**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:

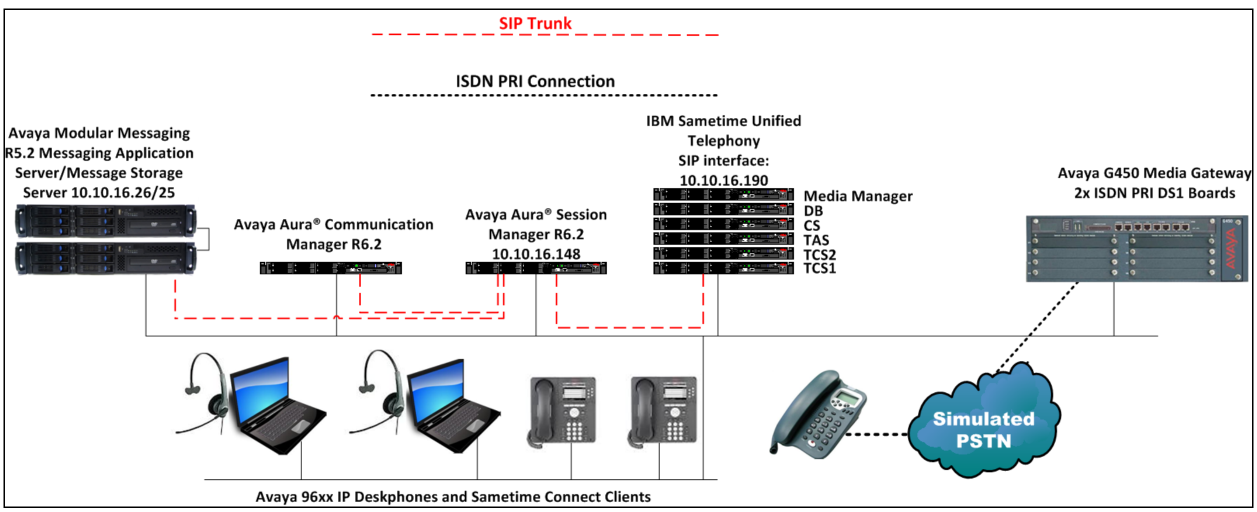
- Where g.729 is configured as the preferred codec on Sametime Unified Telephony, Communication Manager must be configured with only g.729 in order for g.729 to be utilized.
- It was not possible to successfully test TCS failover.

## 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 | • 5.2 Patch 8<br>• MAS - 9.2.150.13 |
| Avaya G450 Media Gateway<br>• MM710<br>• MM712 | 31.22.0<br>• HW5 FW22<br>• HW7 FW14 |
| Avaya 9630 IP Deskphone | • H323 S3.1 SP5<br>• 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
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
   2           10  udp
   3           11  udp
   4           4   udp
   5           4   ext
   6           4   ext
   7           3   dac
   8           4   udp
   9           1   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                         Insert                Node
  Pattern         Len Del          Digits        Net Conv Num
 2                 10  0                         ars  n
 3                 11  0                         aar  n
 4                 4   0                         aar  n
 8                 4   0                         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           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    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

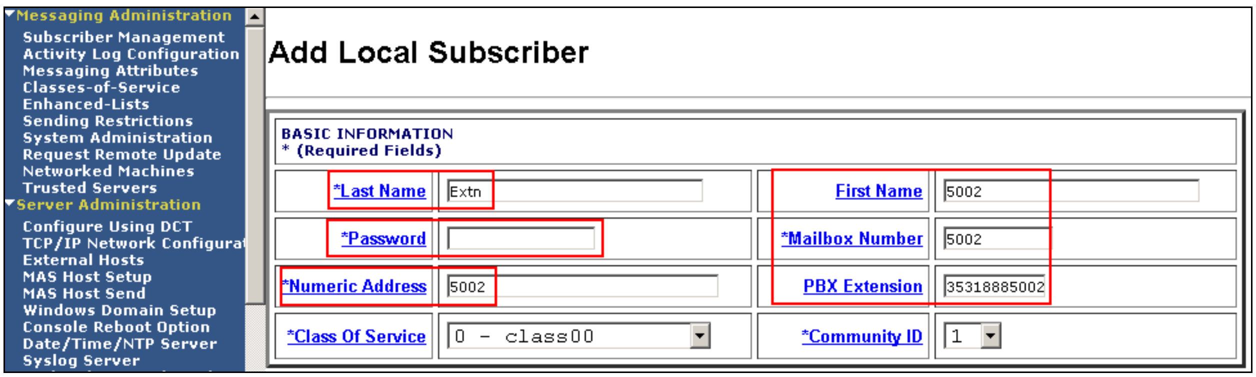
