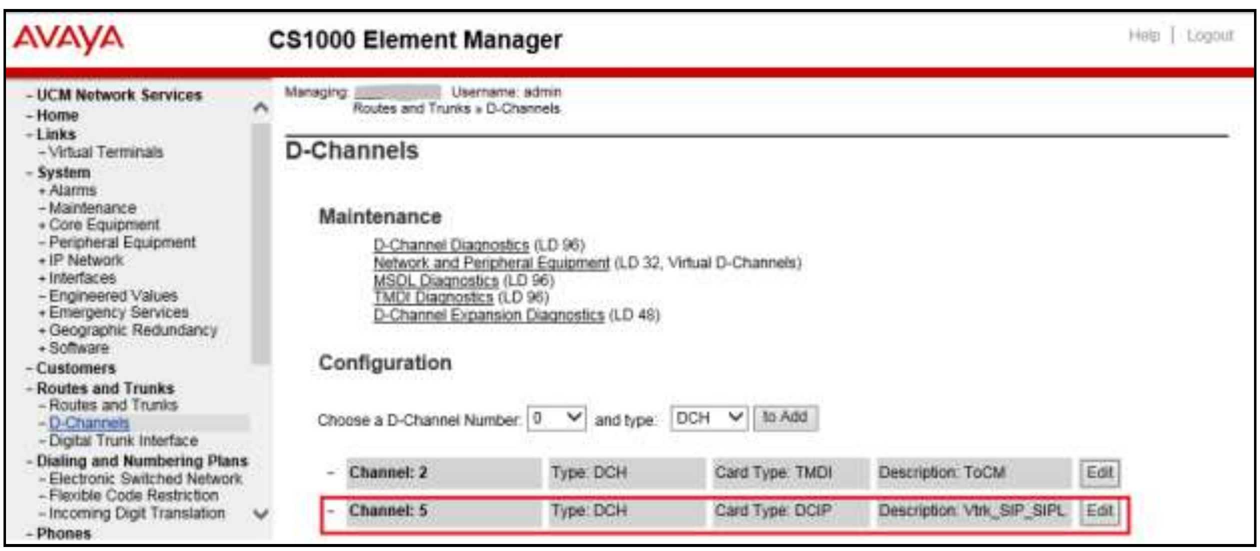
Save and transmit (not shown) these Node properties to complete the SIPGw configuration.

5.3. Configuring D-Channel

This section explains the configuration of a D-Channel for a SIP Trunk. From the EM navigation screen, navigate to **Routes and Trunks → D-Channels** as shown below.



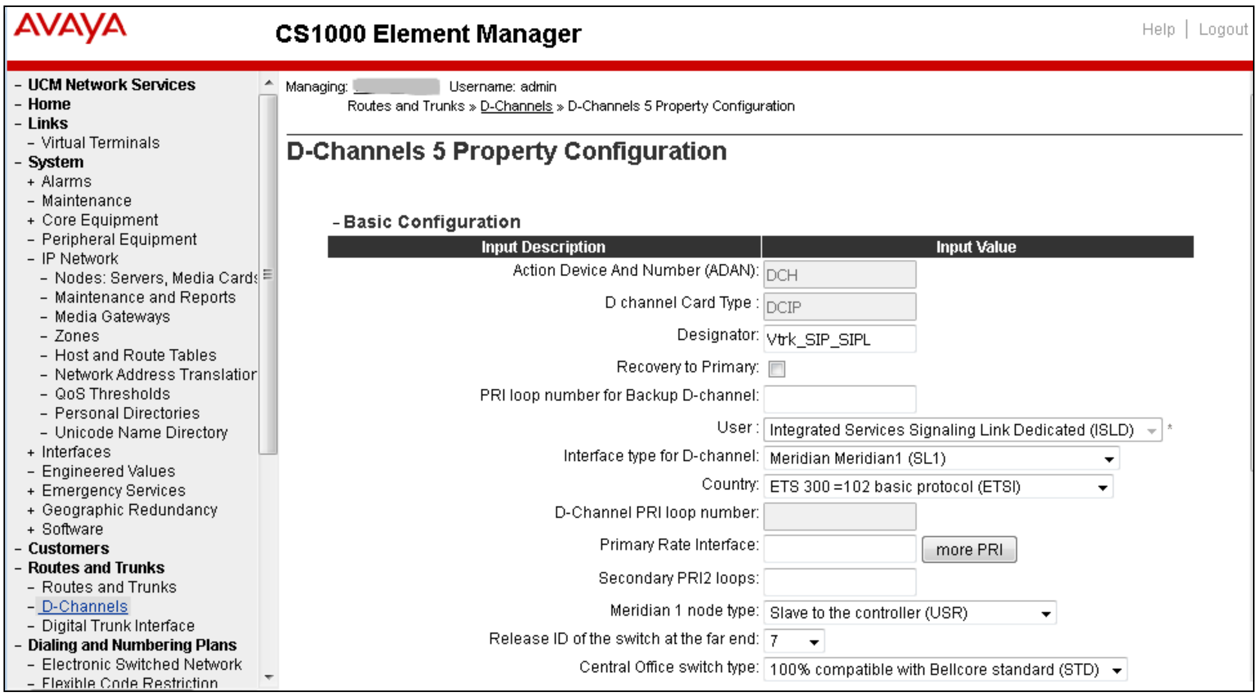
Choose an available D-Channel number to add as shown in the screen below. During compliance testing D-Channel number **5** was configured. Click on **Edit** to view its configuration.



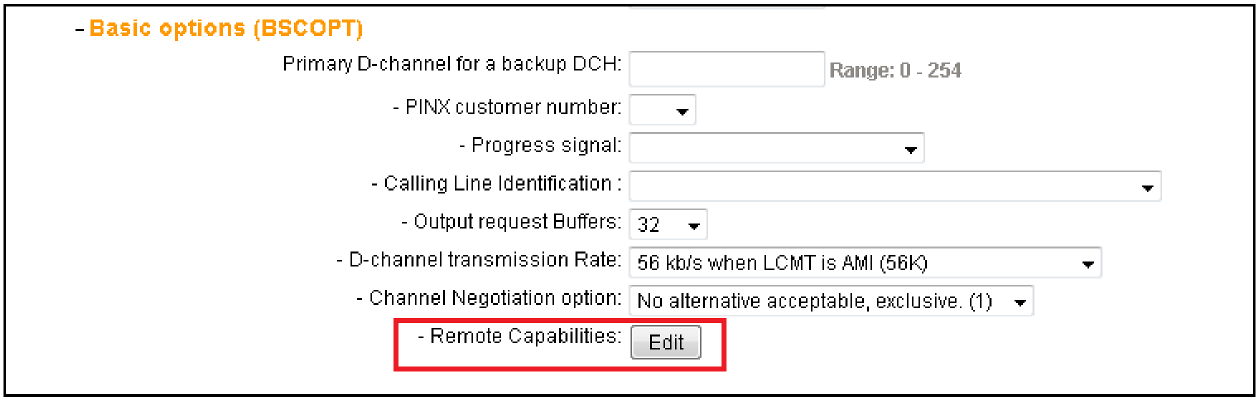
The following values were configured in **Basic Configuration** for the D-Channel as shown below.

- **Action Device And Number (ADAN)**: *DCH*.
- **D channel Card Type**: *DCIP*.
- **Designator**: A descriptive name.
- **Inerface type for D-channel**: Select *Meridian Meridian1 (SL1)* from the drop down menu.
- **Meridian 1 node type**: Select *Salve to the controller (USR)* from the drop down menu.
- **Release ID of the switch at the far end**: Select *7* from the drop down menu.

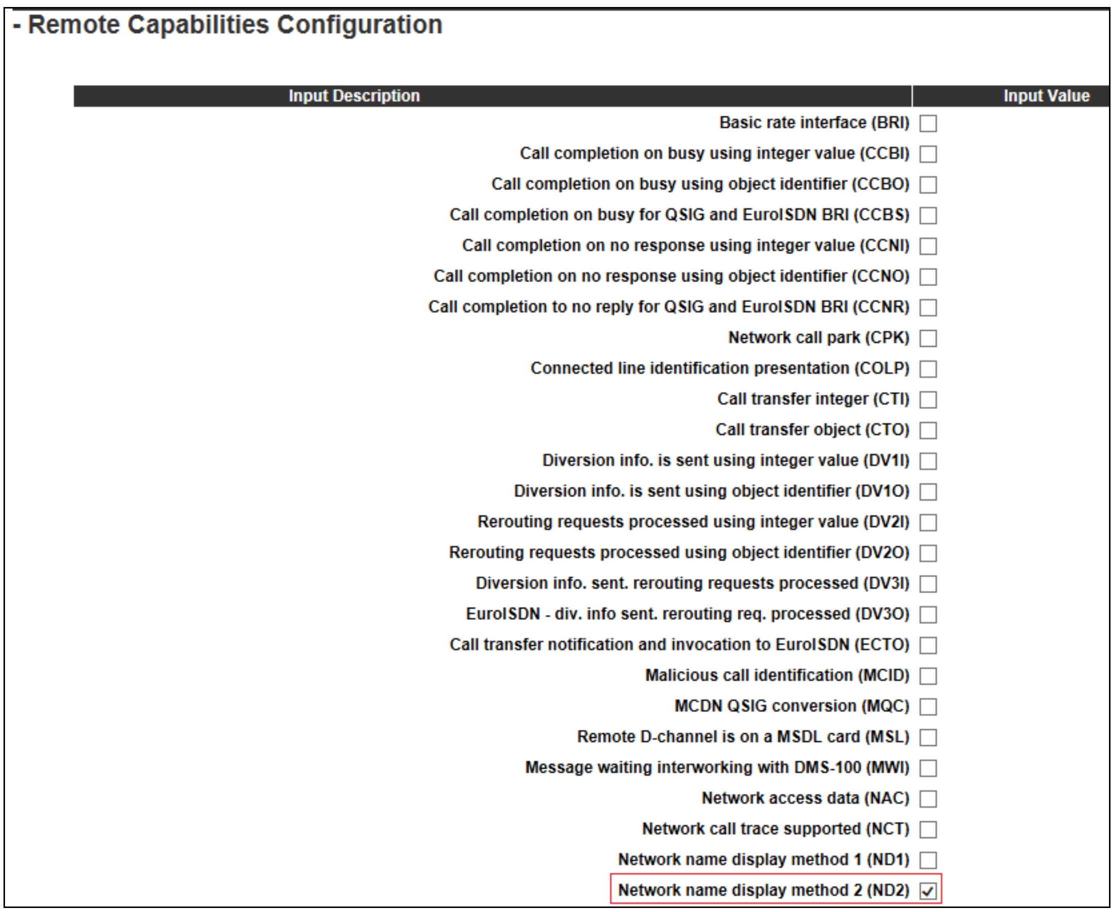
Retain default values for all other fields.



