Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Tango Networks Abrazo Solution with Avaya Communication Manager, Avaya SIP Enabled Services, Avaya Application Enablement Services, Avaya Modular Messaging Server, Avaya IA 770 INTUITY and Avaya IP Telephones - Issue 1.0

Abstract

These Application Notes describe a compliance-tested configuration comprised of the Tango Networks Abrazo Solution connected to an Avaya telephony infrastructure. The Abrazo solution extends enterprise PBX functionality to mobile devices, allowing end users to be accessible when out of the office. The Abrazo solution integrates mobile devices with existing Private Branch Exchanges (PBXs) so that the PBX sees the mobile device as another desk phone. This allows the existing PBX feature set to be applied consistently across both devices. Mobile specific functionality is then layered on top.

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

1. Introduction

These Application Notes describe a compliance-tested configuration comprised of the Tango Networks Abrazo Solution connected to an Avaya telephony infrastructure, including Avaya Communication Manager, Avaya SIP Enablement Services, and Avaya Application 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 unparalleled level of convergence allows mobile phones to offer the same productivity features as a conventional enterprise desk phone.

1.1. Background

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

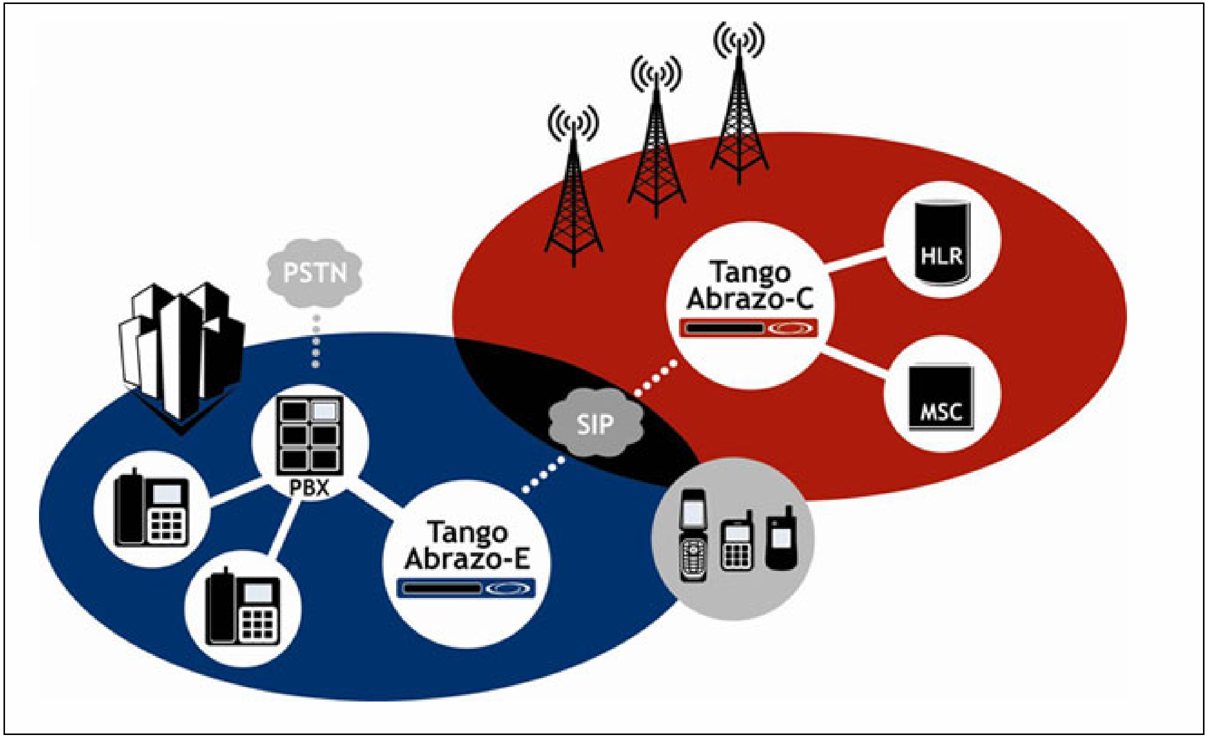


Figure 1: Tango Networks' Architecture Diagram

The Abrazo solution interacts with the Avaya Communication Manager (ACM) via Avaya SIP Enablement Services (SES) and Application Enablement Services (AES) as shown in **Figure 2**.

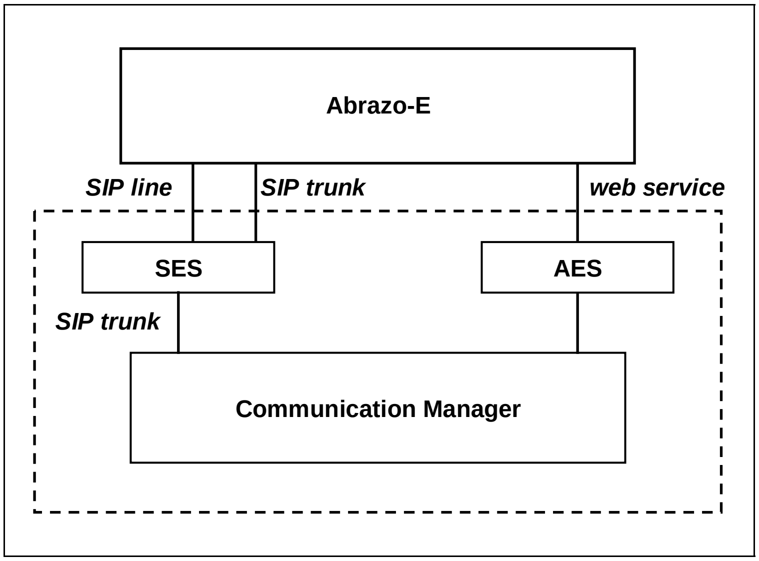


Figure 2: Abrazo-E Interfaces to the Avaya Telephony Infrastructure

The Abrazo solution uses a combination of SIP lines and trunks to integrate with Avaya 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). The Abrazo solution uses the web services telephony interface, which is *Telephony Services Application Programming Interface* (TSAPI) based, to connect with Application Enablement Services for enabling the Call Move service to be originated from the mobile phone.

Mobile Originations

The 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 Communication Manager. Within Avaya 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 Communication Manager. Setting the CLID in the P-Asserted-Identity header ensures that

Avaya SIP Enablement Services recognizes the user and applies originating services (i.e., treats this as a SIP line call).

Mobile Terminations

To receive calls made to a subscriber, the Abrazo solution registers a route pointing to an Abrazo Session Conductor by sending a SIP register message. This is done on a per subscriber basis when subscribers are added to the Abrazo system. Once a subscriber is registered with the Avaya, all calls made to the subscriber fork to 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.

To prevent the Avaya from recognizing the originating subscriber and erroneously applying originating services again, the P-Asserted-ID header is removed from the SIP Invite message. This causes the Avaya to treat this request as a trunk based origination and to route the call to the PSTN without providing any subscriber services. By leaving the user portion of the originating subscriber information intact, CLID information is preserved.

1.2. Solution Configuration

These application notes describe a solution for integrating the Tango Abrazo-E with the Avaya Product Portfolio. **Figure 3** 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.

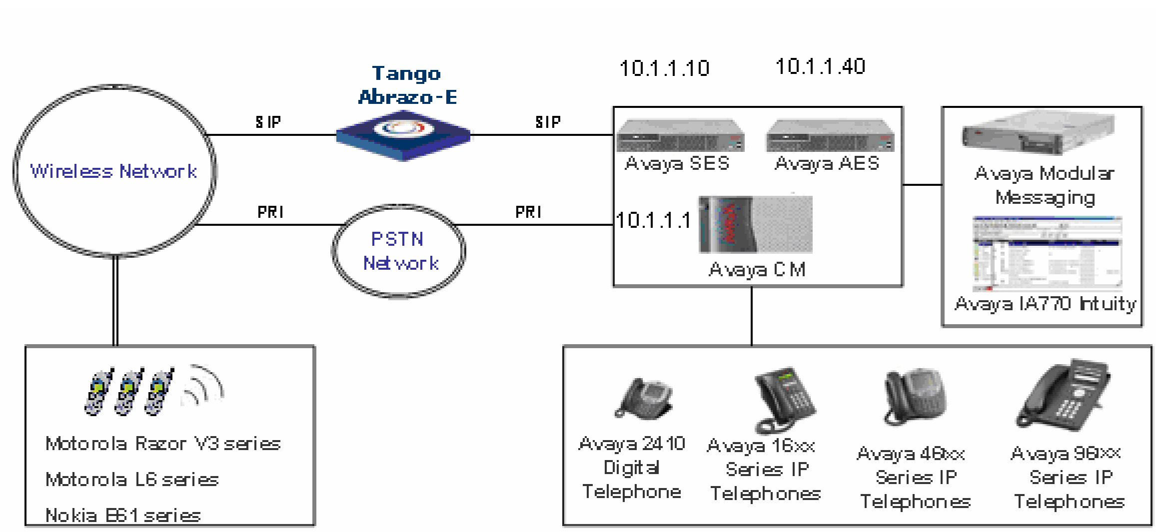


Figure 3: Interoperability Configuration Diagram

1.3. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya PBX Products	
Avaya S8300 Server running Avaya Communication Manager	Avaya Communication Manager 4.0.1 4x.00.1.731.2
Avaya G700 Media Gateway with MM712 DCP Media Module 8	26.31.0 FW 008
Avaya SIP Enabled Services (SES) Server	SES-4.0.0.0-033.6
Avaya Application Enablement Services	4.1
Avaya Messaging (Voice Mai)l Products	
Avaya IA 770 INTUITY	4.0
Avaya Modular Messaging Server	3.1
Avaya Telephony Sets	
Avaya 9600 Series IP Telephones	Avaya one-X Deskphone SIP 1.5
Avaya 9600 Series IP Telephones	Avaya one-X Deskphone Edition 1.2
Avaya 1600 Series IP Telephones	1.23
Avaya 4600 Series IP Telephones	SIP (2.2) H.323 (2.8)
Avaya 2410 Digital Telephone	4.0
Tango Abrazo Products	
Tango Networks Abrazo-Enterprise Release	3.2
Tango Networks Abrazo-Carrier Release	3.2
Mobile Devices	
Nokia	E61 Series
Motorola	L6 Series
Motorola Razor	V3 Series

The configuration tested utilized Avaya Communication Manager running on the Avaya S8300 server; however the solution described in this document is also extensible to other Avaya servers and media gateways.

In addition, the configuration tested interfaced with a GSM wireless network and utilized GSM mobile devices; however, the solution described in this document is also extensible to IS-41 and IMS based networks. Any mobile device may be supported with the Tango Abrazo solution.

2. Configure Avaya Communication Manager

Basic configuration of Avaya Communication Manager and Avaya SES are beyond the scope of these Application Notes. See section 9 for Avaya documentation references. It is assumed that the reader has a basic understanding of the administration of Avaya Communication Manager and has access to the System Access Terminal (SAT).

This section describes the steps required for Avaya Communication Manager to support the configuration in **Figure 3: 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.

2.1. System Parameters Customer Options

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

Step	Description
2.1.1.	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
```

Step	Description
2.1.2.	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

        (NOTE: You must logoff & login to effect the permission changes.)
```

2.2. IP Codec Set

This section describes the steps for administering the codec set in Avaya Communication Manager. This codec set is used in the IP Network Region for communications between Avaya Communication Manager and Avaya SES.

Step	Description
2.2.1.	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:
```

2.3. IP Network Region

This section describes the steps for administering the IP Network Region in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SIP Enablement Services.

Step	Description
2.3.1.	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, set **Codec Set** to the number of the IP codec set configured in Step 1.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1      Authoritative Domain: dev4.com
    Name: 1
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
```

2.4. IP Node Names

This section describes the steps for setting IP node name for Avaya SES in Avaya Communication Manager.

Step	Description
2.4.1.	Enter the **change node-names ip** command. On page **1** of the **change node-names ip** form, enter the name for the SES, **“SES”**, and enter the IP address of the SES, **“10.1.1.10”**.

```
change node-names ip                                                Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
50SES               50.1.1.50
AES-DevCon2         192.45.100.153
G250-314            10.10.200.10
G350                50.1.1.10
SES                 10.1.1.10
default             0.0.0.0
mm                  10.1.1.45
msgserver           10.1.1.20
procr               10.1.1.1
```

2.5. Trunks and Signaling Groups for Avaya SES

This section describes the steps for administering the trunk group and signaling group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

This SIP trunk will carry the SIP signaling sent to the Abrazo-E for mobile originated and terminated calls. This SIP trunk will also provide the trunking for calls originated by the Abrazo-E when acting as a SIP endpoint to support intelligent call delivery.

Step	Description
2.5.1.	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"**

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

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: T0 SES                     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:
                                                     Number of Members: 0
```

Step	Description
2.5.2.	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 a local C-LAN board, or "**procr**" if the local node is an Avaya S8300 Server. • **Near-end Listen Port** – specify the local listen port, typically **5060**. • **Far-end Node Name** – enter the node name of the SES configured in **Step 2.4.1** • **Far-end Listen Port** – specify the local listen port, typically **5060.** • **Far-end Domain – dev4.com** • **Far-end Network Region** – enter the IP network region configured in **Step 2.3.1** • **DTMF over IP** – set to "**rtp-payload**". • **Direct IP-IP Audio Connections** – set to "**y**".

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

 Group Number: 1              Group Type: sip
                        Transport Method: tcp

          IP Video? n




   Near-end Node Name: procr                 Far-end Node Name: SES
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1
       Far-end Domain: dev4.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
```

Step	Description
2.5.3.	Enter the **change trunk-group i** command, where "i" is the number of the trunk group configured in **Step 2.5.1**. **On Page 1** of the **trunk-group** form, configure the following: • **Signaling Group** – enter the Signaling Group number that was used in **step 2.5.2.** • **Number of Members** – set to **24**

```
change trunk-group 1                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: T0 SES                     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: 1
                                                     Number of Members: 24
```

2.6. Dial Plan – AAR and Route Pattern

This section describes the steps for setting the Dialplan, ARS digit analysis and Route Pattern in Avaya Communication Manager for proper routing of calls from Avaya Communication Manager to Avaya SES. These calls are ultimately destined for the Tango Abrazo-E.

