



Application Notes for Configuring Rauland-Borg Responder[®] 5 to Interoperate with Avaya Communication Server 1000 and Avaya Aura[®] Session Manager– Issue 1.0

Abstract

These Application Notes describe a compliance-tested configuration consisting of the Rauland-Borg Responder[®] 5 solution, Avaya Communication Server 1000 and Avaya Aura[®] Session Manager.

The Rauland-Borg Responder[®] 5 solution is a complete nurse call system with associated Staff Management applications ensuring calls for assistance from patient rooms are immediately routed to the proper staff for response.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the 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 a compliance-tested configuration consisting of the Rauland-Borg Responder[®] 5 (hereafter known as Responder) solution, Avaya Communication Server 1000 (hereafter known as Communication Server 1000) and Avaya Aura[®] Session Manager (hereafter known as Session Manager).

The Responder solution is a complete nurse call system with associated Staff Management applications ensuring calls for assistance from patient rooms are immediately routed to the proper staff for response. It should be noted that the solution involves the use of a third party Brekeke SIP Server which is sold and supported by Rauland-Borg and/or Rauland-Borg authorized distributors, as a standard element of any solution involving SIP PBX integrations.

Calls from a patient room could be initiated by a patient (pain, assistance needed, etc.), or hospital staff (room cleaning, linens, etc.) with the push of a button. Staff using Avaya phones can be incorporated into the system so that calls to talk to a nurse for example would route through Session Manager to Communication Server 1000, and to be able to call the patient room in return. This adds the benefit of staff having access to other resources in the hospital using Avaya endpoints.

Hospital staff members who are responsible for direct communication with patient rooms generally roam using wireless phones. During compliance testing only Avaya Desk phones were used.

2. General Test Approach and Test Results

The compliance test focused on the ability for Rauland Responder[®] 5 endpoints to initiate and receive calls to and from Avaya Communication Server 1000 via Avaya Aura[®] Session Manager.

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 compliance test validated the ability of Responder to route calls to and from patient rooms to Avaya endpoints. Additionally, testing validated the ability for the Responder solution to recover from common outages such as network outages and server reboots.

Responder endpoints are designed with limited functionality. Responder endpoints are not designed for multi-line functions like Hold, Conference and Transfer.

2.2. Test Results

The objectives described in **Section 2.1** were verified and passed.

2.3. Support

Information, Documentation and Technical support for Rauland-Borg products can be obtained at:

- Phone: 1-847-590-7130
- Web: <http://www.rauland.com/>

3. Reference Configuration

Figure 1 illustrates the compliance test configuration consisting of:

- Avaya Communication Server 1000 R7.6
- Avaya Aura® Session Manager R7.0
- Avaya Aura® System Manager R7.0
- Various UNISTim and SIP endpoints
- Brekeke SIP Server (registrar)
- Responder® 5 Gateway Server
- Responder® 5 Branch Regional Controller
- Responder® 5 Communication Endpoints

Calls routed to and from the Communication Server 1000 used SIP trunks between the Brekeke SIP server and Session Manager, and in turn SIP trunks between Session Manager and Communication Server 1000.

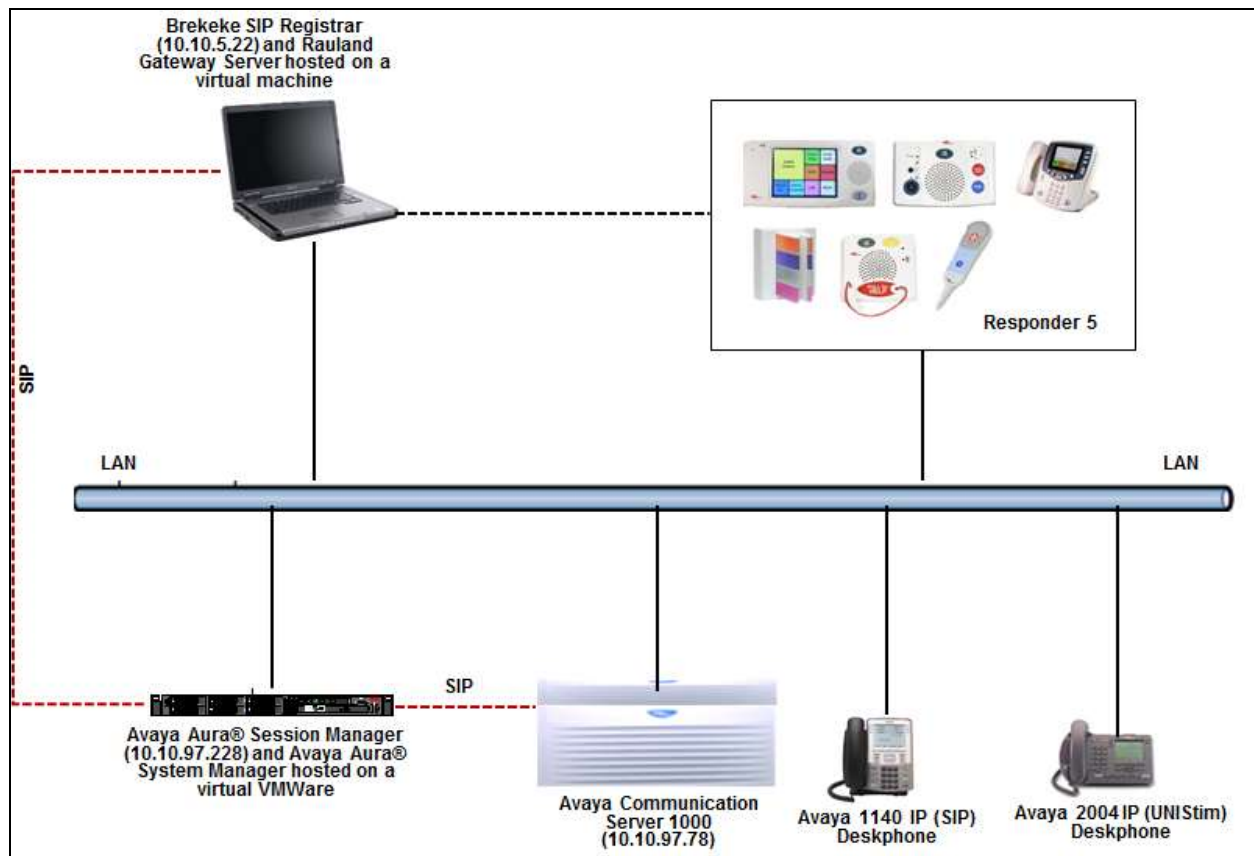


Figure 1 – Rauland-Borg Responder® 5 Compliance Test Configuration

4. Equipment and Software Validated

The following equipment and version were used in the reference configuration described above:

Equipment	Version
Avaya Communication Server 1000	7.65.16 SP7
Avaya Aura® Session Manager	7.0.0.0.700007
Avaya Aura® System Manager	7.0.0.0
Avaya IP Deskphones: 1140 (SIP) 2004P1 (UNISim)	4.03.09 0602B76
Rauland Nurse Call	T15 SP1
Rauland Gateway Server	T15 SP1
Rauland Apps	T15 SP1
Rauland DB	T15 SP1
Brekeke Server (Registrar)	3.3.4.4

5. Configure Avaya Communication Server 1000

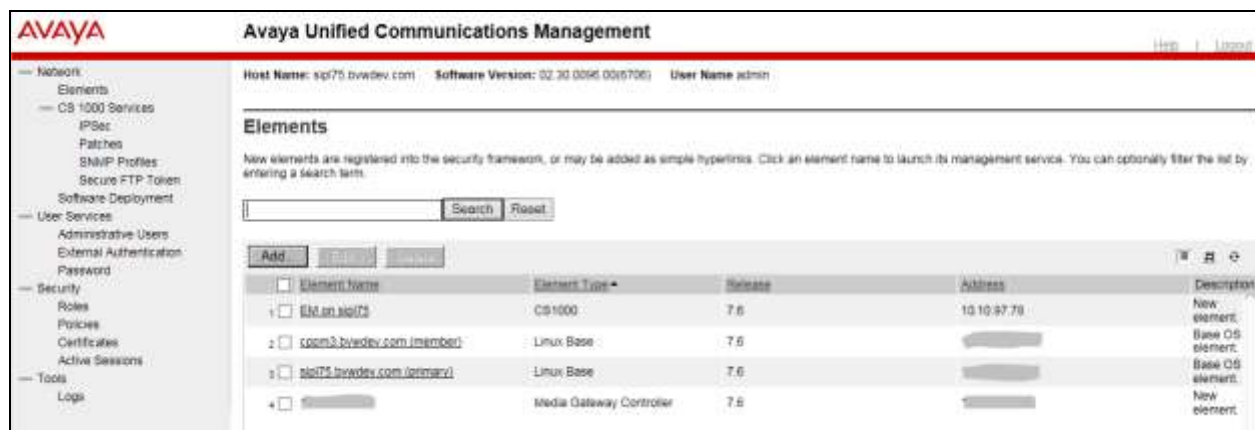
This section describes the Communication Server 1000 configuration necessary to interoperate with Session Manager and Responder. It provides the procedures for configuring Avaya Communication Server 1000 system. The procedures include the following areas:

- 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 Unified Communication Manager


User can login to the Element Manager via System Manager or Unified Communication Manager (UCM). During this compliance testing UCM was used to login to the Element Manager. To login to the UCM, open a browser and type in the IP address of the UCM in the URL (not shown). Screen below shows the main dashboard.



