Avaya Solution & Interoperability Test Lab

Application Notes for the CounterPath eyeBeam Softphone with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Manager, Avaya SIP Enablement Services, and CounterPath Solutions eyeBeam Softphone. CounterPath eyebeam 1.5 is a SIP-based VoIP Softphone. During compliance testing, the eyeBeam Softphone successfully registered with Avaya SIP Enablement Services, placed and received calls to and from SIP and non-SIP telephones, and established conference calls. Information in these Application Notes has been obtained through compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

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1. Introduction

These Application Notes describe a solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services 3.1.1, and CounterPath Solutions eyeBeam Softphone. CounterPath eyeBeam1.5 is a SIP-based VoIP Softphone. During compliance testing, the eyeBeam Softphone successfully registered with Avaya SIP Enablement Services, placed and received calls to and from SIP and non-SIP telephones, and established conference calls.

Figure 1 illustrates a sample configuration consisting of a pair of Avaya S8710 Media Servers, an Avaya G650 Media Gateway, an Avaya SIP Enablement Services (SES) server, and the eyeBeam Softphone. Avaya Communication Manager is installed on the S8710 Media Servers. The solution described herein is also extensible to other Avaya Media Servers and Media Gateways. For completeness, Avaya 4600 Series SIP Telephones, Avaya one-X Desktop Edition, Avaya 4600 Series H.323 IP Telephones, and Avaya 6400 and 8400 Series Digital Telephones, are included in **Figure 1** to demonstrate calls between the SIP-based eyeBeam Softphone and Avaya SIP, H.323, and digital telephones. The analog PSTN telephone is also included to demonstrate calls routed by Avaya Communication Manager between the eyeBeam Softphone and the PSTN.

The eyeBeam Softphone originates a call by sending a call request (SIP INVITE message) to the Avaya SES server. The Avaya SES server routes the call over a SIP trunk to Avaya Communication Manager for origination services. If the call is destined for another local SIP endpoint, such as another eyeBeam Softphone or an Avaya SIP telephone/softphone, then Avaya Communication Manager routes the call back over the SIP trunk to the Avaya SES server, which delivers the call to the destination SIP telephone/softphone. Otherwise, Avaya Communication Manager routes the call to the PSTN, a local Avaya H.323, digital, or analog telephone, an adjunct, a vector, a hunt group, etc., depending on the destination number. For a call arriving to Avaya Communication Manager that is destined for the eyeBeam Softphone, Avaya Communication Manager routes the call over the SIP trunk to the Avaya SES server, which delivers the call to the eyeBeam Softphone.

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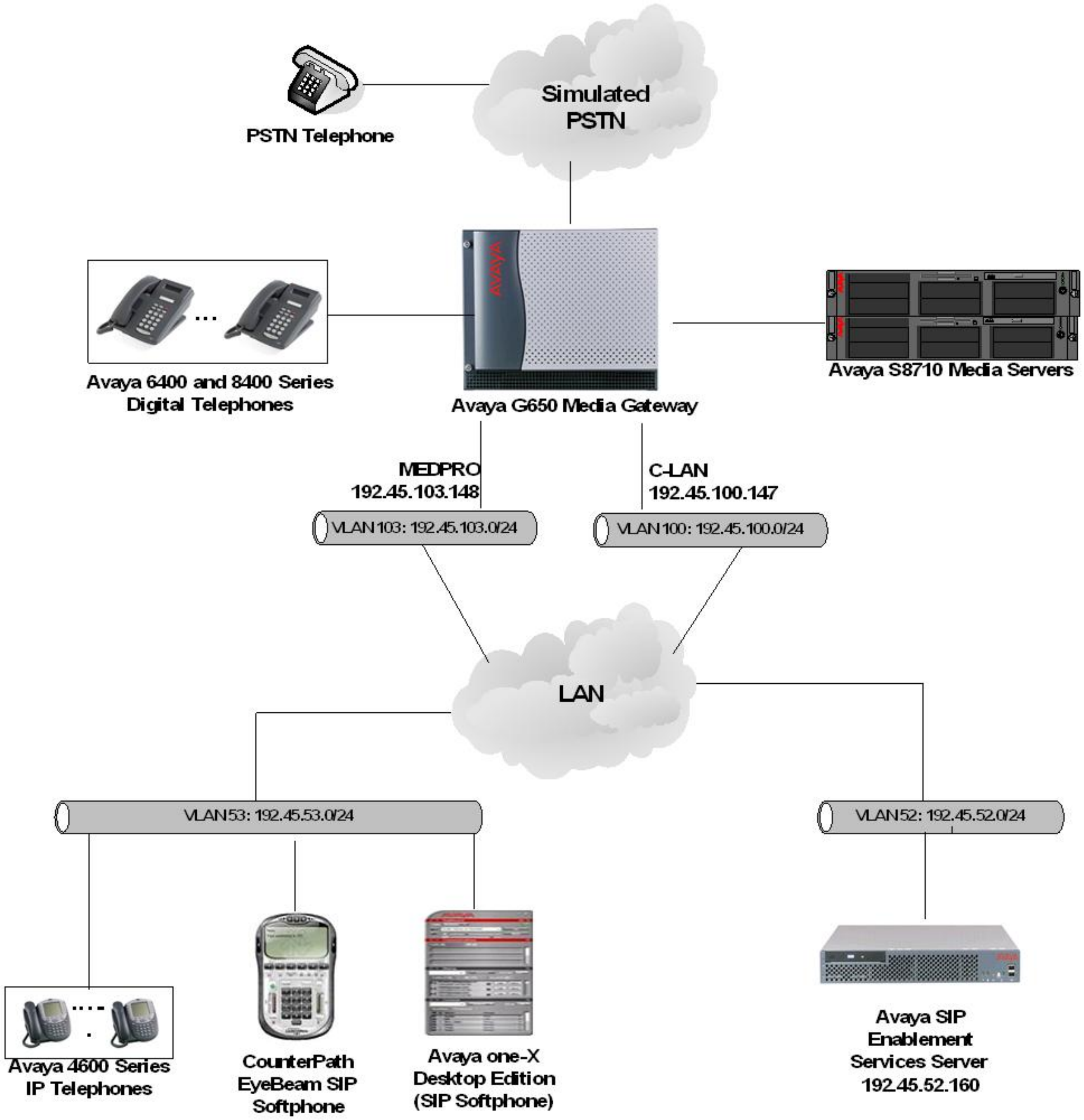

Figure 1: Sample configuration

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2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8710 Media Server	Avaya Communication Manager 3.1.2 (R013x.01.2.632.1)
Avaya G650 Media Gateway	-
TN2312BP IP Server Interface	HW12 FW 31
TN799DP C-LAN Interface	HW01 FW 17
TN2302AP IP Media Processor	HW20 FW 112
Avaya SIP Enablement Services Server	SES 3.1.1(R03.1.1-03.1.114.0)
Avaya 4600 Series IP Telephones	2.3 (4602SW H.323) 2.5 (4625SW H.323) 2.2.2 (4610SW SIP)
Avaya 6400 and 8400 Series Digital Telephones	-
Avaya one-X SIP Softphone	2.1, Build 70
CounterPath eyeBeam SIP Softphone	1.5.13 Build 37982
Analog Telephone	-

3. Configure Avaya Communication Manager

This section describes the steps for configuring IP codec sets and associating SIP telephone numbers with off-PBX telephone stations in Avaya Communication Manager. The steps are performed from the Avaya Communication Manager System Access Terminal (SAT) interface. IP codec sets identify the codecs that may be used in calls involving VoIP telephones. An off-PBX telephone is a phone that Avaya Communication Manager does not control, such as a cellular phone, a home telephone, or a SIP telephone. However, Avaya Communication Manager features and calling privileges can be applied to an off-PBX telephone by associating a local, on-PBX extension with the off-PBX telephone. This approach is taken for SIP Telephones that register with the Avaya SES server and intend to use Avaya Communication Manager for call origination and termination services. Similarly, on the Avaya SES server, the number of the SIP telephone is administratively associated with the extension of the on-PBX station. Throughout the rest of this document, on-PBX stations associated with SIP Telephones in such a manner will be referred to as Outboard Proxy SIP (OPS) stations.

Configuration in the following sections is only for the fields where a value needs to be entered or modified. Default values are used for all other fields.

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3.1. Capacity Verification

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">1.</td><td>Enter the display system-parameters customer-options command. Verify that there are sufficient Maximum Off-PBX Telephones – OPS licenses. If not, contact an authorized Avaya account representative to obtain additional licenses.</td></tr>
<tr><td><pre>
display system-parameters customer-options   Page   1 of 10
                                OPTIONAL FEATURES

     G3 Version: V13
       Location: 1   RFA System ID (SID): 1
       Platform: 8   RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 44000 908
                                      Maximum Stations: 36000 410
                              Maximum XMOBILE Stations: 0   0
                    Maximum Off-PBX Telephones - EC500: 5   0
                    Maximum Off-PBX Telephones -   OPS: 200   50
                    Maximum Off-PBX Telephones - SCCAN: 0   0
</pre></td></tr>
<tr><td rowspan="2">2.</td><td>Proceed to Page 2 of OPTIONAL FEATURES form. Verify that the number of SIP trunks supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses.

Note: Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted.</td></tr>
<tr><td><pre>
display system-parameters customer-options   Page   2 of 10
                                OPTIONAL FEATURES

IP PORT CAPACITIES   USED
                     Maximum Administered H.323 Trunks: 200   148
           Maximum Concurrently Registered IP Stations: 1000 2
             Maximum Administered Remote Office Trunks: 0   0
Maximum Concurrently Registered Remote Office Stations: 0   0
              Maximum Concurrently Registered IP eCons: 0   0
  Max Concur Registered Unauthenticated H.323 Stations: 0   0
                  Maximum Video Capable H.323 Stations: 0   0
                   Maximum Video Capable IP Softphones: 0   0
                       Maximum Administered SIP Trunks: 200   153

   Maximum Number of DS1 Boards with Echo Cancellation: 0   0
                             Maximum TN2501 VAL Boards: 1   1
                    Maximum G250/G350/G700 VAL Sources: 0   0
           Maximum TN2602 Boards with 80 VoIP Channels: 2   0
          Maximum TN2602 Boards with 320 VoIP Channels: 2   1
   Maximum Number of Expanded Meet-me Conference Ports: 0   0
        (NOTE: You must logoff & login to effect the permission changes.)
</pre></td></tr>
</table>

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3.2. IP Codec Set

This section describes the steps for administering a codec set in Avaya Communication Manager. This codec set is used in the IP Network Region for communications between Avaya Communication Manager and Avaya SES.

