



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

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# **Application Notes for Configuring Microsoft Office Communications Server 2007 R2 and Avaya IP Office PSTN Call Routing - Issue 1.0**

### **Abstract**

These Application Notes describe the procedures for configuring Microsoft Office Communications Server (OCS) 2007 R2 with Avaya IP Office. The Avaya infrastructure is used by OCS as a gateway for voice telephone calls to the Public Switched Telephone Network (PSTN). The steps described herein focus on how Microsoft Office Communicator 2007 R2 clients configured for Enterprise Voice mode can utilize the Avaya infrastructure to place and receive telephone calls to and from the PSTN.

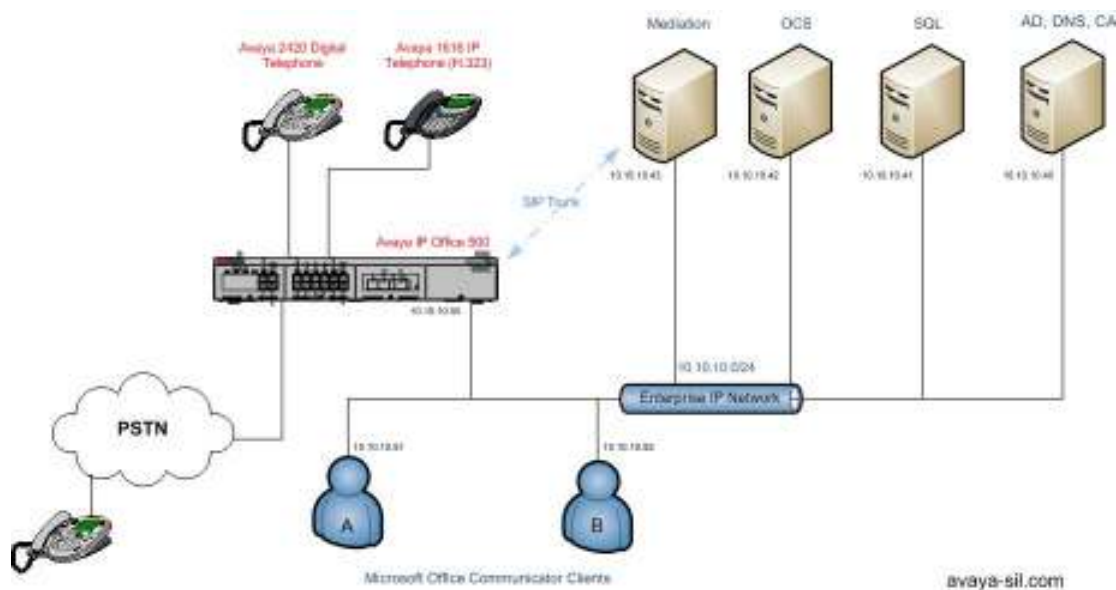
# 1. Introduction

These Application Notes describe the procedures for configuring Microsoft Office Communications Server (OCS) 2007 R2 with Avaya IP Office. The Avaya infrastructure is used by OCS as a gateway for voice telephone calls to the Public Switched Telephone Network (PSTN). The steps described herein focus on how Microsoft Office Communicator 2007 R2 clients configured for Enterprise Voice mode can utilize the Avaya infrastructure to place and receive telephone calls to and from the PSTN.

Microsoft OCS is comprised of several component servers that, in some cases, may run collocated on the same physical Microsoft Windows server or, in other cases, require separate physical servers, depending on the desired capacity, topology, and security. Consult references [2] through [5] for further details on Microsoft OCS architecture and deployment options. Please consult reference [8] for important considerations for Enterprise Voice as it contains relevant information regarding 911 (US) or 999 (UK) emergency calls and emergency services.

## 2. Configuration

The sample configuration described throughout these Application Notes is shown in **Figure 1**. An ISDN/PRI trunk provides inbound and outbound voice call access to the PSTN. Avaya IP Office sends and receives SIP Invites to and from the Microsoft Mediation Server. The Microsoft Mediation Server converts call signaling between standard SIP and Microsoft signaling protocol (MTLS) when routing voice calls to and from Microsoft OCS. The Microsoft Mediation Server also converts call media between G.711 and a proprietary Microsoft codec. The Microsoft Office Communicator (MOC) clients are registered with Microsoft OCS via a front end server pool. The pool can consist of more than one server. However, in the tested configuration, the pool consisted of one front end server. The Microsoft OCS server and Mediation servers are supported by a Microsoft SQL 2005 database server, as well as another Microsoft Windows Server running Active Directory (AD), Domain Name System (DNS) server, and Certificate Authority (CA) roles.



**Figure 1: Network Configuration**

These Application Notes describe one possible approach to configuring PSTN inbound and outbound call routing. The following user experience goals were considered in formulating the approach:

- An Enterprise Voice (EV) client should be able to call an E.164-formatted number. To address this, Microsoft OCS can be configured with one or more “normalization rules<sup>1</sup>” that match the dialed number. For example, a 7-digit dialed number can be converted into an E.164-formatted 11-digit number. On Avaya IP Office, Short Codes can be configured

<sup>1</sup> Normalization rules define matching criteria for various number strings and translations for converting the strings into E.164-formatted numbers.

to delete digits as necessary of the called party numbers in order to obtain the number to be sent to the PSTN.

- An EV client should be able to call to the PSTN from the MOC client Recent Contacts list. Again, “normalization rules” also apply to incoming calling party numbers, thereby generating E.164 numbers in the Recent Contacts list.

The flow for an outbound call from an EV client is as follows. When an EV client dials a number, Microsoft OCS applies normalization rules to the dialed number. If there is a match, Microsoft OCS checks whether the called party number (now converted to E.164 format by the normalization rule) is assigned to another MOC user. If so, Microsoft OCS sends the call to the called user’s MOC client. If not, Microsoft OCS looks up a call routing table for a match of the E.164-formatted called party number. If there is a match, Microsoft OCS routes the call to the Microsoft Mediation Server specified in the matching route. The Microsoft Mediation Server then routes the call to the configured next hop destination, which in the sample configuration, is IP Office. IP Office then routes the call to the PSTN.

For inbound calls from the PSTN, Avaya IP Office receives the incoming call. Based on the called party number, Avaya IP Office looks up the corresponding Short Code and routes the call to the Microsoft Mediation Server.

In the test scenario +35312078XXX E.164 phone numbers were mapped to IP Office extensions 8XXX. 7-digit phone numbers received from the ISDN/PRI E1 trunk or an IP Office extension dial pad matching pattern 656XXXX corresponding to E.164 numbers +3531656XXXX were routed to Microsoft Mediation Server.

### 3. Equipment and Software Validated

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

Equipment	Software Version
Avaya IP Office 500	IP Office 5.0 beta (build 011041)
Avaya 1616 IP Telephone (H.323)	Release 1.1 (ha1616ua1_100.bin)
Avaya 2420 Digital Telephone	-
Microsoft Active Directory, DNS Server, and Certification Authority on Microsoft Windows Server 2003 R2 Standard Edition Service Pack 2	Version 5.2 R2 (Build 3790.srv03_sp2_gdr.090319-1204: Service Pack 2)
Microsoft Enterprise Edition Office Communications Server 2007 R2 on Windows Server 2003 R2 Enterprise Edition x64 Edition Service Pack 2	OCS 2007 R2: 3.5.6907.0 (Volume) w/ KB 972041  Windows OS : Version 5.2 R2 (Build 3790.srv03_sp2_rtm.070216-1710 : Service Pack 2)
Microsoft SQL 2005 SP2 Server on Microsoft Windows Server 2003 R2 Standard Edition Service Pack 2	2005.090.3042.00  Windows OS : Version 5.2 R2 (Build 3790.srv03_sp2_gdr.090319-1204: Service Pack 2)
Microsoft Mediation Server on Microsoft Windows Server 2003 R2 Enterprise Edition x64 Edition Service Pack 2	OCS 2007 R2: 3.5.6907.0 (Volume)
Microsoft Office Communicator 2007 R2 on Microsoft Windows XP Professional Version SP3	R2 : 3.5.6907.0 w/ KB 972042  Windows OS: 2600.xpsp_sp3_gdr.090206-1234 : Service Pack 3
Microsoft Office Communications Server 2007 R2 Attendant	1.0.6907.0

**Table 1: Equipment/Software List**

## 4. Configure Avaya IP Office

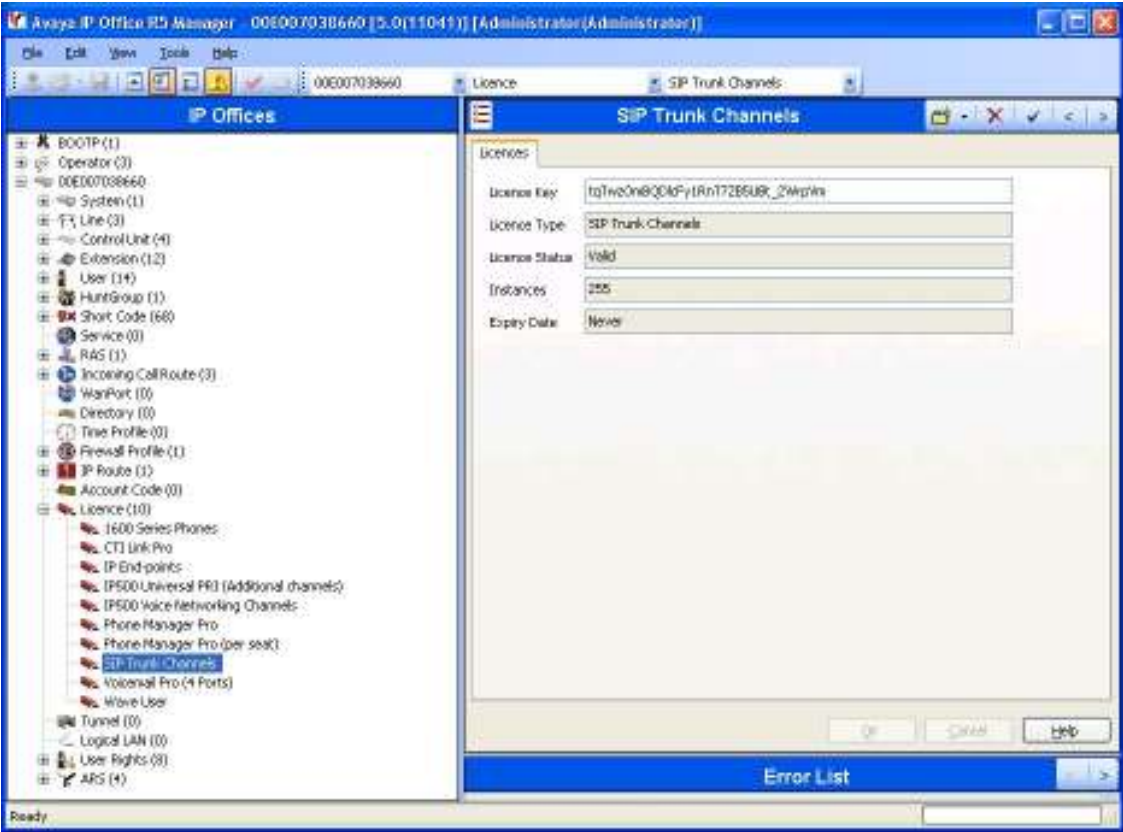
This section describes the steps for configuring call routing on Avaya IP Office. The steps are performed from the IP Office Manager interface. These Application Notes assume that basic Avaya IP Office administration has already been performed, ISDN/PRI E1 line is already configured according to the parameters given by the service provider, and user extensions are administered in the range 8XXX. See reference [1].

