Avaya Solution & Interoperability Test Lab

Configuring SIP Trunking between Microsoft Office Communications Server 2007 R2, Avaya Aura™ Session Manager 6.0 and Avaya Aura™ Communication Manager 6.0 - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Microsoft Office Communications Server (OCS) 2007 R2 with Avaya Aura™ Session Manager 6.0 and Avaya Aura™ Communication Manager 6.0. The topology described herein uses the "Direct SIP" integration model as defined by Microsoft. The focus of these Application Notes is to document a reference configuration for intra-enterprise voice communications between Avaya Aura™ Communication Manager telephones and Microsoft Office Communications Server 2007 R2 clients configured in Enterprise Voice mode as defined by Microsoft. Capabilities that were verified are documented in **Section 1.1**.

Avaya Aura™ Session Manager 6.0 is a core SIP routing and integration engine that connects disparate SIP devices and applications within an enterprise. Avaya Aura™ Communication Manager is a telephony application server used here to process all calls involving traditional Avaya IP (H.323), Avaya IP (SIP), Avaya digital, and analog telephones.

In heterogeneous environments consisting of both Avaya endpoints and Microsoft Office Communicator 2007 R2 clients, Avaya recommends that OCS clients leverage Avaya's voice infrastructure today by using Microsoft's Remote Call Control integration. This TR/87 integration allows a Microsoft Office Communicator (MOC) 2007 R2 client to operate an Avaya Aura™ Communication Manager phone using CSTA CTI call control constructs. With the Avaya Aura™ AES integration, the Microsoft Office Communicator 2007 R2 client can be used exclusively to control an Avaya telephone without the need to use an Avaya client. The RCC integration method is documented in ***Application Notes for Microsoft Office Communicator R2 Remote Call Control (RCC) Integration with Avaya Communication Manager Telephones - Issue 1.0***. This document is available via the *Resource Library* at www.avaya.com or at the following link: http://www.avaya.com/master-usa/en-us/resource/assets/applicationnotes/OCS-ACM-RCC-R2.pdf

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1. Introduction

The focus of these Application Notes is to document a reference configuration for intra-enterprise voice communications between Avaya Aura™ Communication Manager telephones and Microsoft Office Communicator 2007 R2 clients configured in Enterprise Voice mode as defined by Microsoft. The topology described uses the "Direct SIP" integration model as defined by Microsoft. Sample procedures for configuring Microsoft Office Communications Server (OCS) 2007 R2 with Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager are also given. Capabilities that were verified are documented in **Section 1.1**.

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 [14-19]** for further details on Microsoft OCS 2007 R2 architecture and deployment options. Please consult **Reference [20]** for important considerations for Enterprise Voice as it contains relevant information regarding 911 (US) or 999 (UK) emergency calls and emergency services.

1.1. Interoperability Testing and Results

Microsoft OCS clients[1] configured for Enterprise Voice successfully utilized Avaya's communications infrastructure to place and receive telephone calls to and from Avaya endpoints.[2] Test cases that were performed included bi-directional calls between OCS clients and Avaya endpoints, as well as traditional telephony operations and features such as extension dialing, displays, hold/resume, transfer, conferencing, call forwarding, simultaneous ringing[3] and ring no answer.[4]

The testing that resulted in the reference configuration presented in these Application Notes was performed by Avaya.

1.2. Known Limitations

- SIP Direct Media otherwise known as the IP-IP Initial Media feature is not supported for this integration.
- Calls originated from an Avaya 96xx SIP Telephone to an OCS client with simultaneous ringing active will cause the Avaya 96xx SIP Telephone to erroneously activate a second call appearance.
- TLS and SRTP are not supported. While OCS, Session Manager and Communication Manager systems have their respective implementations of TLS and SRTP, the interoperation of these features was not tested and is not supported.

[1] The Microsoft client used in this scenario was Microsoft Office Communicator 2007 R2.
[2] Avaya SIP endpoints were included in the reference configuration.
[3] The OCS client's "Simultaneously Ring" feature can be used to simultaneously ring an Avaya endpoint when a call arrives at an OCS client. Conversely, the Communication Manager EC500 feature can be used to simultaneously ring an OCS client when a call arrives at an Avaya endpoint.
[4] The interoperability testing described here was exploratory in nature, rather than an exhaustive exercise of Microsoft OCS 2007 R2 voice telephony features.

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2. Reference Configuration

The reference configuration described throughout these Application Notes is shown in **Figure 1**.

2.1. Avaya Aura™ Session Manager

Session Manager provides core SIP routing and integration services that enables communications between disparate SIP-enabled entities, e.g., PBXs, SIP proxies, gateways, adjuncts, trunks, applications, etc., across the enterprise using centralized and policy-based routing. Configuration of Session Manager is performed from System Manager. In the reference configuration, Session Manager sends and receives SIP Invites to and from the Microsoft Mediation Server and is the primary integration point between the Avaya telecommunications infrastructure and Microsoft OCS. SIP trunks are programmed between Session Manager and Communication Manager. In addition, a SIP trunk is programmed between Session Manager and the Microsoft Mediation Server.

2.2. Avaya Aura™ Communication Manager

In the reference configuration, the Avaya G450 Media Gateway[5] provides the physical interfaces and resources for Communication Manager. Avaya 9600 Series (H.323) telephones were registered with Communication Manager over IP. Avaya 9600 Series (SIP) telephones were registered with Session Manager and supported by the Communication Manager evolution server. Avaya 2400 Series Digital and Avaya 6200 Series Analog telephones were directly connected to their respective digital and analog media modules on the G450 Media Gateway. Extension numbers 5-digits in length were assigned to Avaya endpoints. In addition, SIP trunks are programmed between Communication Manager and Session Manager.

[5] This solution is extensible to the Avaya G430 Media Gateway using internal media processing capability as well as the Avaya G650 Media Gateway using the TN2602AP Media Processor.

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2.3. Microsoft OCS 2007 R2

In the reference configuration, 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. A SIP trunk is programmed between Session Manager and the Microsoft Mediation Server. The OCS clients are registered with Microsoft OCS via a front end server pool. Extension numbers 5-digits in length were assigned to OCS clients. 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 Front End 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.

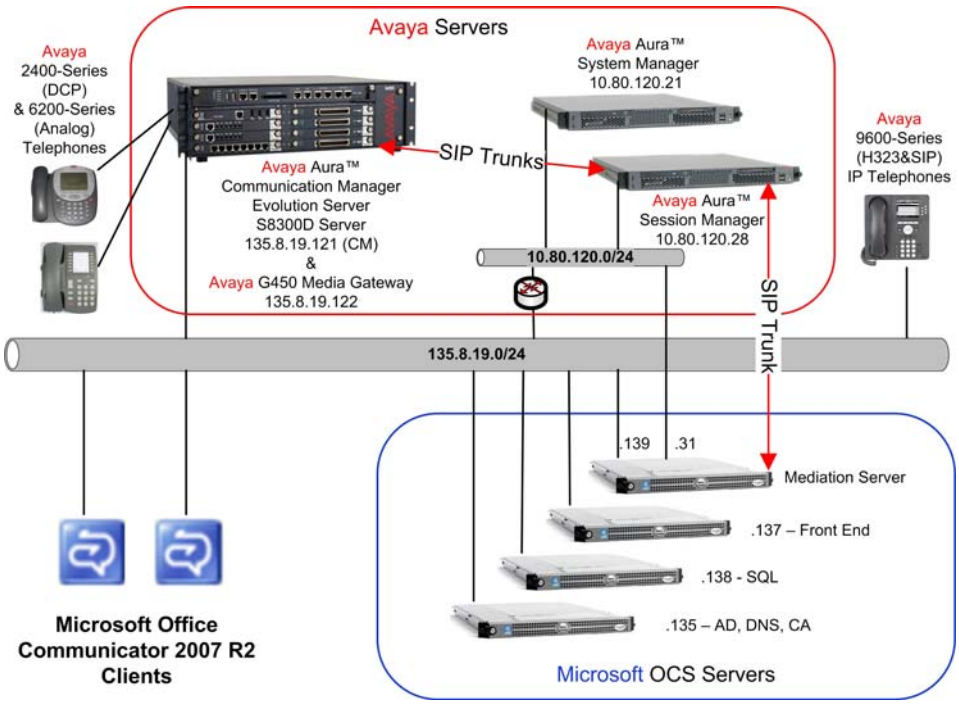

Figure 1: Reference Configuration

These Application Notes describe one possible approach to configuring call routing between Communication Manager endpoints and Microsoft OCS endpoints. The following user experience goals were considered in formulating the approach:

- An OCS client should be able to call an Avaya telephone using extension dialing. To address this, Microsoft OCS can be configured with one or more "normalization rules[6]" that match the dialed number. As shown in **Section 4.2 Step 11**, on Session Manager, a Dial Pattern can be configured to match 5-digit called party numbers and forward the calls to Communication Manager for routing to Avaya endpoints.

[6] Normalization rules define matching criteria for various number strings and translations for converting the strings into private extension numbers.

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- An OCS client should be able to call an Avaya telephone from the OCS client's Recent Contacts list. "Normalization rules" also apply to incoming calling party numbers.
- An Avaya telephone should be able to call an OCS client using extension dialing.[7] To address this, the Uniform Dial Plan (UDP) available with Communication Manager is used to share a common dial plan among a group of servers. UDP provides extension-to-extension dialing between two or more private switching systems. In the reference configuration, UDP is used to route calls from Communication Manager to Session Manager. As shown in **Section 4.2 Step 8**, on Session Manager, a Dial Pattern can be configured to match 5-digit called party numbers and forward the call to the Microsoft Mediation Server for routing to OCS endpoints. For a complete description of the UDP feature, see **Reference [9]**.
- An Avaya telephone should be able to call an OCS client from the Avaya telephone's missed and answered call logs.

The flow for an outbound call from an OCS client is as follows. When an OCS 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 is assigned to another OCS client. If so, Microsoft OCS sends the call to the called user's OCS client. If not, Microsoft OCS looks up a call routing table for a match of the 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 reference configuration, is Session Manager. Based on the called party number and originating location information, Session Manager routes the call to Communication Manager, which delivers the call to the Avaya telephone. The SIP Invite from the Microsoft Mediation Server towards Session Manager may also contain the calling party number information. This information is forwarded by Session Manager to Communication Manager.

When an Avaya telephone is used to call an OCS client, Communication Manager applies dial plan and routing rules to route the call to Session Manager. Based on the called party number and originating location information, Session Manager looks up the corresponding routing policy, applies any necessary adaptations, and routes the call to the Microsoft Mediation Server. The SIP Invite towards the Microsoft Mediation Server may also contain the calling party number information of the Avaya telephone. This information is forwarded by Session Manager to the Microsoft Mediation Server and OCS presents the calling party number to the OCS client for display to the user and for placement in the Recent Contacts list.

[7] **Reference [12]** documents an approach where E.164-formatted telephone numbers are assigned to Microsoft OCS clients.

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3. Equipment and Software Validated

The following equipment and software were used for the reference configuration provided:

Equipment & Software	Version
Avaya S8300D Server	Avaya Aura™ Communication Manager 6.0 (R016x.00.0.345.0 with patch 1002) System Platform 6.0.0.0.11
Avaya G450 Media Gateway	30.11.3/0
Avaya Aura™ Session Manager	6.0.0.0.600020
Avaya Aura™ System Manager	6.0.6.0 w/6.0 GA PATCH 6.0.0.600019
Avaya 9630 IP (H.323) Telephone	3.1
Avaya 9640 IP (SIP) Telephone	2.6.0.0
Avaya 9630 IP (SIP) Telephone	2.6.0.0
Avaya 9620 IP (SIP) Telephone	2.6.0.0
Avaya 2410 (Digital) Telephone	N/A
Avaya 6210 (Analog) Telephone	N/A
Microsoft Active Directory, DNS Server, and Certification Authority on Microsoft Windows Server 2003 R2 Enterprise Edition Service Pack 2	Version 5.2 R2 (Build 3790.srv03_sp2_rtm.070216-1710: 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: 3.5.6907.83 (Volume) Windows OS : Version 5.2 R2 (Build 3790.srv03_sp2_rtm.070216-1710 : Service Pack 2)
Microsoft SQL 2005 Server on Microsoft Windows Server 2003 R2 Enterprise Edition x64 Edition Service Pack 2	9.00.3042.00 Windows OS : Version 5.2 R2 (Build 3790.srv03_sp2_rtm.070216-1710 : Service Pack 2)
Microsoft Mediation Server on Microsoft Windows Server 2003 R2 Enterprise Edition x64 Edition Service Pack 2	OCS 2007 R2: 3.5.6907.37 (Volume)
Microsoft Office Communicator 2007 R2 on Microsoft Windows XP Professional Version SP3	R2 : 3.5.6907.196 Windows OS: 2600.xpsp_sp3_gdr.100216-1514 : Service Pack 3

Table 1: Equipment/Software List

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4. Avaya Aura™ Session Manager

These Application Notes assume basic administration of System Manager and Session Manager has already been performed, and that the necessary integration with Communication Manager has already been implemented according to **References [1-11]**. Configuration of Session Manager is performed from Avaya Aura™ System Manager. To invoke the Avaya Aura™ System Manager Common Console, launch a web browser, enter https://<*IP address of the Avaya Aura™ System Manager server*>/SMGR in the URL, and log in with the appropriate credentials.

4.1. Background

Session Manager serves as a central point for supporting SIP-based communication services in an enterprise. The various SIP network components are represented as "SIP Entities" and the connections/trunks between Session Manager and those components are represented as "Entity Links". Thus, rather than connecting to every other SIP Entity in the enterprise, each SIP Entity simply connects to Session Manager and relies on Session Manager to route calls to the correct destination. This approach reduces the dial plan and trunking administration needed on each SIP Entity, and consolidates said administration in a central place, namely Avaya Aura™ System Manager.

When calls arrive at Session Manager from a SIP Entity, Session Manager applies SIP protocol and numbering modifications to the calls. These modifications, referred to as "Adaptations", are sometimes necessary to resolve SIP protocol differences between disparate SIP Entities, and also serve the purpose of "normalizing" the calls to a common or uniform numbering format, which allows for simpler administration of routing rules in Session Manager. Session Manager then matches the calls against certain criteria embodied in profiles termed "Dial Patterns", and determines the destination SIP Entities based on "Network Routing Policies" specified in the matching Dial Patterns. Lastly, before the calls are routed to the respective destinations, Session Manager again applies Adaptations in order to bring the calls into conformance with the SIP protocol interpretation and numbering formats expected by the destination SIP Entities.

