Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Tango Networks Abrazo Fixed-Mobile Convergence Solution with an Avaya Telephony Infrastructure and WiFi / VCC Support - Issue 1.0

Abstract

These Application Notes describe a compliance-tested configuration comprised of the Tango Networks Abrazo Fixed-Mobile Convergence (FMC) Solution connected to an Avaya telephony infrastructure. The Abrazo solution using dual mode phones extends enterprise PBX functionality to mobile devices and allows roaming seamlessly to and from WiFi to mobile networks., 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 FMC Solution connected to an Avaya telephony infrastructure, including Avaya Communication Manager and Avaya SIP Enablement Services.

Tango Networks' Abrazo fixed mobile convergence (FMC) solution employs components in both the enterprise network and the mobile operator network in order to extend corporate PBX features to the users mobile phone. It can also seamlessly handoffs voice calls to and from Wi-Fi to wireless utilizing Voice Call Continuity (VCC). This convergence allows mobile phones to offer the same productivity features as a conventional enterprise desk phone.

1.1. Background

The Tango Networks Abrazo FMC 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, WiFi network and corporate databases via standard interfaces to extend the enterprise network functionality transparently to the mobile network. Using a dual-mode device, calls can be connected via the wireless network within the enterprise. The Abrazo provides the ability to seamlessly move calls from the WiFi network to the mobile network and vice-versa.

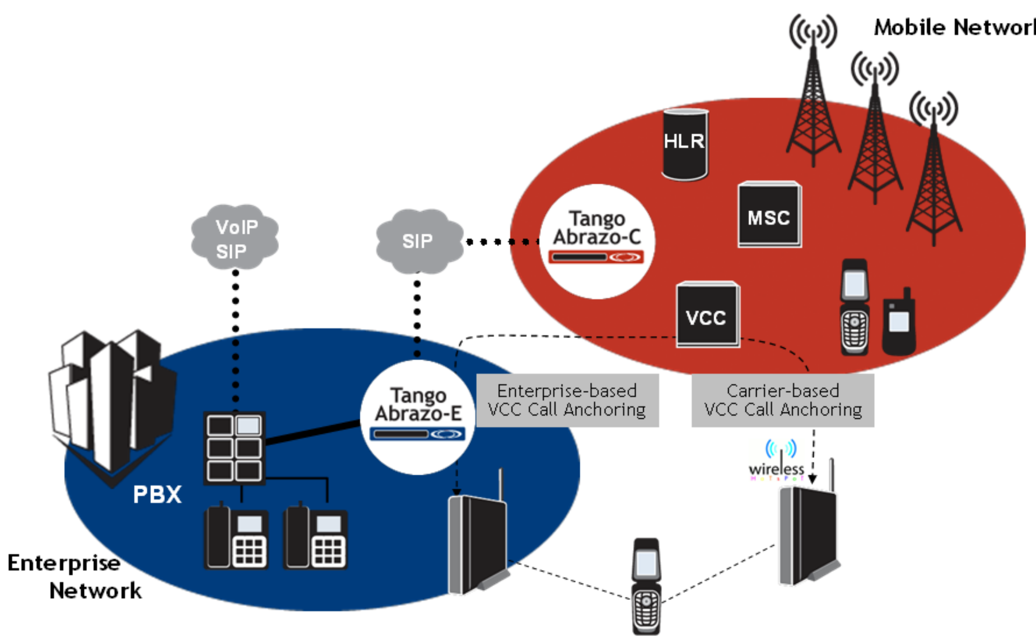


Figure 1: Tango Networks' Architecture Diagram with WiFi/VCC

The Abrazo solution interacts with the Avaya Communication Manager (ACM) via Avaya SIP Enablement Services (SES) **Figure 2**.

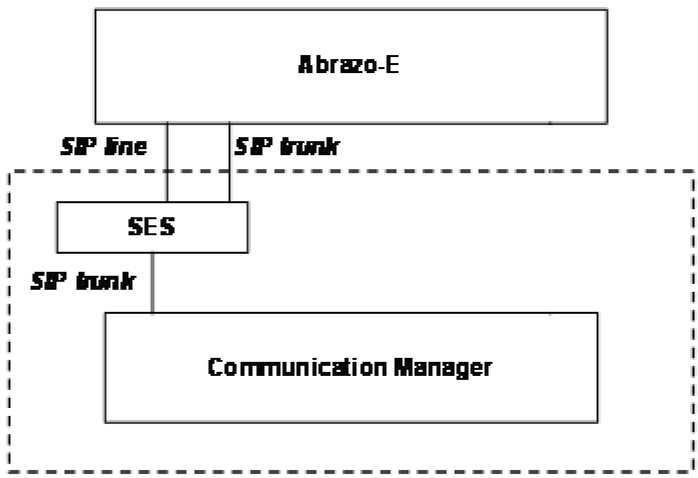


Figure 2: Abrazo-E Interfaces to the Avaya Telephony Infrastructure

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 does not require Voice over Wi-Fi (VoWi-Fi) in order to extend FMC features and functionality to users. However, if an enterprise wants to deploy VoWi-Fi as part of their enterprise communications strategy, the Abrazo seamlessly supports it. Voice Call Continuity (VCC) is a standard currently under development in the 3GPP R7 standards body. It extends IMS network architecture to support both Wireless IP (e.g., Wi-Fi) and cellular coverage and addresses seamless handoff between these environments. Tango takes this basic technology and applies the concepts to support its FMC product.

The Abrazo supports the SIP clients running Wi-Fi on dual-mode mobile phones. It authenticates the clients and proxies the client to the right PBX (i.e., Wi-Fi in multi-vendor PBX environment supported).

Due to its hybrid architecture, Tango offers the ability to have service continuity between cellular and Wi-Fi environments for enterprises choosing VoWi-Fi. The Abrazo-E supports Domain Transfer Functions (DTF). DTF controls the handoffs between the cellular and Wi-Fi networks. The Abrazo's hybrid architecture supports seamless, automatic handoffs from Wi-Fi to wireless as well as wireless to Wi-Fi.

Whether the mobile operator or the enterprise hosts the Wi-Fi, the functionality remains the same. When the call is on the enterprise's Wi-Fi, the enterprise controls the call. When the call is on the mobile network, the operator controls the calls.

While using the Abrazo, the employees are always provided PBX services on their phones no matter where the call starts or what handoffs occur. Abrazo users have access to the same desk features and midcall features whether their calls are on the Wi-Fi network or the wireless network. Features work the same way too. It doesn't matter if the calls are directed to the mobile number or the desk number; the Abrazo provides seamless Wi-Fi support.

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

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.

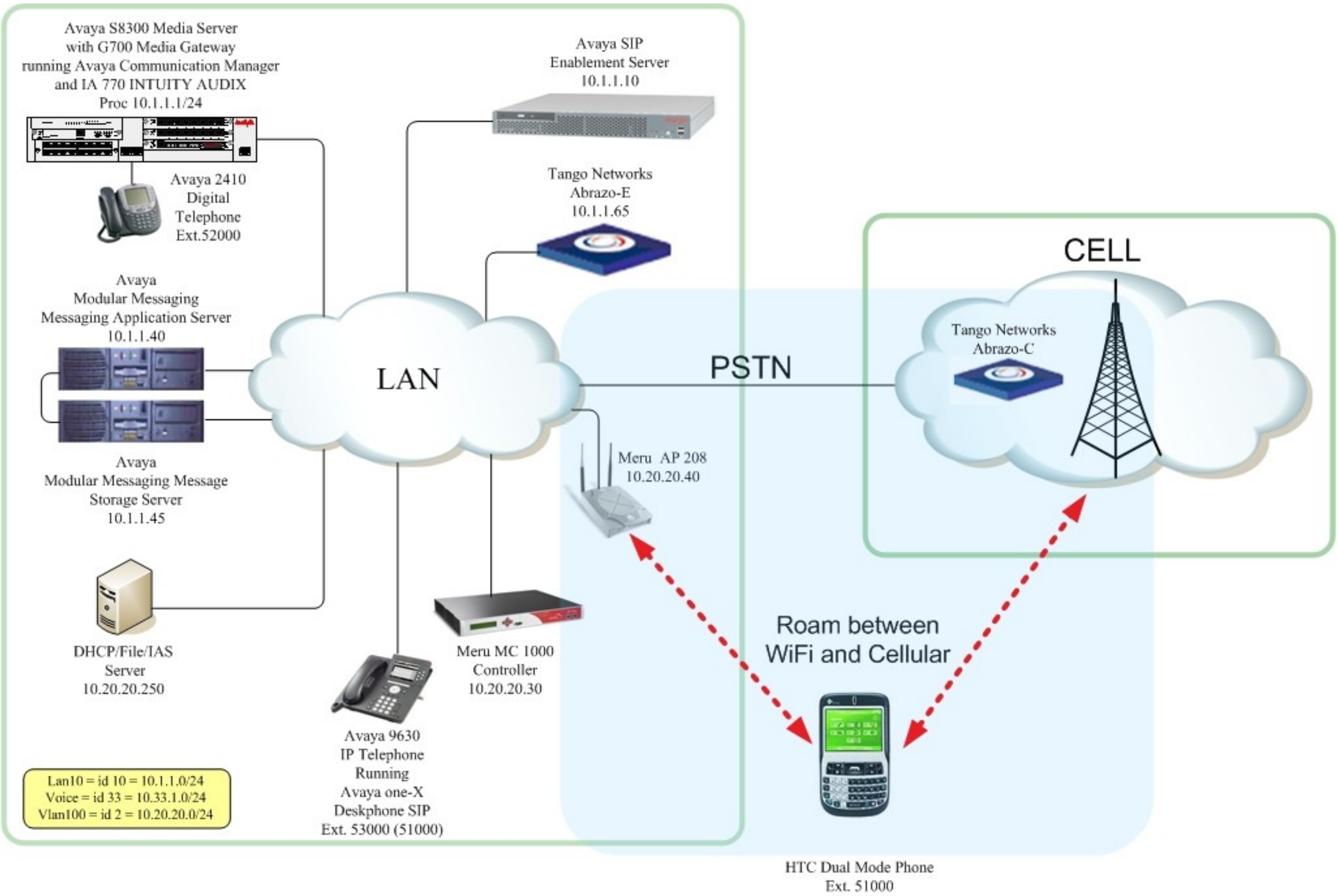


Figure 3: Interoperability Configuration Diagram

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 5.1 - R015x.01.0.414.3
Avaya G700 Media Gateway with MM712 DCP Media Module 8	26.31.0 FW 008
Avaya SIP Enabled Services (SES) Server	SES-5.1.0.0-414.3f
Avaya Application Enablement Services	4.2
Avaya Messaging (Voice Mai)l Products	
Avaya IA 770 INTUITY	5.1
Avaya Modular Messaging Server	4.0
Avaya Telephony Sets	
Avaya 9600 Series IP Telephones	Avaya one-X Deskphone SIP 2.0.5
Avaya 9600 Series IP Telephones	Avaya one-X Deskphone Edition 2.0
Avaya 1600 Series IP Telephones	1.0.3
Avaya 4600 Series IP Telephones	SIP (2.2) H.323 (2.9)
Avaya 2410 Digital Telephone	4.0
Tango Abrazo Products	
Tango Networks Abrazo-Enterprise Release	4.0
Tango Networks Abrazo-Carrier Release	4.0
Mobile Devices	
HTC	P3600i
HTC	Kaiser 100

