Avaya Solution & Interoperability Test Lab

Sample Configuration for a SIP Trunk between Avaya SIP Enablement Services Server and Cisco Unified CM 7.0 – Issue 1.0

Abstract

These Application Notes describe the steps for configuring a SIP trunk between Avaya SIP Enablement Services (SES) Server and Cisco Unified Communications Manager (CUCM).

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 to support calling between an Avaya Communication Manager and a Cisco IP PBX.

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 system consists of Avaya Communication Manager and Avaya SIP Enablement Services (SES) Server supporting a variety of Avaya 4600 and 9600 Series IP Telephones with either H.323 or SIP protocol along with digital and analog fax stations. The Cisco IP PBX system consists of Cisco Unified Communications Manager (CUCM) supporting the Cisco SIP and SCCP stations along with analog fax stations through the use of a Cisco 1751 router/gateway. A SIP trunk is configured between Avaya SES and CUCM to support calling for telephones 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 regardless in this sample network 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 Communication Manager or Avaya SES and use extension 11xxx. 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.

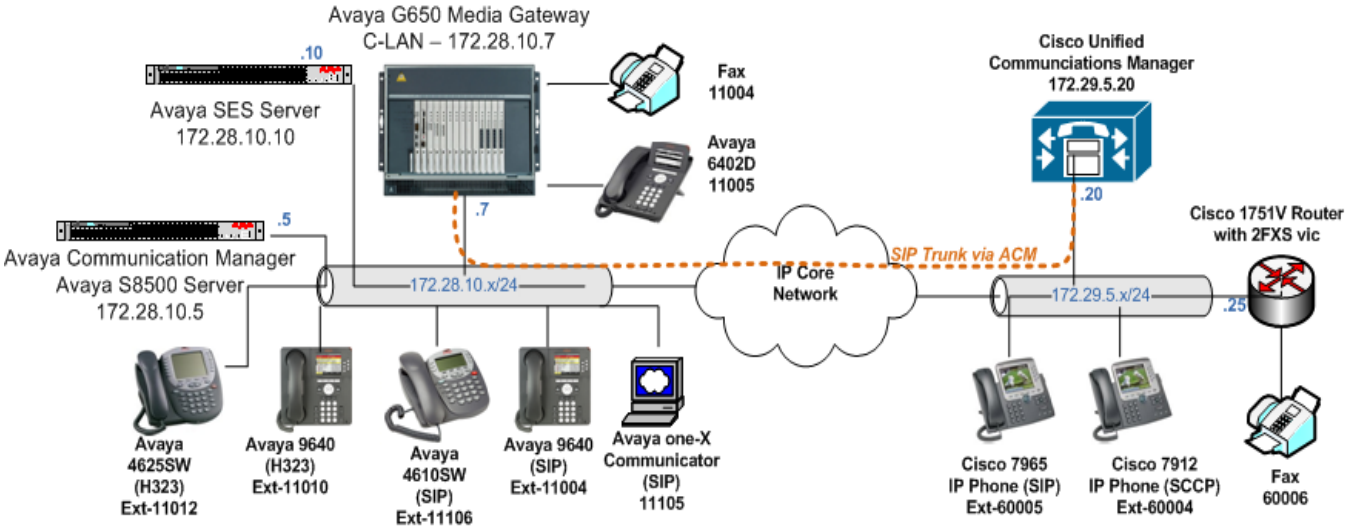


Figure 1: Sample Network Configuration

4. Equipment and Software Validated

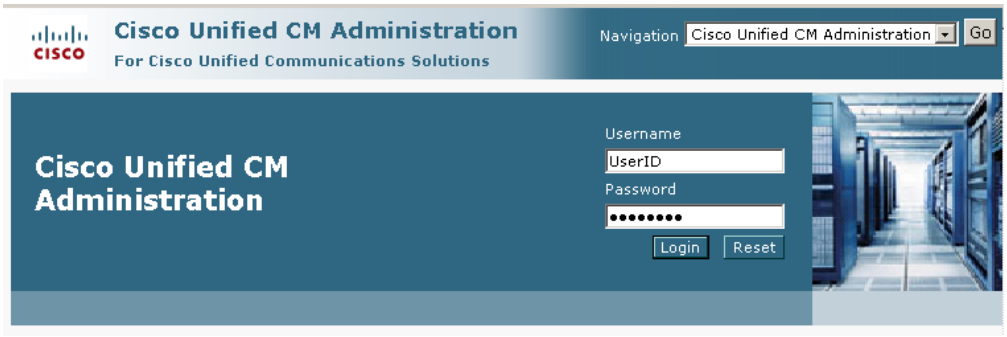
The following equipment and software/firmware were used for the sample configuration:

DEVICE DESCRIPTION	VERSION TESTED
Avaya S8500 Server with G650 Media Gateway	5.1.2
Avaya SIP Enablement Services (SES) Server	5.1.2
Avaya 4625SW IP Telephone (H.323)	2.9
Avaya 9640 IP Telephone (H.323)	2.0
Avaya 4610SW IP Telephone (SIP)	2.2.2
Avaya 9640 IP Telephone (SIP)	2.2.0
Avaya 6402D Digital Telephone	-
Avaya One-X Communicator (SIP)	1.0
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)

