



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Rauland-Borg Responder[®] 5 with Avaya Aura[®] Session Manager and Avaya Aura[®] Communication Manager R7.0 – Issue 1.0

Abstract

These Application Notes describe a compliance-tested configuration consisting of the Rauland-Borg Responder[®] 5 solution, Avaya Aura[®] Session Manager and Avaya Aura[®] Communication Manager R7.0.

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 solution, Avaya Aura[®] Session Manager and Avaya Aura[®] Communication Manager R7.0.

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 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 Manager, 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 Session Manager and Communication 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 for purpose 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 with the following observation.

- The Responder Branch Regional Controller (BRC) media processing unit does not support media shuffling. Attempts by the Avaya Media Gateway, or Media Resource/Processing boards to offer direct audio connections between IP endpoints and the BRC failed. The impact of this was that additional DSP resources were required on the Avaya Media Gateways and Media Resource/Processing boards to accommodate connections to Responder endpoints. A customer should ensure that adequate VoIP resources are available based on expected call traffic.
- Responder only supports G.711MU codec.

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 Aura® Communication Manager R7.0
- Avaya Aura® Session Manager R7.0
- Avaya Aura® System Manager R7.0
- Avaya Aura® Media Server R7.0
- Avaya G450 Media Gateway
- Various H.323 and SIP endpoints
- Brekeke SIP Server
- Rauland-Borg Responder® 5 Branch Regional Controller
- Rauland-Borg Responder® 5 Communication Endpoints

