Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Session Manager and Avaya Aura® Communication Manager with Tango Networks Enterprise Accelerator - Issue 1.0

Abstract

These Application Notes describe the procedure for configuring Tango Networks Enterprise Accelerator to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP trunking.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as any observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

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

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1. Introduction

These Application Notes describe the procedure for configuring Tango Networks Enterprise Accelerator to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP trunking.

Tango Networks Enterprise Accelerator Solution is a fixed mobile convergence (FMC) solution that employs solution components in both the enterprise network and the mobile operator network in order to seamlessly extend the corporate PBX features to the mobile phone. This convergence allows mobile phones to offer the same productivity features as a conventional enterprise desk phone.

Tango Networks Enterprise Accelerator Solution includes the Mobilizer and the Accelerator components. As shown in **Figure 1**, the Mobilizer communicates with the mobile operator network using standard protocols and always resides in the mobile operator's network or a hosting center. The Accelerator 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.

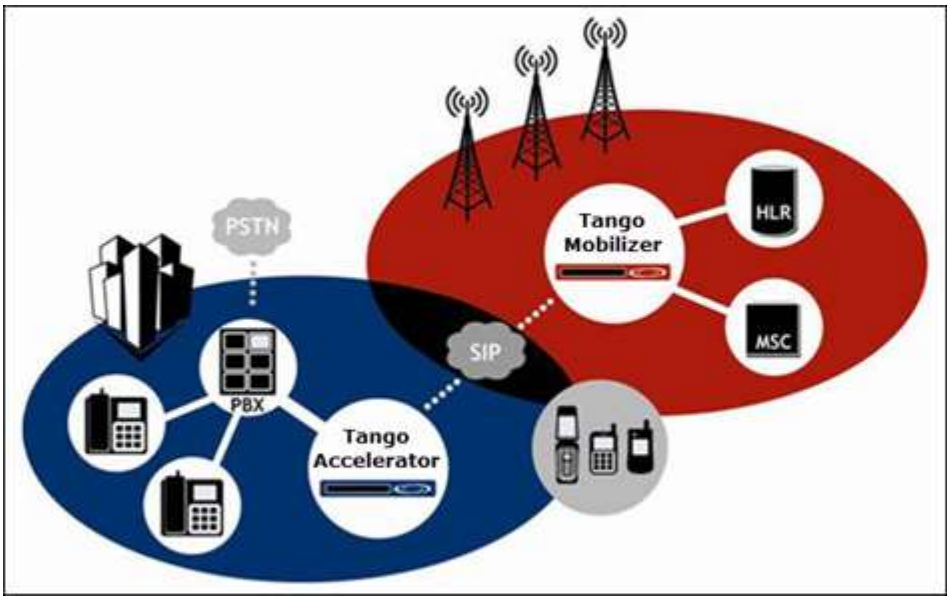


Figure 1: Tango Networks' Architecture Diagram

The Tango Networks Enterprise Accelerator Solution uses a combination of SIP lines and trunks to integrate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. SIP lines are used so that Tango Networks Enterprise Accelerator 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 Tango Networks Enterprise Accelerator solution must terminate a call via the Public Switched Telephone Network (PSTN).

1.1. Mobile Originations

The Tango Networks Enterprise Accelerator Solution captures all mobile originations from a user's mobile device and redirects them into the enterprise. This allows calls made from a mobile device to receive the same originating services (e.g., Abbreviated Dialing, Class of Service, Accounting, etc.) as a desk phone. To do this, the Tango Networks Enterprise Accelerator solution redirects the call in the wireless carrier network to a *Pilot Directory Number* (PDN) (or set of DNs). This Pilot DN is owned by the enterprise (i.e., the PSTN will route calls to it into the enterprise) and must be provisioned to route to Avaya Aura® Communication Manager. Within Avaya Aura® Communication Manager, telephony translations are created that then route these calls to the Tango Networks Enterprise Accelerator solution.

When the Tango Networks Enterprise Accelerator Solution receives calls to a Pilot DN, it replaces the Pilot DN with the original dialed digits for the call and changes the *Calling Line ID* (CLID) from the user's mobile number to the user's enterprise number. The call is then routed back to Avaya Aura® Communication Manager so that originating services can be applied to the call.

1.2. Mobile Terminations

To receive calls made to a subscriber, the Avaya Aura® Communication Manager is configured using the Off-PBX Station Mapping, and Sessing Manager using Multi-Device Access, to alert the Tango Networks Enterprise Accelerator Solution simultaneously whenever Avaya Aura® Communication Manager alerts other client devices, such as the subscriber's desk phone. The Tango Networks Enterprise Accelerator, upon receipt of this forked leg of the call, retrieves the temporary roaming number of the subscriber's mobile device from the wireless network and re-routes the call back to the Avaya Aura® Communication Manager addressed to the retrieved number.

2. General Test Approach and Test Results

The general test approach was to make mobile originating and mobile terminating calls route through the Avaya telephony infrastructure. All feature functionality test cases were performed manually. In addition, testing entailed verifying different types of Avaya Deskphones and system features interacting with the Tango Networks Enterprise Accelerator Solution. Tests were performed focusing on the following calling patterns:

- Mobile originated calls routed through the Avaya telephony infrastructure terminating to a desk phone, mobile device, or the PSTN
- Mobile terminated calls routed through the Avaya telephony infrastructure
- Desktop originated calls routed to mobile devicesand the PSTN.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability.

Feature testing focused on verifying the following:

- Abbreviated Dialing - Communication Manager allows extension dialing or internal dialing from the desktop phone. Tango Networks Enterprise Accelerator allows the user to dial these same abbreviated codes from the mobile phone.
- Call Hold and Retrieve - lets users temporarily disconnect from a call, use the telephone for another call, and then return to the original call. The Tango Networks Enterprise Accelerator solution allows for subscribers to use this service.
- Call Forward All - allows users to forward all calls to another destination, either on net or off net. Users enter a feature access code or press a Call Forward All feature button to activate or deactivate call forwarding. When an Tango Networks Enterprise Accelerator subscriber uses this feature on Communications Manager, all calls will be forwarded to the designated number. The subscriber's mobile will not ring in this scenario. When the forwarded to number is an Tango Networks Enterprise Accelerator subscriber, intelligent call delivery will ensure that both the desk phone and mobile phone ring.
- Calling Line Identification (CLID) - provides the user information about the calling party. Tango Networks Enterprise Accelerator supports calling line identification when it is the called party. Tango Networks Enterprise Accelerator also supports ensuring that the enterprise identity of the caller is preserved when a call is initiated from the mobile phone. In this case although the call is made from a mobile phone, the calling line ID will be that of the Tango Networks Enterprise Accelerator user's desktop phone.
- Calling Name Identification (CNID) - provides the user with calling party name information. When Tango Networks Enterprise Accelerator subscribers make a call from their mobile phone, Tango Networks Enterprise Accelerator adds calling name information to the call so that calling name services are supported from the mobile phone.
- Call Transfer - lets users transfer the calling party in a currently established call from their mobile phone to another destination. This is implemented by the user entering a mid-call feature code followed by the transfer to number. There are two types of call transfers that are supported by this functionality:
    - Blind Call Transfer – where the call is transferred without interaction between the user who initiated the transfer and the transfer destination.
    - Consultative Call Transfer – where the call is transferred allowing interaction between the user who initiated the transfer and the transfer destination.
- The automatic bridged line appearance feature interacts with the call transfer service for subscribers using H.323 and SIP desk phones. When a voice call is established on the desk phone and the subscriber invokes the call transfer service, a bridged line appearance remains on the desk phone. With this capability, the subscriber can simply press the bridged line appearance button to reenter the call from their desk phone.
- Class of Service - allows or denies user access to some system features. The Tango Networks Enterprise Accelerator supports COS for mobile originated calls over SIP lines.
- Class of Restrictions – Defines the restrictions that apply when a user places or receives a call. This is supported for mobile originated calls over SIP lines.
- Direct Inward Dialing – provides the user a separate number for the desk phone that can be accessed from the PSTN. The Tango Networks Enterprise Accelerator solution supports enterprise Direct Inward Dialing.
- Direct Outward Dialing – allows users inside an enterprise to dial directly to an external number. The Tango Networks Enterprise Accelerator solution supports the mobile device dialing directly to an external number.

- Enterprise Dial Tone - provides mobile subscribers with the ability to have their enterprise dial tone.
- Immediate Divert to Voice Mail - allows a user to immediately divert a call to voice mail by using a soft key on the phone. Tango Networks Enterprise Accelerator uses the mobile phone's ability to divert a call to voice mail by using the End button on the phone.
- Intelligent Call Delivery - ensures that both the desk phone and mobile phone ring when the dialed number is an Tango Networks Enterprise Accelerator subscriber.
- Least Cost Routing - For mobile originations and terminations, Tango Networks Enterprise Accelerator ensures that the least cost route is used. This results in the enterprise voice network being used to route the call as much as possible, thus reducing voice costs such as roaming.
- Multiple Calls per Line - allows multiple calls to be delivered to a single number and have the incoming call information displayed to the user. Tango Networks Enterprise Accelerator supports this feature on the mobile phone based on the ability to support call waiting for mobile phone devices.
- Single Number Services - lets a user share one number with others that he or she wishes to communicate with. When this single number is dialed, the subscriber's enterprise desktop phone as well as mobile phone will ring. This service is provided by Tango Networks Enterprise Accelerator and available when interworking with Communication Manager.
- Send All Calls - allows the user to temporarily direct all incoming calls for the desk phone and mobile phone to call coverage regardless of the assigned call-coverage redirection criteria. When Send All Calls is activated, the Tango Networks Enterprise Accelerator service is not invoked.
- Voice Mail Message Waiting Indication - provides a visible indication on the desk phone that there is a message waiting in the voice mail system. Tango Networks Enterprise Accelerator supports supplying a Message Waiting indication on the mobile phone that indicates that there are voice mail messages in the enterprise voice mail system.
- Call Pull (Desk → Mobile Call Move) – Allows a subscriber to move a phone call between the desk phone and mobile phone. Feature is invoked from the mobile phone.
- Call Push (Mobile → Desk Call Move) – Allows a subscriber to move a phone call between the mobile and desk phone. Feature is invoked from the mobile phone.

Serviceability testing focused on verifying the following:

- Business Continuity – allows calling via the mobile network when access to Session Manager is unavailable.
- Network Failure
- Service Conductor Reboot
  - Without Call
  - With Active Call

2.2. Test Results

The test objectives of **Section 2.1** were verified. The Tango Networks Enterprise Accelerator Solution successfully completed all test cases for the features identified in **Section 2.1** and is able to route inbound/outbound calls to/from the Avaya Aura® Telephony Environment with all services tested. Additionally the following behavior was observed during compliance testing.

- With regard to Tango Networks Accelerator, Business Continuity, enabling "Deny New Service" only on one of the three required entity links will not result in fault tolerance. Session Manager itself must be unreachable or set to "Deny New Service".

2.3. Support

Use the following contacts for technical support of Tango Networks Enterprise Accelerator products:

- Web site: http://www.tango-networks.com
- Email: support@tango-networks.com
- Telephone: +1 469-229-6000

3. Reference Configuration

These Application Notes describe a solution for integrating the Tango Networks Enterprise Accelerator with an Avaya Aura® Telephony Infrastructure. **Figure 2** illustrates the configuration used in these Application Notes. The diagram indicates the logical signaling connections between the Tango Networks Enterprise Accelerator and Avaya products. The solution described herein is also extensible to other Avaya Servers and Media Gateways.

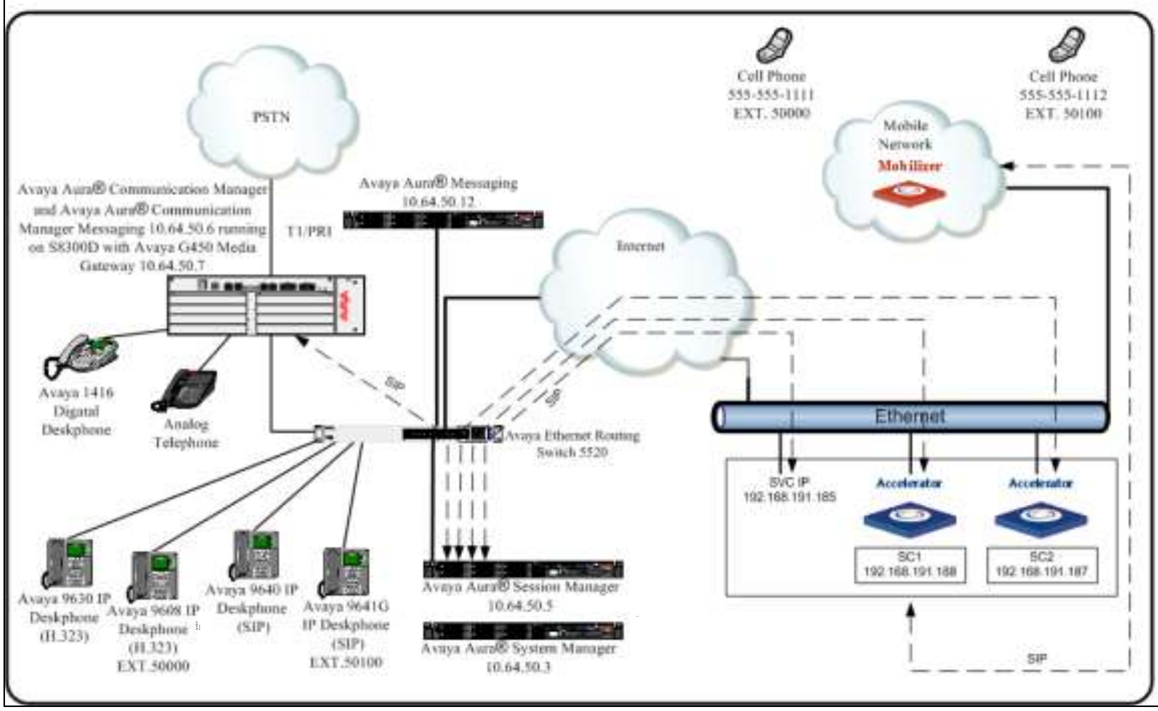


Figure 2: Compliance Test Reference Configuration

For the sample configuration shown in **Figure 2**, Session Manager runs on an HP Proliant GL360 Server and Communication Manager runs on an Avaya S8300D Server circuit board installed in an Avaya G450 Media Gateway. These Application Notes focus on the configuration of the SIP trunks and call routing.

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on Avaya S8300D Server	6.3.10.0-SP10
Avaya Aura® Session Manager running on HP Proliant GL360 Server	6.3.12.0.631208

Avaya Aura® System Manager	6.3.12.9.3022
Avaya Aura® Messaging	6.3.2 SP 2
Avaya 96x0 Deskphone	SIP R2_6_13-141010, H.323 R3_2_4-121214
Avaya 96x1 Deskphone	SIP R6_5_0-121114, H.323 R6_4_0_14-040314
Avaya 6211 and 6221 analog telephone	-
Avaya 1416 Digital Deskphone	Rel. 39.0
Tango Enterprise Accelerator	6.4

5. Configure Avaya Aura® Communication Manager

This section shows the configuration in Communication Manager. All configurations in this section are administered using the System Access Terminal (SAT). These Application Notes assumed that the basic configuration has already been administered. For further information on Communication Manager, please consult with **Reference [1].** The procedures include the following areas:

- Verify Communication Manager License
- Configure System Parameters Features
- Configure Dial Plan, ARS and Route Pattern
    - Configure outbound routing
    - Change dial plan analysis
    - Change feature access code
    - Change incoming call handling treatment
    - Change route pattern
    - Edit ARS table
- Change off PBX station mappings
- Save Changes

5.1. Verify Avaya Aura® Communication Manager License

The steps in this section verify that there are a sufficient number of SIP trunks between Communication Manager and Session Manager and SIP stations. Using the SAT, verify that the Off-PBX Telephones (OPS) and SIP Trunks features are enabled on the **System-Parameters Customer-Options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

5.2. Verify system-parameters customer-options

Issue the command **display system-parameters customer-options** to display the active licensed features. Go to **Page 1** to ensure that the **Maximum Off-PBX Telephones - OPS:** value is equal to or greater than the number of endpoints projected in the configuration.