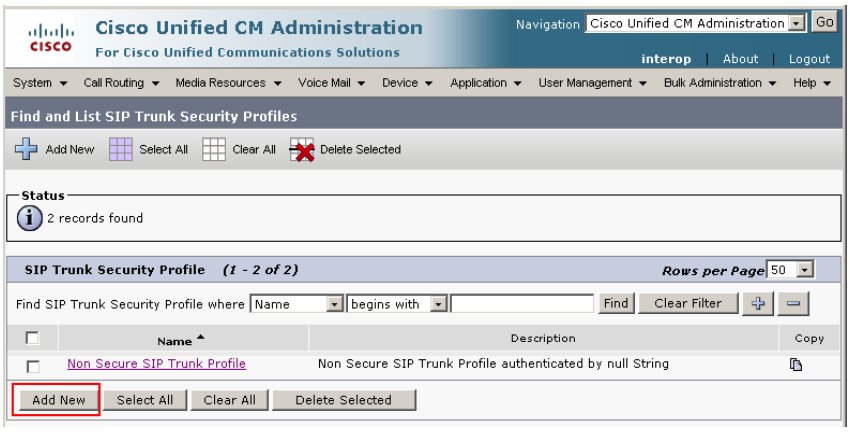
5. Configure Cisco Unified CM

This section describes the SIP Trunk configuration for CUCM as shown in **Figure 1**. 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 references [8] and [9].

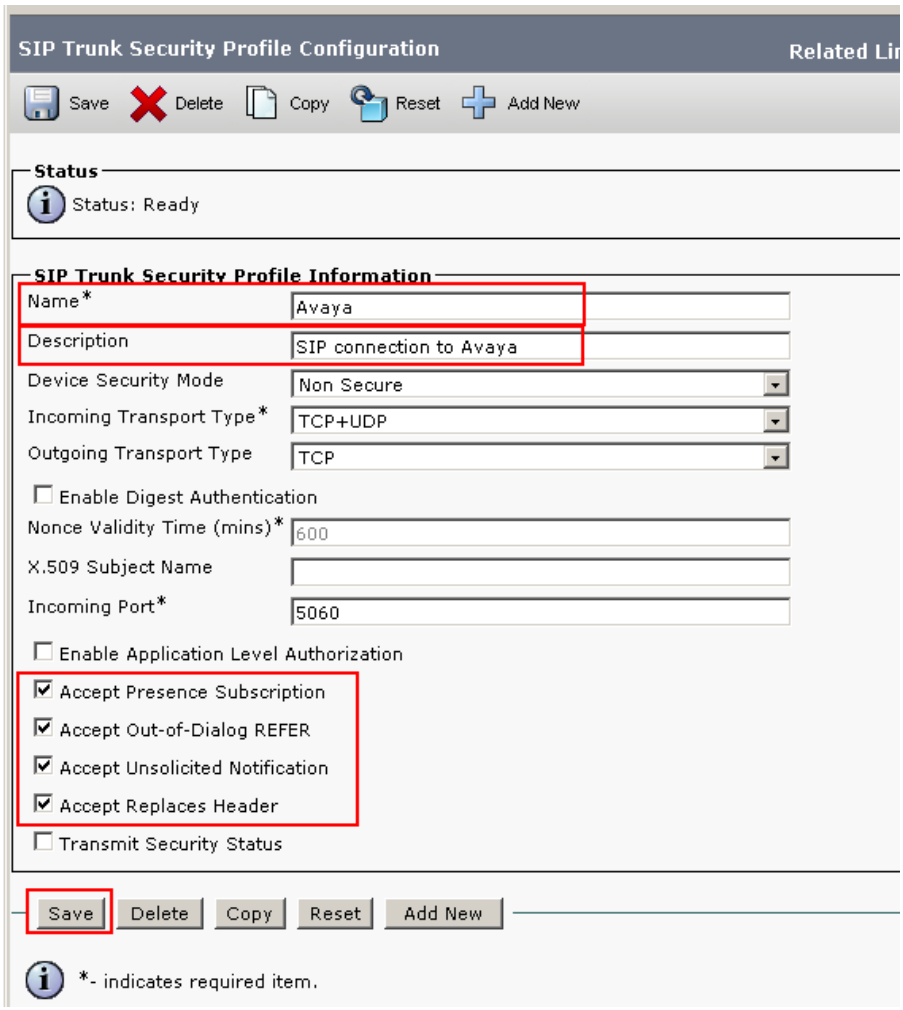
1. Open Cisco Unified CM Administration by entering the IP address of the CUCM into the Web Browser address field, and log in using an appropriate Username and Password.



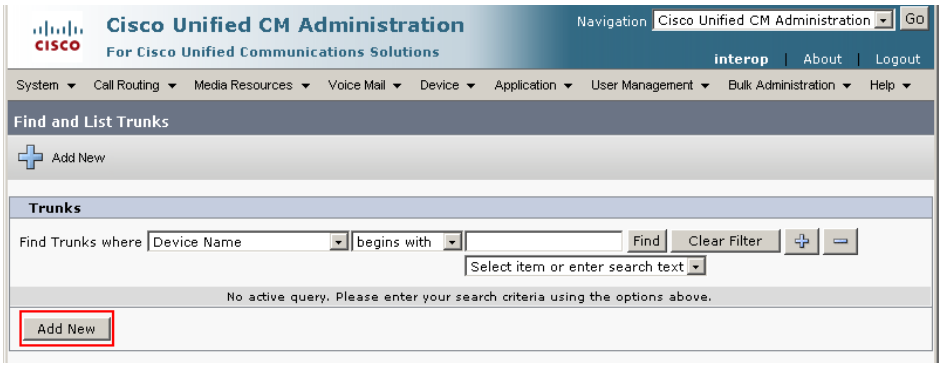
2. Select **System → Security Profile → SIP Trunk Security Profile** from the top menu then click **Add New** to add a new SIP Trunk Security Profile.



