Avaya Solution & Interoperability Test Lab

Application Notes for Configuring the Agito Networks RoamAnywhere Mobility Router with Avaya Communication Manager, Avaya SIP Enablement Services, Avaya Modular Messaging, Avaya IA 770 INTUITY AUDIX and Avaya IP Telephones - Issue 1.0

Abstract

These Application Notes describe a compliance-tested configuration comprised of the Agito Networks RoamAnywhere Mobility Router connected to an Avaya telephony infrastructure. The Agito Networks RoamAnywhere Mobility Router fuses WLAN, Cellular and IP Private Branch Exchanges (PBXs) technology in order to extend enterprise PBX functionality to mobile devices. This allows end users to be accessible when out of the office as well as to leverage WLAN networks to improve wireless coverage and reduce costs. The Agito Networks RoamAnywhere Mobility Router integrates mobile devices with existing PBX 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.

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1. Introduction

These Application Notes describe a compliance-tested configuration comprised of the Agito Networks RoamAnywhere Mobility Router connected to an Avaya telephony infrastructure. The Agito Networks RoamAnywhere Mobility Router fuses WLAN, Cellular and IP PBX technology in order to extend enterprise PBX functionality to mobile devices. This allows end users to be accessible when out of the office as well as to leverage WLAN networks to improve wireless coverage and reduce costs. The Agito Networks RoamAnywhere Mobility Router integrates mobile devices with existing Private Branch Exchanges (PBXs) so that the PBX sees the mobile device as another desk phone. This allows the existing PBX feature set to be applied consistently across both devices. Mobile specific functionality is then layered on top.

The Agito RoamAnywhere Solution uses a combination of SIP lines and trunks to integrate with Avaya Communication Manager. SIP lines are used so that Agito-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 Agito RoamAnywhere solution must terminate a call via the Public Switched Telephone Network (PSTN).

The RoamAnywhere Solution transparently handles all mobile call originations from a user's mobile device and redirects them through the enterprise leveraging the WLAN network when available or routing over cellular when outside of WLAN coverage areas. 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.

1.1. Test Environment

The test environment consisted of an Avaya Communication Manager running on an Avaya S8300 Server with an Avaya G700 Media Gateway, one Avaya SIP Enablement Services server, one Avaya Modular Messaging Application Server, one Avaya Modular Messaging Storage Server, one Avaya 2400 Series Digital Telephone, one Avaya 9630 IP Telephone running Avaya one-X™ Deskphone SIP, one Agito RoamAnywhere Mobility Router, one dual mode cell phone running Agito Networks RoamAnywhere Mobility Dialer, one WiFi controller and access point and one DHCP/File Server.

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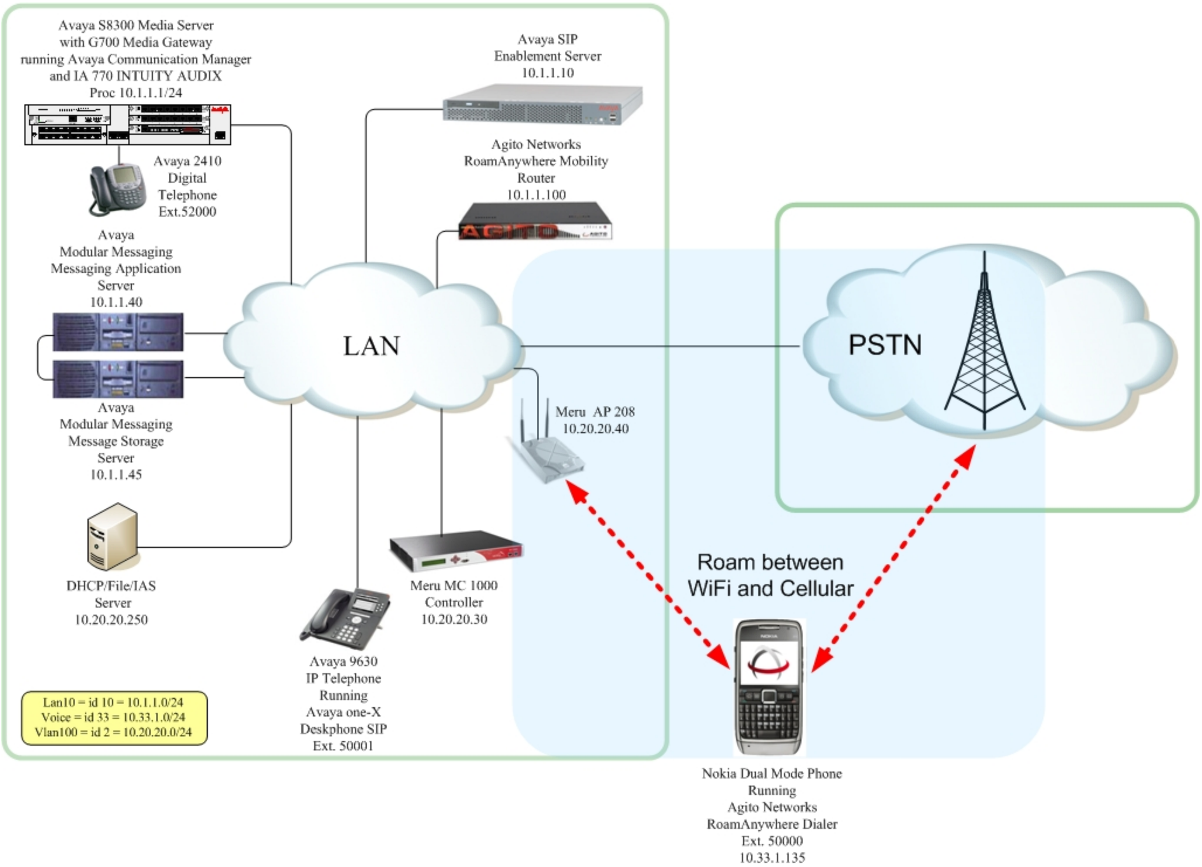

Figure 1:Network Diagram

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8300 Server	Avaya Communication Manager 5.0 - R015x.00.0.825.4
Avaya G700 Media Gateway MGP MM712 DCP Media Module	26.31.0 HW05 / FW08
Avaya SIP Enabled Services (SES) Server	5.0 - SES-5.0.0.0-825.31
Avaya Modular Messaging - Messaging Application Server (MAS)	3.1
Avaya Modular Messaging - Message Storage Server (MSS)	3.1
Avaya IA 770 INTUITY AUDIX	5.0
Avaya 9600 Series IP Telephones	Avaya one-X Deskphone SIP 2.0.3 (SIP)
Avaya 2410 Digital Telephone	5.0
Agito Networks RoamAnywhere Mobility Router	1.0.0.118
Agito Networks RoamAnywhere Mobility Dialer	1.0.0.24
Nokia E71	n/a

3. Configure Avaya Communication Manager

This section describes the steps required for Avaya Communication Manager to support the configuration shown in **Figure 1**. The following pages provide instructions on how to administer the required configuration parameters. The assumption is 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 Communication Manager and has access to the System Administration Terminal (SAT) screen. For detailed information on the installation, maintenance, and configuration of Avaya Communication Manager, please consult references 1 thru 4 in Section 9

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3.1. 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 the SIP trunk between the Avaya Communication Manager and Avaya SES.

Description
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:
```

3.2. IP Node Names

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

Description
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
AES-DevCon2         192.45.100.153
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
```


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3.3. 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.

Description
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
```

3.4. Trunks and Signaling Groups for Avaya SES

This section describes the steps for administering the trunk groups and signaling groups in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES. A second SIP trunk needs to be created because The Agito Mobility Router inputs its own IP address as the domain in the from header of the SIP packet. This is created in section 3.6.

These SIP trunks will carry the SIP signaling sent to the Agito Mobility Router for mobile originated and terminated calls. This SIP trunk will also provide the trunking for calls originated by the Agito Mobility Router when acting as a SIP endpoint to support intelligent call delivery.

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3.5. Create SIP trunk with domain information

Step	Description
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
```


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Step	Description
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 3.2** • **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 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
```


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Step	Description
3.	Enter the **change trunk-group i** command, where "i" is the number of the trunk group configured in **Step 3.5.1**. **On Page 1** of the **trunk-group** form, set configure the following: • **Signaling Group –** enter the Signaling Group number that was used in **step 3.5.2.** • **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
```


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3.6. Create SIP trunk without domain information

Step	Description
4.	Enter the **add trunk-group g** command, where "g" 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 98                                                 Page   1 of  21
                                TRUNK GROUP

Group Number: 98                   Group Type: sip           CDR Reports: y
  Group Name: T0SESB                       COR: 1        TN: 1        TAC: *098
   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
```


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Step	Description
5.	Enter the **add signaling group m** command, where "**m**" 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 Domain – 10.1.1.100** • **Far-end Node Name –** enter the node name of the SES configured in **Step 3.2** • **Far-end Listen Port** – specify the local listen port, typically **5060.** • **Far-end Network Region –** enter the IP network region configured in **Step 3.1** • **DTMF over IP –** set to "**rtp-payload**". • **Direct IP-IP Audio Connections –** set to "**y**".

```
add signaling-group 98                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 98             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: 10.1.1.100

                                             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
```


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Step	Description
6.	Enter the **change trunk-group i** command, where "i" is the number of the trunk group configured in **Step 3.5.1**. **On Page 1** of the **trunk-group** form, set configure the following: • **Signaling Group** – enter the Signaling Group number that was used in **step 3.5.2.** • **Number of Members** – set to **24**

```
change trunk-group 98                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 98                   Group Type: sip           CDR Reports: y
  Group Name: T0 SES                      COR: 1        TN: 1        TAC: *098
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 98
                                                     Number of Members: 24
```

3.7. Configure Off-PBX Telephone Integration information

Step	Description
7.	Every user must be defined as an off-PBX station in order to enable simultaneous ringing. 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. Enter **change off-pbx-telephone station-mapping n**, where **n** is the number of the phone extension where a mobile extension shall be configured. Enter the following information: • **Station Extension = n** • **Application = OPS** • **Phone Number = Phone Number of the new Extension** • **Trunk Selection = Trunk used to the SES** • **Configuration Set = 1**

```
change off-pbx-telephone station-mapping 40000                  Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station     Application   Dial   Phone Number      Trunk       Configuration
 Extension                 Prefix                   Selection   Set
 50000        OPS            -       50000          1          1
                               -
                               -
                               -
```


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3.8. Dial Plan

This section describes the steps for setting 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 Agito Networks RoamAnywhere Mobility Router.

Note: Route handling varies from location to location. The following example was used for compliance testing. Refer to [1] for further options.

From the SAT, enter the following commands and information:

Step	Description
1.	To handle the incoming calls to the Agito Mobility router the dial string need to be altered. This is done with the **change inc-call-handling-trmt trunk-group j** command, where "j" is the trunk group going out to the PSDN**.** For the compliance testing, the Agito was expecting the dial string without the 1 so the 1 was deleted. In addition, Automatic Alternate Routing (AAR) was used to the Agito Router so the AAR feature access code of 3 was inserted. •

```
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  1732555          6
 tie            11  17328522963      1    3
 tie
```

Step	Description
2.	Use the **change aar analysis** command to add an AAR entry for the Agito mobility router.