Scroll down to edit the **Remote Capabilities** of the D-Channel that is seen under the **Basic options (BSCOPT)** section. Click on **Edit** button as shown in the screen below.



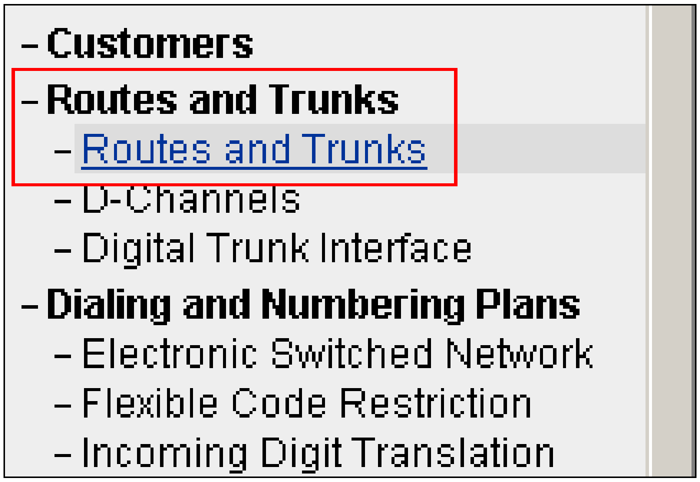
Enable the **Network name display method 2 (ND2)** option. Now click on **Return - Remote Capabilities** button (not shown) to return back to the main screen.



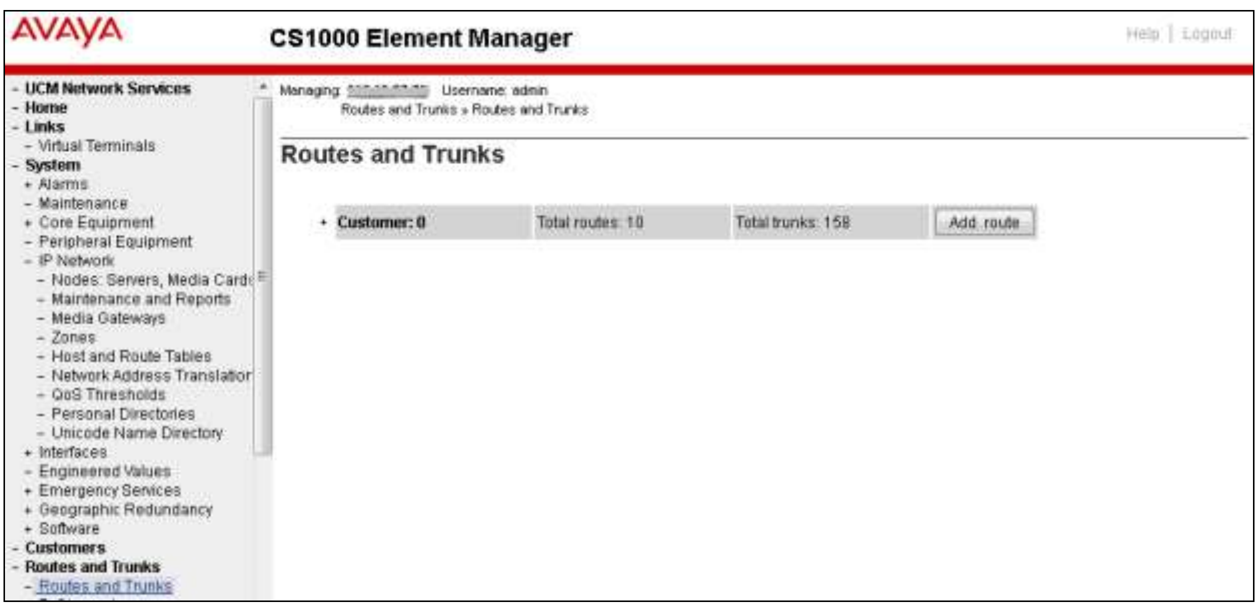
Now click on the **Submit** button (not shown) to complete the D-channel configuration.

5.4. Configuring Route and Trunks

This section explains the configuration of the SIP route and trunks which will be used by Communication Server 1000 to communicate with the Session Manager. To add a new route, navigate to **Routes and Trunks → Routes and Trunks** from the EM left hand navigator window as shown in screen below.



Now from the **Routes and Trunks** screen as shown below click on **Add route** button to start configuring a new route.

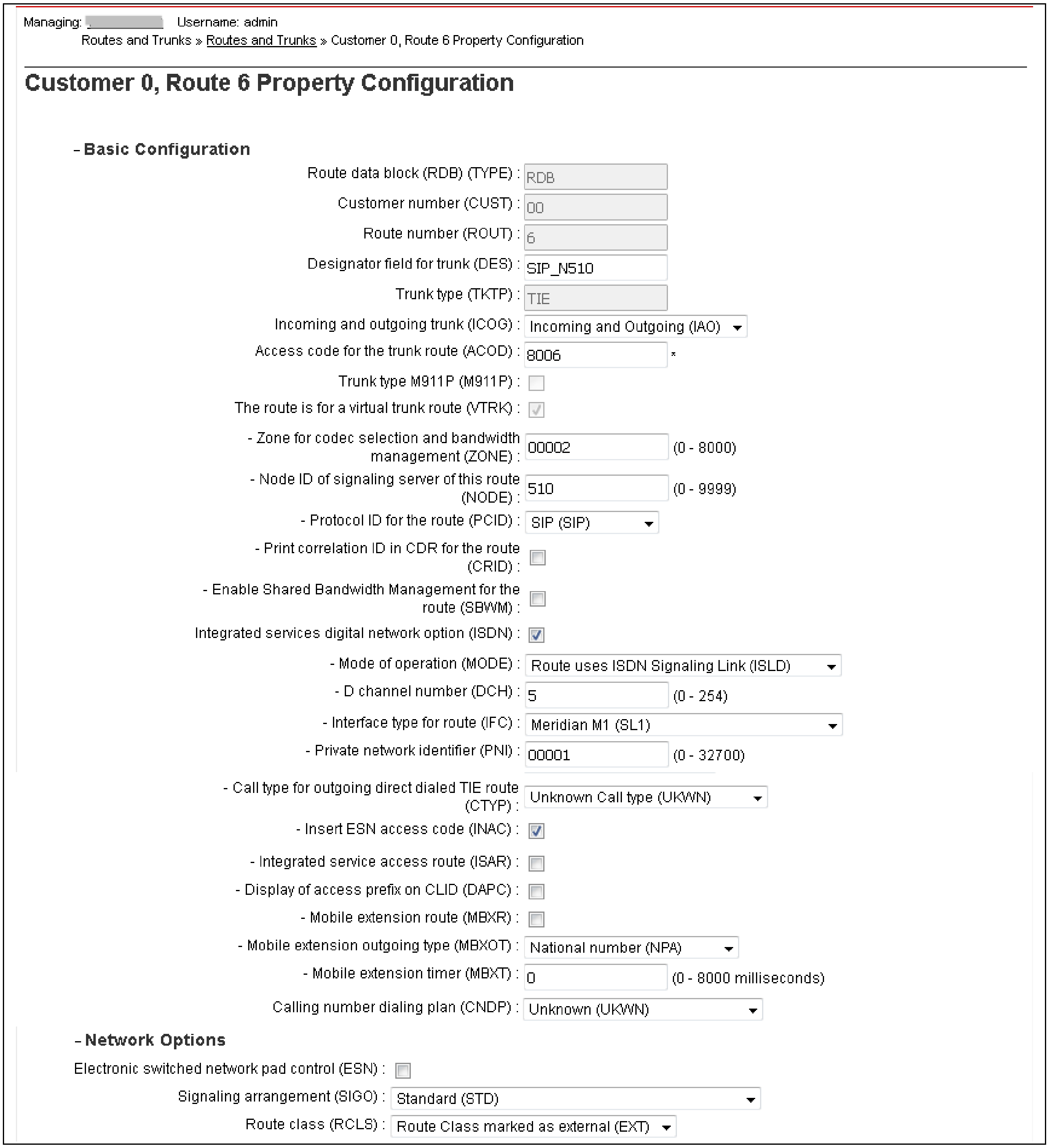


During compliance testing route **6** was added. The next three screens below shows the configuration for route 1 used during compliance testing.

- **Route data block (RDB) (TYPE)**: *RDB*
- **Customer number (CUST)**: *00*
- **Route number (ROUT)**: *6*
- **Designator field for trunk (DES)**: A descriptive name.
- **Trunk type (TKTP)**: *TIE*
- **Incoming and outgoing trunk (ICOG)**: Select *Incoming and Outgoing (IAO)* from the drop down menu.
- **Access code for the trunk route (ACOD)**: An available Directory number from the system.
- **The route is for a virtual trunk route (VTRK)**: Enable the box.
- **Zone for codec selection and bandwidth management (ZONE)**: A number configured in the system.
- **Node ID of signaling server of this route (NODE)**: *510*; this is the same node added in **Section 5.2**.
- **Protocol ID for the route (PCID)**: Select *SIP (SIP)* from the drop down menu.
- **Integrated services digital network option (ISDN)**: Enable the box.
- **D channel number (DCH)**: *5*; this is the same D channel added in **Section 5.3**.
- **Interface type for route (IFC)**: Select *Meridian M1 (SL1)* from the drop down menu.
- **Private network identifier (PNI)**: A value configured in the system.
- **Call type for outgoing direct dialed TIE route (CTYP)**: Select *Unknown Call Type (UKWN)* from the drop down menu.
- **Calling number dialing plan (CNDP)**: Select *Unknown (UKWN)* from the drop down menu.
- **Signaling arrangement (SIGO)**: Select *Standard (STD)* from the drop down menu.
- **Route class (RCLS)**: Select *Route Class marked as external (EXT)* from the drop down menu.

Retain default values for other fields.

Now click on the **Submit** button (not shown) to complete the configuration.

