Avaya Solution & Interoperability Test Lab

Sample Configuration for SIP Trunking between Avaya IP Office R6.1 and Cisco Unified Communications Manager 8.0 – Issue 1.0

Abstract

These Application Notes describe the steps for configuring a SIP trunk between Avaya IP Office R6.1 and Cisco Unified Communications Manager (CUCM) Release 8.0.

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1. Introduction

Session Initiation Protocol (SIP) is a standards-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 1600 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 an optional Cisco VG248 gateway (not shown). 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, digital, SCCP or analog stations.

3. Configuration

Figure 1 illustrates the configuration used in these Application Notes. All IP telephones in the 33.1.1.0/24 IP network are registered with Avaya IP Office and use extension 2xx. All IP telephones in the 10.80.60.0/24 IP network are registered with CUCM and use extension 8xxx. A single SIP trunk between Avaya IP Office and CUCM manages call control between the Avaya and Cisco IP PBX systems.

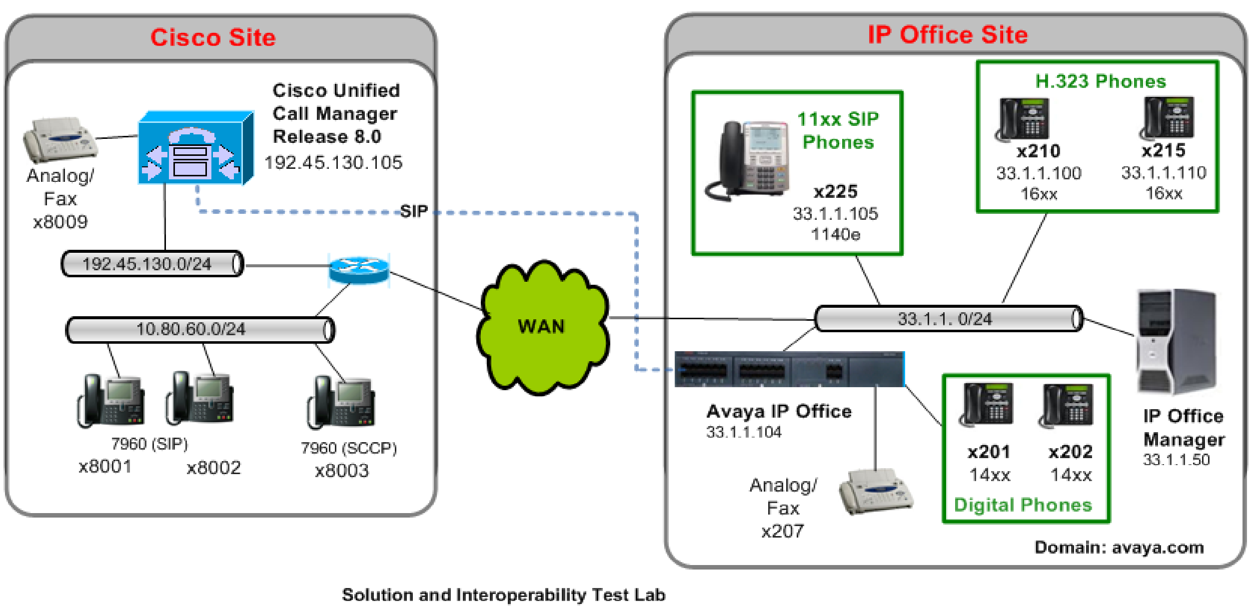


Figure 1: Sample Network Configuration

4. Equipment and Software Validated

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

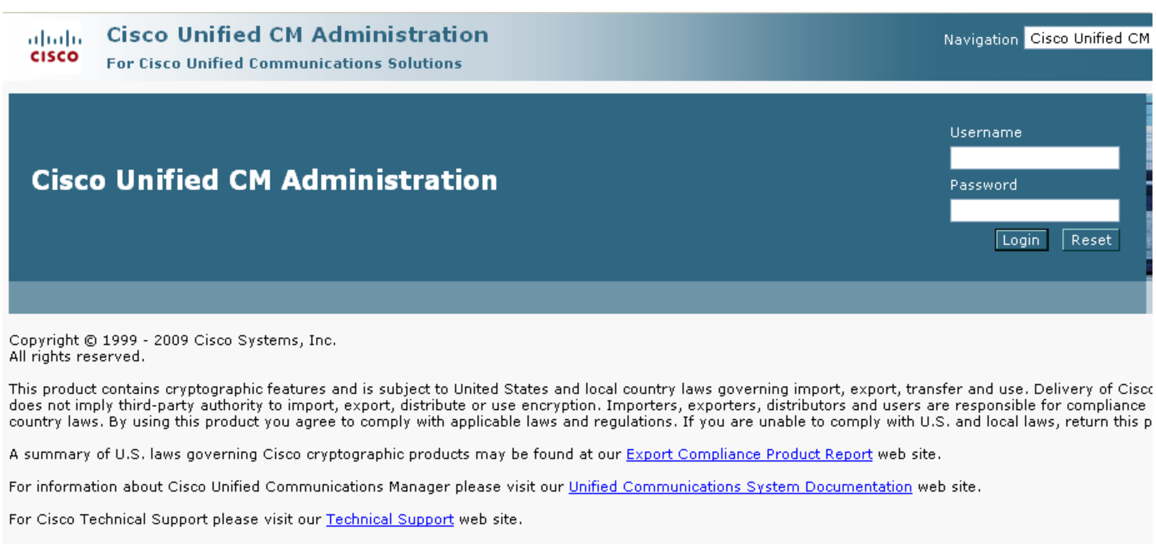
DEVICE DESCRIPTION	VERSION TESTED
Avaya IP Office 500v2	6.1(5)
Avaya IP Office Manager	8.1(5)
Avaya 1608/1616L IP Telephone (H323)	1.22
Avaya 1408 Digital Telephone	n/a
Avaya 1140eSIP	4.0
Cisco Unified Communications Manager	8.0.3.20000-2
Cisco 7960 Unified IP Phone (SIP)	P0S3-8-12-0
Cisco 7960 Unified IP Phone (SCCP)	8.1 (2.0)

5. Configure Cisco Unified CM

This section describes the SIP Trunk configuration for CUCM as shown in **Figure 1**. Fields left using default values are not highlighted. It is assumed that the basic configuration needed to support the VG248 gateway (needed for analog phone and fax support) and support for Cisco IP telephones has been completed. For further information on Cisco UCM, please consult **Section 10**: references [3]-[7].

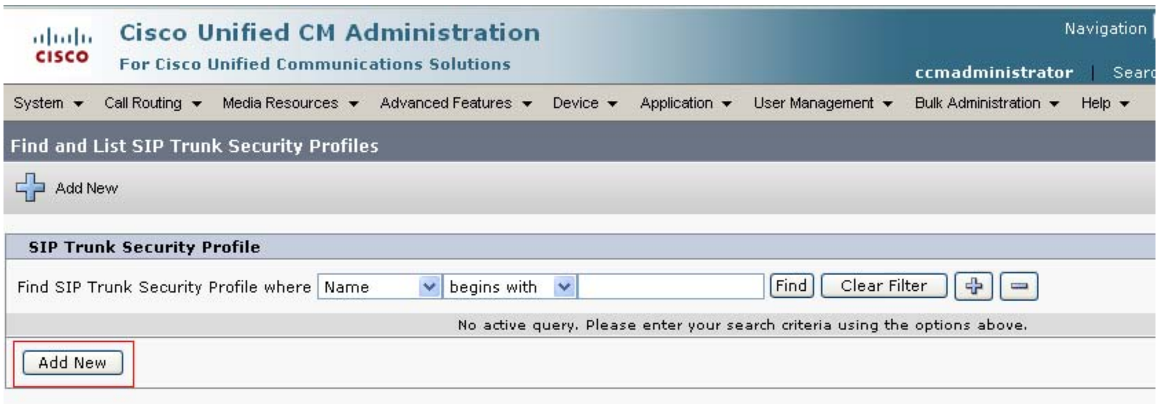
5.1. Login to Cisco Unified CM Administration

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.