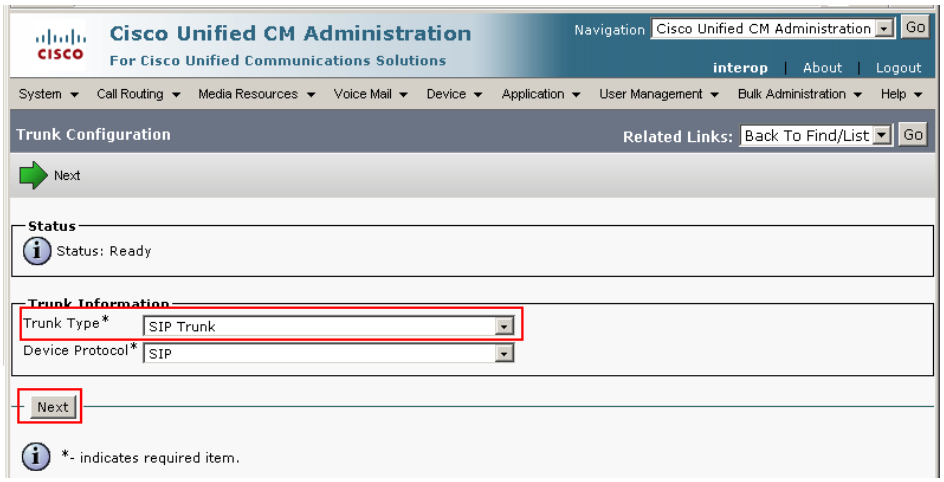
The following is a screen capture of the SIP Trunk Security Profile used in the sample network. Configure the highlighted areas and click **Save** to commit the changes.



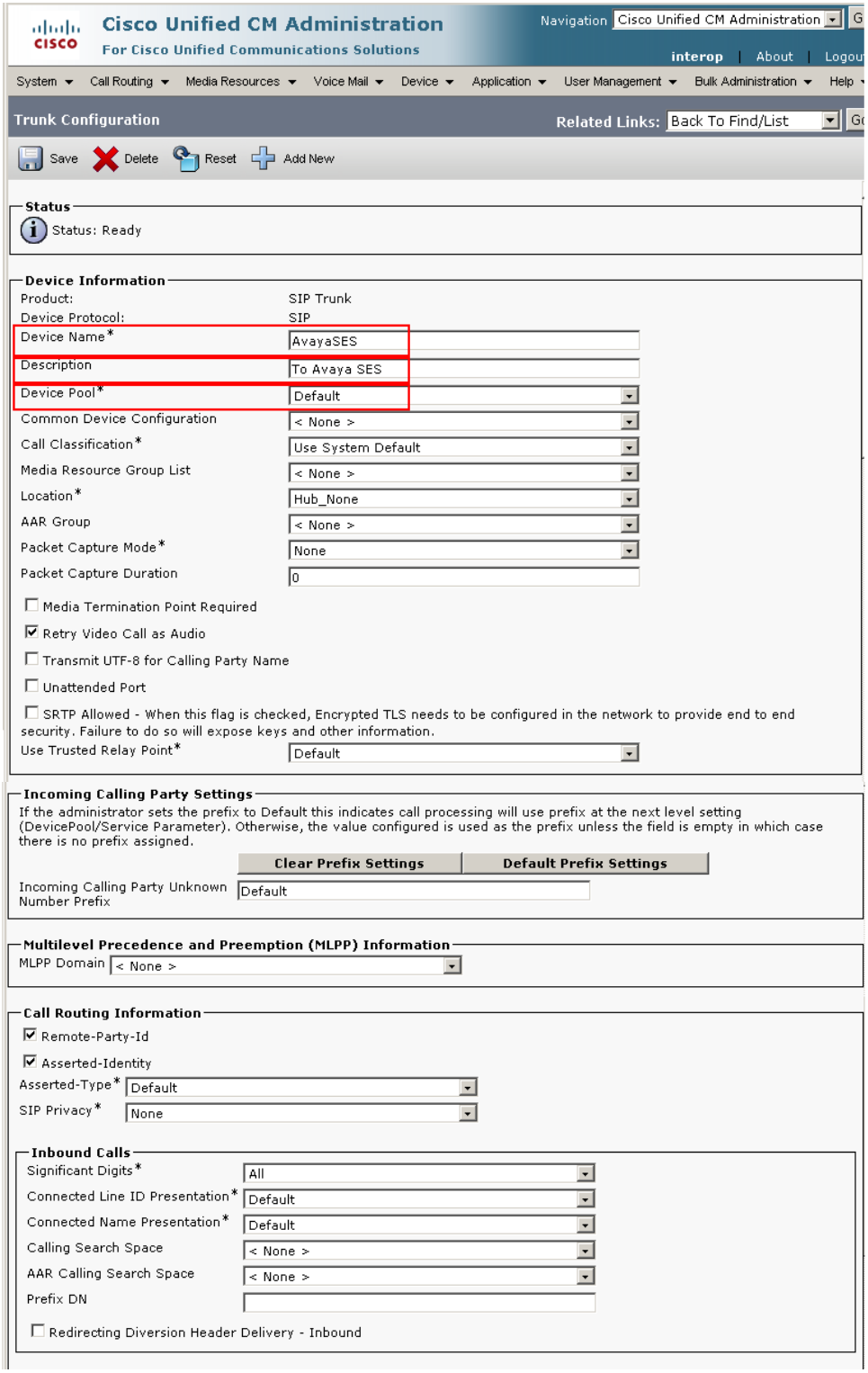
3. Select **Device → Trunk** from the top menu then click **Add New** to begin adding a new SIP trunk.