The screenshot shows the Avaya Unified Communications Management (UCM) dashboard. The top navigation bar includes the Avaya logo, the title "Avaya Unified Communications Management", and links for Help and Logout. The left sidebar contains a tree view of navigation options: Network, Elements, CS 1000 Services, IPSec, Patches, SNMP Profiles, Secure FTP Token, Software Deployment, User Services, Administrative Users, External Authentication, Password, Security, Roles, Policies, Certificates, Active Sessions, Tools, and Logs. The main content area displays the "Elements" page. At the top, it shows "Host Name: sip75.bvwddev.com", "Software Version: 02.30.0066.00(6706)", and "User Name: admin". Below this, a message states: "New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service. You can optionally filter the list by entering a search term." A search bar with "Search" and "Reset" buttons is provided. Below the search bar are "Add", "Edit", and "Delete" buttons. A table lists the elements:

<input type="checkbox"/>	Element Name	Element Type	Release	Address	Description
<input type="checkbox"/>	EM on sip75	CS1000	7.6	10.10.97.78	New element.
<input type="checkbox"/>	com3.bvwddev.com (member)	Linux Base	7.6		Base OS element.
<input type="checkbox"/>	sip75.bvwddev.com (primary)	Linux Base	7.6		Base OS element.
<input type="checkbox"/>		Media Gateway Controller	7.6		New element.

From the **Elements** page of UCM as shown in screen below, click on the Element **EM on sip175**. This is the element which is configured to access the Element Manager (EM) for the Communication Server 1000 Call Server.



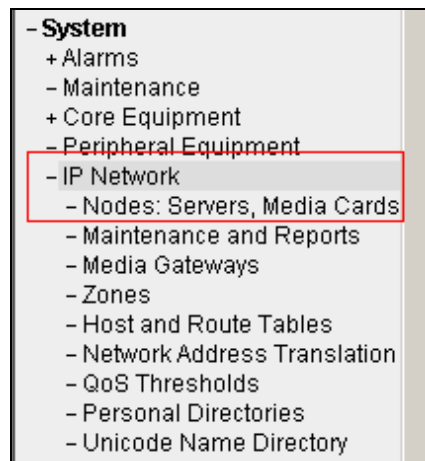
This screenshot is identical to the one above, but with a red box highlighting the first element in the table, "EM on sip75".

<input type="checkbox"/>	Element Name	Element Type	Release	Address	Description
<input type="checkbox"/>	EM on sip75	CS1000	7.6	10.10.97.78	New element.
<input type="checkbox"/>	com3.bvwddev.com (member)	Linux Base	7.6		Base OS element.
<input type="checkbox"/>	sip75.bvwddev.com (primary)	Linux Base	7.6		Base OS element.
<input type="checkbox"/>		Media Gateway Controller	7.6		New element.

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.

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

IP Telephony Nodes

Click the Node ID to view or edit its properties.

Node ID	Components	Enabled Applications	ELAN IP	Node/TLAN IPv4	Node/TLAN IPv6	Status
<input type="checkbox"/> 510	1	LTPS, Gateway (SIPGw)	-	-	-	Synchronized
<input type="checkbox"/> 512	1	SIP Line, LTPS, Presence Publisher	-	-	-	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.

AVAYA CS1000 Element Manager

Managing: [System](#) > [IP Network](#) > [IP Telephony Nodes](#) > [Node Details](#)

Node Details (ID: 510 - LTPS, Gateway (SIPGw))

Node ID: * (0-8899)

Call server IP address: *

TLAN address type: ☒ IPv4 only
☐ IPv4 and IPv6

Embedded LAN (ELAN)

Gateway IP address: *

Subnet mask: *

Telephony LAN (TLAN)

Node IPv4 address: *

Subnet mask: *

Node IPv6 address:

IP Telephony Node Properties

- [Voice Gateway \(V/GW\) and Coders](#)
- [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](#)

* Required Value.

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:** *bvwddev.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.

AVAYA CS1000 Element Manager

Managing: *System > IP Network > IP Telephony Nodes > Node Details > Virtual Trunk Gateway Configuration* Username: admin

Node ID: 510 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

Vtrk gateway application: ☒ Enable gateway service on this node

General

Vtrk gateway application: SIP Gateway (SIPGw) *
SIP domain name: bvwddev.com *
Local SIP port: 5060 * (1 - 65535)
Gateway endpoint name: cppm3 *
Gateway password: *
Application node ID: 510 * (0-9999)

Enable failsafe NRS: ☐
Note: FailSafe NRS will be enabled only on those servers in the node where NRS application is not deployed.

Virtual Trunk Network Health Monitor

☐ Monitor IP addresses (listed below)
Information will be captured for the IP addresses listed below:
Monitor IP: Add
Monitor addresses:
 Remove

* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved.

Save Cancel

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.

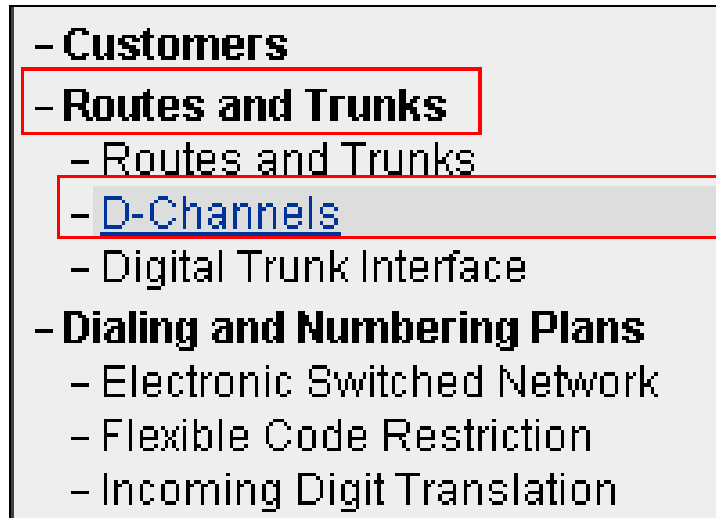
Retain default values for other fields.

The screenshot displays the AVAYA CS1000 Element Manager interface. The left sidebar shows a navigation tree with categories like UCM Network Services, Home, Links, System, Alarms, Core Equipment, Peripheral Equipment, IP Network, Nodes, Servers, Media Call, Maintenance and Reports, Media Gateways, Zones, Host and Route Tables, Network Address Translato, QoS Thresholds, Personal Directories, Unicode Name Directory, Interfaces, Engineered Values, Emergency Services, Geographic Redundancy, Software, Customers, Routes and Trunks, Routes and Trunks, D-Channels, Digital Trunk Interface, Dialing and Numbering Plans, Electronic Switched Network, Flexible Code Restriction, and Incoming Digit Translation. The main content area is titled 'Node ID: 510 - Virtual Trunk Gateway Configuration Details'. It shows the 'General' tab selected, with sub-tabs for 'SIP Gateway Settings' and 'SIP Gateway Services'. The 'Proxy Or Redirect Server' section is highlighted with a red box. It contains the following fields: 'Primary TLAN IP address' (10.10.97.228), 'Port' (5060), 'Transport protocol' (UDP), 'Options' (Support registration, Primary CDS proxy), 'Secondary TLAN IP address' (0.0.0.0), and 'Port' (5060). The 'Transport protocol' is also set to UDP. A note at the bottom states: 'Note: Changes made on this page will NOT be transmitted until the Node is also saved.' There are 'Save' and 'Cancel' buttons at the bottom right.

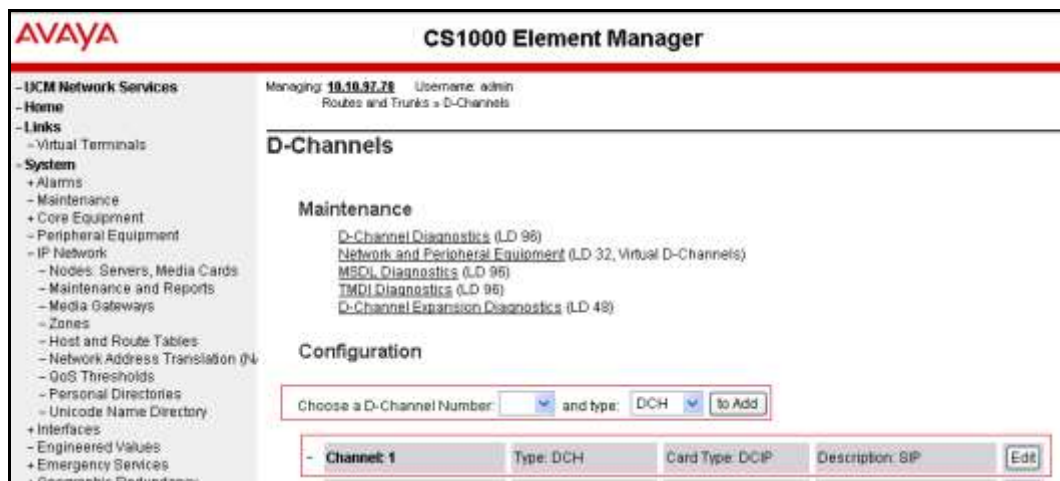
Save and transmit (not shown) these Node properties to complete the SIPGw configuration.

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 **1** 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 25 from the drop down menu.

Retain default values for all other fields.

Input Description	Input Value
Action Device And Number (ADAN):	DCH
D channel Card Type:	DCIP
Designator:	SIP
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 (ETSI)
D-Channel PRI loop number:	
Primary Rate Interface:	<input type="text"/> 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:	25
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: **Edit**

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.