5.2. Add a SIP Trunk Security Profile

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

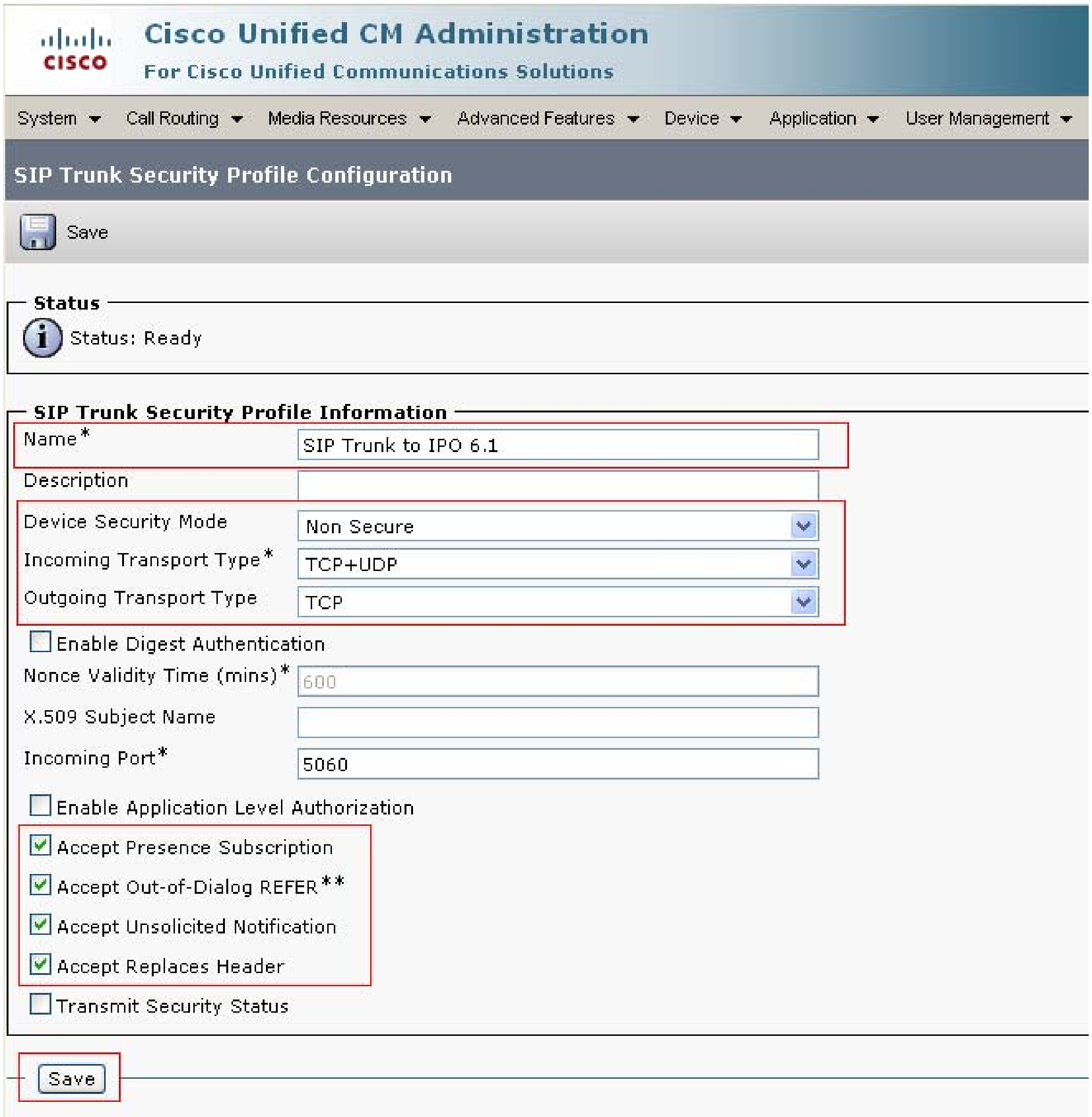


The following is a screen capture of the SIP Trunk Security Profile used in the sample network. The following values were used in the sample configuration:

- Name — A descriptive name for the profile
- Device Security Mode — "**Non Secure**" indicates unencrypted SIP signaling
- Incoming Transport Type — "**TCP+UDP**" indicates CUCM will listen for both protocols
- Outgoing Transport Type — "**TCP**" indicates CUCM will only use TCP to initiate SIP signaling
- Incoming Port — "**5060**". Typical value for UDP and TCP SIP Signaling

- Accept Presence Subscription — Enable

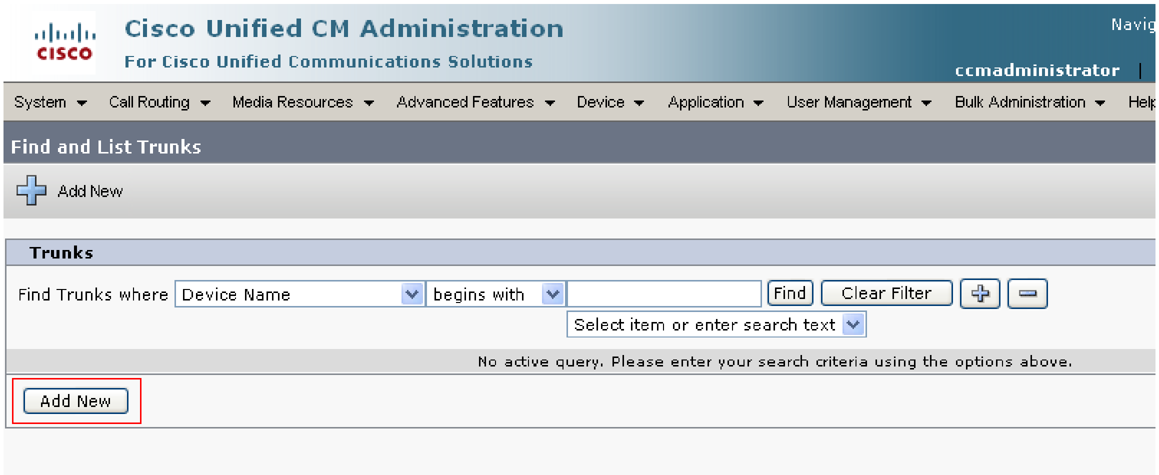
- Accept Out-of-Dialog REFER ** — Enable
- Accept Unsolicted Notification — Enable
- Accept Replaces Header — Enable



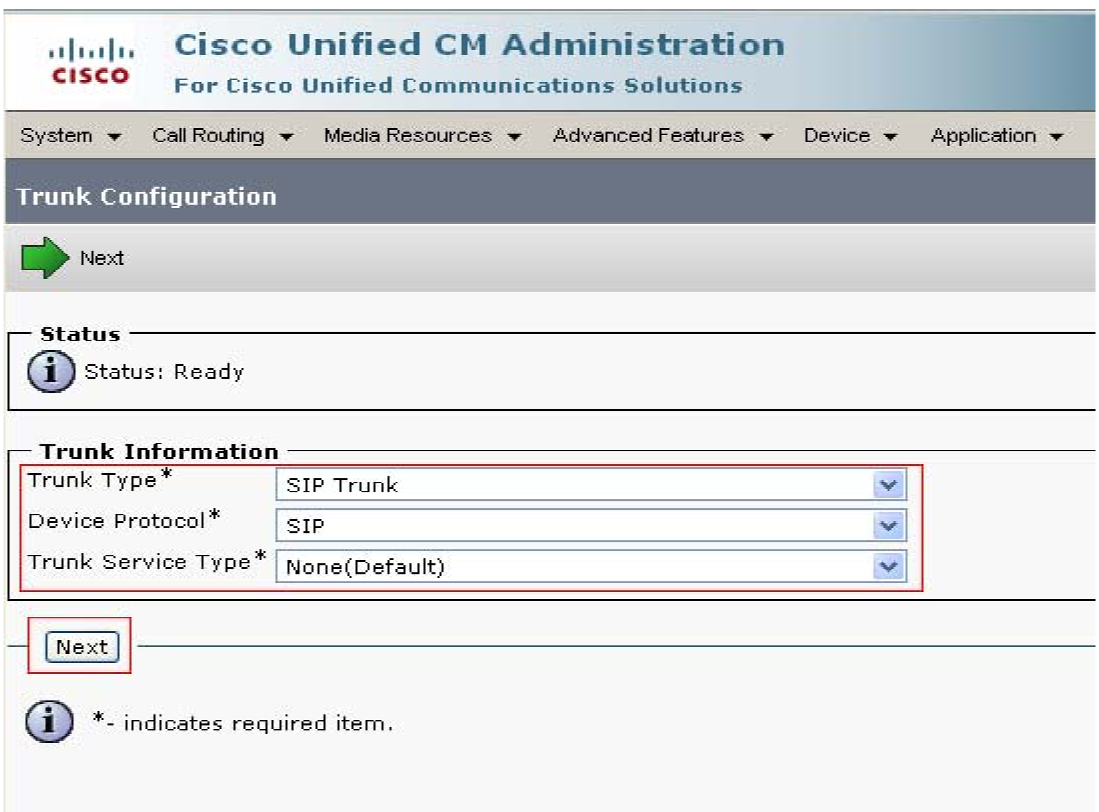
Click **Save** to commit the configuration.

