Avaya Solution & Interoperability Test Lab

Configuring Cisco Catalyst 3750E-24P to provide Power over Ethernet to Avaya 9600, 1600 and 4600 Series IP and SIP Telephones with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services running on Avaya Aura™ Midsize Enterprise Single Server - Issue 1.0

Abstract

These Application Notes describes the configuration steps required to connect Avaya 9600, 1600 and 4600 Series IP and SIP Telephones to a Cisco Catalyst 3750-24P Power over Ethernet Switch with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services running on Avaya Aura™ Midsize Enterprise Single Server. The Application Notes identifies how to configure a vlan on a Cisco Catalyst 3750E switch and to administer a SIP Trunk within Avaya Aura™ Communication Manager to carry voice calls between Avaya IP and SIP endpoints.

1. Introduction

Power over Ethernet (PoE) technology describes a system to safely pass electrical power, along with data, on Ethernet cabling. Power over Ethernet requiries category 5 cable or higher for high power levels, but can operate with category 3 cable for low power levels. Power can come from a power supply within a Power over Ethernet enabled networking device such as an Ethernet switch or from a device built for injecting power onto the Ethernet cabling, dubbed Midspan. The IEEE 802.3af 2003 Power over Ethernet standard provides up to 15.4 Watts of DC power to each device. Only 12.95 Watts is assured to be available at the powered device as some power is dissipated in the cable. The IEEE 802.3at 2009 Power over Ethernet standard provides up to 25.5 Watts of power. Some vendors have announced products that claim to comply with the new 802.3at standard and offer up to 51 Watts of power over a single cable by utilizing all 4 pairs in the category 5 cable. Numerous non-standard schemes had been used prior to Power over Ethernet standardization to provide power over Ethernet cabling.

1.1. Interoperability Compliance Testing

The objective of this interoperability test is to verify that Cisco Catalyst 3750E-24P can provide Power over Ethernet capability to Avaya 9600, 1600 and 4600 Series IP and SIP telephones and interoperate with Avaya Aura™ Communication Manager 5.2.1 and Avaya Aura™ SIP Enablement Services 5.2.1 running on Avaya Aura™ Midsize Enterprise Single Server. It also includes procedures for configuring a vlan in an Extreme Summit x450e router and Cisco Catalyst 3750E switch. Testing was carried out on codec support and negotiation supported by Avaya 9600, 1600 and 4600 IP and SIP telephones and as well as supplementary features such as Call Hold, Forward, Transfer and Conference between the Avaya IP and SIP endpoints.

1.2. Configuration

The configuration used in these Application Notes is shown in **Figure 1**. The Avaya Aura^{TM} Midsize Enterprise software is installed and configured on Avaya Aura^{TM} System Platform on a S8500C Media Server. The Avaya Aura^{TM} Midsize Enterprise Single Server is a template running software applications. These software applications include Avaya Aura^{TM} Communication Manager, Avaya Aura^{TM} SIP Enablement Services and Avaya Aura^{TM} Application Enablement Services. The Avaya Aura^{TM} Midsize Enterprise Media Server is connected to an Extreme Summit x250e 24P switch and is configured in a separate vlan. All IP and SIP telephones are physically connected to a single Cisco Catalyst 3750E-24P switch and are administered in a single subnet. The 9600, 1600 and 4600 IP telephones register to Avaya Aura^{TM} Communication Manager running on the Avaya Aura^{TM} Midsize Enterprise Single Server and are administered as H.323 stations. The 9600 SIP telephones register to Avaya Aura^{TM} SIP Enablement Services running on the Avaya Aura^{TM} Midsize Enterprise Single Server and are administered as an OPS station on Avaya Aura^{TM} Communication Manager. Both the Extreme Summit x250e 24P switch and the Cisco Catalyst 3750E-24P switch are connected to an Extreme Summit x450e 48P router. Each of the two switches is configured with an uplink trunk port to connect to the router.

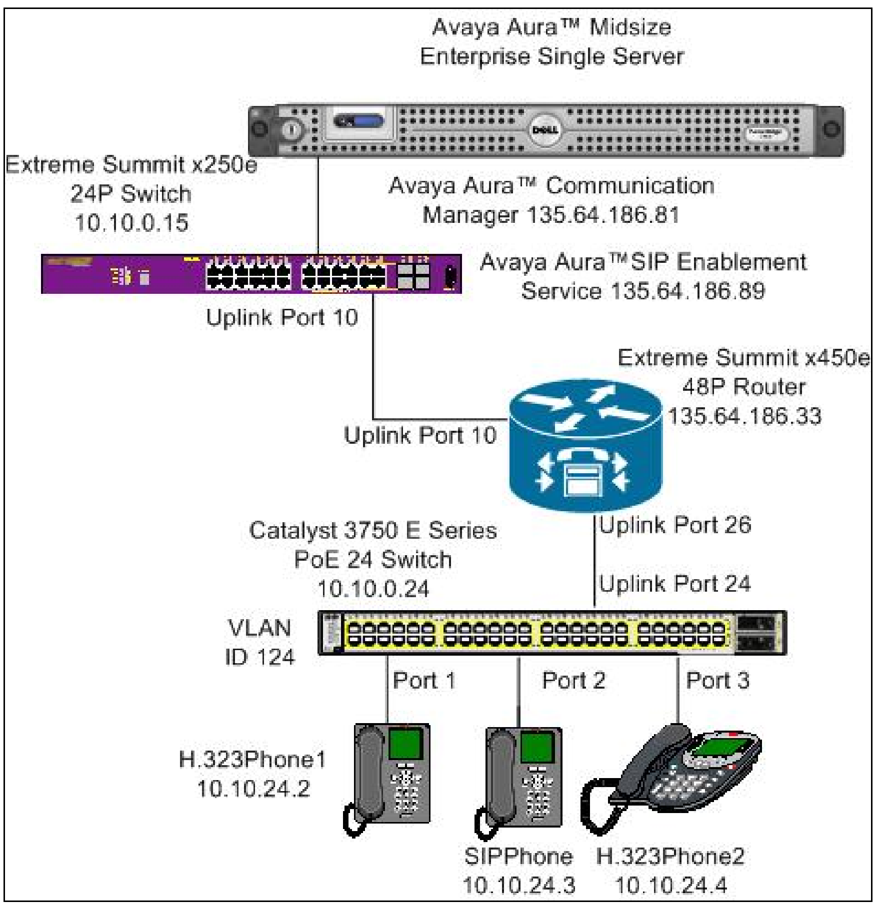


Figure 1: Avaya Aura™ Midsize Enterprise Single Server with Cisco Catalyst 3750E Switch