It is assumed the necessary signaling group and trunk configuration has been completed for the interface between Communication Manager and Session Manager. Enter the command **change signaling-group x** where **x** is the signaling group relevant to the trunk between Communication Manager and Session 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:
                                           Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                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
```

# 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)

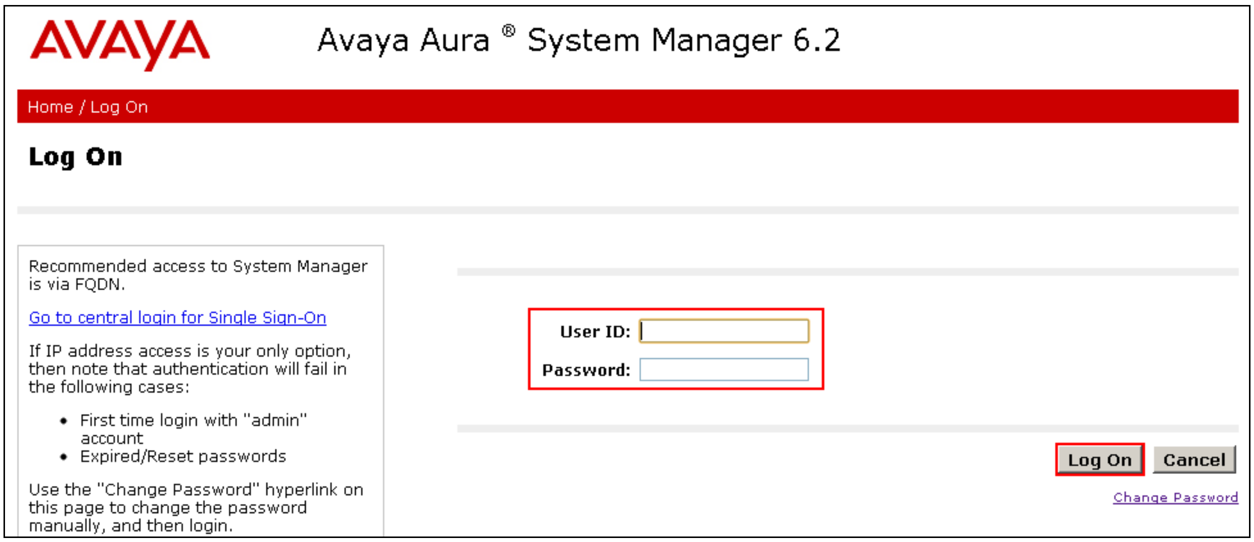
- **Last Name** – an identifying last name
- **First Name** – an identifying first name
- **Password** – a numeric password
- **Mailbox Number & Numeric Address** – enter the extension number configured on SUT
- **PBX Extension** – enter the full number dialed for the SUT user – in this case **35318885002**.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



# 7. Configure Avaya Aura® Session Manager

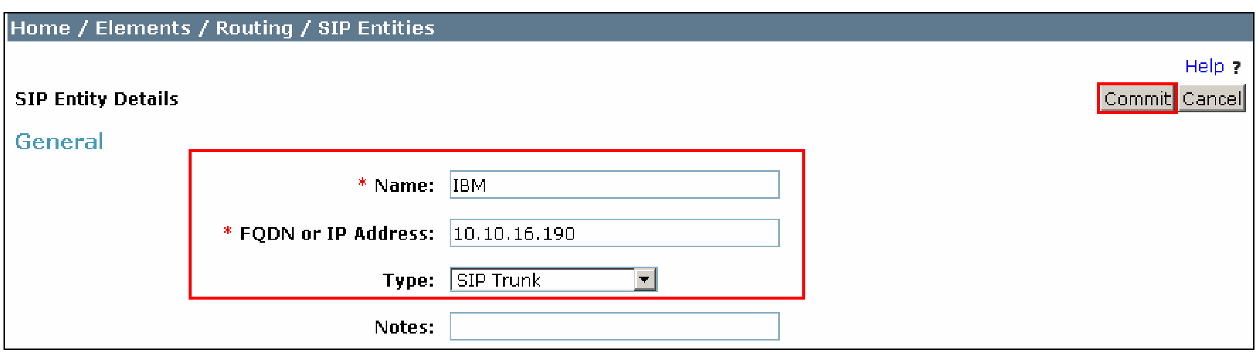
This section illustrates relevant aspects of the Avaya Aura® Session Manager configuration required for interoperating with Sametime Unified Telephony.

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



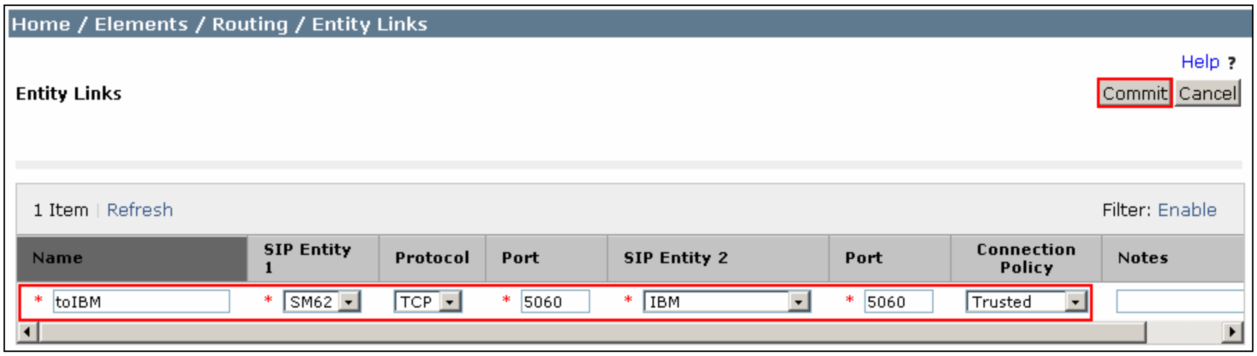

## 7.1. Configure IBM Sametime Unified Telephony SIP Entity

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.



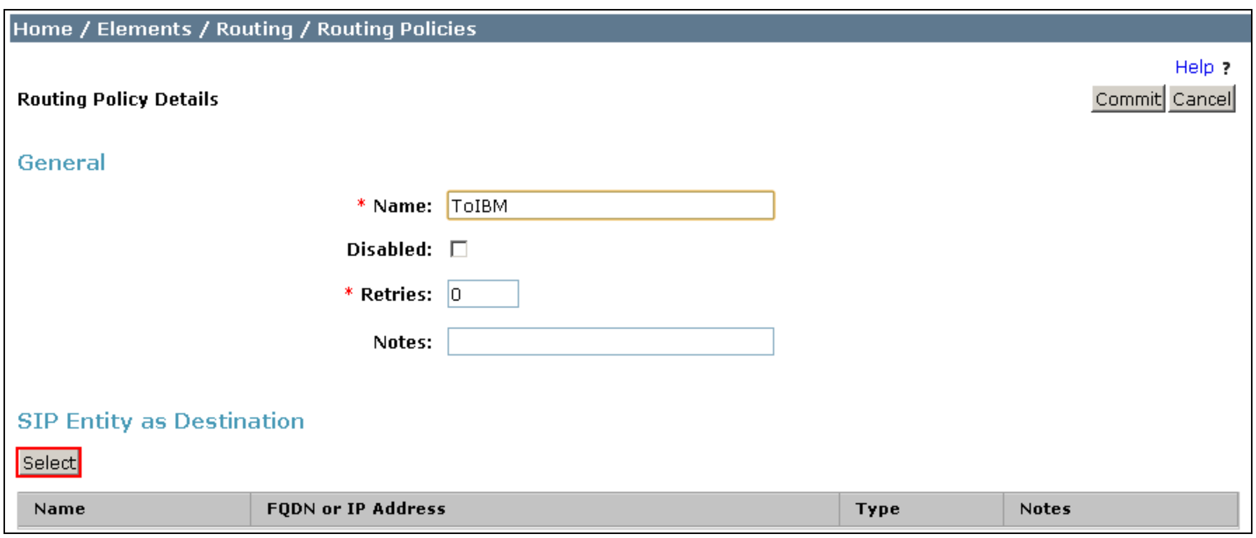
## 7.2. Configure Entity Link

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.

