Avaya Solution & Interoperability Test Lab

Configuring Cisco 7960/7941 SIP Telephones to connect to Avaya Aura™ SIP Enablement Services with Avaya Aura™ Communication Manager running on Avaya Aura™ Midsize Enterprise Single Server Issue – 1.0

Abstract

These Application Notes describes the configuration steps required to connect Cisco 7960/7941 SIP Telephones to Avaya Aura™ SIP Enablement Services with Avaya Aura™ Communication Manager running on Avaya Aura™ Midsize Enterprise Single Server. The Application Notes identifies how to configure a TFTP Server to upload SIP firmware to the Cisco 7960/7941 SIP Telephones. Administration of a SIP Trunk within Avaya Aura™ Communication Manager to carry calls between Cisco SIP endpoints and Avaya SIP and IP endpoints.

1. Introduction

With the introduction of SIP protocol standards that supports telephony as well as a wide range of other communication modes, there is a much broader range of SIP telephones and Gateways available to customers. There will be sales opportunities involving customers who wish to purchase the Avaya SIP Solution, but already own SIP telephones other than those offered by Avaya. Customers may be interested in replacing their existing telephony infrastructure (e.g., Cisco Communication Manager) with Avaya Servers, but wish to re-use the existing telephones. In addition, the Off-PBX Station (OPS) feature set can be extended from Avaya Aura ™ Communication Manager to these SIP telephones, providing enhanced calling features in advance of SIP protocol definitions and implementation by telephone manufacturers.

1.1. Interoperability Compliance Testing

The objective of this interoperability test is to verify that Cisco 7960/7941 SIP telephones can interoperate with Avaya Aura™ Communication Manager 5.2.1 and Avaya Aura™ SIP Enablement Services 5.2 running on Avaya Aura™ Midsize Enterprise Single Server. It also includes procedures for upgrading telephone firmware on Cisco 7960/7941 SIP telephones. Testing was carried out on codec support and negotiation supported by Cisco 7960/7941 SIP telephones and Avaya 9630 SIP telephone with Avaya 1616 IP telephone and as well as supplementary features such as Call Hold, Forward, Transfer and Conference between the Cisco endpoints and the Avaya endpoints.

1.2. Configuration

The configuration used in these Application Notes is shown in **Figure 1**. The Avaya Midsize Enterprise software is installed and configured on Avaya System Platform on a S8500C Media Server. The Avaya Midsize Enterprise Single Server is a template running software applications. These software applications include Avaya Aura™ Communication Manager, Avaya Aura™ SIP Enablement Services and Avaya Aura™ Application Enablement Services. The diagram indicates logical signaling connections. All components are physically connected to a single Avaya C363T-PWR Converged Stackable Switch and are administered in a single subnet. The Cisco 7960/7941 SIP telephones are configured to register to Avaya Aura™ SIP Enablement Services running on the Avaya Midsize Enterprise Single Server and are administered as an OPS station on Avaya Aura™ Communication Manager running on the Avaya Midsize Enterprise Single Server. The 9630 SIP telephone registers to Avaya Aura™ SIP Enablement Services with the 1616 IP telephone registering to Avaya Aura™ Communication Manager. The TFTP Server is used to transfer the SIP firmware to the Cisco SIP telephones.

