Avaya Solution & Interoperability Test Lab

Application Notes for Tango Networks Abrazo 5.3 with Avaya Communication Server 1000E 7.5, Avaya Aura® Session Manager 6.2 and Avaya CallPilot™ 5.0 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Tango Networks Abrazo 5.3 to interoperate with Avaya Communication Server 1000E 7.5, Avaya Aura® Session Manager 6.2 and CallPilot™ 5.0. The Abrazo solution extends enterprise PBX functionality to mobile devices, allowing end users to be accessible when out of the office. The Abrazo solution integrates mobile devices with existing Private Branch Exchanges (PBXs) so that the PBX sees the mobile device as another desk phone. This allows the existing PBX feature set to be applied consistently across both devices. Mobile specific functionality is then layered on top.

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 Tango Networks Abrazo 5.3, Avaya Communication Server 1000E 7.5 (Avaya CS 1000E), Avaya Aura® Session Manager 6.2 and Avaya IP Deskphones.

The Abrazo solution extends enterprise PBX functionality to mobile devices, allowing end users to be accessible when out of the office. The Abrazo solution integrates mobile devices with existing Private Branch Exchanges (PBXs) so that the PBX sees the mobile device as another desk phone. This allows the existing PBX feature set to be applied consistently across both devices. Mobile specific functionality is then layered on top.

The Tango Networks Abrazo Solution includes the Abrazo-C and the Abrazo-E components. As shown in **Figure 1**, the Abrazo-C communicates with the mobile operator network using standard protocols and always resides in the mobile operator's network or a hosting center. The Abrazo-E communicates with the enterprise network components including the PBX, voice mail systems, and corporate databases via standard interfaces to extend the enterprise network functionality transparently to the mobile network.

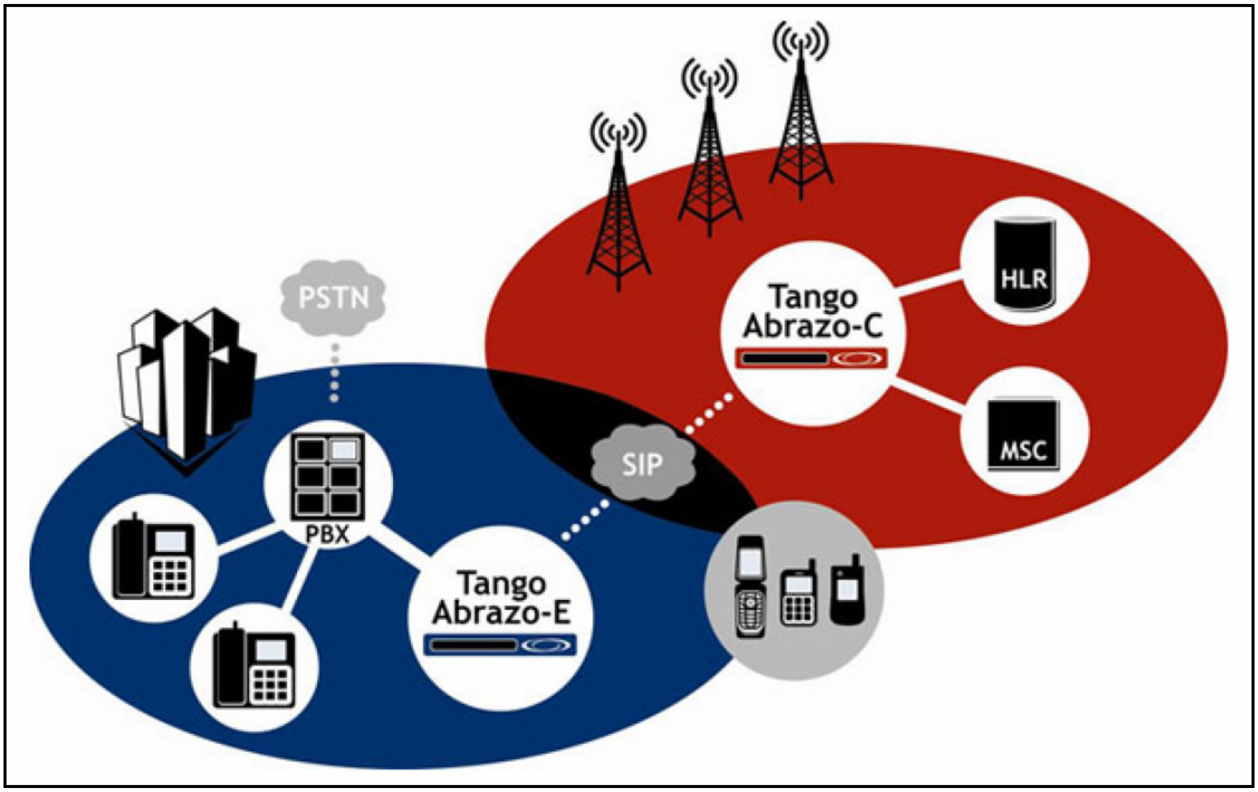


Figure 1: Tango Networks' Architecture Diagram

The Abrazo-E can potentially be provisioned in one of three ways based on your Abrazo-E license key. Your license key dictates whether your enterprise has the ability to enable Mobile UC and/or PSTN Access (SIP Trunking) functionality.

During Abrazo-E provisioning, a Carrier(s) was created that enabled one or both of these services. How you integrate your Avaya CS1000 7.5 PBX with the Abrazo solution depends on how your Carrier(s) is configured on the Abrazo-E.

- **Mobile UC** - The Mobile UC application extends PBX and Unified Communications (UC) features to mobile devices. Examples include Single Number, Single Voicemail, Abbreviated Dialing, and Presence Status. For Mobile UC, the Abrazo uses a combination of SIP lines and trunks to integrate with the Avaya CS1000 7.5 PBX. SIP lines are used so that Abrazo-controlled mobile devices appear as standard SIP phones and therefore benefit from the common set of PBX services offered to such devices. SIP trunks are used when the Abrazo must terminate a call via the Public Switched Telephone Network (PSTN).

- **PSTN Access** (e.g. SIP Trunking Controller) – PSTN Access facilitates interworking between enterprise and SIP entities such as PBXs and PSTN carriers (i.e. SIP Trunking Service Providers) as well as between internal enterprise SIP entities. For PSTN Access, the Abrazo uses only SIP trunk(s) to integrate with the Avaya CS1000 7.5 PBX.

- **Mobile UC and SIP Trunking combination** – It is possible to have *both* the Mobile UC functionality as well as the PSTN Access functionality enabled on your Abrazo-E.

2. General Test Approach and Test Results

The compliance testing focused on the interoperability between the Tango Networks' Abrazo solution and Avaya CS 1000E to ensure that the Mobile phones and Avaya IP Deskphones function as expected.

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

Testing consisted of typical call scenarios involving mobile originating and mobile terminating calls routed through the Avaya CS 1000E telephony infrastructure. All test cases were performed manually. In addition, testing entailed verifying different types of Avaya telephones and system features interacting with the Tango Abrazo solution. Tests were performed focusing on the following calling patterns:

- mobile originated calls routed through the Avaya CS 1000E telephony infrastructure terminating to a desk phone, mobile device, or the PSTN
- mobile terminated calls routed through the Avaya CS 1000E telephony infrastructure
- desktop originated calls routed to mobile devices

Feature testing included, unanswered calls, Abbreviated Dialing, Call Forward All, Call Forward Busy, Call Forward No Answer, Call Hold and Retrieve, Calling Line Identification (CLID), abandoned calls, voice mail deposit and retrieve, Message Waiting Indicator (MWI), Call Moves, Blind and Consultative Transfer scenarios and Blind and Consultative Conference scenarios.

2.2. Test Results

Testing of this solution with Tango Networks Abrazo 5.3, Avaya CS1000E 7.5, Session Manager 6.2 and CallPilot 5.0 was successful with the following observations relating to voicemail.

- Voicemail in a Route Direct scenario does not work properly. The call attempts voicemail retrieval instead of deposit. A ticket is open with Avaya for this problem.
- UNISTIM phones that attempt voicemail retrieval must first enter their desk extension followed by the '#' sign then followed by their password. The call does not automatically recognize the subscriber's mail box number. A ticket is open with Avaya for this problem.
- MWI for UNISTIM subscribers does not update MWI on the subscriber's mobile phone. A ticket is open with Avaya for this problem.

2.3. Support

Technical support for Tango networks Abrazo can be obtained through the following:

- **Web site:** http://www.tango-networks.com
- **Email:** sales@tango-networks.com
- **Telephone:** 469-229-2000

3. Reference Configuration

These application notes describe a solution for integrating the Tango Abrazo-E with the Avaya Product Portfolio. **Figure 2** illustrates the configuration used in these application notes. The diagram indicates the logical signaling connections between the Tango Abrazo and Avaya products. The Abrazo-C and mobile carrier are not shown in this diagram because they are out of the scope of this document.

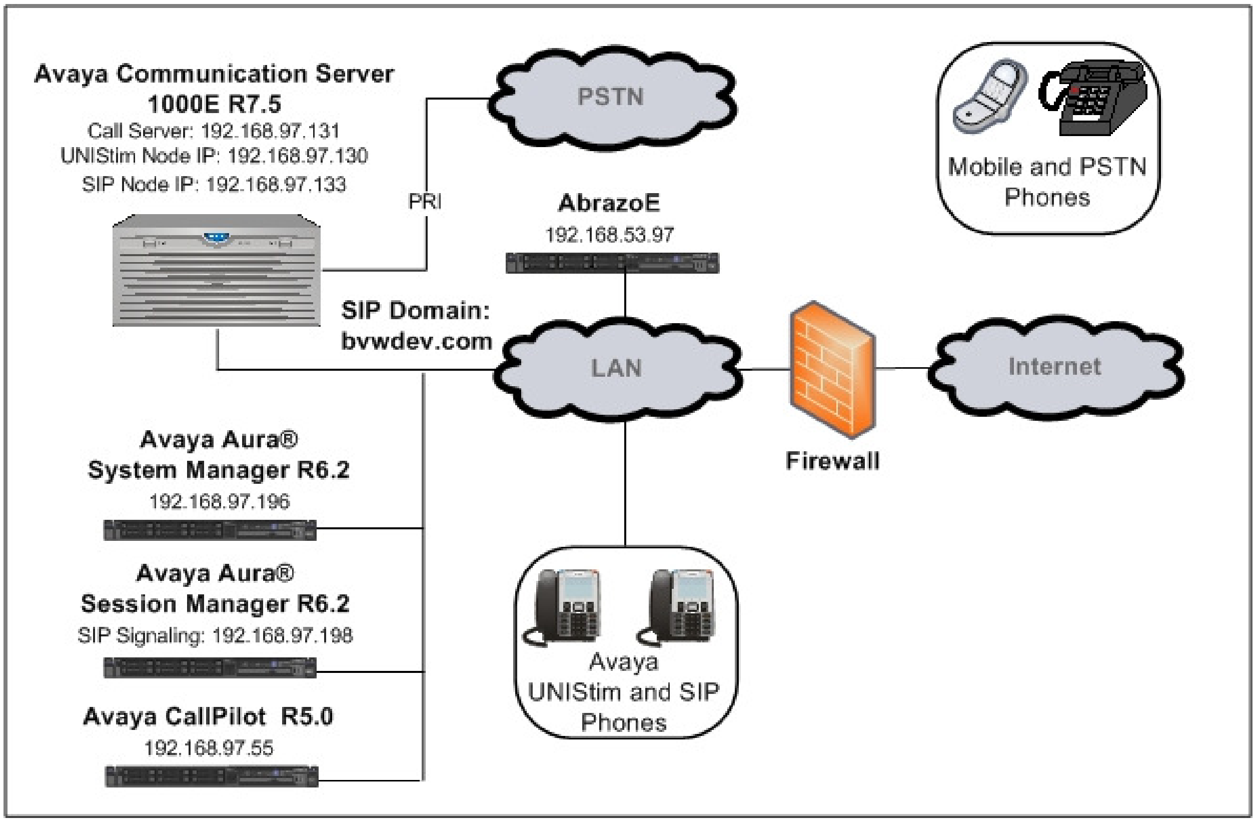


Figure 2: Tango Networks' Abrazo Solution with Avaya Communication Server 1000E

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Communication Server 1000E CPPM co-resident server	Call Server (CPPM): 7.50Q Signaling Server (CPPM): 7.50.17
Avaya Aura® Session Manager running on S8800 Server	Avaya Aura® Session Manager 6.2 Release: 6.2.2.0.622005
Avaya Aura® System Manager running on S8800 Server	Avaya Aura® System Manager 6.2 Build Number 6.2.0.0.15669
Avaya CallPilot™	5.0
Avaya 1120E IP Deskphone (UNIStim)	0624C8L
Avaya 1140E IP Deskphone (UNIStim)	0625C8L
Avaya 1140E IP Deskphone (SIP)	04.03.12.00
Tango Networks Abrazo	5.3

5. Abrazo Integration Processes

The integration with the Abrazo-E can be setup in several ways. During Abrazo-E provisioning, your enterprise selected Carrier types are based on your Abrazo-E enterprise license key. Your license key may be enabled for Mobile UC functionality or it may be enabled for PSTN Access functions, or even *both*.
The sections and steps outlined below will guide your workflow to integrate your Avaya CS1000E with Abrazo-E.