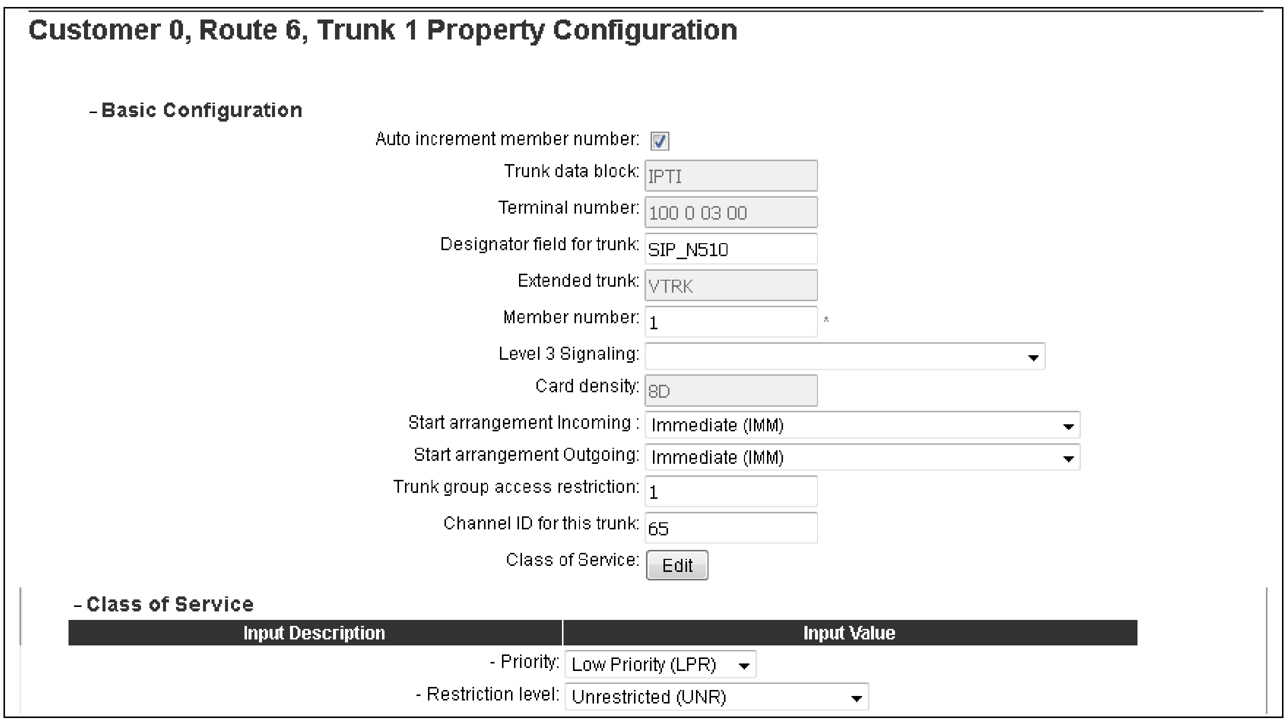

Managing: Username: admin
Routes and Trunks » Routes and Trunks » Customer 0, Route 6 Property Configuration
Customer 0, Route 6 Property Configuration
- Basic Configuration
Route data block (RDB) (TYPE) : RDB
Customer number (CUST) : 00
Route number (ROUT) : 6
Designator field for trunk (DES) : SIP_N510
Trunk type (TKTP) : TIE
Incoming and outgoing trunk (ICOG) : Incoming and Outgoing (IAO)
Access code for the trunk route (ACOD) : 8006 *
Trunk type M911P (M911P) :
The route is for a virtual trunk route (VTRK) :
- Zone for codec selection and bandwidth management (ZONE) : 00002 (0 - 8000)
- Node ID of signaling server of this route (NODE) : 510 (0 - 9999)
- Protocol ID for the route (PCID) : SIP (SIP)
- Print correlation ID in CDR for the route (CRID) :
- Enable Shared Bandwidth Management for the route (SBWM) :
Integrated services digital network option (ISDN) :
- Mode of operation (MODE) : Route uses ISDN Signaling Link (ISLD)
- D channel number (DCH) : 5 (0 - 254)
- Interface type for route (IFC) : Meridian M1 (SL1)
- Private network identifier (PNI) : 00001 (0 - 32700)
- Call type for outgoing direct dialed TIE route (CTYP) : Unknown Call type (UKWN)
- Insert ESN access code (INAC) :
- Integrated service access route (ISAR) :
- Display of access prefix on CLID (DAPC) :
- Mobile extension route (MBXR) :
- Mobile extension outgoing type (MBXOT) : National number (NPA)
- Mobile extension timer (MBXT) : 0 (0 - 8000 milliseconds)
Calling number dialing plan (CNDP) : Unknown (UKWN)
- Network Options
Electronic switched network pad control (ESN) :
Signaling arrangement (SIGO) : Standard (STD)
Route class (RCLS) : Route Class marked as external (EXT)

After the route has been configured, trunks can be added that belongs to this route. The two screens below shows the configuration of the trunks that was used during compliance testing.

- **Auto increment member number**: Enable this box.
- **Trunk data block**: *IPTI*
- **Terminal number**: An available terminal number from the system.
- **Designator field for trunk**: A descriptive name.
- **Extended trunk**: *VTRK*
- **Member number**: *1*; this is the starting member number of the trunk.
- **Start arrangement Incoming**: Select *Immediate (IMM)* from the drop down menu.
- **Start arrangement Outgoing**: Select *Immediate (IMM)* from the drop down menu.
- **Class of Service**: Click on the **Edit** button.
- **Restriction level**: Select *Unrestricted (UNR)* from the drop down menu.

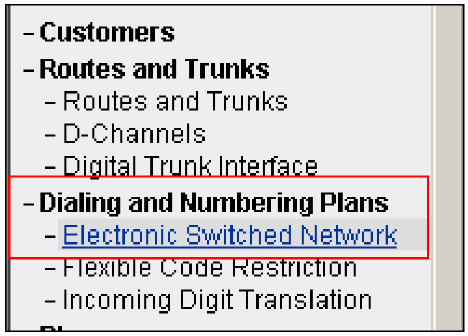
Retain default values for other fields.

Now click on **Return Class of Service** button (not shown) to return to the main page of trunks configuration. Click on **Save** button (not shown) to complete the trunks configuration.

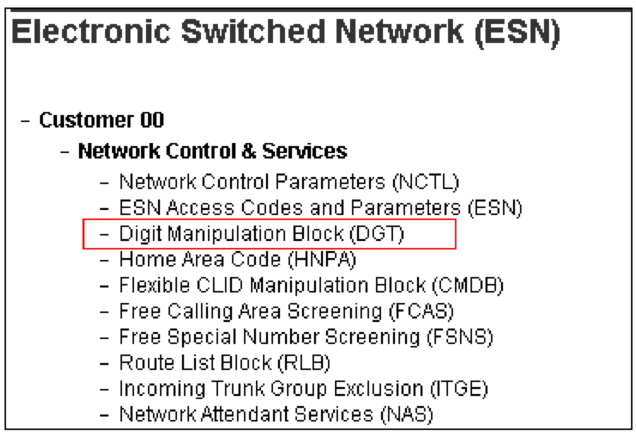


5.5. Configuring Digit Manipulation Block

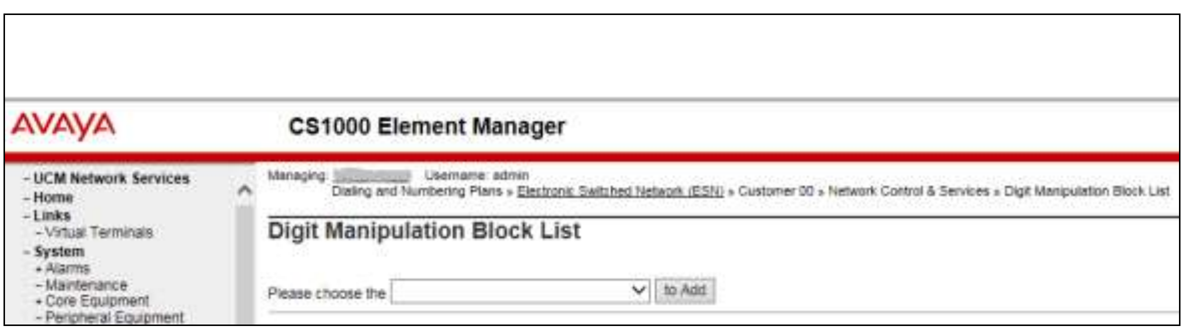
This section explains the digit manipulation block that is to be configured in the Communication Server 1000 dialing plan for its users to communicate with the Responder via the Session Manager. From the EM navigator pane, navigate to **Dialing and Numbering Plans → Electronic Switched Network** as shown below.



Click on **Digit Manipulation Block (DGT)** option as shown below.



Screen below shows the **Digit Manipulation Block List** page where users can add a digit manipulation block index by selecting an available one from the drop down menu. During compliance testing **Digit Manipulation Block Index -- 1** was used.

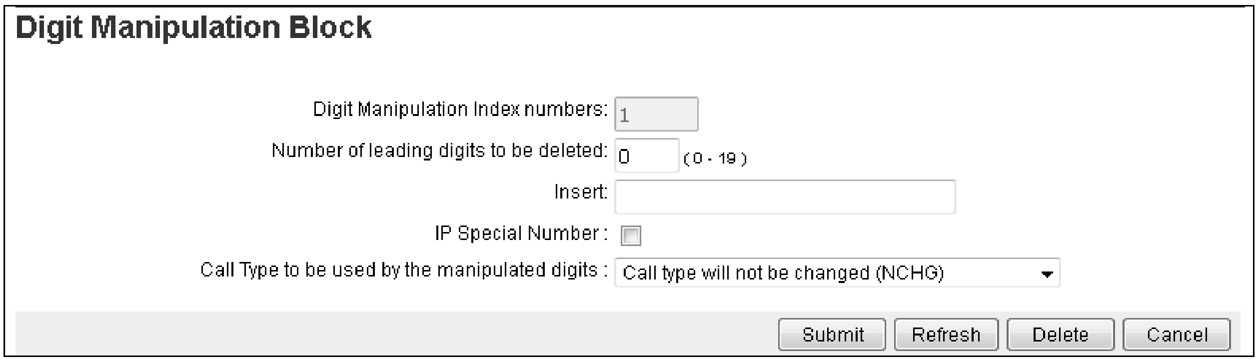


Screen below show the values configured for the digit manipulation block 1 added during compliance testing.

