



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.

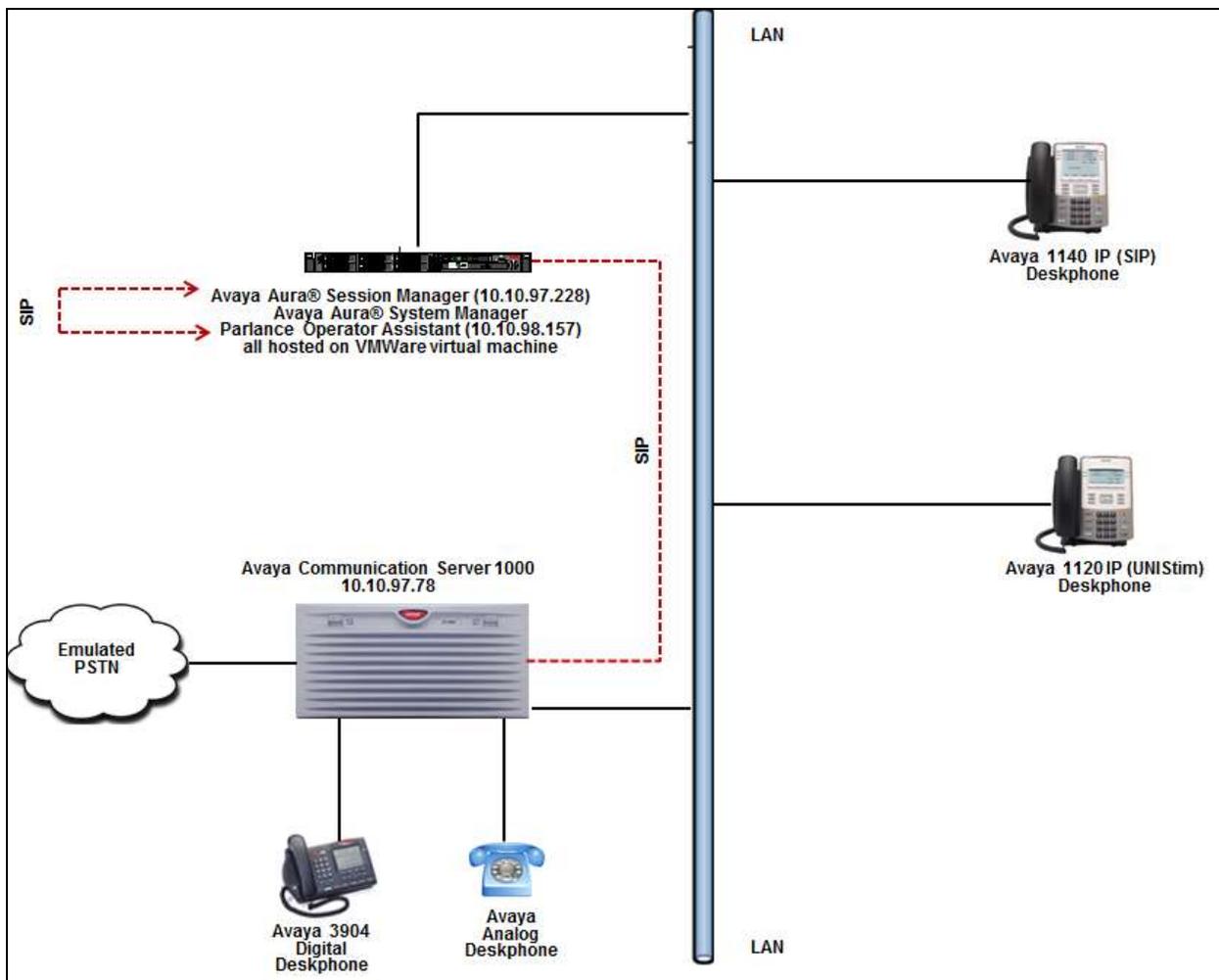


Figure 1: Compliance Testing Configuration

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: <ul style="list-style-type: none">• 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.