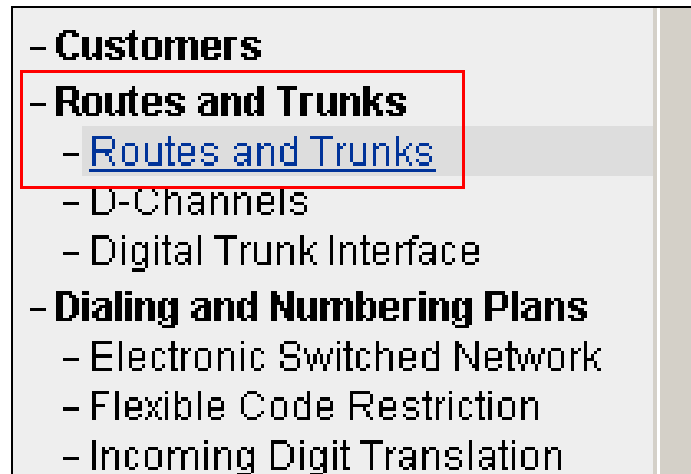
- Remote Capabilities Configuration

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"/>

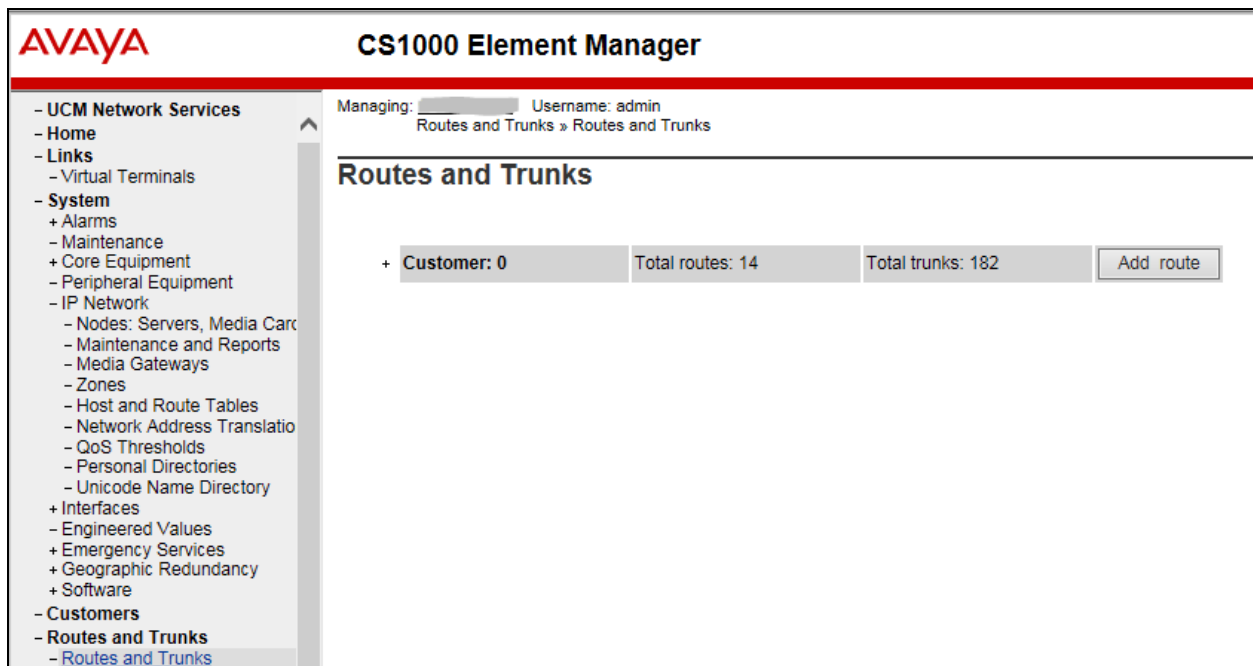
Now click on the **Submit** button (not shown) to complete the D-channel configuration.

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 1 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):** *1*
- **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):** *1*; 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 *Coordinated Dialing Plan (CDP)* from the drop down menu.
- **Calling number dialing plan (CNDP):** Select *Coordinated dialing plan (CDP)* 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 1 Property Configuration

- Basic Configuration

Route data block (RDB) (TYPE) :	<input type="text" value="RDB"/>
Customer number (CUST) :	<input type="text" value="00"/>
Route number (ROUT) :	<input type="text" value="1"/>
Designator field for trunk (DES) :	<input type="text" value="SIP"/>
Trunk type (TKTP) :	<input type="text" value="TIE"/>
Incoming and outgoing trunk (ICOG) :	<input type="text" value="Incoming and Outgoing (IAO)"/>
Access code for the trunk route (ACOD) :	<input type="text" value="8001"/>
Trunk type M911P (M911P) :	<input type="checkbox"/>
The route is for a virtual trunk route (VTRK) :	<input checked="" type="checkbox"/>
- Zone for codec selection and bandwidth management (ZONE) :	<input type="text" value="00002"/> (0 - 8000)
- Node ID of signaling server of this route (NODE) :	<input type="text" value="510"/> (0 - 9999)
- Protocol ID for the route (PCID) :	<input type="text" value="SIP (SIP)"/>
- Print correlation ID in CDR for the route (CRID) :	<input checked="" type="checkbox"/>
- Enable Shared Bandwidth Management for the route (SBWM) :	<input type="checkbox"/>
Integrated services digital network option (ISDN) :	<input checked="" type="checkbox"/>
- Mode of operation (MODE) :	<input type="text" value="Route uses ISDN Signaling Link (ISLD)"/>
- D channel number (DCH) :	<input type="text" value="1"/> (0 - 254)
- Interface type for route (IFC) :	<input type="text" value="Meridian M1 (SL1)"/>
- Private network identifier (PNI) :	<input type="text" value="00001"/> (0 - 32700)
- Network calling name allowed (NCNA) :	<input checked="" type="checkbox"/>
- Call type for outgoing direct dialed TIE route (CTYP) :	<input type="text" value="Coordinated Dialing Plan (CDP)"/>
- Insert ESN access code (INAC) :	<input checked="" type="checkbox"/>
- Integrated service access route (ISAR) :	<input type="checkbox"/>
- Display of access prefix on CLID (DAPC) :	<input type="checkbox"/>
- Mobile extension route (MBXR) :	<input type="checkbox"/>
- Mobile extension outgoing type (MBXOT) :	<input type="text" value="National number (NPA)"/>
- Mobile extension timer (MBXT) :	<input type="text" value="0"/> (0 - 8000 milliseconds)
Calling number dialing plan (CNDP) :	<input type="text" value="Coordinated dialing plan (CDP)"/>

- Network Options

Electronic switched network pad control (ESN) :	<input checked="" type="checkbox"/>
Signaling arrangement (SIGO) :	<input type="text" value="Standard (STD)"/>
Route class (RCLS) :	<input type="text" value="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.

Customer 0, Route 1, 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:

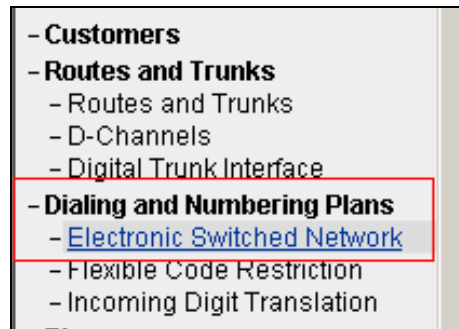
- Priority:

- Restriction level:

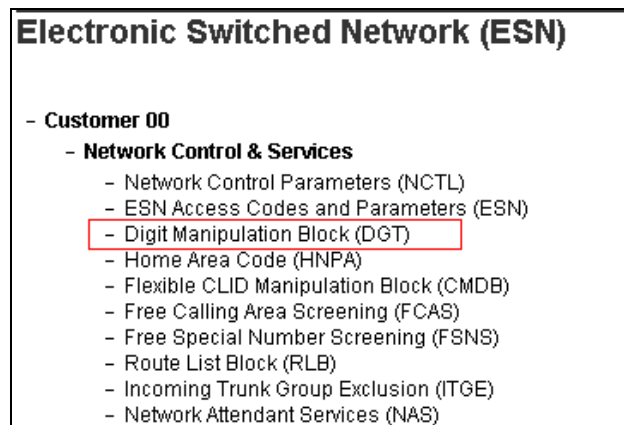
- Reversed Ear Piece:

5.5. Configuring Digit Manipulation Block

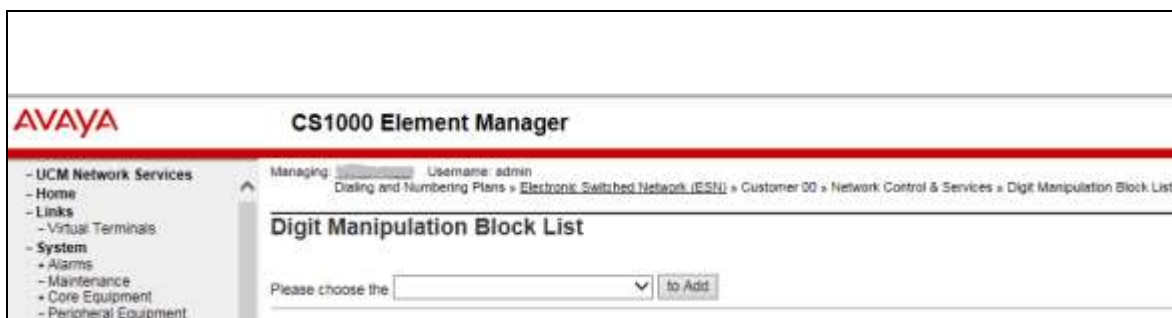
This section explains the digit manipulation block that is to be configured in the Communication Server 1000 dialing plan for its users to communicate with the Responder via the Session Manager. From the EM navigator pane, navigate to **Dialing and Numbering Plans** → **Electronic Switched Network**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



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

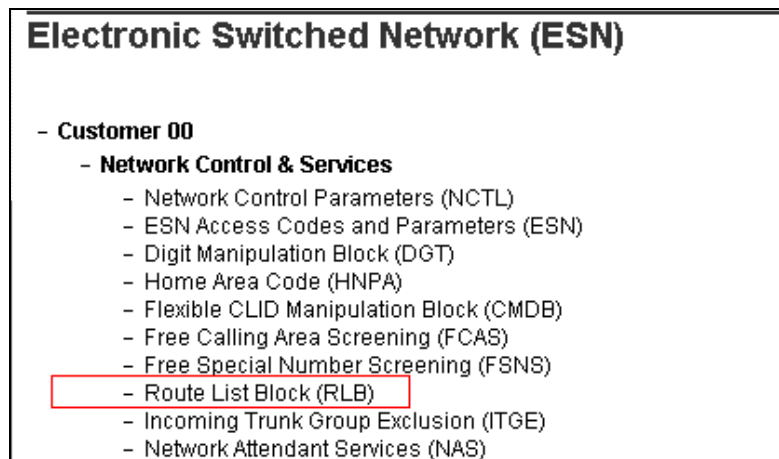


Screen below shows the **Digit Manipulation Block List** page where users can add a digit manipulation block index by selecting an available one from the drop down menu. During compliance testing **Digit Manipulation Block Index -- 0** was used which is already added in the Communication Server 1000 system by default.

