Avaya Solution & Interoperability Test Lab

Sample Configuration for SIP Trunking between Avaya IP Office and Cisco Unified Communications Manager 7.0 – Issue 1.0

Abstract

These Application Notes describe the steps for configuring a SIP trunk between Avaya IP Office and Cisco Unified Communications Manager (CUCM).

1. Introduction

Session Initiation Protocol (SIP) is a standard based communication protocol capable of supporting voice, video, instant messaging and other multi-media communication. These Application Notes will outline a solution for using SIP as a trunking protocol between Avaya IP Office and Cisco Unified Communications Manager.

2. Overview

The sample network shown in **Figure 1** consists of two IP PBX systems each belonging to a different domain with its own dialing plan. The Avaya IP PBX system consists of Avaya IP Office system capable of supporting a variety of Avaya 5600 and 4600 Series IP Telephones along with digital and analog phone/fax stations. The Cisco IP PBX system consists of Cisco Unified Communications Manager (CUCM) supporting Cisco SIP and SCCP stations along with analog Fax station through the use of a Cisco 1751 router/gateway. A SIP trunk is configured between Avaya IP Office and CUCM to support calling between the Avaya and Cisco IP PBX systems. With the use of the SIP trunk trans-coding, media and protocol conversion, calls between any 2 telephones are supported in this sample network regardless of whether they are between SIP, H.323, DCP, SCCP or analog stations.

3. Configuration

Figure 1 illustrates the configuration used in these Application Notes. All telephones in the 172.28.10.0/24 IP network are either registered with Avaya IP Office and use extension 122xx. All IP telephones in the 172.29.5.0/24 IP network are registered with CUCM and use extension 60xxx. A single SIP trunk between Avaya SES and CUCM manages call control between the Avaya and Cisco IP PBX systems.

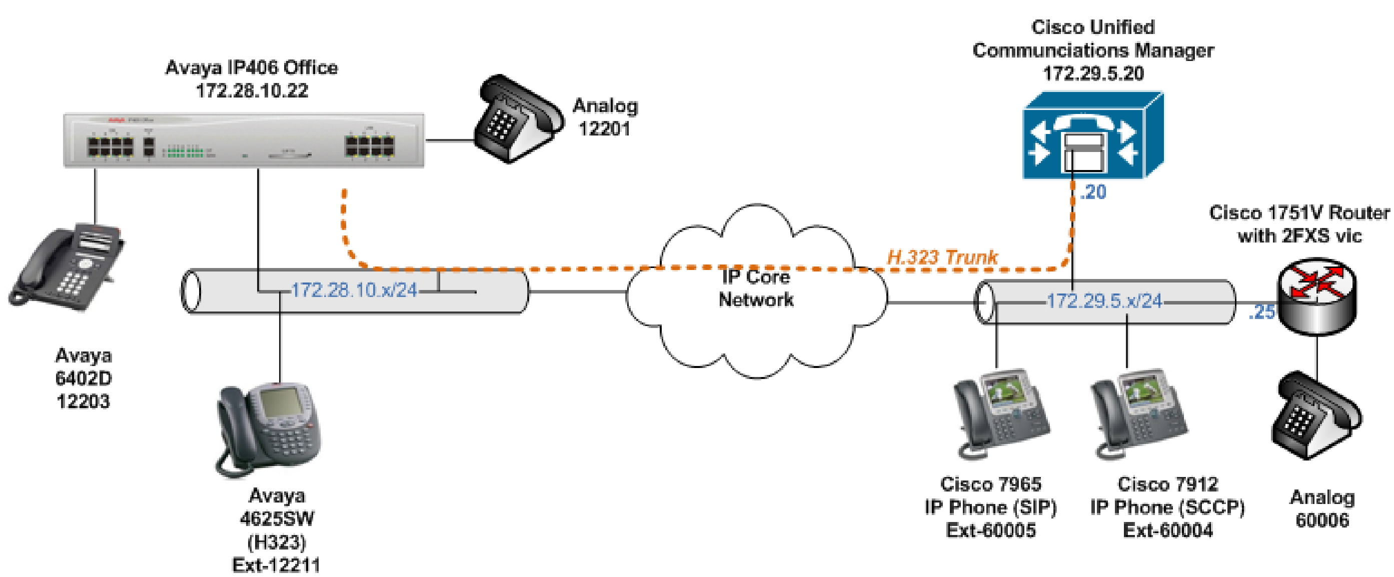


Figure 1: Sample Network Configuration

4. Equipment and Software Validated

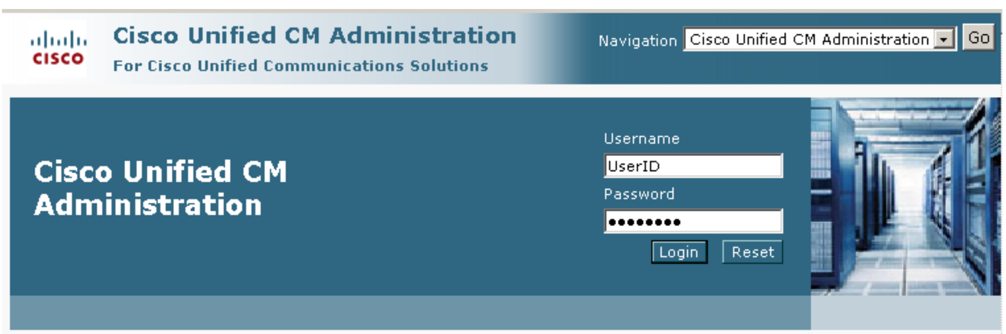
The following equipment and software/firmware were used for the sample configuration:

DEVICE DESCRIPTION	VERSION TESTED
Avaya IP Office 406v2	4.2(11)
Avaya IP Office Manager	6.2(11)
Avaya 4625SW IP Telephone (H323)	2.9
Avaya 6402D Digital Telephone	-
Analog telephone	-
Cisco Unified Communications Manager	7.0.1.1.11000-2
Cisco 7965 Unified IP Phone (SIP)	SIP45.8-4-1S
Cisco 7912 Unified IP Phone (SCCP)	App Load ID CP7912080003SCCP070409A Boot Load ID LD0100BOOT021112A
Cisco 1751v router	IOS 12.4(10a)

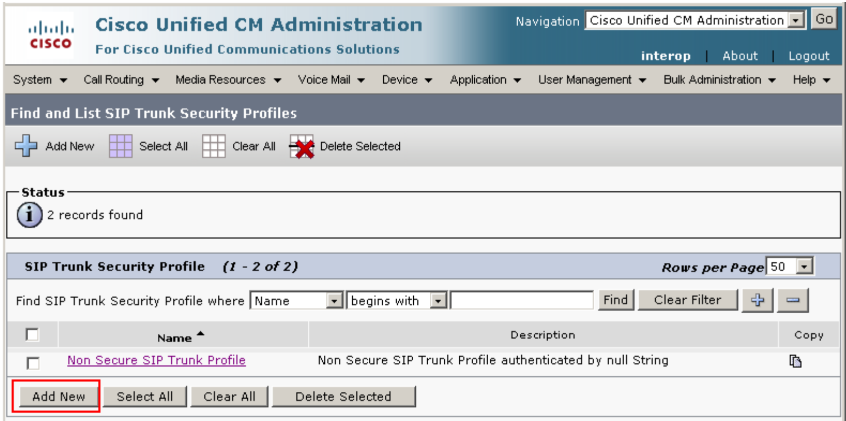
5. Configure Cisco Unified CM

This section describes the SIP Trunk configuration for CUCM as shown in **Figure 1**. Fields left using default value are not highlighted. It is assumed that the basic configuration needed to interoperate with the 1751 router/gateway and support for Cisco IP telephones has been completed. For further information on Cisco Unified CM, please consult reference [2], [3], and [4].