5.3. Create a SIP Trunk

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

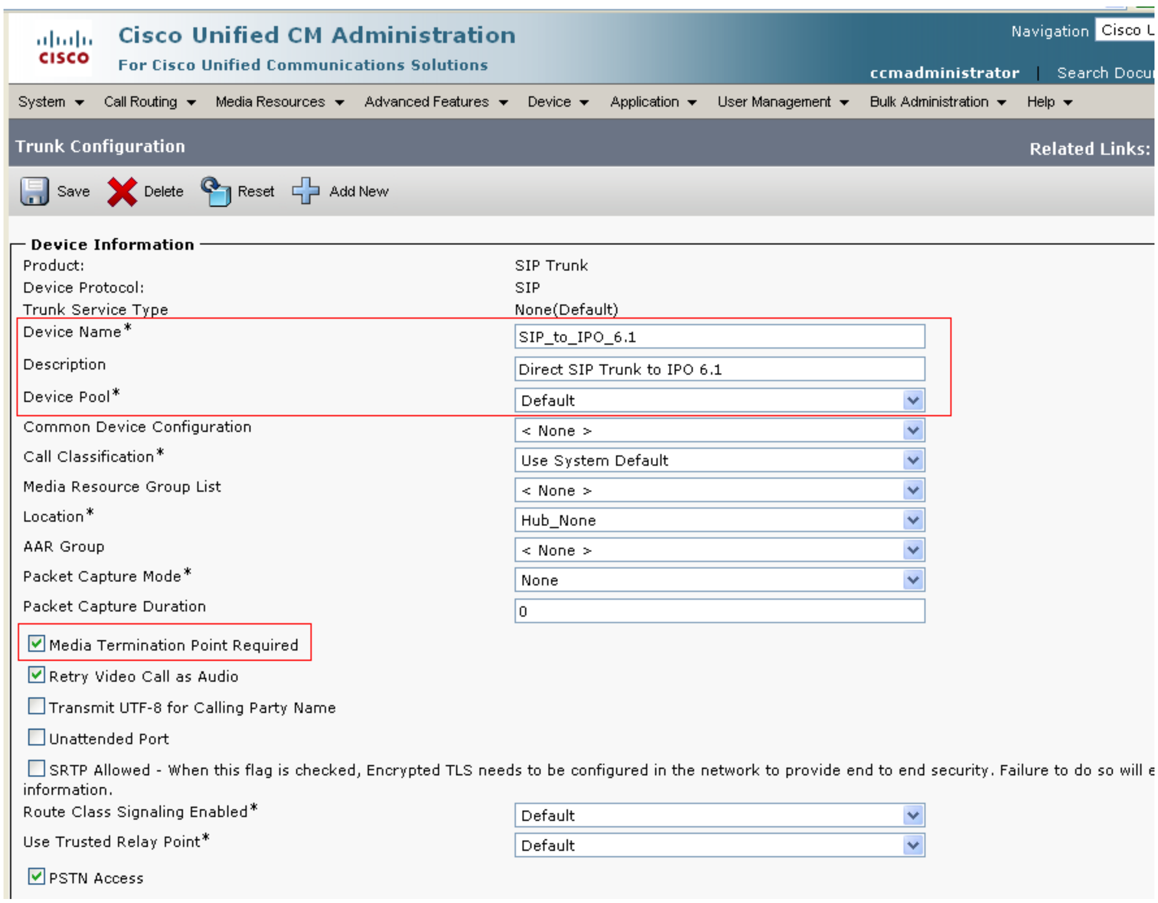


Select **SIP Trunk** as the **Trunk Type** and the **Device Protocol** field will automatically change to **SIP**. Click **Next** to continue.



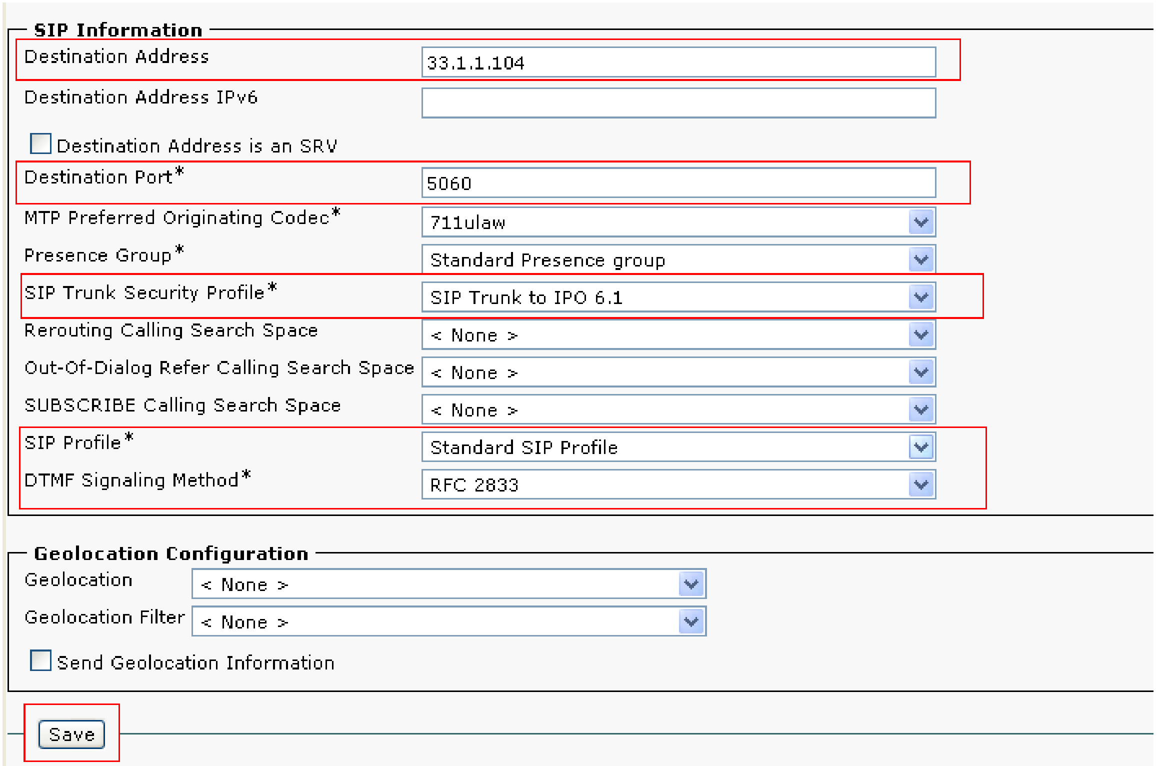
Enter the following information for the SIP Trunk.

- **Device Name** A descriptive name/identifier for the SIP Trunk. (Make sure there are no spaces in the device name).
- **Description** Additional descriptive information about the SIP Trunk
- **Device Pool** Select **Default**
- **Media Termination Point Required** This will cause CUCM to include SDP information in its initial SIP Invite message.



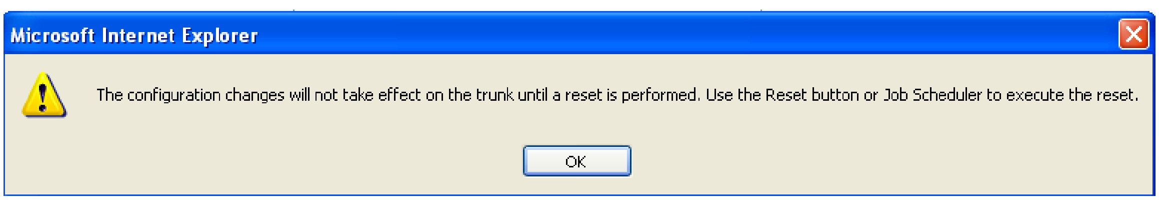
Scroll down to the section titled **SIP Information** and fill in the fields as indicated below.

- **Destination Address** IP Address of IP Office
- **Destination Port** Port 5060 is typically used for TCP and UDP SIP signaling
- **SIP Trunk Security Profile** Use the Security Profile defined in **Section 5.2**
- **DTMF Signaling Method** Select **RFC2833**.



Click **Save** to complete.

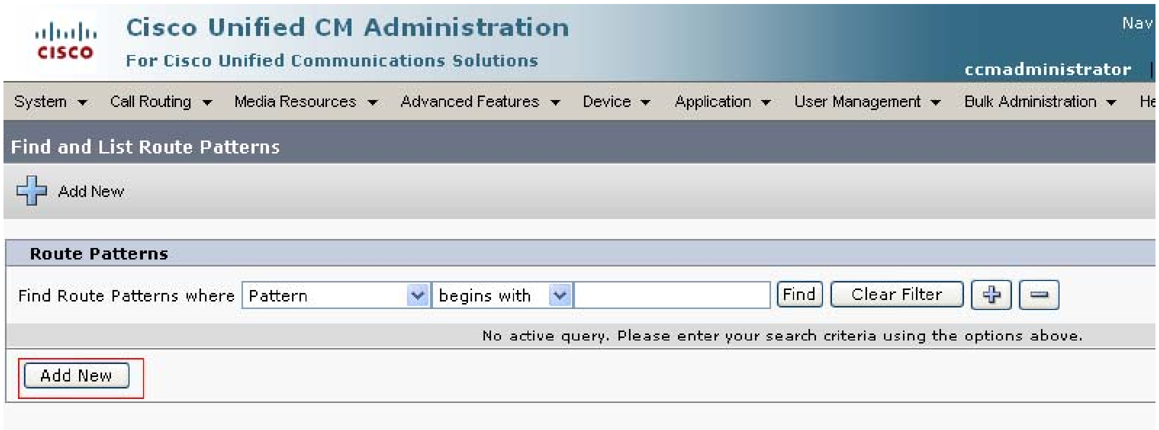
Following screen will appear and click **OK**.