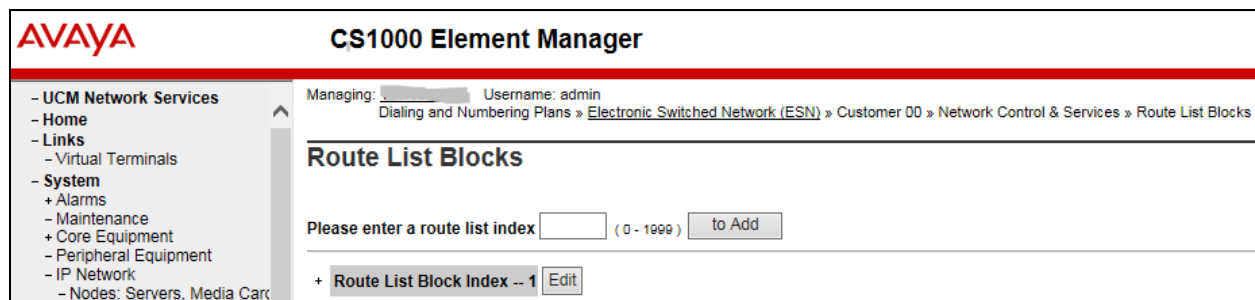


5.6. Configuring Route List Block

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 **1** was added.



Screen below show the values configured for the route list index block 1 added during compliance testing.

- **Digit Manipulation Index:** Select *0* from the drop down menu. This was configured in **Section 5.5**.
- **Route Number:** Select *1* 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: 1

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

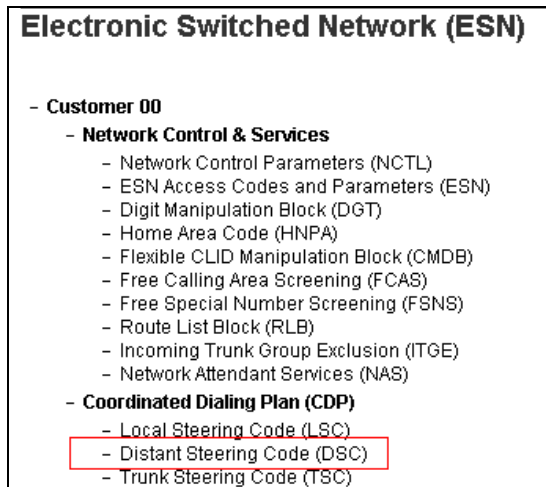
Local Termination entry: ☐

Route Number: ▼

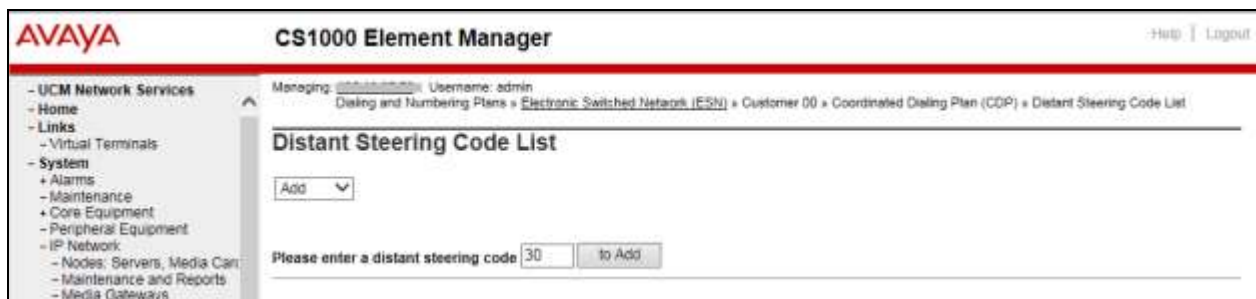
Skip Conventional Signaling: ☐

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 pilot number assigned to Responder 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 Responder is a 5 digit number.
- **Route List to be accessed for trunk steering code:** Select 1 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: 30

Flexible Length number of digits: 5 (0 - 10)

Display: Local Steering Code (LSC) ▼

Remote Radio Paging Access: ☐

Route List to be accessed for trunk steering code: 1 ▼

Collect Call Blocking: ☐

Maximum 7 digit NPA code allowed:

Maximum 7 digit NXX code allowed:

Submit

Refresh

Delete

Cancel

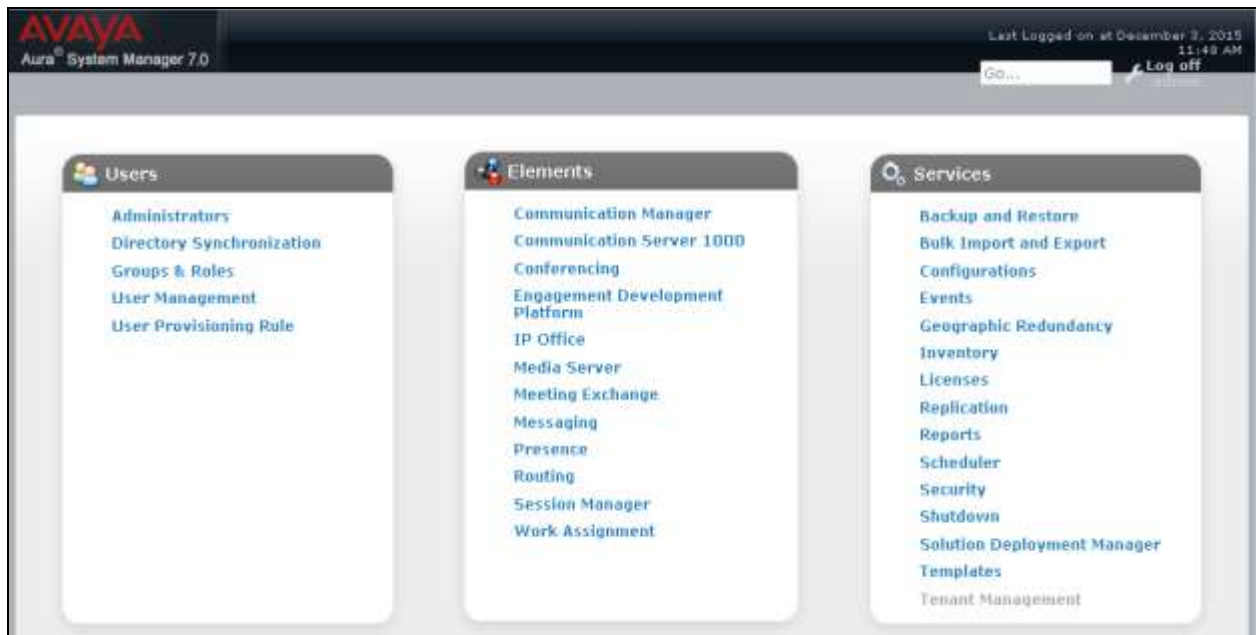
6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring routing using Avaya Aura® System Manager. The procedures include the following areas:

For detail configuration details of the Session Manager refer to **Section 10**

Session Manager is administered via the Avaya Aura® System Manager Web interface. In a browser, navigate to **https://:<hostname>/** and login with appropriate credentials. Use the hostname or IP Address of the System Manager server in the URL.

All navigation is performed by clicking links in the navigation links on the System Manager landing page as shown in the screen below. Click on the **Routing** link to access the Session Manager Routing Administration.



6.1. Configure Session Manager Details

Administration for the solution required the following steps:

- Add a Domain
- Add a Location
- Create an Adaptation Rule
- Add a SIP Entity
- Add an Entity Link
- Create a Routing Policy
- Create a Dial Pattern

6.1.1. Add a Domain

To add a domain, select **Domains** from the left hand window of the Routing screen and click on **New**. Configure a domain name and click on **Commit** (not shown) to complete adding a domain. Screen below shows a domain name of **bvwddev.com** that was added during compliance testing. Additional domains can be added in a similar fashion.



6.1.2. Add a Location

To add a location, select **Locations** from the left hand window of the Routing screen and click on **New**. Configure a location name and click on **Commit** (not shown) to complete adding a location. Screen below shows a location name of **Belleville** that was added during compliance testing. Additional locations can be added in a similar fashion.



6.1.3. Create an Adaptation Rule

Session Manager used an Adaptation rule for two purposes. First, domains in the To and From headers were modified to reconcile differences in the *bvwddev* domain used on Session Manager and Communication Server 1000, and the IP Address of the Brekeke SIP (Rauland) registrar used as the domain on that side of the call flow. For detail configuration details of various adaptations rules refer to **Section 10**.