Step	Description
1.	Enter the **change ip-codec-set <c>** command, where **c** is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 3.3** for configuring IP network region to specify which codec sets may be used within and between network regions. For the compliance testing, **G.711MU** and **G.729AB** were used and **Media Encryption** was set to **none** as encryption is currently not supported for SIP telephony.

```
change ip-codec-set 2                                          Page   1 of 2

                          IP Codec Set

    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2: G.729AB           n         2        20
 3:
 4:
 5:
 6:
 7:

     Media Encryption
 1: none
 2:
 3:
```


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3.3. IP Network Region

This section describes the steps for administering an IP Network Region in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SIP Enablement Services.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Enter the change ip-network-region <n> command, where n is a number between 1 and 250 inclusive and configure the following:

Authoritative Domain – Set to the devconnect.com. This should match the SIP Domain value in Section 4, Step 2.
Intra-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio connectivity between endpoints registered to Avaya Communication Manager or Avaya SES in the same IP network region.
Codec Set – Set the codec set number as provisioned in Section 3.2.
Inter-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio connectivity between endpoints registered to Avaya Communication Manager or Avaya SES in different IP network regions.

<pre>
change ip-network-region 2   Page 1 of 19
                               IP NETWORK REGION
  Region: 2
Location: Authoritative Domain: devconnect.com
    Name:
MEDIA PARAMETERS   Intra-region IP-IP Direct Audio: yes
      Codec Set: 2   Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048   IP Audio Hairpinning? y
   UDP Port Max: 65535
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
</pre></td></tr>
<tr><td>2.</td><td>Proceed to Page 3 of IP network region configuration and enable inter-region connectivity between regions as per below. For this compliance testing, codec set was set to the IP codec set configured in Section 3.2.
<pre>
Page 3 of 19
                     Inter Network Region Connection Management

 src dst codec direct   Total   Video   Dyn
 rgn rgn   set   WAN WAN-BW-limits WAN-BW-limits Intervening-regions CAC IGAR
 2   1   2   y   :NoLimit   n
 2   2   2
 2   3
 2   4
</pre></td></tr>
</table>

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3.4. IP Node Names

This section describes the steps for setting IP node name for Avaya SES in Avaya Communication Manager.

Step	Description
1.	Enter the **change node-names ip** command and administer settings as per below. • Add a node name for Avaya SES along with the IP address. **Note:** Verify that node-names are configured for the CLAN and MEDPRO boards.
	(screen below)

```
change node-names ip                                           Page 1 of 1
                                 IP NODE NAMES
    Name              IP Address
CLAN-1A06           192.45 .100.147
MEDPRO-1A13         192.45 .103.148
SES                 192.45 .52 .160
```

3.5. SIP Signaling

This section describes the steps for administering a signaling group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SIP Enablement Services.

Step	Description
1.	Enter the **add signaling-group <s>** command, where **s** is an available signaling group and configure the following: • **Group Type –** Set to **sip.** • **Transport Method –** Set to **tls**. • **Near-end Node Name** - Set to CLAN name as displayed in **Section 3.4**. • **Far-end Node Name** - Set to Avaya SES name configured in **Section 3.4**. • **Far-end Network Region** - Set to the region configured in **Section 3.3**. • **Far-end Domain** - Set to **devconnect.com**. This should match the **SIP Domain** value in **Section 4**, **Step 2**.

```
add signaling-group 10                                     Page    1  of   5
                                SIGNALING GROUP

 Group Number: 10             Group Type: sip
                        Transport Method: tls






   Near-end Node Name: CLAN-1A06            Far-end Node Name: SES
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 2
       Far-end Domain:devconnect.com

                                          Bypass If IP Threshold Exceeded? n

        DTMF over IP: rtp-payload          Direct IP-IP Audio Connections? y
                                                     IP Audio Hairpinning? n
 Session Establishment Timer (min): 120
```


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3.6. SIP Trunking

This section describes the steps for administering a trunk group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Enter the add trunk-group <t> command, where t is an unallocated Trunk Group and configure the following:

Group Type – Set to same value as Group Type configured in Section 3.5.
TAC (Trunk Access Code) – Set to any number with 1-4 digits; * and # may be used as first digit only.
Signaling Group – Set to same value as Group Number configured in Section 3.5.
Number of Members – Set to a value between 0 and 255.
Group Name – Enter any descriptive name.

Note: Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted.
<pre>
add trunk-group 10   Page   1 of 21
                                TRUNK GROUP

Group Number: 10   Group Type: sip   CDR Reports: y
  Group Name: SIP-SES-DevCon1   COR: 1   TN: 1   TAC: 110
   Direction: two-way   Outgoing Display? n
 Dial Access? n   Night Service:
Queue Length: 0
Service Type: tie   Auth Code? n

                                                       Signaling Group: 10
                                                     Number of Members: 150
</pre>
</td></tr>
</table>

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3.7. SIP Stations

This section describes the steps for administering OPS stations in Avaya Communication Manager and associating the OPS station extensions with the telephone numbers of the eyeBeam Softphones.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Enter the add station <s> command, where s is an available extension in the dial plan, to administer an OPS station. On Page 1 of the STATION form, configure the following:

Type – Set to 6408D+.
Port – Set to X for Administration Without Hardware (AWOH) as SIP stations are not directly connected to Avaya Communication Manager.
Name – Enter any descriptive name.
IP Softphone – Set to y.

<pre>
add station 54008   Page   1 of   3
                                     STATION

Extension: 54008   Lock Messages? n   BCC: 0
     Type: 6408D+   Security Code: TN: 1
     Port: X   Coverage Path 1:   COR: 1
     Name: EYEBEAM   Coverage Path 2:   COS: 1
                                        Hunt-to Station:

STATION OPTIONS
              Loss Group: 2   Personalized Ringing Pattern: 1
             Data Module? n   Message Lamp Ext: 54008
            Speakerphone: 2-way   Mute Button Enabled? y
        Display Language: english

                                                   Media Complex Ext:
                                                       IP SoftPhone? y
</pre></td></tr>
<tr><td>2.</td><td>Proceed to Page 3 of the STATION form and add the required number of call-appr entries in BUTTON ASSIGNMENT field. The number of call appearances should match the Call Limit field value in Step 4.
<pre>
add station 54008   Page   3 of   3
                                     STATION
SITE DATA
      Room: Headset? n
      Jack: Speaker? n
     Cable: Mounting: d
     Floor: Cord Length: 0
  Building: Set Color:

ABBREVIATED DIALING
     LIST1:   List2:   List3:

BUTTON ASSIGNMENTS
 1: call-appr   5:
 2: call-appr   6:
 3:                                     7:
 4:                                     8:
</pre></td></tr>
</table>

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Step	Description
3.	Enter the **add off-pbx-telephone station-mapping** command and configure the following: • **Station Extension –** Set the extension of the OPS station as configured above. • **Application** – Set to **OPS**. • **Phone Number –** Enter the number that the eyeBeam Softphone will use for registration and call termination. In the example below, the **Phone Number** is the same as the **Station Extension**, but is not required to be the same. • **Trunk Selection –** Set to the trunk group number configured in **Section 3.6**.

	<pre>add off-pbx-telephone station-mapping 54008 Page 1 of 2
	                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
	
	 Station Application Dial Phone Number Trunk Configuration
	 Extension Prefix Selection Set
	 54008 OPS - 54008 10 1</pre>

4.	Proceed to **Page 2** of station mapping form and verify that the **Call Limit** field value matches the number of call appearances configured in **Step 2**.

	<pre>add off-pbx-telephone station-mapping 54008 Page 2 of 2
	                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
	