2. Equipment and Software Validated

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

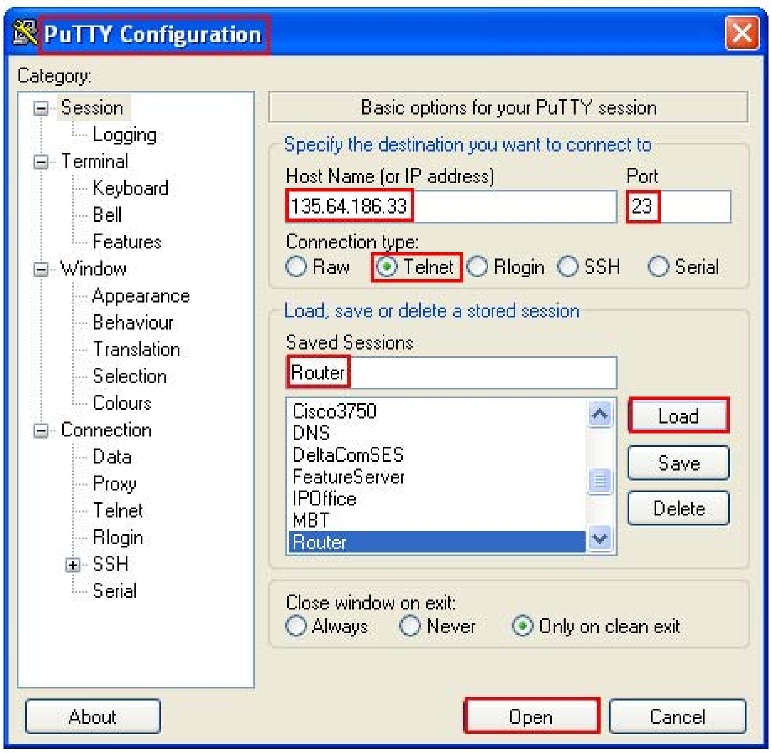
Avaya Aura™	Software
Avaya Avaya™ Midsize Enterprise Single Server on a S8500C Media Server	Avaya Avaya™ Midsize Enterprise R5.2 Release 5.2.1.2.5 Avaya Aura™ Communication Manager R5.2.1 Release 5.2.1 R15x.02.1.016.4 Update: Service Pack 0 Avaya Aura™ SIP Enablement Services R5.2 Release SES05.2.1.016.4 Update: Service Pack 0 Avaya Aura™ Application Enablement Services R5.2 Release 5.2.0.98 Update: Service Pack 0
Avaya one-X® 9600 Series IP Telephones (SIP)	Release 2.5
Avaya one-X® 9600 Series IP Telephones (H.323)	Release 3.1
Avaya one-X® 1600 Series IP Telephones (H.323)	Release 1.22
Avaya one-X® 4600 IP Telephones (H.323)	Release 2.9
Non Avaya Aura™	**Software**
Cisco Catalyst 3750 PoE 24P Switch	SW Ver. 12.2(35)SE5
Extreme Summit x250 24P Switch	IOS 12.2(35)SE5
Extreme Summit x450 48P Router	IOS 12.2(35)SE5

3. Configure Vlan in an Extreme Summit x450e Router

This section describes the steps needed to configure vlans in the Extreme Summit x450e router. In this sample configuration vlan 124 was assigned IP address range 10.10.24.1/24 with the name vlan p124 and the Communication Manager vlan 220 was assigned the IP address range 135.64.186.65/24 the the name vlan s220.

3.1. Access Extreme Summit x450e Router

To access the Extreme Summit x450e router open **Putty Configuration** and input the IP Address of the Extreme Summit x450e router and use the **Telnet** connection type with **port 23**. The **IP Address** of the Extreme Summit x450e router was **135.64.186.33**. Load the following information and press the **Open button.**



Enter the router **Login** : **admin** and **password** and hit the return key. This brings the user to the command line interface of the Extreme Summit x450e router shown as **X450e-48p.1 #.**

```
telnet session telnet0 on /dev/ptyb2

login: admin
password:

ExtremeXOS
Copyright (C) 2000-2008 Extreme Networks. All rights reserved.
Protected by US Patent Nos: 6,678,248; 6,104,700; 6,766,482; 6,618,388; 6,034,957;
6,859,438; 6,912,592; 6,954,436; 6,977,891; 6,980,550; 6,981,174; 7,003,705;
7,017,082; 7,046,665; 7,126,923; 7,142,509; 7,149,217; 7,152,124; 7,154,861.
==============================================================================

Press the <tab> or '?' key at any time for completions.
Remember to save your configuration changes.

X450e-48p.1 #
```

3.2. Create Vlan 124

To create vlan 124 the **create vlan p124** was issued from the command line interface of the router. To configure voice vlan 124 and assign vlan tag 124 to the vlan the following command **configure vlan p124 tag 124** was issued**.** The subnet range 10.10.24.1/24 for voice vlan 124 was used. From the command line interface the **configure vlan p124 ip address 10.10.24.1/24** was issued to assign this range to vlan 124. When new ip interfaces are added to the router IP forwarding is disabled by default and must be enabled. To enable IP forwarding on voice vlan p124 the command **enable ip forwarding vlan p124** was issued from the command line interface of the router.

```
X450e-48p.1 #create vlan p124
X450e-48p.1 #configure vlan p124 tag 124
X450e-48p.1 #configure vlan p124 ip address 10.10.24.1/24
X450e-48p.1 #enable ip forwarding vlan p124
```

3.3. Add Uplink Interface to Vlan 124

This sample configuration uses **port 26** on the Extreme Summit x450e router as the uplink interface that would connect to the Cisco Catalyst 3750 switch. This port needed to be added to voice vlan 124 so voice vlan 124 could communicate with Communication Manager in vlan 220. From the command line interface of the Extreme Summit x450e router the **configure vlan p124 add port 26 tagged** performed this function. The uplink interface port 26 was added to vlan s220 also with the command **configure vlan p220 add port 26 tagged.**

```
X450e-48p.1 #configure vlan p124 add port 26 tagged
X450e-48p.1 #configure vlan s220 add port 26 tagged
```

3.4. Create Vlan 220

To create vlan 220 the **create vlan s220** was issued from the command line interface of the router. To configure vlan 220 and assign vlan tag 220 to the vlan the following command **configure vlan s220 tag 220** was issued**.** The subnet range 135.64.186.68/24 for vlan 220 was used. From the command line interface the **configure vlan s220 ip address 135.64.186.68/24** was issued to assign this range to vlan 220. When new ip interfaces are added to the router IP forwarding is disabled by default and must be enabled. To enable IP forwarding on vlan s124 the command **enable ip forwarding vlan s220** was issued from the command line interface of the router.

```
X450e-48p.1 #create vlan s220
X450e-48p.1 #configure vlan s220 tag 220
X450e-48p.1 #configure vlan s220 ip address 135.64.186.68/24
X450e-48p.1 #enable ip forwarding vlan s220
```