If the connectivity between the wireless carrier and the enterprise is via VoIP, then no incremental dial plan modifications are required on Avaya Communication Manager to route inbound calls to the Tango Abrazo.

If the connectivity between the wireless carrier and the enterprise is via the PSTN, then dial plan and route patterns must be configured on Avaya Communication Manager for both pilot directory numbers and service pilot pools as described in the steps below.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.6.1.</td><td>Calls are routed to route patterns based upon the dialed number. A telephony route pattern must be created that matches the pilot directory numbers and routes requests over the SIP trunk to the SES to the Abrazo-E. The SES will also match the pilot DN and route to the Abrazo-E.

A route pattern should be created for all of the pilot directory numbers. The example below specifies that incoming numbers beginning with 56 with a length of nine digits get routed to Tango.

<pre>
change uniform-dialplan 0   Page 1 of 2
                         UNIFORM DIAL PLAN TABLE
                                                          Percent Full: 0
Matching   Insert   Node Matching   Insert   Node
Pattern Len Del Digits Net Conv Num   Pattern   Len Del Digits   Net
Conv Num
56   9   0   aar   n   n

change aar analysis 0   Page 1 of 2
                        AAR DIGIT ANALYSIS TABLE
                                                          Percent Full: 2
Dialed   Total   Route   Call Node ANI
String   Min Max Pattern Type Num Reqd
   56   9   9   140   aar   n
</pre></td></tr>
</table>

2.6.2. Calls are routed to route patterns based upon the dialed number. A telephony route pattern must be created that matches the service pilot pool numbers and routes requests over the SIP trunk to the SES to the Abrazo-E. The SES will also match the service pilot pool numbers and route to the Abrazo-E.

A route pattern should be created for all of the service pilot pool numbers beginning with the dial plan analysis table. In the example below, our service pilot pool number is 987788011. The leading number 9 is configured as a feature access code in the dial plan analysis table.

NOTE: The leading digit of the service pilot pool numbers must be defined as a feature access code (fac) in the dial plan analysis table.

```
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
    0            3    fac
    1            3    fac
    2            5    ext
    3            1    fac
    4            5    udp
    5            5    ext
    6            5    ext
    7            5    aar
    8            1    fac
    9            1    fac
    *            2    fac
    *            3    fac
    *            4    dac
    #            2    fac
    #            3    fac
```

2.6.3. The feature access code table defines whether Auto Alternate Routing (AAR) or Auto Route Selection (ARS) routing should be utilized. In our example, the 9 indicates that ARS routing should be utilized.

NOTE: ARS is typically used for public translations while AAR is typically used for private translations. Either the AAR or the ARS table may be utilized for the Tango Abrazo service.

```
change feature-access-codes                                     Page   1 of   8
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: 101
         Abbreviated Dialing List2 Access Code: 102
         Abbreviated Dialing List3 Access Code: 103
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: 106
                       Answer Back Access Code: *550
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 3
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: *5     Deactivation: #5
Call Forwarding Activation Busy/DA: *2     All: *551   Deactivation: #2
   Call Forwarding Enhanced Status:        Act:        Deactivation:
                         Call Park Access Code: *552
                       Call Pickup Access Code: *6
CAS Remote Hold/Answer Hold-Unhold Access Code: #6
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code: 090
           Contact Closure   Open Code:        Close Code:
```

2.6.4.	During digit translations, Avaya Communication Manager automatically deletes the feature access code from the digit pattern prior to indexing into the ARS or AAR table. So in our example, the digit pattern for our service pilot number now becomes 877880011. Because of this, the Tango Abrazo uses the digit conversion table to prefix the feature access code back into the digit pattern so that the digit pattern matches the original service pilot pool number. In the example below, the first 3 digits are replaced with 9877. So in our example, the digit pattern is restored to 9877880011.
	
	`change ars digit-conversion 877                                 Page   1 of   2`
	`                       ARS DIGIT CONVERSION TABLE`
	`                                 Location: all               Percent Full: 0`
	
	`  Matching Pattern     Min   Max   Del   Replacement String   Net  Conv ANI Req`
	
	`  877                  10    10    3     9877                 ars   n        n`
	`  9790                 11    11    1                          ars   n        n`
	`                                                                             n`
	`                                                                             n`
	`                                                                             n`
	`                                                                             n`
	
	`                                                                             n`

2.7. SIP Line Configuration

This section describes the steps for setting for Off-PBX-Telephones in Avaya Communication Manager for proper routing of calls from Avaya Communication Manager to Avaya SES, ultimately destined for the Tango Abrazo-E.

Step	Description
2.7.1.	Validate that the system parameters features are administered as required for the Abrazo solution when acting as a SIP telephone. Generic or default settings should be acceptable because no known changes have been identified.
2.7.2.	Determine the feature name extensions and feature access codes by using the **change dial plan analysis** and **change feature access codes** commands. Analyze these to ensure there is no conflict with the Abrazo mid-call services (conference, sacada, transfer).
2.7.3.	Create a Class of Service (COS) and Class of Restriction (COR) set by using the **change class-of-service** and **change class-of-restriction** commands. This essentially defines the Avaya feature set that will be available to the SIP phone.

Step	Description
2.7.4.	Every Abrazo user must be defined as 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 Avaya Communication manager extension to the extension defined in the SES. • Set the **Station Extension** to the station extension of Abrazo-E as configured above (The example which follows uses 34071.) • Set **Application Type** to **OPS** • Set **Phone Number** to the number Abrazo will use for registration and call origination and terminations, which is the user portion of the SIP addresses defined for subscribers on Abrazo-E. This field maps the Avaya media server extension defined on the SES (example: 34071) to this station defined on the Communication Manager. • Set **Trunk Selection** to the number of the SIP trunk group connected to the SES server. • Set **Configuration Set** to the set to be used for IP phone call treatments as defined above. • Set **Mapping Mode** to **both.**

```
change off-pbx-telephone station-mapping 34071            Page 1 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
Station Application Dial Phone Number   Trunk        Configuration
Extension           Prefix              Selection    Set
34071 OPS -         34071               10               1

change off-pbx-telephone station-mapping 34071            Page 2 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
Station    Call   Mapping  Calls    Bridged
Extension  Limit  Mode     Allowed  Calls
  34071     4     both     all      both
```

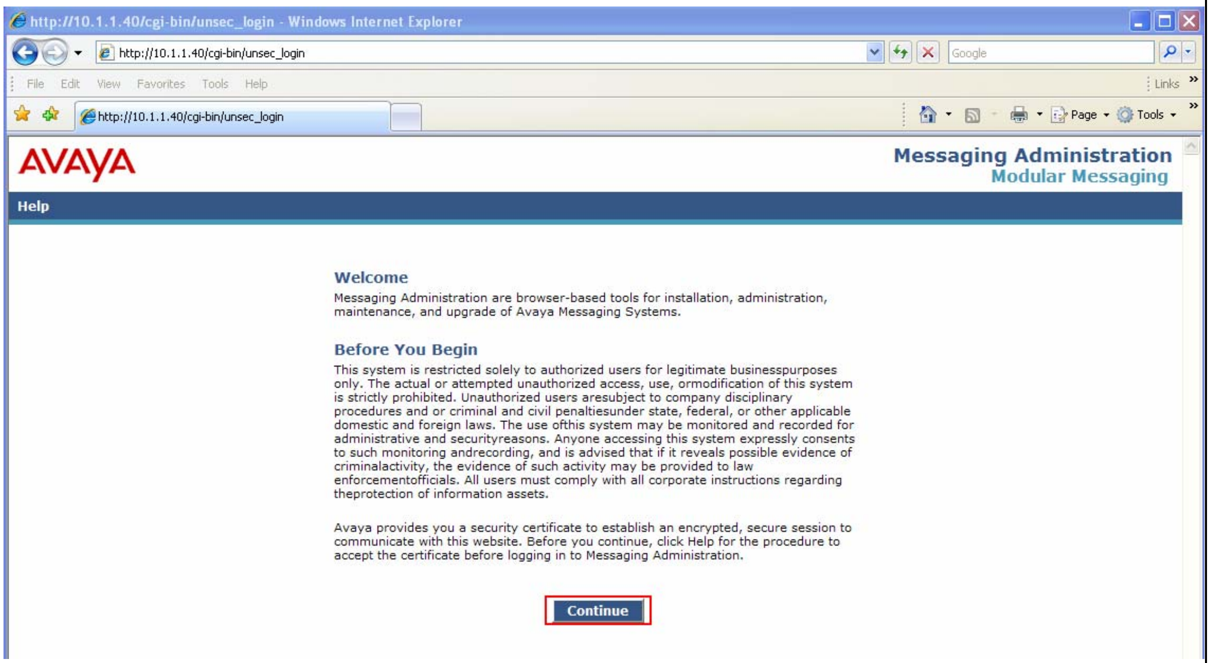
2.8. Voice Mail Configuration

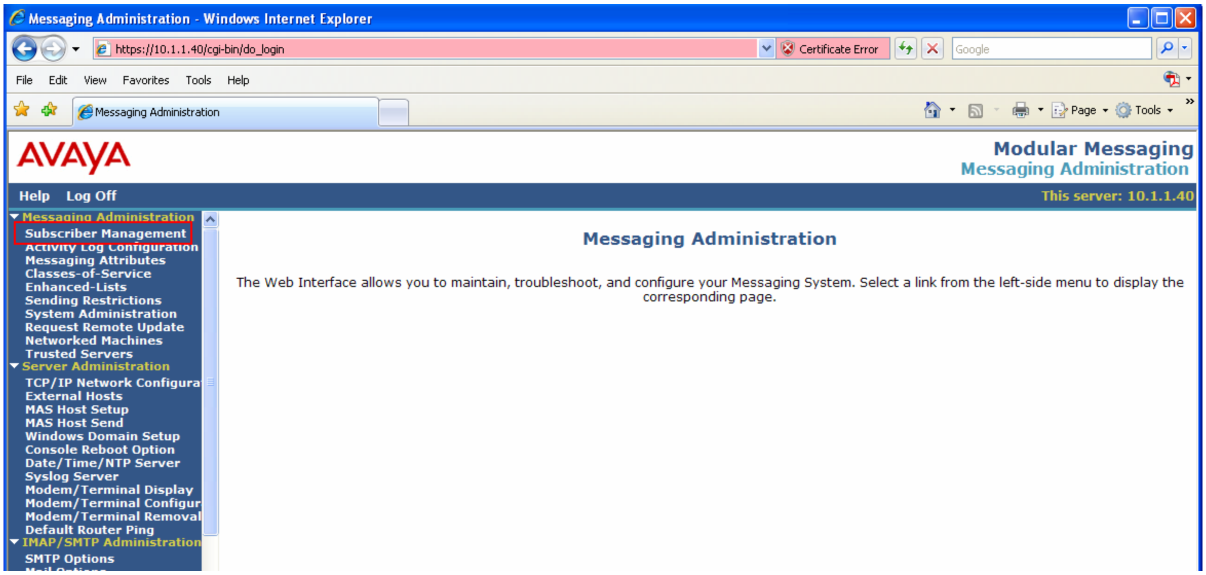
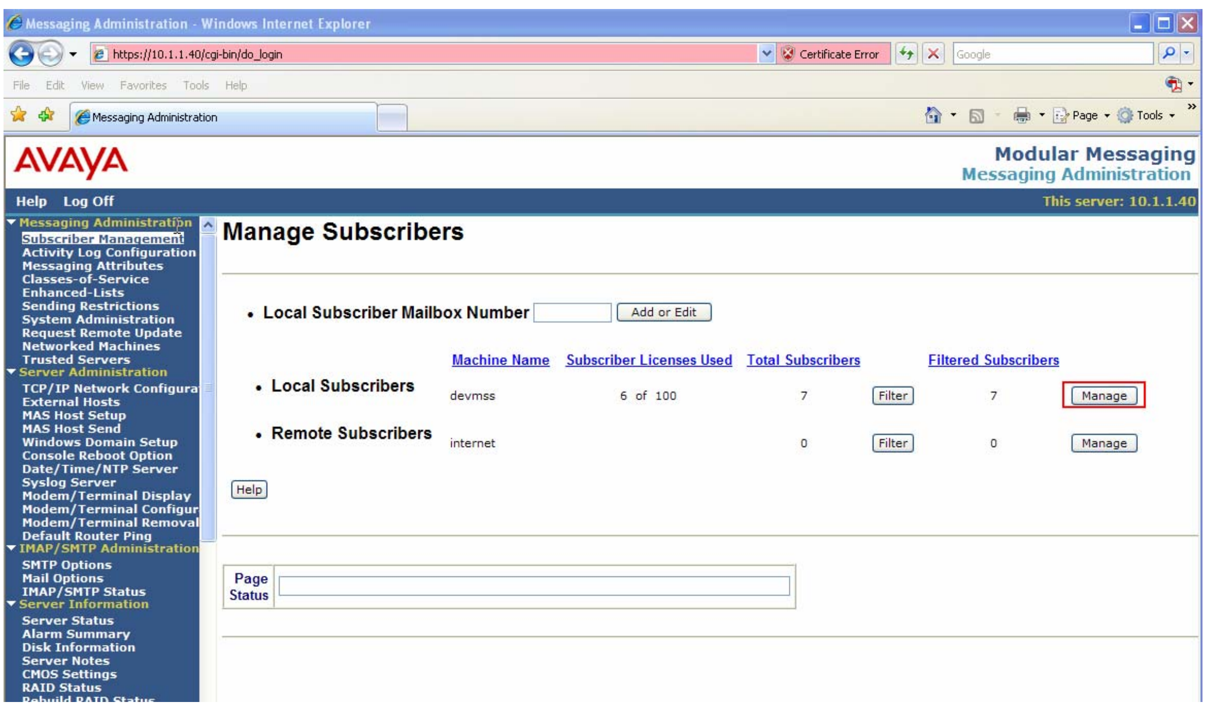
This section describes the steps for configuring voicemail for extensions in the Avaya telephony infrastructure. For informational purposes, steps for both Avaya Modular Messaging and Avaya IA770 INTUITY AUDIX are included in this document. Use the setup information appropriate for the environment being configured.