	 Station Call Mapping Calls Allowed Bridged
	 Extension Limit Mode Allowed Calls
	
	 54008 2 both all both 1</pre>

5.	Repeat **Steps 1 - 4** as necessary to administer additional OPS stations and associations for eyeBeam Softphones.

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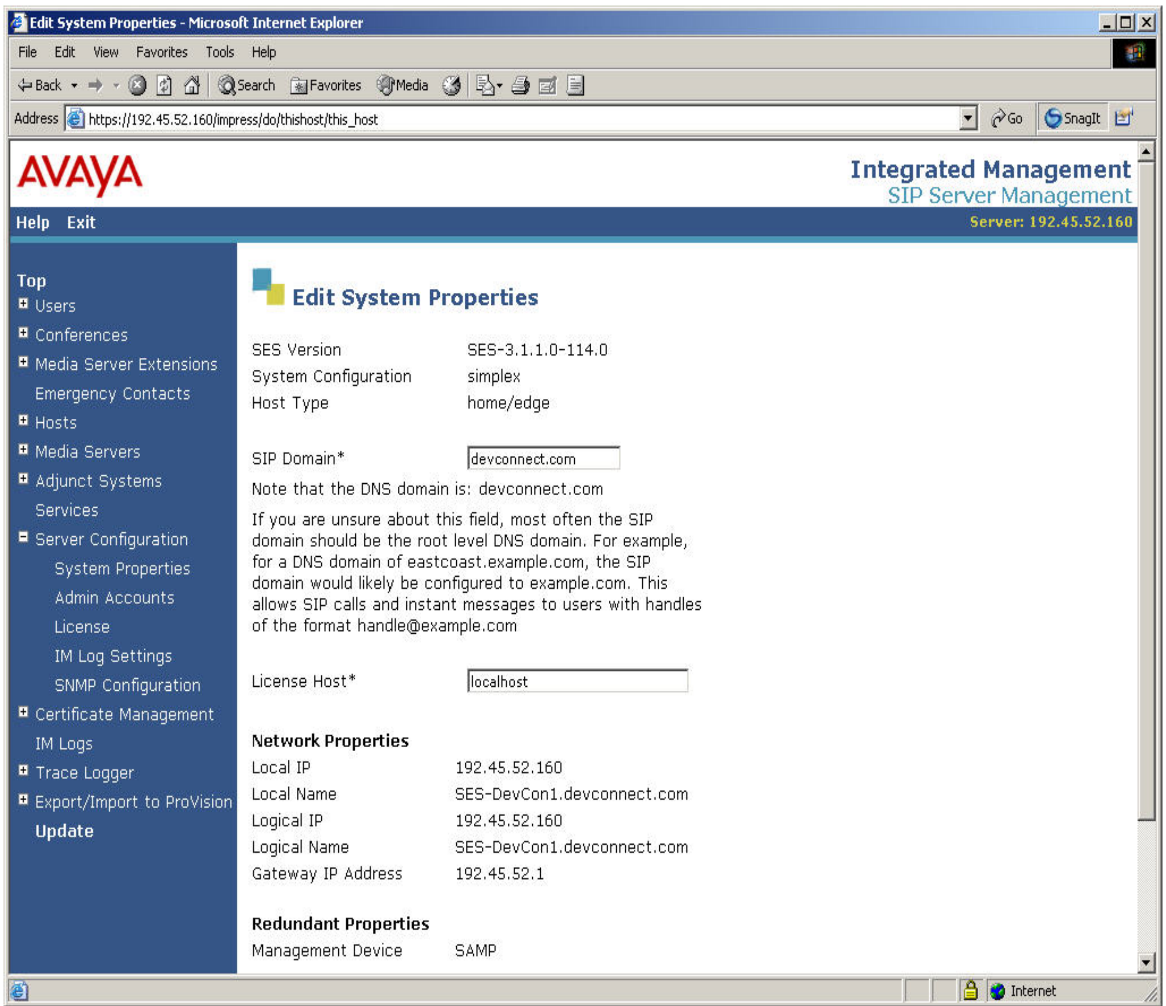
4. Configure Avaya SIP Enablement Services

This section describes the steps for creating SIP user accounts in Avaya SIP Enablement Services (SES) and associating the SIP users with an Avaya Communication Manager OPS station extension. The eyeBeam Softphone will register with Avaya SES using the SIP user accounts.

Configuration in the following steps is only for the fields where a value needs to be entered or modified. Default values are used for all other fields

Step	Description
1.	Open a web browser, enter http://<IP address of Avaya SES server>/admin for the URL, and log in with the appropriate credentials. Click on the "**Launch Administration Web Interface**" link upon successful login.
2.	From the **Administration Web Interface**: • Click the **+** sign to expand the options under **Server Configuration**. • Click **System Properties**. • Verify the **SIP Domain** matches the **Far-end Domain** configured for the signaling group on Avaya Communication Manager in **Section 3.5**.



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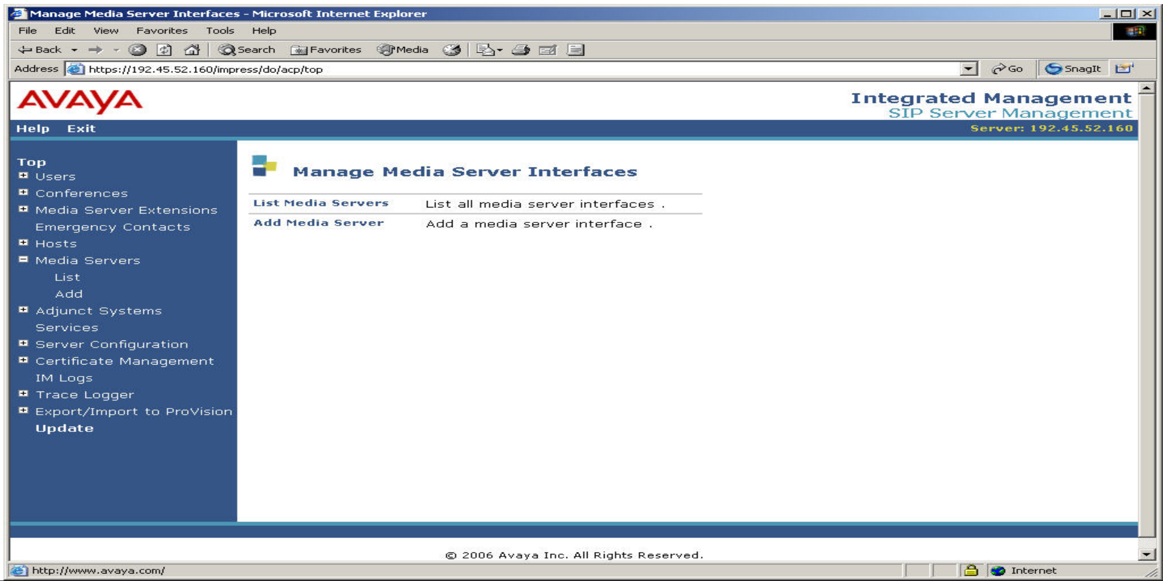
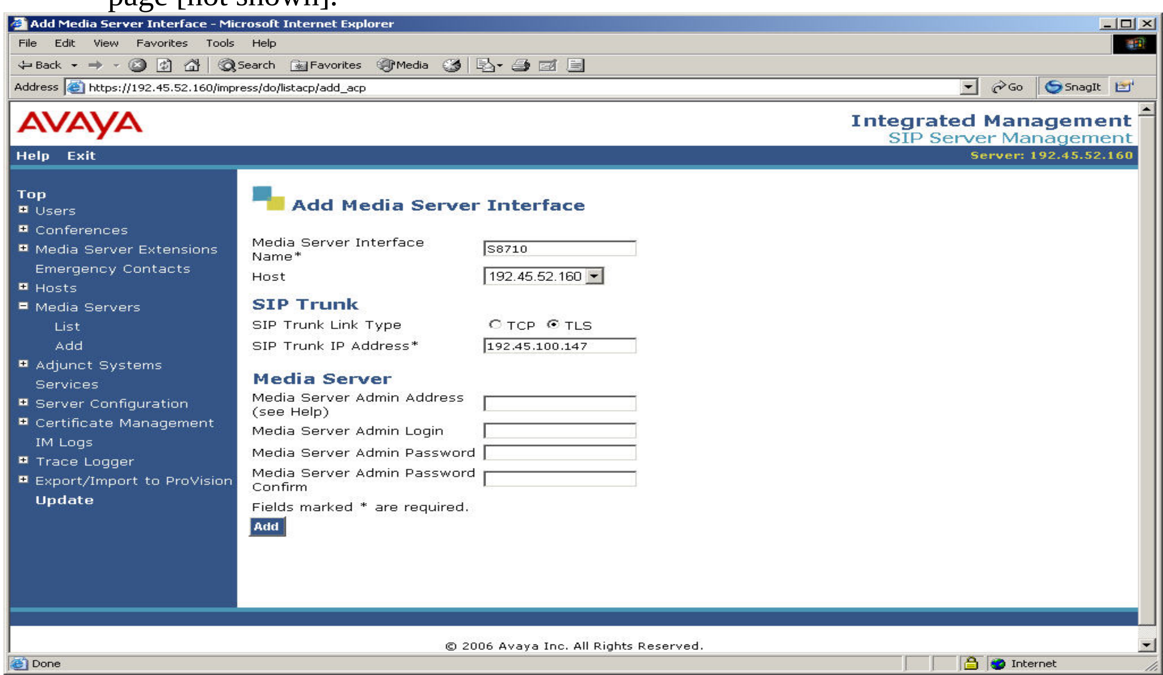
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>
To enable secure SIP trunking between Avaya SES and Avaya Communication Manager, add a Media Server corresponding to Avaya Communication Manager from the Administration Web Interface:

Click the + sign to expand the options under Media Servers.
Click Add.



</td></tr>
<tr><td>4.</td><td>
At the Add Media Server Interface page, provision SIP Trunk parameters as follows for connectivity to Avaya Communications Manager:

Media Server Interface Name – Enter a descriptive name of the Avaya Communication Manager.
SIP Trunk Link Type - Set to the Transport Method field value in Section 3.5.
SIP Trunk IP Address - Set to the CLAN IP address as displayed in Section 3.4.
Click the Add button when finished and then click Continue on the confirmation page [not shown].