The configuration procedures include the following areas:

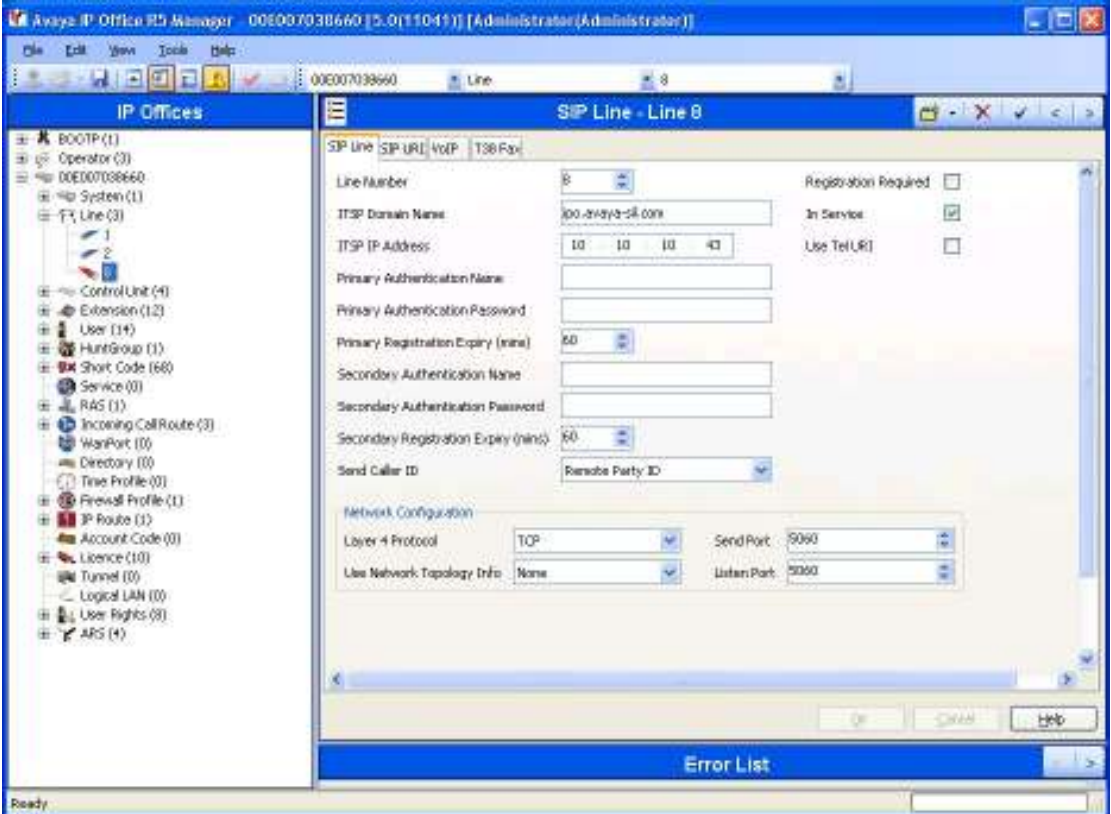
- Verify Avaya IP Office License
- Administer SIP Trunk
- Administer Incoming Call Route
- Administer ARS
- Administer Short Codes

IP Office is configured via the **IP Office Manager** program. Log in the IP Office Manager PC and select **Start → Programs → IP Office → Manager** to launch the **Manager** application. Log in to the **Manager** application using the appropriate credentials.

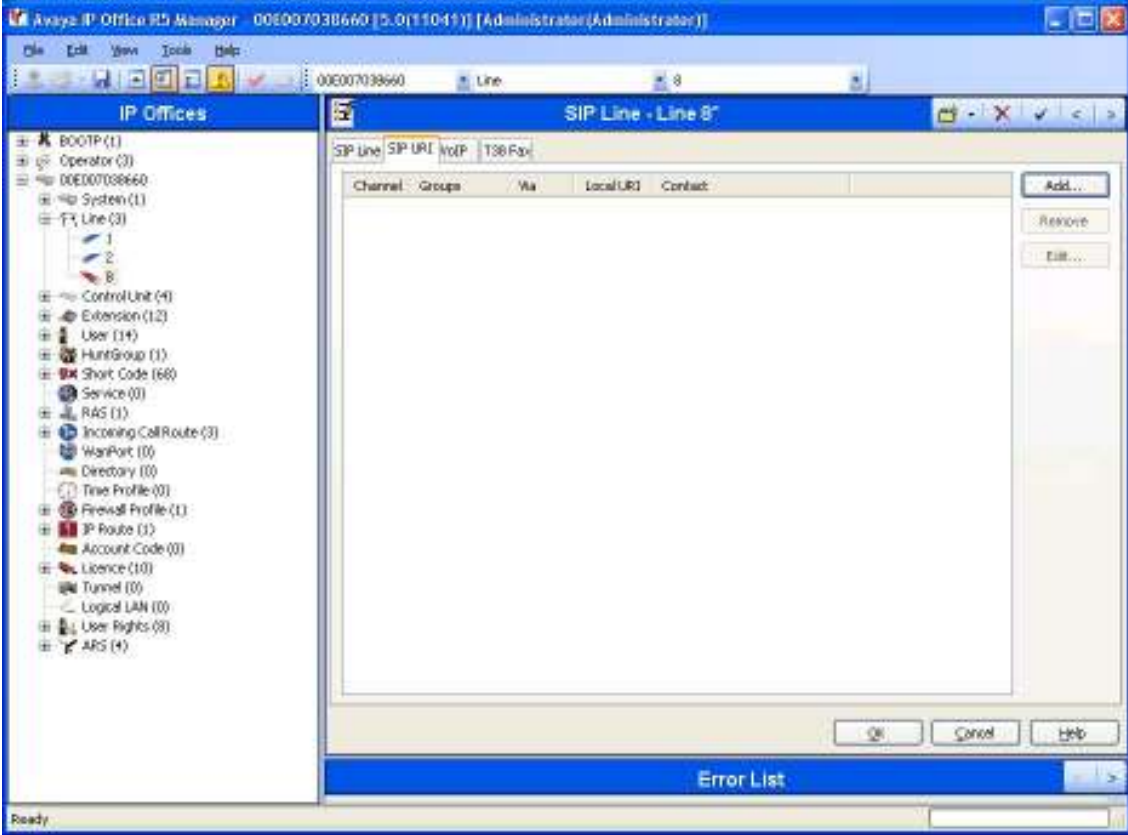
## 4.1. Verify Avaya IP Office License

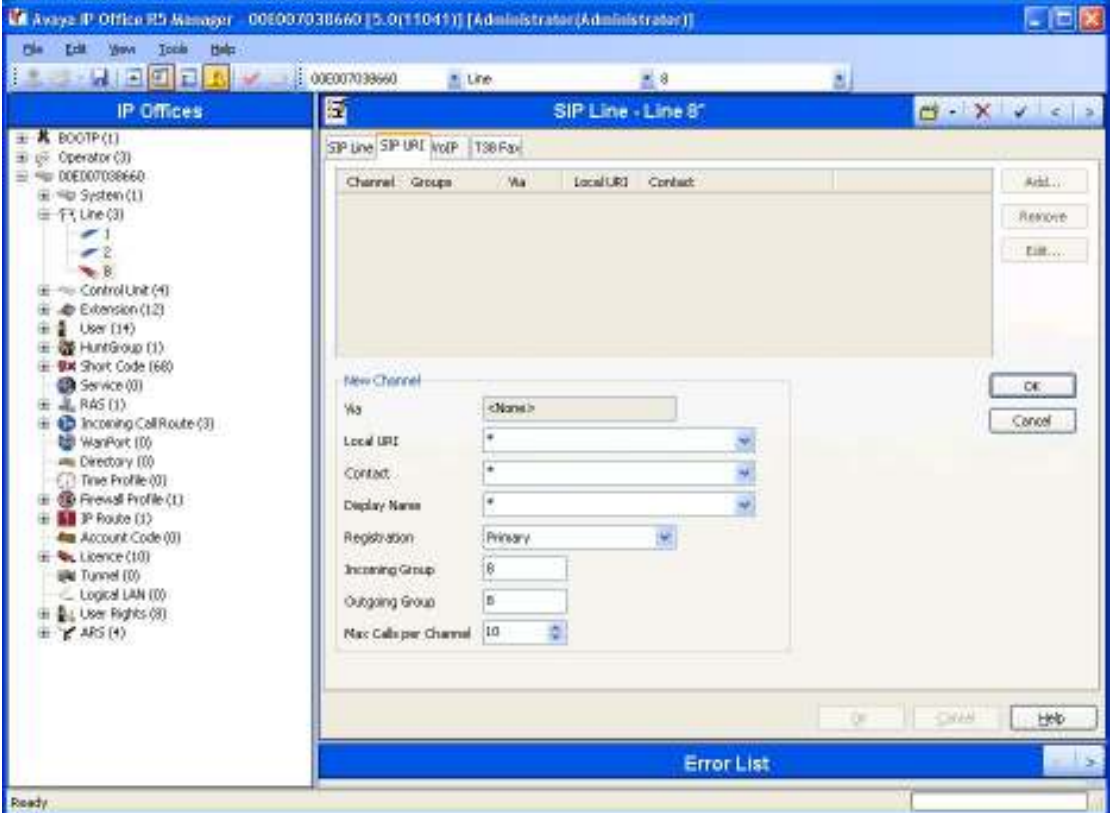
Step	Description												
1.	<p>Verify that there is a <b>SIP Trunk Channels</b> license. Double-click on <b>License</b> in the left panel. Check that there is a <b>SIP Trunk Channels</b> entry. Verify that the following licenses exist for the ISDN/PRI E1 trunk: <b>IP500 Universal PRI (Additional Channels)</b>, <b>IP500 Voice Networking Channels</b> and <b>Wave User</b>. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative or Business Partner to make the appropriate changes.</p>  <p>The screenshot shows the Avaya IP Office R5 Manager interface. The left pane displays a tree view of system components, with 'License (10)' expanded to show various license categories. The right pane shows the 'SIP Trunk Channels' license configuration window. The 'Licenses' section contains the following details:</p> <table border="1"><thead><tr><th>Field</th><th>Value</th></tr></thead><tbody><tr><td>License Key</td><td>fgTwoOnBQdMfyIRnT72B5UR_2WgWk</td></tr><tr><td>License Type</td><td>SIP Trunk Channels</td></tr><tr><td>License Status</td><td>Valid</td></tr><tr><td>Instances</td><td>255</td></tr><tr><td>Expiry Date</td><td>Never</td></tr></tbody></table> <p>At the bottom of the license configuration window, there is an 'Error List' section.</p>	Field	Value	License Key	fgTwoOnBQdMfyIRnT72B5UR_2WgWk	License Type	SIP Trunk Channels	License Status	Valid	Instances	255	Expiry Date	Never
Field	Value												
License Key	fgTwoOnBQdMfyIRnT72B5UR_2WgWk												
License Type	SIP Trunk Channels												
License Status	Valid												
Instances	255												
Expiry Date	Never												

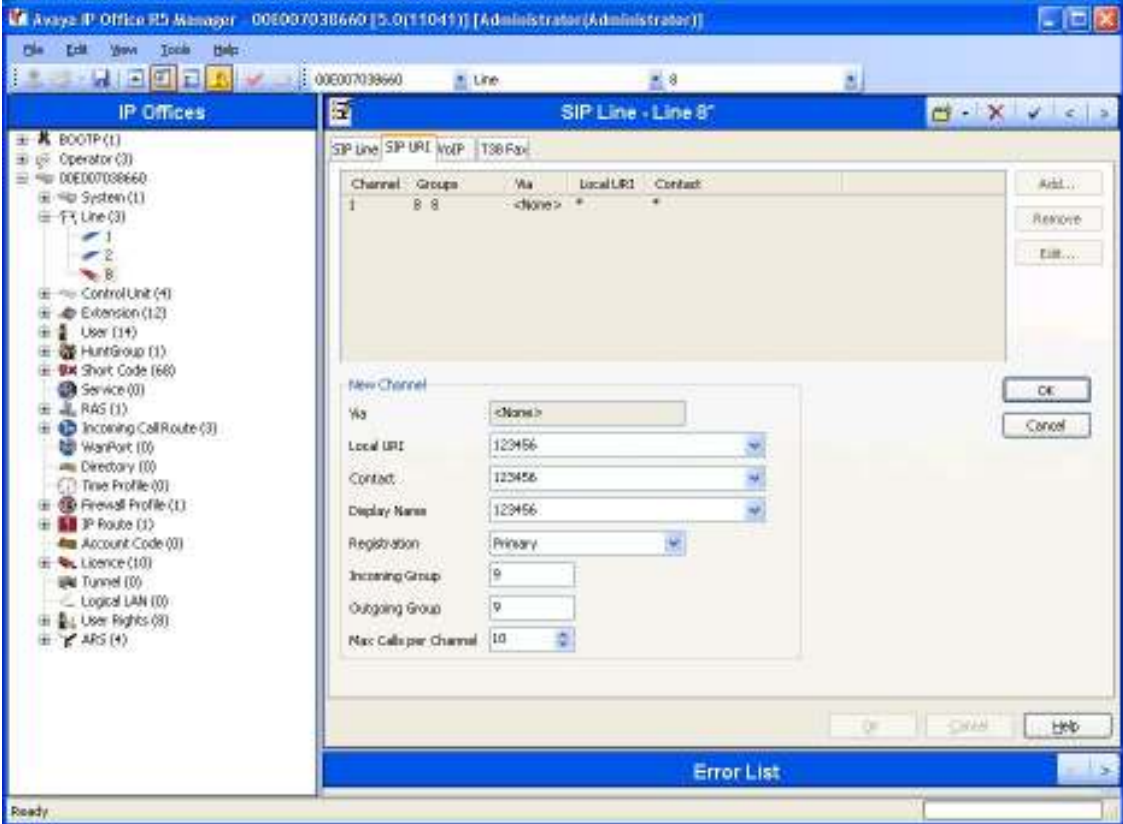
## 4.2. Administer SIP Trunk

Step	Description
1.	<p>Create the SIP line for Microsoft Mediation Server. Select <b>Line</b> in the left panel. Right-click and select <b>New → SIP Line</b>. Enter the SIP Domain Name of IP Office in the <b>ITSP Domain Name</b> field. Enter the Microsoft Mediation Server IP Address in the <b>ITSP IP Address</b> field. Select <b>Remote Party ID</b> in the <b>Send Caller ID</b> field.</p> <p>In the <b>Network Configuration</b> section, select the following:</p> <ul style="list-style-type: none"> <li>• For <b>Layer 4 Protocol</b>, use <b>TCP</b></li> <li>• For <b>Send Port</b>, use <b>5060</b></li> <li>• For <b>Listen Port</b>, use <b>5060</b></li> <li>• For <b>Use Network Topology Info</b>, use <b>None</b></li> </ul> 

