



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Configuring the Agito Networks RoamAnywhere Mobility Router with an Avaya Telephony Infrastructure using Avaya Communication Manager - 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 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 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.

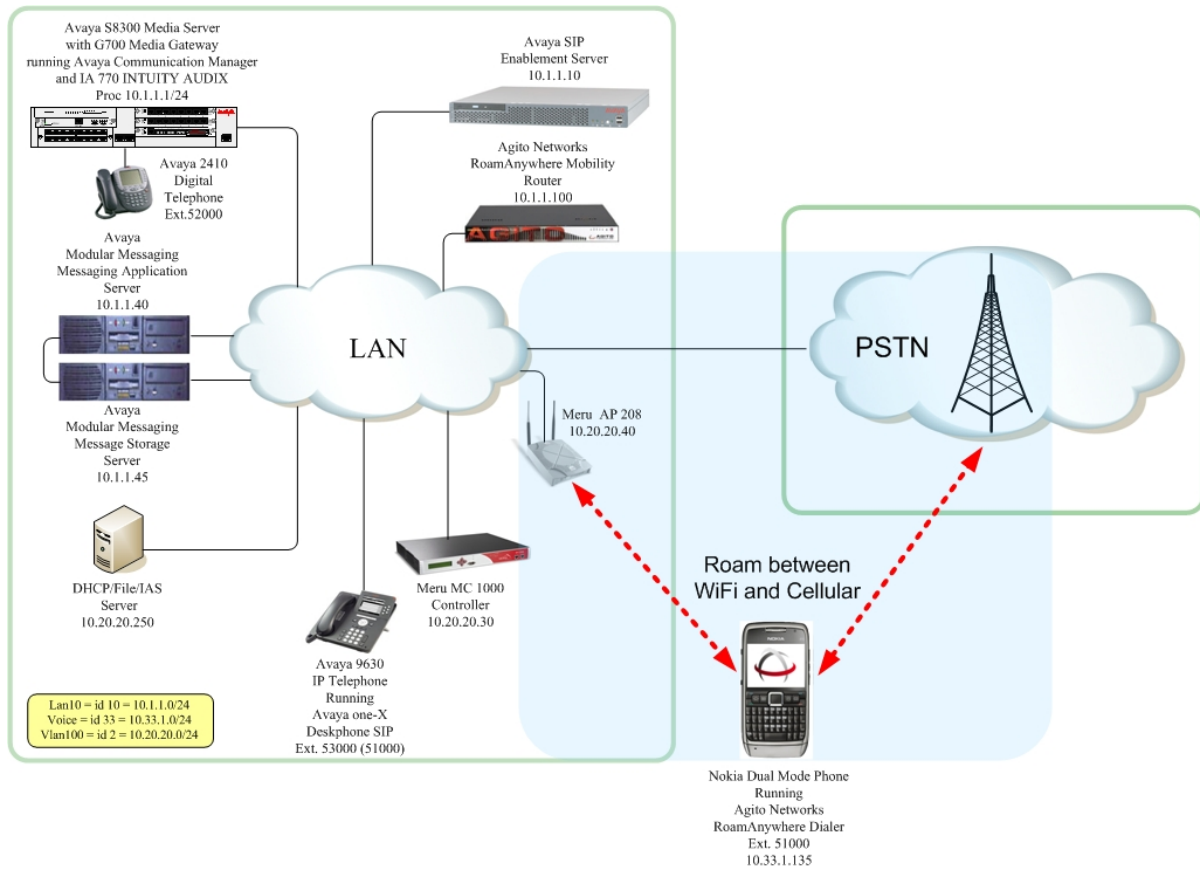
The Agito RoamAnywhere Solution uses a combination of SIP lines and trunks to integrate with Avaya Communication Manager. SIP lines are used so that Agito-controlled mobile devices appear as standard SIP phones and therefore benefit from the common set of PBX services offered to such devices. SIP trunks 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.

Additional Avaya endpoints were tested (see **Section 2**) but are not shown in **Figure 1**.



**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 5.1
Avaya G700 Media Gateway MGP MM712 DCP Media Module	26.31.0 HW05 / FW08
Avaya SIP Enablement Services (SES) Server	5.1
Avaya Modular Messaging - Messaging Application Server (MAS)	4.0
Avaya Modular Messaging - Message Storage Server (MSS)	4.0
Avaya IA 770 INTUITY AUDIX	5.1
Avaya 9600 Series IP Telephones	Avaya one-X Deskphone SIP 2.0.3
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	5.0
Agito Networks RoamAnywhere Mobility Router	2.0.0.137
Agito Networks RoamAnywhere Mobility Dialer	2.0.2.48
Nokia N95 (Dual Mode handset)	<a href="#">V20.0.013</a>
Nokia E71 (Dual Mode handset)	100.07.76

## 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.

### 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 <b>change ip-codec-set g</b> command, where “g” is a number between 1 and 7, inclusive, and enter “ <b>G.711MU</b> ” for <b>Audio Codec</b> . 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 Codec	Silence Suppression
1: <b>G.711MU</b>	n
2:	
Frames Per Pkt	Packet Size(ms)
2	20

### 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 <b>change node-names ip</b> command, On page 1 of the <b>change node-names ip</b> form, enter the name for the SES, “ <b>SES</b> ”, and enter the IP address of the SES, “ <b>10.1.1.10</b> ”.	
change node-names ip	Page 1 of 2
IP NODE NAMES	
Name	IP Address
AES-DevCon2	192.45.100.153
<b>SES</b>	<b>10.1.1.10</b>
default	0.0.0.0
mm	10.1.1.45
msgserver	10.1.1.20
procr	10.1.1.1

### 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 <b>change ip-network-region h</b> command, where “h” is a number between 1 and 250, inclusive. On page 1 of the <b>ip-network-region</b> form, set <b>Codec Set</b> 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	
<b>Codec Set: 1</b>	
UDP Port Min: 2048	
UDP Port Max: 3329	
DIFFSERV/TOS PARAMETERS	
Call Control PHB Value: 46	
Audio PHB Value: 46	
Video PHB Value: 26	
Intra-region IP-IP Direct Audio: yes	
Inter-region IP-IP Direct Audio: yes	
IP Audio Hairpinning? n	
RTCP Reporting Enabled? y	
RTCP MONITOR SERVER PARAMETERS	
Use Default Server Parameters? y	

### 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.

### 3.5. Create SIP Trunk with Domain Information

Step	Description
1.	<p>Enter the <b>add trunk-group i</b> command, where “i” is an available trunk group number. On Page 1 of the <b>trunk-group</b> form, configure the following:</p> <ul style="list-style-type: none"> <li>• <b>Group Type</b> – set to “<b>sip</b>”</li> <li>• <b>Group Name</b> – enter a meaningful name/description.</li> <li>• <b>TAC</b> – enter a Trunk Access Code that is valid under the provisioned dial plan.</li> <li>• <b>Service Type</b> – set to “<b>tie</b>”</li> </ul>
	<pre> 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 </pre>

Step	Description
2.	<p>Enter the <b>add signaling group j</b> command, where “j” is an available signaling group number. On <b>Page 1</b> of the <b>signaling-group</b> form, configure the following:</p> <ul style="list-style-type: none"> <li>• <b>Group Type</b> – set to “<b>sip</b>”</li> <li>• <b>Transport Method</b> – set to “<b>tcp</b>”</li> <li>• <b>Near-end Node Name</b> – enter the node name of a local C-LAN board, or “<b>procr</b>” if the local node is an Avaya S8300 Server.</li> <li>• <b>Near-end Listen Port</b> – specify the local listen port, typically <b>5060</b>.</li> <li>• <b>Far-end Node Name</b> – enter the node name of the SES configured in <b>Section 3.2</b></li> <li>• <b>Far-end Listen Port</b> – specify the local listen port, typically <b>5060</b>.</li> <li>• <b>Far-end Domain</b> – <b>dev4.com</b></li> <li>• <b>Far-end Network Region</b> – enter the IP network region configured in <b>Section 3.3</b></li> <li>• <b>DTMF over IP</b> – set to “<b>rtp-payload</b>”.</li> <li>• <b>Direct IP-IP Audio Connections</b> – set to “<b>y</b>”.</li> </ul>
	<pre> 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 Domain: dev4.com       Far-end Network Region: 1                                  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 </pre>



Step	Description
3.	<p>Enter the <b>change trunk-group i</b> command, where “i” is the number of the trunk group configured in <b>Step 3.5.1</b>. On <b>Page 1</b> of the <b>trunk-group</b> form, set configure the following:</p> <ul style="list-style-type: none"> <li>• <b>Signaling Group</b> – enter the Signaling Group number that was used in <b>Step 3.5.2</b>.</li> <li>• <b>Number of Members</b> – set to <b>24</b></li> </ul>
	<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>

### 3.6. Create SIP Trunk without Domain Information

Step	Description
1.	<p>Enter the <b>add trunk-group g</b> command, where “g” is an available trunk group number. On <b>Page 1</b> of the <b>trunk-group</b> form, configure the following:</p> <ul style="list-style-type: none"><li>• <b>Group Type</b> – set to “<b>sip</b>”</li><li>• <b>Group Name</b> – enter a meaningful name/description.</li><li>• <b>TAC</b> – enter a Trunk Access Code that is valid under the provisioned dial plan.</li><li>• <b>Service Type</b> – set to “<b>tie</b>”</li></ul>
	<pre>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     Queue Length: 0   Service Type: tie              Auth Code? n                                     Signaling Group:                                    Number of Members: 0</pre>

Step	Description
2.	<p>Enter the <b>add signaling group m</b> command, where “<b>m</b>” is an available signaling group number. On <b>Page 1</b> of the <b>signaling-group</b> form, configure the following:</p> <ul style="list-style-type: none"> <li>• <b>Group Type</b> – set to “<b>sip</b>”</li> <li>• <b>Transport Method</b> – set to “<b>tcp</b>”</li> <li>• <b>Near-end Node Name</b> – enter the node name of a local C-LAN board, or “<b>procr</b>” if the local node is an Avaya S8300 Server.</li> <li>• <b>Near-end Listen Port</b> – specify the local listen port, typically <b>5060</b>.</li> <li>• <b>Far-end Domain</b> – <b>10.1.1.100</b></li> <li>• <b>Far-end Node Name</b> – enter the node name of the SES configured in <b>Section 3.2</b></li> <li>• <b>Far-end Listen Port</b> – specify the local listen port, typically <b>5060</b>.</li> <li>• <b>Far-end Network Region</b> – enter the IP network region configured in <b>Section 3.3</b></li> <li>• <b>DTMF over IP</b> – set to “<b>rtp-payload</b>”.</li> <li>• <b>Direct IP-IP Audio Connections</b> – set to “<b>y</b>”.</li> </ul>
	<pre> add signaling-group 98                                      SIGNALING GROUP                                      Page 1 of 1  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 Domain: 10.1.1.100    Far-end Network Region: 1                            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 </pre>

Step	Description
3.	<p>Enter the <b>change trunk-group i</b> command, where “i” is the number of the trunk group configured in <b>Step 3.6.1</b>. On <b>Page 1</b> of the <b>trunk-group</b> form, set configure the following:</p> <ul style="list-style-type: none"> <li>• <b>Signaling Group</b> – enter the Signaling Group number that was used in <b>Step 3.6.2</b>.</li> <li>• <b>Number of Members</b> – set to <b>24</b></li> </ul>
	<pre> 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 </pre>

### 3.7. Configure Off-PBX Telephone Integration Information

Step	Description																								
1.	<p>Every user must be defined as an off-PBX station in order to enable simultaneous ringing. To do this, go to the <b>Stations with Off-PBX Telephone Integration</b> screen and map the Avaya Communication Manager extension to the extension defined in the SES.</p> <p>Enter <b>change off-pbx-telephone station-mapping n</b>, where <b>n</b> is the number of the phone extension where a mobile extension shall be configured. Enter the following information:</p> <ul style="list-style-type: none"><li>• <b>Station Extension = n</b></li><li>• <b>Application = OPS</b></li><li>• <b>Phone Number = Phone Number of the new Extension</b></li><li>• <b>Trunk Selection = Trunk used to the SES</b></li><li>• <b>Configuration Set = 1</b></li></ul> <p><b>Go to page 2</b></p>																								
	<div>change off-pbx-telephone station-mapping 51000<div>STATIONS WITH OFF-PBX TELEPHONE INTEGRATION</div></div> <div>Page 1 of 2</div> <table><tr><th>Station Extension</th><th>Application</th><th>Dial Prefix</th><th>Phone Number</th><th>Trunk Selection</th><th>Configuration Set</th></tr><tr><td>51000</td><td>OPS</td><td>-</td><td>51000</td><td>1</td><td>1</td></tr><tr><td></td><td></td><td>-</td><td></td><td></td><td></td></tr><tr><td></td><td></td><td>-</td><td></td><td></td><td></td></tr></table>	Station Extension	Application	Dial Prefix	Phone Number	Trunk Selection	Configuration Set	51000	OPS	-	51000	1	1			-						-			
Station Extension	Application	Dial Prefix	Phone Number	Trunk Selection	Configuration Set																				
51000	OPS	-	51000	1	1																				
		-																							
		-																							