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V16                              Software Package: Enterprise
       Location: 2                                 System ID (SID): 1
       Platform: 28                                Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 65000 58
                                      Maximum Stations: 41000 4
                              Maximum XMOBILE Stations: 41000 0
                    Maximum Off-PBX Telephones - EC500: 41000 0
                    Maximum Off-PBX Telephones -   OPS: 41000 4
                    Maximum Off-PBX Telephones - PBFMC: 41000 0
                    Maximum Off-PBX Telephones - PVFMC: 41000 0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   0
```

On **Page 2** verify that the **Maximum Administered SIP trunks** supported by the system is sufficient.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 0
           Maximum Concurrently Registered IP Stations: 18000 0
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 41000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 32
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                            Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

5.3. Configure Dial Plan, ARS, and Route Pattern

This section describes the steps for setting the Dial Plan, ARS digit analysis and Route Pattern in Communication Manager for proper routing of calls from Communication Manager destined for the PSTN via an ISDN-PRI trunk and Tango Networks Enterprise Accelerator via Session Manager.

5.3.1. Configure Outbound Routing

In these Application Notes, Automatic Route Selection (ARS) feature is used to route outbound calls via an ISDN-PRI trunk to the PSTN and to reach the Tango Networks Enterprise Accelerator PDN's via Session Manager. In the sample configuration, the single digit 9 is used as the ARS access code. Enterprise callers will dial 9 to reach an "outside line". The common configuration is illustrated below with little elaboration.

5.3.2. Change dialplan analysis

Use the **change dialplan analysis** command to define a dialed string beginning with 9 for ARS of length 1 as a feature access code (**fac**).

```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all           Percent Full: 0

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
   1           3   dac
   5           5   ext
   8           1   fac
   9           1   fac
   *           3   fac
```

5.3.3. Change feature-access-codes

Use the **change feature-access-codes** command to configure **9** as the **Auto Route Selection (ARS) – Access Code:**.

```
change feature-access-codes                                     Page   1 of  10
                              FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA:        All:        Deactivation:
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

On Page 3 set **Per Call CPN Blocking Code Access Code** to ***22** and **Per Call CPN Unblocking Code Access Code** to ***23**. These codes are used as calling id restriction codes on the Mobile phone.

```
change feature-access-codes                                     Page   3 of  10
                              FEATURE ACCESS CODE (FAC)
             Leave Word Calling Send A Message:
           Leave Word Calling Cancel A Message:
   Limit Number of Concurrent Calls Activation:        Deactivation:
               Malicious Call Trace Activation:        Deactivation:
         Meet-me Conference Access Code Change:
          Message Sequence Trace (MST) Disable:

 PASTE (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:        Dissociate Code:
        Per Call CPN Blocking Code Access Code: *22
      Per Call CPN Unblocking Code Access Code: *23
                  Posted Messages Activation:        Deactivation:
                  Priority Calling Access Code:
                           Program Access Code:

       Refresh Terminal Parameters Access Code:
              Remote Send All Calls Activation:        Deactivation:
               Self Station Display Activation:
                     Send All Calls Activation:        Deactivation:
         Station Firmware Download Access Code:
```

5.3.4. Change inc-call-handling-trmt

Change inc-call-handling-trmt, this will insert the FAC for ARS in front of the Pilot DN dialed number so calls will be routed to the Tango Networks Enterprise Accelerator via Session Manager. Additionally Direct Inward Dial (DID) numbers for the Tango Networks Enterprise Accelerator enabled stations configured on Session Manager and Communication Manager are also configured on this form. The appropriate 5 digit extension is inserted for each DID number.
Use the command **change inc-call-handling-trmt trunk-group 1**.**7205551111** is used as an example DID number. Enter the following information:

- **Number Len** should be set to 10 (the length of the DID number)
- **Number Digits** should be set to the DID number configured for the Avaya Deskphone
- **Del** should be set to **all**
- **Insert** should be set to the extension number configured for both the Avaya Deskphone and Tango Networks Enterprise Accelerator.
- Additionally an entry with **Number Digits blank,** and **Insert** set to **9** was used to insert the FAC for ARS for reaching Tango Networks Enterprise Accelerators' Pilot DN's. In the example shown below any 10 digit number other than the two DID's would be treated as a PDN.

```
change inc-call-handling-trmt trunk-group 1                     Page   1 of  30
                      INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert            Per Call Night
 Feature        Len      Digits                           CPN/BN   Serv
 public-ntwrk    10 7205550001      all 50000
 public-ntwrk    10 7205550002      all 50100
 public-ntwrk    10                      9
```

5.3.5. Change route pattern

A route pattern must be created so calls to the pilot DN are routed to the Tango Networks Enterprise Accelerator. Any number not currently in use can be used for the route pattern, for compliance testing **2** was used. Use the command **change route-pattern 2** and configure the following attributes;

- **Grp No** should be set to the value for the SIP trunk between the Communication Manager and Session Manager. In our example **2** is the trunk number for the SIP trunk.
- **FRL** should be set to **0**
- All other values can be left at their default values

```
change route-pattern 2                                          Page   1 of   3
                    Pattern Number: 2      Pattern Name: publicSM
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                             DCS/ IXC
    No          Mrk Lmt List Del  Digits                               QSIG
                             Dgts                                      Intw
 1: 2    0                                                              n   user
 2:                                                                     n   user
 3:                                                                     n   user
 4:                                                                     n   user
 5:                                                                     n   user
 6:                                                                     n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                         none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

5.3.6. Edit the ARS table

Edit the ARS table to include the translations to the route pattern, which will route the call to the Tango Networks Enterprise Accelerator. Issue the command **change ars analysis.** In our example, executed **change ars analysis 720** and enter the following:

- **Dialed String** should be set to the pilot DN
- **Min** and **Max** should be set to the length of the pilot DN number
- **Route Pattern** should be set to the number of the route pattern just created
- **Call Type** should be set to **hnpa**

The rest of the values can be left at their defaults.

```
change ars analysis 720                                         Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location: all            Percent Full: 0

          Dialed           Total      Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
           720            10   10      2        hnpa         n
```

5.4. Off-PBX Station Mapping

Every Tango Networks Enterprise Accelerator subscriber must have an off-PBX station in order to enable simultaneous ringing to the Tango Networks Enterprise Accelerator. To do this, go to the **Stations with Off-PBX Telephone Integration** screen and map the Communication Manager extension to the extension defined in the Tango Networks Enterprise Accelerator.

5.4.1. H.323 Phone Off-PBX Station Mapping

In the example below, the H.323 station extension is 5000 and will need an off-PBX station entry to enable simultaneous ringing to the endpoint off of the Tango Networks Enterprise Accelerator. Use the **change off-pbx-telephone station-mapping 50000** command to configure the station.

- Set **Application** to **OPS**
- Set **Phone** Number to the number Tango Networks Enterprise Accelerator will use for call originations and terminations, which is the user portion of the SIP address defined for the subscriber on the Tango Networks Enterprise Accelerator.
- Set **Trunk Selection** to **aar**
- Set **Configuration Set** to the set to be used for IP phone call treatments

```
change off-pbx-telephone station-mapping 50000                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 50000             OPS          -     50000            aar        1
```

5.4.2. SIP Phone Off-PBX Station Mapping

Avaya SIP Deskphones also require and off-pbx-telephone station-mapping, however this will be configured in **Section 6** using System Manager. The screen below is the result of the configuration performed on System Manager.

```
change off-pbx-telephone station-mapping 50100                  Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                     Selection   Set     Mode
 50100             OPS          -     50100            aar        1
```

5.4.3. Change Off-PBX Feature Name Extensions

Off-pbx-telephone feature-name-extensions are required for use by Tango Networks Enterprise Accelerator Solution and a feature name extension should be configured for Active Appearance and Transfer to Voice Mail. Use the **change off-pbx-telephone feature-name-extensions set 1** command to set the **Active Appearance Select** to **50990.**

```
change off-pbx-telephone feature-name-extensions set 1          Page   1 of   2
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
                     Set Name:

        Active Appearance Select: 50990
             Automatic Call Back:
      Automatic Call-Back Cancel:
                Call Forward All:
     Call Forward Busy/No Answer:
             Call Forward Cancel:
                       Call Park:
           Call Park Answer Back:
                    Call Pick-Up:
            Calling Number Block:
          Calling Number Unblock:
   Conditional Call Extend Enable:
  Conditional Call Extend Disable:
             Conference Complete:
            Conference on Answer:
           Directed Call Pick-Up:
           Drop Last Added Party:
```

On Page 2 set **Transfer to Voice Mail** to **59991**.

```
change off-pbx-telephone feature-name-extensions set 1          Page   2 of   2
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

       Exclusion (Toggle On/Off):
      Extended Group Call Pickup:
          Held Appearance Select:
          Idle Appearance Select:
              Last Number Dialed:
            Malicious Call Trace:
     Malicious Call Trace Cancel:
             Off-Pbx Call Enable:
            Off-Pbx Call Disable:
                   Priority Call:
                          Recall:
                  Send All Calls:
           Send All Calls Cancel:
               Transfer Complete:
             Transfer On Hang-Up:
          Transfer to Voice Mail: 59991
         Whisper Page Activation:
```

5.5. Change OPTIM Failure on Trunk Group between Communication Manager and Session Manager

The Redirect On OPTIM Failure parameter should be increased to 30 seconds. Lower values may cause routing of calls for off-pbx stations to the Tango Networks Enterprise Accelerator to be cancelled. Use the **change trunk-group 2** command and advance to **Page 2** to set the **Redirect On OPTIM Failure** parameter to **30000** milliseconds.

```
change trunk-group 2                                          Page   2 of  21
     Group Type: sip

TRUNK PARAMETERS

    Unicode Name: auto

                                    Redirect On OPTIM Failure: 30000
```

5.6. Save Changes

Use the **save translation** command to save all changes.

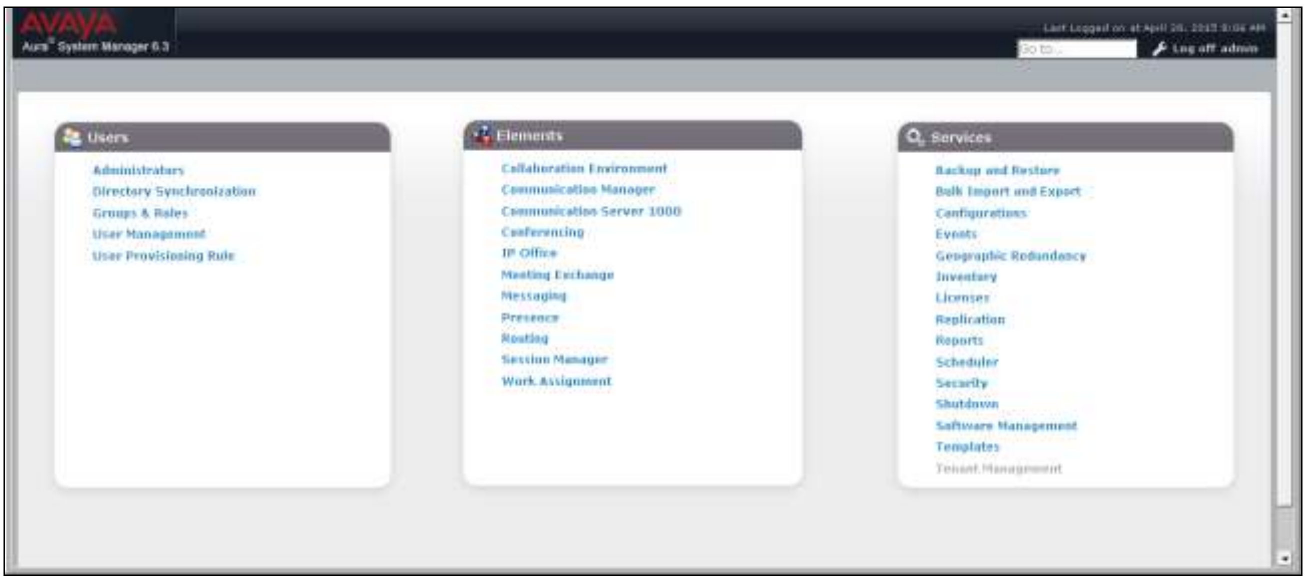
```
save translation

