



Avaya Solution & Interoperability Test Lab

Application Notes for configuring Avaya Aura® Communication Manager R6.2 and Avaya Aura® Session Manager R6.2 with IBM Sametime Unified Telephony v8.5.2 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for IBM Sametime Unified Telephony to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. IBM Sametime Unified Telephony provides users with one number to reach them at any device. In the compliance testing, IBM Sametime Unified Telephony connected to Avaya Aura® Session Manager over a SIP trunk.

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

1. Introduction

These Application Notes describe the configuration steps required for IBM Sametime Unified Telephony to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

IBM Sametime Unified Telephony software integrates telecommunications into the existing unified communications (UC) and collaboration environment. IBM Sametime Unified Telephony software gives users one unified number to reach them at any device. The software turns the Sametime Connect client into a softphone, which allows users to make and receive calls from their computer from virtually anywhere. Incoming call management, telephony presence, click-to-call and click-to-conference are all provided within a consistent communications experience.

The connection between IBM Sametime Unified Telephony and Avaya Aura® Session Manager uses a TCP SIP trunk.

2. General Test Approach and Test Results

The general test approach was to validate correct handling of calls in a variety of call scenarios and recovery from network interruption. Parties involved in calls, clarity of audio and accurate call times, call parties and durations were verified. The resumption of service following outages of various components of the solution was also checked.

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

Interoperability compliance testing consisted of the successful placement, handling and termination of a variety of call activities using the Sametime Connect application and a variety of devices for the speech path, as well as recovery from failure in the following scenarios:

- Call to/from PSTN/PBX Deskphone/Computer SUT and non-SUT users
- Call forward to/from PSTN/PBX Deskphone/Computer SUT and non-SUT users
- Call deflect to/from PSTN/PBX Deskphone/Computer SUT and non-SUT users
- Call decline to voicemail from PSTN/PBX Deskphone/Computer SUT and non-SUT users
- Calling/called transfer to/from PSTN/PBX Deskphone/Computer SUT and non-SUT users
- Conferencing and Drag and Drop Conferencing with PSTN/PBX Deskphone/Computer SUT and non-SUT users
- Off-hook presence of SUT users
- Codec Negotiation

- Call history/dial from call history
- Voicemail Cover
- Conference bridge and DTMF
- Personal Call Routing
- Merge Consultation Calls

2.2. Test Results

All test cases passed successfully with the following observations:

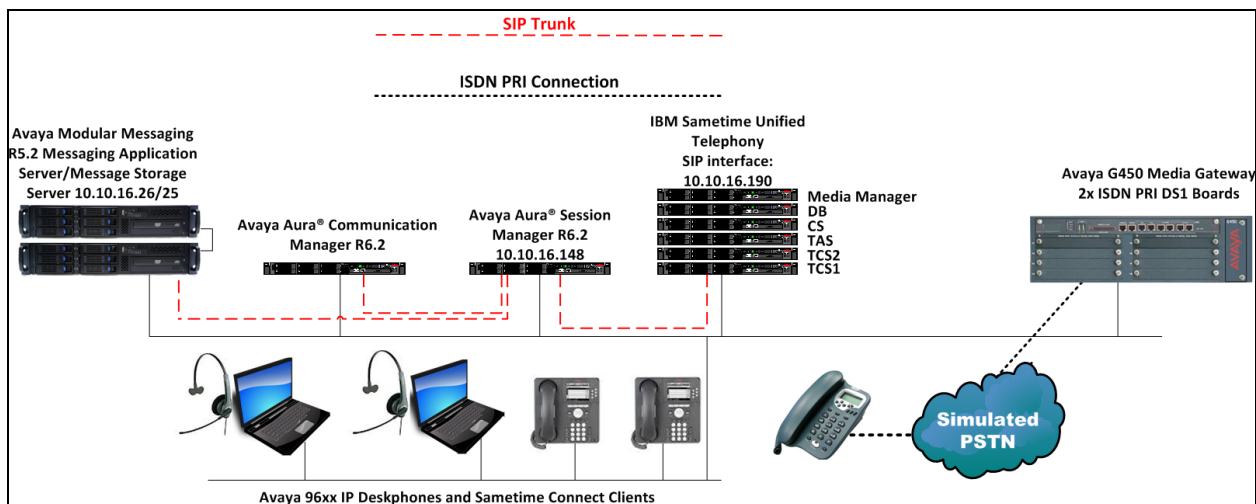
- Where g.729 is configured as the preferred codec on Sametime Unified Telephony, Communication Manager must be configured with only g.729 in order for g.729 to be utilized.
- It was not possible to successfully test TCS failover.