For a Mobile UC and PSTN Access enabled Abrazo-E, execute both sets of steps outlined below in **Sections 6** and **7**. It does not matter which set you execute first, meaning it does not matter if you provision PSTN Access first or Mobile UC first.

6. Provisioning Avaya Products for Mobile Unified Communications

This section covers provisioning for the Session Manager, Avaya CS 1000 and CallPilot for integration with the Abrazo-E Mobile UC services.

6.1. Avaya Aura® Session Manager Provisioning

This section describes the steps required to provision Session Manager to interoperate with Abrazo-E. It is assumed that Session Manager has already been installed and is functioning. For more information refer to documents listed in **Section 11**.

The following is a summary of the steps to provision Session Manager to interoperate with the Abrazo-E.

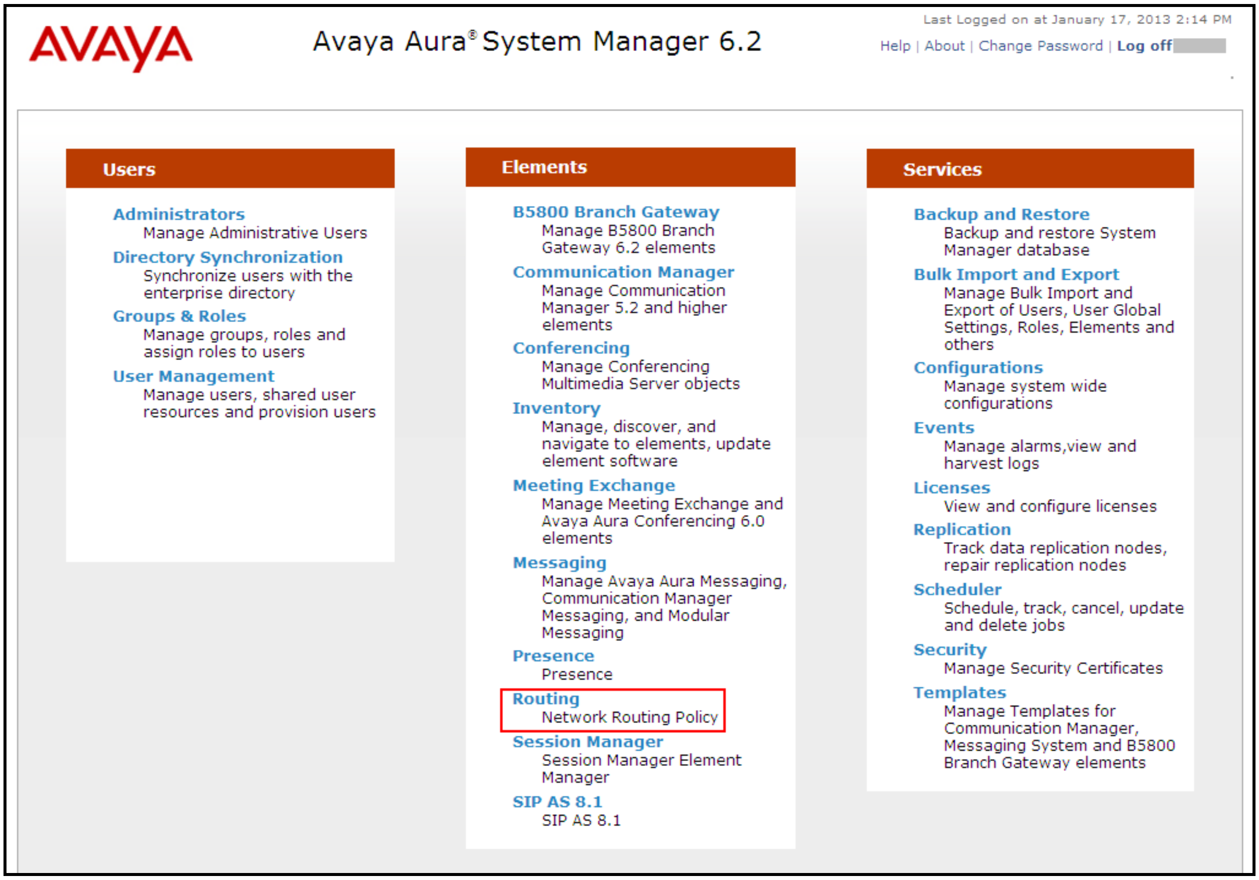
- Define SIP Domains
- Define Locations
- Define SIP Entities:
  - Abrazo-E
  - Avaya CS1000 PBX
- Define Entity Links:
  - CS1000 – TCP and UDP Entity Links
  - Abrazo-E – TCP and UDP Entity Links
- Define Routing Policy
- Define Dial Pattern

6.1.1. Avaya Aura® Session Manager Provisioning Access

This section provides the procedures for configuring Session Manager to route calls between the Avaya CS1000 and the Abrazo-E.

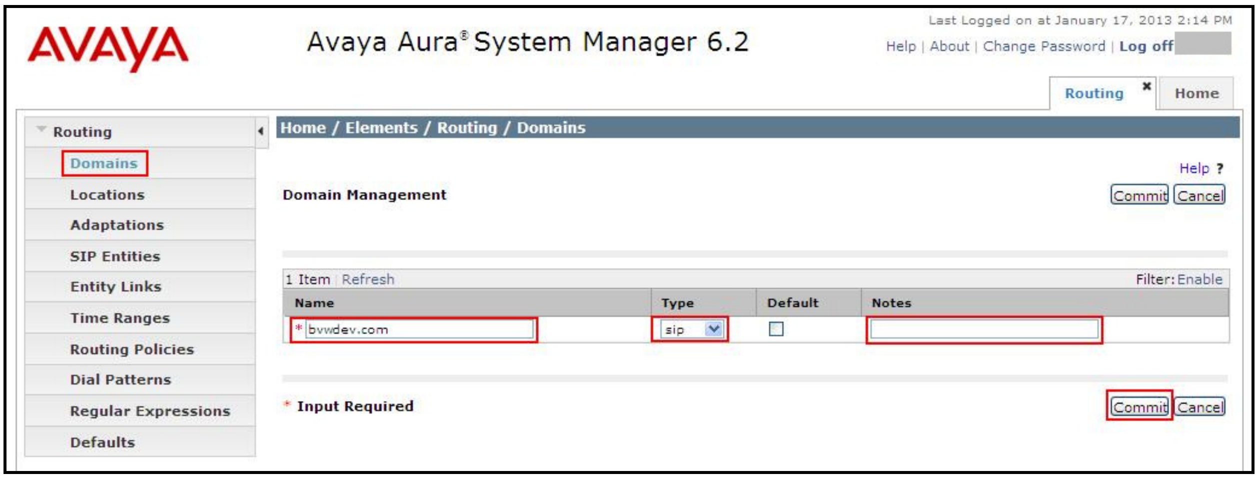
Access the browser-based GUI of System Manager, using the **URL http://<FQDN >/SMGR**, where <FQDN> is the fully qualified domain name of System Manager. Log in to System Manager with the appropriate credentials (not shown).

From the main System Manager page click on the **Routing** link as shown below.



6.1.2. Define SIP Domains

1. Expand the **Routing** menu topic and select **Domains** as shown below.
2. Click **New** (not shown). Enter the following values and use default values for remaining fields:
   - **Name** – Enter the Authoritative Domain Name. In the sample configuration, **bvwdev.com** was used.
   - **Type** – Select '**sip**'.
   - **Notes** – *(optional)* Add a description.
3. **Click Commit to save**. The figure below shows the SIP Domain defined for the bvwdev.com domain.

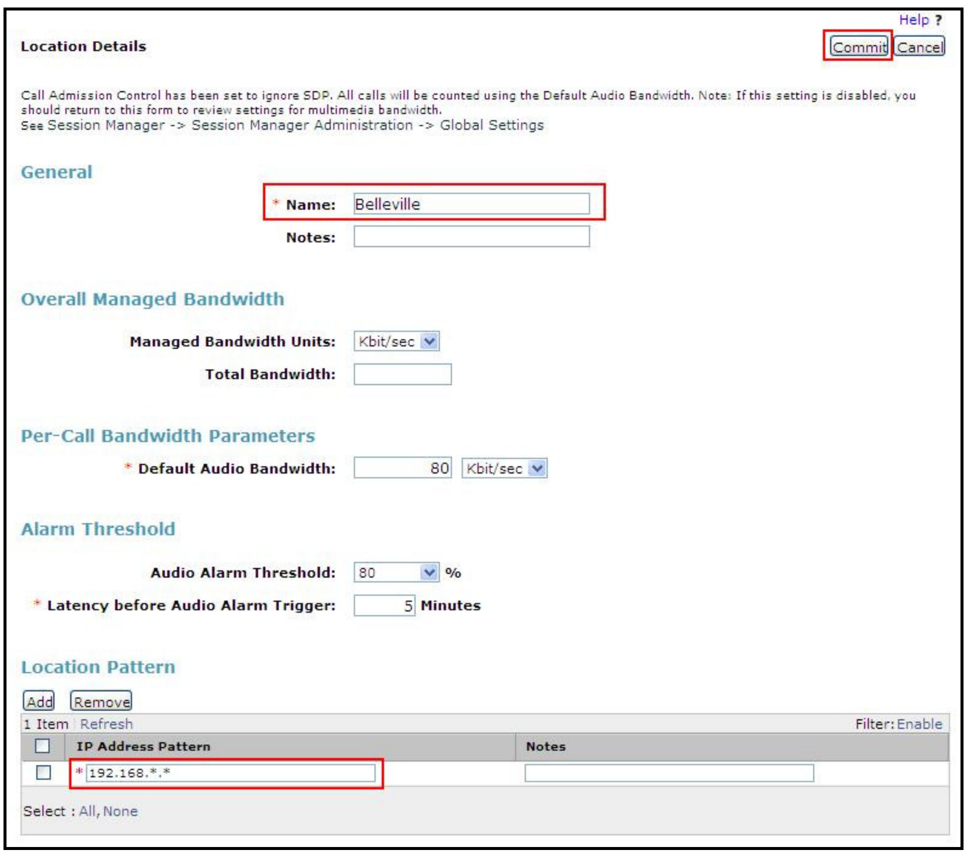


6.1.3. Define Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth or location-based routing.

1. Expand **Routing** and select **Locations** from the left menu.
2. Click New (not shown).
3. In the **General** section, enter the following values and use default values for remaining fields:
    - **Name** – Enter a descriptive name for the location.
    - **Notes** – *(optional)* – Add a brief description
4. In **the Location Pattern** section, click **Add** and enter the following values:
    - **IP Address Pattern** – Enter the logical pattern used to identify the location. For this sample configuration, **192.168.*.*** was used.
    - **Notes** – *(optional)* – Add a brief description.
5. Click **Commit** to save.

The figure below shows the Location defined in this sample configuration.

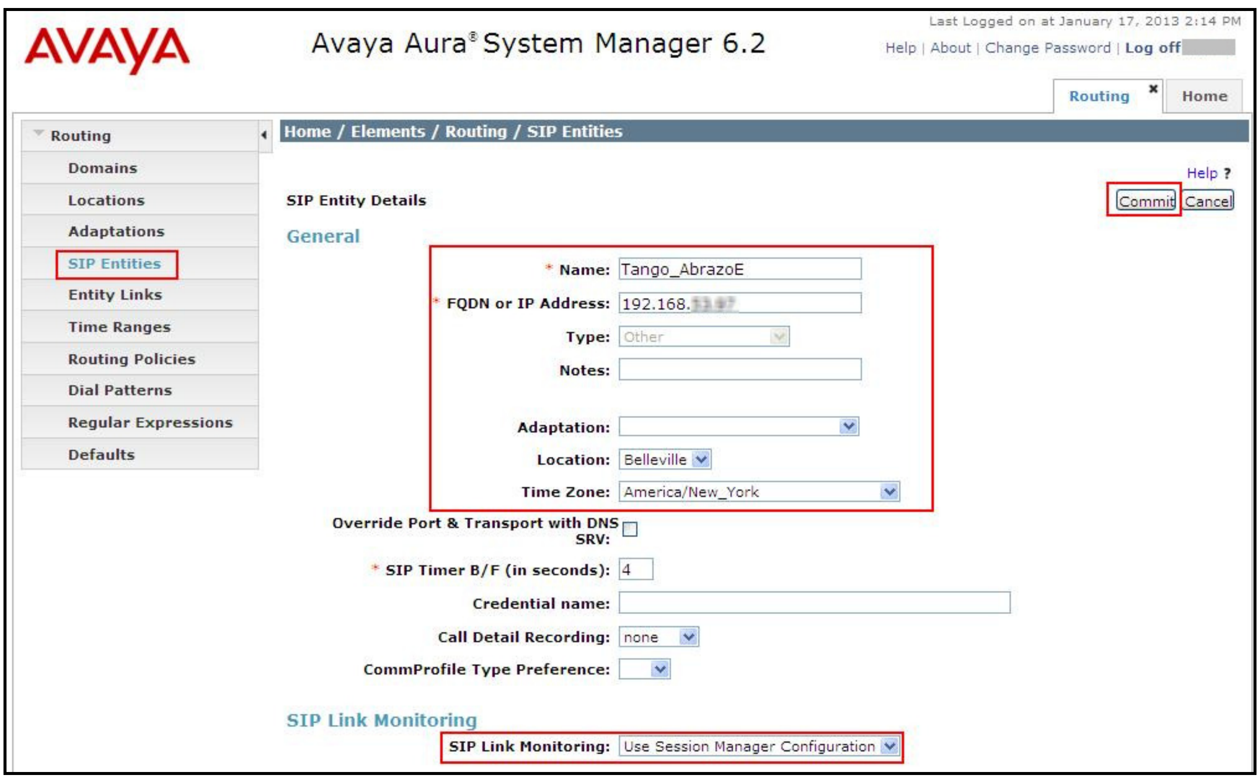


6.1.4. Define SIP Entities – Abrazo-E

A SIP Entity must be added for each telephony system connected to the Session Manager over SIP trunks.