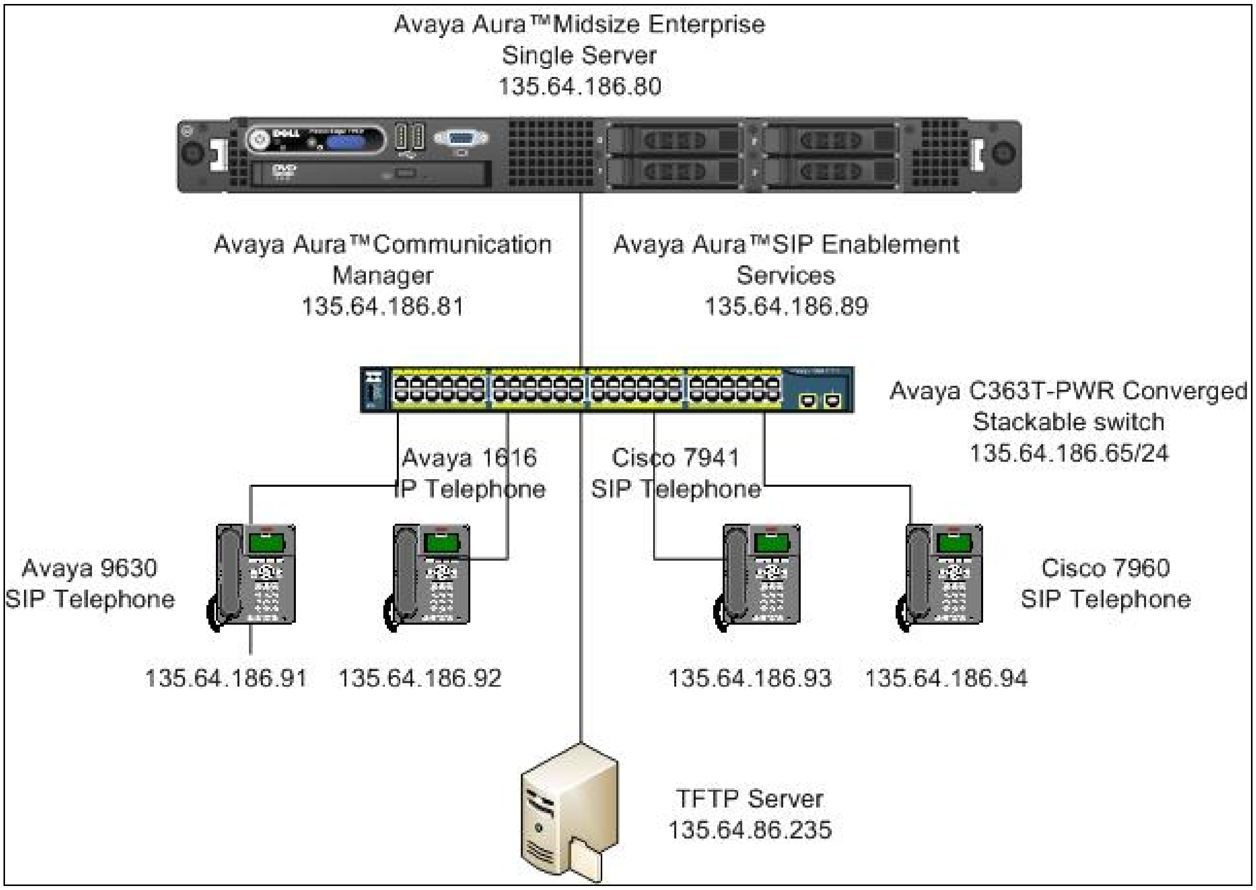

Figure 1: Avaya Aura™ Midsize Enterprise Single Server

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 Release 5.2.1.2.5 Avaya Aura™ Communication Manager Release 5.2.1 R15x.02.1.016.4 Update: Service Pack 0 Avaya Aura™ SIP Enablement Services Release 5.2.1 SES05.2.1.016.4 Update: Service Pack 0 Avaya Aura™ Application Enablement Services Release 5.2.0.98 Update: Service Pack 0
Avaya Avaya™ C363T–PWR Converged Stackable Switch	Release 4.5.14
Avaya one-X® 9600 Series IP Telephones (SIP)	Rel. 2.6.x
Avaya one-X® 1616 IP Telephones (H.323)	Rel. 1.2110
Non Avaya Aura™	**Software**
Cisco 7960 IP Telephone (SIP)	Rel. POS3-08-12-00
Cisco 7941 IP Telephone (SIP)	Rel. SIP41.8-3-1S
Trivial File Transfer Protocol Server	Microsoft Windows XP Professional Workstation Version 2002 Update: Service Pack 2 Pumpkin Release 2.7.2

