



## **Configuring Avaya SIP Enablement Services, Avaya Communication Manager, and Cisco IOS Survivable Remote Site Telephony (SRST) to Support Cisco 7940/7960 SIP Telephones - Issue 1.0**

### **Abstract**

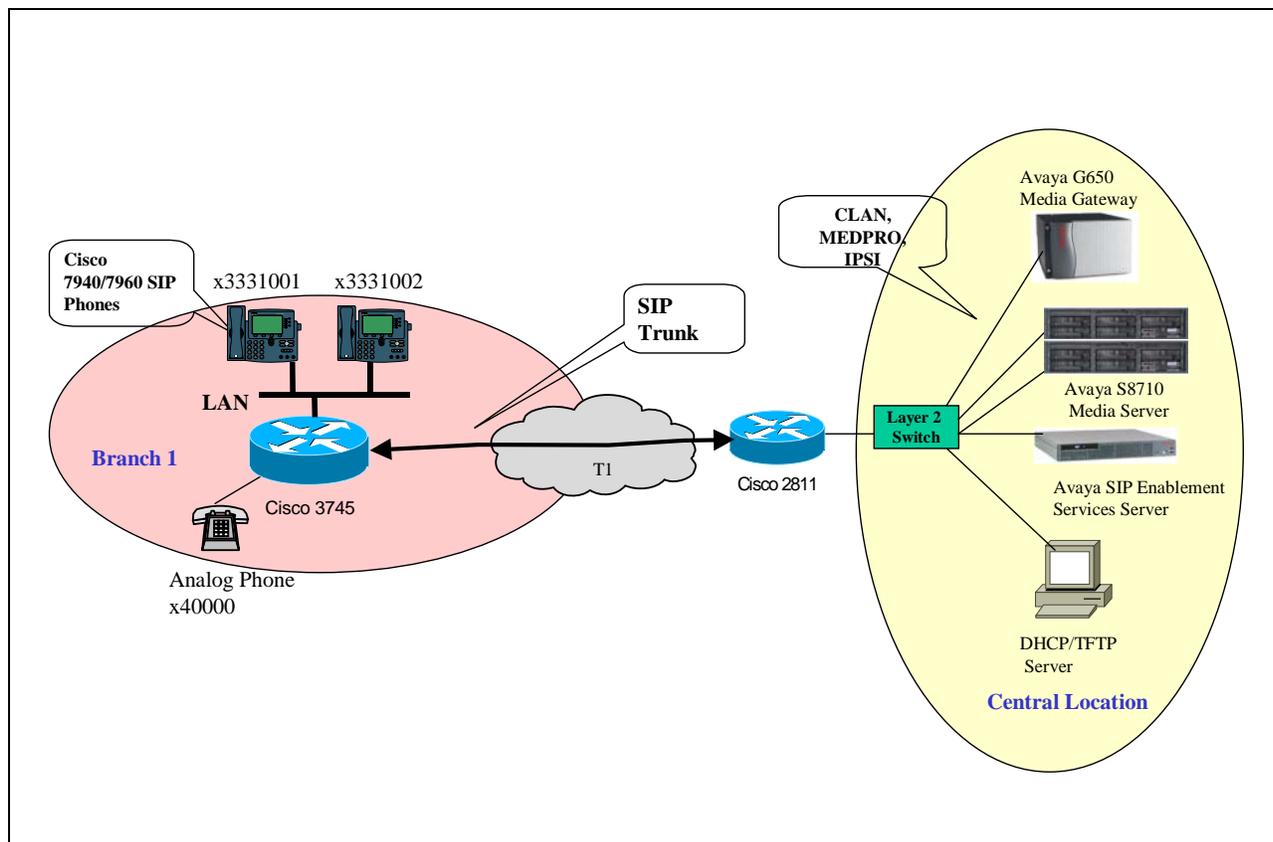
These Application Notes describe the procedure for configuring the Avaya SIP Enablement Services (SES) server, Avaya Communication Manager, and a Cisco SRST enabled router to support Cisco 7940/7960 SIP Telephones. Under normal operation, the Cisco 7940/7960 SIP Telephones are registered with the Avaya SES server as the primary proxy and can leverage the Avaya Outboard Proxy SIP (OPS) features provided by Avaya Communication Manager. In the sample configuration, a Cisco 3745 router with SRST is configured as a backup proxy for the Cisco SIP Telephones. If the Avaya SES server becomes unreachable (e.g., due to network failure), the Cisco SIP Telephones are served by the SRST feature of the router. When the Avaya SES server becomes reachable (e.g., network is restored), the Cisco SIP Telephones are once again served by the Avaya SES server and Avaya Communication Manager. These Application Notes were written at the request of an Avaya customer.