1. Define the **Tango Abrazo-E SIP Entity** by expanding the **Routing** topic and select **SIP Entities** from the left menu.
2. Click **New** (not shown).
3. In the **General** section, enter the following values and use default values for remaining fields:
   - **Name** – Enter an identifier for the SIP Entity.
   - **FQDN** or **IP Address** – Enter IP address of the Abrazo-E.
   - **Type** – Select '**Other**'.
   - **Notes** *(optional)* – Enter a brief description
   - **Adaptation** – No adaptation was used.
   - **Location** – Select the Location defined in **Section 6.1.3**.
   - **Time Zone** – Enter the appropriate Time Zone.
4. **SIP Link Monitoring** – Select 'Use Session Manager Configuration'.
5. Click **Commit** to save the definition of the new SIP Entity.

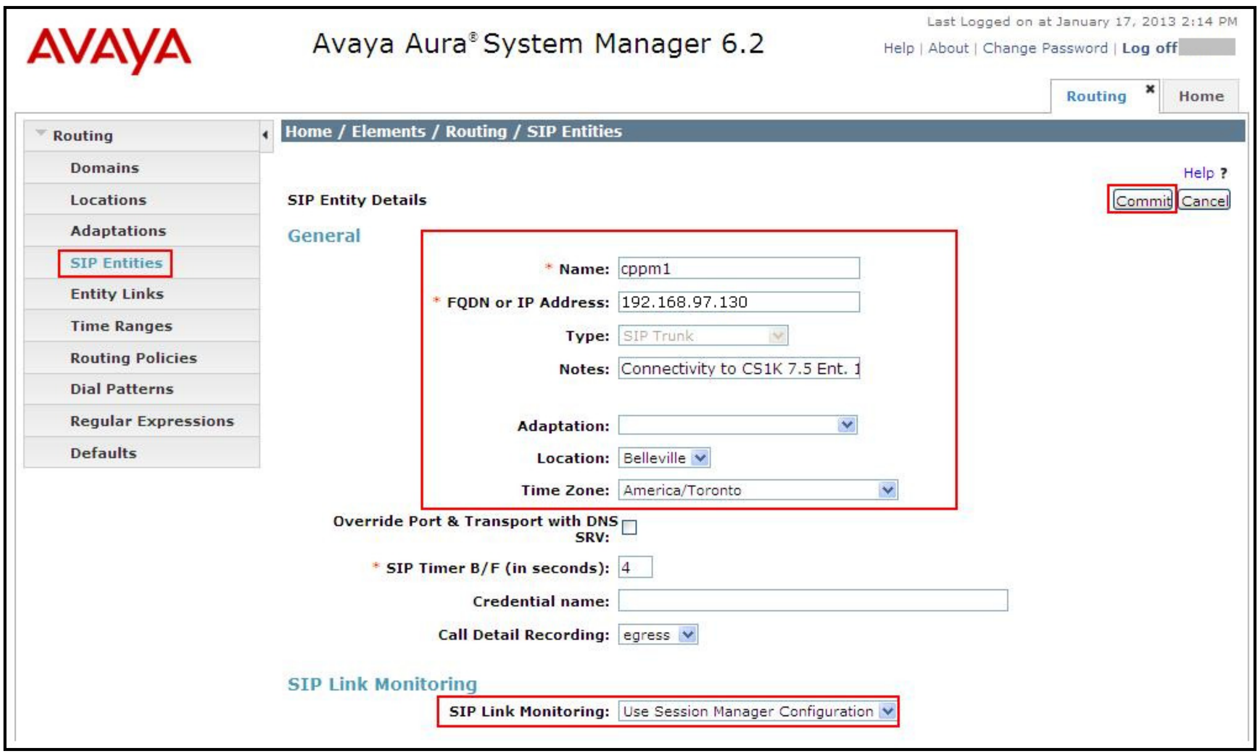
The following figure shows the SIP Entity defined for the Abrazo-E in the sample configuration.



6.1.5. Define SIP Entities – Avaya Communication Server 1000E

1. Define the **Avaya CS1000 SIP Entity** by expanding the **Routing** topic and select **SIP Entities** from the left menu.
2. Click **New** (not shown).
3. In the **General** section, enter the following values and use default values for remaining fields:
    - **Name** – Enter an identifier for the SIP Entity.
    - **FQDN** or **IP Address** – Enter IP address of the Avaya CS1000.
    - **Type** – Select '**SIP Trunk**'.
    - **Notes** (optional) – Enter a brief description
    - **Adaptation** – No adaptation was used.
    - **Location** – Select the Location defined in **Section 6.1.3**.
    - **Time Zone** – Enter the appropriate Time Zone.
4. **SIP Link Monitoring** – Select 'Use Session Manager Configuration'.
5. Click **Commit** to save the definition of the new SIP Entity.

The following figure shows the SIP Entity defined for Avaya CS1000 in the sample configuration.



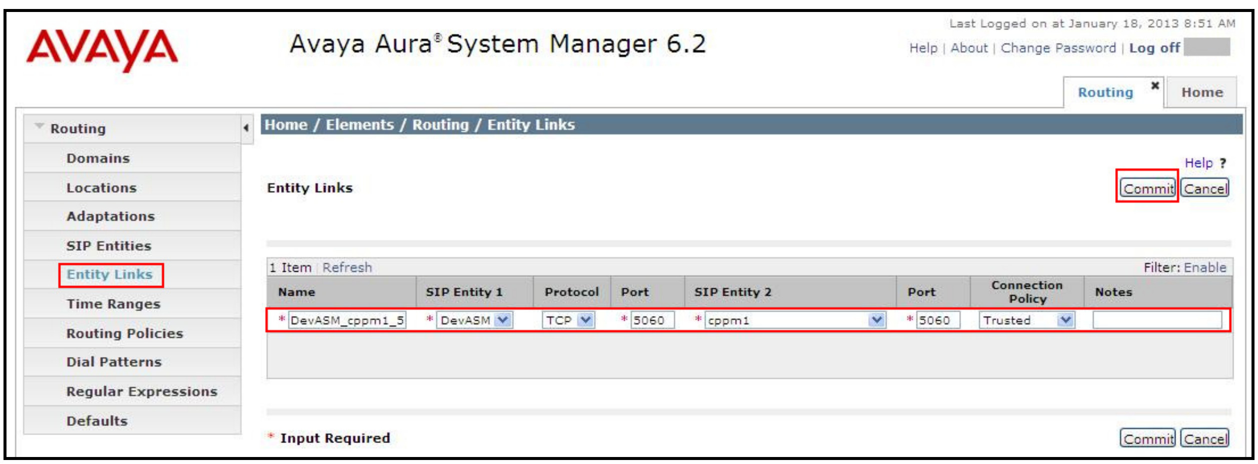
6.1.6. Define Entity Links

A SIP trunk between the Session Manager and a telephony system is described by an Entity Link. In the sample configuration, SIP Entity Links were added between Session Manager and each telephony system.

Define Entity Link – Avaya CS1000E TCP Entity Link

1. Expand **Elements -> Routing** and select **Entity Links** from the left menu.
2. Click **New** (not shown).
3. Enter the following values to configure an Entity Link for the CS1000:
   - **Name** – Enter an identifier for the link.
   - **SIP Entity 1** – Select SIP Entity defined for the Session Manager.
   - **SIP Entity 2** – Select the CS1000 SIP Entity defined in **Section 6.1.5**.
   - **Protocol** - After selection SIP Entity 1 and 2, select '**TCP**' as the required protocol.
   - **Port** – Verify **Port** for both SIP entities is the default listen port. For the sample configuration, default listen port is '**5060**'.
   - **Connection Policy** – From the drop-down menu, select '**Trusted**'.
   - **Notes** *(optional)* – Enter a brief description.
4. Click **Commit** to save Entity Link definition.

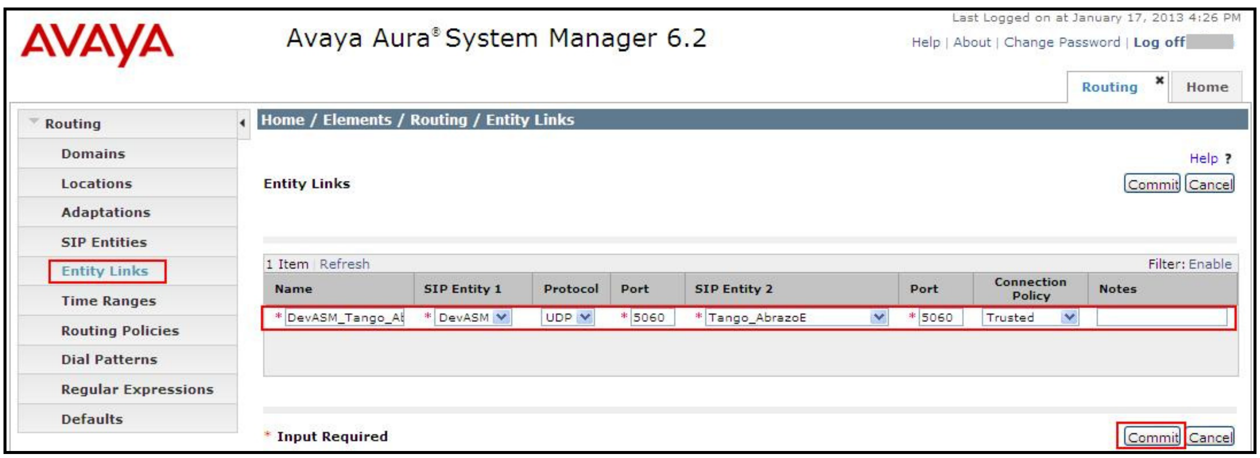
The following figure shows the TCP entity link defined for the SIP trunk between the Session Manager and Avaya CS1000.



Define Entity Link – CS1000 UDP Entity Link

1. Expand **Elements -> Routing** and select **Entity Links** from the left menu.
2. Click **New** (not shown).
3. Enter the following values to configure an Entity Link for the CS1000:
   - **Name** – Enter an identifier for the link.
   - **SIP Entity 1** – Select SIP Entity defined for the Session Manager.
   - **SIP Entity 2** – Select the CS1000 SIP Entity defined in **Section 6.1.5**.
   - **Protocol** - After selection SIP Entity 1 and 2, select '**UDP**' as the required protocol.
   - **Port** – Verify **Port** for both SIP entities is the default listen port. For the sample configuration, default listen port is '**5060**'.
   - **Connection Policy** – From the drop-down menu, select '**Trusted**'.
   - **Notes** *(optional)* – Enter a brief description.
4. Click **Commit** to save Entity Link definition.

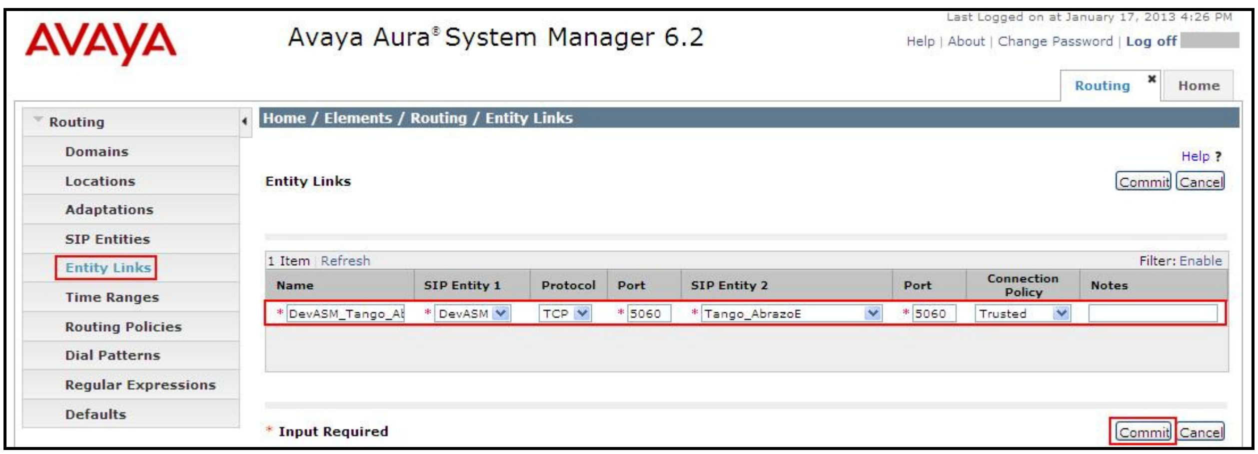
The following figure shows the UDP entity link defined for the SIP trunk between the Session Manager and the Avaya CS1000.