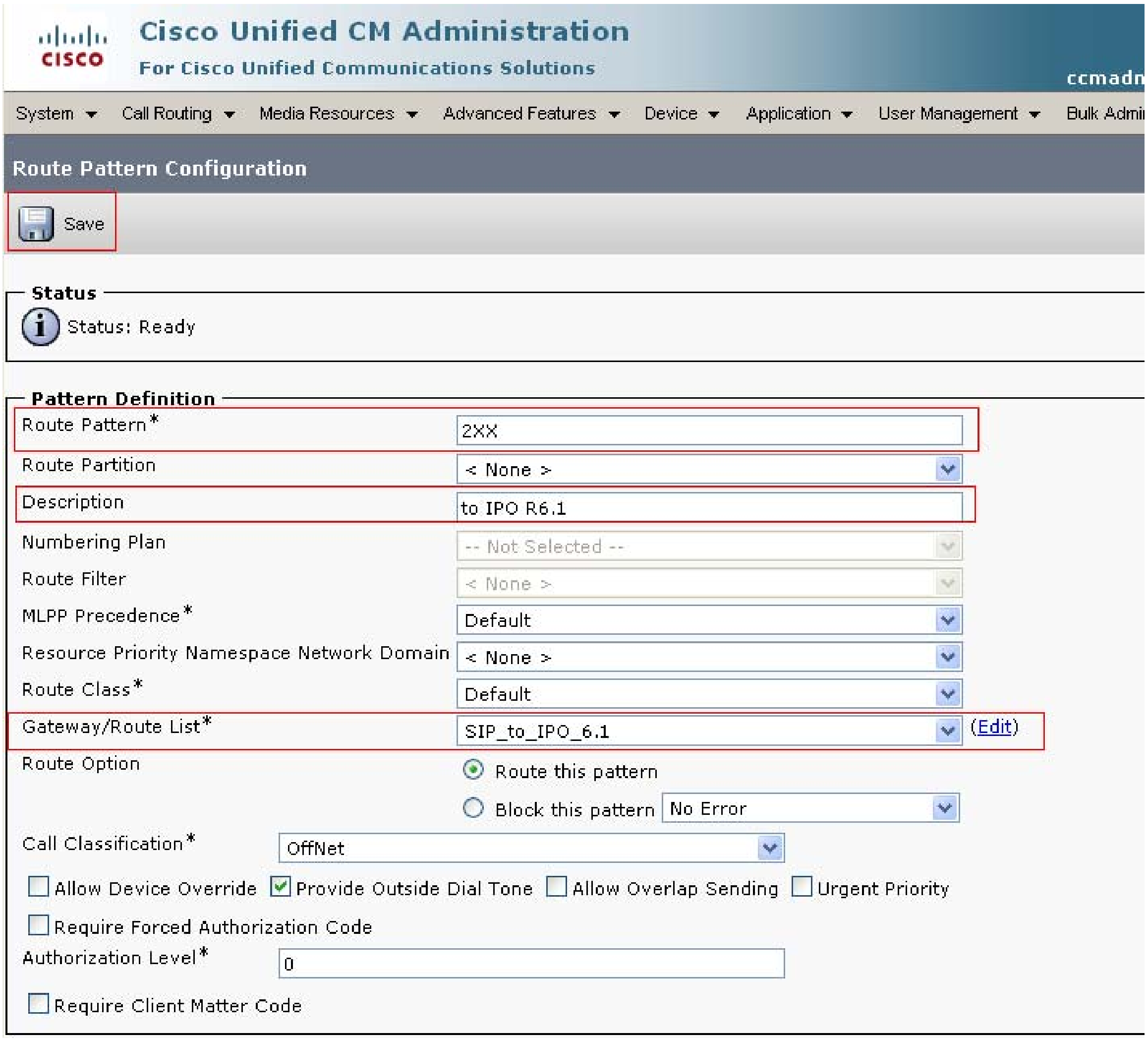
Follow the instructions from **Section 10**, Reference 5and perform a reset for the Cisco Call Manager.

Create a Route Pattern

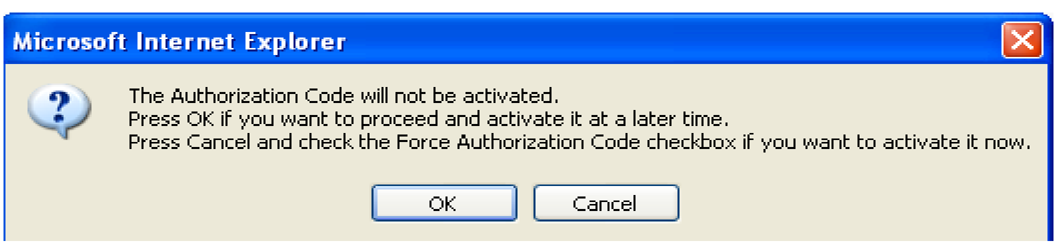
Select **Call Routing → Route/Hunt → Route Pattern** then click **Add New** to add a new route pattern for extension **2xx** which are for telephones registered with Avaya IP Office.



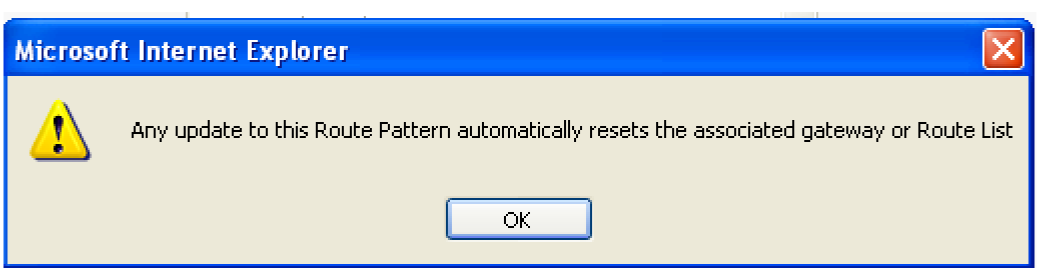
The following screen shows the route pattern used in the sample network. The route pattern **2xx** will cause all 3-digit calls beginning with “2” to be routed to the SIP Trunk defined in **Section 5.3**. Click **Save** to complete.



Following screen will appear and click **OK**.



Following screen will appear and click **OK**.

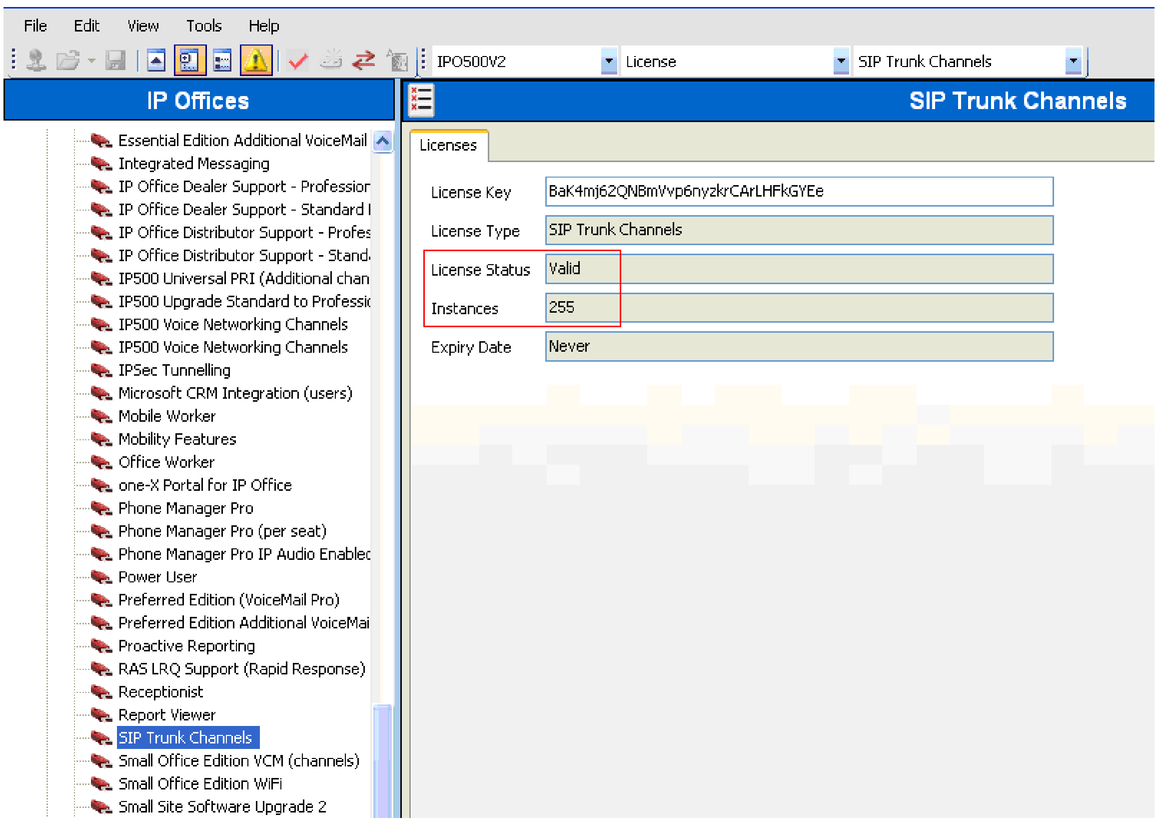


6. Configure Avaya IP Office

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 need to be configured are highlighted, all other fields are left with their default value. For further information on Avaya IP Office, please consult **Section 10**: Reference [1].