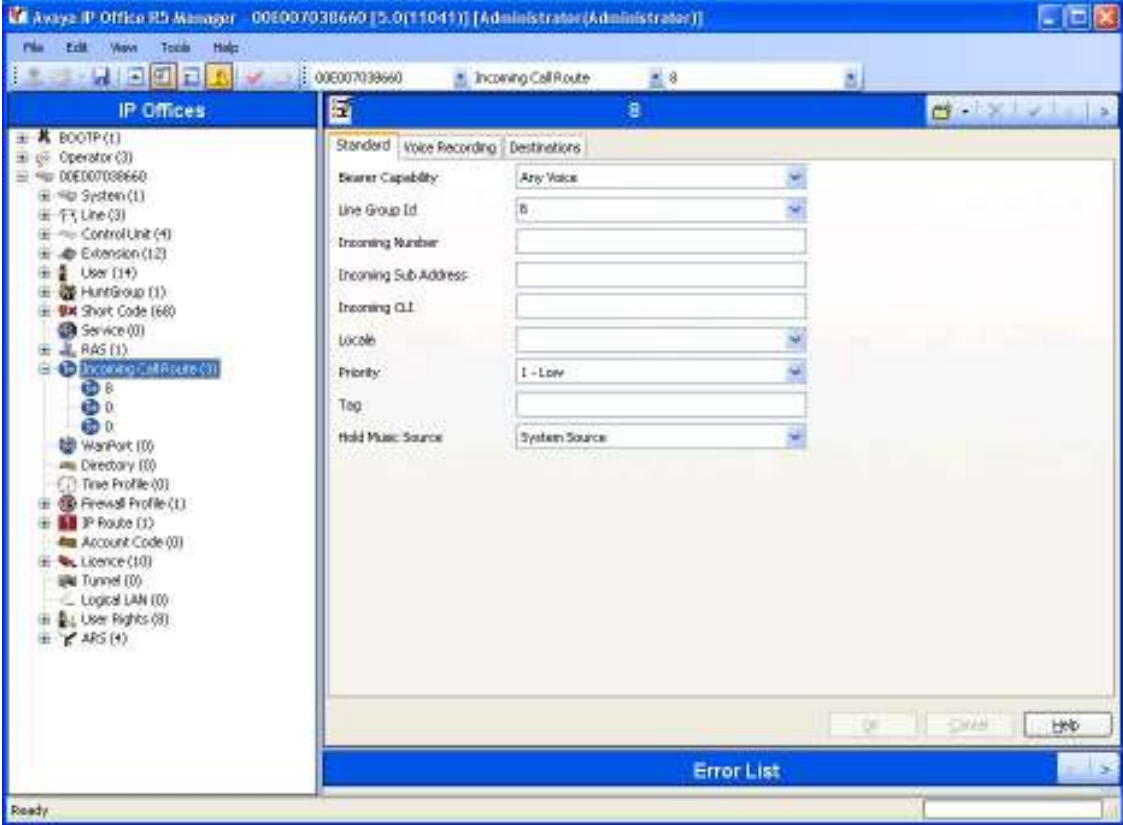


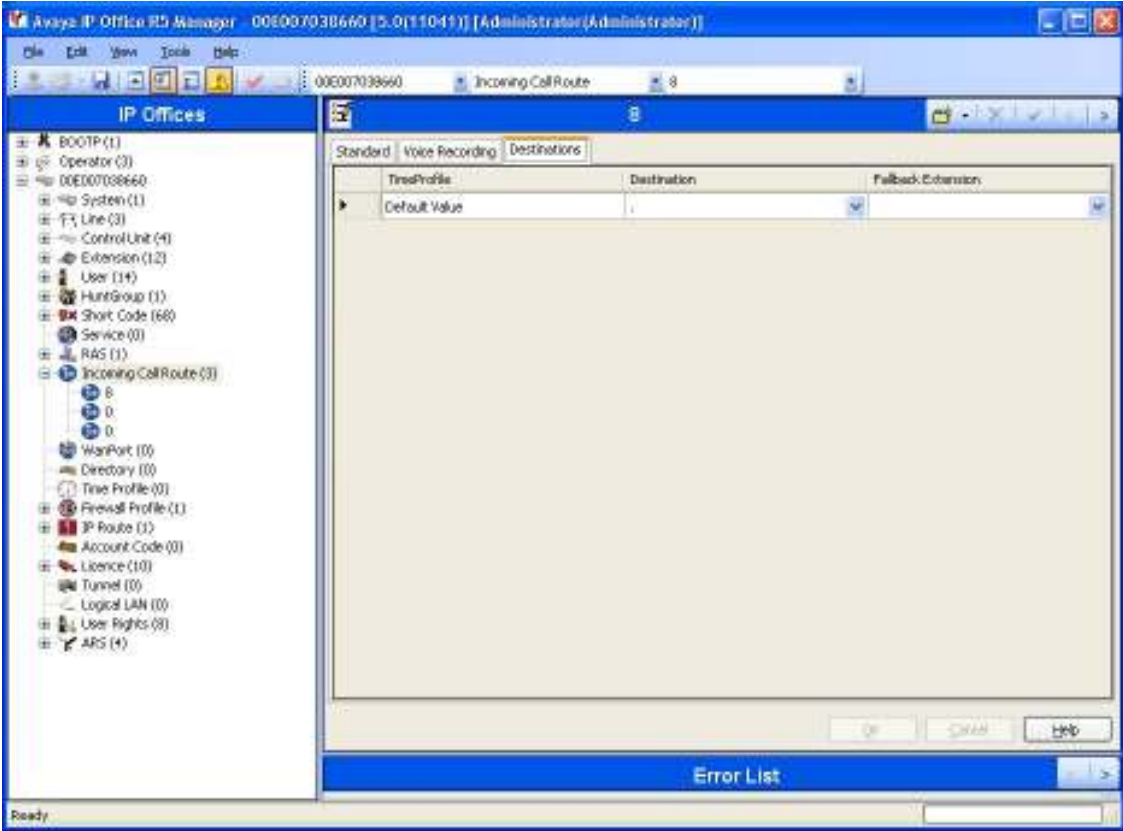
Step	Description
2.	<p>URI parameters for the SIP line. Select the <b>SIP URI</b> tab and click on <b>Add</b>.</p>  <p>The screenshot shows the Avaya IP Office R5 Manager interface. On the left is a tree view of the system configuration, including categories like IP Offices, Control Unit, Extension, User, Hunt Group, Short Code, Service, RAS, Incoming Call Route, WanPort, Directory, Time Profile, Firewall Profile, IP Route, Account Code, Licence, Tunnel, Logical LAN, User Rights, and ARS. The main window displays the configuration for 'SIP Line - Line 8'. The 'SIP URI' tab is selected, showing a table with columns for Channel, Group, Via, Local URI, and Contact. An 'Add...' button is visible on the right side of the table. At the bottom of the window, there is an 'Error List' section.</p>

Step	Description
3.	<p>Create a primary SIP URI. Enter a unique number for the <b>Incoming Group</b> and <b>Outgoing Group</b> fields. Enter * for the <b>Local URI</b>, <b>Contact</b> and <b>Display Name</b> fields. Use defaults for all other field. Press the <b>OK</b> button.</p>  <p>The screenshot shows the Avaya IP Office 8.0 Manager interface. The main window is titled 'SIP Line - Line 8'. On the left is a tree view of 'IP Offices' containing various system components like BOOTP, Operator, System, Line, Control Unit, Extension, User, HuntGroup, Short Code, Service, RAS, Incoming CallRoute, WanPort, Directory, Time Profile, Firewall Profile, IP Route, Account Code, Licence, Tunnel, Logical LAN, User Rights, and ARS. The main area displays a table for 'SIP Line SIP URI VoIP T38 Fax' with columns for Channel, Group, Wa, Local URI, and Contact. Below the table is a 'New Channel' configuration dialog. The dialog fields are: Wa (set to &lt;None&gt;), Local URI (set to *), Contact (set to *), Display Name (set to *), Registration (set to Primary), Incoming Group (set to 8), Outgoing Group (set to 5), and Max Calls per Channel (set to 10). There are 'Add...', 'Remove', and 'Edit...' buttons above the table, and 'OK' and 'Cancel' buttons to the right of the dialog. At the bottom, there is an 'Error List' section.</p>

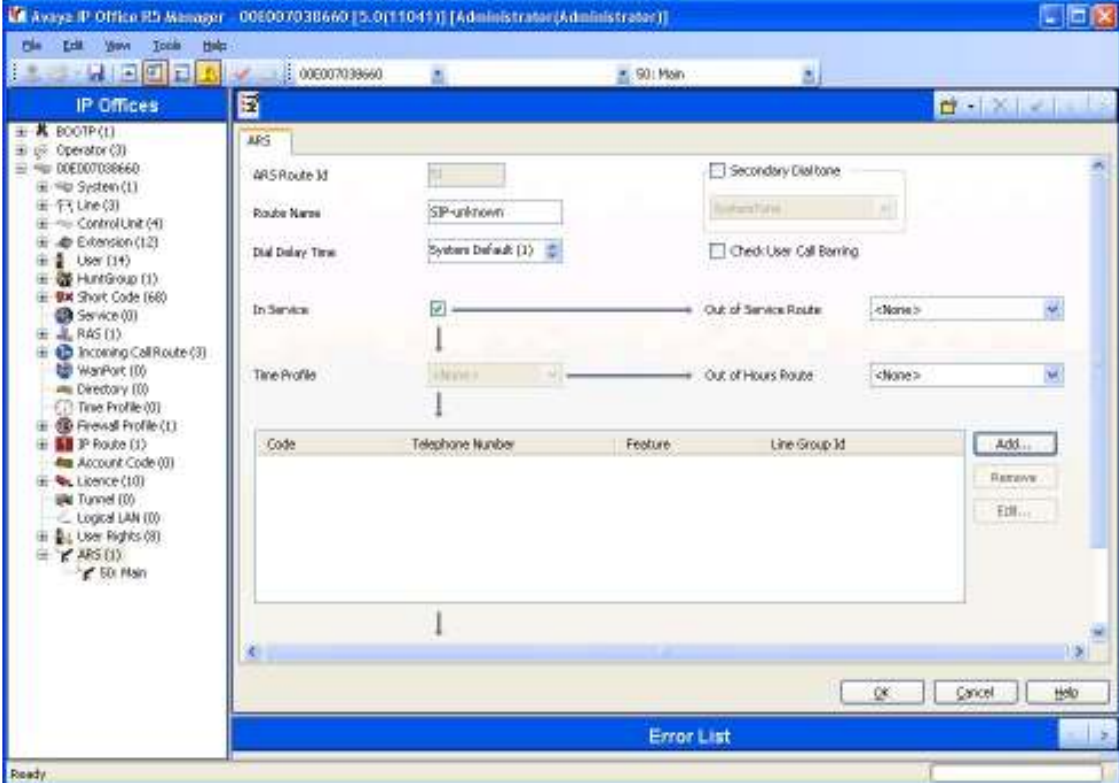
Step	Description
4.	<p>Create a SIP URI for calls received from the PSTN with withheld caller ID. Select the <b>SIP URI</b> tab and click on <b>Add</b> again. Enter a unique number for the <b>Incoming Group</b> and <b>Outgoing Group</b> fields. Enter <b>123456</b> for the <b>Local URI</b>, <b>Contact</b> and <b>Display Name</b> fields. Calls received with hidden caller ID from the PSTN will be shown as coming from this number on the MOC client. Use defaults for all other field. Press the <b>OK</b> button.</p> 