Define Entity Link – Abrazo-E TCP Entity Link

1. Expand **Elements -> Routing** and select **Entity Links** from the left menu.
2. Click **New** (not shown).
3. Enter the following values to configure an Entity Link for the Abrazo-E:
   - **Name** – Enter an identifier for the link.
   - **SIP Entity 1** – Select SIP Entity defined for the Session Manager.
   - **SIP Entity 2** – Select the Abrazo-E SIP Entity defined in **Section 6.1.4**.
   - **Protocol** - After selection SIP Entity 1 and 2, select '**TCP**' as the required protocol.
   - **Port** – Verify **Port** for both SIP entities is the default listen port. For the sample configuration, default listen port is '**5060**'.
   - **Connection Policy** – From the drop-down menu, select '**Trusted**'.
   - **Notes** *(optional)* – Enter a brief description.
4. Click **Commit** to save Entity Link definition.

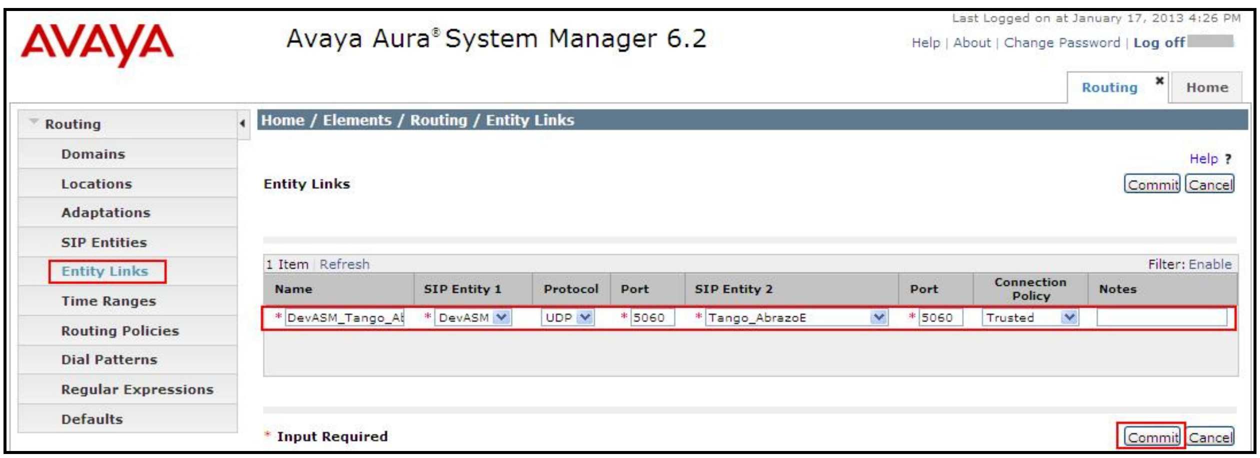
The following figure shows the TCP entity link defined for the SIP trunk between the Session Manager and the Abrazo-E.



Define Entity Link – Abrazo-E UDP Entity Link

1. Expand **Elements -> Routing** and select **Entity Links** from the left menu.
2. Click **New** (not shown).
3. Enter the following values to configure an Entity Link for the Abrazo-E:
   - **Name** – Enter an identifier for the link.
   - **SIP Entity 1** – Select SIP Entity defined for the Session Manager.
   - **SIP Entity 2** – Select the Abrazo-E SIP Entity defined in **Section 6.1.4**.
   - **Protocol** - After selection SIP Entity 1 and 2, select '**UDP**' as the required protocol.
   - **Port** – Verify **Port** for both SIP entities is the default listen port. For the sample configuration, default listen port is '**5060**'.
   - **Connection Policy** – From the drop-down menu, select '**Trusted**'.
   - **Notes** *(optional)* – Enter a brief description.
4. Click **Commit** to save Entity Link definition.

The following figure shows the UDP entity link defined for the SIP trunk between the Session Manager and the Abrazo-E.



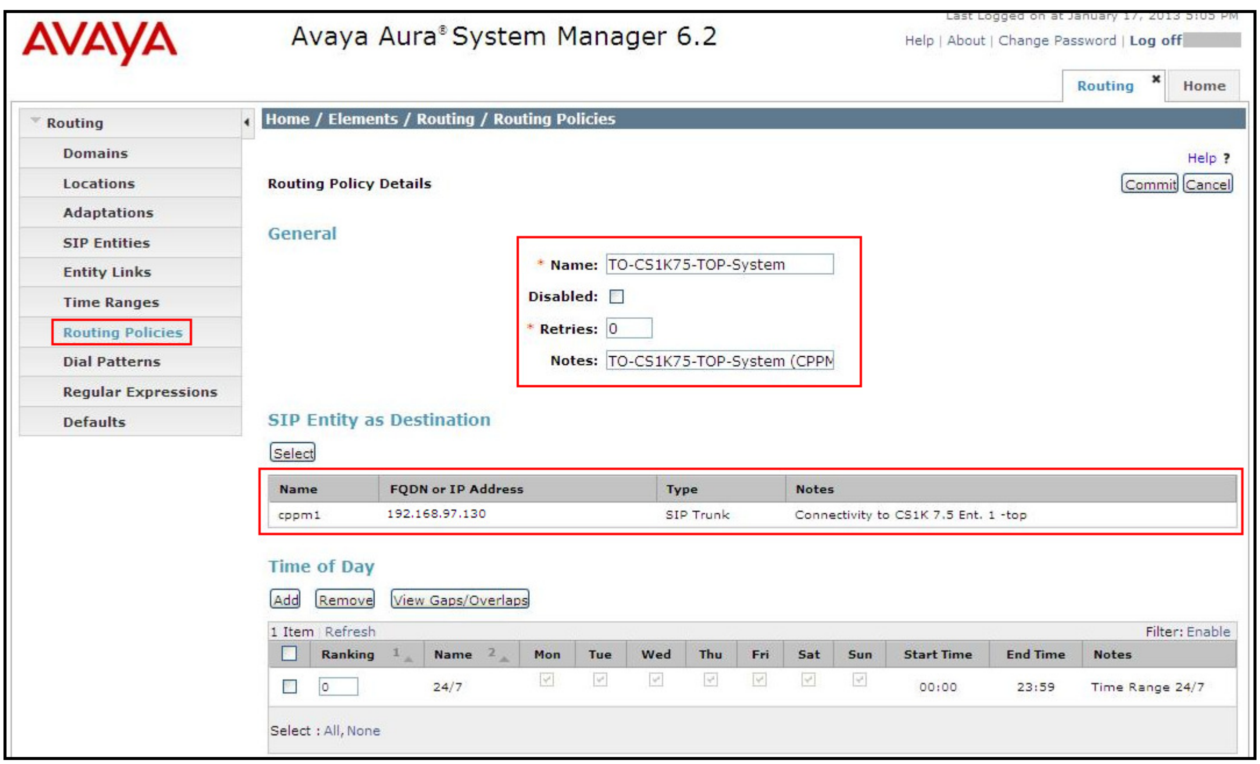
6.1.7. Define Routing Policy

Routing policies describe the conditions under which calls are routed to the SIP Entities specified in **Section 6.1.4 (Define SIP Entities – Abrazo-E)** and in **Section 6.1.5 (Define SIP Entities – Avaya CS1000)**. Two routing policies must be added, one for the CS1000 and one for Abrazo-E.

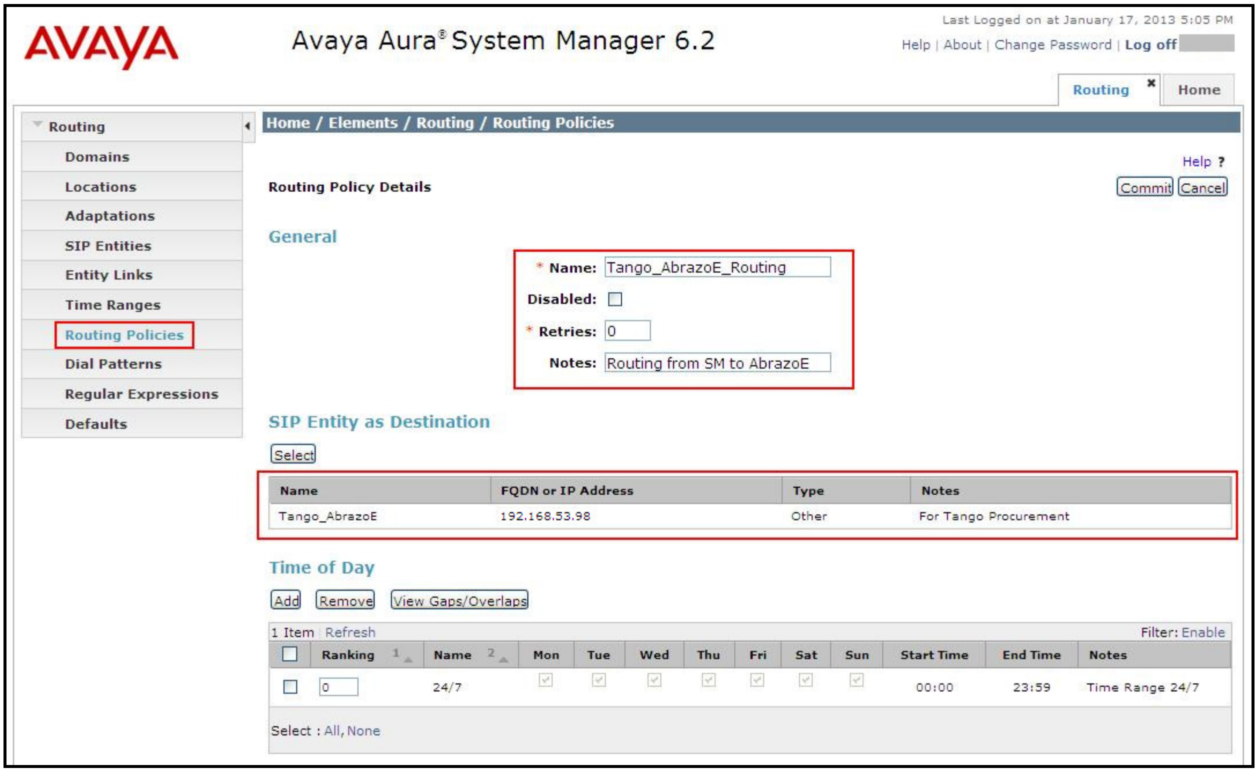
1. To add a routing policy, expand **Elements** → **Routing** and select **Routing Policies**.
2. Click **New** (not shown).
3. In the **General** section, enter the following values:
    - **Name -** Enter an identifier to define the routing policy.
    - **Disabled -** Leave unchecked.
    - **Notes** *(optional)* **-** Enter a brief description.
    - In the **SIP Entity as Destination** section, click **Select**. The SIP Entity List page opens (not shown).
    - Select the SIP Entity associated with CS1000 defined in **Section 6.1.5** and click **Select**
    - The selected SIP Entity displays on the **Routing Policy Details** page.
4. Use default values for the remaining fields. Click **Commit** to save the Routing Policy definition.

The routing policy defined in this section is an example. Other routing policies may be appropriate for other networks.

The following figure shows the Routing Policy for Avaya CS1000 in this sample configuration.