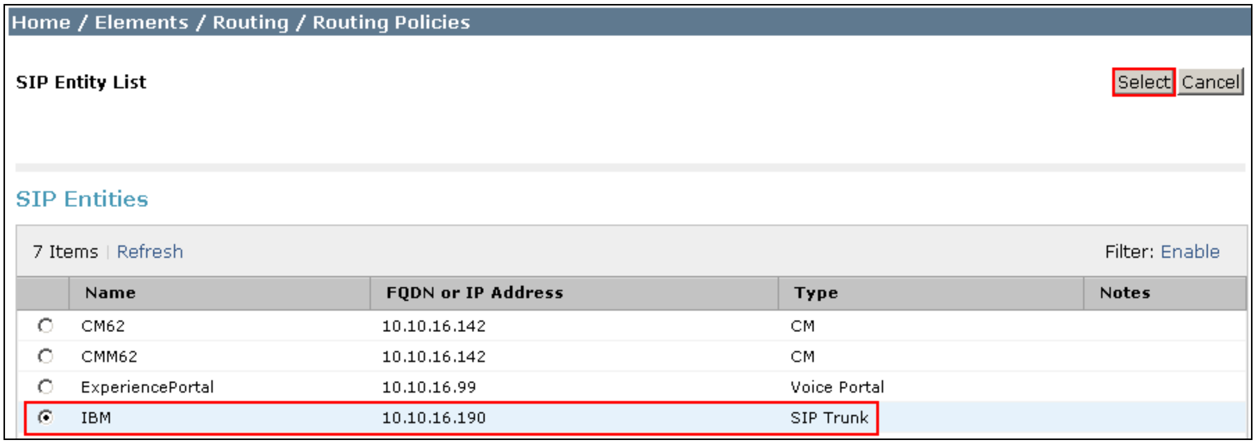



## 7.3. Create Routing Policy

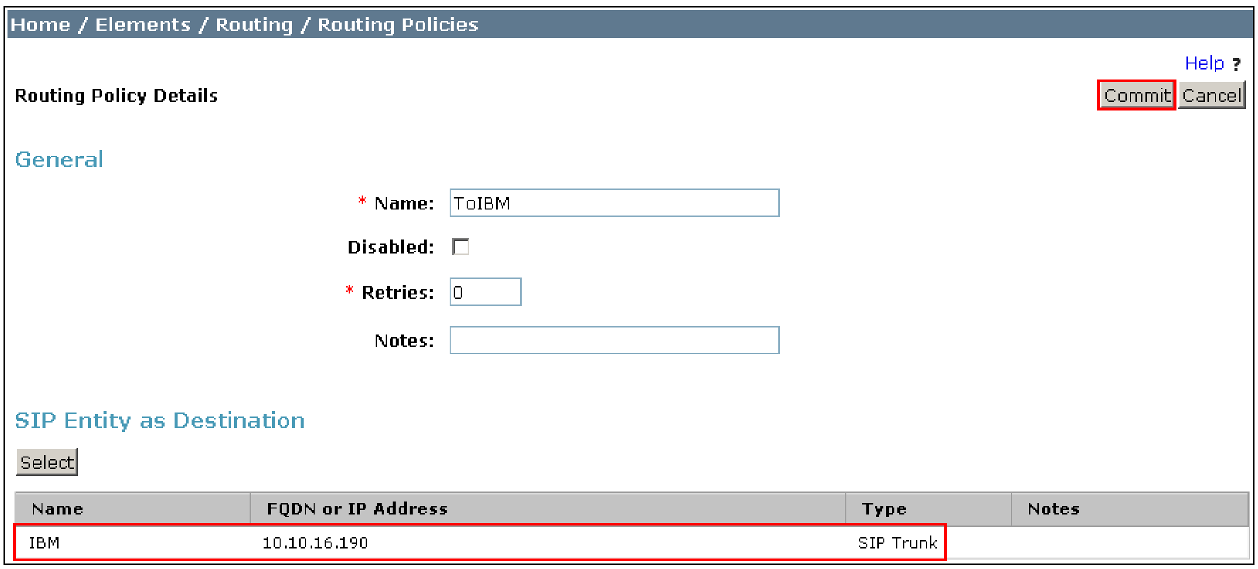
Click **Routing → Routing Policies → New** (not shown). Enter a descriptive **Name** for the route. Under the **SIP Entity as Destination** section, click on **Select**.