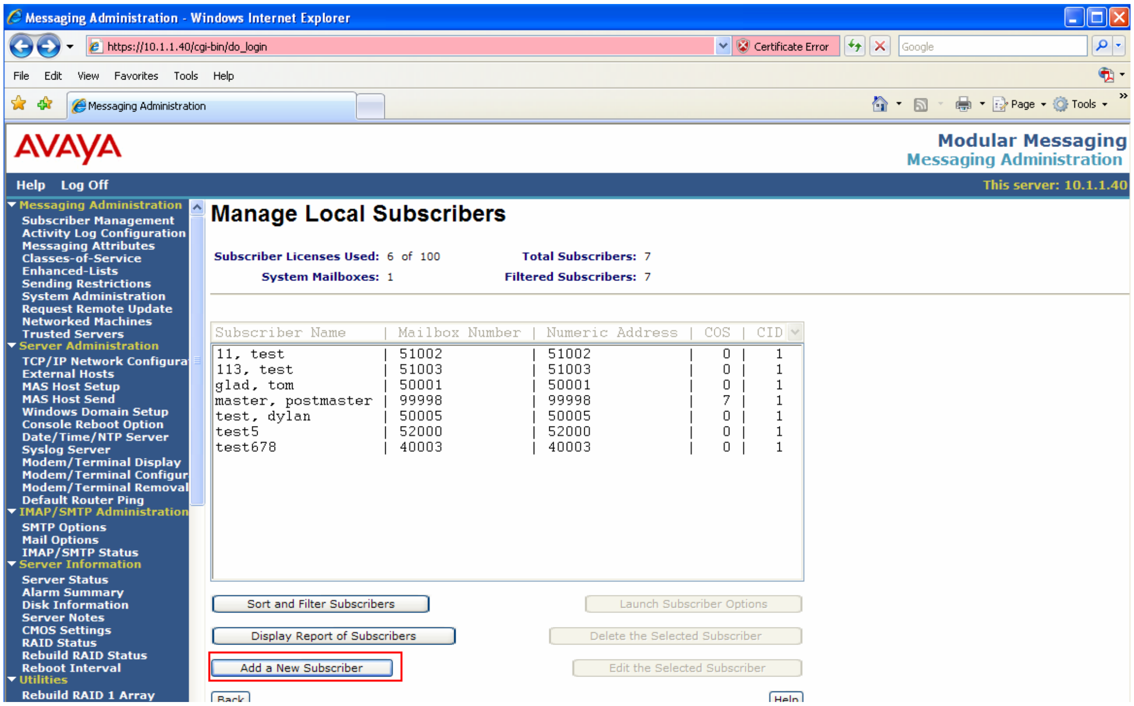
Note: It is recommended that at least four rings be used to route a call to voice mail.

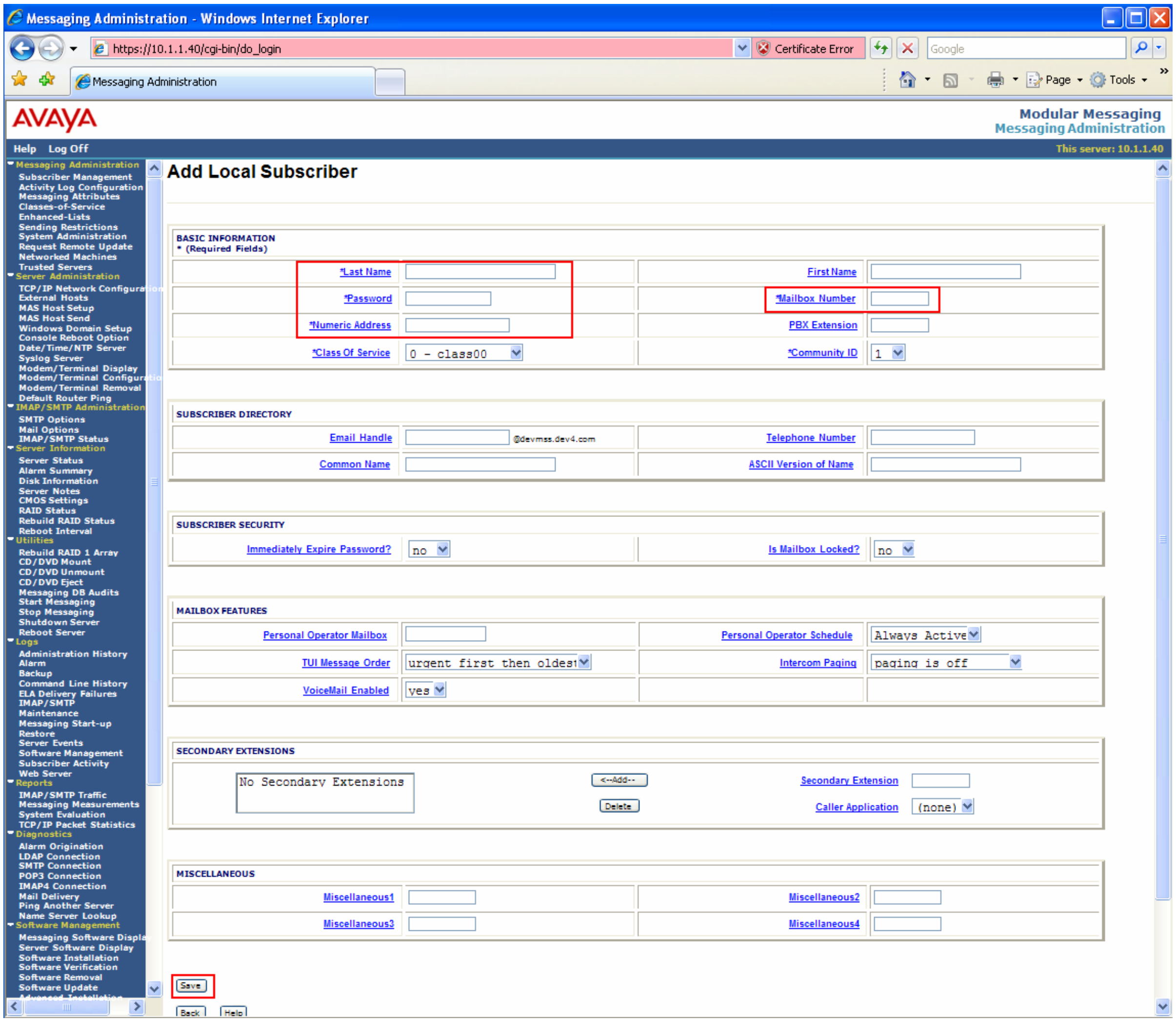
2.8.1. Configure Subscriber on Avaya Modular Messaging

2.8.2 Connect to the Modular Messaging Administration page, For this example http://10.1.1.40/ was used. Select **Continue.** Enter the appropriate Username and Password information, click Login to proceed.

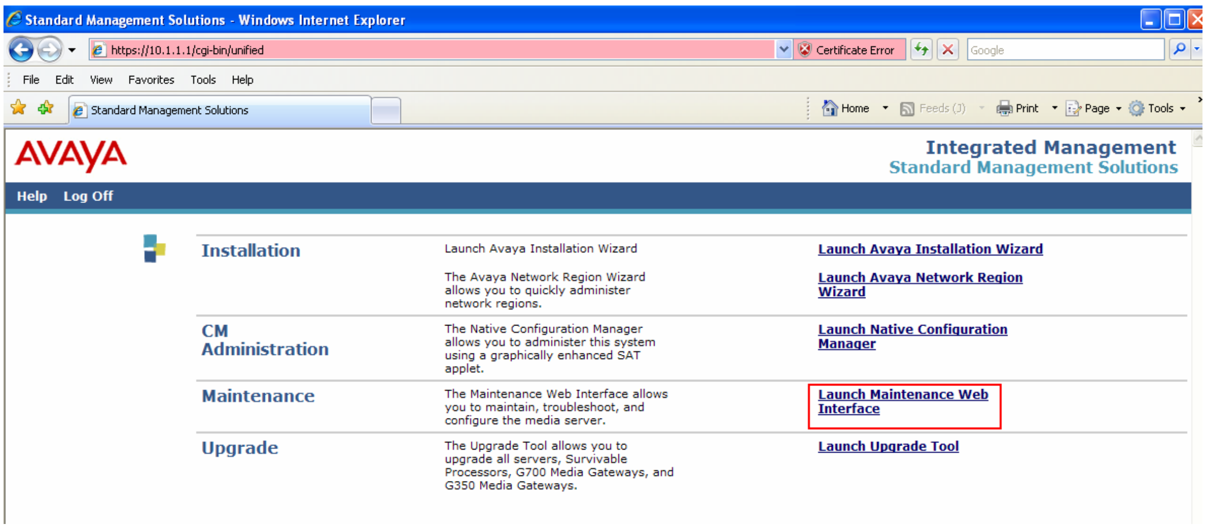


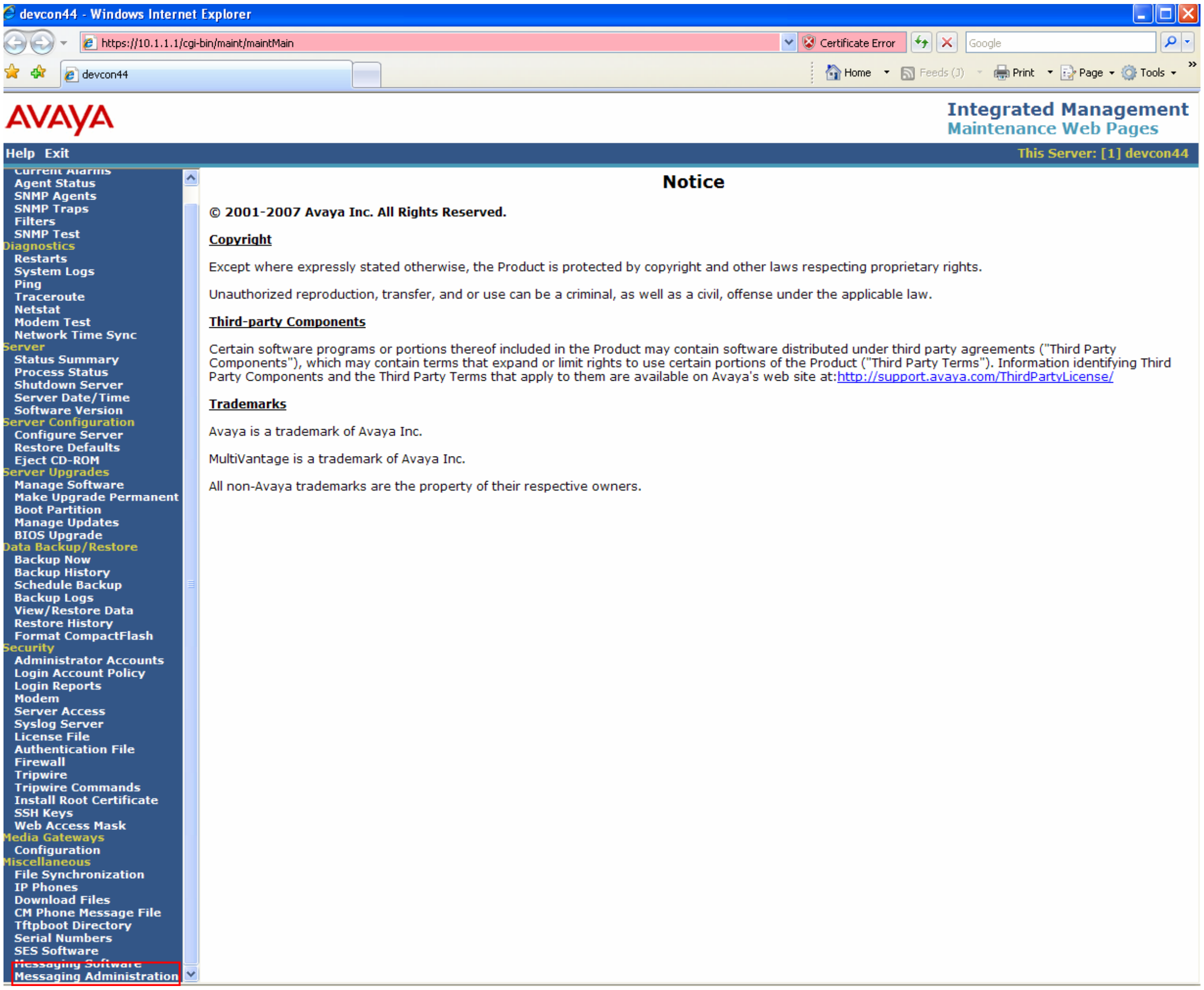
Step	Description
2.8.3.	Select **Subscriber Management**. 
2.8.4.	Select **Manage**. 

