Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Tango Networks™ Abrazo™ Solution with an Avaya Aura™ Telephony Infrastructure - Issue 1.0

Abstract

These Application Notes describe a compliance-tested configuration comprised of the Tango Networks™ Abrazo™ Solution connected to an Avaya Aura™ Telephony Infrastructure.

The Tango Abrazo solution extends enterprise PBX functionality to mobile devices, allowing end users to be accessible when out of the office. The Tango 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 compliance testing and additional technical discussions.

1. Introduction

These Application Notes describe a compliance-tested configuration comprised of the Tango Networks Abrazo Solution connected to an Avaya Aura™ Telephony Infrastructure, consisting of Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services.

Tango Networks' Abrazo 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.

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.

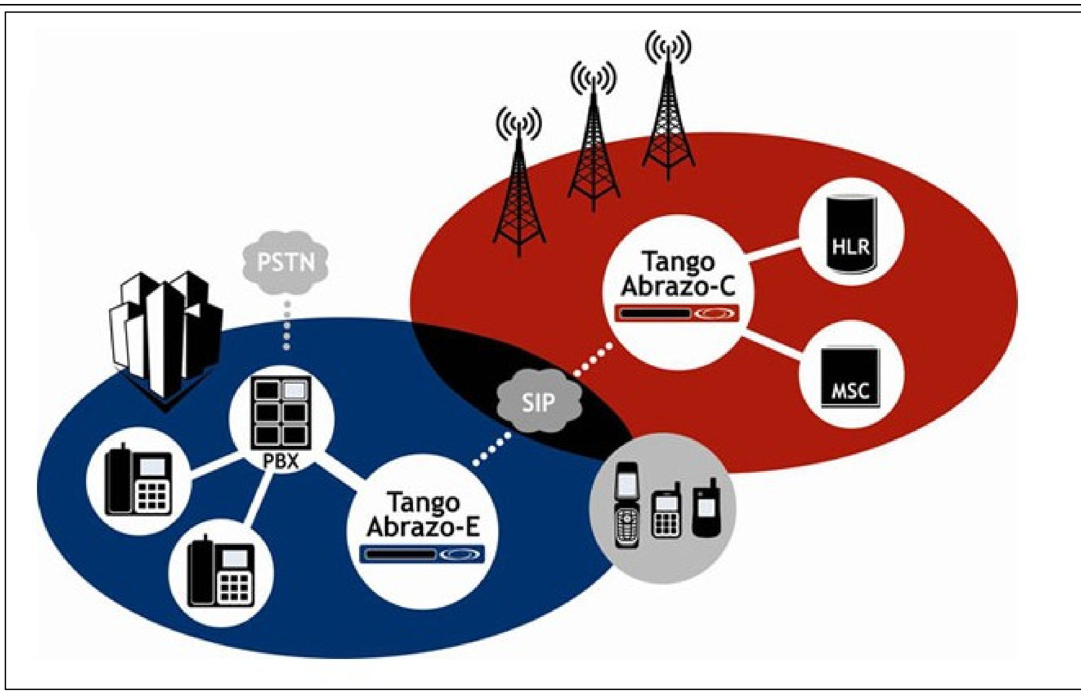


Figure 1: Tango Networks' Architecture Diagram

The Tango Networks™ Abrazo™ Solution uses a combination of SIP lines and trunks to integrate with Avaya Aura™ Communication Manager. 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 solution must terminate a call via the Public Switched Telephone Network (PSTN).

1.1.1. Mobile Originations

The Tango Networks™ Abrazo™ 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 Abrazo 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 all calls to the Pilot DN to the Abrazo solution.

When the Abrazo 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.

1.1.2. Mobile Terminations

To receive calls made to a subscriber the Avaya is configured using the Off-PBX Station Mapping to alert the Abrazo solution simultaneously whenever the Avaya alerts other client devices, such as the subscriber's desk phone. The Abrazo, 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 addressed to the retrieved number.

1.2. Interoperability Compliance Testing

Testing was conducted via the DevConnect Program. Compliance testing verified the integration between Avaya Aura™ Telephony Infrastructure and Tango Networks Abrazo Solution and the ability for an enterprise user to be accessible via one business number whether the user is in the office or mobile.

1.3. Support

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

- Web site: http://www.tango-networks.com
- Email: sales@tango-networks.com
- Telephone: +1 469-229-6024

2. Reference Configuration

These Application Notes describe a solution for integrating the Tango Abrazo-E 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 Abrazo and Avaya products. The solution described herein is also extensible to other Avaya Servers and Media Gateways.