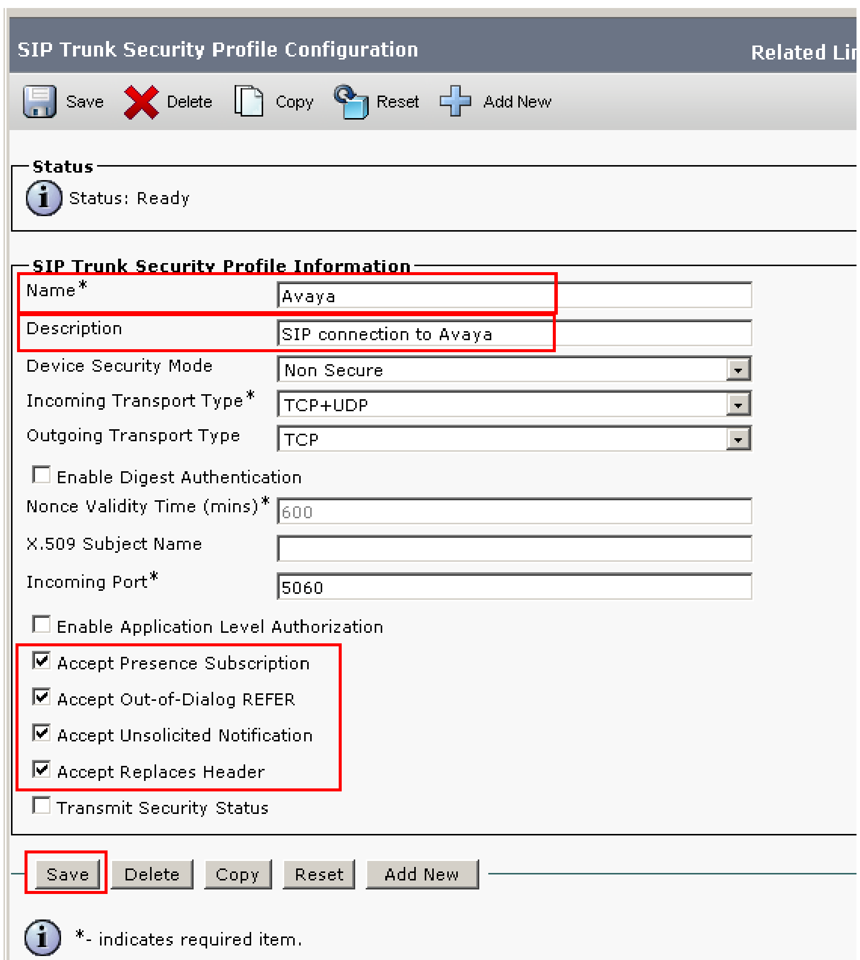
1. Open Cisco Unified CM Administration by entering the IP address of the CUCM into the Web Browser address field, and log in using an appropriate Username and Password.



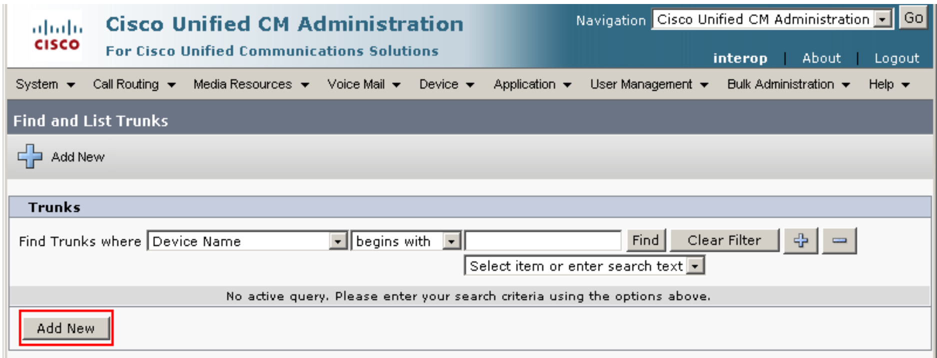
2. Select **System → Security Profile → SIP Trunk Security Profile** from the top menu then click **Add New** to add a new SIP Trunk Security Profile.



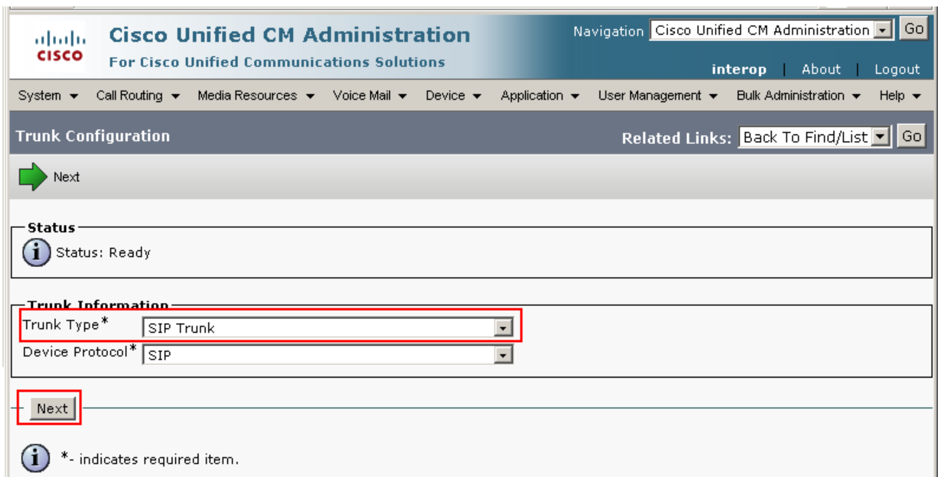
The following is a screen capture of the SIP Trunk Security Profile used in the sample network. Click **Save** to commit the configuration.