2.3. Support

http://www.ibm.com/support/entry/portal/overview/software/lotus/ibm_sametime_unified_telephony

3. Reference Configuration

An Avaya S8800 Server running Avaya Aura® Communication Manager R6.2 serving H323 endpoints with an Avaya G450 Media Gateway connected to a simulated PSTN was configured along with Avaya Aura® Session Manager R6.2 hosted on an Avaya S8800 Server providing SIP endpoints and SIP trunks to IBM Sametime Unified Telephony and Avaya Aura® Communication Manager. IBM Sametime Unified Telephony was configured on the same IP network for connection to the Avaya solution and hosted on a VMware 5.0 ESXi host. Avaya Modular Messaging provided Voicemail.



Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Avaya Modular Messaging with IBM Sametime Unified Telephony Solution

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Avaya S8800 Server	R6.2 SP4 build R016x.02.0.823.0-20199
Avaya Aura® Session Manager running on Avaya S8800 Server	R6.2 SP3
Avaya Modular Messaging running on Avaya S3500 Servers	<ul style="list-style-type: none">• 5.2 Patch 8• MAS - 9.2.150.13
Avaya G450 Media Gateway <ul style="list-style-type: none">• MM710• MM712	31.22.0 <ul style="list-style-type: none">• HW5 FW22• HW7 FW14
Avaya 9630 IP Deskphone	<ul style="list-style-type: none">• H323 S3.1 SP5• SIP 2.6 SP8
Avaya 2420 Digital Deskphone	2420 Rel 6.00 HWT=51H HWV=1 FWV=6
IBM Sametime Unified Telephony	8.5.2 IFR 1

5. Configure Avaya Aura® Communication Manager

The configuration and verification operations illustrated in this section were all performed using the Communication Manager System Access Terminal (SAT). It is assumed that the relevant dialplan, hunt groups, stations, trunks and call routing have been configured. The connection from Communication Manager to Session Manager is not specific to the test environment and is therefore not detailed below.

The information provided in this section describes the configuration of Communication Manager for this solution. For all other provisioning information such as installation and configuration, please refer to the product documentation in **Section 11**.

5.1. Configure Dialplan

In order that calls are routed to the extensions configured on the SUT solution, the dialplan must be configured accordingly using the **change dialplan analysis** command. In this case an **11** digit **Dialed String** beginning with **3** is routed to the uniform-dialplan (**udp**) table.

```

change dialplan analysis                                     Page 1 of 1
12
                                     DIAL PLAN ANALYSIS TABLE
                                     Location: all                               Percent Full: 1

   Dialed   Total   Call   Dialed   Total   Call   Dialed   Total   Call
   String   Length Type   String   Length Type   String   Length Type
1          3     fac          3          3     fac
2          10    udp          2          10    udp
3         11  udp         3         11  udp
4          4     udp          4          4     udp
5          4     ext          5          4     ext
6          4     ext          6          4     ext
7          3     dac          7          3     dac
8          4     udp          8          4     udp
9          1     fac          9          1     fac
*          3     fac          *          3     fac
  