3. Configure the Cisco 7960/7941 SIP Telephone

This section describes steps needed to configure and connect Cisco SIP telephones to SIP Enablement Services. It will explain the configuration files needed to register the Cisco SIP telephones and the settings needed to be configured on each on the two configurations files. It will discuss the SIP firmware files needed for the Cisco SIP telephones. The SIP firmware used on the Cisco 7960/7941 SIP telephones were the latest version to date from the release of this Application Notes. An explanation of the TFTP settings configured to upload the SIP firmware to the Cisco SIP telephones is also discussed.

3.1. Cisco Configuration Files

The Cisco 7960/7941 SIP telephones will register to SIP Enablement Services. In order for the Cisco telephones to register correctly with SIP Enablement Services certain parameters on the Cisco telephones need to mirror certain parameters on SIP Enablement Services. Two configuration files are used to configure these parameters which are the **SIPDefault.cnf** and **SIP<macaddress>.cnf** configuration files. **Appendices A** and **B** contain sample configurations of the SIPDefault.cnf and SIP001200B4743E.cnf files. SIPDefault.cnf is a default configuration file containing parameter settings that apply to all telephones, and a telephone specific configuration file, SIP<macaddress>.cnf, containing settings applicable only to that telephone.

SIP<macaddress>.cnf where <macaddress> is the mac address of the Cisco telephone. With a few exceptions, most parameters can be specified in the configuration files as this is the preferred method for maintaining a large number of telephones. In the SIP Default Configuration file, the image version P0S3-8-12-00 must match the SIP firmware version that is being uploaded to the Cisco telephones. The Cisco telephone will upgrade to that version when the files are available under the root directory of the TFTP Server. The proxy1_address in the SIP Default configuration file must be the same IP Address as the SIP Enablement Server in order for the Cisco SIP telephones to register to the SIP Enablement Server. The proxy1_address in the SIP Default configuration file was set to 135.64.186.89 to mirror the SIP Enablement Server IP Address which was also set to 135.64.186.89. In the SIP<macaddress>.cnf file the Line1_authname and Line1_password must match the user name and password of the user to be added in SIP Enablement Service. In the SIP001200B4743E configuration file, where the MAC Address of the Cisco telephone is 001200B4743E the Line1-authname was set to 40030 and the Line1_password was set to 123456 to mirror the SIP Enablement Server user 40030 and password 123456 that has been added as a user to SIP Enablement Services. The extension 40030 is also configured as an Off PBX Station Mapping extension in Communication Manager. Theses values were used in order to register the Cisco SIP telephone to the SIP Enablement Server. The dialplan.xml configuration file is used to determine when enough digits have been pressed to complete dialing. The dial plan can also be used to specify the local region dial tone to be played locally on the phone. If no dial tone configuration is specified the default US dial tone is used.

3.2. Cisco SIP Firmware

Cisco 7960 and 7941 IP telephones can support either the MGCP, SCCP, or SIP protocols, and require a different application firmware version for each. This section describes the steps required to convert a Cisco IP telephone from SCCP to Cisco SIP telephone. The latest version of SIP firmware was obtained from the Cisco website. The SIP firmware was downloaded as a zip file with sample configuration POS3-08-12-00.zip. These files were extracted into the **TFTP Directory** on the TFTP Server which will be accessed by the Cisco telephones during the boot process as shown in **Figure 2**. The Cisco SIP firmware was placed in a folder in directory **C:\tftpboot**. The **P0S3-08012-00.loads** file is the SIP firmware file that will be uploaded to the Cisco IP telephone to make it become a Cisco SIP telephone.