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

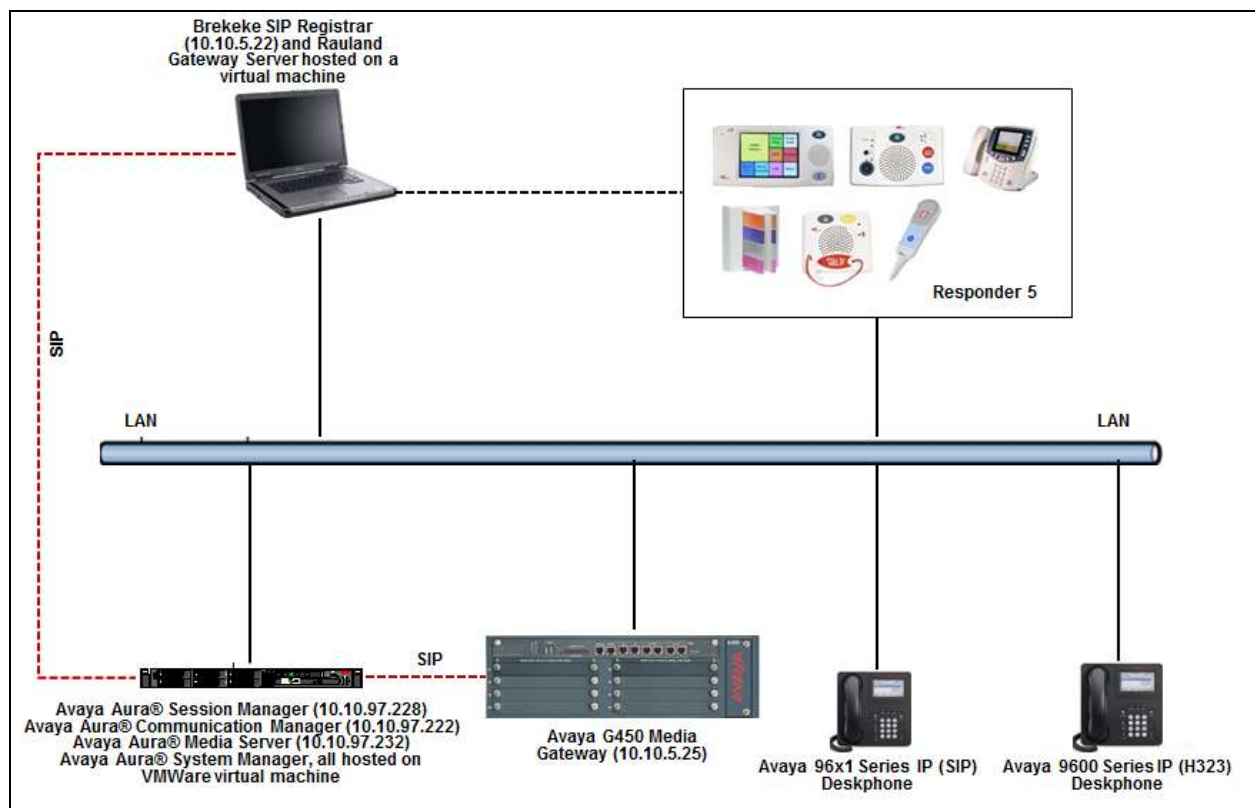


Figure 1 – Rauland-Borg Responder® 5 Compliance Test Configuration

4. Equipment and Software Validated

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

Equipment	Version
Avaya Aura® Communication Manager	7.0.0.1.0-SP1 (R017x.00.0.441.0)
Avaya Aura® Session Manager	7.0.0.0.700007
Avaya Aura® System Manager	7.0.0.0
Avaya Aura® Media Server	7.7.0.226
Avaya G450 Media Gateway	37 .19 .0 /1
Avaya IP Deskphones: 9620 (SIP) 9641 (H323)	2.6.4 6.0.1
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 Aura® Communication Manager

Configuration of Communication Manager required standard station administration which will not be covered in these Application Notes. In addition, routing was configured to enable calls originating from Communication Manager and Session Manager registered endpoints to be able to reach the Responder endpoints.

5.1. Configure Communication Manager Details

Calls were routed to Rauland endpoints using a 5 digit 30xxx pattern. All calls routed via a SIP trunk between Communication Manager and Session Manager using TCP transport. Existing SIP Trunks were in place in the environment. The steps below outline modifications made to accommodate the Responder solution. Therefore, some details required for SIP trunks may be omitted.

Administration for the solution required the following steps:

- Confirm Licensing
- Add node-names
- Add SIP Signaling Group
- Add SIP Trunk Group
- Change Route Pattern
- Change AAR Analysis
- Confirm IP codecs

5.1.1. Confirm Licensing

Using the **display system-parameters customer-options** command, confirm that the system has capacity for additional SIP Trunks. If additional license are required, contact an authorized Avaya Sales or Reseller representative.

display system-parameters customer-options		Page	2 of
12			
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
Maximum Administered H.323 Trunks:	4000	10	
Maximum Concurrently Registered IP Stations:	2400	3	
Maximum Administered Remote Office Trunks:	4000	0	
Maximum Concurrently Registered Remote Office Stations:	2400	0	
Maximum Concurrently Registered IP eCons:	68	0	
Max Concur Registered Unauthenticated H.323 Stations:	100	0	
Maximum Video Capable Stations:	2400	0	
Maximum Video Capable IP Softphones:	2400	3	
Maximum Administered SIP Trunks:	4000	24	
Maximum Administered Ad-hoc Video Conferencing Ports:	4000	0	
Maximum Number of DS1 Boards with Echo Cancellation:	80	0	

5.1.2. Add Node Names

Communication Manager uses the node-names ip table as a host lookup table. Host names used in subsequent steps will refer to these. Using the **change node-names ip** command, entries were added for Session Manager (**SM-VM**) and the processor Ethernet interface on Communication Manager (**procr**).

change node-names ip

Page 1 of 2

IP NODE NAMES

Name	IP Address
AVAYA-RD TT	10.10.98.71
DevvmAES	10.10.97.224
DevvmAMS	10.10.97.232
GW-G450	10.10.4.25
Loopback	10.10.97.222
SM-VM	10.10.97.228
TFTP-Server	10.10.98.72
default	0.0.0.0
procr	10.10.97.222
procr6	::

5.1.3. Add SIP Signaling Group

A signaling group was added using the **add signaling group 1** command with the following settings (settings not highlighted are default):