```

5.2. Call Routing

The Application Notes assume that the relevant digital, SIP and H323 stations are configured and routing to Session Manager and the PSTN is in place. Use the **change uniform-dialplan 0** command and configure as shown below, where a matching pattern of **3** with a **Length** of **11** digits is sent to the **aar** table.

```

change uniform-dialplan 0                                 Page 1 of 2
                                     UNIFORM DIAL PLAN TABLE
                                     Percent Full: 0

   Matching   Len Del   Insert   Node
   Pattern    Len Del   Digits   Net Conv Num
2            10 0      2        ars  n
3           11 0      3       aar n
4            4 0       4        aar  n
8            4 0       8        aar  n
                                     n
  
```

Use the **change aar analysis 0** command. Assign values for this command as shown in the following table. In this case the **Dialed String 11** digits in length beginning with **3** is routed using **Route Pattern 1** where route pattern 1 is a preconfigured route to Session Manager.

```

change aar analysis 0                                     Page 1 of 2
                                     AAR DIGIT ANALYSIS TABLE
Location: all                                           Percent Full: 0

```

Dialed String	Total		Route Pattern	Call Type	Node Num	ANI Reqd
	Min	Max				
3	11	11	1	unku		n
4	4	4	1	unku		n
402	4	4	4	aar		n
5	4	4	1	aar		n
5999	4	4	1	unku		n
6000	4	4	1	unku		n
6001	4	4	1	unku		n
6002	4	4	1	unku		n
6003	4	4	1	unku		n
8000	4	4	1	unku		n
8897	4	4	1	aar		n