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

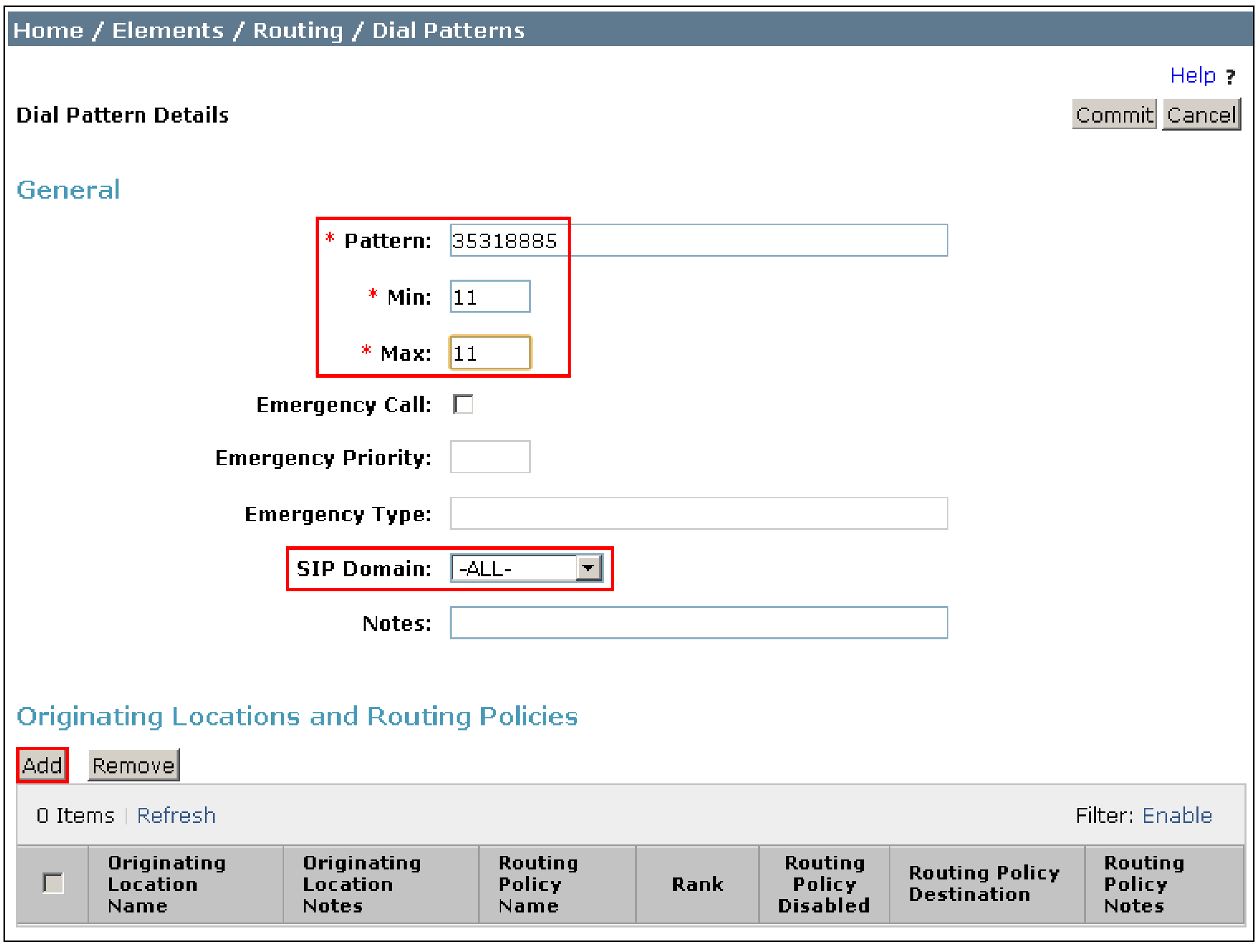



Review the configuration and click **Commit** when done.

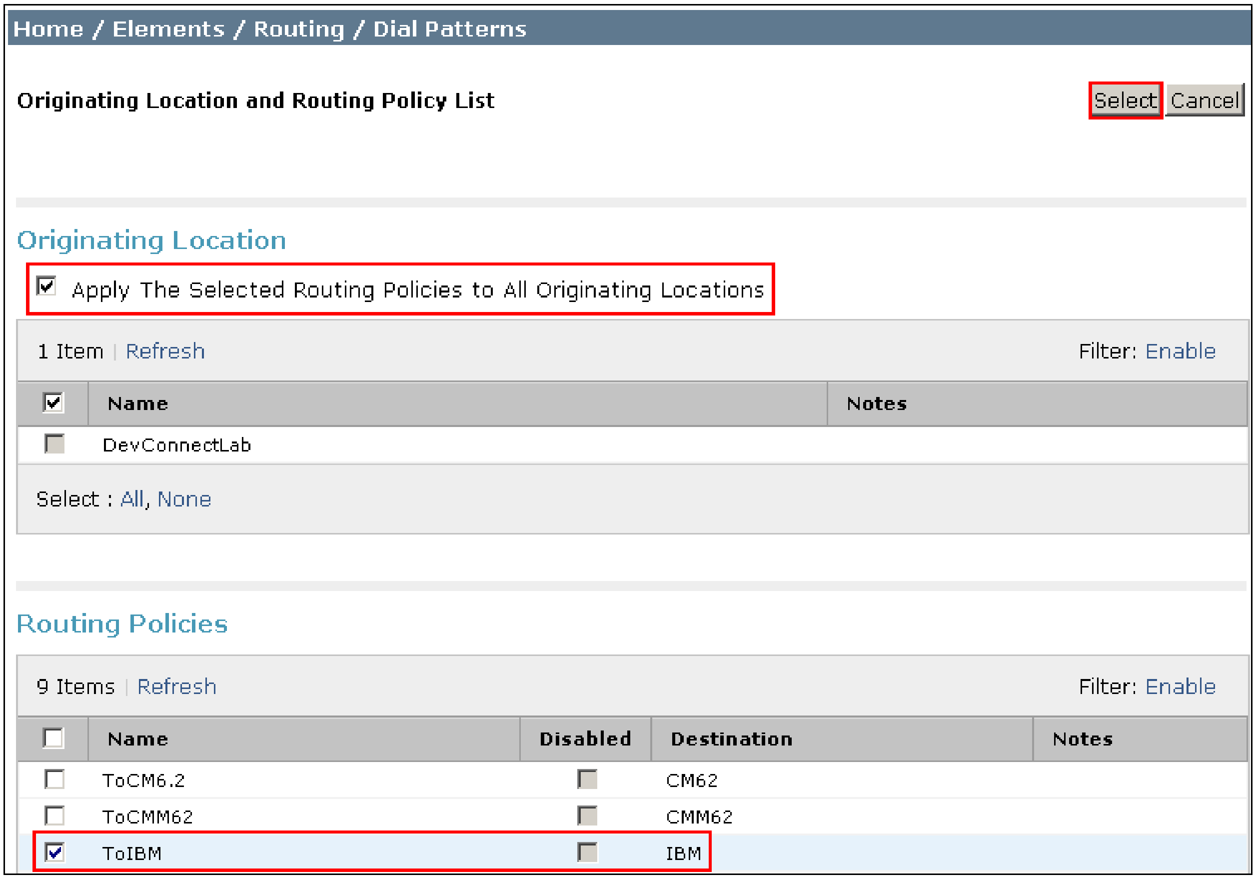


## 7.4. Administer Dial Patterns

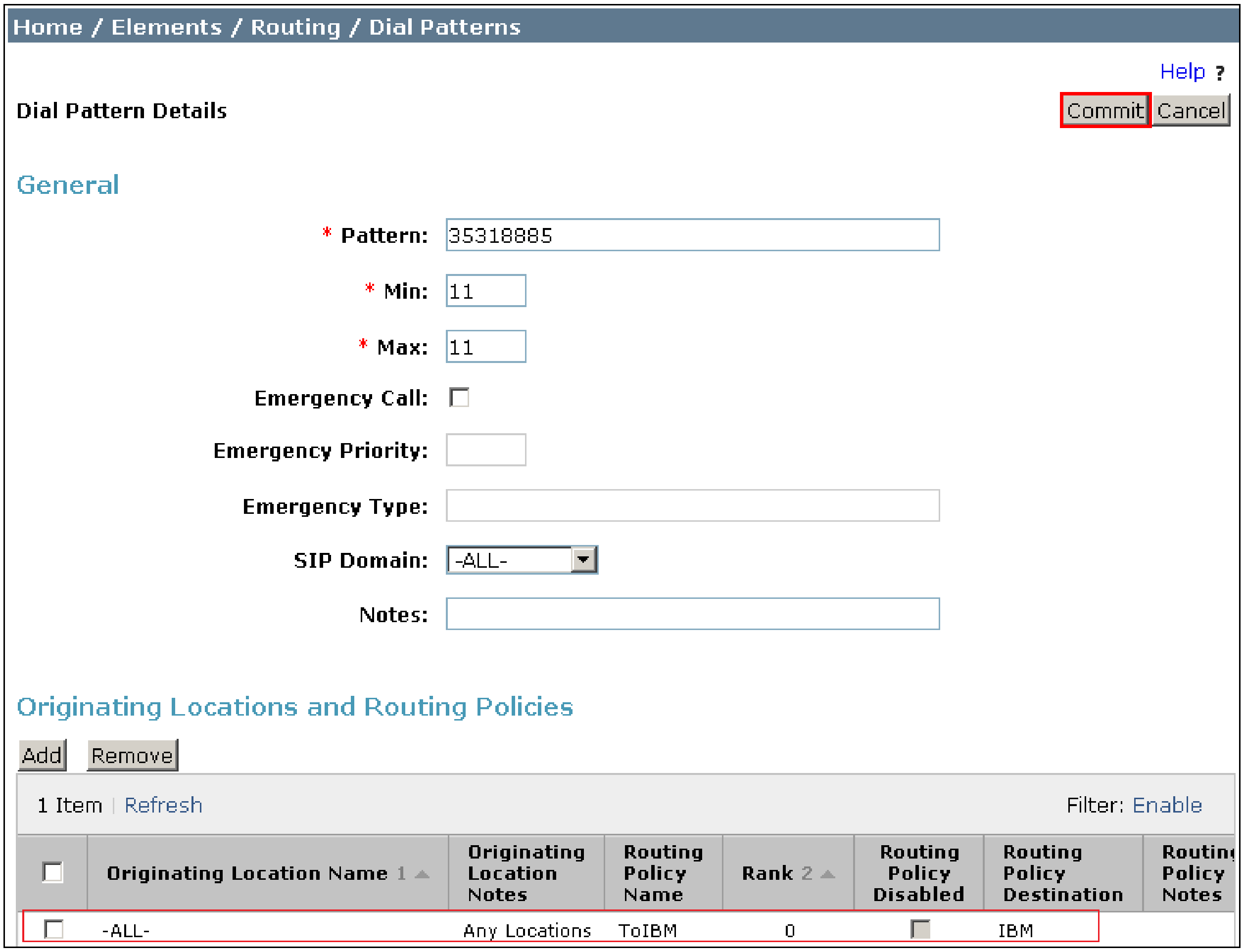
Session Manager routes SIP traffic between connected devices. Dial Patterns are created as part of the configuration to manage SIP traffic routing, which will direct calls based on the number dialed to the appropriate destination. In **Section 5.2** Communication Manager is configured to route 11 digit 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**.