```
change aar analysis 0                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location:  all         Percent Full:    0

          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    5555552963            10   10    24        aar          n
                                                            n
                                                            n
```


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Step	Description
3.	Use the **change route-pattern** command to associate a route pattern to the SIP trunk which is used to access the Agito Mobility Router.

```
change route-pattern 24                                         Page   1 of   3
                    Pattern Number: 24  Pattern Name:
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                             DCS/ IXC
    No          Mrk Lmt List Del  Digits                               QSIG
                             Dgts                                      Intw
 1: 1    0                                                              n   user
 2:                                                                     n   user
 3:                                                                     n   user
none
```

4. Configure Avaya SIP Enablement Services

This section describes the steps required for Avaya SIP Enablement Services to support the configuration in **Figure 1**. 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 Agito.

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 Administrator web interface.

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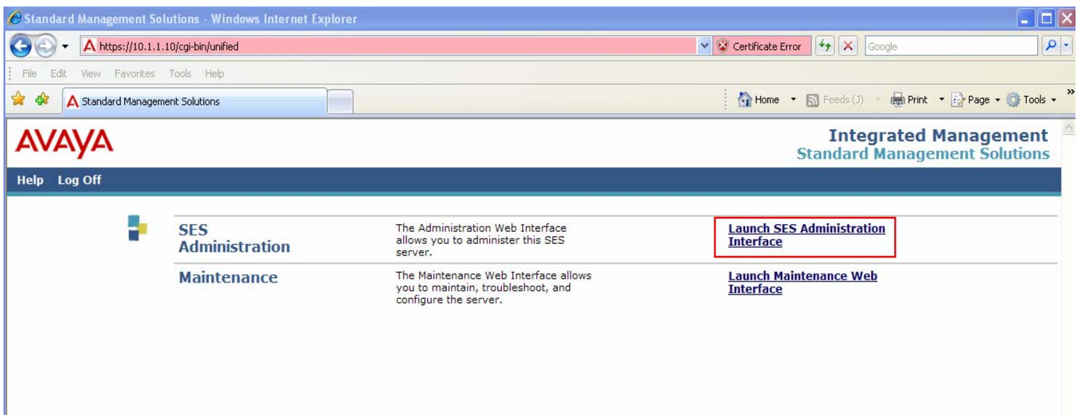
4.1. SIP Trunk Configuration

On the SES, the Agito Mobility Router needs to be configured using both a SIP trunk and a SIP line. The SIP trunk interface(s) are used by the Agito Mobility Router to terminate a call to the wireless operator's network. A SIP trunk is also used by Avaya Communication Manager to route mobile calls through the SES to the Agito Mobility Router using the a Direct Inward Dialing (DID) number assigned to the Agito Mobility Router.