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4.2. Network Routing Policies (NRP)

Network Routing Policies define how Session Manager routes calls between SIP network elements. A Network Routing Policy is dependent on the administration of several inter-related items including:

- SIP Domains – SIP Domains are the domains for which Session Manager is authoritative in routing SIP calls. In other words, for calls to such domains, Session Manager applies Network Routing Policies to route those calls to SIP Entities. For calls to other domains, Session Manager routes those calls to another SIP proxy (either a pre-defined default SIP proxy or one discovered through DNS).
- Locations – Locations define the physical and/or logical locations in which SIP Entities reside. Call Admission Control (CAC) / bandwidth management may be administered for each location to limit the number of calls to and from a particular Location.
- Adaptations – Adaptations are used to apply any necessary protocol adaptations, e.g., modify SIP Invite, and apply any necessary digit conversions for the purpose of inter-working with specific SIP Entities.
- SIP Entities – SIP Entities represent SIP network elements such as Session Manager instances, Communication Manager systems, Microsoft Mediation Servers, Session Border Controllers, SIP gateways, SIP trunks, and other SIP network devices.
- Entity Links – Entity Links define the SIP trunk/link parameters, e.g., ports, protocol (UDP/TCP/TLS), and trust relationship, between Session Manager instances and other SIP Entities.
- Time Ranges – Time Ranges specify customizable time periods, e.g., Monday through Friday from 9AM to 5:59PM, Monday through Friday 6PM to 8:59AM, all day Saturday and Sunday, etc. A Network Routing Policy may be associated with one or more Time Ranges during which the Network Routing Policy is in effect. For example, for a Dial Pattern administered with two Network Routing Policies, one Network Routing Policy can be in effect on weekday business hours and the other Network Routing Policy can be in effect on weekday off-hours and weekends. In the reference configuration no restrictions were placed on calling times.
- Routing Policies – Routing policies identify the SIP Entity to which calls should be routed as well as the applicable Time of Day range.
- Dial Patterns – A Dial Pattern specifies a set of criteria and a set of Network Routing Policies for routing calls that match the criteria. The criteria include the called party number and SIP domain in the Request-URI, and the Location from which the call originated. For example, if a call arrives at Session Manager and matches a certain Dial Pattern, then Session Manager selects one[8] of the Network Routing Policies specified in the Dial Pattern. The selected Network Routing Policy in turn specifies the SIP Entity to which the call is to be routed. Note that Dial Patterns are matched after ingress Adaptations have already been applied.

[8] The Network Routing Policy in effect at that time with the highest ranking is attempted first. If that Network Routing Policy fails, then the Network Routing Policy with the next highest ranking is attempted, and so on.

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To view the sequenced steps required for configuring network routing policies, click on "**Routing**" in the left pane of the Avaya Aura™ System Manager (see **Figure 2**).

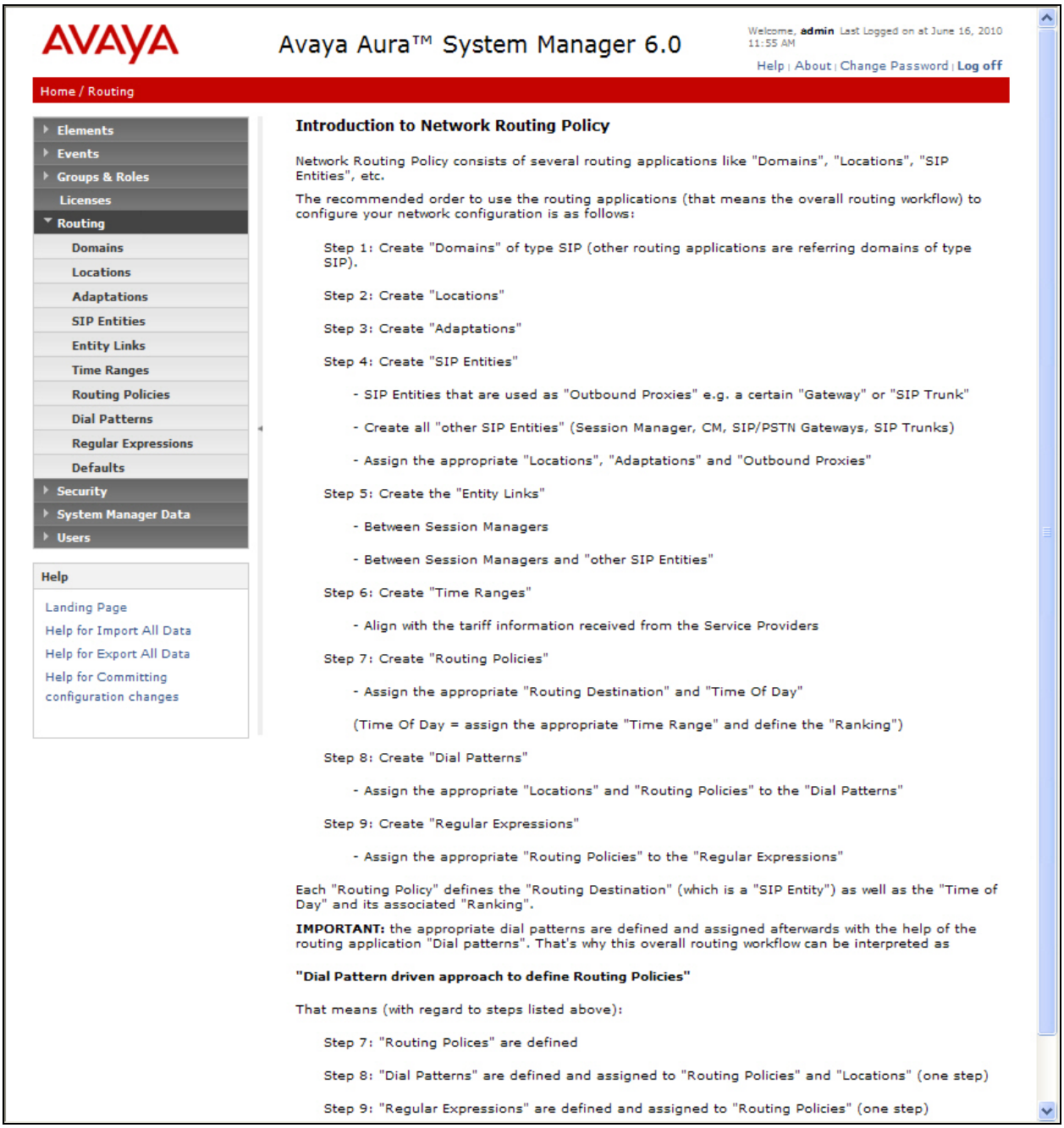

Figure 2: Introduction to Network Routing Policy (NRP) Page

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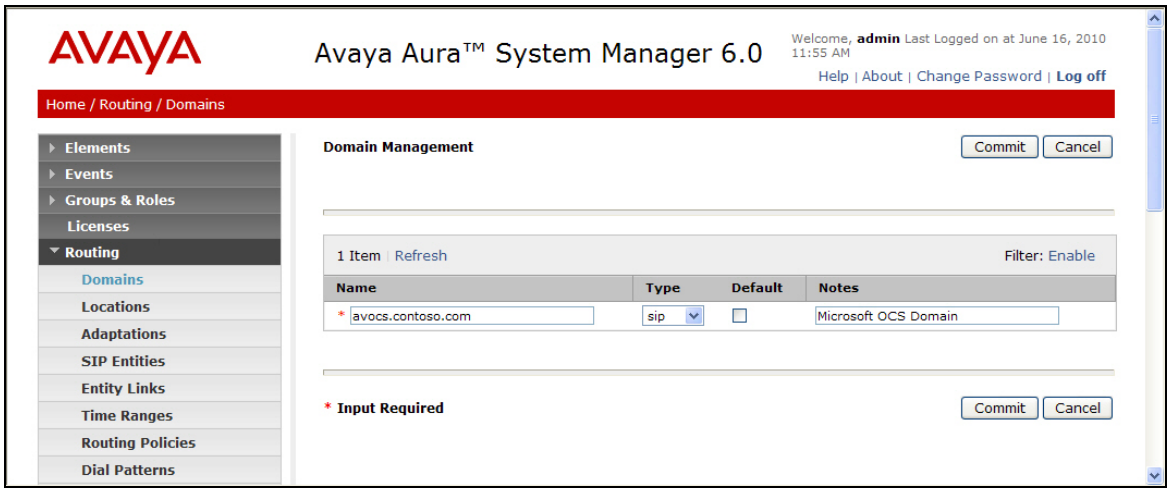
Step	Description
1.	Create a new **SIP Domain** for Microsoft OCS. Select **Routing → Domains**. Click **New** and enter the domain used by Microsoft OCS in the **Name** field, verify that "**sip**" is selected in the **Type** drop-down menu, and enter any optional notes in the **Notes** field. In the reference configuration, the SIP domain was "**avocs.contoso.com"**. Note that the domain used by Microsoft OCS can be identified by following **Steps 15** and **16** in **Section 6.** Click **Commit**. 

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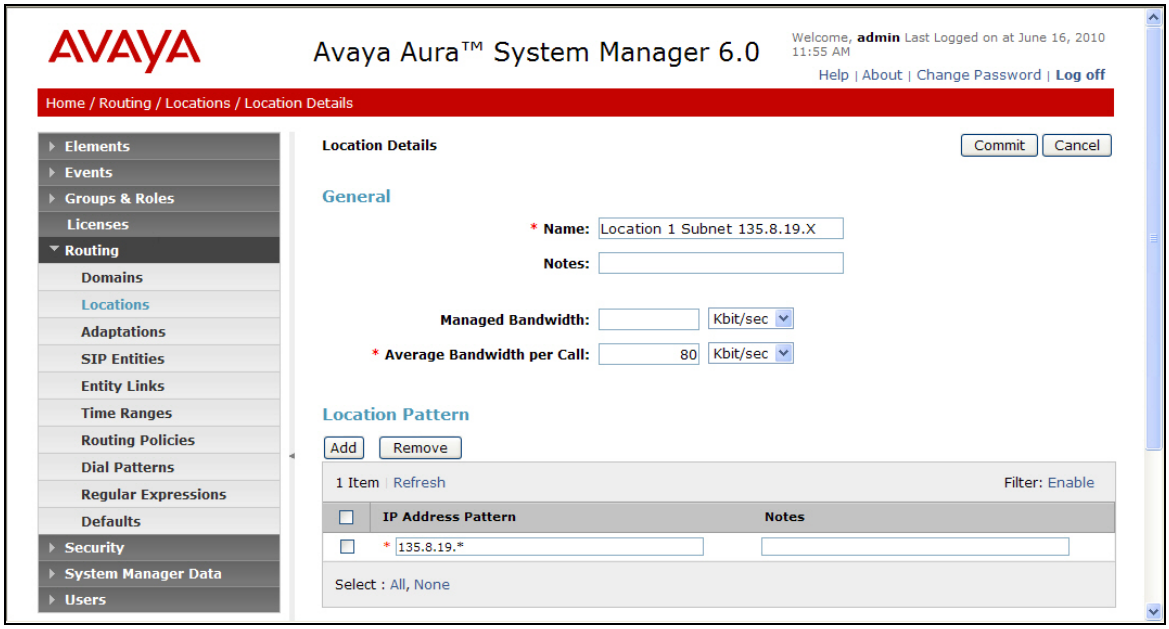
Step	Description
2.	Create two new **Locations**. Select **Routing → Locations**. Click **New** and enter an appropriate name such as *Location 1 Subnet 135.8.19.X* in the **Name** field and optional notes in the **Notes** field. Then, under **Location Pattern**, click **Add**. In the **IP Address Pattern** field, enter the IP address pattern associated with this location. In the reference configuration, **135.8.19.*** was entered in the **IP Address Pattern** field. This address pattern was associated with the Communication Manager. Click **Commit**.



Select **Routing → Locations**. Click **New** and enter an appropriate name such as *Location 1 Subnet 10.80.120.X* in the **Name** field and optional notes in the **Notes** field. Then, under **Location Pattern**, click **Add**. In the **IP Address Pattern** field, enter the IP address pattern associated with this location. In the reference configuration, **10.80.120.*** was entered in the **IP Address Pattern** field. This address pattern was associated with the OCS Mediation Server. Click **Commit** (not shown).

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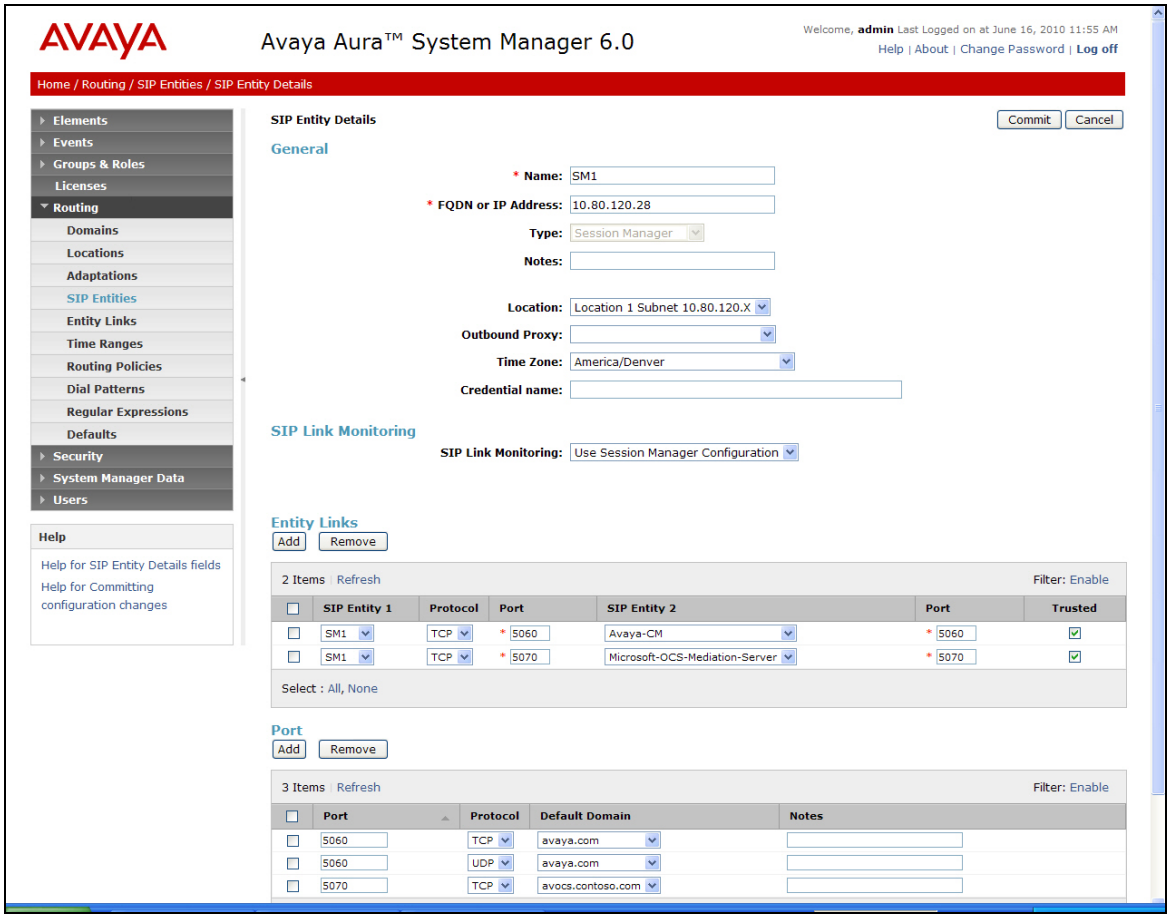
Step	Description
3.	Assign the newly created **SIP Domain** to an available TCP/IP Port on the Session Manager. In a single SIP domain environment, port 5060 can be used as the receive port for all SIP messaging. However, as one approach to accommodate multiple domains in a shared Session Manager environment, port 5070 was used rather than 5060 for the SIP entity associated with the Microsoft Mediation Server. Session Manager used "avocs.contoso.com" as the default SIP domain for SIP TCP messages received on port 5070. When Session Manager receives a request where the host part of the request-URI is the IP address of the Session Manager, it associates one of the administered domains with the TCP/IP port on which the request was received. On the existing Session Manager **SIP Entity Details** form, add a port to handle the new domain used by Microsoft OCS. Select **Routing → SIP Entities.** Select the entity associated with the existing Session Manager and click **Edit**. Under the **Port** section, click **Add**. When the new Port line appears, enter "**5070"** in the **Port** field, select TCP for the **Protocol** field, select "**avocs.contoso.com**" from the Default Domain drop-down menu, and enter any optional notes in the **Notes** field. In the reference configuration, the SIP domain was "**avocs.contoso.com"**. Note that the domain used by Microsoft OCS can be identified by following **Steps 15** and **16** in **Section 6.** Note that this screenshot also shows a pre-existing **Entity Link** from Session Manager (SIP Entity SM1) to Communication Manager (SIP Entity Avaya-CM). Click **Commit**. 