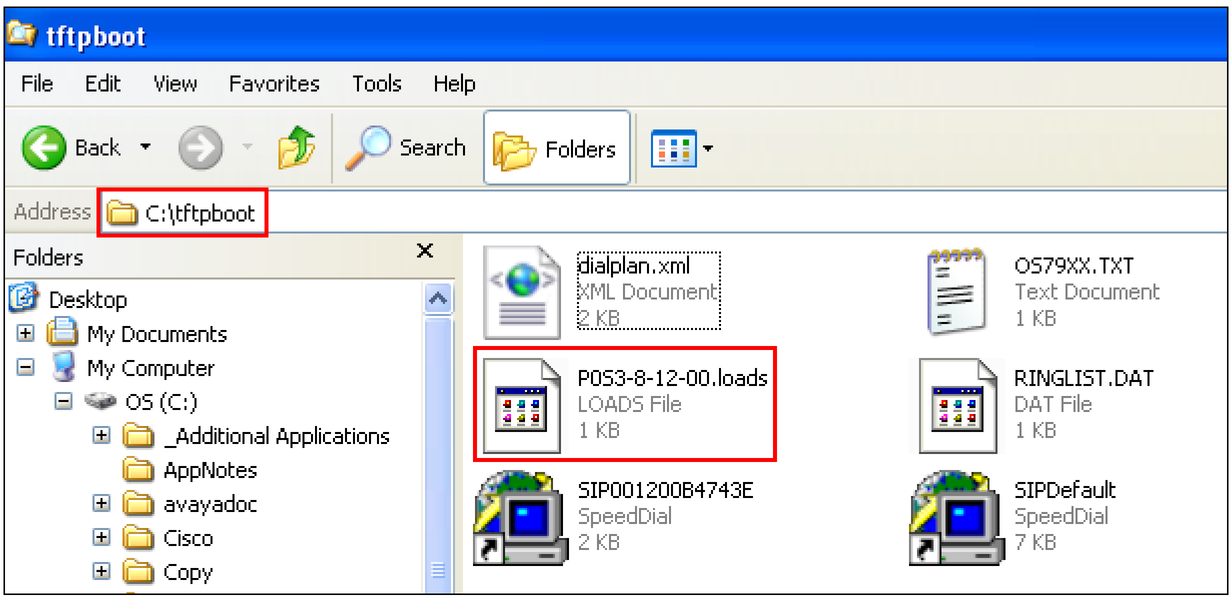

Figure 2: TFTP Directory

A configuration file **SIPDefault.cnf** containing parameter settings is needed in the SIP firmware boot process as shown below. This configuration file contains the image version of the SIP firmware P0S3-8-12-00. A second configuration file SIP<macaddress>.cnf is used to reference the MAC address of the individual Cisco telephone. The MAC address for the Cisco telephone is 001200B4743E so the configuration file will be labeled **SIP001200B4743E.cnf**. The **dialplan.xml** file is used to determine when enough digits have been pressed to complete dialing. The **RINGLIST.DAT** file contains the ringlist information for the Cisco telephone. The **OS79XX.TXT** configuration file is already configured with the text P0S3-8-12-00 will be used in the upgrade process of the SIP firmware.