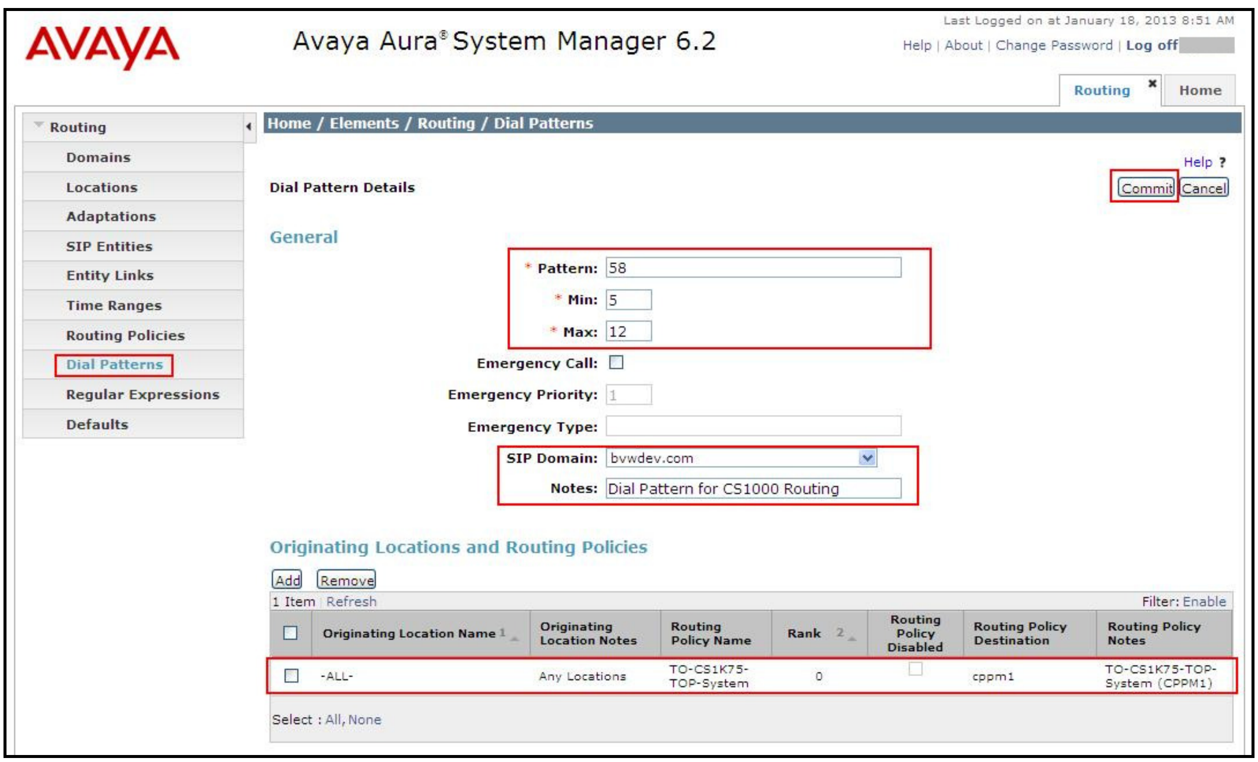
5. **Repeat all the steps in this section** and define the routing policy to the Abrazo-E as shown in the following screen.



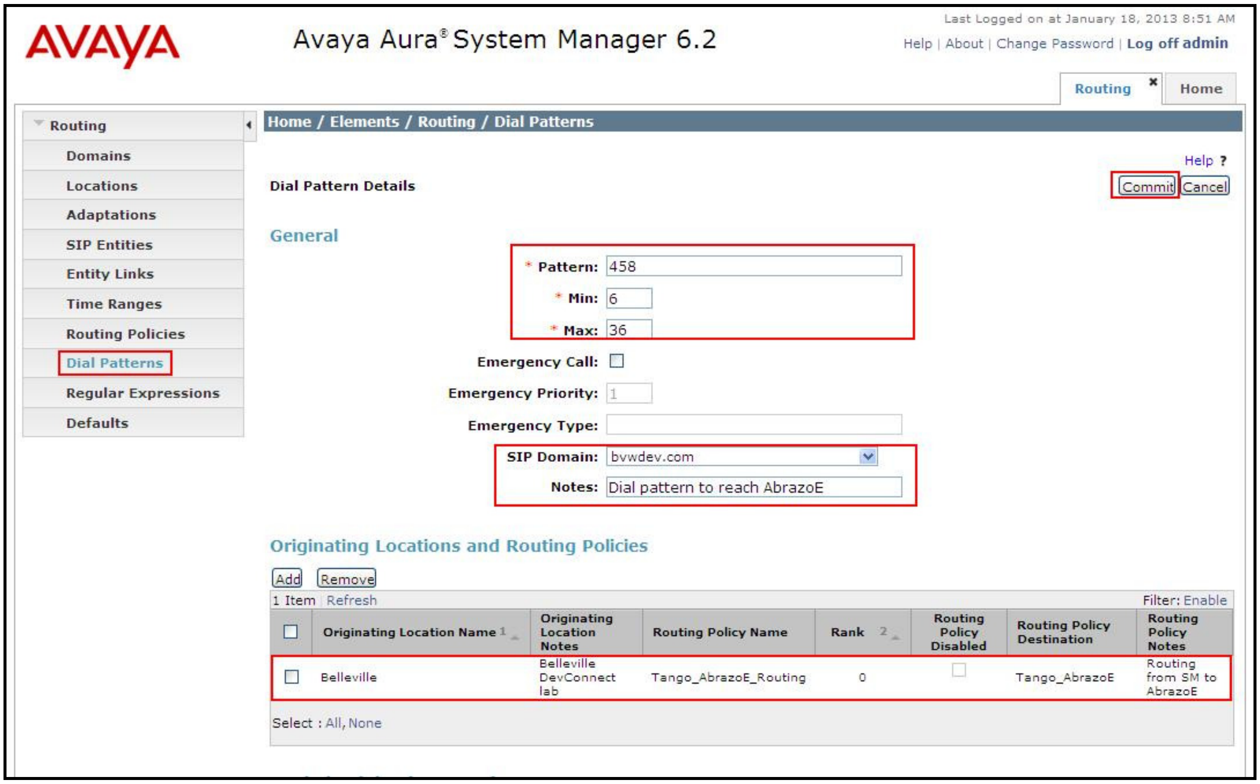
6.1.8. Define Dial Pattern

1. To define a dial pattern, expand **Routing** → select **Dial Patterns** (not shown).
2. Click **New** (not shown).
3. In the **General** section, enter the following values and use default values for the remaining fields:
   - **Pattern** – Enter dial pattern for calls to the CS1000.
   - **Min** – Enter the minimum number of digits that must be dialed.
   - **Max** – Enter the maximum number of digits that may be dialed.
   - **SIP Domain** – Select the SIP Domain from the drop-down menu or select '**All**' if the Session Manager should accept incoming calls from all SIP Domains.
   - **Notes** *(optional)* – Enter a brief description.
4. In the **Originating Locations and Routing Policies** section, click Add.
5. The Originating Locations and Routing Policy List page opens (not shown).
6. In Originating Locations table, select 'Belleville'.
7. In **Routing Policies** table, select the Routing Policy defined for CS1000 in **Section 6.1.7**.
8. Click **Select** to save these changes and return to the **Dial Pattern Details** page.
9. Click **Commit** to save.

The figure below shows the Dial Pattern defined for the sample configuration to route calls to the Avaya CS1000.



The following figure shows the Dial Pattern defined for the sample configuration to route calls to the Abrazo-E.



6.2. Avaya Communication Server 1000E Provisioning

This section describes the steps required to configure Avaya CS1000 to interoperate with Abrazo-E. It is assumed that Avaya CS1000 has already been installed and is functioning. For more information refer to documents listed in **Section 11**. To use Abrazo services, subscribers need to be provisioned with certain features. The following sections detail the steps required to enable these services.

6.2.1. Required License Keys

To enable the subscriber features required for Abrazo service, the following license keys must be enabled and have sufficient resources available (one unit per Abrazo subscriber):

- Personal Call Assistant (PCA)
- AST
- SIP CTI TR87

The Avaya CS1000 also requires a sufficient number of SIP Access Port (i.e., SIP Trunk) licenses to facilitate communication with the Abrazo. SIP Access Port licenses are calculated based on the maximum number of concurrent calls allowed via the SIP protocol.

The following steps explain how an administrator can determine if the Avaya CS1000 system has a sufficient number of licenses:

Use an SSH terminal emulator to connect to Avaya CS1000E and log in with the appropriate credentials. Enter **LD 22** to enter overlay 22 and then enter the appropriate data as shown in bold below.

```
LD 22          ← Load Overlay 22
REQ: SLT       ← Print out system limits
```

A list similar to the following will be displayed. Verify that the license requirements can be met. The four relevant license types and the number of IP users are highlighted below in bold. Each license type needs to have a number greater than 0 in the **LEFT** column.

```
System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz

IPMGs Registered:             1
IPMGs Unregistered:           0
IPMGs Configured/unregistered:  0

TRADITIONAL TELEPHONES 32767    LEFT 32751    USED    16
DECT USERS             32767    LEFT 32767    USED     0
IP USERS               32767    LEFT 32706    USED    61
BASIC IP USERS         32767    LEFT 32765    USED     2
TEMPORARY IP USERS     32767    LEFT 32767    USED     0
DECT VISITOR USER      10000    LEFT 10000    USED     0
ACD AGENTS             32767    LEFT 32760    USED     7
MOBILE EXTENSIONS      32767    LEFT 32761    USED     6
TELEPHONY SERVICES     32767    LEFT 32767    USED     0
CONVERGED MOBILE USERS 32767    LEFT 32767    USED     0
AVAYA SIP LINES        32767    LEFT 32751    USED    16
THIRD PARTY SIP LINES  32767    LEFT 32744    USED    23

PCA                    32767    LEFT 32766    USED     1
ITG ISDN TRUNKS        32767    LEFT 32767    USED     0
H.323 ACCESS PORTS     32767    LEFT 32767    USED     0
AST                    32767    LEFT 32757    USED    10
SIP CONVERGED DESKTOPS 32767    LEFT 32767    USED     0
SIP CTI TR87           32767    LEFT 32745    USED    22
SIP ACCESS PORTS       32767    LEFT 32703    USED    64
RAN CON                32767    LEFT 32767    USED     0
MUS CON                32767    LEFT 32767    USED     0

IP RAN CON             16384    LEFT 16384    USED     0
IP MUS CON             16896    LEFT 16896    USED     0
IP MEDIA SESSIONS      35842    LEFT 35842    USED     0
…
TRADITIONAL TRUNKS     32767    LEFT 32740    USED    27
ELC ACCESS PORTS       32767    LEFT 32767    USED     0
DCH                      255    LEFT   250    USED     5
```

The output in the above example shows 32,766 licenses available for **PCA**, 32,757 for **AST**, and 32,745 for **SIP CTI TR87**. This means that up to 32,745 Avaya CS1000 users can be provisioned for Abrazo services. (Each Abrazo user requires one license for each of the three PCA, AST, and SIP CTI TR87 services.)

6.2.2. Enable Personal Call Assistant (PCA) Feature – Mobile UC

The Personal Call Assistant (PCA) feature allows a second DN to be alerted whenever a subscriber receives an incoming call. The second DN is alerted in parallel (simultaneously) with the device(s) associated with the primary DN. The Abrazo requires this service in order to alert the subscriber's mobile device in parallel with the user's desk phone.

Enable PCA at Customer Level

The PCA feature must be enabled at the customer level before enabling it for a given subscriber. If PCA is not already enabled, set **PCA** to **ON** as shown below for the required customer.

Use an SSH terminal emulator to connect to Avaya CS1000E and log in with the appropriate credentials. Enter **LD 15** to enter overlay 15 and then enter the appropriate data as shown in bold below. In this sample configuration defaults were used for the remaining prompts (not shown).

LD 15 ← Load Overlay 15.
REQ: **CHG**
TYPE: **FTR**
CUST: **0**
…
PCA **ON** ← Enables Personal Call Assistant.

Configure PCA for Phones

Per-TN configuration is required to define a PCA TN (Terminal Number) for each Abrazo-enabled subscriber. Before executing this procedure, however, allocate a range of numbers to use as the target DN for the PCA feature (i.e., the DN that will be alerted in parallel to the primary DN). Using the DN and adding a unique prefix is the recommended alias DN format. Calls terminating to this range will be configured to route via a SIP Trunk to the Abrazo-E in a later step.

Use an SSH terminal emulator to connect to Avaya CS1000E and log in with the appropriate credentials. Enter **LD 11** to enter overlay 11 and then enter the appropriate data as shown in bold below. In this sample configuration defaults were used for the remaining prompts (not shown).

LD 11	← Go to Overlay 11.
REQ: **NEW**	
TYPE: **PCA**	
TN: **x x x x**	← Enter an appropriate TN, where **x x x x** is the TN
CUST: **xx**	← Enter the customer number defined in LD 15
CLS: **AHA**	← Automatic Hold Allowed
KEY: **0 aaa yyyy**	← Primary PCA DN where aaa = MCN, MCR, SCN, or SCR where yyyy = the primary DN
KEY: **1 HOT P nn yyyy**	← Target PCA DN where nn = length of the PCA DN (32 maximum) where yyyy=the target DN (the number to ring simultaneously with primary DN)

Example KEY entries:

KEY: 0 MCR 58713
KEY: 1 HOT P 6 458713

In this example, if a call terminates to 58713 the Avaya CS1000 will ring 58713 and 458713 simultaneously.

For more information, refer to the Personal Call Assistant section in the Avaya NN43001-106 document as listed in **Section 11**.

6.2.3. Enabling SIP CTI via TR87

The Abrazo solution uses the CS1000's SIP CTI link to provide the Call Move feature. This feature allows an active call to be moved from a desk phone to a mobile and vice-versa. Each Abrazo-enabled subscriber who has the Call Move feature must therefore have the SIP CTI service activated. Each subscriber requires an AST license and a SIP CTI license in order to activate SIP CTI. (See **Section 6.2.1** (**Required License Keys**))

To enable SIP CTI for a subscriber, use the following one time setup procedures.