# 1. Introduction

These Application Notes describe the procedure for configuring the Avaya SIP Enablement Services (SES) server, Avaya Communication Manager, and a Cisco SRST enabled router to support Cisco 7940/7960 SIP Telephones. In the sample configuration shown in **Figure 1**, Cisco 7940/7960 SIP Telephones were dual registered with the Avaya SES Server as the primary proxy and with the Cisco 3745 router as the backup proxy. Avaya Communication Manager OPS station feature were made available to the Cisco SIP Telephones to complement the natively supported features. If the WAN link fails, the Cisco SRST router will act as a proxy server to provide basic call service for the Cisco SIP Telephones. These Application Notes cover the following areas:

- Avaya SES and Avaya Communication Manager SIP related configuration
- Cisco router IOS SRST configuration

**Figure 1** shows the network configuration used for verification.



**Figure 1: Network Configuration**

**Table 1** lists the IP address information for the tested devices.

Device	IP Address/Mask	Gateway
Avaya S8710 Media Server 1	5.1.1.2 /24	5.1.1.1
Avaya S8710 Media Server 2	5.1.1.3/24	5.1.1.1
Active Server	5.1.1.34/24	
Avaya G650 Media Gateway		
• IPSI	5.1.1.7/24	5.1.1.1
• C-LAN	5.1.1.8/24	5.1.1.1
• MEDPRO	5.1.1.9/24	5.1.1.1
Avaya SES Server	5.1.1.14/24	5.1.1.1
Cisco 3745 Router		
Fast Ethernet 0/0	120.1.1.1/24	
Serial 0/0	150.1.1.2/24	
Cisco 2811 Router		
Fast Ethernet 0/0	5.1.1.1/24	
Serial 0/0	150.1.1.1/24	

**Table 1: Device IP Address and Gateway Configuration**

## 2. Equipment and Software Validated

**Table 2** lists the equipment and software version used for the configuration.

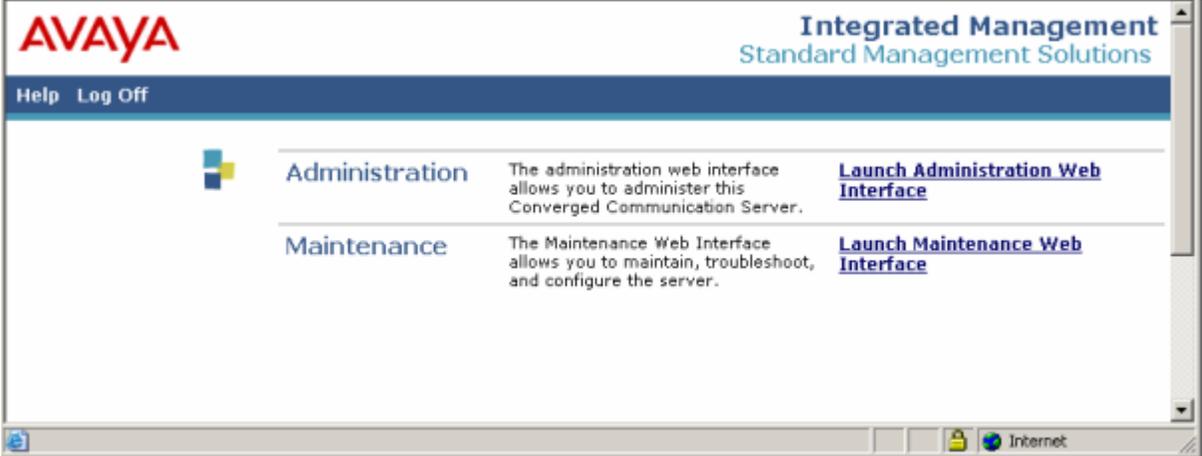
Equipment	Software
Avaya S8710 Media Server/G650 Media Gateway	Communication Manager 3.0 (Load 340.3)
Avaya SIP Enablement Services Server	R3.0
Cisco 7960/7940 SIP Telephones	V 7.4
Cisco 3745 Router	IOS 12.4(4)T
Cisco 2811 Router	IOS 12.3
DHCP/TFTP Server	
▪ Windows 2003 Advanced Server	Version 2003