Step	Description																		
2.	Change the following:																		
	<ul style="list-style-type: none"><li>• <b>Mapping Mode = both</b></li><li>• <b>Bridged Calls = both</b></li></ul>																		
	change off-pbx-telephone station-mapping 51000																		
	STATIONS WITH OFF-PBX TELEPHONE INTEGRATION																		
	<table><tr><td>Station</td><td>Call</td><td>Mapping</td><td>Calls</td><td>Bridged</td><td>Location</td></tr><tr><td>Extension</td><td>Limit</td><td>Mode</td><td>Allowed</td><td>Calls</td><td></td></tr><tr><td>51000</td><td>4</td><td><b>both</b></td><td>all</td><td><b>both</b></td><td></td></tr></table>	Station	Call	Mapping	Calls	Bridged	Location	Extension	Limit	Mode	Allowed	Calls		51000	4	<b>both</b>	all	<b>both</b>	
Station	Call	Mapping	Calls	Bridged	Location														
Extension	Limit	Mode	Allowed	Calls															
51000	4	<b>both</b>	all	<b>both</b>															

### 3.8. Configure Station Information for SIP Desktop.

This step is required if the desktop Avaya IP Telephone is a SIP station. Because Avaya SES will only allow one SIP endpoint to register at a time, another station ID needs to be created. To keep the button appearance consistent on both the Agito handset and the Avaya SIP desktop, the Agito endpoint will login into the SES as the primary phone number and the Avaya IP telephone (SIP) will login using the secondary phone number. For this example, station 51000 is the primary number and 53000 is the secondary number. There are no special settings for station 51000 so the configuration will not be shown.

Step	Description
1.	<p>Enter <b>change station 53000</b>, Enter the following information:</p> <ul style="list-style-type: none"><li>• <b>Station Extension = 53000</b></li><li>• <b>Type = 9620</b></li><li>• <b>Name = User Name</b></li><li>• <b>Message Lamp Ext: = 51000</b></li></ul> <p><b>Go to page 4</b></p> <div>change station 53000<div>STATION<div>Extension: 53000Type: 9620Port: S00014Name: User Name</div><div>Lock Messages? nSecurity Code: 123456Coverage Path 1: 99Coverage Path 2:Hunt-to Station:</div><div>BCC: 0TN: 1COR: 1COS: 1</div></div><div>STATION OPTIONS<div>Loss Group: 19Speakerphone: 2-wayDisplay Language: englishSurvivable GK Node Name:Survivable COR: internalSurvivable Trunk Dest? y</div><div>Time of Day Lock Table:Personalized Ringing Pattern: 1Message Lamp Ext: 51000Mute Button Enabled? yExpansion Module? nMedia Complex Ext:IP SoftPhone? nIP Video? n</div><div>Customizable Labels? y</div></div><div>Page 1 of 6</div></div>

Step	Description
2.	<p>Change the <b>BUTTON ASSIGNMENTS</b> to use the primary station number, <b>51000</b>.</p> <p>1: brdg-appr B:1 E:51000  2: brdg-appr B:2 E:51000  3: brdg-appr B:3 E:51000</p> <hr/> <div> <div>change station 53000</div> <div>STATION</div> <div> <div>SITE DATA</div> <div>Room: Headset? n</div> <div>Jack: Speaker? n</div> <div>Cable: Mounting: d</div> <div>Floor: Cord Length: 0</div> <div>Building: Set Color:</div> </div> <div>ABBREVIATED DIALING</div> <div> <div>List1:</div> <div>List2:</div> <div>List3:</div> </div> <div> <div>BUTTON ASSIGNMENTS</div> <div> <div>1: brdg-appr B:1 E:51000 4:</div> <div>2: brdg-appr B:2 E:51000 5:</div> <div>3: brdg-appr B:3 E:51000 6:</div> </div> </div> </div>

### 3.9. Dial Plan

This section describes the steps for setting the 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.	<p>To handle the incoming calls to the Agito Mobility Router the dial string need to be altered. This is done with the <b>change inc-call-handling-trmt trunk-group j</b> command, where “j” is the trunk group going out to the PSTN. 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.</p> <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 tie          11  1732555      6 <b>tie</b>         11  <b>17328522963</b>    1   3 tie </pre>

Step	Description
2.	<p>Use the <b>change aar analysis</b> command to add an AAR entry for the Agito Mobility Router.</p> <pre> 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      Req'd <b>732582963</b>    10  10    <b>24</b>      <b>aar</b>      <b>n</b>                                      n                                      n </pre>



Step	Description
3.	<p>Use the <b>change route-pattern</b> command to associate a route pattern to the SIP trunk which is used to access the Agito Mobility Router.</p> <pre> change route-pattern 24 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 2: 3: none </pre> <p>Page 1 of 3</p> <p>n user n user n user</p>

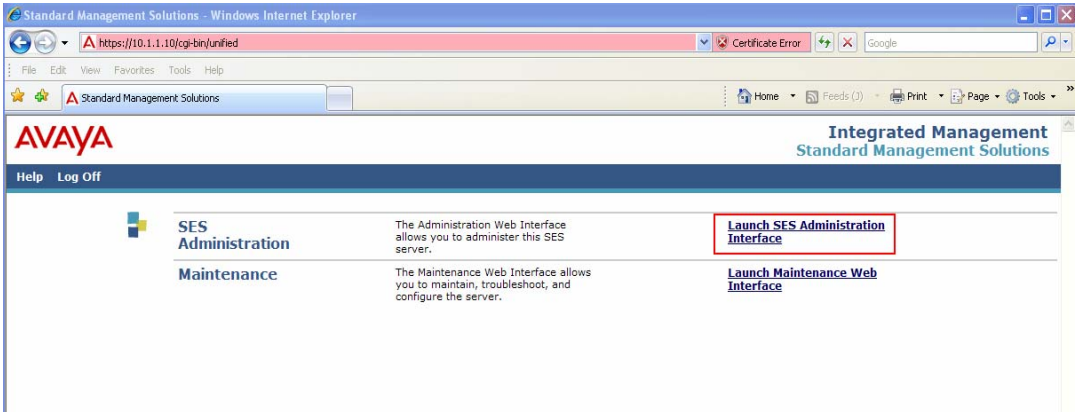
## 4. Configure Avaya SIP Enablement Services

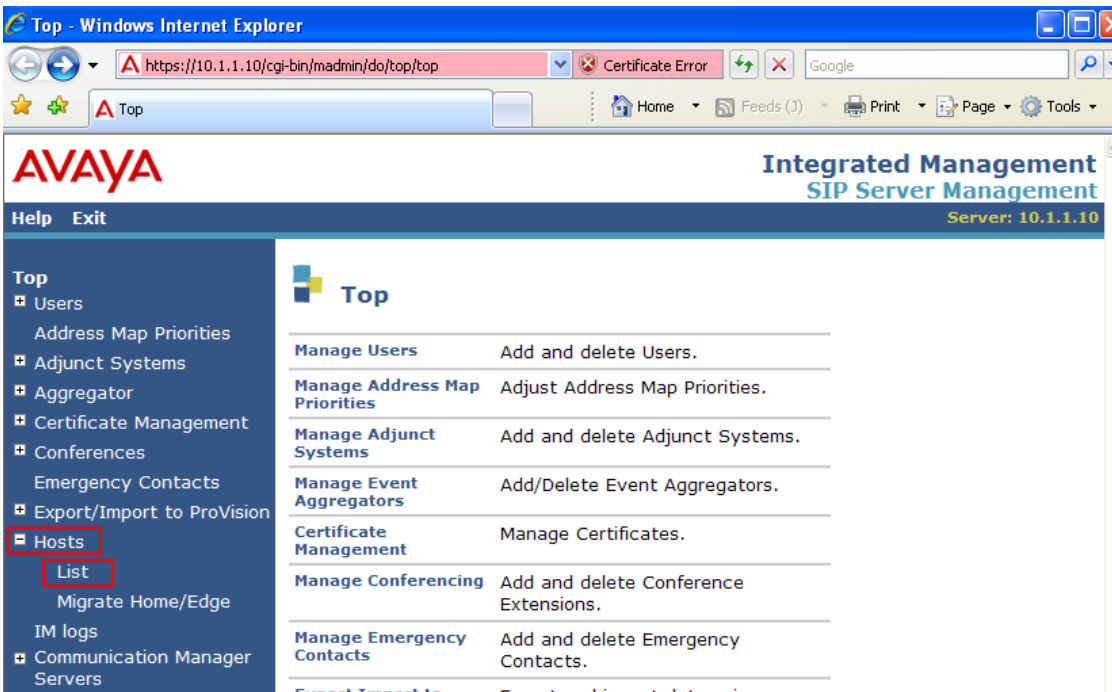
This section describes the steps required for Avaya SIP Enablement Services to support the configuration in **Figure 1**. The following pages provide step-by-step instructions on how to create the media server entry, define the host address map entry along with contact information for the Agito.

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.

## 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 Direct Inward Dialing (DID) number assigned to the Agito Mobility Router.

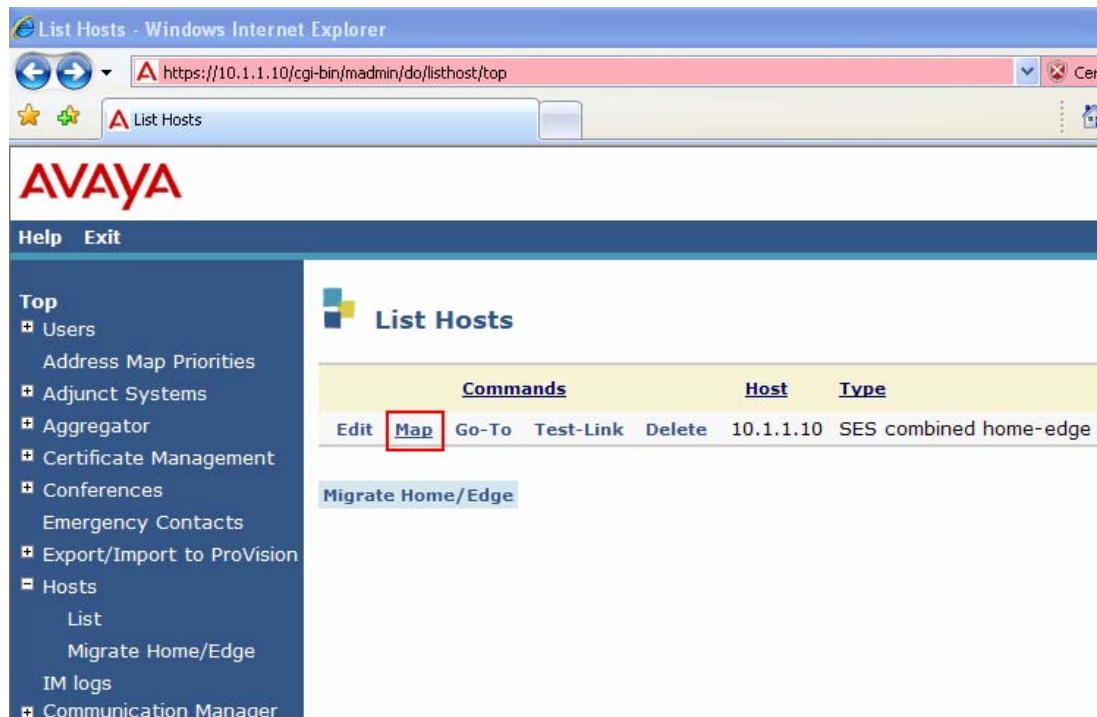
Step	Description
1.	<p>Access the SES administration web interface by using the URL <b>HTTP://ip-address/ADMIN</b> in an Internet browser window, where <b>ip-address</b> is the IP address of the SES server. Log in with the appropriate credentials. The first screen of the interface is displayed. Select <b>Launch Administration Web Interface</b>.</p> 