### 4.3. Administer Incoming Call Route

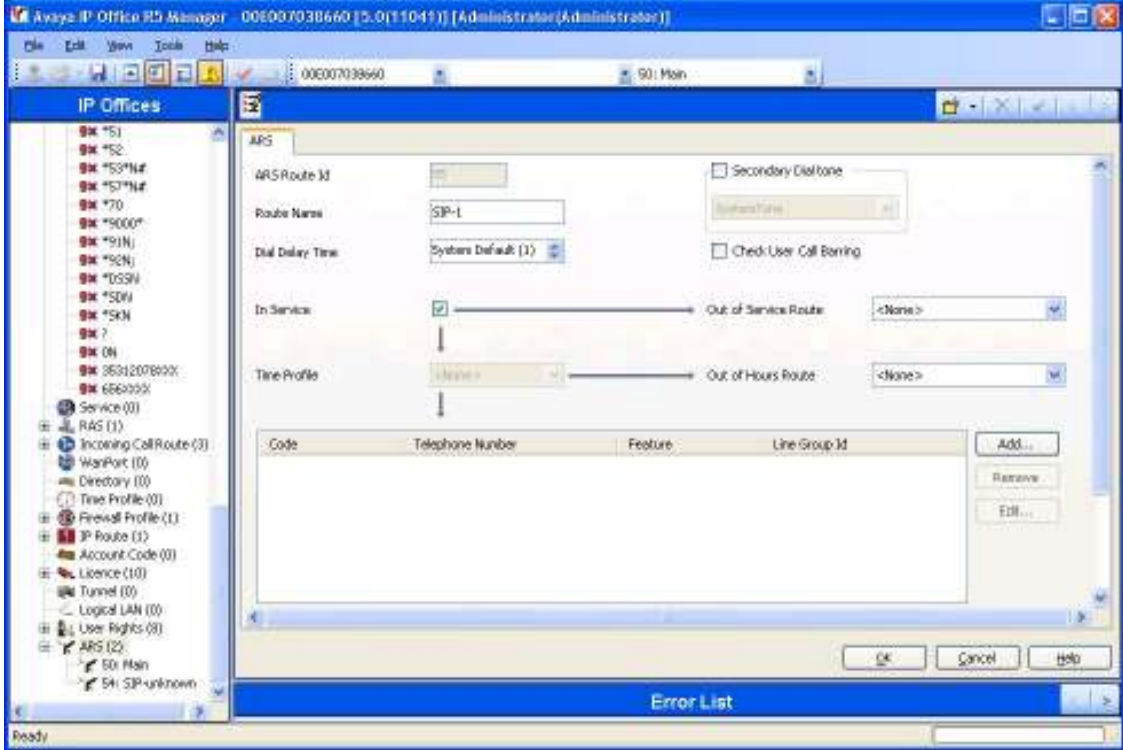
Step	Description
1.	<p>Create an Incoming Call Route for the SIP calls. Select <b>Incoming Call Route</b> in the left panel. Right-click and select <b>New</b>. Select the <b>Incoming Group</b> from the drop down list, created for receiving SIP calls for any destination in the <b>Line Group Id</b> field.</p> 

Step	Description
2.	<p>Click on the <b>Destinations</b> tab. Enter “.” in the <b>Destination</b> field. Use default values for all other fields. Press the <b>OK</b> button.</p> 

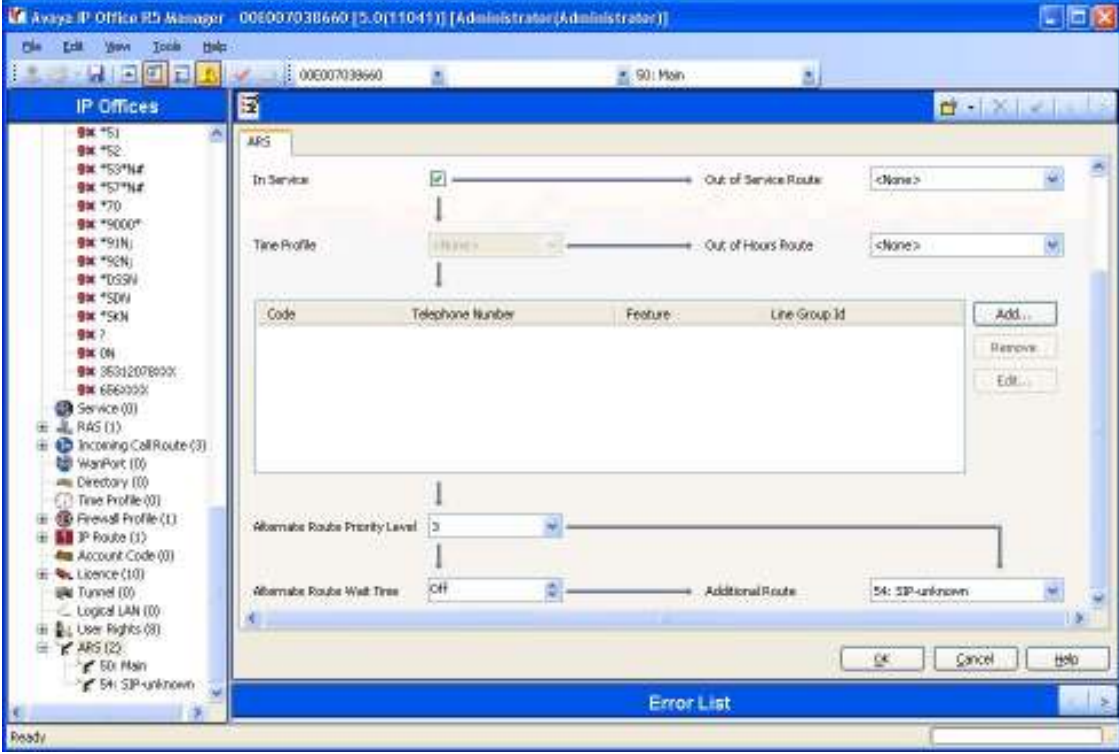
## 4.4. Administer ARS

Step	Description
1.	<p>Create the secondary Alternate Route Selection (ARS) for routing calls with withheld caller ID from the PSTN. Select <b>ARS</b> in the left panel. Right-click and select <b>New</b>. Enter a unique identifier for the route in the <b>Route Name</b> field (e.g. <b>SIP-unknown</b>) and use defaults for all other field on the <b>ARS</b> tab. Click on <b>Add</b> button.</p> 

Step	Description
2.	<p>The <b>New Short Code</b> pop-up window appears. Enter a code matching OCS phone numbers in the <b>Code</b> field, select <b>Dial</b> in the <b>Feature</b> field and enter +&lt;<b>E.164 prefix</b>&gt;<b>N</b>"@&lt;<b>Microsoft Mediation Server IP address</b>&gt;" i.e. +<b>3531656N</b>"@<b>10.10.10.43</b>". Select <b>Outgoing Group</b> from the <b>Line Group Id</b> drop down list, created for calls received from the PSTN with withheld caller ID field. Click the <b>OK</b> button. Press the <b>OK</b> button on the <b>ARS</b> tab.</p> 

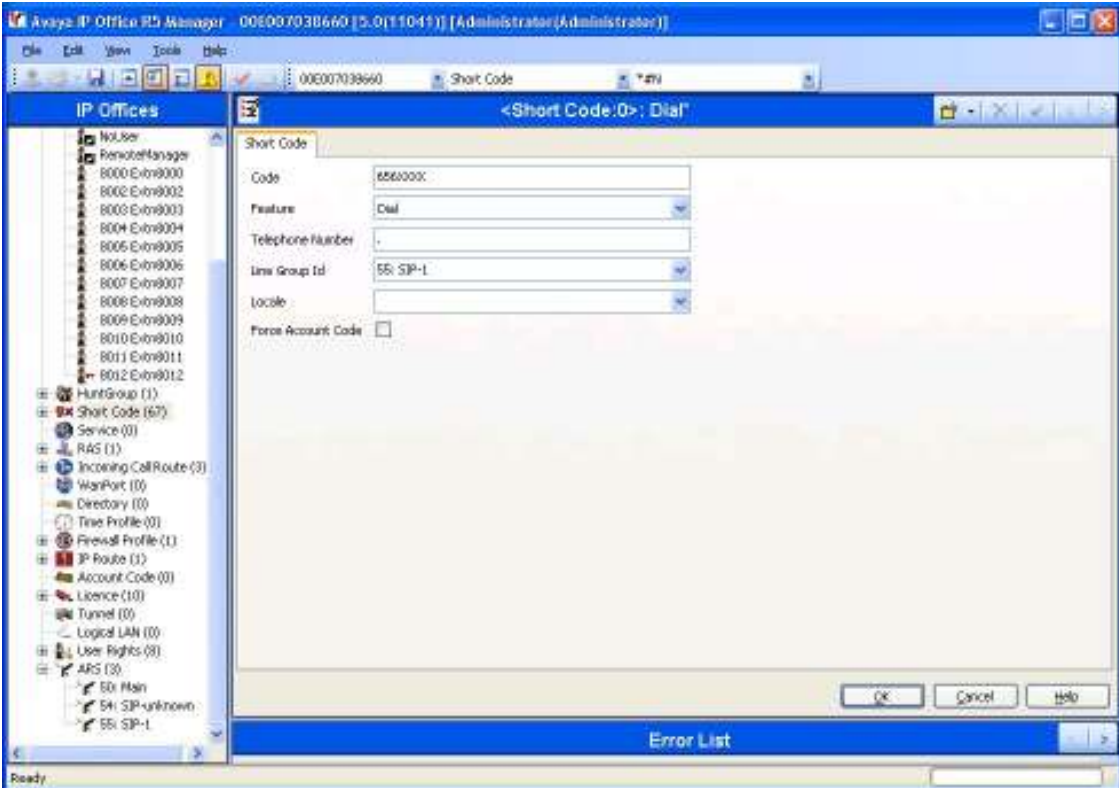
Step	Description
3.	<p>Create the primary ARS for routing calls with withheld caller ID from the PSTN. Select <b>ARS</b> in the left panel. Right-click and select <b>New</b>. Enter a unique identifier for the route in the <b>Route Name</b> field (e.g. SIP-1).</p> 

