



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.

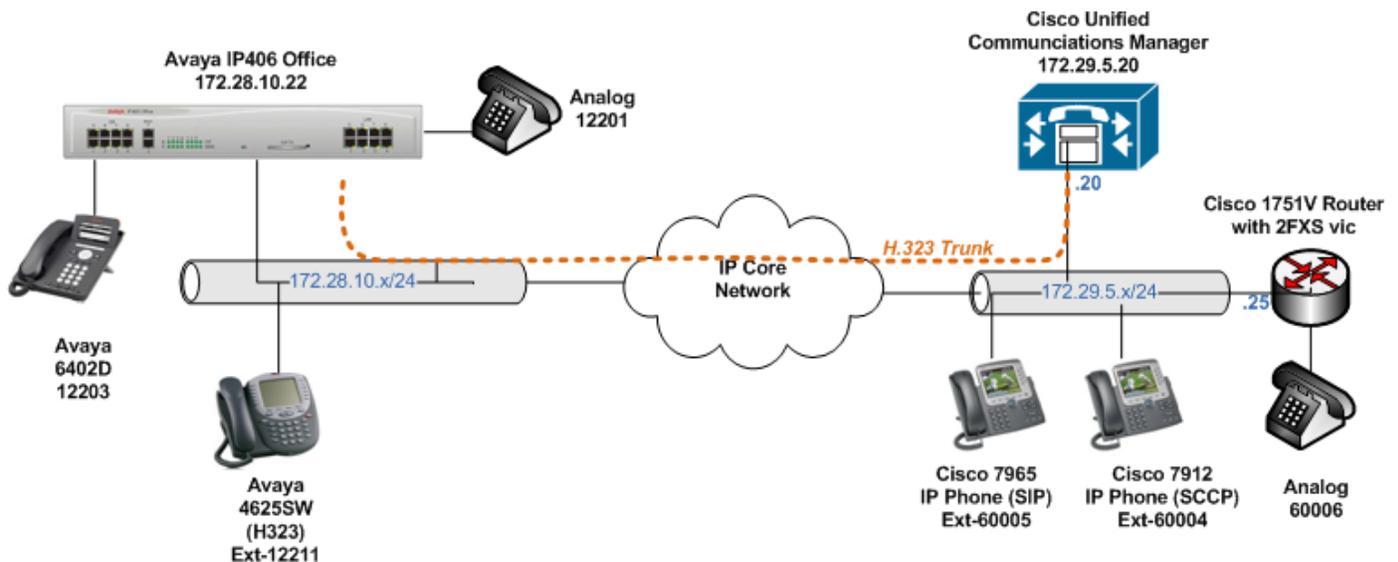


Figure 1: Sample Network Configuration

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)

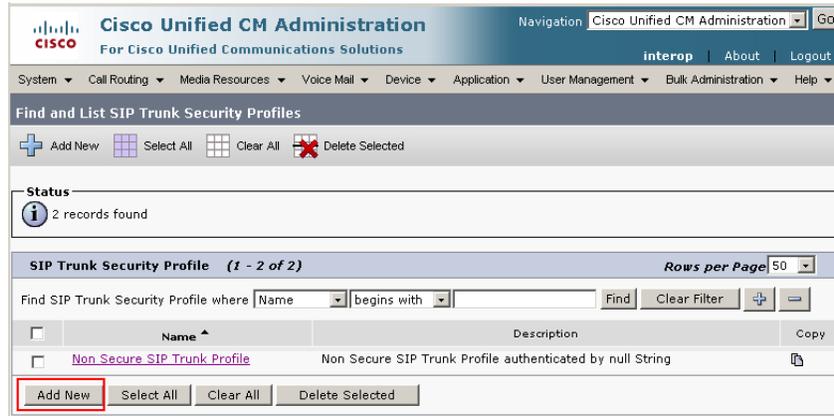
5. Configure Cisco Unified CM

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].