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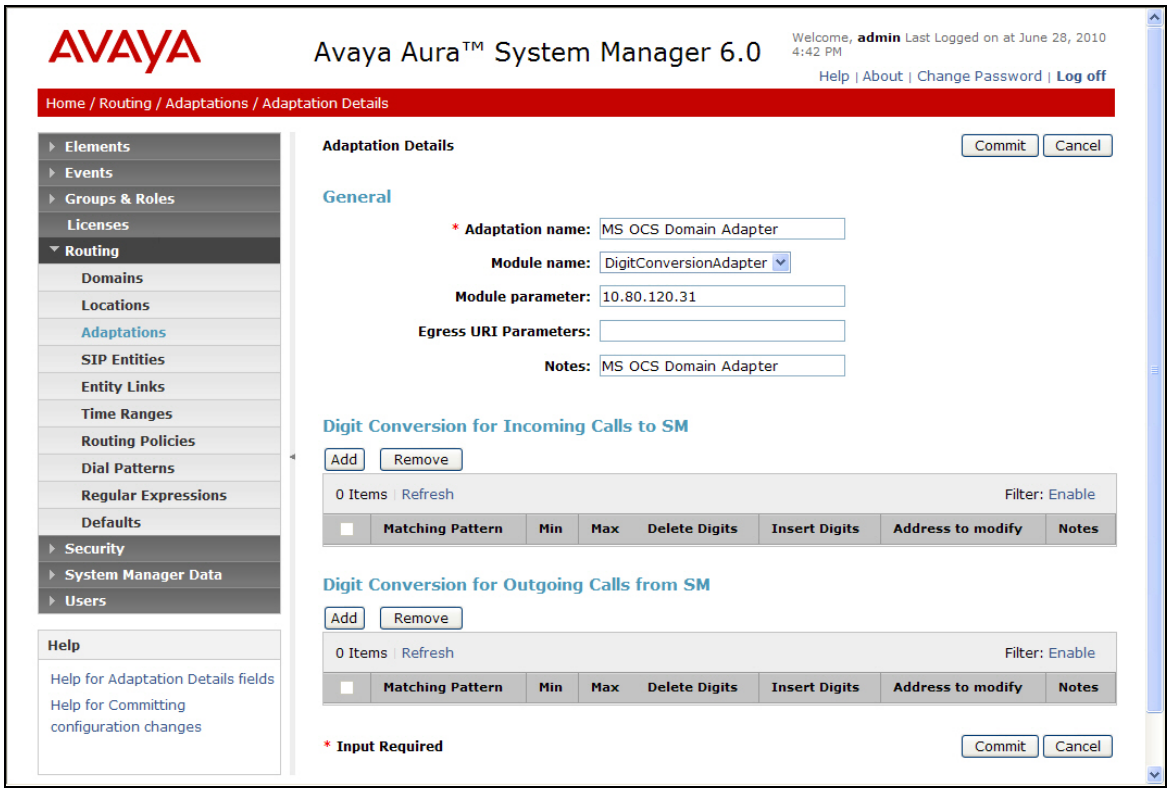
Step	Description
4.	Create a new **Adaptation.** Select **Routing → Adaptations**. Click **New** and enter an appropriate name in the **Adaptation Name** field, select <click to add module> in the **Module Name** field, enter **DigitConversionAdapter** in the **New module name:** field from the **Module name** drop-down menu, enter the IP address of the Microsoft Mediation Server in the **Module parameter** field, and enter optional notes in the **Notes** field. This adaptation is created to change the domain part of the Request-URI in the SIP Invite to an IP address. In addition, the **Module parameter** field can be used to implement source and destination SIP Domain overrides when multiple SIP Domains are implemented in the topology.[9] Although not required in the reference configuration, called party number digit manipulation (deletion and insertion of digits) can be performed using the section **Digit Conversion for Outgoing Calls from SM.** Click **Commit**. 

[9] Ingress Domain Modification Parameters such as ingressOverrideDestinationDomain could be used to replace the domain in the Request-URI with the given value for ingress only and ingressOverrideSourceDomain could be used to replace the domain in the P-Asserted-Identity header and calling part of the History-Info header with the given value for ingress only. See **Reference [3]** for more information on SIP Domain overrides.

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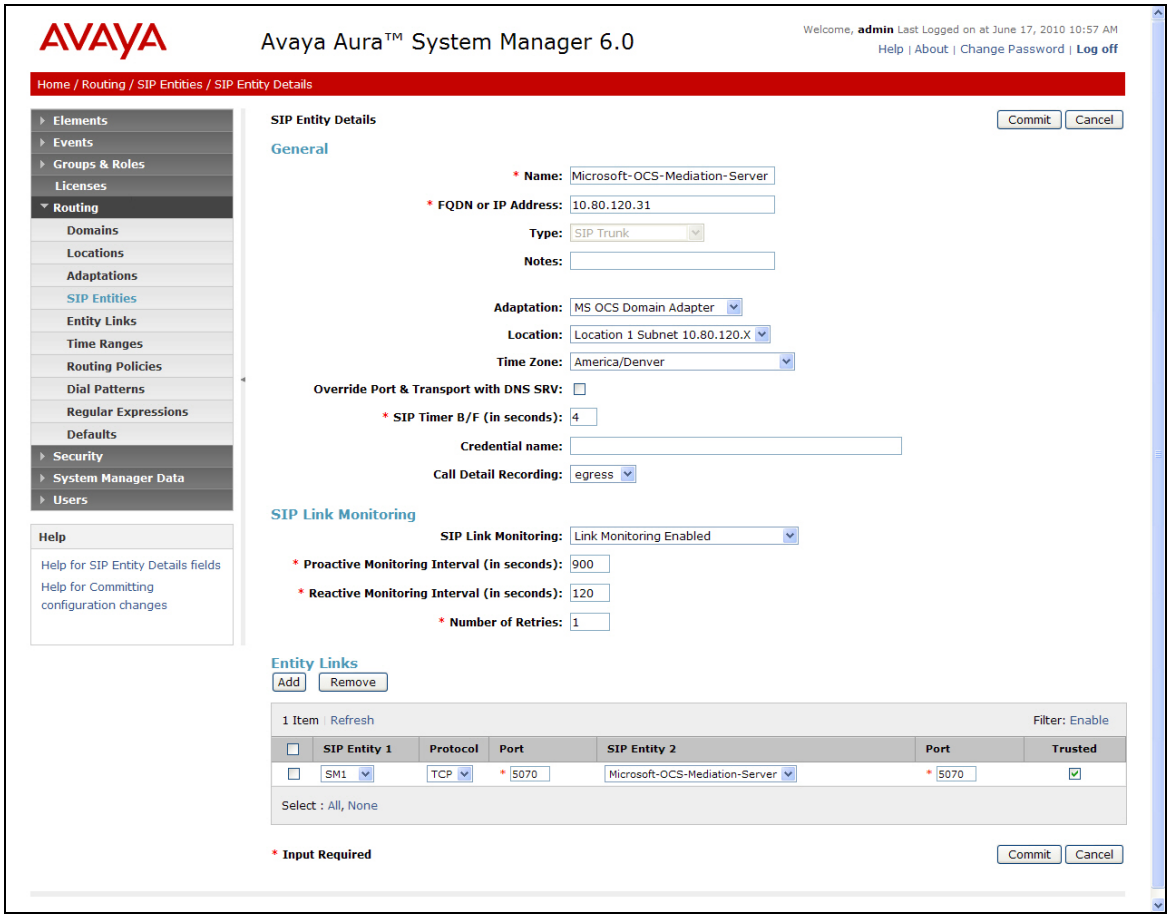
Step	Description
5.	Create a new **SIP Entity** for the Microsoft Mediation Server. Select **Routing → SIP Entities**. Click **New** and enter an appropriate name in the **Name** field, such as **Microsoft-OCS-Mediation-Server**. Enter the FQDN or IP address of the Microsoft Mediation Server. Note that the address entered here should be the address of the local IP interface used to communicate with the Avaya Aura™ Session Manager. Select **SIP Trunk** from the **Type:** drop-down menu. Enter optional notes in the **Notes** field, select **MS OCS Domain Adapter** from the **Adaptation** drop-down menu, select **Location 1 Subnet 10.80.120.X** from the **Location** drop-down menu, and select an appropriate time zone from the **Time Zone:** drop-down menu. Then, under **SIP Link Monitoring**, select **Link Monitoring Enabled** from the **SIP Link Monitoring** drop-down menu.[10] The rest of the values can be left at their default values. Note that when a new **SIP Entity** is created, there will be no **Entity Links** assigned. However, the screen below was captured after performing **Step 6**, hence it displays an associated **Entity Link**. Click **Commit.** 

[10] The Microsoft Mediation Server responds to SIP OPTIONS from Session Manager with a "405 Method Not Allowed". This is sufficient for Session Manager to consider the monitored link entity to be "up".

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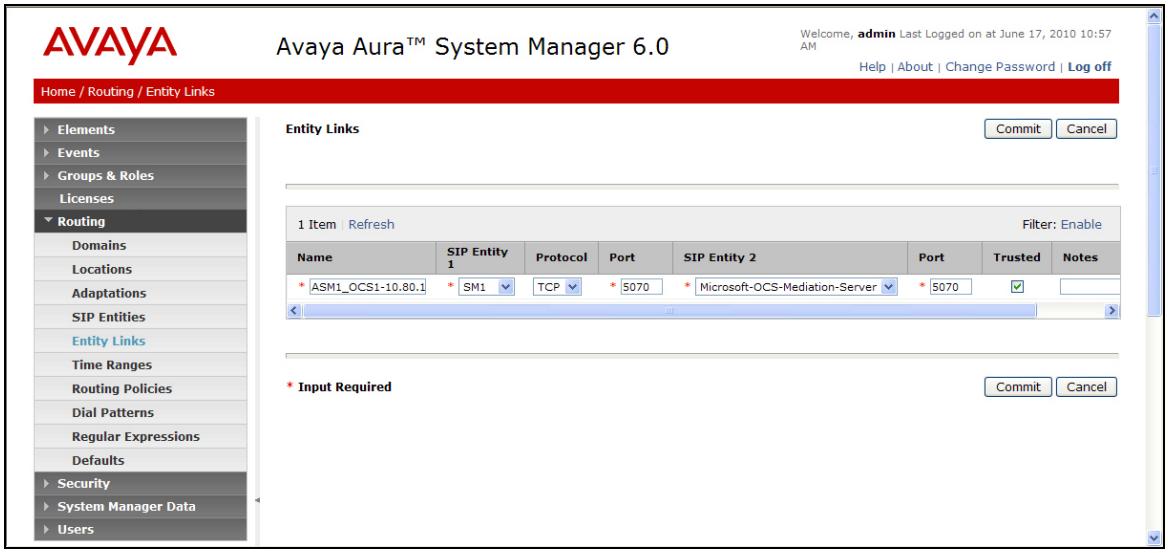
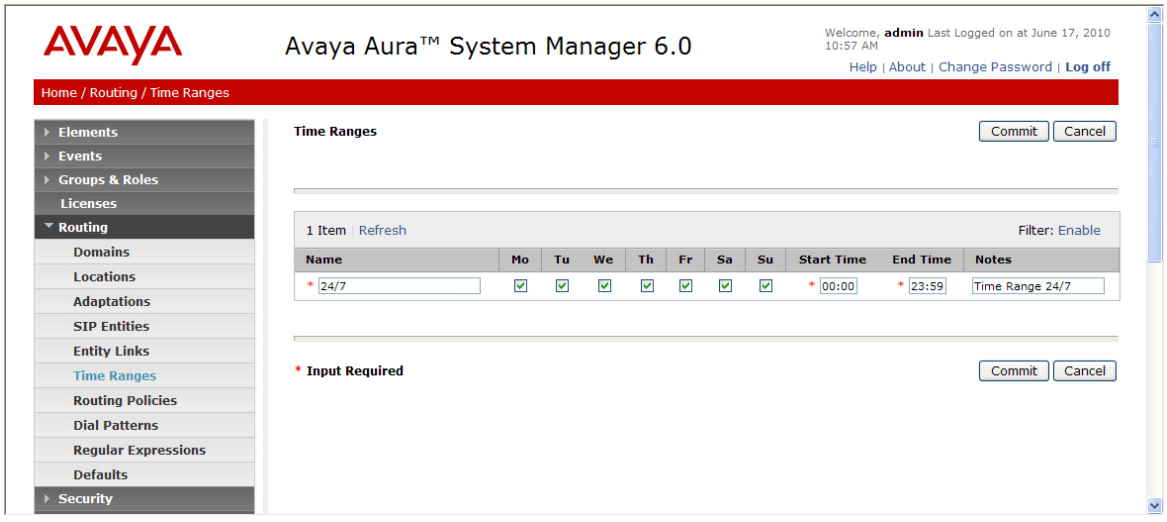
Step	Description
6.	Create a new **Entity Link** between the Session Manager and the Microsoft Mediation Server. Select **Routing → Entity Links**. Click **New** and enter an appropriate name in the **Name** field, such as **ASM1_OCS1-10.80.120.31_5070_TCP.** For **SIP Entity 1,** select the existing Session Manager, and for **SIP Entity 2**, select **Microsoft-OCS-Mediation Server.** For **Protocol** select **TCP**[11]**,** and for both **Port** fields enter "**5070**". Check the **Trusted** checkbox. Click **Commit**. 
7.	Create a new **Time Range**, if needed. Select **Routing → Time Ranges**. Click **New** and enter an appropriate name in the **Name** field, such as **24/7**, check all days of the week and enter "**00:00**" for **Start Time** and "**23:59**" for **End Time**, and enter any optional notes in the **Notes** field. 

[11] The TLS protocol was not tested and is not supported for interoperability in the reference configuration.