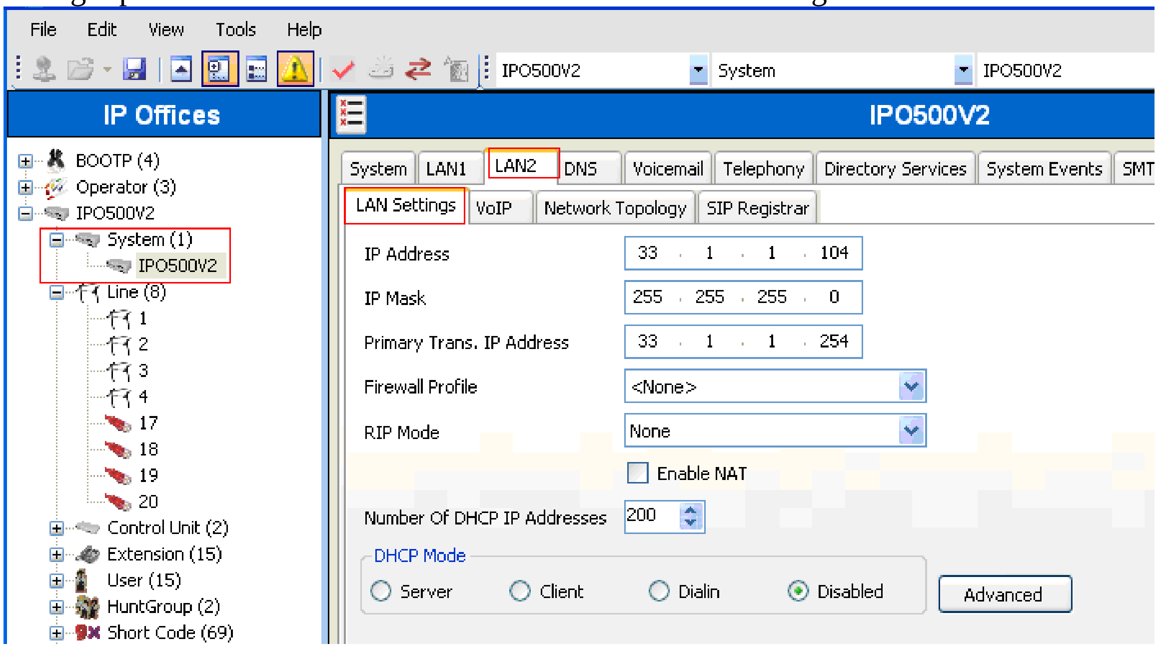
6.1. Verify SIP License

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



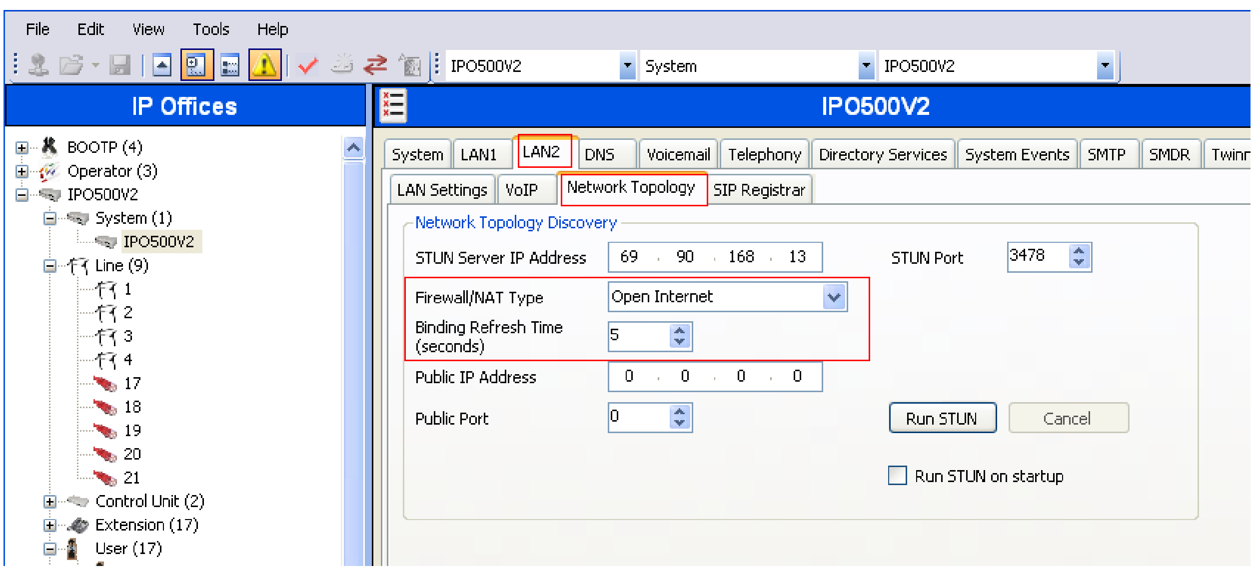
6.2. Obtain LAN IP Address

From the configuration tree in the left pane, select **System** to display the **IPO500V2** screen in the right pane. Select the **LAN2** tab, followed by the **LAN Settings** sub-tab in the right pane. This **IP Address** is used in **Section 5.3** to configure SIP Trunks.



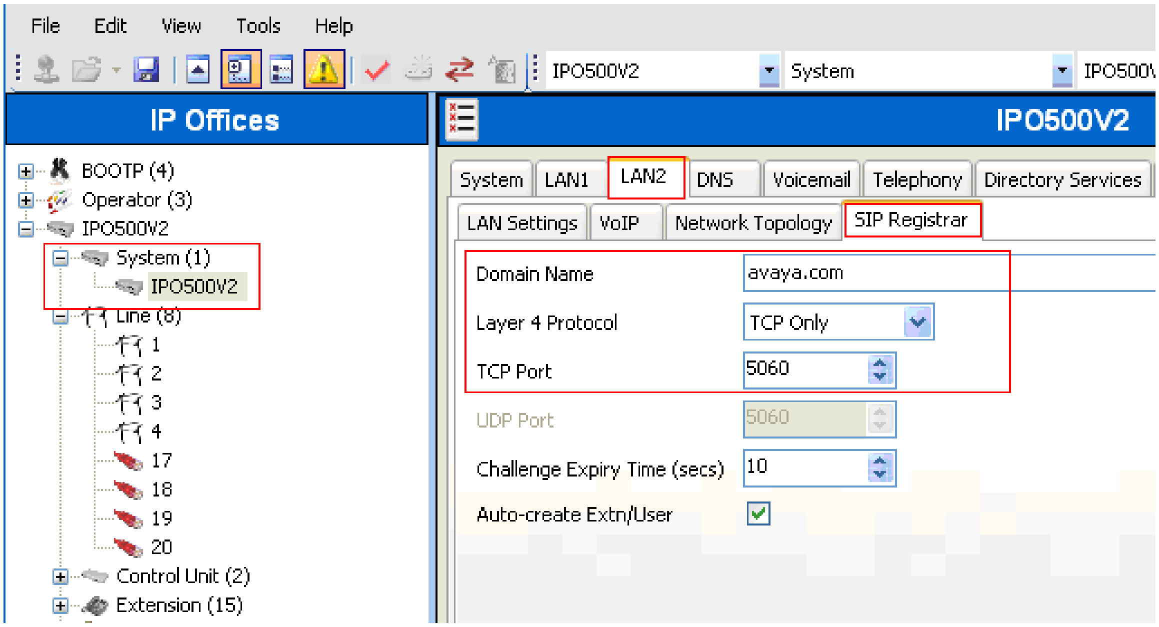
6.3. Configure Network Topology

From the configuration tree in the left pane, select **System** to display the **IPO500V2** screen in the right pane. Select the **LAN2** tab, followed by the **Network Topology** sub-tab in the right pane. Configure **Firewall/NAT Type** to "Open Internet". Configure **Binding Refresh Time** to "5". Click **OK**.



6.4. Administer SIP Registrar

Select **SIP Registrar** sub-tab in the right pane. Enter a valid **Domain Name**. Select **TCP only** from the drop down menu for **Layer 4 Protocol.** Make a note of the **TCP Port** number. These will be used later to configure SIP endpoints. Click **OK** (not shown).

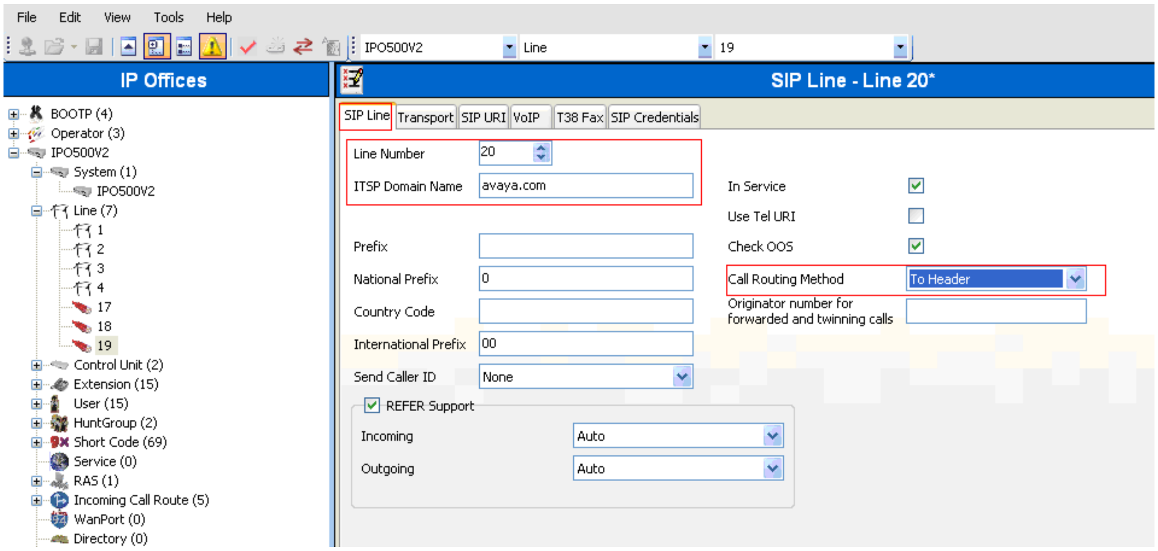


6.5. Create a SIP Line

Select **Line** from the left panel menu and then right-click and select **New → SIP Line** to create an SIP line to CUCM.