Step	Description																
2.	<p>The following screen is displayed. From the left pane, select <b>Hosts</b> → <b>List</b>.</p>  <p>The screenshot shows a Windows Internet Explorer browser window displaying the Avaya Integrated Management SIP Server Management interface. The address bar shows the URL <a href="https://10.1.1.10/cgi-bin/madmin/do/top/top">https://10.1.1.10/cgi-bin/madmin/do/top/top</a>. The page features a blue header with the Avaya logo and the text 'Integrated Management SIP Server Management'. Below the header, there is a navigation menu on the left with various options. The 'Hosts' option is highlighted, and its sub-menu is open, showing 'List' as the selected option. The main content area on the right displays a table of management tasks.</p> <table border="1"> <thead> <tr> <th>Task</th> <th>Description</th> </tr> </thead> <tbody> <tr> <td>Manage Users</td> <td>Add and delete Users.</td> </tr> <tr> <td>Manage Address Map Priorities</td> <td>Adjust Address Map Priorities.</td> </tr> <tr> <td>Manage Adjunct Systems</td> <td>Add and delete Adjunct Systems.</td> </tr> <tr> <td>Manage Event Aggregators</td> <td>Add/Delete Event Aggregators.</td> </tr> <tr> <td>Certificate Management</td> <td>Manage Certificates.</td> </tr> <tr> <td>Manage Conferencing</td> <td>Add and delete Conference Extensions.</td> </tr> <tr> <td>Manage Emergency Contacts</td> <td>Add and delete Emergency Contacts.</td> </tr> </tbody> </table>	Task	Description	Manage Users	Add and delete Users.	Manage Address Map Priorities	Adjust Address Map Priorities.	Manage Adjunct Systems	Add and delete Adjunct Systems.	Manage Event Aggregators	Add/Delete Event Aggregators.	Certificate Management	Manage Certificates.	Manage Conferencing	Add and delete Conference Extensions.	Manage Emergency Contacts	Add and delete Emergency Contacts.
Task	Description																
Manage Users	Add and delete Users.																
Manage Address Map Priorities	Adjust Address Map Priorities.																
Manage Adjunct Systems	Add and delete Adjunct Systems.																
Manage Event Aggregators	Add/Delete Event Aggregators.																
Certificate Management	Manage Certificates.																
Manage Conferencing	Add and delete Conference Extensions.																
Manage Emergency Contacts	Add and delete Emergency Contacts.																

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.

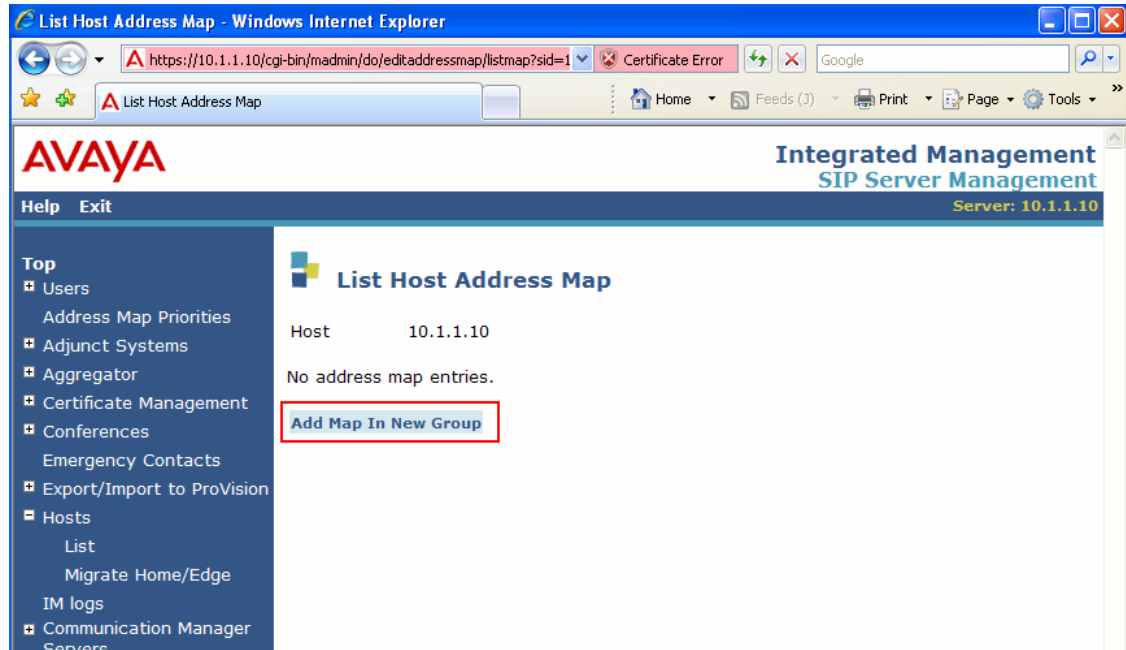
3.

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



4.

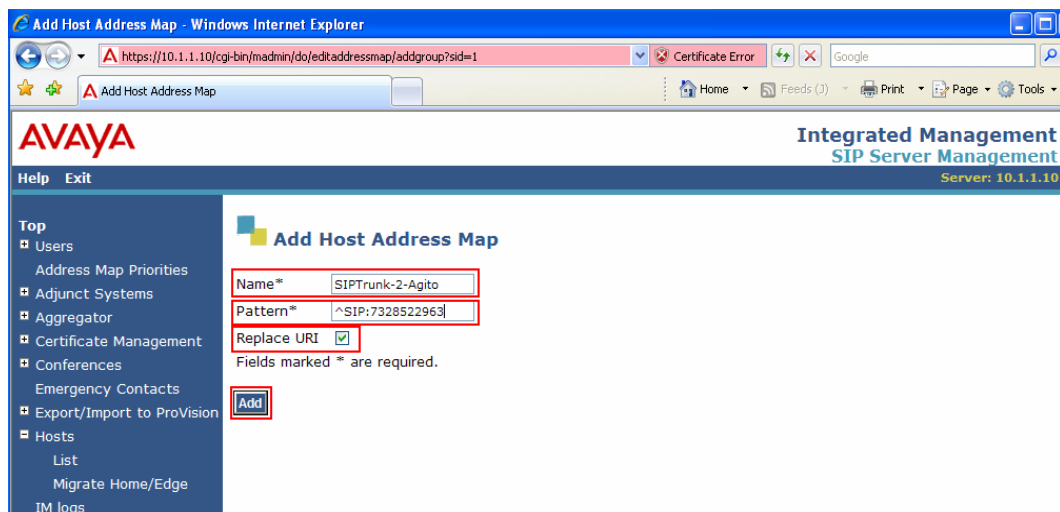
The **List Host Address Map** screen is displayed. Select, **Add Map In New Group**.



5.

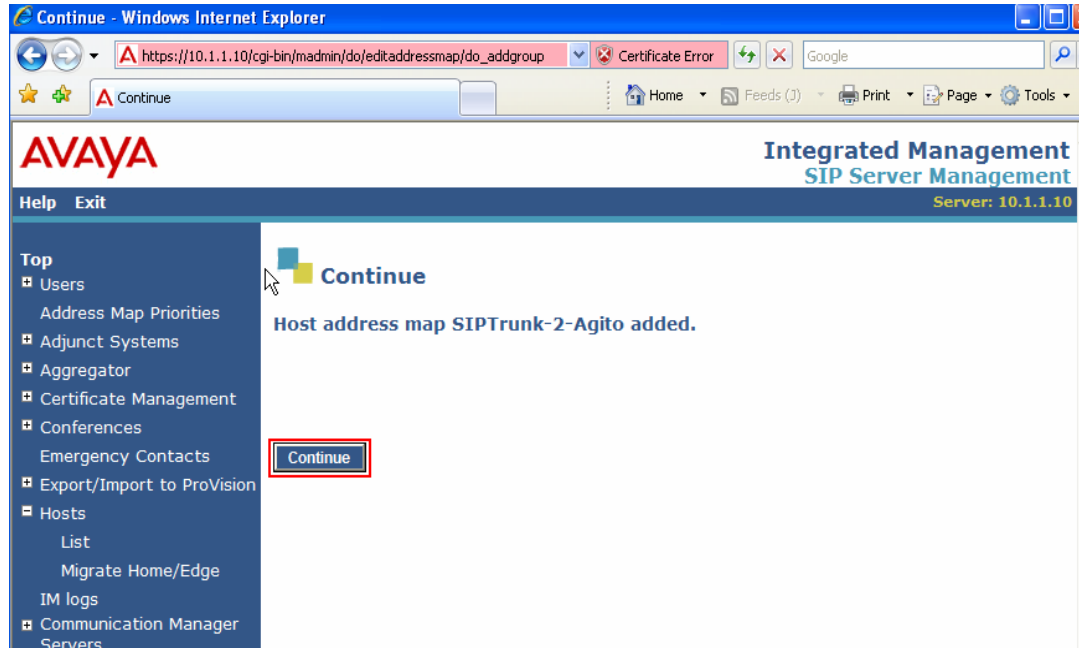
The **Add Host Address Map** screen is displayed. Enter the following:

- 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**.



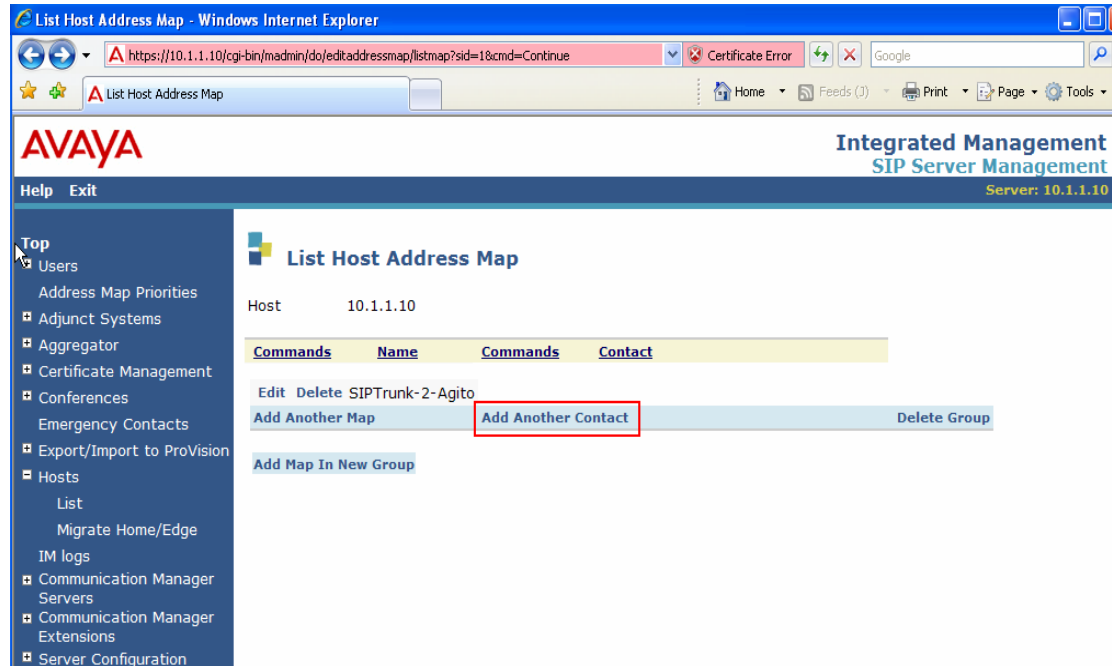
6.

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



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.



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.

**Add Host Contact - Windows Internet Explorer**

https://10.1.1.10/cgi-bin/madmin/do/editaddressmap/addcontact?handle=SIPTrunk-2-Agito&sid=1&c

**AVAYA**

Help Exit

**Top**

- Users
- Address Map Priorities
- Adjunct Systems
- Aggregator
- Certificate Management
- Conferences
- Emergency Contacts
- Export/Import to ProVision
- Hosts
  - List
  - Migrate Home/Edge
  - IM logs
- Communication Manager Servers

**Add Host Contact**

Handle SIPTrunk-2-Agito

Contact\* sip:\$(user)@10.1.1.100:5060;transport=udf

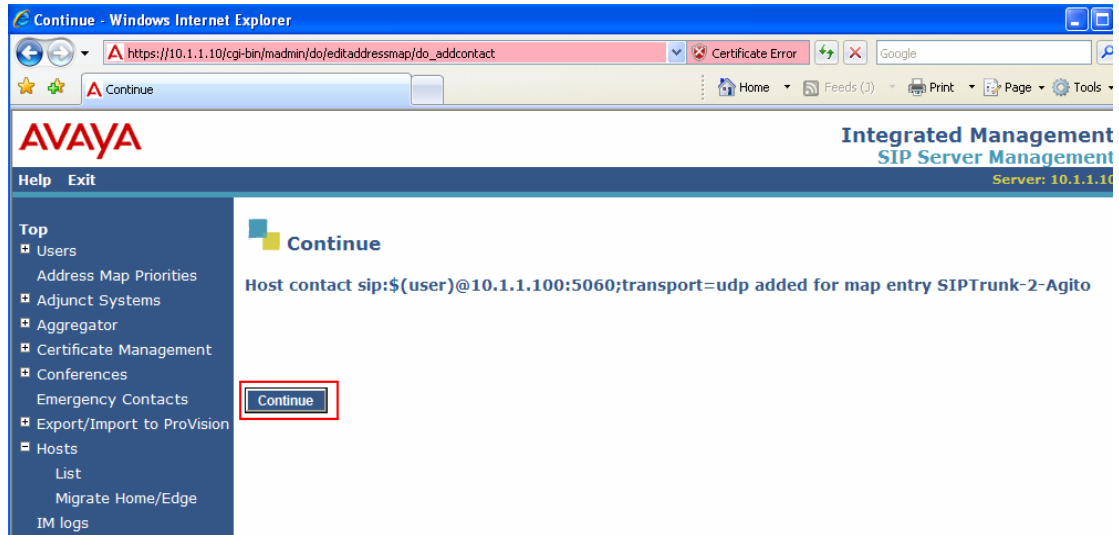
Fields marked \* are required.

