Avaya Solution & Interoperability Test Lab

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.

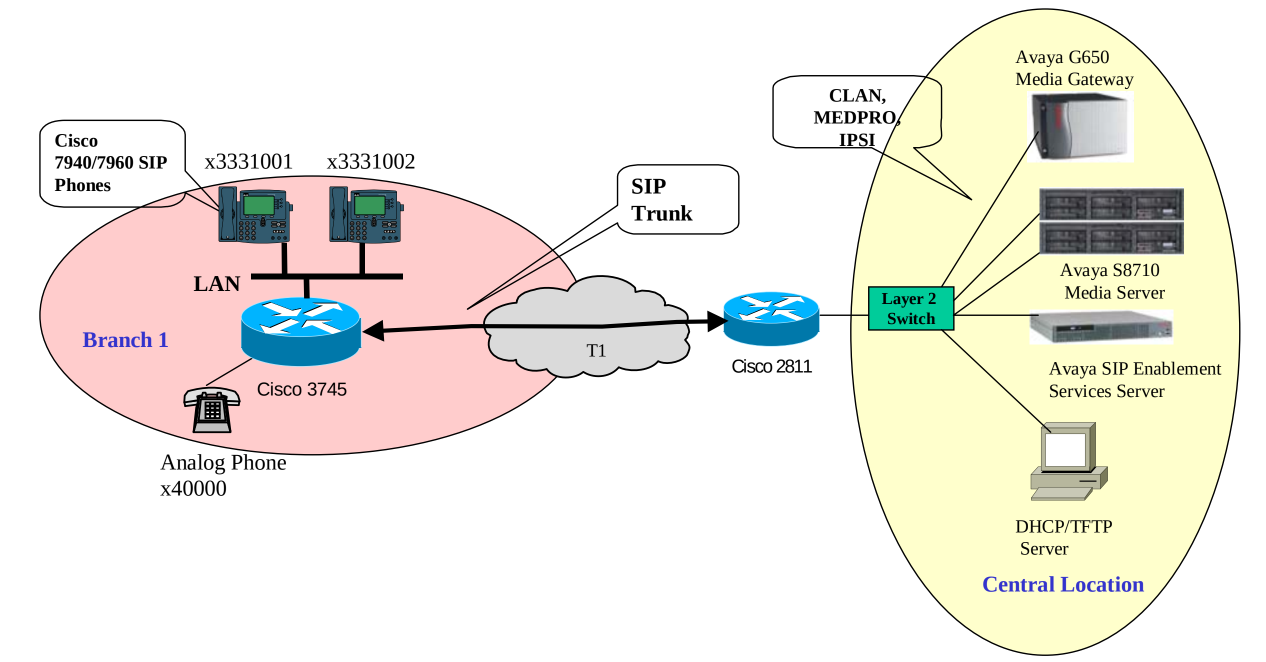


Figure 1: Network Configuration

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

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

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.