3.5. Add Uplink Interface to Vlan 220

This sample configuration uses **port 10** on the Extreme Summit x450e router as the uplink interface that would connect to the Extreme Summit x250e switch. This port needed to be added to vlan s220 so vlan 220 could communicate with vlan 124. From the command line interface of the Extreme Summit x450e router the **configure vlan s220 add port 10 tagged** performed this function. The uplink interface port 10 was added to vlan p124 also with the command **configure vlan p124 add port 10 tagged.**

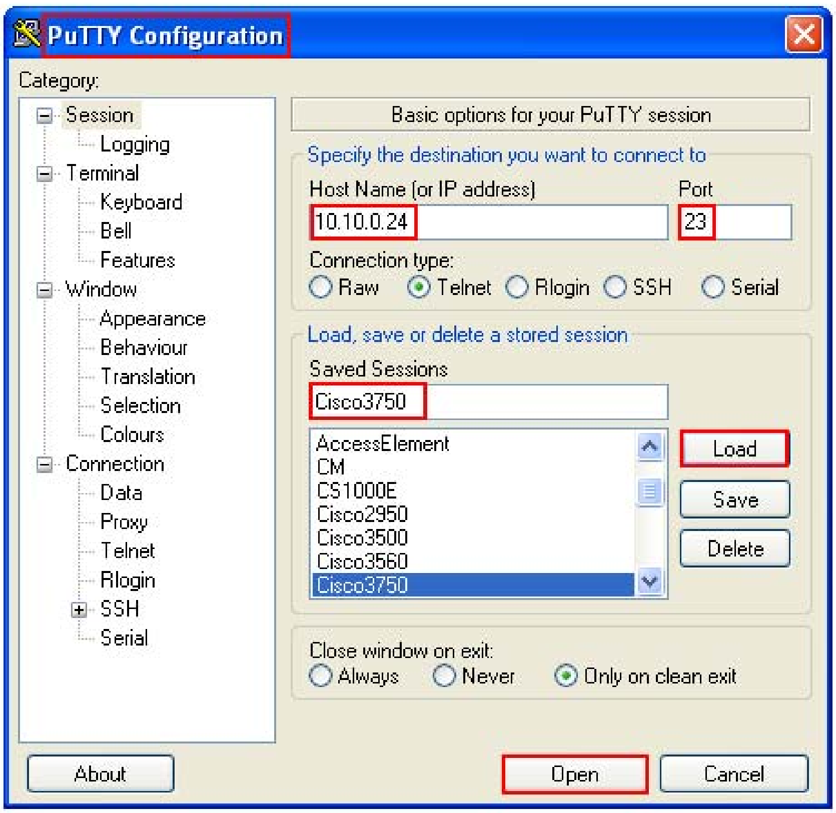
```
X450e-48p.1 #configure vlan s220 add port 10 tagged
X450e-48p.1 #configure vlan p124 add port 10 tagged
```

4. Configure Vlan in Cisco Catalyst 3750E Switch

This section describes steps needed to configure a vlan in the Cisco Catalyst 3750E switch. This sample configuration uses vlan with IP address range 10.10.24.1/24.

4.1. Access Cisco Catalyst 3750E Switch

To access the Cisco Catalyst 3750E switch open **Putty Configuration** and input the IP Address of the Cisco Catalyst 3750E switch and use the **Telnet** Connection type with **port 23**. The **IP Address** of the Cisco Catalyst switch was **10.10.0.24**. Load the following information and press the **Open button.**



Type **enable** at the command line interface. The user is then prompted to enter a password. Type the password and the user is brought to the hash prompt of the command line interface known as priviledged mode.

```
cisco3750>enable
Password:
cisco3750#
```

4.2. Create Vlan 124

To create vlan 124 in the Cisco Catalyst 3750E switch the user must enter VLAN database configuration mode from priviledged mode. This is completed by typing **VLAN database** from priviledged mode. From VLAN database configuration mode the user can assign a **vlan id** and a **vlan name** to the vlan. **Vlan id 124 with vlan name p124 are** created below.

```
cisco3750#vlan database
cisco3750(vlan)#vlan 124 name p124
cisco3750(vlan)#exit
```

4.3. Assign IP Address to Vlan 124

In order to make changes to the Cisco Catalyst 3750E switch the user needs to be in global configuration mode of the switch. This is achieved be entering the command **configure terminal** from priviledged mode of the switch. The IP address range 10.10.24.0/24 was assigned to vlan 124 by issuing the following command **ip address 10.10.24.1 255.255.255.0** from configuration mode of the Cisco Catalyst 3750E switch. The user must then enter configuration interface mode by issuing the command **interface vlan 124** from configuration mode and then assign the address ranges to the desired vlan id.

```
cisco3750#configure terminal
cisco3750(config)#interface vlan 124
cisco3750(config-if)#ip address 10.10.24.1 255.255.255.0
cisco3750(config-if)#no shut
```

4.4. Assign Interface to Vlan 124

In this sample configuration the interface ports were configured in non trunking mode. Again the user enters configuration mode and enters the command **interface gigabitethernet1/0/1.** This means the user is selecting port 1 as the interface on the Cisco Catalyst 3750E switch. This is the interface the IP telephone was connected to. To put the interface, access port, into permanent nontrunking mode and negotiates to convert the link into a nontrunk link the following command **switchport mode access** was issued from the command line interface. The **switchport access vlan 124** allows vlan 124 across this interface. The **switchport voice vlan 124** makes vlan 124 a voice vlan. Since we a using Avaya IP and SIP telephones Cisco discovery protocol was disabled using the following command **no cdp enable.**

```
cisco3750#configure terminal
cisco3750(config)#interface gigabitethernet1/0/1
cisco3750(config-if)#switchport mode access
cisco3750(config-if)#switchport access vlan 124
cisco3750(config-if)#switchport voice vlan 124
cisco3750(config-if)#no cdp enable
```

The sample configuration in **Section 4.4** as detailed above was used to configure interface 2, 3 and 4 on the Cisco Catalyst 3750E switch. The IP and SIP telephones were connected to interface 2 and 3 on the Cisco Catalyst switch.