Step	Description
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. Select **Launch Administration Web Interface**. 

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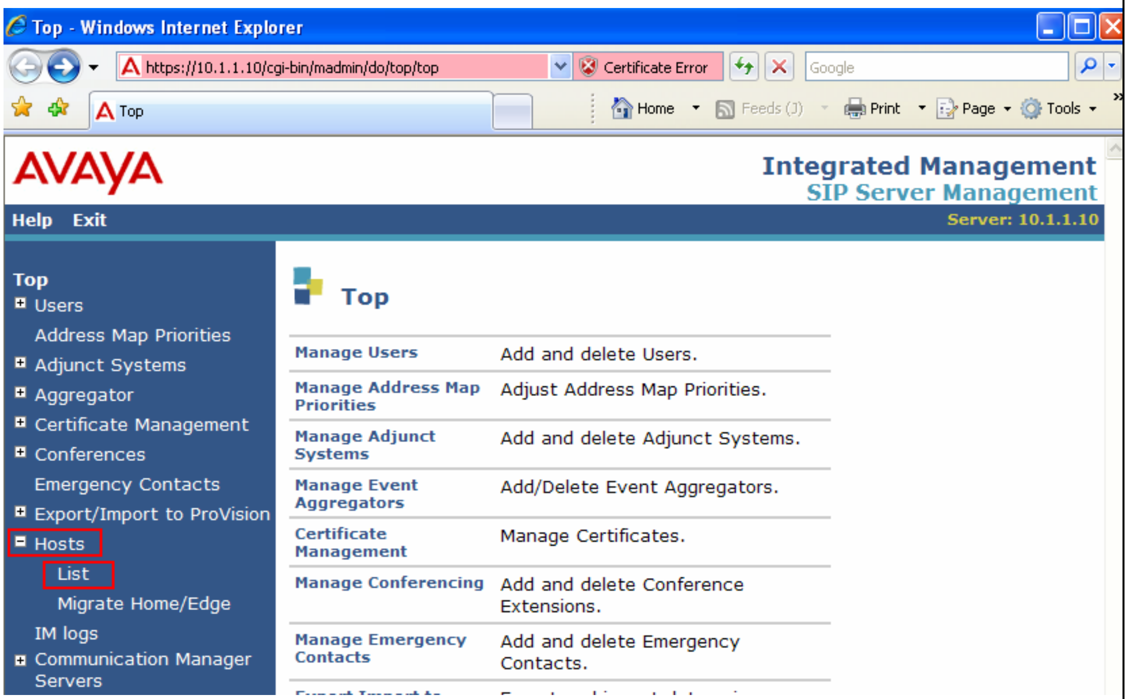
Step	Description
2.	The following screen is displayed. From the left pane, select **Hosts → List.** 

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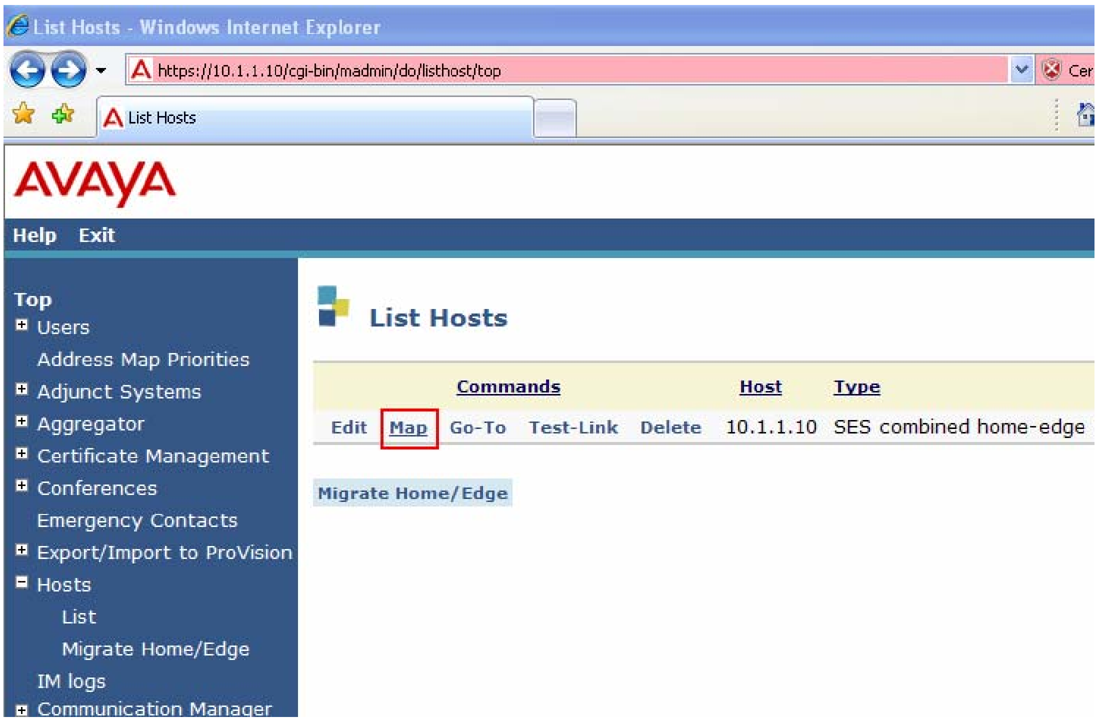
Outbound calls are first routed by Avaya Communication Manager to the SIP trunk group. These calls are then subject to further routing decisions determined by Host Address Maps in the Avaya SES.

3. The **List Hosts** screen is displayed, click on **Map** in the right pane.



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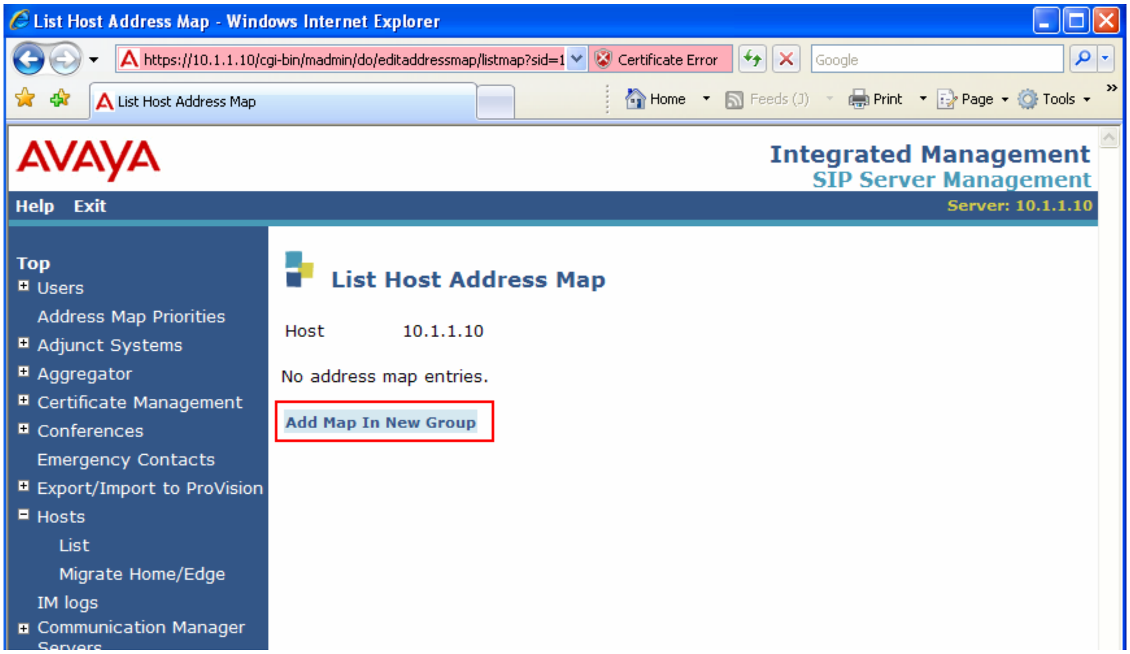
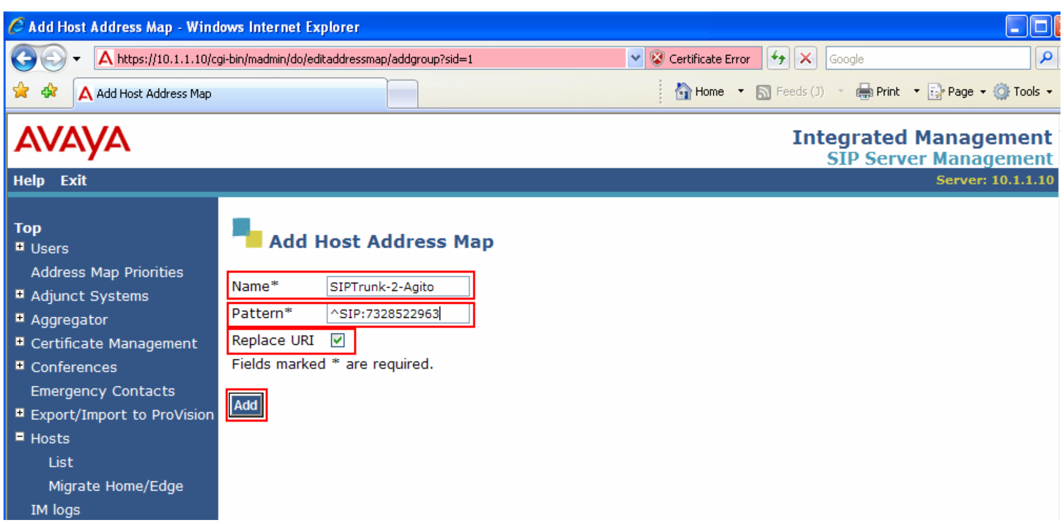
<table>
<tr>
<td>4.</td>
<td>The List Host Address Map screen is displayed. Select, Add Map In New Group.



</td>
</tr>
<tr>
<td>5.</td>
<td>The Add Host Address Map screen is displayed. Enter the following;

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 DID number used to route mobile calls into the Agito Mobility Router.
Retain the check in Replace URI, and click Add.



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

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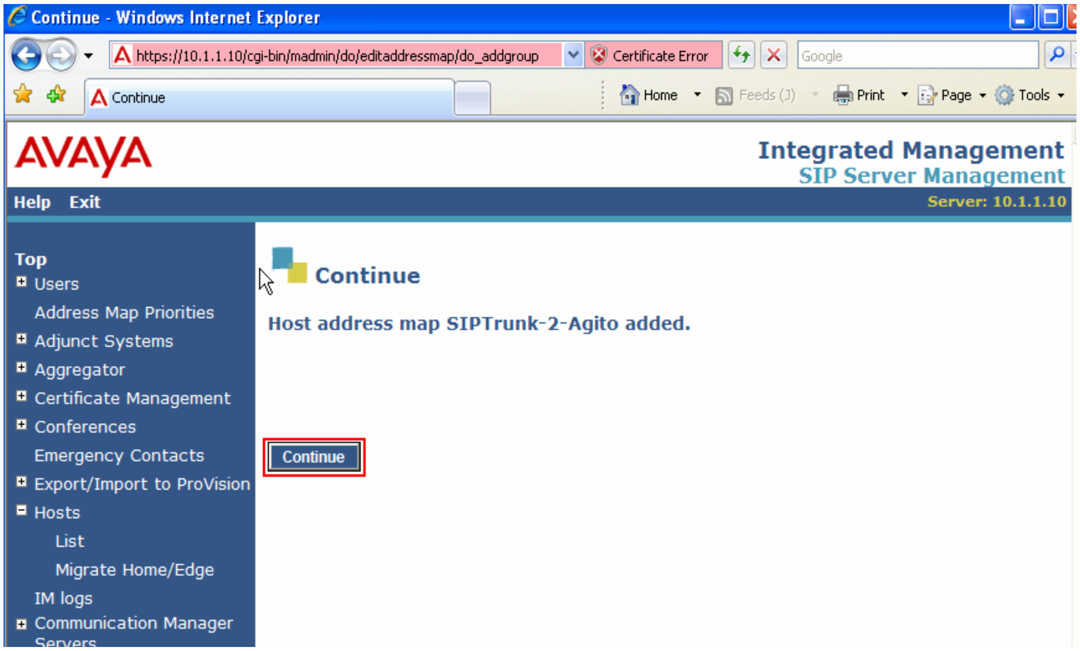
6.	A **Continue** screen is displayed to confirm the addition. Click **Continue** to continue. 

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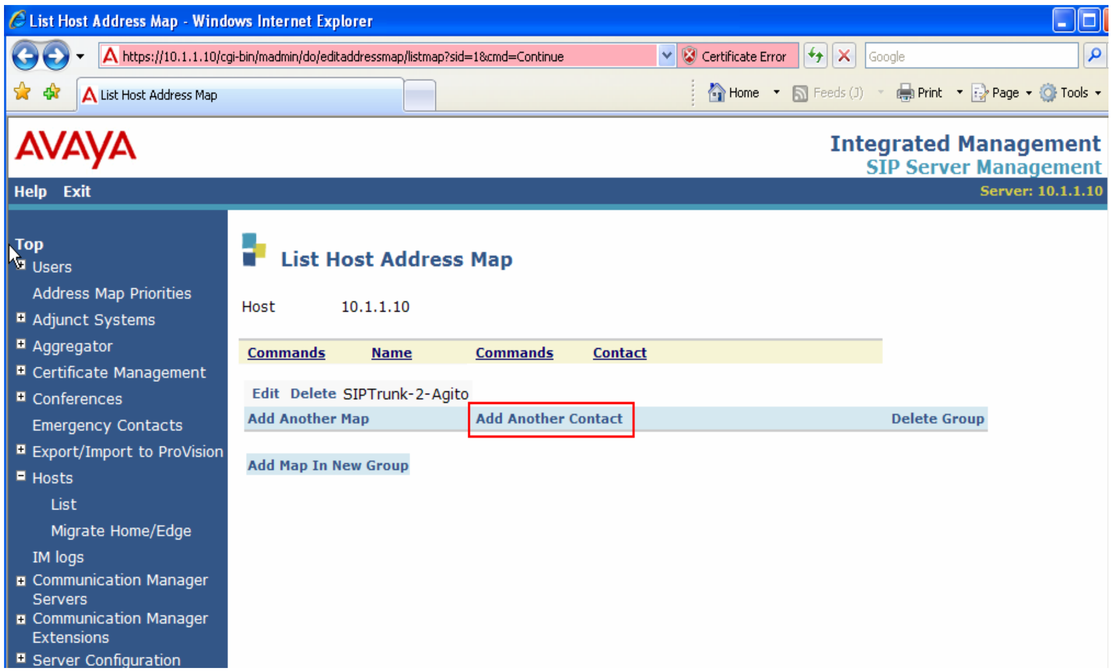
7.	The **List Host Address** Map screen is redisplayed, showing the newly added item. On the **List Host Address** Map screen, define the contact address for the Agito by clicking on **Add Another Contact** on the line below SIPTrunk-2-Agito. 