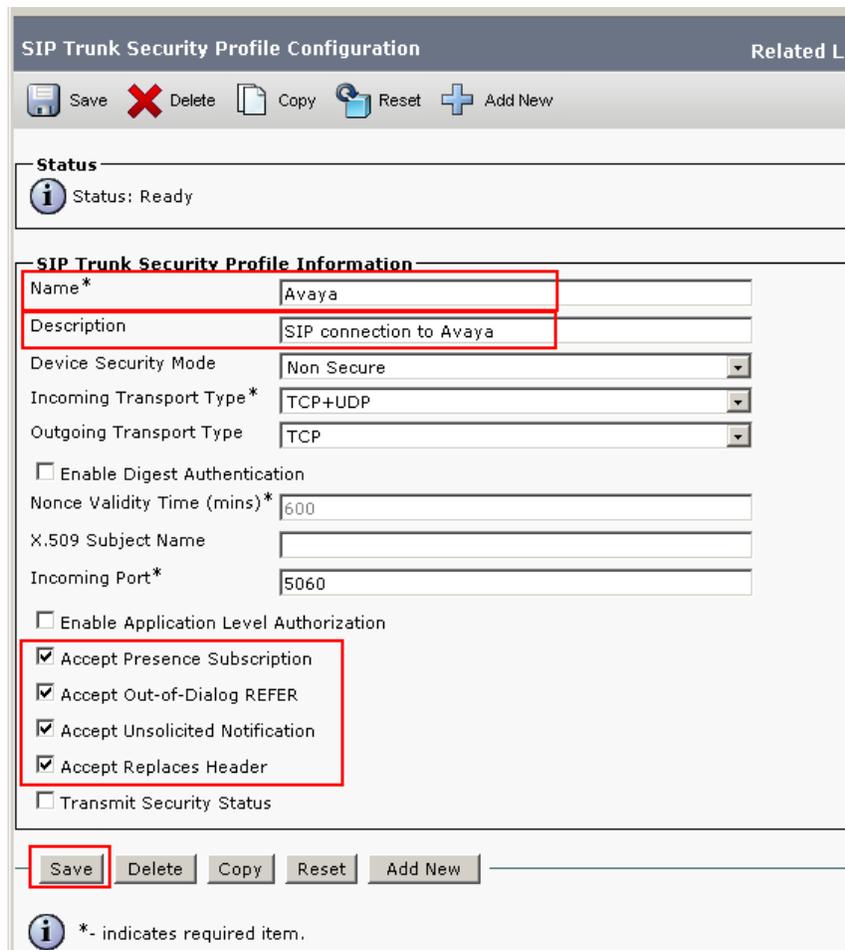
1. Open Cisco Unified CM Administration by entering the IP address of the CUCM into the Web Browser address field, and log in using an appropriate Username and Password.



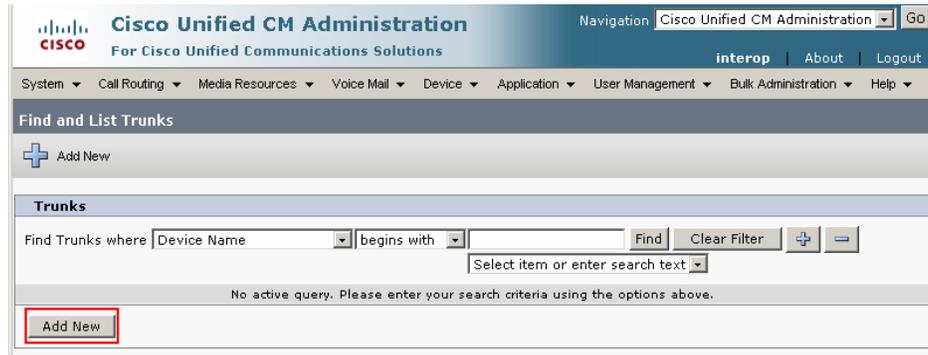
2. Select **System** → **Security Profile** → **SIP Trunk Security Profile** from the top menu then click **Add New** to add a new SIP Trunk Security Profile.



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.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration

Trunk Configuration | Related Links: Back To Find/List

Save | Delete | Reset | Add New

Status
Status: Ready

Device Information

Product: SIP Trunk
 Device Protocol: SIP
 Device Name*: To_IPOffice
 Description: SIP Trunk to Avaya IP Office
 Device Pool*: Default
 Common Device Configuration: < None >
 Call Classification*: Use System Default
 Media Resource Group List: < None >
 Location*: Hub_None
 AAR Group: < None >
 Packet Capture Mode*: None
 Packet Capture Duration: 0
 Media Termination Point Required
 Retry Video Call as Audio
 Transmit UTF-8 for Calling Party Name
 Unattended Port
 SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.
 Use Trusted Relay Point*: Default