To add an adaptation, select **Adaptations** from the left hand window of the Routing screen. Now click on **New** (not shown) to add an Adaptation rule. Screen below shows the adaptation details used during compliance testing.

- **Adaption Name:** *For_Rauland* – Any Descriptive name.
- **Module name:** *DigitConversionAdapter* – Selected from the drop down menu.
- **Module Parameter Type:** *Name-value Parameter* – Selected from the drop down menu and values added as follows,
fromto=true
iodstd=bvwddev.com
iosrcd=bvwddev.com
odstd=10.10.5.22

This defines a rule to modify domains in SIP headers. 10.10.5.22 is the IP address of the Brekeke SIP (Rauland) registrar used during compliance testing.

Click **Commit** to save the changes, then add the adaptation rule to the SIP Entity form that will be described in **Section 6.1.4**.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The left-hand navigation menu shows 'Routing' selected, with sub-items: Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Adaptation Details' and contains the following fields and controls:

- Adaptation Name:** For_Rauland
- Module Name:** DigitConversionAdapter (selected from a dropdown)
- Module Parameter Type:** Name-Value Parameter (selected from a dropdown)
- Parameter Table:** A table with columns 'Name' and 'Value'. It contains three rows:
 - fromto: true
 - iodstd: bvwddev.com
 - iosrcd: bvwddev.com
- Buttons:** 'Add' and 'Remove' buttons are located above the parameter table.
- Footer:** 'Egress URI Parameters:' and 'Notes:' fields are at the bottom.

Screen below shows the Adaptation rule after it was Committed.

AVAYA
Aura® System Manager 7.0

Last Logged on at December 3, 2015 11:48 AM
Log off

Home Routing

Home / Elements / Routing / Adaptations

Adaptations

New Edit Delete Duplicate More Actions

2 Items Filter Enable

<input type="checkbox"/>	Name	Module Name	Module Parameters	Egress URI Parameters	Notes
<input type="checkbox"/>	CS1000Adapter	CS1000Adapter	fronto=true		CS1000 adapter for Phone Context
<input type="checkbox"/>	For Rauland	DigitConversionAdapter	fronto=true iadstd=bvwddev.com iosrcd=bvwddev.com odstd=10.10.5.22		

Select: All, None

6.1.4. Add a SIP Entity

It is assumed that user has already configured SIP entities for Session Manager and Communication Server 1000. This application notes only describes below the SIP entity configured for the Brekeke SIP Registrar that is being used by Responder to connect to Session Manager.

To add a SIP entity, select **SIP Entities** from the left hand window of the Routing screen and click on **New** (not shown). On the SIP Entity Details screen shown below which appears when the New button is pressed, enter the following values.

- **Name:** Enter a descriptive name for the entity (*Rauland*).
- **FQDN or IP Address:** *10.10.5.22* was the address used by the Brekeke SIP registrar during compliance testing.
- **Type:** Select *Other* from the drop down menu.
- **Notes:** Useful for quick glance identification on other screens.
- **Adaptation:** Select *For_Rauland* from the drop down menu. This adaptation rule was created in **Section 6.1.3**.
- **Location:** Select *Belleville* from the drop down menu. This was created in **Section 6.1.2**.
- **SIP Link Monitoring:** Select *Link Monitoring Disabled* from the drop down menu. The Brekeke SIP registrar does not use link monitoring.
- **Entity Links:** This was added in a subsequent edit to the Entity record using the **Add** button but is described here for brevity purposes. See **Section 6.1.5** for how the Entity Link was created.

Retain default values for other fields.

Click **Commit** to complete the entries on this screen.

6.1.5. Add Entity Links

It is assumed that user has already configured Entity links for Communication Server 1000. This application notes only describes below the Entity links configured for the Brekeke SIP registrar that is being used by Responder to connect to Session Manager.

To add an Entity Link, select **Entity Links** from the left hand window of the Routing screen and click on **New** (not shown). On the **Entity Links** screen shown below which appears when the New button is pressed, enter the following values.

- **Name:** *DevvmSM_Rauland_5060_UDP* - A Descriptive name for the Entity Link.
- **SIP Entity 1:** Select *DevvmSM* from the drop down menu – This is the existing Session Manager SIP Entity.
- **SIP Entity 2:** Select *Rauland* from the drop down menu – This is the newly created SIP entity in **Section 6.1.4**.
- **Protocol:** Select *UDP* from the drop down menu.
- **Port:** *5060* – Port 5060 is the standard listen port for the UDP SIP transport protocol.
- **Connection Policy:** Select *trusted* from the drop down menu.

Retain default values for other fields.

Click **Commit** to save the entries.

The screenshot displays the 'Entity Links' configuration page in the Avaya Aura System Manager 7.0 interface. The left-hand navigation pane shows 'Entity Links' selected under the 'Routing' section. The main content area features a table with the following data:

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connection Policy
* DevvmSM_Rauland_E	* DevvmSM	UDP	* 5060	* Rauland	<input type="checkbox"/>	* 5060	trusted

At the bottom of the table, there is a 'Select' dropdown menu currently set to 'All, None'. The page includes 'Commit' and 'Cancel' buttons at the top right and bottom right.

6.1.6. Create a Routing Policy

Routing Policies require definition of a Routing Policy, and definition of Dial Patterns. A new Routing Policy is created first, leaving the Dial Pattern undefined, then a Dial Pattern is defined, then the Dial Pattern is applied to the Routing Policy.

It is assumed that user has already configured routing policies for Communication Server 1000. This application notes only describes below the routing policy configured for the Brekeke SIP registrar that is being used by Responder to connect to Session Manager.

To add a routing policy, select **Routing Policies** from the left hand window of the Routing screen and click on **New** (not shown). On the **Routing Policy Details** screen shown below which appears when the New button is pressed, enter the following values.

- **Name** and **Notes** as desired for the policy.
- Click the **Select** button to select the **SIP Entity as Destination** (not shown). The *Rauland* SIP Entity was selected as the Destination.

Retain default values for other fields.

Click **Commit** to save the entries.

Note that the **Dial Patterns** shown below was added when the **Dial Pattern** was defined in **Section 6.1.7** but is shown here for brevity.

AVAYA

Aura System Manager 7.0

Last Logged on at December 3, 2015 11:45 AM

Log off admin

Home

Routing

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit

Cancel

Help ?

General

Name: Route_to_Rauland_Server

Disabled: ☐

Retries: 0

Notes: Route to a partner testing server

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Rauland	10.10.5.22	Other	SIP entity for a partner testing

Time of Day

Add

Remove

View Gaps/Overlaps

1 Item

Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select: All, None

Dial Patterns

Add

Remove

1 Item

Filter: Enable

Pattern	Min	Max	Emergency Call	SIP Domain	Originating Location	Notes
30	\$	\$	<input type="checkbox"/>	brvdex.com	Belleville	Dial pattern to reach Rauland server

6.1.7. Create a Dial Pattern

It is assumed that user has already configured dial pattern for Communication Server 1000. This application notes only describes below the dial pattern configured for the Brekeke SIP Registrar that is being used by Responder to connect to Session Manager.

To add a dial pattern, select **Dial Patterns** from the left hand window of the Routing screen and click on **New** (not shown). On the **Dial Pattern Details** screen shown below which appears when the New button is pressed, enter the following values.

- **Pattern:** 30 – Pilot number to reach the Rauland was defined as 30xxx during compliance testing.
- **Min and Max:** 5 – The number of digits in the dialed number to match.
- **SIP Domain:** Select *bvwddev.com* from the drop down menu – The SIP Domain was configured in **Section 6.1.1**.
- **Originating Locations and Routing Policies:** See the next page for details of this step.

Retain default values for other fields.

Click on the **Commit** button to save the entries after the step on the following page is completed.

AVAYA
Aura System Manager 7.0

Last Logged on at December 3, 2015 11:45 AM
Go... Element: admin

Home Routing

Home / Elements / Routing / Dial Patterns

Dial Pattern Details [Commit] [Cancel] [Help ?]

General

* Pattern: 30
* Min: 5
* Max: 5
Emergency Call: ☐
Emergency Priority: 1
Emergency Type:
SIP Domain: bvwddev.com
Notes: Dial pattern to reach Rauland server

Originating Locations and Routing Policies

[Add] [Remove]

Filter: Enable

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 DevConnect Lab	Route_to_Rauland_Server	0	<input type="checkbox"/>	Rauland	Route to a partner testing server

Select: All, None

When the **Add** button is clicked on the **Originating Locations and Routing Policies** section for the **Dial Pattern Details** page, the screen shown below will appear.

The **Originating Location** can be defined as any location that originates a SIP request. In the compliance test, the location **Belleville** was used and therefore this option was selected. The *Route_To_Rauland_Server* policy defined in **Section 6.1.6** was selected in the **Routing Policies** section.

Click the **Save** button (not shown) to save these changes and return to the **Dial Pattern Details** page.