                              SAVE TRANSLATION

          Command Completion Status                                    Error
Code

          Success                                                      0








Command successfully completed
Command:
```

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager, assuming it has been installed and licensed as described in **Reference [2]**. The procedures include adding the following items:

- Add SIP Domain
- Add SIP Entities and Entity Links
- Add Routing Policies
- Add Dial Patterns
- Add Users for Tango Subscribers

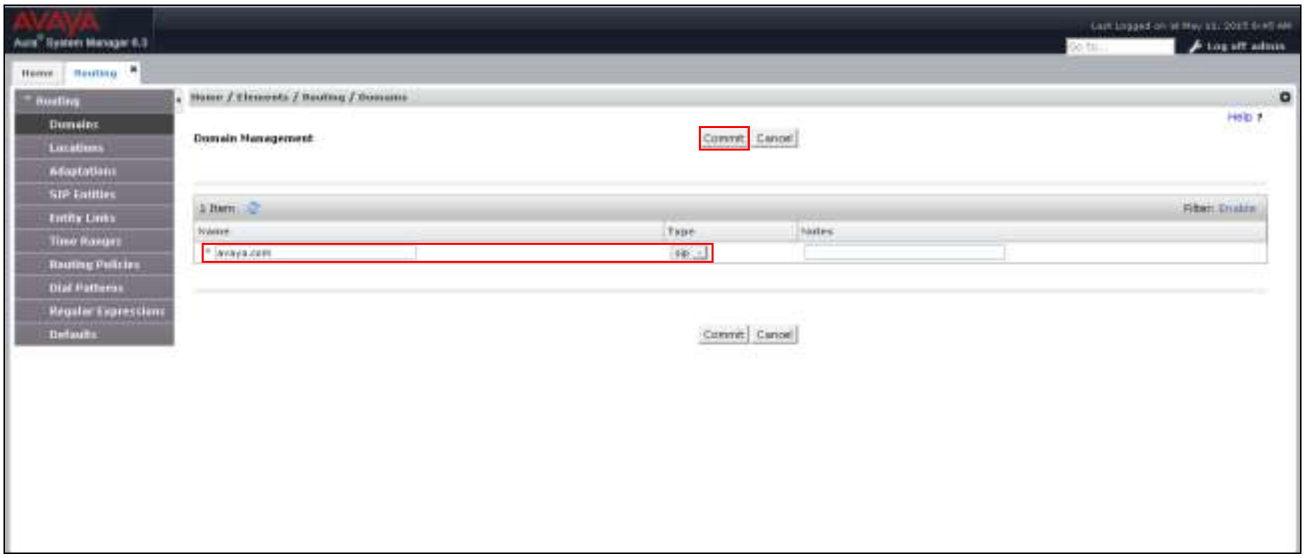
Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR**, where **<ip-address>** is the IP address of System Manager. Log in with the appropriate credentials and accept the Copyright Notice (not shown). The home screen as shown below is displayed. Expand the **Routing** Link under **Elements**.



6.1. Add SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button on the right. The following screen will then be shown. Fill in the following fields and click **Commit**.

- **Name:** The authoritative domain name (e.g. **avaya.com**)
- **Type** Select **sip**
- **Notes:** Descriptive text (optional)



6.2. Add SIP Entities and SIP Entity Links

A SIP Entity is required for each SIP-based telephony system wishig to communicate with Session Manager for call routing. During compliance testing the Tango Networks Enterprise Accelerator was provisioned as a fault tolerant system with three components and required three SIP Entities to be configured on Session Manager. The three components included the Service IP (SVC IP) for communicating to the Tango Networks Enterprise Accelerator and two Session Conductors (SC) for communicating from the Tango Networks Enterprise Accelerator to Session Manager.

Note: *When the Tango Networks Enterprise Accelerator is provisioned as a single node solution it will be identified by a single IP Address and only one SIP Entity configuration is required in System Manager.*

6.2.1. Adding SIP Entity Link for the Tango Networks Enterprise Accelerator SVC IP

Navigate to **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right.

Under **General**:

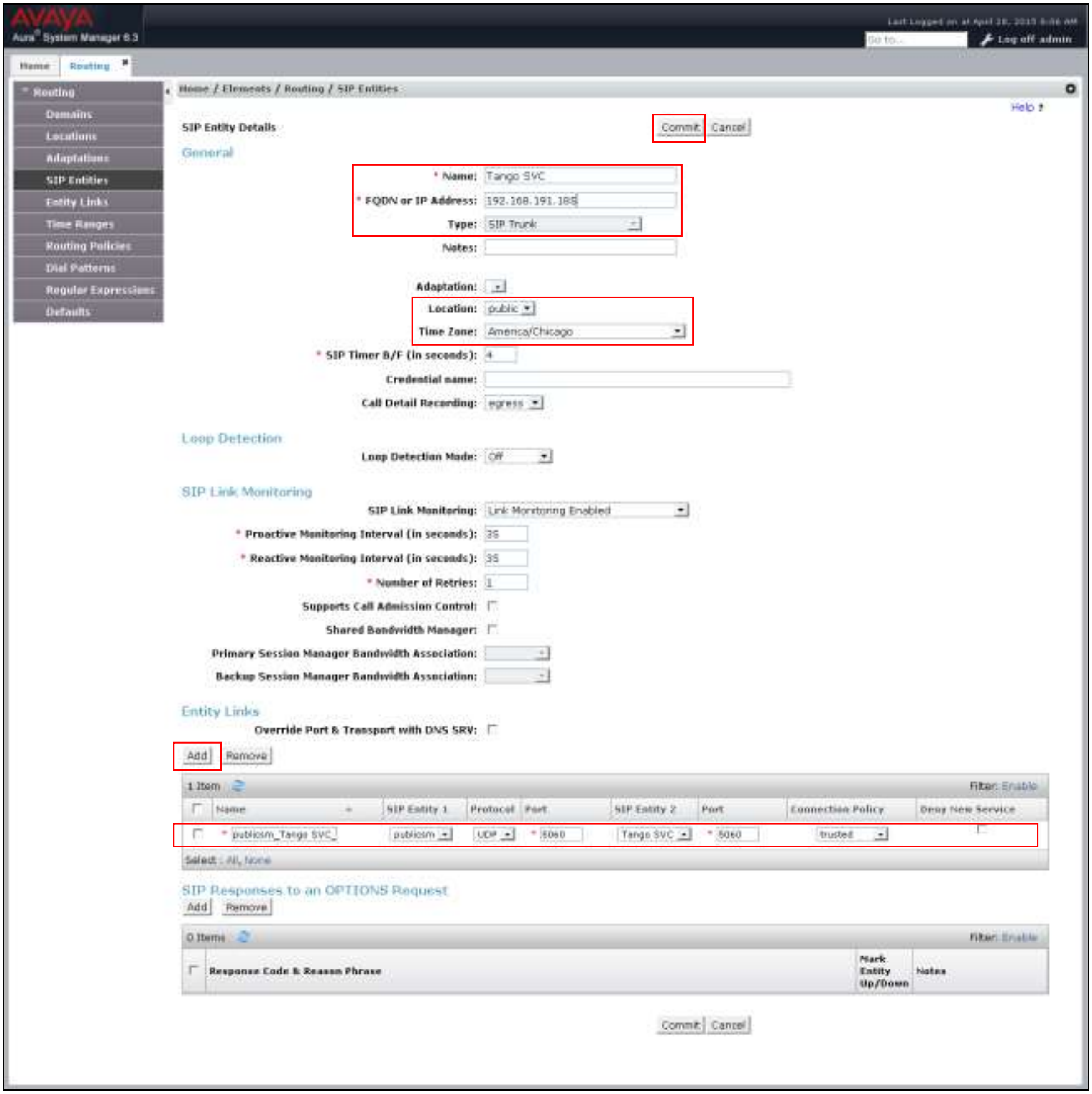
- **Name:** A descriptive name, e.g.. **Tango SVC**
- **FQDN or IP Address:** IP address of the Tango Accelerator SVC IP i.e. **192.168.191.185**
- **Type:** Select **SIP Trunk**

- **Location:** Select the appropriate location (e.g. **public**)
- **Time Zone:** Time zone for this entity

Add Entity Links. Under **Entity Links**, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Name** Will be populated automatically
- **SIP Entity 2** Will be populated automatically with the name of this SIP Entity.
- **SIP Entity 1** Select Session Manager from the pull down box
- **Protocol** Select the **UDP** from the pull down box
- **Port** Enter **5060** for the Entity Link
- **Connection Policy** Select **trusted** from the pull down box

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition. The following screen shows the addition of the SIP Entity for **Tango SVC**.


AVAYA
Aura System Manager 6.3
Log off admin
Home / Elements / Routing / SIP Entities
SIP Entity Details
Commit
Cancel
General
Name: Tango SVC
FQDN or IP Address: 192.168.191.188
Type: SIP Trunk
Notes:
Adaptation:
Location: public
Time Zone: America/Chicago
SIP Timer B/F (in seconds): 4
Credential name:
Call Detail Recording: egress
Loop Detection
Loop Detection Mode: Off
SIP Link Monitoring
SIP Link Monitoring: Link Monitoring Enabled
Proactive Monitoring Interval (in seconds): 35
Reactive Monitoring Interval (in seconds): 35
Number of Retries: 1
Supports Call Admission Control:
Shared Bandwidth Manager:
Primary Session Manager Bandwidth Association:
Backup Session Manager Bandwidth Association:
Entity Links
Override Port & Transport with DNS SRV:
Add
Remove
1 Item
Name
SIP Entity 1
Protocol
Port
SIP Entity 2
Port
Connection Policy
Deny New Service
publicsm_Tango SVC_
publicsm
UDP
5060
Tango SVC
5060
trusted
SIP Responses to an OPTIONS Request
0 Items
Response Code & Reason Phrase
Mark Entity Up/Down
Notes

6.2.2. Adding SIP Entity Link for the Tango Networks Enterprise Accelerator SC1

Navigate to **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right.

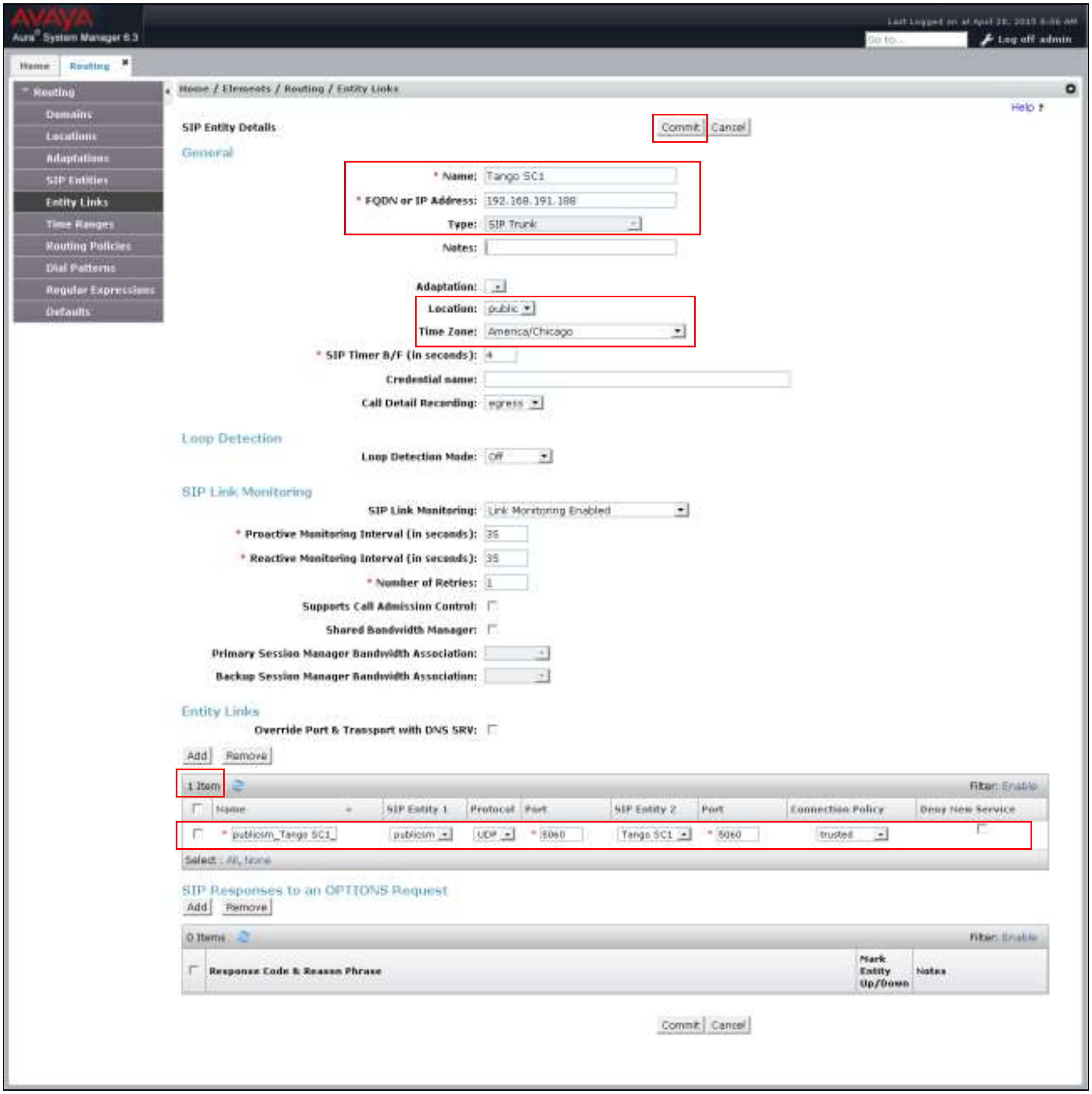
Under **General**:

- **Name:** A descriptive name, i.e. **Tango SC1**
- **FQDN or IP Address:** IP address of the Tango Accelerator SVC IP i.e. **192.168.191.188**
- **Type:** Select **SIP Trunk**
- **Location:** Select the appropriate location (e.g. **public**)
- **Time Zone:** Time zone for this entity

Add Entity Links. Under **Entity Links**, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Name** Will be populated automatically
- **SIP Entity 2** Will be populated automatically with the name of this SIP Entity.