Incoming Calling Party Settings
 If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
 Clear Prefix Settings | Default Prefix Settings
 Incoming Calling Party: Default
 Unknown Number Prefix: Default

Multilevel Precedence and Preemption (MLPP) Information
 MLPP Domain: < None >

Call Routing Information
 Remote-Party-Id
 Asserted-Identity
 Asserted-Type*: Default
 SIP Privacy*: Default

Inbound Calls
 Significant Digits*: All
 Connected Line ID Presentation*: Default
 Connected Name Presentation*: Default
 Calling Search Space: < None >
 AAR Calling Search Space: < None >
 Prefix DN:
 Redirecting Diversion Header Delivery - Inbound

Outbound Calls

Called Party Transformation CSS < None >

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS

Calling Party Selection* Originator

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Caller ID DN

Caller Name

Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination Address 172.28.10.22

Destination Address is an SRV

Destination Port* 5060

MTP Preferred Originating Codec* 711ulaw

Presence Group* Standard Presence group

SIP Trunk Security Profile* Avaya

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile* Standard SIP Profile

DTMF Signaling Method* RFC 2833

Save Delete Reset Add New

- 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.

Cisco Unified CM Administration Navigation Cisco Unified CM Administration Go

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System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Find and List Route Patterns

+ Add New Select All Clear All Delete Selected

Status

1 records found

Route Patterns (1 - 1 of 1) Rows per Page 50

Find Route Patterns where Pattern begins with Find Clear Filter + -

Pattern	Description	Partition	Route Filter	Associated Device	Copy

Add New Select All Clear All Delete Selected

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.

The screenshot displays the 'Route Pattern Configuration' window. At the top, there are navigation icons for Save, Delete, Copy, and Add New. The 'Status' section shows 'Status: Ready'. The 'Pattern Definition' section contains the following fields:

- Route Pattern*: 122XX
- Route Partition: < None >
- Description: To Avaya IPO
- Numbering Plan: -- Not Selected --
- Route Filter: < None >
- MLPP Precedence*: Default
- Resource Priority Namespace Network Domain: < None >
- Gateway/Route List*: To_IPOffice (with an Edit link)
- Route Option: Route this pattern, Block this pattern (No Error)
- Call Classification*: OffNet
- Checkboxes: Allow Device Override, Provide Outside Dial Tone, Allow Overlap Sending, Urgent Priority, Require Forced Authorization Code, Require Client Matter Code.
- Authorization Level*: 0

The 'Calling Party Transformations' section includes:

- Use Calling Party's External Phone Number Mask
- Calling Party Transform Mask: [Empty field]
- Prefix Digits (Outgoing Calls): [Empty field]
- Calling Line ID Presentation*: Default
- Calling Name Presentation*: Default
- Calling Party Number Type*: Cisco CallManager
- Calling Party Numbering Plan*: Cisco CallManager

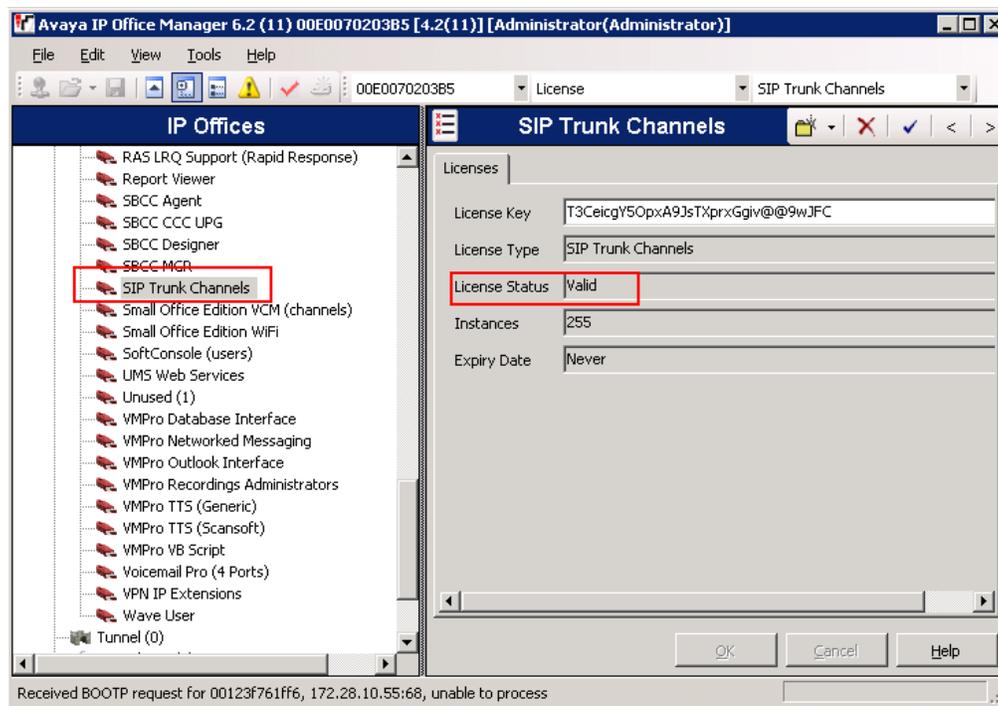
Connected Party Transformations		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
Called Party Transformations		
Discard Digits	< None >	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
ISDN Network-Specific Facilities Information Element		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service P
-- Not Selected --	< Not Exist >	
Save	Delete	Add New