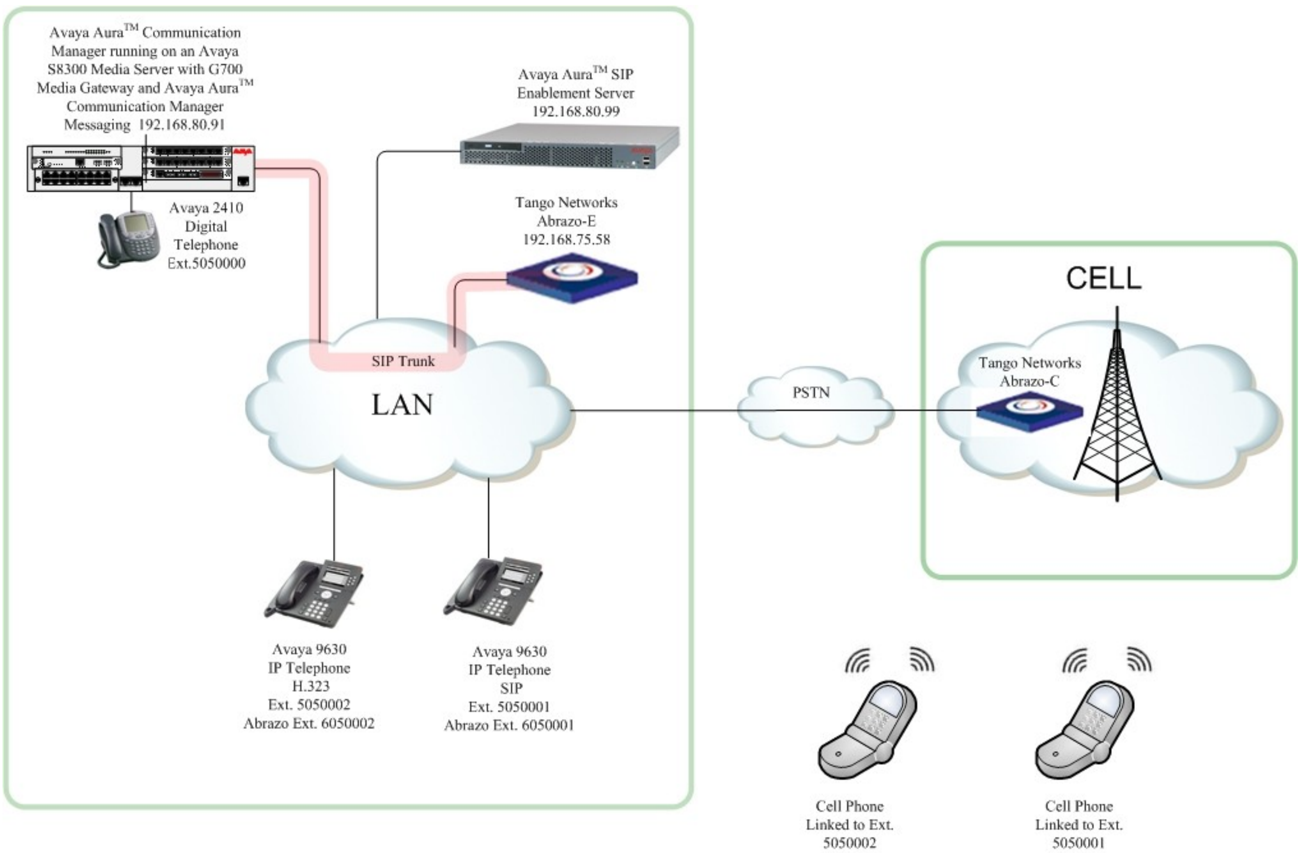


Figure 2: Interoperability Configuration Diagram

3. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya PBX Products	
Avaya S8300 Server running Avaya Aura™ Communication Manager	Avaya Aura™ Communication Manager 5.2.1
Avaya G700 Media Gateway (Corporate Site) MGP MM712 DCP Media Module	 28.22.0 HW9
Avaya Aura™ SIP Enablement Services (SES)	
Avaya S8300 Server running Avaya Aura™ SIP Enablement Services	Avaya Aura™ SIP Enablement Services 5.2.1
Avaya Messaging (Voice Mail) Products	
Avaya S8300 Server running Avaya Aura™ Communication Manager Messaging (CMM)	Avaya Aura™ Communication Manager Messaging (CMM) 5.2.1-13.0
Avaya Telephony Sets	
Avaya 9600 Series IP Telephones	Avaya one-X® Deskphone Edition 3.1.1
Avaya 9600 Series IP Telephones	Avaya one-X® Deskphone SIP 2.6
Avaya 4600 Series IP Telephones	SIP (2.2) & H.323 (2.9)
Avaya 2410 Digital Telephone	5.0
Tango Abrazo Products	
Tango Networks Abrazo-Enterprise Release	4.4
Tango Networks Abrazo-Carrier Release	4.4
Mobile Devices	
Various Mobile Cell phones used for testing	N/A

4. Avaya Aura™ Communication Manager

Basic configuration of Communication Manager and SIP Enablement Services are beyond the scope of these Application Notes. See **Section 8** for Avaya documentation references. It is assumed that the reader has a basic understanding of the administration of Communication Manager and has access to the System Access Terminal (SAT).

This section describes the steps required to configure Communication Manager to support the configuration in **Figure 2: Interoperability Configuration Diagram**. The following pages provide step-by-step instructions on how to administer the required configuration parameters. The steps are performed from the SAT interface.

4.1. System Parameters Customer Options

The steps in this section verify that there are a sufficient number of SIP trunks and stations between Communication Manager and SES. 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.

4.1.1. 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  10
                                OPTIONAL FEATURES

     G3 Version: V13
       Location: 1                            RFA System ID (SID): 1
       Platform: 13                           RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 900   80
                                      Maximum Stations: 450   29
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 100   0
                    Maximum Off-PBX Telephones -   OPS: 100   21
                    Maximum Off-PBX Telephones - SCCAN: 0     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  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 450   50
           Maximum Concurrently Registered IP Stations: 450   4
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 40    0
                        Maximum Video Capable Stations: 40    0
                   Maximum Video Capable IP Softphones: 40    0
                       Maximum Administered SIP Trunks: 100   20
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 30    0
                             Maximum TN2501 VAL Boards: 0     0
                     Maximum Media Gateway VAL Sources: 50    0
           Maximum TN2602 Boards with 80 VoIP Channels: 0     0
          Maximum TN2602 Boards with 320 VoIP Channels: 0     0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

4.2. Provision Avaya Aura™ Communication Manager to interoperate with Tango Abrazo

4.2.1. Add IP Node Names

This section describes the steps for setting the IP node name for the Tango Abrazo-E with Communication Manager. Enter the **change node-names ip** command. On **Page 1** of the **change node-names ip** form, enter the **Name** and the **IP Address** for the Abrazo. Enter the following:

- **Name - Abrazo**,
- **IP Address - 192.168.75.58**

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
AvayaSES            192.168.80.99
default             0.0.0.0
Audix               192.168/80.94
procr               192.168.80.91
Abrazo             192.168.75.58
```

4.2.2. Add IP Codec Set

This section describes the steps for administering the codec set in Communication Manager. Enter the **change ip-codec-set g** command, where **g** is a number between 1 and 7, inclusive, and enter **G.711MU** for **Audio Codec**. This IP codec set will be selected later in the IP Network Region form to define which codecs may be used within an IP network region.

```
change ip-codec-set 1                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n          2        20
 2:
```

4.2.3. Add IP Network Region

This section describes the steps for administering the IP Network Region in Communication Manager. Enter the **change ip-network-region h** command, where **h** is a number between 1 and 250, inclusive. On **Page 1** of the **ip-network-region** form. Configure the following:

- **Codec Set** – Set to the number of the IP codec set configured in **Section 4.2.2**
- **Region** – **67** was used for compliance testing
- **Authoritative Domain** – **mpc.com** was used for compliance testing
- **Name** – **TangoAbrazo** was used for compliance testing

```
change ip-network-region 67                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 67
Location: 1       Authoritative Domain: mpc.com
    Name: TangoAbrazo
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

4.3. Create and configure the Trunk and Signaling Group between Avaya Aura™ Communication Manager and the Tango Abrazo-E

This section describes the steps for administering the trunk group and signaling group between Communication Manager and the Tango Abrazo.

4.3.1. Add signaling group

Enter the **add signaling group j** command, where **j** is an available signaling group number. On **Page 1** of the **signaling-group** form, configure the following:

- **Group Type** – set to **sip**
- **Transport Method –** set to **tcp**
- **Near-end Node Name** – enter the node name of **procr**
- **Near-end Listen Port** – specify the local listen port, typically **5060**
- **Far-end Node Name** – enter the node name of the Abrazo configured in **Section 4.2.1**
- **Far-end Listen Port** – specify the local listen port, typically **5060**
- **Far-end Domain – mpc.com**
- **Far-end Network Region** – enter the IP network region configured in **Section 4.2.3**
- **DTMF over IP** – set to **rtp-payload**"
- **Direct IP-IP Audio Connections** – set to **y**

```
add signaling-group 67                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 67              Group Type: sip
                        Transport Method: tcp

          IP Video? n




   Near-end Node Name: procr                  Far-end Node Name: Abrazo
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                         Far-end Network Region: 1
       Far-end Domain: mpc.com

                                              Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
                                                         IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 120
```

4.3.2. Add trunk-group

Enter the **add trunk-group i** command, where "i" is an available trunk group number. On **Page 1** of the **trunk-group** form, configure the following:

- **Group Type** – set to **sip**
- **Group Name** – enter a meaningful name/description
- **TAC** – enter a Trunk Access Code that is valid under the provisioned dial plan
- **Service Type** – set to **tie**
- **Signaling Group** – enter the Signaling Group number that was created in **step 4.3.1**
- **Number of Members** – set to **150**

```
add trunk-group 67                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: T0 Tango                  COR: 1        TN: 1        TAC: *001
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                    Signaling Group: 67
                                                  Number of Members: 150
```

Go to **Page 4**, enter the following:

- **Telephone Event Payload – 127** (based on the customer's network)

```
add trunk-group 67                                            Page   4 of  21
                              PROTOCOL VARIATIONS

                       Mark Users as Phone? n
             Prepend '+' to Calling Number? n
      Send Transferring Party Information? n

                     Send Diversion Header? n
                   Support Request History? y
              Telephone Event Payload Type: 127
```

4.4. Provision Class Of Service

This section describes the steps for administering the IP Network Region in Communication Manager. Enter the **change cos** command on **Page 1** of the form, configure the following:

- **Auto Callback** set to **y**
- **Call Fwd-All Calls** set to **y**
- **Data Privacy** set to **y**
- **Priority Calling** set to **y**
- **Restrict Call Fwd-Off Net** set to **n**
- **Call Forwarding Busy/DA** set to **y**
- **Trk-to-Trk Transfer Override** set to **y**

```
change cos                                                     Page   1 of   2
CLASS OF SERVICE

                                0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
 Auto Callback                  n  y  y  n  y  n  y  n  y  n  y  n  y  n  y  n
 Call Fwd-All Calls             n  y  n  y  y  n  n  y  y  n  n  y  y  n  n  y
 Data Privacy                   n  y  n  n  n  y  y  y  y  n  n  n  n  y  y  y
 Priority Calling               n  y  n  n  n  n  n  n  n  y  y  y  y  y  y  y
 Console Permissions            n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Off-hook Alert                 n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Client Room                    n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Restrict Call Fwd-Off Net      y  n  y  y  y  y  y  y  y  y  y  y  y  y  y  y
 Call Forwarding Busy/DA        n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Personal Station Access (PSA)  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding All        n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding B/DA       n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Trk-to-Trk Transfer Override   n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 QSIG Call Offer Originations   n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Contact Closure Activation     n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

4.5. Configure the Dial Plan, AAR 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 Tango Abrazo-E. If the connectivity between the wireless carrier and the enterprise is via the PSTN, then dial plan and route patterns must be configured on the Communication Manager for pilot directory numbers. In addition, the Abrazo-E can query into the wireless network to obtain a routing number (TLDN) for least cost routing functionality. When this occurs, the Communication Manager must have outgoing routes defined for these TLDNs via the PSTN network.

4.5.1. Configure Outbound Routing

In these Application Notes, the Automatic Route Selection (ARS) feature is used to route outbound calls via the SIP trunk to the Tango Abrazo. 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.

4.5.2. Change dialplan analysis

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

```
change dialplan analysis                                       Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all         Percent Full:    3

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     0            3    fac
     1            3    fac
     2            5    ext
     3            1    fac
     4            5    aar
     5            5    ext
     6            5    ext
     7            5    aar
     8            1    fac
     9            1    fac
     *            2    fac
```

4.5.3. Change feature-access-codes

Use the **change feature-access-codes** command to configure **9** as the **Auto Route Selection (ARS) – Access Code 1**.

```
change feature-access-codes                                        Page   1 of   9
                              FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: *600
         Abbreviated Dialing List2 Access Code: *601
         Abbreviated Dialing List3 Access Code: *602
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: *604
                       Answer Back Access Code: *650
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 3
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: *605   Deactivation: *606
Call Forwarding Activation Busy/DA: *607   All: *608   Deactivation: *609
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code: *652
                       Call Pickup Access Code: #6
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:
```

4.5.4. Change inc-call-handling-trmt

Change inc-call-handling-trmt, this will insert the FAC in front of the dialed number so the call will be routed to the Abrazo-E, for Compliance testing, trunk group 80 was used as the PRI trunk. Use the command **change inc-call-handling-trmt trunk-group 80**. **2125550000** is used as an Example Pilot number. Enter the following information:

- **Called Len** should be set to 10 (the length of the pilot DN number)
- **Called Number** should be set to the Pilot DN number configured on the Abrazo-E
- **Insert** should be set to the FAC configured for this trunk
- **Per Call CPN/BN** should be set to none

```
change inc-call-handling-trmt trunk-group 80                       Page   1 of   3

                       INCOMING CALL HANDLING TREATMENT
 Service/      Number   Number      Del Insert            Per Call Night
 Feature        Len      Digits                           CPN/BN   Serv
 tie             10   2125550000         9                none
 tie
```

4.5.5. Add route pattern

A route pattern must be created so calls to the pilot DN are routed to the Abrazo-E. Any number not currently in use can be used for the route pattern, for compliance testing **3** was used. Use the command **change route-pattern 3** and configuring the following attributes;

- **Grp No** should be set to the value for the SIP trunk between the Communication Manager and Abrazo-E. In our example **67** 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 3                                          Page   1 of   3
                    Pattern Number: 3   Pattern Name:
                             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: 67   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
```

4.5.6. Edit the ARS table

Edit the ARS table to include the translations to the route pattern, which will route the call to the Abrazo-E. Issue the command **change ars analysis.** In our example, executed **change ars analysis 212** and enter the following:

- **Dialed String** should be set to the pilot DN previously configured
- **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 212                                         Page   1 of   2
                             ARS DIGIT ANALYSIS TABLE
                                   Location:  all           Percent Full:    3

          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    212555000             10   10     3        hnpa         n
                                                            n
```

4.6. Off-PBX Station Mapping

Every Abrazo user must have an off-PBX station in order to enable simultaneous ringing to the Abrazo-E. 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 Abrazo-E. One entry is needed for H.323 extensions, two entries are needed for SIP extensions, one for the primary number and one for the mobile endpoint.

4.6.1. H.323 Phone Off-PBX Station Mapping

In the example below, the H.323 station extension is 505-0002 and will need an off-PBX station entry to enable simultaneous ringing to the endpoint off of the Abrazo-E. Use the **change off-pbx-telephone station-mapping 5050002** command to configure the station.

- Set **Application** to CSP
- Set **Phone** Number to the number Abrazo-E will use for call originations and terminations, which is the user portion of the SIP address defined for the subscriber on the Abrazo-E. This can be any unique number assigned to the user and should not be the same as their station number. Extension **6050002** was used for compliance testing
- Set **Trunk Selection** to the number of the SIP trunk group connected to the Abrazo-E
- Set **Configuration Set** to the set to be used for IP phone call treatments

```
change off-pbx-telephone station-mapping 5050002                 Page   1 of  3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                  Prefix                     Selection   Set     Mode
 5050002          CSP           -      6050002          67         1
                                -
```

Go to **Page 2** and change the following:

- Set the **Call Limit** to match the number of **call appearance** button assignments on the stations. This can be found on tab 4 of the station form. Count the number of button appearances set to **call-appr**, and that is the number that should be used in the field. In order to support Abrazo call move services – this number may need to be incremented by at least one. The example is set to **4**
- Set **Mapping Mode** to both
- Set **Calls Allowed** to all
- Set **Bridged Calls** appropriately. See the Avaya documentation for more details on this field. The example is set to **both**

```
change off-pbx-telephone station-mapping 51000                  Page   2 of  3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Appl    Call        Mapping    Calls        Bridged      Location
 Extension      Name    Limit       Mode       Allowed      Calls
 5050002        CSP      4        both         all          both
```

4.6.2. SIP Phone Off-PBX Station Mapping

In the example below, the SIP station extension is **5050001** and will need an additional off-PBX station entry added to enable simultaneous ringing to the endpoint off of the Abrazo-E. Use the **change off-pbx-telephone station-mapping 5050001** command to configure the station.

- Add duplicate **Station Extension** to the station extension field

Set **Application** to **CSP**

- Set **Phone Number** to the number Abrazo-E will use for call originations and terminations, which is the user portion of the SIP address defined for the subscriber on the Abrazo-E. This can be any unique number assigned to the user and should not be the same as their station number. Extension **6050001** was used for compliance testing
- Set **Trunk Selection** to the number of the SIP trunk group connected to the Abrazo-E
- Set **Configuration Set** to the set to be used for IP phone call treatments

```
change off-pbx-telephone station-mapping 5050001               Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number     Trunk        Config  Dual
 Extension                 Prefix                      Selection    Set     Mode
 5050001          OPS          -      5050001           2           1
 5050001          CSP          -      6050001           67          1
```

Go to **Page 2** and change the following:

- Set the **Call Limit** to match the number of **call appearance** button assignments on the stations. This can be found on tab 4 of the station form. Count the number of button appearances set to **call-appr**, and that is the number that should be used in the field. In order to support Abrazo call move services – this number may need to be incremented by at least one. The example is set to **4**
- Set **Mapping Mode** to **both**
- Set **Calls Allowed** to **all**
- Set **Bridged Calls** appropriately. See the Avaya documentation for more details on this field. The example is set to **both**.