Originating Location
☐ Apply The Selected Routing Policies to All Originating Locations

1 Item

Filter: [Enable](#)

<input type="checkbox"/>	Name	Notes
<input type="checkbox"/>	Belleville	Belleville DevConnect Lab

Select : All, None

Routing Policies

9 Items

Filter: [Enable](#)

<input type="checkbox"/>	Name	Disabled	Destination	Notes
<input type="checkbox"/>	Route_to_CS1K_CPPM3	<input type="checkbox"/>	CS1K_Bottom	Route to the bottom CS1000
<input type="checkbox"/>	Route to DevAAM	<input type="checkbox"/>	DevAAM	
<input type="checkbox"/>	Route_to_DevCM	<input type="checkbox"/>	DevCM	Route to the main CM in the lab
<input type="checkbox"/>	RouteToDevvmCM	<input type="checkbox"/>	DevvmCM	
<input type="checkbox"/>	Route_to_IPO_36	<input type="checkbox"/>	IPO_36	Route to the top IP Office
<input type="checkbox"/>	Route_to_IPO_41_Server	<input type="checkbox"/>	IPO_41_Server	Route to the IP Office Server Edition
<input type="checkbox"/>	Route_to_IPO_44_Exp	<input type="checkbox"/>	IPO_44_Exp	Route to the IP Office Expansion
<input type="checkbox"/>	[REDACTED]	<input checked="" type="checkbox"/>	[REDACTED]	Route to a partner testing server
<input type="checkbox"/>	Route_to_Rauland_Server	<input type="checkbox"/>	Rauland	Route to a partner testing server

Select : All, None

7. Configure Responder[®] 5

The Responder solution is typically implemented by Rauland engineers or their resale partners. When integrated with a third party SIP PBX, it is always deployed with a Brekeke SIP registrar which serves two purposes. First, Brekeke SIP registrar is commonly deployed with a variety of SIP capable PBX solutions giving the Responder equipment a common and predictable SIP interface that is adaptable to many environments. Second, the Brekeke SIP registrar is capable of providing registrar services without requiring provisioning for each Responder endpoint thus significantly reducing the implementation and ongoing administration of the solution.

The Responder equipment will be provisioned completely by Rauland engineers based on site requirements, and will be configured to use the Brekeke SIP server for all calls destined to endpoints outside of the Responder endpoints.

The focus of this section will be on administration of the Responder applications, and configuration of the Brekeke SIP Server to properly route SIP calls and RTP.

7.1. Responder 5 Configuration Details

Administration for the solution required the following steps:

- Configure Endpoints
- Assign Endpoints to User
- User Login and Device Assignment
- Assign Staff to Patient Rooms

7.1.1. Configure Endpoints

Typically, hospital staff use wireless phones to enable instant communications with staff and patient rooms. In the tested confirmation, a variety of H.323 and SIP wireless devices which were previously configured on Communication Server 1000 were administered in the Responder applications to associate the endpoints with the hospital staff.

The Responder applications are accessed from the Windows PC used by a staff administrator and/or at nurse stations throughout the hospital. These PCs are used by staff to clock in and manage patient room assignments. The applications are launched from **Start → All Programs → Responder 5 Applications**.

In the top left corner is a drop down list that navigates to the various applications. Each requires an appropriate login (not shown). Select **Administration – Devices** in the upper left drop down list (not shown) to add or modify phones. Enter the appropriate **Device Name/Extension, Type**, and a **Description**. The illustration below shows a number of devices used in the test environment, extensions 56xxx were UNISTim and SIP devices administered on Communication Server 1000.

Click **OK** at the bottom of the screen to complete edits on this screen.

Device Name/Extension	Type	Description	Barcode
56104	Wireless Phone	Avaya test	
56204	Wireless Phone		

7.1.2. Assign Endpoints to User

Select **Administration – Devices** in the upper left drop down list (not shown) to add or modify users and to assign devices to the users. This task is only necessary for statically assigned device assignments. Users who share devices are able to enter the device they are using for a shift when they login as described in **Section 7.1.3**.

Users can be created or modified on the **User – Creation** tab (user creation is beyond the scope of these application notes, see Responder documentation for details of this task). Devices (phones) are created on the **User – General** tab as shown below.

Click **OK** to complete edits on this screen.



Employee #	User Name	Password	First Name	Middle Name	Last Name	Staff Level	Role	Permanent Device	Permanent Badge
10000001									
10001			Sharon		Steele	PCT	Surgery Clinical		
10003			Amy		Steele	LPN	Surgery Clinical		
10006			Louise		Steele	PCT	Surgery Clinical		
10008			Ray		Steele	PCT	Surgery Clinical		
10007			Dyllis		Steele	PCT	Surgery Clinical		
10006			P	A	Steele	EVS	EVS		
10009			Pam		Steele	EVS	EVS		
1001			Barbara		Steele	PCT	Surgery Clinical		
10010			Geoff		Steele	EVS	EVS		
10016			Ralph		Steele	EVS Supervisor	EVS		
10017			Todd		Steele	RN	Surgery Clinical		
10018			Holly		Steele	RN	Surgery Clinical		
10019			Patricia		Steele	RN	Surgery Clinical		
10023			Tim		Steele	RN	Surgery Clinical		
10024			C	J	Steele	Nurse Manager	Nurse Manager		
10028			Jonathan		Steele	LPN	Surgery Clinical		
10028			Ray		Steele	LPN	Surgery Clinical		
10028			Maria		Steele	Unit Secretary	Surgery Clinical		
10029			Harold		Steele	LPN	Surgery Clinical		
10030			Deryll		Steele	LPN	Surgery Clinical		
10034			J	C	Steele	PCT	S East Clinical		
10036			Kathleen		Steele	PCT	S East Clinical		
10042			Zayd		Steele	PCT	S East Clinical		
10043			Gertrud		Steele	LPN	S East Clinical		
10044			Ramona		Steele	RN	S East Clinical		
10048			Nina		Steele	RN	S East Clinical		

7.1.3. User Login and Device Assignment

At the beginning of a shift, or return to duty from breaks, users will scan their Hospital ID badge bar code with a scanner connected to the PC which will automatically log them in to the **My Profile** screen.

From this screen, a **Wireless Phone** and/or **Pager** number can be entered; duty status updated, and break status entered. The **My Assignments** and **My Preferences** tabs are available for staff to review the patient rooms they are assigned to and modify user preferences. The details of these tasks are beyond the scope of these Application Notes.

Click **Update** or **Update and Exit** to commit the changes.

The screenshot displays the 'My Profile' application window. The top navigation bar includes 'My Home', 'My Assignments', and 'My Preferences'. The main content area is divided into two sections: 'User' and 'My Status'.

User Section: Displays user information for 'Katie' with ID '19944'. It includes fields for 'Call', 'Text', and 'Pager' numbers, and a 'Close' button.

My Status Section: Contains a 'Device' section with a 'Phone' field (value: 55281) and a 'Wireless Phone' field (value: 55281). Below this are 'Update' and 'Update and Exit' buttons. The 'Duty' section shows 'Duty' status as 'On' (green dot) and 'Off' (grey dot). The 'Break' section includes a 'I am going to break for' dropdown (value: 15 minutes) and a 'My current break coverage' field (value: 15:21). A 'Code' field is also present.

This task is typically performed by shift supervisors. Staff can be assigned to patient rooms on the **Staff Assignment** screen which is accessed from the drop down menu at the upper left of the Responder 5 Applications. In the illustration below, 56201 is assigned to room like 501-1 by clicking on the Staff name in the left column, then clicking on the assignment space below the patient name. The staff members initials will appear as below when the staff member has been successfully assigned to a patient.

7.2. Configure Brekeke SIP Registrar

All administration is performed via web browser by navigating to the hostname or IP Address of the Brekeke server. Administration for the solution required the following steps:

- Configure SIP Server System Tab
- Configure SIP Server SIP Tab
- Configure SIP Server RTP Tab
- Configure Dial Plan Routing Rules

7.2.1. Configure SIP Server System Tab

The following system properties were pre-configured for the test environment.

The screenshot displays the Brekeke SIP Server configuration interface. The top navigation bar includes tabs for System, SIP, RTP, Database/Radius, and Advanced. The left sidebar lists various configuration categories, with 'SIP Server' and 'SYSTEM' highlighted. The main content area is titled 'System' and is divided into several sections:

- General:** Contains fields for Server Name (your-sip-sv), Server Description (your SIP Server), Server Location (your-place), Administrator SIP URI (your-sip-uri), Administrator Email Address, and Start up (radio buttons for manual and auto, with auto selected).
- Network:** Contains fields for Interface address 1 through 5 (10.10.5.34, and four empty fields), Remote Address Pattern 1 through 5, and Auto interface discovery (radio buttons for on and off, with off selected).
- IPv6:** Contains fields for IPv6 (radio buttons for on and off, with off selected) and RFC3484's policy table for Address Selection (radio buttons for on and off, with on selected).
- DNS:** Contains fields for DNS SRV (radio buttons for on and off, with on selected), DNS AAAA (radio buttons for on and off, with on selected), DNS Server (empty field), DNS SRV Failover (radio buttons for on and off, with on selected), and Caching period for resolved name (sec) (3600).

MENU	Caching period for unknown name (sec)	600
	Caching period for error (sec)	10
	UPnP	
	Enable/Disable	<input type="radio"/> enable <input checked="" type="radio"/> disable
	Default router IP address	
	Cache size	24
	Cache period (sec,0=disable)	86400
	Refresh Interval (sec,0=disable)	30
	Java	
	Java VM arguments	
<input type="button" value="Save"/> Your changes will be in effect after restart.		

7.2.2. Configure SIP Server SIP Tab

The following sip properties were pre-configured for the test environment.