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 dual-mode 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, however only dual-mode (WiFi and cellular) devices can take advantage of the Abrazo's WiFi / VCC capabilities.

2. Configure Avaya Communication Manager

Basic configuration of Avaya Communication Manager and Avaya SES 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 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.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">2.1.1.</td><td>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.</td></tr>
<tr><td><pre>
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
</pre></td></tr>
<tr><td rowspan="2">2.1.2.</td><td>On Page 2 verify that the Maximum Administered SIP trunks supported by the system is sufficient.</td></tr>
<tr><td><pre>
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.)
</pre></td></tr>
</table>

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.
	<pre>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:</pre>

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 2.2.1**.

```
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.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">2.5.1.</td><td>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".</td></tr>
<tr><td>

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

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: TO 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
```

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

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
```

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">2.5.3.</td><td>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”.</td></tr>
<tr><td><pre>
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
</pre></td></tr>
</table>

2.6. Dial Plan – AAR and Route Pattern

This section describes the steps for setting the Dialplan, AAR 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 732852 with a length of ten digits get routed to Tango.

Use the change inc-call-handling-trmt trunk-group g command, where “g” is the trunk assigned to the PSTN trunk. The example below specifies the trunk ID as 56.
<pre>
change inc-call-handling-trmt trunk-group 56

                Page   1 of   3
                         INCOMING CALL HANDLING TREATMENT
 Service/   Called   Called   Del Insert   Per Call Night
 Feature   Len   Number   CPN/BN   Serv
 tie   11   1732852   1   3
</pre>
Use the change aar analysis 0 command, and enter the dial string need to be matched.
<pre>
change aar analysis 0   Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all   Percent Full: 1

          Dialed   Total   Route   Call   Node ANI
          String   Min Max Pattern   Type   Num   Reqd

          732852   10   10   1   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 **7328522963**. The leading number **3** 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
```

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 **3** indicates that **AAR** 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:
```

2.6.4.	Create a route pattern that will send the piolet number to the SES SIP. Use the **change route-pattern *n*** command, where ***n*** is the number of an unused route pattern. Enter a descriptive name for the **Pattern Name** field. Set the **Grp No** field to the trunk group number created for the SES SIP trunk. Set the Facility Restriction Level (**FRL**) field to a level that allows access to this trunk for all users that require it. The value of ***0*** is the least restrictive level. The default values may be retained for all other fields.

```
change route-pattern 1                                            Page   1 of   3
                    Pattern Number: 1   Pattern Name: ToTango
                             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: 1    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
none
```

2.7. SIP Line Configuration

This section describes the steps for setting 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.	Using the **change system-parameters customer-options** command, validate that **OPS** has available licenses for the Abrazo solution because acts as a SIP telephone.
	
	`change system-parameters customer-options                        Page   1 of  11`
	`OPTIONAL FEATURES`
	`G3 Version: V15                         Software Package: Standard`
	`Location: 1                         RFA System ID (SID): 1`
	`Platform: 7                         RFA Module ID (MID): 1`
	`USED`
	`Platform Maximum Ports: 900   147`
	`Maximum Stations: 450   29`
	`Maximum XMOBILE Stations: 10    0`
	`Maximum Off-PBX Telephones - EC500: 450   1`
	`Maximum Off-PBX Telephones -   OPS: 450   12`
	`Maximum Off-PBX Telephones - PBFMC: 0     0`
	`Maximum Off-PBX Telephones - PVFMC: 0     0`
	`Maximum Off-PBX Telephones - SCCAN: 100   0`
	
	`(NOTE: You must logoff & login to effect the permission changes.)`

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.

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. One OPS entry is need for H.323 extensions, two entries are needed for SIP extensions, one for the primary number and one for the mobile endpoint. For more information, refer to **Section 3.3**.

2.7.4.

Example for H.323 extension:

- Set the **Station Extension** to the station extension of Abrazo-E as configured above (The example which follows uses 34071.)
- Set **Application** 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 Avaya 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**".
- Set **Call Limit** to "4".