```
change off-pbx-telephone station-mapping 51000               Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 5050001       OPS      4        both       all         both
 5050001       CSP      4        both       all         both
```

4.6.3. Change off-pbx-telephone configuration-set

Use the **change off-pbx-telephone configuration-set h** command, where **h** is a number between 1 and 99. **1** was used for compliance testing. Change the following:

- **Configuration Set Description: SIP PHONE**
- **Calling Number Style: pbx**
- **Fast Connect on Origination? n**

```
change off-pbx-telephone configuration-set 1                    Page   1 of   1

                                     CONFIGURATION SET: 1

                         Configuration Set Description: SIP PHONE
                                  Calling Number Style: pbx
                                    CDR for Origination: phone-number
                     CDR for Calls to EC500 Destination? y
                           Fast Connect on Origination? n
                          Post Connect Dialing Options: dtmf
                         Cellular Voice Mail Detection: timed  (seconds): 4
                                        Barge-in Tone? n
                           Calling Number Verification? y
              Call Appearance Selection for Origination: primary-first
                                      Confirmed Answer? n

 Use Shared Voice Connections for Second Call Answered? n
Use Shared Voice Connections for Second Call Initiated? n
```

4.7. Change off-pbx-telephone feature-name-extensions set

Use the **change off-pbx-telephone feature-name-extensions set 1.** Change the following:

- **Transfer to Voice Mail: 7077** (7077 was used for compliance testing)