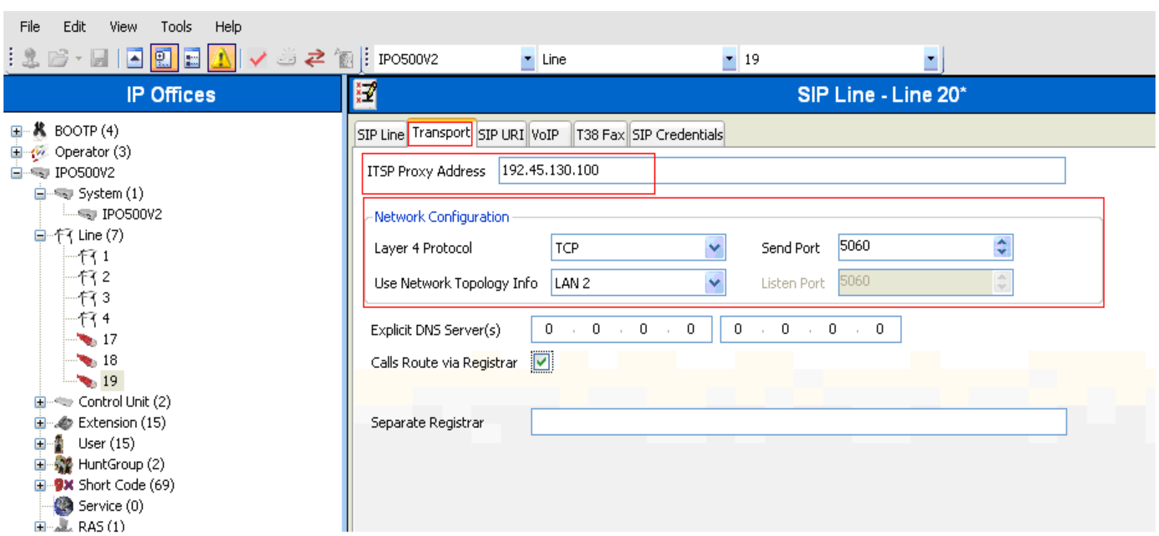
In the **SIP Line** tab, enter the following

- **ITSP Domain Name**: Enter the domain name from **Section 6.4**
- **Call Routing Method:** Select "To Header" from drop down menu



In the **Transport** tab, enter the following

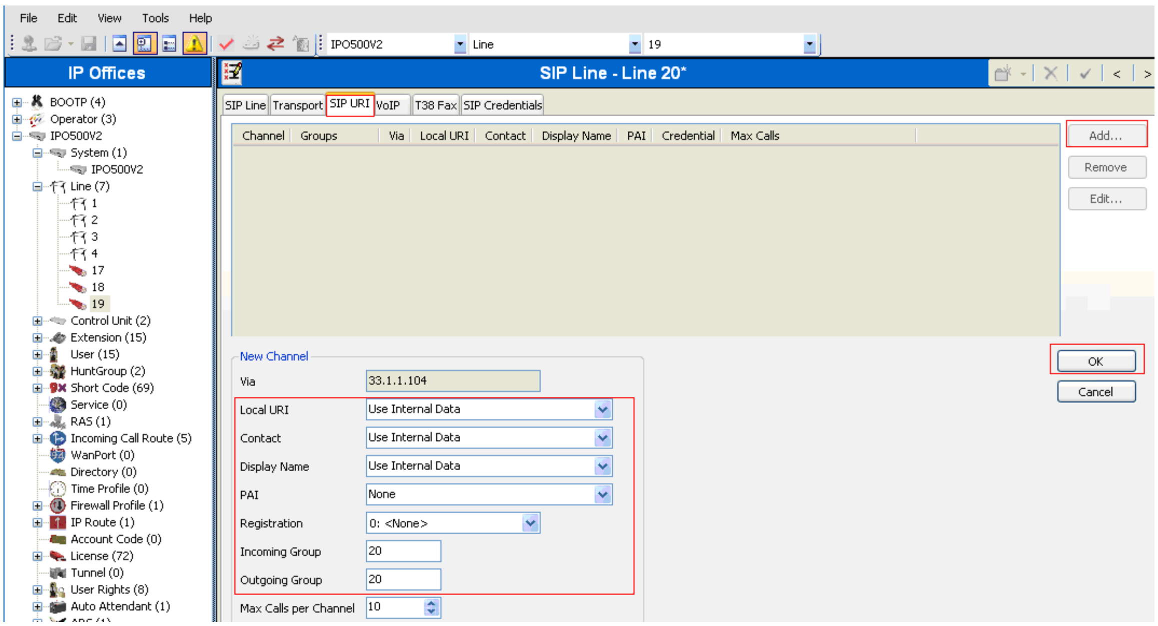
- **ITSP Proxy Address:** Enter the IP address of CUCM 8.0.
- **Layer 4 Protocol:** Select "TCP" from drop down menu
- **Send Port:** Select "5060" from drop down menu
- **Use Network Topology Info:** Select the LAN port from **Section 6.2**



In the **SIP URI** tab, select **Add** button and enter the following:

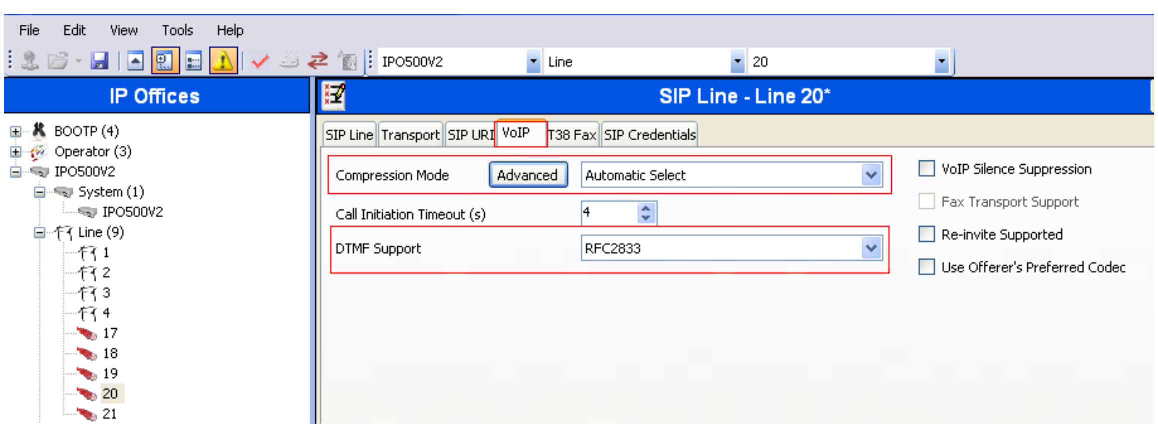
- **Local URI:** Select "Use Internal Data" from drop down menu
- **Contact:** Select "Use Internal Data" from drop down menu
- **Display Name:** Select "Use Internal Data" from drop down menu
- **Incoming Group:** Enter the line number created above
- **Outgoing Group:** Enter the line number created above

Select the **OK** button when done.



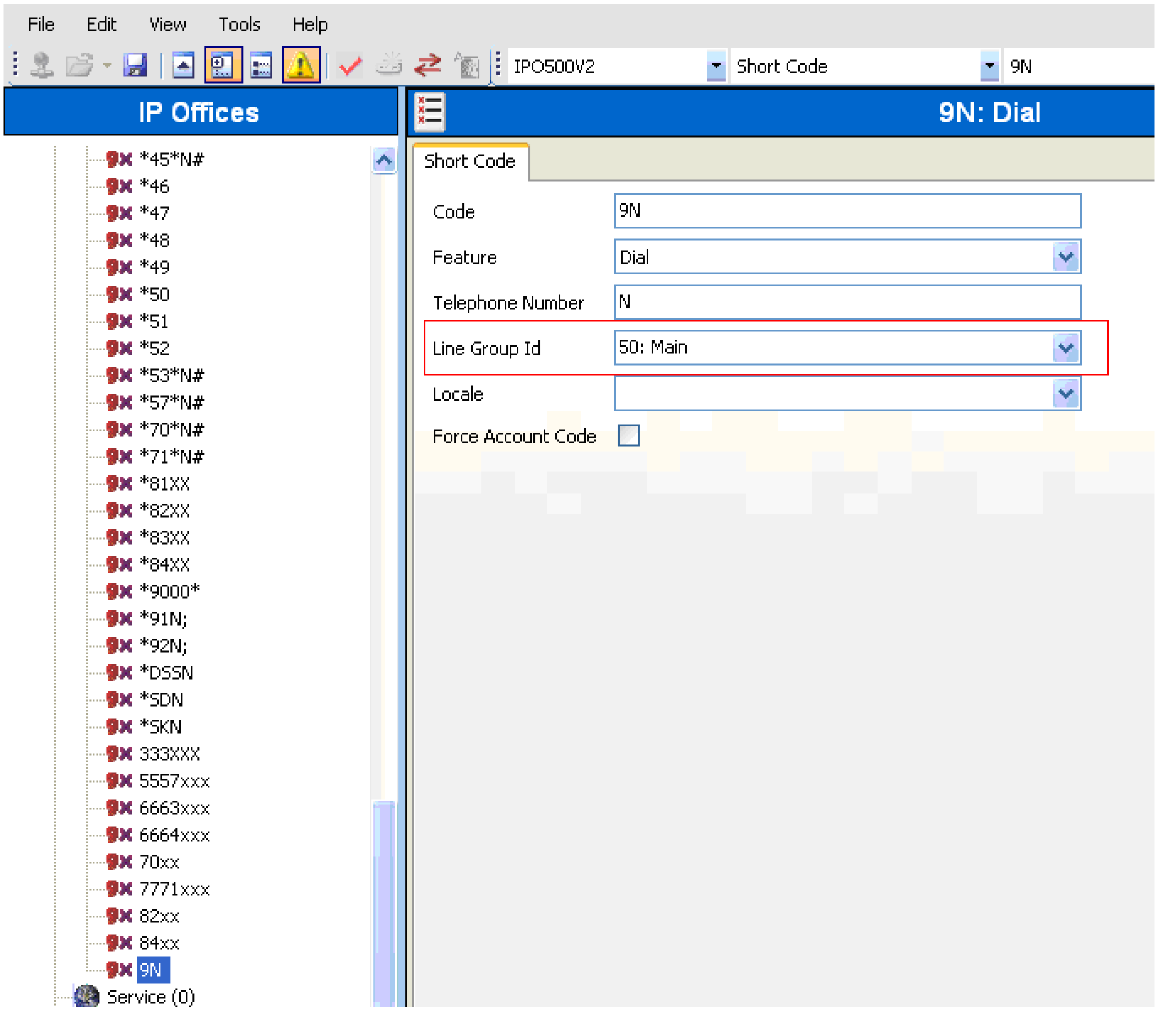
In the **VoIP** tab:

- Select **Automatic Select** for **Compression Mode**.
- **DTMF Support** should be set for **RFC2833**.
- Select the **OK** button (not shown) at the bottom of the screen once all changes have been made.