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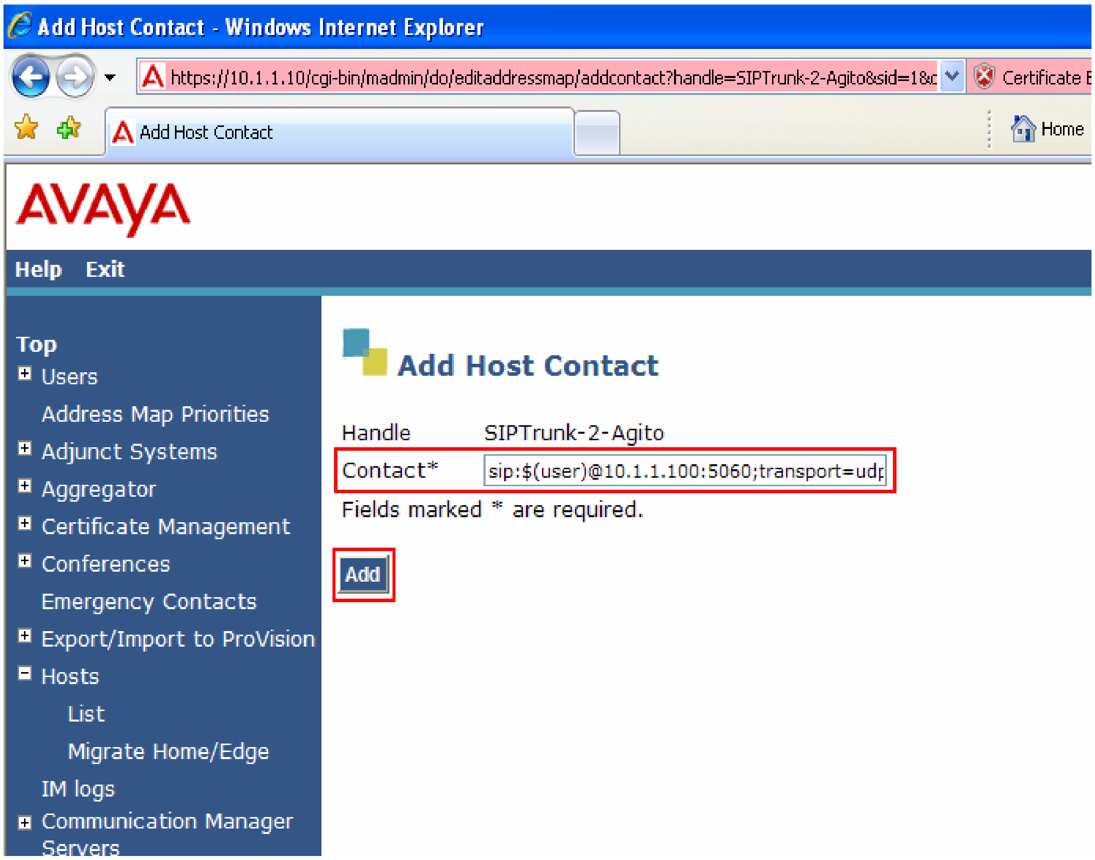
8.	The **Add Host Contact** screen is displayed. The Contact field specifies the destination for the call. Populate the **Contact** field with the service IP address of the Agito that the SES should substitute into the required URI before sending a message to the Agito. The Avaya SES replaces **$(user)** with the user portion of the request URI before sending the message. Click the **Add** button.



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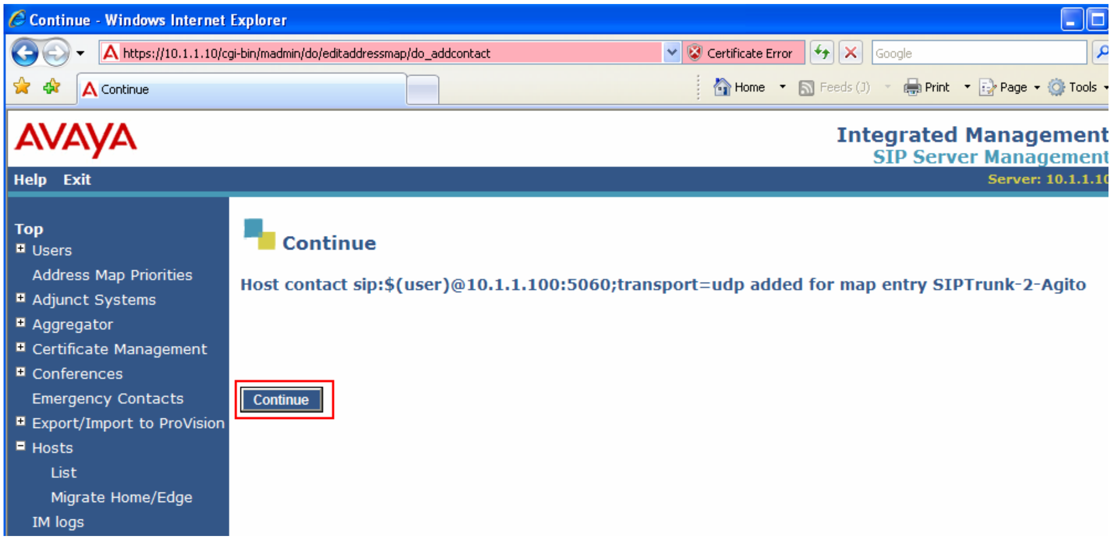
9. A **Continue** screen is displayed to confirm the addition. Click the **Continue** button.



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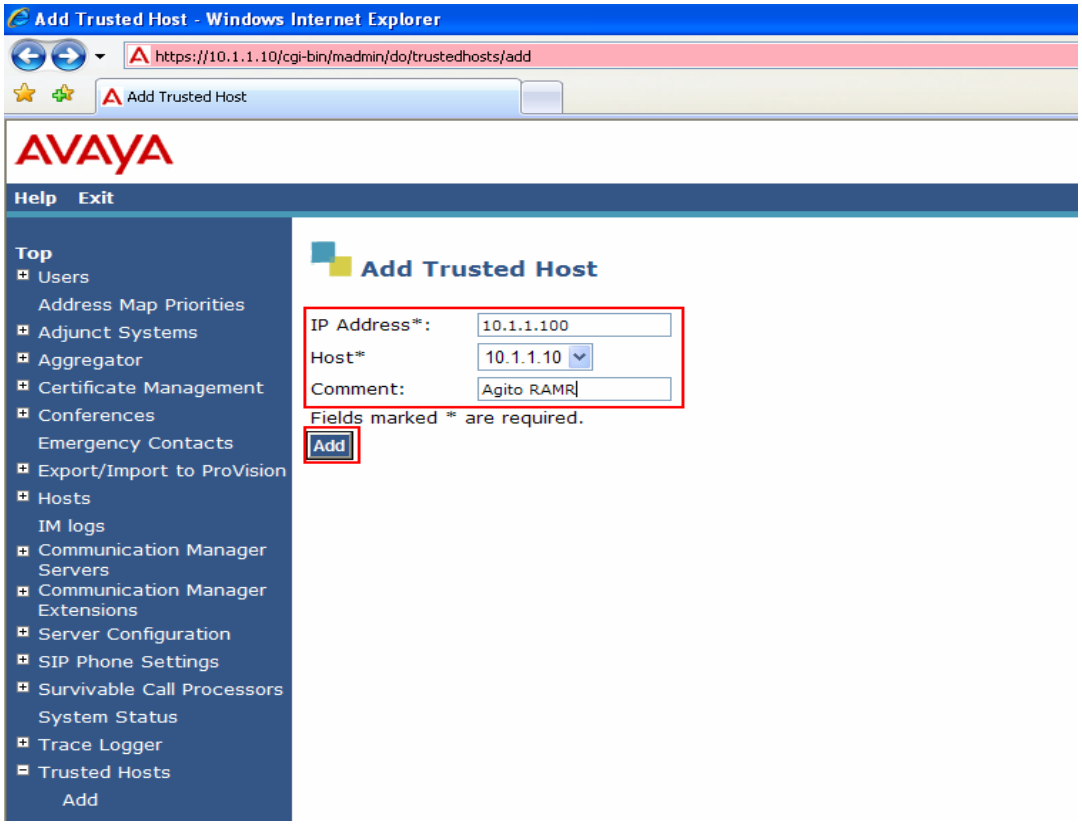
10.	Administer the Agito as a trusted host so that the SES will not challenge SIP messages from the Agito. From the left pane, select **Trusted Hosts → Add Trusted Host**. Enter the IP address of the Agito Mobility Router , click **Add** to continue. 

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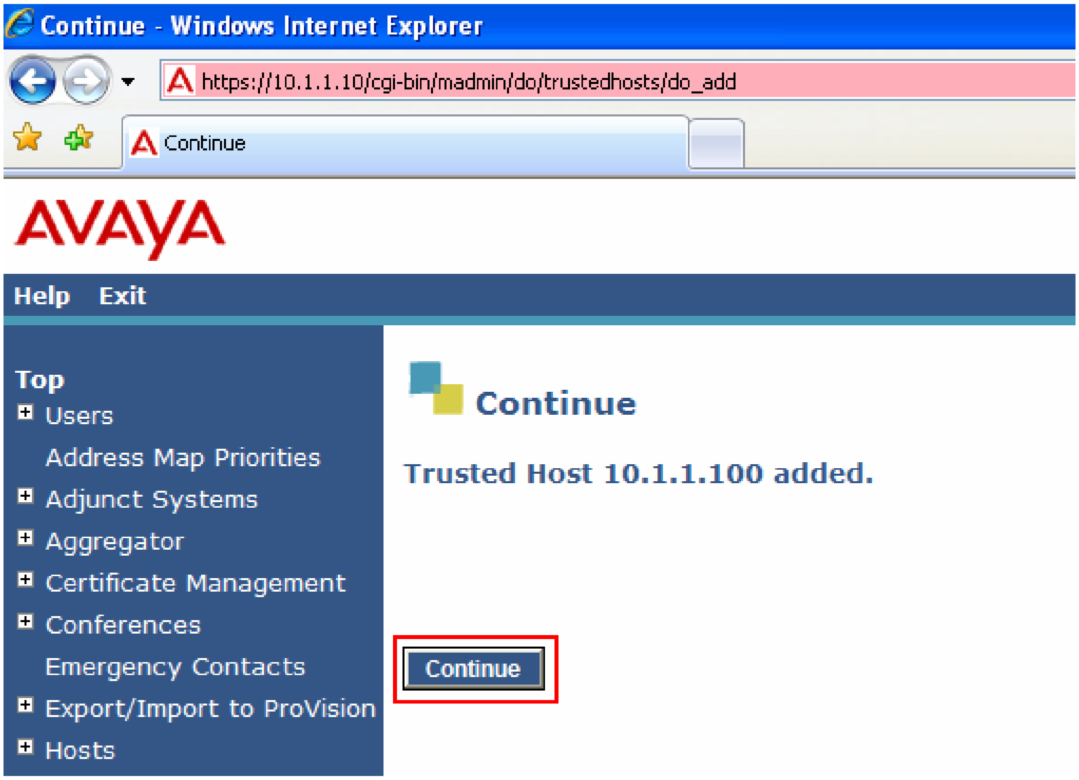
11.	A **Continue** screen is displayed to confirm the addition. Click the **Continue** button. 

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5. Configure the Agito RoamAnywhere Mobility Router

The following configuration steps outline the required settings to enable the Agito RoamAnywhere Mobility Router to interoperate with the Avaya telephony infrastructure.

Step	Description
1.	Refer to RoamAnywhere Quick Start document to perform basic system configuration settings (IP, password) from the serial console port. Access the administrator interface at: **https://<ramr-ip-address>/adm** Follow the configuration steps shown in part 1 of the RoamAnywhere Administrator's Guide, references 5 thru 7 in Section 9