4.5. Configure Uplink Interface on Cisco Catalyst 3750E Switch

Port 24 on the Cisco Catalyst 3750E switch was used as the uplink interface to the Extreme Summit x450e router. It was configured as a trunking port to carry traffic between the Cisco Catalyst 3750E switch and the Extreme Summit x450e router. Enter configuration mode by issuing the command **configure terminal** at the command line interface. The command **Interface gigabitethernet1/0/24** was issued as port 24 was used as the interface port to the Extreme summit x450 router. Permanent trunking was set on port 24 with the command **switchport trunk mode** with 802.1q encapsulation with the command **switchport trunk encapsulation dot1q**. To allow traffic from the vlan 124 across the interface to the Extreme Summit x450 router the command **switchport trunk allowed vlan 124** was entered.

```
cisco3750#configure terminal
cisco3750(config)#interface gigabitethernet1/0/24
cisco3750(config-if)#switchport mode trunk
cisco3750(config-if)#switchport trunk encapsulation dot1q
cisco3750(config-if)#switchport trunk allowed vlan 124
```

5. Administer Avaya Aura™ Communication Manager

This section highlights the important commands for registering Avaya IP telephones within Communication Manager and administering an IP network region and IP codecs to carry calls between Avaya IP and SIP endpoints. It also highlights the important commands for defining Avaya SIP telephones as an Off-PBX Station (OPS) and administering a SIP Trunk and Signaling Group to carry calls between Avaya IP and SIP telephones.

5.1. Verify OPS Capacity

Use the **display system-parameters customer-options** command to verify that **Maximum Off-PBX Telephones** has been set to the value that has been licensed, and that this value will accommodate addition of the SIP telephones. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya Sales representative to obtain additional capacity.

```
display system-parameters customer-options                    Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V15                                 Software Package: Standard
       Location: 2                                RFA System ID (SID): 1
       Platform: 25                               RFA Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 44000 181
                                      Maximum Stations: 2400  9
                              Maximum XMOBILE Stations: 2400  0
                    Maximum Off-PBX Telephones - EC500: 2400  2
                    Maximum Off-PBX Telephones -   OPS: 2400  5
                    Maximum Off-PBX Telephones - PBFMC: 2400  2
                    Maximum Off-PBX Telephones - PVFMC: 2400  0
```

Verify that **Maximum Concurrently Registered IP Stations** has been set to the value that has been licensed, and that this value will accommodate addition of IP telephones. Verify that the **Maximum Administered SIP Trunks** has been set to accommodate addition of SIP Trunks.

```
display system-parameters customer-options                      Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 8000  12
           Maximum Concurrently Registered IP Stations: 18000 5
             Maximum Administered Remote Office Trunks: 8000  0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 100   3
                       Maximum Administered SIP Trunks: 5000  80
  Maximum Administered Ad-hoc Video Conferencing Ports: 8000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 10    1
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0
```

5.2. Administer Dial Plan Analysis

This section describes the **Dial Plan Analysis** screen**.** This is Communication Manager's way of translating digits dialed by the user. The user can determine the beginning digits and total length for each type of call that Communication Manager needs to interpret. The **Dialed String** beginning with the number **4** and with a **Total Length** of **5** digits will be used to administer the **extension** range used for the IP telephones.

```
display dialplan analysis                                       Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location:  all          Percent Full:    0

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     0            1    fac
     1            5    ext
     2            5    aar
     209          5    ext
     3            5    aar
     4            5    ext
     5            5    ext
     6            5    ext
     7            5    aar
     8            5    aar
     9            1    fac
     *            3    dac
     #            3    dac
```

5.3. Administer IP Node-Name

This section describes **IP Node-Name** form. This is where Communication Manager assigns the IP Address and node-name to the SIP Enablement Server. The node-name of the SIP Enablement Server is **ses1** and the IP Address of the SIP Enablement Server is **135.64.186.89** within Communication Manager. Communication Manager automatically populates a processor node name to the IP Address of Communication Manager. This node name is **procr** with IP Address **135.64.186.81**.

```
list node-names all

                        NODE NAMES

Type     Name              IP Address
IP       AES1              135.64.186.88
IP       CMM               135.64.186.82
IP       MedSvcsMedpro1    135.64.186.84
IP       MedSvcsMedpro2    135.64.186.85
IP       MedSvcsMedpro3    135.64.186.86
IP       MedSvcsMedpro4    135.64.186.87
IP       procr             135.64.186.81
IP       ses1              135.64.186.89
```

5.4. Administer Signaling Group

This section describes the **Signaling Group** screen. The **Group Type** was set to **sip** and the **Transport Method** was set to **tls**. Since the sip trunk is between Communication Manager and SIP Enablement Services the **Near-end Node Name** is the node name of Communication Manager, **procr**. The **Far-end Node Name** is the node name the SIP Enablement Services. This is **ses1**. The **Near-end Listen Port** and **Far-end Listen Port** are both set to port number **5061**. The **Far-end Network-Region** was set to **1.** The **Far-end Domain** is **silstack.com**, the domain name of the SIP Enablement Server.

```
display signaling-group 3
                                SIGNALING GROUP

 Group Number: 3              Group Type: sip
                        Transport Method: tls
  IMS Enabled? n
     IP Video? n




   Near-end Node Name: procr                 Far-end Node Name: ses1
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                        Far-end Network Region: 1
Far-end Domain: silstack.com

                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? n                        Direct IP-IP Early Media? n
```

5.5. Administer Trunk Group

This section describes **Trunk Group** used to carry calls between SIP telephones. Trunk Group 3 was configured as a SIP Trunk with the **Group Type** set as **sip.** The trunk **Group Name** was set to **SIP Trunk to SES**. The **Direction** of the calls was set to **two-way** as there will be calls to and from Avaya IP and SIP telephones. The **Service Type** was set to **tie** since the trunk is configured as an internal trunk between Communication Manager and SIP Enablement Services. The **Signaling Group** number assigned to this trunk is **3**. The **Number of Members** assigned to this trunk group is **100**. All other fields on this page are left as default.

```
display trunk-group 3                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 3                    Group Type: sip           CDR Reports: y
  Group Name: SIP Trunk to SES           COR: 1        TN: 1        TAC: *03
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                     Signaling Group: 3
                                                   Number of Members: 100
```

5.6. Administer IP Network Region

This section describes **IP Network Region** screen**.** The sample configuration places all IP and SIP endpoints in the one network region. The **Authoritative Domain** must mirror the domain name of the SIP Enablement Server. This was **silstack.com**.The codecs used on the IP and SIP endpoints were placed in **Codec Set 1**. IP Shuffling was turned on so both **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** were set to **yes.**

```
display ip-network-region 1                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: silstack.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
```

5.7. Administer IP Codec Set

This section describes the **IP Codec Set** screen. It was decided to use IP Codec **G.711MU, G.711A** and **G.729** for testing purposes with the IP and SIP endpoints.

```
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: G.711A            n          2        20
 3: G.729             n          2        20
 4:
```