**Add**



9.

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



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.

**Add Trusted Host - Windows Internet Explorer**

Address Bar: <https://10.1.1.10/cgi-bin/madmin/do/trustedhosts/add>

**AVAYA**

Help Exit

**Top**

- Users
  - Address Map Priorities
- Adjunct Systems
- Aggregator
- Certificate Management
- Conferences
  - Emergency Contacts
- Export/Import to ProVision
- Hosts
  - IM logs
- Communication Manager Servers
- Communication Manager Extensions
- Server Configuration
- SIP Phone Settings
- Survivable Call Processors
  - System Status
- Trace Logger
- Trusted Hosts
  - Add

**Add Trusted Host**

IP Address\*: 10.1.1.100

Host\*: 10.1.1.10

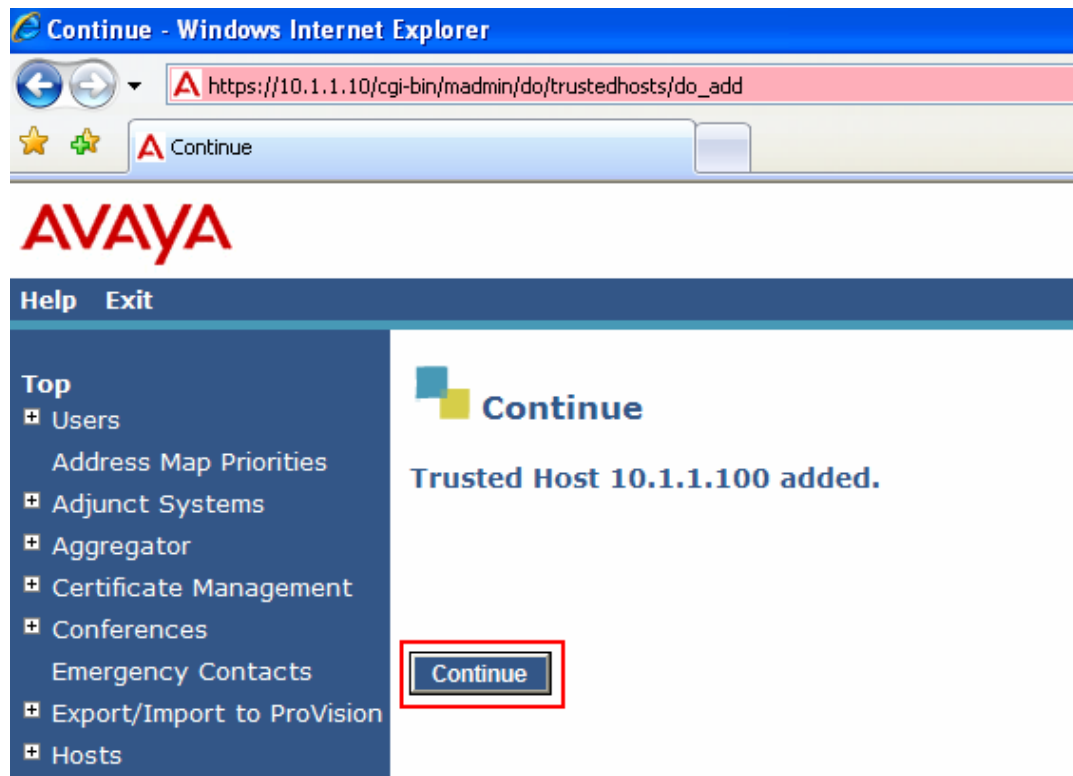
Comment: Agito RAMR

Fields marked \* are required.

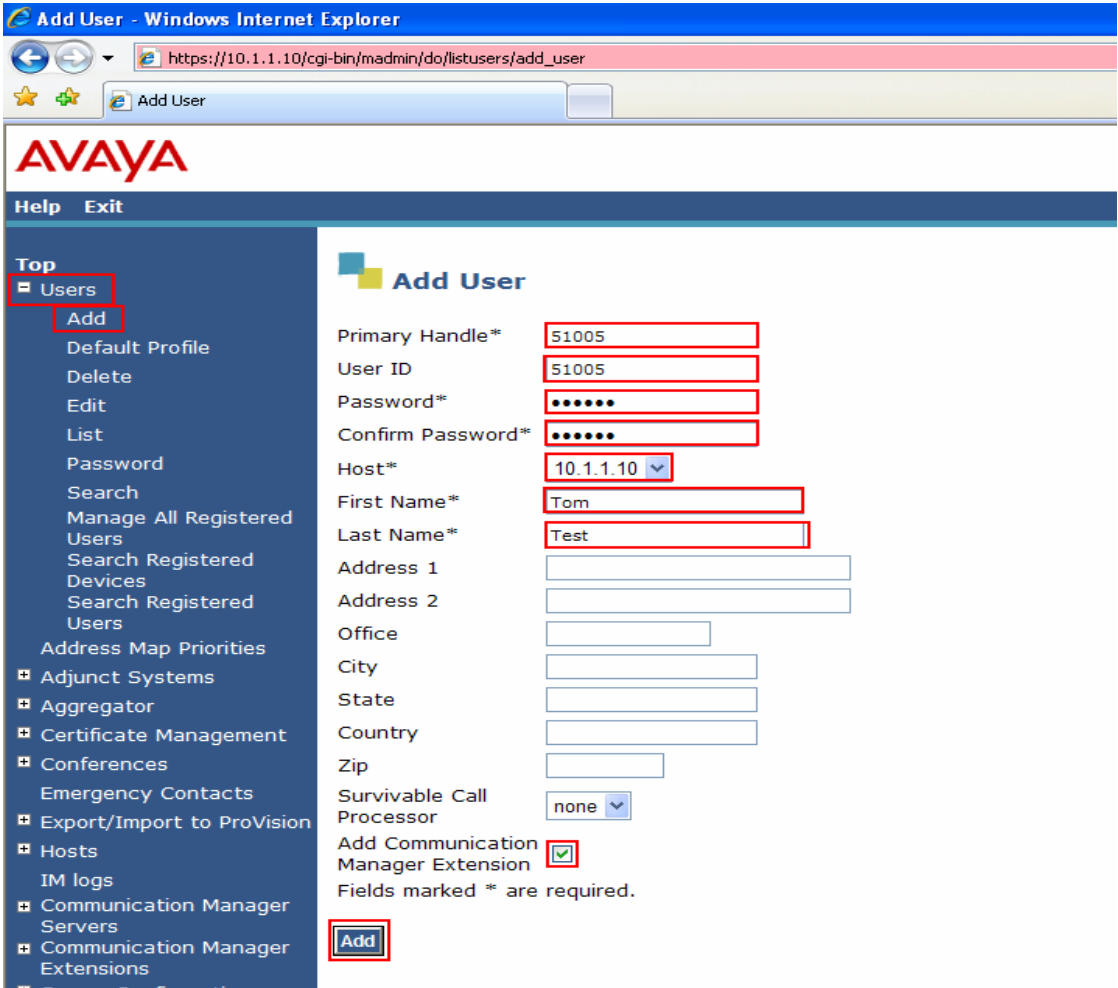
**Add**

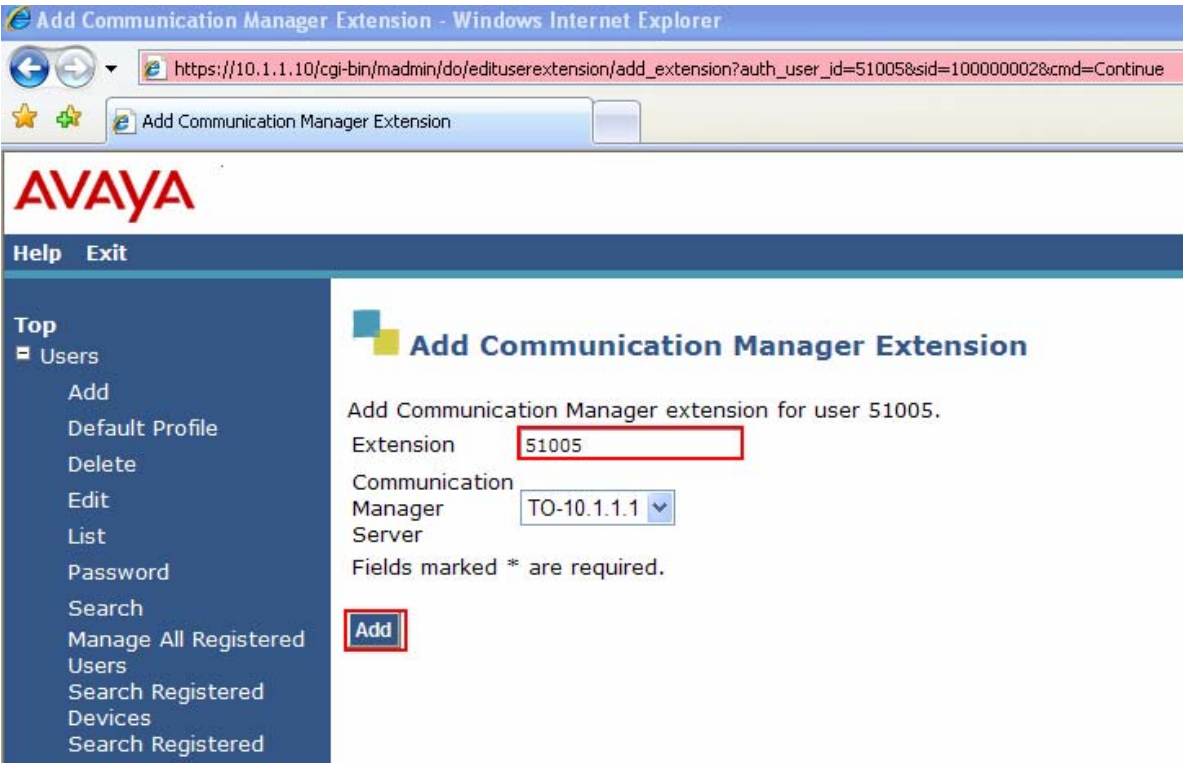
11.