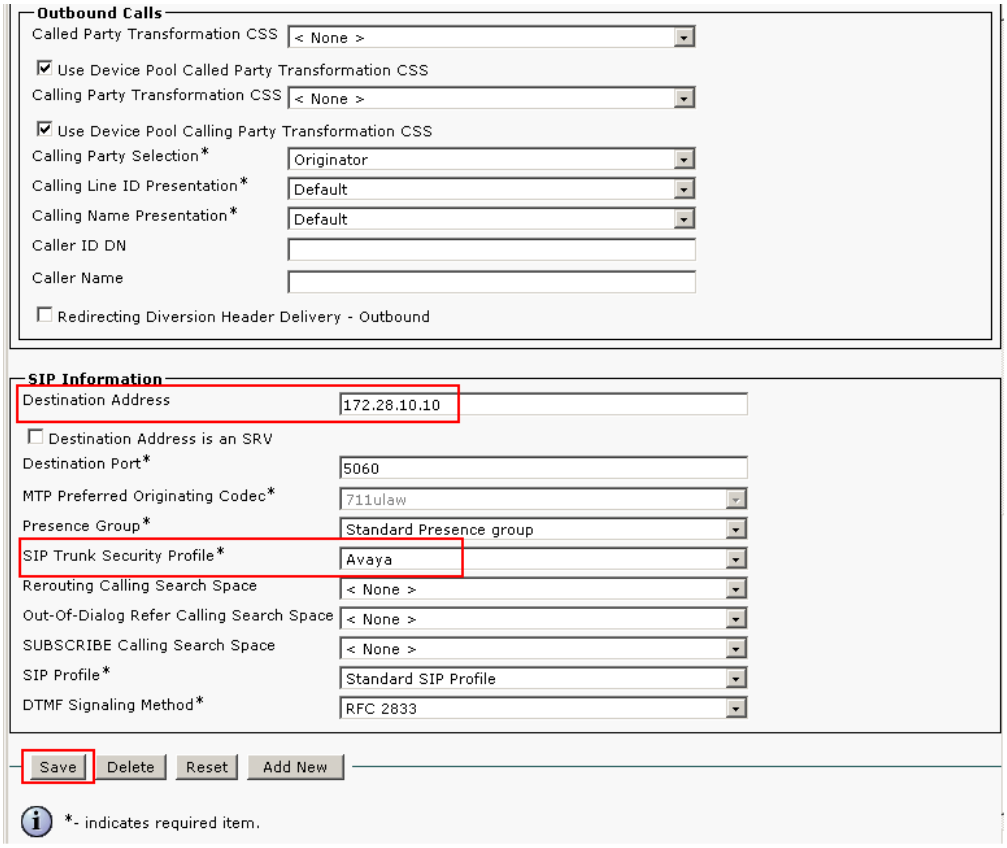


Select **SIP Trunk** as the **Trunk Type** and the **Device Protocol** field will automatically be change to SIP. Click **Next** to continue.

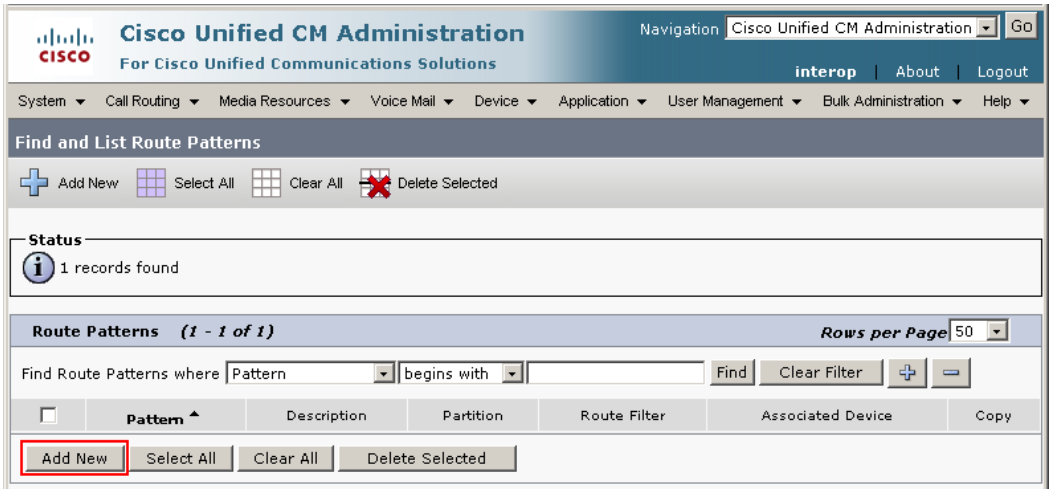


Enter the appropriate information for the SIP Trunk. The following screen capture shows the configuration used in the sample network. Click **Save** to complete.