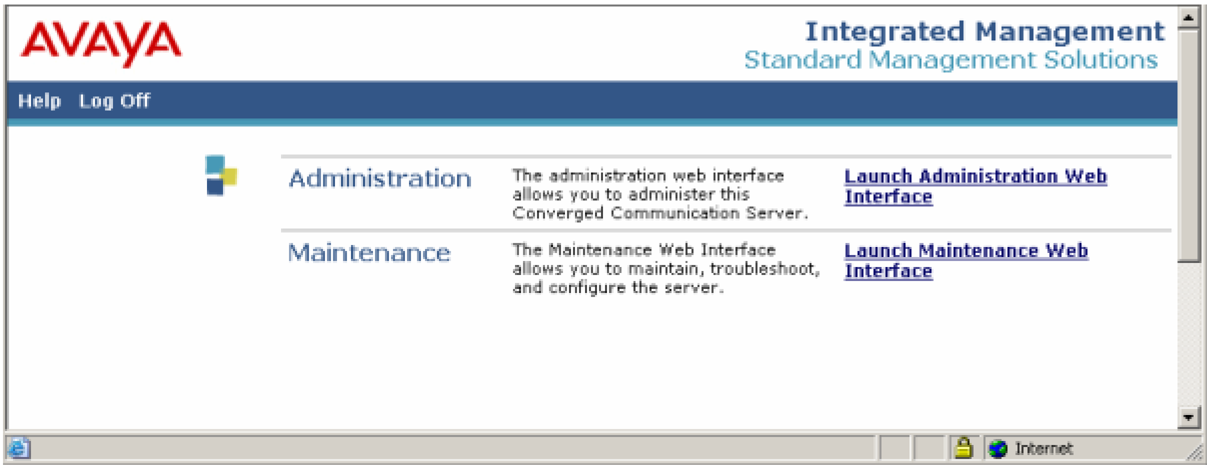
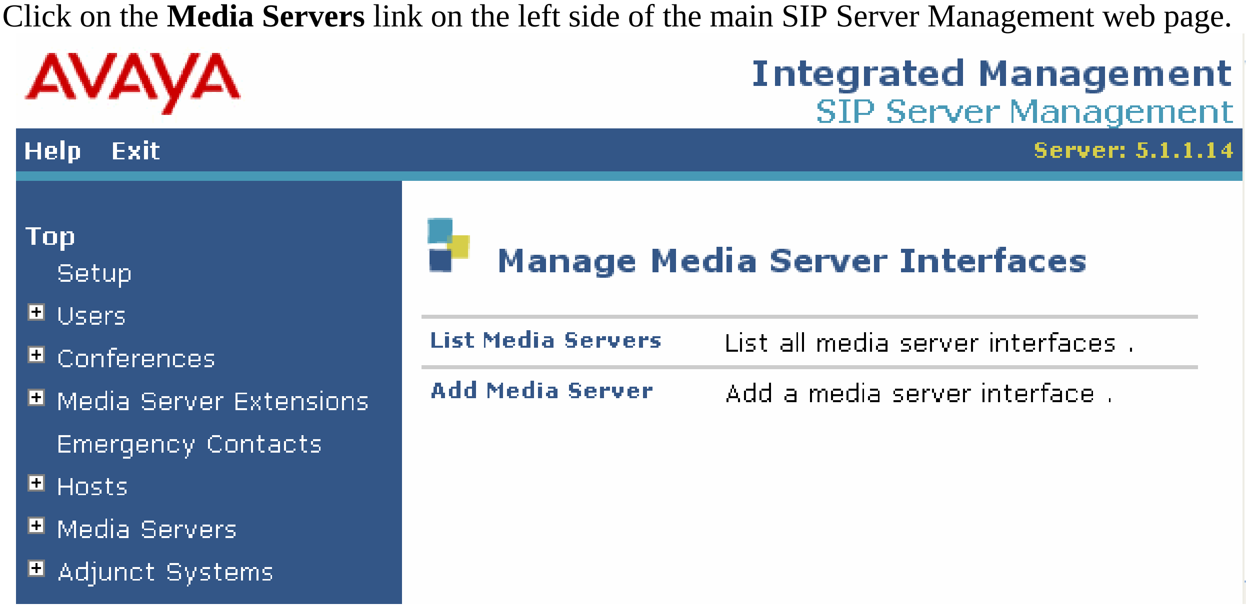
Equipment	Software
Avaya S8710 Media Server/G650 Media Gateway	Communication Manager 3.0 (Load 340.3)
Avaya SIP Enablement Services Server	R3.0
Cisco 7960/7940 SIP Telephones	V 7.4
Cisco 3745 Router	IOS 12.4(4)T
Cisco 2811 Router	IOS 12.3
DHCP/TFTP Server ▪ Windows 2003 Advanced Server	 Version 2003