- Configure Application Module Link (AML) (used for SIP CTI communication)
- Configure Value Added Server (VAS) such as to set up an IP server
- Configure SIP CTI on Element Manager

After the above one-time setup procedures are executed, use the next procedure for each subscriber that will have access to the Abrazo Call Move feature:

- Configure Phones for SIP CTI services

Each of these procedures is explained next.

Configure Application Module Link (AML)

LD 17 ← Load Overlay 17.
Press Enter for all prompts other than those listed below.
REQ: **CHG**
TYPE: **ADAN**
ADAN: **NEW ELAN xx** ← A new AML link; the link is an ELAN type.

For Small System: 32 to 47
For Large System: 32 to 127.

An AML link number in the above range implies that the transport is over a TCP link, and denotes a "logical" AML ELAN type.

CTYP: **ELAN**

Configure Value Added Server (VAS)

LD 17 ← LOAD Overlay 17

Press Enter for all prompts other than those listed below.

REQ: **CHG**

TYPE: **VAS**

VAS: **NEW**

VSID: **vas#** ← Enter VAS ID

For Small System: 32 to 47
For Large System: 32 to 127

A VAS number in the above range denotes a "logical" VAS ID.

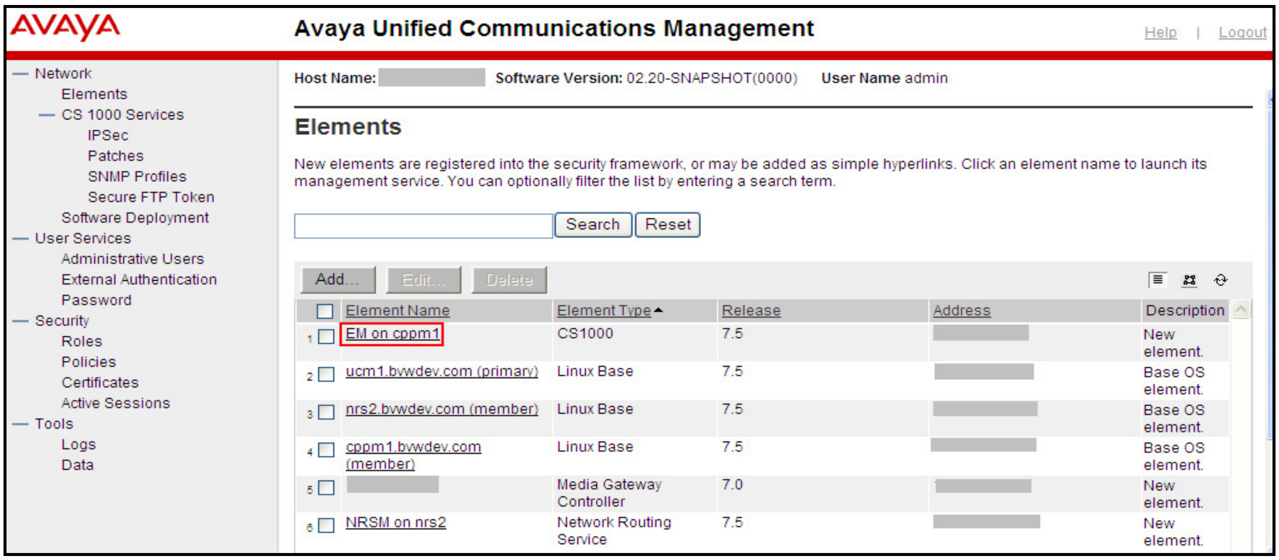
ELAN **link#** ← The AML ELAN link number provisioned when the AML link was created.

SECU: **No** ← Enables security for Meridian Link Applications.

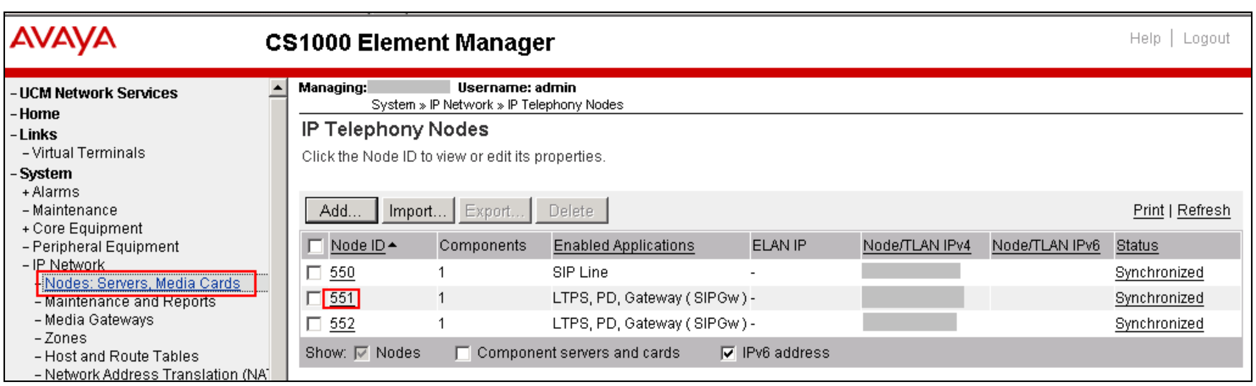
6.2.4. Element Manager Provisioning - Configure SIP CTI

To complete these steps, you must log into Unified Communications Manager (UCM). Open an instance of a web browser and connect to the UCM GUI at the following address: **http://<node IP address>** or **http://<UCM IP address>**. Log in using an appropriate Username and Password (not shown).

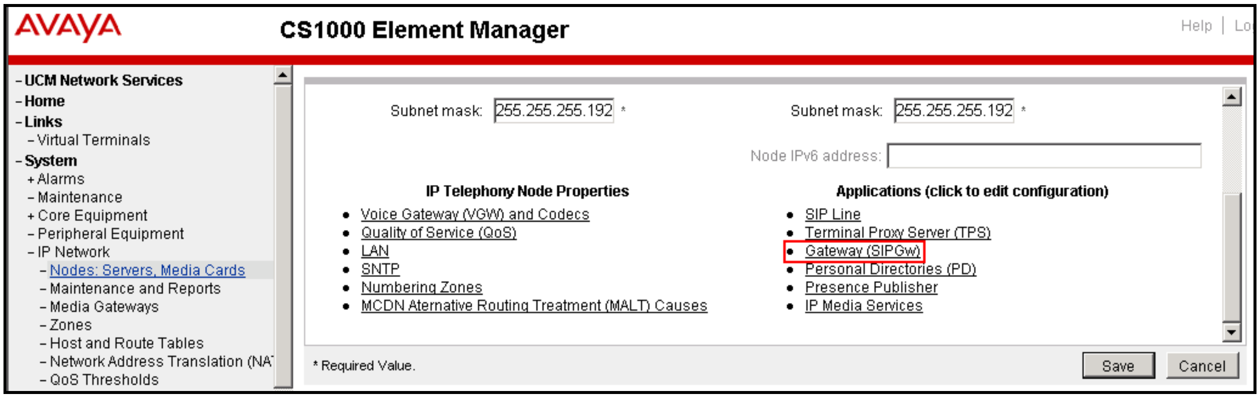
The Unified Communications Management screen is then displayed. Click on the **Element Name** of the Avaya CS1000E Element Manager as in the screen below. The following screen is displayed. Select the link for appropriate Element Manager. In the example configuration, **EM on cppm1** was selected.



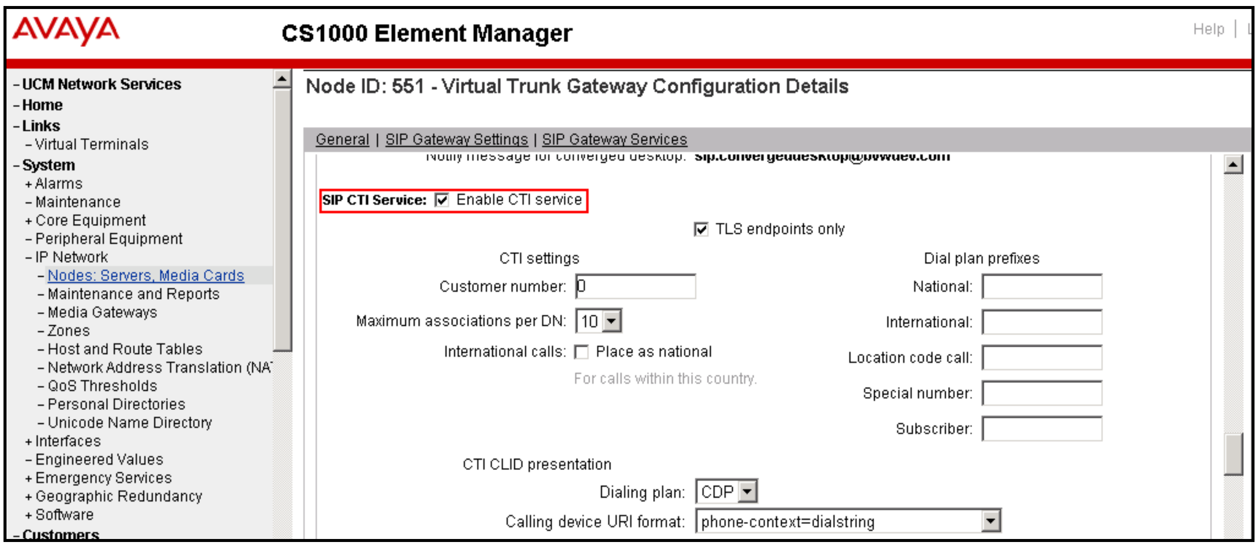
As shown below, select `System` → `IP Network` → `Nodes: Servers, Media` and then select the appropriate node from the right pane. In this example, Node 551 was selected.



In the right pane, scroll down and click on **Gateway (SIPGw)**.

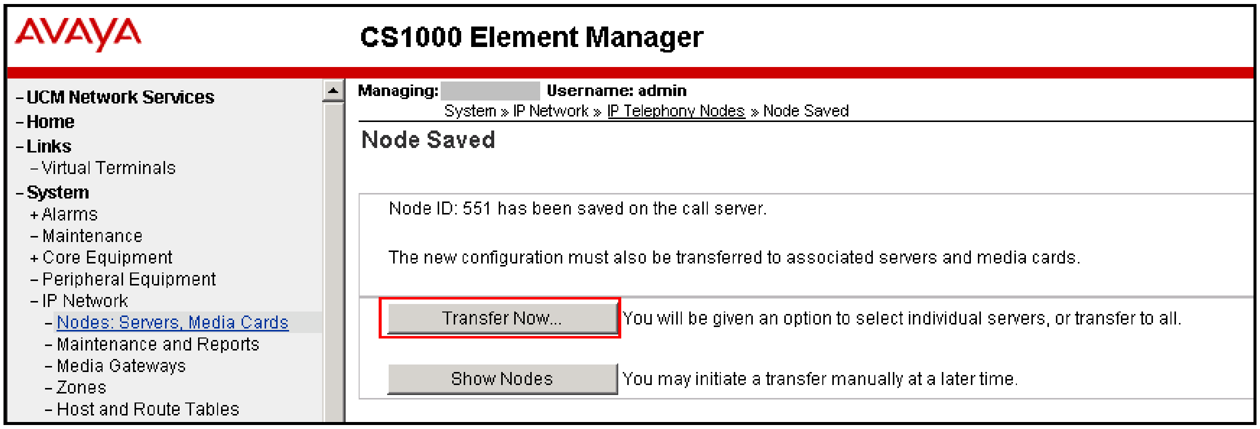


Ensure that the **SIP CTI Service: Enable CTI service** is enabled (checked). Click **Save** (not shown).



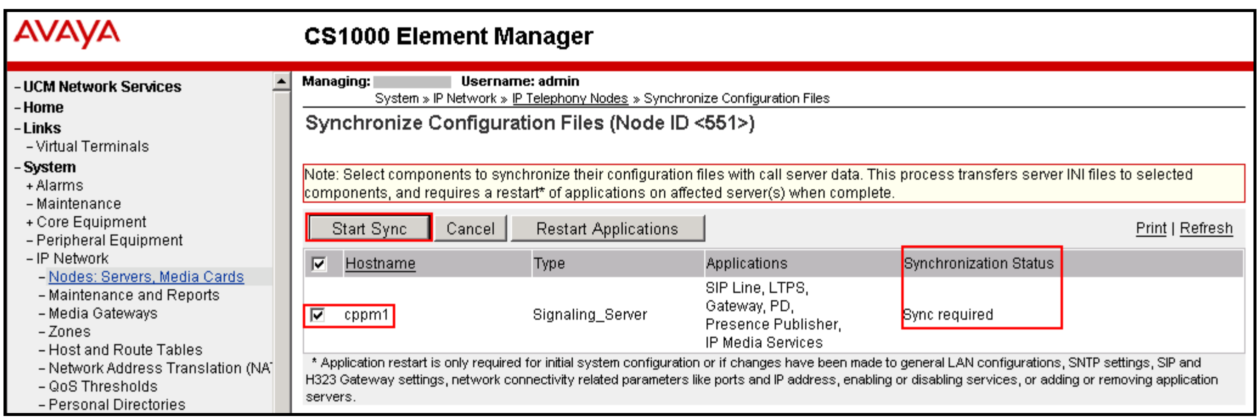
Click **Save** on the **Node Details** screen (not shown).