AVAYA
Aura® System Manager 7.0

Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

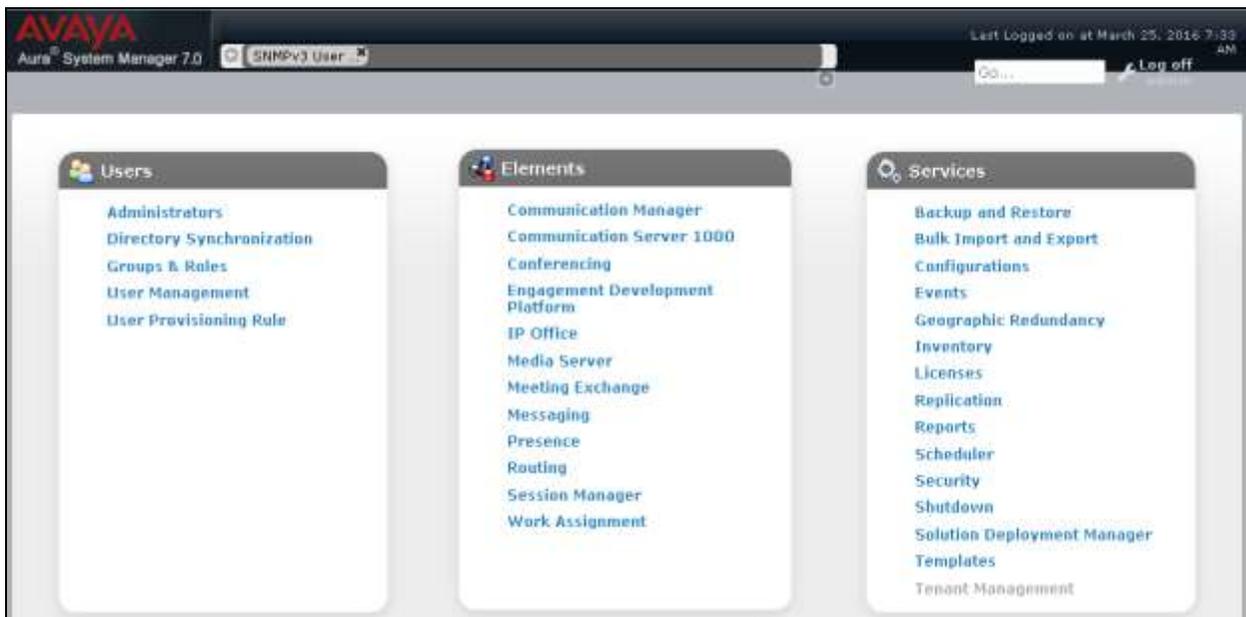
Use the "Change Password" hyperlink on this page to change the password manually, and then login.

User ID:

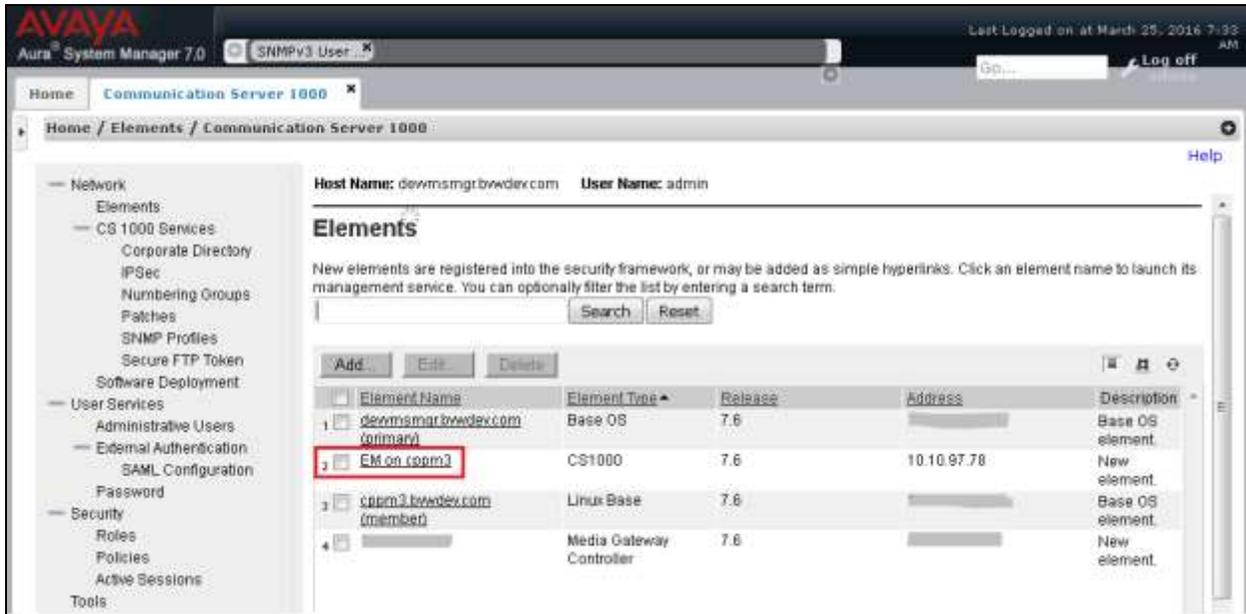
Password:

[Change Password](#)

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



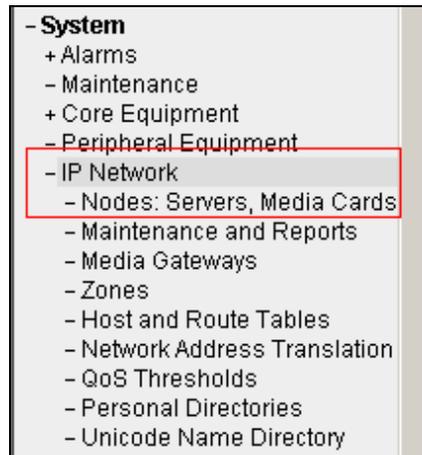
From the **Elements** page of System Manager as shown in screen below, click on the Element **EM on cppm3**. This is the element which is configured to access the Element Manager (EM) for the Communication Server 1000 Call Server.



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.

AVAYA CS1000 Element Manager

Managing: Username: admin
System > IP Network > IP Telephony Nodes

IP Telephony Nodes

Click the Node ID to view or edit its properties.

Buttons: Add... Import... Export... Delete... Print | Refresh

Node ID	Components	Enabled Applications	ELAN IP	Node/TLAN IPv4	Node/TLAN IPv6	Status
510	1	SIP Line, LTSP, PD, Gateway (SIPGW)				Synchronized

Show Nodes: Component servers and cards IPv6 address

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

The screenshot displays the AVAYA CS1000 Element Manager interface. The top navigation bar includes the AVAYA logo, the title "CS1000 Element Manager", and "Help | Logout" links. Below the navigation bar, the user is logged in as "admin". The breadcrumb trail shows the path: "System » IP Network » IP Telephony Nodes » Node Details".