The screenshot displays the Brekeke SIP Server configuration interface. The left sidebar contains a menu with the following items: Registered Clients, Active Sessions, User Authentication, Dial Plan, Aliases, Logs, Push Notification, Domains, Configuration (highlighted), SYSTEM (+), MAINTENANCE (-), Start/Shutdown, and Software Maintenance. The main content area is titled 'SIP' and is divided into several sections:

- SIP Server** (Header)
- SIP SERVER** (Section Header)
- SIP exchanger**
 - Session Limit (-1=unlimited): -1
 - Local Port: 5060
 - B2B-UA mode: ☐ on ☒ off
 - Check Maximum UDP packet size: ☐ on ☒ off
 - Maximum UDP packet size: 1500
- NAT traversal**
 - Keep address/port mapping: ☐ on ☒ off
 - Interval (ms): 12000
 - Method: ☒ Blank packet ☐ OPTIONS
 - Add 'rport' parameter (Send): ☒ on ☐ off
 - Add 'rport' parameter (Receive): ☒ on ☐ off
- Authentication**
 - REGISTER: ☐ on ☒ off
 - INVITE: ☐ on ☒ off
 - MESSAGE: ☐ on ☒ off
 - SUBSCRIBE: ☐ on ☒ off
 - Realm (ex: domain name):
 - Auth-user=user in "To:" (Register): ☐ yes ☒ no
 - Auth-user=user in "From:": ☐ yes ☒ no
 - FQDN only: ☐ yes ☒ no
 - Nonce Expires (seconds): 60
- Registration**
 - Adjusted Expires:
- Upper Registration**
 - On/Off: ☐ on ☒ off
 - Register Server:
 - Protocol: ☒ UDP ☐ TCP ☐ TLS

<ul style="list-style-type: none"> Registered Clients Active Sessions User Authentication Dial Plan Aliases Logs Push Notification Domains Configuration SYSTEM + MAINTENANCE - Start/Shutdown Software Maintenance 	<h3>Thru Registration</h3> <p>On/Off <input checked="" type="radio"/> on <input type="radio"/> off</p>
	<h3>Timeout (0=unlimited)</h3> <p>Ringing Timeout (ms) <input type="text" value="240000"/></p> <p>Talking Timeout (ms) <input type="text" value="259200000"/></p> <p>Upper/Thru Timeout(ms) <input type="text" value="30000"/></p>
	<h3>Dial Plan</h3> <p>Maximum history records <input type="text" value="10"/></p>
	<h3>Miscellaneous</h3> <p>100 Trying <input type="radio"/> any requests <input checked="" type="radio"/> only for initial INVITE</p> <p>Check Request-URI's validity <input type="radio"/> yes <input checked="" type="radio"/> no</p> <p>Server/User-Agent <input type="text"/></p>
	<h3>TCP</h3> <p>TCP-handling <input checked="" type="radio"/> on <input type="radio"/> off</p> <p>Queue Size <input type="text" value="50"/></p> <p>Maximum Active Connections (0=unlimited) <input type="text" value="0"/></p>
	<h3>TLS</h3> <p>TLS-handling <input type="radio"/> on <input checked="" type="radio"/> off</p> <p>Queue Size <input type="text" value="50"/></p> <p>Maximum Active Connections <input type="text"/></p>
<h3>WS (WebSocket)</h3> <p>WS-handling <input type="radio"/> on <input checked="" type="radio"/> off</p> <p>Listen port <input type="text" value="10080"/></p> <p>Queue Size <input type="text" value="50"/></p> <p>Maximum Active Connections <input type="text"/></p>	

Start/Shutdown
Software Maintenance

WSS (WebSocket over TLS)

WSS-handling ☐ on ☒ off

Listen port

Queue Size

Maximum Active Connections

Key and Certificate

Peer Certification Validation ☒ on ☐ off

File Type ☒ Certificate (.pem .der .cer .crt .cert) and Key (.pem .key .der) ☐ JKS ☐ PKCS#12 (.p12 .pfx)

Private Key File No File

Certificate File No File

Performance Optimization (Proxy)

Initial threads Upgrade required

Maximum Sessions per thread Upgrade required

Performance Optimization (Registrar)

Initial threads Upgrade required

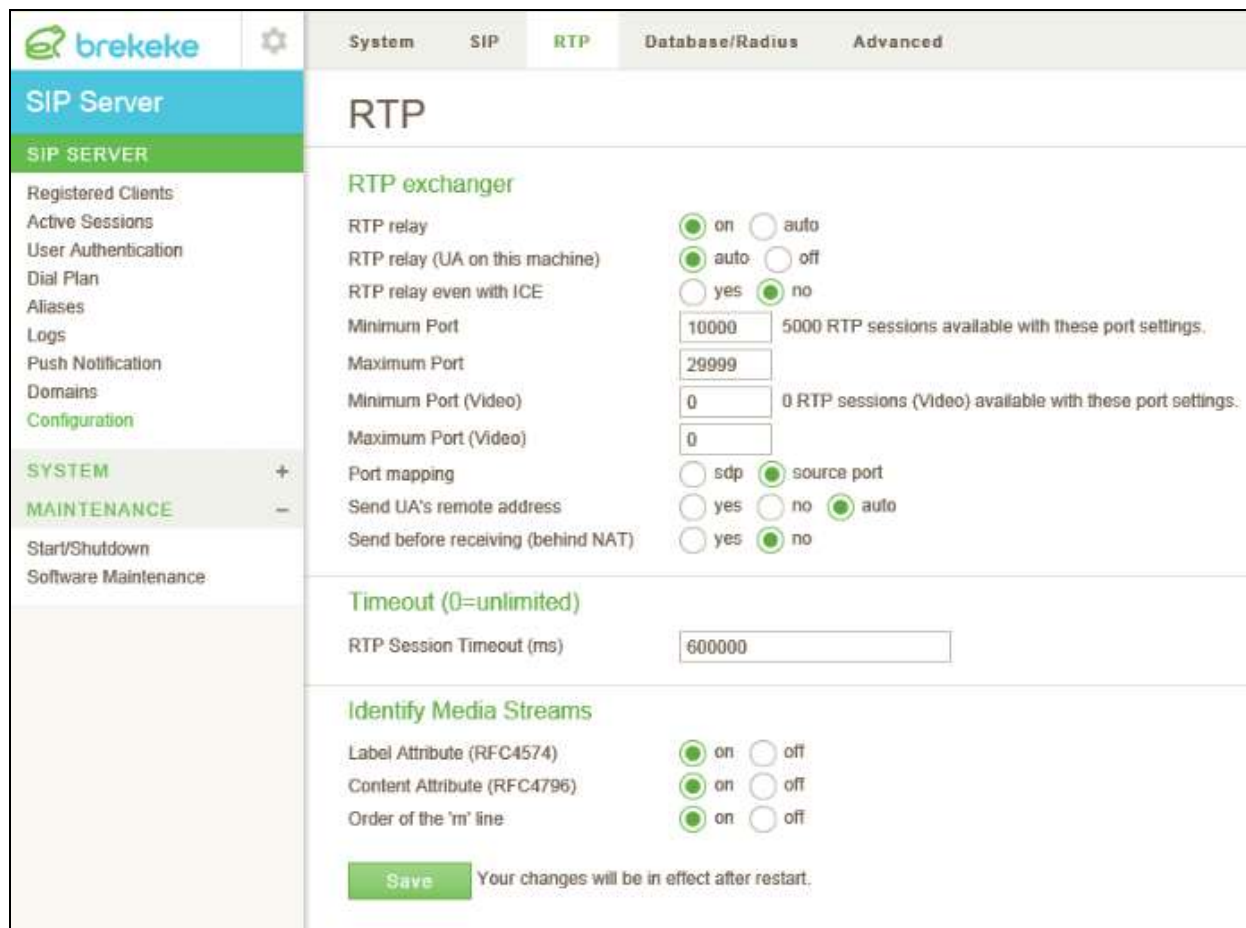
Maximum Sessions per thread Upgrade required

MENU

Your changes will be in effect after restart.

7.2.3. Configure SIP Server RTP Tab

On the **Configuration → RTP** screen, set **RTP Relay** to *on*, **RTP relay (UA on this machine)** to *auto*, **Port mapping** to *source port* and click **Save** to complete entries. Note, the **Minimum** and **Maximum Port** range settings should be sufficient to handle the maximum number of concurrent RTP sessions between systems.