**Table 2: Equipment and Software Validated**

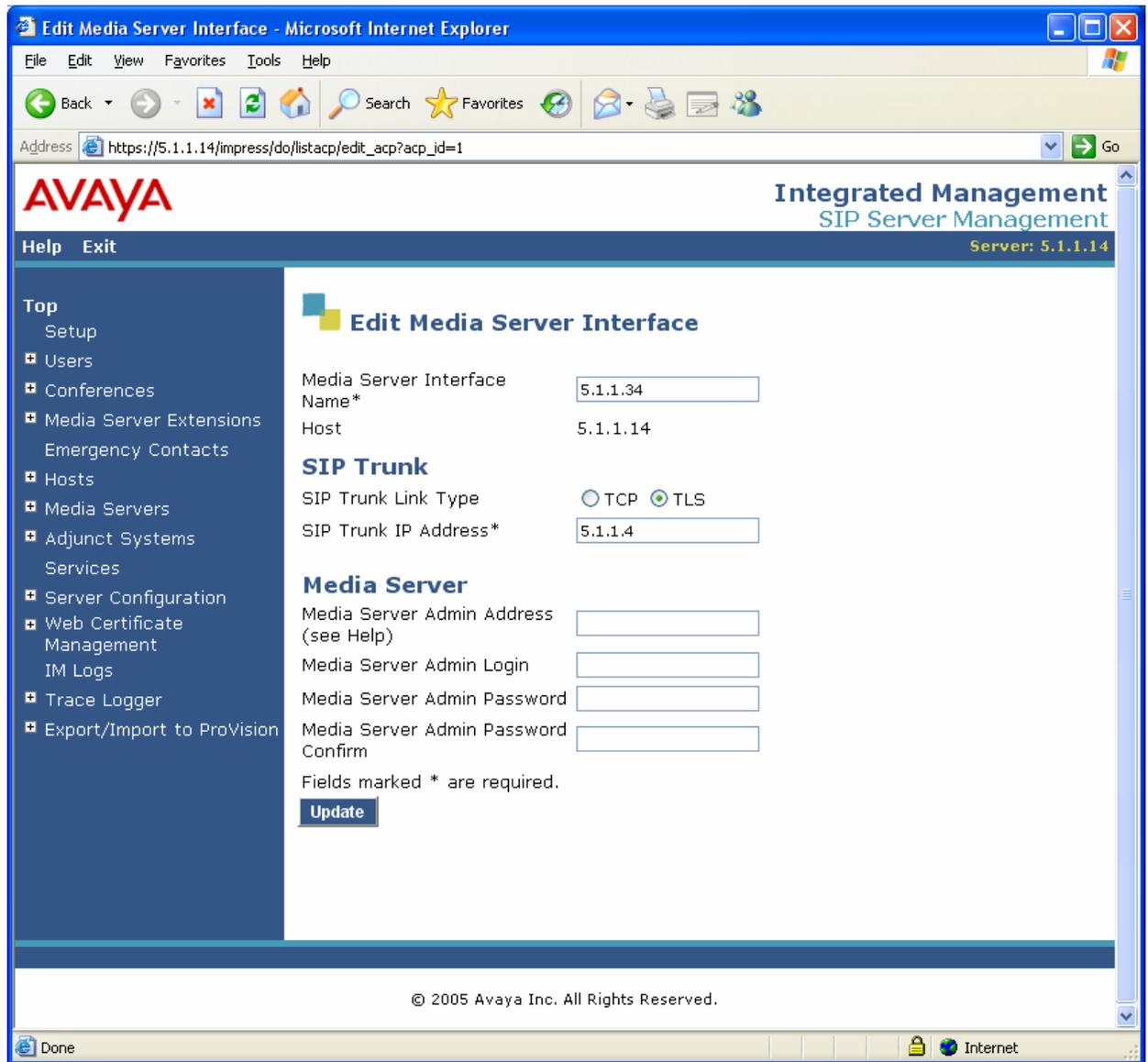
## 3. Configure Avaya SES Server

Installation and initial configuration of the Avaya SES Server is beyond the scope of this document. Section 7 lists references to detailed product documentation and related Application Notes. This section illustrates the configuration of the SIP trunk, Media Server, and Host

Address Mapping relevant to the network shown in **Figure 1**.

Step	Description
<p><b>1.</b></p>	<p><b>Log in to Avaya SES Server</b></p> <ul style="list-style-type: none"> <li>• Launch a web browser with the URL of the Avaya SES Server (e.g., http://5.1.1.14). Log in with proper user name and password.</li> <li>• Click <b>Launch Administration Web Interface</b> as shown below.</li> </ul> 
<p><b>2.</b></p>	<p><b>Add Media Server</b></p> <p>Avaya Communication Manager must be configured as a Media Server on the Avaya SES Server. Click on the <b>Media Servers</b> link on the left side of the main SIP Server Management web page.</p> 

- Click on **Add Media Server** link under **Manage Media Server Interfaces**.
- Enter the S8710 active server's IP address **5.1.1.34**.
- Click on **TLS** as **SIP Trunk Link Type**.
- Enter C-LAN IP address **5.1.1.4** in **SIP Trunk IP Address** field.
- Click on **Update**.