Table 2: Equipment and Software Validated

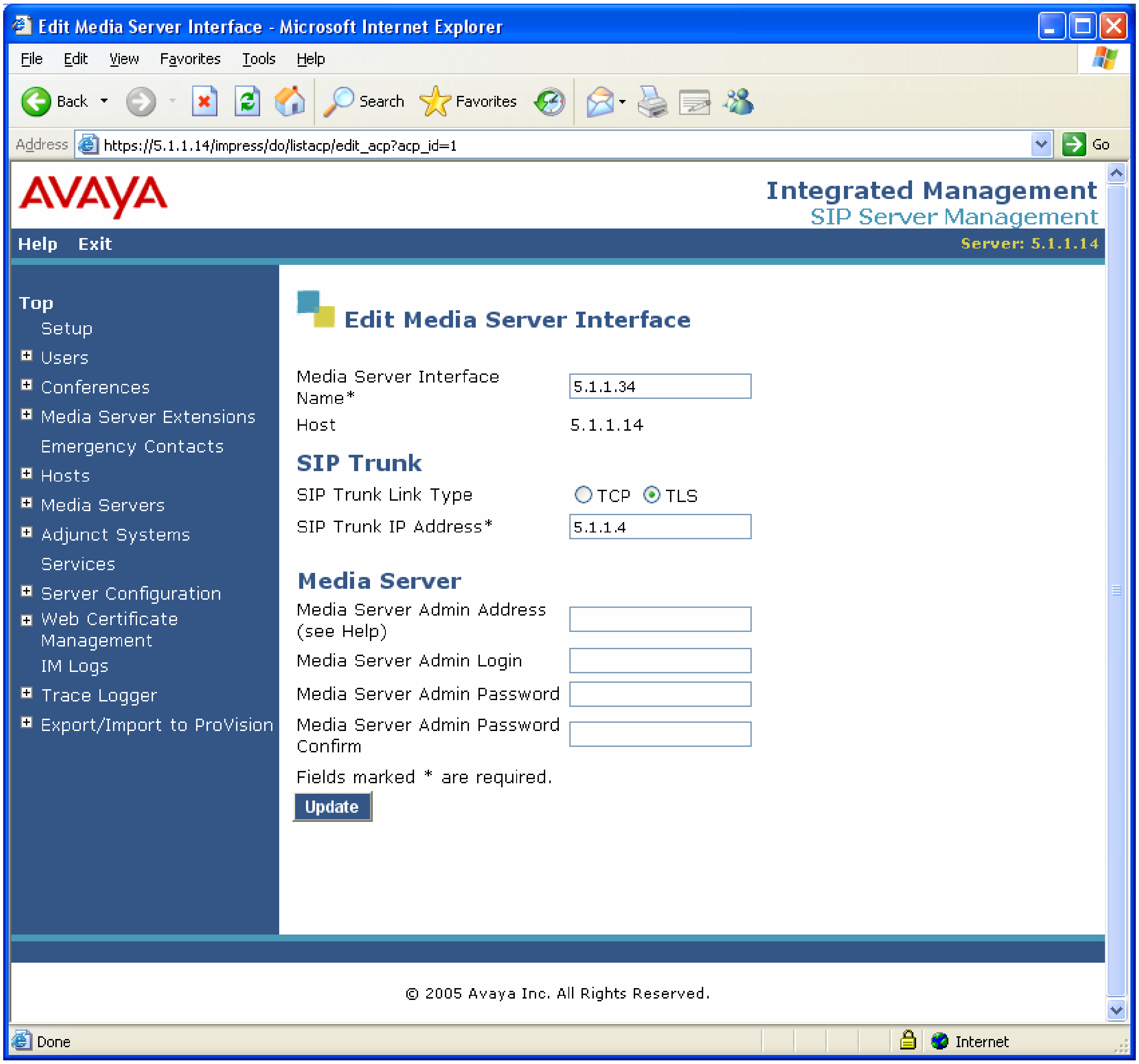
3. Configure Avaya SES Server

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
1.	**Log in to Avaya SES Server** • 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. • Click **Launch Administration Web Interface** as shown below. 
2.	**Add Media Server** Avaya Communication Manager must be configured as a Media Server on the Avaya SES Server. Click on the **Media Servers** link on the left side of the main SIP Server Management web page. 

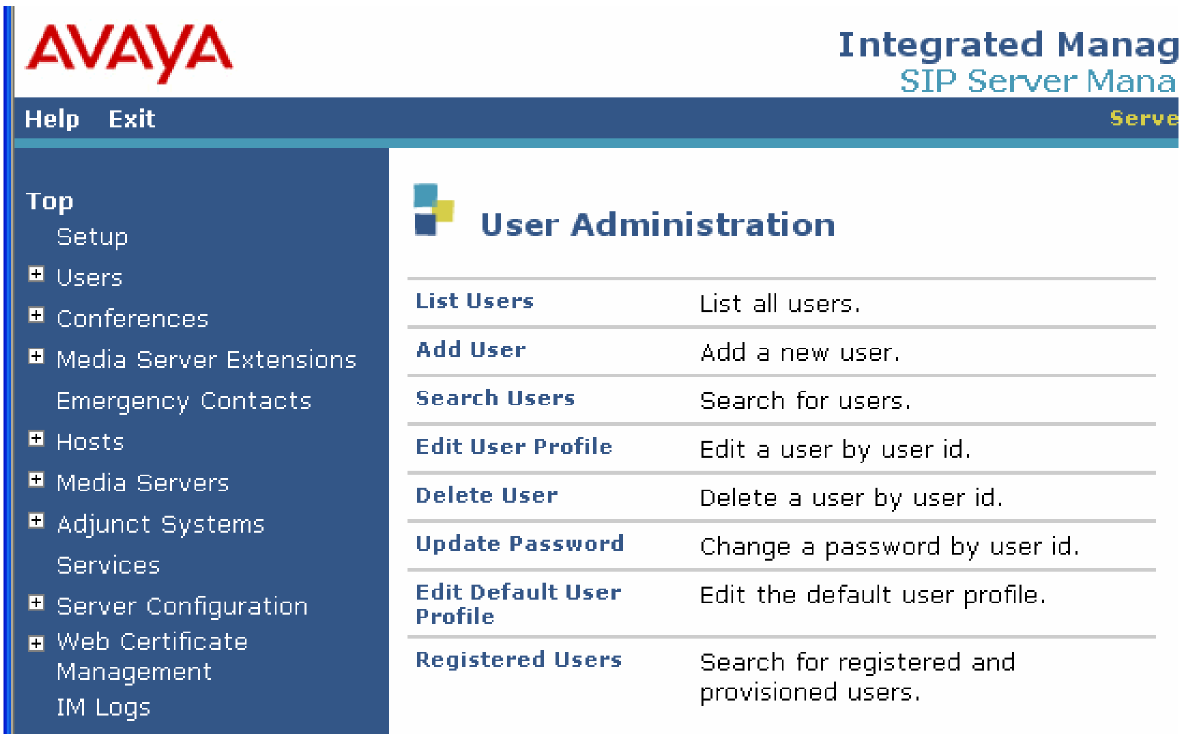
- Click on **Add Media Server** link under **Manage Media Server Interfaces**.
- Enter the S8710 active server's IP address **5.1.1.34**.
- Click on **TLS** as **SIP Trunk Link Type**.
- Enter C-LAN IP address **5.1.1.4** in **SIP Trunk IP Address** field.
- Click on **Update**.