```
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          1              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
```

2.7.5.	Example for SIP extension: Note: link the second SIP extension (34081) to the first extension (34080) so both station will ring when 34080 is called.

- Set the **Station Extension** to the station extension of Abrazo-E as configured above (The example which follows uses 34080 and 34081.)
- Set **Application** 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: 34080 and 34081) 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**".
- Set **Call Limit** to "4".

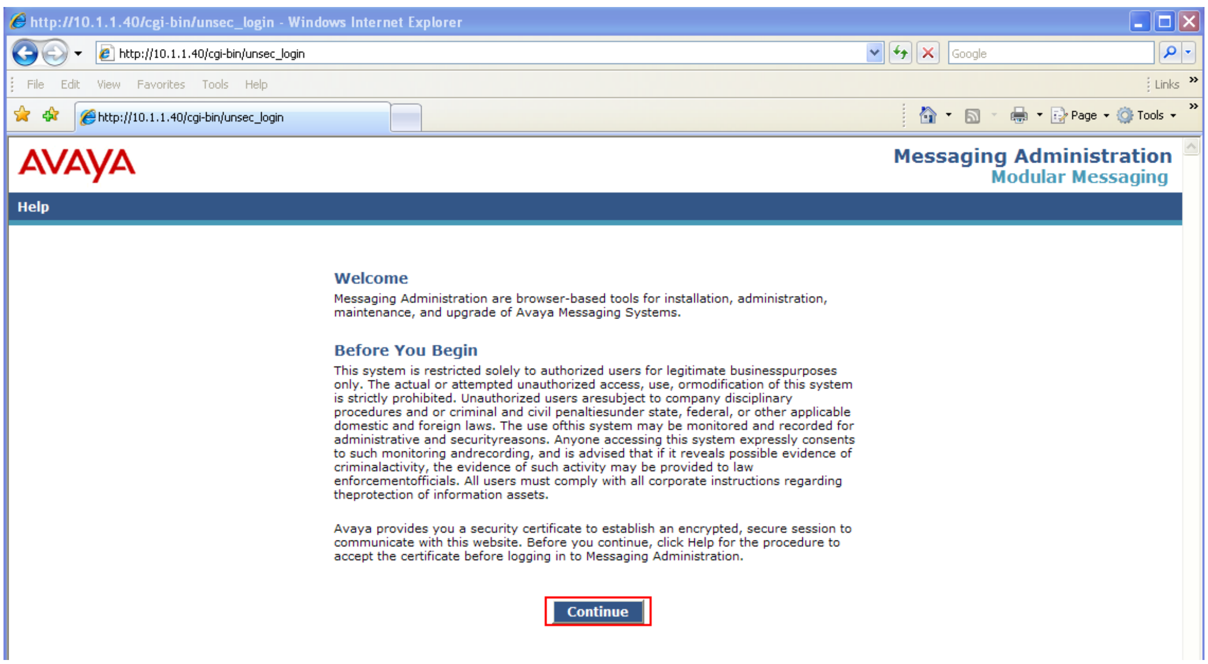
```
change off-pbx-telephone station-mapping 34071            Page 1 of 2
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
Station Application Dial Phone Number   Trunk        Configuration
Extension           Prefix              Selection    Set
34080 OPS -              34080          1              1
34080 OPS -              34081          1              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
 34080     4      both     all      both
 34080     4      both     all      both
```

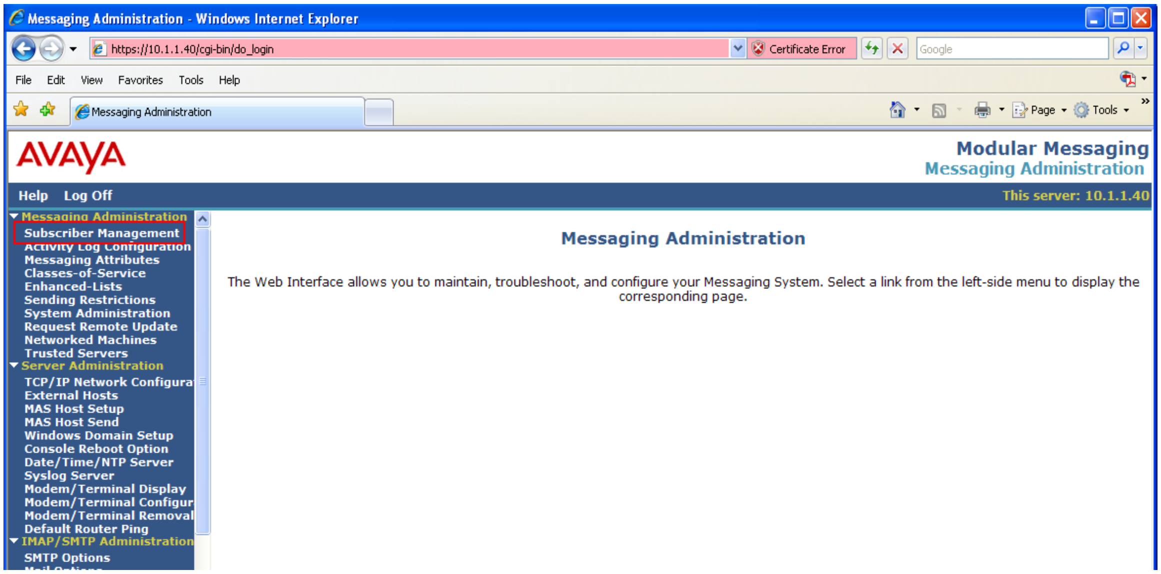
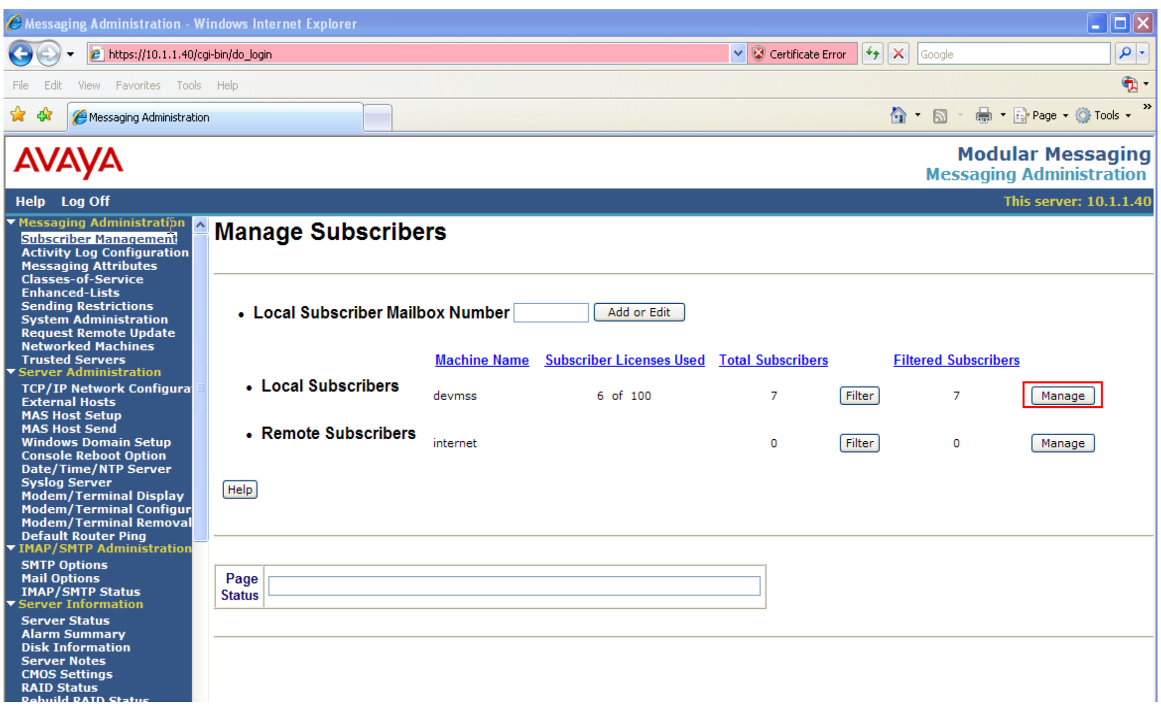
2.8. Voice Mail Configuration

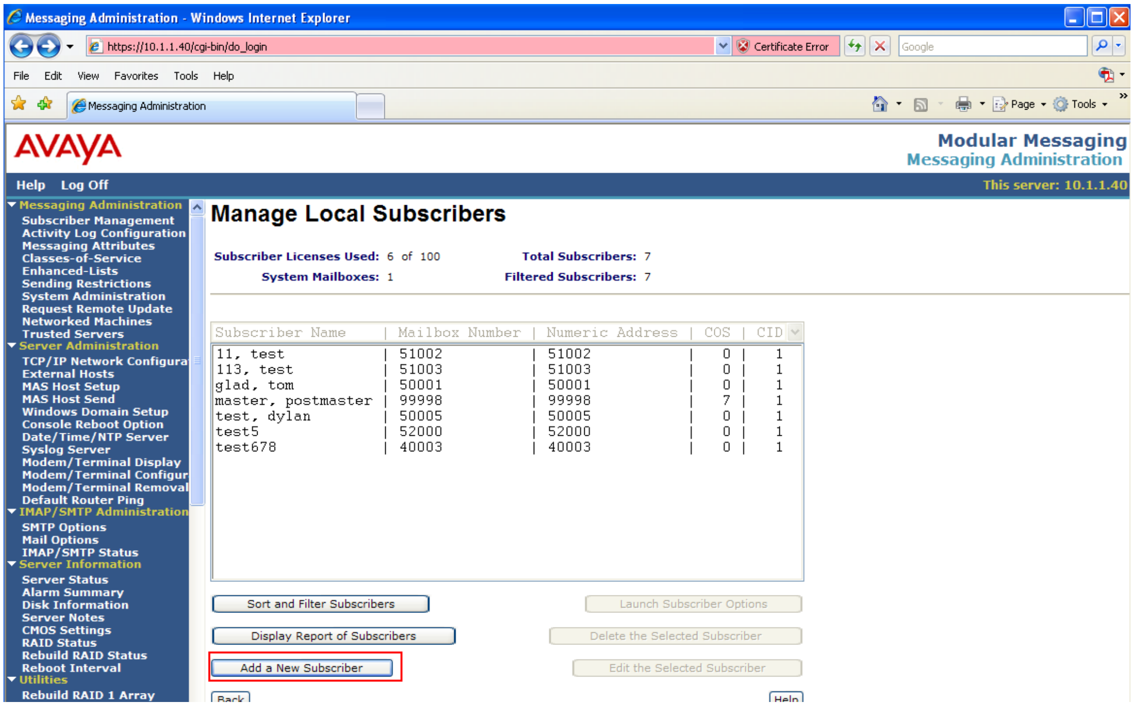
This section describes the steps for configuring voicemail for extensions in the Avaya telephony infrastructure. For informational purposes, steps for both Avaya Modular Messaging and Avaya IA770 INTUITY AUDIX are included in this document. Use the setup information appropriate for the environment being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

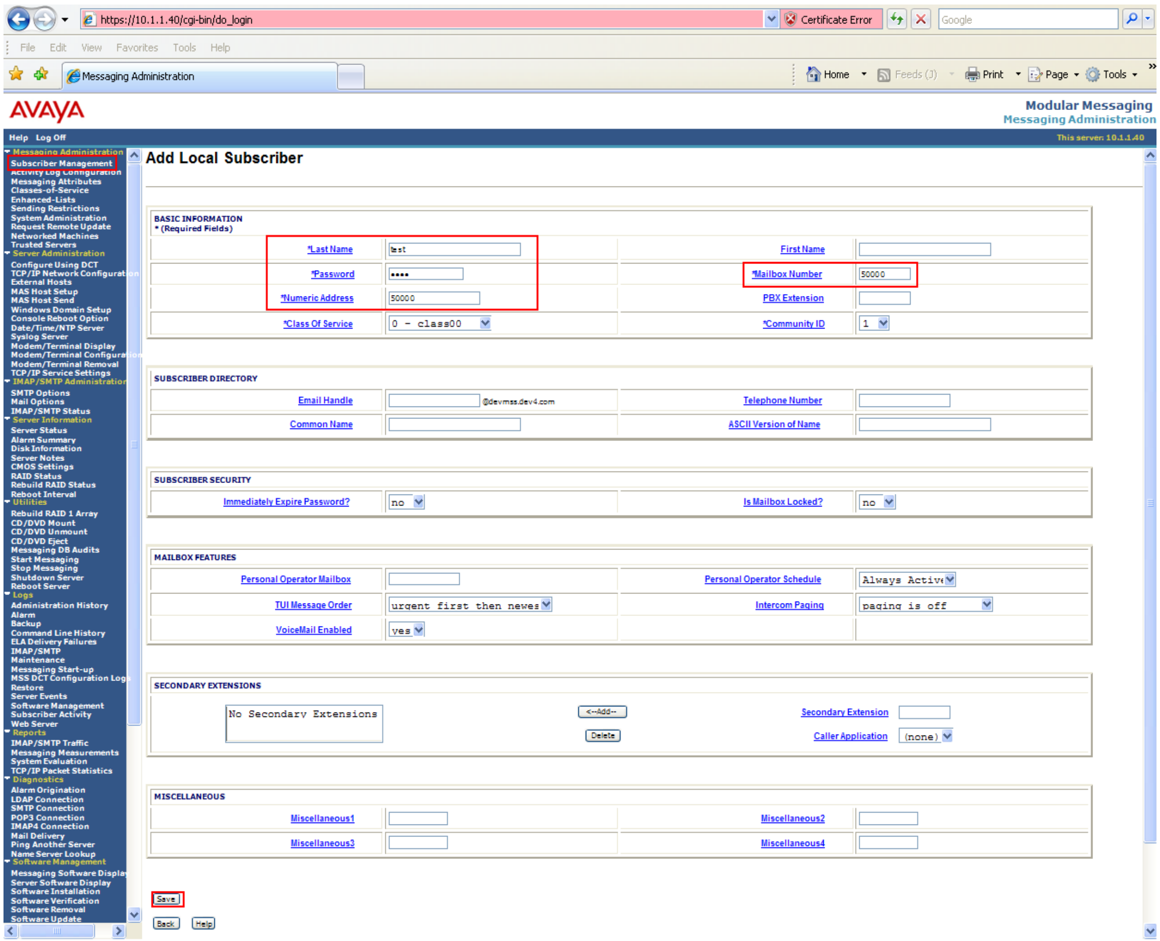
Note: It is recommended that at least four rings be used to route a call to voice mail.

Configure Subscriber on Avaya Modular Messaging

2.8.1.	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. 

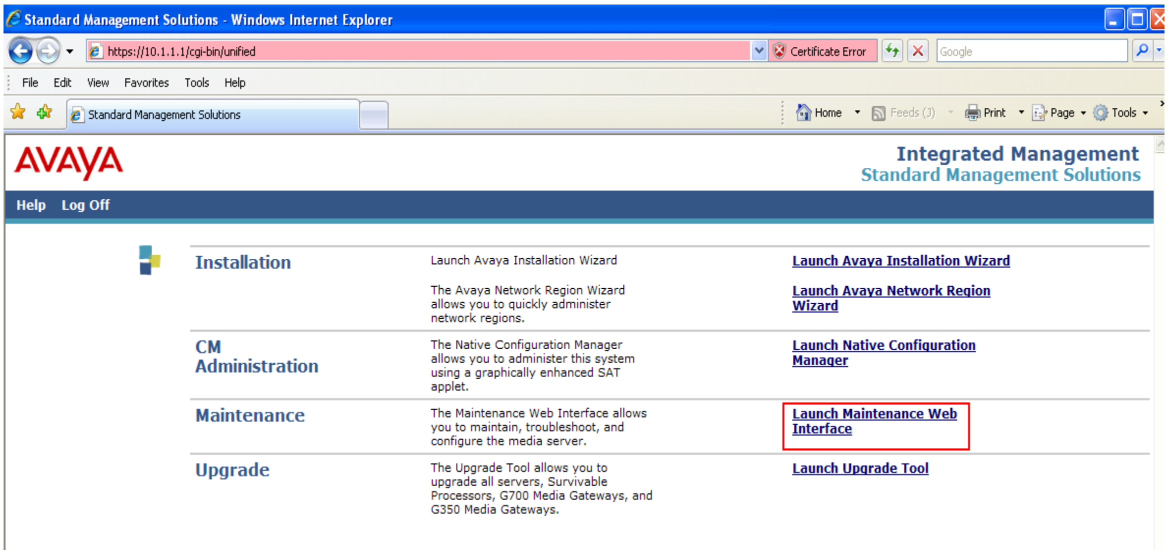
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.8.2.</td><td>Select Subscriber Management.

</td></tr>
<tr><td>2.8.3.</td><td>Select Manage.

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

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

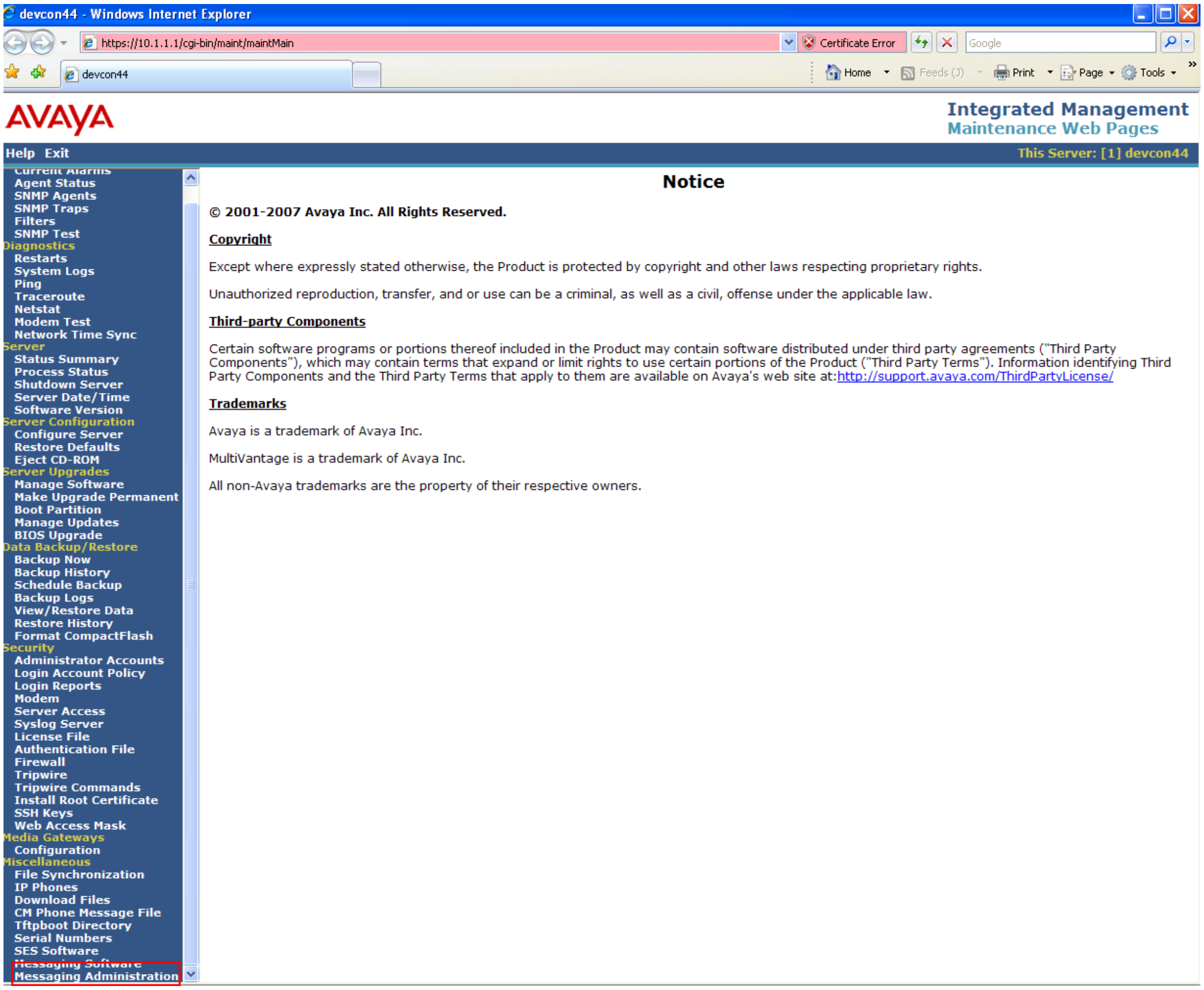
Step	Description
2.8.5.	Enter the following user information: **Last Name, Password**, **Mailbox Number**, **Numeric Address**. Select **Save** to continue. 

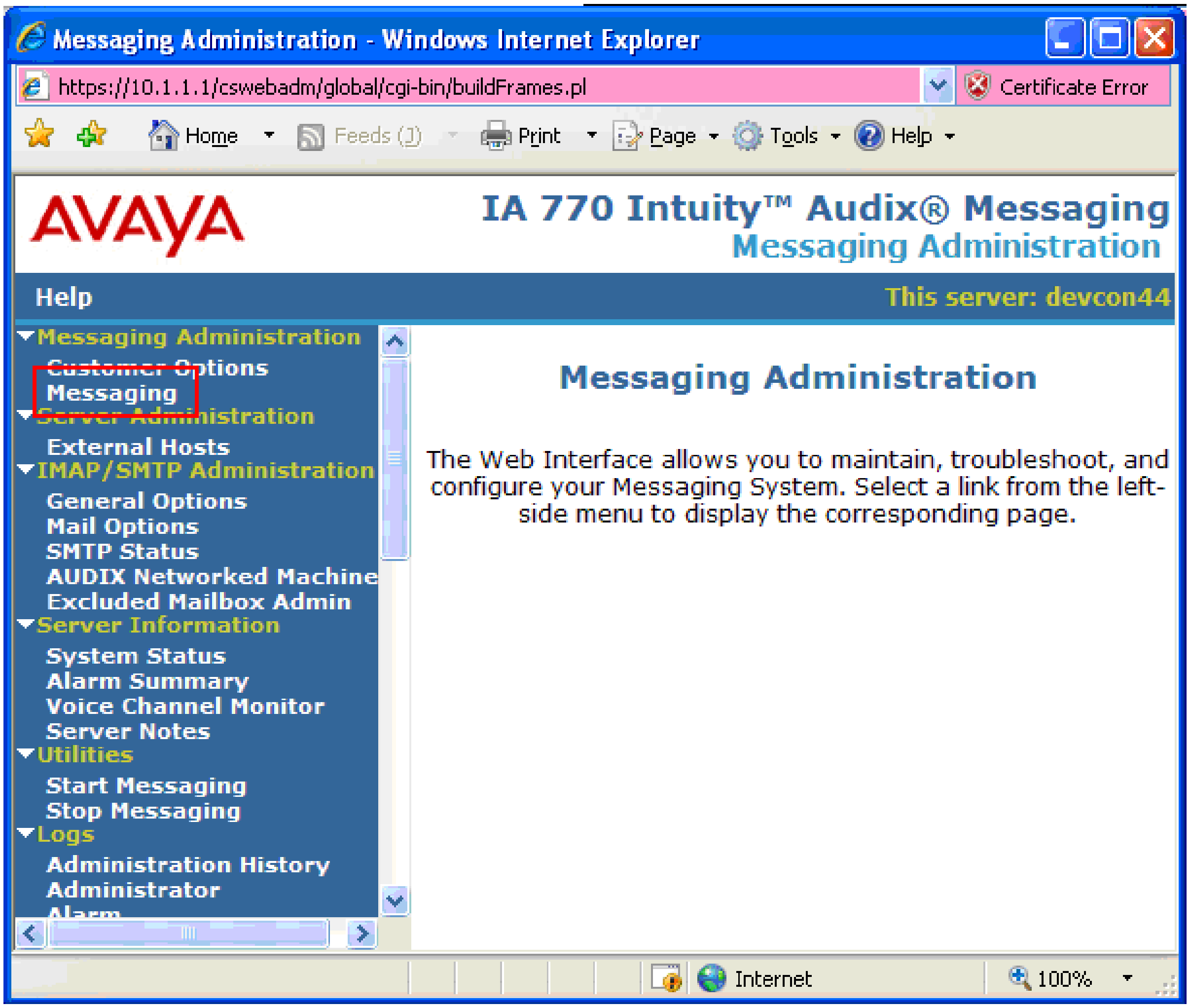
Configure Subscriber on Avaya IA770 INTUITY AUDIX