### 3. Add Users

Note that all Cisco 7940/7960 SIP Telephones must be added as users in Avaya SES Server with a media server extension. The following configuration displays the procedure to add user 3331001.

- Click on **Users** link on left side configuration panel.
- Click on **Add User** under **User Administration**.

The screenshot displays the Avaya Integrated Management SIP Server Administration interface. The top left features the Avaya logo, and the top right shows the text 'Integrated Management SIP Server Administration'. Below the logo is a navigation menu with 'Help' and 'Exit' links. The main content area is titled 'User Administration' and contains a list of actions:

Action	Description
List Users	List all users.
Add User	Add a new user.
Search Users	Search for users.
Edit User Profile	Edit a user by user id.
Delete User	Delete a user by user id.
Update Password	Change a password by user id.
Edit Default User Profile	Edit the default user profile.
Registered Users	Search for registered and provisioned users.

- Enter user information as shown below.
- Check box **Add Media Server Extension**
- Click on **Add**



Help Exit

Top

- Setup
- ▣ Users
  - List
  - Add
  - Search
  - Edit
  - Delete
  - Password
  - Default Profile
  - Registered Users
- ▣ Conferences
- ▣ Media Server Extensions
  - Emergency Contacts
- ▣ Hosts
- ▣ Media Servers
- ▣ Adjunct Systems
- Services
- ▣ Server Configuration
- ▣ Web Certificate Management
  - IM Logs
- ▣ Trace Logger
- ▣ Export/Import to ProVision

Add User

Primary Handle*	<input type="text" value="3331001"/>
User ID	<input type="text"/>
Password*	<input type="password" value="•••••"/>
Confirm Password*	<input type="password" value="•••••"/>
Host*	<input type="text" value="5.1.1.14"/> ▾
First Name*	<input type="text" value="SIP"/>
Last Name*	<input type="text" value="Telephone 1"/>
Address 1	<input type="text"/>
Address 2	<input type="text"/>
Office	<input type="text"/>
City	<input type="text"/>
State	<input type="text"/>
Country	<input type="text"/>
Zip	<input type="text"/>
Add Media Server Extension	<input checked="" type="checkbox"/>

Fields marked \* are required.

**Add**

Click on **Continue**.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top header includes the Avaya logo, the title "Integrated Management SIP Server Management", and the server version "Server: 5.1.1.14". A navigation menu on the left lists "Top", "Setup", "Users" (expanded), "List", "Add", "Search", "Edit", "Delete", and "Password". The main content area displays a confirmation message: "User ID 3331001 added." with a "Continue" button below it.

From **Add Media Server Extension** screen:

- Enter **3331001** as **Extension**
- Select the appropriate **Media Server**
- Click **Add**

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top header includes the Avaya logo, the title "Integrated Management SIP Server Management", and the server version "Server: 5.1.1.14". A navigation menu on the left lists "Top", "Setup", "Users" (expanded), "List", "Add", "Search", "Edit", "Delete", and "Password". The main content area displays the "Add Media Server Extension" form. The form title is "Add Media Server Extension". Below the title, it says "Add Media Server extension for user 3331001." The form has two input fields: "Extension\*" with the value "3331001" and "Media Server" with a dropdown menu showing "5.1.1.34". Below the form, it says "Fields marked \* are required." and there is an "Add" button.

- Click **Continue**.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. At the top left is the Avaya logo. At the top right, the text reads "Integrated Management SIP Server Management" with "Server: 5.1.1.14" below it. A dark blue navigation bar contains "Help" and "Exit" on the left, and "Server: 5.1.1.14" on the right. A dark blue sidebar menu on the left lists "Top", "Setup", "Users" (with a sub-menu icon), "List", "Add", "Search", "Edit", and "Delete". The main content area displays a confirmation message: "Extension 3331001 added for user 3331001". Below the message is a blue button labeled "Continue".

Click **Update** from the left configuration panel.

The screenshot shows the Avaya Integrated Manager SIP Server Manager interface. The top left features the Avaya logo. The top right displays 'Integrated Manager SIP Server Manager' and 'Server: 5.1.1.'. Below the header is a navigation menu with 'Help' and 'Exit'. The main content area is titled 'List Media Server Extensions' and shows 'Media Server extensions for user 3331001.'. A table lists the extensions with columns for 'Commands', 'Extension', 'User', 'Media Server', and 'Host'. The table contains one row with the following data: 'Free', 'Edit User', 'Delete', '3331001', '3331001', '5.1.1.34', and '5.1.1.14'. The left sidebar contains a tree view with categories like 'Users', 'Conferences', 'Media Server Extensions', 'Hosts', 'Media Servers', 'Adjunct Systems', 'Server Configuration', 'Web Certificate Management', 'Trace Logger', and 'Export/Import to ProVision'. The 'Update' button is highlighted at the bottom of the sidebar.