5. This concludes the configuration for CUCM.

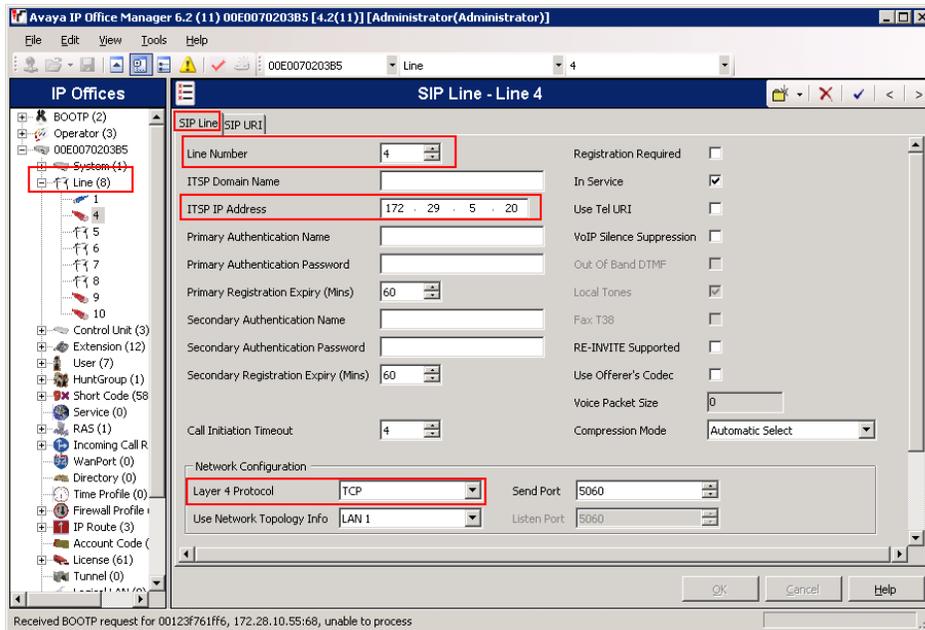
6. Avaya IP Office

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].

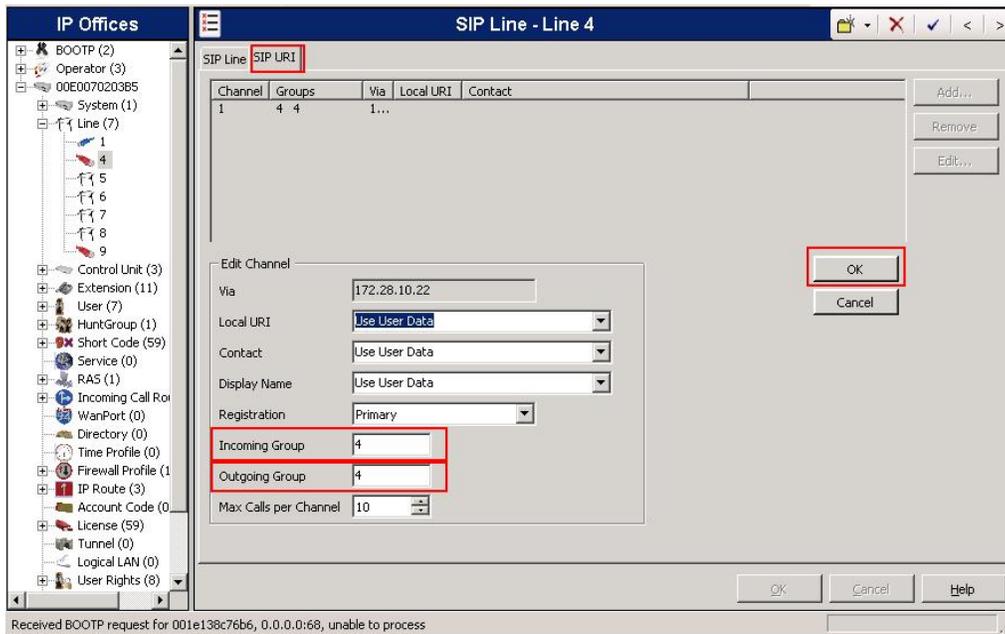
1. Select **License** → **SIP Trunk Channels** from the left panel menu and verify that there is a valid **SIP Trunk Channels** license. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.