5.8. Administer Off PBX Telephone Station Mapping

This section show the **off-pbx-telephone station-mapping**. The Avaya SIP telephone extension **40126** uses off pbx **Application OPS** which is used for SIP enabled telephones. The SIP **Trunk Selection** is **3** as Trunk Group 3 was configured. The **Config Set** which is the desired call treatment was set to **1**.

```
display off-pbx-telephone station-mapping 40030                 Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk      Config  Dual
 Extension                 Prefix                     Selection  Set     Mode
 40126            OPS          -      40126            3         1
```

The **Call Limit** is set to **6** as shown below. This is the maximum amount of simultaneous calls for extension 40126. The **Mapping Mode** field was set to **both** in this configuration setup. This is used to control the degree of integration between SIP telephones. The **Calls Allowed** field was set to **all**. This identifies the call filter type for a SIP Phone. The **Bridged Calls** field was set to **none** as it was not needed for testing purposes.

```
display off-pbx-telephone station-mapping 40030                 Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 40126         OPS      6         both         all         none
```

5.9. Administer Station Screen

This screen describes the **station** form setup for the SIP telephone in Communication Manager as shown. The **Extension** used was **40126** with phone **Type 9630**. The **Name** of the phone was set to **PoE test** and all other values on **Page 1** of the station form were left as default.

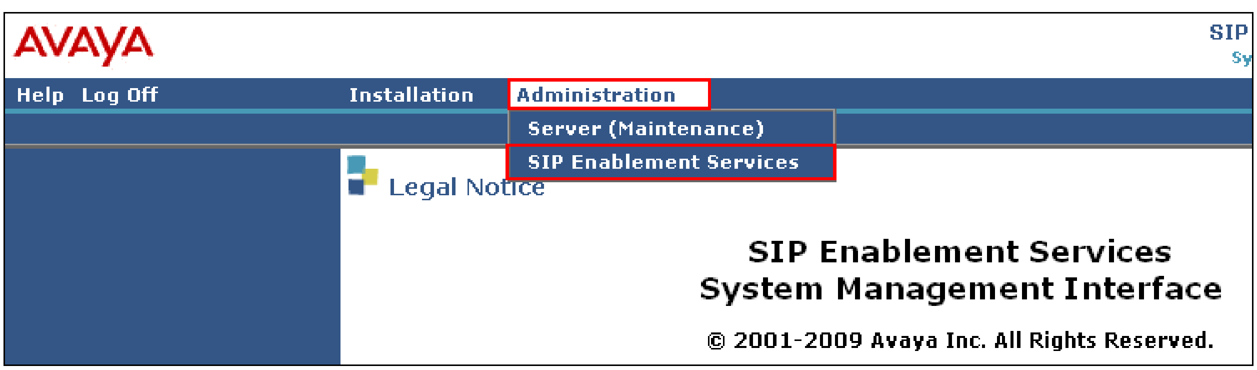
```
display station 40126                                           Page   1 of   5
                                     STATION

Extension: 40126                         Lock Messages? n               BCC: 0
     Type: 9630                          Security Code:                  TN: 1
     Port: S00010                      Coverage Path 1:                 COR: 1
     Name: PoE test                    Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 40030
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english                 Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n

                                                          IP Video? n
```

6. Administer Avaya Aura™ SIP Enablement Services

The following steps describe configuration of SIP Enablement Services to allow Avaya SIP telephones to register to SIP Enablement Services.

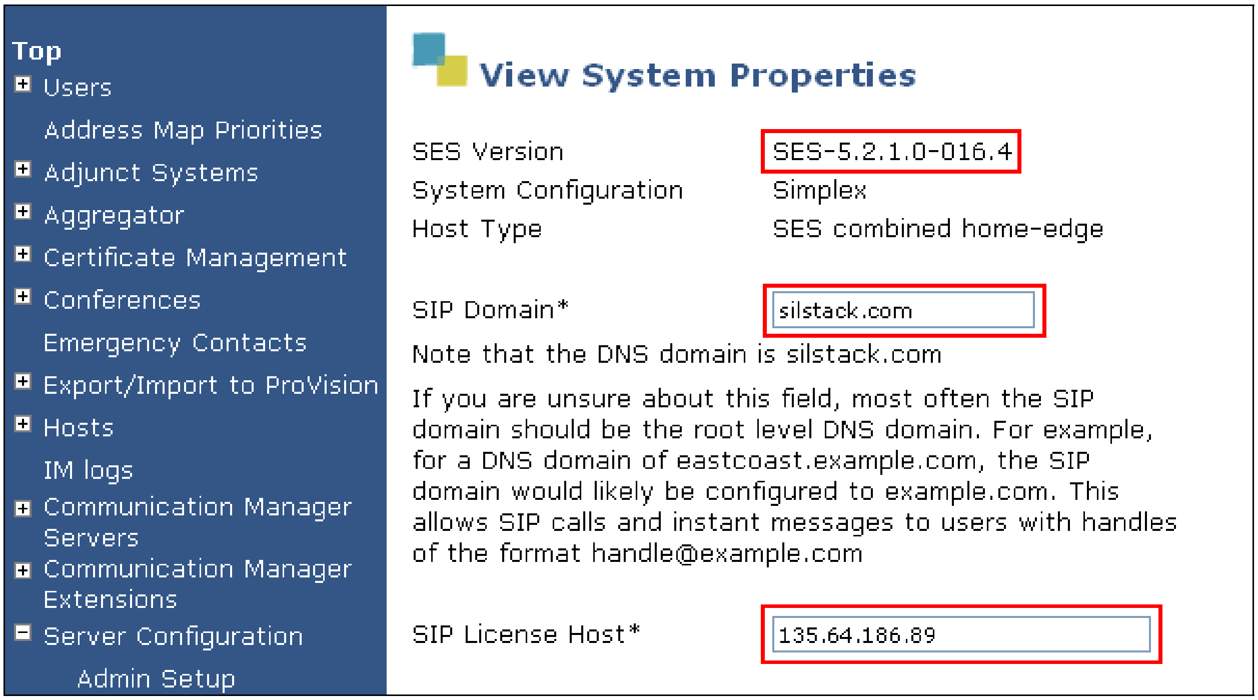
6.1. Access Avaya Aura™ SIP Enablement Services

Access the SES Administration web interface, by entering **http://<ip-addr>/admin** as the URL in Internet browser, where *<ip-addr>* is the IP address of the SIP Enablement Services server. Log in with the appropriate credentials and then select the **Administration** link and then **SIP Enablement Services** from the main screen.