The main content area is titled "Node Details (ID: 510 - SIP Line, LTPS, PD, Gateway (SIPGw))". It contains several configuration sections:

- Node ID:** 510 (with a note: *(0-9999))
- Call server IP address:** [Redacted]
- TLAN address type:** IPv4 only, IPv4 and IPv6
- Embedded LAN (ELAN):**
 - Gateway IP address:** [Redacted]
 - Subnet mask:** [Redacted]
- Telephony LAN (TLAN):**
 - Node IPv4 address:** [Redacted]
 - Subnet mask:** [Redacted]
 - Node IPv6 address:** [Redacted]
- IP Telephony Node Properties:**
 - Voice Gateway (VGW) and Codecs
 - Quality of Service (QoS)
 - LAN
 - SNTP
 - Numbering Zones
 - MCDN Alternative Routing Treatment (MALT) Causes
- Applications (click to edit configuration):**
 - SIP Line
 - Terminal Proxy Server (TPS)
 - Gateway (SIPGw)
 - Personal Directories (PD)
 - Presence Publisher
 - IP Media Services

The "Gateway (SIPGw)" application is highlighted with a red box in the "Applications" section.

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.

The screenshot displays the AVAYA CS1000 Element Manager interface. The main window is titled "Node ID: 510 - Virtual Trunk Gateway Configuration Details". The left sidebar shows a navigation tree with categories like "UCM Network Services", "System", "Interfaces", "Customers", "Routes and Trunks", and "Dialing and Numbering Plans". The main content area is divided into "General" and "Virtual Trunk Network Health Monitor" sections. In the "General" section, the "Vtrk gateway application" is set to "SIP Gateway (SIPGw)", "SIP domain name" is "bvwdev.com", "Local SIP port" is "5060", "Gateway endpoint name" is "cppm3", and "Application node ID" is "510". The "Enable gateway service on this node" checkbox is checked. The "Virtual Trunk Network Health Monitor" section has the "Monitor IP addresses" checkbox unchecked. At the bottom, there are "Save" and "Cancel" buttons.

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.

The screenshot displays the AVAYA CS1000 Element Manager interface. The main title is "CS1000 Element Manager". The breadcrumb navigation shows: "System > IP Network > IP Telephony Nodes > Node Details > Virtual Trunk Gateway Configuration". The page title is "Node ID: 510 - Virtual Trunk Gateway Configuration Details". The interface is divided into several sections: "General", "SIP Gateway Settings", and "SIP Gateway Services". The "Proxy Or Redirect Server" section is highlighted with a red box. It contains the following fields and options:

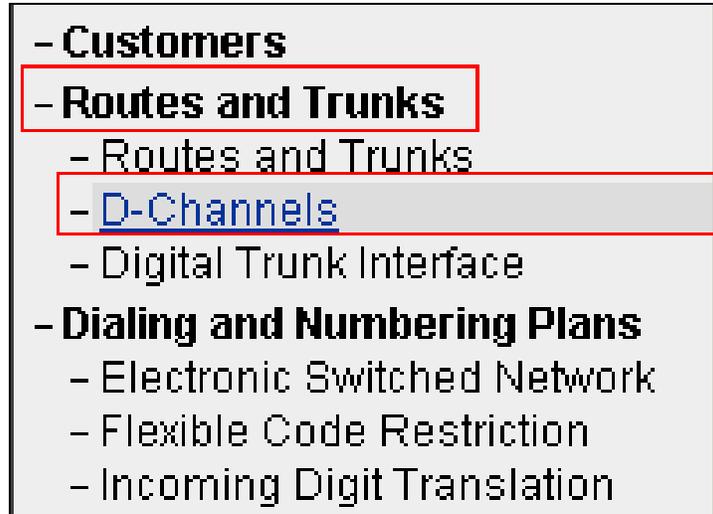
- Primary TLAN IP address:** 10.10.97.228 (with a note: "The IP address can have either IPv4 or IPv6 format based on the value of 'TLAN address type'")
- Port:** 5060 (with a range of 1 - 65535)
- Transport protocol:** UDP (selected from a dropdown menu)
- Options:** Support registration, Primary CDS proxy

Below the highlighted section, there are fields for "Secondary TLAN IP address" (0.0.0.0) and "Port" (5060), along with a "Transport protocol" dropdown menu (UDP). At the bottom, there are "Save" and "Cancel" buttons, and a note: "Note: Changes made on this page will NOT be transmitted until the Node is also saved."