- **Group Type:** *sip*
- **Transport Method:** *tcp*
- **IP Video:** *n*
- **Near-end Node Name:** *procr*
- **Far-end Node Name:** *SM-VM*
- **Near-end Listen Port:** *5060*
- **Far-end Listen Port:** *5060*
- **Far-end Domain:** *bvwdev.com* (Match the domain on Session Manager).
- **Direct IP-IP Audio Connections:** *n*. (Responder does not support media shuffling)
- **DTMF over IP:** *rtp-payload*

```
add signaling-group 1                                     Page 1 of 3
                                                         SIGNALING GROUP

Group Number: 1                                           Group Type: sip
IMS Enabled? n                                           Transport Method: tcp
Q-SIP? n
IP Video? n                                              Enforce SIPS URI for SRTP? y
Peer Detection Enabled? y Peer Server: SM
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
Near-end Node Name: procr                               Far-end Node Name: SM-VM
Near-end Listen Port: 5060                             Far-end Listen Port: 5060
                                                         Far-end Network Region: 1

Far-end Domain: bvwdev.com

Incoming Dialog Loopbacks: eliminate                    Bypass If IP Threshold Exceeded? n
                                                         RFC 3389 Comfort Noise? n
DTMF over IP: rtp-payload                               Direct IP-IP Audio Connections? n
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
Enable Layer 3 Test? y
                                                         Alternate Route Timer(sec): 6
```

5.1.4. Add SIP Trunk Group

Using the **add trunk-group 1** command, trunk group 1 was created with the following settings (settings not highlighted are default):

- **Group Type:** *sip*
- **Group Name:** *Trunk to SM on VM*
- **TAC:** *#001*
- **Direction:** *two-way*
- **Service Type:** *tie*
- **Signaling Group:** *1*
- **Number of Members:** *24*

```
add trunk-group 1                                     Page 1 of 21
                                     TRUNK GROUP

Group Number: 1                                     Group Type: sip          CDR Reports: y
  Group Name: Trunk to SM on VM          COR: 1          TN: 1          TAC: #001
    Direction: two-way          Outgoing Display? y
  Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie          Auth Code? n
                                     Member Assignment Method: auto
                                     Signaling Group: 1
                                     Number of Members: 24
```

In page 3, **Numbering Format:** *private*

```
add trunk-group 1                                     Page 3 of 21
TRUNK FEATURES
  ACA Assignment? n          Measured: none
                                     Maintenance Tests? y

                                     Numbering Format: private
                                     UUI Treatment: service-provider
```


5.1.5. Change Route Pattern

Route Pattern 1 was configured to use Trunk Group 1 for calls to Responder and Session Manager registered endpoints using the **change route-pattern 1** command with the following settings (settings not highlighted are default):

- **Pattern Name:** *To SM on VM*
- **Grp No:** *1* (This specifies the Trunk Group to use)
- **FRL:** *0* (This can be used as a security setting to restrict access to trunks based on Class Of Restriction, 0 is least restrictive).

change route-pattern 1										Page 1 of 3			
Pattern Number: 1										Pattern Name: To SM on VM			
SCCAN? n		Secure SIP? n		Used for SIP stations? n									
Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted		DCS/	IXC			
No			Mrk	Lmt	List	Del	Digits		QSIG				
							Dgts		Intw				
1:	1	0							n	user			
2:										n	user		
3:										n	user		
4:										n	user		
5:										n	user		
6:										n	user		
BCC VALUE		TSC		CA-TSC		ITC		BCIE Service/Feature		PARM	Sub	Numbering	LAR
0 1 2 M 4 W				Request							Dgts	Format	
1:	y	y	y	y	y	n	n	rest				lev0-pvt	none
2:	y	y	y	y	y	n	n	rest					none
3:	y	y	y	y	y	n	n	rest					none
4:	y	y	y	y	y	n	n	rest					none
5:	y	y	y	y	y	n	n	rest					none
6:	y	y	y	y	y	n	n	rest					none

5.1.6. Change AAR Analysis

Using the **change aar analysis 0** command, dialed strings of 5 digits beginning with a 30 were instructed to use the *Route Pattern 1* configured in **Section 5.1.5**. Note all Responder endpoints used a 5 digit 30xxx extension.