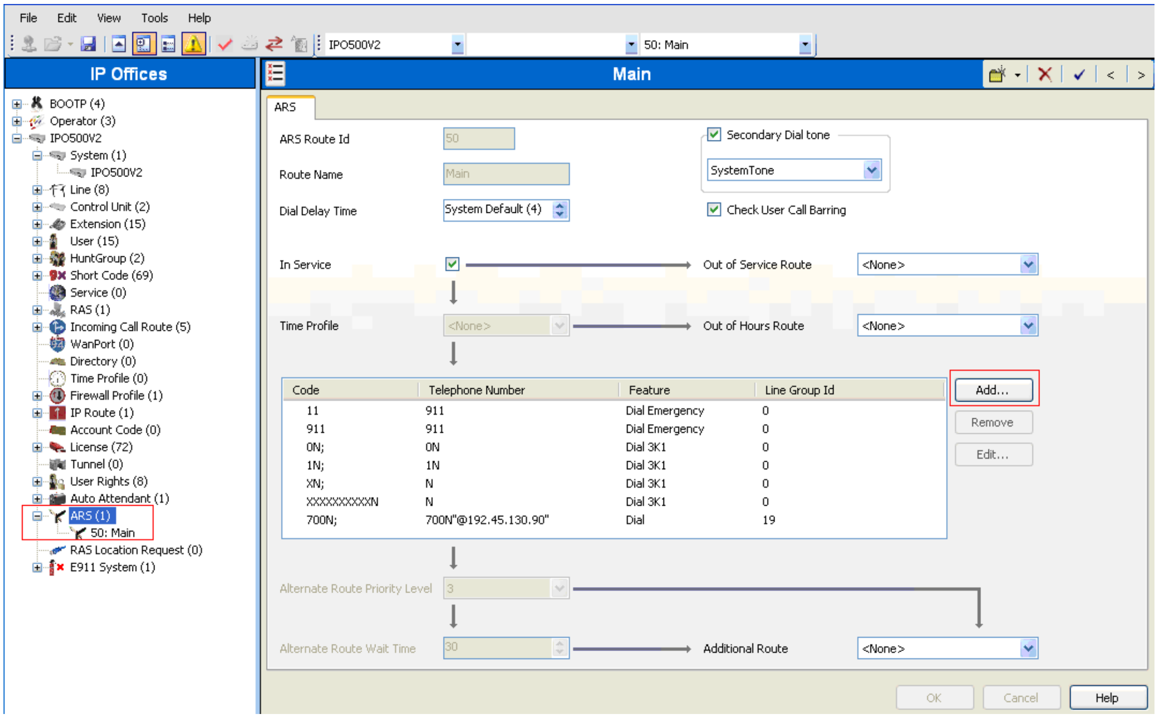


6.6. Create Outgoing Routing Entry for Calls to Cisco UCM

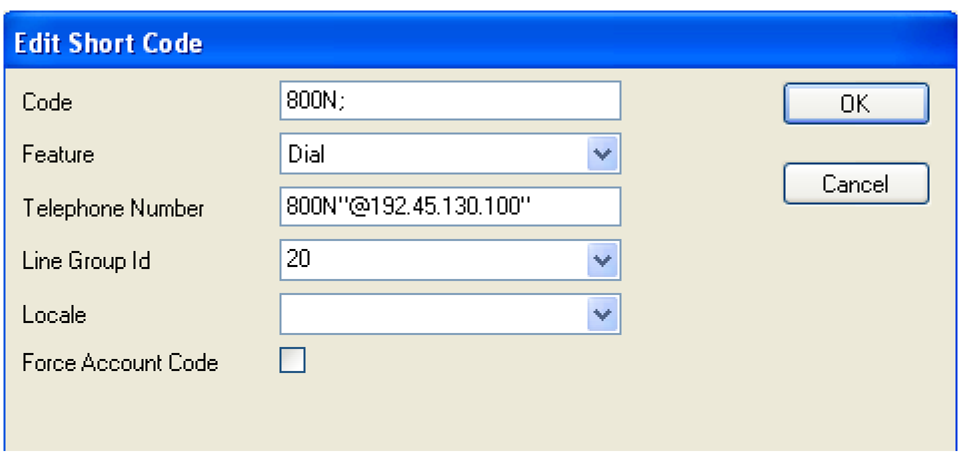
In the left pane, under **9x Short Codes,** by default there should be a short code for **9N** that routes calls to a default ARS group called **Main**. These Application Notes will use ARS to route call to CUCM. The screen capture below shows the default **9N** Short Code.



1. Select **ARS → Main** from the left panel menu, and then click on **Add** to create a new Code entry to route calls to CUCM. Note: 50:Main is the default Line Group Id for ARS.

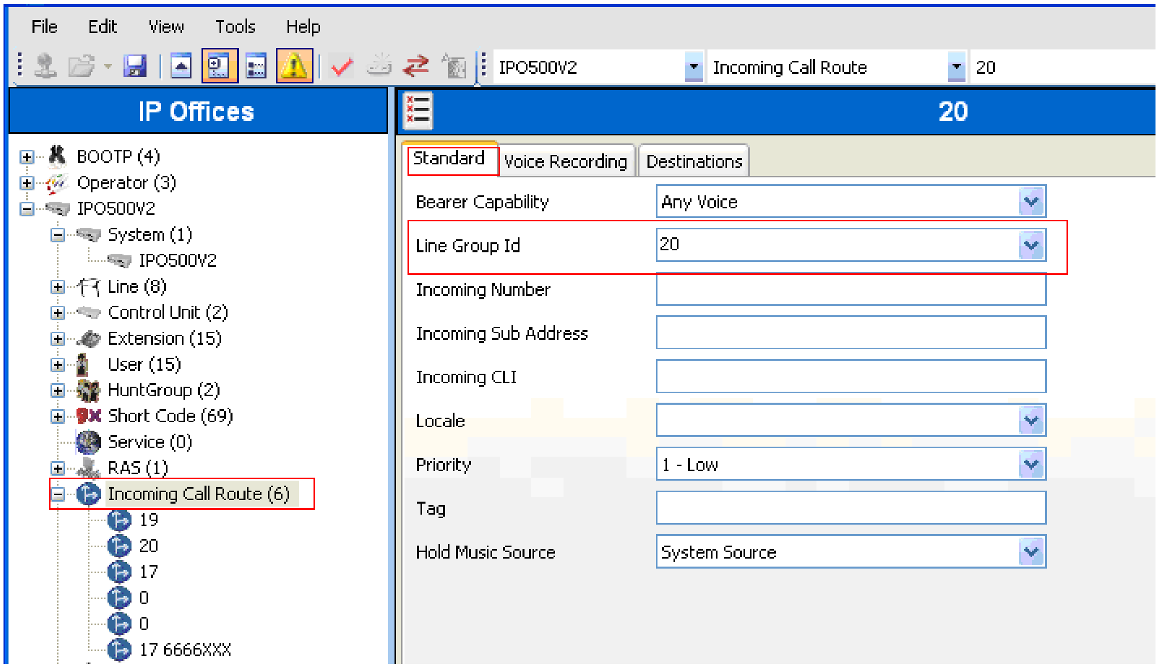


2. Enter the appropriate information for the Code entry. The following screen capture shows a portion of the Cisco dialing plan "800" is being used as part of the Code. The Telephone Number is composed of the called phone number appended with "@" and the CUCM IP Address. **Line Group ID** created in **Section 6.5** will be used to send out the call.

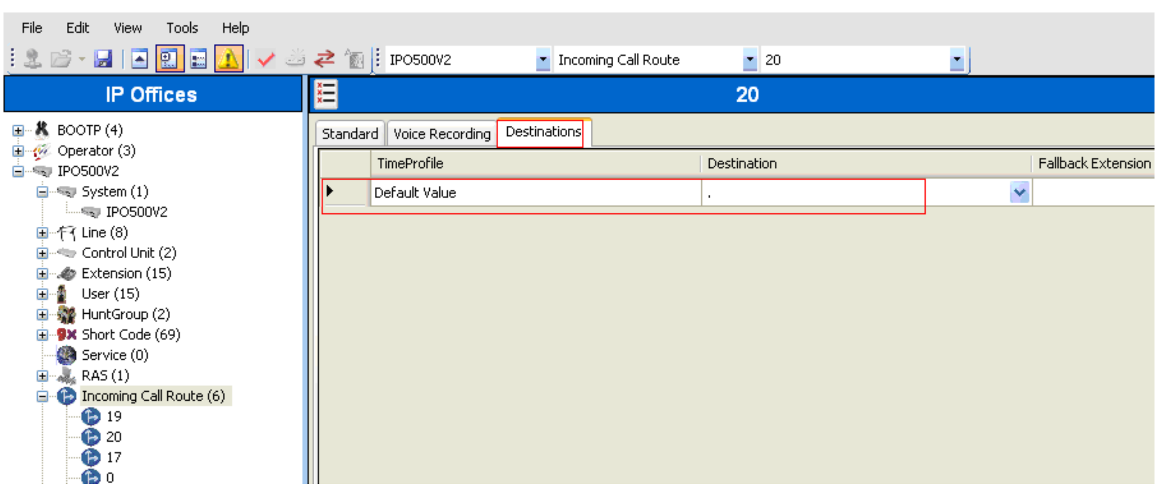


6.7. Create Incoming Routing Entry for Calls From Cisco UCM

1. Select **Incoming Call Route** from the left panel menu and then right-click it and select **New** (not shown) to create a new Incoming Call Route. Under the **Standard** tab, select the Line Group number created in **Section 6.5** in the **Line Group Id** field. The following screen shows the setting used in the sample network.