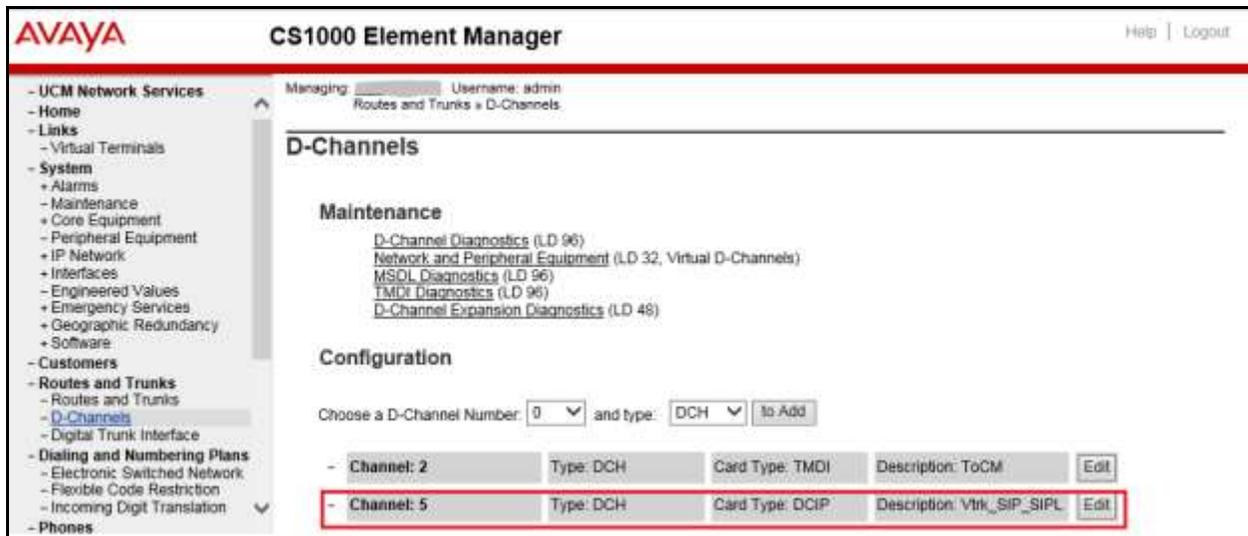
Save and transmit (not shown) these Node properties to complete the SIPGw configuration.

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.
- **Interface type for D-channel:** Select *Meridian Meridian1 (SL1)* from the drop down menu.
- **Meridian 1 node type:** Select *Slave 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.

The screenshot shows the AVAYA CS1000 Element Manager interface. The left sidebar contains a navigation menu with categories like UCM Network Services, System, Interfaces, Customers, and Routes and Trunks. The main content area is titled 'D-Channels 5 Property Configuration' and shows the 'Basic Configuration' section. The configuration fields are as follows:

Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type:	DCIP
Designator:	Vtrk_SIP_SIPL
Recovery to Primary:	<input type="checkbox"/>
PRI loop number for Backup D-channel:	
User:	Integrated Services Signaling Link Dedicated (ISLD)
Interface type for D-channel:	Meridian Meridian1 (SL1)
Country:	ETS 300 =102 basic protocol (ETS)
D-Channel PRI loop number:	
Primary Rate Interface:	<input type="text"/> <input type="button" value="more PRI"/>
Secondary PRI2 loops:	<input type="text"/>
Meridian 1 node type:	Slave to the controller (USR)
Release ID of the switch at the far end:	7
Central Office switch type:	100% compatible with Bellcore standard (STD)

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.

- Basic options (BSCOPT)

Primary D-channel for a backup DCH: Range: 0 - 254

- PINX customer number:

- Progress signal:

- Calling Line Identification :

- Output request Buffers: 32

- D-channel transmission Rate: 56 kb/s when LCMT is AMI (56K)

- Channel Negotiation option: No alternative acceptable, exclusive. (1)

- Remote Capabilities:

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.