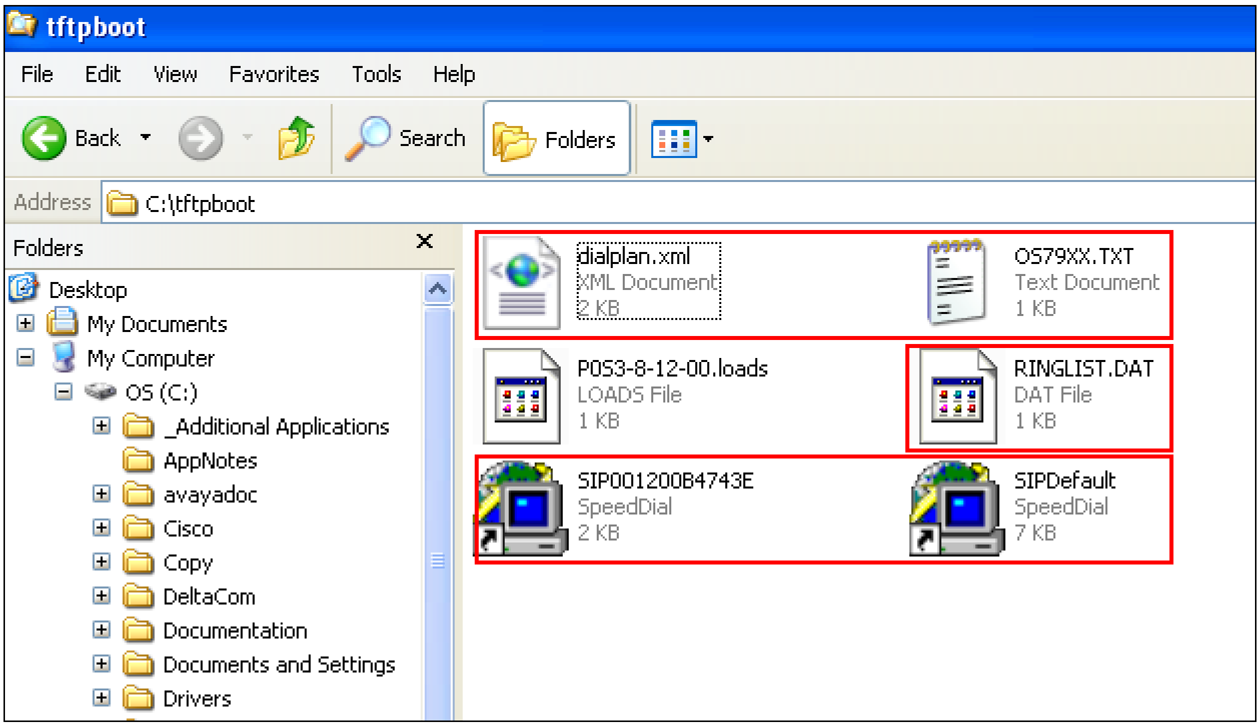

Figure 3: TFTP Directory

3.3. Configure TFTP Server

The SIP firmware files are transferred to the Cisco telephone using a TFTP Server. It was decided to use TFTP Server software compatible with Windows XP as shown below. Under the options settings a **TFTP File system root** folder was created on the TFTP Server. This is where the TFTP Server will tell the Cisco telephones to goto to get the SIP firmware files. A folder was created in directory **C:\tftpboot**. Set the **Read Request Behaviour** to **Give all files** and the **Write Request Behaviour** to **Take all files**. These settings are activated to avoid being prompted to give devices permission every time the device requests a file. The SIP firmware files have already being populated in the TFTP File system root folder labeled tftpboot.