Commands		Extension	User	Media Server	Host	
Free	Edit User	Delete	3331001	3331001	5.1.1.34	5.1.1.14

#### 4. Add Host Address Map

Address maps specify how incoming SIP calls are to be routed, based on the dialed number. The Avaya SES Server will route calls to Avaya Communication Manager. If the called station is a local registered station, Avaya Communication Manager will instruct the Avaya SES Server to ring the station. If the extension is not an OPS extension on Avaya Communication Manager, then the SES Server must route the call. Routing instructions for incoming calls to non-local destinations must be configured. In the sample configuration shown in **Figure 1**, the analog telephone connected to the Cisco 3745 router is an example of a non-local destination.

- Click the **Hosts** link on the left side of the main Avaya SES web page.
- Click **List** under **Hosts** on the left side of the main SES web page.
- Click **Add Another Map**.

The screenshot shows the Avaya SES web interface. At the top left is the Avaya logo, and at the top right is the text 'Integrate SIP Ser'. Below the logo is a navigation menu with 'Help' and 'Exit' links. The main content area is titled 'List Host Address Map' and displays a host 'TF-SES-SIP.avaya.com'. Below this is a table with the following structure:

Commands	Name	Commands	Contact
Add Another Map		Add Another Contact	

Below the table is a button labeled 'Add Map In New Group'.

Since the analog telephone is using extension 40000, set the **Pattern** to match the leading digit “4” to route the call to the Cisco 3745 router.

- Enter values as shown in the screen below.
- Click **Add**.

**AVAYA** Integrat  
SIP

Help Exit

Top

- Setup
- + Users
- + Conferences
- + Media Server Extensions
- Emergency Contacts
- Hosts
  - List
  - Migrate Home/Edge
- + Media Servers

### Add Host Address Map

Host TF-SES-SIP.avaya.com

Name\*

Pattern\*

Replace URI

Fields marked \* are required.

**Add**

**5. Add Contact**

In addition to the Host map, the SIP **Contact** must be specified. Click **Add Another Contact** (under **List Host Address Map** shown in Step 4) and enter the configuration shown below. The analog telephone is connected to the Cisco 3745 router with IP Address 120.1.1.1 (see **Table 1**).

**AVAYA** Integrated Management  
SIP Server Management

Help Exit Server: 5.1.1.14

Top

- Setup
- + Users
- + Conferences
- + Media Server Extensions
- Emergency Contacts
- Hosts
  - Update All
  - List
  - Migrate Home/Edge

### Add Host Contact

Host TF-SES-SIP.avaya.com

Handle ToCiscoAnalog

Contact\*

Fields marked \* are required.

**Add**

Click **Add**.

**AVAYA** Integrated Management  
SIP Server Management  
Server: 5.1.1.14

Help Exit

Top  
Setup  
Users  
Conferences  
Media Server Extensions  
Emergency Contacts  
Hosts  
Update All

**Continue**

Host contact sip:\$(user)@120.1.1.1:5060;transport=udp added for map entry ToCiscoAnalog

**Continue**

- Click **Continue**
- Click **Update All**

### 3.1. Configure SIP on Avaya Communication Manager

This section presents the SIP related configuration in Avaya Communication Manager. SIP signaling groups and trunk groups are required to support SIP endpoints. The Avaya SES Server always consults Avaya Communication Manager to determine how to route a call upon receiving a SIP Invite.

Step	Description																																								
1.	<p>From the Avaya Communication Manager SAT interface, use the <b>change node-names ip</b> command to add a node name for the Avaya SES Server.</p> <pre>change node-names ip</pre> <p style="text-align: right;">Page 1 of 1</p> <table border="1"> <thead> <tr> <th colspan="2"></th> <th colspan="4">IP NODE NAMES</th> <th colspan="2"></th> </tr> <tr> <th>Name</th> <th>IP Address</th> <th>Name</th> <th colspan="3">IP Address</th> <th colspan="2"></th> </tr> </thead> <tbody> <tr> <td><b>Avaya-SES</b></td> <td>5 .1 .1 .14</td> <td></td> <td>.</td> <td>.</td> <td>.</td> <td colspan="2"></td> </tr> <tr> <td><b>c-lan</b></td> <td>5 .1 .1 .8</td> <td></td> <td>.</td> <td>.</td> <td>.</td> <td colspan="2"></td> </tr> <tr> <td><b>medpro</b></td> <td>5 .1 .1 .9</td> <td></td> <td>.</td> <td>.</td> <td>.</td> <td colspan="2"></td> </tr> </tbody> </table>			IP NODE NAMES						Name	IP Address	Name	IP Address					<b>Avaya-SES</b>	5 .1 .1 .14		.	.	.			<b>c-lan</b>	5 .1 .1 .8		.	.	.			<b>medpro</b>	5 .1 .1 .9		.	.	.		
		IP NODE NAMES																																							
Name	IP Address	Name	IP Address																																						
<b>Avaya-SES</b>	5 .1 .1 .14		.	.	.																																				
<b>c-lan</b>	5 .1 .1 .8		.	.	.																																				
<b>medpro</b>	5 .1 .1 .9		.	.	.																																				