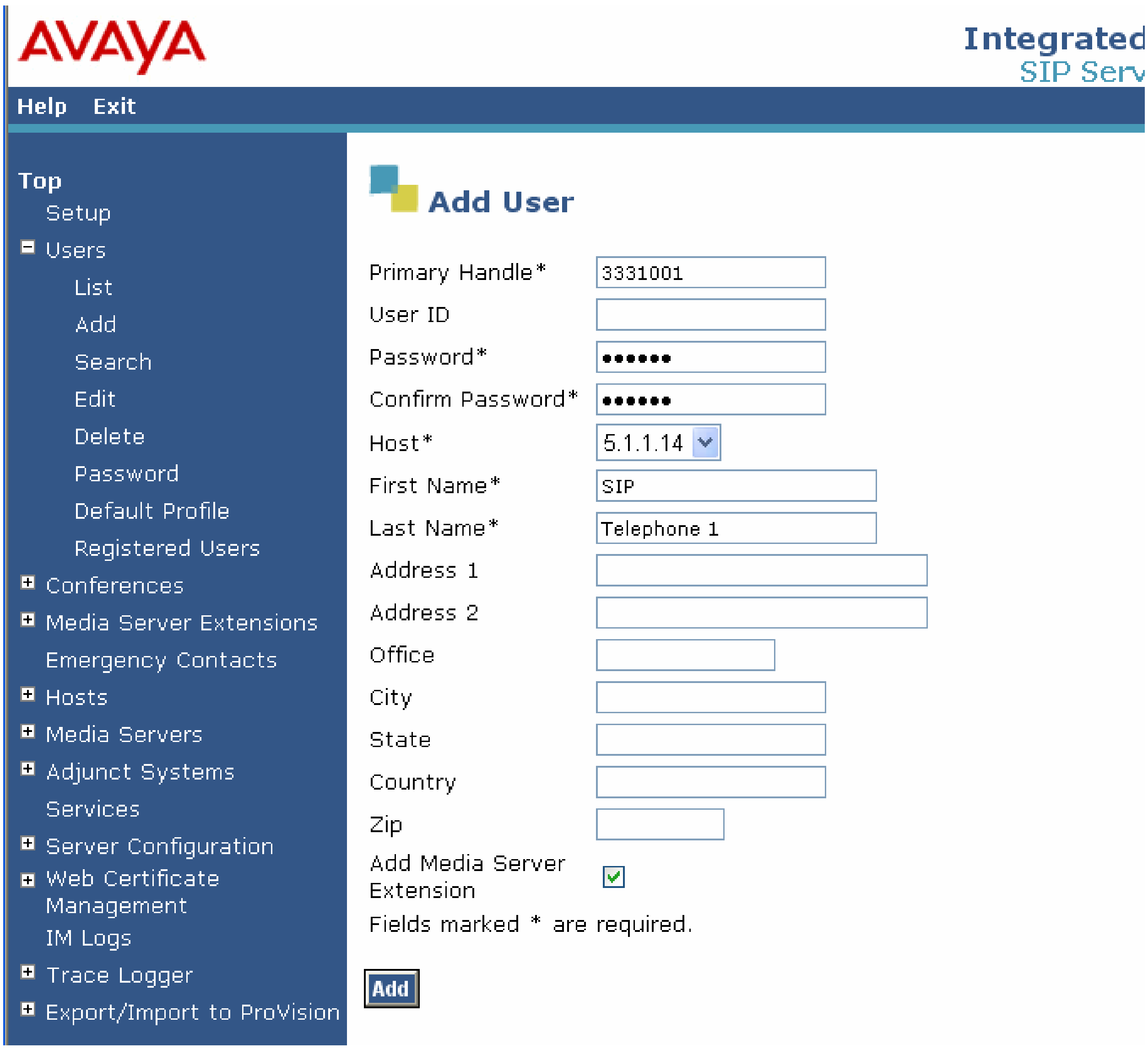
3. **Add Users**

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.