2.8.6. 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.

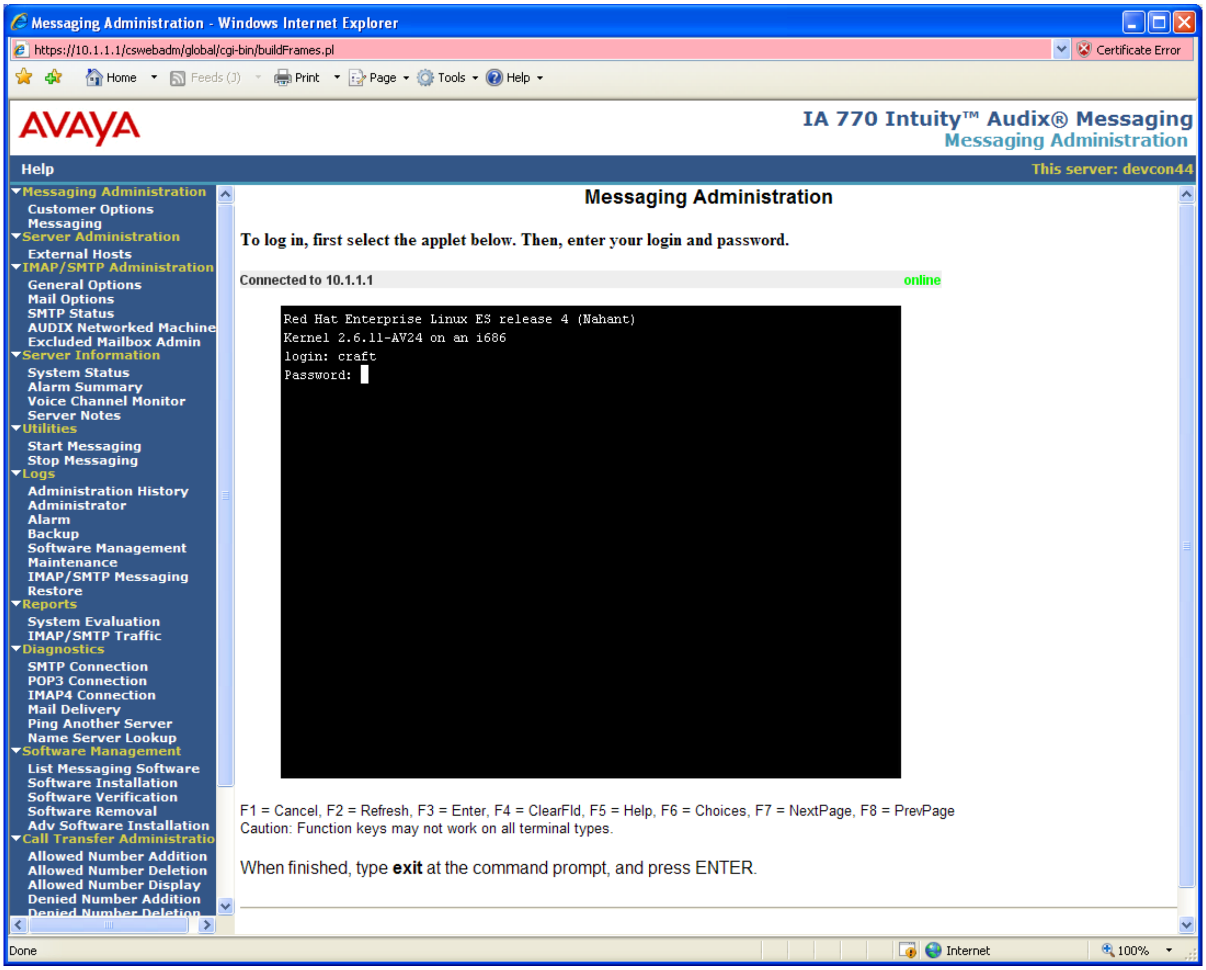


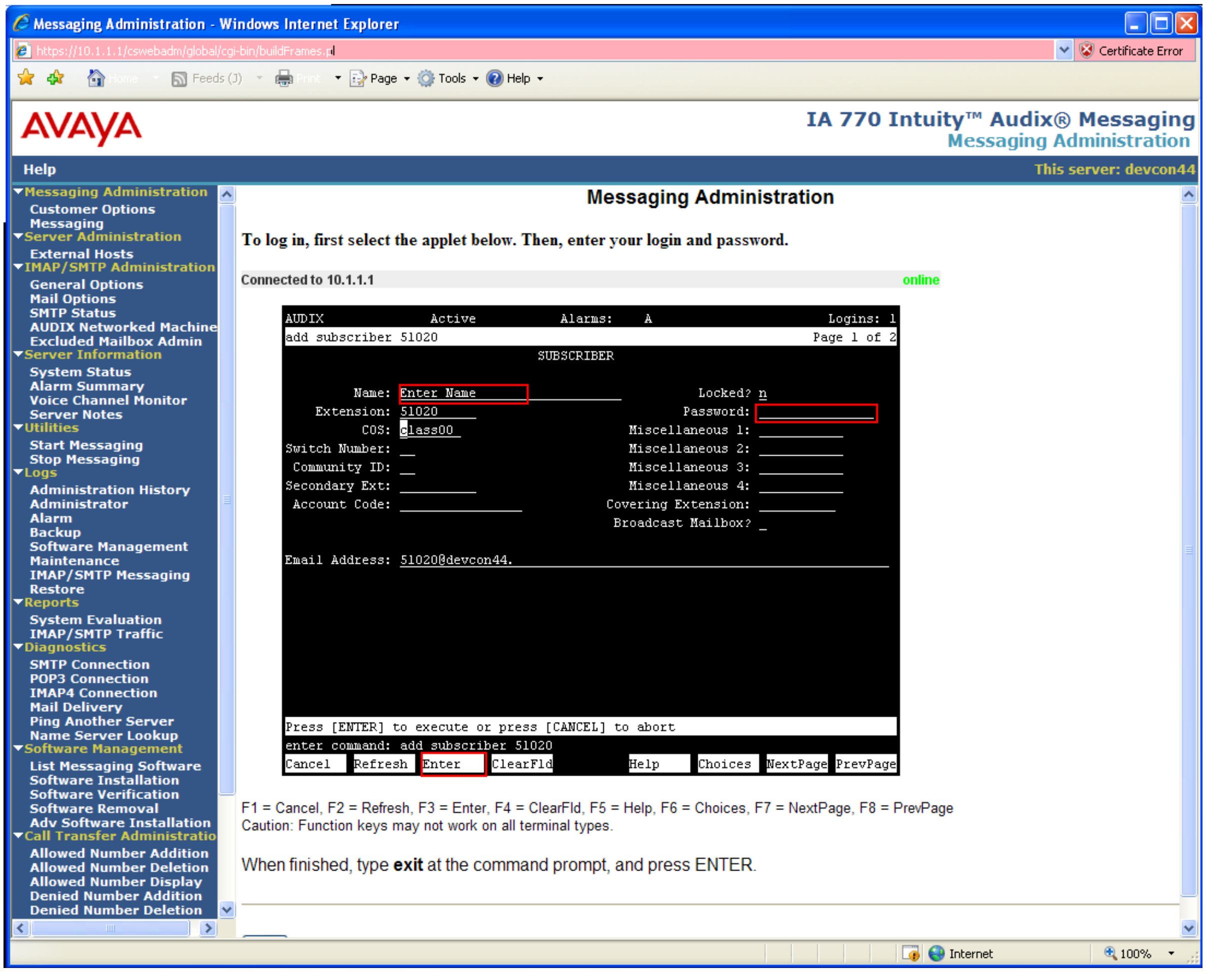
2.8.7.	Click on **Miscellaneous → Messaging Administration**.



2.8.8.	Click on **Messaging**. 

2.8.9. Login with the appropriate **Login** and **Password**.



2.8.10.	At the prompt, type **add subscriber X** where **X** is the desktop 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 Avaya Communication Manager:

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.

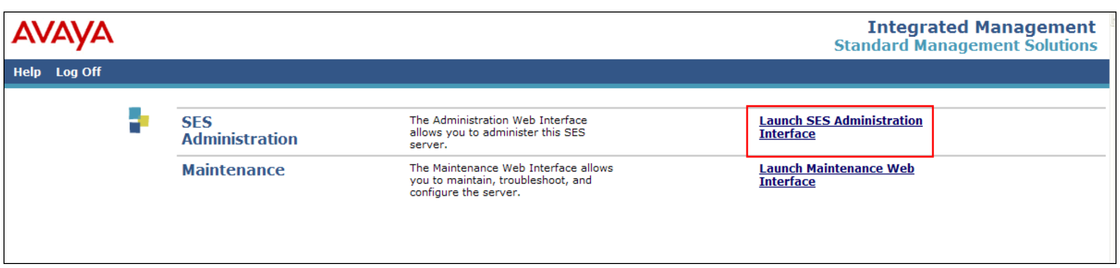
3.1. SES Software Configuration

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

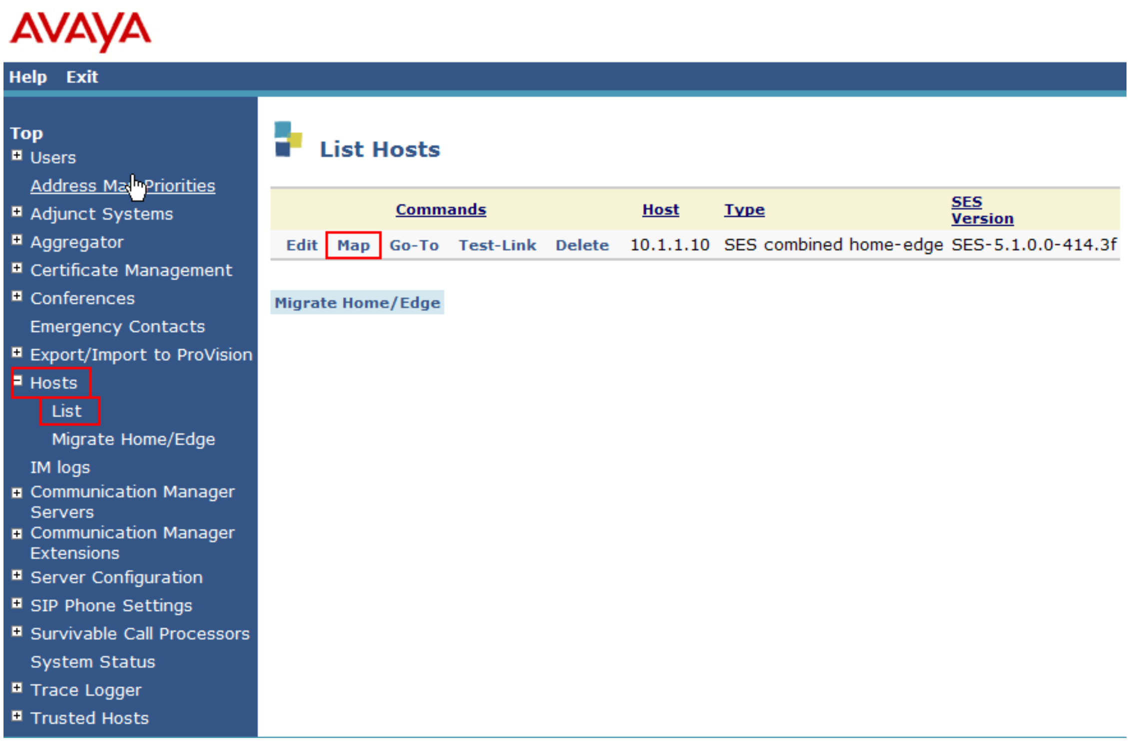
On the SES, the Abrazo solution needs to be configured using both a SIP trunk and a SIP line. The SIP trunk interface(s) are used by the Abrazo solution to terminate a call to the wireless operator's network. A SIP trunk is also used by the PBX to route mobile calls to the Abrazo solution via the enterprise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

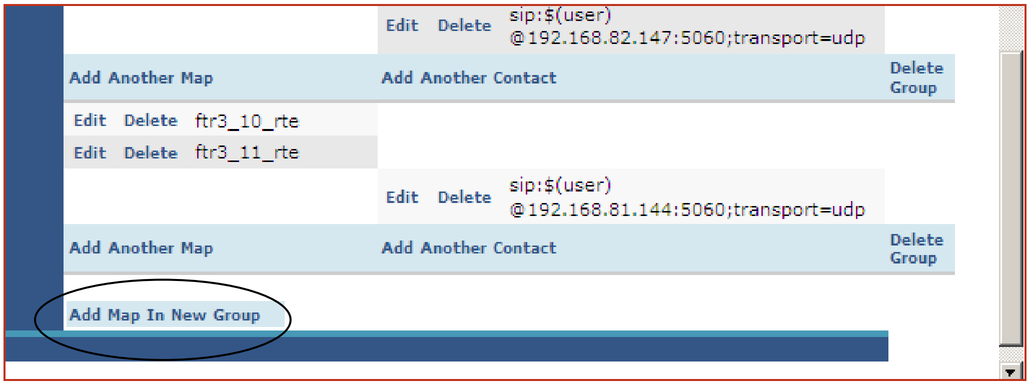
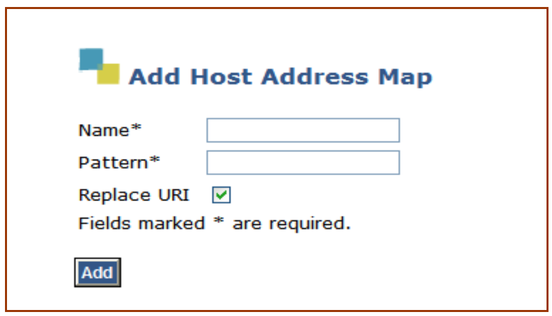
Step	Description
3.2.1.	Access the SES administration web interface by using the URL **HTTP://ip-address/ADMIN** in an Internet browser window, where **ip-address** is the IP address of the SES server. Log in with the appropriate credentials. The first screen of the interface is displayed. 

Step	Description
3.2.2.	The following screen is displayed.

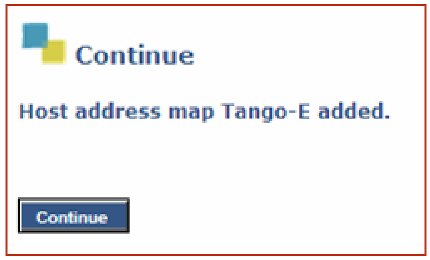
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.2.3.</td><td>Outbound calls are first routed by Avaya Communication Manager to the SIP trunk group. These calls are then subject to further routing decisions determined by Host Address Maps in the Avaya SES.

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.

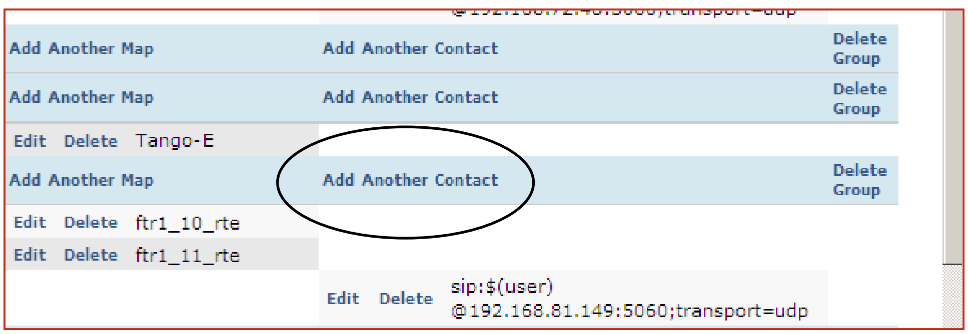

</td></tr>
</table>

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.  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**.

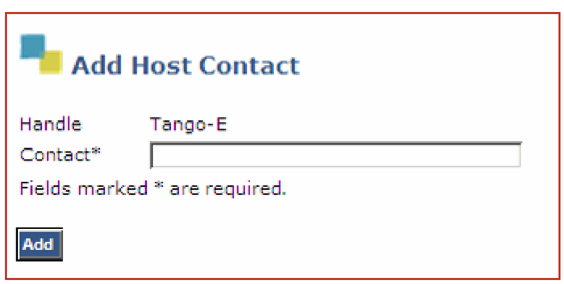
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.2.6.</td><td>A Continue screen is displayed to confirm the addition.



</td></tr>
<tr><td>3.2.7.</td><td>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.</td></tr>
<tr><td>3.2.8.</td><td>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.

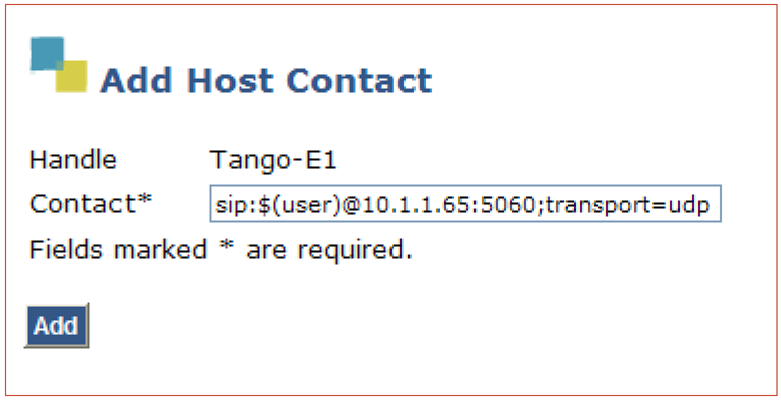


</td></tr>
<tr><td>3.2.9.</td><td>The Add Host Contact screen is displayed.



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