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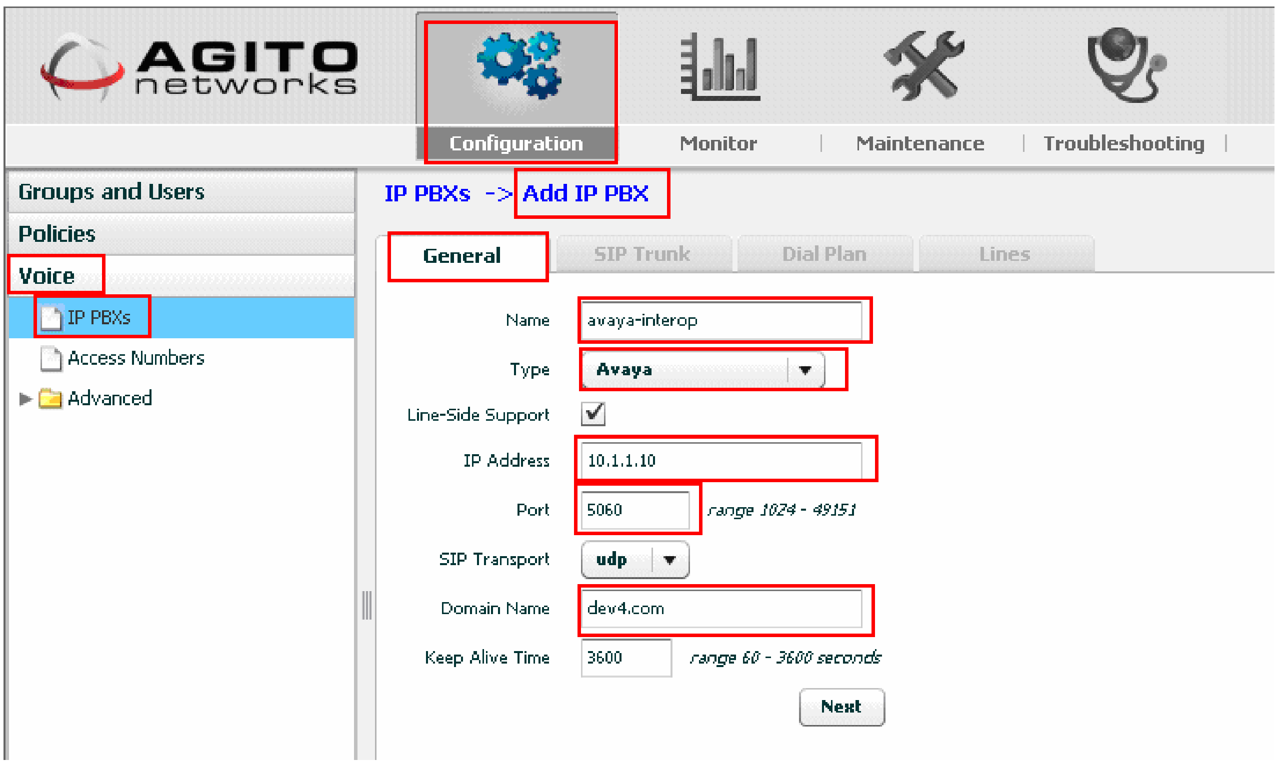
Step	Description
2.	From the **Configuration** tab, select **Voice → IP PBXs → Add IP PBX → General,** enter the settings for the PBX similar to the configuration below. Ensure the IP address and domain name match the Avaya SES and the domain name deployment in steps 3.1 and 3.2.. Set the following: • **Name** to **Avaya-interop** • **Type** to **Avaya** • **IP Interface to 10.1.1.10** • **Port** to **5060** • **Domain Name** to **dev4.com** Select **Next** to continue. 

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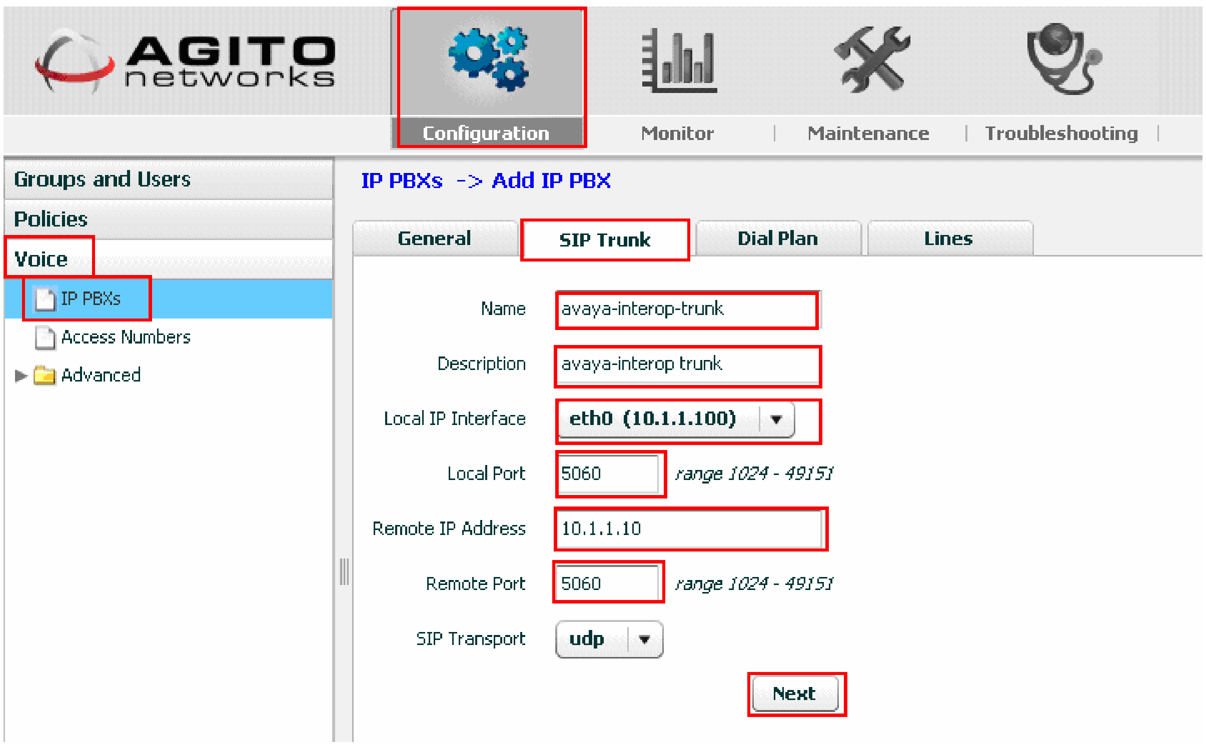
Step	Description
3.	From the **Configuration** tab, select **Voice → IP PBXs → Add IP PBX → SIP Trunk**, Ensure the IP address is set to the IP of the Avaya SES. Set the following: • **Name** to **avaya-interop-trunk** • **Description** to **avaya-interop-trunk** • **Local IP Interface** to **eth0 (10.1.1.100)** • **Local Port** to **5060** • **Remote IP Address** to **10.1.1.10** Select **Next** to continue. 

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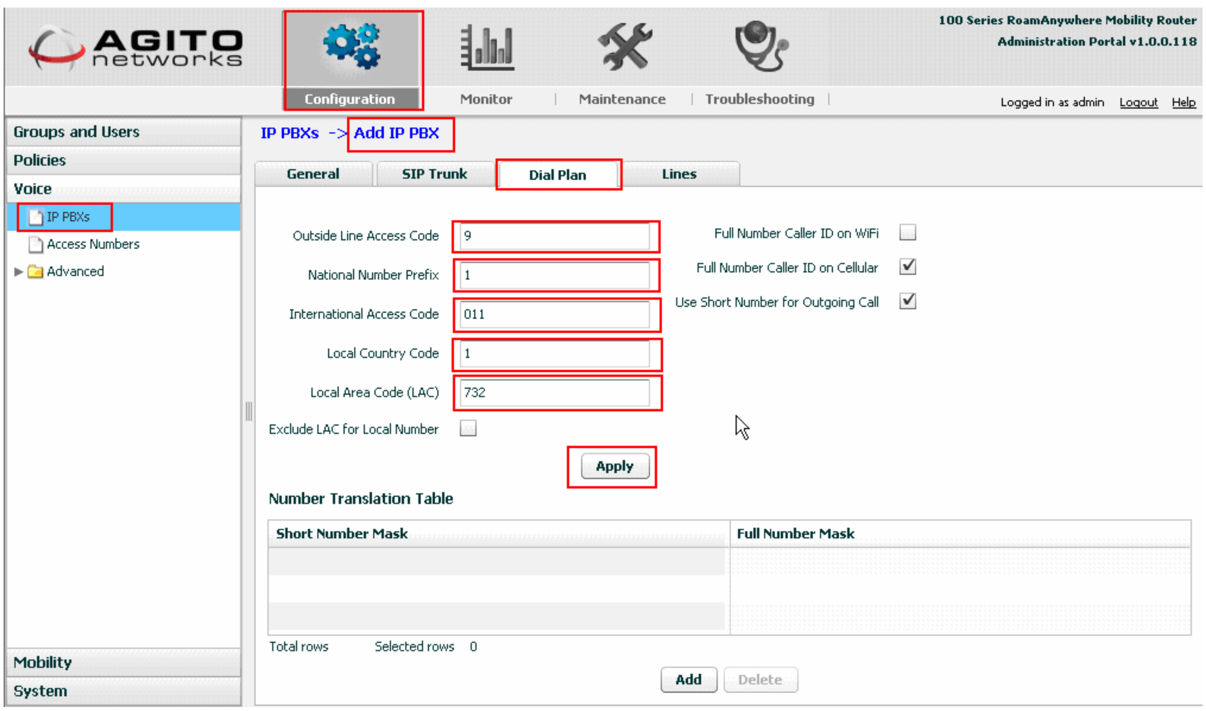
Step	Description
4.	From the **Configuration** tab, select **Voice → IP PBXs → Add IP PBX → Dial Plan,** set the Dial Plan to match that used by Avaya Communication Manager for routing calls. Ensure the Local Area Code (LAC) matches the enterprise location and whether or not the LAC is required for local external calls. • **Outside Line Access Code** to **9** • **National Number Prefix** to **1** • **International access Code** to **011** • **Local Country Code** to **1** • **Local Area Code** to **732** Select **Apply** to commit the settings and complete the addition of the IP PBX settings. 

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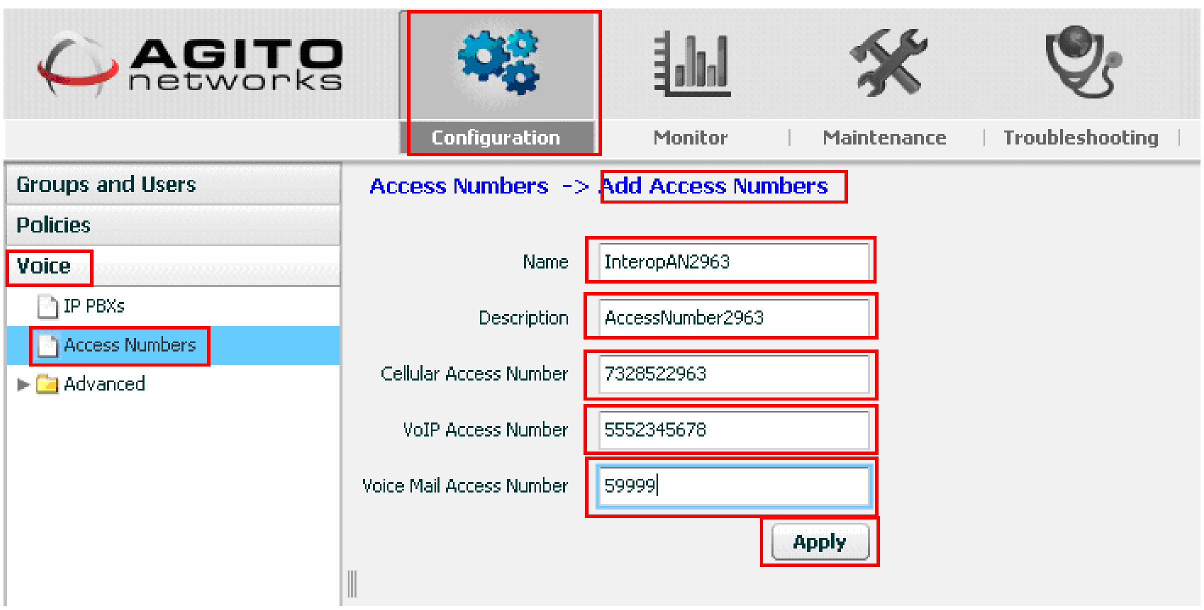