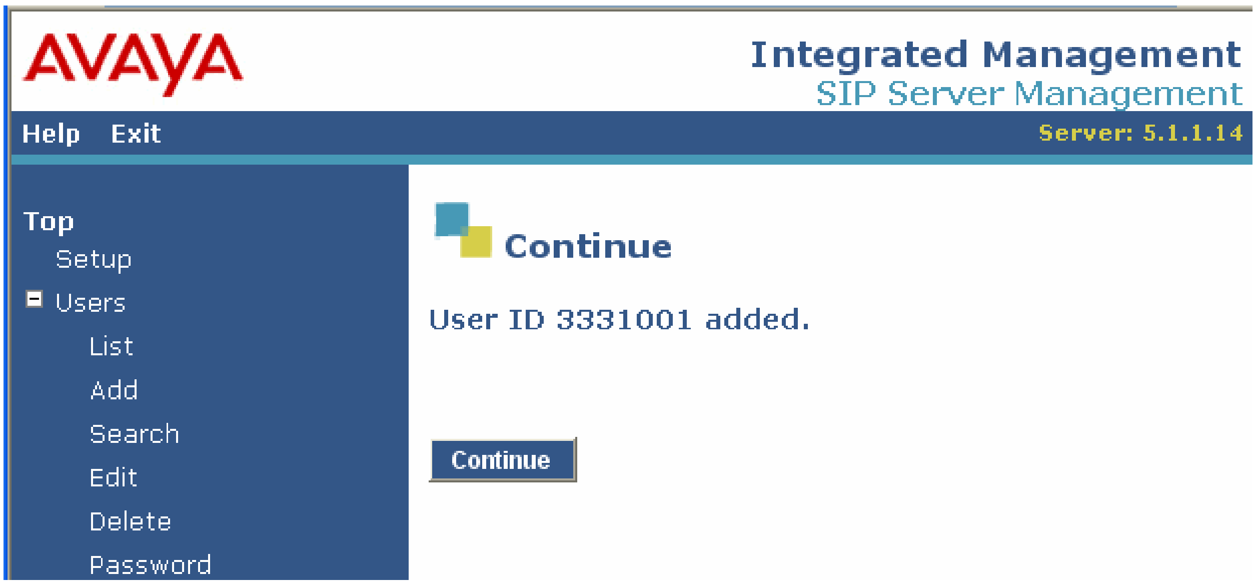
- Click on **Users** link on left side configuration panel.
- Click on **Add User** under **User Administration.**