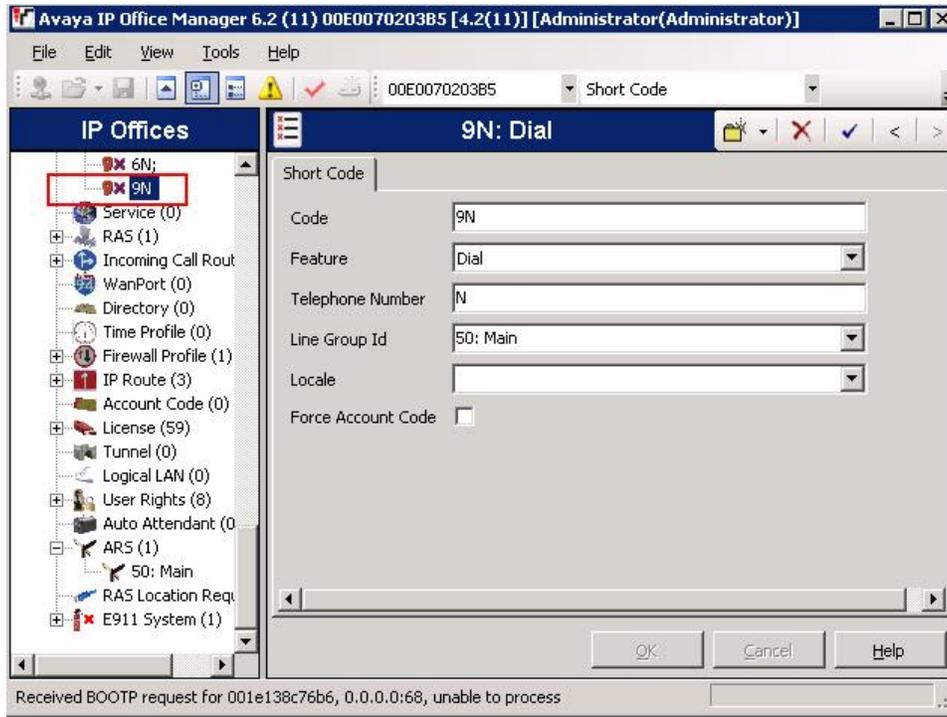
- Select **Line** from the left panel menu and then right-click and select **New → SIP Line** to create an SIP line to CUCM. Enter the IP address of CUCM in the ITSP Address field. The screen capture below shows the configuration used in the sample network.