In order to be able to access the enterprise features and dial plan when outside the enterprise on the cellular network as well as activating enterprise voicemail, configure an access number for the incoming trunk connection. The **Cellular Access Number** should be a PSTN DID allocated to the enterprise that routes through the IP PBX to the SIP trunk terminating on the RoamAnywhere Mobility Router. The **VoIP Access Number** should be configured as a valid digit pattern in the IP PBX dial plan that terminates on the SIP trunk connected to the Mobility Router. Ensure that your **Voice Mail Access Number** is also configured to match the digits used within your enterprise.

5. From the **Configuration** tab, select **Voice → Access Numbers → Add Access Numbers,** set the follows options:

- **Name** to **InteropAN2963**
- **Description** to **AccessNumber2963**
- **Cellular Access Number** to **7328522963**
- **VoIP Access Number** to **5552345678**
- **Voice Mail Access Number to 59999**

Select **Apply** to commit the settings.



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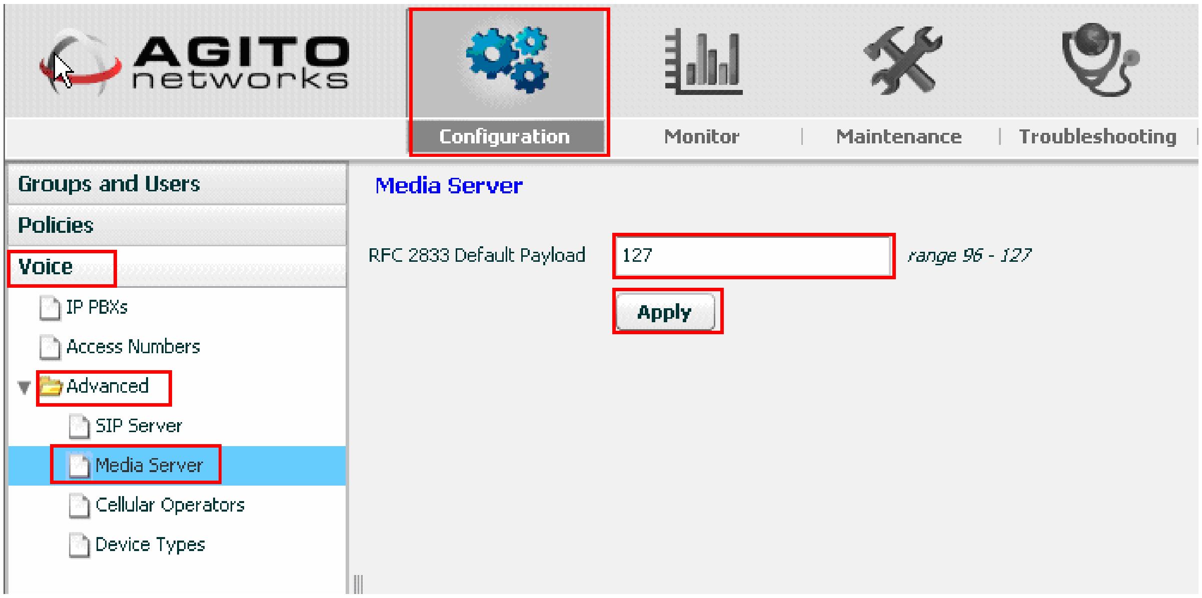
6.	From the **Configuration** tab, select **Voice → Advanced → Media Server.** Set the **RFC 2833 Default Payload** type to be "127" for DTMF processing of digits. Select **Apply** to commit the settings. 

5.1. 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.

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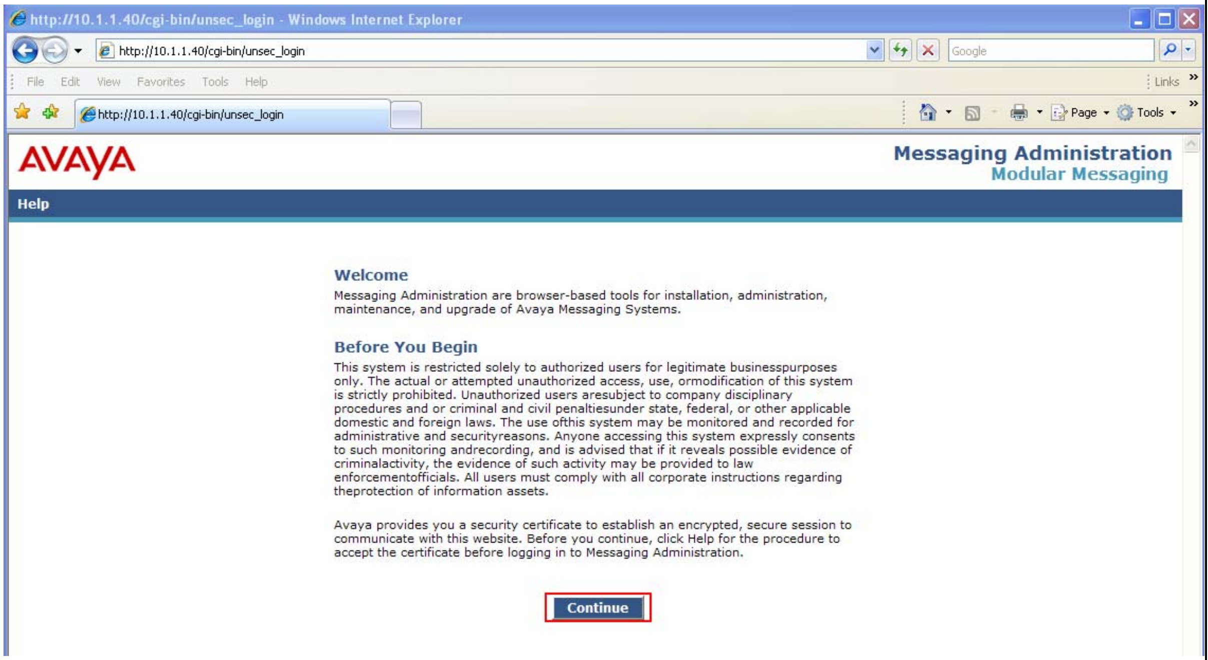
5.2. Configure Subscriber on Avaya Modular Messaging

<table>
<tr>
<td>1.</td>
<td>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.



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

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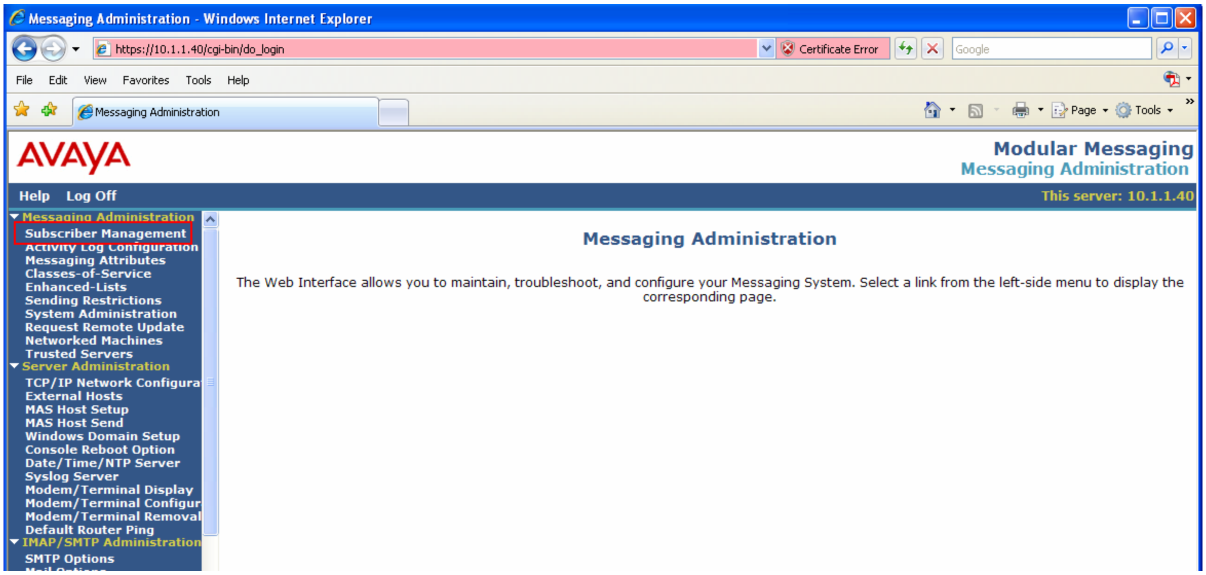
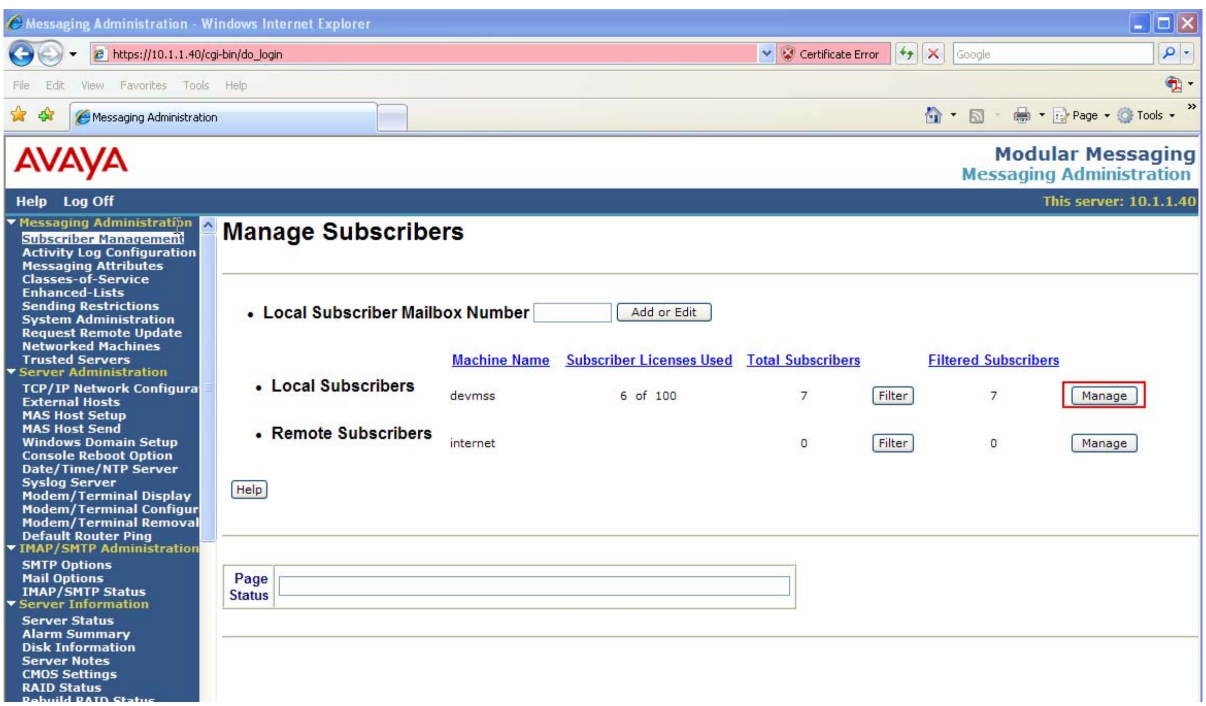
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.</td><td>Select Subscriber Management.

</td></tr>
<tr><td>3.</td><td>Select Manage aligned with Local Subscribers..