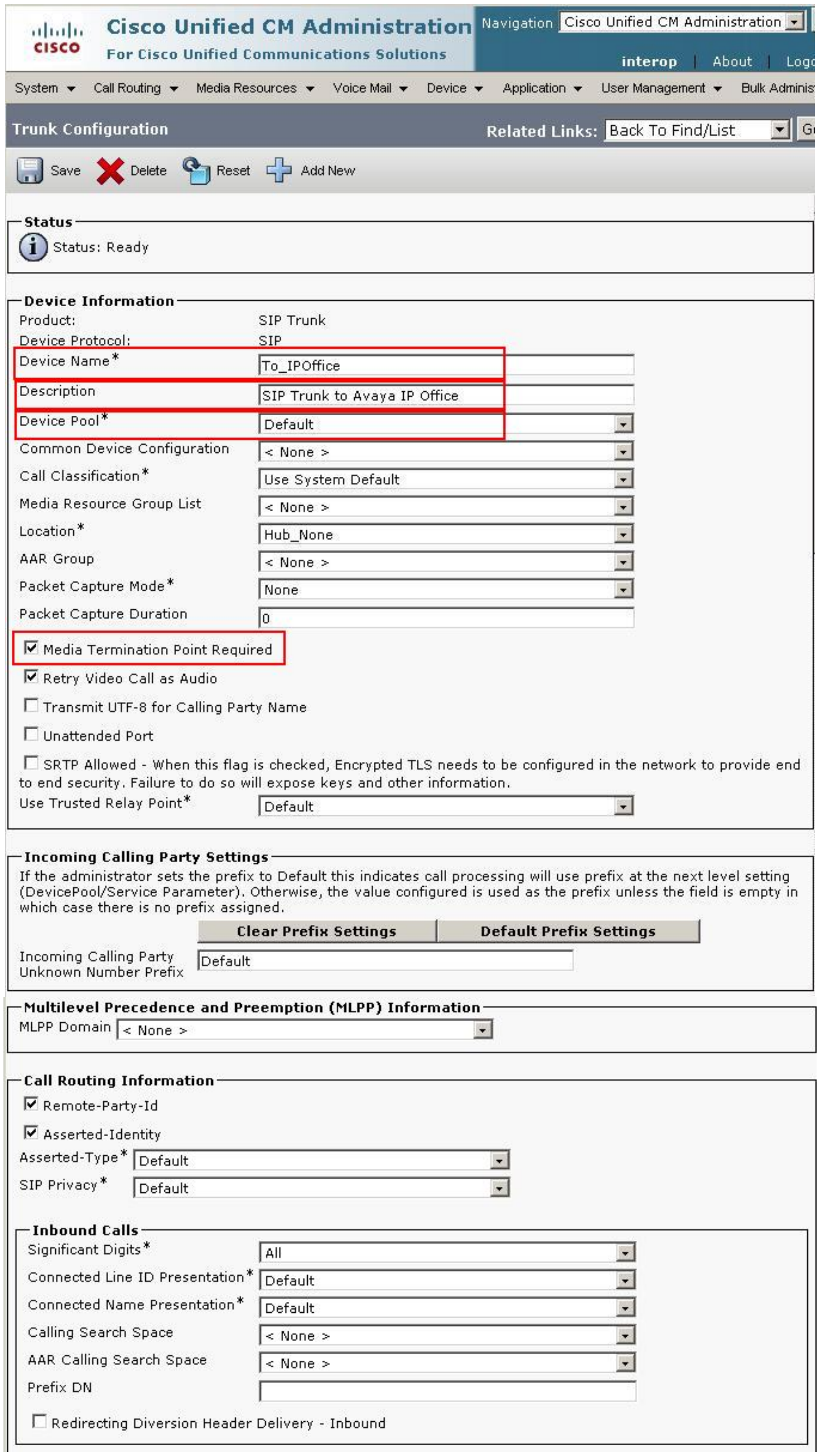
3. Select **Device → Trunk** from the top menu then click **Add New** to begin adding a new SIP trunk.

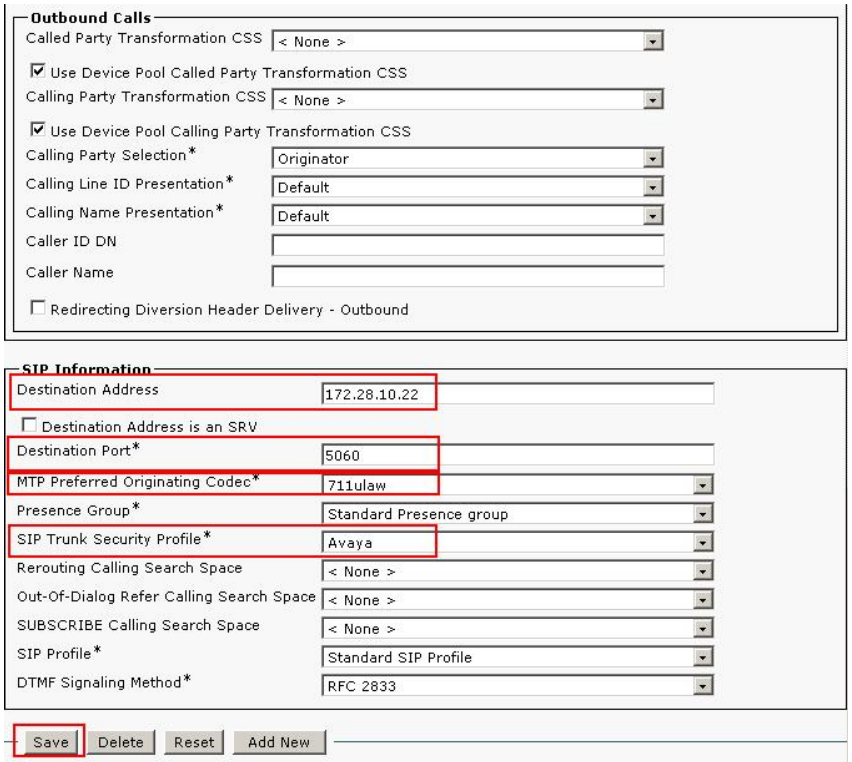


Select **SIP Trunk** as the **Trunk Type** and the **Device Protocol** field will automatically be change to **SIP**. Click **Next** to continue.