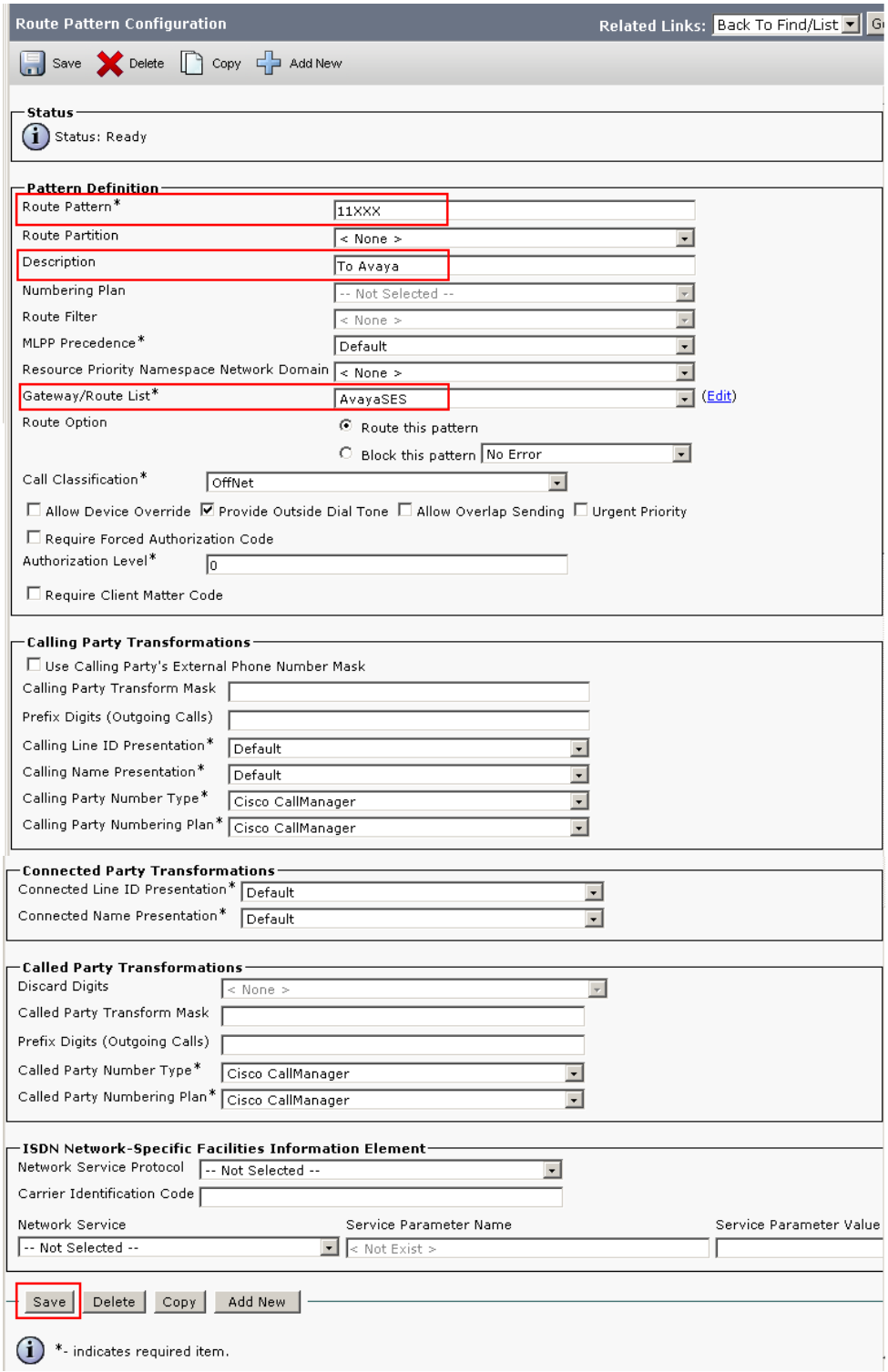




4. Select **Call Routing → Route/Hunt → Route Pattern** then click **Add New** to add a new route pattern for extension 11xxx which are for telephones registered with Avaya Communication Manager.



The following screen capture shows the route pattern used in the sample network. The route pattern "11xxx" will cause all 5 digit calls beginning with "11" to be routed through the "AvayaSES" SIP Trunk defined in **Step 3**. Click **Save** to complete.



6. Configure Avaya Communication Manager

This section shows the configuration of Avaya Communication Manager. All configurations in this section are administered using the System Access Terminal (SAT). These Application Notes assume that the basic configuration between Avaya Communication Manager and Avaya SIP Enablement Services (SES) Server has already been completed and working. For further information on Avaya Communication Manager, please consult with references [1], [2] and [3].