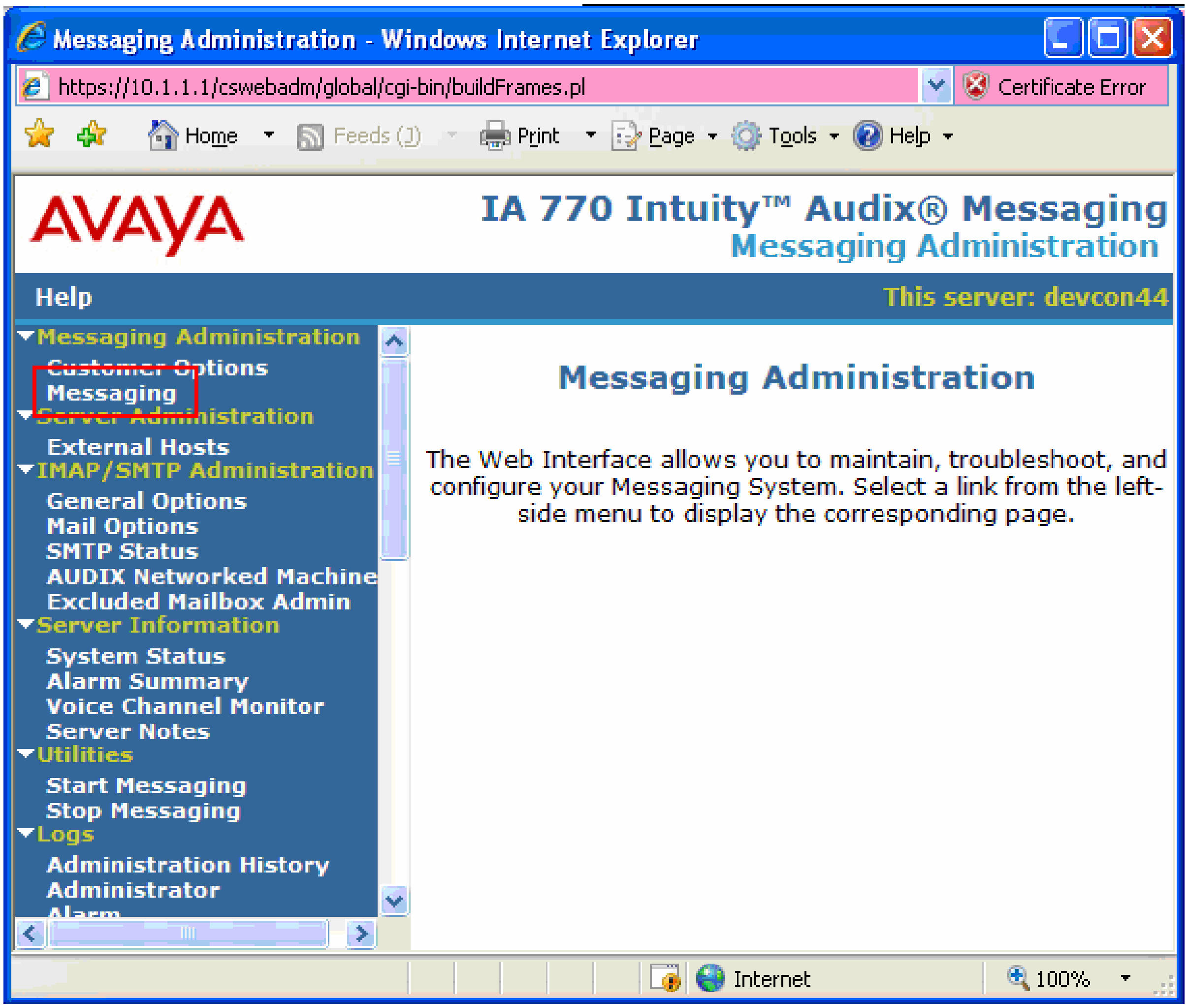
Step	Description
2.8.5.	Select **Add a New Subscriber**. 

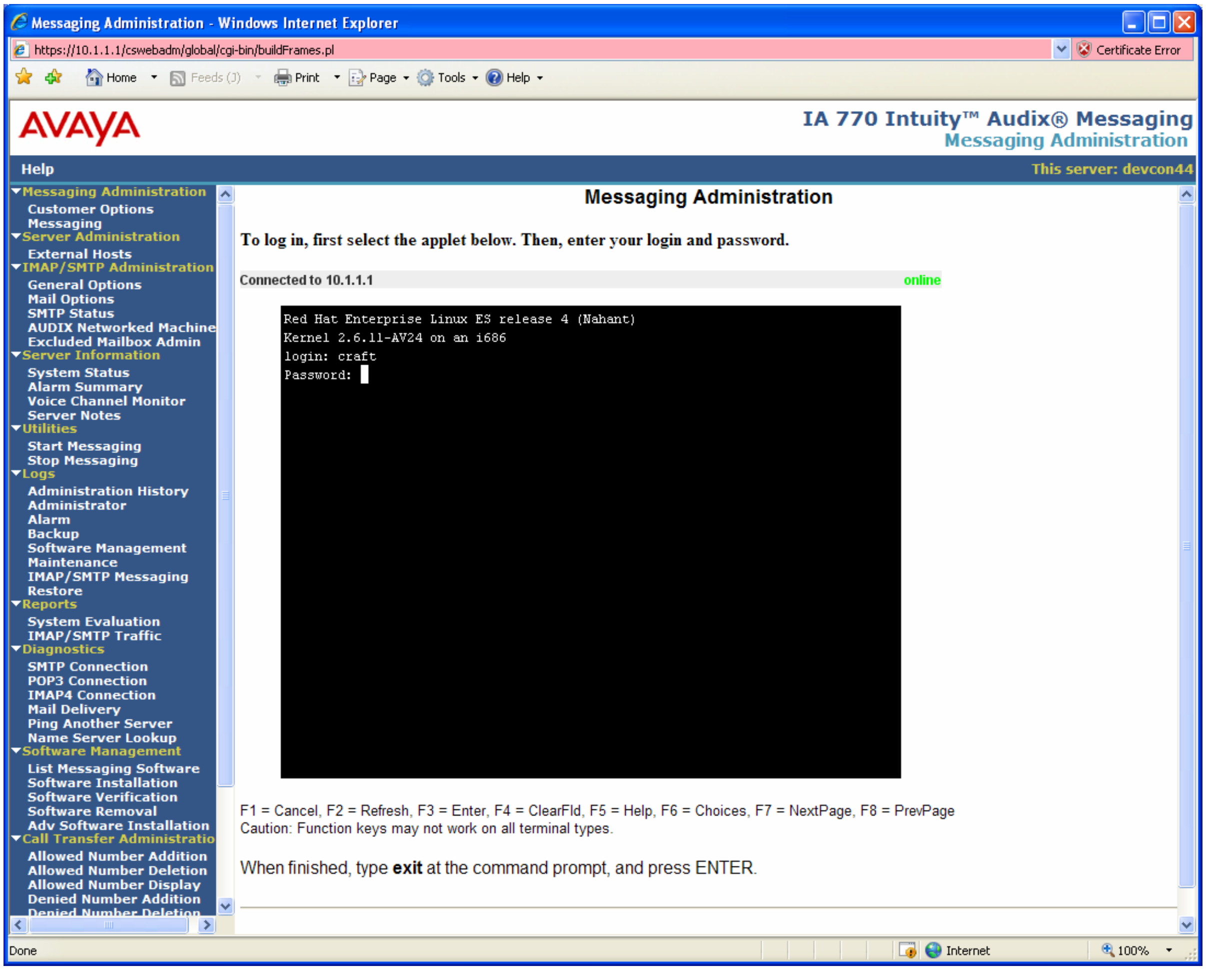
Step	Description
2.8.6.	Enter the Follow User Information: **Last Name, Password Mailbox Number, Numeric Address**. Select **Save** to continue. 

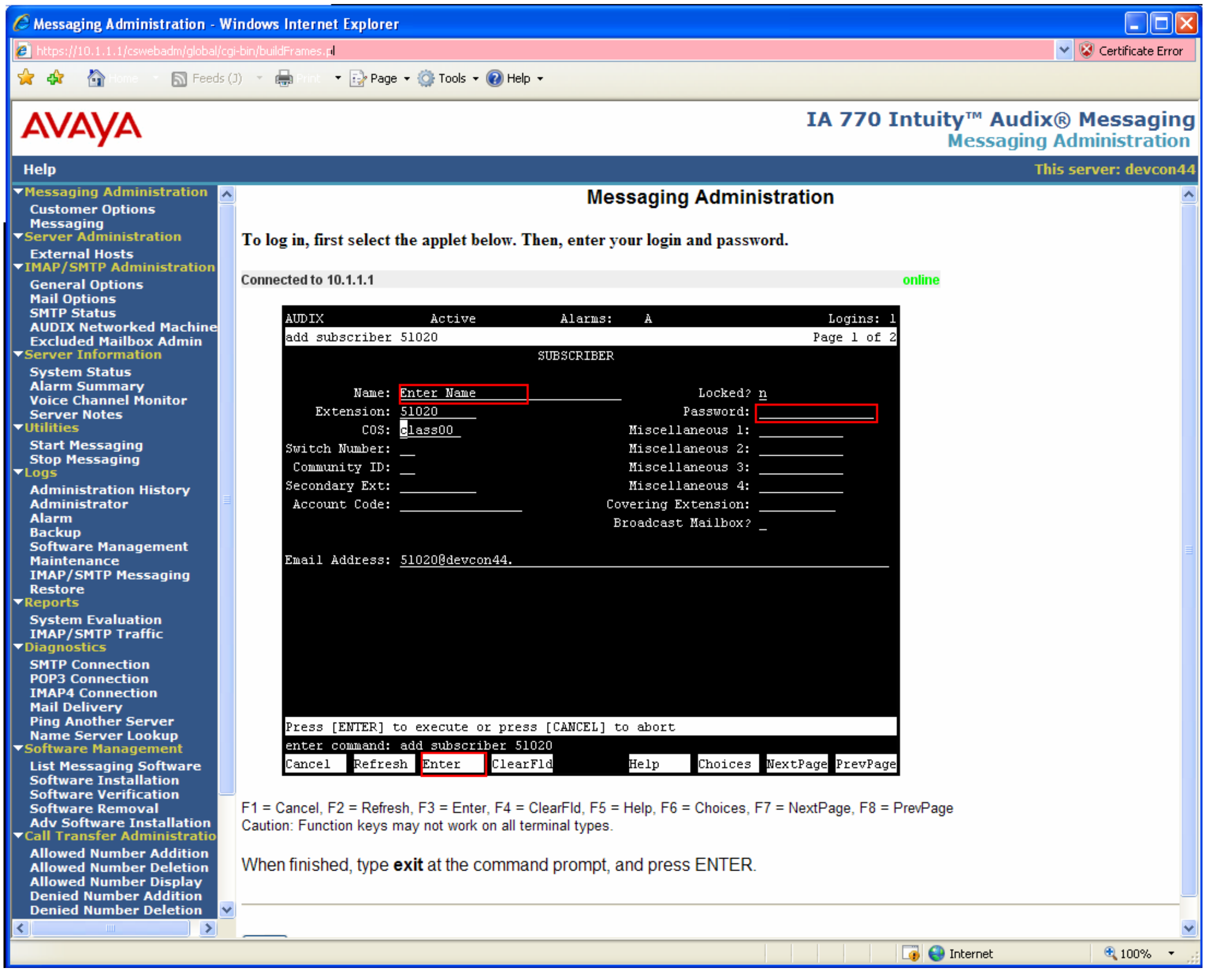
2.8.7. Configure Subscriber on Avaya IA770 INTUITY AUDIX

2.8.	Connect to Avaya Communication Manager; for this example http://10.1.1.1/ was used. Select **Continue**. Enter the appropriate Logon ID and Password information and click **Login**. Click **Launch Maintenance Web Interface** to continue. 

2.8.9	Click on **Messaging Administration**. 

2.8.10	Click on **Messaging**. 

2.8.11	Login with the appropriate Login and Password. 

2.8.12	At the prompt, Type, **add subscriber X** where **X** is the extension, and press **Enter**. Enter the following: **Name**, **Password,** (the Password will disappear after being entered). Click **Enter** to continue. 

2.9. Verification Steps

Use the following steps to verify the configuration on the Avaya CM:

Step	Description
2.9.1.	From the Avaya Communication Manager SAT, use the **status trunk n** command where **n** is the trunk group number to verify that the SIP trunk group is in service.
2.9.2.	From the Avaya Communication Manager SAT, use the **status signaling-group n** command where **n** is the signaling group number to verify that the SIP signaling group is in service.

3. Configure Avaya SIP Enablement Services

This section describes the steps required for Avaya SIP Enablement Services to support the configuration in **Figure 3: Interoperability Configuration Diagram.** . The following pages provide step-by-step instructions on how to create the media server entry, define the host address map entry along with contact information for the Tango Abrazo-E.

Note: It is assumed that that the appropriate license and authentication files have been installed on the servers and that login and password credentials are available. It is assumed that the reader has a basic understanding of the administration of Avaya SIP Enablement Services and has access to the SES web browser.

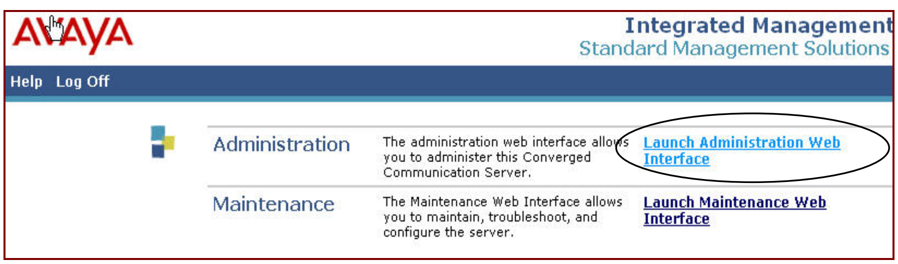
3.1. SES Software Configuration

The Abrazo solution needs the ability to trigger originating and terminating services. Avaya has provided a way to do this using the standard SIP P-Asserted-Identity header. If the header is detected in the Invite requests, the SES marks the transaction for origination services, otherwise it assumes termination services. For this functionality to be enabled on Avaya, do the following steps.