Step	Description
2.	<p data-bbox="289 268 1398 338">Use <b>add signaling-group</b> to add a signaling group to the Avaya SES Server, as shown below.</p> <pre data-bbox="289 373 1484 890"> add signaling-group 1                                     Page 1 of 1                                  SIGNALING GROUP  Group Number: 1                Group Type: <b>sip</b>                                 Transport Method: <b>tls</b>  Near-end Node Name: <b>c-lan</b>           Far-end Node Name: <b>Avaya-SES</b> Near-end Listen Port: <b>5061</b>         Far-end Listen Port: <b>5061</b>                                 Far-end Network Region: 1  Far-end Domain: avaya.com                                  Bypass If IP Threshold Exceeded? n  DTMF over IP: rtp-payload      Direct IP-IP Audio Connections? y                                 IP Audio Hairpinning? y  SESSion Establishment Timer(min): 120 </pre>

Step	Description
3.	<p data-bbox="289 233 1481 302">Use the <b>add trunk group</b> command to add a SIP trunk between Avaya Communication Manager and the Avaya SES Server. The following screens illustrate the trunk configuration.</p> <pre data-bbox="289 331 1481 783"> display trunk-group 1                                     Page 1 of 19                                      TRUNK GROUP Group Number: 1                Group Type: sip                CDR Reports: y Group Name: sip-trunk          COR: 1                TN: 1                TAC: 101   Direction: two-way          Outgoing Display? y   Dial Access? n                Busy Threshold: 255          Night Service: Queue Length: 0 Service Type: tie                Auth Code? n                                      Signaling Group: 1                                      Number of Members: 15 TRUNK PARAMETERS Unicode Name? y                Redirect On OPTIM Failure: 5000   SCCAN? n                    Digital Loss Group: 18 </pre> <pre data-bbox="289 856 1481 1140"> display trunk-group 1                                     Page 2 of 19 TRUNK FEATURES   ACA Assignment? n                Measured: internal                                      Maintenance Tests? y                                      Numbering Format: public                                      Replace Unavailable Numbers? n </pre> <pre data-bbox="289 1192 1481 1822"> display trunk-group 1                                     Page 3 of 19                                      TRUNK GROUP Administered Members (min/max): 1/15 GROUP MEMBER ASSIGNMENTS          Total Administered Members: 15 Port                               Name 1: T00001                          sip-trunk 2: T00002                          sip-trunk 3: T00003                          sip-trunk 4: T00004                          sip-trunk 5: T00005                          sip-trunk 6: T00006                          sip-trunk 7: T00007                          sip-trunk 8: T00008                          sip-trunk 9: T00009                          sip-trunk 10: T00010                         sip-trunk 11: T00011                         sip-trunk 12: T00012                         sip-trunk 13: T00013                         sip-trunk 14: T00014                         sip-trunk 15: T00015                         sip-trunk </pre>

Step	Description
<p><b>4.</b></p>	<p>As shown in <b>Figure 1</b>, there is an analog telephone connected to the Cisco 3745 router FXS port with extension 40000. Use the <b>change uniform-dialplan</b> command to enable Avaya Communication Manager to route a call to extension 40000 via its AAR table.</p> <pre data-bbox="289 373 1481 772"> change uniform-dialplan 4                                     Page 1 of 2                                  UNIFORM DIAL PLAN TABLE                                  Percent Full: 0  Matching          Insert          Node Matching          Insert          Node Pattern   Len Del  Digits  Net Conv Num  Pattern   Len Del  Digits  Net Conv Num <b>4</b>         5  0          aar   n      n      n         n      n      n      n      n           n      n      n      n      n      n         n      n      n      n      n           n      n      n      n      n      n         n      n      n      n      n           n      n      n      n      n      n         n      n      n      n      n </pre>
<p><b>5.</b></p>	<p>Use the <b>change route-pattern</b> command to configure a route-pattern containing trunk group 1. As shown in the following step, route-pattern 1 will be used for calls to the analog telephone with extension 40000.</p> <pre data-bbox="289 951 1481 1255"> change route-pattern 1                                     Page 1 of 3                                  Pattern Number: 1   Pattern Name: SIP                                 SCCAN? n           Secure SIP? n  Grp FRL NPA Pfx Hop Toll No.  Inserted          DCS/ IXC No   Lmt List Del  Digits          QSIG                                 Dgts          Intw 1:  1   0 2: 3:                                 n  user                                 n  user                                 n  user </pre>
<p><b>6.</b></p>	<p>Use the <b>change aar analysis</b> command to configure route pattern 1 for numbers that are five digits in length, starting with the digit 4.</p> <pre data-bbox="289 1394 1481 1749"> change aar analysis 1                                     Page 1 of 2                                  AAR DIGIT ANALYSIS TABLE                                  Percent Full: 1  Dialed          Total          Route          Call          Node          ANI String          Min  Max  Pattern  Type  Num  Reqd 2              7   7   999     aar   n    n 3              7   7   999     aar   n    n <b>4</b>           5   5   1       aar   n    n 5              7   7   999     aar   n    n 6              7   7   999     aar   n    n </pre>