- **Number of leading digits to be deleted**: Enter *0*.
- **Insert**: Keep this value blank.

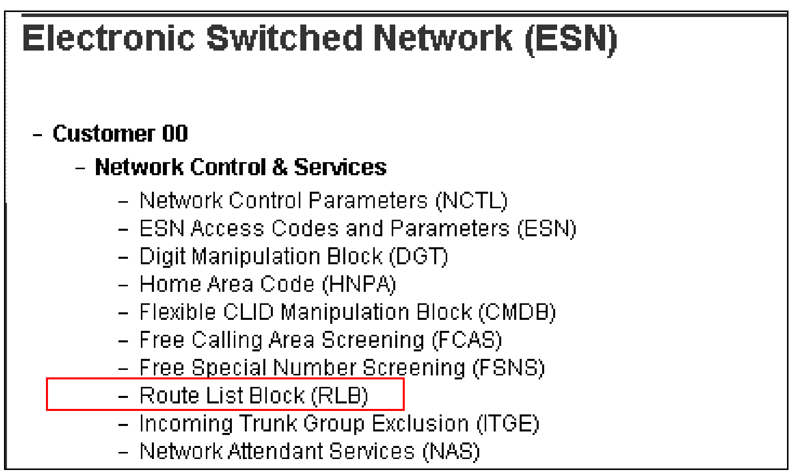
Retain default values for other fields.

Click on **Submit** to complete the configuration.

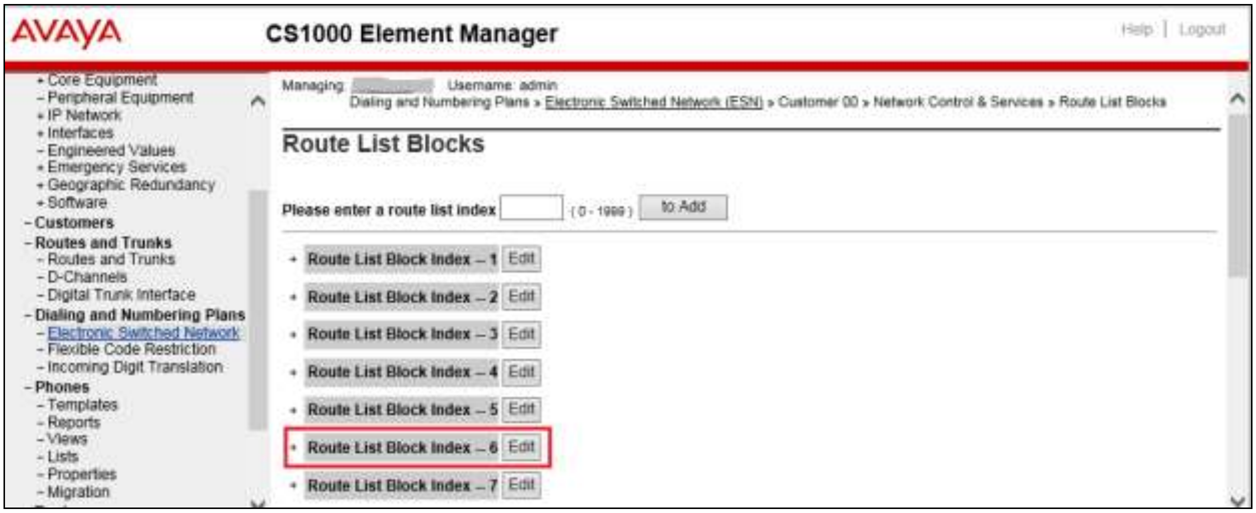


5.6. Configuring Route List Block

This section explains the route list block that is to be configured in the Communication Server 1000 dialing plan for its users to communicate with the Responder via Session Manager. From the EM navigator pane, navigate to **Dialing and Numbering Plans → Electronic Switched Network** as shown in **Section 5.5**. Click on **Route List Block (RLB)** option as shown below.



To add a route list index, enter a valid number in the **Please enter a route list index** box and click on **to Add** button as shown in the screen below. During compliance testing a route list block index of **6** was added.

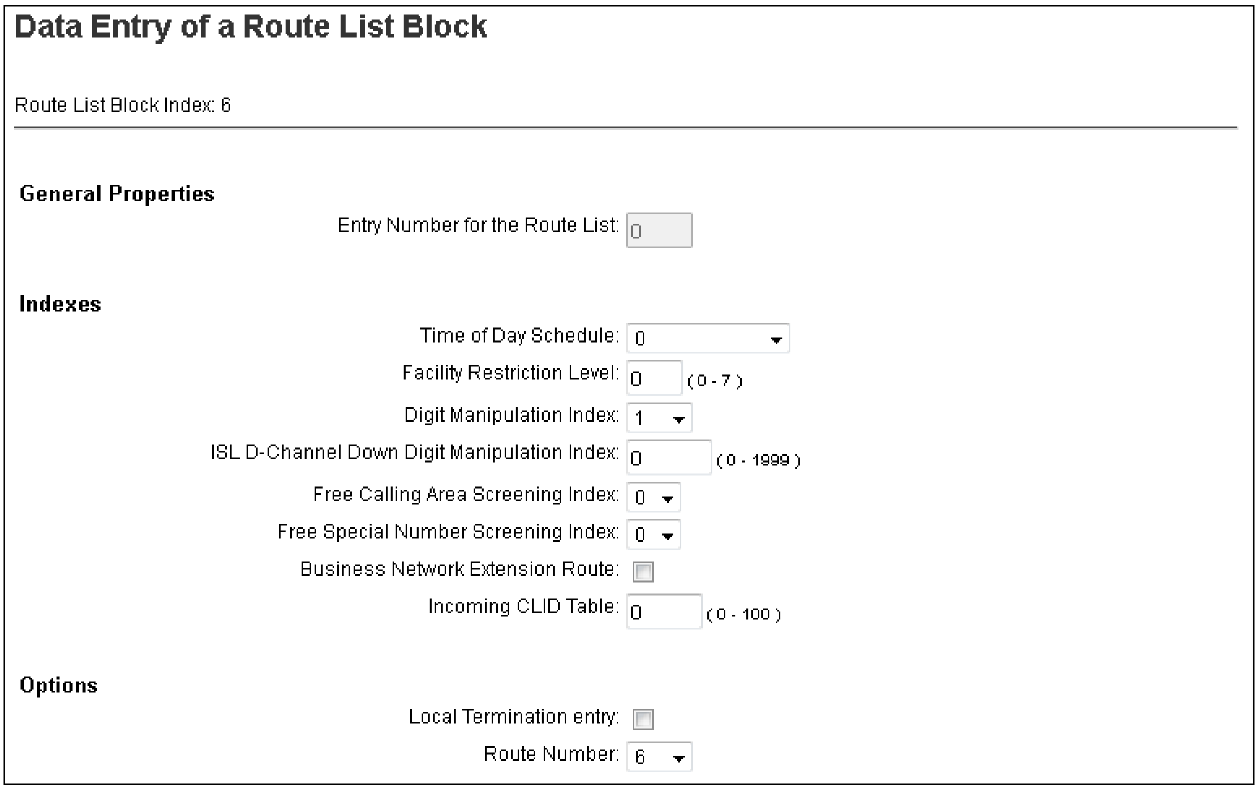


Screen below show the values configured for the route list index block 6 added during compliance testing.

- **Digit Manipulation Index**: Select *1* from the drop down menu. This was configured in **Section 5.5**.
- **Route Number**: Select *6* from the drop down menu. This was configured in **Section 5.4**.

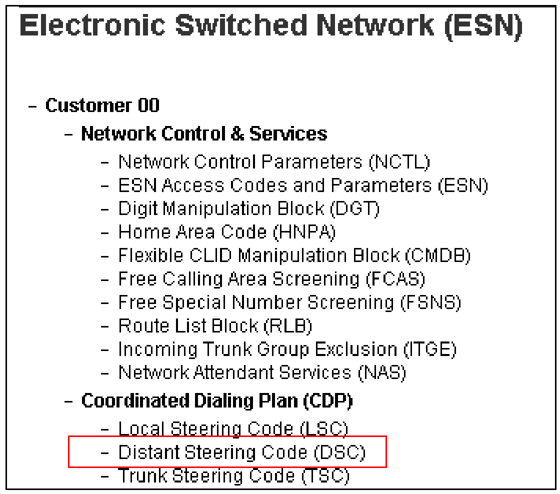
Retain default values for other fields.

Click on **Submit** to complete the configuration.

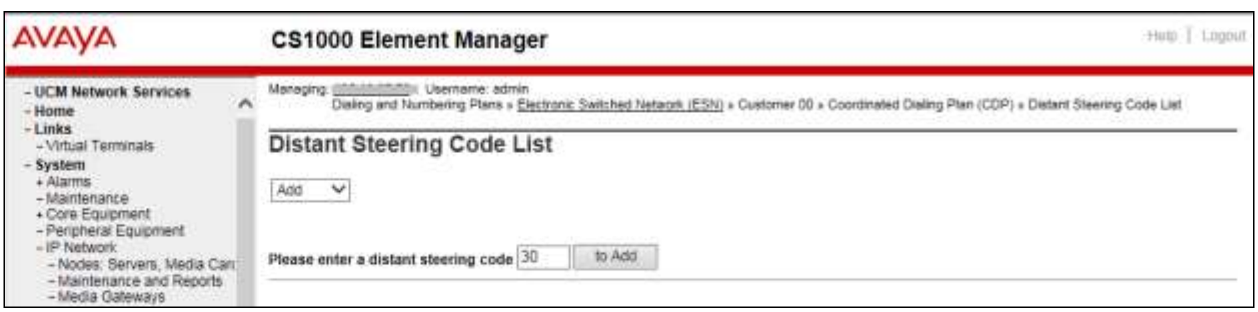


5.7. Configuring Distant Steering Code

This section explains the distant steering code that is to be configured in the Communication Server 1000 dialing plan for its users to communicate with the Responder via Session Manager. From the EM navigator pane, navigate to **Dialing and Numbering Plans → Electronic Switched Network** as shown in **Section 5.5**. Click on **Distant Steering Code (DSC)** option as shown below.