5.3. Configure Signaling Group

It is assumed the necessary signaling group and trunk configuration has been completed for the interface between Communication Manager and Session Manager. Enter the command **change signaling-group x** where **x** is the signaling group relevant to the trunk between Communication Manager and Session Manager, in this case **1**. Ensure that the items highlighted below are configured accordingly in order that shuffling is not configured for calls using this trunk.

```

change signaling-group 1                                 Page 1 of 2
                                     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
Near-end Node Name: procr	Far-end Node Name: sm62sigint
Near-end Listen Port: 5060	Far-end Listen Port: 5060
	Far-end Network Region: 1
Far-end Domain:	
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? n
Enable Layer 3 Test? y	IP Audio Hairpinning? n
	Alternate Route Timer(sec): 6

6. Configure Avaya Modular Messaging

Modular Messaging was used for the deposit and retrieval of voicemail. Login to the web interface of Modular Messaging and click **Subscriber Management**, enter the extension number to be added, in this case **5002** and click Add or Edit (not shown). Enter the following information and click Save (not shown)

- **Last Name** – an identifying last name
- **First Name** – an identifying first name
- **Password** – a numeric password
- **Mailbox Number & Numeric Address** – enter the extension number configured on SUT
- **PBX Extension** – enter the full number dialed for the SUT user – in this case **35318885002**. **Note:** this is especially necessary in order that voicemails are left on the correct mailbox.

BASIC INFORMATION * (Required Fields)			
*Last Name	Extn	First Name	5002
*Password		*Mailbox Number	5002
*Numeric Address	5002	*PBX Extension	35318885002
*Class Of Service	0 - class00	*Community ID	1

7. Configure Avaya Aura® Session Manager

This section illustrates relevant aspects of the Avaya Aura® Session Manager configuration required for interoperating with Sametime Unified Telephony.

Session Manager is managed via Avaya Aura® System Manager. Using a web browser, access **https://<ip-addr of System Manager>/SMGR**. In the **Log On** screen, enter appropriate **User ID** and **Password** and press the **Log On** button

AVAYA Avaya Aura® System Manager 6.2

Home / Log On

Log On

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

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

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

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

User ID:

Password:

[Change Password](#)

7.1. Configure IBM Sametime Unified Telephony SIP Entity

A SIP Entity must be created for the SUT SIP interface. Click **Routing** → **SIP Entities** → **New** (not shown). Enter a descriptive **Name**, the **FQDN or IP Address** for the SUT SIP interface, set the **Type** to **SIP Trunk**, and click **Commit** when done.

Home / Elements / Routing / SIP Entities

Help ?

SIP Entity Details

General

* Name:

* FQDN or IP Address:

Type:

Notes:

7.2. Configure Entity Link

The configuration of an Entity Link connects the Session Manager SIP Entity with the SUT SIP Entity. Click **Routing** → **Entity Links** → **New**(not shown). Enter a descriptive **Name**, choose the entity assigned to the preconfigured Session Manager SIP Signaling Interface as **SIP Entity 1**, set the **Protocol** as **TCP**, enter **5060** for the Port, choose the SUT SIP entity as **SIP Entity 2** and set the **Port** to **5060**, select **Trusted** from the **Connection Policy** drop down box. Click **Commit** when done. This establishes the Session Manager end of the SIP Trunk to SUT.

The screenshot shows the 'Entity Links' configuration page. The breadcrumb is 'Home / Elements / Routing / Entity Links'. The page title is 'Entity Links'. There are 'Commit' and 'Cancel' buttons. Below the title, there is a table with one item. The table has columns: Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Connection Policy, and Notes. The row contains: * toIBM, * SM62, TCP, * 5060, * IBM, * 5060, Trusted, and an empty Notes field. A red box highlights the row.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* toIBM	* SM62	TCP	* 5060	* IBM	* 5060	Trusted	