Step	Description
3.1.1.	Log into the SES as root.
3.1.2.	Edit the file: /usr/impress/sip-server/etc/ccs.conf
3.1.3.	Under the Proxy section remove the # before the line: EnableThirdPartyOriginatingProcessing=true
3.1.4.	Stop the server by entering: stop –a
3.1.5.	Verify all components are shutdown by entering: statapp
3.1.6.	Once all are shutdown restart the server by entering: start –a

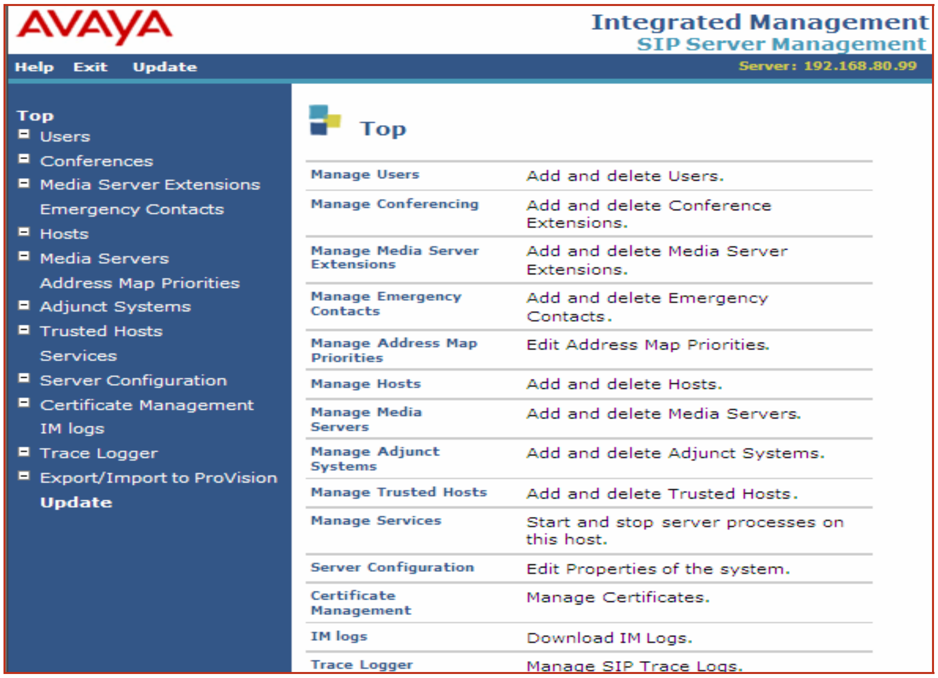
3.2. SIP Trunk Configuration

On the SES, the Abrazo solution needs to be configured using both a SIP trunk and a SIP line. The SIP trunk interface(s) are used by the Abrazo solution to terminate a call to the wireless operator's network. A SIP trunk is also used by the PBX to route mobile calls to the Abrazo solution via the enterprise using Pilot Directory Numbers or Service Pool Numbers.

Step	Description
3.2.1.	Access the SES administration web interface by using the URL **HTTP://ip-address/ADMIN** in an Internet browser window, where **ip-address** is the IP address of the SES server. Log in with the appropriate credentials. The first screen of the interface is displayed. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.2.2.</td><td>

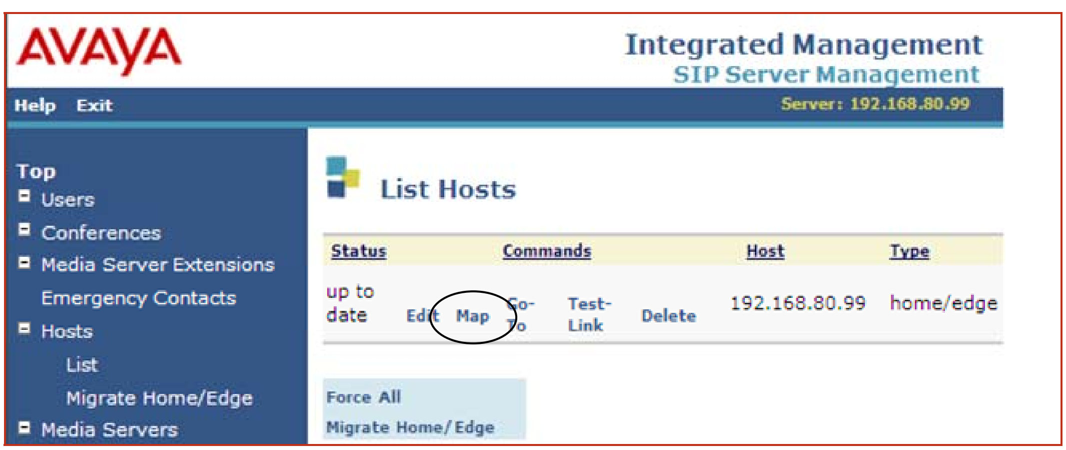
Select **Launch Administration Web Interface**. The following screen is displayed.



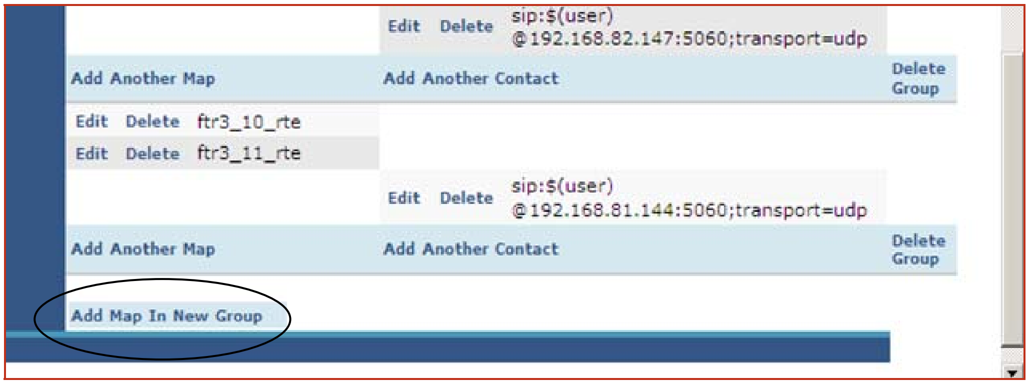
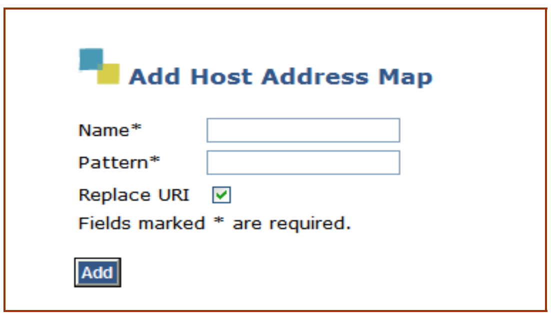
</td></tr>
<tr><td>3.2.3.</td><td>

Outbound calls are first routed by Avaya Communication Manager to the SIP trunk group. These calls are then subject to further routing decisions determined by Host Address Maps in the Avaya SES.

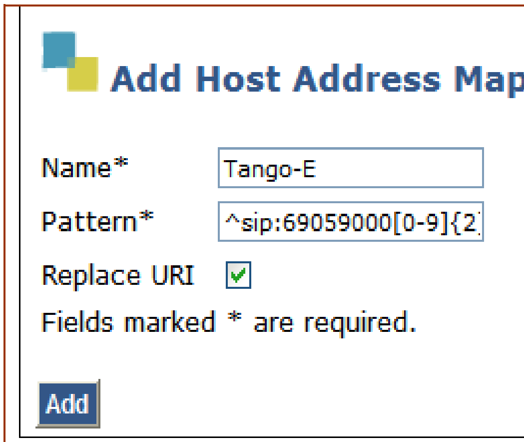
Navigate to the **Add Host Address Map** screen by selecting **Hosts > List** from the left pane. The **List Hosts** screen is displayed. Click on **Map** in the right pane.



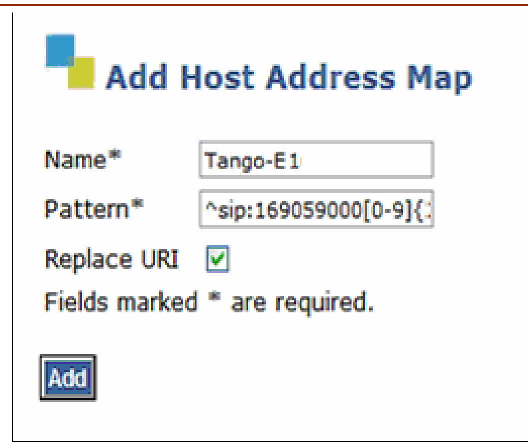
</td></tr>
</table>

Step	Description
3.2.4.	The **List Host Address Map** screen is displayed. Scroll to the bottom of the screen if needed. Click on **Add Map in New Group**. 
3.2.5.	The **Add Host Address Map** screen is displayed. 

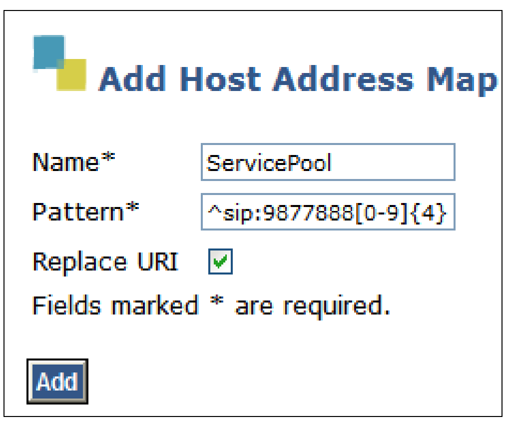
Step	Description
3.2.6.	Use the **Add Host Address Map** screen to create host address map patterns on the SES to specify what calls should be routed to the Abrazo-E. • For the **Name** field, enter a descriptive name to denote the routing pattern. • For the **Pattern** field, define an appropriate syntax for address mapping that matches the format of the PDN/Service Pilot Pool Numbers that are used to route mobile calls into the Abrazo-E. • Retain the check in **Replace URI**, and click **Add**. The Screens below illustrate National, International, and Service Pool address maps.



National PDN Host Address Map Pattern



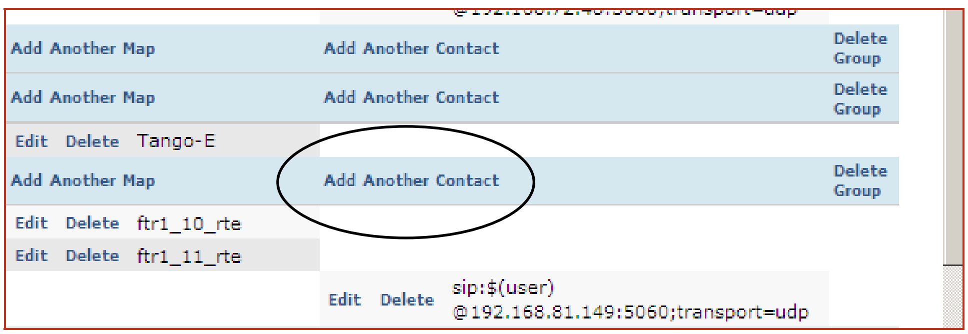
International PDN Host Address Map Pattern



Service Pool Host Address Map Pattern

Note: Multiple patterns should be defined to ensure that both national and international PDN numbers route to the Abrazo solution through the PBX.

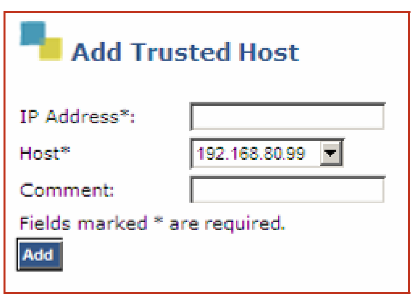
In the PDN examples above, the PDN is 69059000 followed by any two{2} digits[0-9].

Step	Description
3.2.7.	A **Continue** screen is displayed to confirm the addition. 
3.2.8.	Host address map patterns must be defined for Pilot Directory Number (PDNs) and Service Pilot Pool numbers used by the Abrazo solution. Repeat steps above as required.
3.2.9.	Click the **Continue** button. The **List Host Address** screen is redisplayed, showing the newly added item. Define the contact address for the Abrazo-E (Tango-E) by clicking on **Add Another Contact** on the line below Tango-E. 
3.2.10.	The **Add Host Contact** screen is displayed. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.2.11.</td><td>The Contact field specifies the destination for the call. Populate the Contact field with the service IP address that the SES should substitute into the required URI before sending a message to the Abrazo Tango-E. The Avaya SES replaces $(user) with the user portion of the request URI before sending the message. Click the Add button.

</td></tr>
<tr><td>3.2.12.</td><td>A Continue screen is displayed to confirm the addition. Click the Continue button.

</td></tr>
<tr><td>3.2.13.</td><td>A Host Contact must be defined for each of the Host Address Maps provisioned for the Abrazo PDNs and Service Pool numbers. Repeat steps above as required.</td></tr>
<tr><td>3.2.14.</td><td>Administer the Abrazo-E as a trusted host so that the SES will not challenge SIP messages from the Abrazo-E. Select Trusted Hosts → Add Trusted Host. The Add Trusted Host screen is displayed. Enter the IP address of the Abrazo-E and, if desired, a descriptive Comment. Click Add to continue.



</td></tr>
</table>

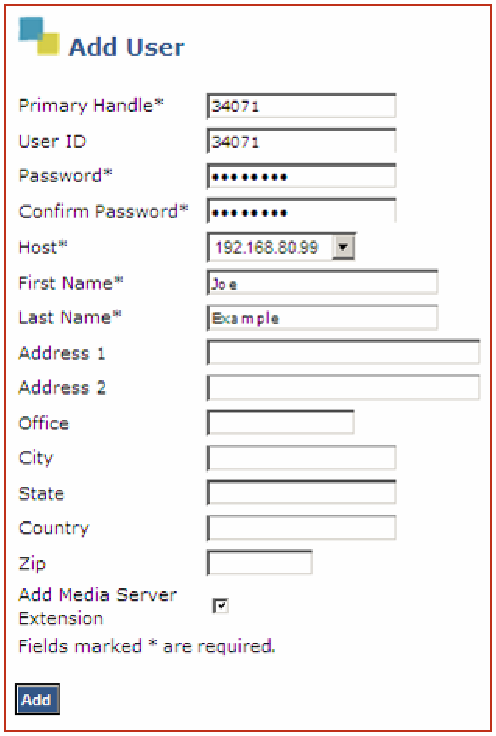
Step	Description
3.2.15.	A **Continue** screen is displayed to confirm the addition. Click the **Continue** button 
3.2.16.	To apply the changes in the above steps, click **Update** at the bottom of the left pane. This link appears on the current page whenever updates are outstanding, and can be used at any time to save the administrative changes performed to that point.