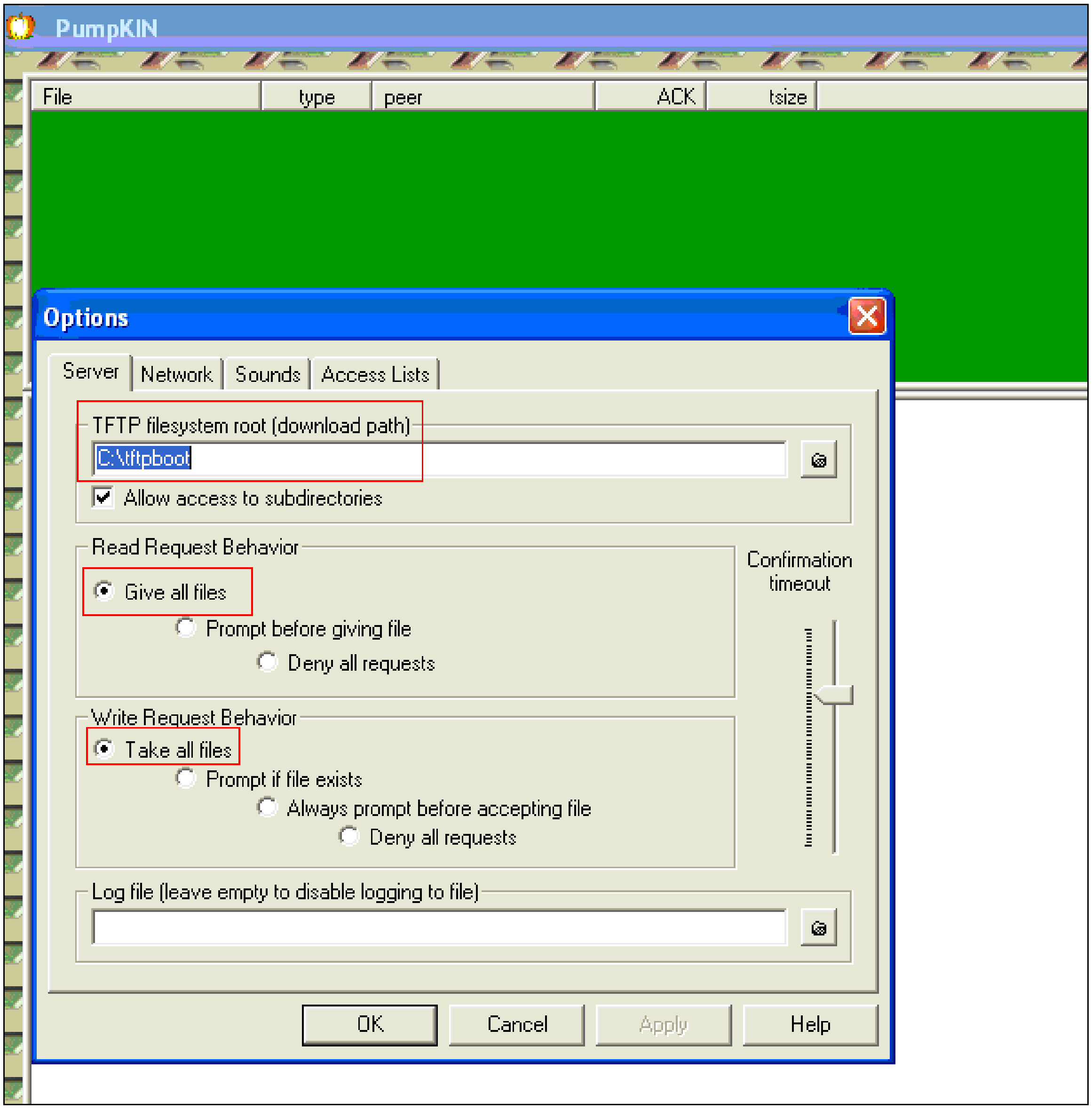

Figure 4: TFTP Server

Connect the Cisco telephone to the network to power up the **Cisco telephone** as shown in **Figure 5**. Press the **Settings** button and **Select Network Configuration**.