1. Use the **add signaling-group** command to add a new signaling group into the system. The sample network uses signaling groups 29 and 80. Signaling group 29 is the signaling group configured for communication between Avaya Communication Manager and Avaya SES Server that should have been configured as part of the basic configuration. This signaling group has interop.com which is the same domain as Avaya Communication Manager configured as its Far-end Domain.

```
display signaling-group 29
                                SIGNALING GROUP

 Group Number: 29             Group Type: sip
                        Transport Method: tls


   Near-end Node Name: CLAN                 Far-end Node Name: SES
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                    Far-end Network Region: 9
       Far-end Domain: interop.com

                                         Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload        Direct IP-IP Audio Connections? y
                                                  IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 3          Alternate Route Timer(sec): 6
```

The following signaling-group 80 is added to Avaya Communication Manager for the SIP Trunk that is configured for traffic between Avaya SES and CUCM. The Far-end Domain is left blank to allow for incoming SIP calls from domains not matching interop.com. Alternatively, a specific domain that the CUCM system is in can also be entered.

```
display signaling-group 80
                                SIGNALING GROUP

 Group Number: 80             Group Type: sip
                        Transport Method: tls

   Near-end Node Name: CLAN                 Far-end Node Name: SES
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                    Far-end Network Region: 9
       Far-end Domain:

                                         Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload        Direct IP-IP Audio Connections? y
                                                  IP Audio Hairpinning? n
         Enable Layer 3 Test? n
 Session Establishment Timer(min): 3          Alternate Route Timer(sec): 6
```

2. Use the **add trunk-group** command to add a new trunk group into the system. The sample network uses trunk groups 29 and 80. Trunk group 29 is the trunk group configured for communication between Avaya Communication Manager and Avaya SES Server that should have been configured as part of the basic configuration. This trunk group is configured to use signaling group 29 shown in **Step 1**. The following is a screen capture for trunk group 29.

```
display trunk-group 29                                        Page   1 of  21
                              TRUNK GROUP

Group Number: 29                   Group Type: sip           CDR Reports: y
  Group Name: To-SES                       COR: 1        TN: 1        TAC: 129
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                                      Signaling Group: 29
                                                    Number of Members: 50
```

Trunk group 80 is added to Avaya Communication Manager for the SIP Trunk that is configured for traffic between Avaya SES and Cisco Unified CM. Trunk group 80 is configured to use signaling group 80 shown in **Step 1**. The following is a screen capture of trunk group 80. It is important to note that the trunk group used for domains other than interop.com has a higher trunk group number than the trunk group having the native domain, in this case trunk group 29 using interop.com. This helps match domain information on incoming SIP calls to the native domain when multiple SIP trunks are configured between Avaya Communication Manager and Avaya SES.

```
display trunk-group 80                                        Page   1 of  21
                              TRUNK GROUP

Group Number: 80                   Group Type: sip           CDR Reports: y
  Group Name: To-CUCM7                     COR: 1        TN: 1        TAC: 180
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                                      Signaling Group: 80
                                                    Number of Members: 10
```

3. Use the **change ip-network-region** form to configure the ip-network-region and codec. The following screen capture shows ip-network-region 9 used in the sample network.

```
display ip-network-region 9                                   Page   1 of  19
                              IP NETWORK REGION
  Region: 9
Location:         Authoritative Domain: interop.com
    Name:
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: 3329
DIFFSERV/TOS PARAMETERS                RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        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
```

```
display ip-network-region 9                                   Page   3 of  19

                  Inter Network Region Connection Management

 src dst codec direct   WAN-BW-limits   Video      Intervening      Dyn
 rgn rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC IGAR AGL
 9   1    1     y    NoLimit                                          n
 9   2
 9   3
 9   4
 9   5
 9   6
 9   7
 9   8
 9   9    1                                                                all
```

4. Use the **change ip-codec-set** form to configure the audio codec. The following screen capture shows ip-codec-set 1 used in the sample network.

```
display 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:
```

5. Use **change dialplan analysis** to define any 5 digit call beginning with 60 as an aar Call Type. The following screen capture shows the dialplan used in the sample network.

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

      Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
      String   Length Type    String   Length Type     String   Length Type
    1           3    dac
    11          5    ext
    60          5    aar
    8           1    fac
    9           1    fac
    *           3    fac
    #           3    fac
```

6. Use **change aar analysis** form to configure the appropriate route pattern for the 5 digit dial string that begins with 60. The following screen capture shows the calls to 60xxx are routed using Route Pattern 29 in the sample network.

```
display aar analysis 60                                      Page   1 of   2
                           AAR DIGIT ANALYSIS TABLE
                                 Location:  all          Percent Full:    1