3.3. SIP Line Configuration

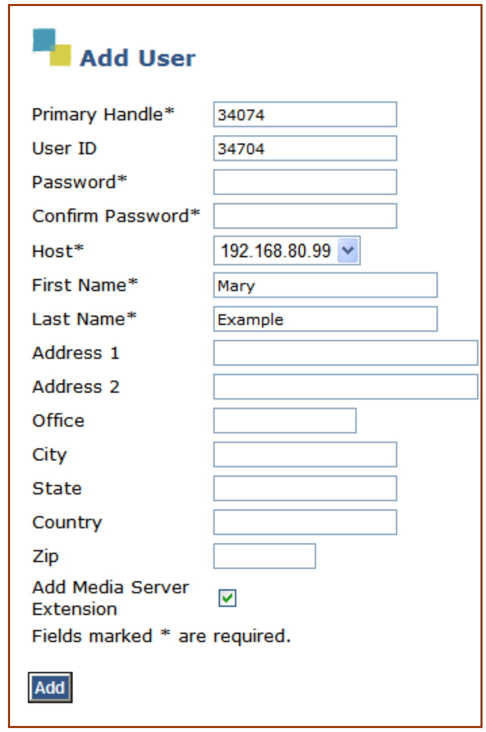
On the SES, the Abrazo solution needs to be configured using both a SIP trunk and a SIP line. The SIP line interface(s) are used to offer originating PBX services to the mobile user. The configuration of the Avaya desk phone dictates how the Abrazo user is added to the SES.

- If the user has an H.323 or digital desk phone, then an SES ID must be created with the same extension that was created for the desk station.
- If the user has a SIP desk phone, he or she will need a new ID created. (Since the desk phone is SIP, the user already has an SES ID created.)

Step	Description
3.3.1	Select **Users** → **Add**. Fill in the screens as follows depending on the user's type of desk phone.



Adding Joe (H.323)



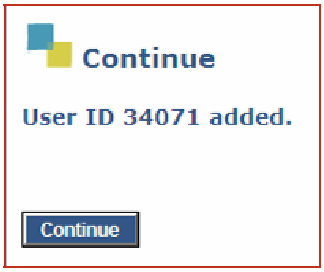
Adding Mary (SIP)

Ensure the following fields are populated as described below:

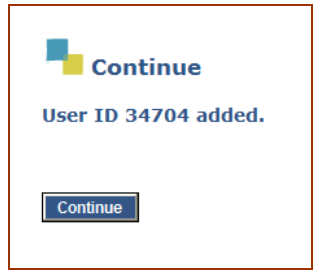
- For the **Primary handle** field, enter the phone number (OPS station number) of the Abrazo subscriber (example: 34071).
- For the **User ID** field of a H.323 or digital user, enter the user's extension\station number. For a SIP user, create a new SIP number. The **UserID** field must match the Abrazo SIP number provisioned for the subscriber.
- For the **Password** field, set the password to be used by the Abrazo solution during registration with the SES. The **Password** field is a required field for SIP users that are not configured as trusted nodes. For trusted nodes, like the Tango Abrazo, the password should be entered; however, it is not included in the SIP registration message.
- For the **Host** field, enter the IP address of the Avaya SES with which Abrazo will register.
- Click the **Add Media Server Extension** check box.

Step	Description

Click **Add**. A confirmation screen is displayed.

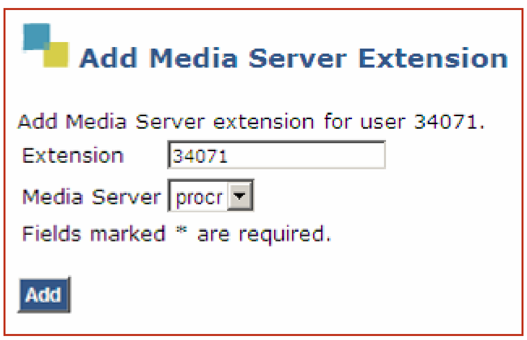


Confirming the Addition of Joe



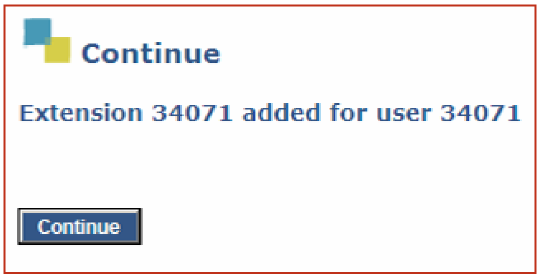
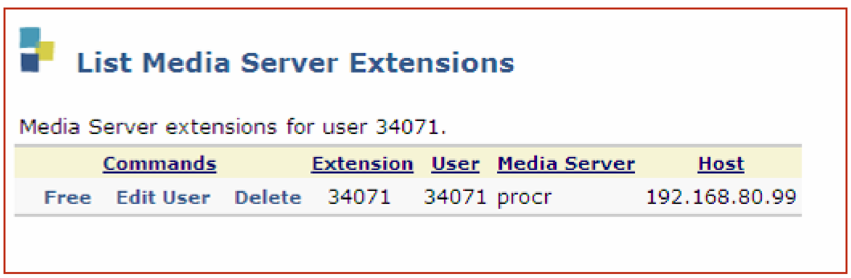
Confirming the Addition of Mary

Click **Continue**. The **Add Media Server Extension** screen is displayed.



Adding Media Server Extension for Joe

Adding Media Server Extension for Mary

Step	Description
3.3.2	Use the **Add Media Server Extension** screen to set the corresponding telephone extension. For the **Extension** field, enter the extension of corresponding OPS station (same one used for primary handle when adding user). For the **Media Server** field, select the media server on which the desk phone is configured. The SES should automatically populate or default to this field. Click **Add**. A confirmation screen is displayed.  Click **Continue**. A list of media server extensions for that user is displayed. 
3.3.3	To apply the changes in the above steps, click **Update** at the bottom of the left pane. This link appears on the current page whenever updates are outstanding, and can be used at any time to save the administrative changes performed to that point.

3.4. Verification Steps

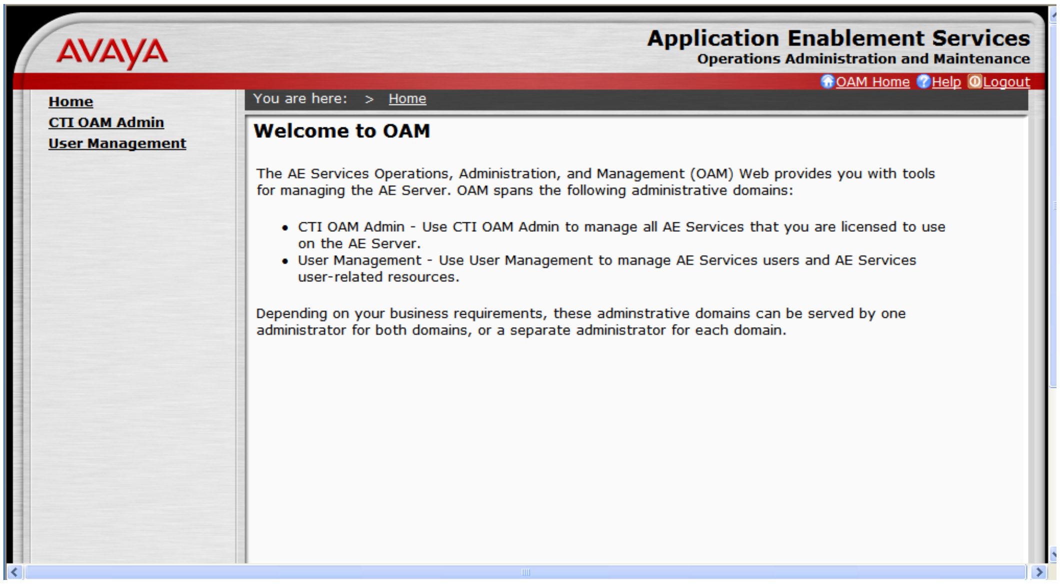
Use the following steps to verify the configuration on the Avaya SES:

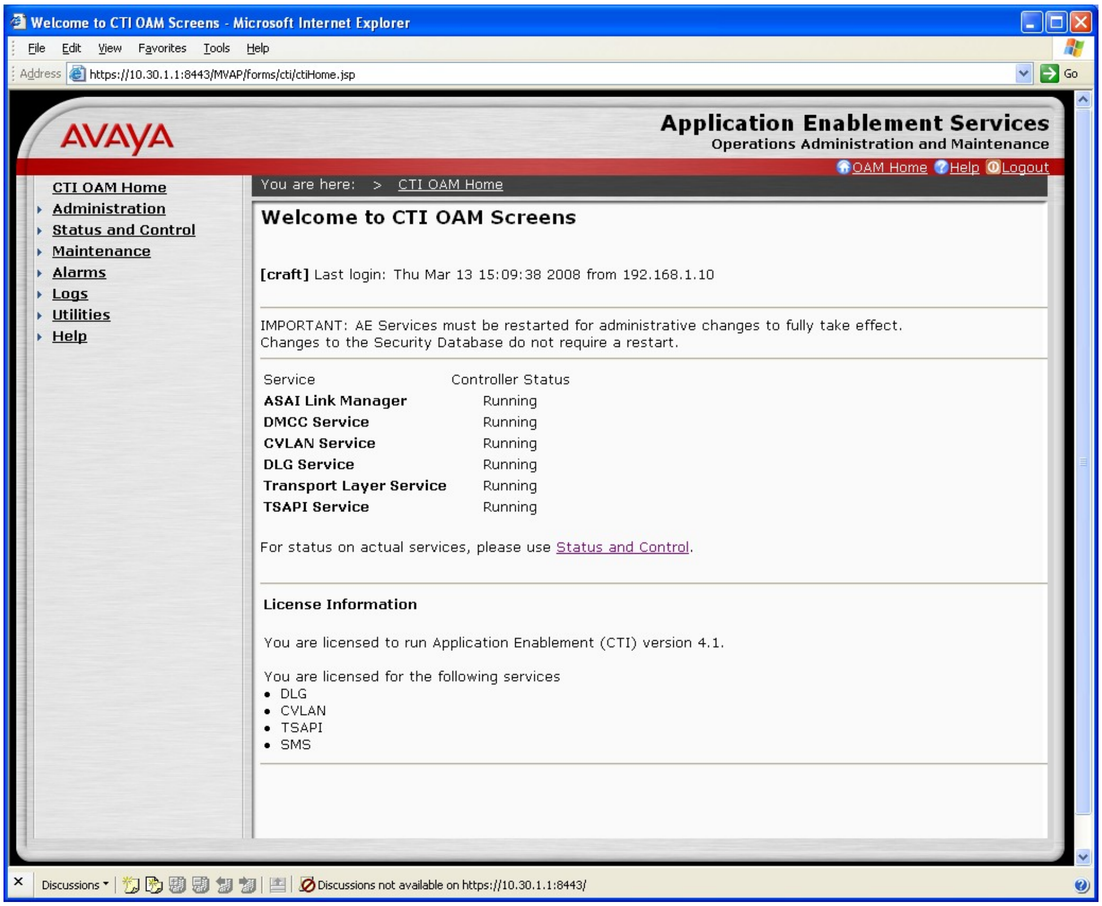
Step	Description
3.4.1	From the Avaya SIP Server Management interface, select **Trusted Host→ List** to verify that the IP address of the Abrazo Tango-E is listed as a trusted node.
3.4.2	From the Avaya SIP Server Management interface, navigate to **User → Registered Users** to verify that all SIP endpoints are registered with their respective Avaya SES.

4. Configure Avaya Application Enablement Server

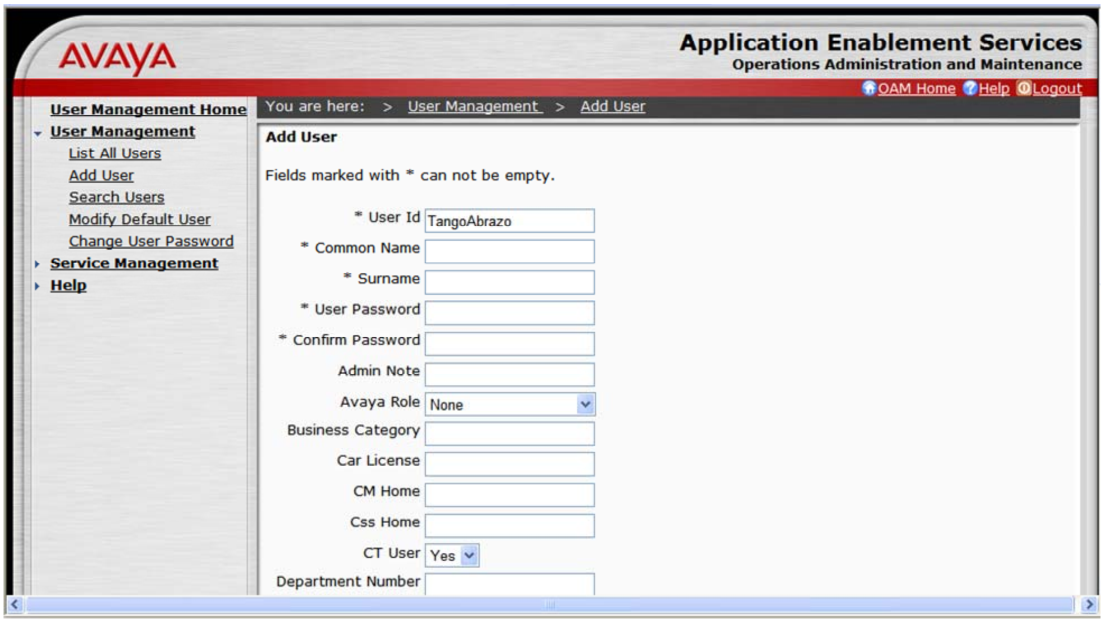
4.1. Configure CTI Interface

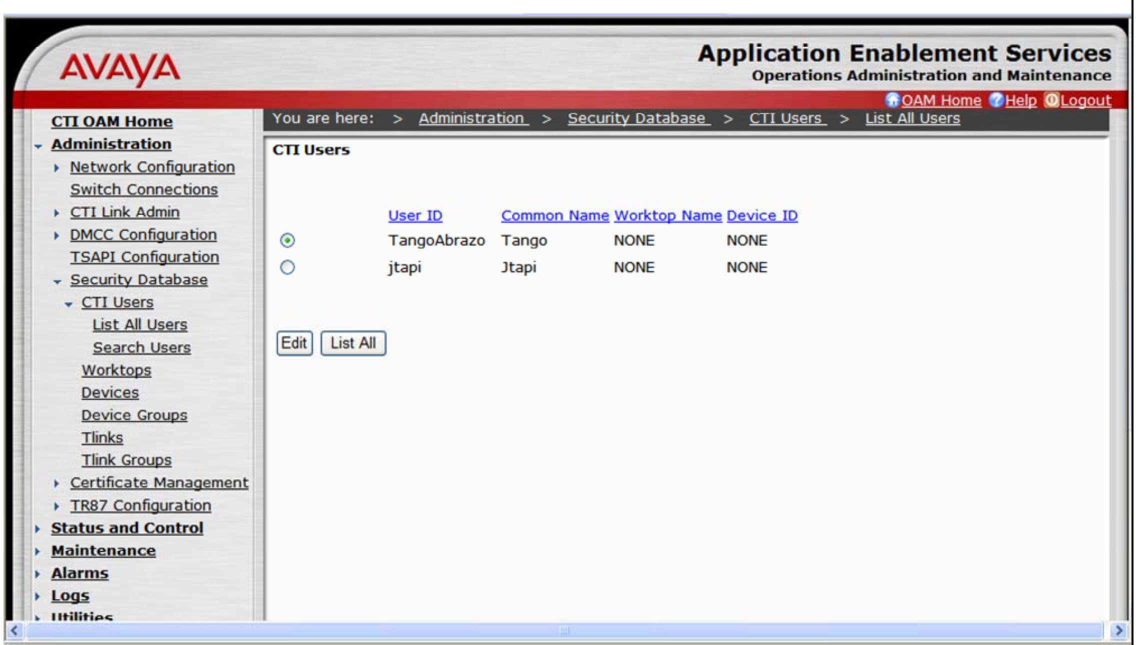
This section describes the steps for setting the CTI interface in the Avaya Application Enablement Server for enabling the Abrazo Call Move service.