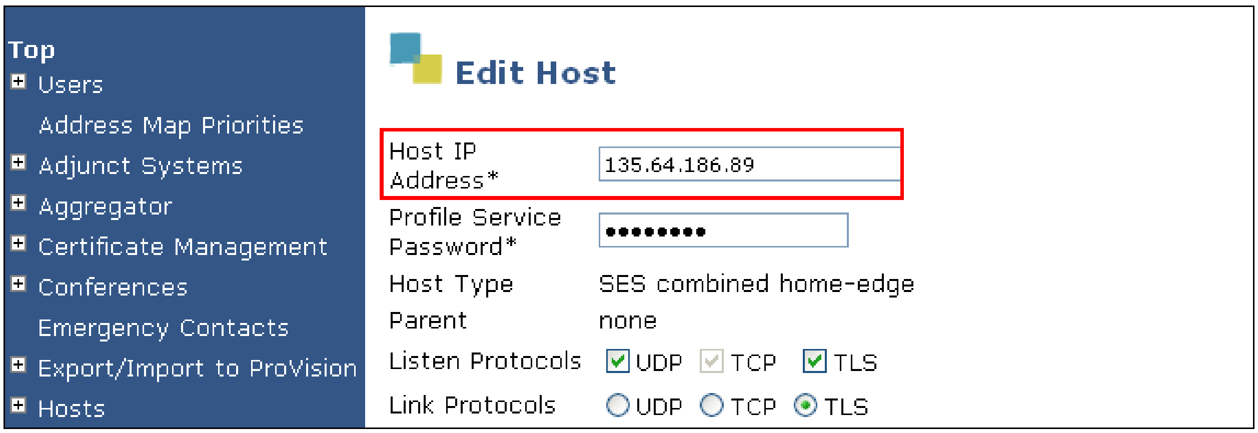
6.2. System Properties

The **View System Properties** screen defines the server's type and domain. This **SES Version** field displays the release number, the current load and build number of the Avaya software that is running on this SES server. The **System Configuration** field identifies the SES server as being a **Simplex** machine. The **Host Type** field identifies the SES server as a home/edge type server. The **SIP Domain** field indicates the domain name assigned to the SIP Enablement Services Configuration. This was set to **silstack.com**. The **SIP License Host** field requires the IP address of the SES server that is running the WebLM application and has the associated license file installed. This entry shows the IP address of the SIP Enablement Server was entered as **135.64.186.89**.



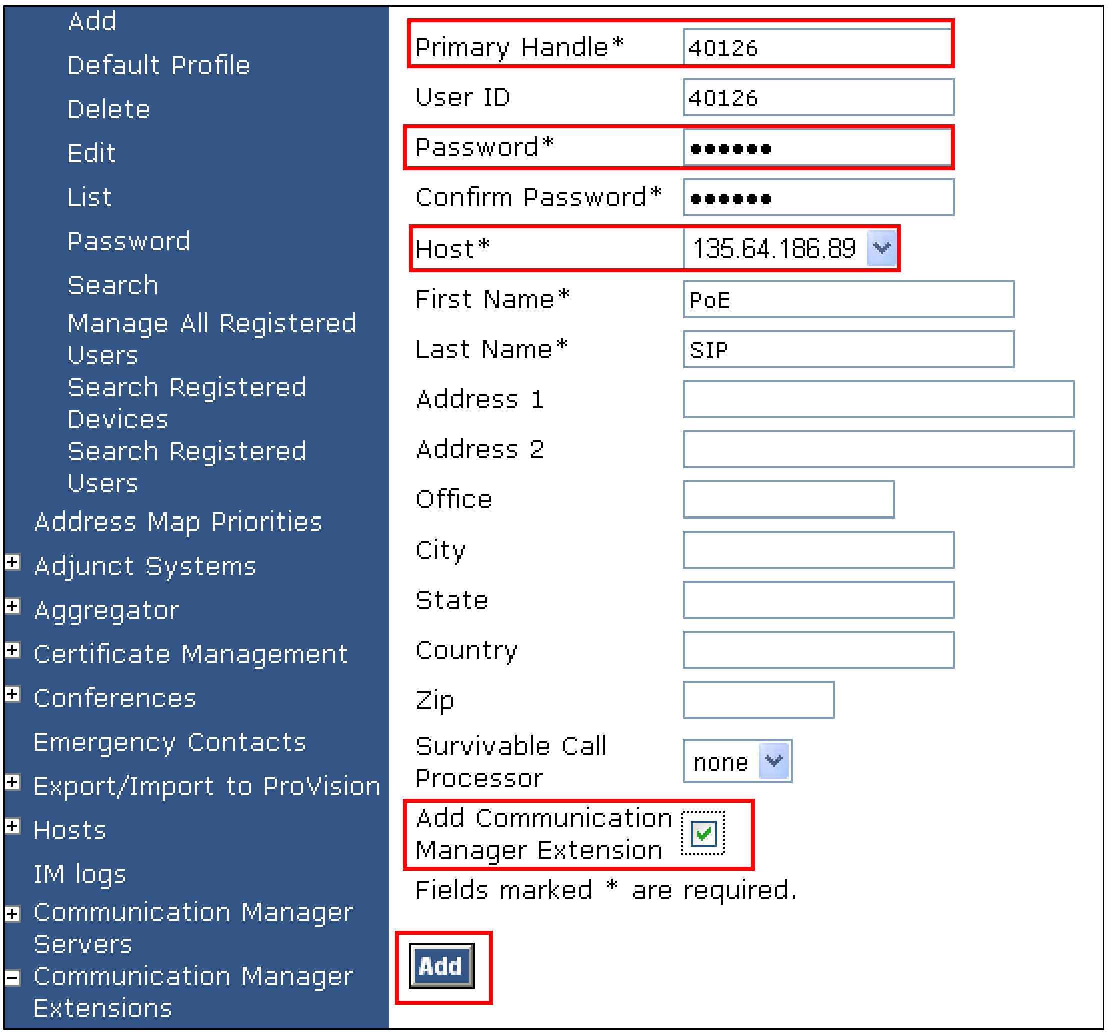
6.3. Add Host Screen

The **Host IP Address** field contains the IP address for this combined home/edge server. This was **135.64.186.89**. The **Profile Service Password** is for permissions between SES hosts. This is not used by the administrator; it is used by internal software components for secure communication between SES servers and the master administration system. The **Host Type** functions as a **CM combined home-edge** server. In the **Listen Protocol** fields **UDP** and **TLS** were selected. The **Link Protocols** field refers to the trunk signaling between SIP Enablement Services and Communication Manager. Typically, the selection here matches the Signal Group value on Communication Manager. This was **TLS.** For third-party proxy servers you may select to link to SES with TLS, TCP or UDP.