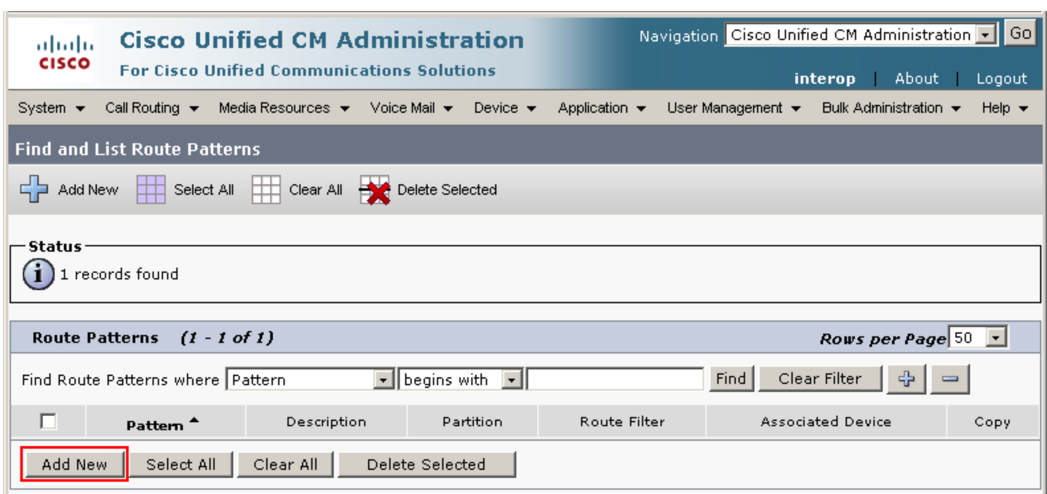


Enter the appropriate information for the SIP Trunk. The following screen capture shows the configuration used in the sample network. Click **Save** to complete. Make sure Media Termination Point Required is checked. This will cause CUCM to include SDP information in its initial SIP Invite message.

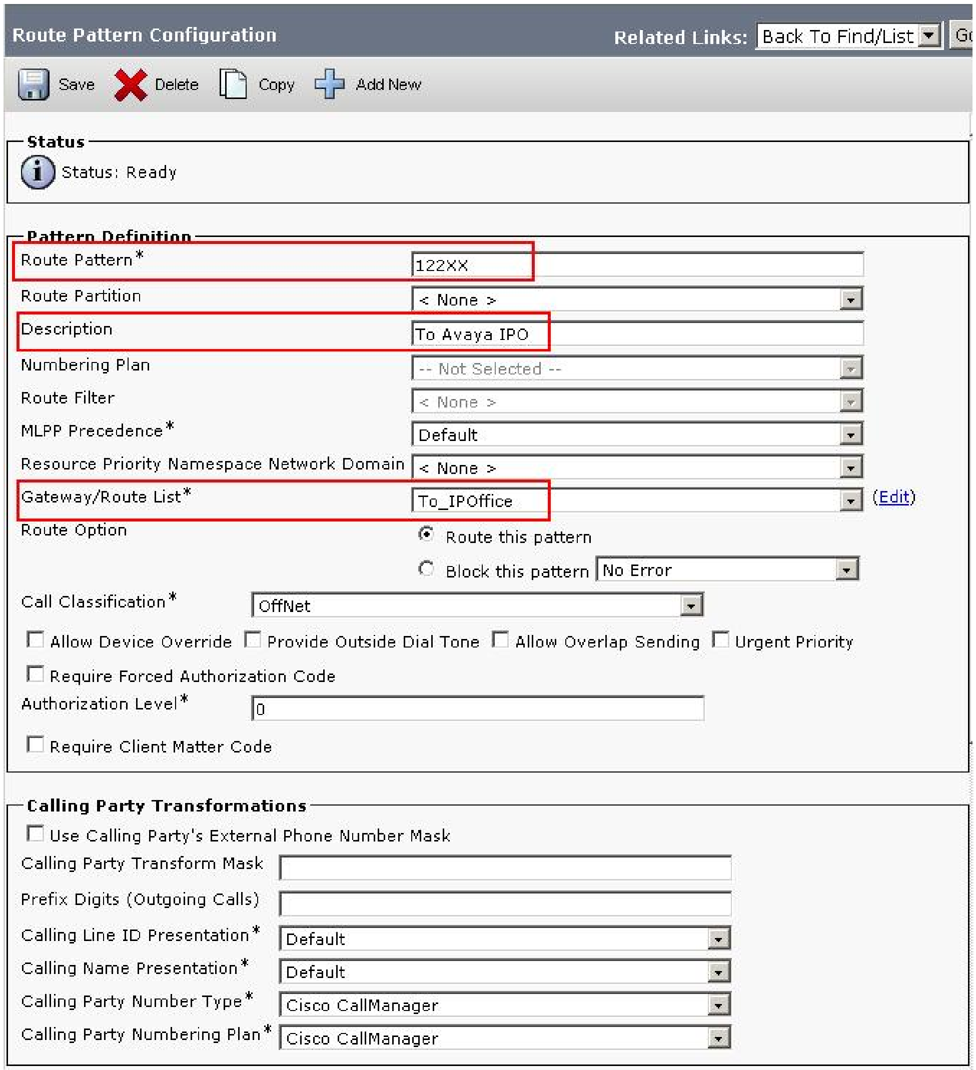


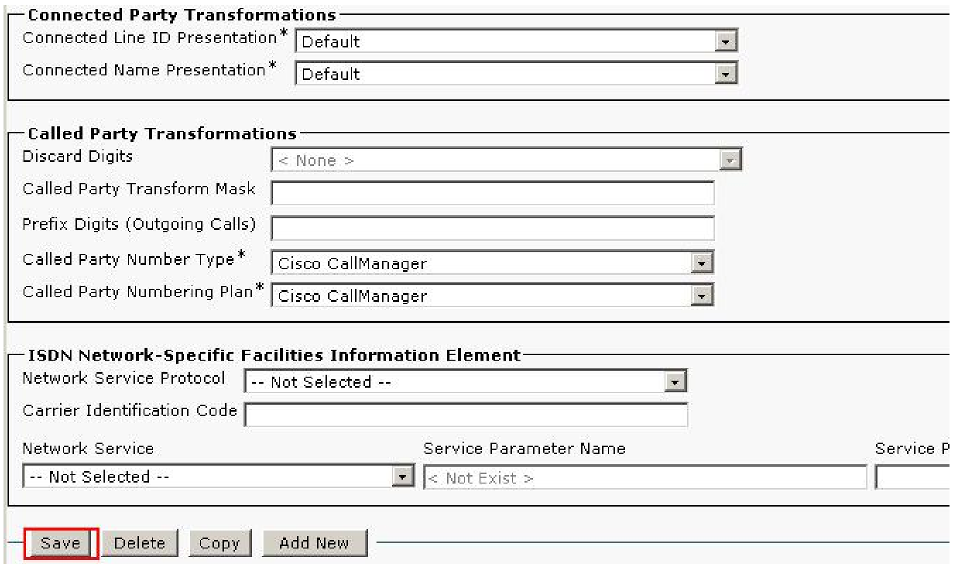


4. Select **Call Routing → Route/Hunt → Route Pattern** then click **Add New** to add a new route pattern for extension 122xx which are for telephones registered with Avaya IP Office.



The following screen capture shows the route pattern used in the sample network. The route pattern "122xx" will cause all 5 digit calls beginning with "122" to be routed to the "To_IPOffice" SIP Trunk defined in **Step 3**. Click **Save** to complete.