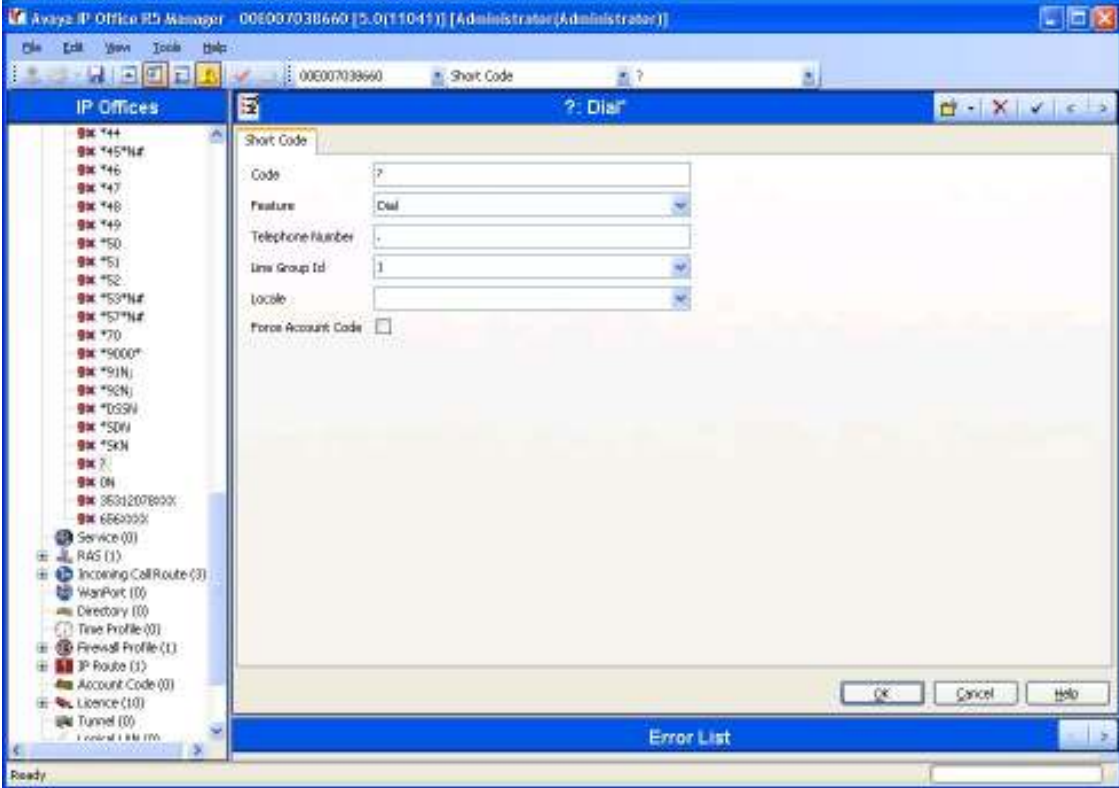


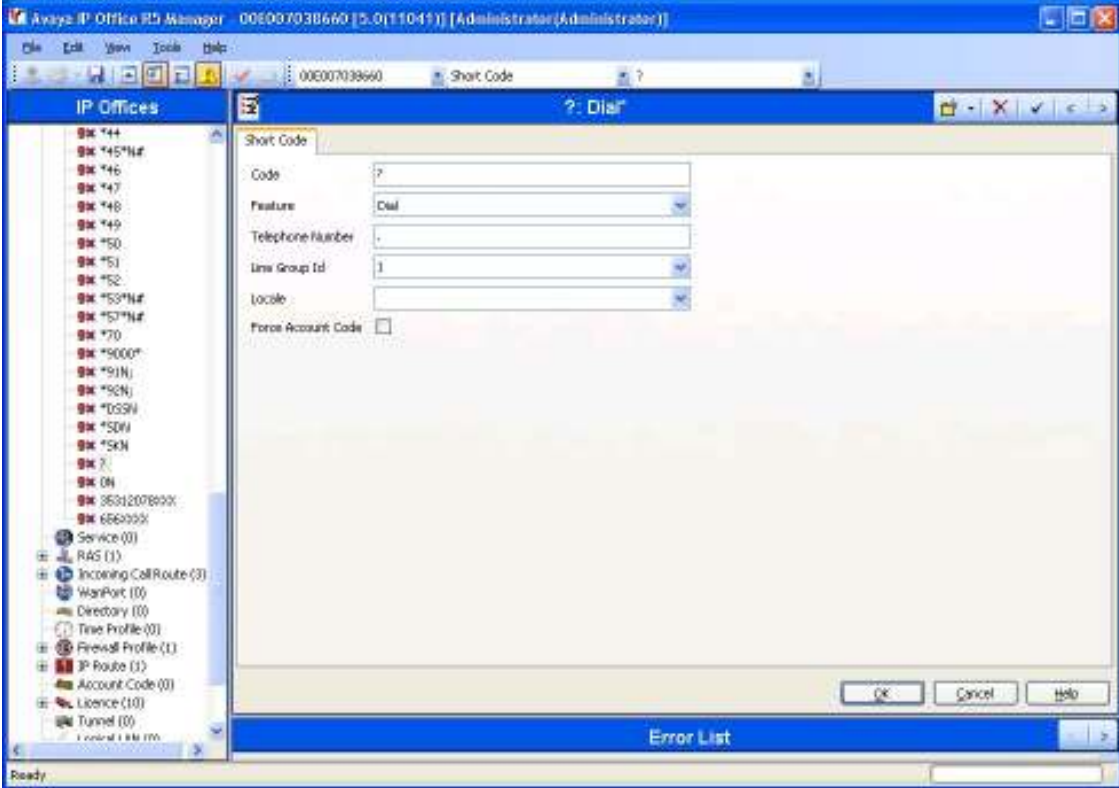
Step	Description
4.	<p>Scroll down to the bottom of the ARS tab. Select the secondary ARS created in the previous step from the <b>Additional Route</b> field (e.g. SIP-unknown). Set the <b>Alternate Route Wait Time</b> field to <b>Off</b>. Use defaults for all other fields on the <b>ARS</b> tab. Click on the <b>Add</b> button.</p>  <p>The screenshot shows the Avaya IP Office RS Manager interface. On the left is a tree view of IP Offices, with '54: SIP-unknown' selected under 'ARS (2)'. The main window displays the configuration for this ARS. Key settings include: 'In Service' checked, 'Out of Service Route' set to '&lt;None&gt;', 'Time Profile' set to '&lt;None&gt;', 'Out of Hours Route' set to '&lt;None&gt;', 'Alternate Route Priority Level' set to '3', and 'Alternate Route Wait Time' set to 'Off'. The 'Additional Route' dropdown is set to '54: SIP-unknown'. There are 'Add...', 'Remove', and 'Edit...' buttons for the route list, and 'OK', 'Cancel', and 'Help' buttons at the bottom.</p>

Step	Description
5.	<p>The <b>New Short Code</b> pop-up window appears. Enter a code matching OCS phone numbers in the <b>Code</b> field, select <b>Dial</b> in the <b>Feature</b> field and enter <b>+&lt;E.164 prefix&gt;N"@&lt;Microsoft Mediation Server IP address&gt;"</b>. Select <b>Outgoing Group</b> from the <b>Line Group Id</b> drop down list. Matching the configuration in section 4.3 above. Click the <b>OK</b> button. Press the <b>OK</b> button on the <b>ARS</b> tab.</p> <div data-bbox="435 449 1295 848" data-label="Form"> </div>

## 4.5. Administer Short Codes

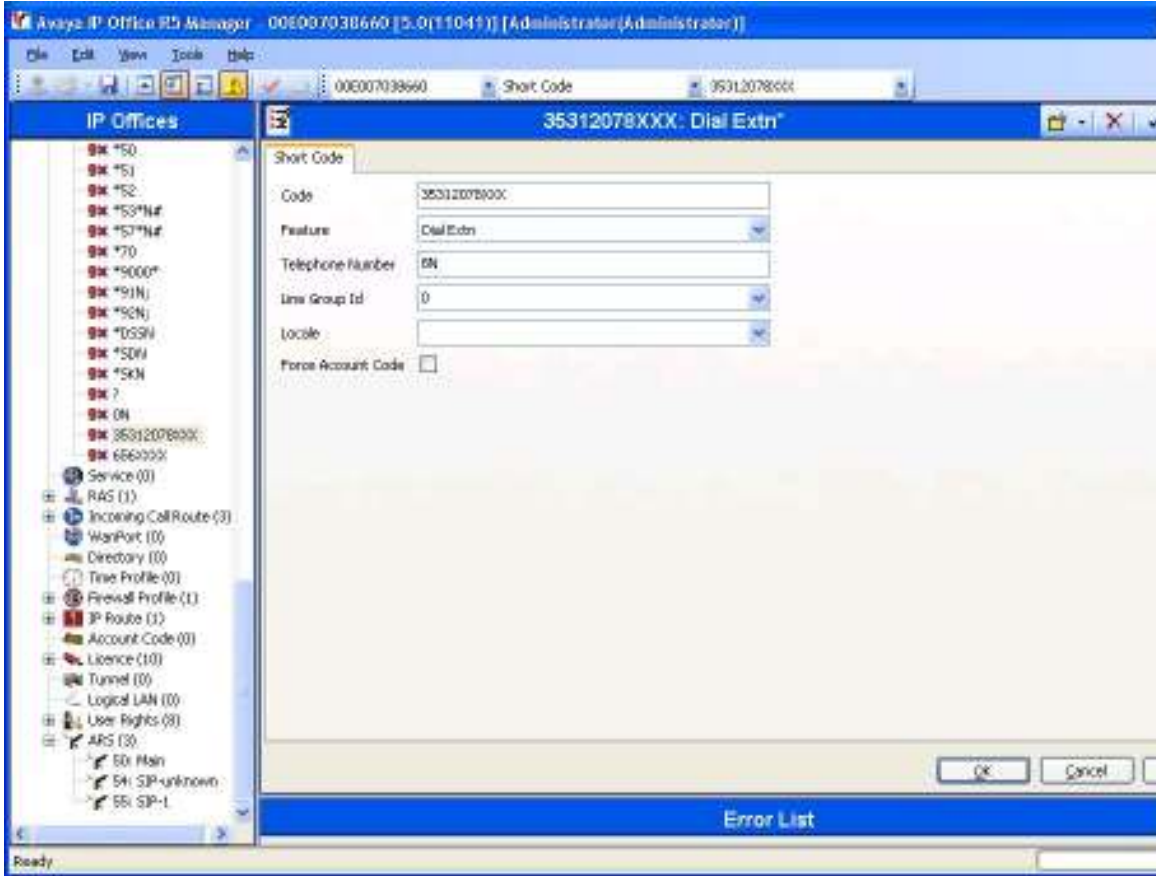
Step	Description
1.	<p>Create a short code to route calls to Microsoft Mediation Server. Select <b>Short Code</b> in the left panel. Right-click and select <b>New</b>. Enter a unique code in the <b>Code</b> field that matches phone numbers routed to Microsoft Mediation Server. Select <b>Dial</b> for the <b>Feature</b>. Select the primary ARS created previously from the <b>Line Group Id</b> drop down list. Enter “.” for the <b>Telephone Number</b> field. Use default values for all other fields. Press the <b>OK</b> button.</p> 

Step	Description
2.	<p>Create a short code to route calls to the PSTN. Select <b>Short Code</b> in the left panel. Right-click and select <b>New</b>. Enter “?” in the <b>Code</b> field. This short code will be matched for any number as a last resort if no other short code is matched. Select <b>Dial</b> for the <b>Feature</b>. Select the ISDN/PRI E1 line Outgoing Group Id from the <b>Line Group Id</b> drop down list. Enter “.” for the <b>Telephone Number</b> field. Use default values for all other fields. Press the <b>OK</b> button.</p> 

Step	Description
3.	<p>Create a short code to route calls to the PSTN. Select <b>Short Code</b> in the left panel. Right-click and select <b>New</b>. Enter “?” in the <b>Code</b> field. This short code will be matched for any number as a last resort if no other short code is matched. Select <b>Dial</b> for the <b>Feature</b>. Select the ISDN/PRI E1 line Outgoing Group Id from the <b>Line Group Id</b> drop down list. Enter “.” for the <b>Telephone Number</b> field. Use default values for all other fields. Press the <b>OK</b> button.</p>  <p>The screenshot shows the Avaya IP Office R5 Manager interface. On the left, a tree view under 'IP Offices' is expanded to 'Short Code'. The main window displays the 'Short Code' configuration dialog for a new entry with Code '?', Feature 'Dial', Telephone Number '.', and Line Group Id '1'. The 'Force Account Code' checkbox is unchecked. The dialog has 'OK', 'Cancel', and 'Help' buttons at the bottom right. An 'Error List' pane is visible at the bottom of the application window.</p>