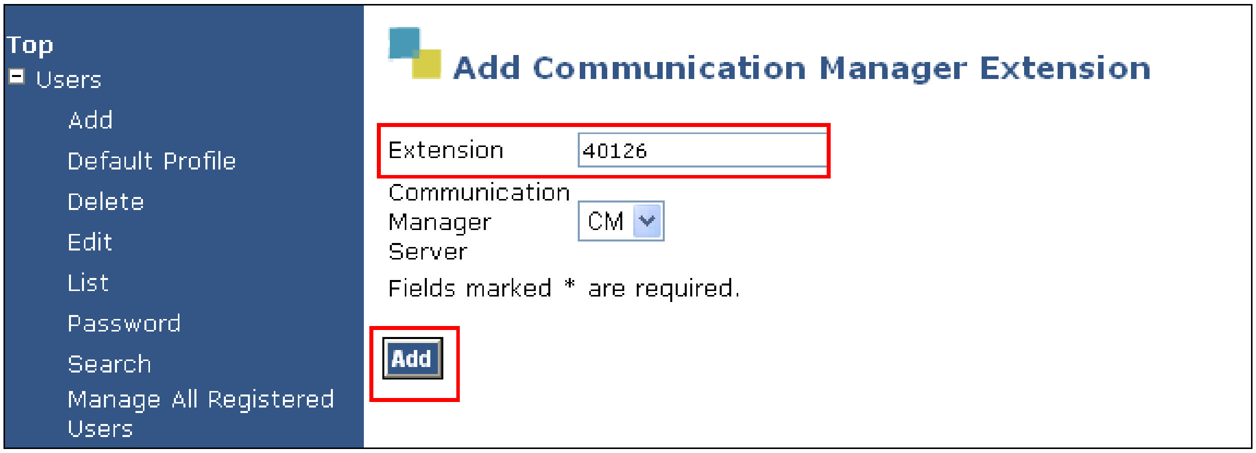


6.4. Administer Avaya SIP Telephones on SES

This screen allows Avaya SIP telephone users to be added to the SES. Users are added one at a time with this screen. A handle identifies the user on the SES system. Users **Primary Handle** must be the same as the **User ID**'s. In this example the **Primary Handle** and **User ID** is **40126**. The **Password** needs to be six characters long and was set to **123456**. This password is needed when the Avaya SIP telephone registers to the SES server after the extension of the SIP phone is inputted. The **Host** IP address is populated automatically to **135.64.186.89**. The name of the Avaya SIP telephone was **PoE SIP** (**First Name**, **Last Name**). Check the **Add Communication Manager Extension**. Press the **Add** button at the bottom of the screen. The SIP Phone extension 40126 must be added to Communication Manager also.



Once the **Add Communication Manager Extension** field is ticked the screen below appears. Confirm that extension **40126** is the **Communication Manager Extension** and press **Add**.



7. Verification Steps

The following verification steps were tested using the sample configuration.

From the Cisco Catalyst 3750E, verify the Avaya IP and SIP telephones received sufficient power from the Cisco Catalyst 3750E switch as shown below.

```
cisco3750#show power inline

Module   Available     Used     Remaining
          (Watts)     (Watts)    (Watts)
------   ---------   --------   ---------
1           450.0       28.0       422.0
Interface Admin  Oper       Power   Device              Class Max
                            (Watts)
--------- ------ ---------- ------- ------------------- ----- ----
Gi1/0/1   auto   on         7.0     Ieee PD             2     15.4
Gi1/0/2   auto   on         7.0     Ieee PD             2     15.4
Gi1/0/3   auto   on         7.0     Ieee PD             2     15.4
Gi1/0/4   auto   on         7.0     Ieee PD             2     15.4
Gi1/0/5   auto   off        0.0     n/a                 n/a   15.4
Gi1/0/6   auto   off        0.0     n/a                 n/a   15.4
Gi1/0/7   auto   off        0.0     n/a                 n/a   15.4
Gi1/0/8   auto   off        0.0     n/a                 n/a   15.4
Gi1/0/9   auto   off        0.0     n/a                 n/a   15.4
Gi1/0/10  auto   off        0.0     n/a                 n/a   15.4
Gi1/0/11  auto   off        0.0     n/a                 n/a   15.4
Gi1/0/12  auto   off        0.0     n/a                 n/a   15.4
Gi1/0/13  auto   off        0.0     n/a                 n/a   15.4
Gi1/0/14  auto   off        0.0     n/a                 n/a   15.4
Gi1/0/15  auto   off        0.0     n/a                 n/a   15.4
Gi1/0/16  auto   off        0.0     n/a                 n/a   15.4
Gi1/0/17  auto   off        0.0     n/a                 n/a   15.4
Gi1/0/18  auto   off        0.0     n/a                 n/a   15.4
```

From Avaya Communication Manager, verify the IP telephones registered to Communication Manager as shown in **Figure 2**.

```
list registered-ip-stations

                            REGISTERED IP STATIONS

Station Ext    Set Type/ Prod ID/   TCP Station IP Address/
or Orig Port   Net Rgn   Release    Skt Gatekeeper IP Address
------------- --------- ---------- --- ---------------------------------------
40020          9640      IP_Phone    y  10.10.99.11
               1         3.1000         135.64.186.81
40124          9640      IP_Phone    y  10.10.24.2
               1         3.1000         135.64.186.81
40125          4621      IP_Phone    y  10.10.24.4
               1         2.9            135.64.186.81
40127          1608      IP_Phone    y  10.10.24.5
               1         1.2200         135.64.186.81
40128          9630      IP_Phone    y  10.10.24.6
               1         3.1000         135.64.186.81
```

Figure 2: IP Telephone Registration

To see what endpoints are registered to the SIP Enablement Server access the **Search Registered Users** on the left hand side of the System Management Interface menu. **Figure 6** shows the Avaya SIP telephone 40126 registered to SIP Enablement Services.