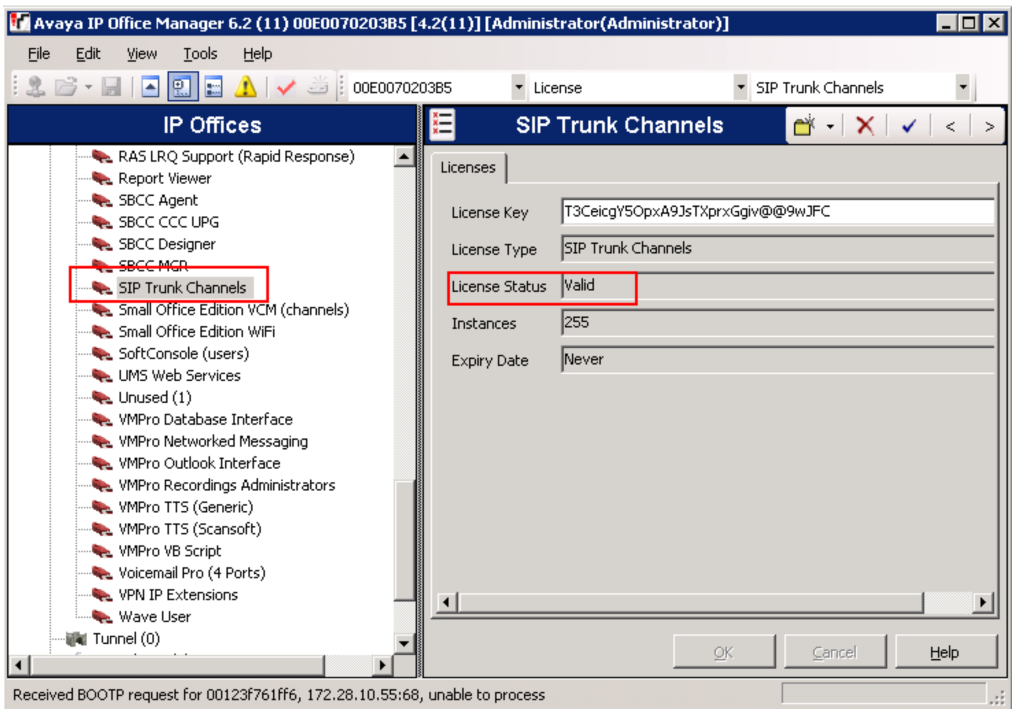


5. This concludes the configuration for CUCM.

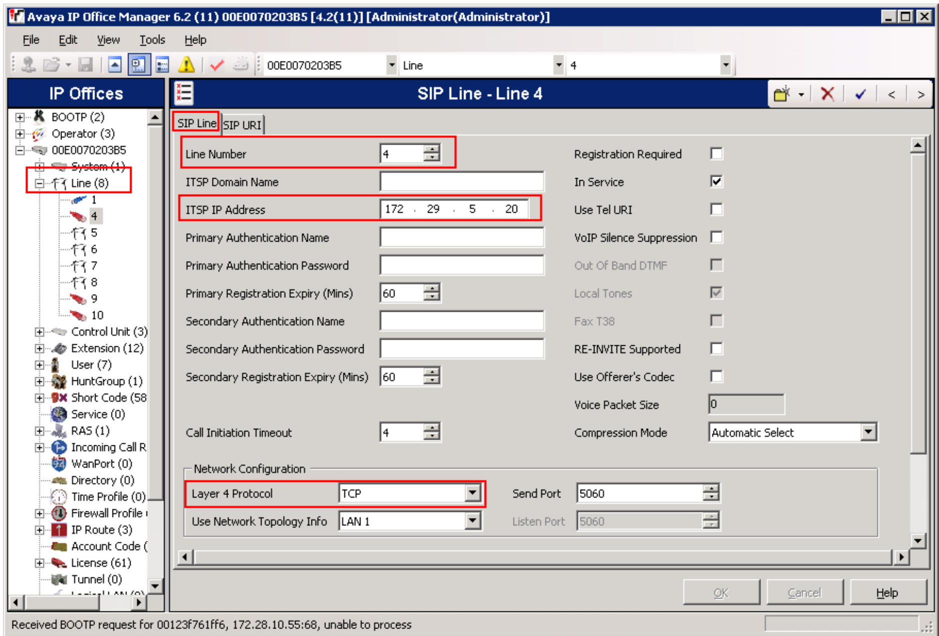
6. Avaya IP Office

This section describes the SIP Trunk configuration for Avaya IP Office as shown in **Figure 1**. It is assumed that the basic configuration has been completed and Avaya IP Office is accessible from the network. Begin by connecting to the Avaya IP Office using the Avaya IP Office Manager and log in using an appropriate user name and password. Fields that needs to be configured are highlighted, all other fields are left with their default value. For further information on Avaya IP Office, please consult reference [1].

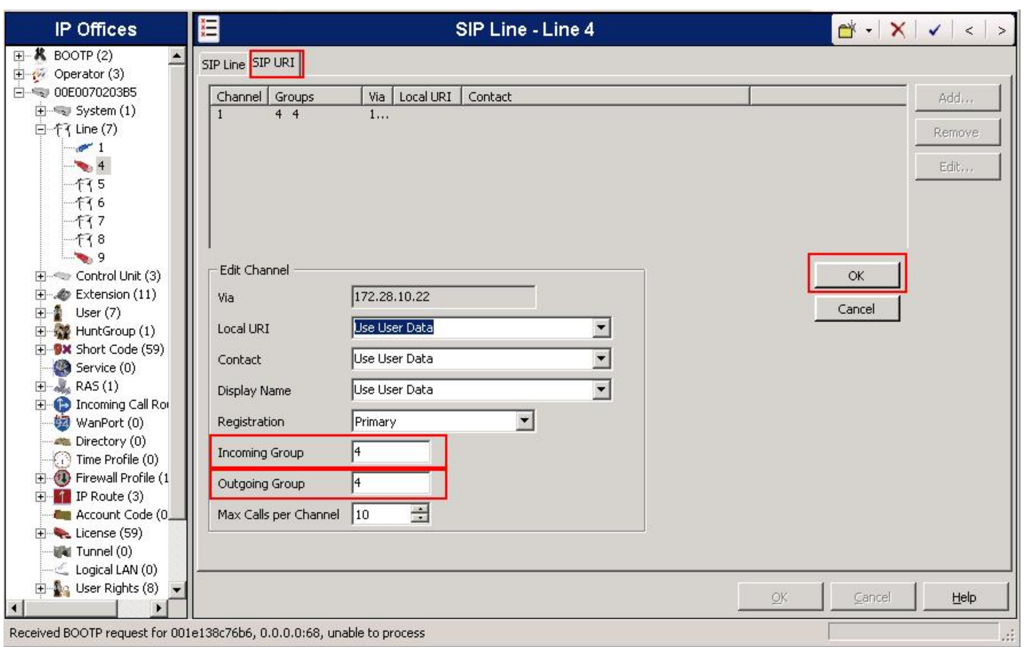
1. Select **License →** SIP Trunk Channels from the left panel menu and verify that there is a valid **SIP Trunk Channels** license. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.