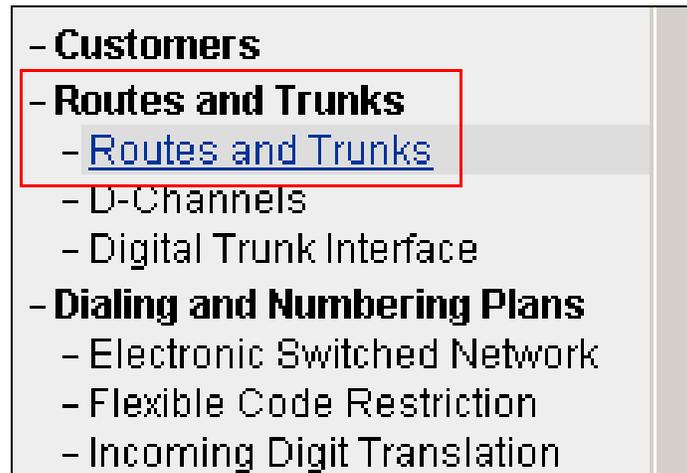
- Remote Capabilities Configuration

Input Description	Input Value
Basic rate interface (BRI)	<input type="checkbox"/>
Call completion on busy using integer value (CCBI)	<input type="checkbox"/>
Call completion on busy using object identifier (CCBO)	<input type="checkbox"/>
Call completion on busy for QSIG and EuroISDN BRI (CCBS)	<input type="checkbox"/>
Call completion on no response using integer value (CCNI)	<input type="checkbox"/>
Call completion on no response using object identifier (CCNO)	<input type="checkbox"/>
Call completion to no reply for QSIG and EuroISDN BRI (CCNR)	<input type="checkbox"/>
Network call park (CPK)	<input type="checkbox"/>
Connected line identification presentation (COLP)	<input type="checkbox"/>
Call transfer integer (CTI)	<input type="checkbox"/>
Call transfer object (CTO)	<input type="checkbox"/>
Diversion info. is sent using integer value (DV1I)	<input type="checkbox"/>
Diversion info. is sent using object identifier (DV1O)	<input type="checkbox"/>
Rerouting requests processed using integer value (DV2I)	<input type="checkbox"/>
Rerouting requests processed using object identifier (DV2O)	<input type="checkbox"/>
Diversion info. sent. rerouting requests processed (DV3I)	<input type="checkbox"/>
EuroISDN - div. info sent. rerouting req. processed (DV3O)	<input type="checkbox"/>
Call transfer notification and invocation to EuroISDN (ECTO)	<input type="checkbox"/>
Malicious call identification (MCID)	<input type="checkbox"/>
MCDN QSIG conversion (MQC)	<input type="checkbox"/>
Remote D-channel is on a MSDL card (MSL)	<input type="checkbox"/>
Message waiting interworking with DMS-100 (MWI)	<input type="checkbox"/>
Network access data (NAC)	<input type="checkbox"/>
Network call trace supported (NCT)	<input type="checkbox"/>
Network name display method 1 (ND1)	<input type="checkbox"/>
Network name display method 2 (ND2)	<input checked="" type="checkbox"/>

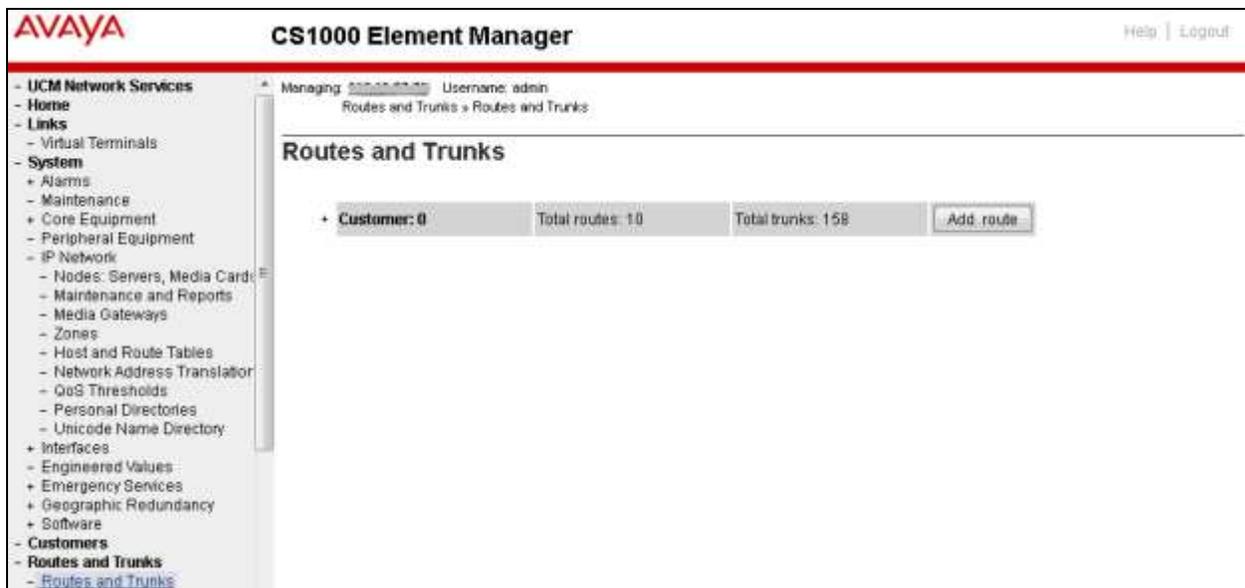
Now click on the **Submit** button (not shown) to complete the D-channel configuration.

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.

Customer 0, Route 6 Property Configuration

- Basic Configuration

Route data block (RDB) (TYPE):

Customer number (CUST):

Route number (ROUT):

Designator field for trunk (DES):

Trunk type (TKTP):

Incoming and outgoing trunk (ICOG):

Access code for the trunk route (ACOD): *

Trunk type M911P (M911P):

The route is for a virtual trunk route (VTRK):

- Zone for codec selection and bandwidth management (ZONE): (0 - 8000)

- Node ID of signaling server of this route (NODE): (0 - 9999)

- Protocol ID for the route (PCID):

- 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):

- D channel number (DCH): (0 - 254)

- Interface type for route (IFC):

- Private network identifier (PNI): (0 - 32700)

- Call type for outgoing direct dialed TIE route (CTYP):

- 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):

- Mobile extension timer (MBXT): (0 - 8000 milliseconds)

Calling number dialing plan (CNDP):

- Network Options

Electronic switched network pad control (ESN):

Signaling arrangement (SIGO):

Route class (RCLS):

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.