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



## 4.2. Add User on Avaya SES

Step	Description
1.	<p>Select <b>Users</b> → <b>Add</b>. Fill in the screens as follows.</p> <ul style="list-style-type: none"> <li>• <b>Primary handle</b> to <b>51005</b></li> <li>• <b>User ID</b> to <b>51005</b></li> <li>• <b>Password</b> to (create User password)</li> <li>• <b>Confirm Password</b></li> <li>• <b>Host</b> to <b>10.1.1.10</b></li> <li>• <b>First Name</b> to <b>Tom</b></li> <li>• <b>Last Name</b> to <b>Test</b></li> <li>• Click the <b>Add Communication Manager Extension</b> check box.</li> <li>• Select <b>Add</b></li> <li>• A dialogue box appears, Click <b>Continue</b> to continue.</li> </ul> 

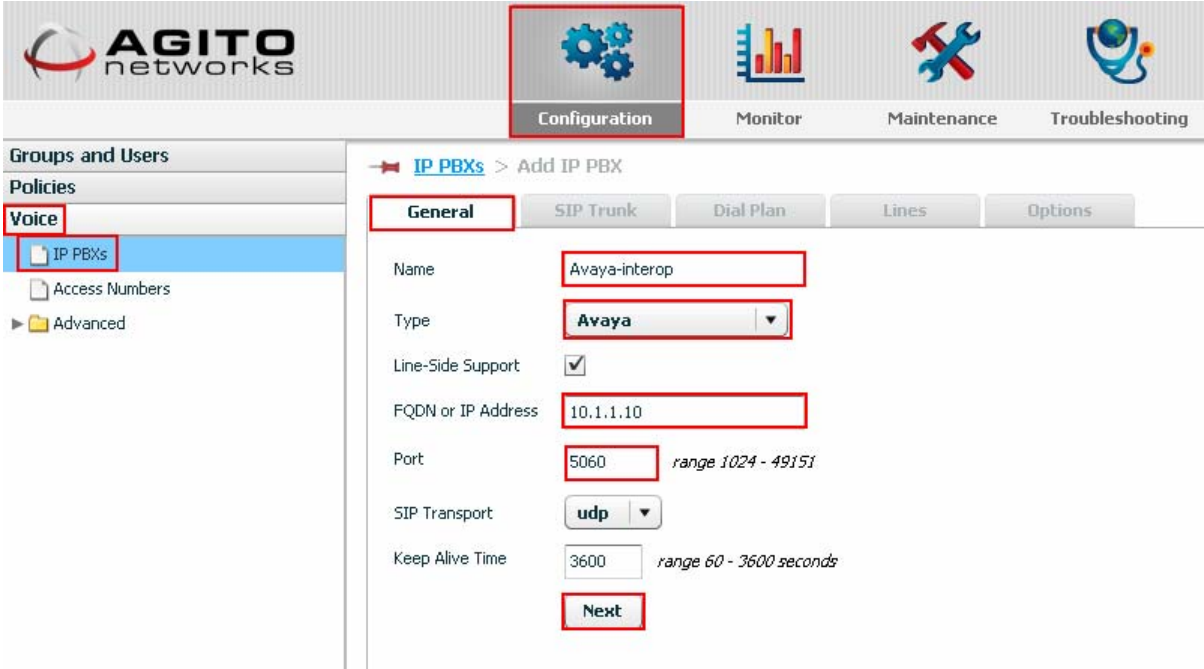
Step	Description
2.	<p>The <b>Add Communication Manage</b> Extension page appears, enter the <b>Extension</b>. Click <b>Add</b> to continue.</p> 

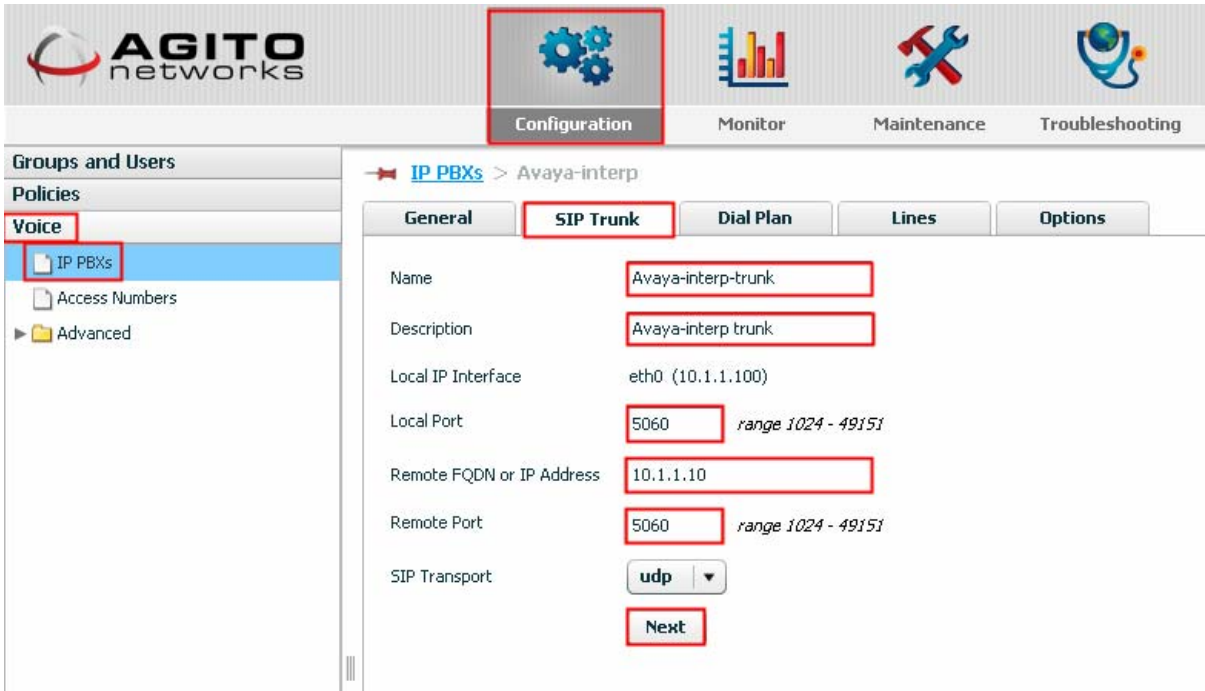
## 5. Configure the Agito RoamAnywhere Mobility Router

### 5.1. Configure Trunk to Avaya SES

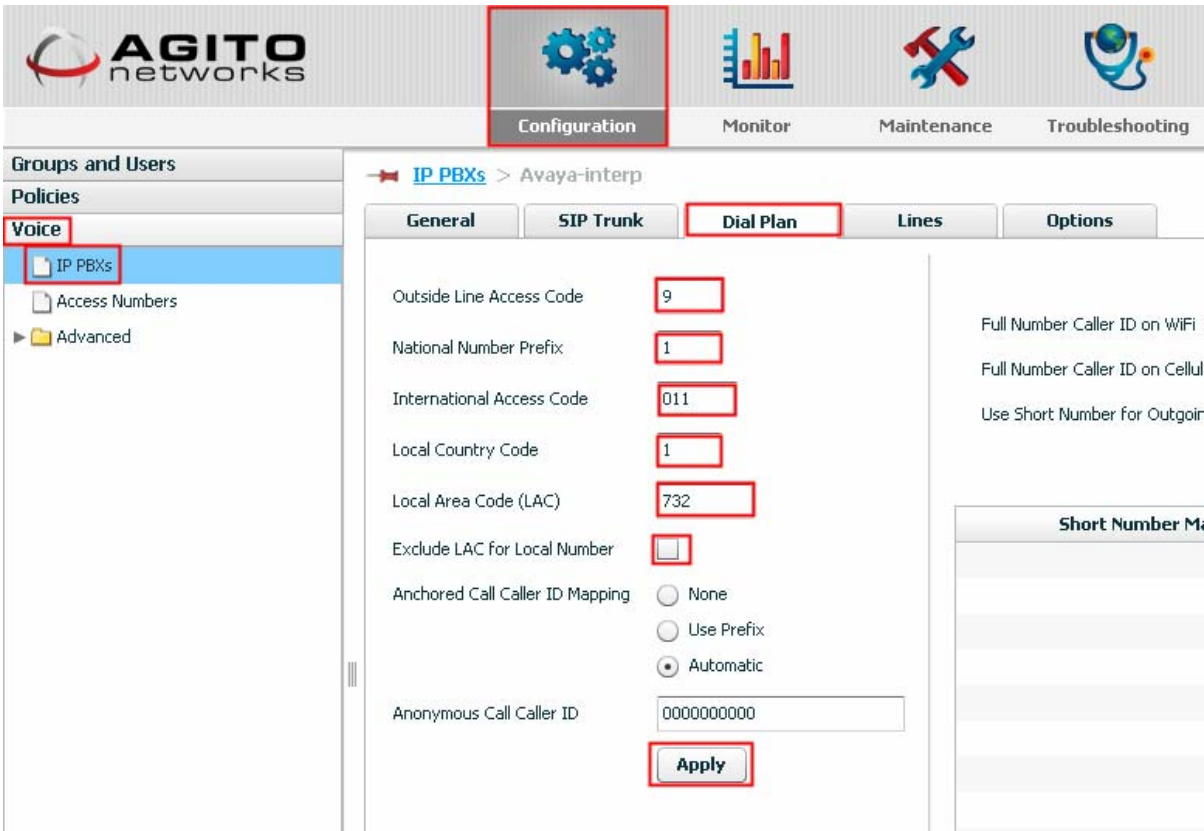
The following configuration steps outline the required settings to enable the Agito RoamAnywhere Mobility Router to interoperate with the Avaya telephony infrastructure.

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

Step	Description
2.	<p>From the <b>Configuration</b> tab, select <b>Voice</b> → <b>IP PBXs</b> → <b>Add IP PBX</b> → <b>General</b>. 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 Sections 3.3 and 3.3. Set the following:</p> <ul style="list-style-type: none"> <li>• <b>Name</b> to <b>Avaya-interop</b></li> <li>• <b>Type</b> to <b>Avaya</b></li> <li>• <b>IP Interface</b> to <b>10.1.1.10</b></li> <li>• <b>Port</b> to <b>5060</b></li> <li>• <b>Domain Name</b> to <b>dev4.com</b></li> </ul> <p>Select <b>Next</b> to continue.</p> 

Step	Description
3.	<p>From the <b>Configuration</b> tab, select <b>Voice</b> → <b>IP PBXs</b> → <b>Add IP PBX</b> → <b>SIP Trunk</b>. Ensure the IP address is set to the IP of the Avaya SES. Set the following:</p> <ul style="list-style-type: none"> <li>• <b>Name</b> to <b>avaya-interop-trunk</b></li> <li>• <b>Description</b> to <b>avaya-interop-trunk</b></li> <li>• <b>Local Port</b> to <b>5060</b></li> <li>• <b>Remote FQDN or IP Address</b> to <b>10.1.1.10</b></li> <li>• <b>Remote Port</b> to <b>5060</b></li> </ul> <p>Select <b>Next</b> to continue.</p>  <p>The screenshot shows the Agito Networks web interface. The top navigation bar includes 'Configuration' (highlighted with a red box), 'Monitor', 'Maintenance', and 'Troubleshooting'. On the left sidebar, 'Voice' is selected, and 'IP PBXs' is highlighted. The main panel shows the configuration for 'Avaya-interp' under the 'SIP Trunk' tab. The following fields are highlighted with red boxes: 'Name' (Avaya-interp-trunk), 'Description' (Avaya-interp trunk), 'Local Port' (5060), 'Remote FQDN or IP Address' (10.1.1.10), 'Remote Port' (5060), and the 'Next' button. The 'Local IP Interface' is set to 'eth0 (10.1.1.100)' and 'SIP Transport' is set to 'udp'.</p>



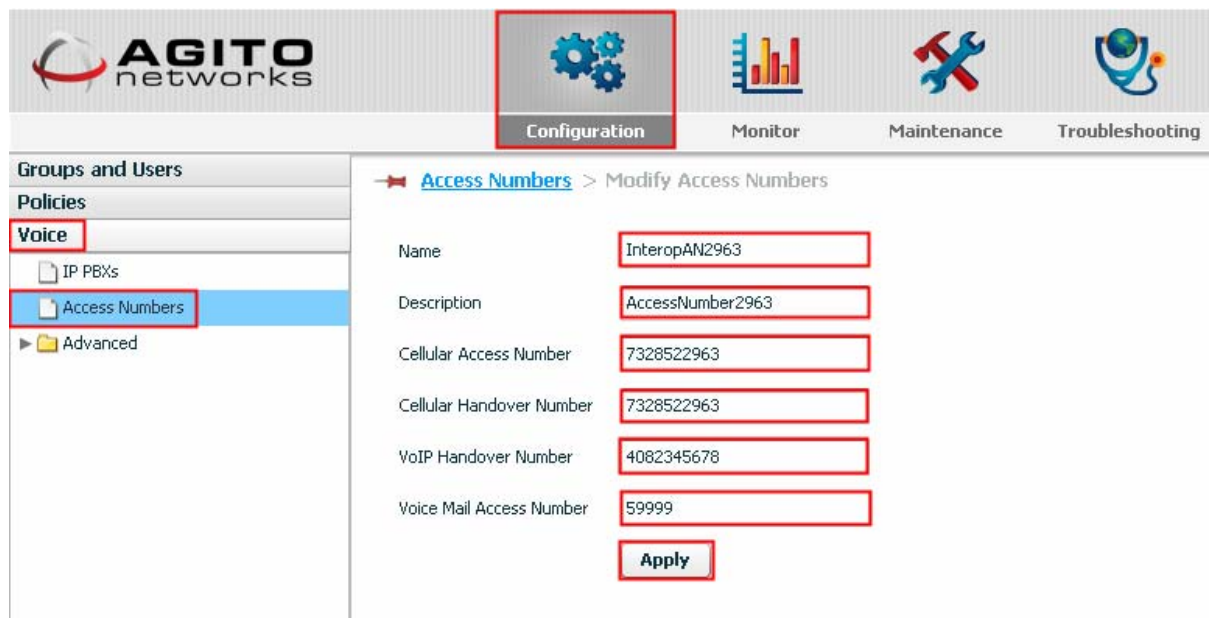
Step	Description
4.	<p>From the <b>Configuration</b> tab, select <b>Voice → IP PBXs → Add IP PBX → Dial Plan</b>. Set the Dial Plan to match that used by Avaya Communication Manager for routing calls. Ensure the Local Area Code (LAC) matches the enterprise location and whether or not the LAC is required for local external calls.</p> <ul style="list-style-type: none"> <li>• <b>Outside Line Access Code</b> to <b>9</b></li> <li>• <b>National Number Prefix</b> to <b>1</b></li> <li>• <b>International access Code</b> to <b>011</b></li> <li>• <b>Local Country Code</b> to <b>1</b></li> <li>• <b>Local Area Code</b> to <b>732</b></li> <li>• <b>Exclude LAC for Local Number</b> to <b>Uncheck</b></li> </ul> <p>Select <b>Apply</b> to commit the settings and complete the addition of the IP PBX settings.</p> 

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 following options:

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

Select **Apply** to commit the settings.