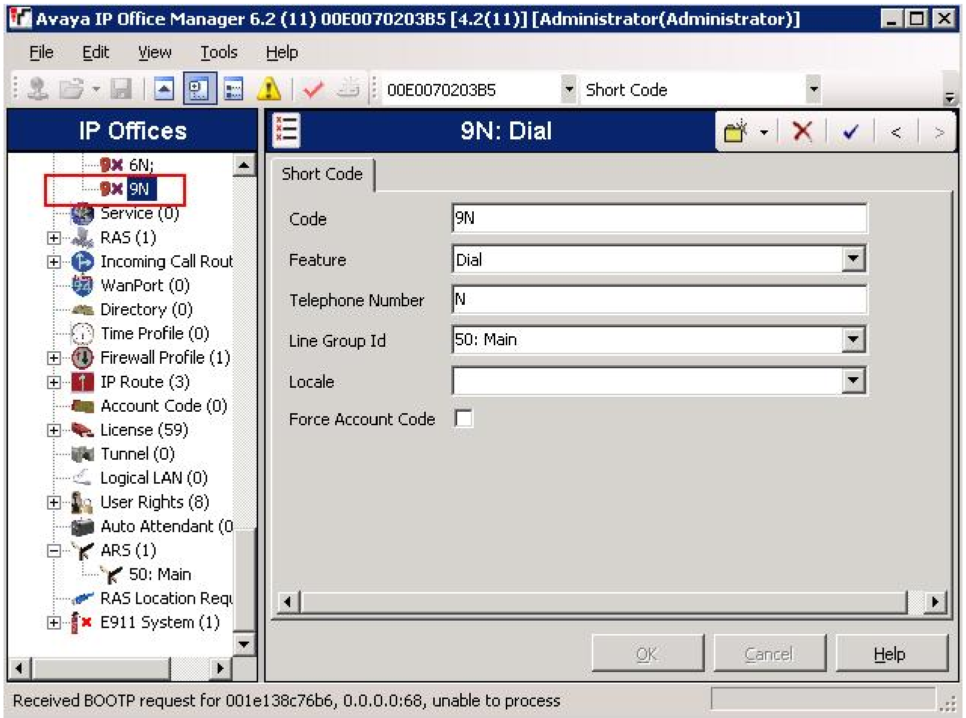
2. Select **Line** from the left panel menu and then right-click and select **New → SIP Line** to create an SIP line to CUCM. Enter the IP address of CUCM in the ITSP Address field. The screen capture below shows the configuration used in the sample network.



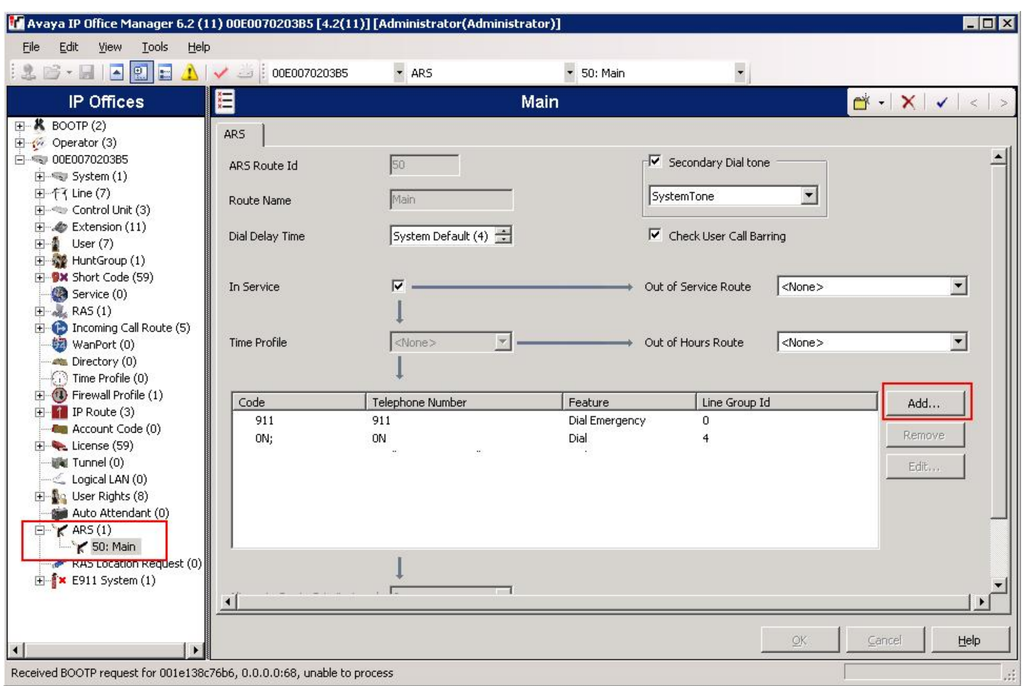
In the SIP URI tab, enter the line number created above in the **Incoming Group** and **Outgoing Group** fields.



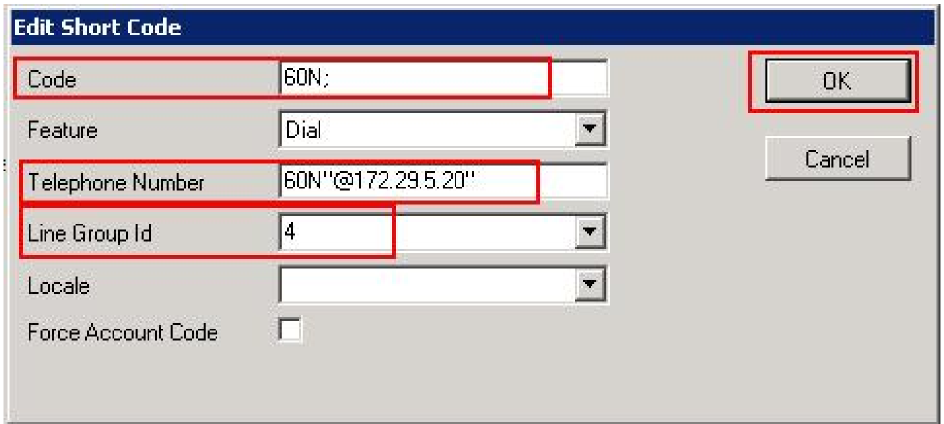
3. By default there should be Short Code for **9N** that routes calls to a default ARS group call **Main**. These Application Notes will use ARS to route call to CUCM. The screen capture below shows the default **9N** Short Code.