         Dialed           Total     Route    Call   Node  ANI
         String          Min  Max  Pattern   Type   Num   Reqd
   60                    5    5     29       aar          n
   7                     7    7     999      aar          n
   8                     7    7     999      aar          n
   9                     7    7     999      aar          n
```

7. Use the **change route-pattern** form to configure the appropriate trunk group. The following screen capture shows that route pattern 29 directs calls to use trunk group 29 in the sample network.

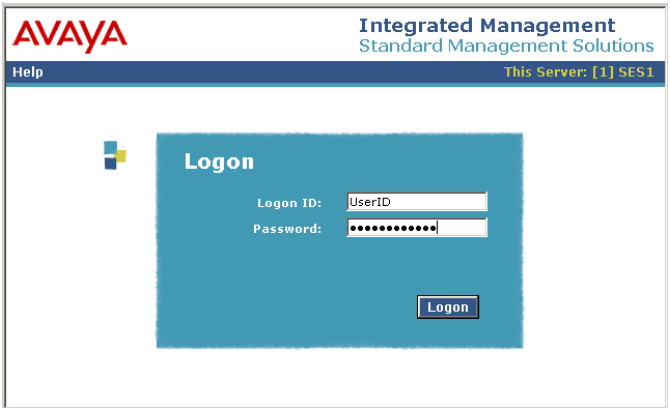
```
change route-pattern 29                                      Page   1 of   3
                    Pattern Number: 29  Pattern Name: To-CUCM-7
                              SCCAN? n     Secure SIP? n
   Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
   No          Mrk Lmt List Del  Digits                          QSIG
                             Dgts                                 Intw
1: 29   0                                                         n   user
2:                                                                n   user
3:                                                                n   user
4:                                                                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                                     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
```

7. Configuring Avaya SES Server

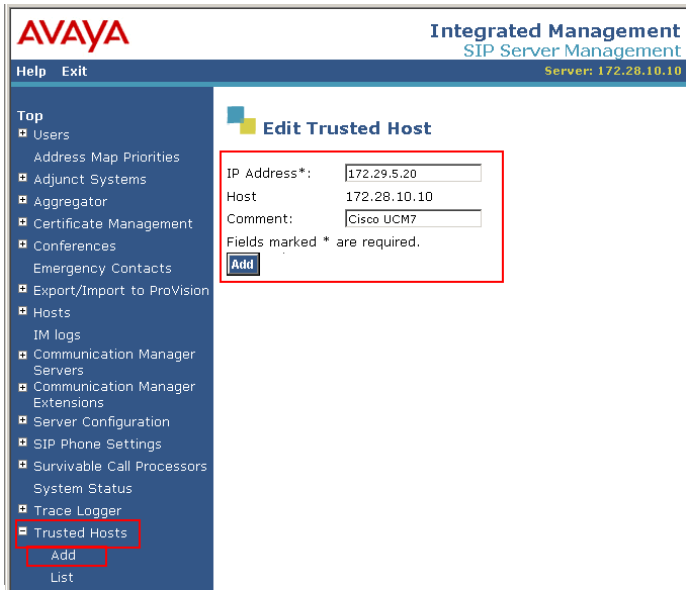
This section shows the configuration of Avaya SES Server. These Application Notes assume that the basic configuration between Avaya Communication Manager and Avaya SES Server has already been completed and working. For further information on Avaya SES, please consult references [4], [5] and [6].

1. Open the Avaya SES administration interface by entering the IP address of Avaya SES Server into a Web Browser in the form of http://IP address/admin. Log in using appropriate credentials.

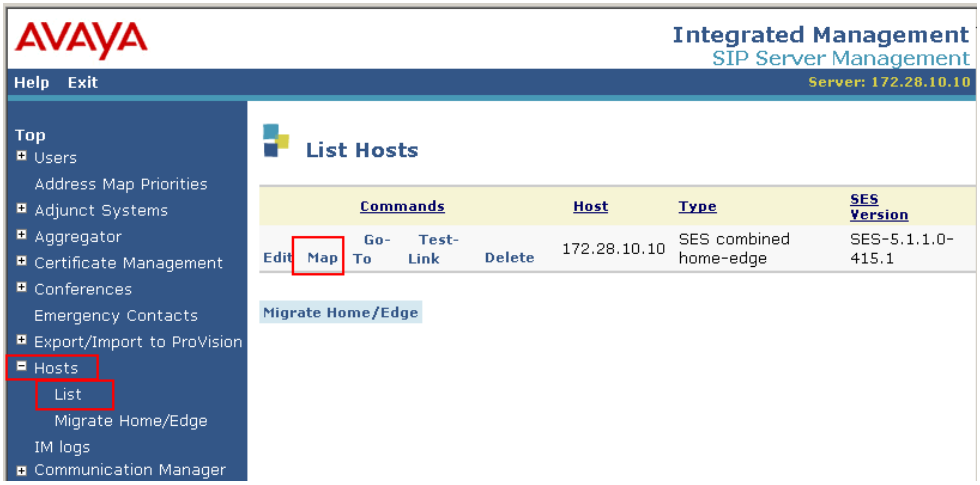
Upon successfully logging into Avaya SES, select Launch SES Administration Interface (not shown below).



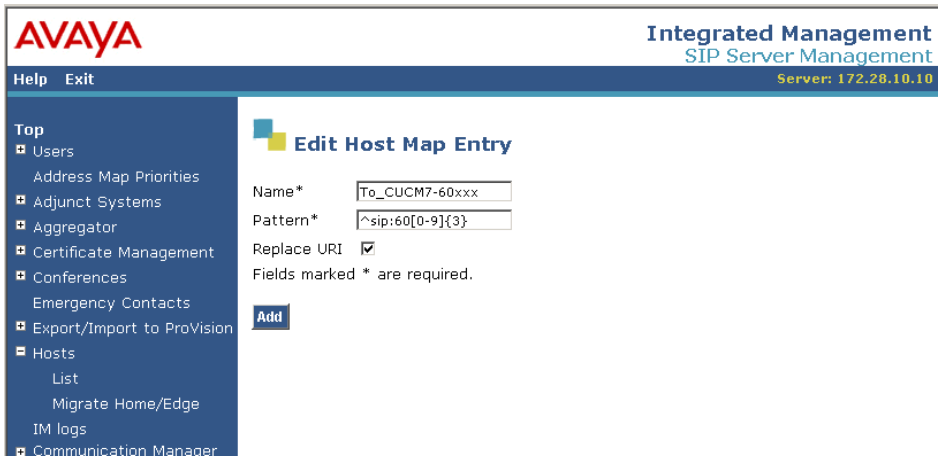
2. Select **Trusted Hosts → Add** to add the CUCM as a trusted host. The following is a screen capture of the Trusted Host information. Click **Add** to complete



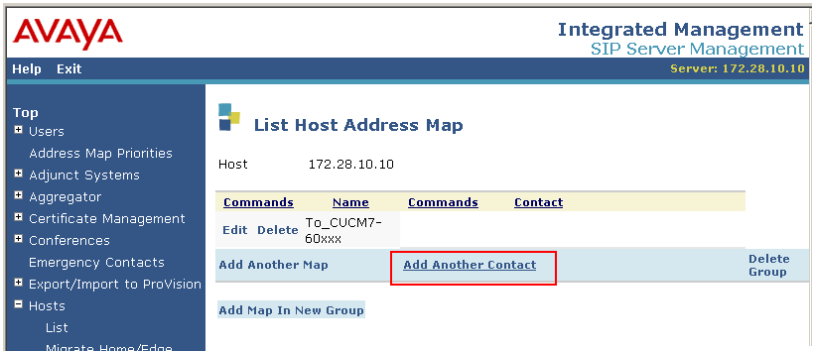
3. Select **Hosts → List** from the left menu panel to display a list of hosts then click on **Map** to begin adding a new address map.



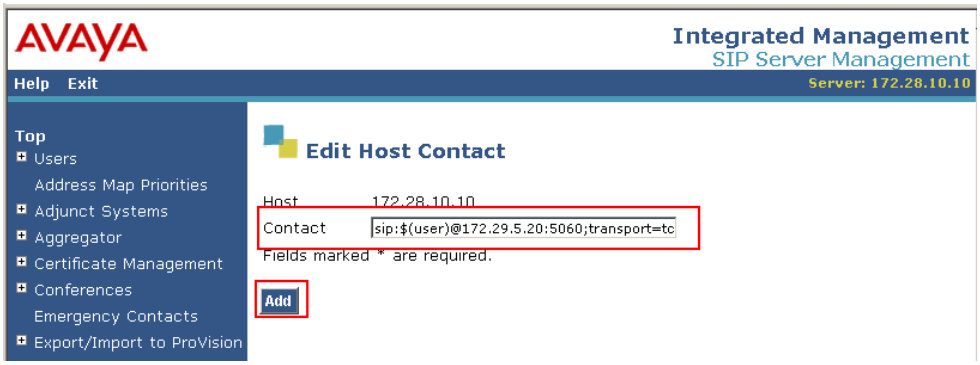
The following screen capture shows the host map to direct any 5 digit number beginning with "60" to Cisco Unified CM. Click **Add** to complete.