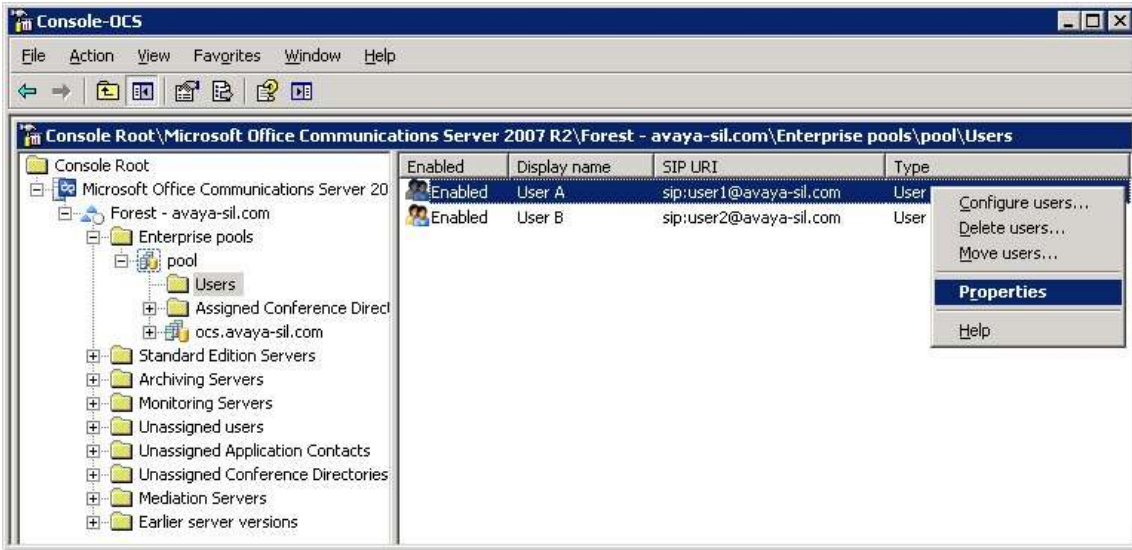
Step	Description
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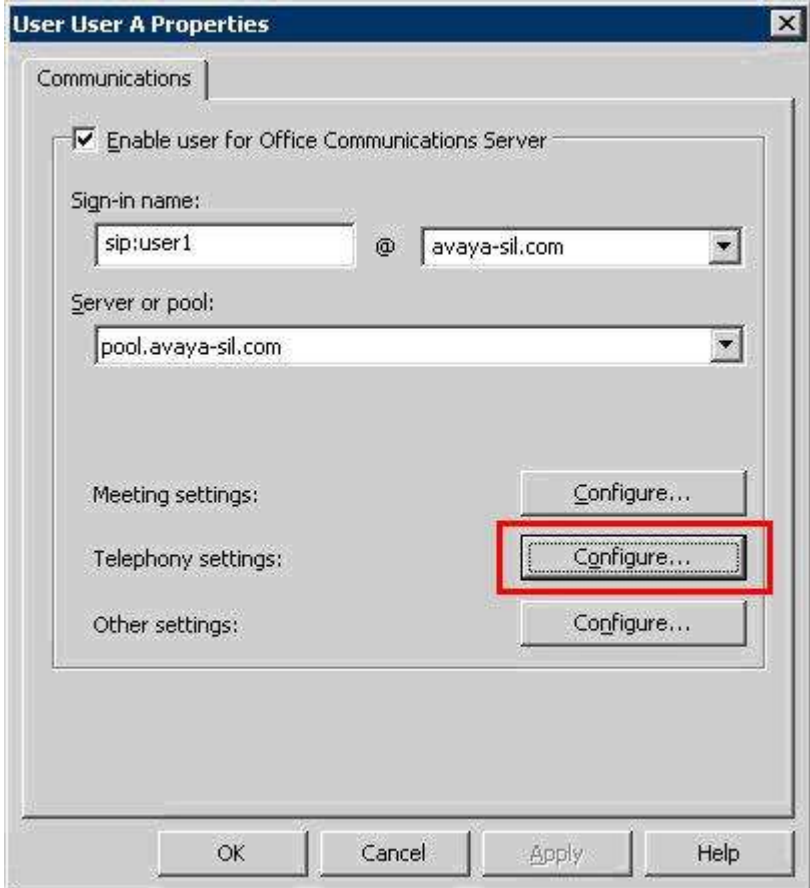
<p>4.</p>	<p>Create a short code to route calls received on the SIP line to IP Office extensions. Select <b>Short Code</b> in the left panel. Right-click and select <b>New</b>. Enter <b>35312078XXX</b> in the <b>Code</b> field. Select <b>Dial Extn</b> for the <b>Feature</b>. Enter <b>8N</b> for the <b>Telephone Number</b> field. Use default values for all other fields. Press the <b>OK</b> button.</p>
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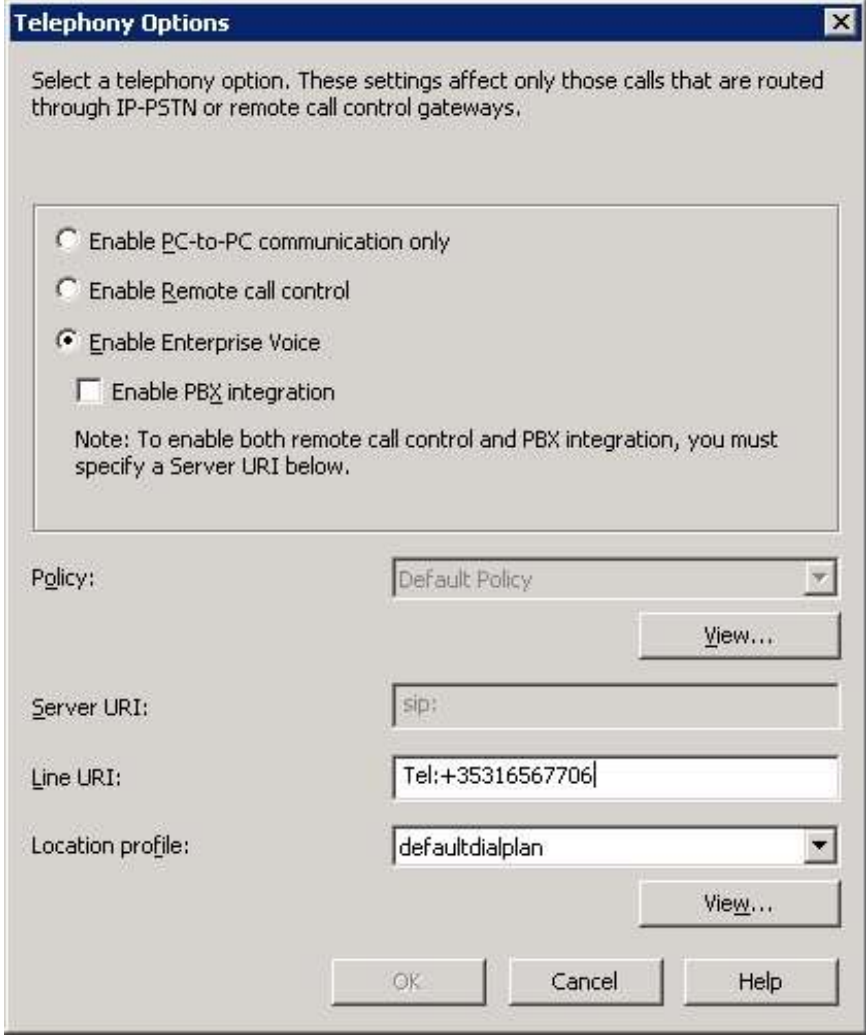
## 5. Configure Microsoft Office Communications Server


This section highlights the Microsoft Office Communications Server (OCS) 2007 R2 configuration for routing calls to and from Avaya IP Office. These Application Notes assume that basic Microsoft OCS server and Mediation Server installation and configuration have already been performed according to the guidelines provided in references [2] through [4]. These Application Notes further assume that user accounts have been created in Microsoft Active Directory and enabled for OCS.

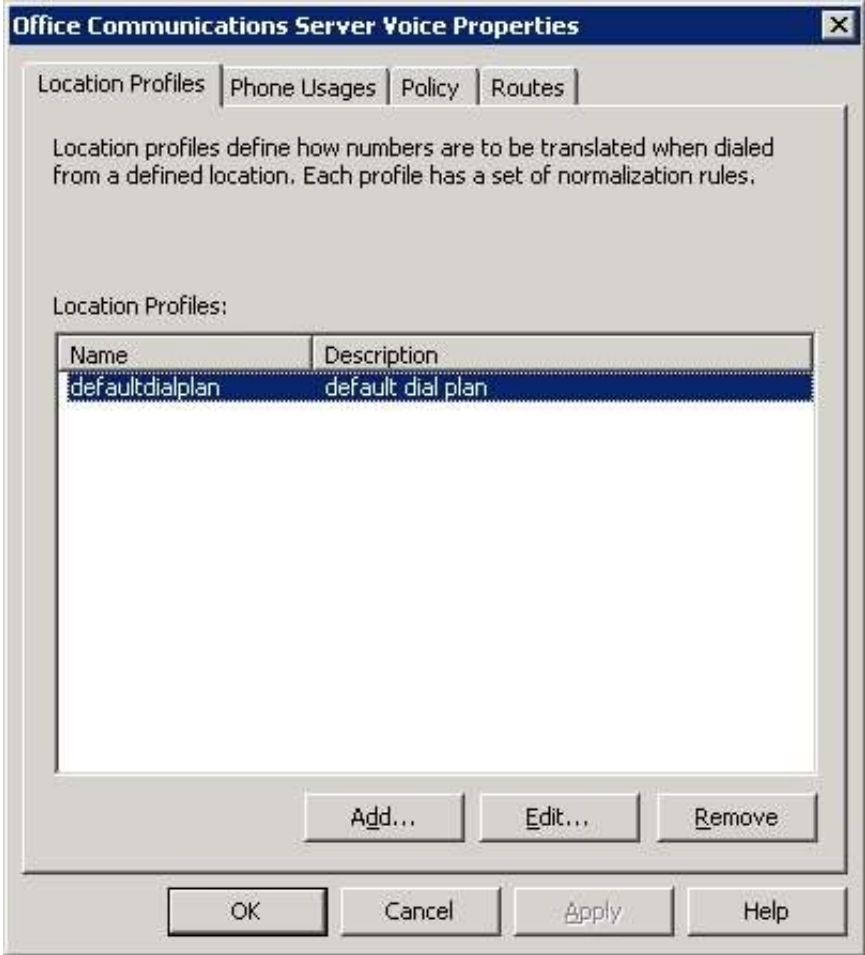
Step	Description
1.	<p>On the Microsoft OCS server, launch the <b>Microsoft Office Communications Server 2007 R2</b> Microsoft Management Console (MMC) snap-in. In the left pane, expand the <b>Forest</b> node down to the Users level (<b>Forest → Enterprise pools → &lt;name of Pool&gt; → Users</b>). In the right pane, right-click on a user (one that is to be configured as an Enterprise Voice (EV) user) and select <b>Properties</b>.</p> 

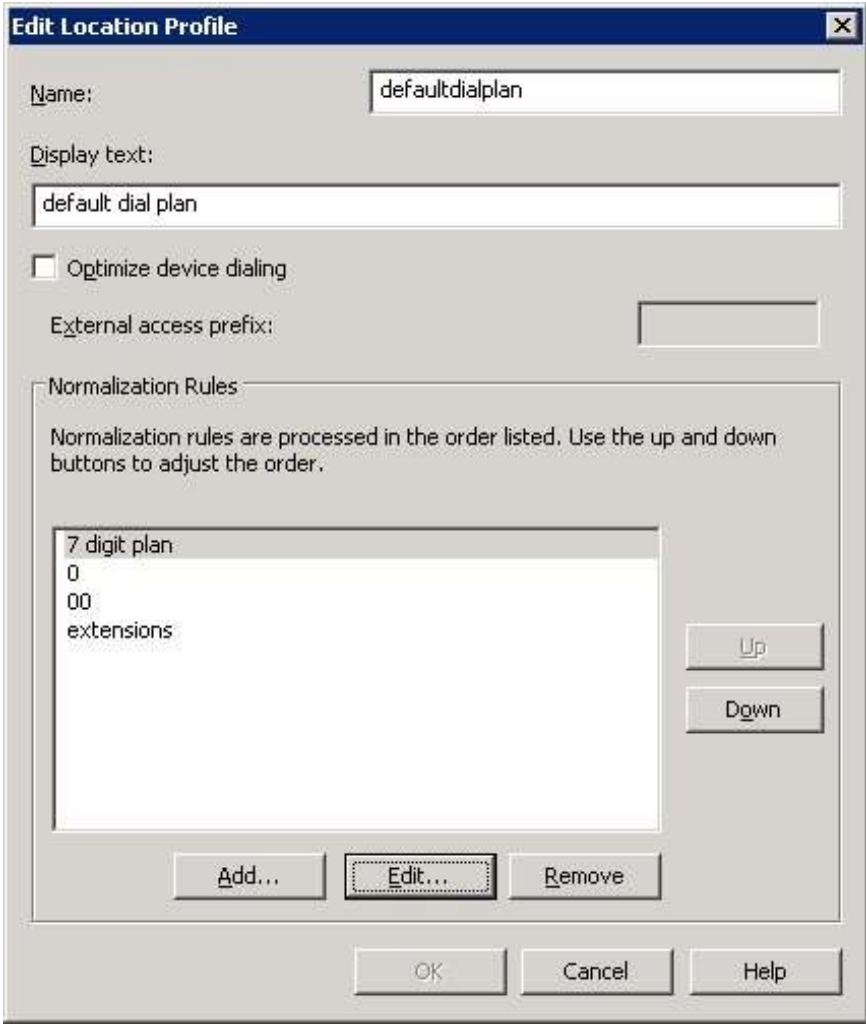
Step	Description
2.	<p>In the selected user's <b>Properties</b> dialog box, click on <b>Configure</b>.</p>  <p>The screenshot shows the 'User User A Properties' dialog box with the 'Communications' tab selected. The 'Enable user for Office Communications Server' checkbox is checked. The 'Sign-in name' field contains 'sip:user1' and the domain dropdown is 'avaya-sil.com'. The 'Server or pool' dropdown is 'pool.avaya-sil.com'. There are three 'Configure...' buttons for Meeting, Telephony, and Other settings. The Telephony settings 'Configure...' button is highlighted with a red rectangle.</p>