The screenshot shows the Agito Networks web interface. The top navigation bar includes the Agito Networks logo and four tabs: Configuration (highlighted with a red box), Monitor, Maintenance, and Troubleshooting. The left sidebar shows a tree view with 'Groups and Users', 'Policies', 'Voice' (highlighted with a red box), 'IP PBXs', 'Access Numbers' (highlighted with a red box), and 'Advanced'. The main content area is titled 'Access Numbers > Modify Access Numbers'. It contains a form with the following fields, all of which are highlighted with red boxes: Name (InteropAN2963), Description (AccessNumber2963), Cellular Access Number (7328522963), Cellular Handover Number (7328522963), VoIP Handover Number (4082345678), and Voice Mail Access Number (59999). An 'Apply' button is located at the bottom right of the form, also highlighted with a red box.

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.

The screenshot displays the Agito Networks web interface. At the top, there is a navigation bar with the Agito Networks logo and four tabs: Configuration (highlighted with a red box), Monitor, Maintenance, and Troubleshooting. Below the navigation bar, the left sidebar shows a tree structure under 'Policies' with 'Voice' selected (highlighted with a red box). Under 'Voice', there is an 'Advanced' folder icon, and within it, 'Media Server' is selected (highlighted with a red box). The main content area shows the 'Media Server' configuration page. It includes a red minus sign icon and the text 'Media Server'. There are two settings: 'RFC 2833 Default Payload' with a text input field containing '127' (highlighted with a red box) and a range indicator 'range 96 - 127'; and 'Inband DTMF Detection' with an unchecked checkbox. At the bottom of the settings, there is an 'Apply' button (highlighted with a red box).

## 5.2. Create the Avaya-GRP Group

1. From the **Configuration** tab, select **Groups** → **General**. Set the following options:

- **Name** to **Avaya-GRP**
- **IP PBX** to **Avaya-GRP**
- **Access Numbers** to **InteropAN2963**

Select **Next** to continue.

The screenshot shows the Agito Networks web interface. At the top, there is a navigation bar with icons for Configuration (gears), Monitor (bar chart), Maintenance (wrench), and Troubleshooting (headset). Below this is a sub-navigation bar with tabs for Groups and Users. The 'Groups' tab is selected, and the 'General' sub-tab is active. The main form area is titled 'Add Group' and contains the following fields:

- Name:** A text input field containing 'Avaya-GRP'.
- IP PBX:** A dropdown menu showing 'Avaya-interop' with an 'Add' link next to it.
- Access Numbers:** A dropdown menu showing 'InteropAN2963' with an 'Add' link next to it.
- External Group Authorization:** A checkbox that is currently unchecked.

At the bottom of the form is a 'Next' button. The 'Groups and Users' sidebar on the left shows a tree view with 'Groups' and 'Users' folders, where 'Groups' is selected.

2. From the **Configuration** tab, select **Groups → Security**. Set the following options:

- Check **Digest**
- Check **Same as RA user ID**
- **Password** to get the password from PBX admin
- **Client-Side Security (WiFi)** to **None**
- **Client-Side Security (Cellular)** to **None**

Select **Options** to continue.

The screenshot shows the AGITO networks web interface. The top navigation bar includes the AGITO logo and four tabs: Configuration (highlighted with a red box), Monitor, Maintenance, and Troubleshooting. Below the navigation bar, the left sidebar shows 'Groups and Users' with 'Groups' (highlighted with a red box) and 'Users'. The main content area is titled 'Groups > Avaya-GRP' and contains four sub-tabs: General, Security (highlighted with a red box), Options (highlighted with a red box), Calling Rules, and Users. The Security tab is active, showing the following settings:

- PBX-Side Security:** Radio buttons for None, Digest (selected and highlighted with a red box), and User ID. The User ID dropdown is set to 'Same as RA user ID' (highlighted with a red box). A 'Default' option is also present.
- Password:** A text field containing '\*\*\*\*\*' (highlighted with a red box).
- Client-Side Security (WiFi):** Radio buttons for None (selected and highlighted with a red box), Certificate, and Default.
- Client-Side Security (Cellular):** Radio buttons for None (selected and highlighted with a red box), Generate random pin per user, and Default Pin.
- Apply:** A button (highlighted with a red box) and a checkbox labeled 'Apply to all existing users in this group'.

3. From the **Configuration** tab, select **Groups → Options**. Set the following options:

- Check **Allow VoIP from remote WiFi networks**
- Check **Allow from cellular data networks** if you want all users to have the ability to access Directory Query utilizing the cellular data connection
- Check **Allow from remote WiFi networks**
- Check **Route all outgoing calls through the enterprise**
- Check **Allow outbound calls directly over cellular**

Select **Next** to continue.

AGITO networks

Configuration Monitor Maintenance Troubleshooting

Groups and Users

Groups Users

Groups > Avaya-GRP

General Security Options Calling Rules Users

Secure Remote Voice ☒ Allow VoIP from remote WiFi networks [Remote Access](#)

Secure Enterprise Services ☒ Allow from cellular data networks  
☒ Allow from remote WiFi networks

Enterprise Cellular Call Routing ☐ Forward all direct incoming cellular calls through the enterprise (this could increase mobile costs)  
☒ Route all outgoing calls through the enterprise  
☐ Route all outgoing calls directly over cellular  
☐ Route all outgoing calls directly over cellular except  
☒ From nearby home or office location  
☒ Enterprise extensions  
☒ International numbers

Personal Call Routing ☒ Allow outbound calls directly over cellular

Call Ignore ☐ Ignore call on all shared line devices (stops ringing your desk phone)

**Apply** ☐ Apply to all existing users in this group

### 5.3. Create users

1. From the **Configuration** tab, select **Users** → **General**. Set the following options:

- **Group** to **Avaya-GRP**
- **User ID** to **51005**
- **Full Name** to **Tom Test**
- **Password** to (enter the password for the
- **Access Numbers** to **InteropAN2963**

Select **Next** to continue.

The screenshot shows the AGITO networks web interface. At the top, there are four main tabs: Configuration (highlighted with a red box), Monitor, Maintenance, and Troubleshooting. Below these, there's a 'Groups and Users' section on the left with 'Groups' and 'Users' (highlighted with a red box). The main area is titled 'Users > Add User' and has several sub-tabs: General (highlighted with a red box), Line, Mobile Device, Calling Rules, Home Locations, and Options. The 'General' tab contains the following fields: 'Group' (a dropdown menu set to 'Avaya-GRP' with an 'Add' link), 'User ID' (text box with '51005'), 'Full Name' (text box with 'Tom Test'), 'Local User' (checkbox checked), 'Password' (text box with '\*\*\*\*\*'), and 'Enabled' (checkbox checked). At the bottom of the form is a 'Next' button (highlighted with a red box).

2. From the **Configuration** tab, select **Users** → **Line**. Set the following options:

- **Directory Number** to **51005**
- **Enterprise Number** to (obtain information from Agito)
- Check **PBX-Side Security** to **None**
- Check **Client-Side Security (WiFi)** to **None**
- Check **Client-Side Security (Cellular)** to **None**

Select **Mobile Device** to continue.

The screenshot displays the Agito Networks web interface. At the top, the 'Configuration' tab is selected, indicated by a red box around its gear icon. Below the navigation bar, the 'Groups and Users' sidebar on the left has the 'Users' option highlighted with a red box. The main content area shows the configuration for user '51005'. The 'Mobile Device' tab is selected and highlighted with a red box. Within this tab, the following fields are visible: 'Directory Number' is set to '51005'; 'Enterprise Number' is a redacted field with a red box around it and a red note 'This field is required'; 'PBX-Side Security' has 'None' selected with a red box around the radio button; 'Client-Side Security (WiFi)' has 'None' selected with a red box around the radio button; and 'Client-Side Security (Cellular)' has 'None' selected with a red box around the radio button. An 'Apply' button is located at the bottom of the configuration area.

## 5.4. Voice Mail Configuration

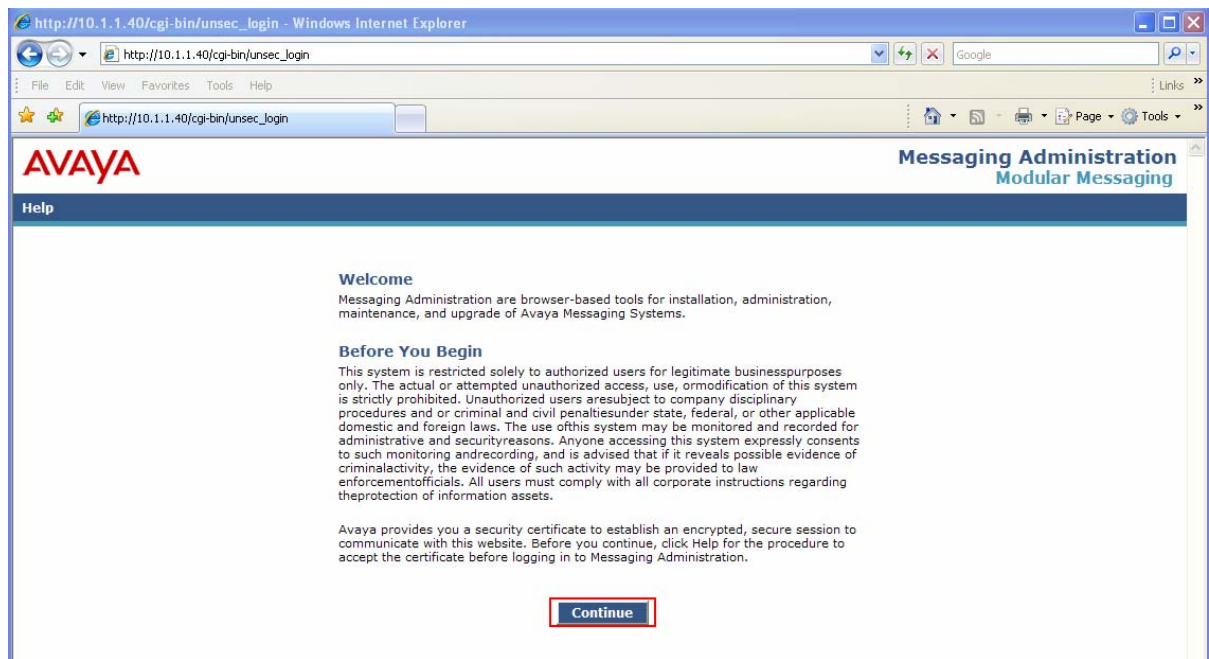
This section describes the steps for configuring voicemail for extensions in the Avaya telephony infrastructure. For informational purposes, steps for both Avaya Modular Messaging and Avaya IA770 INTUITY AUDIX are included in this document. Use the setup information appropriate for the environment being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