Customer 0, Route 6, Trunk 1 Property Configuration

- Basic Configuration

Auto increment member number:

Trunk data block:

Terminal number:

Designator field for trunk:

Extended trunk:

Member number: *

Level 3 Signaling:

Card density:

Start arrangement Incoming:

Start arrangement Outgoing:

Trunk group access restriction:

Channel ID for this trunk:

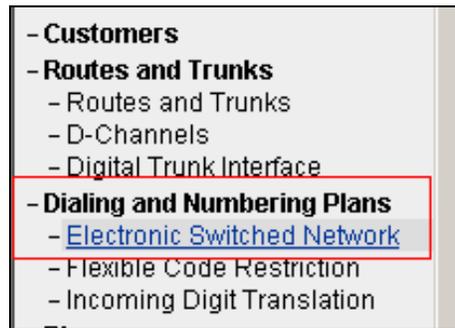
Class of Service:

- Class of Service

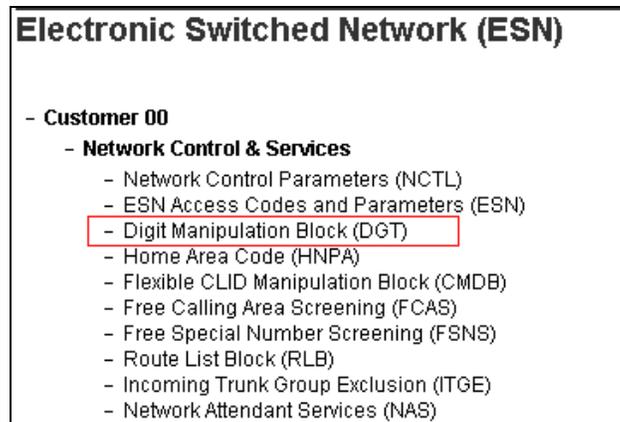
Input Description	Input Value
- Priority:	Low Priority (LPR) ▼
- Restriction level:	Unrestricted (UNR) ▼

5.5. Configuring Digit Manipulation Block

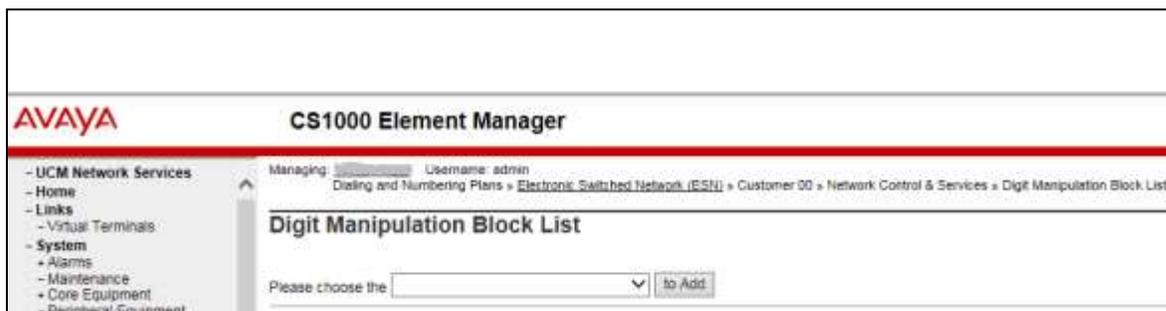
This section explains the digit manipulation block that is to be configured in the Communication Server 1000 dialing plan for its users to communicate with the Responder via the Session Manager. From the EM navigator pane, navigate to **Dialing and Numbering Plans** → **Electronic Switched Network**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

Digit Manipulation Block

Digit Manipulation Index numbers:

Number of leading digits to be deleted: (0 - 19)

Insert:

IP Special Number :

Call Type to be used by the manipulated digits :

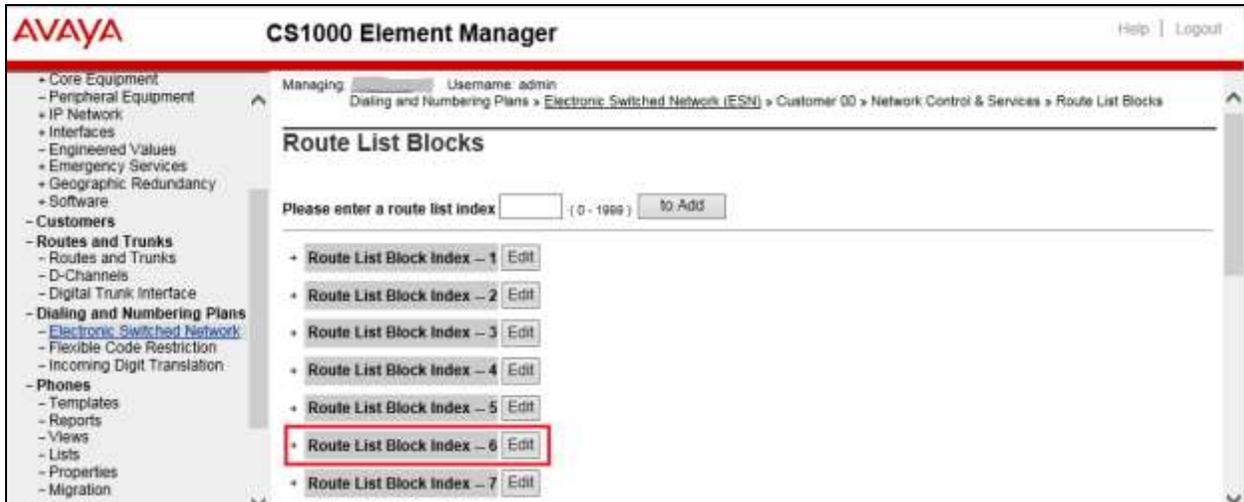
5.6. Configuring Route List Block

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.