```
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): 7064
      Extended Group Call Pickup:
          Held Appearance Select: 7066
          Idle Appearance Select: 7067
              Last Number Dialed:
            Malicious Call Trace:
     Malicious Call Trace Cancel:
             Off-Pbx Call Enable: 7071
            Off-Pbx Call Disable: 7072
                   Priority Call:
                          Recall:
                  Send All Calls:
           Send All Calls Cancel: 7080
               Transfer Complete:
             Transfer On Hang-Up: 7076
          Transfer to Voice Mail: 7077
         Whisper Page Activation:
```

5. Provision the Tango Networks Abrazo-E

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

- Enterprise information
- Wireless carrier information

The integration process includes the following steps:

- Define a Trunk Dial Plan for the Communication Manager in the Abrazo system.
- Define the Communication Manager in the Abrazo system.
- Define a SIP Trunk Group/Trunk to route traffic from the Communication Manager to the Abrazo system.
- Define a SIP Line Group/Line to route traffic from the Abrazo system to the Communication Manager.
- Define Pilot Numbers (optional) - PSTN routable number(s) that represent the mobile route into the enterprise network. The enterprise network must be provisioned to route calls for each pilot number to the Abrazo-E.
- Define Least Cost Route information for SIP trunk traffic from the Abrazo system to the Avaya PBX. (Optional)
- Define the Feature Access Codes in the Abrazo system.
- Define the Voice Mail system used with the Avaya PBX.
- Define Subscriber Dial Plans in the Abrazo system.
- Define Abrazo subscribers that use the Avaya PBX.

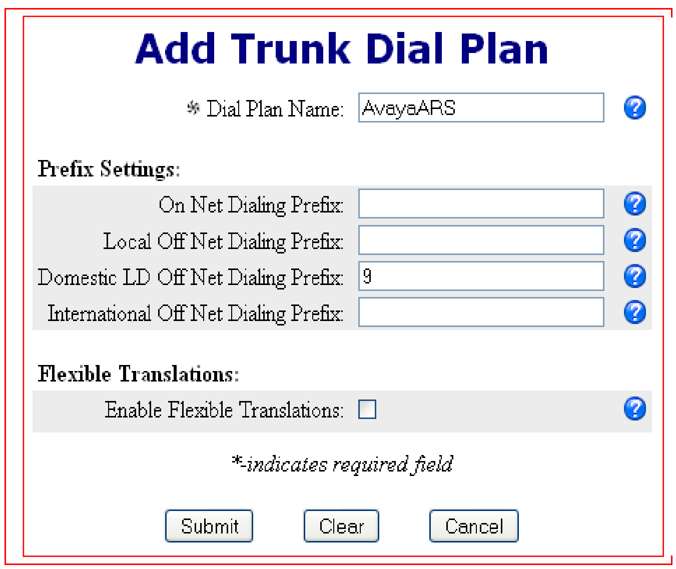
The steps below describe the unique configuration areas needed to integrate the Communication Manager with the Abrazo solution. Refer to the *Abrazo-E Provisioning Guide* for a comprehensive explanation of Abrazo provisioning.

5.1. Voice Network: PBX

5.1.1. Create a Dial Plan

Create a Dial Plan to support routing prefixes as defined in the ARS table on the Communication Manager. Select **Voice Network → PBX → Trunk Dial Plan → Add.** Change the following:

- The Dial Plan Name should be unique.
- On Net Dialing Prefix should not be set.
- Local Off Net Dialing Prefix should not be set.
- Domestic LD Off Net Dialing Prefix set to the fac routing prefix defined in the Avaya ARS table.
- International Off Net Dialing Prefix should not be set.
- Enable Flexible Translations should not be checked for this dial plan.

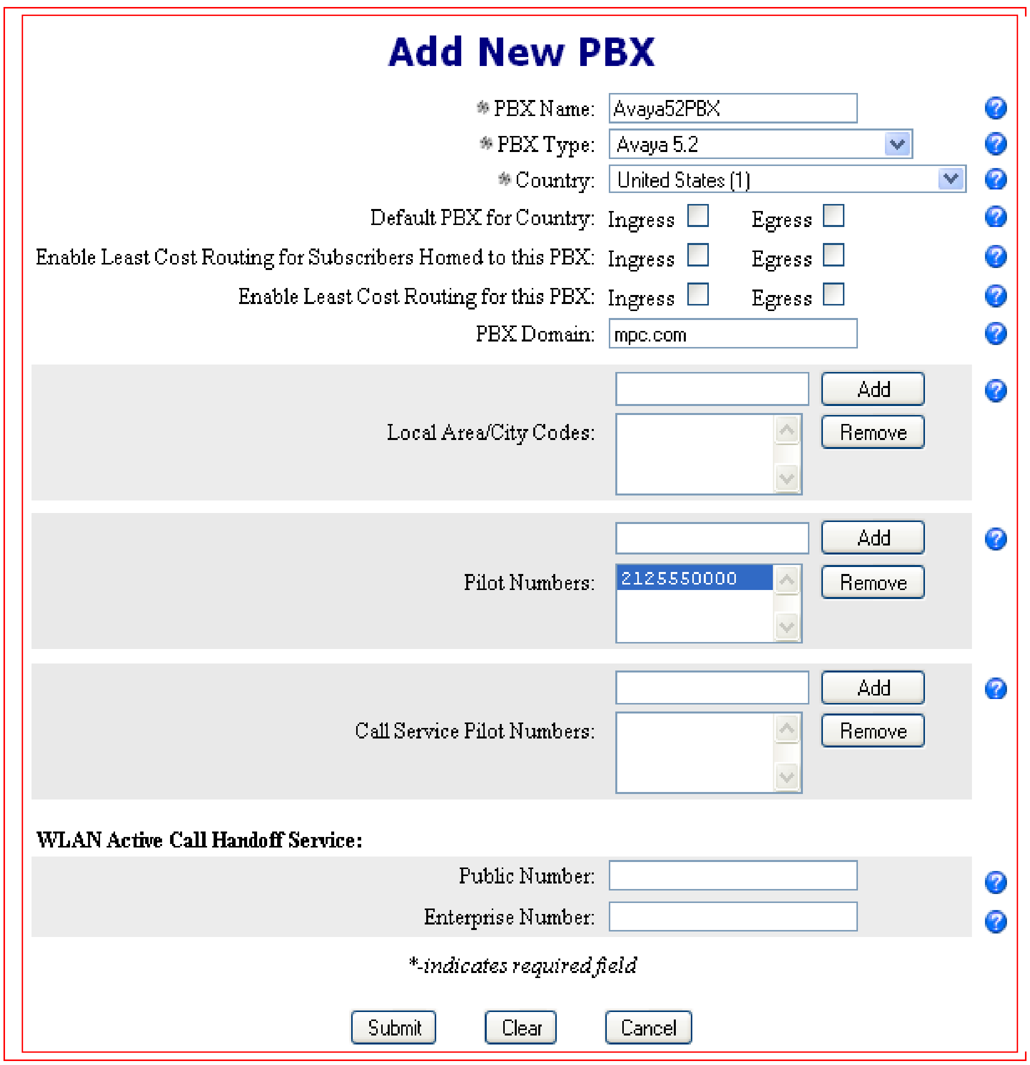


5.1.2. Add an entry for Communication Manager on the Abrazo-E

Select **Voice Network → PBX → Add.** Change the following:

- The **PBX Name** field provides a name for the PBX on the Abrazo-E. It should be a unique identifier for this PBX.
- **PBX Type** should be Avaya 5.2.
- The **Country** 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).
- The **Default PBX for Country** fields are used to identify this node as the default entry point or exit point to be used if no match is found for the area/city code for inbound and outbound routing. Only one default entry/exit point may be selected per country code. This can be a PBX or a Gateway. To define this as the default entry point – the ingress box should be selected. To define this as the default exit point – the egress box should be selected.
- The **Enable Least Cost Routing for Subscribers Homed to this PBX** field allows subscribers to route incoming or outgoing calls through other PBXs within the enterprise. When it is not selected, the subscribers' calls on this PBX will route locally. To allow least cost routing for inbound calls to the enterprise – the ingress box should be selected. To allow least cost routing for calls out of the enterprise – the egress box should be selected.