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

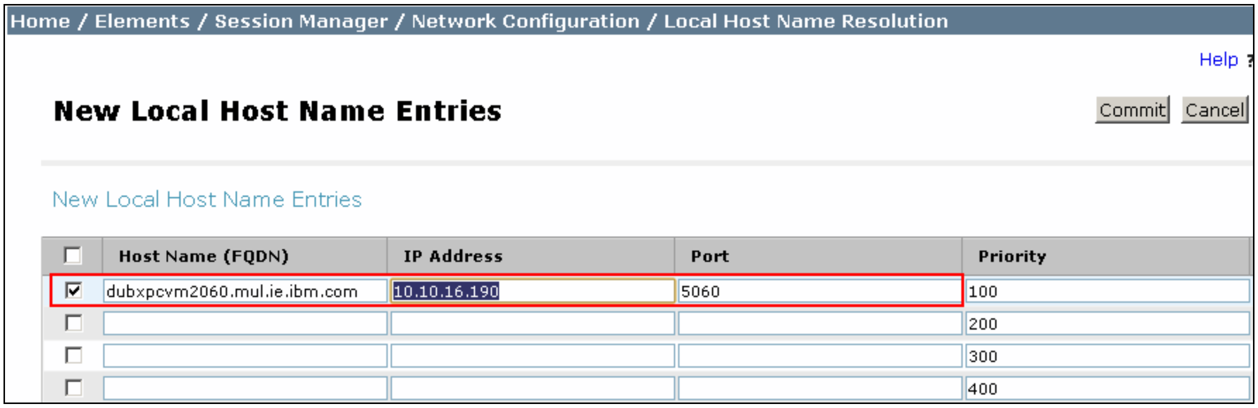


Review the configuration and click **Commit** when done.

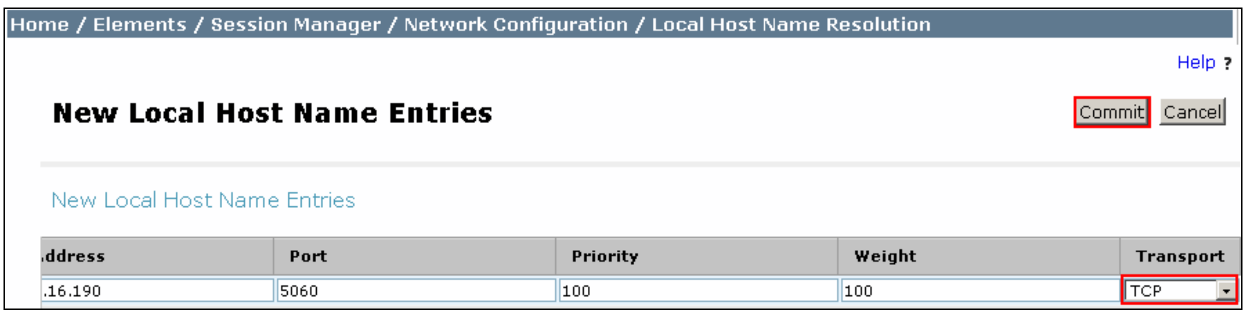


## 7.5. Configure Local Host Name Resolution

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.




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



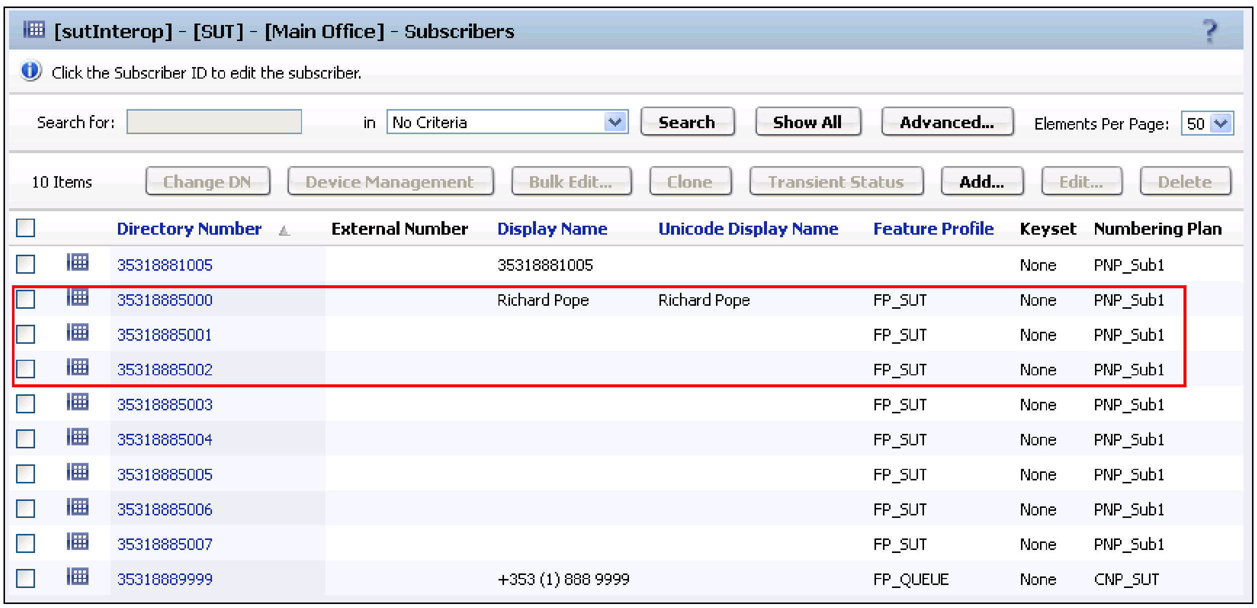
# 8. Configure IBM Sametime Unified Telephony

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

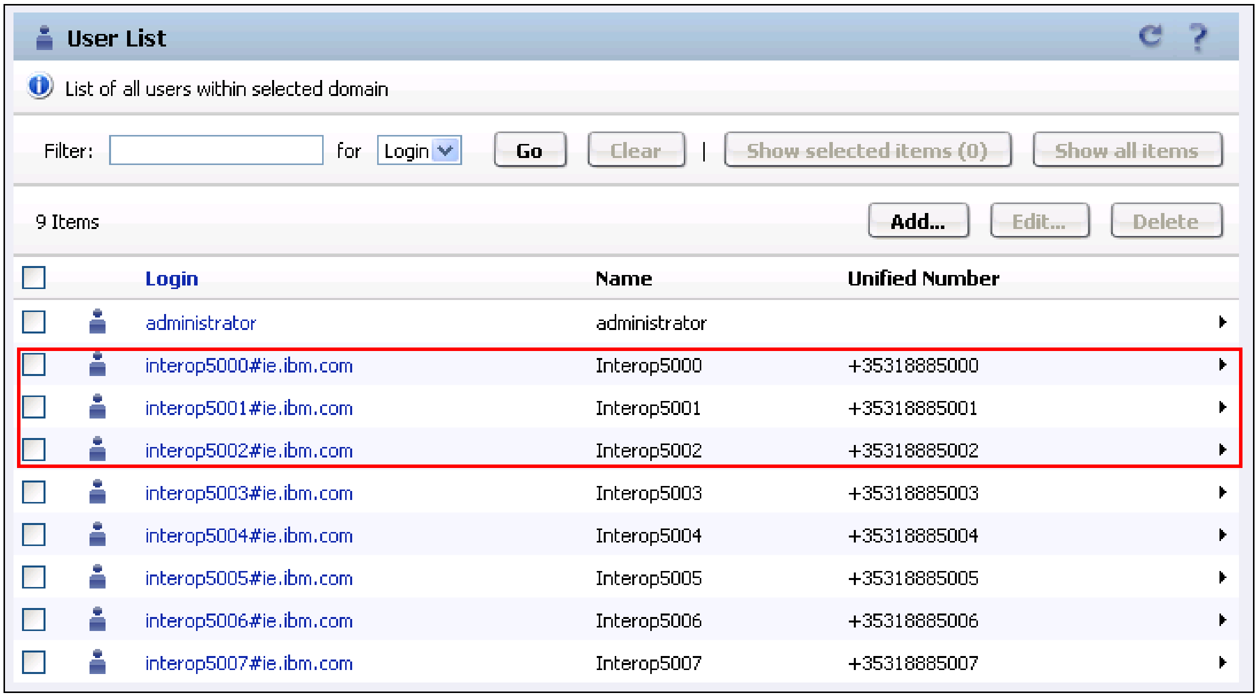
Unified Numbers must be configured for each user on the Sametime Unified Telephony platform. The screen below shows the configured **Directory Numbers** which are configured in **Section 7** to route to Sametime Unified Telephony.