</td></tr>
</table>

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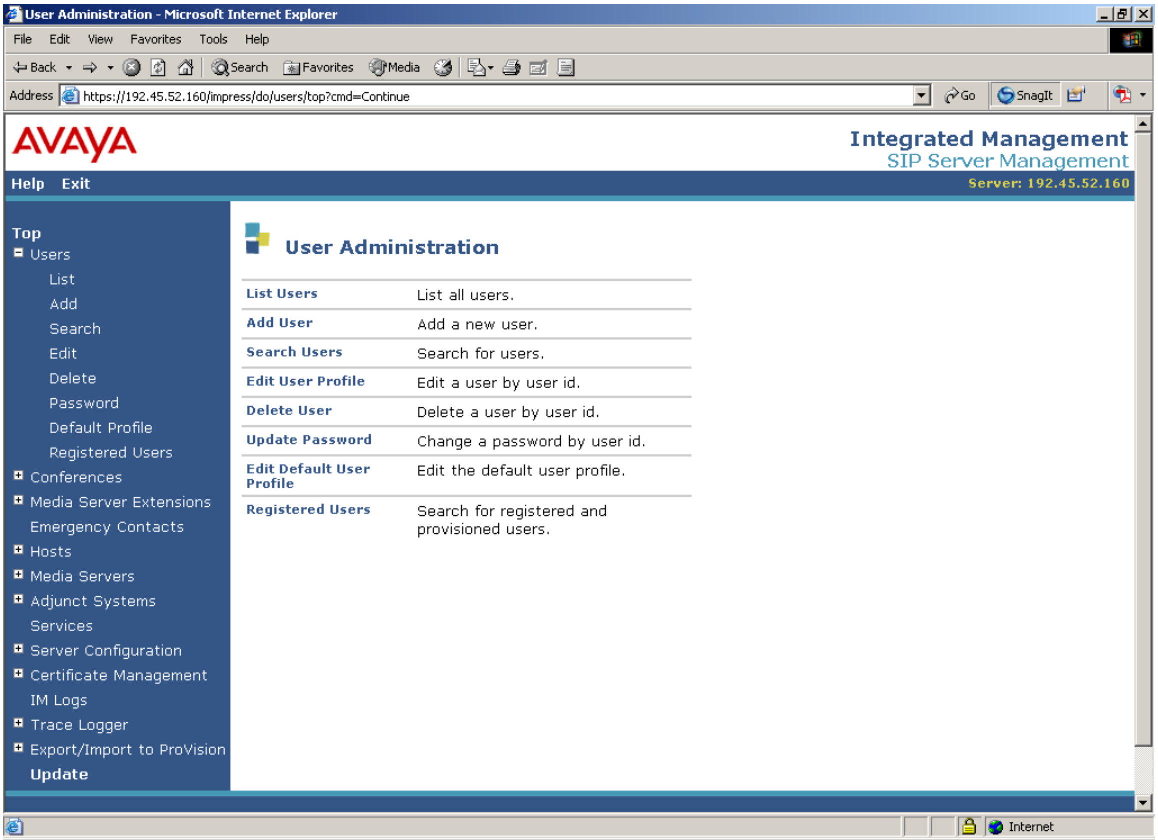
Step	Description
5.	In the left pane of the SES Administration Web Interface, expand **Users** and click **Add**.



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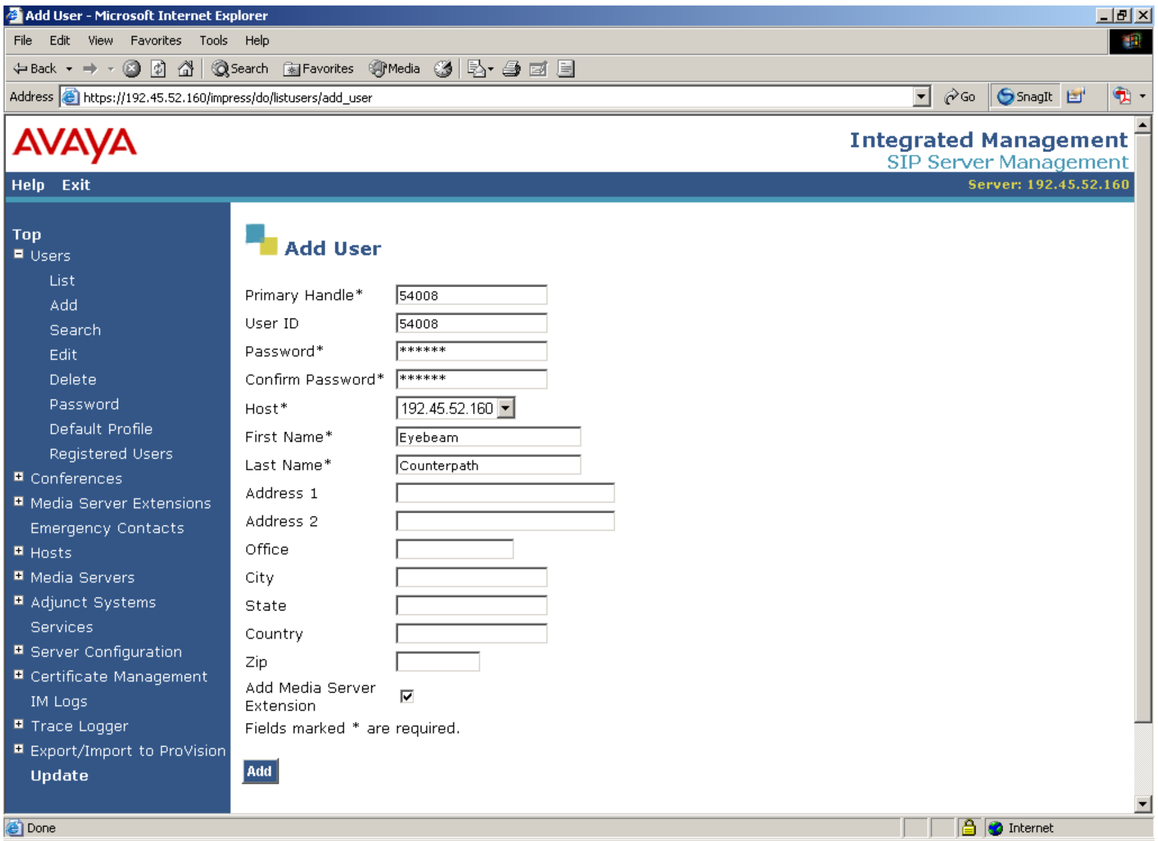
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>

At the **Add User** page, configure the following:

- **Primary Handle –** Enter the phone number of the eyeBeam Softphone. The number must match the **Phone Number** field value entered in **Section 3.7**, **Step 2**.
- **Password** and **Confirm Password –** Specify a password that the eyeBeam Softphone must use to successfully register with Avaya SES.
- **Host –** Select the IP address or Fully Qualified Domain Name (FQDN) of the Avaya SES server.
- **First Name** and **Last Name –** Enter descriptive names.
- Check the **Add Media Server Extension** checkbox.
- Click **Add**.
- Click **Continue** on the next page [not shown].



</td></tr>
</table>

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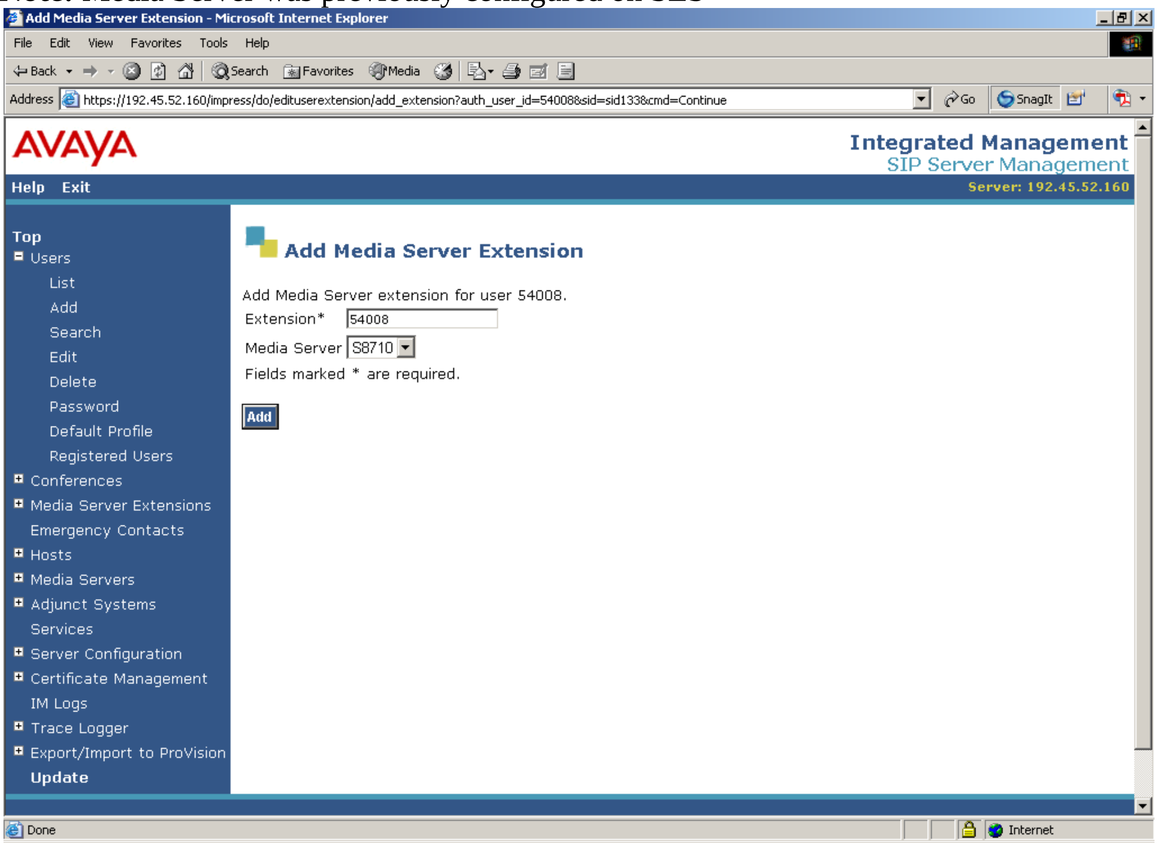
Step	Description
7.	At the **Add Media Server Extension** screen, configure the following: • **Extension –** Set it to the corresponding Avaya Communication Manager OPS station configured in **Section 3.7**, **Step 2**. • **Media Server** – Set to the Media Server where this OPS station is configured. • Click **Add**. • Click **Continue** on the next page [not shown]. **Note:** Media Server was previously configured on SES 
8.	Repeat **Steps 5 – 7** as necessary to configure SIP users for additional eyeBeam Softphones.