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

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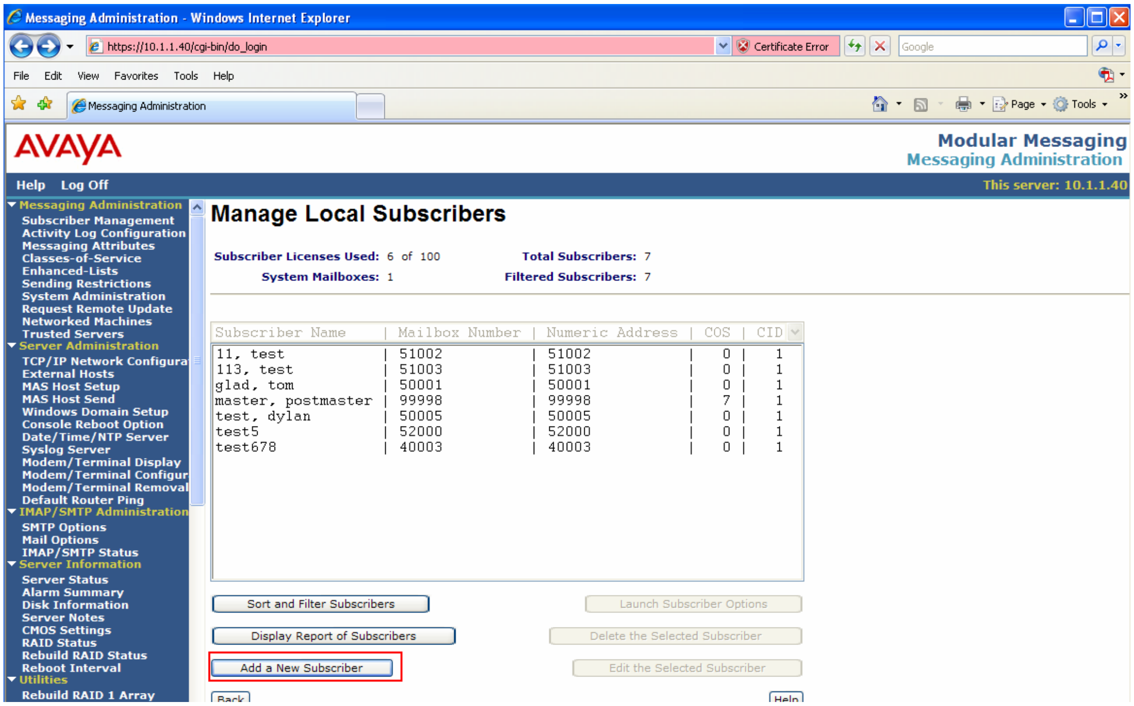
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>Select Add a New Subscriber.



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

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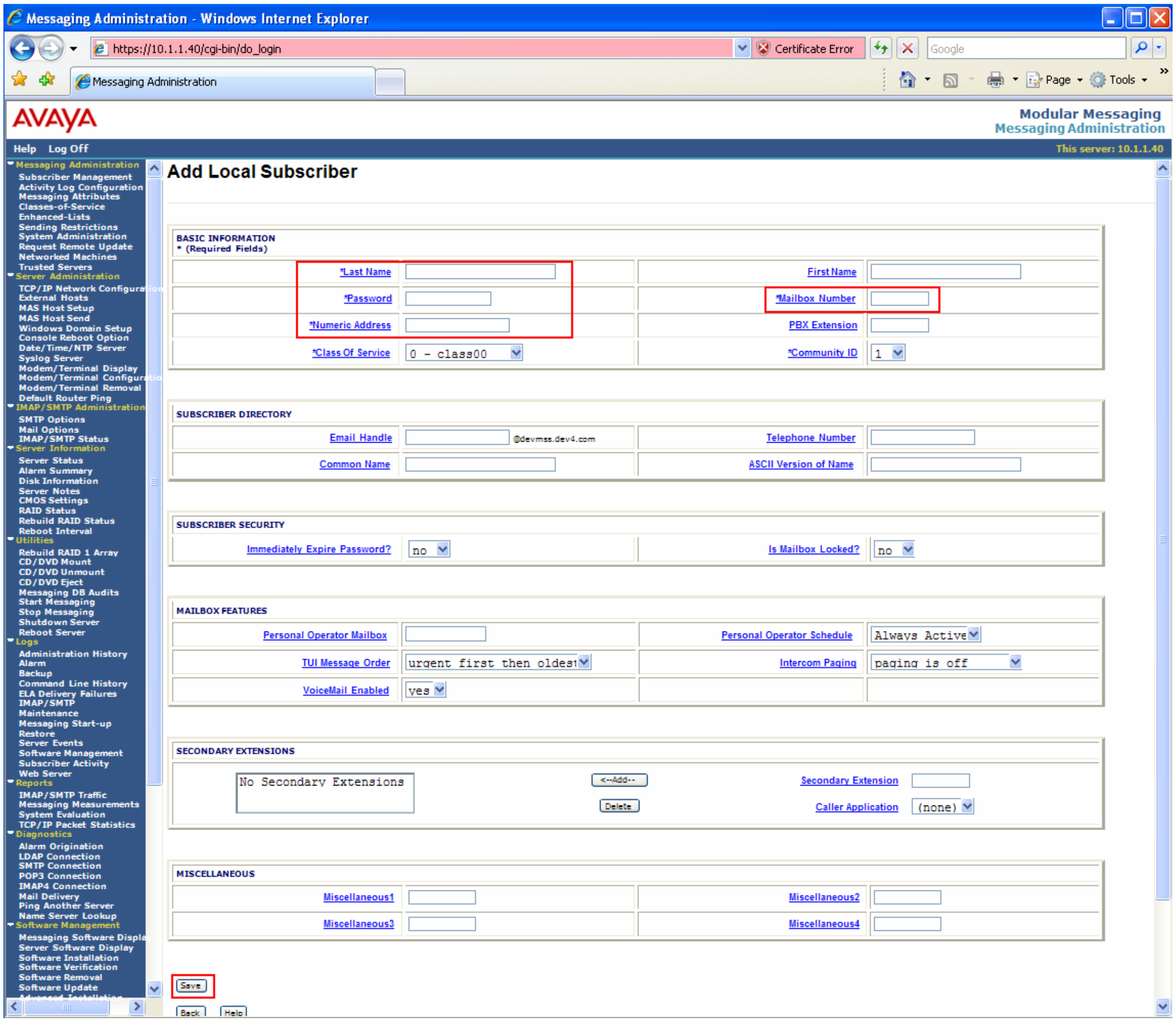
Step	Description
5.	Enter the Follow User Information: **Last Name, Password Mailbox Number, Numeric Address**. Select **Save** to continue. 

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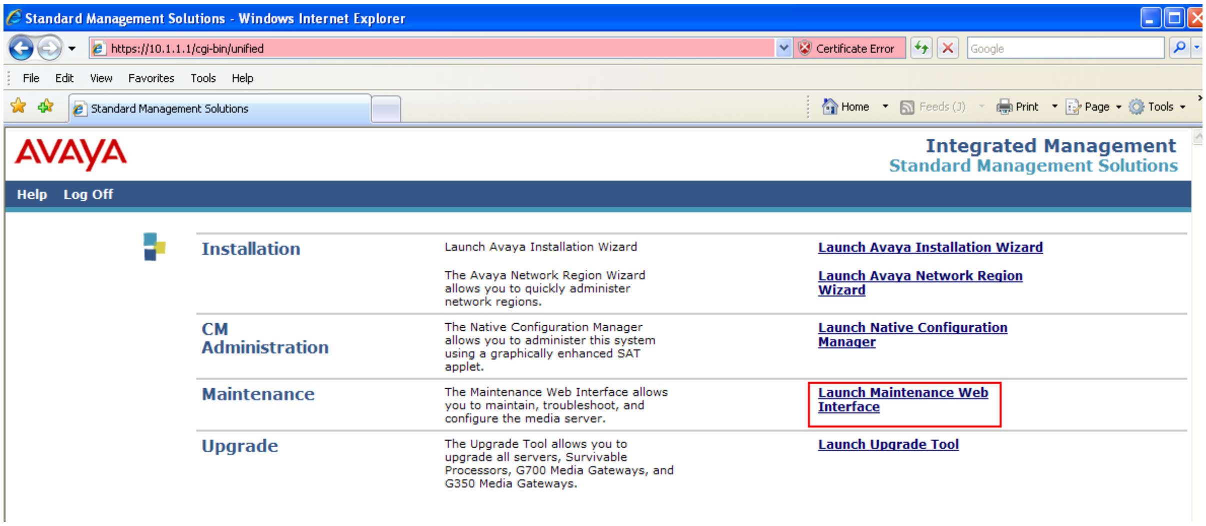
5.3. Configure Subscriber on Avaya IA770 INTUITY AUDIX

1. 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.



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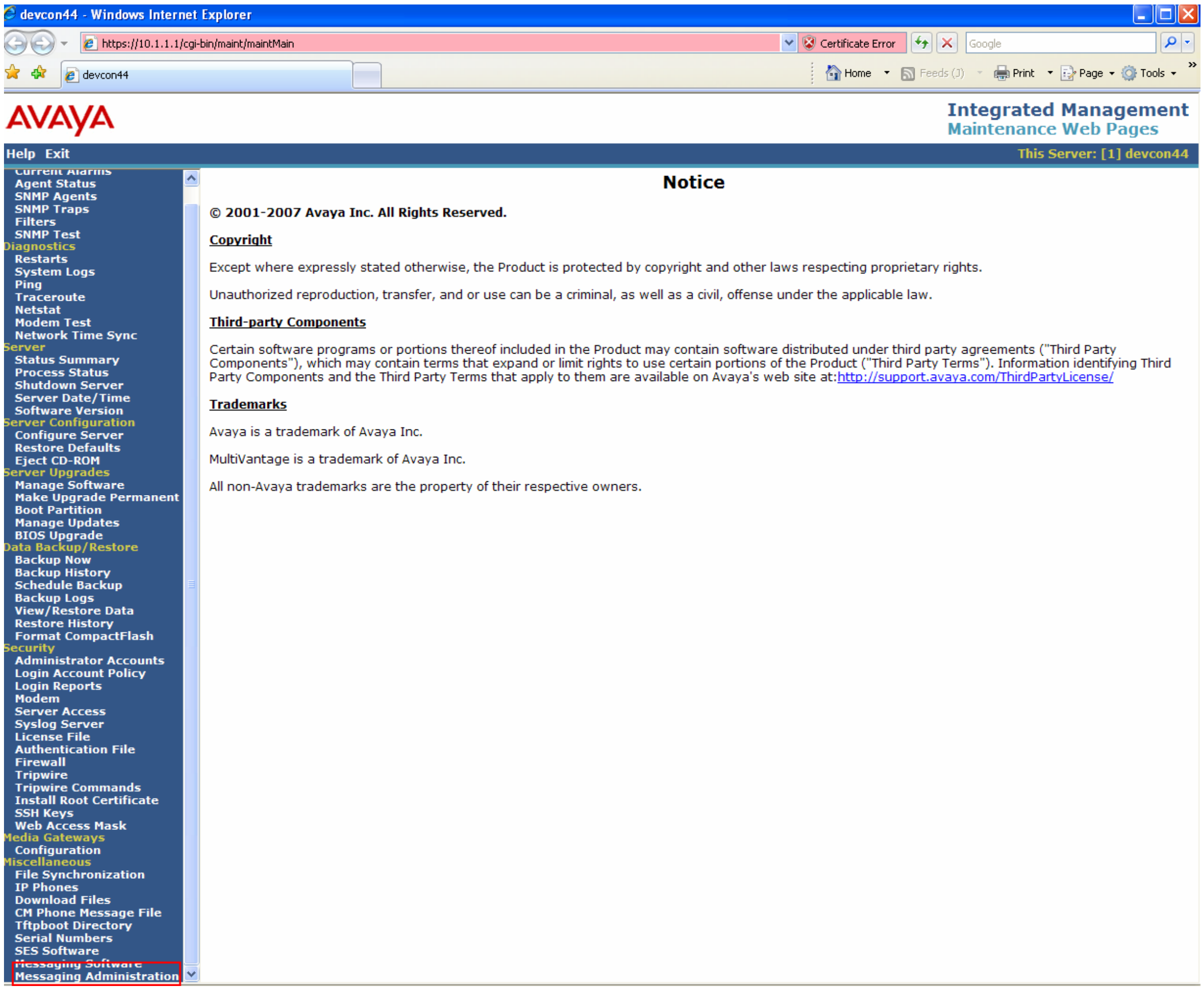
2. Click on **Messaging Administration.**



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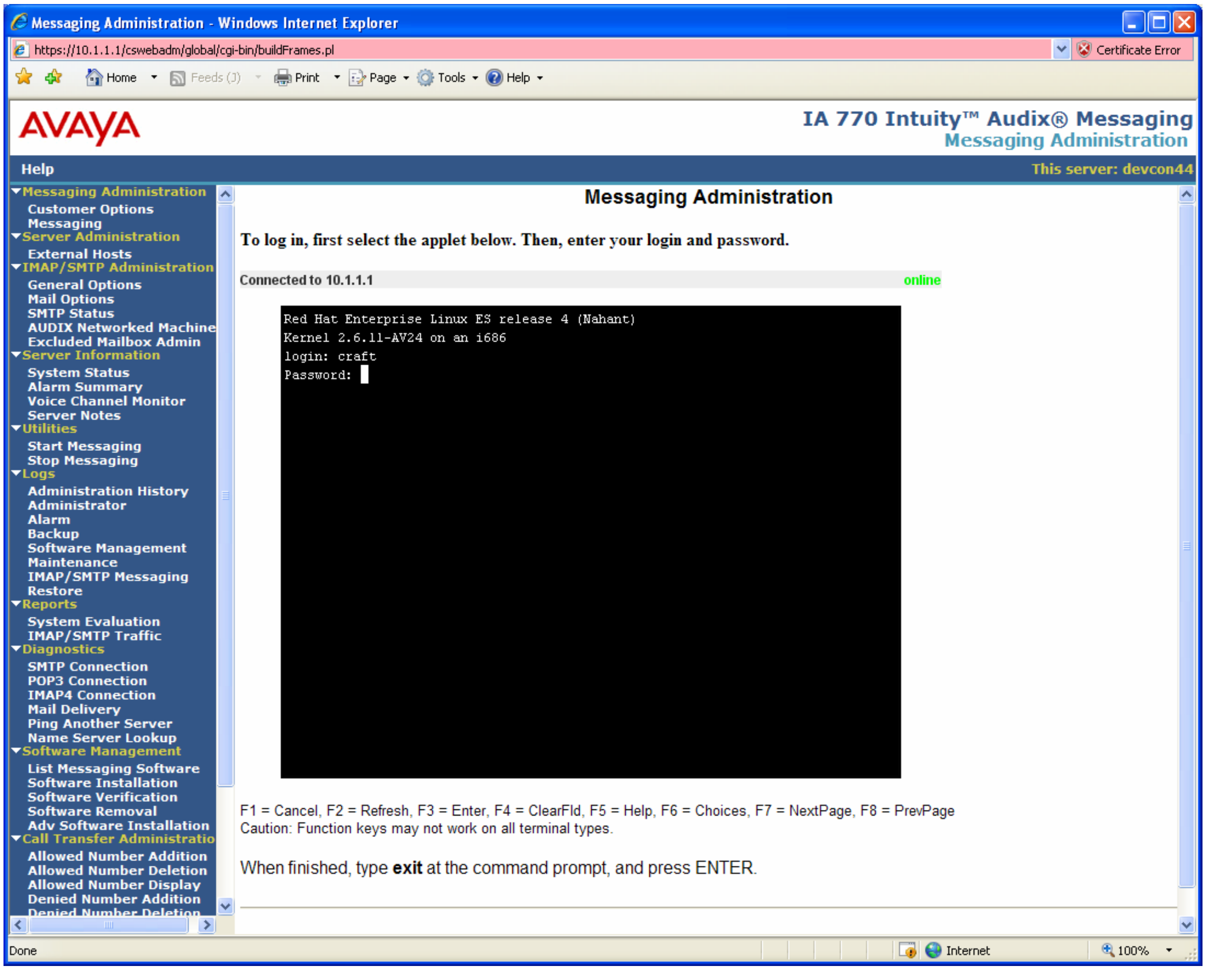
3.	Click on **Messaging.**
4.	Login with the appropriate Login and Password.



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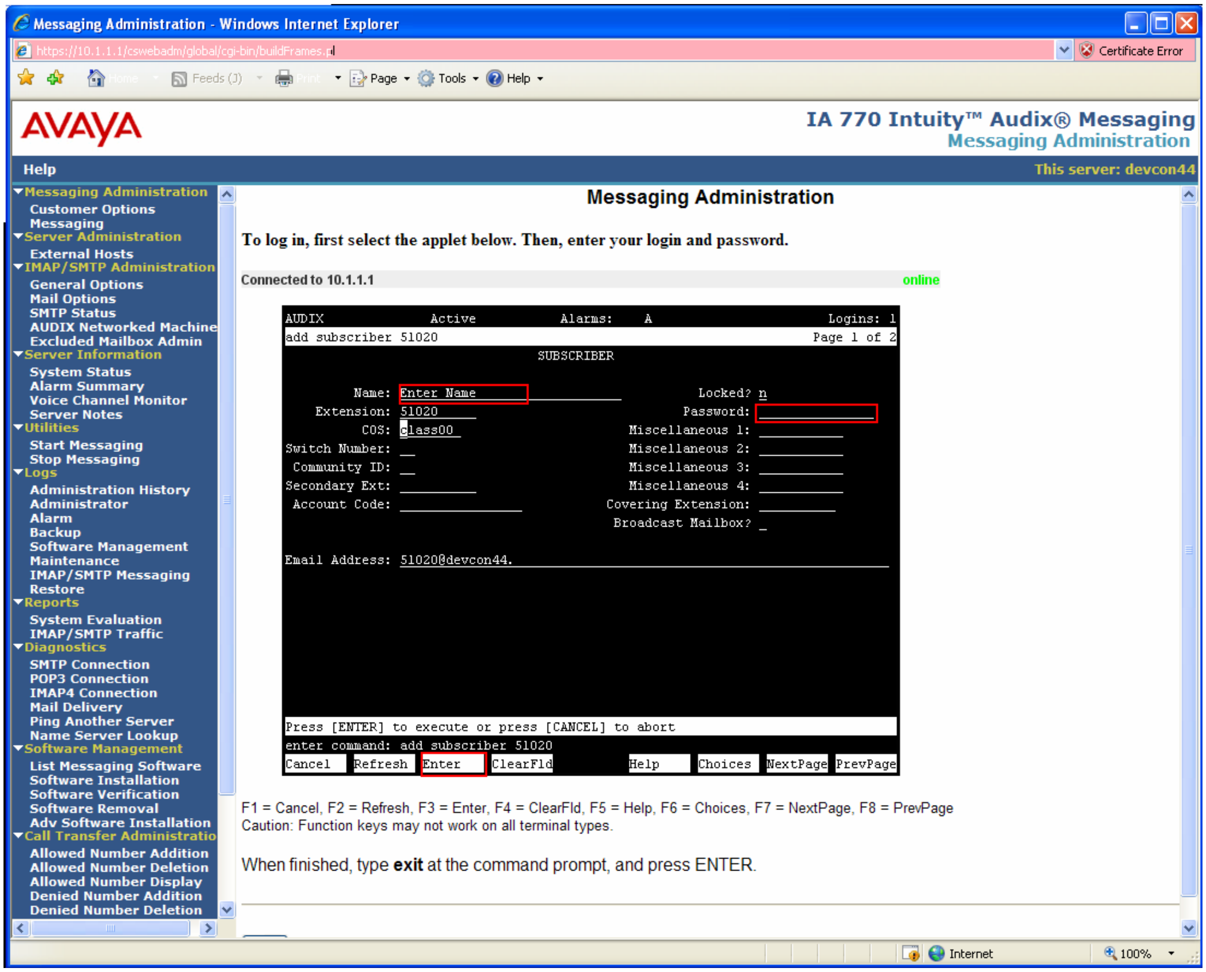
5.	At the prompt, Type, **add subscriber X** where **X** is the extension, and press **Enter**. Enter the following: **Name**, **Password,** (the Password will disappear after being entered). Click **Enter** to continue. 

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 an Avaya telephony infrastructure and Agito Networks RoamAnywhere Mobility Router and the ability for an enterprise user to be accessible via one business number whether the user is in the office or mobile.

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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 Agito 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 PSTN]
- mobile terminated calls routed through the Avaya telephony infrastructure
- desktop originated calls routed to mobile devices