| Directory Number | External Number | Display Name | Unicode Display Name | Feature Profile | Keyset | Numbering Plan |
|---|---|---|---|---|---|---|
| 35318881005 | | 35318881005 | | | None | PNP_Sub1 |
| 35318885000 | | Richard Pope | Richard Pope | FP_SUT | None | PNP_Sub1 |
| 35318885001 | | | | FP_SUT | None | PNP_Sub1 |
| 35318885002 | | | | FP_SUT | None | PNP_Sub1 |
| 35318885003 | | | | FP_SUT | None | PNP_Sub1 |
| 35318885004 | | | | FP_SUT | None | PNP_Sub1 |
| 35318885005 | | | | FP_SUT | None | PNP_Sub1 |
| 35318885006 | | | | FP_SUT | None | PNP_Sub1 |
| 35318885007 | | | | FP_SUT | None | PNP_Sub1 |
| 35318889999 | | +353 (1) 888 9999 | | FP_QUEUE | None | CNP_SUT |

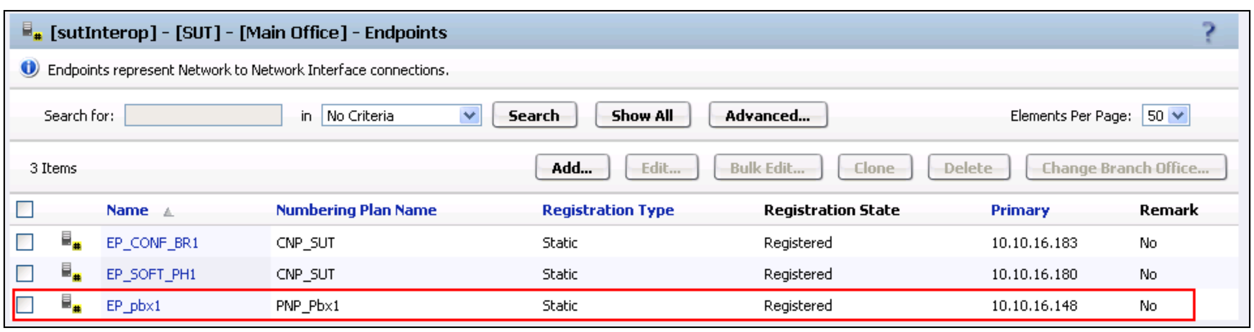
## 8.2. Configure IBM Sametime Unified Telephony Users

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



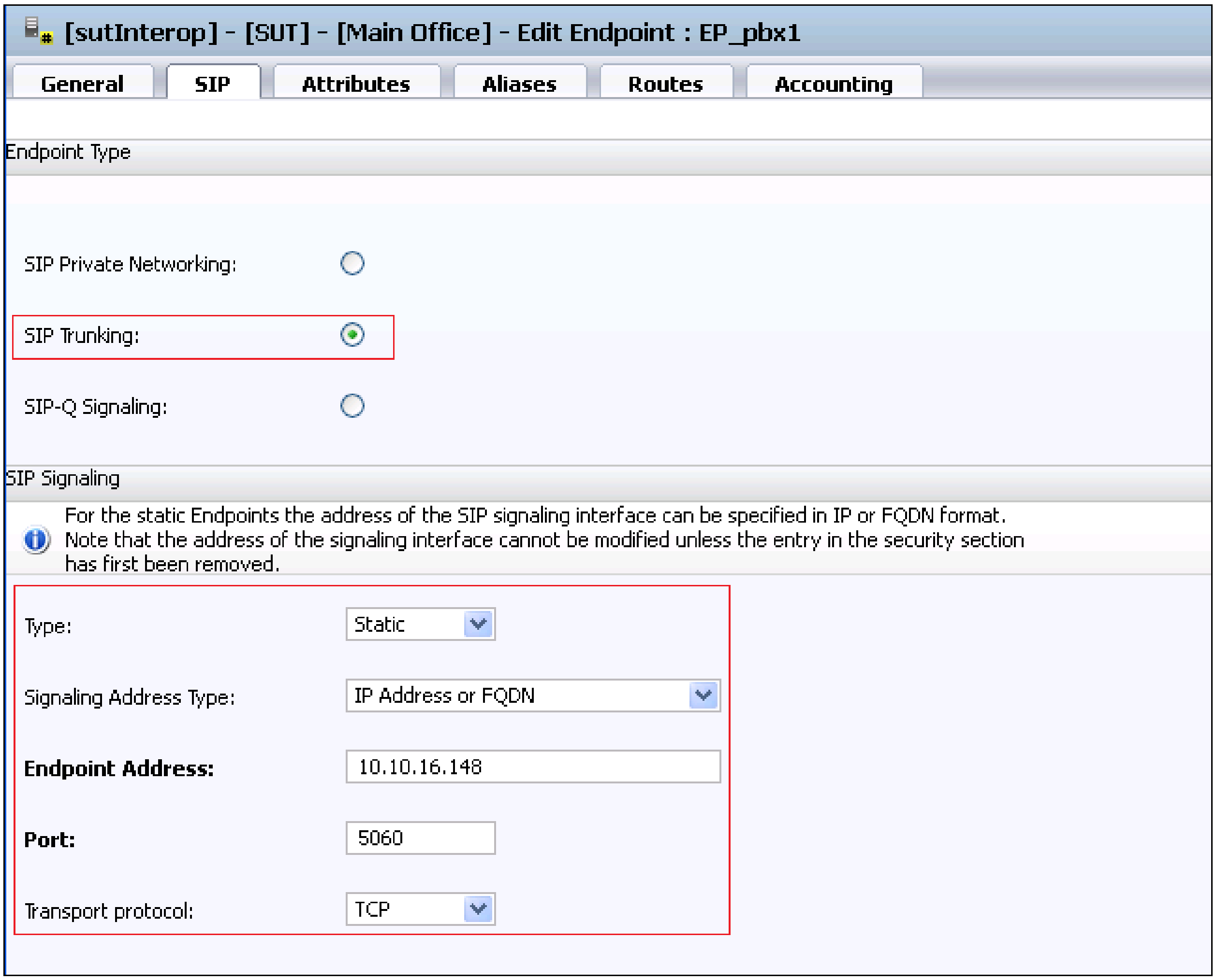