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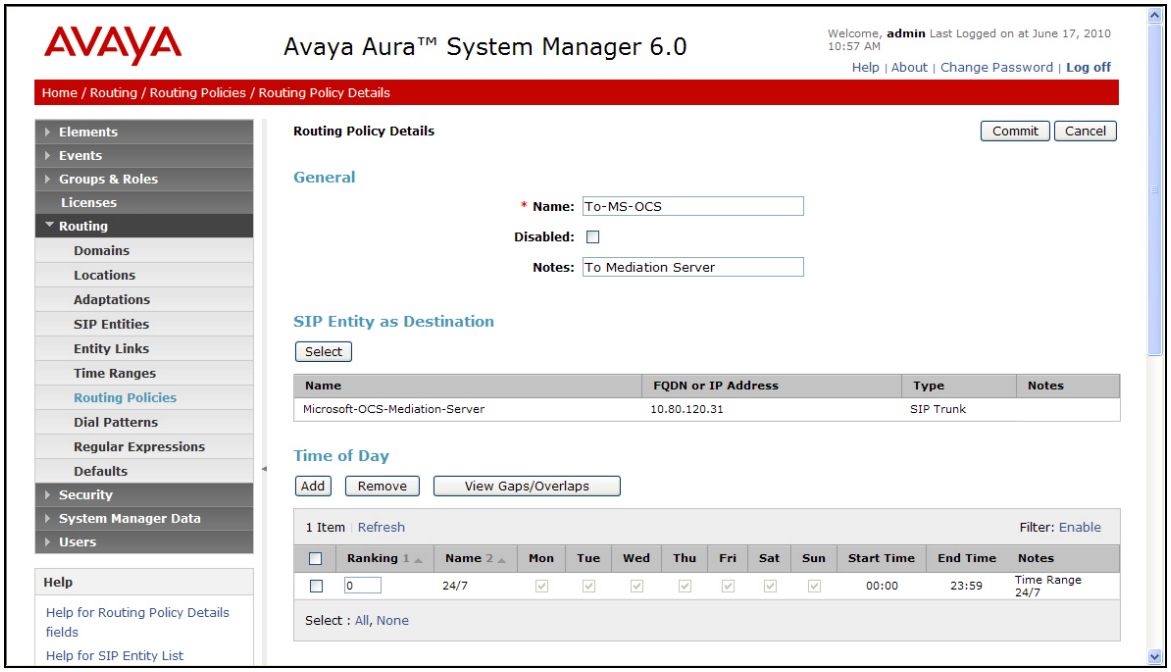
Step	Description
8.	Create a new **Routing Policy** to route calls to Microsoft OCS clients. Select **Routing → Routing Policies**. Click **New** and enter an appropriate name in the **Name** field, such as **To-MS-OCS**, leave the **Disabled:** field unchecked, and enter any optional notes in the **Notes** field. Then, under **SIP Entity as Destination**, click **Select.** Click the radio button next to **Microsoft-OCS-Mediation-Server** (not shown) and click **Select.** Under **Time of Day**, select the **24/7** time range and click **Select** (not shown). Click **Commit.** 

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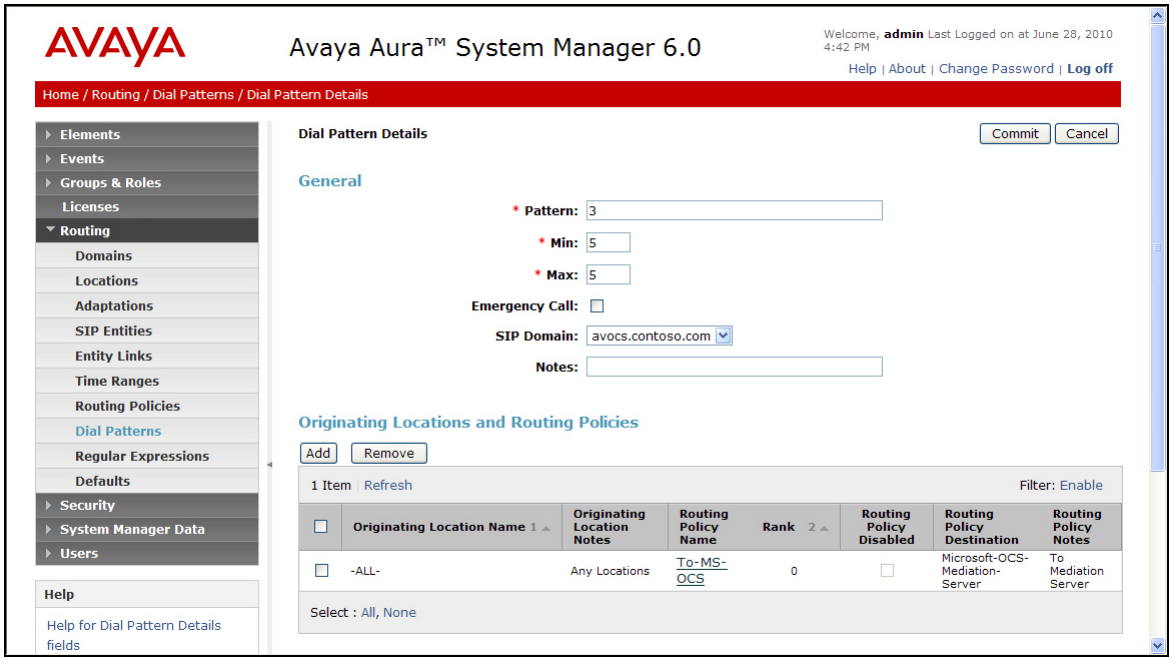
Step	Description
9.	Create a new **Dial Pattern** to route calls to Microsoft OCS clients. Select **Routing → Dial Patterns**. OCS clients were assigned extension numbers in the 3xxxx range. Click **New** and enter **"3**" in the **Pattern:** field, "5" in the **Min** and **Max** fields, select "**avocs.contoso.com**" from the **SIP Domain:** drop-down menu, and enter optional notes in the **Notes** field. Then, under **Originating Locations and Routing Policies**, click **Add**. To route calls to this number from any location, under **Originating Location**, select **Apply The Selected Routing Policies to All Originating Locations** (not shown). Under **Routing Policies**, select **To-MS-OCS** (not shown) and click **Select**. Click **Commit**. 

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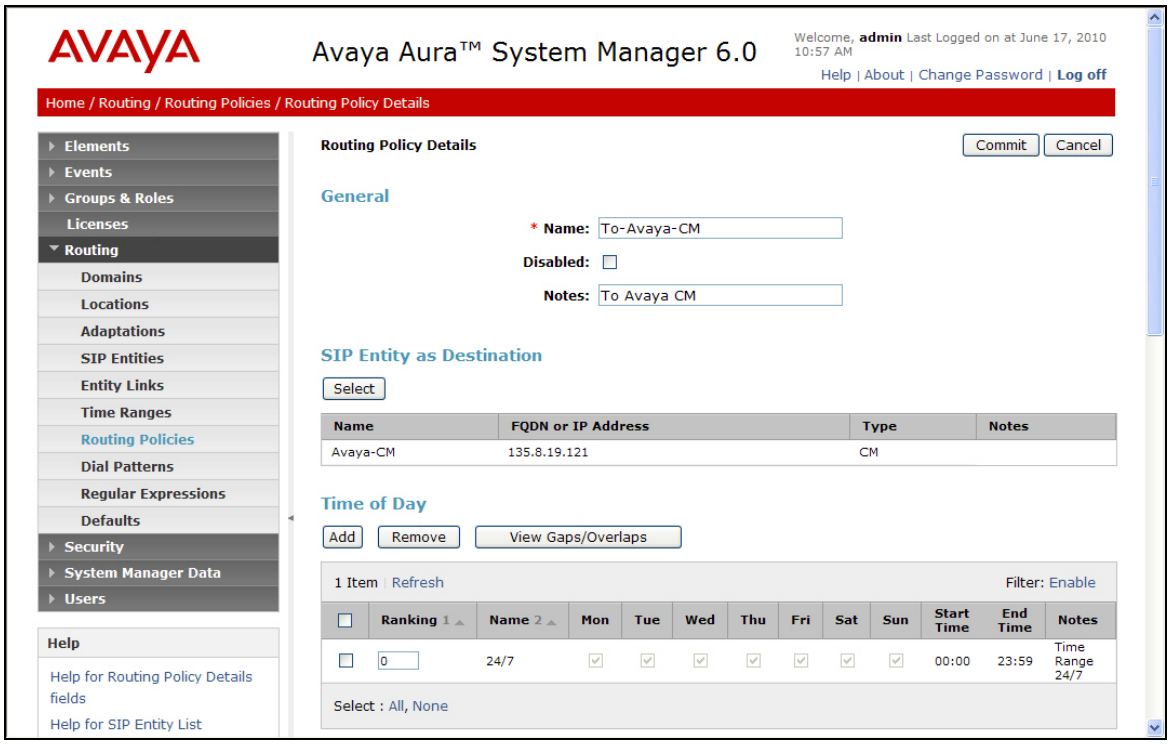
Step	Description
10.	Create a new **Routing Policy** to route calls to Communication Manager. Select **Routing → Routing Policies**. Click **New** and enter an appropriate name in the **Name** field, such as **To-Avaya-CM**, leave the **Disabled:** field unchecked, and enter any optional notes in the **Notes** field. Then, under **SIP Entity as Destination**, click **Select.** In the reference configuration, the existing **SIP Entity** associated with Communication Manager was named **Avaya-CM**. Click the radio button next to **Avaya-CM** (not shown) and click **Select.** Under **Time of Day**, select the **24/7** time range. Click **Commit.**



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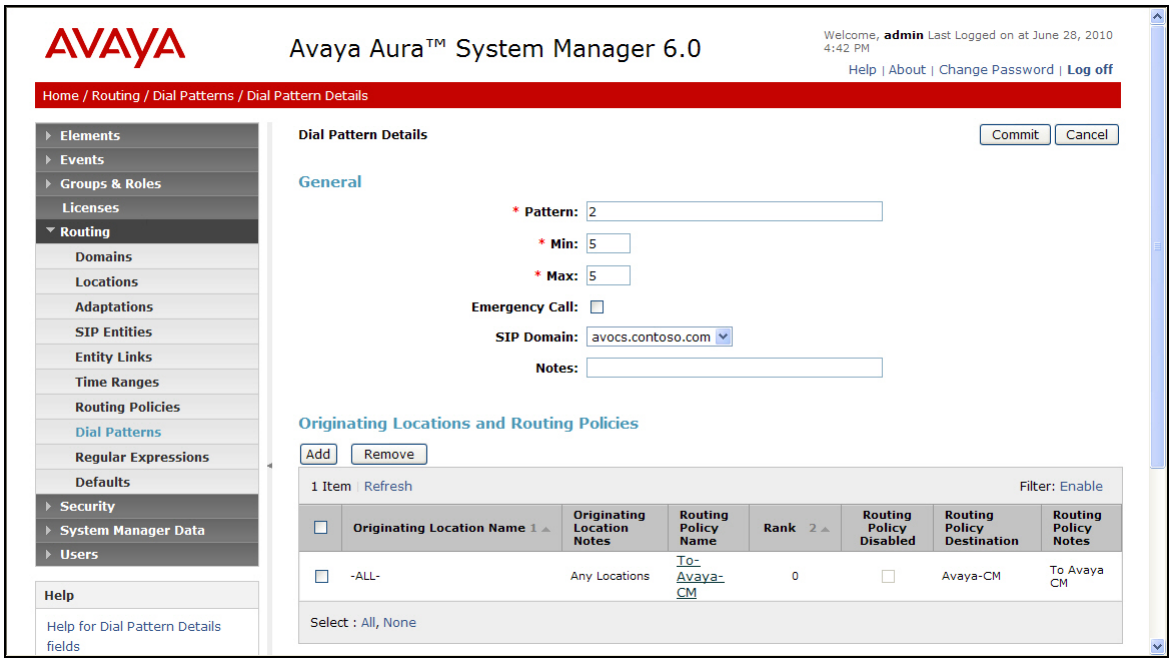
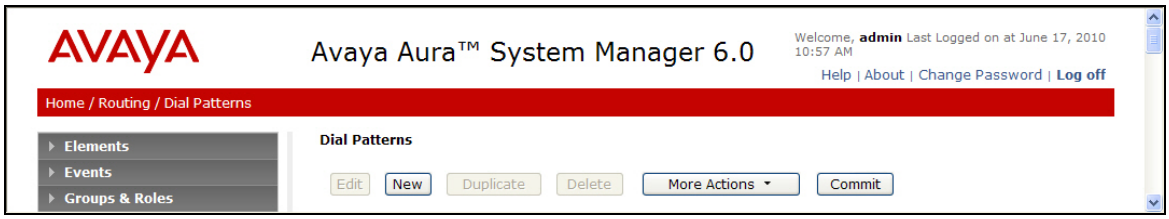
Step	Description
11.	Create a new **Dial Pattern** to route calls to Communication Manager endpoints. Select **Routing → Dial Patterns**. Avaya endpoints were assigned extension numbers in the 2xxxx range. Click **New** and enter **"2**" in the **Pattern:** field, "**5**" in the **Min** field and "**5**" in the **Max** field, select "**avocs.contoso.com**" from the **SIP Domain:** drop-down menu, and enter optional notes in the **Notes** field. Then, under **Originating Locations and Routing Policies**, click **Add**. To route calls to this number from any location, under **Originating Location**, select **Apply The Selected Routing Policies to All Originating Locations** (not shown). Under **Routing Policies** (not shown), select **To-Avaya-CM** and click **Select**. Click **Commit**. 
12.	Click **Commit** to distribute changes to the Session Manager system. 

5. Avaya Aura™ Communication Manager

This section describes the steps for configuring call routing on Communication Manager. The steps are performed from the Communication Manager System Access Terminal (SAT) interface. These Application Notes assume that basic Communication Manager administration has already been performed, and that the necessary integration with Session Manager has already been implemented. See **References [1-11]**.

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5.1. System Parameters

This section reviews the licenses and features that are required for the reference configuration described in these Application Notes. For required licenses that are not enabled in the **system-parameters customer-options** form discussed below, contact an authorized Avaya account representative to obtain the licenses.

Step	Description
1.	Enter the **display system-parameters customer-options** command. On **Page 2** of the **system-parameters customer-options** form, verify that the **Maximum Administered SIP Trunks** number has not been reached.

```
display system-parameters customer-options                    Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 4000  12
          Maximum Concurrently Registered IP Stations: 2400  3
            Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
              Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                       Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 2400  0
                      Maximum Administered SIP Trunks: 4000  60
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
                            Maximum TN2501 VAL Boards: 10    0
                    Maximum Media Gateway VAL Sources: 50    1
          Maximum TN2602 Boards with 80 VoIP Channels: 128   0
         Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

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


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Step	Description
2.	On **Page 5** of the **system-parameters customer-options** form, verify that the **Private Networking** and **Uniform Dialing Plan** features are enabled as shown below.