change aar analysis 0							Page	1 of	2
AAR DIGIT ANALYSIS TABLE									
Location: all							Percent Full: 2		
	Dialed	Total		Route	Call	Node	ANI		
	String	Min	Max	Pattern	Type	Num	Reqd		
2		10	10	2	aar		n		
30		5	5	1	aar		n		
4		7	7	254	aar		n		
50		5	5	1	aar		n		
54		5	5	1	aar		n		
56		5	5	1	aar		n		
6		7	7	254	aar		n		
7		7	7	254	aar		n		
8		7	7	254	aar		n		
9		7	7	254	aar		n		

5.1.7. Confirm IP Codecs

Use the **change ip-codec-set n** command to add or change RTP codecs. In the test environment, codec set 1 was used for all endpoints and trunks. **G.711MU** was used for all calls with responder end points; the Responder BRC does not support G.729. As the media gateway or media server was required to be connected to all calls, the gateways/media server were able to transcode RTP, enabling different codecs to be used for each leg of the call.

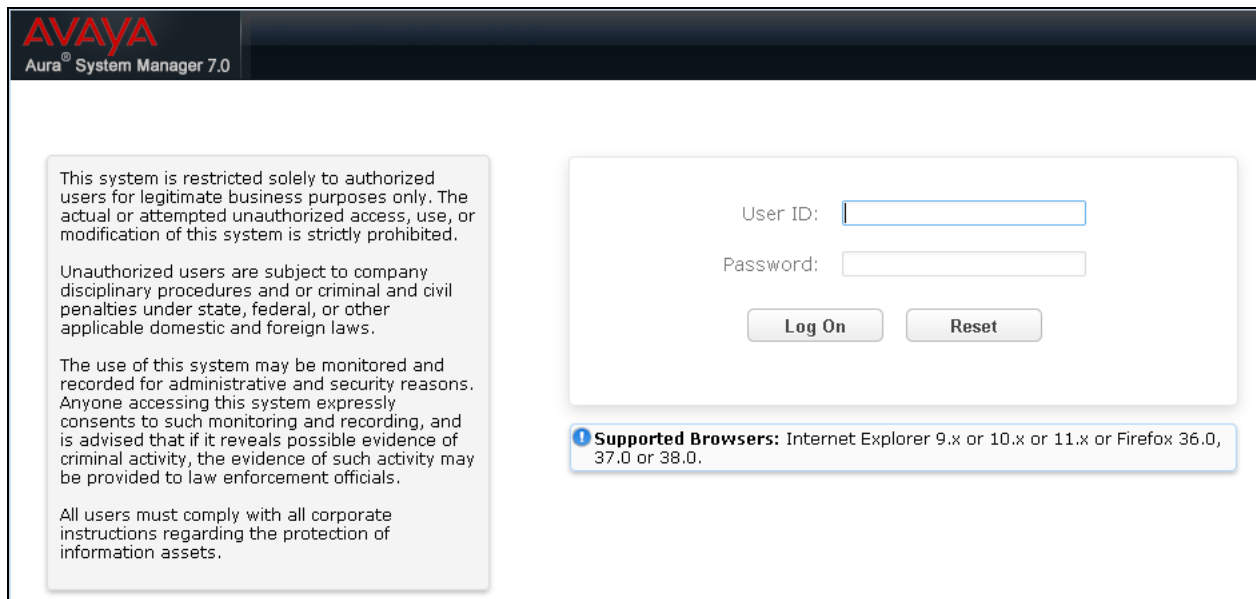
change ip-codec-set 1							Page	1 of	2
IP CODEC SET									
Codec Set: 1									
Audio	Silence	Frames		Packet					
Codec	Suppression	Per	Pkt	Size (ms)					
1: G.711MU	n	2		20					

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.



AVAYA
Aura® System Manager 7.0

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

All users must comply with all corporate instructions regarding the protection of information assets.