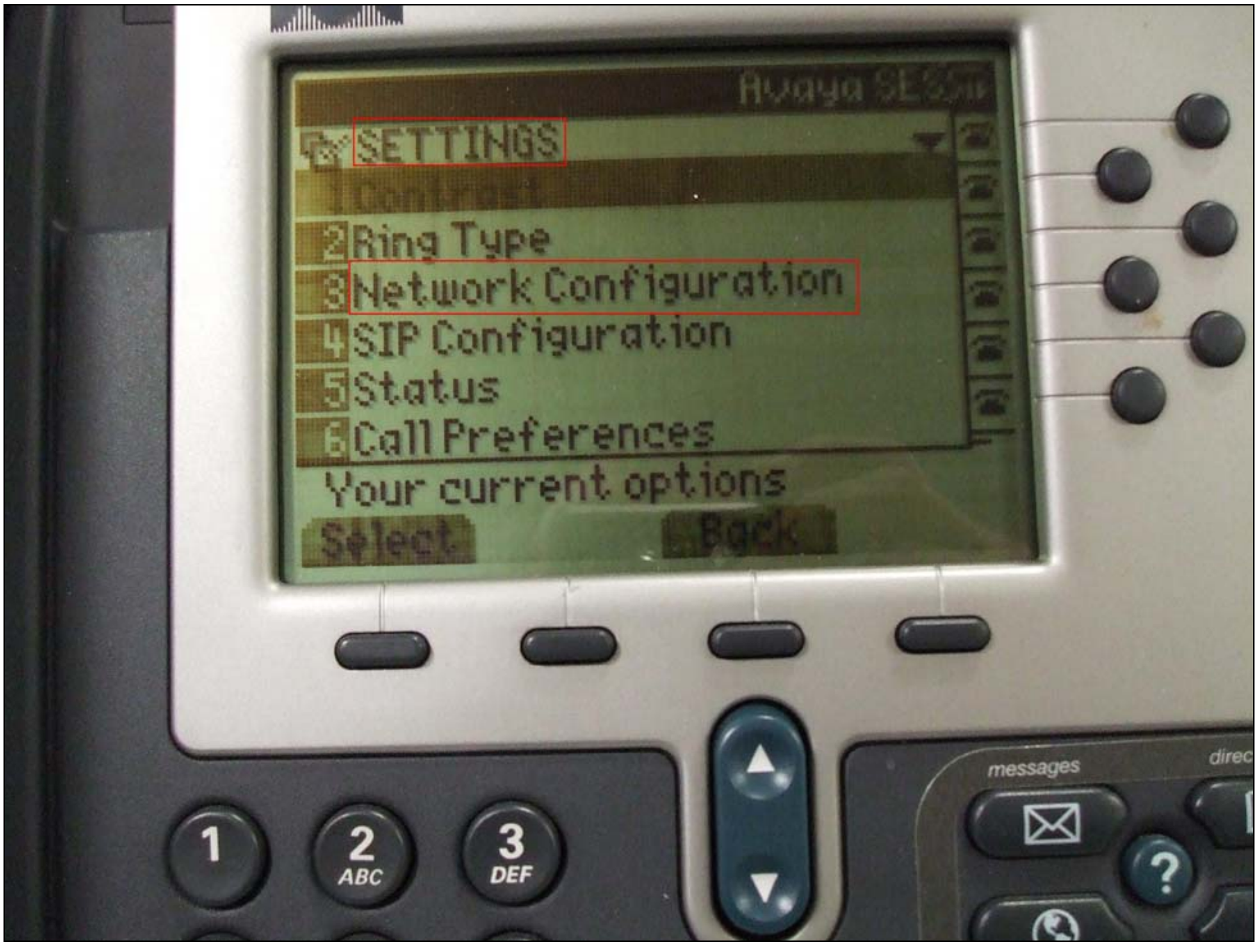

Figure 5: Cisco Telephone

Use the down arrow key on the Cisco telephone to scroll down to the **TFTP Server IP Address** and configure the **TFTP Server IP Address** in Network Setup to the IP Address of the TFTP Server where the SIP firmware files are located. The IP Address of the TFTP Server was **135.64.186.235**.