- **SIP Entity 1** Select Session Manager from the pull down box
- **Protocol** Select the **UDP** from the pull down box
- **Port** Enter **5060** for the Entity Link
- **Connection Policy** Select **trusted** from the pull down box

- Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition. The following screen shows the addition of the SIP Entity for **Tango SC1**.



6.2.3. Adding SIP Entity Link for the Tango Networks Enterprise Accelerator SC2

Navigate to **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right.

Under **General**:

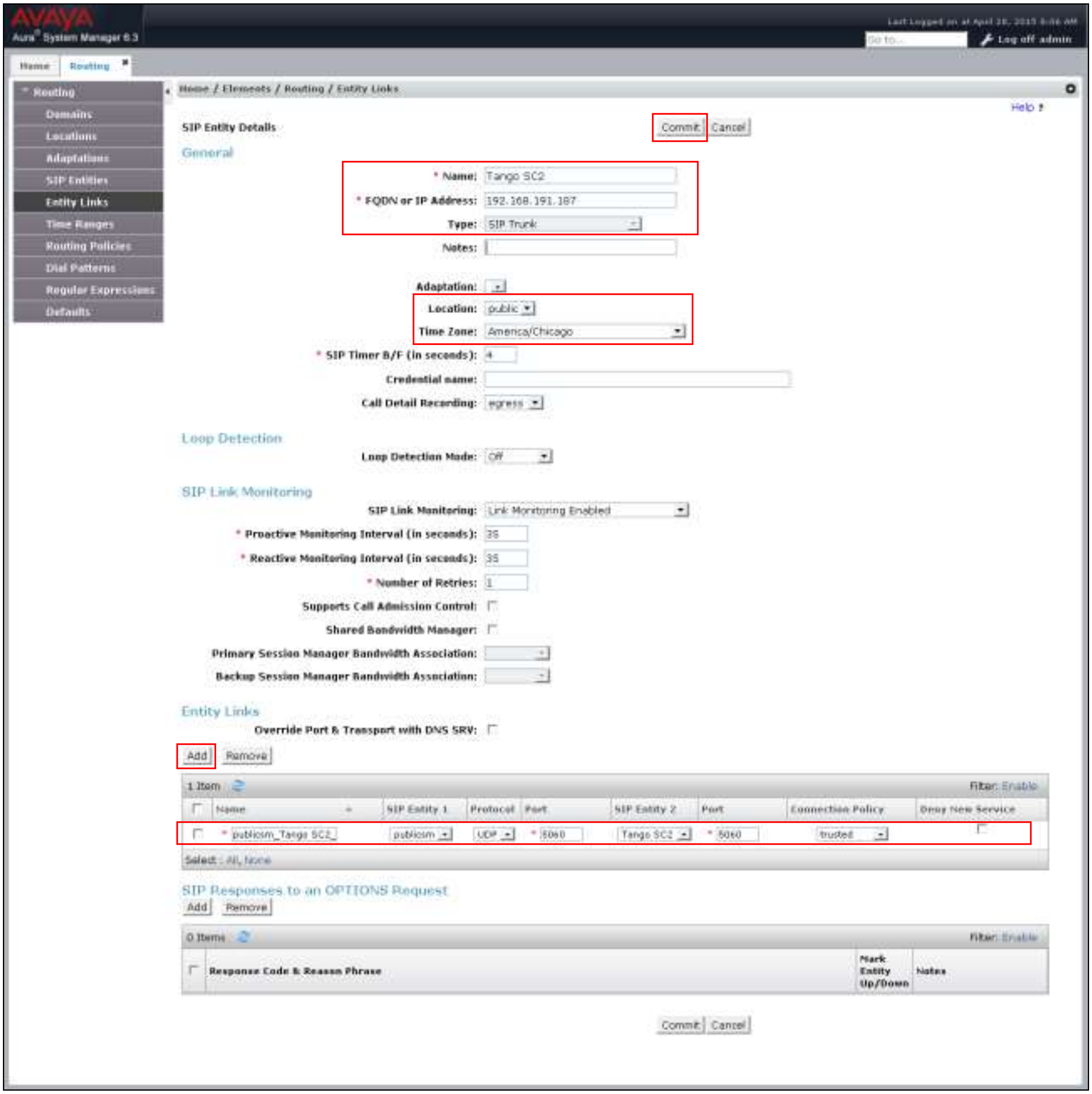
- **Name:** A descriptive name, i.e. **Tango SC2**
- **FQDN or IP Address:** IP address of the Tango Accelerator SVC IP i.e. **192.168.191.187**
- **Type:** Select **SIP Trunk**

- **Location:** Select the appropriate location (e.g. **public**)
- **Time Zone:** Time zone for this entity

Add Entity Links. Under **Entity Links**, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Name** Will be populated automatically
- **SIP Entity 2** Will be populated automatically with the name of this SIP Entity.
- **SIP Entity 1** Select Session Manager from the pull down box
- **Protocol** Select the **UDP** from the pull down box
- **Port** Enter **5060** for the Entity Link
- **Connection Policy** Select **trusted** from the pull down box

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition. The following screen shows the addition of the SIP Entity for **Tango SC2**.



AVAYA
Aura® System Manager 6.3
Log off admin
Home
Routing
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Home / Elements / Routing / Entity Links
Help ?
SIP Entity Details
Commit
Cancel
General
* Name: Tango SC2
* FQDN or IP Address: 192.168.191.187
Type: SIP Trunk
Notes:
Adaptation:
Location: public
Time Zone: America/Chicago
* SIP Timer B/F (in seconds): 4
Credential name:
Call Detail Recording: egress
Loop Detection
Loop Detection Mode: Off
SIP Link Monitoring
SIP Link Monitoring: Link Monitoring Enabled
* Proactive Monitoring Interval (in seconds): 35
* Reactive Monitoring Interval (in seconds): 35
* Number of Retries: 1
Supports Call Admission Control:
Shared Bandwidth Manager:
Primary Session Manager Bandwidth Association:
Backup Session Manager Bandwidth Association:
Entity Links
Override Port & Transport with DNS SRV:
Add
Remove
1 Item
Filter: Enable
Name
SIP Entity 1
Protocol
Port
SIP Entity 2
Port
Connection Policy
Deny New Service
* publicsm_Tango SC2_
publicsm
UDP
* 5060
Tango SC2
* 5060
trusted
Select : All, None
SIP Responses to an OPTIONS Request
Add
Remove
0 Items
Filter: Enable
Response Code & Reason Phrase
Mark Entity Up/Down
Notes
Commit
Cancel

6.3. Add Routing Policies

Routing policies describe the condition under which calls will be routed to the SIP Entities specified in **Section 6.2**. During compliance testing a routing policy was be added for the Tango Pilot DN's. To add a routing policy, select **Routing Policies** on the left and click on the **New** button on the right. The following screen is displayed. Fill in the following:

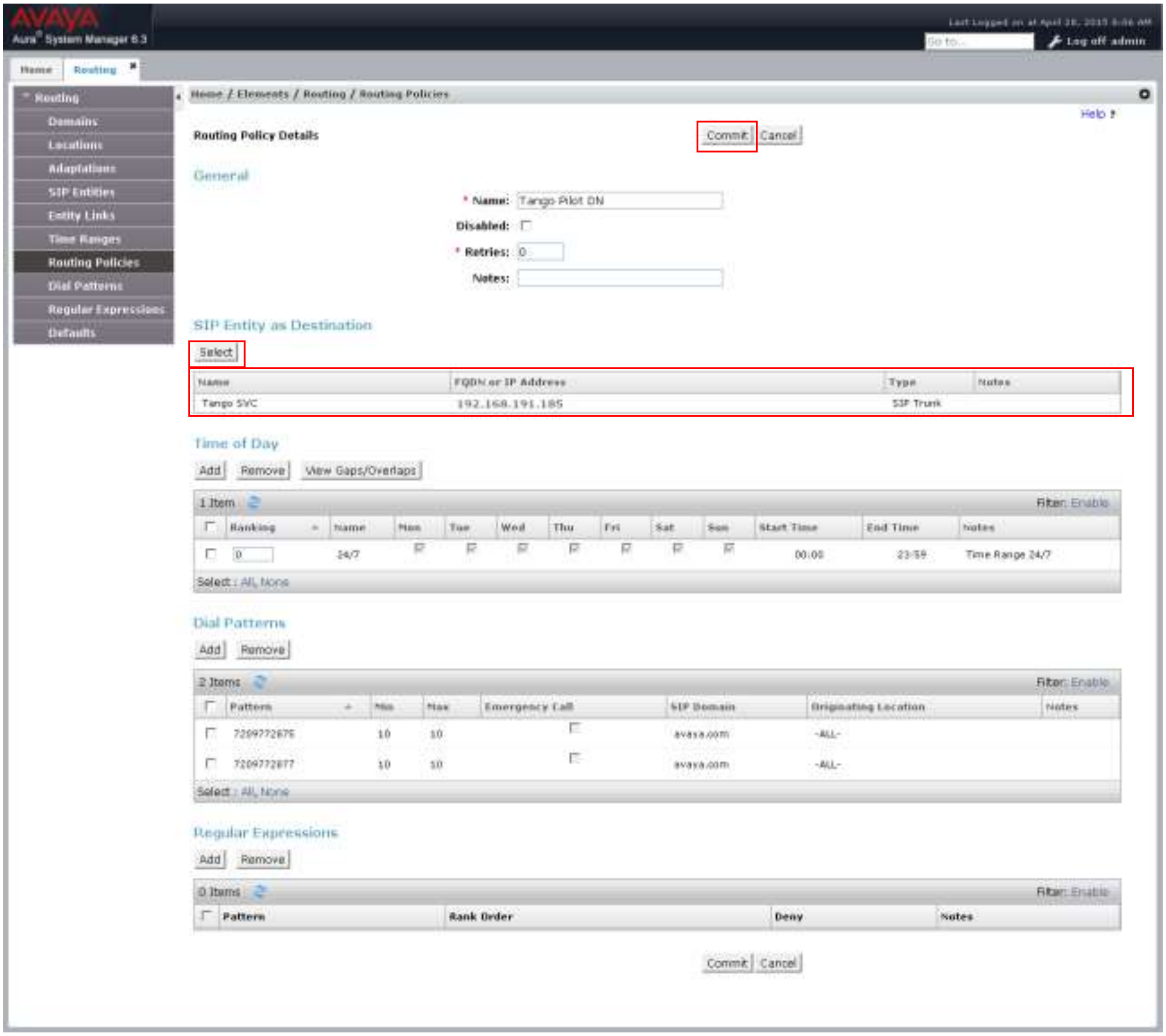
Under **General**

- Enter a descriptive **Name** i.e., **Tango Pilot DN**

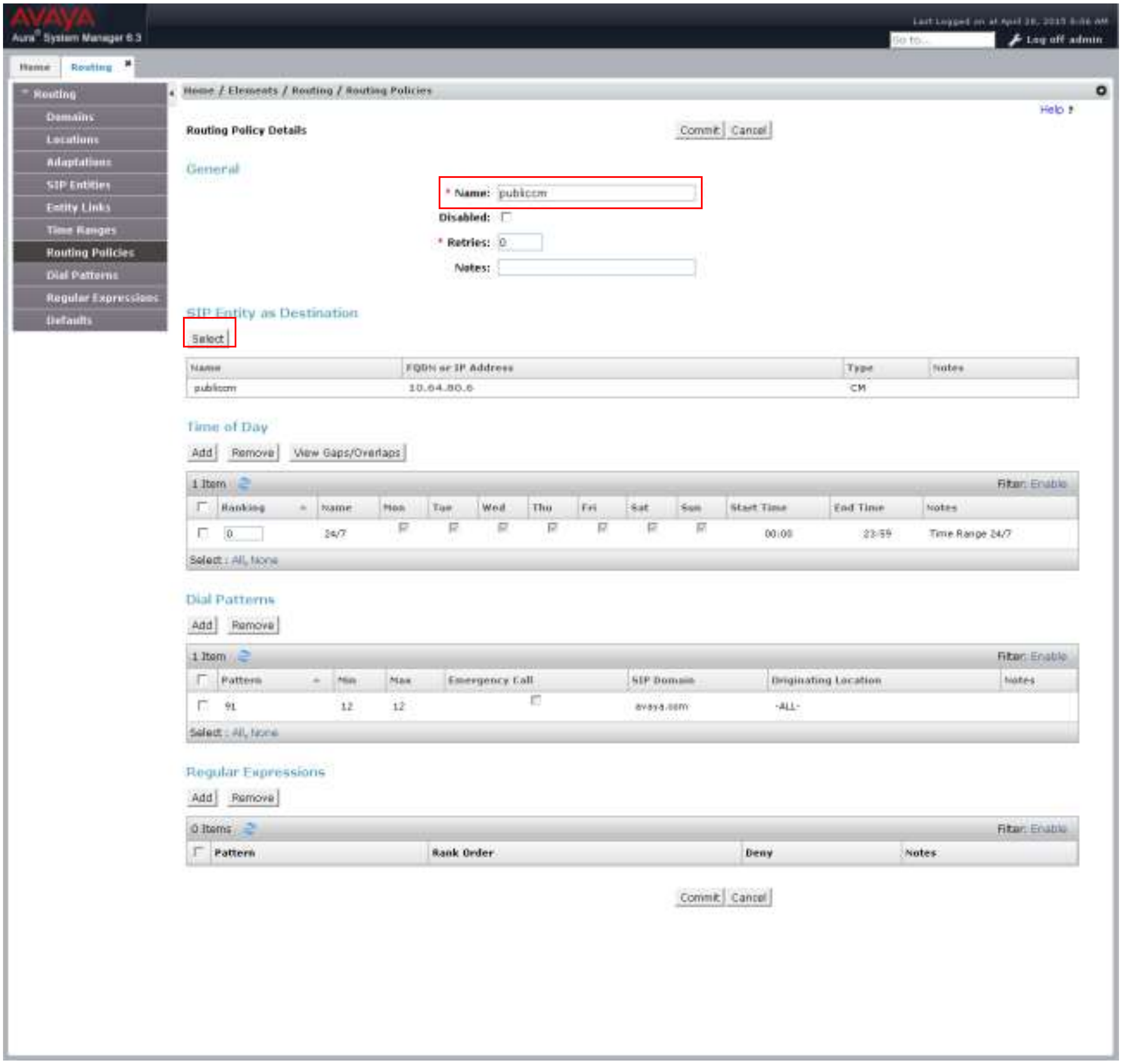
Under **SIP Entity as Destination**

- Click **Select**, and then select the **Tango SVC** SIP entity.

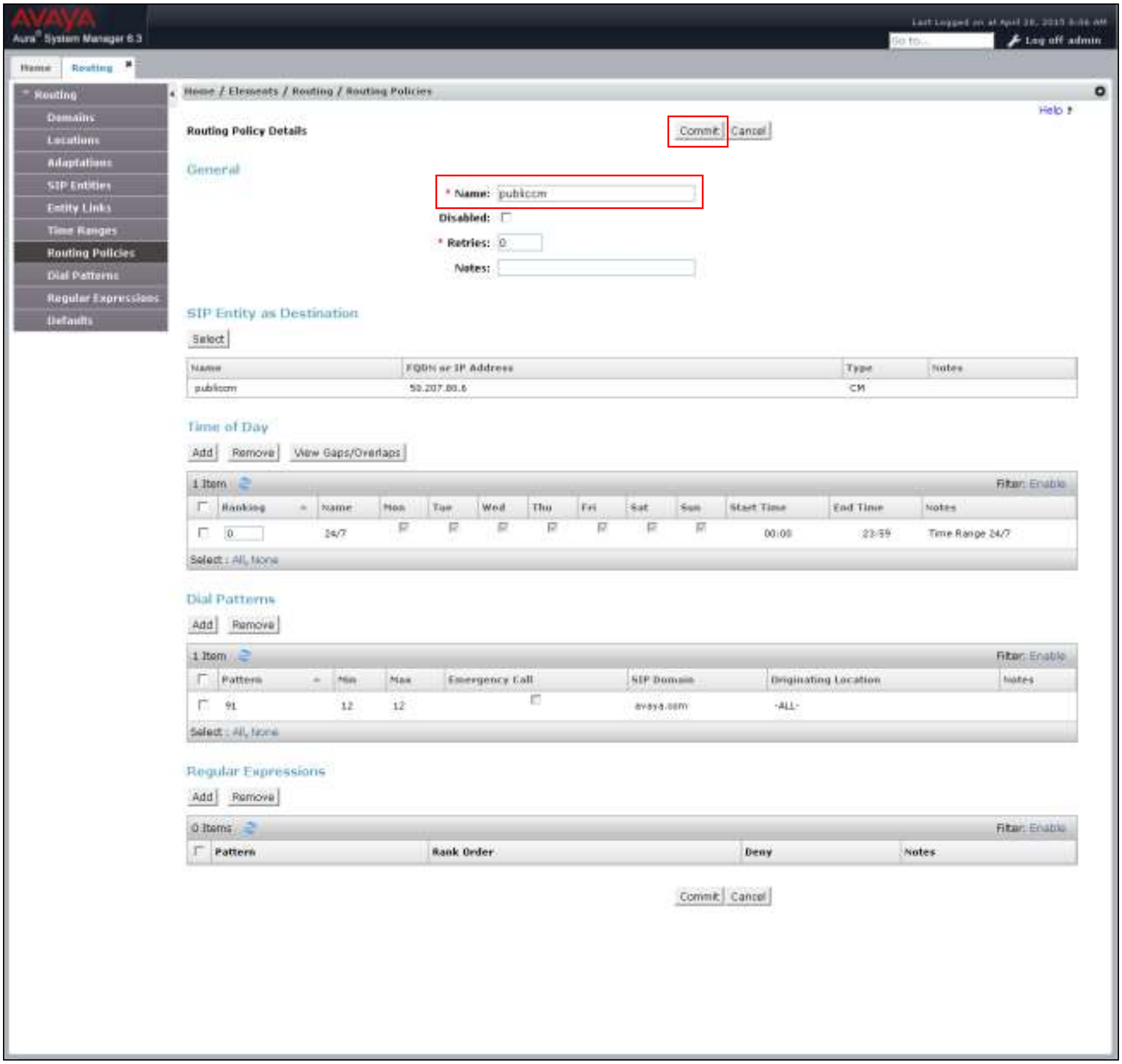
Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screen shows the Routing Policies for **Tango SVC**. Note that **Dial Patterns** (to be configured in **Section 6.6**), when configured, will be automatically displayed in the **Routing Policy Details** page.



AVAYA
Aura System Manager 6.3
Log off admin
Home / Elements / Routing / Routing Policies
Routing
Domains
Locations
Adaptations
SIP Entities
Entity Links
Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults
Routing Policy Details
Commit
Cancel
General
Name: Tango Pilot DN
Disabled:
Retries: 0
Notes:
SIP Entity as Destination
Select
Tango SVC
192.168.191.185
SIP Trunk
Time of Day
Add
Remove
View Gaps/Overlaps
24/7
00:00
23:59
Time Range 24/7
Dial Patterns
7209772876
7209772877
avaya.com
-ALL-
Regular Expressions



Note: *The existing Routing Policy shown below was used on Session Manager for routing PSTN calls.*



6.4. Add Dial Patterns

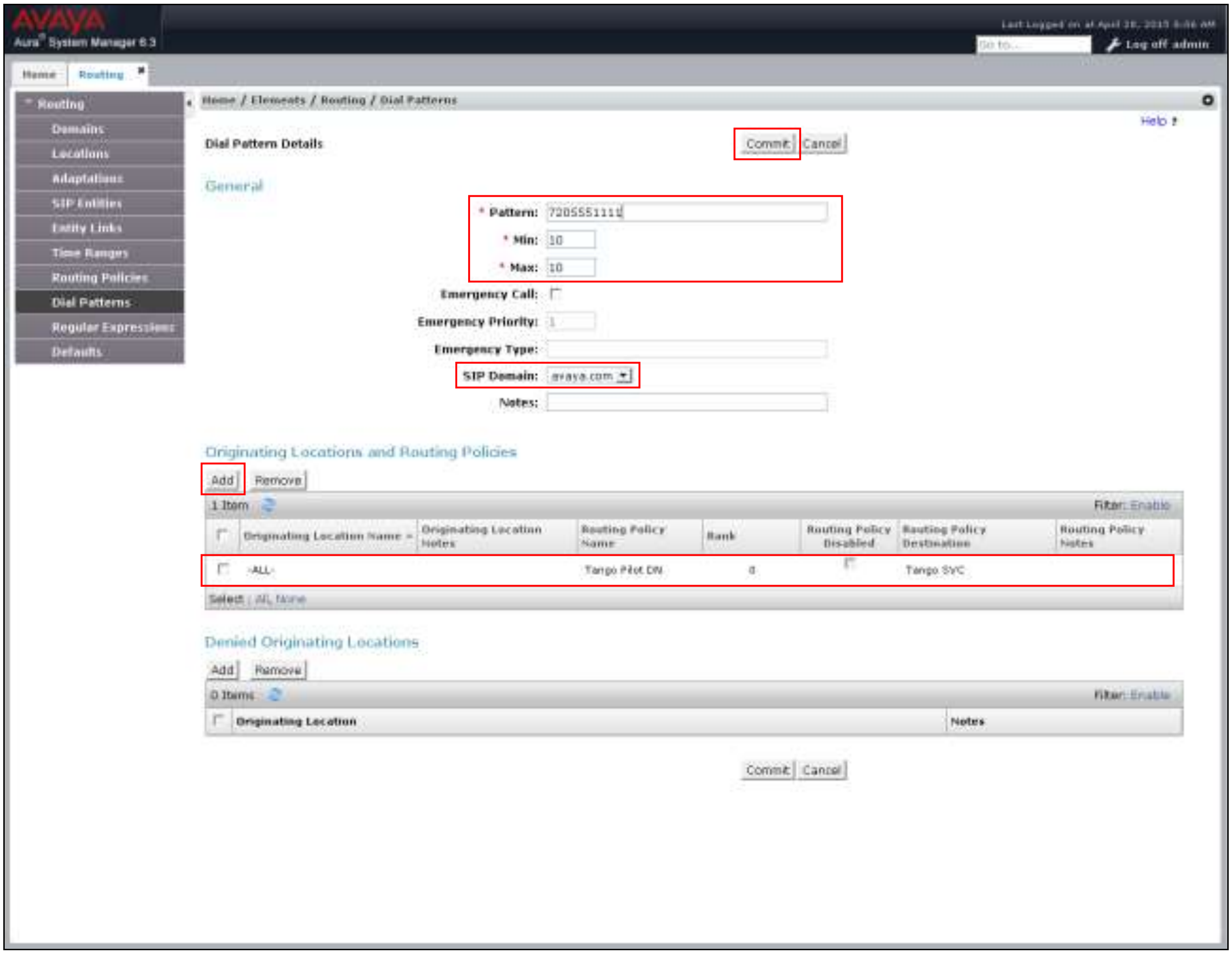
Dial patterns must be defined that will direct calls to the appropriate SIP Entity. During compliance testing two Dial Patterns were added for routing calls to the Pilot DN's to the Tango Networks Enterprise Accelerator. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button on the right. Fill in the following, as shown in the screen below, which corresponds to one of the dial patterns used for routing calls to the Tango Networks Enterprise Accelerator.

Under **General**:

- **Pattern:** Dialed number or prefix i.e. **7205551111**
- **Min:** Minimum length of dialed number i.e. **10**
- **Max:** Maximum length of dialed number i.e. **10**
- **SIP Domain:** Select **avaya.com**

Under **Originating Locations and Routing Policies**, click **Add**, and then select the appropriate location and routing policy from the list. In this example **All** was selected for **Originating Location Name** and **Tango Pilot DN** was selected for **Routing Policy Name**. Default values can be used for the remaining fields. Click **Commit** to save this dial pattern.

The following screen shows the dial pattern definition for calls to the Tango Networks Enterprise Accelerator.

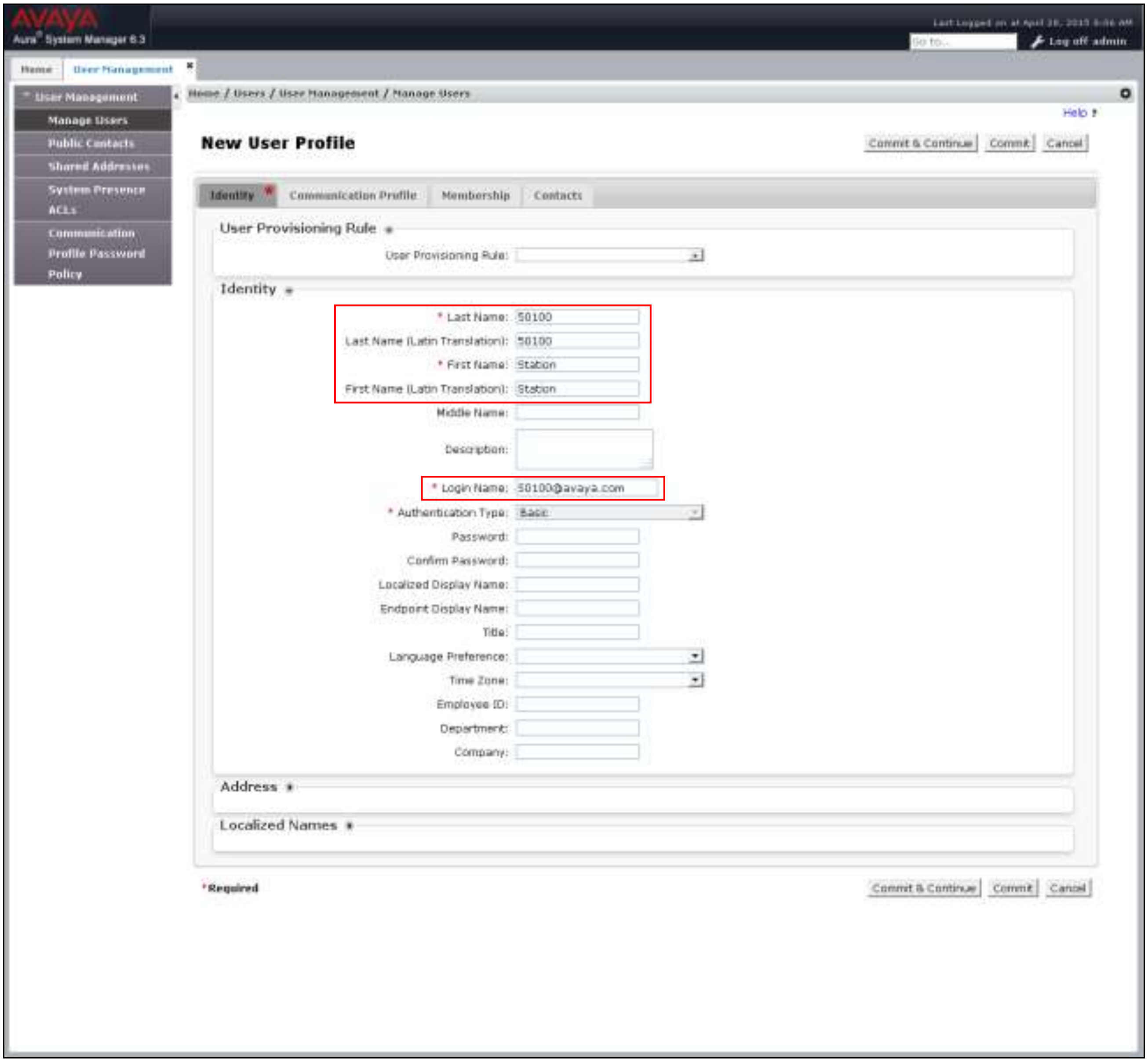


6.5. Add Users

From the home screen select **Users → User Management → Manage Users** to display the **User Management** screen (not shown). Click **New** to add a user.

6.5.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter "n@z", where "n" is the user extension and "z" is the domain name, in this case "**avaya.com**" used for compliance testing. Retain the default values in the remaining fields.


AVAYA
Aura® System Manager 6.3
Log off admin
Home
User Management
User Management
Manage Users
Public Contacts
Shared Addresses
System Presence ACLs
Communication Profile Password Policy
Home / Users / User Management / Manage Users
Help ?
New User Profile
Commit & Continue
Commit
Cancel
Identity
Communication Profile
Membership
Contacts
User Provisioning Rule
User Provisioning Rule:
Identity
* Last Name: 50100
Last Name (Latin Translation): 50100
* First Name: Station
First Name (Latin Translation): Station
Middle Name:
Description:
* Login Name: 50100@avaya.com
* Authentication Type: Basic
Password:
Confirm Password:
Localized Display Name:
Endpoint Display Name:
Title:
Language Preference:
Time Zone:
Employee ID:
Department:
Company:
Address
Localized Names
*Required
Commit & Continue
Commit
Cancel

6.5.2. Communication Profile

Select the **Communication Profile** tab. For **Communication Profile Password** and **Confirm Password**, enter the desired password for the SIP user to use for registration. Scroll down to the **Communication Address** sub-section, and click **New** to add a new address.

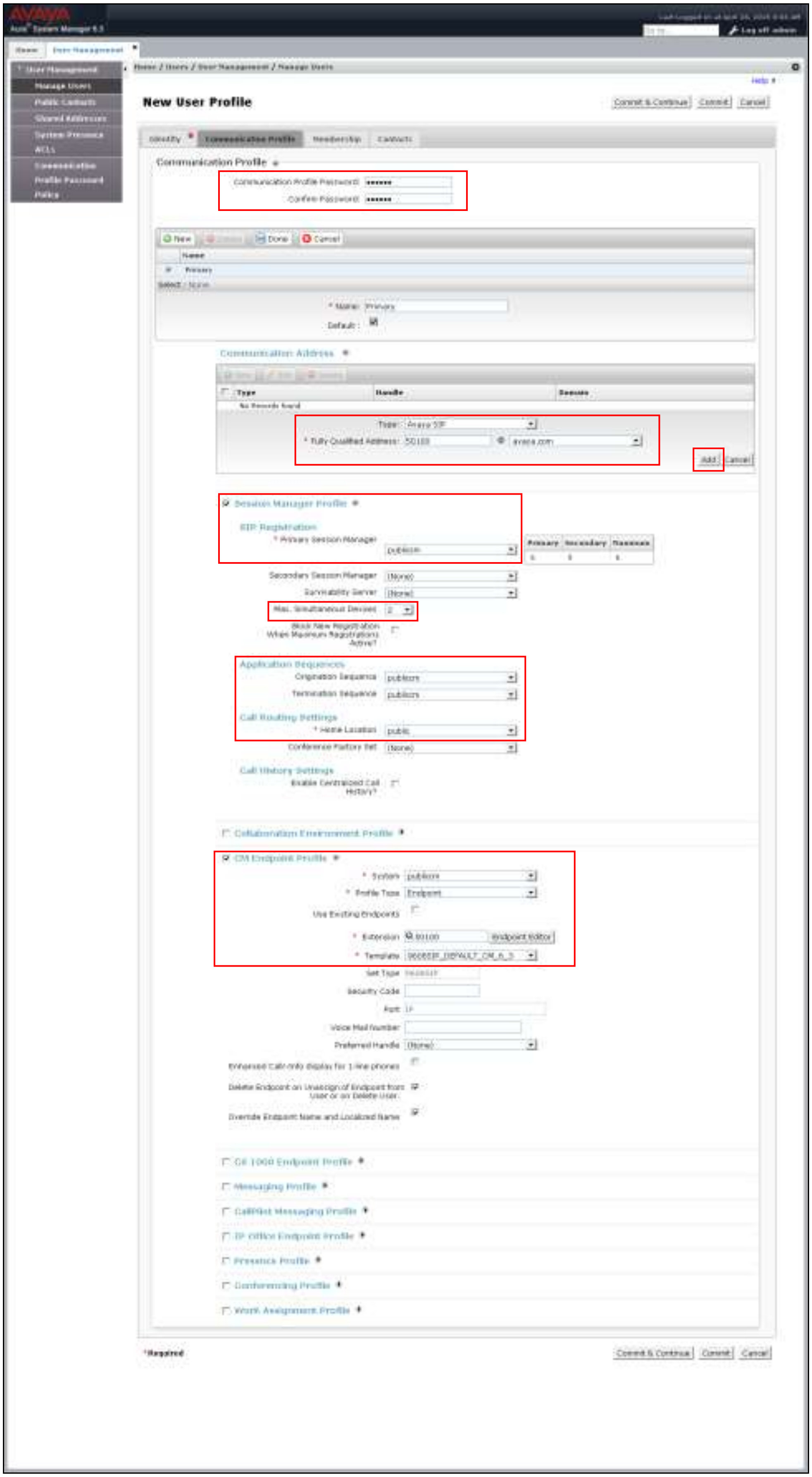
For **Type**, retain "**Avaya SIP**". For **Fully Qualified Address**, enter and select the SIP user extension and domain configured in **Section 6.5.1**. Click **Add**.

Scroll down to check and expand **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager. SIP line integration with the Tango Networks Enterprise Accelerator requires that **Max. Simultaneous Devices** be incremented by one. This value is set to **1** by default. During compliance testing Avaya SIP Deskphones were set to **2**. Retain the default values in the remaining fields. These settings are configured during the initial setup of Session Manager.

Note: *Incrementing **Max. Simultaneous Devices** is not required for H.323 Deskphones.*

Scroll down to check and expand **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter or select the SIP user extension configured in **Section 6.5.1**. For **Template**, select corresponding Telephone type. Retain the default values in the remaining fields.

Click **Commit** to complete the creation of the new user.


New User Profile
Commit & Continue
Commit
Cancel
Identity
Communication Profile
Membership
Contacts
Communication Profile
Communication Profile Password:
Confirm Password:
New
Done
Cancel
Name
Primary
* Name:
Default :
Communication Address
Type
Handle
Domain
Type:
* Fully Qualified Address:
Add
Cancel
Session Manager Profile
SIP Registration
* Primary Session Manager
Primary
Secondary
Maximum
Secondary Session Manager
Survivability Server
Max. Simultaneous Devices
Application Sequences
Origination Sequence
Termination Sequence
Call Routing Settings
* Home Location
Conference Factory Set
Call History Settings
Collaboration Environment Profile
CM Endpoint Profile
* System
* Profile Type
Use Existing Endpoints
* Extension
Endpoint Editor
* Template
Set Type
Security Code
Port
Voice Mail Number
Preferred Handle
CS 1000 Endpoint Profile
Messaging Profile
CallPilot Messaging Profile
IP Office Endpoint Profile
Presence Profile
Conferencing Profile
Work Assignment Profile
*Required
Commit & Continue
Commit
Cancel

7. Tango Networks Enterprise Accelerator

This document assumes that the Tango Networks Enterprise Accelerator has already been provisioned with:

- Enterprise information
- Wireless carrier information

The integration process includes the following steps:

- Create a Trunk Dial Plan
- Add Session Manager
- Add a Trunk Group/Trunk
- Add a Line Group/Line
- Feature Access Codes
- Add Voice Mail Server
- Add Subscriber Dial Plan
- Add Subscriber