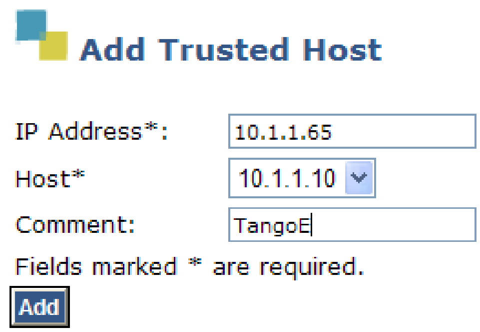
Step	Description
3.2.10.	The Contact field specifies the destination for the call. Populate the **Contact** field with the Abrezo IP address that the SES should substitute into the required URI before sending a message to the Abrazo Tango-E. The Avaya SES replaces **$(user)** with the user portion of the request URI before sending the message. Click the **Add** button.



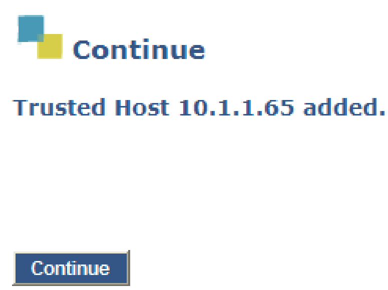
Step	Description
3.2.11.	A **Continue** screen is displayed to confirm the addition. Click the **Continue** button.

Step	Description
3.2.12.	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.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.2.13.</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>
<tr><td>3.2.14.</td><td>A Continue screen is displayed to confirm the addition. Click the Continue button



</td></tr>
<tr><td>3.2.15.</td><td>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.</td></tr>
</table>

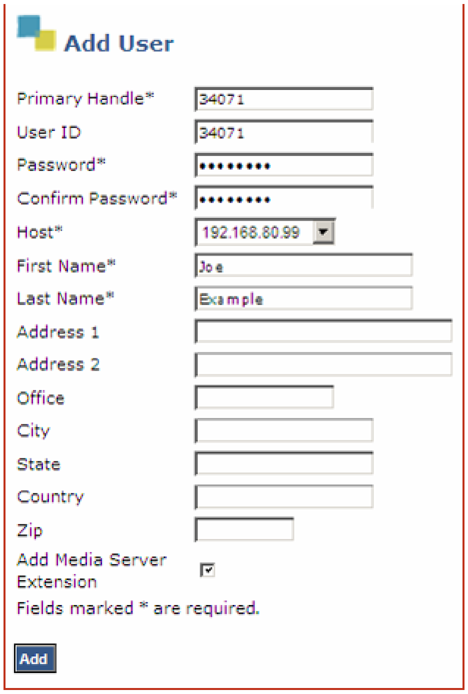
3.3. SIP Line Configuration

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.

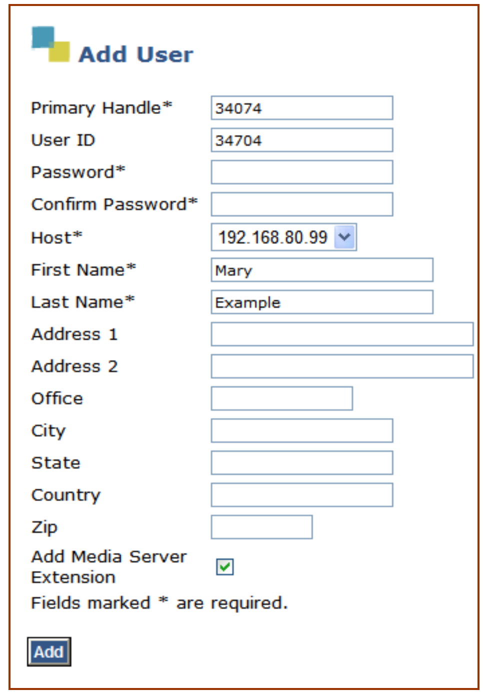
- If the user has an H.323 or digital desk phone, then an SES ID must be created with the same extension that was created for the desk station.
- If the user has a SIP desk phone, he or she will need a new ID created. (Since the desk phone is SIP, the user already has an SES ID created.)

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.3.1.</td><td>

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)

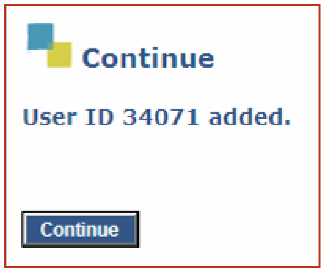
Ensure the following fields are populated as described below:

- 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.

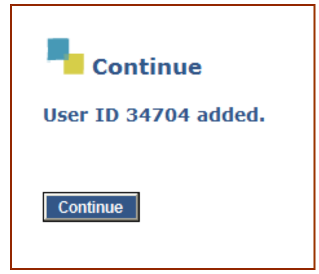
</td></tr>
</table>

Step	Description

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



Confirming the Addition of Joe



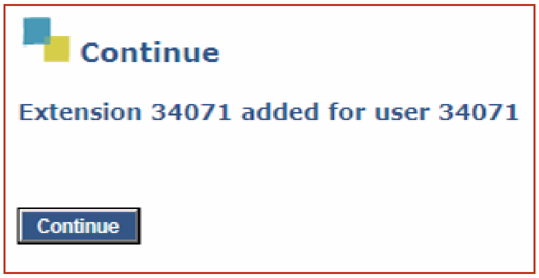
Confirming the Addition of Mary

Click Continue**. The Add Communication Manager Extension screen is displayed.**

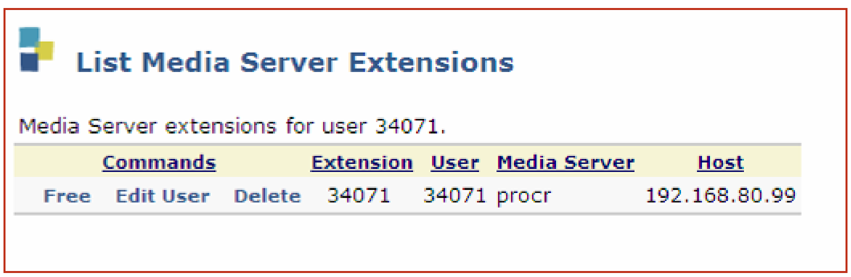
Adding Media Server Extension for Joe

Adding Media Server Extension for Mary

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.3.2.</td><td>
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.



</td></tr>
<tr><td>3.3.3.</td><td>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.</td></tr>
</table>

3.4. Verification Steps

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

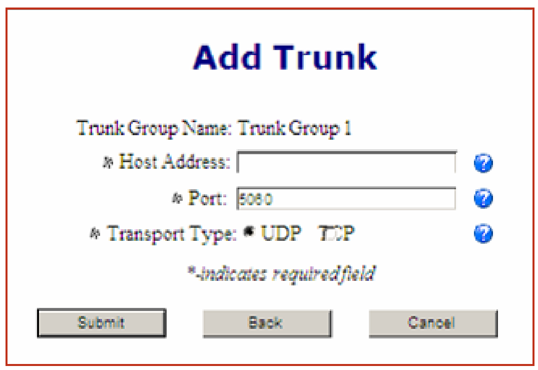