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

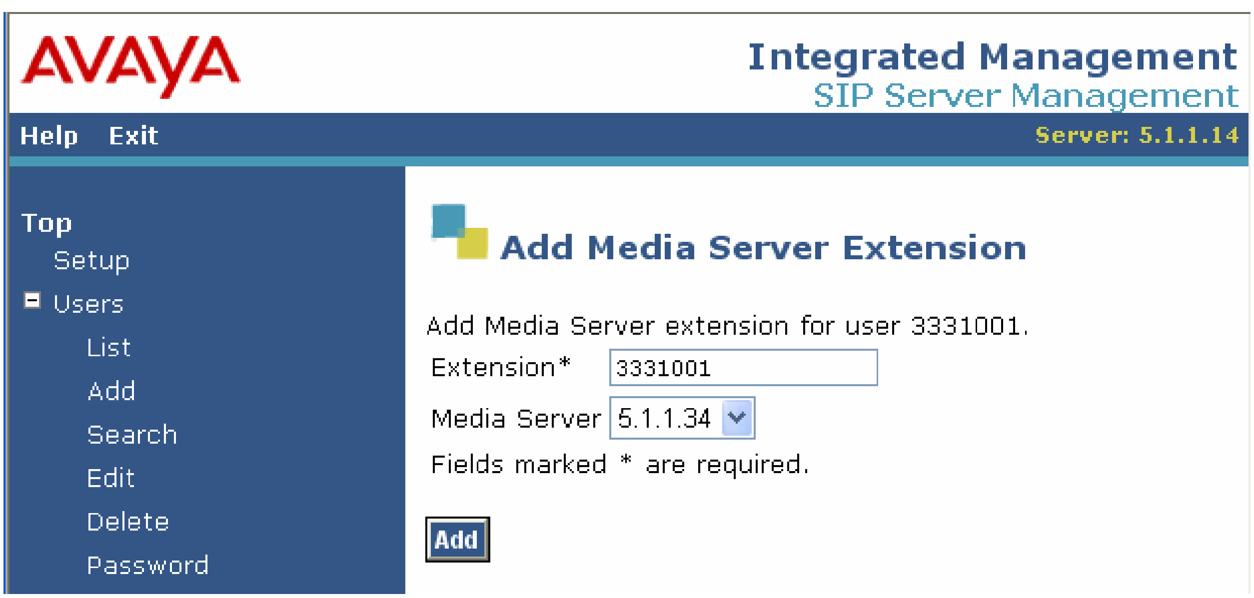


Click on **Continue.**



From **Add Media Server Extension** screen:

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



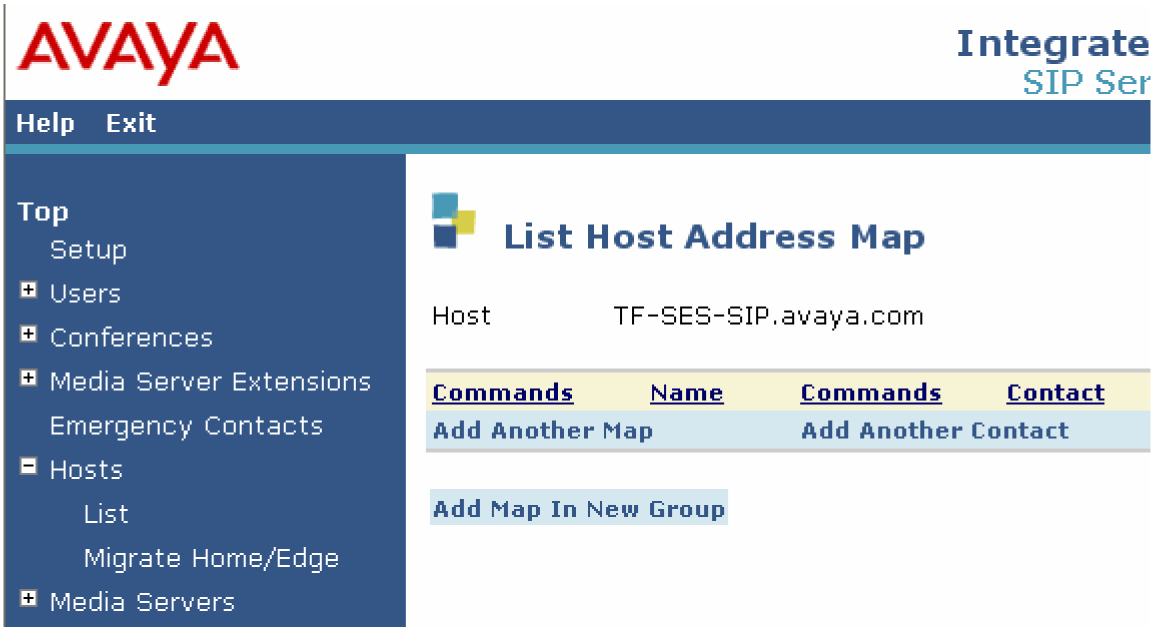
- Click **Continue.**

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

4. **Add Host Address Map**

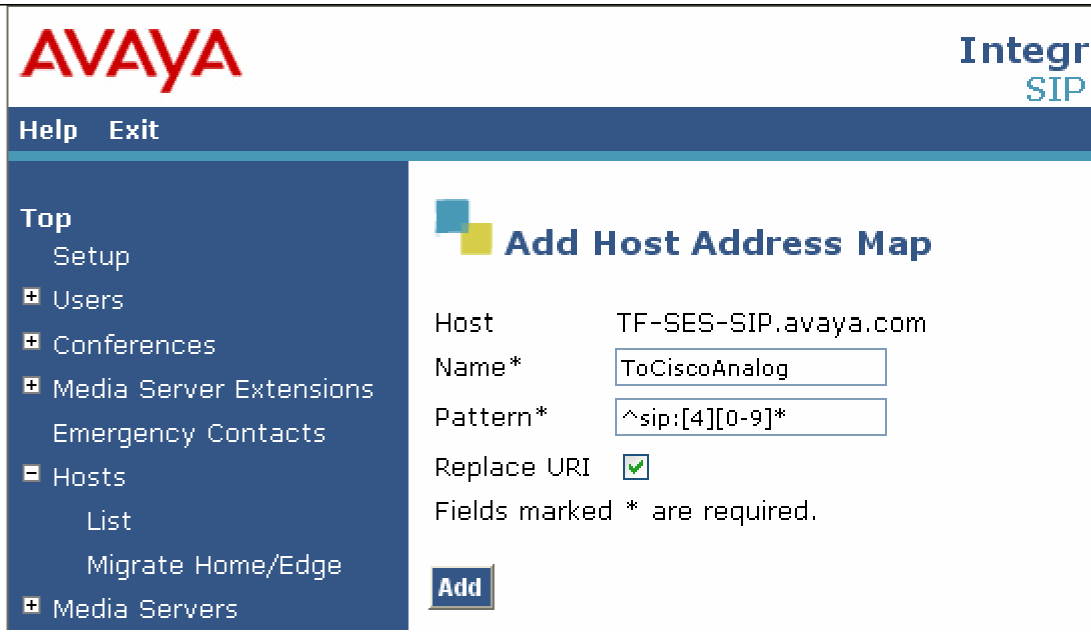
Address maps specify how incoming SIP calls are to be routed, based on the dialed number. The Avaya SES Server will route calls to Avaya Communication Manager. If the called station is a local registered station, Avaya Communication Manager will instruct the Avaya SES Server to ring the station. If the 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**.



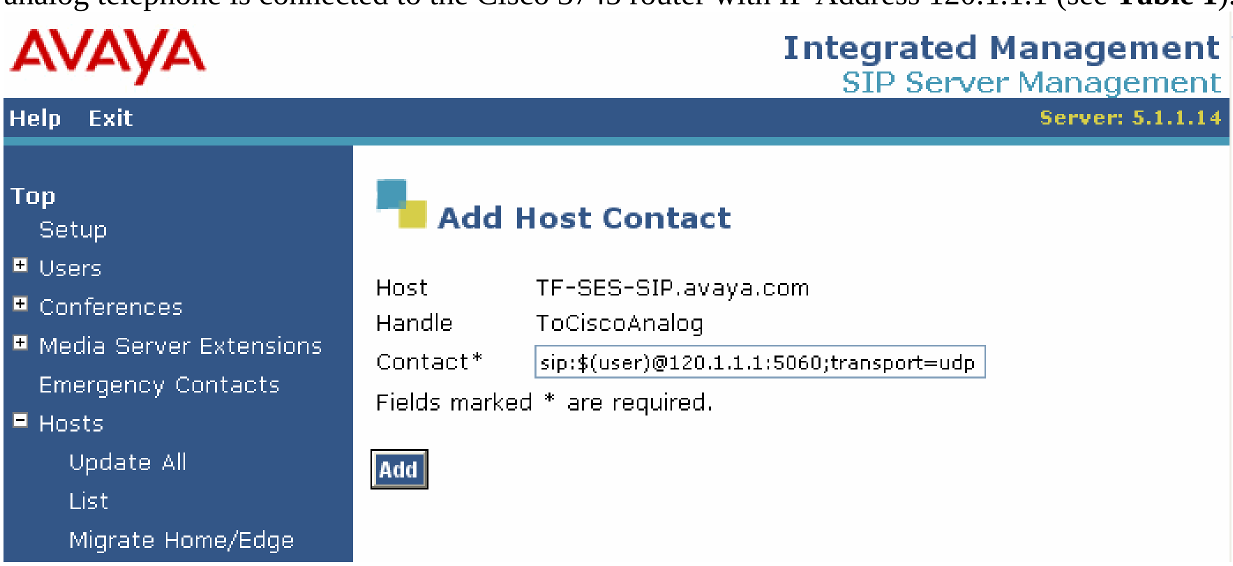
Since the analog telephone is using extension 40000, set the **Pattern** to match the leading digit "4" to route the call to the Cisco 3745 router.