Electronic Switched Network (ESN)

- Customer 00
 - Network Control & Services
 - Network Control Parameters (NCTL)
 - ESN Access Codes and Parameters (ESN)
 - Digit Manipulation Block (DGT)
 - Home Area Code (HNPA)
 - Flexible CLID Manipulation Block (CMDB)
 - Free Calling Area Screening (FCAS)
 - Free Special Number Screening (FSNS)
 - Route List Block (RLB)
 - Incoming Trunk Group Exclusion (ITGE)
 - Network Attendant Services (NAS)

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.

Data Entry of a Route List Block

Route List Block Index: 6

General Properties

Entry Number for the Route List:

Indexes

Time of Day Schedule: ▼

Facility Restriction Level: (0 - 7)

Digit Manipulation Index: ▼

ISL D-Channel Down Digit Manipulation Index: (0 - 1999)

Free Calling Area Screening Index: ▼

Free Special Number Screening Index: ▼

Business Network Extension Route:

Incoming CLID Table: (0 - 100)

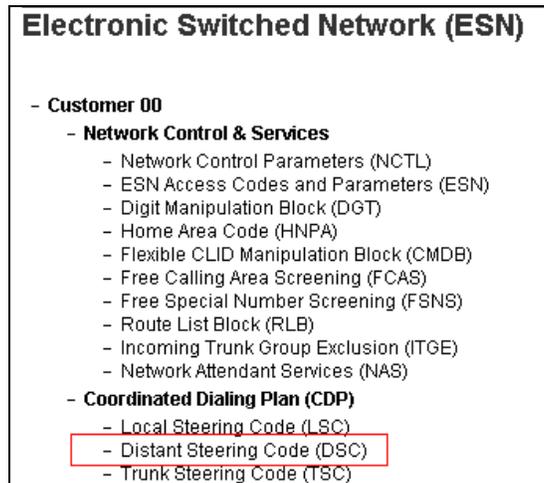
Options

Local Termination entry:

Route Number: ▼

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 **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.

Distant Steering Code

Distant Steering Code:

Flexible Length number of digits: (0 - 10)

Display:

Remote Radio Paging Access:

Route List to be accessed for trunk steering code:

Collect Call Blocking:

Maximum 7 digit NPA code allowed:

Maximum 7 digit NXX code allowed:

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.

AVAYA
Aura® System Manager 7.0

Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

User ID:

Password:

[Change Password](#)

6.2. Administer 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.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with options like Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Location Details' and has a breadcrumb path of 'Home / Elements / Routing / Locations'. The 'General' sub-section is selected, displaying the following fields:

- Name:** Belleville
- Notes:** Belleville DevConnect Lab
- Dial Plan Transparency in Survivable Mode:** Enabled:
- Listed Directory Number:** [Empty field]
- Associated CM SIP Entity:** [Empty field]

Buttons for 'Commit' and 'Cancel' are visible in the top right corner of the form area.

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.

The screenshot displays two sections of the configuration interface:

Alarm Threshold

- Overall Alarm Threshold: 80 %
- Multimedia Alarm Threshold: 80 %
- * Latency before Overall Alarm Trigger: 5 Minutes
- * Latency before Multimedia Alarm Trigger: 5 Minutes

Location Pattern

Buttons: Add, Remove

3 Items Filter: Enable

IP Address Pattern	Notes
* 10.10.98.0	
* 10.10.97.0	

Select: All, None

Buttons: Commit, Cancel

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:** Select “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.

The screenshot shows the 'Adaptation Details' screen in Avaya Aura System Manager 7.0. The left navigation pane is expanded to 'Routing', and the main content area is titled 'Adaptation Details' with a 'General' tab. The form contains the following fields:

- Adaptation Name:** CS1000Adapter
- Module Name:** CS1000Adapter
- Module Parameter Type:** Name-value Parameter

Below these fields is a table for parameters:

Name	Value
fromto	true

At the bottom of the form, there is a **Notes** field with the text: CS1000 adapter for Phone Context.

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.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The top navigation bar includes the Avaya logo, the text "Aura System Manager 7.0", and a "Last Logged on at March 11, 2016 11:51 AM" timestamp. A "Log off" button is visible in the top right. The main content area is titled "SIP Entity Details" and is divided into several sections:

- General:**
 - Name:** Parlance_OperatorAssistant
 - FQDN or IP Address:** 10.10.98.157
 - Type:** Other (selected from a dropdown menu)
 - Notes:** SIP entity for a partner testing
 - Adaptation:** (empty dropdown)
 - Location:** Belleville (selected from a dropdown menu)
 - Time Zone:** America/Fortaleza (selected from a dropdown menu)
 - SIP Timer B/F (in seconds):** 4
 - Credential name:** (empty text field)
 - Securable:**
 - Call Detail Recording:** none (selected from a dropdown menu)
 - CommProfile Type Preference:** (empty dropdown)
- Loop Detection:**
 - Loop Detection Mode:** On (selected from a dropdown menu)
 - Loop Count Threshold:** 5
 - Loop Detection Interval (in msec):** 200
- SIP Link Monitoring:**
 - SIP Link Monitoring:** Use Session Manager Configuration (selected from a dropdown menu)

Buttons for "Commit" and "Cancel" are located at the top right of the configuration area. A "Help ?" link is also present.

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.