7.3. Create Routing Policy

Click **Routing** → **Routing Policies** → **New** (not shown). Enter a descriptive **Name** for the route. Under the **SIP Entity as Destination** section, click on **Select**.

The screenshot shows the 'Routing Policy Details' configuration page. The breadcrumb is 'Home / Elements / Routing / Routing Policies'. The page title is 'Routing Policy Details'. There are 'Commit' and 'Cancel' buttons. Below the title, there is a 'General' section with fields for Name (ToIBM), Disabled (checkbox), Retries (0), and Notes. Below that is the 'SIP Entity as Destination' section with a 'Select' button. At the bottom, there is a table with columns: Name, FQDN or IP Address, Type, and Notes.

General

* Name:

Disabled:

* Retries:

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
------	--------------------	------	-------

Choose the SUT Entity configured in **Section 7.1** and click **Select**.

Home / Elements / Routing / Routing Policies

SIP Entity List [Select](#) [Cancel](#)

SIP Entities

7 Items | [Refresh](#) Filter: [Enable](#)

	Name	FQDN or IP Address	Type	Notes
<input type="radio"/>	CM62	10.10.16.142	CM	
<input type="radio"/>	CMM62	10.10.16.142	CM	
<input type="radio"/>	ExperiencePortal	10.10.16.99	Voice Portal	
<input checked="" type="radio"/>	IBM	10.10.16.190	SIP Trunk	

Review the configuration and click **Commit** when done.

Home / Elements / Routing / Routing Policies

Routing Policy Details [Help ?](#)
[Commit](#) [Cancel](#)

General

* Name:

Disabled:

* Retries:

Notes:

SIP Entity as Destination

[Select](#)

Name	FQDN or IP Address	Type	Notes
IBM	10.10.16.190	SIP Trunk	

7.4. Administer Dial Patterns

Session Manager routes SIP traffic between connected devices. Dial Patterns are created as part of the configuration to manage SIP traffic routing, which will direct calls based on the number dialed to the appropriate destination. In **Section 5.2** Communication Manager is configured to route 11 digit strings beginning with 3 to Session Manager. To create a Dial Pattern to route these digits from Session Manager to SUT click **Routing → Dial Patterns → New** (not shown). Under **General** enter the extension (full or partial) presented to Session Manager by Communication Manager in the **Pattern** box. Set the **Min** and **Max** digit string length, and set **SIP Domain** to **ALL**. In the **Originating Locations and Routing Policies** section of the web page, click **Add**.

Home / Elements / Routing / Dial Patterns [Help ?](#)

Dial Pattern Details

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call:

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

0 Items | [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
--------------------------	---------------------------	----------------------------	---------------------	------	-------------------------	----------------------------	----------------------

Check the **Apply The Selected Routing Policies to All Originating Locations** check box, and select the **Routing Policy** created in **Section 7.3**. Click **Select** when done

Home / Elements / Routing / Dial Patterns

Originating Location and Routing Policy List Select Cancel

Originating Location

Apply The Selected Routing Policies to All Originating Locations

1 Item | Refresh Filter: Enable

<input checked="" type="checkbox"/>	Name	Notes
<input type="checkbox"/>	DevConnectLab	

Select : All, None

Routing Policies

9 Items | Refresh Filter: Enable

<input type="checkbox"/>	Name	Disabled	Destination	Notes
<input type="checkbox"/>	ToCM6.2	<input type="checkbox"/>	CM62	
<input type="checkbox"/>	ToCMM62	<input type="checkbox"/>	CMM62	
<input checked="" type="checkbox"/>	ToIBM	<input type="checkbox"/>	IBM	

Review the configuration and click **Commit** when done.

Home / Elements / Routing / Dial Patterns

[Help ?](#)

Dial Pattern Details **Commit** **Cancel**