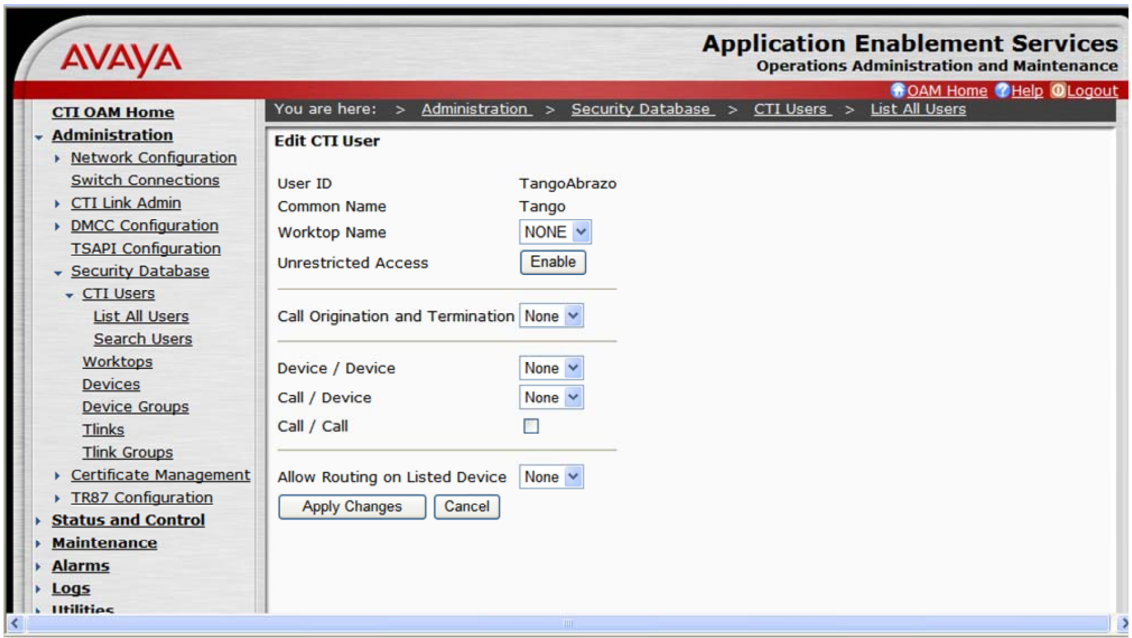
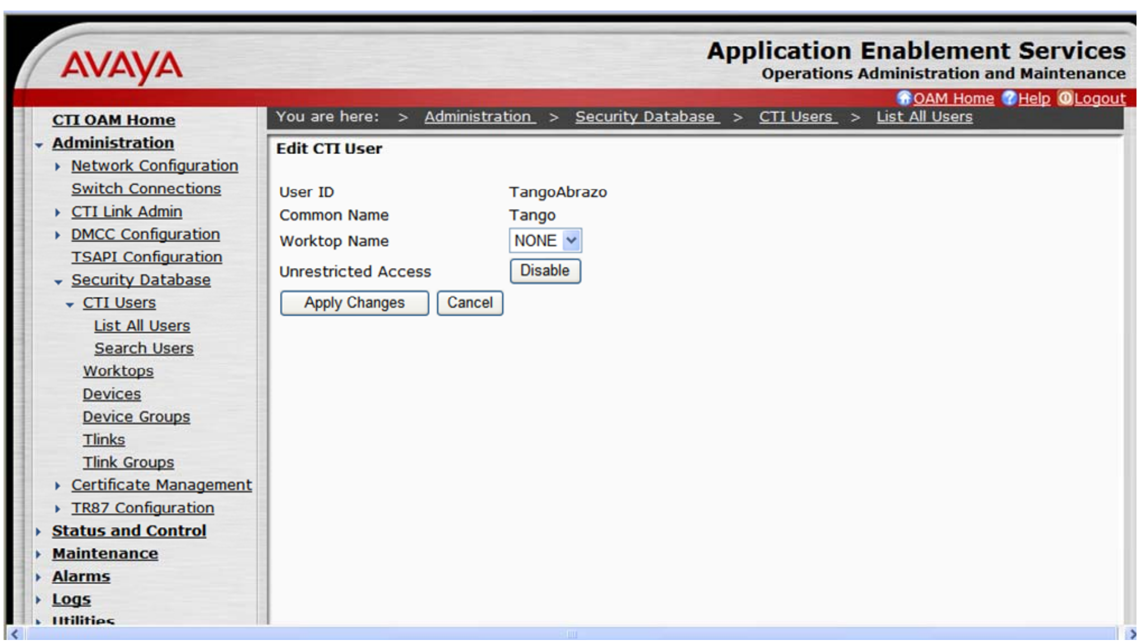
Step	Description
4.1.1.	Access the AES administration web interface by using the URL HTTP://ip-address/ in an Internet browser window, where **ip-address** is the IP address of the AES server. Log in with the appropriate credentials. The first screen of the interface is displayed. 

Step	Description
4.1.2.	Verify the Avaya Application Enablement Services license. From the left panel, select **CTI OAM Admin**. The **License Information** must be visible as displayed in the **Welcome to CTI OAM Screens** below. Verify that the Avaya Application Enablement Services license has proper permissions for the features illustrated in these Application Notes by ensuring the TSAPI service is licensed. If the TSAPI service is not licensed, then contact the Avaya sales team or business partner for a proper license file 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.1.3</td><td>Click on the User Management link and Log in with the appropriate credentials. Then, click on the Add User link to create a generic user account for the Tango Abrazo solution. Ensure that the CT user field is set to ‘yes’ for this account.


</td></tr>
<tr><td>4.1.4</td><td>Click on the CTI OAM Home link and Log in with the appropriate credentials. Select the Administration link. Under Administration link, select Security Database The following screen is displayed. Select the List All Users link and the CTI userid (e.g., TangoAbrazo) created in the step above should be displayed..


</td></tr>
</table>

Step	Description
4.1.5	Select the **Edit** button for the CTI Userid (e.g., TangoAbrazo) and the following screen is displayed. Enable unrestricted access for this CTI userid by selecting the **Enable** button and apply the changes by selecting the **apply changes** button. 
4.1.6	Once changes are applied, the CTI User ID should be displayed. This change allows the Tango Abrazo to perform third party call control for all Abrazo subscribers on Avaya Communication Manager. 

4.2. Verification Steps

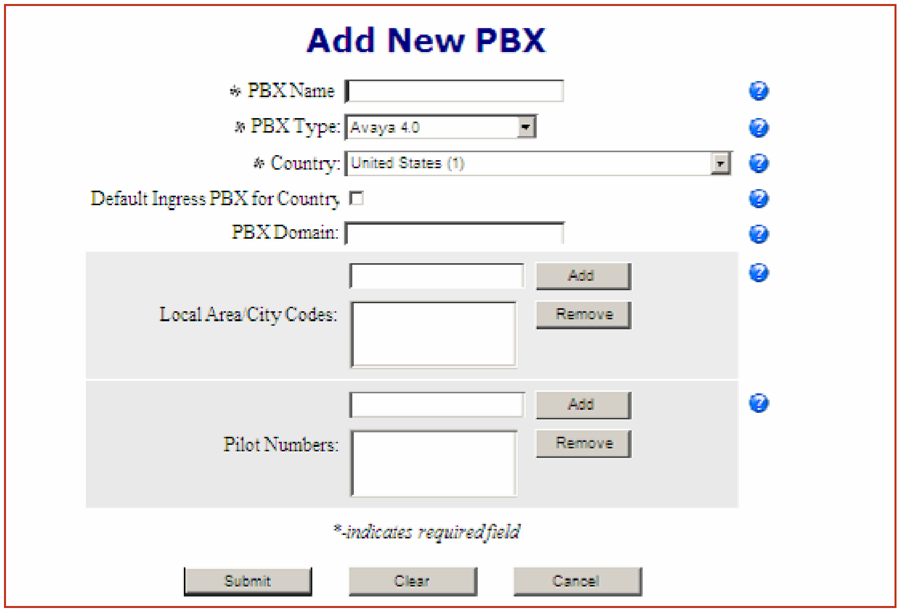
Use the following steps to verify the configuration on the Avaya AES:

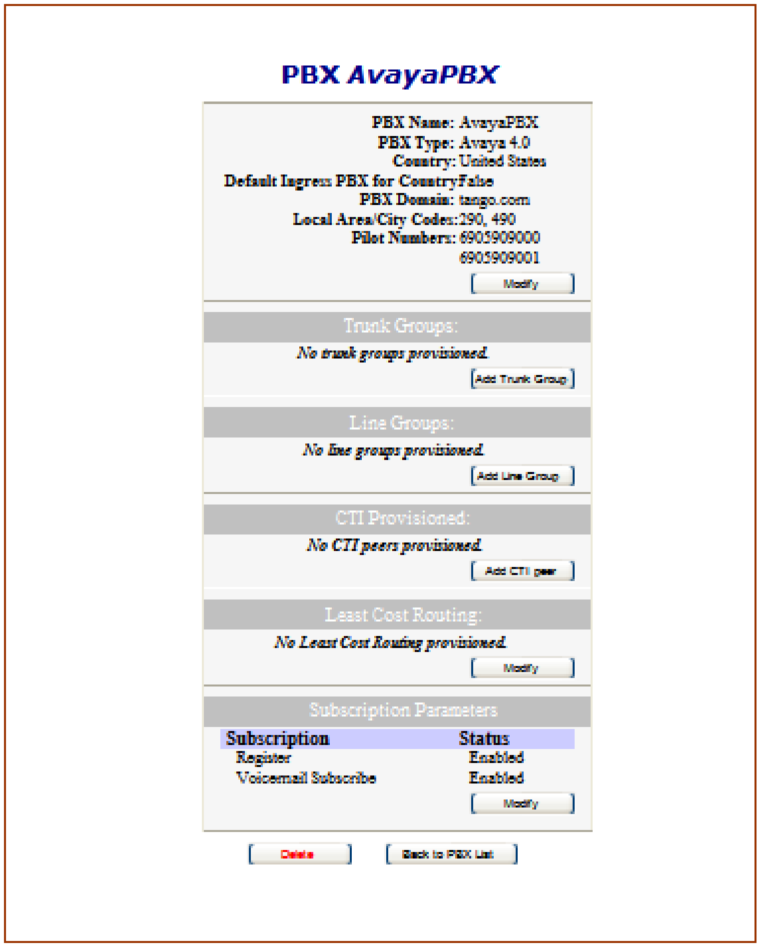
Step	Description
4.2.1.	From the Avaya AES Server Management interface, navigate to **CTI OAM Home -> Administration -> CTI Users** to verify that third party call control is enabled.

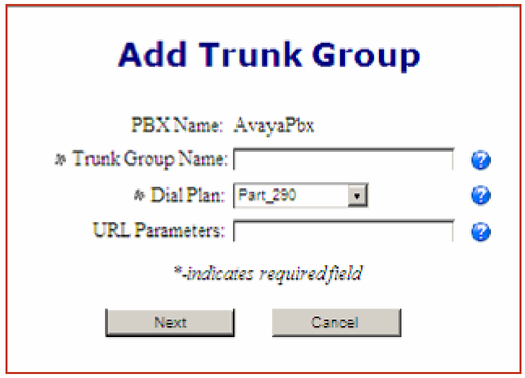
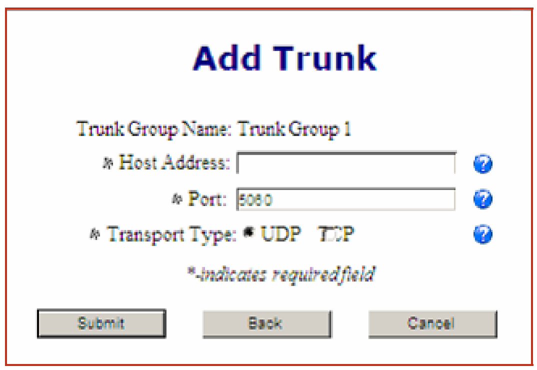
5. Provision the Tango Networks Abrazo-E

This section describes the processes required to integrate the Abrazo solution with a SIP-enabled Avaya Communication Manager. This document assumes that the Abrazo solution has already been provisioned with enterprise information and wireless carrier information. The integration process includes the following steps:

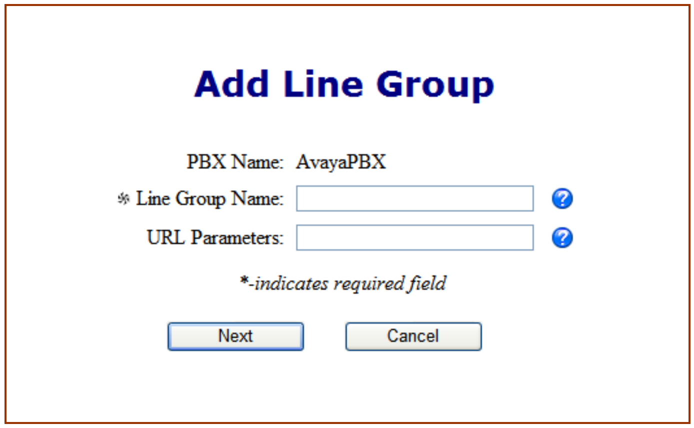
- Provision Avaya Communication Manager in the Abrazo system.
- Create an Avaya outbound SIP trunk to route traffic from the Avaya telephony infrastructure to the Abrazo system.
- Create inbound SIP lines to route traffic from the Abrazo system to the Avaya telephony infrastructure.
- Create the web services connection to enable Call Move service.
- Provision the service pool numbers used for the Call Move service.
- Provision the dial plans for the Avaya telephony infrastructure in the Abrazo system.
- Provision the voicemail system used with the Avaya telephony infrastructure.
- Provision Abrazo subscribers that use the Avaya telephony infrastructure.