```
display system-parameters customer-options                      Page   5 of  11
                                OPTIONAL FEATURES

                 Multinational Locations? n           Station and Trunk MSP? y
 Multiple Level Precedence & Preemption? n     Station as Virtual Extension? y
                     Multiple Locations? n
                                                System Management Data Transfer? n
          Personal Station Access (PSA)? y              Tenant Partitioning? y
                       PNC Duplication? n       Terminal Trans. Init. (TTI)? y
                 Port Network Support? n              Time of Day Routing? y
                      Posted Messages? y       TN2501 VAL Maximum Capacity? y
                                                       Uniform Dialing Plan? y
                    Private Networking? y     Usage Allocation Enhancements? y
               Processor and System MSP? y
                     Processor Ethernet? y               Wideband Switching? y
                                                                 Wireless? n
                           Remote Office? y
          Restrict Call Forward Off Net? y
                  Secondary Data Module? y


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


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5.2. Dial Plan

This section describes dial plan settings for the reference configuration described in these Application Notes.

Step	Description
1.	Enter the **change dialplan analysis** command. Add two entries for OCS client extension numbers and Avaya endpoint extension numbers as follows: • **Dialed String** - enter the leading digit(s) of the 5-digit extension numbers assigned to OCS clients. Also, enter the leading digit(s) of the 5-digit extension numbers assigned to Avaya endpoints. Note that appropriate **Dialed String** entries may already exist. • **Total Length** – enter "**5**". • **Call Type** – enter "**ext**". **Note:** In the example below, the **Dialed String** entry "**2**" of **Total Length** "**5**" and of **Call Type** "**ext**" was used to define extension numbers for Avaya telephones. The **Dialed String** entry "**3**" of **Total Length** "**5**" and of **Call Type** "**ext**" was used to define uniform dialing plan numbers for OCS clients.

```
change dialplan analysis                                    Page   1 of  12
                           DIAL PLAN ANALYSIS TABLE
                                 Location: all            Percent Full: 3

    Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
    String   Length Type     String   Length Type     String   Length Type
   1           3   dac
   2           5   ext
   3           5   ext
   30          5   ext
   4           5   ext
   5           5   ext
   8           1   fac
   9           1   fac
   *           3   fac
   #           3   fac
```


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Step	Description
2.	Enter the **change uniform-dialplan 3** command. Add an entry as follows: • **Matching Pattern** - enter the leading digits(s) of the 5-digit numbers assigned to OCS clients. In the example below, any 5-digit dialed numbers that begin with "**3**" will be routed by the Automatic Alternate Routing (AAR) feature. • **Len** – enter "**5**". • **Del** – enter "**0**". • **Net** – enter "**aar**". Complete programming of the AAR feature is out of the scope of these Application Notes and is covered in standard user documentation.

```
change uniform-dialplan 3                                    Page   1 of   2
                      UNIFORM DIAL PLAN TABLE
                                                           Percent Full: 0

  Matching                    Insert              Node
  Pattern       Len Del       Digits     Net Conv Num
 3               5   0                    aar  n
```

5.3. SIP Trunks to Session Manager

The following section describes the configuration of SIP signaling groups and SIP trunk groups for handling inbound and outbound calls. It should be noted that for inbound calls from Microsoft OCS 2007 R2, the host portion of the From header URI is set to the Fully Qualified Domain Name (FQDN) of the Microsoft Mediation Server and not necessarily the Fully Qualified Domain Name of the Session Manager. Due to the domain mismatch, an additional SIP signaling/trunk group must be configured to handle such inbound calls. In the reference configuration, separate signaling groups and separate trunk groups were used for inbound and outbound calls.

5.3.1. Calling and Called Party Numbers

As discussed in **Section 2**, this reference configuration implements 5-digit extension dialing.

Calls from OCS Clients to Avaya telephones contain 5-digit calling party numbers, as defined by the **Line URI:** field associated with the calling OCS user's telephony options (see **Section 6 Step 3**), and 5-digit called party numbers. Communication Manager will present the 5-digit calling party number on the display of Avaya endpoints, if available. Also, Communication Manager will automatically match the 5-digit called party number to extension numbers assigned to local telephones. No further configuration on Communication Manager is required to route these calls to Avaya telephones. However, should there be a need to manipulate the called party number as presented to Communication Manager, the **Incoming Call Handling Treatment** form, available within Communication Manager, can be used to specify call handling. For more information on the **Incoming Call Handling Treatment** form, see the **Screen Reference** section of **Reference [8]**.

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Calls from Avaya telephones to OCS clients contain 5-digit calling party numbers, as defined by the **Numbering – Private Format** table on Communication Manager, and 5-digit called party numbers. OCS will present the 5-digit calling party number to OCS clients. Also, OCS will search for an OCS client whose **Line URI:** field (see **Section 6, Step 3**) matches the 5-digit called party number.

Step	Description
1.	Enter the **change private-numbering 2** command. Add an entry as follows: • **Ext Len**- enter "**5**". • **Ext Code** – enter the leading digits(s) of the 5-digit numbers assigned to Avaya endpoints. • **Total Len** – enter "**5**". In the example below, for any 5-digit calling party number that begins with "**2**", Communication Manager will generate a 5-digit calling party number (CPN) and transmit this CPN with the call.
	change private-numbering 2 Page 1 of 2 NUMBERING - PRIVATE FORMAT **Ext Ext** Trk Private **Total** **Len Code** Grp(s) Prefix **Len** **5 2** **5** Total Administered: 2

5.3.2. Calls from Microsoft OCS Clients to Avaya Endpoints

This section describes the configuration steps required on Communication Manager for routing calls originated by OCS clients to Avaya endpoints. Note that these calls arrive at Communication Manager from Session Manager as incoming SIP calls.

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1. Enter the **add signaling-group s** command, where **s** is an unused signaling group number. Set the following values as shown below:

- IMS Enabled? - **"n"**
- Group Type: - **"sip"**
- Transport Method: - **"tcp"**
- Peer Detection Enabled? - **"y"**
- Initial IP-IP Direct Media? - **"n"**

In the reference configuration, the **Far-end Network Region** was set to **2**. Leave the **Far-end Domain:** field empty to allow this SIP signaling group to handle inbound calls from any domain. If stricter control of the domains for which Communication Manager can handle inbound calls is desired, then the **Far-End Domain:** field may be set to the FQDN of the Microsoft Mediation Server. In the reference configuration, the FQDN of the Microsoft Mediation Server was **contosomed1.avocs.contoso.com** (see **Section 6 Step 13**). However, note that if there are multiple Microsoft Mediation Servers, then multiple corresponding SIP signaling/trunks groups would need to be configured. Check with your system administrator to identify the pre-existing **Far-end Node Name** for the Session Manager. The **Far-end Node Name:** should correspond to the Session Manager's SIP routing network interface. In addition, the **Far-end Listen Port** should correspond to the TCP/IP port number on which the Session Manager is listening for SIP messages from the pre-existing Communication Manager SIP domain. Set the **RFC 3389 Comfort Noise** field to "**y**" so that Comfort Noise support is signaled and Comfort Noise packets are generated in the RTP stream as needed.[12] To enable audio shuffling support, set the **Direct IP-IP Audio Connections** field to "**y**". Set the **Initial IP-IP Direct Media** field to "**n**".

```
add signaling-group 2                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 2              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
       Q-SIP? n                                          SIP Enabled LSP? n
    IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM



   Near-end Node Name: procr                 Far-end Node Name: SM1
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                       Far-end Network Region: 2

Far-end Domain:
                                            Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                 RFC 3389 Comfort Noise? y
         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                    IP Audio Hairpinning? n
         Enable Layer 3 Test? n                    Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n             Alternate Route Timer(sec): 6
```

[12] Note that when the **RFC 3389 Comfort Noise?** field is enabled on the signaling group form, silence suppression is implemented and the **Silence Suppression** field on **the ip-codec-set** form used for the call is ignored by Communication Manager.

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2. Enter the **add trunk-group t** command, where **t** is an unused trunk group number. On **Page 1** of the trunk group form, set the values as shown below. In the reference configuration, the **Direction:** field was set to **"incoming**". Set the **Service Type:** field to "**tie**". The **Signaling Group:** field should be set to the signaling group number configured in **Step 1**.

```
add trunk-group 2                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 2                     Group Type: sip           CDR Reports: y
  Group Name: From OCS via SM1            COR: 1        TN: 1        TAC: 102
   Direction: incoming       Outgoing Display? n
 Dial Access? n                                    Night Service:

Service Type: tie                   Auth Code? n
                                            Member Assignment Method: auto
                                                      Signaling Group: 2
                                                    Number of Members: 50
```

While not required, on **Page 4**, the **Telephone Event Payload Type** field can be set to "**101**", if desired. The **Telephone Event Payload Type:** field can also be left blank, in which case, the Telephone Event Payload Type will be negotiated automatically during SIP call establishment.

```
change trunk-group 2                                           Page   4 of  21
                             PROTOCOL VARIATIONS

                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                  Network Call Redirection? n
                     Send Diversion Header? n
                   Support Request History? n
              Telephone Event Payload Type: 101


       Convert 180 to 183 for Early Media? n
 Always Use re-INVITE for Display Updates? n
                             Enable Q-SIP? n
```


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3. In the reference configuration, the **Far-End Network Region:** for the signaling group in **Step 1** above was set to a value of **2**. However, the final call destinations were Avaya endpoints assigned to hardware in IP Network Region 1. In this scenario, Communication Manager examines the **Authoritative Domain:** field of the IP Network Region associated with the final destination of the call. If the SIP Invite request-URI domain matches the **Authoritative Domain:** specified in the ip-network-region form, Communication Manager will accept the call. Therefore, in this scenario, IP Network Region 1 must contain the domain used by Microsoft OCS (**avocs.contoso.com**) in the **Authoritative Domain:** field as shown below. Otherwise, Communication Manager will attempt to route the call based on the route pattern value of the **Proxy Select Route Pattern** field of the **Locations** form (not shown). See **Section 6 Steps 15** and **16** to identify the OCS SIP domain.

```
change ip-network-region 1                                    Page   1 of  20
                              IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avocs.contoso.com
    Name: Location 1
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                     IP Audio Hairpinning? n
   UDP Port Max: 3029
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                       RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```


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4. An appropriate IP Codec Set should be defined for calls that traverse IP Network Regions. Enter the **change ip-codec-set x** command, where x is an available IP Codec Set number. Verify that **G.711MU** is an available codec set as shown below. IP codec sets supported for the base integration are G.711MU and G.711A.

```
change ip-codec-set 1                                          Page   1 of   2

                          IP Codec Set

   Codec Set: 1

   Audio         Silence      Frames   Packet
   Codec         Suppression  Per Pkt  Size(ms)
 1: G.711MU           n          2        20
 2:
 3:
 4:
 5:
 6:
 7:
```

5. When calls traverse IP Network Regions, as was the case in the reference configuration, the IP Network Regions must also be inter-connected. Enter the **change ip-network-region 2** command and enter the IP codec set number configured in **Step 4** above in the **codec set** field corresponding to destination region 1 as shown below.

```
change ip-network-region 2                                     Page   4 of  20

 Source Region: 2     Inter Network Region Connection Management     I       M
                                                                    G  A    t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    e
 1    1     y    NoLimit                                          n       t
```

5.3.3. Calls from Avaya Endpoints to Microsoft OCS Clients

This section describes the configuration steps on Communication Manager for routing calls from Avaya endpoints to OCS clients. Note that Communication Manager delivers these calls as outbound calls to Session Manager.

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Step	Description
1.	Enter the **add signaling-group s** command, where **s** is an unused signaling group number. Set the following values as shown below: a. IMS Enabled? n b. Group Type: sip c. Transport Method: tcp d. Peer Detection Enabled? y e. Initial IP-IP Direct Media? n In the reference configuration, the **Far-end Domain:** field was populated with the domain used by Microsoft OCS as shown below, and the **Far-end Network Region** was set to **2**. See **Section 6 Steps 15** and **16** to identify the domain used by Microsoft OCS. Set the **RFC 3389 Comfort Noise** field to "**y**" so that Comfort Noise support is signaled and Comfort Noise packets are generated in the RTP stream as needed.[13] To enable audio shuffling support, set the **Direct IP-IP Audio Connections** field to "**y**". The **Far-end Node Name:** should correspond to the Session Manager's SIP routing network interface. In addition, the **Far-end Listen Port** should correspond to the TCP/IP port number on which the Session Manager is listening for SIP messages from the pre-existing Communication Manager SIP domain.

```
add signaling-group 3                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 3              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                          SIP Enabled LSP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM




   Near-end Node Name: procr                 Far-end Node Name: SM1
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                     Far-end Network Region: 2

Far-end Domain: avocs.contoso.com
                                              Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                   RFC 3389 Comfort Noise? y
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                     Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

[13] Note that when the **RFC 3389 Comfort Noise?** field is enabled on the signaling group form, silence suppression is implemented and the **Silence Suppression** field on **the ip-codec-set** form used for the call is ignored by Communication Manager.

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Step	Description
2.	Enter the **add trunk-group t** command, where **t** is an unused trunk group number. In the reference configuration, the **Direction:** field was set to **"outgoing**". Set the **Service Type:** field to "**tie**". Set the **Signaling Group: field** to the signaling group number configured in **Step 1**.
	add trunk-group 3 **Page 1** of 21 TRUNK GROUP Group Number: 3 Group Type: sip CDR Reports: y Group Name: To OCS via SM1 COR: 1 TN: 1 TAC: 103 **Direction: outgoing** Outgoing Display? n Dial Access? n Queue Length: 0 **Service Type: tie** Member Assignment Method: auto **Signaling Group: 3** Number of Members: 10
	On **Page 3**, set the **Numbering Format:** field to "**private**".
	add trunk-group 3 **Page 3** of 21 TRUNK FEATURES ACA Assignment? n Measured: none Maintenance Tests? y **Numbering Format: private** UUI Treatment: service-provider Replace Restricted Numbers? n Replace Unavailable Numbers? n Modify Tandem Calling Number: no Show ANSWERED BY on Display? y
	While not required, on **Page 4**, the set **Telephone Event Payload Type** field can be set to "**101**", if desired. The **Telephone Event Payload Type** field can also be left blank, in which case, the Telephone Event Payload Type will be automatically negotiated during SIP call establishment. Note that in the reference configuration, the **Prepend '+' to Calling Number?** field was set to "**n**" because no '+' sign was expected by the normalization rule defined on OCS.
	add trunk-group 3 **Page 4** of 21 PROTOCOL VARIATIONS Mark Users as Phone? n **Prepend '+' to Calling Number? n** Send Transferring Party Information? n Network Call Redirection? n Send Diversion Header? n Support Request History? n **Telephone Event Payload Type: 101** Convert 180 to 183 for Early Media? n Always Use re-INVITE for Display Updates? n Enable Q-SIP? n
VV; R SPOC	