- The **Enable Least Cost Routing for this PBX** field enables the PBX to route incoming or outgoing calls from subscribers on other PBXs. If it is not selected, only calls from local subscribers will be routed through this PBX. To allow this PBX to be used as part of the least cost routing algorithm for calls into the enterprise – the ingress box should be selected. To allow this PBX to be used as part of the least cost routing algorithm for calls out of the enterprise – the egress box should be selected.
- The **PBX Domain** field value should match the domain defined on the Communication Manager as the Authoritative Domain name. See IP Network Region, **Section 4.2, Step 4.2.3.**
- **Local Area/City Codes** refer to the area/city codes that are considered to be a local call for this PBX (i.e. considered local based on the Local Exchange Carrier LEC the PBX is connected).
- Configure the appropriate **Pilot Numbers** on the **Add New PBX** interface. Pilot Numbers are used to route calls from the mobile operator network into the enterprise and ultimately to the Abrazo-E. These numbers must therefore be routable to the enterprise by the PSTN if a PSTN interconnect to the mobile operator is used. Please see the *Abrazo-E Provisioning Guide* for additional details on Pilot DNs.
- **Call Service Pilot Numbers** – (Optional) This field is not used for integration with the Avaya 5.2 PBX.

- **WLAN Active Call Handoff Service** – Used when dual-mode mobile devices access Abrazo services:
    - Public Number – Publicly routable number used by the mobile device client to handoff an active call. The number should generally be in national format but should match the number as it appears when routed into the Abrazo from the enterprise PBX or PSTN Gateway. Must be a valid number consisting of digits 0-9.
    - Enterprise Number – Internal enterprise representation of the Public Handoff Number if the enterprise PBX or PSTN GW converts the number into a private format (including on-net prefix if appropriate). Must be a valid number consisting of digits 0-9.



5.1.3. Define a new trunk group

Define a new trunk group and add trunk group members to communicate with the Communication Manager. To define a new trunk group, select the PBX to added to the Trunk Group, (**Voice Network → PBX → List all**). Select the PBX from the list. The following screen will be displayed. Click the **Add Trunk Group button**.

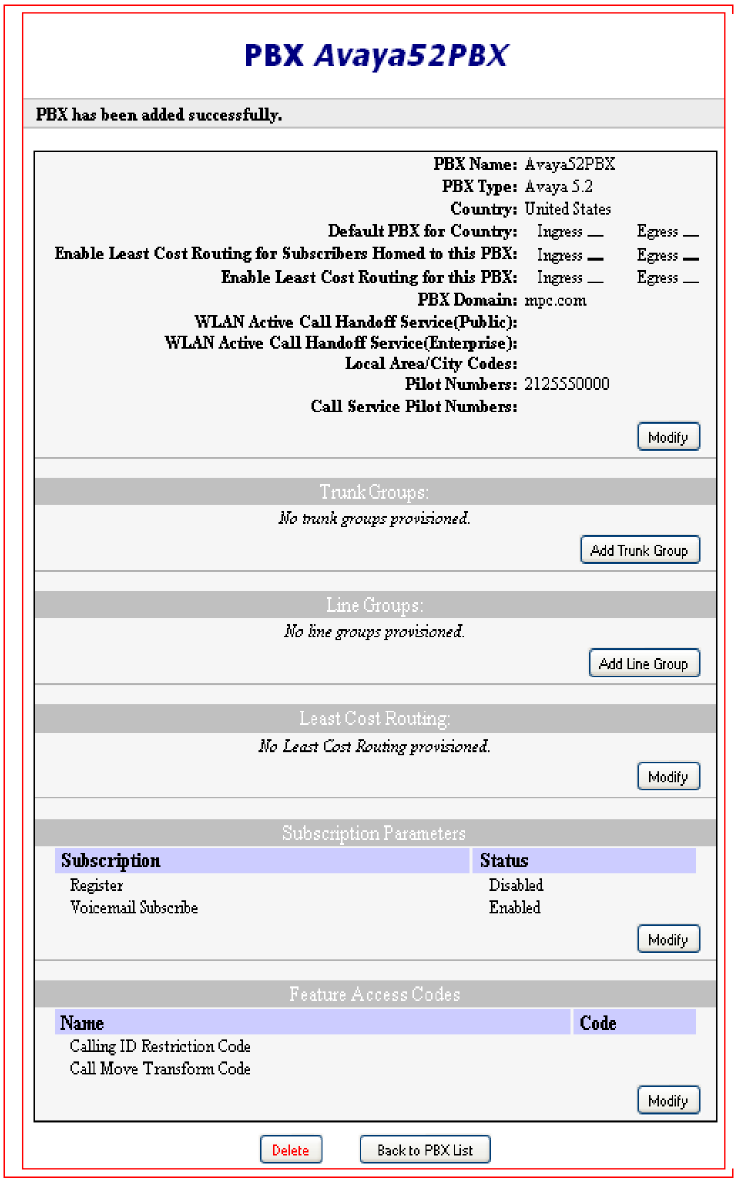


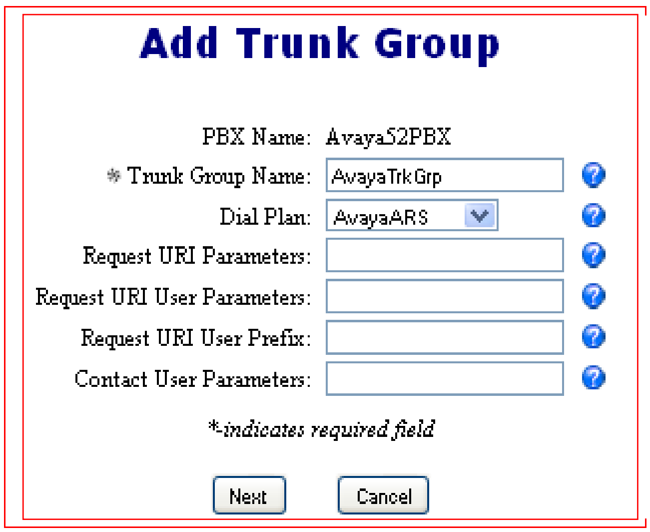
Figure 3

The Add Trunk Group screen is displayed. Change the following:

- The **Trunk Group Name** field provides a name for the trunk on the Abrazo-E. It should be a unique identifier for this trunk.
- **Dial Plan** should be set to the dial plan configured for the Avaya routing prefixes.
- **URI Parameters** are optional fields and not required for integration with the Avaya.

Note: Only one trunk group can be data-filled for the Communication Manager.

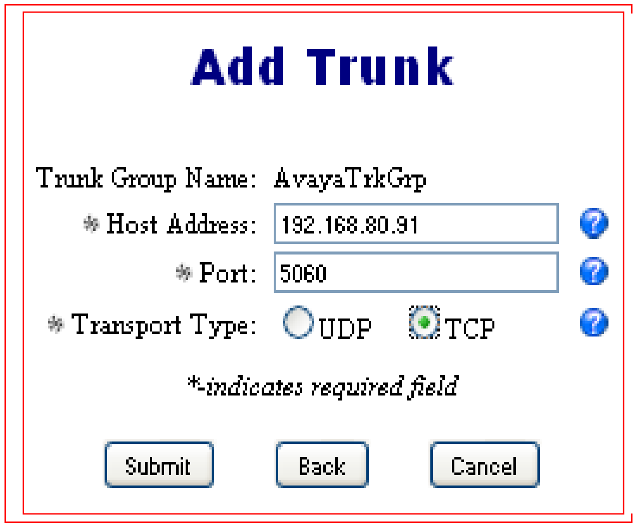
Click **Next** to continue.



The **Add Trunk** screen is displayed. Change the following:

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

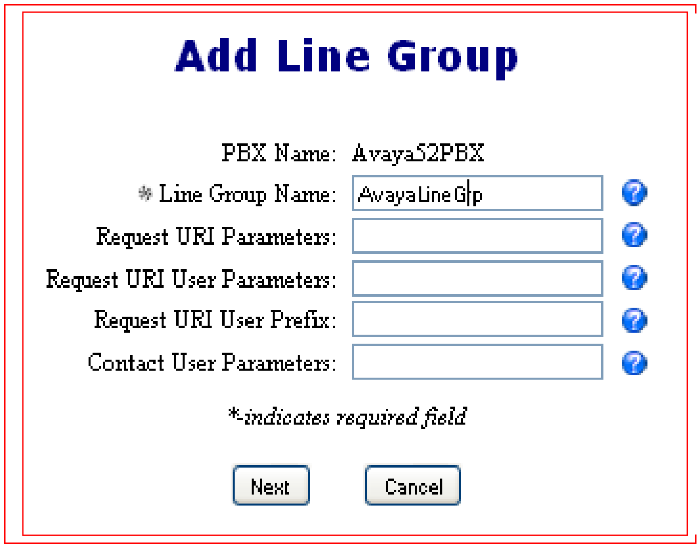
Click **Submit** to continue.



Select **Add Line Group** on the Selected PBX Screen shown in **Section 5.1.3, Figure 3**, to create the SIP line group to interface with the SES. Change the following:

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

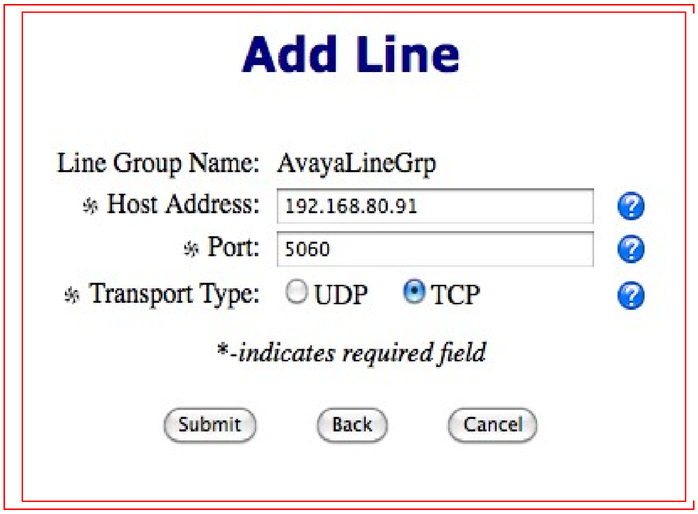
Select **Next** to add individual lines within the group.



Add individual lines within the group. Change the following:

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

Select **Submit** to add the line members.

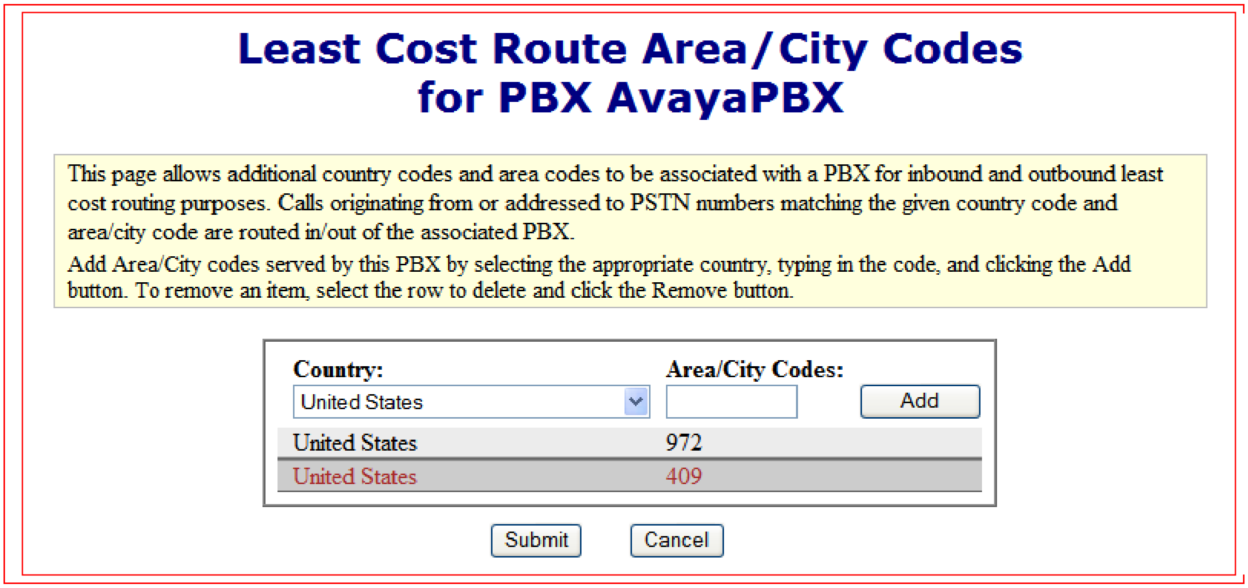


Add Pilot Numbers which are PSTN routable number(s) that represent the mobile route into the enterprise network. The enterprise network must be provisioned to route calls for each pilot number to the Abrazo-E. In the example, 2125550000 was added as a Pilot Number.

5.1.3.1 Least Cost Routing

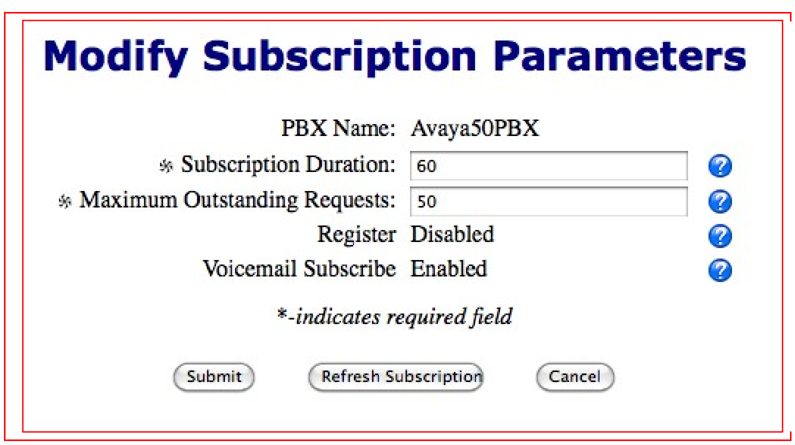
If desired, Least Cost Routing rules can be provisioned. To set up least cost routing rules, select **Modify** in the Least Cost Routing section of the Selected PBX Screen shown **Section 5.1.3, Figure 3**. The following screen will be displayed:

- Add all appropriate least cost routing area/city codes and press **Submit.**



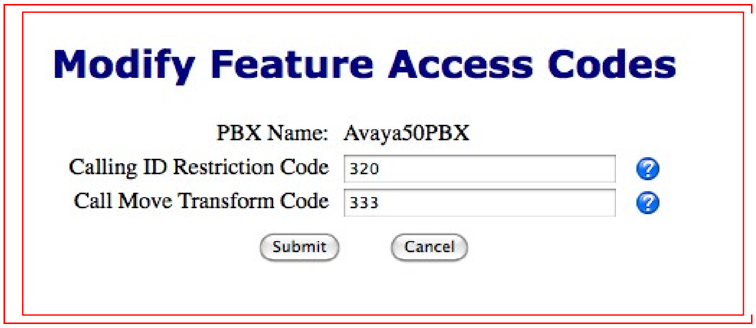
5.1.3.2 Subscription Parameters

Subscription Parameters can be changed by selecting **Modify** in the Subscription Parameters section of the Selected PBX Screen shown **Section 5.1.3, Figure 3**. These values should not be changed. The screen is being shown for informational purposes only.



Feature Access Codes can be changed by selecting **Modify** in the Feature Access Codes section of the Selected PBX Screen shown **Section 5.1.3, Figure 3**. These values should be the same as the ones provisioned on the Communication Manager. Change the following:

- **Calling ID Restriction Code** must match the Avaya field **Per Call CPN Blocking Codes Access Code** in table **feature-access-codes**, tab 3.
- **Call Move Transform Code** must match the Avaya field **Active Appearance Select** in table **off-pbx-telephone feature-name-extensions**, tab 1.



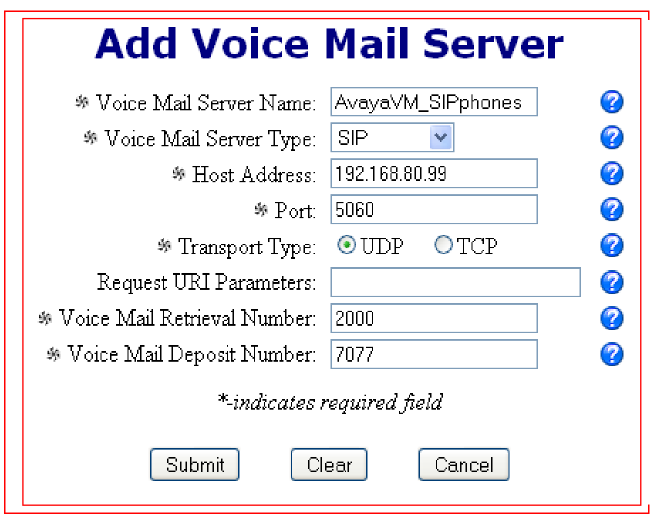
5.1.4. Voice Network : Voice Mail

Provision the voice mail server used with the Communication Manager so the Abrazo can provide a single voice mail solution. If the enterprise has a mix of SIP and H.323 or digital desk phones, separate voice mail instances must be defined in the Abrazo.

5.1.4.1 SIP Desk Phones Only

To add a Voice Mail Server for SIP desk phones only, select **Voice Network → Voice mail → Add**. Select **SIP** as the **Voice Mail Server Type**. The following screen is displayed. Change the following:

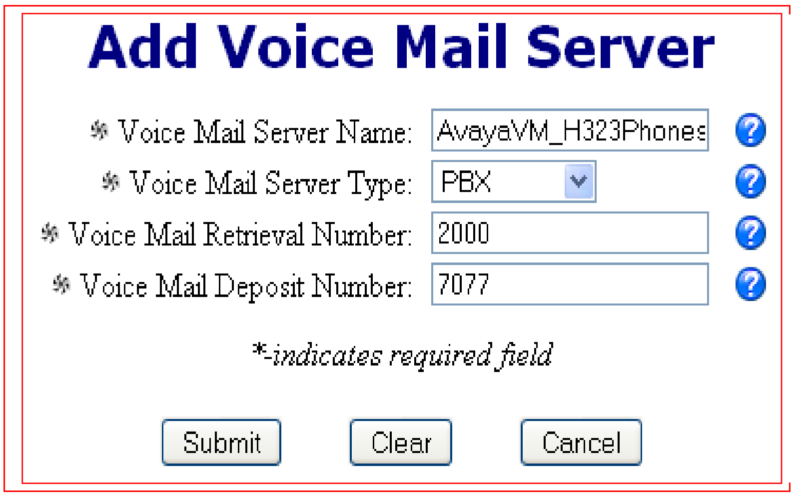
- The **Voice Mail Server Name** should be unique.
- The **Voice Mail Server Type** should be set to **SIP**.
- The **Host Address** and **Port** fields should match the settings of the SES.
- **Voice Mail Retrieval Number** should be set to the number that routes callers to their voicemail.
- **Voice Mail Deposit Number** should set to the feature code, defined in **Section 3.6,** to transfer the call to voice mail.



5.1.4.2 H.323 or Digital Desk Phones Only

To add a Voice Mail Server for SIP desk phones only, select **Voice Network → Voice mail → Add**. Select **PBX** as the Voice Mail Server Type. The following screen is displayed. Change the following:

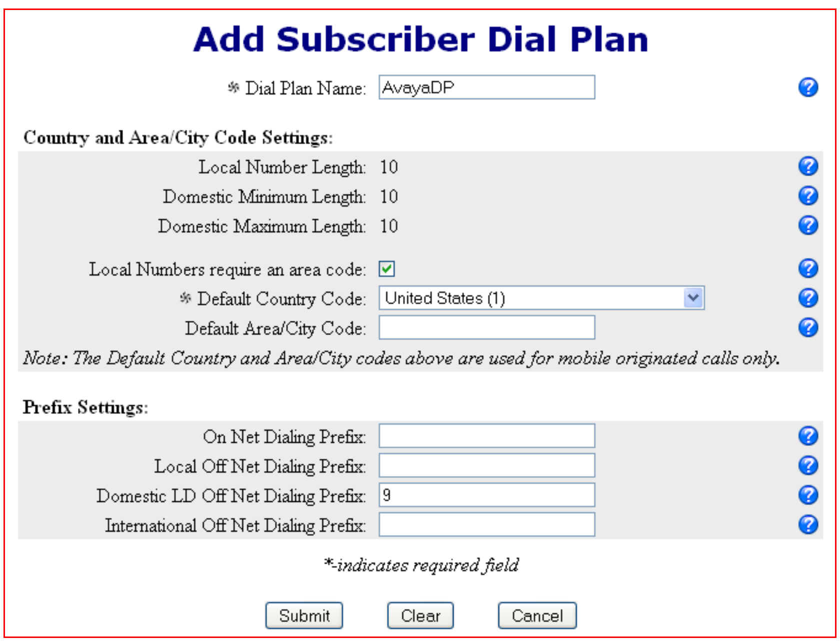
- The **Voice Mail Server Name** should be unique.
- The **Voice Mail Server Type** should be set to **PBX**.
- **Voice Mail Retrieval Number** should be set to the number that routes callers to their voicemail.
- **Voice Mail Deposit Number** should set to the feature code, defined in **Section 3.6,** to transfer the call to voice mail.



5.1.5. Add a Subscriber Dial plan

Before subscribers can be added to the Abrazo system, a Subscriber Dial Plan must first be defined. Add a Subscriber Dial Plan - select **Subscriber → Subscriber Dial Plan → Add.**

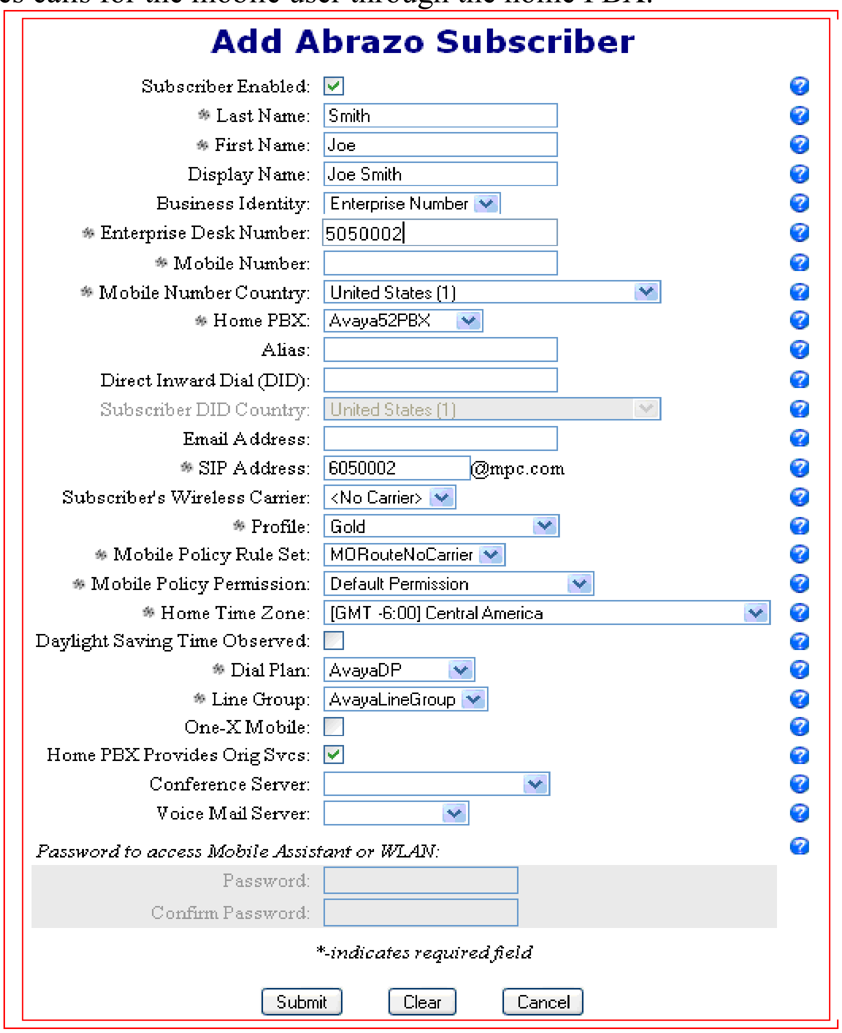
- The **Dial Plan Name** should be unique.
- The **Local Number Requires an Area Code** Used to indicate whether dialing an Area Code is necessary for local numbers.
- **Default Country Code** – The Country Code to be used if none is dialed by the subscriber.
- **Default Area/City Code** – Area Code 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.
- **Local Off Net Dialing Prefix** – The Off Net prefix to indicate calls that are intended for routing outside of the enterprise.
- **Domestic LD Off Net Dialing Prefix** – The Off Net prefix used for routing Domestic Long Distance calls. The example shows "9" which correlates to Avaya's fac code configured in **Section 3.4**.



5.1.6. Add a Subscriber Dial plan

The following steps describe the Abrazo configuration required when the desk phone is SIP, H.323 or digital. The example below illustrates how to add a H.323 subscriber. Add or modify Abrazo subscribers to associate them with the Communication Manager. To add subscribers, select. **Subscriber→Add.** The following screen is displayed.

- Select Avaya52PBX as the user's **HomePBX** field.