Step	Description
3.4.1.	From the Avaya SIP Server Management interface, select **Trusted Host→ List** to verify that the IP address of the Abrazo Tango-E is listed as a trusted node.
3.4.2.	From the Avaya SIP Server Management interface, navigate to **User → Registered Users** to verify that all SIP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

4. Provision the Tango Networks Abrazo-E

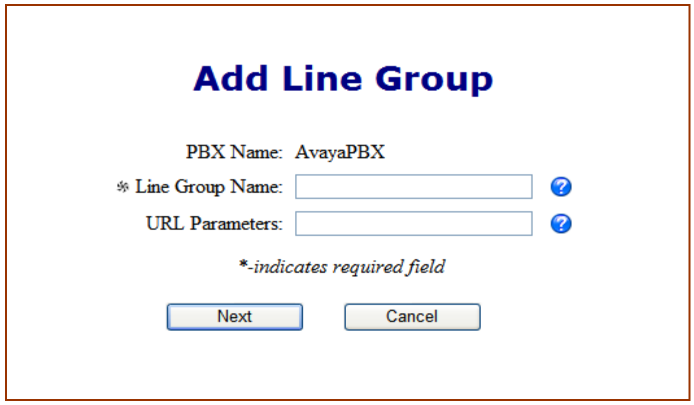
This section describes the processes required to integrate the Abrazo solution with a SIP-enabled Avaya Communication Manager. This document assumes that the Abrazo solution has already been provisioned with enterprise information and wireless carrier information. The integration process includes the 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.
- Provision the access numbers to enable handoff between the mobile network and WiFi network.
- 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
4.1.1.	To add the Avaya telephony infrastructure to the Abrazo-E, select **Voice Network → PBX → Add**. Enter a **PBX Name** (e.g., "AvayaPBX"). Select **Submit** to continue. 
4.1.2.	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
4.1.3.	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>4.1.4.</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.

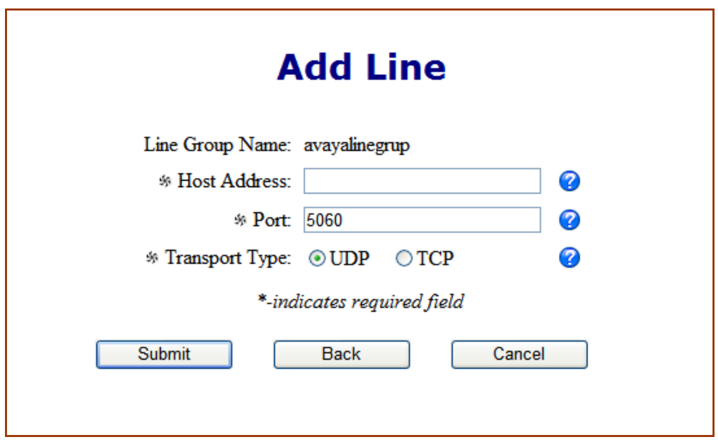
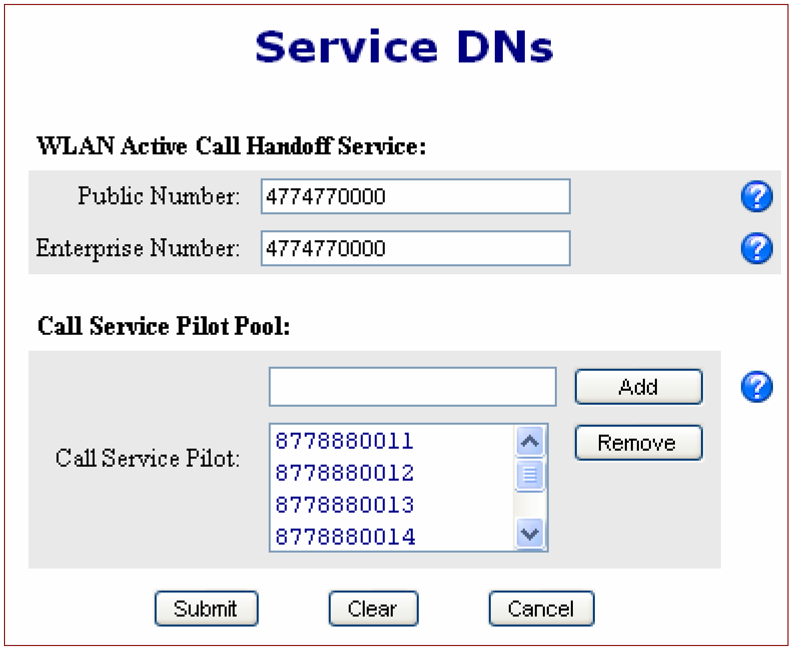


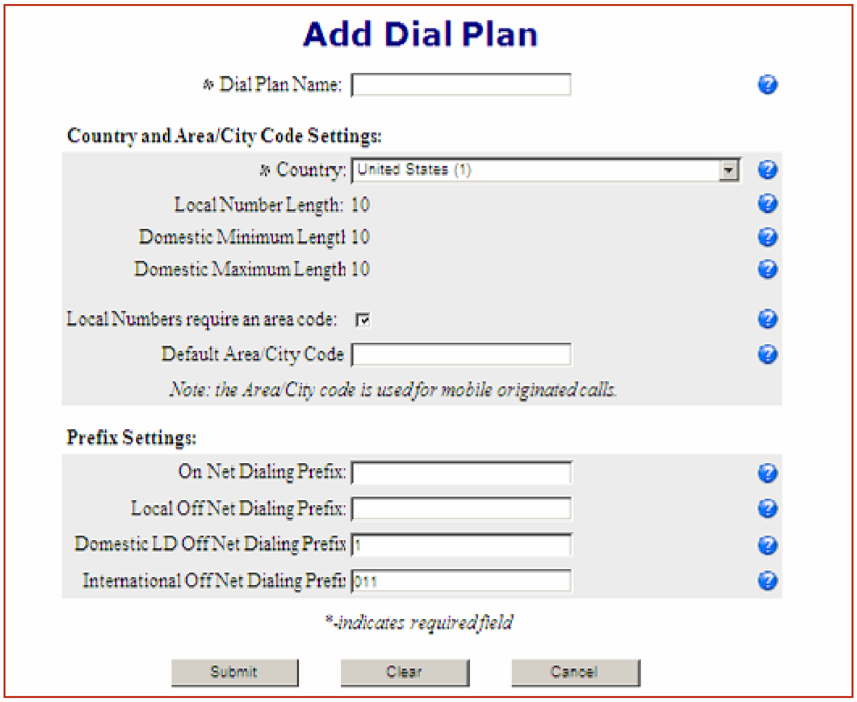
Figure 4 Add Line Screen

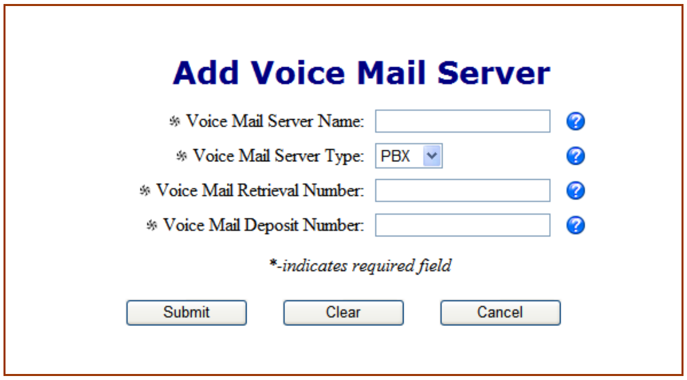
Enter line information on the Add Line screen. Select Submit to add the line members.
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.1.5.</td><td>
Define the Pilot numbers to be used to handoff from the cellular network to the WiFi network (Abrazo). To add service pool numbers, select Services → Service DNs. To save, click on Submit.



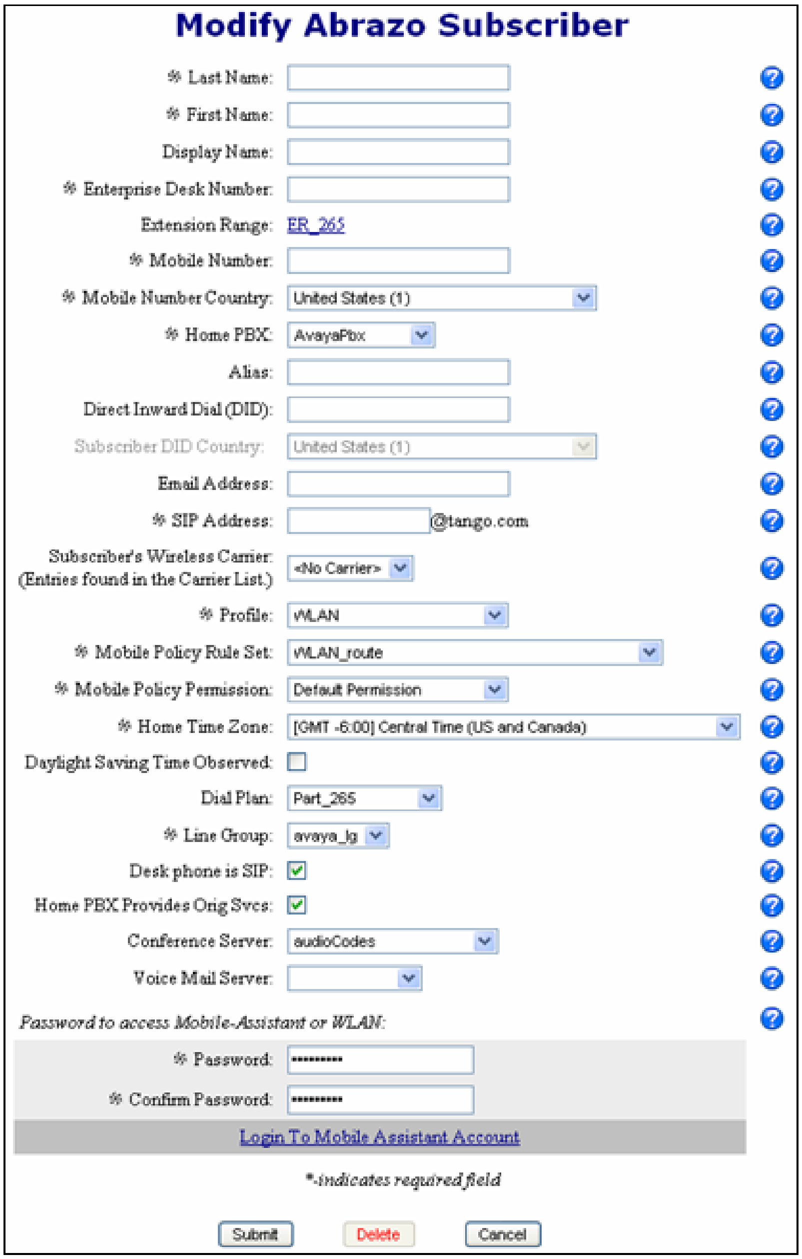
The actual service pilot number can be any format; however it must be prefixed with the feature access code for public translations configured on Avaya Communication Manager. In the example above, the number 9 is the feature access code.
</td></tr>
</table>

Step	Description
4.1.6.	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. 

Step	Description
4.1.7.	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.

4.1.8. 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 **Submit** 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.

Step	Description