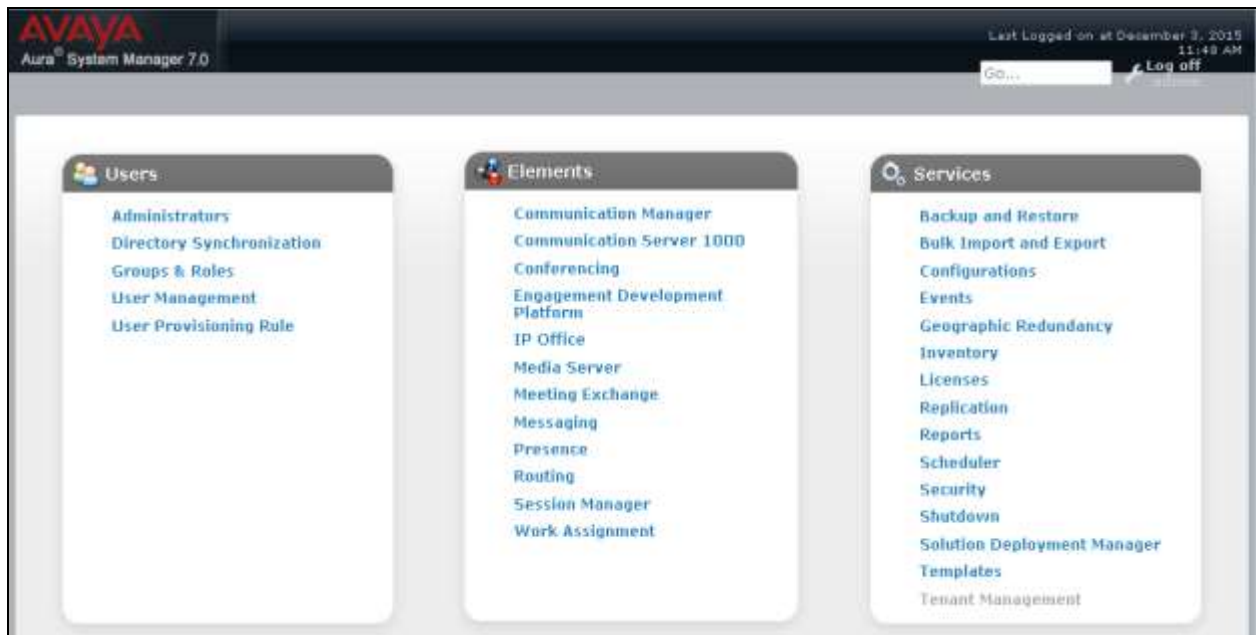
User ID:

Password:

Log On **Reset**

Supported Browsers: Internet Explorer 9.x or 10.x or 11.x or Firefox 36.0, 37.0 or 38.0.

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 **bvwdev.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 Manager, 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 shows the Avaya Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with 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 has a 'General' tab. The form includes the following fields and values:

- Adaptation Name:** For_Rauland
- Module Name:** DigitConversionAdapter
- Module Parameter Type:** Name-Value Parameter

Below these fields is a table with columns 'Name' and 'Value':

Name	Value
fromto	true
iodstd	bvwddev.com
iosrcd	bvwddev.com

At the bottom of the form are fields for 'Egress URI Parameters' and 'Notes'. The top right of the interface shows the user is logged in as 'Left Logged on at December 3, 2016 11:40 AM' with a 'Log off' button.

Screen below shows the Adaptation rule after it was Committed.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The top navigation bar includes the Avaya logo, 'Aura System Manager 7.0', and a 'Last Logged on at December 3, 2015 11:48 AM' status. The main menu on the left lists various configuration areas: Home, Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The 'Routing' section is expanded, showing a breadcrumb trail: Home / Elements / Routing / Adaptations. The 'Adaptations' page title is displayed, along with a 'Help' link. Below the title is a toolbar with buttons for 'New', 'Edit', 'Delete', 'Duplicate', and a 'More Actions' dropdown. A table lists two items, with the second item, 'For Rauland', highlighted by a red box. The table has columns for Name, Module Name, Module Parameters, Egress URI Parameters, and Notes. The 'For Rauland' row shows the module 'DigitConversionAdapter' with parameters 'fronto=true', 'iodstd=bvwddev.com', and 'iosrod=bvwddev.com odstd=10.10.5.22'. The 'Notes' column for this row contains the text 'CS1000 adapter for Phone Context'. At the bottom of the table, there is a 'Select: All, None' option.