The steps below describe the unique configuration areas needed to integrate Communication Manager and Session Manager with the Tango Networks Enterprise Accelerator solution. Refer to the Tango Networks Enterprise Accelerator Provisioning Guide for a comprehensive explanation of Tango Networks Enterprise Accelerator provisioning.

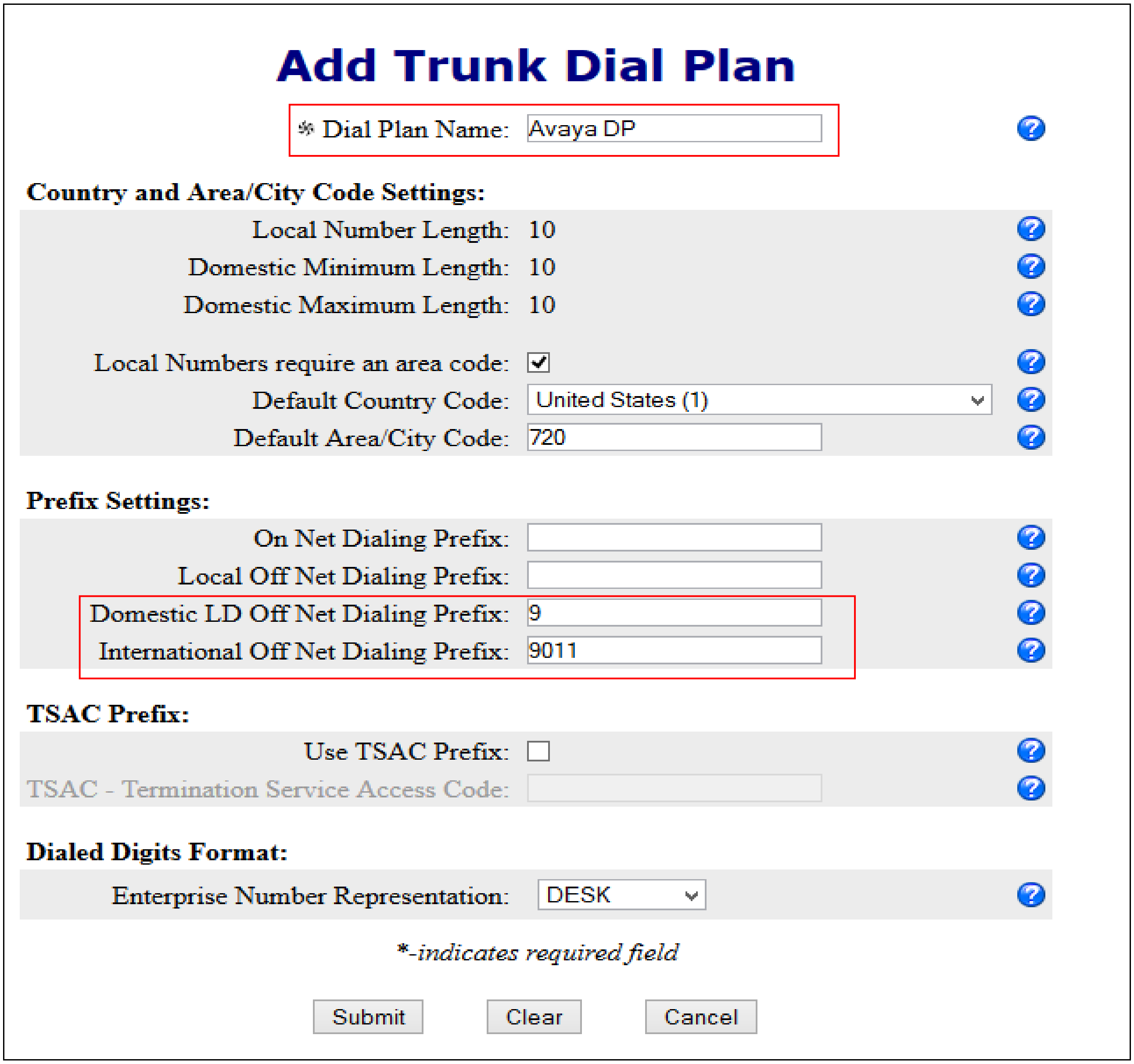
Configuration is accomplished by accessing the browser-based GUI of the Tango Accelerator, using the URL **http://<ip-address>:8443/provisioning**, where **<ip-address>** is the IP address of primary Tango provisioning node.

7.1.1. Create a Trunk Dial Plan

Create a Trunk Dial Plan to support routing prefixes as defined in the ARS table in Communication Manger. To add a new dial plan select **Voice Network → PBX → Trunk Dial Plan → Add** (not shown).

- **Dial Plan Name** – Something unique to identify this dial plan.
- **Domestic LD Off Net Dialing Prefix** was set to **9** this was ARS Access Code prefix defined in Communication Manager.
- **International Off Net Dialing Prefix** was set to 9011.
- All remaining fields can remain set to their default values.

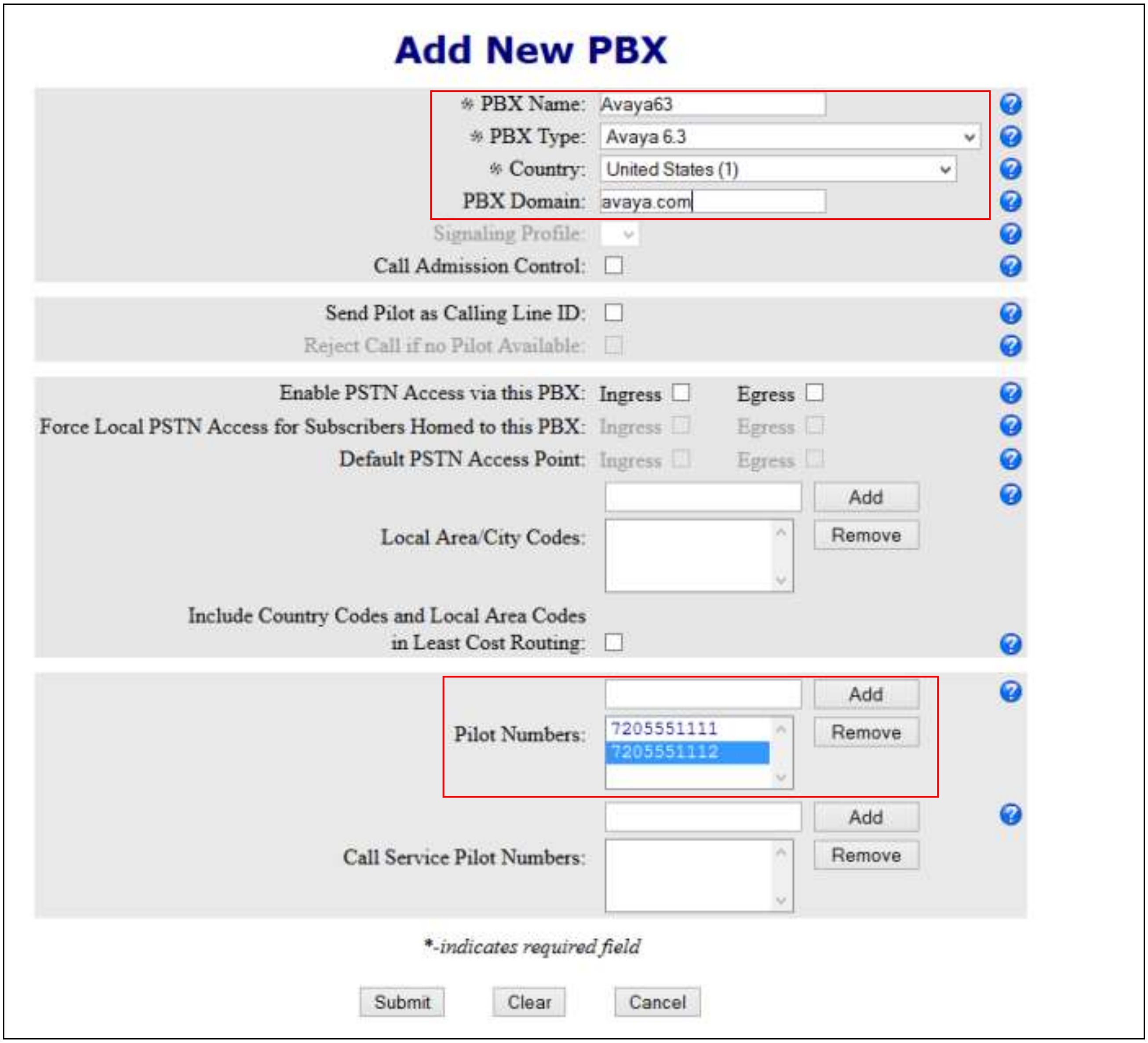
Click **Submit**.



7.1.2. Add Session Manager as PBX

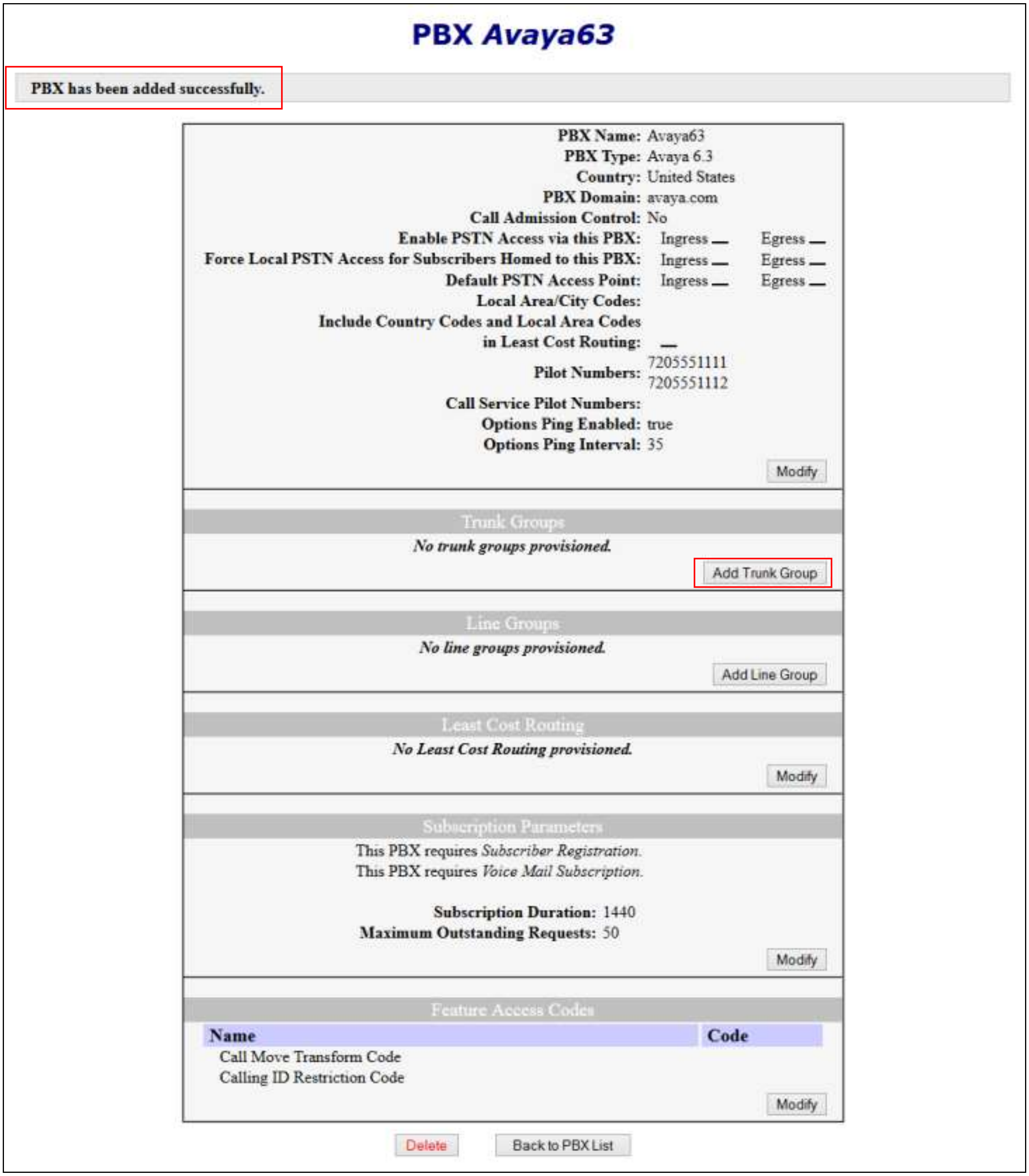
To add Session Manager to the Accelerator, select **Voice Network → PBX → Add**.

- **PBX Name** A unique name for the Session Manager.
- **PBX Type** Should be **Avaya 6.3**
- **Country** This field is used for Least Cost Routing purposes and indicates which country the PBX provides services in (this generally corresponds to where it is physically located).
- **PBX Domain** field value should match the domain defined configured on Session Manager in **Section 6.1**.
- **Pilot DN Numbers** Add the Pilot DN Numbers used in **Section 5.3.4**.



7.1.3. Add a Trunk Group/Trunk

Define a new trunk group and add trunk group members to communicate with Session Manager. To define a new trunk group, select the PBX created in **Section 7.1.2**. Select **Voice Network → PBX → List all**). Click the **Add Trunk Group** button.

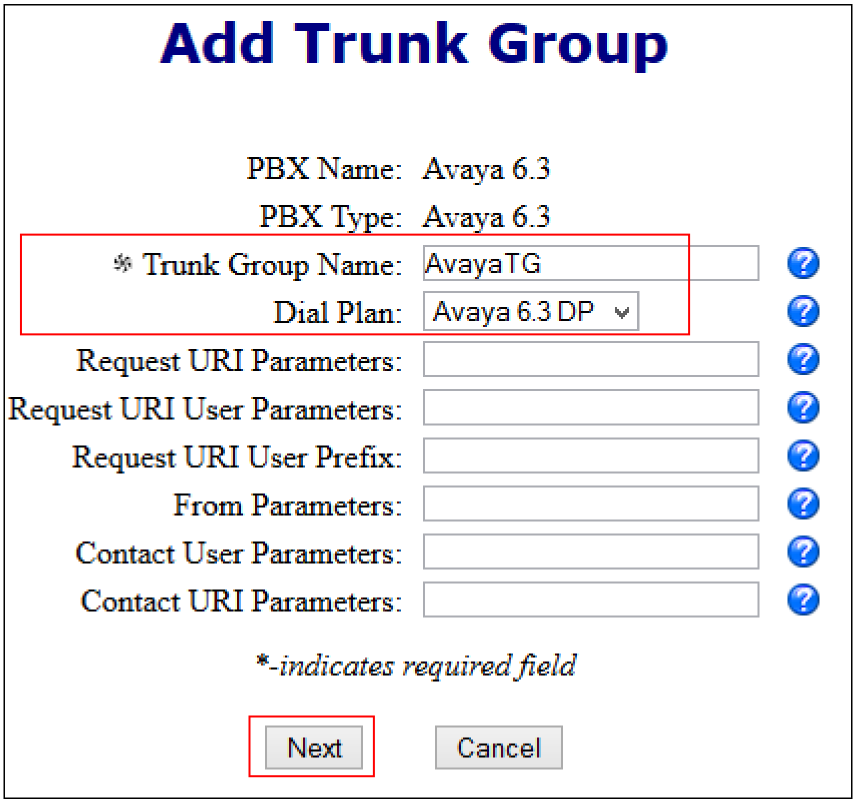


The **Add Trunk Group** screen is displayed.

- The **Trunk Group Name** field provides a name for the trunk on the Accelerator. It should be a unique identifier for this trunk.

- **Dial Plan** should be set to **Avaya 6.3 DP** which is the dial plan configured for the Avaya routing prefixes.
- **URI Parameters** are optional fields and are not required for integration with Avaya.

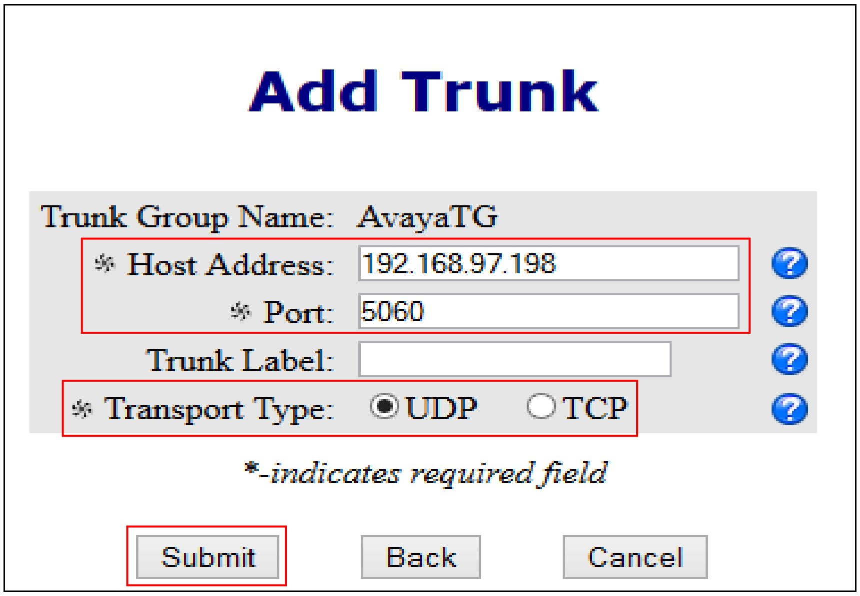
Note: *Only one trunk group can be data-filled for the Avaya PBX.*



Click **Next**. The **Add Trunk** screen is displayed.

- The **Host Address** should be the hostname or IP address of Session Manager.
- **Port** should match the value configured on Session Manager.
- **Transport Type** should be set to **UDP**.

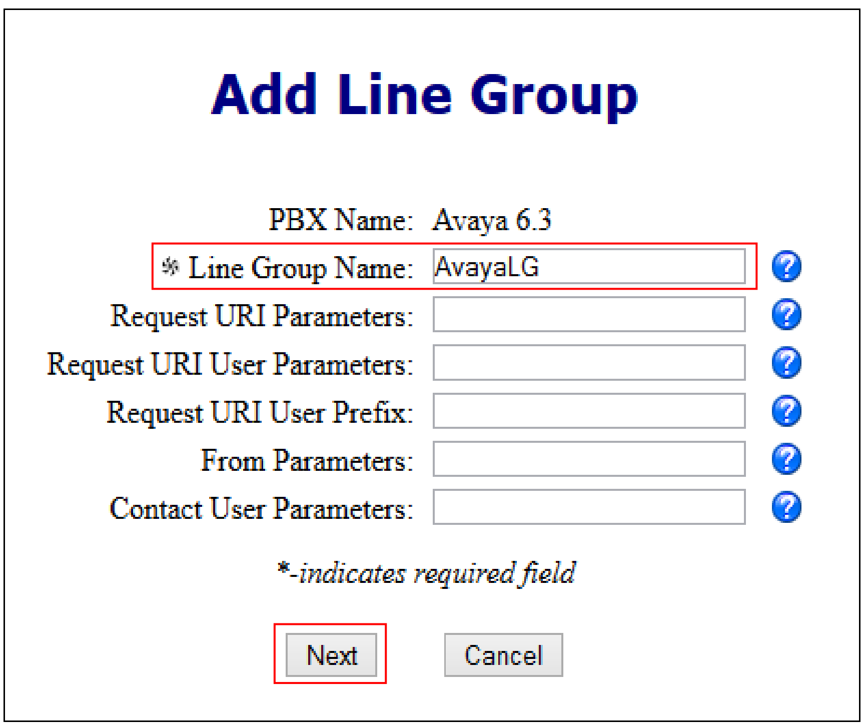
Click **Submit**.



7.1.4. Add a Line Group/Line

Select **Add Line Group** on the Selected PBX Screen to create the SIP line group to interface with Session Manager.

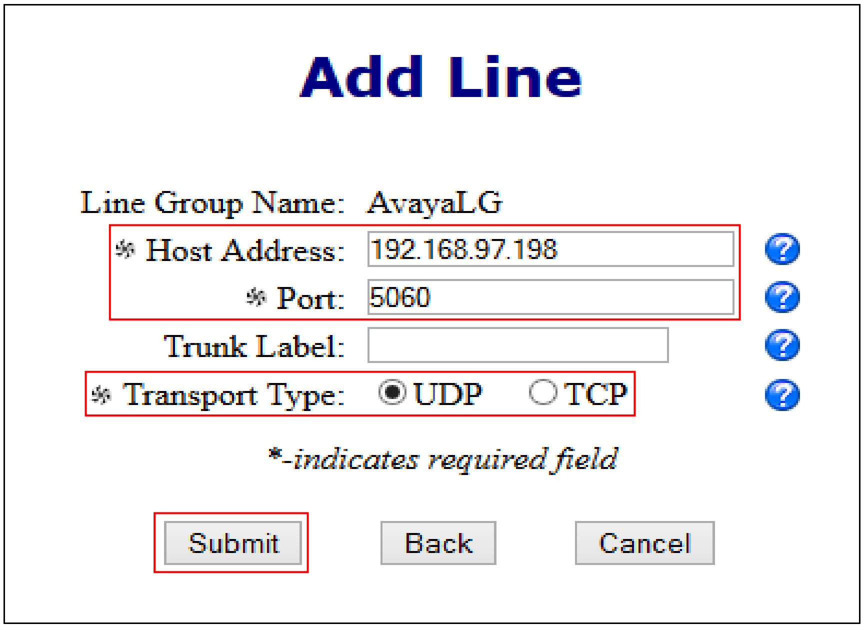
- The **Line Group Name** should be a unique identifier for this line group.
- URI Parameters are optional fields and are not required for integration.



Select **Next** to add individual lines within the group.

- The **Host Address** should be the hostname or IP address of Session Manager.
- **Port** should match the value configured on Session Manager.
- **Transport Type** should be **UDP**.

Click **Submit**.



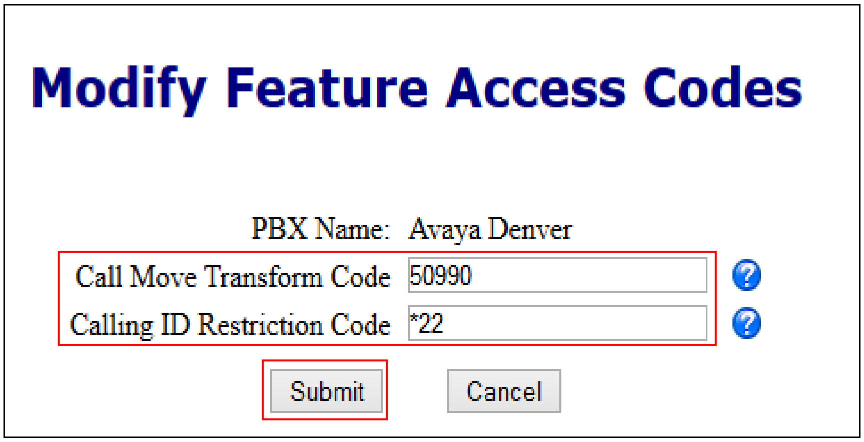
7.1.5. Feature Access Codes

Feature Access Codes can be changed by selecting **Modify** in the **Feature Access Codes** section of the PBX screen shown earlier. These values should be the same as the ones provisioned on Session Manager.

Call Move Transform Code must match the Avaya field **Active Appearance Select** configured in **Section 5.4.3**

Calling ID Restriction Code must match the **Per Call CPN Blocking Codes Access Code** configured on Communication Manager in **Section 5.3.3**.

Enter the values in the appropriate fields and click **Submit**.

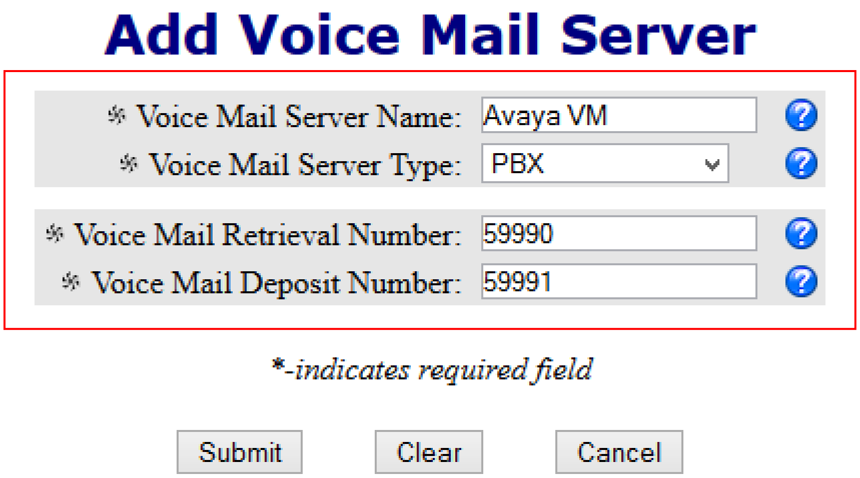


7.1.6. Add Voice Mail Server

Provision the voice mail server used with the Avaya PBX so the Accelerator can provide a single voice mail solution. To add a Voice Mail Server, select **Voice Network → Voice Mail → Add**. Select **PBX** as the **Voice Mail Server Type**.

- The **Voice Mail Server** Name should be unique.
- The **Voice Mail Server Type** should be set to **PBX**.
- The **Voice Mail Retrieval Number** should be set to 59990 which is the number that routes callers to their voicemail.
- The **Voice Mail Deposit Number** should be set to the feature code defined on Communication Manager in **Section 5.4.3.**for Transfer Call to Voice Mail.