To add a distant steering code, select **Add** from the drop down menu and enter an available distant steering code in the **Please enter a distant steering code** box and click on **to Add** button to finish adding one as shown in the screen below. During compliance testing a code of **30** was added since the number assigned to reach Operator Assistant was 30xxx.

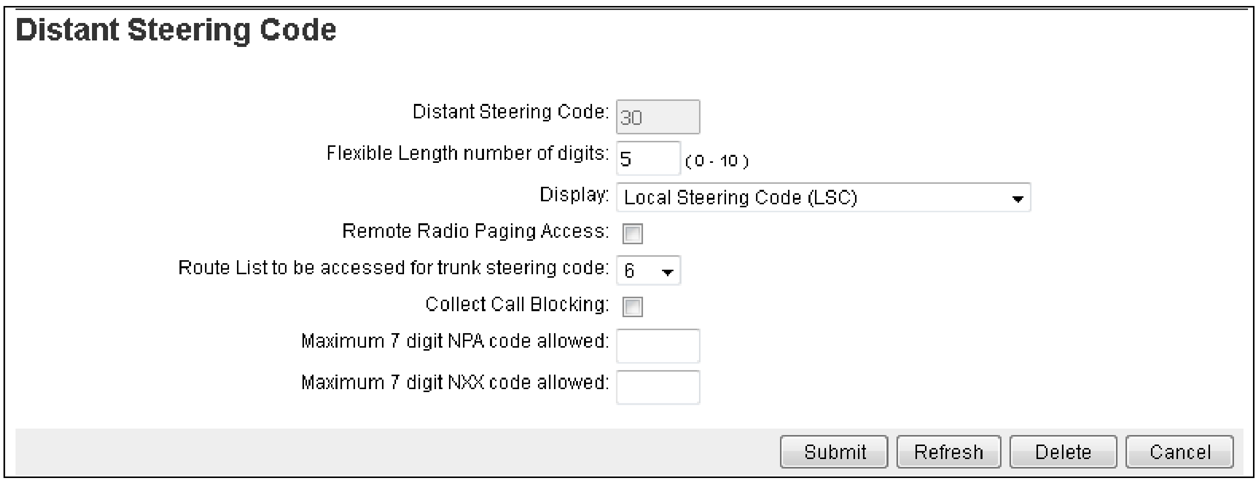


Screen below show the values configured for the distant steering code of 30 added during compliance testing.
Enter the values as shown in screen below.

- **Flexible Length number of digits**: *5*; since 30xxx the number to dial Operator Assistant is a 5 digit number.
- **Route List to be accessed for trunk steering code**: Select *6* from the drop down menu. This was configured in **Section 5.6**.

Retain default values for other fields.

Click on **Submit** to complete the configuration.



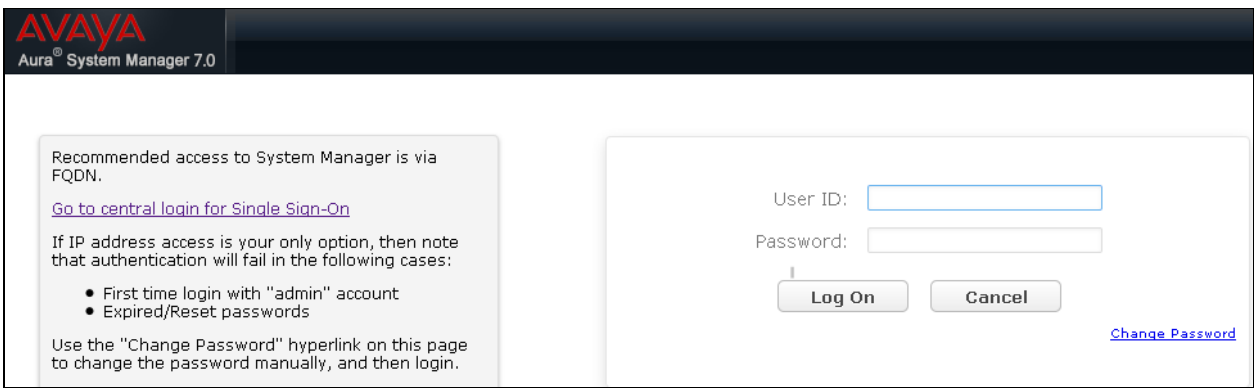
6. Configure Avaya Aura® Session Manager

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

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

6.1. Launch System Manager

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



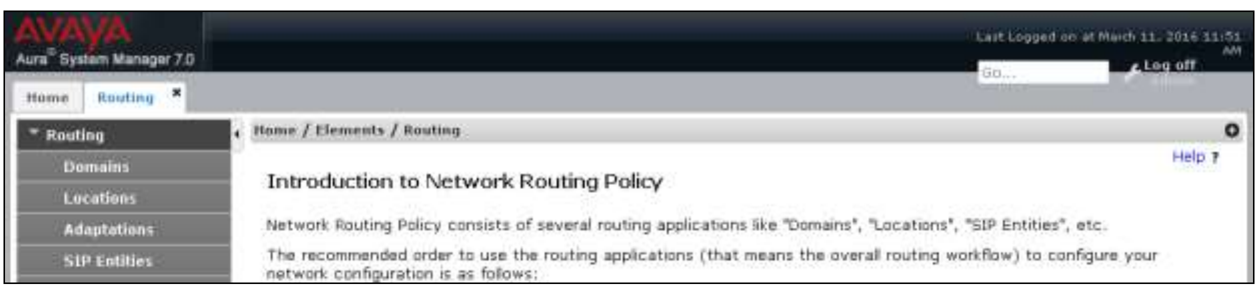
6.2. Adminsiter Domains

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

The **Domain Management** screen is displayed. In the **Name** field enter the domain name, select *sip* from the **Type** drop down menu and provide any optional **Notes**.

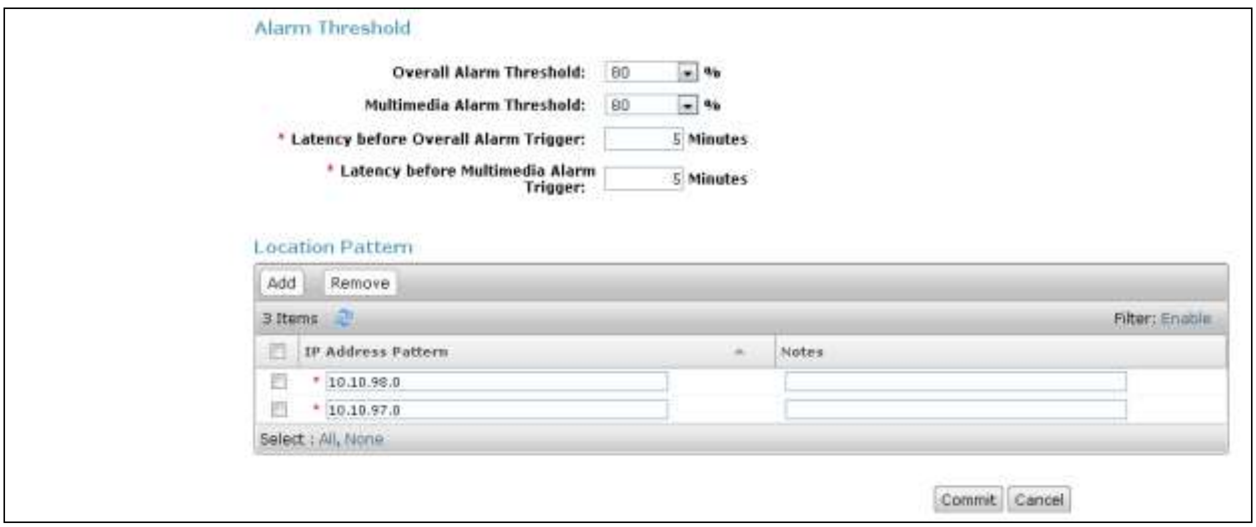
6.3. Administer Locations

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



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

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

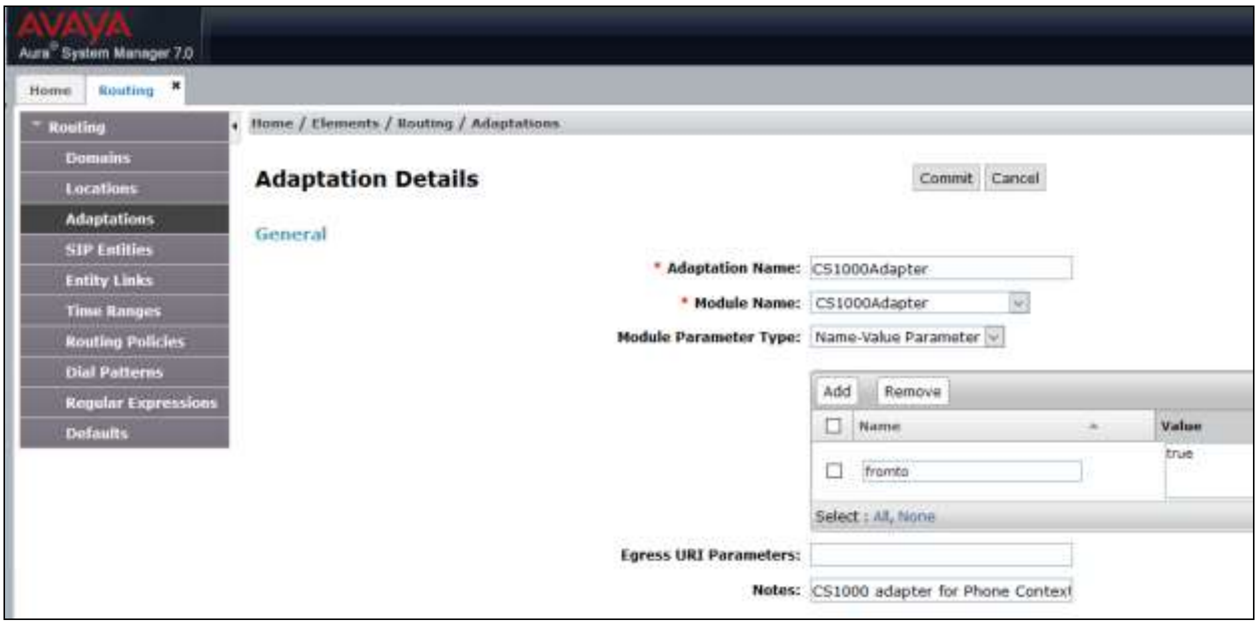


6.4. Administer Adaptations

Select **Routing → Adaptations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new Adaptation module

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

- **Adaptation Name:** A descriptive name.
- **Module Name:** Seelct "CS1000Adapter" from the drop down menu.
- **Module Parameter Type:** Select "Name-Value Parameter" from the drop down menu and then a rule, "fromto=true".
- **Notes:** Any desired notes.