4.1.9.	If the user's desk phone is H.323 or digital 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. The following steps describe the Abrazo configuration required when the desk phone is H.323 or digital. 1. Set the Abrazo **Enterprise Desk Number** to the extension defined for the user's station in the Avaya Communication Manager. 2. Set the Abrazo **SIP Address** to the user's ID created on the SES. In this case the ID on the SES must match the extension for the user's H.323 station. The Abrazo solution will use this address for registrations. 3. Leave the option **Desk phone is SIP** unchecked. 4. Ensure the option **Home PBX Provides Orig Svs** is checked. When checked, Abrazo originates calls for the mobile user through the home PBX. 5. For WiFi users, a WiFi enabled profile must be set, and a password must be set for the user's Wireless LAN account. This password is specific to the individual user and is used to authenticate access from the wireless client.

Step	Description
4.1.10.	If the user's desk phone is SIP If the user's desk phone is SIP, the SES will contain two IDs for this user. One ID will match the extension for the station and this will be used by the SIP phone. A second ID is created that is used by the Abrazo. In this case, the Abrazo will register using this second ID. Since this ID does not match the station's extension, the Abrazo will also subscribe for voice mail events. 1. The following steps describe the Abrazo configuration required when the desk phone is SIP. 2. Set the Abrazo **Enterprise Desk Number** to the extension defined for the user's SIP station. The Abrazo solution will use this for voice mail subscriptions. 3. Set the Abrazo **SIP Address** to the user's ID created on the SES. Because the user's desk phone is SIP, there will be two SES IDs created for this user—one for the SIP desk phone and one for the Abrazo SIP Address. The Abrazo solution will use this address for registrations. 4. Check the option for **Desk phone is SIP**. 5. For WiFi users, a WiFi enabled profile must be set, and a password must be set for the user's Wireless LAN account. This password is specific to the individual user and is used to authenticate access from the wireless client. Ensure the option **Home PBX Provides Orig Svs** is checked. By checking this field, the Abrazo always originates the call on behalf of the mobile user for mobile originations into their home PBX within the enterprise network.
4.1.11.	Repeat **Step 4.1.7** thru **9** for each user to be added to the system.