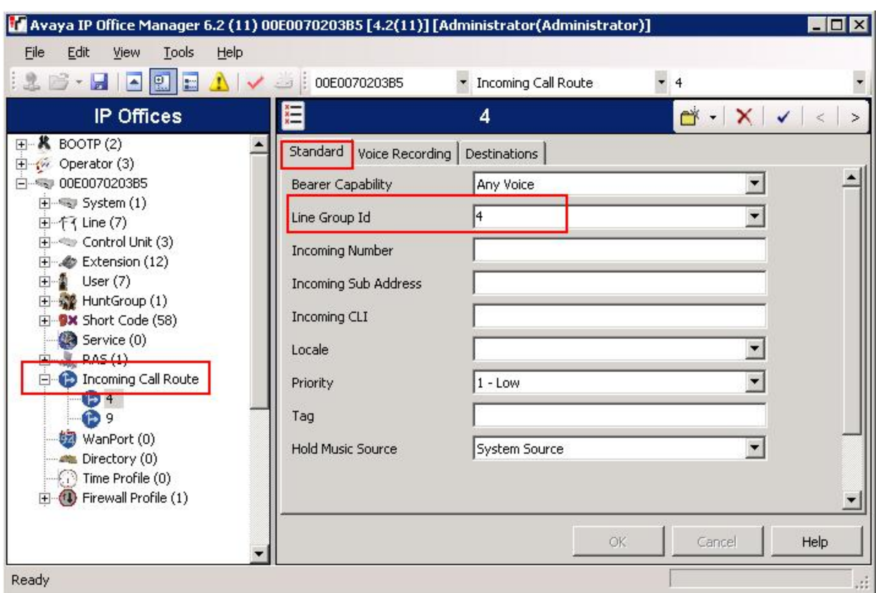
4. Select **ARS → Main** from the left panel menu, and then click on **Add** to create a new Code entry to route call to CUCM.



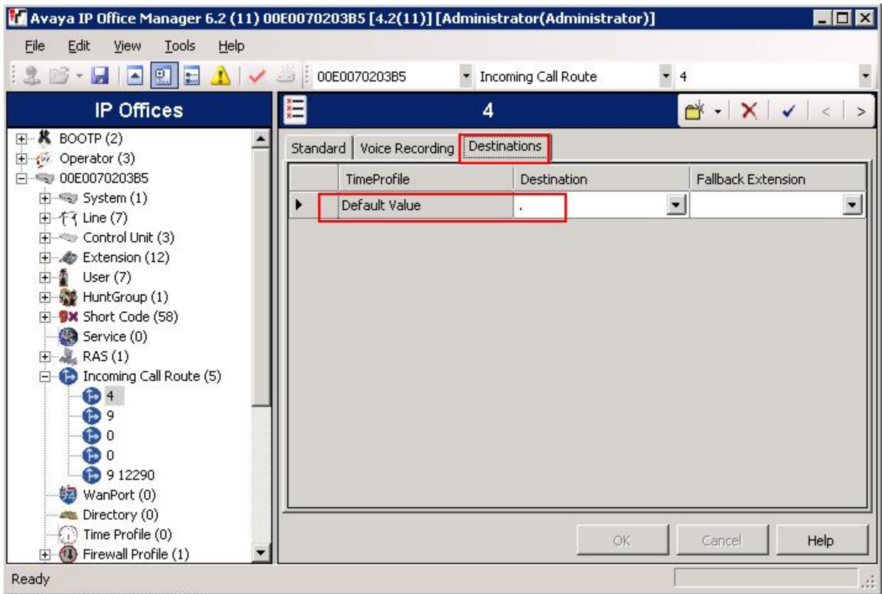
Enter the appropriate information for the Code entry. The following screen capture shows a portion of the Cisco dialing plan “60” is being used as part of the Code. The Telephone Number is composed of the called phone number appended with “@” and the CUCM IP Address. **Line Group ID** 4 created in **Step 2** will be used to send out the call.



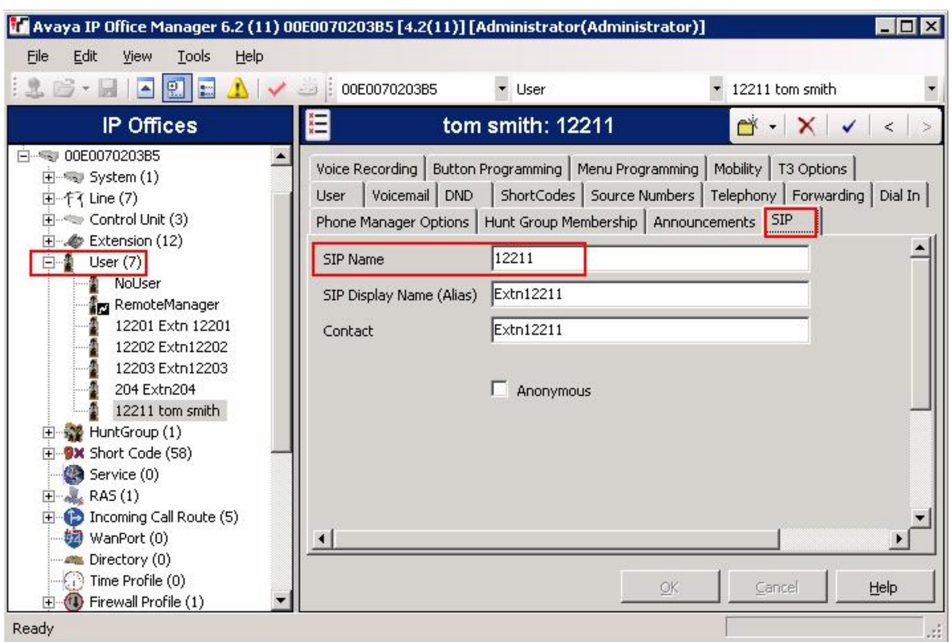
5. Select **Incoming Call Route** from the left panel menu and then right-click and select **New** to create a new Incoming Call Route. Under the **Standard** tab, select the Line Group number created in **Step 2** in the **Line Group Id** field. The following screen capture shows the setting used in the sample network.



Under the **Destination** tab, enter "." as the **Default Value**. The "." indicate the incoming call can be route to any extension. The following screen capture shows the setting used.



6. Select **User** from the left panel menu. After selecting a user that need to be configured to use the SIP trunk, select the **SIP** tab on right panel window. Modify the **SIP Name** to be the same as the user's extension number. The other field can be left as default.