.

6.5. Administer SIP Entities

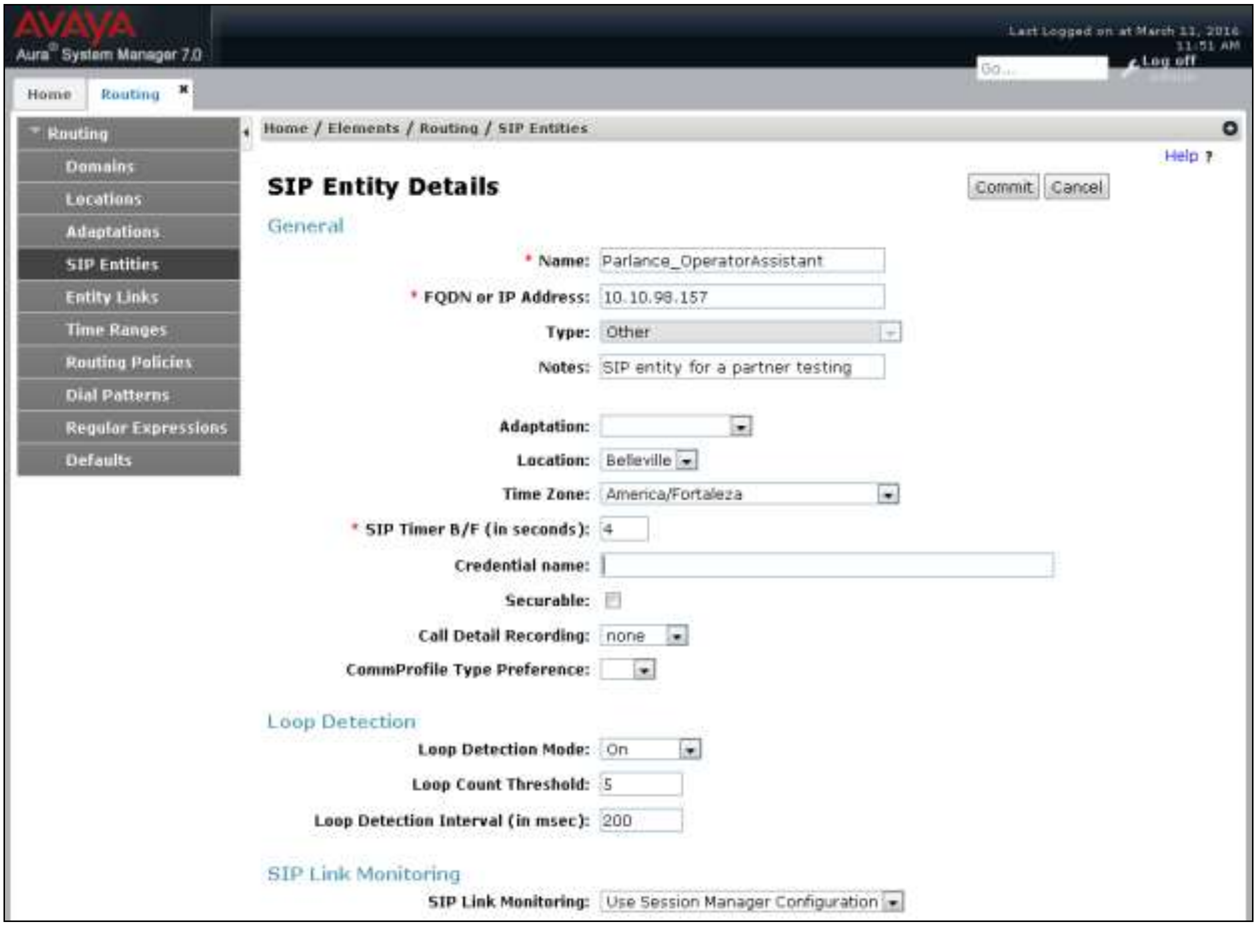
Add two new SIP entities, one for Operator Assistant and one for the new SIP trunks with Communication Server 1000.

6.5.1. SIP Entity for Operator Assistant

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

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

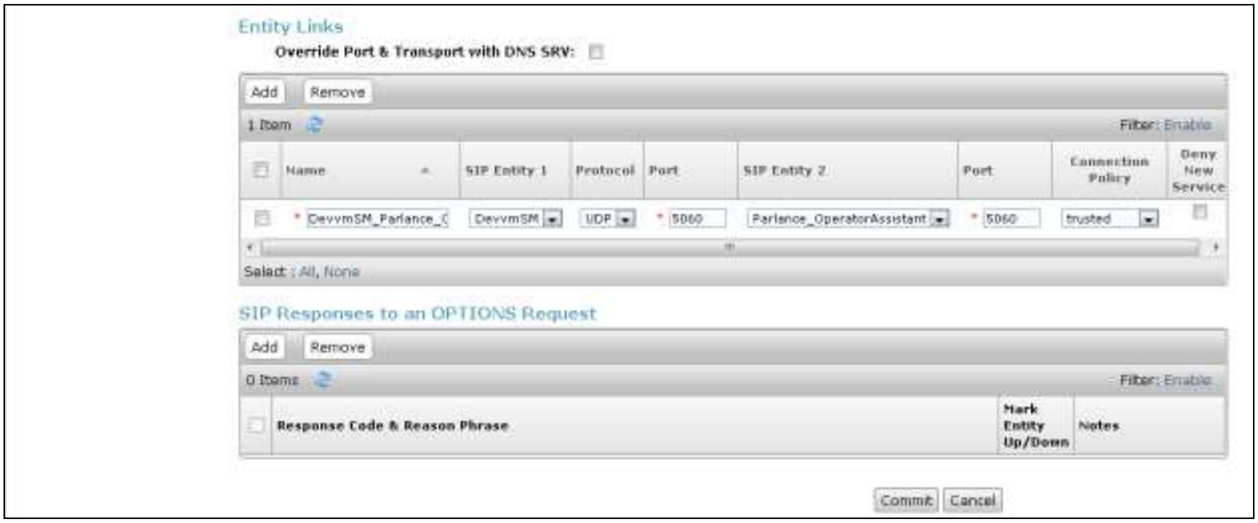
- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the Operator Assistant server.
- **Type:** "Other"
- **Notes:** Any desired notes.
- **Location:** Select the Operator Assistant location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.



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

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case "DevvmSM".
- **Protocol:** "UDP"
- **Port:** "5060"
- **SIP Entity 2:** The Operator Assistant entity name from this section.
- **Port:** "5060"
- **Connection Policy:** "trusted"

Note that Operator Assistant can only support UDP protocol.

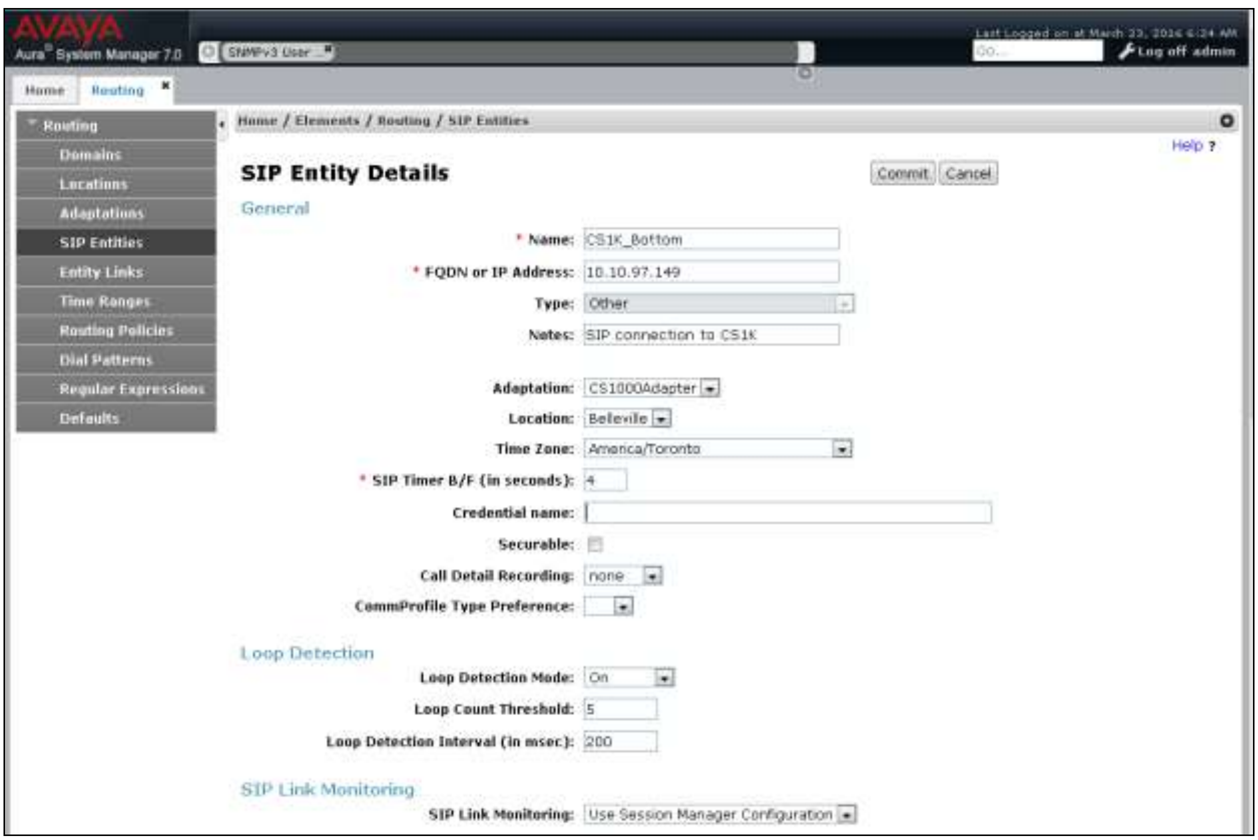


6.5.2. SIP Entity for Communication Server 1000

Select **Routing → SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Communication Server 1000. Note that this SIP entity is used for integration with Operator Assistant.

