Avaya Solution & Interoperability Test Lab

Sample Configuration for SIP Trunking between Avaya IP Office R8.0 and Cisco Unified Communications Manager 8.6.2 – Issue 1.0

Abstract

These Application Notes describe the steps for configuring a SIP trunk between Avaya IP Office R8.0 and Cisco Unified Communications Manager (CUCM) Release 8.6.2.
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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. Interoperability Testing

2.1. 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.

2.2. 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 6.3.
3. A number of telephone display anomalies were observed while testing call-transfer and call-forwarding scenarios. In several test scenarios, it was observed that phones on both CUCM and IP Office would not update their display with the ‘connected to’ name and/or number.
4. IP Office incorrectly fills in the refer-to header with the user dialed number when REFER is enabled. The following problems have been observed if ARS short code is dialed to transfer calls:
   a. Avaya SIP phone displays “Transfer Failed” after attended transfer to a CUCM endpoint. However, the transferred-to phone was successfully connected and has two-way talk path.
   b. The CUCM transferee will remain connected after transferred-to CUCM phone hangs up the call. This occurred after successfully transfers an H323 endpoint to another CUCM endpoint.

   Note: To work around the transfer failure, user can create a short code to dial the CUCM extension number as shown in Section 7.6.
5. CUCM sends “500 Internal Server error” to IPO after blind transfers an H323 endpoint to another CUCM endpoint. However, the call transfer is successful. This problem is under investigation.
6. IP Office SIP phones do not block the calling number when receive an anonymous call.
7. IP Office fails to send T38 fax to CUCM. CUCM rejects the T38 fax after receives G.711 PCMU in the 200 OK message from IP Office.

3. 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 1100 Series SIP Telephones, 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.

4. Configuration
Figure 1 illustrates the configuration used in these Application Notes. All IP telephones in the 192.45.2.0/24 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 720-567-8xxx. A single SIP trunk was configured and connected between Avaya IP Office and CUCM. All inter-system calls are carried over this SIP trunk.

![Figure 1: Sample Network Configuration](image-url)
5. Equipment and Software Validated

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

<table>
<thead>
<tr>
<th>DEVICE DESCRIPTION</th>
<th>VERSION TESTED</th>
</tr>
</thead>
<tbody>
<tr>
<td>Avaya IP Office 500v2</td>
<td>8.0(16)</td>
</tr>
<tr>
<td>Avaya IP Office Manager</td>
<td>10.0(16)</td>
</tr>
<tr>
<td>Avaya 1618 IP Telephone (H323)</td>
<td>1.300B</td>
</tr>
<tr>
<td>Avaya 9630G IP Telephone (H323)</td>
<td>3.186a</td>
</tr>
<tr>
<td>Avaya 1408 Digital Telephone</td>
<td>n/a</td>
</tr>
<tr>
<td>Avaya 1140eSIP</td>
<td>04.03.09.00</td>
</tr>
<tr>
<td>Avaya 1230eSIP</td>
<td>04.03.09.00</td>
</tr>
<tr>
<td>Cisco Unified Communications Manager</td>
<td>8.6.2.20000-2</td>
</tr>
<tr>
<td>Cisco 7975 Unified IP Phone (SIP)</td>
<td>75.9-2-1S</td>
</tr>
<tr>
<td>Cisco 7965 Unified IP Phone (SCCP)</td>
<td>45.9-2-1S</td>
</tr>
</tbody>
</table>

6. 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 CUCM, please consult Section 10, References [3]-[6].

6.1. 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.
6.2. 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 Unsolicited Notification**: Enable
- **Accept Replaces Header**: Enable
Click **Save** to commit the configuration.
6.3. 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 6.2

- **DTMF Signaling Method**  
  Select RFC2833.

Click **Save** to complete.

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

Follow the instructions from Section 10, Reference 5 and perform a reset for the Cisco Call Manager.
6.4. 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 6.3. Click Save to complete.
Following screen will appear and click **OK**.

![Message from webpage](image1.png)

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

![Message from webpage](image2.png)

### 7. 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]**.

#### 7.1. 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.
7.2. Obtain LAN2 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 6.3** to configure SIP Trunks.

Note: The **LAN1** IP Address is used for the LAN port of the IP Office control unit. The **LAN1** interface was used to connect the Avaya IP Office to the enterprise site IP network. The LAN1 interface configuration is not directly relevant to the interface with the Cisco Call Manager, and therefore it is not described in these Application Notes.
7.3. 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 “321”. Click OK.

7.4. 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:** Select “To Header” from drop down menu
  Enter the domain name. “avaya.com” was used in the sample configuration.

- **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. “192.45.130.100” was used in the sample configuration. (Administrative screens is not shown)

- **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 7.2**

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 **System Default** for Codec Selection.
- Select **G.711** for Fax Transport Support.
- DTMF Support should be set for RFC2833.
- Check **Re-invite Supported**
- Select the **OK** button (not shown) at the bottom of the screen once all changes have been made.
7.5. Create Outgoing Routing Entry for Calls to Cisco UCM

In the left pane, under 9N 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 “720567” 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 7.4** will be used to send out the call.
7.6. Create Short Code using extension number

When REFER is enabled, dial ARS short code to transfer calls from IP Office to CUCM will fail. To work around the problem, create a short code using the CUCM extension numbers.

In the left pane, right click 9X Short Code and then select New (not shown) to create a new Code entry to route calls to CUCM. In the sample configuration 10-digit extension numbers (720-567-8xxx) were assigned to CUCM.

![Image of short code configuration](image)

7.7. 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 7.4 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.
8. 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.

![IP Office System Status](image.png)
Once logged in, in the left-pane expand **Trunks** and select the appropriate SIP Trunk. In the sample configuration this is **Line 17**. The screen below shows 2 active calls and several idle channels on Line 17.
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 Cisco **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 Cisco Unified Communications 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 [6].

![Real Time Monitoring Tool Screen Capture](image)

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](http://support.avaya.com)

[1] *Avaya IP Office Release 8.0 Manager 10.0, Document Number 156010011*
[2] *Avaya IP Office 8.0: IP Office Installation, Document Number 156010042*

Product documentation for Cisco Systems products may be found at [http://www.cisco.com](http://www.cisco.com)
[3] Cisco Unified Communications Manager Documentation Guide for Release 8.6(2)
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