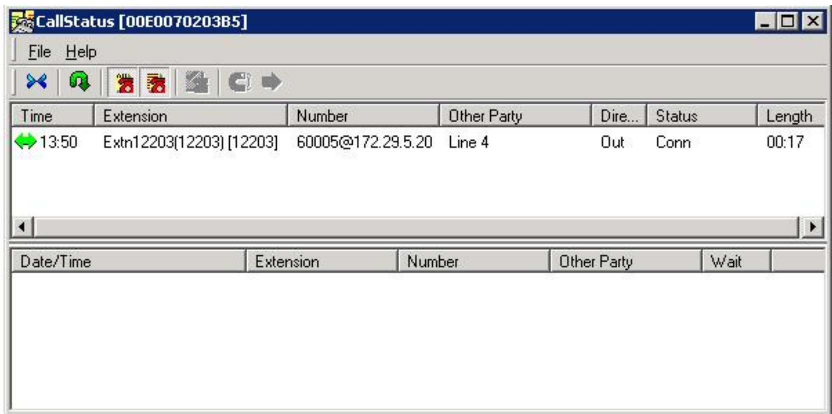


7. Repeat **Step 6** for all users on the system to complete the IP Office provisioning.

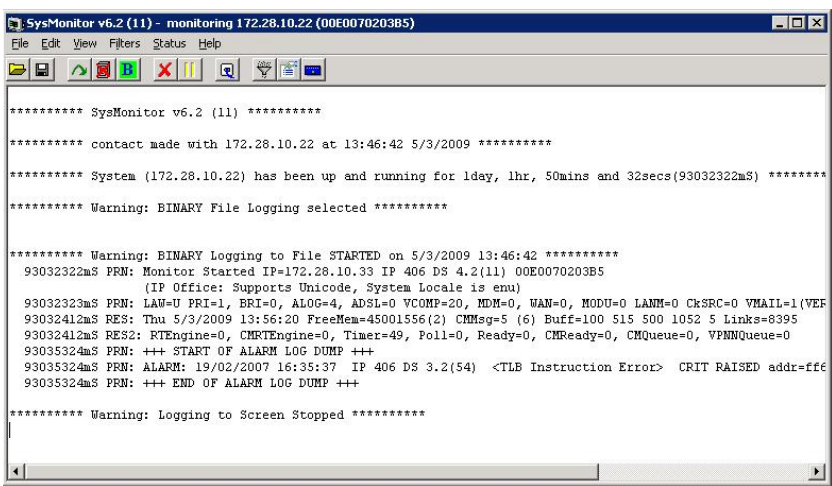
7. Verification

The following steps may be used to verify the configuration:

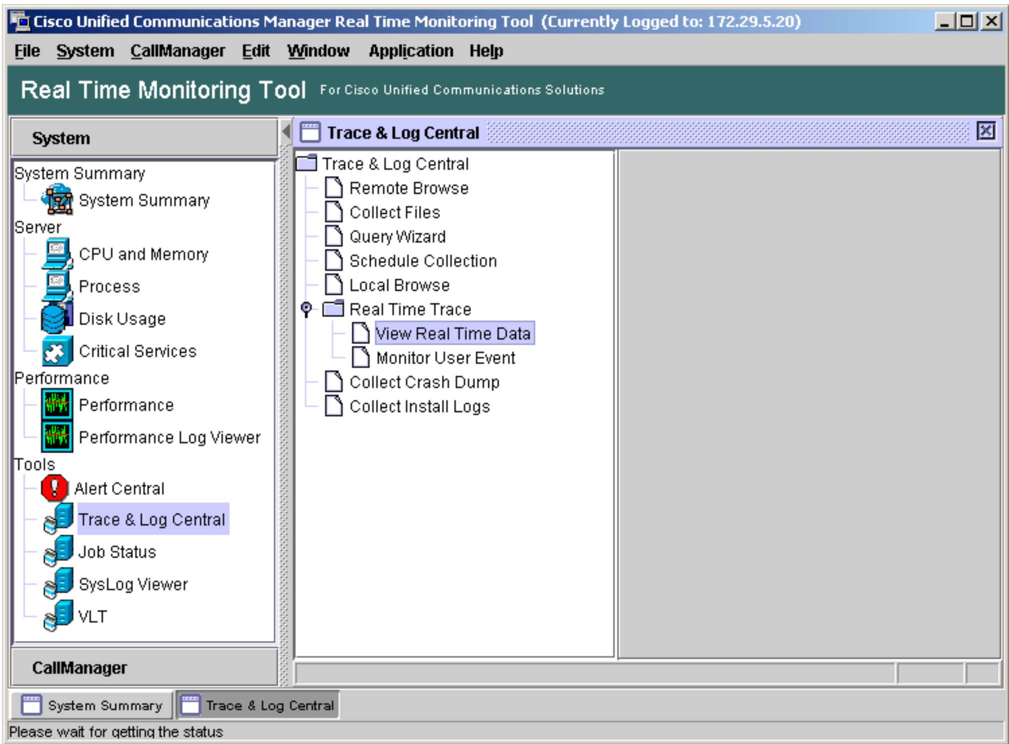
1. Call status can be monitored using **Start → Programs → IP Office → Call Status**. The following is a screen capture shows a out going call being made from extension 12203 to 60005 using Line Group 4.



2. From the computer where IP Office Manager is installed, select **Start → Programs → IP Office → Monitor** to view Avaya IP Office debugging information. The following is a screen capture of the sysMonitor window.



3. The Real Time Monitoring Tool (RTMT) can be use to monitor events on CUCM. This tool can be downloaded by selecting **Application → Plugins** from the top menu of the Cisco Unified CM Administration Web interface. The following is a screen capture of the Csico Unified Communications Manager Real Teim Monitoring Tool. For further information on this tool, please consult with reference [5].



8. Conclusion

These Application Notes described the administrative steps required to configure a SIP trunk to support calls between Avaya IP Office and a Cisco Unified Communications Manager system. Basic calling including Hold, Transfer, Conference and Fax Pass-through as well as supplemental features such as Call Forward All, Call Park/Unpark are supported by this configuration. Please note that the version of IP Office shown in these Application only support initial SIP Invite message that contain SDP information, which is not the default configuration for CUCM. One way to configure CUCM to include SDP with its initial SIP Invite message is to enable Media Terminal Point Required option as shown in **Section 5**, **Step 3**.

9. Additional References

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

[1] *Avaya IP Office 4.2 Manager 6.2*, Part Number: 15-601011, Issue 22k, September 09 2008

Product documentation for Cisco Systems products may be found at http://www.cisco.com

[2] *Cisco SIP IP Phone Administrator Guide*, *Release 6.0, 6.1*, *7.0*, *7.1*, May 2004,

[3] *Cisco Unifed Communications Manager Administration Guide for Cisco Unifed Communications Manager Business Edition*, Release 7.0(1), Part Number: OL-15405-01

[4] *Cisco Unified Communitions Manager Features and Services Guide for Cisco Unified Communications Manager Business Edition*, Release 7.0(1), Part Number: OL-15409-01

[5] *Cisco Unified Real-Time Monitoring Tool Administration Guide*, Release 7.0(1), Part Number: OL-14994-01

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com