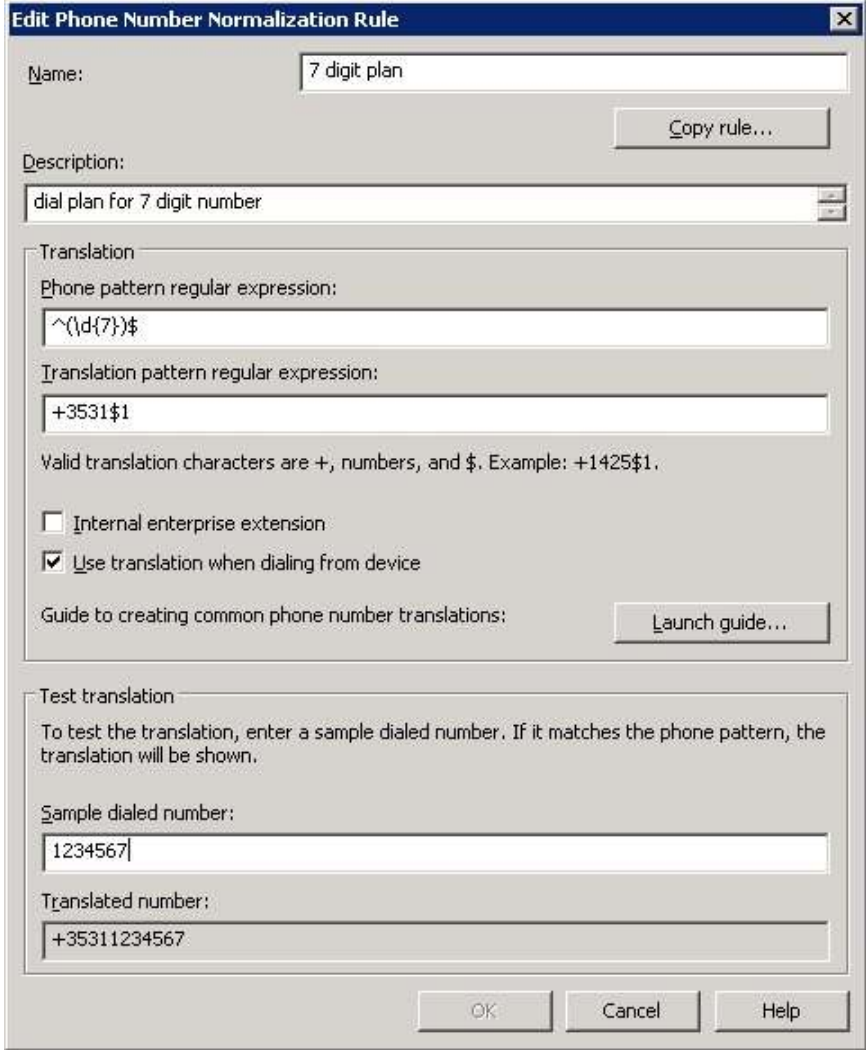


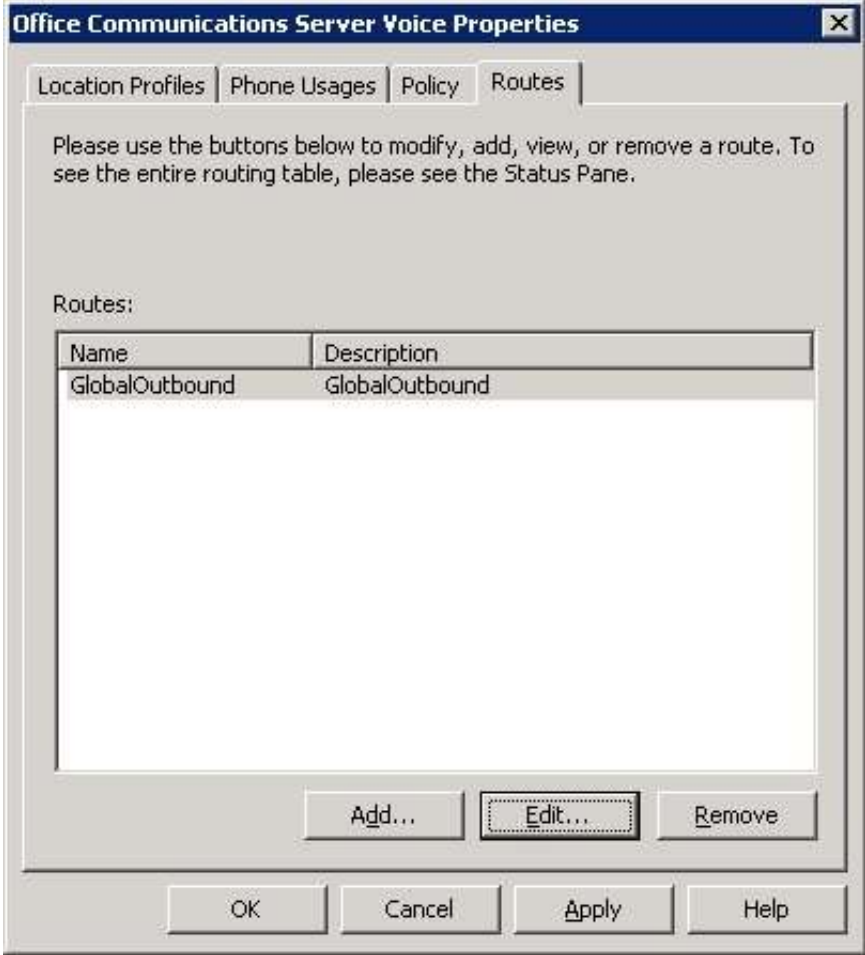
Step	Description
3.	<p>In the <b>Telephony Options</b> dialog box, select <b>Enable Enterprise Voice</b> and enter an E.164 Tel URI for <b>Line URI</b>. In the sample configuration, users were configured with Line URIs in the form of <b>Tel:+3531656xxxx</b>, where +3531656xxxx is the E.164 11-digit number assigned to the user. Click <b>OK</b>.</p> 
4.	Back in the selected user's <b>Properties</b> dialog box, click on <b>OK</b> .
5.	Repeat Steps 1 – 4 for other Microsoft OCS users to be configured as EV users.

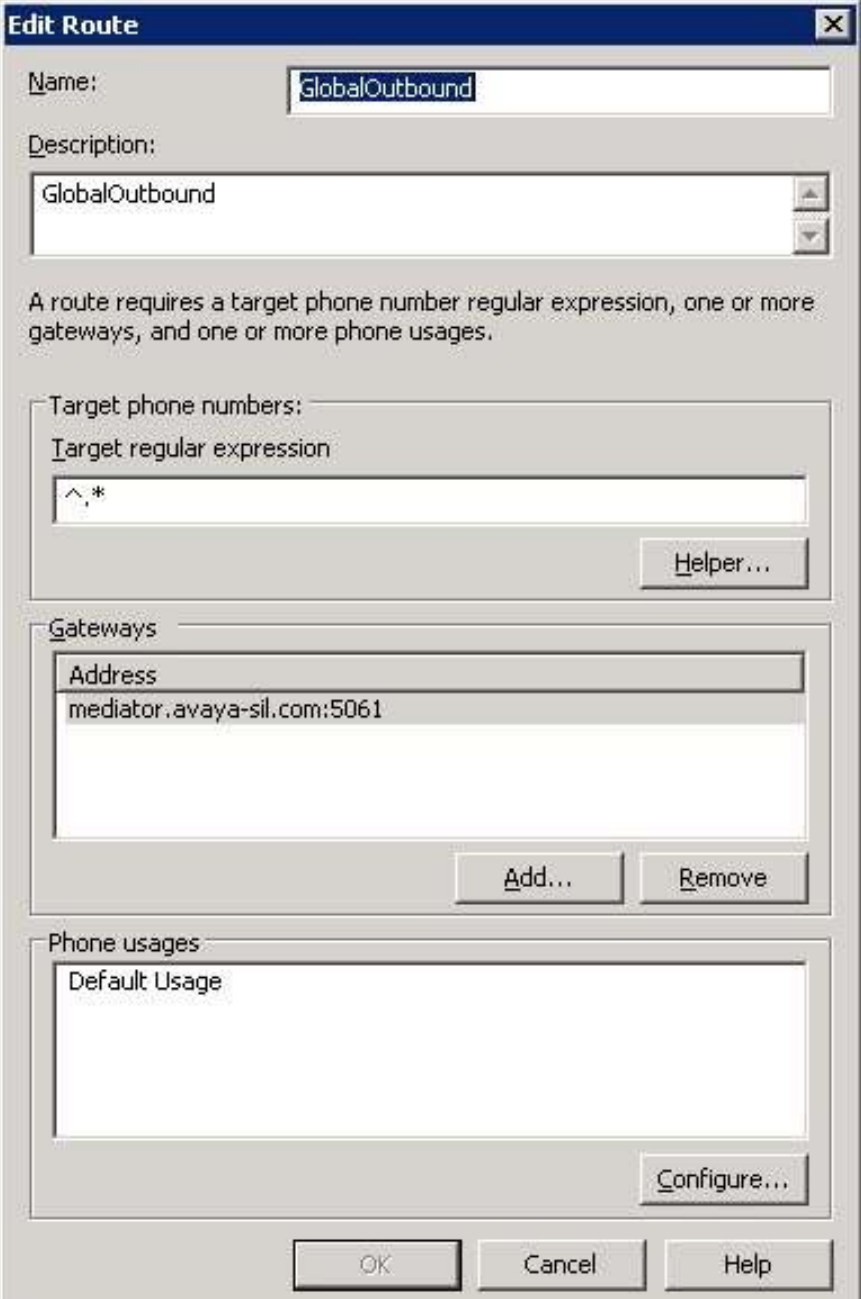
Step	Description
6.	<p>In the left pane of the <b>Microsoft Office Communications Server 2007 R2</b> MMC snap-in, right click on the <b>Forest</b> node and select <b>Properties</b> → <b>Voice Properties</b>.</p>  <p>The screenshot shows the 'Console-DCS' window with the following details:</p> <ul style="list-style-type: none"> <li><b>Left Pane:</b> Console Root &gt; Microsoft Office Communications Server 2007 R2 &gt; Forest (selected)</li> <li><b>Context Menu:</b> Properties (selected), View, New Window from Here, New Taskpad View..., Refresh, Help</li> <li><b>Sub-menu:</b> Global Properties, Voice Properties (selected), Conferencing Attendant Properties, general settings</li> <li><b>Main Pane:</b> Microsoft Office Communications Server 2007 R2, Voice Task Flow, Conferencing Attendant, Forest: Information not available in this view, Schema version: Information not available in this view, Prep state: Information not available in this view, Supported Domains, Default Routing Domain: avaya-sil.com, Meeting Settings</li> </ul>