Select **Transfer Now** on the **Node Saved** page as shown below.



Once the transfer is complete, the **Synchronize Configuration Files (Node ID <id>)** page is displayed.

Select the **Checkbox** associated with the appropriate Call Server and click **Start Sync.** The screen will automatically refresh until the synchronization is finished. The **Synchronization Status** field will update from **Sync required** (as shown) to **Synchronized** (not shown).



6.2.5. Configure Phones for SIP CTI Services

Per-TN configuration is required to define which TNs are used for SIP CTI and to define the specific DN keys on each TN that are available for control by SIP CTI applications. Use the Call Server console command line as follows. AST specifies the primary phone (Key 00) which is the only phone line that can be specified for a call move.

LD 20 ← Load Overlay 20

Press **Enter** for all prompts other than those listed below to turn on T87A.

REQ: **CHG**	
TYPE: **xxxx**	← Where xxxx is the digital phone type of the DN, e.g., i2004, 1120, 1140, etc., depending on the primary desk phone type
TN: **x x x x**	← Enter an appropriate TN, where **x x x x** is the TN
ECHG: **YES**	← Enter Easy Change mode
ITEM: **CLS T87A**	← Enables TR87 Allow Remote Call Control
ITEM: **AST 0**	← Configure AST for Key 0
ITEM: Press **Enter** to exit	

REQ: **PRT** Print the TN to validate the change

You should see output similar to the following. Validate that **T87A** and **AST 00** show up under the CLS section. (Each is highlighted in bold in the following sample output.)

```
REQ: prt
TYPE: 1120
TN   96 0 01 14
DATE
PAGE
DES
MODEL_NAME
EMULATED
KEM_RANGE

DES  TANGO
TN   096 0 01 14  VIRTUAL
TYPE 1120
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00001
CUR_ZONE 00001
```

```
MRT
ERL  0
ECL  0
FDN  58888
TGAR 1
LDN  NO
NCOS 0
SGRP 0
RNPG 0
SCI  0
SSU
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS  CTD FBD WTA LPR MTD FNA HTD TDD HFD CRPD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LND CNDD
     CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDD CFXD ARHD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
     UDI RCC HBTD AHD IPND  DDGA NAMA MIND PRSD NRWD NRCD NROD
     DRDD EXR0
     USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY
DNO3 MCBN
     FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87A SBMD
     KEM3 MSNV FRA  PKCH MUTA MWTD DVLD CROD ELCD
CPND_LANG ENG
RCO  0
HUNT
PLEV 02
PUID
UPWD
DANI NO
AST  00
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 MCR 58713 0     MARP
        CPND
          CPND_LANG ROMAN
```

```
          NAME Tango 11201
          XPLN 13
          DISPLAY_FMT FIRST,LAST
   01 MCR 58713 0
      CPND
        CPND_LANG ROMAN
          NAME Tango 11201
          XPLN 13
          DISPLAY_FMT FIRST,LAST
   02
   …
   30
   31
```

6.2.6. Enabling Trunk Features

To enable the Abrazo Call Move feature, two trunk features must be enabled (value = YES) on the Call Server:

- **TRNX** - Allow transfer on ringing of supervised external trunks across the network.
- **EXTT** - Allow unconditional external trunk-to-trunk transfer. A caller can transfer a call made to one outgoing trunk to another outgoing trunk without restrictions based on supervision. This prompt applies only to situations involving two calls originated by the same caller.

> **Note:** Even though both the TRNX and EXTT trunk features need to be enabled (there by allowing external trunk transfers), you can still maintain security on the CS1000 by using trunk-to-trunk access restriction features like *Trunk Group Access Restriction* (TGAR) / *Trunk Access Restriction Group* (TARG) and Trunk Barring. If the TRNX and EXTT features are not enabled, then phones cannot transfer calls externally from the CS1000, which would prevent the Abrazo Call Move feature from working.
>
> TGAR controls access to the exchange network, tie trunks, and other trunks and services on the CS1000. Phones are assigned a TGAR code to block access to certain trunk groups. When a phone tries to access a trunk route, the system first checks the Class of Service of the terminal. If access is allowed, the system then checks the TGAR code. If the TGAR code of the phone matches one of the Trunk Access Restriction Group codes programmed against the trunk group, access is denied. For example, a phone with a TGAR code of 1 cannot use trunks that have a TARG code of 1.

To enable these features, perform the following steps:

LD 15 ← Load Overlay 15.

Press `Enter` for all prompts other than those listed below to change the customer data block.

REQ: **CHG**
TYPE: **NET**
TYPE NET_DATA
CUST **0**
TRNX **YES**
EXTT **YES**

6.3. CallPilot Voicemail Provisioning– Mobile UC

The Abrazo solution supports the Avaya CS1000/CallPilot voicemail solution. The Avaya CS1000 notifies the Abrazo-E by sending a SIP NOTIFY for voicemail deposit and retrieval. The CallPilot entry for the CS1000 subscriber is configured with the alias DN of an Avaya CS1000 subscriber.

The following will be covered in this section:

- Configure Message Waiting Indicator (MWI) (enables SIP NOTIFY communication)
- Configure CallPilot subscriber with a MWI DN (this is the alias DN of the Avaya CS1000 subscriber)

6.3.1. Configure Message Waiting Indicator (MWI)

LD 17 ← Load Overlay 17

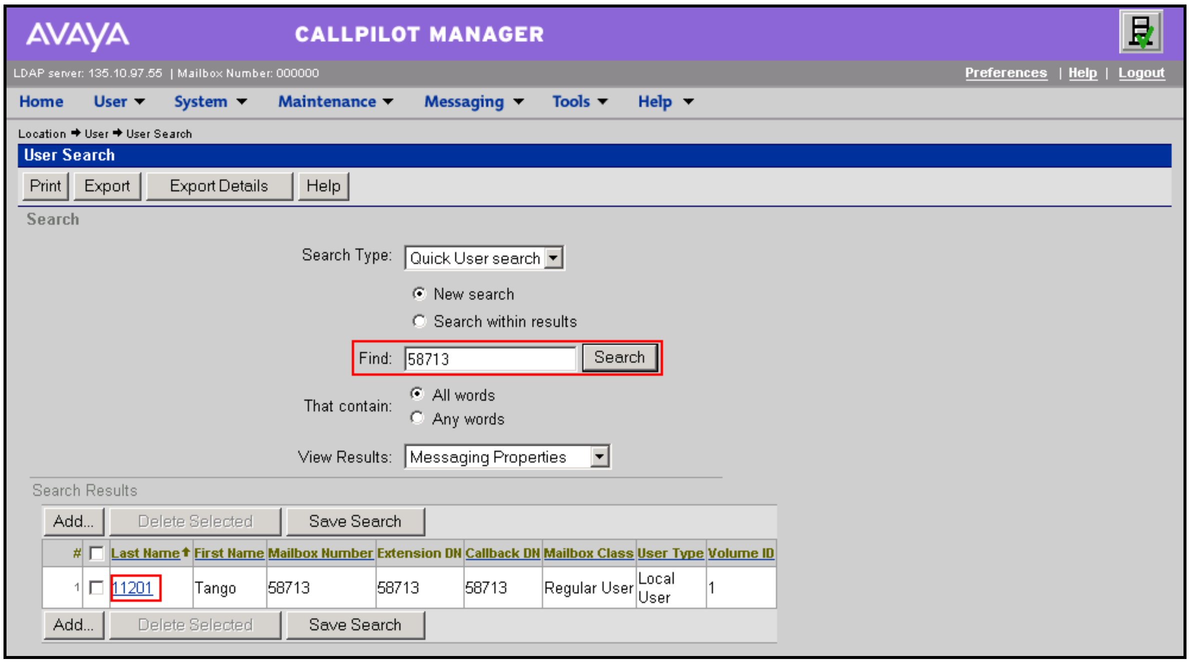
Press Enter for all prompts other than those listed below and enter the text in bold.

REQ: **CHG**
TYPE: **ADAN**
ADAN: **CHG DCH XX** ← Where XX is the D-Channel to Session Manager
RCAP: **MWI** ← Add MWI as a Remote Capability of the far-end

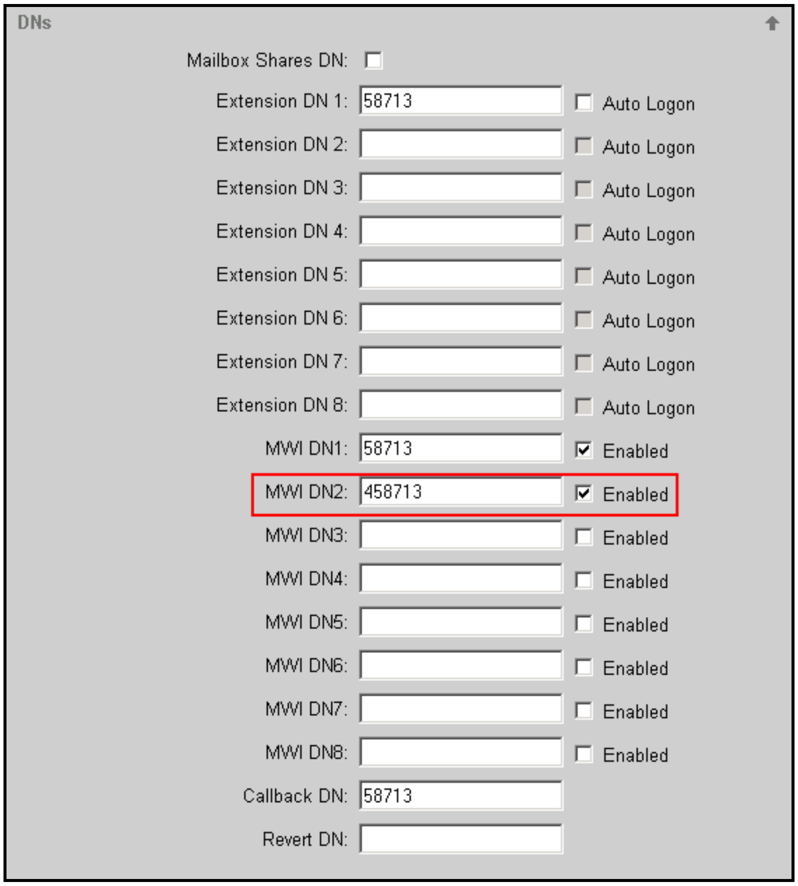
6.3.2. Configure CallPilot subscriber

For the Abrazo-E to receive MWI messages (Message Waiting Indicator) for a subscriber, the PCA alias DN of that Avaya CS1000 subscriber must be added in the CallPilot MWI DN field.

1. Log into the CallPilot web provisioning interface (not shown).
2. From **User → User Search**, perform a search for a desired Extension. In the figure below DN 58713 was used as an example.
3. Next, click on the **Last Name.** In this example 11201 was configured as the last name for the subscriber.



4. Now add the Avaya CS1000 alias (PCA) DN of the subscriber to the subscriber's CallPilot account as shown below.



The CallPilot subscriber DNs page describes the DNs that are alerted upon a voicemail notification as shown by the MWI DN fields. The figure above shows a subscriber DN of 58713, and an alias DN of 458713.

6.3.3. Interaction with Private Network Identifier (PNI)

When a PNI other than 0 is defined in the Route Data Block (RDB) of the Avaya CS1000, a voicemail deposit results in a SIP INVITE being sent to the Abrazo-E (in addition to the SIP NOTIFY). When this SIP INVITE message is received by the Abrazo-E, it is rejected with a SIP "488 Not Acceptable Here" final response message. The voicemail notification operation is not affected by this extraneous SIP INVITE message exchange and can be ignored.
The following procedure can be used to view the PNI configuration on the Avaya CS1000.

LD 21 ← Load Overlay 21
REQ: **PRT**
TYPE: **RDB**
CUST **0**
ROUT **xx** ← Where xx is the route number

6.3.4. Distance Steering Codes

In the section Route Entries to Abrazo-E Gateway Endpoint, the Alias DN, Pilot DN, and Service DN prefix routes were added to go to the Session Manager for SIP trunking. With the SIP/PRI Gateway option, the Pilot DN now needs to be routed to the appropriate PRI trunk to the SIP/PRI Gateway. This is done via Distance Steering Codes (DSC).

The Pilot DN prefix should already have a DSC defined in Overlay 87. The DSC has a Route List Index (RLI) that points to a Route List Blocks (RLB). Each RLB will have Route Numbers (ROUT) which points to a Route Data Block (RDB) that either contains a SIP or PRI trunk. The DSC needs to have the RLI changed to from the SIP trunk to the PRI trunk.

To display the current DSC setting:

LD 87 ← Load Overlay 87

Press Enter for all prompts other than those listed below.

REQ: **prt**
CUST: **0**
FEAT: **cdp**
TYPE: **dsc** ←Distance Steering Code
DSC: **xx** ← Where xx DSC Prefix or blank to list all

Example of print output:
DSC 737437
FLEN 10
DSP LSC
RRPA NO
RLI 10
CCBA NO
NPA
NXX

To change the RLI, enter **chg** under REQ and press **Enter** until the RLI line is reached.