Name	Module Name	Module Parameters	Egress URI Parameters	Notes
CS1000Adapter	CS1000Adapter	fronto=true		CS1000 adapter for Phone Context
For Rauland	DigitConversionAdapter	fronto=true iodstd=bvwddev.com iosrod=bvwddev.com odstd=10.10.5.22		

6.1.4. Add a SIP Entity

It is assumed that user has already configured SIP entities for Session Manager and Communication Manager. 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.

Note: Communication Manager SIP Entity was pre-configured and is not shown in this document as mentioned above. Communication Manager SIP Entity was configured in similar manner with the exception of **Type**; it was set to *CM*. Also there was no Adaptation rule used.

6.1.5. Add Entity Links

It is assumed that user has already configured Entity links for Communication Manager. 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.

Note: Communication Manger SIP Entity link was pre-configured and is not shown in this document as mentioned above. Communication Manager SIP Entity was configured in similar manner with the exception of **Protocol**; it was set to *tcp*.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with the following items: Home, Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links (selected), Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Entity Links' and shows a table with one entry. The table has columns for Name, SIP Entity 1, Protocol, Port, SIP Entity 2, DNS Override, Port, and Connection Policy. The entry is 'DevvmSM_Rauland_E', linking 'DevvmSM' and 'Rauland' via 'UDP' on port '5060' with a 'trusted' connection policy. Buttons for 'Commit' and 'Cancel' are visible at the top and bottom of the configuration area.

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

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 Manager. 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.

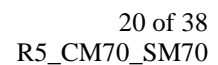
To add a routing policy, select **Routing Policies** from the left hand window of the Routing screen and click on **New** (not shown). On the **Routing Policy Details** screen shown below which appears when the New button is pressed, enter the following values.

- **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.



6.1.7. Create a Dial Pattern

It is assumed that user has already configured dial pattern for Communication Manager. 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 checked="" 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"/>		<input checked="" type="checkbox"/>		Route to a partner testing server
<input checked="" 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 Manager 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 H.323 and SIP devices administered on Communication Manager.

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		Sharon	PCT	Surgery Clinical		
10003			Amy		Amy	LPN	Surgery Clinical		
10006			Louise		Louise	PCT	Surgery Clinical		
10008			Kay		Kay	PCT	Surgery Clinical		
10009			Dyllis		Dyllis	PCT	Surgery Clinical		
10006			P	A	Sharon	EVS	EVS		
10009			Pam		Pam	EVS	EVS		
1001			Barbara		Barbara	PCT	Surgery Clinical		
10010			Geoff		Geoff	EVS	EVS		
10016			Ralph		Ralph	EVS Supervisor	EVS		
10017			Todd		Todd	RN	Surgery Clinical		
10018			Holly		Holly	RN	Surgery Clinical		
10019			Patricia		Patricia	RN	Surgery Clinical		
10023			Tim		Tim	RN	Surgery Clinical		
10024			C	J	Sharon	Nurse Manager	Nurse Manager		
10028			Jonathan		Jonathan	LPN	Surgery Clinical		
10028			Ray		Ray	LPN	Surgery Clinical		
10028			Maria		Maria	Unit Secretary	Surgery Clinical		
10029			Harold		Harold	LPN	Surgery Clinical		
10030			Deryll		Deryll	LPN	Surgery Clinical		
10034			J	C	Sharon	PCT	S East Clinical		
10036			Kathleen		Kathleen	PCT	S East Clinical		
10042			Zayd		Zayd	PCT	S East Clinical		
10043			Gertrud		Gertrud	LPN	S East Clinical		
10044			Ramona		Ramona	RN	S East Clinical		
10048			Nina		Nina	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 set to '555' and a 'Wireless Phone' field set to '555'. Below this are 'Update' and 'Update and Exit' buttons. The 'Duty' section shows 'Duty' and 'Code' fields, both set to '100'. The 'Break' section includes a 'I am going to break for' dropdown menu and a 'My current break coverage' field set to '10:00'. A 'Code' field is also present.