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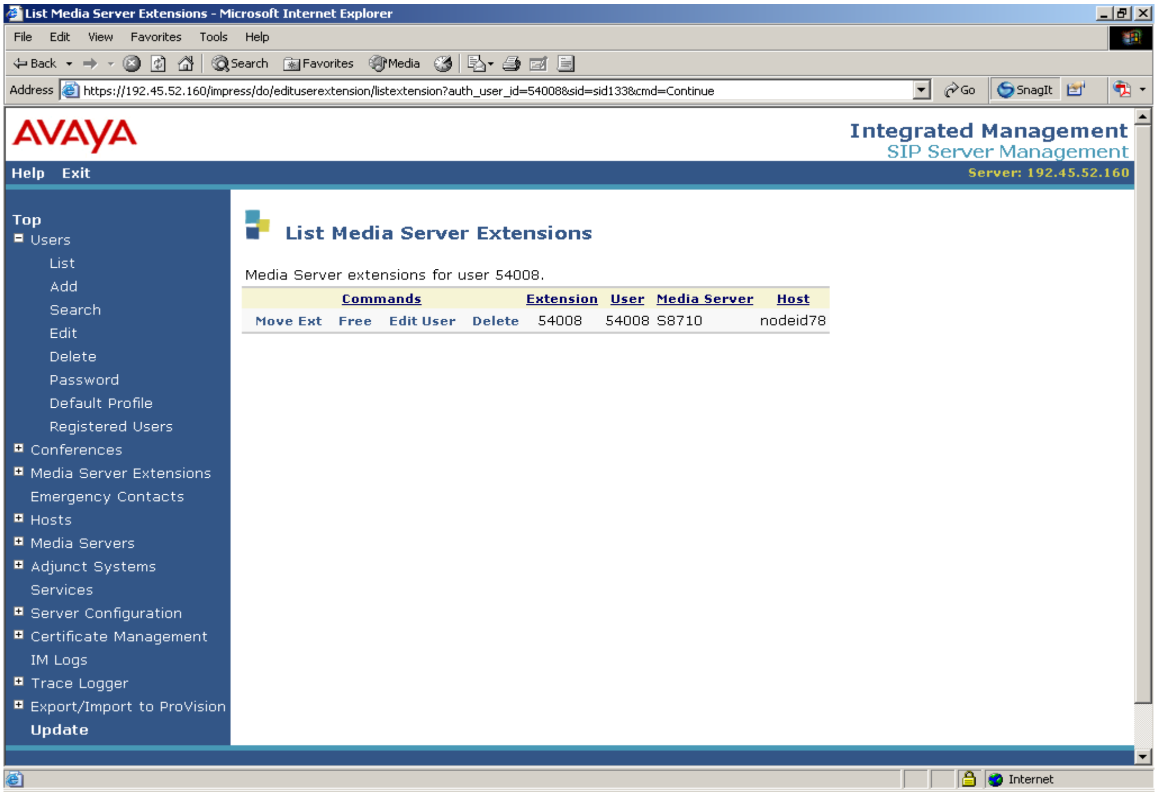
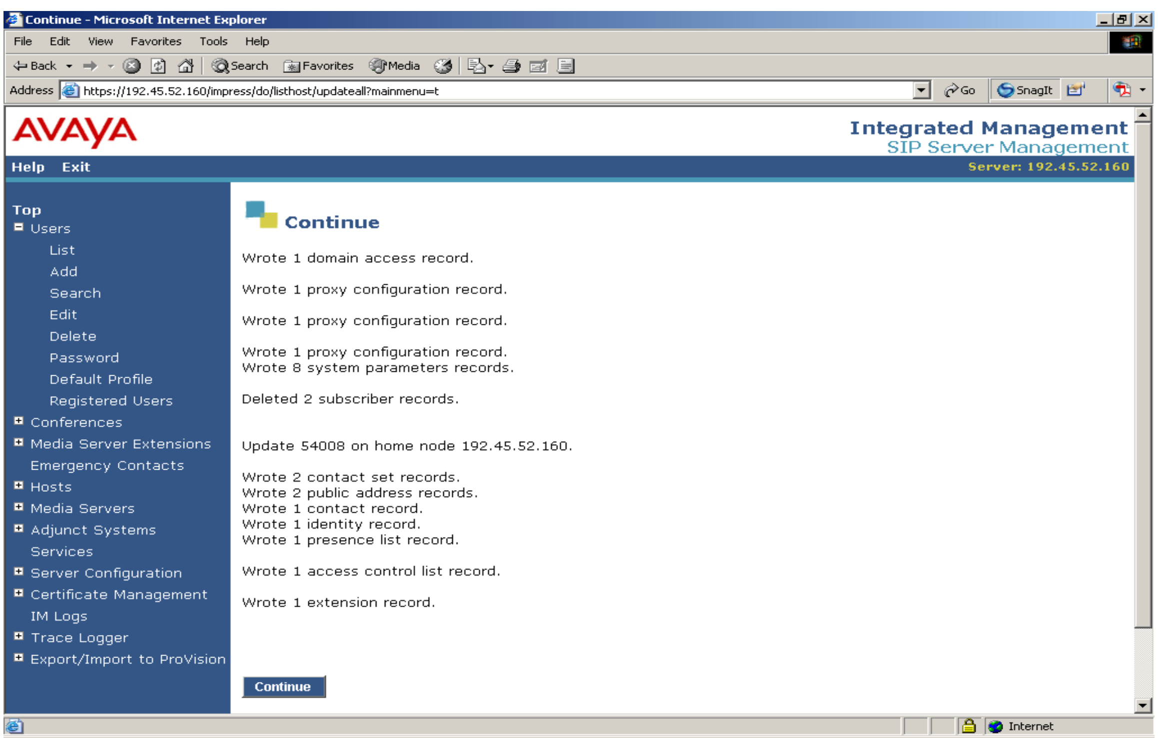
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>9.</td><td>Click Update at the bottom of the left pane.


</td></tr>
<tr><td>10.</td><td>Click Continue at the bottom of the right panel.


</td></tr>
</table>

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5. Configure the eyeBeam Softphone

This section describes the steps for configuring the eyeBeam Softphone. This section assumes that the eyeBeam Softphone is already installed and the IP address of the computer where the softphone is installed has already been configured.

Step	Description
1.	Click on the eyeBeam Softphone icon and the following screen is displayed. This screen with **No SIP accounts are enabled** message is only displayed when the softphone has not been configured. 
2.	Click on the down arrow button at the top and select **SIP Account Settings…** from the drop down menu to configure SIP account for eyeBeam Softphone. 

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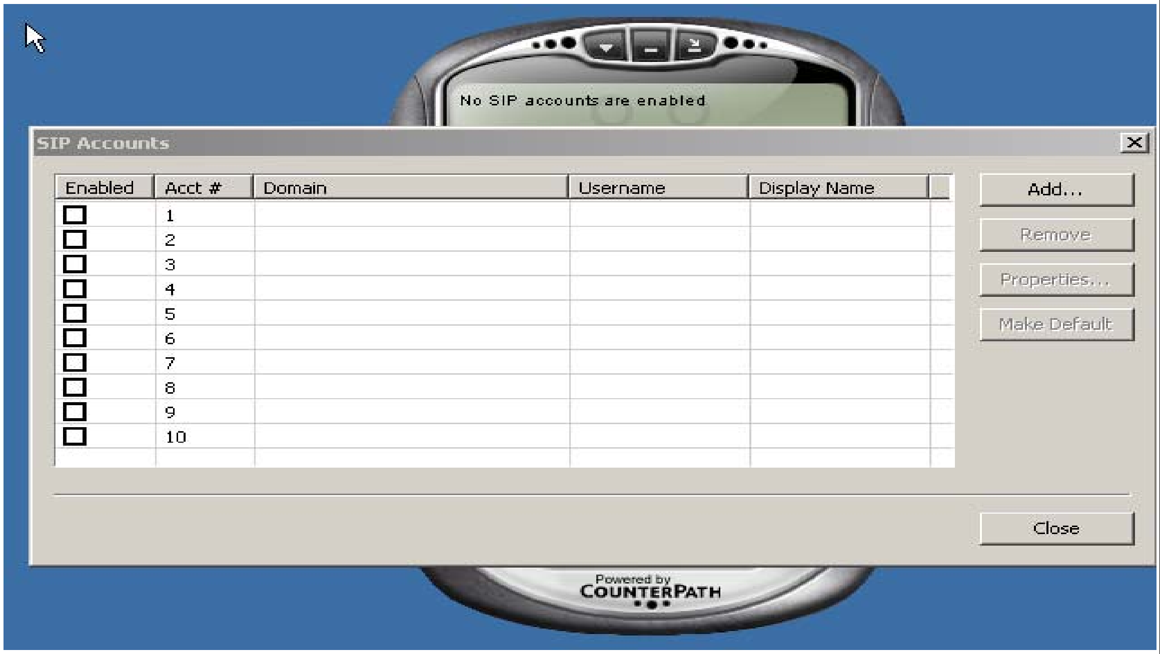
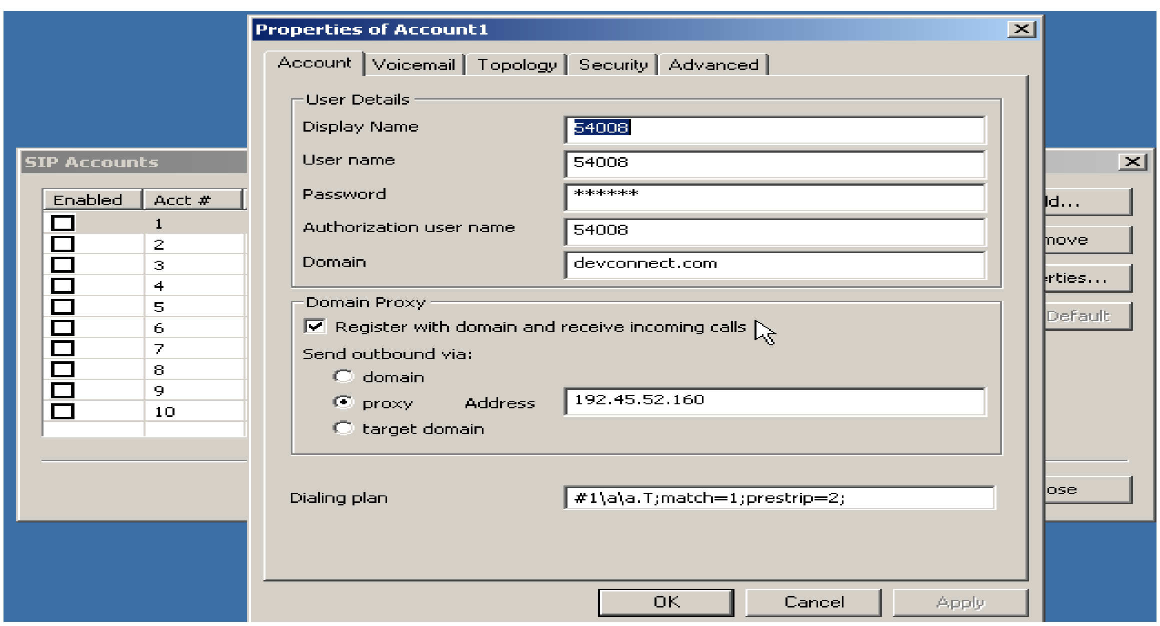
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>Click Add on SIP Accounts screen to add account information.