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

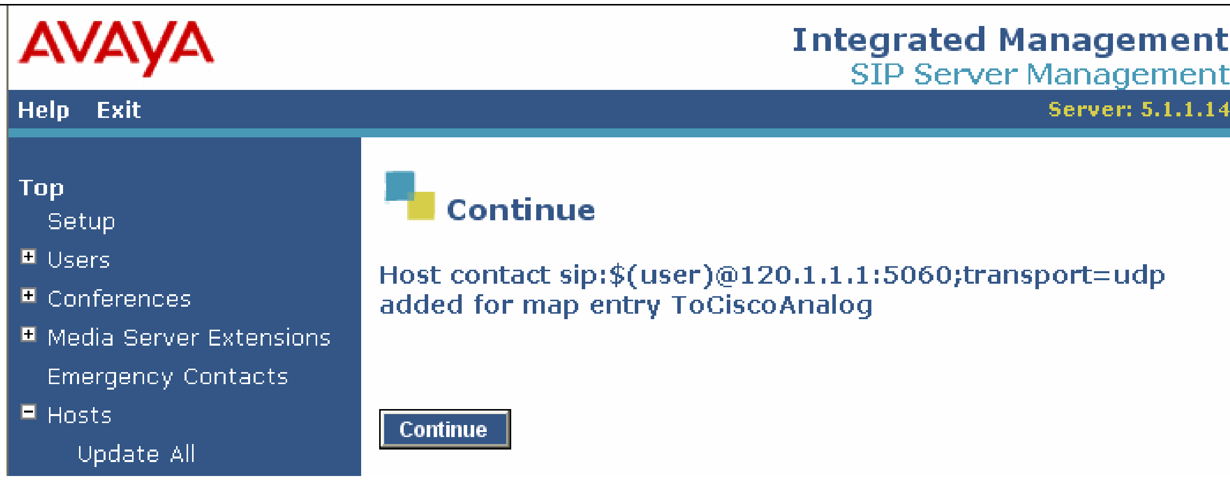


5. **Add Contact**

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**).



Click **Add**.



- 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.	From the Avaya Communication Manager SAT interface, use the **change node-names ip** command to add a node name for the Avaya SES Server.

```
change node-names ip                                          Page   1 of   1
                                  IP NODE NAMES
    Name                  IP Address               Name              IP Address
Avaya-SES               5  .1   .1   .14                              .    .    .
c-lan                   5  .1   .1   .8                               .    .    .
medpro                  5  .1   .1   .9                               .    .    .
```

Step	Description

2.	Use **add signaling-group** to add a signaling group to the Avaya SES Server, as shown below.
	`add signaling-group 1                                 Page   1 of   1`
	`SIGNALING GROUP`
	`Group Number: 1              Group Type: sip`
	`Transport Method: tls`
	`Near-end Node Name: c-lan                Far-end Node Name: Avaya-SES`
	`Near-end Listen Port: 5061              Far-end Listen Port: 5061`
	`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`

Step	Description
3.	Use the **add trunk group** command to add a SIP trunk between Avaya Communication Manager and the Avaya SES Server. The following screens illustrate the trunk configuration.

```
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
```

```
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
```

```
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
```

Step	Description

4.	As shown in **Figure 1**, there is an analog telephone connected to the Cisco 3745 router FXS port with extension 40000. Use the **change uniform-dialplan** command to enable Avaya Communication Manager to route a call to extension 40000 via its AAR table.
	<pre>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
	4 5 0 aar n n
	                              n n
	                              n n
	                              n n
	                              n n</pre>

5.	Use the **change route-pattern** 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.
	<pre>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 Mrk Lmt List Del Digits QSIG
	                             Dgts Intw
	 1: 1 0 n user
	 2: n user
	 3: n user</pre>

6.	Use the **change aar analysis** command to configure route pattern 1 for numbers that are five digits in length, starting with the digit 4.
	<pre>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
	    3 7 7 999 aar n
	    4 5 5 1 aar n
	    5 7 7 999 aar n
	    6 7 7 999 aar n</pre>

Step	Description

7.	To configure the Cisco SIP Telephones as OPS stations on Avaya Communication Manager, use the **change off-pbx-telephone station-mapping** command. The following screen shows an example for configuring OPS station mapping for extension 3331001.
	``
	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 Prefix Selection Set
	 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
	``
	
	For additional OPS configuration, refer to the Application Notes listed in Section 7.

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                          enable VoIP service on router
 allow-connections sip to sip               enable local call transfer
 redirect ip2ip                             enable redirect IP to IP call
 sip
  registrar server expires max 600 min 60
  redirect contact order best-match         enable VoIP service on router
!
voice class codec 15                        configure voice class codec pool for audio
 codec preference 1 g711ulaw                add G711 ulaw as first choice
 codec preference 2 g729br8                 add G729 as second choice

voice register pool  1                         create a voice register pool
 id network 120.1.1.0 mask 255.255.255.0    allow IP end points with IP address at range
                                            120.1.1.0 to register with router
 application session                        enable Application SIP
 preference 2
 proxy 5.1.1.14 preference 1 monitor probe icmp-ping    define Avaya SES (5.1.1.14)
                                                        as primary proxy.
 dtmf-relay rtp-nte
 voice-class codec 15                                   use voice-class 15 defined above

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                  set call fallback active for SRST to fallback to primary
```

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)*

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.