Step	Description
3.	Enter the **change aar analysis 3** command. On **Page 2**, add an entry as follows: • **Dialed String** - enter the leading digits(s) of the 5-digit numbers assigned to OCS clients. In the example below, any 5-digit dialed numbers that begin with "**3**" will be routed by the Automatic Alternate Routing (AAR) feature to the Microsoft Mediation Server via route pattern 3. • **Total Min** – enter "**5**". • **Total Max** – enter "**5"**. • **Route Pattern** – enter "**3**". • **Call Type** – enter "**aar**". Complete programming of the AAR feature is out of the scope of these Application Notes and is covered in standard user documentation.

```
change aar analysis 3                                          Page   2 of   2
                            AAR DIGIT ANALYSIS TABLE
                                  Location:  all           Percent Full:    3

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    3                     5    5     3        aar          n
```

4.	Enter the **change route-pattern r** command, where **r** is the number of the route pattern specified in **Step 3** for calls to OCS clients. Add a routing preference entry as follows: • **Grp No** – enter the number of the trunk group defined in **Step 2**. • **FRL** – assign a Facility Restriction Level to this routing preference. "**0**" is the least restrictive. • **Numbering Format** – enter **lev0-pvt**. This entry is required in order to send the calling party number as defined in **Section 5.3.1**.

```
change route-pattern 3                                         Page   1 of   3
                    Pattern Number: 3   Pattern Name: UDP-to-OCS
                                SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                                  Dgts         x                        Intw
 1: 3    0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                   Dgts Format
                                                           Subaddress
 1: y y y y y n  n           rest                                   lev0-pvt  none
 2: y y y y y n  n           rest                                             none
 3: y y y y y n  n           rest                                             none
 4: y y y y y n  n           rest                                             none
 5: y y y y y n  n           rest                                             none
 6: y y y y y n  n           rest                                             none
```


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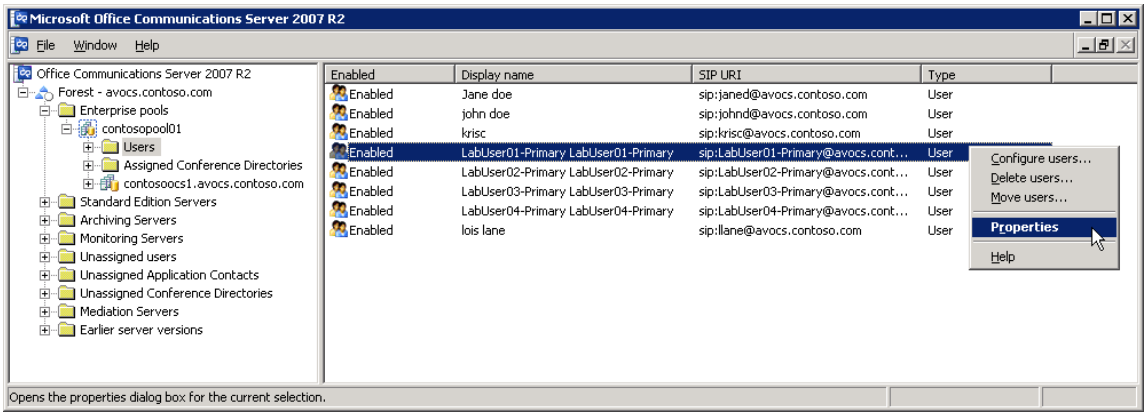
6. Microsoft OCS 2007 R2

This section highlights the Microsoft Office Communications Server (OCS) configuration for routing calls to and from Communication Manager via Session Manager. 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 [14-19]**. These Application Notes further assume that user accounts have been created in Microsoft Active Directory and enabled for OCS.

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

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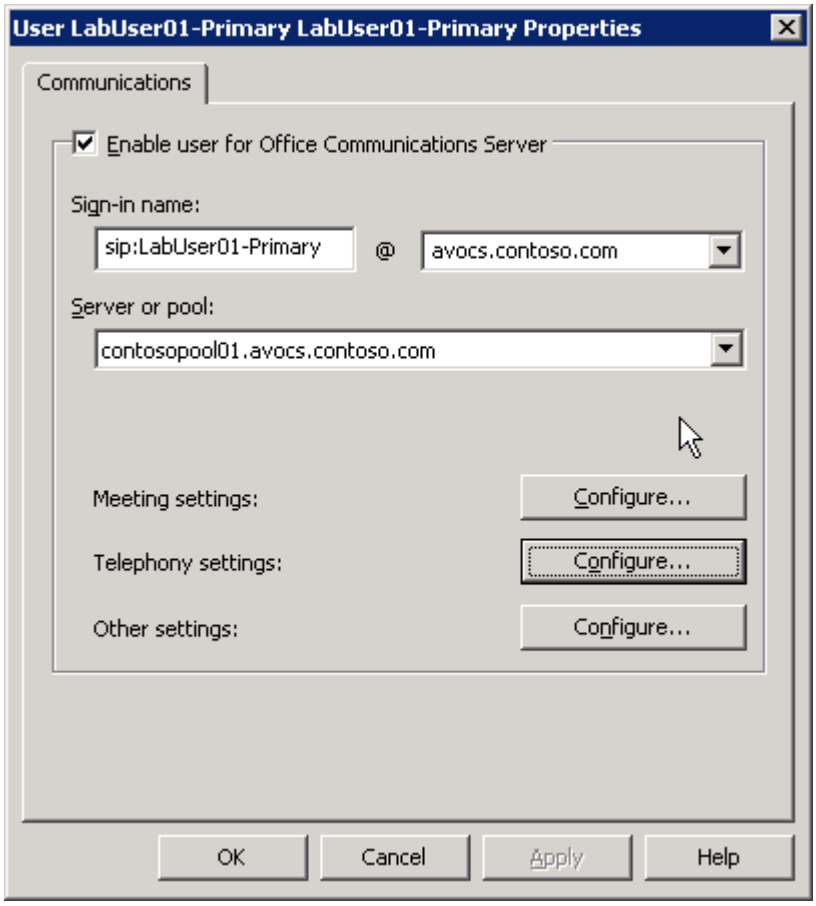
Step	Description
2.	In the selected user's **Properties** dialog box, next to **Telephony settings:** click on "**Configure**". 

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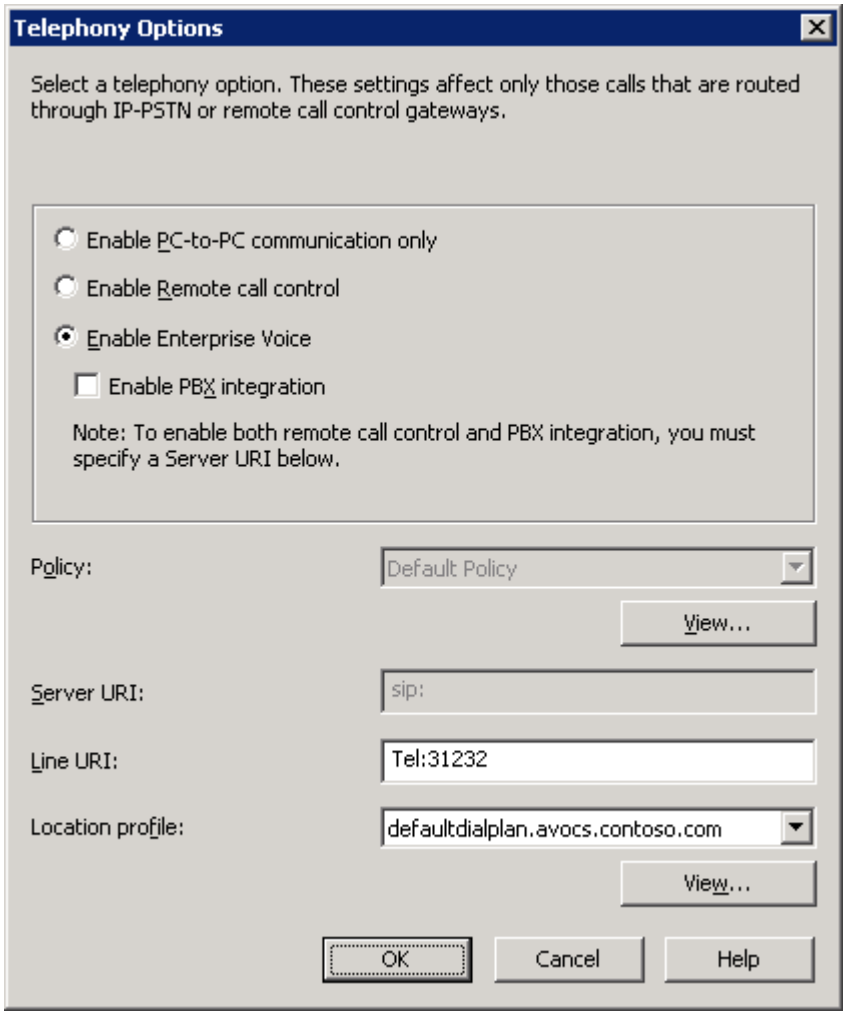
Step	Description
3.	In the **Telephony Options** dialog box, select "**Enable Enterprise Voice**" and enter a 5-digit extension in the **Line URI** field. Verify that a location profile is also selected. The OCS system automatically generates a default location profile (e.g. defaultdialplan.avocs.contoso.com). As shown below, for this user, this default location profile was selected from the drop-down menu in the **Location profile** field. In the reference configuration, users were configured with Line URIs in the form of "Tel:31232", where "31232" is the 5-digit extension number assigned to the user. Click on "**OK**". 
4.	Back in the selected user's **Properties** dialog box, click on "**OK**".
5.	Repeat **Steps 1 – 4** for other Microsoft OCS users to be configured as Enterprise Voice (EV) users.

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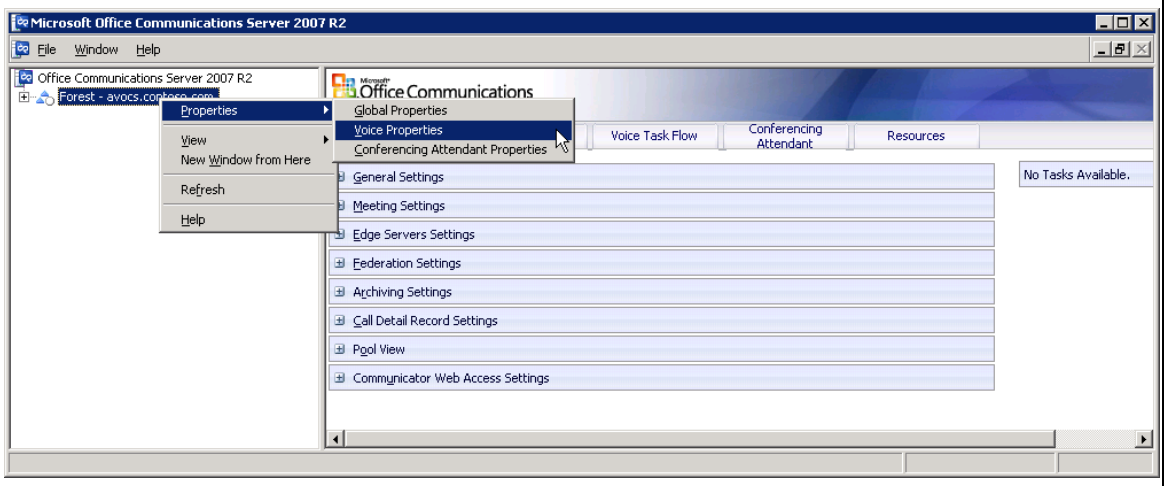
Step	Description
6.	In the left pane of the **Microsoft Office Communications Server 2007 R2** MMC snap-in, right-click on the **Forest** node and select **Properties → Voice Properties**. 

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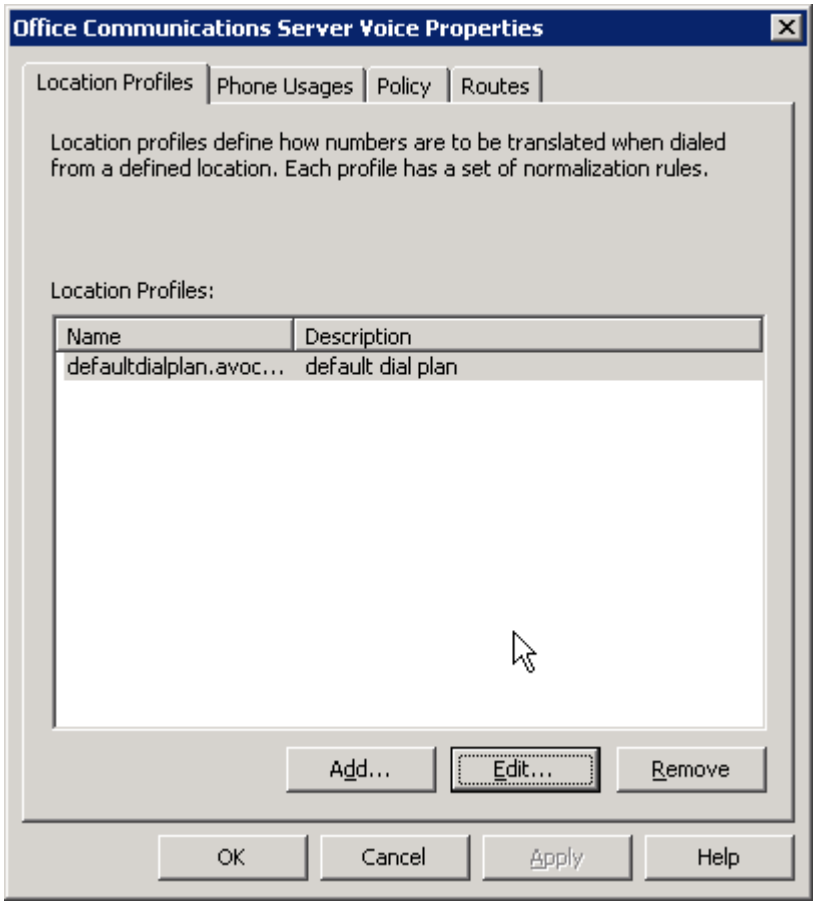
Step	Description
7.	In the **Office Communications Server Voice Properties** dialog box, edit the location profile. This should be the location profile assigned to the user in **Step 3** above. Note that in the reference configuration, the default location profile **defaultdialplan.avocs.contoso.com** was used. Click on "**Edit".** 

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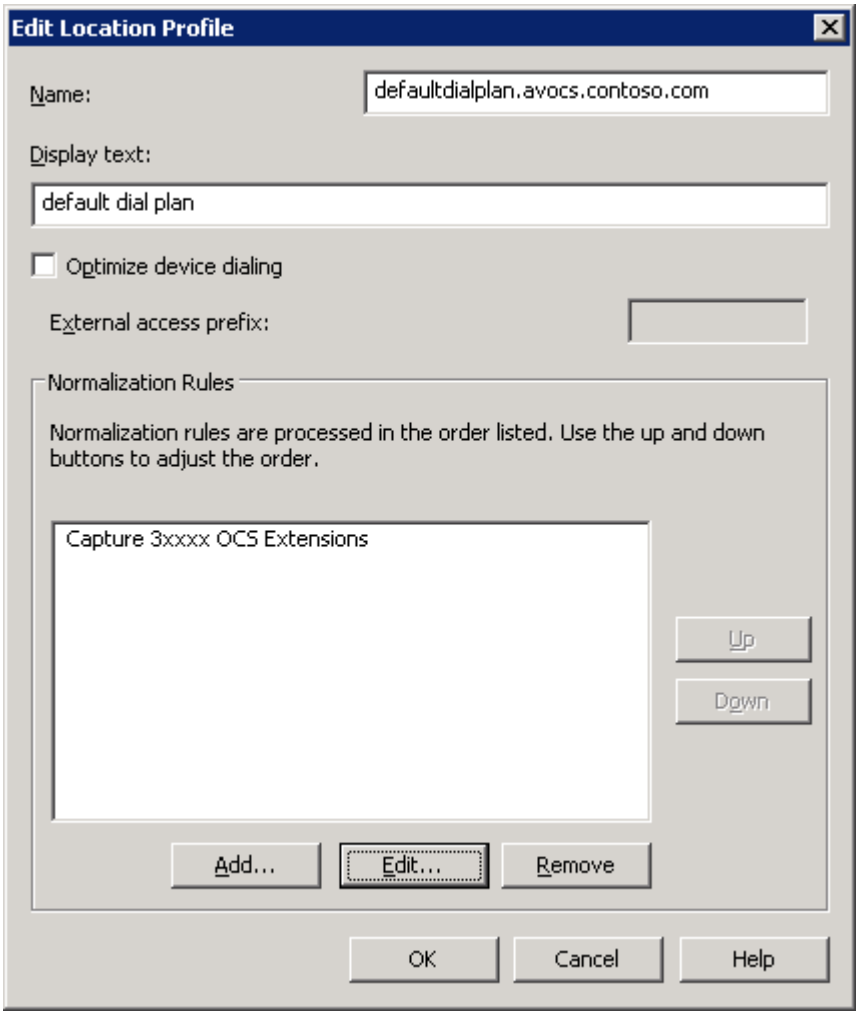
Step	Description
8.	Microsoft OCS location profiles define how OCS entities, such as OCS servers, enterprise pools and mediation servers, interpret and modify phone numbers. Each location profile contains an ordered set of normalization rules that translates phone numbers expressed in various formats. In the reference configuration, normalization rules are used to match dialed numbers. Verify that a normalization rule exists for 5-digit numbers associated with Enterprise Voice users or add one as appropriate. In the **Normalization Rules** section, click on "**Add**" or "**Edit**" as appropriate. 