7.1.4. Assign Staff to Patient Rooms

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 member's 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. Ensure that **B2B-UA mode** is set to *on*.

The screenshot displays the Brekeke SIP Server configuration interface. The sidebar on the left includes the Brekeke logo, a settings gear icon, and a list of navigation items: SIP Server, SIP SERVER, Registered Clients, Active Sessions, User Authentication, Dial Plan, Aliases, Logs, Push Notification, Domains, Configuration, SYSTEM, MAINTENANCE, Start/Shutdown, and Software Maintenance. The main content area is titled 'SIP' and features several configuration sections:

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

<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

10081

Queue Size

50

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

Browse...

Certificate File

No File

Browse...

Performance Optimization (Proxy)

Initial threads

Upgrade required

Maximum Sessions per thread

Upgrade required

Performance Optimization (Registrar)

Initial threads

Upgrade required

Maximum Sessions per thread

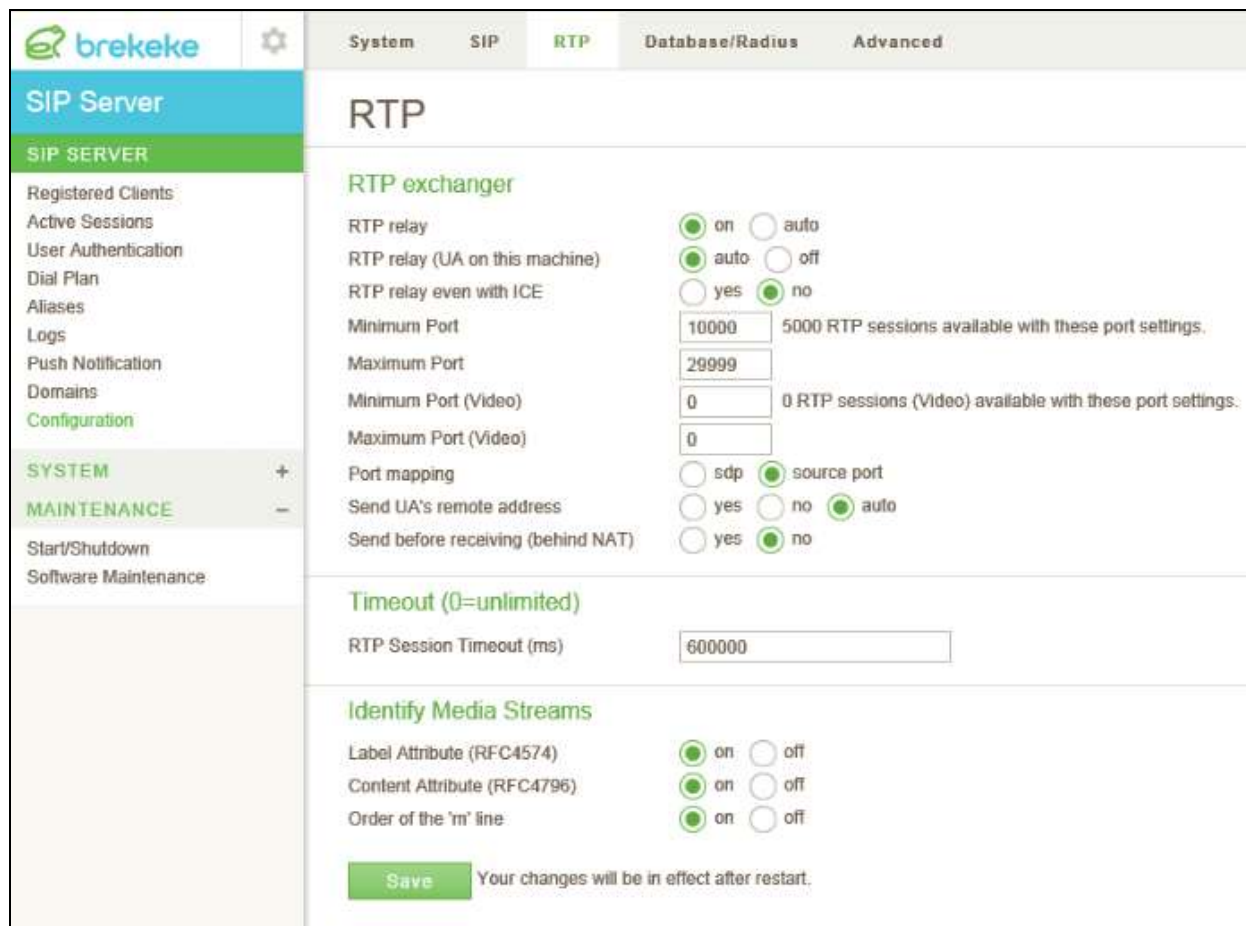
Upgrade required

Save

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, 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				
Registered Clients				
Active Sessions				
User Authentication				
Dial Plan				
Aliases				
Logs				
Push Notification				
Domains				
Configuration				
SYSTEM +				
MAINTENANCE -				
Start/Shutdown				
Software Maintenance				

RTP

RTP exchanger

RTP relay ☒ on ☐ auto

RTP relay (UA on this machine) ☒ auto ☐ off

RTP relay even with ICE ☐ yes ☒ no

Minimum Port 5000 RTP sessions available with these port settings.

Maximum Port

Minimum Port (Video) 0 RTP sessions (Video) available with these port settings.

Maximum Port (Video)

Port mapping ☐ sdp ☒ source port

Send UA's remote address ☐ yes ☐ no ☒ auto

Send before receiving (behind NAT) ☐ yes ☒ no

Timeout (0=unlimited)

RTP Session Timeout (ms)

Identify Media Streams

Label Attribute (RFC4574) ☒ on ☐ off

Content Attribute (RFC4796) ☒ on ☐ off