Step	Description
5.1.1.	To add the Avaya telephony infrastructure to the Abrazo-E, select **Voice Network → PBX → Add**. Enter a **PBXName** (e.g., "AvayaPBX"). Select **Submit** to continue. 

Step	Description
5.1.2	Define a new trunk group and add trunk group members to communicate with the Avaya telephony infrastructure. To define a new trunk group, select the PBX to which the Trunk Group is to be added from the list displayed by **Voice Network→ PBX→ List all**, as shown below. Select the PBX you want from the list. Click the **Add Trunk Group** button. 

Step	Description
5.1.3	The Add Trunk Group screen is displayed. Select **Next** to continue.  Enter trunk group information on the Add Trunk Group screen. Click on **Next** to add trunk group members, clicking on **Submit**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.1.4</td><td>
Define a new line group and add line group members to communicate with the Avaya telephony infrastructure. To define a new line group, select the PBX to which you want to add the Trunk Group (Voice Network→ PBX→ List all). Select the PBX you want from the list. Click the Add Line Group button. You’ll see the screen shown below.



Enter line group information on the Add Line Group screen. Select Next to add line group members. You’ll see the screen shown below.

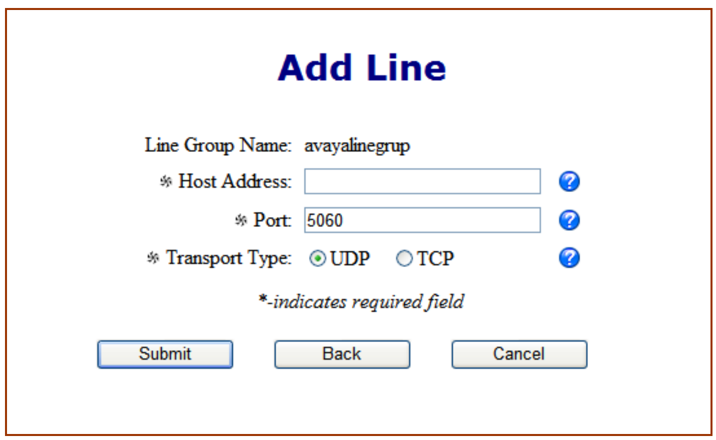
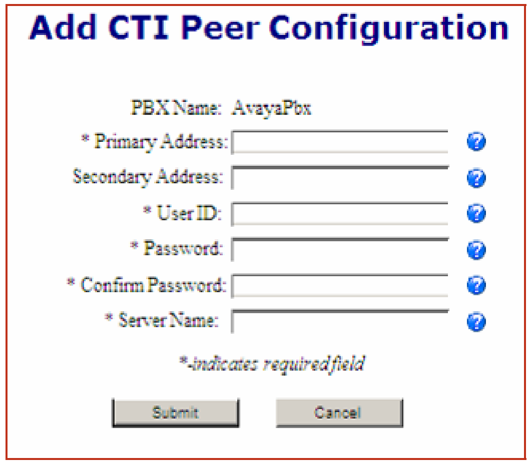


Figure 4 Add Line Screen

Enter line information on the Add Line screen. Select Submit to add the line members.
</td></tr>
</table>

Step	Description
5.1.5.	Select **Add CTI peer** on the Selected PBX Screen shown in Step 5.1.2 to create the web services connection.  Ensure that the **Server Name** field used for CTI integration is the same as the server connection name, which is already defined in the Application Enablement Server. The server connection name can be verified by logging into the AES server, and performing one of the following steps: 1. Select **CTI OAM->Home Administration** and view the CTI connections for the switch connections. 2. Select **CTI OAM->Home Administration**, click on the **Utilities** link, and click on **TSAPI Test**. The server connection name is part of the TLink name.

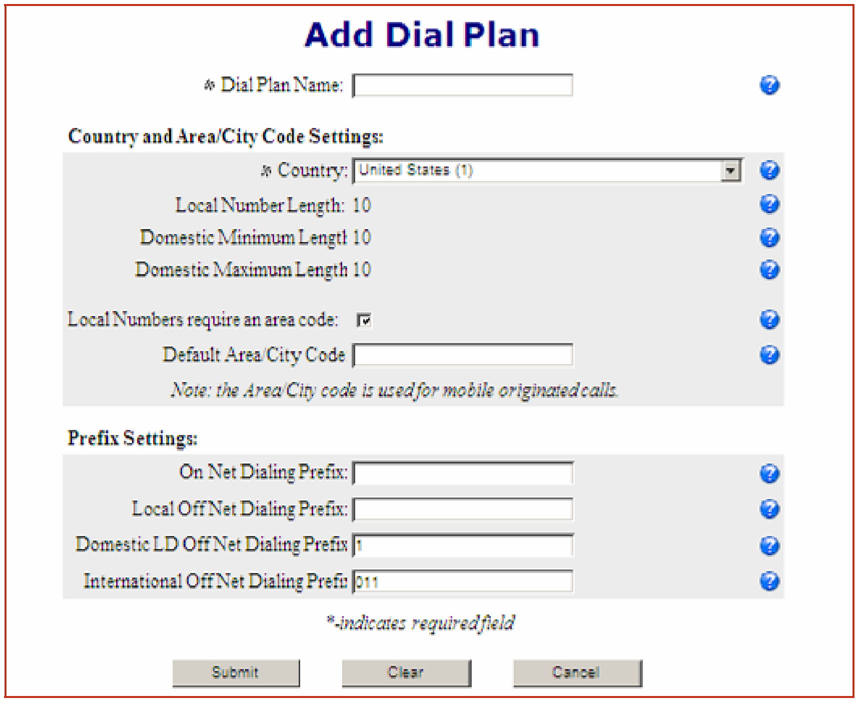
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.1.6</td><td>

Define the service pool numbers that will be used for the Call Move service. To add service pool numbers, select **Voice Network → Service Pool → Add**. To save, click on **Submit**.

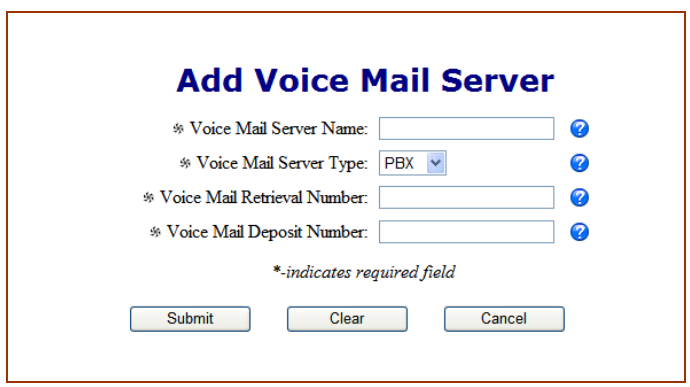
The actual service pilot number can be any format; however it must be prefixed with the feature access code for public translations configured on Avaya Communication Manager. In the example above, the number 9 is the feature access code.

</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.1.7</td><td>Update the voice network topic for the newly configured Avaya telephony infrastructure with the enterprise’s voice network layout. At a minimum, add the dial plan information to enable abbreviated dialing support as well as forced on-net services. To add dial plan information, select Voice Network → Dial Plan → Add. Select Submit to continue.



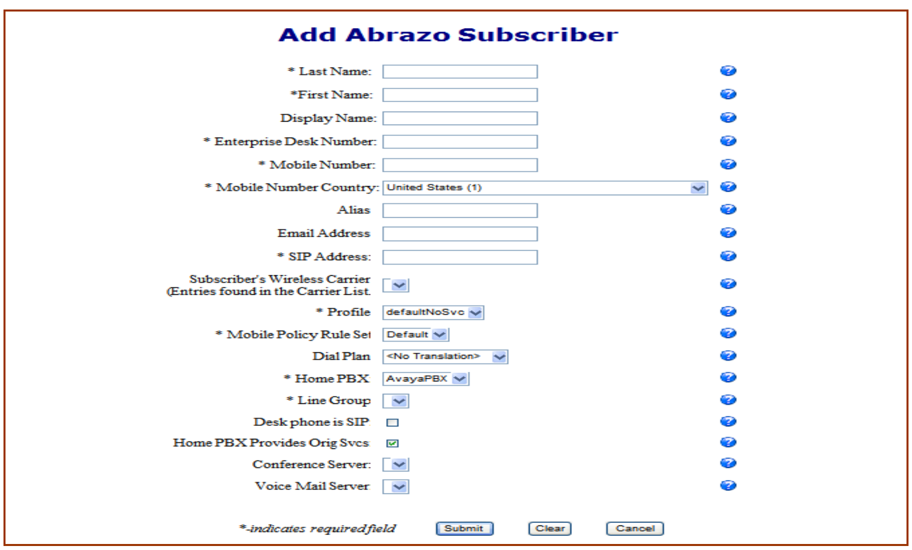
</td></tr>
<tr><td>5.1.8</td><td>The voice mail server used with the Avaya telephony infrastructure must be specified so the Abrazo solution can provide a single voice mail solution. To add a Voice Mail Server, select Voice Network -> Voice mail -> Add. Select PBX as the Voice Mail Server Type. The following screen is displayed.



For Voice Mail Retrieval Number, enter the number that routes callers to their voicemail. For Voice Mail Deposit Number, enter the feature code defined on the PBX to transfer the call to voice mail.</td></tr>
</table>

Step	Description
5.1.9.	Add or modify Abrazo subscribers to associate them with the Avaya telephony infrastructure. To add subscribers, select **Subscriber→ Add**. The following screen is displayed.

Select AvayaPBX as the user's **HomePBX** field. The screen is modified to display additional fields (**Line Group**, **Desk phone is SIP**) as shown below. Select **Submi**t to continue.



The Avaya user's desk phone could be H.323, digital, or SIP. The Abrazo subscriber data should be added based on the following guidelines which depend on the protocol of the user's desk phone.

<u>If the user's desk phone is H.323 or digital</u>

If the user's desk phone is H.323 or digital, the SIP user added to the SES must match the extension set up for the station. This allows the Abrazo solution to register using the user ID as the station. When these IDs are the same, the Abrazo solution does not need to subscribe for voice mail events. The Avaya will send notifications based on the registration matching the station extension.

Step	Description
5.1.10	Repeat step 8 for each user to be added to the system.

6. Interoperability Compliance Testing

Testing was conducted via the *DevConnect* Program at the Avaya Solution and Interoperability Test Lab. Compliance testing verified the integration between Avaya Communication Manager 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.

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** - Avaya 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.
- **Call Coverage** - automatically sequentially reroutes incoming calls to up to six alternate telephone numbers when the called party is unavailable. The subscriber's mobile will not ring in this scenario because Abrazo will not be involved in the call termination.
- **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.

- **Call Forward Busy** - allows users to forward calls to another destination when their device is busy. Users activate or deactivate the Call Forward Busy capability with a feature access code or a Call Forward Busy feature button. When an Abrazo subscriber's desk phone is busy, calls will be forwarded. When the forwarded to number is an Abrazo subscriber, intelligent call delivery ensures that both the desk phone and mobile phone ring.
- **Call Forward No Answer** - allows users to forward calls to another destination when their device is not answered. Users activate or deactivate the Call Forward No Answer capability with a feature access code or a Call Forward No Answer feature button. When an Abrazo subscriber's desk phone is not answered, calls will be forwarded. When the forwarded to number is an Abrazo subscriber, intelligent call delivery ensures that both the desk phone and mobile phone ring.
- **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.
- **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.
- **Call Move** - (not supported on Avaya SIP phones) allows an Abrazo subscriber to move a phone call between the desk phone and the mobile phone by entering a feature code such as **72. The automatic bridged line appearance feature interacts with the call move service for subscribers using H.323 desk phones. When a voice call is established on the desk phone and the subscriber invokes the call move service in order to move the call from the desk phone to the mobile phone, 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.
- **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.
- **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.
- **Flexible Dialing Support** - Abrazo has a flexible dialing plan enabling PBX services to be provided to mobile users.
- **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 Avaya 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.

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 Avaya Communication Manager with all services tested.

7. 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: +1 972-301-9316

8. 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 Communication Manager with the wireless connectivity to the GSM network. The functionality of the Avaya/ Tango Networks Abrazo Solution was validated via the *DeveloperConnection* Program at the Avaya Solution and Interoperability Test Lab. All feature functionality test cases passed.

9. Additional References

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

[1] *Administrator Guide for Avaya Communication Manager*, February 2007, Issue 3.1, Document Number 03-300509
[2] *Installing and Administering SIP Enablement Services*, March 2007, Issue 2.1, Document Number 03-600768
[3] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide*
[4] *Messaging Application Server (MAS) Administration Guide Release 3.1*, February 2007

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.