- Set the Abrazo **Enterprise Desk Number** to the extension defined for the user's station in the Communication Manager (5050002 in our example). If the subscriber were SIP, the example would reflect 5050000 instead.
- Set the **SIP Address** to the user's off-pbx-telephone station-mapping. The example shows 6050002@mpc.com for the subscriber. If the subscriber were SIP, the example would show 6050000@mpc.com.
- **One-X Mobile** – Leave this field unchecked. One-X Mobile will be supported in a future Abrazo release
- Ensure the option **Home PBX Provides Orig Svcs** is checked. When checked, Abrazo originates calls for the mobile user through the home PBX.



6. General Test Approach and Test Results

6.1. General Test Approach

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 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 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 devices

The following system features were tested to be available on the mobile device using the Abrazo service:

- Abbreviated Dialing - Communication Manager allows extension dialing or internal dialing from the desktop phone. Abrazo allows the user to dial these same abbreviated codes from the mobile phone.
- Ad Hoc Conferencing External to PBX – Allows an Abrazo subscriber to initiate a reservation-less conference from a mobile phone using an external media server. In the certification testing an AudioCodes was used.
- 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 Abrazo 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 Abrazo subscriber uses this feature on Avaya 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 Abrazo 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. Abrazo supports calling line identification when it is the called party. Abrazo 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 Abrazo user's desktop phone.
- Calling Name Identification (CNID) - provides the user with calling party name information. When Abrazo subscribers make a call from their mobile phone, Abrazo 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 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 Abrazo-E 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 Abrazo solution supports enterprise Direct Inward Dialing.
- Direct Outward Dialing – allows users inside an enterprise to dial directly to an external number. The Abrazo 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. Abrazo 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 Abrazo subscriber.
- Least Cost Routing - For mobile originations and terminations, Abrazo 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. Abrazo 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 Abrazo 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 Abrazo 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. Abrazo 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 desk and mobile phone. Feature is invoked from the mobile phone.

6.2. Test Results

The test objectives of **Section 6.1** were verified. The Tango Networks Abrazo Solution successfully completed all test cases for the features identified in **Section 6.1**. Tango Networks is able to route inbound/outbound calls to/from Communication Manager with all services tested.

7. Conclusion

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

8. Additional References

The documents referenced below were used for additional support and configuration information. This section references the Avaya documentation relevant to these Application Notes. The following Avaya product documentation is available at http://support.avaya.com.

[1] *Administering Avaya Aura™ Communication Manager,* May 2009, Issue 5.0.0, Document Number 03-300509.

[2] *Installing, Administering, Maintaining, and Troubleshooting Avaya Aura™ SIP Enablement Services*, November 2009, Issue 8.0, Document 03-600768.

[3] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide Release 3.1,* Document Number 16-300698.

[4] *Avaya one-X Deskphone SIP for 9600 Series IP Telephones Administrator Guide, Release 2.6,* Document Number 16-601944.

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

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.