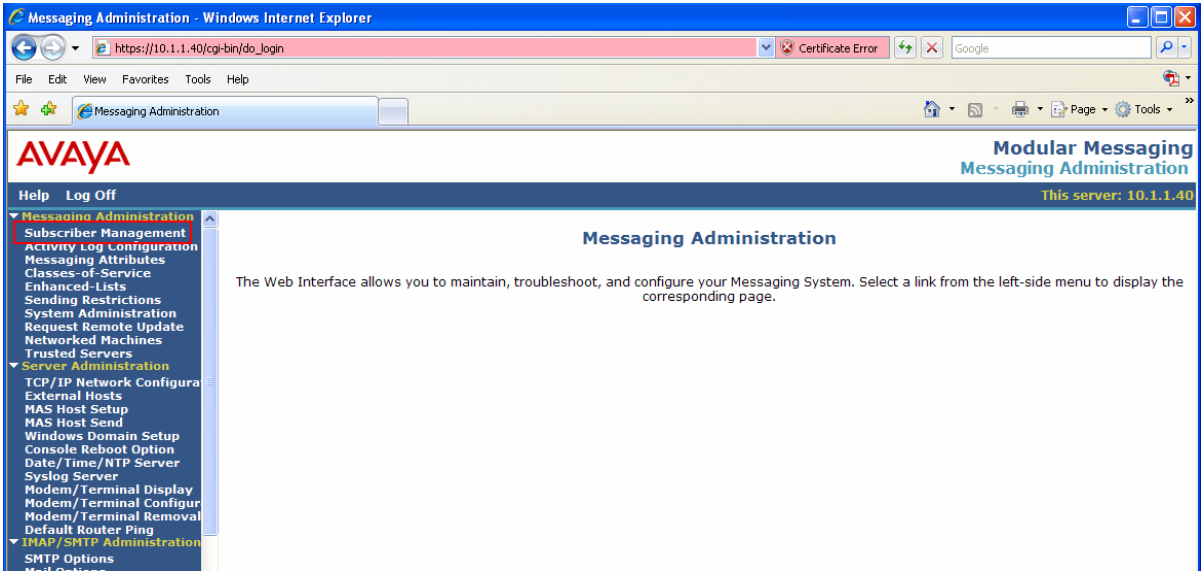
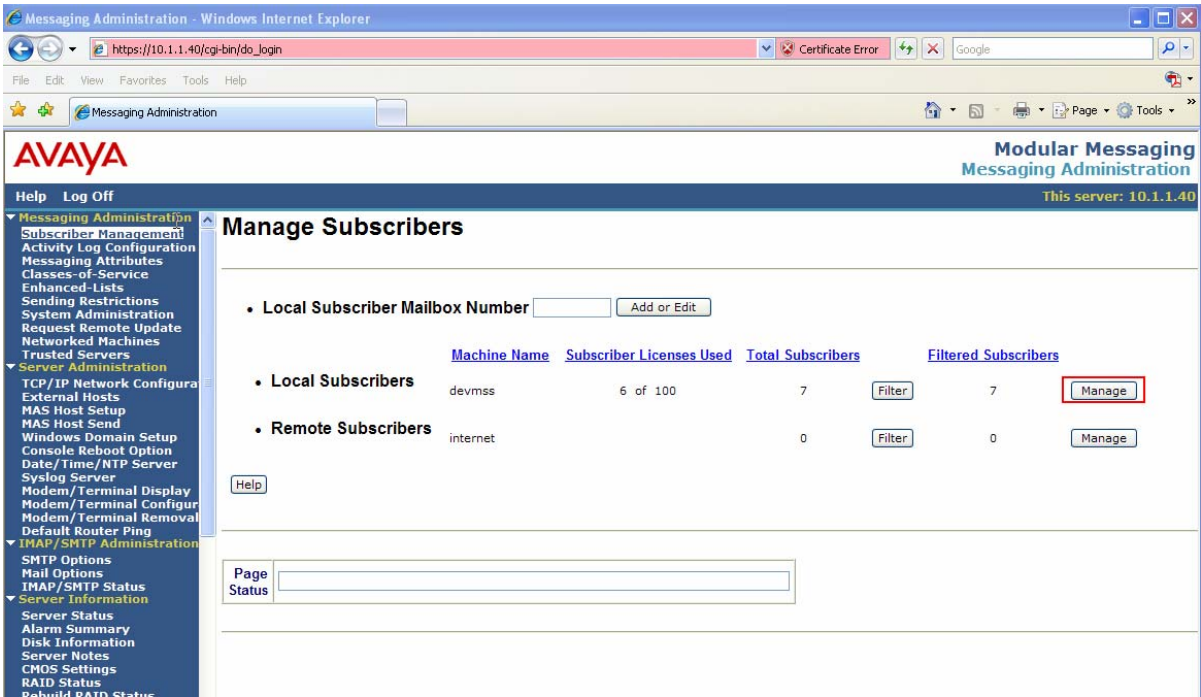
**Note:** It is recommended that at least four rings be used to route a call to voice mail.

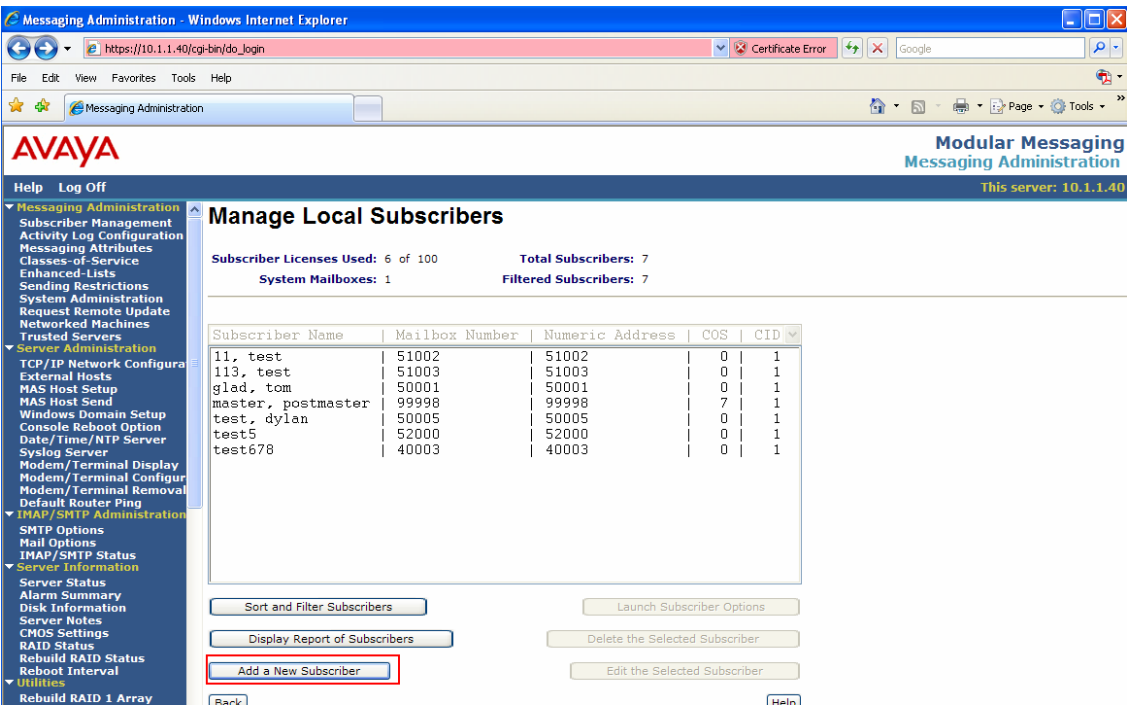


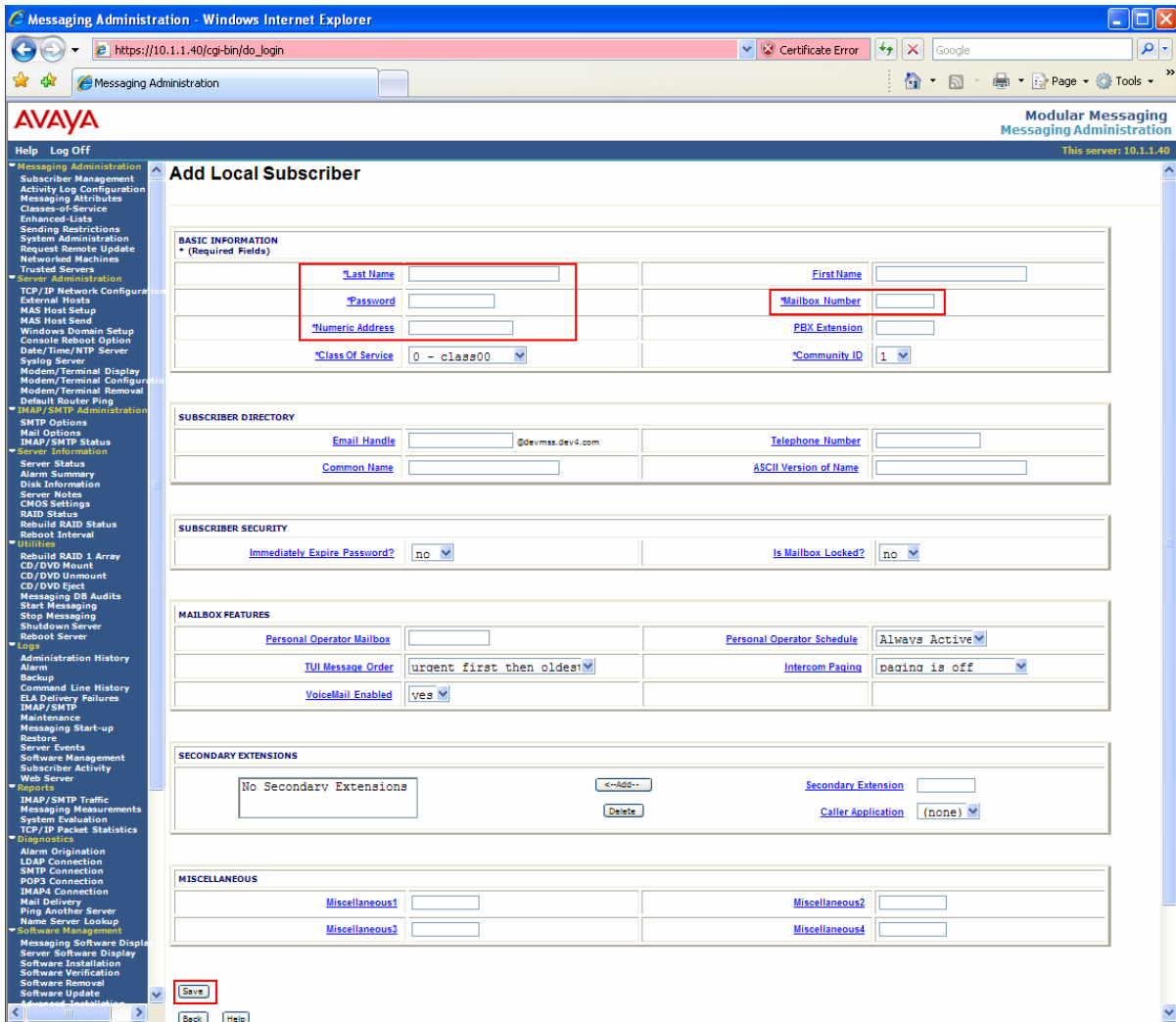
## 5.5. Configure Subscriber on Avaya Modular Messaging

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.