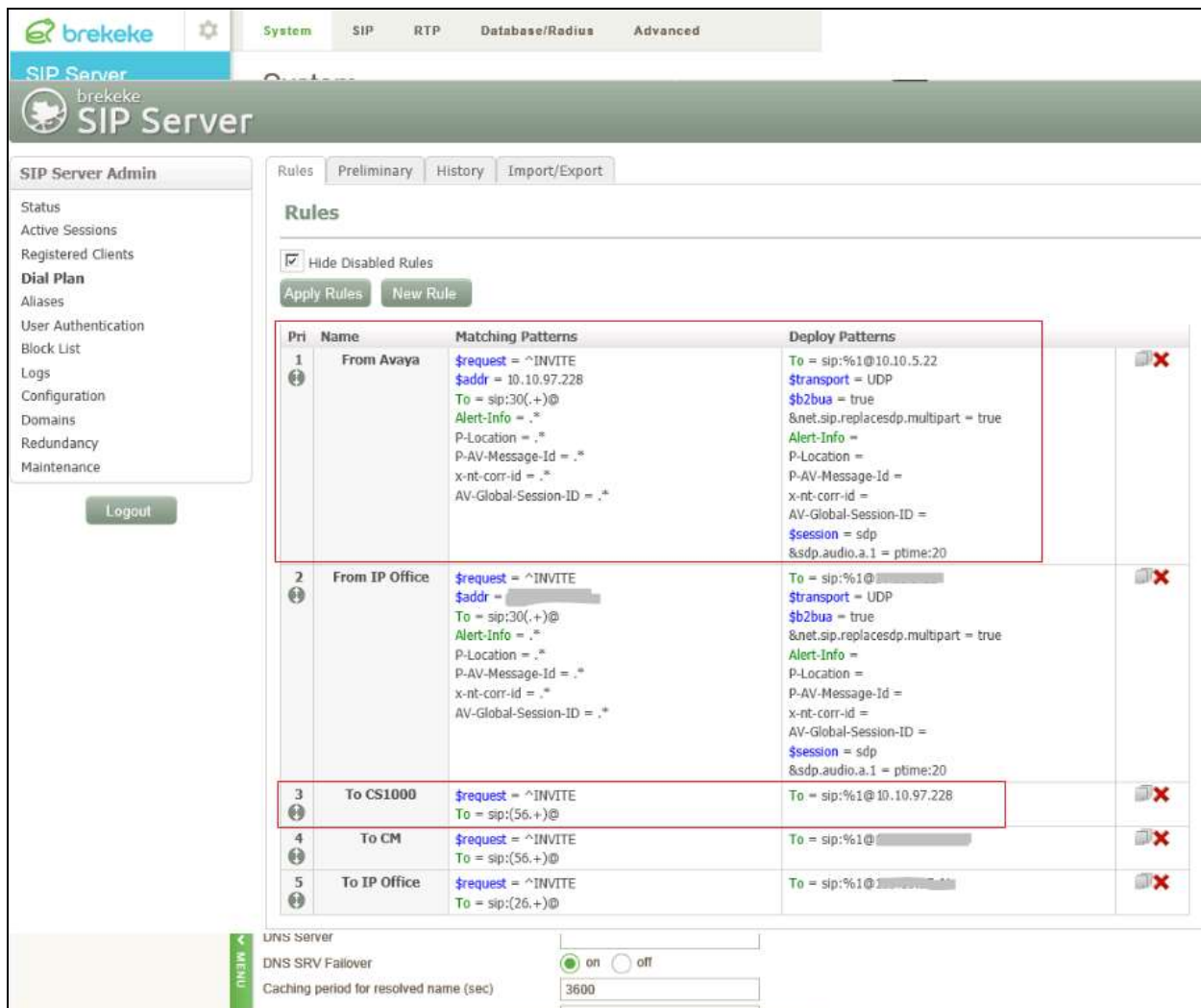


The screenshot shows the Brekeke SIP Server configuration interface, specifically the RTP tab. The left sidebar contains a menu with options like Registered Clients, Active Sessions, User Authentication, Dial Plan, Aliases, Logs, Push Notification, Domains, Configuration (highlighted), SYSTEM, and MAINTENANCE. The main content area is titled 'RTP' and contains several sections: 'RTP exchanger' with settings for RTP relay (on), RTP relay (UA on this machine) (auto), RTP relay even with ICE (no), Minimum Port (10000), Maximum Port (29999), Minimum Port (Video) (0), Maximum Port (Video) (0), Port mapping (source port), Send UA's remote address (auto), and Send before receiving (behind NAT) (no); 'Timeout (0=unlimited)' with RTP Session Timeout (ms) set to 600000; and 'Identify Media Streams' with Label Attribute (RFC4574) (on), Content Attribute (RFC4796) (on), and Order of the 'm' line (on). A 'Save' button is at the bottom, with a note that changes will be in effect after a restart.

System	SIP	RTP	Database/Radius	Advanced
SIP Server				
SIP SERVER				
RTP				
RTP exchanger				
RTP relay <input checked="" type="radio"/> on <input type="radio"/> auto				
RTP relay (UA on this machine) <input checked="" type="radio"/> auto <input type="radio"/> off				
RTP relay even with ICE <input type="radio"/> yes <input checked="" type="radio"/> no				
Minimum Port <input type="text" value="10000"/> 5000 RTP sessions available with these port settings.				
Maximum Port <input type="text" value="29999"/>				
Minimum Port (Video) <input type="text" value="0"/> 0 RTP sessions (Video) available with these port settings.				
Maximum Port (Video) <input type="text" value="0"/>				
Port mapping <input type="radio"/> sdp <input checked="" type="radio"/> source port				
Send UA's remote address <input type="radio"/> yes <input type="radio"/> no <input checked="" type="radio"/> auto				
Send before receiving (behind NAT) <input type="radio"/> yes <input checked="" type="radio"/> no				
Timeout (0=unlimited)				
RTP Session Timeout (ms) <input type="text" value="600000"/>				
Identify Media Streams				
Label Attribute (RFC4574) <input checked="" type="radio"/> on <input type="radio"/> off				
Content Attribute (RFC4796) <input checked="" type="radio"/> on <input type="radio"/> off				
Order of the 'm' line <input checked="" type="radio"/> on <input type="radio"/> off				
Save Your changes will be in effect after restart.				

7.2.4. Configure Dial Plan Routing Rules

Dial Plan rules that was used is illustrated below. For calls routing from Session Manager, the **From Avaya** rule was used. For calls routing to Communication Server 1000, the **To CS1000** rule was used.



The screenshot displays the Brekeke SIP Server Admin interface. The top navigation bar includes links for System, SIP, RTP, Database/Radius, and Advanced. The left sidebar contains a 'SIP Server Admin' menu with options like Status, Active Sessions, Registered Clients, Dial Plan, Allases, User Authentication, Block List, Logs, Configuration, Domains, Redundancy, and Maintenance. The main content area is titled 'Rules' and includes a 'Hide Disabled Rules' checkbox and buttons for 'Apply Rules' and 'New Rule'. A table lists five rules:

Pri	Name	Matching Patterns	Deploy Patterns
1	From Avaya	<code>\$request = ^INVITE</code> <code>\$addr = 10.10.97.228</code> <code>To = sip:30(.+)@</code> <code>Alert-Info = ,*</code> <code>P-Location = ,*</code> <code>P-AV-Message-Id = ,*</code> <code>x-nt-corr-id = ,*</code> <code>AV-Global-Session-ID = ,*</code>	<code>To = sip:%1@10.10.5.22</code> <code>\$transport = UDP</code> <code>\$b2bua = true</code> <code>&net.sip.replacesdp.multipart = true</code> <code>Alert-Info =</code> <code>P-Location =</code> <code>P-AV-Message-Id =</code> <code>x-nt-corr-id =</code> <code>AV-Global-Session-ID =</code> <code>\$session = sdp</code> <code>&sdp.audio.a.1 = ptime:20</code>
2	From IP Office	<code>\$request = ^INVITE</code> <code>\$addr =</code> <code>To = sip:30(.+)@</code> <code>Alert-Info = ,*</code> <code>P-Location = ,*</code> <code>P-AV-Message-Id = ,*</code> <code>x-nt-corr-id = ,*</code> <code>AV-Global-Session-ID = ,*</code>	<code>To = sip:%1@</code> <code>\$transport = UDP</code> <code>\$b2bua = true</code> <code>&net.sip.replacesdp.multipart = true</code> <code>Alert-Info =</code> <code>P-Location =</code> <code>P-AV-Message-Id =</code> <code>x-nt-corr-id =</code> <code>AV-Global-Session-ID =</code> <code>\$session = sdp</code> <code>&sdp.audio.a.1 = ptime:20</code>
3	To CS1000	<code>\$request = ^INVITE</code> <code>To = sip:(56.+)@</code>	<code>To = sip:%1@10.10.97.228</code>
4	To CM	<code>\$request = ^INVITE</code> <code>To = sip:(56.+)@</code>	<code>To = sip:%1@</code>
5	To IP Office	<code>\$request = ^INVITE</code> <code>To = sip:(26.+)@</code>	<code>To = sip:%1@</code>

At the bottom, there are server status options: 'DNS SRV Failover' (on/off) and 'Caching period for resolved name (sec)' (3600).

8. Verification Steps

Calls were placed to and from Responder endpoints, and two-way audio was confirmed. The nature of these devices is simple, one-way communications with Hospital staff; complex calls like transfer and conference are not supported on the patient room devices.

On the Brekeke SIP Server, the **Registered Clients** → **View Clients** screen will confirm if Responder endpoints are successfully registered as shown below.



The screenshot displays the Brekeke SIP Server Admin interface. On the left is a sidebar with navigation links: Status, Active Sessions, **Registered Clients**, Dial Plan, Allases, User Authentication, Block List, Logs, Configuration, Domains, Redundancy, and Maintenance. A 'Logout' button is at the bottom of the sidebar. The main content area is titled 'Registered Clients' and includes a 'Show Filter' button and an 'Unregister' button. A status bar indicates 'Registered: 210' and 'Pages: 1 2 3 ... 19 20 21'. Below this is a table with three columns: 'User', 'Contact URI (Source IP Address)', and 'Detail'.

User	Contact URI (Source IP Address)	Detail
<input type="checkbox"/> 5*501	sip:5*501@10.0.0.1:5064 (10.0.0.1:5064)	Expires : 3600 Priority : 1000 User Agent : Transport : UDP Time Update : Thu Dec 03 11:01:37 CST 2015
<input type="checkbox"/> 5*501*1	sip:5*501*1@10.0.0.1:5064 (10.0.0.1:5064)	Expires : 3600 Priority : 1000 User Agent : Transport : UDP Time Update : Thu Dec 03 11:01:37 CST 2015
<input type="checkbox"/> 5*501*101	sip:5*501*101@10.0.0.1:5064 (10.0.0.1:5064)	Expires : 3600 Priority : 1000 User Agent : Transport : UDP Time Update : Thu Dec 03 11:01:37 CST 2015

9. Conclusion

These Application Notes describe the procedures required to configure Rauland-Borg Responder® 5 to interoperate with endpoints registered to Avaya Communication Server 1000 via Avaya Aura® Session Manager using a Brekeke SIP Server as a SIP registrar and Proxy for the Responder 5 side of the solution.

All feature functionality test cases described in **Section 2.1** were passed.

10. Additional References

Product documentation for Avaya products may be found at <http://support.avaya.com>.

Avaya

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.

Rauland-Borg

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