5. 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 FMC Solution and the ability for an enterprise user to be accessible via one business number and be able to roam between WiFi and mobile networks whether the user is in the office or mobile.

5.1. 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
- WiFi originated calls routed through the Avaya telephony infrastructure terminating to a desk phone, mobile device, or the PSTN
- WiFi terminated calls routed through the Avaya telephony infrastructure
- in-call handoff from WiFi to mobile network and mobile network to WiFi network

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 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.
- **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** (while in WiFi mode) - 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 works for subscribers using H.323 desk phones and 9600 Series IP phones running SIP. When a voice call is established on the desk 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.
- **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.
- **Flexible Dialing Support** - Abrazo has a flexible dialing plan enabling PBX services to be provided to mobile users.
- **Intelligent Call Delivery** - ensures that both the desk phone and mobile phone ring when the dialed number is an Abrazo subscriber.
- **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.

- **VCC capabilities** – the Abrazo enables subscribers with dual-mode phones running an Abrazo certified client to seamlessly hand off calls between the mobile and WiFi networks. Calls can be handed off from the WiFi network to the cellular network and vice-versa.
- **Voice Mail Message Waiting Indication** was NOT verified.

5.2. Test Results

The test objectives of **Section 5.1** were verified. The Tango Networks Abrazo FMC Solution successfully completed all test cases for the features identified in **Section 5.1**. Tango Networks is able to hand off calls between the mobile and WiFi networks route inbound/outbound calls to/from Avaya Communication Manager with all services tested.

6. 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

7. Conclusion

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

8. Additional References

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

[1] *Administrator Guide for Avaya Communication Manager,* January 2008, Issue 4.0, Document Number 03-300509

[2] *Installing and Administering SIP Enablement Services,* January 2008, Issue 5.0, 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.