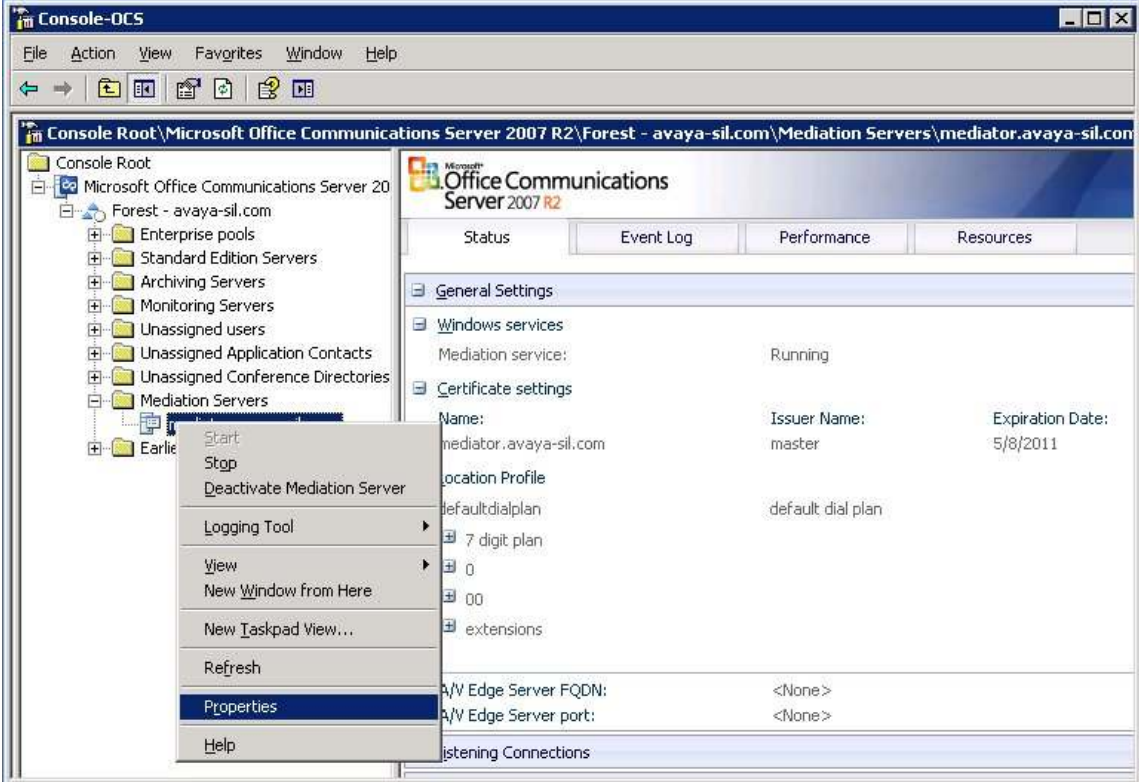
Step	Description				
7.	<p>In the <b>Office Communications Server Voice Properties</b> dialog box verify that a location profile exists in the <b>Location Profiles</b> tab. Click on <b>Add</b> or <b>Edit</b> as appropriate.</p>  <p>The screenshot shows the 'Office Communications Server Voice Properties' dialog box with the 'Location Profiles' tab selected. The dialog contains a text box explaining that location profiles define how numbers are translated when dialed from a defined location. Below this is a table titled 'Location Profiles:' with two columns: 'Name' and 'Description'. The table contains one entry: 'defaultdialplan' with the description 'default dial plan'. At the bottom of the dialog are buttons for 'Add...', 'Edit...', 'Remove', 'OK', 'Cancel', 'Apply', and 'Help'.</p> <table border="1" data-bbox="488 667 1243 1108"> <thead> <tr> <th>Name</th> <th>Description</th> </tr> </thead> <tbody> <tr> <td>defaultdialplan</td> <td>default dial plan</td> </tr> </tbody> </table>	Name	Description	defaultdialplan	default dial plan
Name	Description				
defaultdialplan	default dial plan				

Step	Description
8.	<p>Microsoft OCS location profiles define how OCS entities, such as OCS servers, enterprise pools and mediation servers, interpret and modify phone numbers. Each location profile contains an ordered set of normalization rules that translates phone numbers expressed in various formats into E.164-formatted numbers. The normalization to E.164 format provides a consistent reference for reverse number lookup (retrieving the SIP-URI associated with a user's number; see Steps 2-3) and call routing purposes. In the sample configuration, normalization rules are used to match and convert the dialed number to E.164-formatted numbers.</p> <p>Verify that a normalization rule exists or add one as appropriate. In the <b>Normalization Rules</b> section, click on <b>Add</b> or <b>Edit</b> as appropriate.</p> 

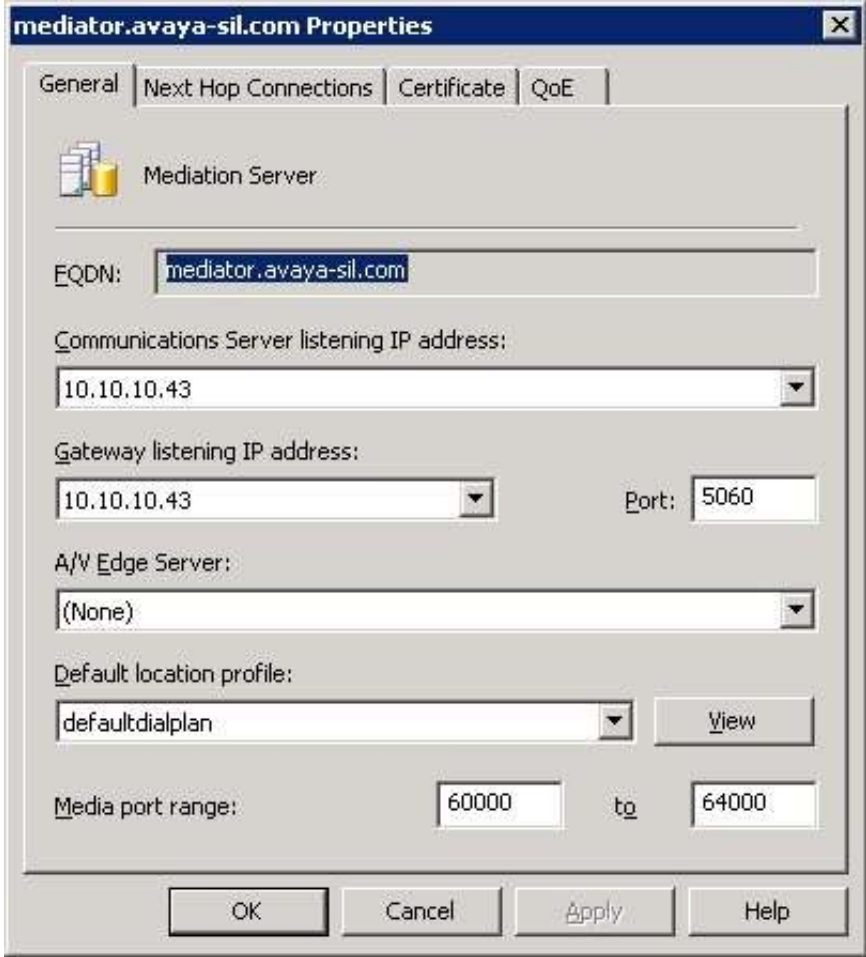
Step	Description
9.	<p>In the <b>Translation</b> section of the normalization rule, enter a <b>Phone pattern regular expression</b> that matches the PSTN number, and a <b>Translation pattern regular expression</b> that converts the matched extensions to E.164-formatted 11-digit numbers. In the example below, the normalization rule matches any 7-digit number, and prefixes the matched number with <b>+3531</b>. To confirm that the translation is configured correctly, enter a number in the <b>Sample dialed number</b> field and confirm that the number in the <b>Translated number</b> field is correct. Other values can be left at their defaults. Click on <b>OK</b>.</p> 


Step	Description				
10.	<p>In the <b>Office Communications Server Voice Properties</b> dialog box, select the <b>Routes</b> tab, and click on <b>Add</b> or <b>Edit</b> as appropriate.</p>  <p>The screenshot shows the 'Office Communications Server Voice Properties' dialog box with the 'Routes' tab selected. The dialog contains a table with the following data:</p> <table border="1" data-bbox="488 667 1243 1108"> <thead> <tr> <th>Name</th> <th>Description</th> </tr> </thead> <tbody> <tr> <td>GlobalOutbound</td> <td>GlobalOutbound</td> </tr> </tbody> </table> <p>Below the table are buttons for 'Add...', 'Edit...' (highlighted), and 'Remove'. At the bottom of the dialog are 'OK', 'Cancel', 'Apply', and 'Help' buttons.</p>	Name	Description	GlobalOutbound	GlobalOutbound
Name	Description				
GlobalOutbound	GlobalOutbound				

Step	Description
11.	<p>Verify that an outbound route exists. If not, configure as shown below. Enter a suitable <b>Name</b> and <b>Description</b>. In the <b>Target regular expression</b> use <code>^.*</code> as a catch all entry. In the <b>Gateways</b> section, select <b>Add</b>, then from the drop down list select the entry for the OCS Mediation server. Click <b>OK</b>.</p> 

Step	Description
12.	<p>In the left pane of the Microsoft Office Communications Server 2007 R2 MMC snap-in, expand the <b>Forest</b> node down to the Mediation Servers level (<b>Forest → Mediation Servers</b>). Right-click on the Mediation Server named as the gateway in Step 11 and select <b>Properties</b>.</p>  <p>The screenshot shows the Microsoft Office Communications Server 2007 R2 MMC console. The left pane displays a tree view of the console root, expanded to 'Forest - avaya-sil.com' &gt; 'Mediation Servers'. A context menu is open over the 'mediator.avaya-sil.com' server, with 'Properties' selected. The right pane shows the 'Properties' dialog for this server, with the 'General Settings' tab active. The 'Mediation service' is shown as 'Running'. The 'Certificate settings' section shows the 'Name' as 'mediator.avaya-sil.com', 'Issuer Name' as 'master', and 'Expiration Date' as '5/8/2011'. Other settings include 'Location Profile' as 'defaultdialplan', '7 digit plan' as '0', and 'extensions' as '00'. The 'A/V Edge Server FQDN' and 'A/V Edge Server port' are both set to '&lt;None&gt;'. The 'Listening Connections' section is also visible at the bottom.</p>



Step	Description
13.	<p>In the Mediation Server <b>Properties</b> dialog box, verify the settings as shown below.</p>  <p>The screenshot shows the 'mediator.avaya-sil.com Properties' dialog box with the following settings:</p> <ul style="list-style-type: none"> <li>EQDN: mediator.avaya-sil.com</li> <li>Communications Server listening IP address: 10.10.10.43</li> <li>Gateway listening IP address: 10.10.10.43</li> <li>Port: 5060</li> <li>A/V Edge Server: (None)</li> <li>Default location profile: defaultdialplan</li> <li>Media port range: 60000 to 64000</li> </ul>

Step	Description
14.	<p>Select the <b>Next Hop Connections</b> tab. In the <b>PSTN Gateway next hop</b> section, enter the IP address of the Avaya IP Office and <b>5060</b> in the <b>Port</b> field. In this sample configuration TCP was used; TLS was not supported. Click on <b>OK</b>.</p> 

## 6. Verification Steps

The following steps may be used to verify the configuration:

- Place a call from a PSTN phone to a Microsoft EV client using the EV client's full telephone number. Verify that the call is established with two-way audio and that the calling party number displayed on the EV client is an E.164-formatted 11-digit number
- From the EV client, place a call back to the PSTN phone by double-clicking on the PSTN phone number in the MOC client Recent Contacts list. Verify that the call is established with two-way audio

## 7. Conclusion

These Application Notes described the procedures for configuring call routing between Avaya IP Office and Microsoft Office Communications Server (OCS). The call routing configuration enabled voice communications between Enterprise Voice mode MOC clients and PSTN telephones.

The following issues were observed from sanity testing of basic telephony functionality:

- Calls routed to the PSTN do not show the real caller ID of the MOC client.
- Calls cannot be muted or put on hold by MOC clients
- Calls from the PSTN with withheld caller ID display a caller ID which was created in the SIP URI configuration in **Section 4.2**
- E.164 numbers may clash with existing IP Office extension numbers

## 8. Additional References

This section references the product documentation relevant to these Application Notes.

The following documentation may be obtained from <http://support.avaya.com/>.

- [1] "Avaya IP Office 5.0 Manager 7.0", Document 15-601011, Issue 23h, 16 July 2009

The following documentation may be obtained from <http://www.microsoft.com/>.

- [2] "Microsoft Office Communications Server 2007 R2 Technical Overview".  
[3] "Microsoft Office Communications Server 2007 R2 Planning and Architecture".  
[4] "Microsoft Office Communications Server 2007 R2 Deploying Enterprise Voice".  
[5] "Microsoft Office Communications Server 2007 R2 Planning for Voice".  
[6] "Microsoft Office Communications Server 2007 R2 Administering Office Communications Server 2007 R2".  
[7] "Integrating Enterprise Telephony with Office Communications Server 2007 R2", March 2009.  
[8] "Microsoft Office Communications Server 2007 R2 Important Considerations for Enterprise Voice: Please Read".

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