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call:

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

1 Item | [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	ToIBM	0	<input type="checkbox"/>	IBM	

7.5. Configure Local Host Name Resolution

The IP address and hostname of SUT must be resolvable by Session Manager. Click **Session Manager → Network Configuration → Local Host Name Resolution → New** (not shown). Enter the **Host Name (FQDN)**, **IP Address** and **Port** used by the SUT server as shown below.

<input type="checkbox"/>	Host Name (FQDN)	IP Address	Port	Priority
<input checked="" type="checkbox"/>	dubxpcvm2060.mul.ie.ibm.com	10.10.16.190	5060	100
<input type="checkbox"/>				200
<input type="checkbox"/>				300
<input type="checkbox"/>				400

Scroll across the bottom of the page, so that the **Transport** drop down box is visible and select **TCP** to match the protocol configured in **Section 7.2**. Click **Commit** when done

Address	Port	Priority	Weight	Transport
.16.190	5060	100	100	TCP

8. Configure IBM Sametime Unified Telephony

The installation and commissioning of IBM Sametime Unified Telephony is managed by IBM. The configuration significant to the integration with the Avaya solution can be summarized as follows, for further specific configuration information see **Section 11**

- Configure Unified Numbers
- Configure IBM Sametime Unified Telephony Users
- Configure SIP Trunk
- Configure Destinations

8.1. Configure Unified Numbers

Unified Numbers must be configured for each user on the Sametime Unified Telephony platform. The screen below shows the configured **Directory Numbers** which are configured in **Section 7** to route to Sametime Unified Telephony.

	Directory Number	External Number	Display Name	Unicode Display Name	Feature Profile	Keyset	Numbering Plan
<input type="checkbox"/>	35318881005		35318881005			None	PNP_Sub1
<input type="checkbox"/>	35318885000		Richard Pope	Richard Pope	FP_SUT	None	PNP_Sub1
<input type="checkbox"/>	35318885001				FP_SUT	None	PNP_Sub1
<input type="checkbox"/>	35318885002				FP_SUT	None	PNP_Sub1
<input type="checkbox"/>	35318885003				FP_SUT	None	PNP_Sub1
<input type="checkbox"/>	35318885004				FP_SUT	None	PNP_Sub1
<input type="checkbox"/>	35318885005				FP_SUT	None	PNP_Sub1
<input type="checkbox"/>	35318885006				FP_SUT	None	PNP_Sub1
<input type="checkbox"/>	35318885007				FP_SUT	None	PNP_Sub1
<input type="checkbox"/>	35318889999		+353 (1) 888 9999		FP_QUEUE	None	CNP_SUT

8.2. Configure IBM Sametime Unified Telephony Users

The screen below shows the configured Sametime Unified Telephony **User List Logins** along with their corresponding **Unified Number** as configured in **Section 8.1**.

User List
List of all users within selected domain

Filter: for Login |

9 Items

<input type="checkbox"/>	Login	Name	Unified Number
<input type="checkbox"/>	administrator	administrator	
<input type="checkbox"/>	interop5000#ie.ibm.com	Interop5000	+35318885000
<input type="checkbox"/>	interop5001#ie.ibm.com	Interop5001	+35318885001
<input type="checkbox"/>	interop5002#ie.ibm.com	Interop5002	+35318885002
<input type="checkbox"/>	interop5003#ie.ibm.com	Interop5003	+35318885003
<input type="checkbox"/>	interop5004#ie.ibm.com	Interop5004	+35318885004
<input type="checkbox"/>	interop5005#ie.ibm.com	Interop5005	+35318885005
<input type="checkbox"/>	interop5006#ie.ibm.com	Interop5006	+35318885006
<input type="checkbox"/>	interop5007#ie.ibm.com	Interop5007	+35318885007

8.3. Configure SIP Trunk

The screen below shows the **Endpoints** configured on Sametime Unified Telephony. The item highlighted **EP_pbx1** represents the SIP trunk to Session Manager, the IP address in the **Primary** column relates to the Session Manager SIP Signaling Interface.

[sutInterop] - [SUT] - [Main Office] - Endpoints
Endpoints represent Network to Network Interface connections.

Search for: in No Criteria Elements Per Page: 50

3 Items

<input type="checkbox"/>	Name	Numbering Plan Name	Registration Type	Registration State	Primary	Remark
<input type="checkbox"/>	EP_CONF_BR1	CNP_SUT	Static	Registered	10.10.16.183	No