The screenshot displays the 'Entity Links' configuration page. At the top, there is a checkbox for 'Override Port & Transport with DNS SRV'. Below this, there are 'Add' and 'Remove' buttons. The main area shows a table with one item. The table has columns for Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Connection Policy, and Deny New Service. The item in the table is: Name: DevvmSM_Parlance_C, SIP Entity 1: DevvmSM, Protocol: UDP, Port: 5060, SIP Entity 2: Parlance_OperatorAssistant, Port: 5060, Connection Policy: trusted. Below the table, there is a 'Select: All, None' dropdown. Underneath, there is a section for 'SIP Responses to an OPTIONS Request' with 'Add' and 'Remove' buttons and a 'Filter: Enable' option. The table below this section has columns for Response Code & Reason Phrase, Mark Entity Up/Down, and Notes. At the bottom right, there are 'Commit' and 'Cancel' buttons.

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.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left navigation pane is expanded to 'Routing', and 'SIP Entities' is selected. The main content area displays the 'SIP Entity Details' form for a new entity named 'CS1K_Bottom'. The form includes the following fields and values:

- Name:** CS1K_Bottom
- FQDN or IP Address:** 10.10.97.149
- Type:** Other
- Notes:** SIP connection to CS1K
- Adaptation:** CS1000Adapter
- Location:** Belleville
- Time Zone:** America/Toronto
- SIP Timer B/F (in seconds):** 4
- Credential name:** (empty)
- Securable:**
- Call Detail Recording:** none
- CommProfile Type Preference:** (empty)
- Loop Detection:**
 - Loop Detection Mode:** On
 - Loop Count Threshold:** 5
 - Loop Detection Interval (in msec):** 200
- SIP Link Monitoring:** Use Session Manager Configuration

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”

The screenshot displays the 'Entity Links' configuration window. At the top, there is a checkbox for 'Override Port & Transport with DNS SRV'. Below this, there are 'Add' and 'Remove' buttons. The main area shows a table with one item:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
* LinktoCS1K_Bottom	DevvmSM	UDP	* 5060	CS1K_Bottom	* 5060	trusted	<input type="checkbox"/>

Below the table, there is a 'Select: All, None' dropdown. Underneath is the 'SIP Responses to an OPTIONS Request' section, which is currently empty. It has 'Add' and 'Remove' buttons and a 'Filter: Enable' option. At the bottom of the window, there are 'Commit' and 'Cancel' buttons.

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.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The top navigation bar includes the Avaya logo, the text "Aura System Manager 7.0", and a "Last Logged on" timestamp of "at March 11, 2016 11:51 AM". A "GO..." search box and a "Log off" button are also present. The left sidebar shows a tree view with "Routing" selected, and its sub-items: Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies (highlighted), Dial Patterns, Regular Expressions, and Defaults. The main content area is titled "Routing Policy Details" and includes "Commit" and "Cancel" buttons. The "General" section has the following fields: "Name" (Route_To_Parlan...), "Disabled" (checkbox), "Retries" (0), and "Notes" (Route to a partner testing server). The "SIP Entity as Destination" section features a "Select" button and a table with the following data:

Name	FQDN or IP Address	Type	Notes
Parlan...	10.10.98.157	Other	SIP entity for a partner testing

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.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left navigation pane is expanded to 'Routing' and 'Routing Policies' is selected. The main content area displays the 'Routing Policy Details' form. The 'General' section contains the following fields:

- Name:** Route_to_CS1K_Bottom
- Disabled:**
- Retries:** 2
- Notes:** (empty)

The 'SIP Entity as Destination' section features a 'Select' button and a table with the following data:

Name	FQDN or IP Address	Type	Notes
CS1K_Bottom	10.10.97.149	Other	SIP connection to CS1K

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.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The left navigation pane shows 'Routing' selected, with 'Dial Patterns' highlighted. The main content area is titled 'Dial Pattern Details' and includes a 'General' sub-section with the following fields:

- Pattern:** 30
- Min:** 5
- Max:** 5
- Emergency Call:**
- Emergency Priority:** 1
- Emergency Type:** (empty)
- SIP Domain:** bvwdev.com
- Notes:** Dial pattern to reach Parlane Office Assistant

Below the 'General' section is the 'Originating Locations and Routing Policies' section, which includes an 'Add' button and a table with the following data:

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Enabled	Routing Policy Destination	Routing Policy Notes
<input checked="" type="checkbox"/> Belleville	Belleville DevConnect Lab	Route_To_Parlane_OperatorAssistant	0	<input type="checkbox"/>	Parlane_OperatorAssistant	Route to a partner testing server

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).

The screenshot displays the Avaya Aura System Manager 7.0 interface. The left navigation pane shows the 'Routing' menu expanded to 'Dial Patterns'. The main content area is titled 'Dial Pattern Details' and contains the following fields:

- Pattern: 54
- Min: 5
- Max: 36
- Emergency Call:
- Emergency Priority: 1
- Emergency Type:
- SIP Domain: bvwddev.com
- Notes:

Below the 'General' section is the 'Originating Locations and Routing Policies' section, which includes an 'Add' button and a table with one item:

Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/> Belleville	Belleville Dev-Connect Lab	Route_to_CS1K_Bottom	0	<input type="checkbox"/>	CS1K_Bottom	

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.

Session Manager Name	SIP Entity Resolved IP	Port	Proto	Busy	Conn. Status	Reason Code	Link Status
DexvmsM	10.10.98.157	5060	UDP	FALSE	UP	200 OK	UP

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**.

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