Step	Description
7.	<p data-bbox="289 235 1482 346">To configure the Cisco SIP Telephones as OPS stations on Avaya Communication Manager, use the <b>change off-pbx-telephone station-mapping</b> command. The following screen shows an example for configuring OPS station mapping for extension 3331001.</p> <pre data-bbox="289 373 1482 877"> change off-pbx-telephone station-mapping 3331001           Page 1 of 2                 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION    Station      Application  Dial   Phone Number      Trunk Configuration Extension 3331001        OPS                - 3331001          1          1                 -                 - change off-pbx-telephone station-mapping 3331001           Page 2 of 2                 STATIONS WITH OFF-PBX TELEPHONE INTEGRATION    Station      Call      Mapping   Calls      Bridged Extension     Limit   Mode      Allowed   Calls 3331001       3       both     all       both </pre> <p data-bbox="289 940 1482 968">For additional OPS configuration, refer to the Application Notes listed in Section 7.</p>

## 4. Configure SRST Feature on Cisco 3745 Router

This section illustrates the commands necessary to configure the SRST feature in the Cisco 3745 router. SIP registrar functionality on Cisco IOS enables the Cisco router to become a backup SIP proxy and accept SIP registration messages from SIP phones. A registrar accepts SIP register requests and dynamically builds VoIP dial peers, allowing the Cisco IOS Voice Gateway software to route calls to SIP phones.

Under normal operation, the Cisco SIP Telephones are registered with the Avaya SES Server as the primary proxy, and with the Cisco 3745 router as the backup proxy. If the Avaya SES Server is not available (e.g., a WAN failure), the Cisco 3745 router will function as an active proxy to route calls for the Cisco SIP Telephones. This “fail-over” happens almost immediately after the router loses connection to the primary proxy. Once the primary proxy (Avaya SES Server) is reachable again (e.g., WAN is restored), the Cisco SIP Telephones will automatically “fall back” to re-register with the primary proxy server. The timer configured on SIP phones controls the time for these telephones to re-register (fallback).

Description	
ip subnet-zero	
ip cef	
!	
no ip domain lookup	
voice service voip	<i>enable VoIP service on router</i>
allow-connections sip to sip	<i>enable local call transfer</i>
redirect ip2ip	<i>enable redirect IP to IP call</i>
sip	
registrar server expires max 600 min 60	
redirect contact order best-match	<i>enable VoIP service on router</i>
!	
voice class codec 15	<i>configure voice class codec pool for audio</i>
codec preference 1 g711ulaw	<i>add G711 ulaw as first choice</i>
codec preference 2 g729br8	<i>add G729 as second choice</i>
voice register pool 1	<i>create a voice register pool</i>
id network 120.1.1.0 mask 255.255.255.0	<i>allow IP end points with IP address at range 120.1.1.0 to register with router</i>
application session	<i>enable Application SIP</i>
preference 2	
proxy 5.1.1.14 preference 1 monitor probe icmp-ping	<i>define Avaya SES (5.1.1.14) as primary proxy.</i>
dtmf-relay rtp-nte	
voice-class codec 15	<i>use voice-class 15 defined above</i>
interface FastEthernet0/0	
ip address 120.1.1.1 255.255.255.0	
duplex auto	
speed auto	
interface Serial0/0	
ip address 150.1.1.2 255.255.255.0	
encapsulation ppp	
no fair-queue	
service-module t1 clock source internal	
router ospf 1	
log-adjacency-changes	
network 150.1.1.0 0.0.0.255 area 0	
network 120.1.1.0 0.0.0.255 area 0	
call fallback active	<i>set call fallback active for SRST to fallback to primary</i>