Step	Description
2.	<p>Select <b>Subscriber Management</b>.</p> 
3.	<p>Select <b>Manage</b> aligned with <b>Local Subscribers</b>.</p> 

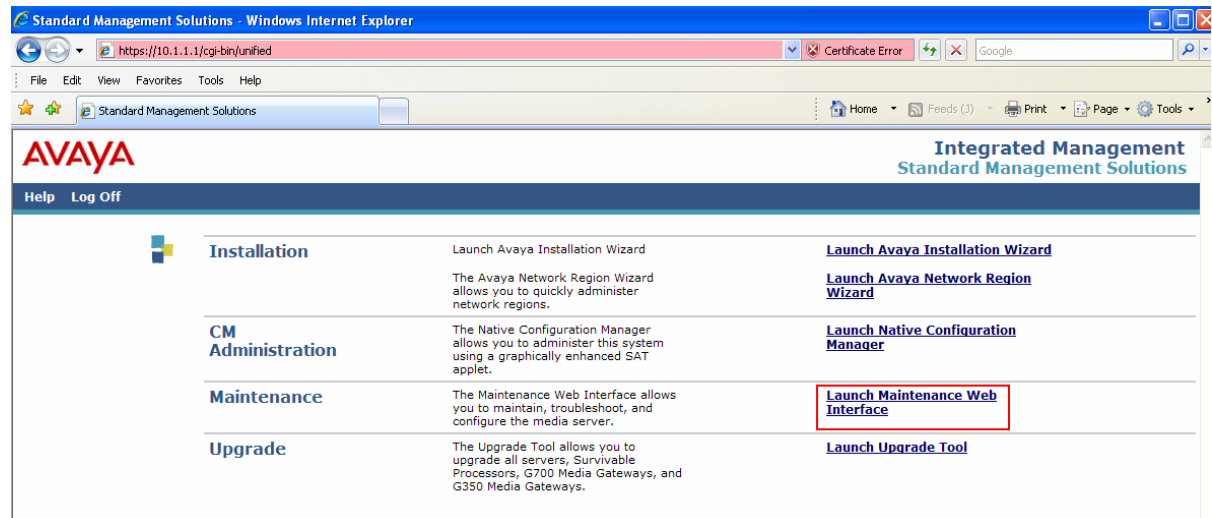
Step	Description
4.	<p>Select <b>Add a New Subscriber</b>.</p>  <p>The screenshot displays the Avaya Modular Messaging Administration web interface. The browser window shows the URL <code>https://10.1.1.40/cgi-bin/do_login</code>. The page title is "Modular Messaging Administration" and it indicates "This server: 10.1.1.40". The left sidebar contains a navigation menu with categories like "Messaging Administration", "Server Administration", "IMAP/SMTP Administration", and "Utilities". The main content area is titled "Manage Local Subscribers" and shows statistics: "Subscriber Licenses Used: 6 of 100", "Total Subscribers: 7", and "Filtered Subscribers: 7". Below these is a table of subscribers with columns: Subscriber Name, Mailbox Number, Numeric Address, COS, and CID. The table lists 8 subscribers. At the bottom of the page, there are several buttons: "Sort and Filter Subscribers", "Launch Subscriber Options", "Display Report of Subscribers", "Delete the Selected Subscriber", "Add a New Subscriber" (highlighted with a red box), and "Edit the Selected Subscriber".</p>

Step	Description
5.	<p>Enter the Follow User Information:  <b>Last Name, Password Mailbox Number, Numeric Address.</b> Select <b>Save</b> to continue.</p>  <p>The screenshot shows the 'Add Local Subscriber' form in the Avaya Modular Messaging Administration interface. The form is divided into several sections:         <ul style="list-style-type: none"> <li><b>BASIC INFORMATION</b>: Contains fields for *Last Name, *Password, *Mailbox Number, *Numeric Address, *Class Of Service (set to 0 - class00), First Name, PBX Extension, and *Community ID (set to 1).</li> <li><b>SUBSCRIBER DIRECTORY</b>: Contains fields for Email Handle (set to @devms.dev4.com), Telephone Number, Common Name, and ASCII Version of Name.</li> <li><b>SUBSCRIBER SECURITY</b>: Contains fields for Immediately Expire Password? (set to no) and Is Mailbox Locked? (set to no).</li> <li><b>MAILBOX FEATURES</b>: Contains fields for Personal Operator Mailbox, Personal Operator Schedule (set to Always Active), TUI Message Order (set to urgent first then oldest), Intercom Paging (set to paging is off), and Voicemail Enabled (set to yes).</li> <li><b>SECONDARY EXTENSIONS</b>: Contains a list of secondary extensions (currently empty) and fields for Secondary Extension and Caller Application (set to none).</li> <li><b>MISCELLANEOUS</b>: Contains four miscellaneous fields (Miscellaneous1, Miscellaneous2, Miscellaneous3, Miscellaneous4).</li> </ul>         A red box highlights the 'Save' button at the bottom of the form.       </p>

## 5.6. Configure Subscriber on Avaya IA770 INTUITY AUDIX

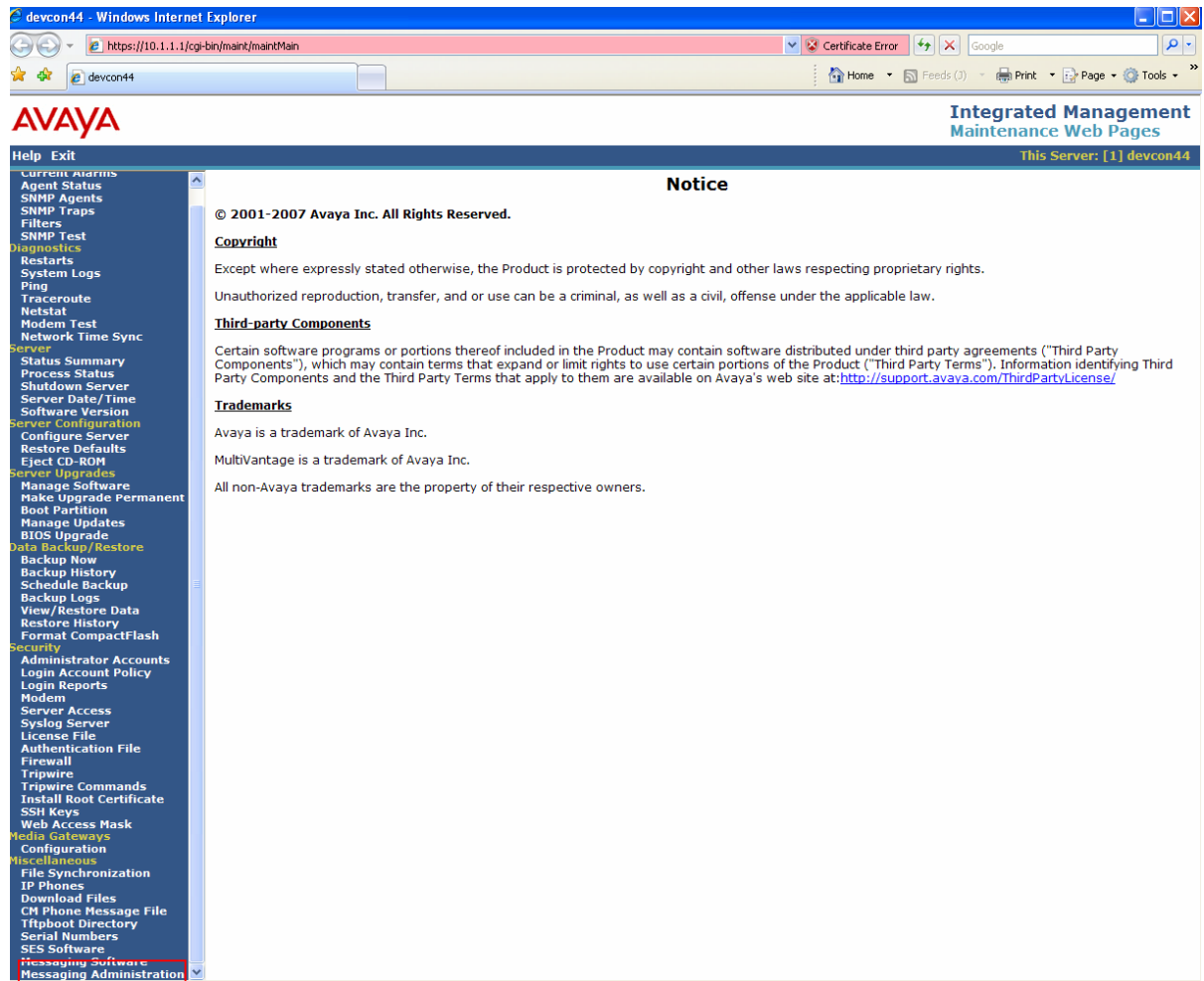
1.

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.



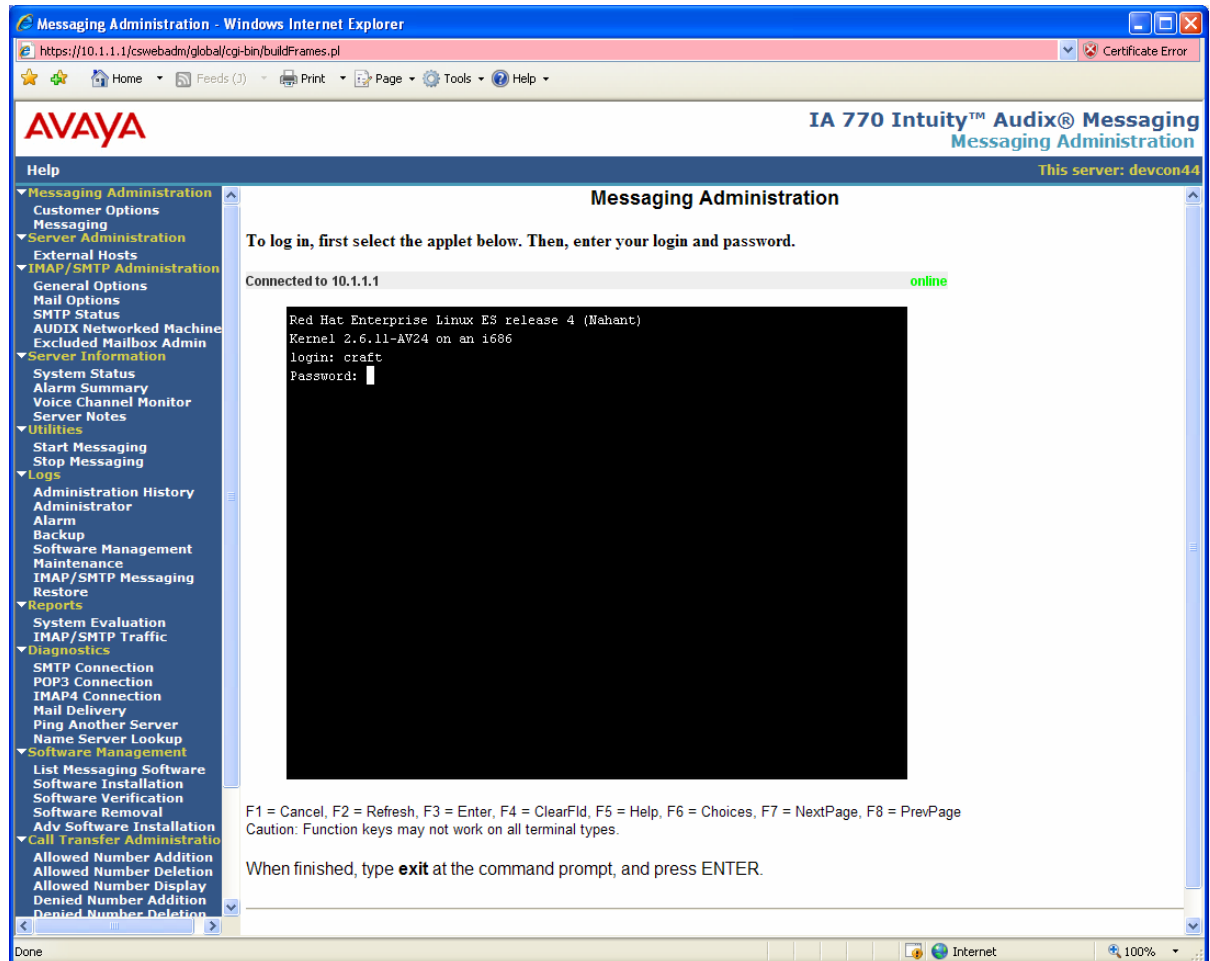
2.

Click on **Messaging Administration** then click on **Messaging**.



3.

Login with the appropriate Login and Password.



4.

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.

**AVAYA** IA 770 Intuity™ Audix® Messaging Administration  
This server: devcon44

**Messaging Administration**

To log in, first select the applet below. Then, enter your login and password.

Connected to 10.1.1.1 online

AUDIX	Active	Alarms:	A	Logins:	1
add subscriber 51020					

Page 1 of 2

**SUBSCRIBER**

Name: Enter Name Locked? n  
 Extension: 51020 Password:   
 COS: Class00 Miscellaneous 1:   
 Switch Number:  Miscellaneous 2:   
 Community ID:  Miscellaneous 3:   
 Secondary Ext:  Miscellaneous 4:   
 Account Code:  Covering Extension:   
 Broadcast Mailbox?

Email Address: 51020@devcon44.

Press [ENTER] to execute or press [CANCEL] to abort  
 enter command: add subscriber 51020  
 Cancel Refresh **Enter** ClearFld Help Choices NextPage PrevPage

F1 = Cancel, F2 = Refresh, F3 = Enter, F4 = ClearFld, F5 = Help, F6 = Choices, F7 = NextPage, F8 = PrevPage  
 Caution: Function keys may not work on all terminal types.

When finished, type **exit** at the command prompt, and press ENTER.

## 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.



## 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 IVR 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 6.1. 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:

- Web site: <http://www.agitonetworks.com/>
- Email: [support@agitonetworks.com](mailto:support@agitonetworks.com)
- Telephone: (408) 919-8000

## 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 2.0, September 2008.
- [7] *RoamAnywhere User's Guide*, Version 2.0, September 2008.
- [8] *RoamAnywhere Release Notes, Version 2.0, October 2008*.

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