Order of the 'm' line ☒ on ☐ 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 Manager, the **To CM** rule was used.

SIP Server Admin

- Status
- Active Sessions
- Registered Clients
- Dial Plan**
- Aliases
- User Authentication
- Block List
- Logs
- Configuration
- Domains
- Redundancy
- Maintenance

[Logout](#)

Rules | Preliminary | History | Import/Export

☒ Hide Disabled Rules

[Apply Rules](#) [New Rule](#)

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 = ptim: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 = ptim:20</code>	
3	To CS1000	<code>\$request = ^INVITE</code> <code>To = sip:(54.+)@</code>	<code>To = sip:%1@</code>	
4	To CM	<code>\$request = ^INVITE</code> <code>To = sip:(56.+)@</code>	<code>To = sip:%1@10.10.97.228</code>	
5	To IP Office	<code>\$request = ^INVITE</code> <code>To = sip:(26.+)@</code>	<code>To = sip:%1@</code>	

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.

brekeke SIP Server

SIP Server Admin

- Status
- Active Sessions
- Registered Clients**
- Dial Plan
- Aliases
- User Authentication
- Block List
- Logs
- Configuration
- Domains
- Redundancy
- Maintenance

[Logout](#)

Registered Clients

Show Filter [Unregister](#) Registered: 210 Pages: 1 2 3 ... 19 20 21

<input type="checkbox"/>	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 Aura® Communication Manager 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 with the observations pointed in **Section 2.2**.

10. Additional References

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

Avaya

1. *Implementing Avaya Aura® Session Manager* Document ID 03-603473.
2. *Administering Avaya Aura® Session Manager*, Doc ID 03-603324.
3. *Deploying Avaya Aura® System Manager*, Release 7.0.
4. *Administering Avaya Aura® System Manager for Release 7.0*, Release 7.0.
5. *Quick Start Guide to Using the Avaya Aura® Media Server with Avaya Aura® Communication Manager*.
6. *Deploying and Updating Avaya Aura® Media Server Appliance*, Release 7.7.
7. *Administering Avaya Aura® Communication Manager*, Release 7.0, 03-300509.
8. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 7.0, 555-245-205.

Rauland-Borg

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