Enter the values in the appropriate fields and click **Submit**.

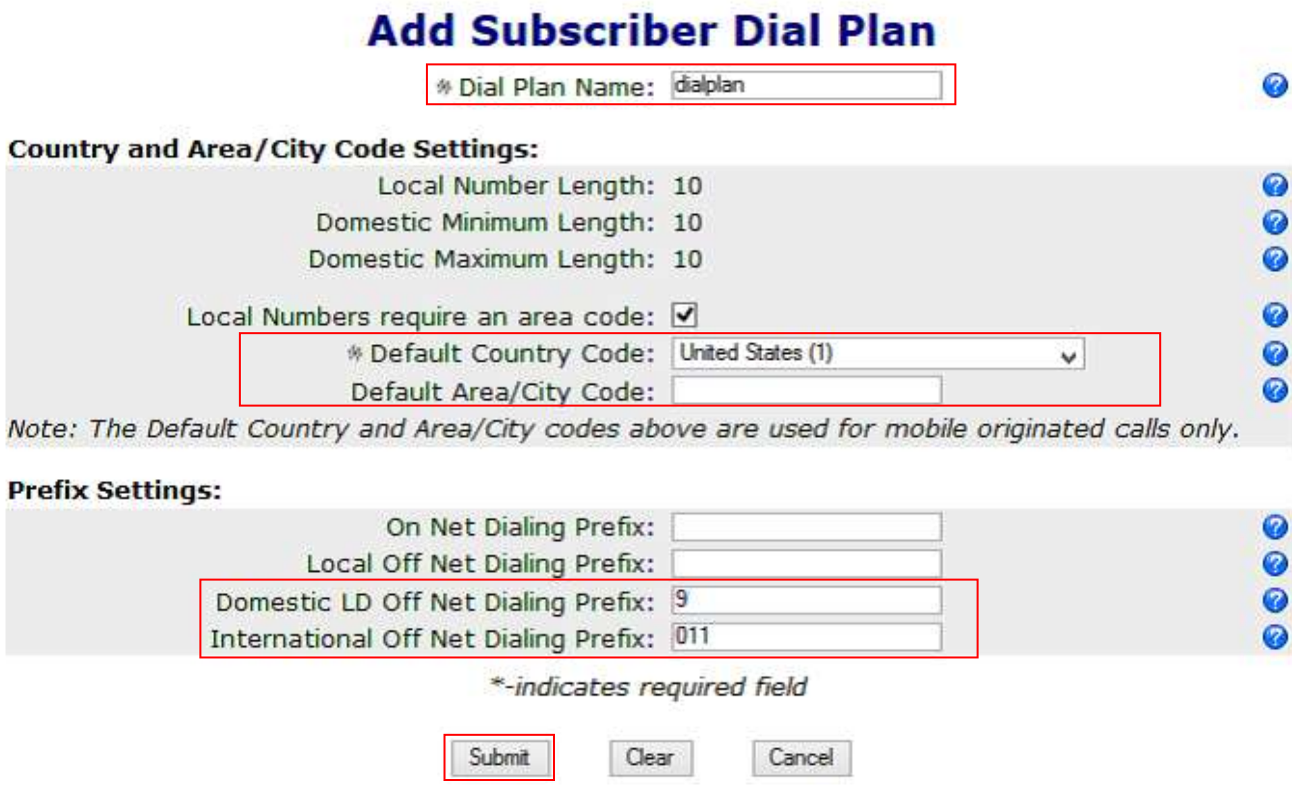


7.1.7. Add Subscriber Dial Plan

Before subscribers can be added to the Accelerator, a Subscriber Dial Plan must first be defined. **Subscriber → Subscriber Dial Plan → Add.**

- The **Dial Plan Name** should be unique.
- The **Local Number Requires an Area Code** should be checked to indicate dialing an area code is necessary for local numbers.
- **Default Country Code** of **United States(1)** was used when none is dialed by the subscriber.
- **Default Area/City Code** Area Code for the Subscribers. Maximum length is 5 digits, except in the United States and Canada where the Area Code must be 3 digits.
- **On Net Dialing Prefix** The On Net prefix that is prepended to dial strings outside the user's home PBX.
- **Domestic LD Off Net Dialing Prefix** The Off Net prefix used for routing Domestic Long Distance Calls. This should be set to the ARS Access Code configured on Communication Manger in **Section 5.3.3**.
- **International Off Net Dialing Prefix** The Off Net prefix used for routing International Long Distance Calls.

Enter the values in the appropriate fields and click Submit.



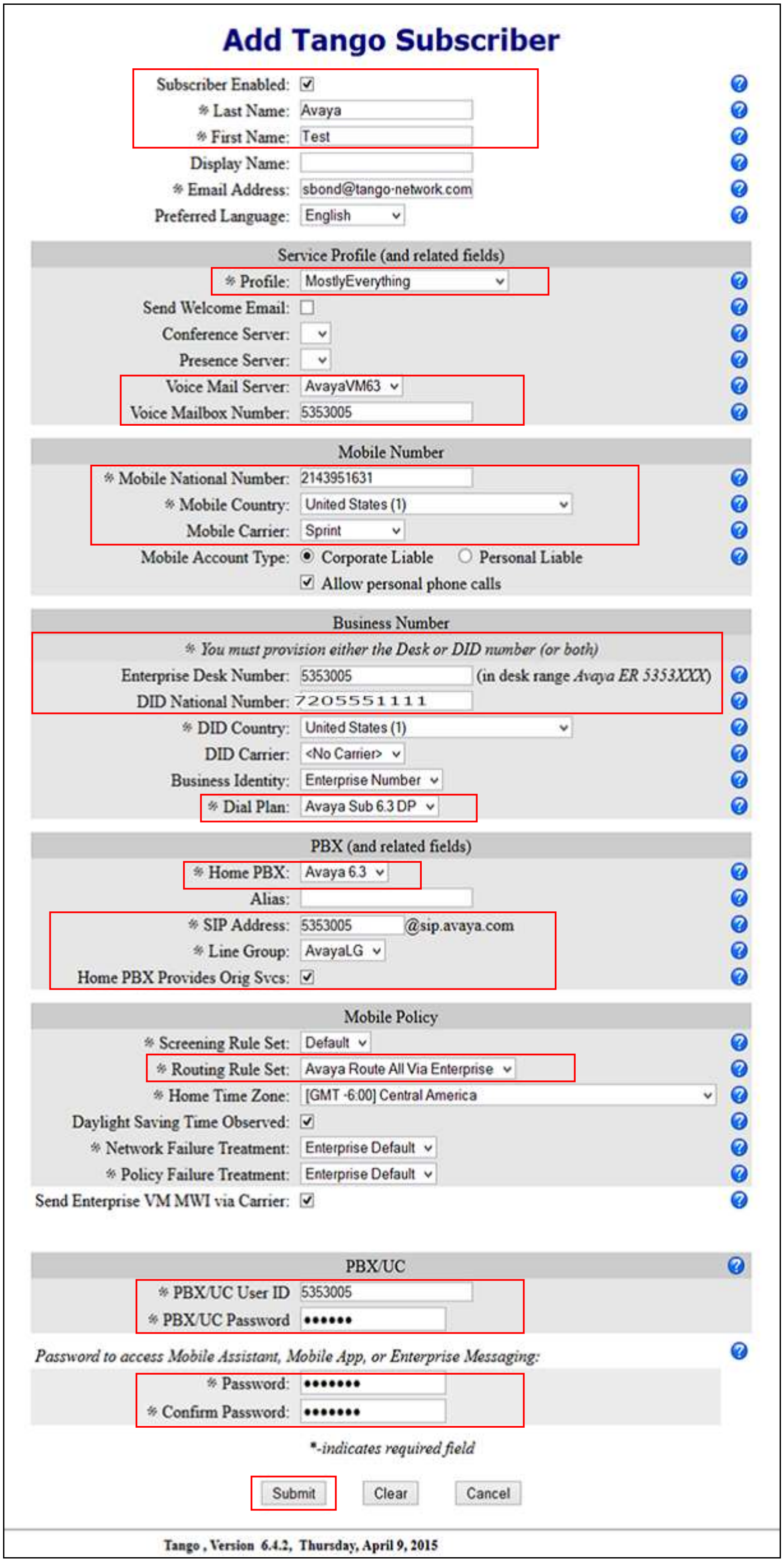
7.1.8. Add Subscriber

The following steps describe the Accelerator configuration required when the desk phone is SIP, H.323, Analog or digital. To add subscribers, select **Subscriber → Add.**

- Select the appropriate Service Profile from the drop down menu. **Mostly Everything** was used for compliance testing.
- Select the appropriate **Voice Mail Server** defined earlier on the Accelerator from the drop down and data fill the mailbox number. (**AvayaVM63** in our example.)
- Set the **Mobile National Number** to that of the provisioned mobile phone and select the appropriate **Mobile Carrier** from the drop down.
- Set the Accelerator **Enterprise Desk Number** to the extension defined for the user's station on the Session Manager (535-3005 in our example).
- Select **Avaya 6.3** as the user's **HomePBX** field.
- Select the **Dial Plan** defined earlier on the Accelerator. (**Avaya 6.3 Sub DP** in our example.)
- Set the **SIP Address** to the user's off-pbx-telephone station-mapping. The example shows 5353005@sip.avaya.com for the subscriber.
- Select the **Line Group** defined earlier on the Accelerator. (**AvayaLG** in our example.)
- Ensure the option **Home PBX Provides Orig Svcs** is checked. When checked, Accelerator originates calls for the mobile user through the home PBX.

- Set the **PBX/UC User ID** and **PBX/UC Password** to match user credentials configured on Session Manager in **Section 6.5.2**. This is what the Accelerator uses to register the line.

Add all other required fields. See the *Accelerator Provisioning Guide* for more information.



Add Tango Subscriber

Subscriber Enabled: ☑
* Last Name: Avaya
* First Name: Test
Display Name:
* Email Address: sbond@tango-network.com
Preferred Language: English

Service Profile (and related fields)

* Profile: MostlyEverything
Send Welcome Email: ☐
Conference Server:
Presence Server:
Voice Mail Server: AvayaVM63
Voice Mailbox Number: 5353005

Mobile Number

* Mobile National Number: 2143951631
* Mobile Country: United States (1)
Mobile Carrier: Sprint
Mobile Account Type: ◉ Corporate Liable ○ Personal Liable
☑ Allow personal phone calls

Business Number

* *You must provision either the Desk or DID number (or both)*
Enterprise Desk Number: 5353005 (in desk range *Avaya ER 5353XXX*)
DID National Number: 7205551111
* DID Country: United States (1)
DID Carrier: <No Carrier>
Business Identity: Enterprise Number
* Dial Plan: Avaya Sub 6.3 DP

PBX (and related fields)

* Home PBX: Avaya 6.3
Alias:
* SIP Address: 5353005 @sip.avaya.com
* Line Group: AvayaLG
Home PBX Provides Orig Svcs: ☑

Mobile Policy

* Screening Rule Set: Default
* Routing Rule Set: Avaya Route All Via Enterprise
* Home Time Zone: [GMT -6:00] Central America
Daylight Saving Time Observed: ☑
* Network Failure Treatment: Enterprise Default
* Policy Failure Treatment: Enterprise Default
Send Enterprise VM MWI via Carrier: ☑

PBX/UC

* PBX/UC User ID 5353005
* PBX/UC Password ••••••

Password to access Mobile Assistant, Mobile App, or Enterprise Messaging:
* Password: •••••••
* Confirm Password: •••••••

*-*indicates required field*

Submit Clear Cancel

Tango , Version 6.4.2, Thursday, April 9, 2015

8. Verification Steps

This section provides the verification steps that may be performed to verify the configuration.

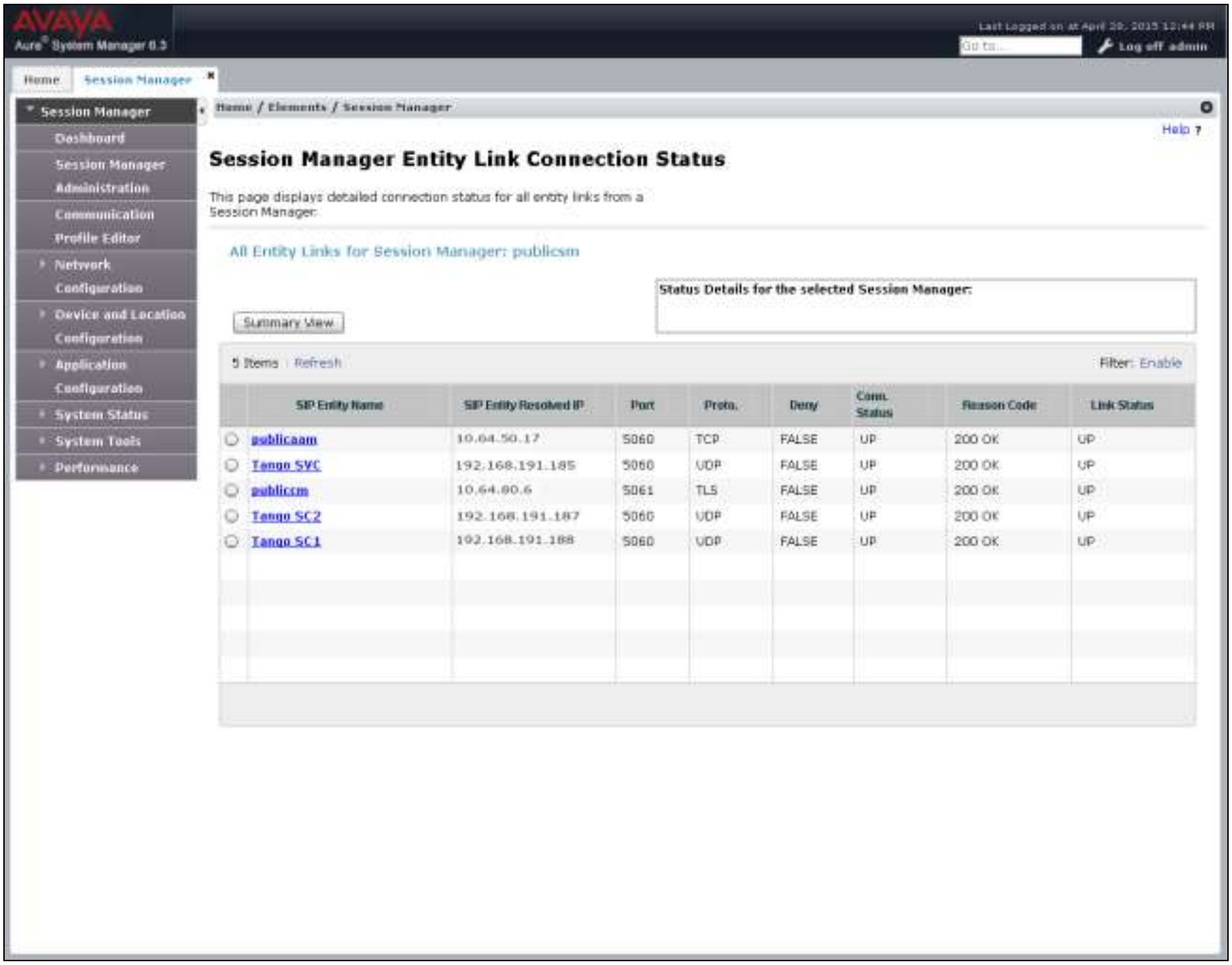
8.1. Verify Avaya Aura® Communication Manager Trunk Status

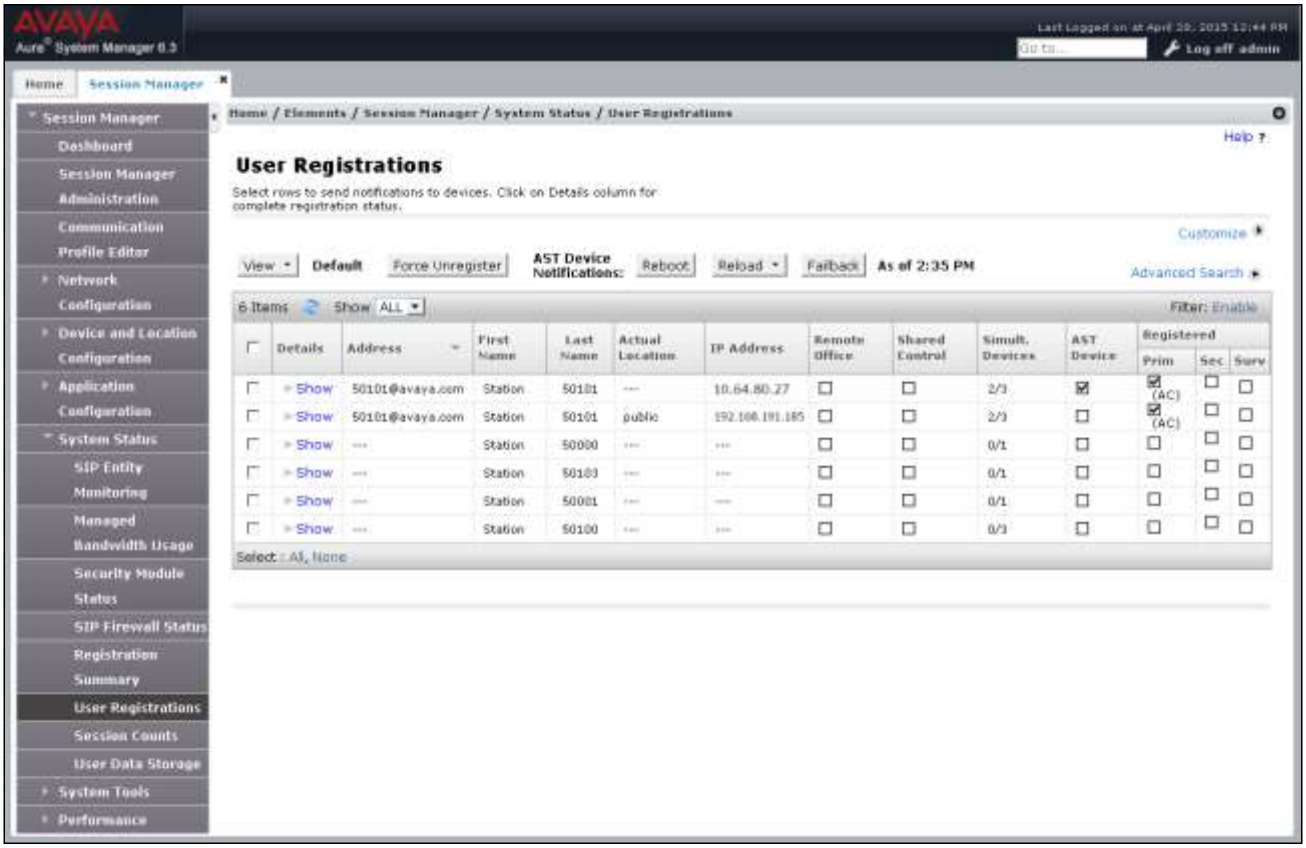
On Communication Manager, ensure that all the signaling groups are in service by issuing the command status **signaling-group n** where **n** is the signaling group number.