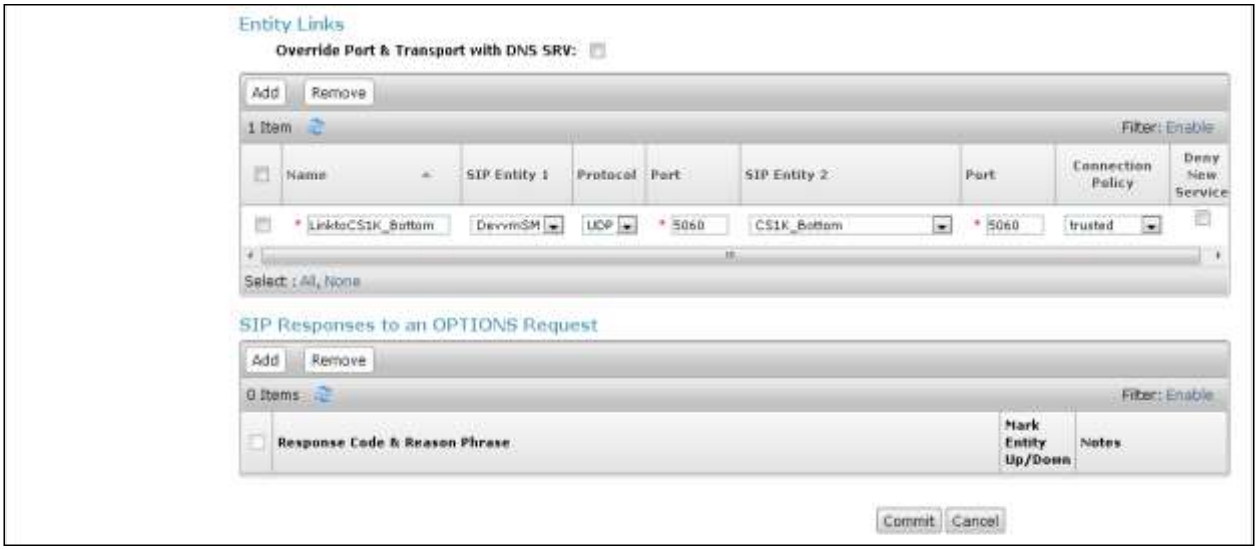
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the SIP Signaling Gateway interface.
- **Type:** "Other"
- **Notes:** Any desired notes.
- **Adaptation:** Select the applicable adaptation for Communication Server 1000 if any. During compliance testing "CS1000Adapter" was used to manipulate phone-context in SIP messages which was configured in **Section 6.4**.
- **Location:** Select the applicable location for Communication Server 1000.
- **Time Zone:** Select the applicable time zone.



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

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case "DevvmSM".
- **Protocol:** The signaling transport method from **Section 5.2**.
- **Port:** The signaling listen port number from **Section** Error! Reference source not found.**2**.
- **SIP Entity 2:** The Communication Server 1000 entity name from this section.
- **Port:** The signaling group listen port number from **Section** Error! Reference source not found.**2**.
- **Connection Policy:** "trusted"



6.6. Administer Routing Policies

Add two new routing policies, one for Operator Assistant and one for the new SIP trunks with Communication Server 1000.

6.6.1. Routing Policy for Operator Assistant

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

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

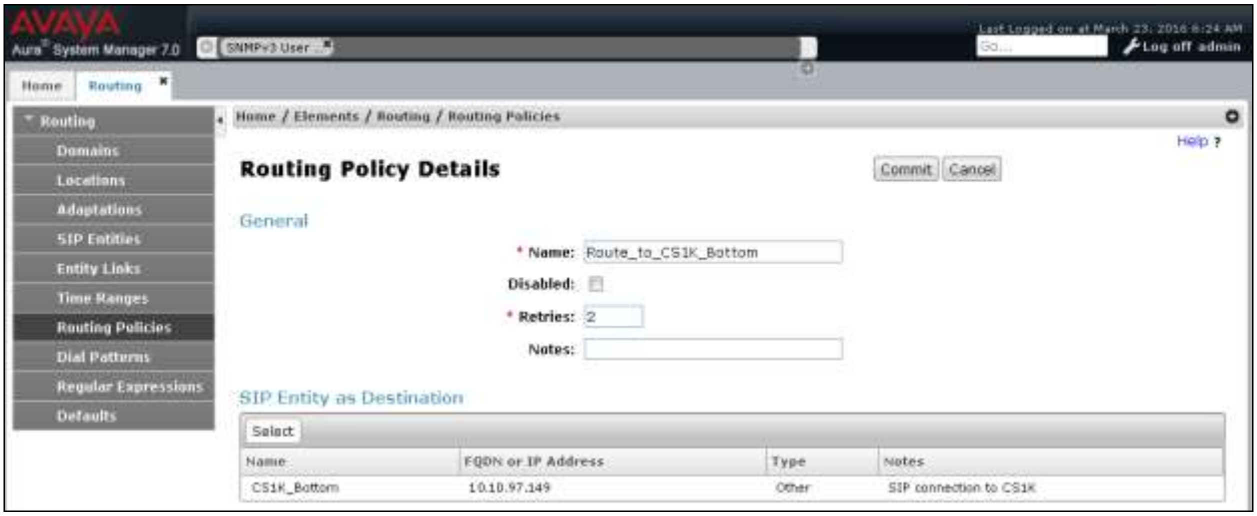
In the **SIP Entity as Destination** sub-section, click **Select** and select the Operator Assistant entity name from **Section 6.5.1**. The screen below shows the result of the selection.

6.6.2. Routing Policy for Communication Server 1000

Select **Routing → Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for Communication Server 1000.

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

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communication Server 1000 entity name from **Section 6.5.2**. The screen below shows the result of the selection.



6.7. Administer Dial Patterns

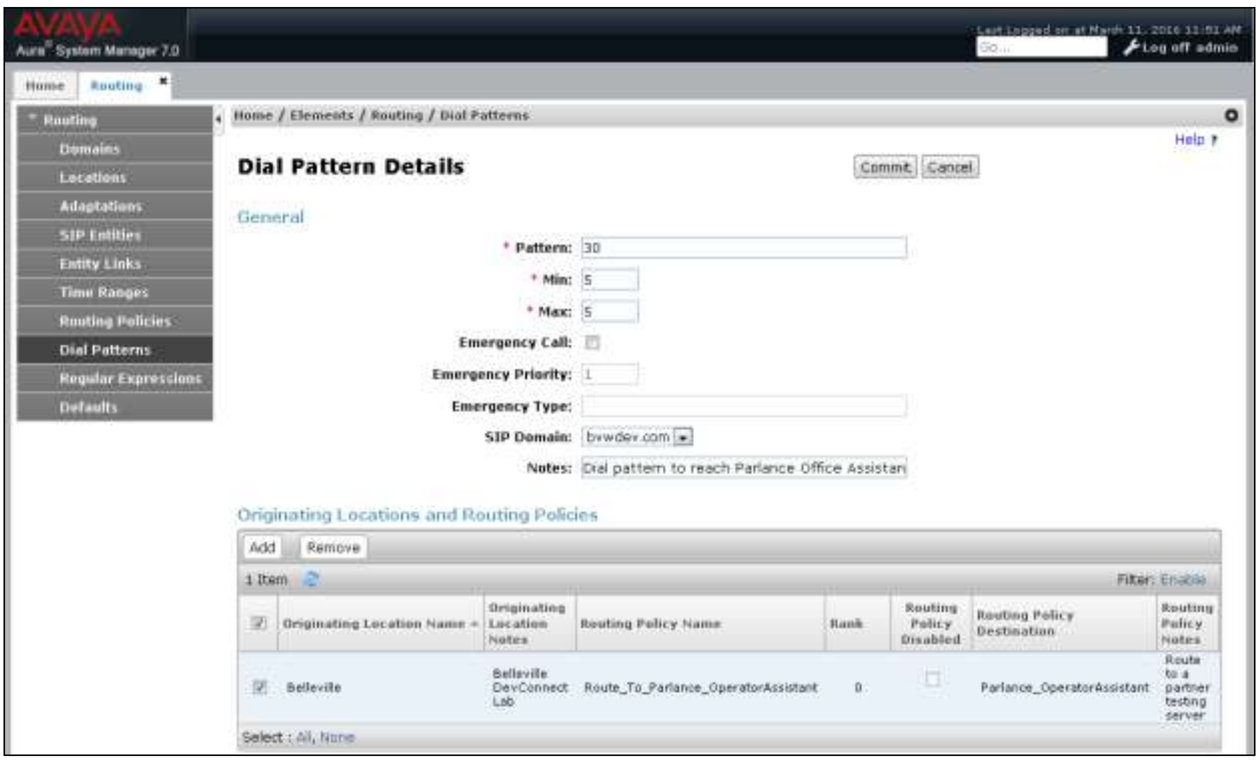
Add a new dial pattern for Operator Assistant, and update existing dial patterns for Communication Server 1000.

6.7.1. Dial Pattern for Operator Assistant

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

- **Pattern:** A dial pattern to match, in this case "30".
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The signaling domain name from **Section 5.2**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Operator Assistant. In the compliance testing, the entry allowed for call originations from all Communication Server 1000 endpoints in locations "Belleville". The Operator Assistant routing policy from **Section 6.6.1** was selected as shown below.