## 8.3. Configure SIP Trunk

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.



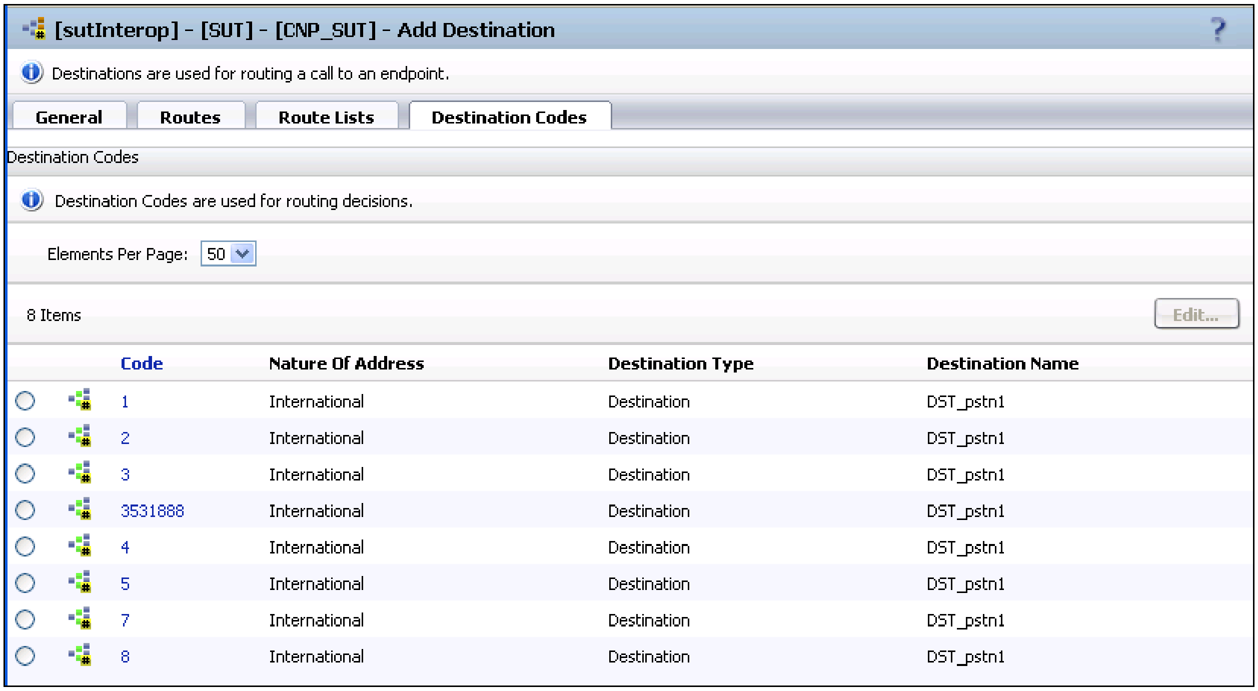
The screen below shows the configuration of the **EP_pbx1** SIP trunk to Session Manager where:

- **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**



## 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 |
|---|---|---|---|
| 1 | International | Destination | DST_pstn1 |
| 2 | International | Destination | DST_pstn1 |
| 3 | International | Destination | DST_pstn1 |
| 3531888 | International | Destination | DST_pstn1 |
| 4 | International | Destination | DST_pstn1 |
| 5 | International | Destination | DST_pstn1 |
| 7 | International | Destination | DST_pstn1 |
| 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**.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.