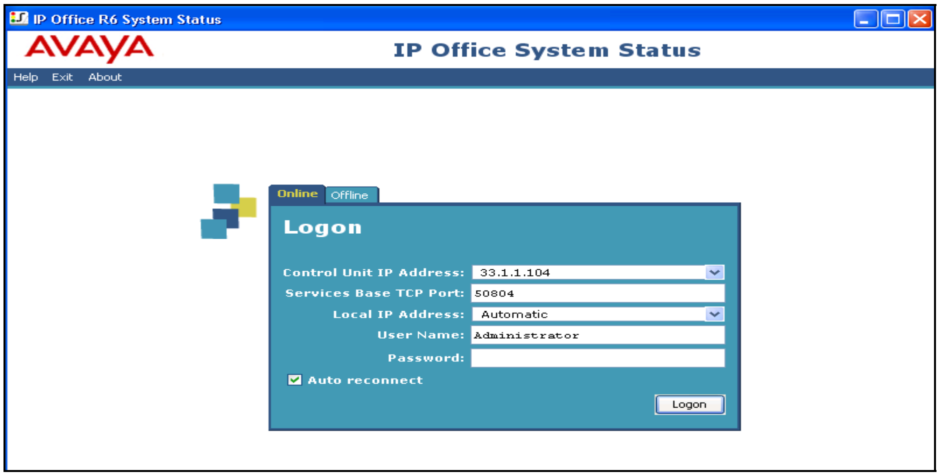
2. Under the **Destination** tab, enter "**.**" as the **Default Value**. The "**.**" indicates the incoming call can be routed to the extension specified by the caller. The following screen shows the setting used. Select the **OK** button when complete.



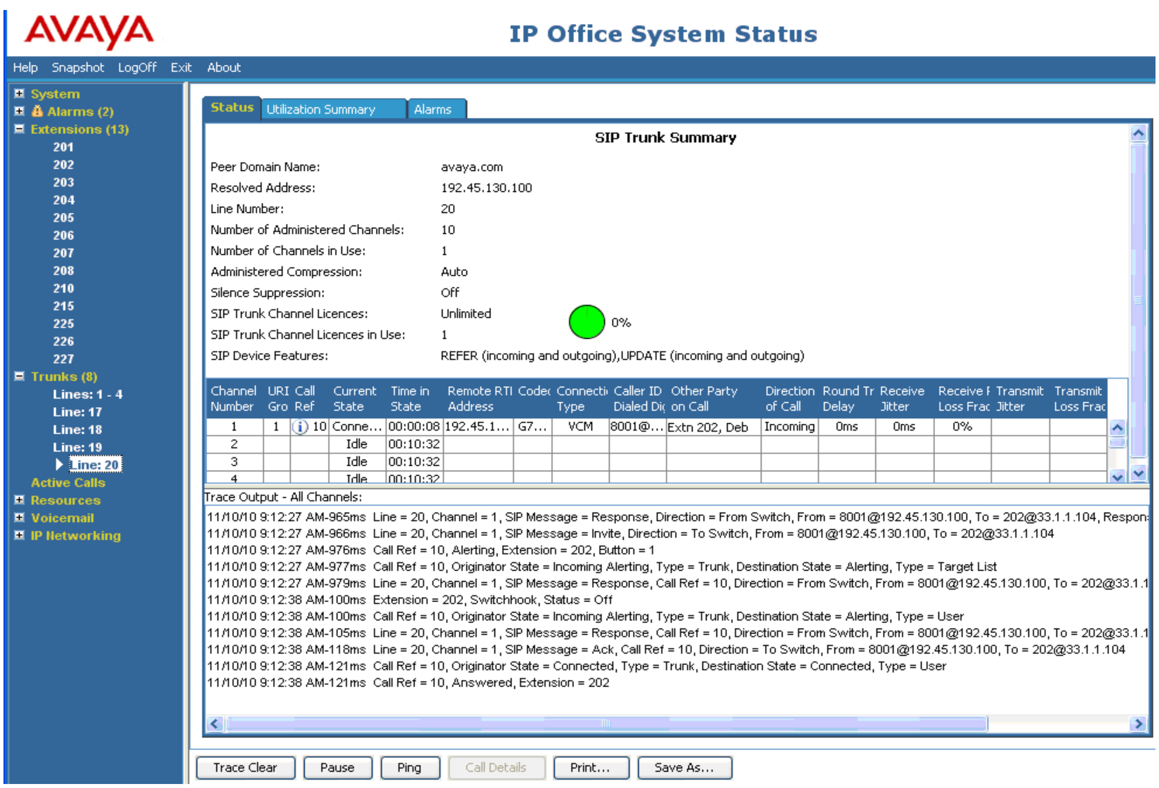
7. Verification

The following steps may be used to verify the configuration:

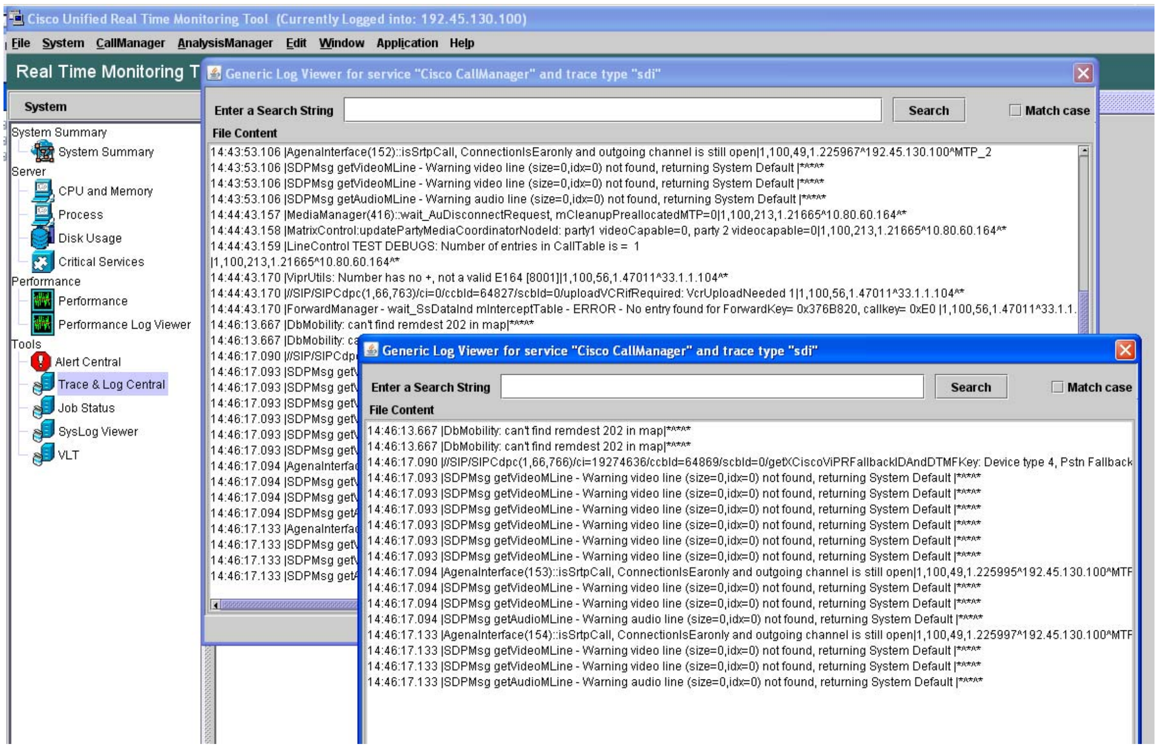
1. Call and trunk status (among other things) can be monitored using **IP Office System Status**. From IP Office Manager select the **File** menu → **Advanced** → **System Status**. Log in with appropriate credentials.



Once logged in, in the left-pane expand **Trunks** and select the appropriate SIP Trunk. In the sample configuration this is **Line 20**. The screen below shows 1 active call and several idle channels on Line 20.



2. The Cisco **Real Time Monitoring Tool** (RTMT) can be use to monitor events on Cisco UCM. 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 Communcations Manager Real Time Monitoring Tool showing a call being traced in real time. For further information on this tool, please consult with reference **Section 10**: reference [7].



8. Features Tested

Basic calling features are supported including Hold, Transfer, Conference and Fax Pass-through. Supplemental features such as Call Forward All, Call Park/Unpark are also supported by this configuration.

8.1. Known Limitations

During interoperability testing, several functional limitations were observed:

1. G.729 Codec is not supported with this solution.
2. The version of IP Office shown in these Application Notes only supports an initial SIP Invite message that contains 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 the **Media Terminal Point Required** option as shown in **Section 5.3.**
3. A number of telephone display anomalies were observed while testing call-transfer and call-forwarding scenarios. In several test scnearios it was observed

that phones on both Cisco UCM and IP Office would not update their display with the 'connected to' name and/or number.

9. 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.

10. Additional References

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

[1] *Avaya IP Office Release 6.1 Manager 8.0*

[2] *Avaya IP Office 6.1: IP Office Installation*

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

[3] *Cisco Unified IP Phones 7960G/7940G Administration Guide for Cisco Unified Communications Manager 7.0 (SCCP)*, Part Number: OL-15498-01

[4] *Cisco Unified IP Phones 7960G/7940G Administration Guide for Cisco Unified Communications Manager 7.0 (SIP)*, Part Number: OL-15499-01

[5] *Cisco Unified Communications Manager Administration Guide 7.1(2),* Release 7.1(2), Part Number: OL-18611-01

[6] *Cisco Unified Communications Manager Features and Services Guide*, Release 7.1(2), Part Number: OL-18610-01

[7] *Cisco Unified Real-Time Monitoring Tool Administration Guide*, Release 7.1(2), Part Number: OL-18620-01

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