7. PSTN Access Enabled Abrazo-E

To make the Abrazo-E the ingress/egress point for off net calls between the PBX and a SIP Trunk provider, modify your PBX's routing tables to send off-net calls over a SIP Trunk to the Abrazo-E on the SIP Trunking port (default of 5080). This SIP Trunk should also allow incoming calls from the Abrazo-E. Make sure any required Class of Service attributes are set appropriately on the PBX. On-net calls between PBX's can also be configured to route via the Abrazo-E using a similar process. It is often desirable to create a second SIP Trunk on the PBX in order to specify different Class of Service attributes for on-net calls. Consult Avaya CS1000 and Session Manager configuration documentation for specific details on SIP Trunk configuration.

8. Abrazo Provisioning

This document assumes that the Abrazo has already been provisioned with:

- Enterprise information
- Wireless carrier information. The Carrier(s) should be enabled for Mobile UC and/or PSTN Access.

8.1. Mobile UC Provisioning

The section discusses the integration process for a Mobile UC enabled Abrazo-E.

The steps below describe the unique configuration areas needed to integrate the Avaya CS1000E with the Abrazo solution. Refer to the Abrazo-E Provisioning Guide for a comprehensive explanation of Abrazo provisioning.

8.1.1. Voice Network : PBX

1. **Add Trunk Dial Plan** –There are no unique configuration items for Trunk Dial Plans and the Avaya CS1000. Refer to the Abrazo-E Provisioning Guide, Voice Networks, PBXs, Add Trunk Dial Plan section.
2. **Add PBX** – The Domain name field must match what was provisioned on the Avaya CS1000. See **Section 6.1.2** (Define SIP Domains). For all other PBX fields, refer to the Abrazo-E Provisioning Guide, Voice Networks, PBX section.
3. **Add Trunk Groups/Trunk** – Create a trunk group according to the Abrazo-E Provisioning Guide. Phones that do not register with Avaya CS1000 are considered 'trunk' type phones.
   - **Host Address, Port** and **Transport Type** should match the value configured for the Node IP of Avaya CS1000.
   - Trunk Group **Request URI** parameters are not used by the Avaya CS1000 7.5 PBX and therefore do not need to be provisioned.
   - Also note that only one trunk group can be data filled for Avaya CS1000. Refer to the Abrazo-E Provisioning Guide, Voice Networks, PBXs (Add/Modify), Add Trunk Groups/Trunk section.
4. **Add Line Groups/Line** – Create a line group according to the Abrazo-E Provisioning Guide Voice Networks, PBXs (Add/Modify), Add Line Groups/Trunk section. Phones that register directly to the PBX are considered 'line' type phones.
   - **Host Address, Port** and **Transport Type** should match the value configured for the Node IP of the Avaya CS1000 7.5 platform.
   - **Trunk Group Request URI** parameters are not used by the Avaya CS1000 and therefore do not need to be provisioned.
   - Also note that only one line group can be data filled for the Avaya CS1000.
5. **Add Pilot Numbers** (optional but recommended) - No unique configuration areas required for provisioning Pilot Numbers to the Abrazo system. Refer to the Abrazo-E Provisioning Guide, Voice Networks, PBXs (Add/Modify) section.
6. **Add Call Service Pilot Number** (optional but recommended for the Call Move service) - No unique configuration items for Call Service Pilot Numbers. Refer to the Abrazo-E Provisioning Guide, Voice Networks, PBXs (Add/Modify) section.
7. **Add Least Cost Routing in the Abrazo system for Avaya CS1000**. No unique configuration items for Least Cost Routing. Refer to the Abrazo-E Provisioning Guide, Voice Networks, Least Cost Routes, Add Least Cost Routes section.

8.1.2. Voice Network : Voice Mail

1. **Add Voice Mail for Avaya CS1000** in the Abrazo system. Select **SIP** for the **Voice Mail Server Type**. Provide a voicemail **retrieval** and **deposit** number in addition to an **MWI host address** and **port number**. This item assumes that a CallPilot Voice Mail server is used with the Avaya CS1000. Refer to the Abrazo-E Provisioning Guide, Voice Networks, Voice Mail Servers, Add Voice Mail Server section.

8.1.3. Subscriber

1. Add Subscriber Dial Plan. No unique configuration items for adding Subscriber Dial Plans. Refer to the Abrazo-E Provisioning Guide, Subscribers, Add Subscriber Dial Plan section.
2. Add Subscribers. There are several items to note when provisioning subscribers. For more information, refer to the Abrazo-E Provisioning Guide, Subscribers, Subscribers section:
   - Home PBX – select the newly created Avaya CS1000 as the subscriber's Home PBX.
   - Home PBX Provides Orig Svcs – enable this setting for those phones that will register as a line interface to the PBX. When checked, the Abrazo always originates the call on behalf of the mobile user for mobile originations into their home PBX within the enterprise. For non-line phones (i.e. SIP phones), there is no need to check this box as it will only affect routing and not present the call as a 'line' type call.
   - Alias – Refer to **Section 6.2.2** (**Enable Personal Call Assistant (PCA) Feature – Mobile UC**) for information on provisioning the Alias field.

8.2. PSTN Access Enabled Abrazo-E Provisioning

This section is for those Abrazo-Es that have only PSTN Access enabled or have both Mobile UC and PSTN Access enabled.

If your configuration is Mobile UC only, refer to **Section 8.1**.

The steps below describe the unique configuration areas needed to integrate the Avaya CS1000 with the Abrazo solution. Refer to the Abrazo-E Provisioning Guide for a comprehensive explanation of Abrazo provisioning.

8.2.1. Voice Network : PBX

1. **Add Trunk Dial Plan** (optional) –There are no unique configuration items for Trunk Dial Plans and the Avaya CS1000. Refer to the Abrazo-E Provisioning Guide, Voice Networks, PBXs, Add Trunk Dial Plan section.

2. **Add PBX** –The Domain name field must match what was provisioned on the Avaya CS1000. See **Section 6.1.2** (**Define SIP Domains**). For all other PBX fields, refer to the Abrazo-E Provisioning Guide, Voice Networks, PBX section

3. **Add Trunk Groups/Trunk** (required)
   - **Host Address, Port** and **Transport Type** should match the value configured for the Node IP of the Avaya CS1000.
   - Trunk Group **Request URI** parameters are not used by the Avaya CS1000 and therefore do not need to be provisioned.
   - Note that only one trunk group can be data filled for the Avaya CS1000.
   - Additional trunks may be added to the Trunk Group.
   - Refer to the Abrazo-E Provisioning Guide, Voice Networks, PBXs (Add/Modify), Add Trunk Groups/Trunk section.

4. **Add Least Cost Routing in the Abrazo system for the Avaya CS1000** (optional). No unique configuration items for Least Cost Routing. Refer to the Abrazo-E Provisioning Guide, Voice Networks, Least Cost Routes, Add Least Cost Routes section.

9. Verification Steps

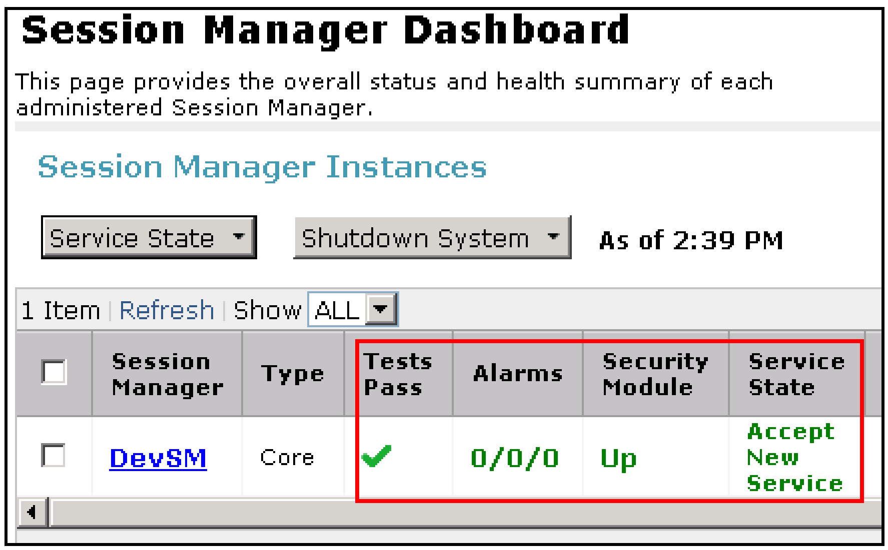
This section provides tests that can be performed to verify proper configuration of Avaya CS1000E, Session Manager and Abrazo.

9.1. Verify Avaya Aura® Session Manager

Log into System Manager with appropriate credentials and navigate **to Elements → Session Manager → Dashboard** (not shown) to verify the overall system status for Session Manager.

Specifically, verify the status of the following fields as shown below:

- Alarms: 0/0/0
- Tests Pass: ✔
- Security Module: Up
- Service State: Accept New Service

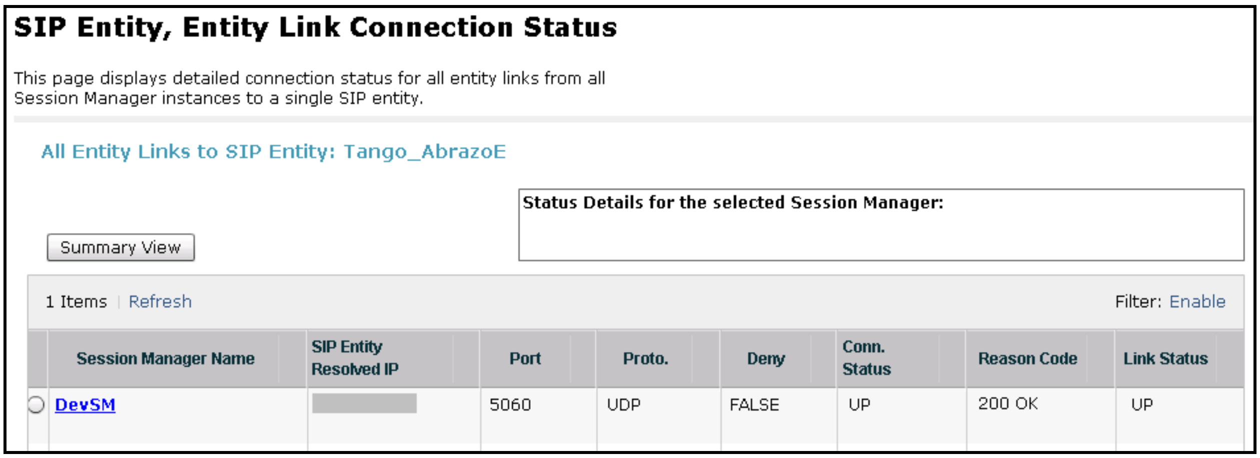


9.2. Verify Avaya Aura® Session Manager SIP Entity Link Status

From System Manager, navigate to **Elements → Session Manager → System Status → SIP Entity Monitoring** (not shown) to view more detailed status information for one of the SIP Entity Links.

Select the SIP Entity for **Abrazo-E** from the **All Monitored SIP Entities** table (not shown) to open the **SIP Entity, Entity Link Connection Status** page.

In the **All Entity Links to SIP Entity: Tango_AbrazoE**, verify the **Conn. Status** for the link is "**UP**", the **Reason Code** is "**200 OK**" and the **Link Status** is "**UP**" as shown below.



SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

All Entity Links to SIP Entity: Tango_AbrazoE

Status Details for the selected Session Manager:

Summary View

1 Items | Refresh

Filter: Enable

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

9.3. Perform Test Calls

1. From any phone on the Avaya CS1000E place a call to a desk phone that has been pared with a mobile phone. Both phones should ring. Answer the phone at the desk phone and verify the voice path. The Mobile phone should stop ringing.

2. Repeat step 1, but this time answer the mobile phone and verify the voice path. The Desk phone should stop ringing.

3. From any phone place a call to one of the paired Mobile phones. Both phones should ring. Answer the phone at the desk phone and verify the voice path. The Mobile phone should stop ringing.

4. Repeat step 3, but this time answer the mobile phone and verify the voice path. The Desk phone should stop ringing.

10. Conclusion

These Application Notes describe the configuration steps required for Tango Networks Abrazo-E 5.3 to interoperate with Avaya Communication Server 1000E, Avaya Aura® Session Manager and Avaya CallPilot™ 5.0.The testing for this sample configuration was successful. Refer to **Section 2.2** for details.

11. Additional References

This section references the product documentation relevant to these Application Notes.

Documentation for Avaya products may be found at http://support.avaya.com.

Avaya Communication Server 1000E

1) *Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000E Release 7.5*, Document Number NN43001-116
2) *Features and Services Fundamentals - Book 2 of 6 (C), Avaya Communication Server 10007.5,* Document Number NN43001-106

For Access to Tango documentation send an email to support@tango-networks.com requesting Self Service Portal access.

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.