</td></tr>
<tr><td>4.</td><td>At Account screen, configure the following:

Display Name – Enter any descriptive name.
User Name – Set to the User Id configured in Section 4, Step 6.
Password – Set to the Password configured in Section 4, Step 6.
Authorization user name – Set to the User Id configured in Section 4, Step 6.
Domain – Set to the SIP Domain configured in Section 4, Step 2.
Set outbound via – Set to proxy.
Address – Set to Avaya SES IP address.



</td></tr>
</table>

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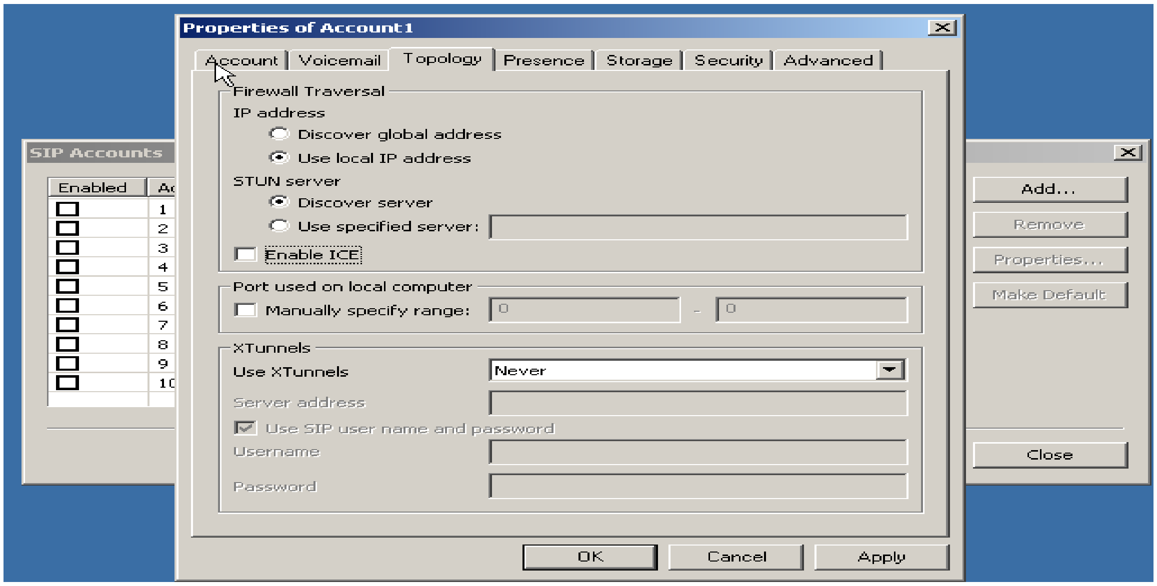
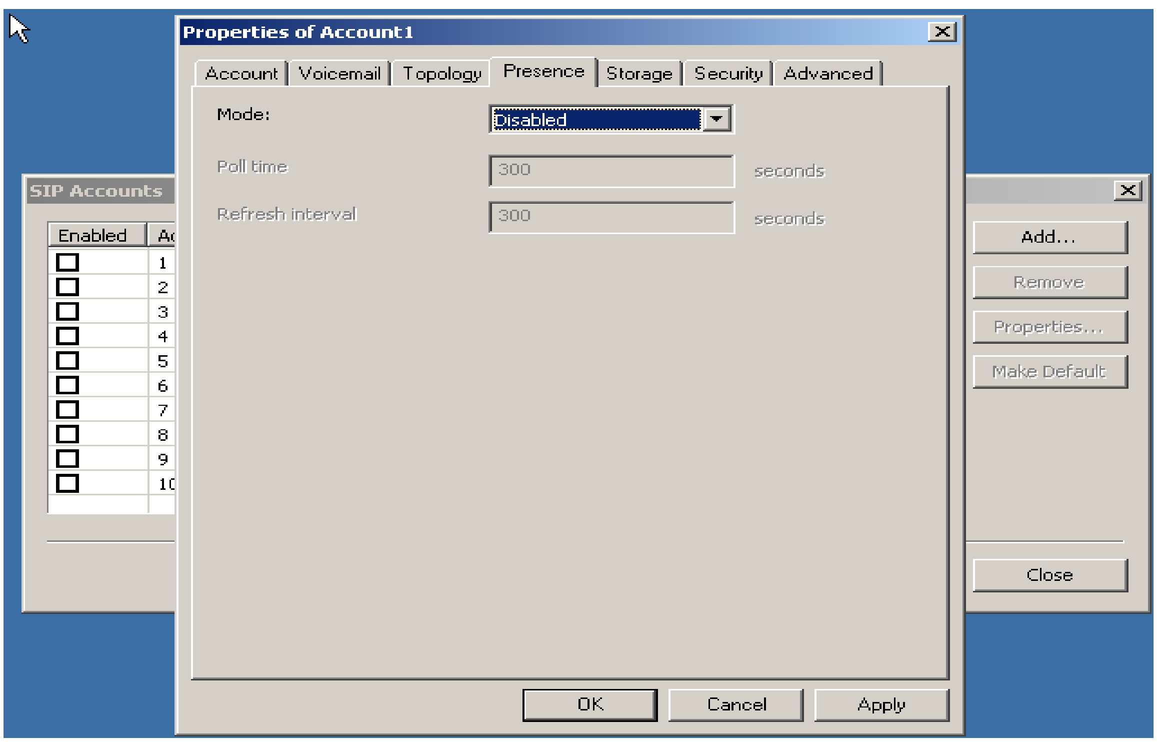
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>5.</td><td>At the Topology screen, do the following:

Select Use local IP address
Clear Enable ICE selection



</td></tr>
<tr><td>6.</td><td>At the Presence screen, set Mode to Disabled.


</td></tr>
</table>

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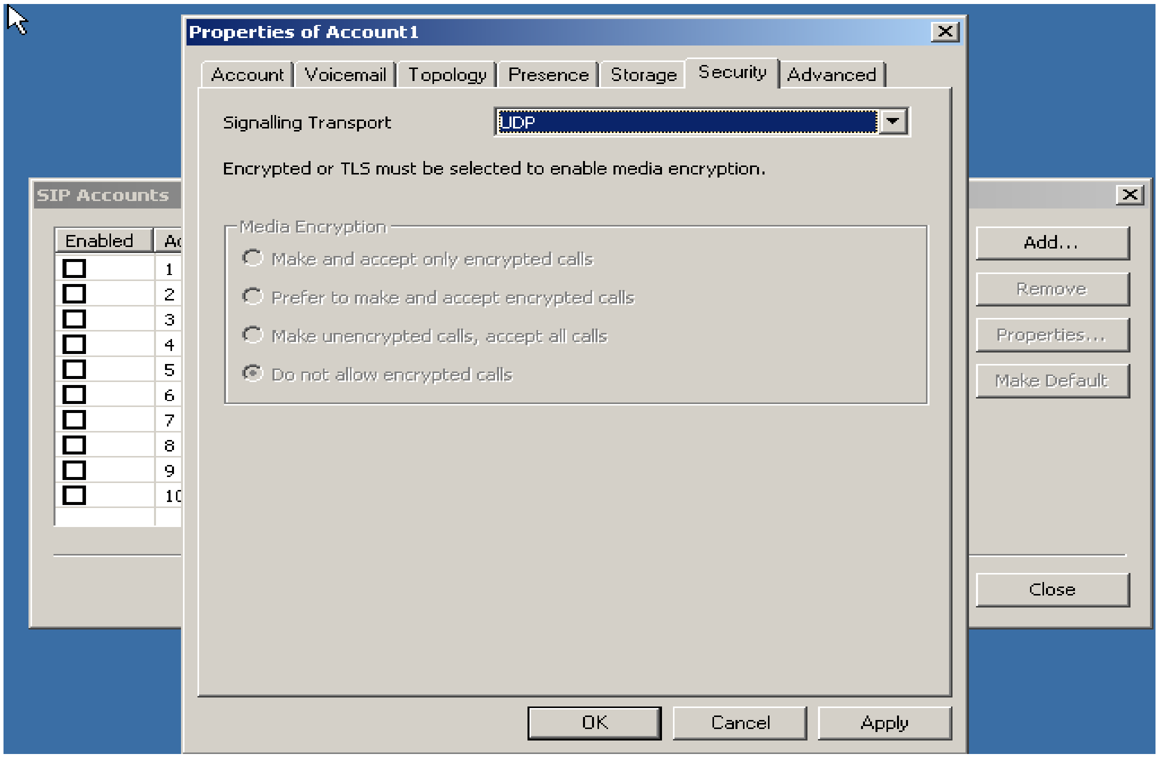
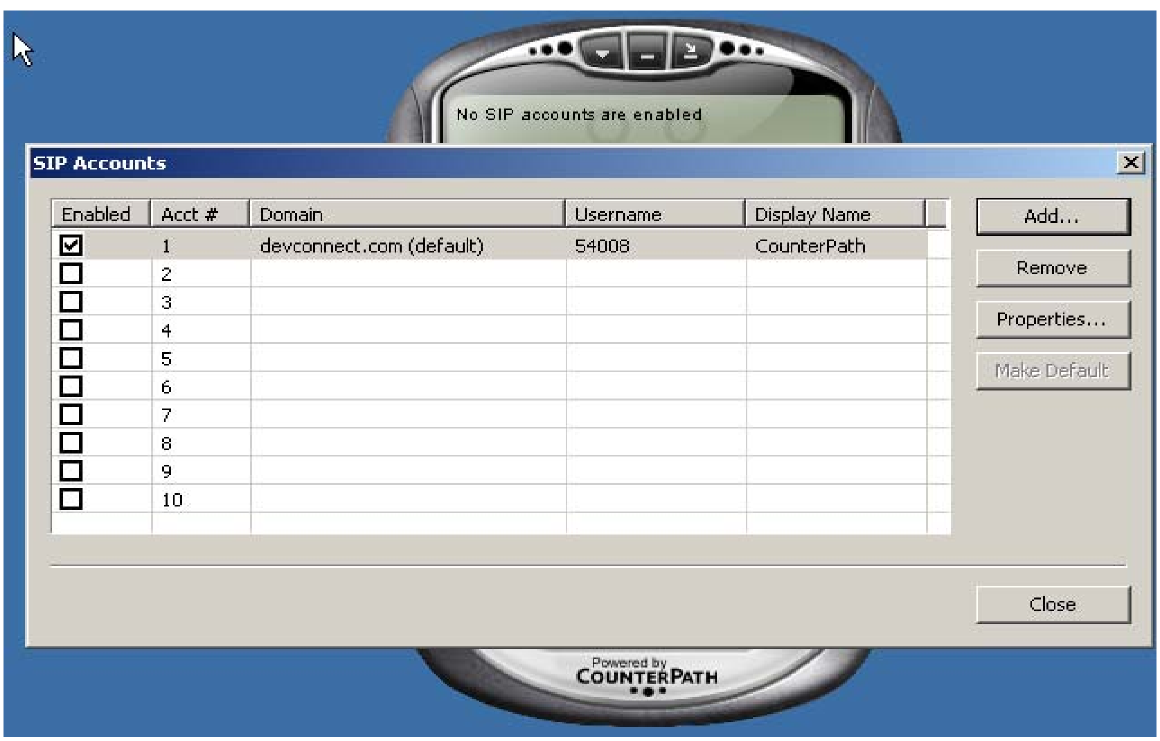
Step	Description
7.	At the **Security** screen, set the Signaling Transport to **UDP** and click the **OK** button. 
8.	Click the **close** button to complete SIP configuration. 