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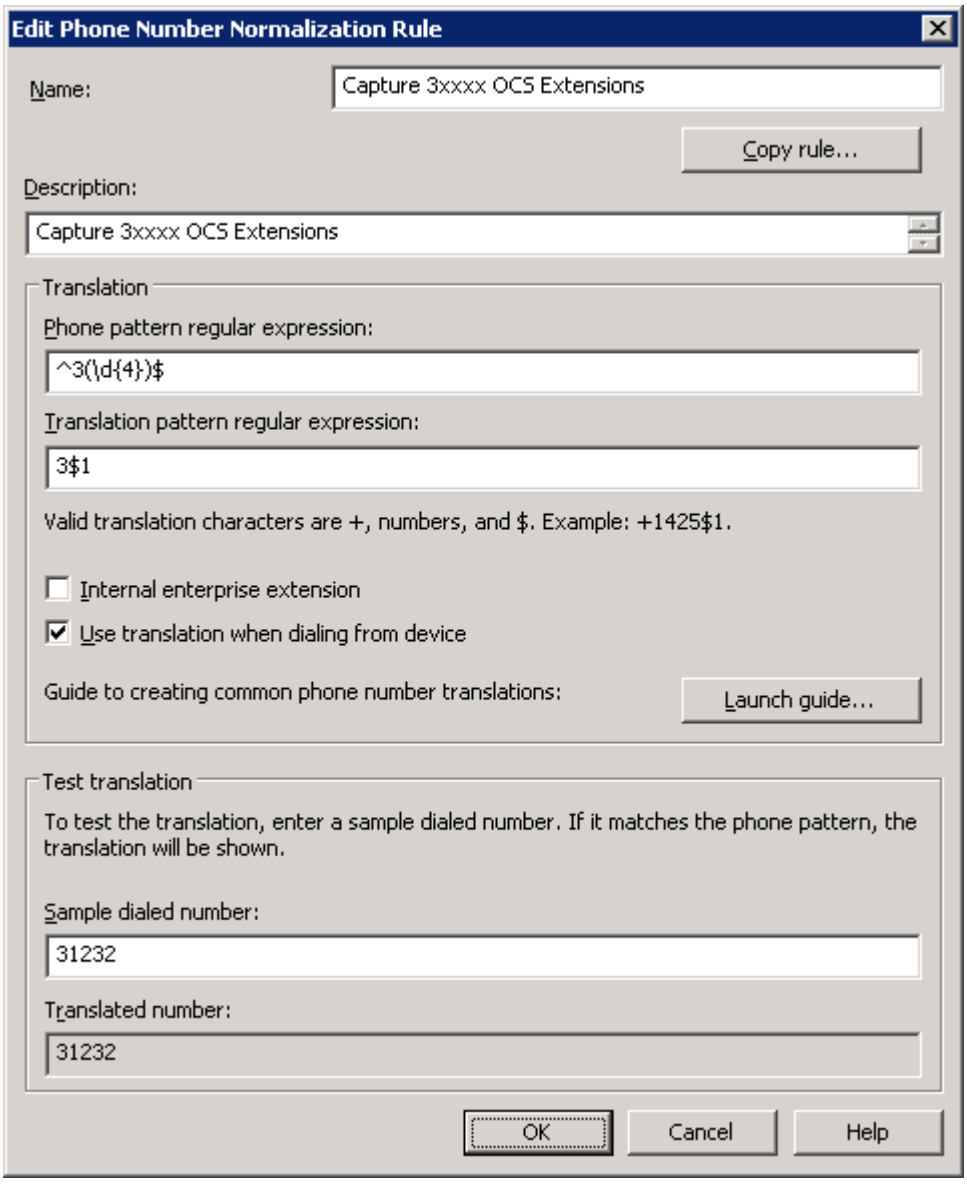
Step	Description
9.	In the **Translation** section of the normalization rule, enter a **Phone pattern regular expression** that matches a 5-digit number that begins with a "3" followed by any four digits, and a **Translation pattern regular expression** as shown below**.** In the example below, the normalization rule matches any 5-digit number that begins with a "3". To confirm that the translation is configured correctly, enter a 5-digit telephone number in the **Sample dialed number** field and confirm that the translation in the **Translated number** field is correct, as shown below. For more information regarding the definition of regular expressions for the **Phone pattern regular expression** field, click on "**Launch guide**" or "**Help**". Other values can be left at their defaults. Click on "**OK**".



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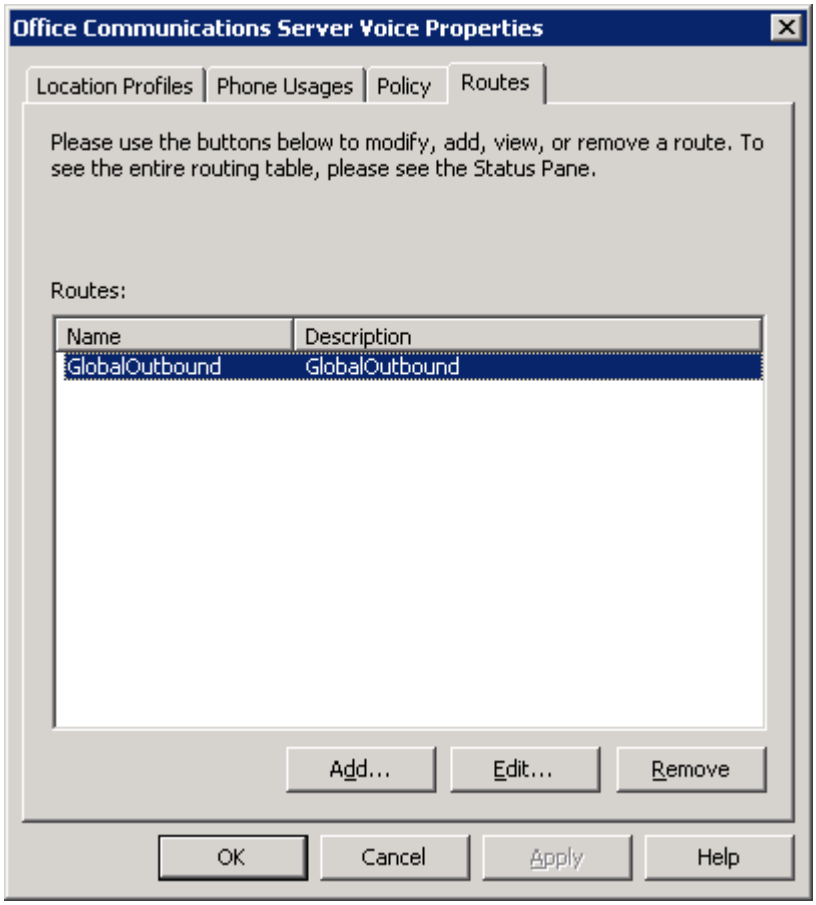
Step	Description
10.	In the **Office Communications Server Voice Properties** dialog box, select the **Routes** tab, and click on "**Add**" or "**Edit**" as appropriate. 

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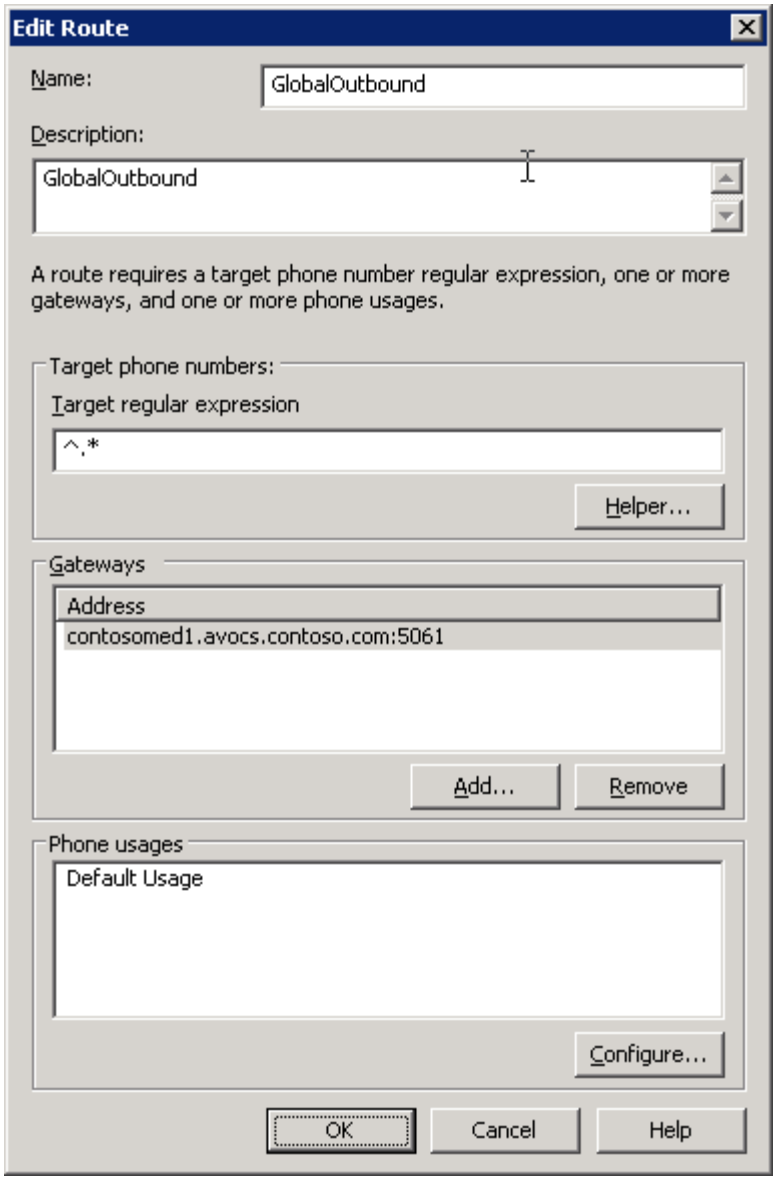
Step	Description
11.	Verify that an outbound route exists as shown below. The **Target regular expression** field under **Target phone numbers**, is typically pre-populated with a default regular expression of "**^.***", which matches all phone numbers. For more information on the **Target regular expression** field, click on "**Helper**". The address listed in the **Gateways** section identifies the FQDN and TCP/IP port of the Microsoft Mediation Server that can handle routing for the target phone numbers.



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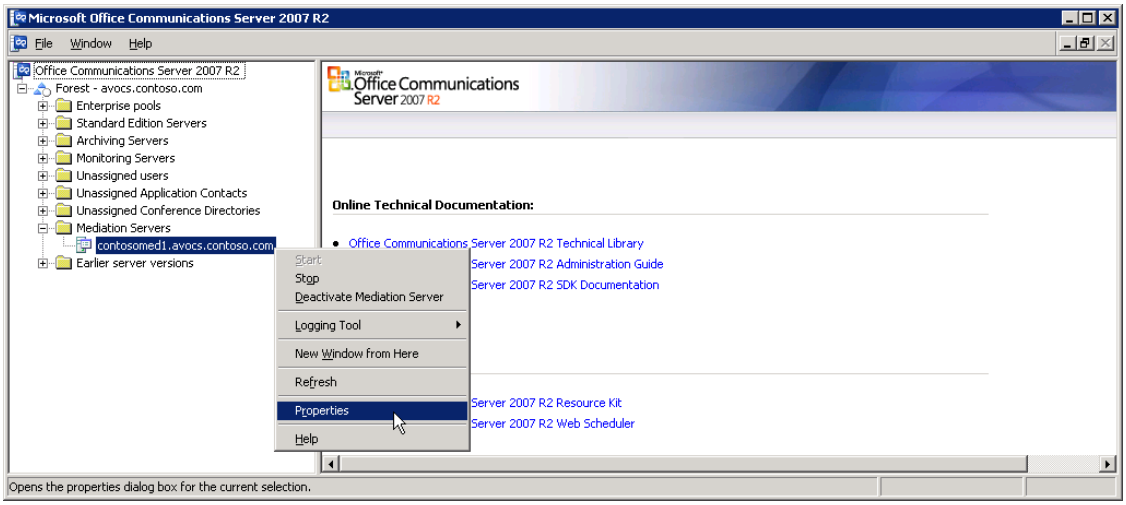
Step	Description
12.	In the left pane of the **Microsoft Office Communications Server 2007 R2** MMC snap-in, expand the **Forest** node down to the Mediation Servers level (**Forest → Mediation Servers**). Right-click on the Mediation Server named as the gateway in **Step 11** and select **Properties**. 