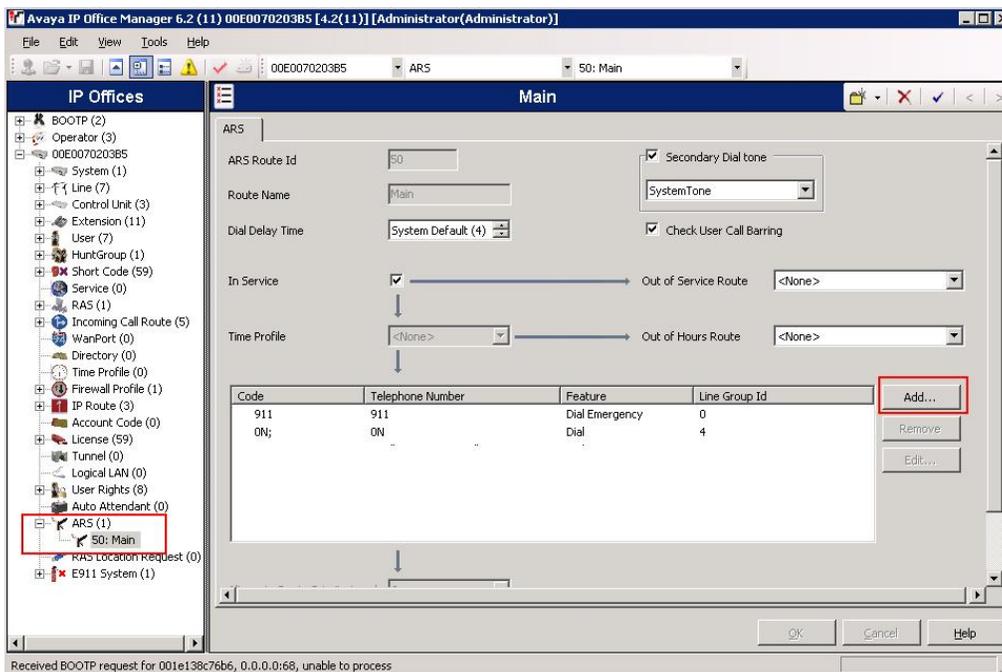
In the SIP URI tab, enter the line number created above in the **Incoming Group** and **Outgoing Group** fields.



- 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.



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



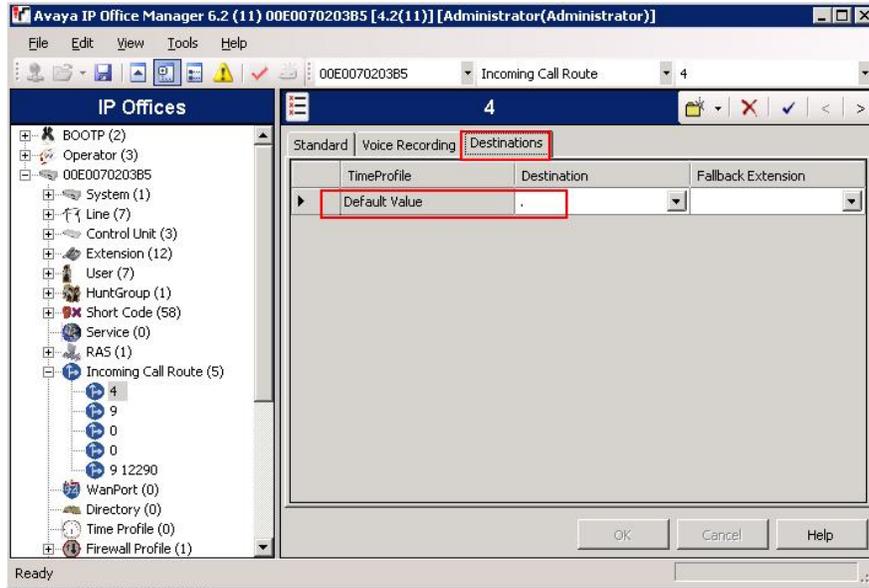
Enter the appropriate information for the Code entry. The following screen capture shows a portion of the Cisco dialing plan “60” is being used as part of the Code. The Telephone Number is composed of the called phone number appended with “@” and the CUCM IP Address. **Line Group ID** 4 created in **Step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

Code	60N;	OK
Feature	Dial	Cancel
Telephone Number	60N"@172.29.5.20"	
Line Group Id	4	
Locale		
Force Account Code	<input type="checkbox"/>	