After adding the new host map, click **Add Another Contact** to configure the contact information.



The following screen capture shows the host contact used in the sample network. The Contact entry "sip:$(user)@172.29.5.20:5060;transport=tcp" directs the SIP call to the CUCM IP address of 172.29.5.20 using TCP as the transport protocol. Click **Add** to complete.



8. Verification

The following steps may be used to verify the configuration:

1. "list trace station" may be used in Avaya Communication Manager via SAT to verify whether calls from Avaya H.323, digital, or analog telephones are being routed to the correct trunk group.
2. The Real Time Monitoring Tool (RTMT) can be use to monitor events on Cisco Unified CM. This tool can be downloaded by selecting **Application → Plugins** from the top menu of the Cisco Unified CM Administration Web interface. For further information on this tool, please consult with reference [10].

9. Conclusion

These Application Notes describe the administrative steps required to configure a SIP trunk to support calls between an Avaya IP PBX and a Cisco IP PBX 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.

Due to implementation differences between Cisco SIP and SCCP telephone, there is a certain limitation when using Music-on-Hold between an Avaya telephone and Cisco SCCP telephone. Music can not be heard on an Avaya telephone when a call is placed on Hold by the Cisco SCCP telephone. However, Music-on-Hold does work when the Hold is placed by an Avaya telephone. This limitation does not apply to the Cisco SIP telephone used in this configuration.

10. Additional References

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

[1] *Administrator Guide for Avaya Communication Manager*, Doc # 03-300509, Issue 4.0, Release 5.0, January 2008

[2] *Avaya Communication Manager Advanced Administration Quick Reference*, Doc # 03-300364, Issue 4, Release 5.0, January 2008

[3] *Administration for Network Connectivity for Avaya Communication Manager*, Doc # 555-233-504, Issue 13, January 2008

[4] *Administering SIP Enablement Services on the Avaya S8300 Server*, Doc # 03-602508, Issue 1.0, January 2008

[5] *SIP Support in Avaya Communication Manager Running on Avaya S8xxx Servers*, Doc # 555-245-206, Issue 8, January 2008

[6] *Configuring Avaya SIP Telephony Users on Avaya SIP Enablement Services and Avaya Communciation Manager*, Issue 1.0

[7] *one-X Communicator Adminsitration Quick Setup*, Doc # 16-602603, Issue 3, December 2008

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

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

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

[10] *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