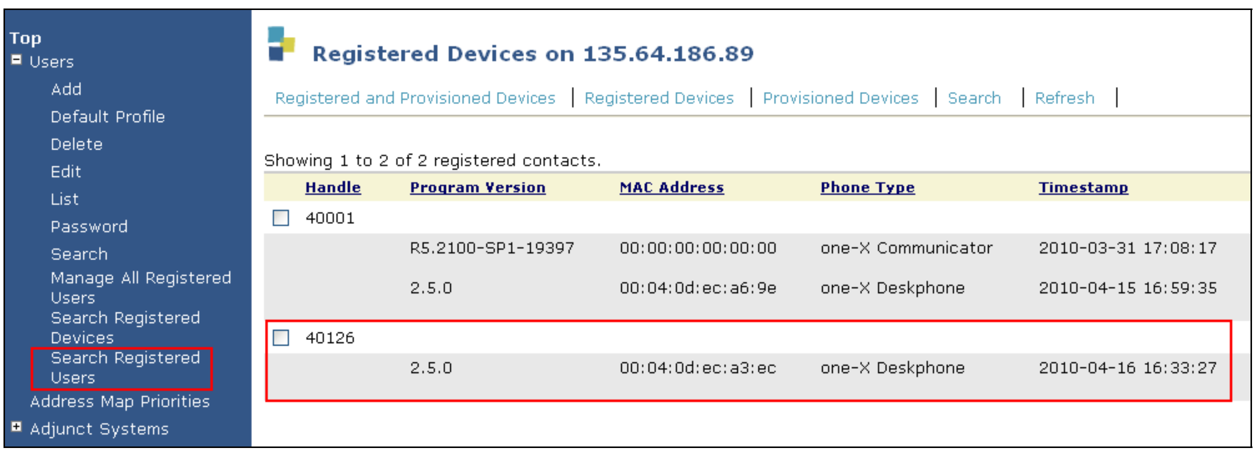


Figure 6: SIP Telephone Registration

Verify calls can be made with clear audio from the Avaya IP telephone to the SIP telephone. Verify the calls are seen to be active within Communication Manager.

```
status station 40124                                            Page   1 of   8
                              GENERAL STATUS
     Administered Type: 9640               Service State: in-service/off-hook
        Connected Type: 9640           TCP Signal Status: connected
             Extension: 40124

status station 40124                                            Page   4 of   8
                                CALL CONTROL SIGNALING
Port: S00025         Switch-End IP Signaling Loc: PROCR    H.245 Port:
            IP Address                               Port  Node Name       Rgn
Switch-End: 135.64.186.81                            61444                 1
   Set End: 10.10.24.2                               1720                  1
H.245 Near:
 H.245 Set:

status station 40124                                            Page   5 of   8

                          AUDIO CHANNEL Port: S00025
G.711MU       Switch-End Audio Location:
            IP Address                               Port  Node Name       Rgn
Other-End: 10.10.24.3                                2192                  1
  Set-End: 10.10.24.2                                3034                  1
Audio Connection Type: ip-direct
```

Verify supplementary features such as Call Hold, Call Forward, Conference and Transfer can be completed between the Cisco endpoints and the Avaya endpoints.

8. Conclusion

These Application Notes describe the administration steps required to configure a vlan on an Extreme Summit x450e router and Cisco Catalyst 3750E switch. Administration of Avaya IP and SIP telephones within Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services to result in H.323 and SIP registration. Configuration of IP network-region and IP codecs, and administration of a SIP Trunk and Signaling Group to carry calls between Avaya IP and SIP endpoints. The power consumption for each IP and SIP telephone was within the required specification of 15.4Watts. All of these functions were verified successfully.

9. Additional References

This section references the Avaya documentation relevant to these Application Notes. Additional Avaya product documentation is available at http://support.avaya.com.

[1] ExtremeXOS Command Reference Guide, May 2007, Document Number 100261-00 available at www.xtremenetworks.com

[2] Avaya Aura™ SIP Enablement Services (SES) Implementation Guide, May 2009, Document Number 16-300140

[3] Administering Avaya Aura™ Communication Manager, Document Number 03-300509

[4] Catalyst 3750 Switch Software Configuration Guide 12.2(35) SE Configuring VoiceVLANs, available at www.cisco.com

[5] Configuring 802.1Q Trunking between a Catalyst 3550/3560/3570 and Catalyst Switches that run Cisco IOS software, available at www.cisco.com

Appendices A

Sample default configuration file for Cisco Catalyst 3750E switch.

Using 5942 out of 524288 bytes

version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption

hostname cisco3750

enable secret 5 1Pb8u$/RR6UVRFCcPMMqZ2WAVkt.
enable password cisco

no aaa new-model
switch 1 provision ws-c3750e-24pd
system mtu routing 1500
ip subnet-zero

no file verify auto
spanning-tree mode pvst
spanning-tree extend system-id

vlan internal allocation policy ascending

interface FastEthernet0
no ip address
no ip route-cache
shutdown

interface GigabitEthernet1/0/1
switchport mode access
switchport access vlan 124
switchport voice vlan 124
no cdp enable

i**nterface GigabitEthernet1/0/2**
switchport mode access
switchport access vlan 124
switchport voice vlan 124
no cdp enable

interface GigabitEthernet1/0/3
switchport mode access
switchport access vlan 124
switchport voice vlan 124
no cdp enable

interface GigabitEthernet1/0/4
switchport mode access
switchport access vlan 124
switchport voice vlan 124
no cdp enable

interface GigabitEthernet1/0/5

interface GigabitEthernet1/0/6

interface GigabitEthernet1/0/7

interface GigabitEthernet1/0/8

interface GigabitEthernet1/0/9

interface GigabitEthernet1/0/10

interface GigabitEthernet1/0/11

interface GigabitEthernet1/0/12

interface GigabitEthernet1/0/13

interface GigabitEthernet1/0/14

interface GigabitEthernet1/0/15

interface GigabitEthernet1/0/16

interface GigabitEthernet1/0/17

interface GigabitEthernet1/0/18

interface GigabitEthernet1/0/19

interface GigabitEthernet1/0/20

interface GigabitEthernet1/0/21

interface GigabitEthernet1/0/22
 switchport access vlan 100

interface GigabitEthernet1/0/23

interface GigabitEthernet1/0/24
switchport trunk encapsulation dot1q
switchport trunk allowed vlan 124
switchport mode trunk
no cdp enable
spanning-tree portfast

interface GigabitEthernet1/0/25

```
interface GigabitEthernet1/0/26

interface GigabitEthernet1/0/27

interface GigabitEthernet1/0/28

interface TenGigabitEthernet1/0/1

interface TenGigabitEthernet1/0/2

interface Vlan1
 no ip address
 no ip route-cache
 shutdown

interface Vlan100
 ip address 10.10.0.23 255.255.255.0
 no ip route-cache

ip default-gateway 10.10.0.1
ip classless
ip http server


control-plane


line con 0
line vty 0 4
 password cisco
 login
 length 0
line vty 5 15
password cisco
 login
monitor session 1 source interface Gi1/0/1
monitor session 1 destination interface Gi1/0/6 encapsulation replicate
monitor session 2 source interface Gi1/0/2
monitor session 2 destination interface Gi1/0/5
end
```

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