## Description

*proxy when the proxy is available.*

```
!  
voice-port 4/0/0  
!  
voice-port 4/0/1  
!  
! Create an extension 40000 on router for analog telephone. Note this analog telephone is  
! registered with the Cisco 3745 router only.  
  
dial-peer voice 40000 pots      configure dial-peer for analog telephone on router FXS port.  
destination-pattern 40000      route call to port 4/0/0 for incoming call to 40000.  
port 4/0/0  
  
!  
line con 0  
line aux 0  
line vty 0 4  
password cisco  
login  
!  
end
```

## 5. Verification Steps

The following verification steps were used in these Application Notes to verify correct system operation:

- Power up Cisco SIP Telephones and verify that all telephones register with the Avaya SES Server.
- Verify that the Cisco 3745 router generates two dial-peers for the SIP IP telephones. One dial-peer is associated with the Avaya SES server, and the other is associated with the router itself. The dial-peer configuration is displayed in the session target statement. Note that the Avaya SES server has preference 1. The following command displays example dial-peers generated on the router.

```
C3745#show voice register dial-peers  
dial-peer voice 40001 voip  
  application session  
  preference 2  
  destination-pattern 3331001  
  redirect ip2ip  
  session target ipv4:120.1.1.101:5060  
  session protocol sipv2  
  voice-class codec 15
```

```

dial-peer voice 40002 voip
  application session
  preference 1
  destination-pattern 3331001
  redirect ip2ip
  session target ipv4:5.1.1.14:5060
  session protocol sipv2
  voice-class codec 15
  monitor probe icmp-ping 5.1.1.14

dial-peer voice 40003 voip
  application session
  preference 2
  destination-pattern 3331002
  redirect ip2ip
  session target ipv4:120.1.1.102:5060
  session protocol sipv2
  voice-class codec 15

dial-peer voice 40004 voip
  application session
  preference 1
  destination-pattern 3331002
  redirect ip2ip
  session target ipv4:5.1.1.14:5060
  session protocol sipv2
  voice-class codec 15
  monitor probe icmp-ping 5.1.1.14

```

- Display the registered users from the Cisco 3745 router and verify that all phones are registered.

**Cisco3745# show sip status register**

Line	destination	expires(sec)	contact
3331001	120.1.1.101	300	120.1.1.101
3331002	120.1.1.102	149	120.1.1.102

- Make a call from the Cisco 7960 SIP Telephone to the analog telephone on the router and verify that the call is successful.
- Make a call from the analog telephone to the Cisco 7960 SIP Telephone and verify that the call is successful.
- Make a call from the Cisco 7960 SIP Telephone to the Cisco 7940 SIP Telephone and verify that the call is successful.

- While the call is up, disconnect the T1 link on Cisco 3745 router. Verify that the connection stays up. The audio path is “IP-direct” between the two SIP phones. The disconnection of the T1 will not affect the local connection between the phones.
- To verify fail-over to the SRST router, leave the T1 link disconnected. Make a new phone call between the two Cisco SIP Telephones. Verify that the Cisco 3745 router handles the call locally using SRST capability. Verify that the SIP (300) re-direct message is shown on the phone that initiated the call.
- Make a call from the analog telephone to the Cisco 7960 SIP Telephone and verify that the call is successful.
- To verify “fall back” to the primary proxy, restore the T1 link and verify that the Cisco SIP Telephones re-register with the Avaya SES Server. Make a call between the two Cisco SIP Telephones; verify that the SIP trunk between the Avaya SES Server and Avaya Communication Manager is active while the call is up.

## 6. Conclusion

These Application Notes have provided the detailed configuration for SIP call routing among Avaya SES Server, Avaya Communication Manager, and a Cisco SRST enabled router. Avaya Communication Manager OPS features are available to Cisco SIP Telephones registered with the Avaya SES Server.

## 7. Additional References

Use URL <http://www.avaya.com> to access the following Application Notes and product documentation.

- [1] *Configuring SIP Call Routing among Avaya Converged Communications Server and Cisco SIP Telephony Network Devices – Issue 1.0*
- [2] *Configuring SIP IP Telephony Using Avaya Converged Communication Server, Avaya Communication Manager, and Polycom SoundPoint IP 500/600 SIP Telephones - Issue 1.0*
- [3] *Configuring SIP IP Telephony Using Avaya SIP Enablement Services, Avaya Communication Manager, and Cisco 7940/7960 SIP Telephones – Issue 1.0*
- [4] *Avaya SIP enablement Services Installation and Administration.*

Use URL <http://www.cisco.com> to access the following document:

- [5] *SIP Survivable Remote Site Telephony (SRST)*

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