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Step	Description
9.	The following screen is displayed after the eyeBeam Softphone registers with the Avaya SES. 
10.	To configure QOS settings, select the Options button on this screen. 

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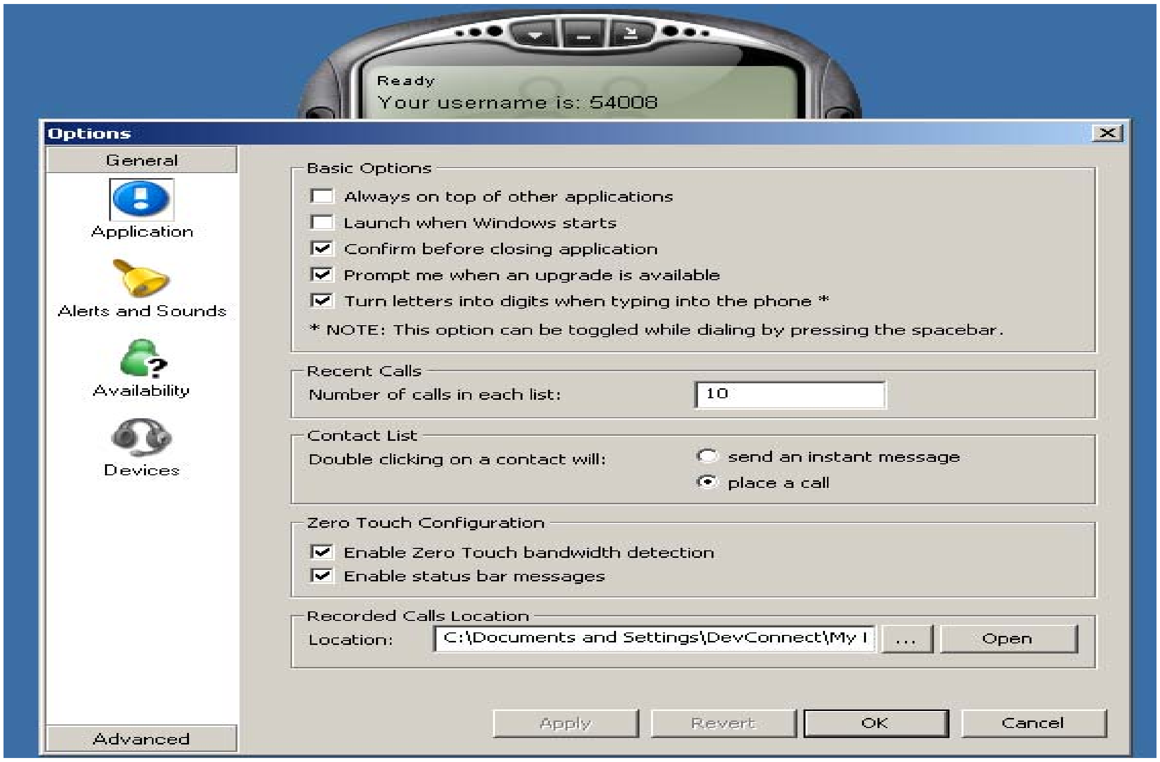
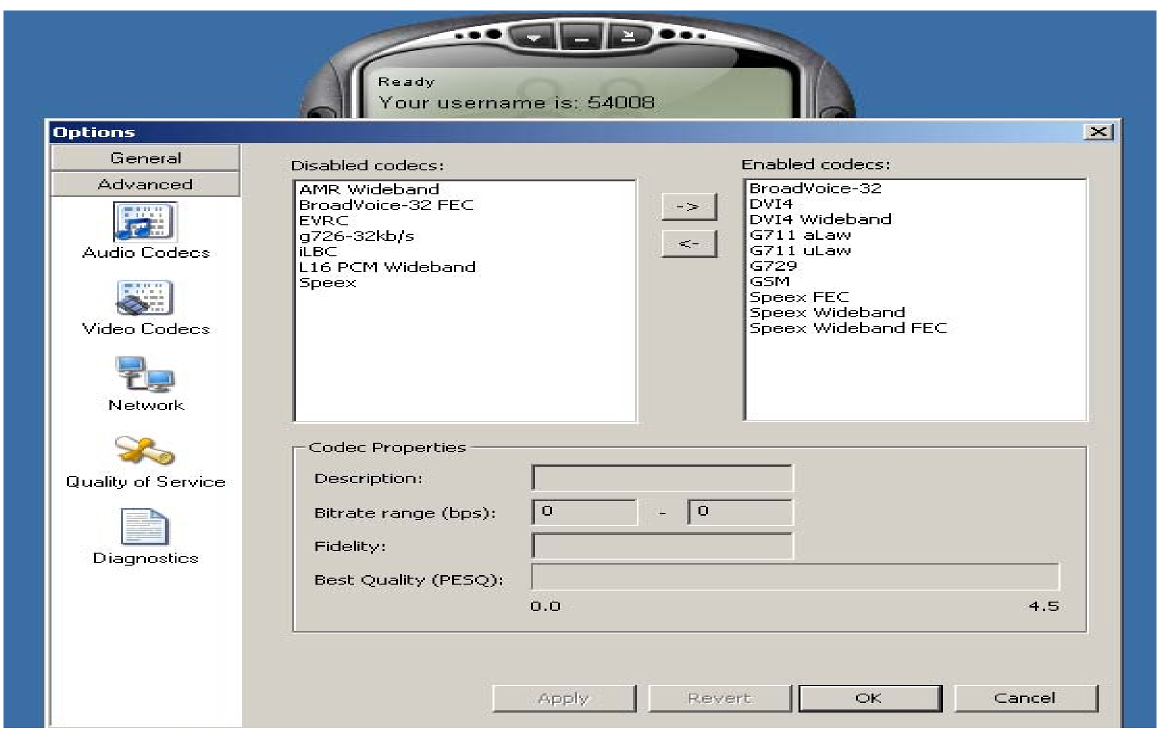
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>11.</td><td>At the Options screen, click on the Advanced button in the left pane at the bottom of the screen for QOS settings.



</td></tr>
<tr><td>12.</td><td>At the Advanced Options screen, select Quality of Service.