```
status signaling-group 2
                         STATUS
   SIGNALING GROUP

        Group ID: 2
      Group Type: sip

     Group State: in-service
```

8.2. SIP Monitoring on Avaya Aura® Session Manager

From System Manager's Home screen, navigate to Elements →**Session Manager**→**System Status**→**SIP Entity Monitoring**. Verify that none of the links to the defined SIP entities are down, indicating that they are all reachable for call routing. The screen below shows the link status between Session Manager and the Tango Networks Enterprise Accelerator.

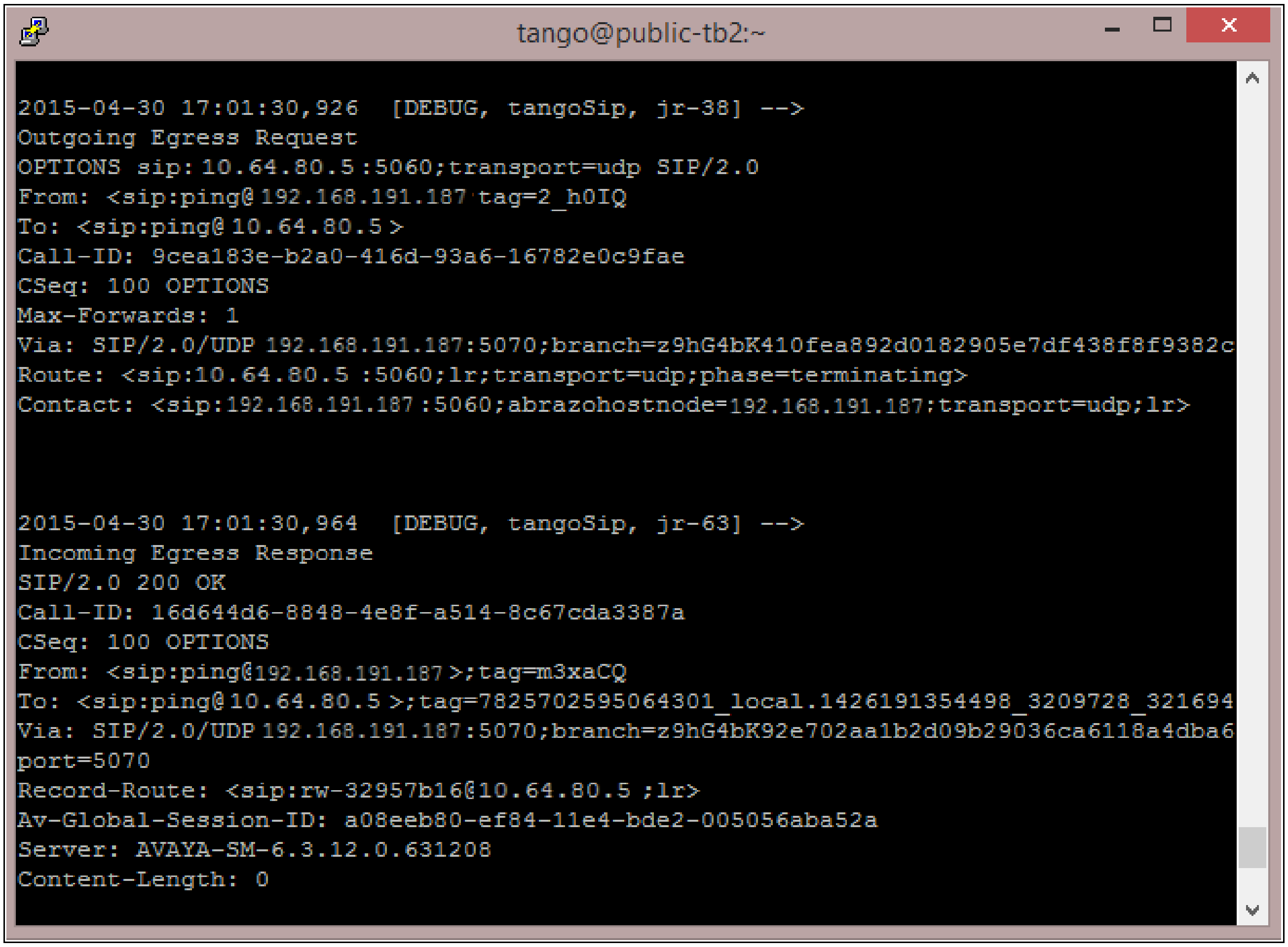



AVAYA
Aura® System Manager 6.3
Log off admin
Home
Session Manager
Home / Elements / Session Manager / System Status / User Registrations
Help ?
User Registrations
Select rows to send notifications to devices. Click on Details column for complete registration status.
Customize
View
Default
Force Unregister
AST Device Notifications:
Reboot
Reload
Failback
As of 2:35 PM
Advanced Search
6 Items
Show ALL
Filter: Enable
Details
Address
First Name
Last Name
Actual Location
IP Address
Remote Office
Shared Control
Simult. Devices
AST Device
Registered
Prim
Sec
Surv
Show
50101@avaya.com
Station
50101
10.64.80.27
2/3
(AC)
public
50000
0/1
50103
50001
50100
Select : All, None
Session Manager
Dashboard
Session Manager Administration
Communication Profile Editor
Network Configuration
Device and Location Configuration
Application Configuration
System Status
SIP Entity Monitoring
Managed Bandwidth Usage
Security Module Status
SIP Firewall Status
Registration Summary
User Registrations
Session Counts
User Data Storage
System Tools
Performance

8.3. Verifying Status on the Tango Networks Enterprise Accelerator

8.3.1. Check SIP Connection Between Tango Accelerator and Avaya Session Manager.

Launch a PuTTY session and browse to the /var/tango/sessionconductor/log directory and tail the latest debug log. Watch for an Options Ping to the Session Manager and ensure the 200 ok is returned.



```
tango@public-tb2:~

2015-04-30 17:01:30,926  [DEBUG, tangoSip, jr-38] -->
Outgoing Egress Request
OPTIONS sip:10.64.80.5:5060;transport=udp SIP/2.0
From: <sip:ping@192.168.191.187 tag=2_h0IQ
To: <sip:ping@10.64.80.5>
Call-ID: 9cea183e-b2a0-416d-93a6-16782e0c9fae
CSeq: 100 OPTIONS
Max-Forwards: 1
Via: SIP/2.0/UDP 192.168.191.187:5070;branch=z9hG4bK410fea892d0182905e7df438f8f9382c
Route: <sip:10.64.80.5 :5060;lr;transport=udp;phase=terminating>
Contact: <sip:192.168.191.187:5060;abrazohostnode=192.168.191.187;transport=udp;lr>



2015-04-30 17:01:30,964  [DEBUG, tangoSip, jr-63] -->
Incoming Egress Response
SIP/2.0 200 OK
Call-ID: 16d644d6-8848-4e8f-a514-8c67cda3387a
CSeq: 100 OPTIONS
From: <sip:ping@192.168.191.187>;tag=m3xaCQ
To: <sip:ping@10.64.80.5>;tag=7825702595064301_local.1426191354498_3209728_321694
Via: SIP/2.0/UDP 192.168.191.187:5070;branch=z9hG4bK92e702aa1b2d09b29036ca6118a4dba6
port=5070
Record-Route: <sip:rw-32957b16@10.64.80.5 ;lr>
Av-Global-Session-ID: a08eeb80-ef84-11e4-bde2-005056aba52a
Server: AVAYA-SM-6.3.12.0.631208
Content-Length: 0
```

8.3.2. Check Line Registration

From the browser-based GUI of the Tango Accelerator, go to **Subscriber → List All →** Select the subscriber and then click on the **Status** tab. Under **PBX Status** the Registration Status should be **Active** and if the subscriber is provisioned for voice mail, the Voice Mail Subscription Status should be **Active.**

9. Conclusion

These Application Notes describe the configuration steps required for integrating the Tango Networks Enterprise Accelerator Solution into an Avaya telephony infrastructure. For the configuration described in these Application Notes, the Tango Networks Enterprise Accelerator Solution was responsible for bridging landline connectivity to Avaya Aura® Communication Manager with the wireless connectivity to the mobile network. The functionality of the Avaya/ Tango Networks Enterprise Accelerator Solution was validated via the DevConnect Program at the Avaya Solution and Interoperability Test Lab. All feature functionality test cases passed.

10. Additional References

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

[1] *Administering Avaya Aura® Communication Manager*, Document 03-300509
[2] *Administering Avaya Aura® Session Manager*, Document 03-603324

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

Product documentation for Tango Networks products may be found at http://www.tango-networks.com/faqs/.

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.