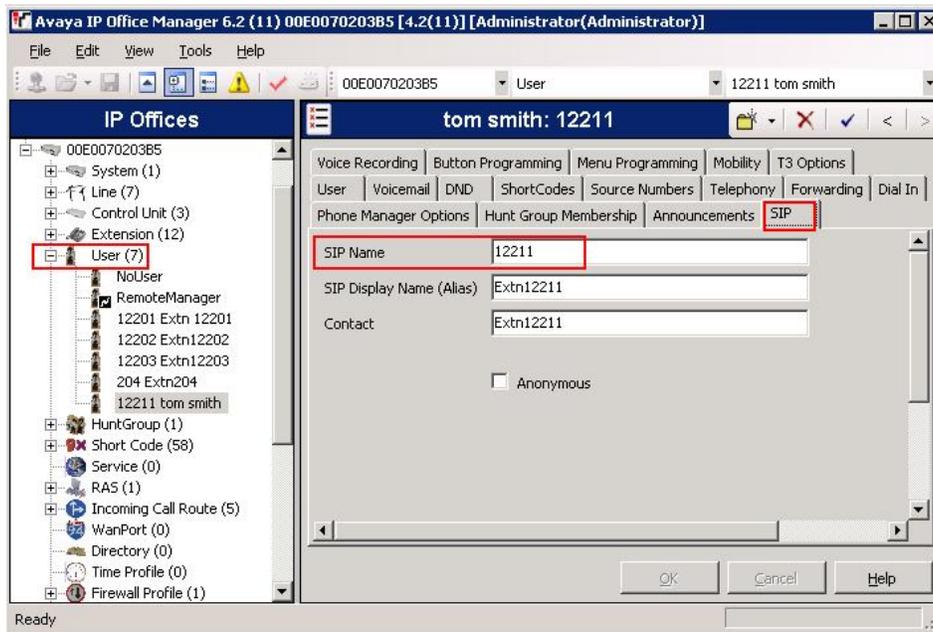
5. Select **Incoming Call Route** from the left panel menu and then right-click and select **New** to create a new Incoming Call Route. Under the **Standard** tab, select the Line Group number created in **Step 2** in the **Line Group Id** field. The following screen capture shows the setting used in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

Standard	Voice Recording	Destinations
Bearer Capability	Any Voice	
Line Group Id	4	
Incoming Number		
Incoming Sub Address		
Incoming CLI		
Locale		
Priority	1 - Low	
Tag		
Hold Music Source	System Source	

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.

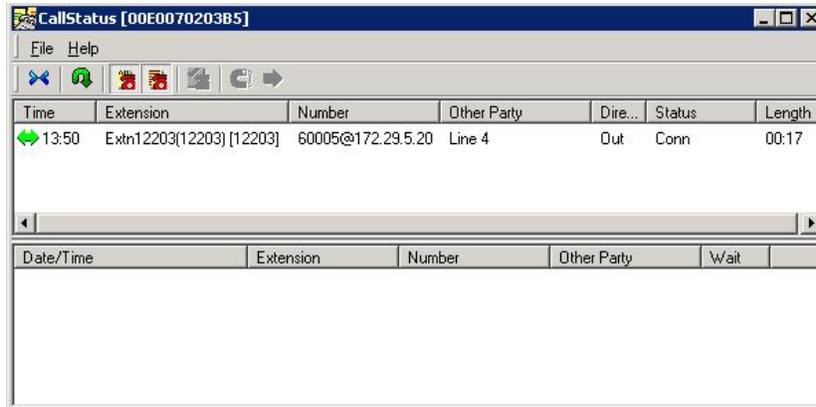


7. Repeat **Step 6** for all users on the system to complete the IP Office provisioning.

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.

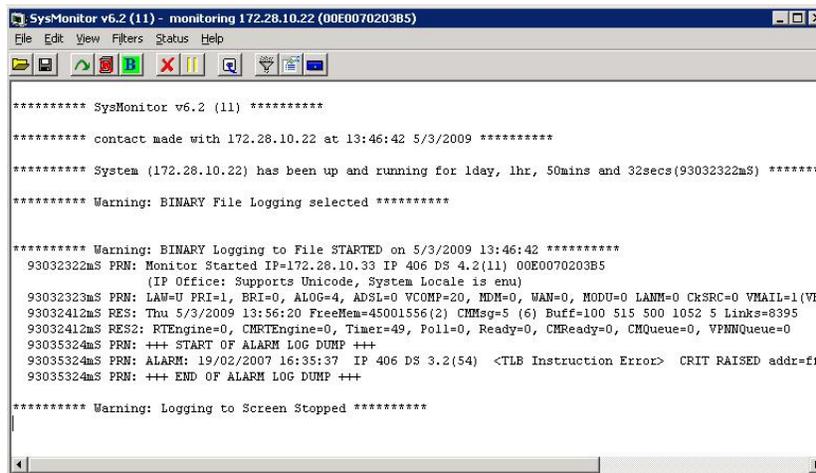


The screenshot shows a window titled "CallStatus [00E0070203B5]". It contains a table with the following data:

Time	Extension	Number	Other Party	Dir...	Status	Length
13:50	Extn12203(12203)	[12203]	60005@172.29.5.20	Line 4	Out Conn	00:17

Below the table is another table with headers: Date/Time, Extension, Number, Other Party, Wait.