</td></tr>
</table>

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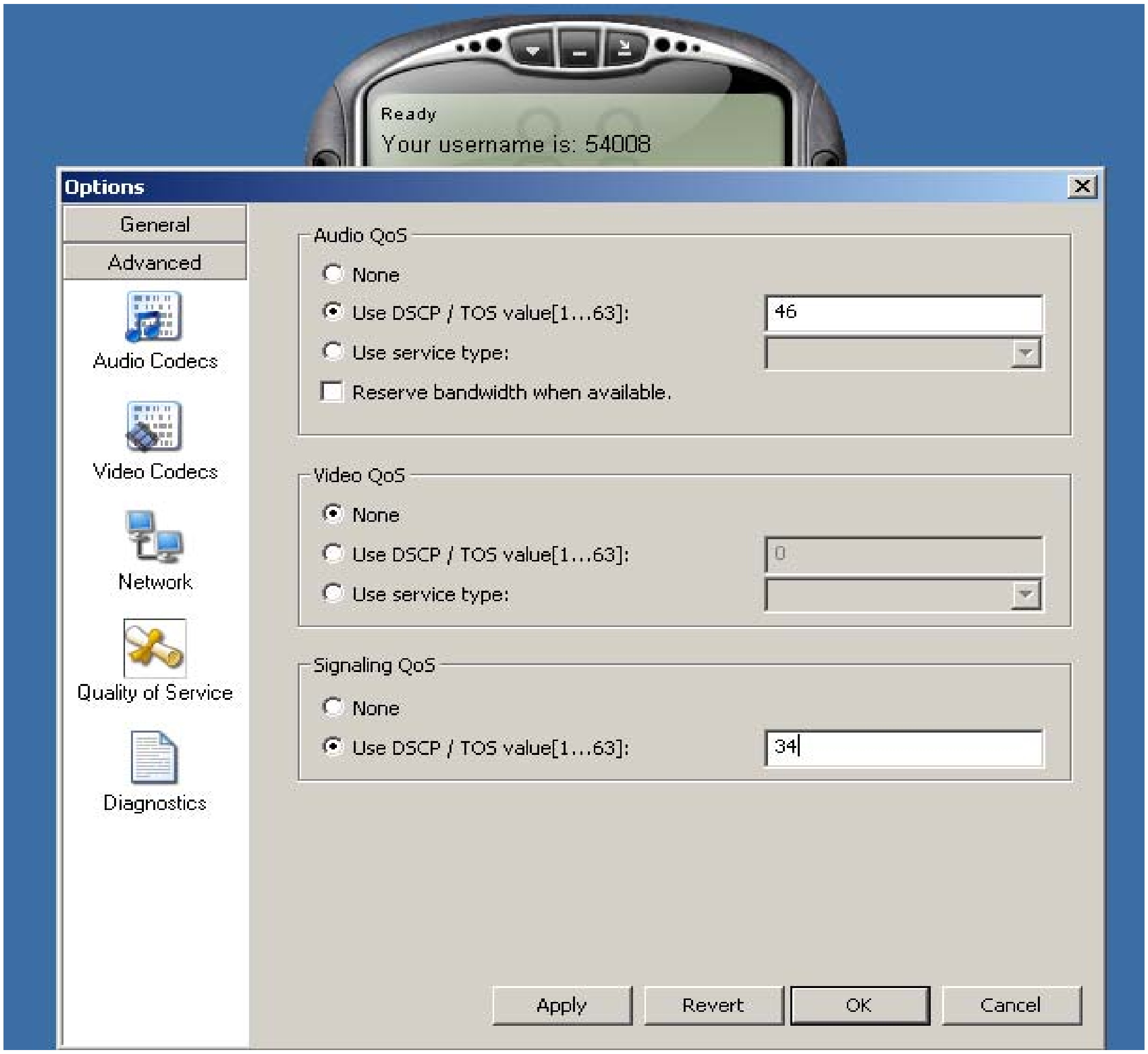
Step	Description
13.	At the **Quality of Service** Options screen, configure the following fields: • **Use DSCP/TOS[1…63] (Audio QoS) –** Set to the desired value between 1 and 63. • **Use DSCP / TOS value[1…63] (Signaling QoS) –** Set to the desired value between 1 and 63. Click **OK** to save this configuration. 

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6. Interoperability Compliance Testing

The focus of the interoperability compliance testing was on verifying call establishment on the eyeBeam Softphones and operations such as dialing methods (manual, re-dial, and phone book), hold, mute, and conference. Verified eyeBeam Softphone interactions with Avaya SIP Enablement Services (SES), Avaya Communication Manager, and Avaya SIP, H.323, and digital telephones.

6.1. General Test Approach

The general test approach was to place calls to and from the eyeBeam Softphone and exercise basic telephone operations on the eyeBeam Softphone. The main objectives were to verify:

- The eyeBeam Softphone successfully registers with Avaya SES.
- The eyeBeam Softphone successfully establishes calls with Avaya SIP, H.323, and digital telephones attached to Avaya SES or Avaya Communication Manager.
- The eyeBeam Softphone successfully establishes calls with PSTN telephones through Avaya Communication Manager.
- The eyeBeam Softphone successfully handles concurrent calls.
- The eyeBeam Softphone successfully negotiates the right codec.
- The eyeBeam Softphone successfully shuffles for VOIP calls.
- The eyeBeam Softphone successfully transmits DTMF during a call.
- The eyeBeam Softphone successfully handles layer-3 (DiffServ) QoS for Signaling and Audio.
- The eyeBeam Softphone successfully handles layer-2 (802.1p) QoS for Audio.

For serviceability testing, failures such as cable pulls and hardware resets were applied. For performance testing, a conference call involving two eyeBeam Softphones and two Avaya telephones was formed as follows. A call was established between an Avaya telephone and an eyeBeam Softphone. The eyeBeam Softphone then used its second line to establish a call with another eyeBeam Softphone, and conferenced the two lines together, forming a 3-party conference. The second eyeBeam Softphone then used its second line to establish a call with another Avaya telephone, and conferenced its two lines together, effectively forming a 4-party conference.

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6.2. Test Results

The test objectives of **Section 6.1** were verified. For serviceability testing, the eyeBeam Softphones operated properly after recovering from failures such as cable disconnects, and resets of the eyeBeam Softphones, the Avaya SES server, and Avaya Communication Manager. For performance testing, the conference call was successfully maintained for approximately two hours. eyeBeam Softphone successfully shuffles to communicate directly with the other telephone. eyeBeam Softphone successfully handles Layer-3 QOS both for signaling and audio.

The following observations were made during testing:

- eyeBeam Softphone is unable to shuffle when multiple codecs are configured on Avaya Communication Manager.
- eyeBeam Softphone has limited interoperatability with Avaya SES and Avaya Communication Manager when transport mode is TCP. Since TCP mode is not fully compliant with Avaya Communication Manager, UDP should be preferred transport mode.
- eyeBeam Softphone's TLS transport mode with Avaya SES and was not tested due to certificate configuration issue.
- eyeBeam Softphone does not support Layer-2 QOS for audio.
- eyeBeam Softphone does not support same Presence and Instant Messaging standards as supported by Avaya SES.
- eyeBeam Softphone needs to be configured with RTP_INACTIVITY_TIMER disabled, otherwise it is unable to place a call on hold or establish a conference call. For configuration of this timer, contact CounterPath Solutions directly for support.

CounterPath Solutions expects to resolve the above observations in future releases. Contact CounterPath Solutions (www.counterpath.com) for further updates.

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7. Verification Steps

The following steps may be used to verify the configuration:

- Verify that the eyeBeam Softphones successfully register with the Avaya SES server by following the **Users →Registered Users** links on the SES Administration Web Interface.
- Place calls to and from the eyeBeam Softphone and verify that the calls are successfully established with two-way talk path.
- From the Avaya Communication Manager System Access Terminal (SAT) interface, use following steps to verify that the calls successfully shuffled between two SIP telephones:

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="3">1.</td><td>Check the ports which are active for the SIP trunk being used by using the following command:
"status trunk 10"
Observe the members in active state. In our example, 10/2 and 10/6 are active.</td></tr>
<tr><td><pre>
Status trunk 10

                          TRUNK GROUP STATUS

Member   Port   Service State   Mtce Connected Ports
                                    Busy

0010/001 T00046 in-service/idle   no
0010/002 T00047 in-service/active no   T0051
0010/003 T00048 in-service/idle   no
0010/004 T00049 in-service/idle   no
0010/005 T00050 in-service/idle   no
0010/006 T00051 in-service/active no   T0047
0010/007 T00052 in-service/idle   no
0010/008 T00053 in-service/idle   no
0010/009 T00054 in-service/idle   no
0010/010 T00055 in-service/idle   no
</pre></td></tr>
<tr><td></td></tr>
</table>

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Step	Description
2.	Issue the following command for the ports in the active state: • **“status trunk 10/2”** • Note the **Near-end IP Addr** and **Far-end IP Addr** for **Audio** are talking on the same port and **Audio Connection Type** is **ip-direct**. This signifies that the endpoints have shuffled and talking to each other directly.

```
status trunk 10/2                                               Page 1 of 2

                              TRUNK STATUS

 Trunk Group/Member: 0010/002          Service State: in-service/active
              Port: T00047          Maintenance Busy? No
Signalling Group ID:



   Connected Ports: T0051

           Port     Near-end IP Addr  : Port    Far-end IP Addr  : Port
Signaling: 01A0617  192. 45.100.147   : 5061    192. 45. 52.160  : 5061

G.711MU   Audio:    192. 45. 53.101   : 34008   192. 45. 53.102  : 34008
          Video:
    Video Codec:
                                     Authentication Type: None
  Audio Connection Type: ip-direct
```

Step	Description
3.	Note on the second page of the status screen, it verifies that both endpoints are using the same codec **g711u**.

```
status trunk 10/2                                               Page 2 of 2
                     SRC PORT TO DEST PORT TALKPATH
src port: T00047
T00047:TX:192.45.53.101:34008/g711u/20ms
T00051:TX:192.45.53.102:34008/g711u/20ms











Dst port: T00051
```


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8. Support

For technical support on eyeBeam Softphones, consult the support pages at http://www.counterpath.com or contact CounterPath Solutions technical support at:

- Telephone: 1.604.320.3344
- E-mail: support@counterpath.com

9. Conclusion

These Application Notes describe a solution comprised of Avaya Communication Manager 3.1.2, Avaya SIP Enablement Services (SES) 3.1.1, and CounterPath Solutions Softphone. CounterPath eyeBeam 1.5 is a SIP-based VoIP Softphone. During compliance testing, the eyeBeam Softphone successfully registered with Avaya SES, placed and received calls to and from SIP and non-SIP telephones, and established conference calls.

10. Additional References

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

[1] *Administrator Guide for Avaya Communication Manager*, Issue 2.1, May 2006, Document Number 03-300509

[2] *Administration for Network Connectivity for Avaya Communication Manager*, Issue 11, February 2006, Document Number 555-233-504

[3] *SIP Support in Release 3.1 of Avaya Communication Manager*, Issue 6, February 2006, Document Number 555-245-206

[4] *Installing and Administering SIP Enablement Services R3.1.1*, Issue 2.0, August 2006, Document Number 03-600768

Product documentation for CounterPath Solutions products may be found at http://www.counterpath.com

[5] *eyeBeam 1.5 User Guid*e (eyeBeam_1.5_User_Guide.pdf)

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