<input type="checkbox"/>	EP_SOFT_PH1	CNP_SUT	Static	Registered	10.10.16.180	No
<input type="checkbox"/>	EP_pbx1	PNP_Pbx1	Static	Registered	10.10.16.148	No

The screen below shows the configuration of the **EP_pbx1** SIP trunk to Session Manager where:

- **Endpoint Type – SIP Trunking**
- **Under SIP Signaling:**
 - **Type – Static**
 - **Signaling Address Type – IP Address or FQDN**
 - **Endpoint Address – 10.10.16.148** which relates to the Session Manager SIP Signaling Interface
 - **Port – 5060** and **Transport protocol - TCP** as configured in **Section 7.2**

[sutInterop] - [SUT] - [Main Office] - Edit Endpoint : EP_pbx1

General SIP Attributes Aliases Routes Accounting

Endpoint Type

SIP Private Networking:

SIP Trunking:

SIP-Q Signaling:

SIP Signaling

For the static Endpoints the address of the SIP signaling interface can be specified in IP or FQDN format.
Note that the address of the signaling interface cannot be modified unless the entry in the security section has first been removed.

Type: Static

Signaling Address Type: IP Address or FQDN

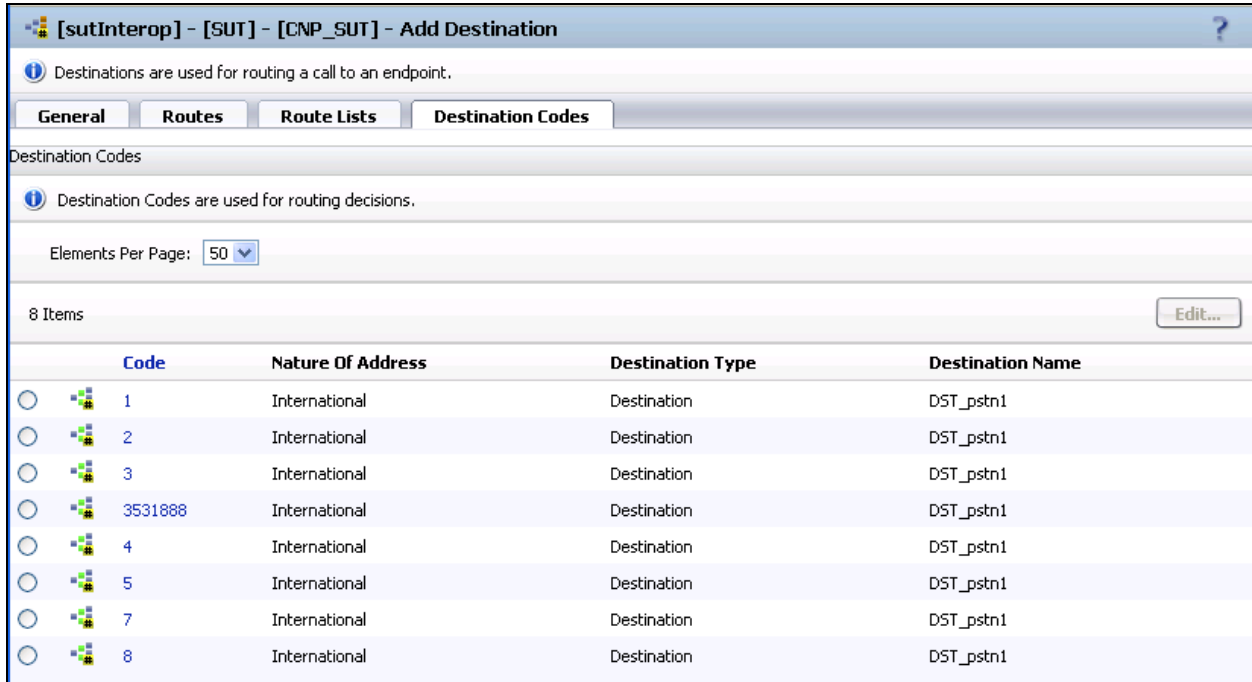
Endpoint Address: 10.10.16.148

Port: 5060

Transport protocol: TCP

8.4. Configure Destinations

Destinations are configured to define how strings dialed from the Sametime Connect Client are routed. The screen below shows the routing configured on Sametime Unified Telephony where DST_pstn1 is the same as EP_pbx1 configured in **Section 8.3**.



[sutInterop] - [SUT] - [CNP_SUT] - Add Destination

Destinations are used for routing a call to an endpoint.

General Routes Route Lists Destination Codes

Destination Codes

Destination Codes are used for routing decisions.

Elements Per Page: 50

8 Items [Edit...](#)

	Code	Nature Of Address	Destination Type	Destination Name
<input type="radio"/>	1	International	Destination	DST_pstn1
<input type="radio"/>	2	International	Destination	DST_pstn1
<input type="radio"/>	3	International	Destination	DST_pstn1
<input type="radio"/>	3531888	International	Destination	DST_pstn1
<input type="radio"/>	4	International	Destination	DST_pstn1
<input type="radio"/>	5	International	Destination	DST_pstn1
<input type="radio"/>	7	International	Destination	DST_pstn1
<input type="radio"/>	8	International	Destination	DST_pstn1

9. Verification Steps

The correct installation and configuration of the Sametime Unified Telephony solution can be verified by performing the following steps shown below.

9.1. Verify Communication Manager SIP Trunk

Using the SAT terminal, enter the **status signaling-group <n>** command, where <n> is the number of the SIP signaling group which connects to Session Manager. Verify that the signaling **Group State** is **in-service**.

```
status signaling-group 1
                        STATUS SIGNALING GROUP

      Group ID: 1
      Group Type: sip

      Group State: in-service
```

9.2. Verify Sametime Connect Softphone Functionality

Place a call to/from a Sametime Connect user, ensure the call can be answered, controlled and terminated as expected using a variety of preferred devices. Leave a voicemail on the mailbox of the Sametime Connect extension, ensure that the message is delivered to the correct mailbox and can be retrieved using the Sametime Connect Softphone.

10. Conclusion

These Application Notes describe the compliance testing of the IBM Sametime Unified Telephony with Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Avaya Modular Messaging. Sametime Unified Telephony passed all of the tests performed with observations noted in **Section 2.2**.

11. References

This section references documentation relevant to these Applications. Avaya product documentation, including the following, is available at <http://support.avaya.com>

- *Administering Avaya Aura® Communication Manager, Release 6.2*, 03-300509, Issue 7.0 December 2012
- *Implementing Avaya Aura® Session Manager, Release 6.2*, 03-603473, July 2012

IBM Sametime Unified Telephony documentation and support can be accessed using the details in **Section 2.3**.

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