- DTMF digit support for voicemail, conference and IVT calls
- Abbreviated Dialing
- Call Forward All
- Call Forward Cancel
- Call Hold / Unhold
- Send All Calls
- Send All Calls Cancel
- Shared Line Appearance
- Transfer
- Transfer To Desk
- Transfer On Hang-Up

6.2. Test Results

The test objectives of section 6.1 were verified. The Agito Networks RoamAnywhere Mobility Router successfully completed all test cases for the features identified in section 8.1. The Agito Networks RoamAnywhere Mobility Router 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 Agito Networks RoamAnywhere Mobility products, contact us one of three ways:

- Web site: http://www.agitonetworks.com/
- Email: support@agitonetworks.com
- Telephone: (408) 919-8000

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8. Conclusion

These Application Notes describe the configuration steps required for integrating the Agito Networks RoamAnywhere Mobility Router into an Avaya telephony infrastructure. For the configuration described in these Application Notes, the Agito Networks RoamAnywhere Mobility Router was responsible for bridging landline connectivity to an Avaya telephony infrastructure with the wireless connectivity to the GSM network. The functionality of the combined Avaya and Agito Networks solution was validated via the *DevConnect* 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] *Messaging Application Server (MAS) Administration Guide Release 3.1,* February 2007

[4] *Avaya one-X Deskphone Edition for 9600 Series IP Telephones Administrator Guide*

The Agito Networks product documentation can be found at:
http://www.agitonetworks.com.

[5] *RoamAnywhere Mobility Router Administrator's Guide, Version 1.0*, February 2008

[6] *2000 Series and 4000 Series RoamAnywhere Mobility Router Quick Start*, Version 1.0, February 2008.

[7] *RoamAnywhere User's Guide*, Version 1.0, February 2008.

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