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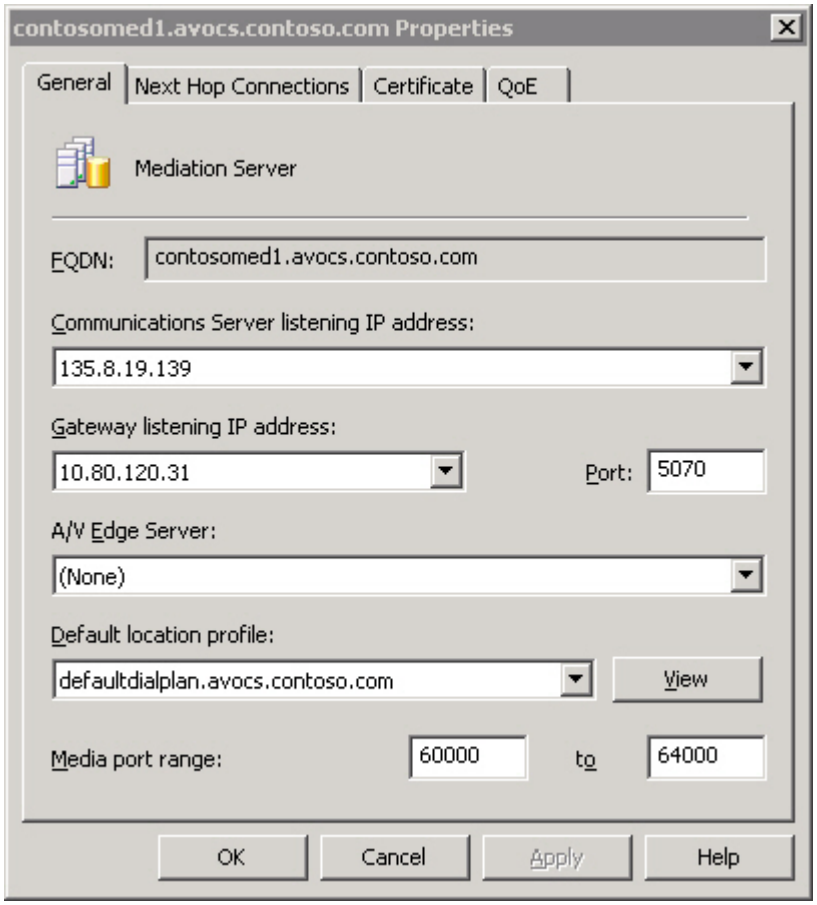
Step	Description
13.	In the Mediation Server **Properties** dialog box, verify the settings as shown below. The **Communications Server listening IP address** should be set to the IP address of the Mediation Server's IP interface that is used to listen for requests from the OCS front end server. The **Gateway listening IP address** should be set to the IP address of the Mediation Server's IP interface that is used to listen for SIP requests from Avaya Aura™ Session Manager. See **Figure 1.** 

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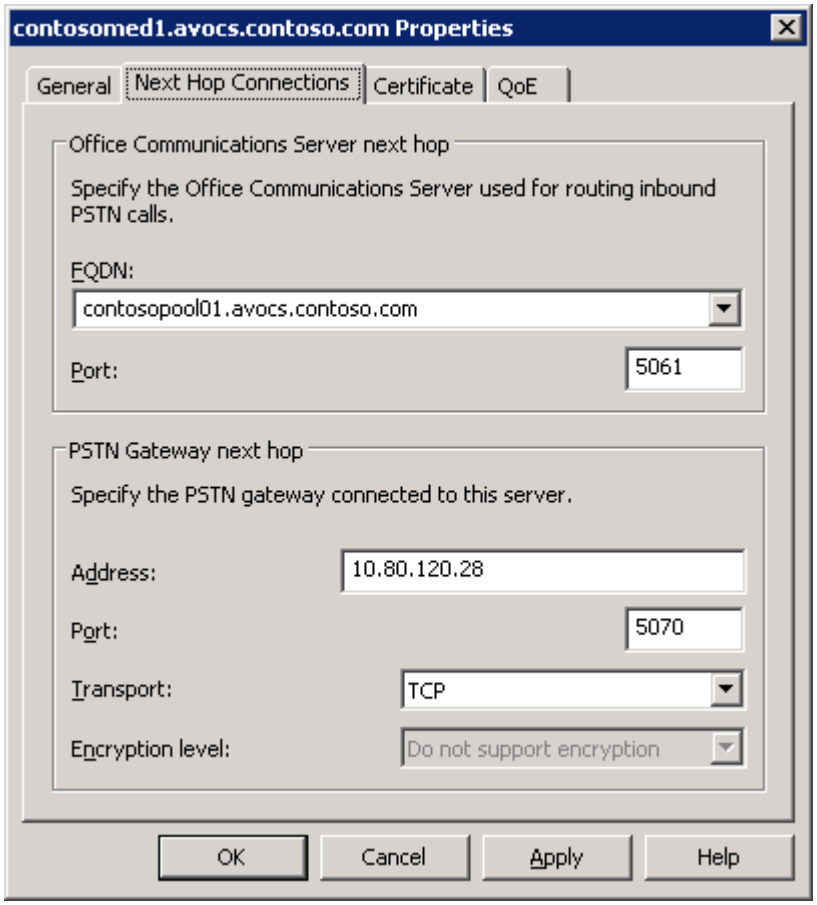
Step	Description
14.	Select the **Next Hop Connections** tab. In the **PSTN Gateway next hop** section, enter the **IP address** of the Avaya Aura™ Session Manager's SIP routing network interface into the **Address** field. Enter "**5070**" in the **Port** field, and select **TCP**[14] in the Transport drop-down menu. See **Section 4.2 Steps 3** and **6**. Note that these values were used in the reference configuration and will vary depending on the implementation. Click on "**OK**". 

[14] The TLS transport was not tested and is not supported for interoperability in the reference configuration.

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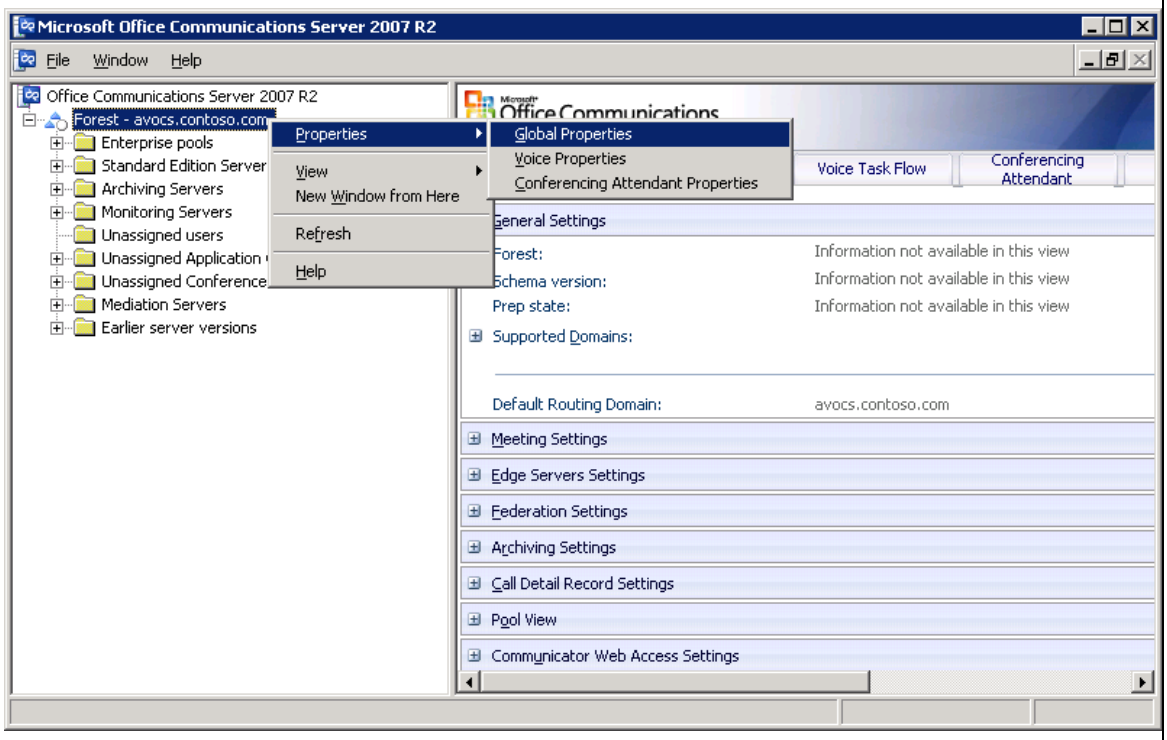
Step	Description
15.	In the left pane of the **Microsoft Office Communications Server 2007 R2** MMC snap-in, right-click on the **Forest** node and select **Properties → Global Properties**. 

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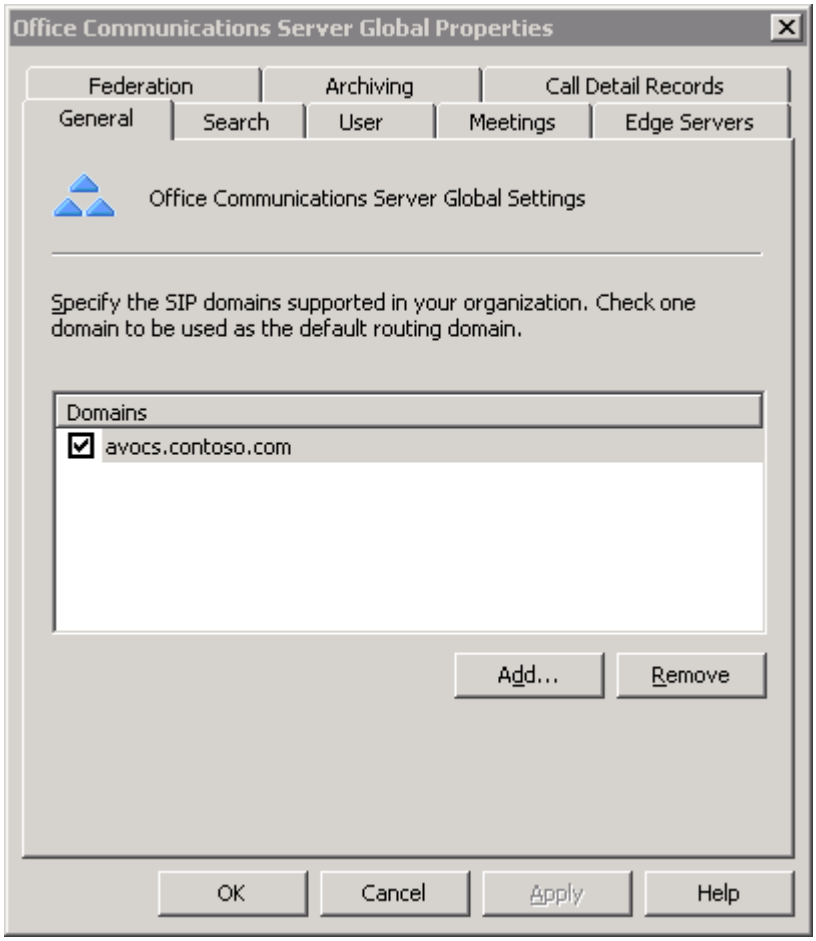
Step	Description
16.	In the Office Communications Server Global Properties dialog box, identify the domain identifier being used by OCS. In the reference configuration, the domain is **avocs.contoso.com** as shown below. This domain will be used when configuring Avaya Aura™ Session Manager. See **Section 4.2 Step 1**. 

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7. Verification Steps

7.1. Verification Tests

The following steps may be used to verify the configuration:

- Place a call from an Avaya telephone to a Microsoft OCS client using the OCS client's 5-digit extension number. Verify that the call is established with two-way audio and that the calling party number on the OCS client is the 5-digit number of the calling Avaya telephone.
- From the OCS client, place a call back to the Avaya telephone by double-clicking on the 5-digit number in the OCS client's Recent Contacts list. Verify that the call is established with two-way audio.

7.2. Troubleshooting Tools

The Communication Manager "list trace tac", and/or "status trunk-group" commands are helpful diagnostic tools to verify correct operation and to troubleshoot problems. MST (Message Sequence Trace) diagnostic traces (performed by Avaya Support) can be helpful in understanding specific interoperability issues.

The logging and reporting functions within the Avaya Aura™ System Manager Common Console may be used to examine the details of Session Manager calls. In addition, if port monitoring is available, a SIP protocol analyzer such as Wireshark (a.k.a. Ethereal) can be used to capture SIP traces at the Session Manager and/or Microsoft Mediation Server interfaces. SIP traces can be instrumental in understanding SIP protocol issues resulting from configuration problems.

8. Conclusion

These Application Notes describe the procedures for configuring Microsoft Office Communications Server (OCS) 2007 R2 with Avaya Aura™ Session Manager 6.0 and Avaya Aura™ Communication Manager 6.0. The topology described herein provides a reference configuration for intra-enterprise voice communications for users of Avaya Aura™ Communication Manager telephones and Microsoft Office Communication Server 2007 R2 clients configured in Enterprise Voice mode as defined by Microsoft.

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9. Additional References

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

[1] "Avaya Aura™ Session Manager Overview", Document Number 03-603323, Issue 3, Release 6.0, June 2010
[2] "Installing and Configuring Avaya Aura™ Session Manager", Document Number 03-603473, Issue 1.0, Release 6.0, June 2010
[3] "Administering Avaya Aura™ Session Manager", Document Number 03-603324, Issue 3, Release 6.0, June 2010
[4] "Maintaining and Troubleshooting Avaya Aura™ Session Manager, Document Number 03-603325, Issue 1.0, Release 6.0, June 2010
[5] "Installing and Configuring Avaya Aura™ System Platform", Release 6.0, June 2010
[6] "Installing and Upgrading Avaya Aura™ System Manager", Release 6.0, June 2010
[7] "Avaya Aura™ Communication Manager Overview", Document Number 03-300468, Issue 7, Release 6.0, June 2010
[8] "Administering Avaya Aura™ Communication Manager", Document Number 03-300509, Issue 6.0, Release 6.0, June 2010
[9] "Avaya Aura™ Communication Manager Feature Description and Implementation", Document Number 555-245-205, Issue 8.0, Release 6.0, June 2010
[10] "Administering Network Connectivity on Avaya Aura™ Communication Manager", Document Number 555-233-504, Issue 14, May 2009
[11] "SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers", Document Number 555-245-206, Issue 9, May 2009"
[12] "Application Notes for Microsoft Office Communicator 2007 R2 Remote Call Control (RCC) Integration with Avaya Communication Manager Telephones", Issue 1.0, June 24, 2009
[13] "Application Notes for Call Routing Between Avaya Communication Manager and Microsoft Office Communications Server", Issue 1.3, September 2009
http://www.avaya.com/master-usa/en-us/resource/assets/applicationnotes/OCS-ACM-CALL.pdf

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

[14] "Microsoft Office Communications Server 2007 R2 Technical Overview"
[15] "Microsoft Office Communications Server 2007 R2 Planning and Architecture"
[16] "Microsoft Office Communications Server 2007 R2 Deploying Enterprise Voice"
[17] "Microsoft Office Communications Server 2007 R2 Planning for Voice"
[18] "Microsoft Office Communications Server 2007 R2 Administering Office Communications Server 2007 R2"
[19] "Integrating Enterprise Telephony with Office Communications Server 2007 R2", March 2009
[20] "Microsoft Office Communications Server 2007 R2 Important Considerations for Enterprise Voice: Please Read"

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