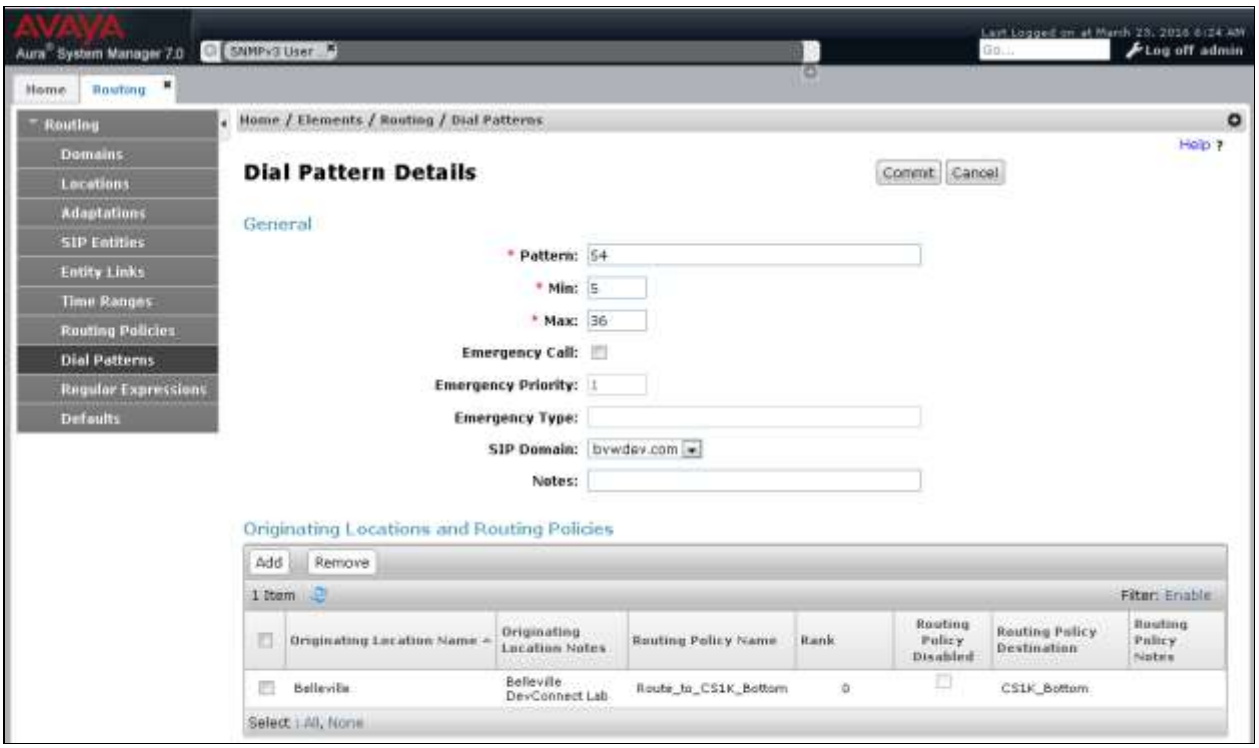


6.7.2. Dial Pattern for Communication Server 1000

Select **Routing → Dial Patterns** from the left pane, and click on the first existing dial pattern for Communication Server 1000 in the subsequent screen, in this case dial pattern "54" (not shown). The **Dial Pattern Details** screen is displayed.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy as necessary for calls from Operator Assistant. In the compliance testing, the new policy allowed for call origination from the Operator Assistant location from **Section 6.2**, and the Communication Server 1000 routing policy from **Section 6.6.2** was selected as shown below. Retain the default values in the remaining fields.

Follow the procedures in this section to make similar changes to the applicable Communication Server 1000 dial pattern to reach the PSTN. In the compliance testing, Operator Assistant will add the prefix "9" for outbound calls to the PSTN, and therefore the existing dial pattern for "9" was also changed (not shown below).



7. Configure Parlance Operator Assistant

The Parlance Operator Assistant will be provisioned completely by Parlance engineers based on site requirements and therefore no configuration details will be provided in these application notes.

To obtain information on Operator Assistant configuration, refer to **Section 2.3**.

8. Verification Steps

This section provides tests that can be performed to verify proper configuration of Communication Server 1000 and Session Manager.

8.1. Verify Avaya Aura® Communication Server 1000

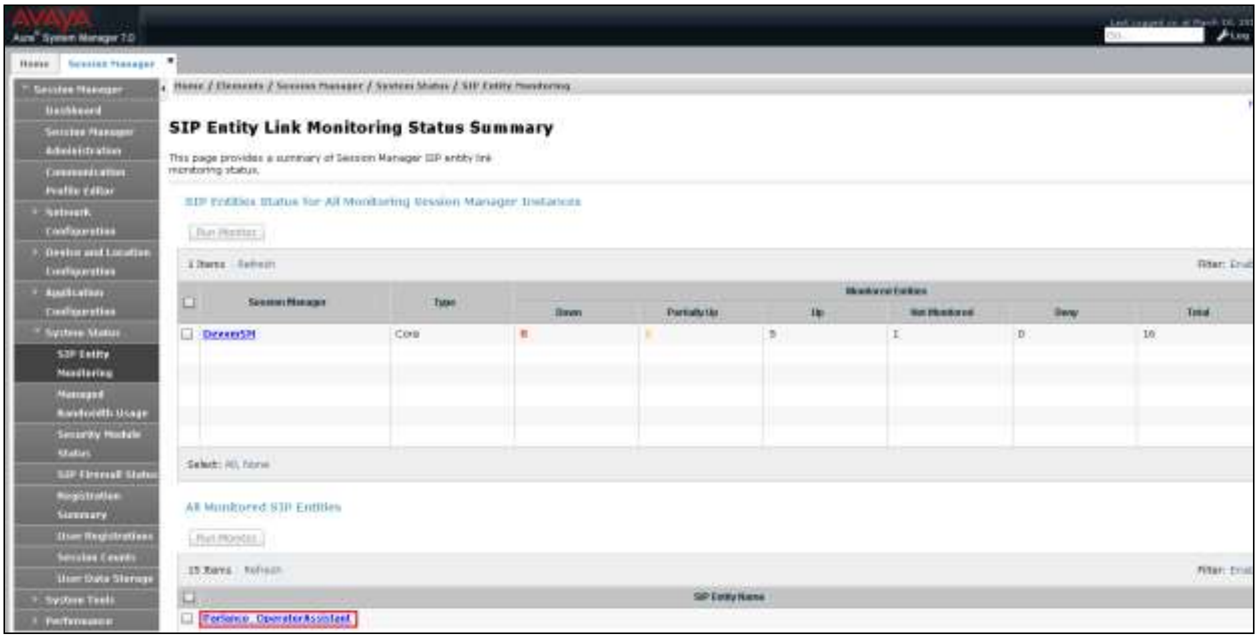
From the CLI interface, verify the status of the SIP trunks by using the "stat" command followed by the Terminal Number in LD 32. During compliance testing it is, "stat 100 0 3". Verify that all trunk units are in the "IDLE" state and the D-CH is in "EST ACTV" state as shown below.

```
>ld 32
NPR000
.stat 100 0 3
00 = UNIT 00 = IDLE            (ISL TRK)(TIE IP  IMM /IMM )
    D-CH 5  EST  ACTV

01 = UNIT 01 = IDLE            (ISL TRK)(TIE IP  IMM /IMM )
    D-CH 5  EST  ACTV

02 = UNIT 02 = IDLE            (ISL TRK)(TIE IP  IMM /IMM )
    D-CH 5  EST  ACTV

03 = UNIT 03 = IDLE            (ISL TRK)(TIE IP  IMM /IMM )
    D-CH 5  EST  ACTV
```

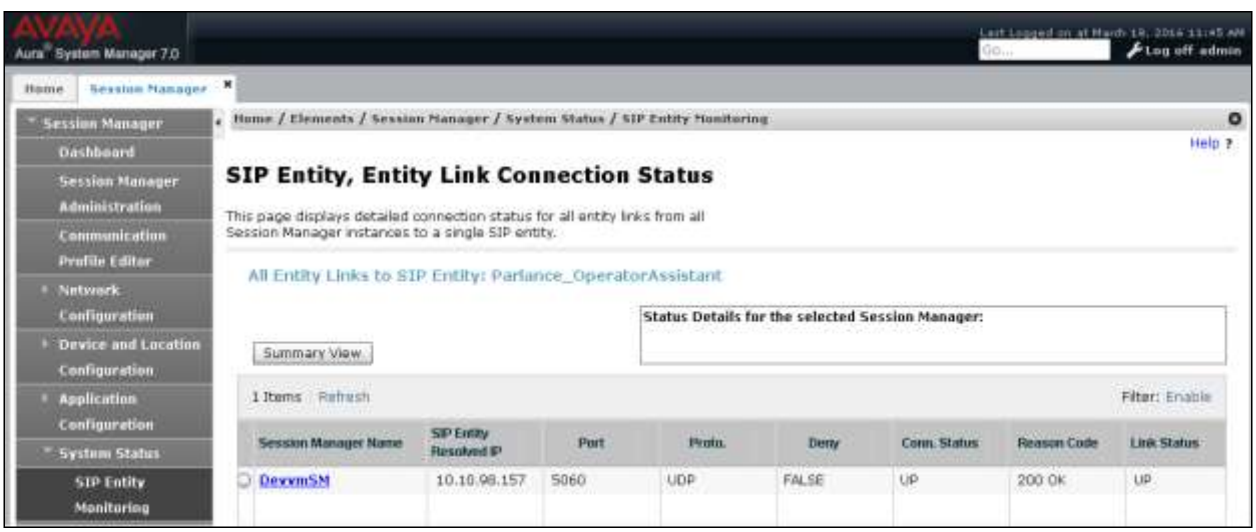
8.2. Verify Avaya Aura® Session Manager

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

Select **Session Manager → System Status → SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click the Operator Assistant entity name from **Section 6.5.1**.



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



9. Conclusion

These Application Notes describe the configuration steps required for Parlance Operator Assistant to successfully interoperate with Avaya Aura® Session Manager 7.0 and Avaya Communication Server 1000 7.0 using SIP trunks. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

10. Additional References

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

1. *Communication Server 1000E Installation and Commissioning*, Release 7.6, NN43041-310
2. *Element Manager System Reference – Administration - Avaya Communication Server 1000*, Release 7.6, NN43001-632.
3. *Avaya Communication Server 1000 Co-resident Call Server and Signaling Server Fundamentals* Release 7.6, *NN43001*-509.
4. *Avaya Communication Server 1000 Unified Communications Management Common Services Fundamentals* -, Release 7.6, NN43001-116.
5. *Avaya Communication Server 1000 - Software Input Output Reference — Administration* Release 7.6, NN43001-611.
6. *Avaya Communication Server 1000 - ISDN Primary Rate Interface Installation and Commissioning*, Release 7.6, NN43001-301.
7. *Implementing Avaya Aura® Session Manager* Document ID 03-603473.
8. *Administering Avaya Aura® Session Manager*, Doc ID 03-603324.
9. *Deploying Avaya Aura® System Manager*, Release 7.0.
10. *Administering Avaya Aura® System Manager for Release 7.0*, Release 7.0.

To obtain information on documents related to Parlance Operator Assistant, refer to **Section 2.3**.

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.