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.

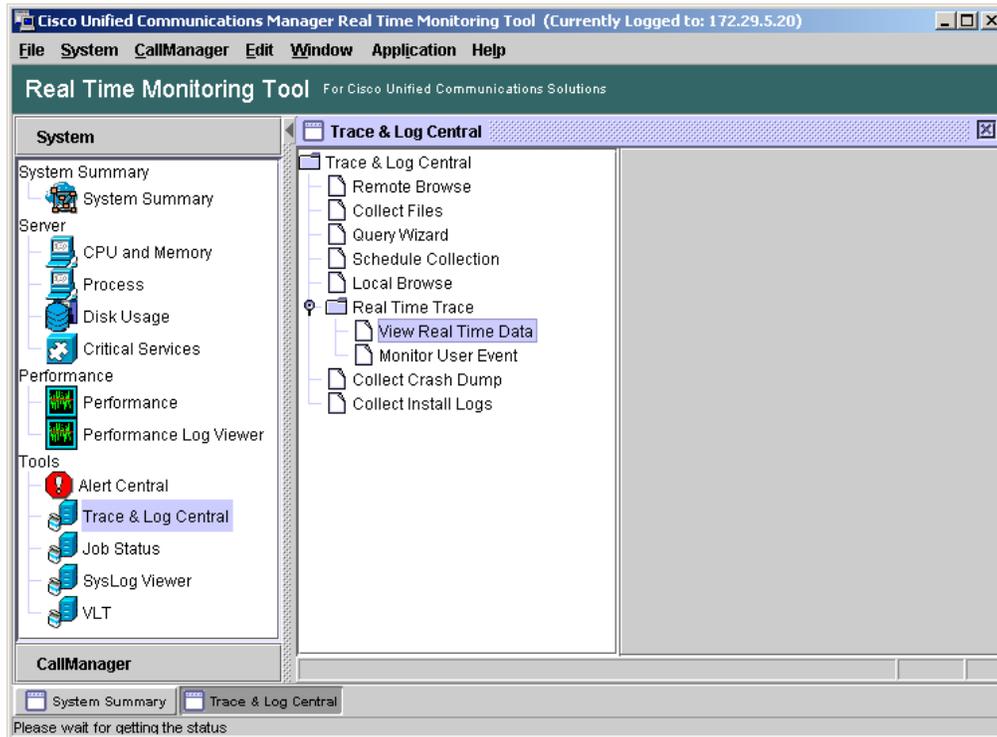


```
***** SysMonitor v6.2 (11) *****
***** contact made with 172.28.10.22 at 13:46:42 5/3/2009 *****
***** System (172.28.10.22) has been up and running for 1day, 1hr, 50mins and 32secs(93032322mS) *****
***** Warning: BINARY File Logging selected *****

***** Warning: BINARY Logging to File STARTED on 5/3/2009 13:46:42 *****
93032322mS PRN: Monitor Started IP=172.28.10.33 IP 406 DS 4.2(11) 00E0070203B5
(IP Office: Supports Unicode, System Locale is emu)
93032323mS PRN: LAN=U PRI=1, BRI=0, ALOG=4, ADSL=0 VCOMP=20, MDN=0, WAN=0, MODU=0 LAMN=0 CkSRC=0 VMAIL=1(VEF
93032412mS RES: Thu 5/3/2009 13:56:20 FreeMem=45001556(2) CMMsg=5 (6) Buff=100 515 500 1052 5 Links=8395
93032412mS RES2: RTEngine=0, CMREngine=0, Timer=49, Poll=0, Ready=0, CMReady=0, CMQueue=0, VFNQueue=0
93035324mS PRN: +++ START OF ALARM LOG DUMP +++
93035324mS PRN: ALARM: 19/02/2007 16:35:37 IP 406 DS 3.2(54) <TLB Instruction Error> CRIT RAISED addr=fff
93035324mS PRN: +++ END OF ALARM LOG DUMP +++

***** Warning: Logging to Screen Stopped *****
```

3. The Real Time Monitoring Tool (RTMT) can be used 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 Cisco Unified Communications Manager Real Time 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 Unified Communications Manager Administration Guide for Cisco Unified Communications Manager Business Edition, Release 7.0(1)*, Part Number: OL-15405-01
- [4] *Cisco Unified Communications 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

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