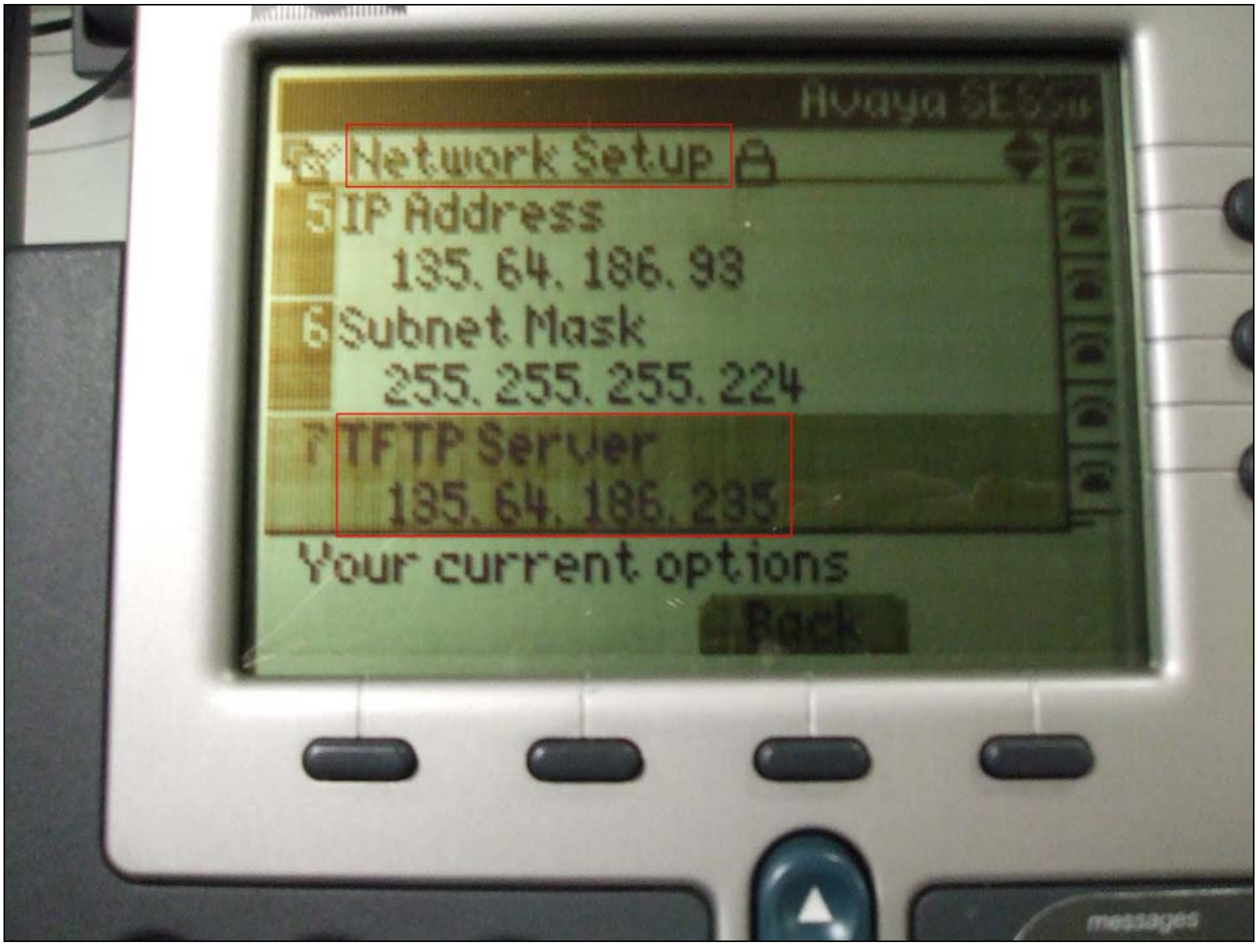

Figure 6: Cisco Telephone

With the SIP firmware files in the TFTP root directory and the **Cisco telephone** in **Figure 7** pointing to the IP Address of the TFTP Server. Disable DHCP on the Cisco telephone and set **DHCP Enabled** to **NO** in the Network Configuration.

Figure 7: Cisco Telephone

Manually program the **IP Address** of the Cisco telephone to **135.64.186.93** in **Figure 8**. Program the **Subnet Mask** as **255.255.255.224** and the **Default Gateway** as 135.64.186.65 (not shown). Press **Save** at the bottom of the screen to save IP Address information.

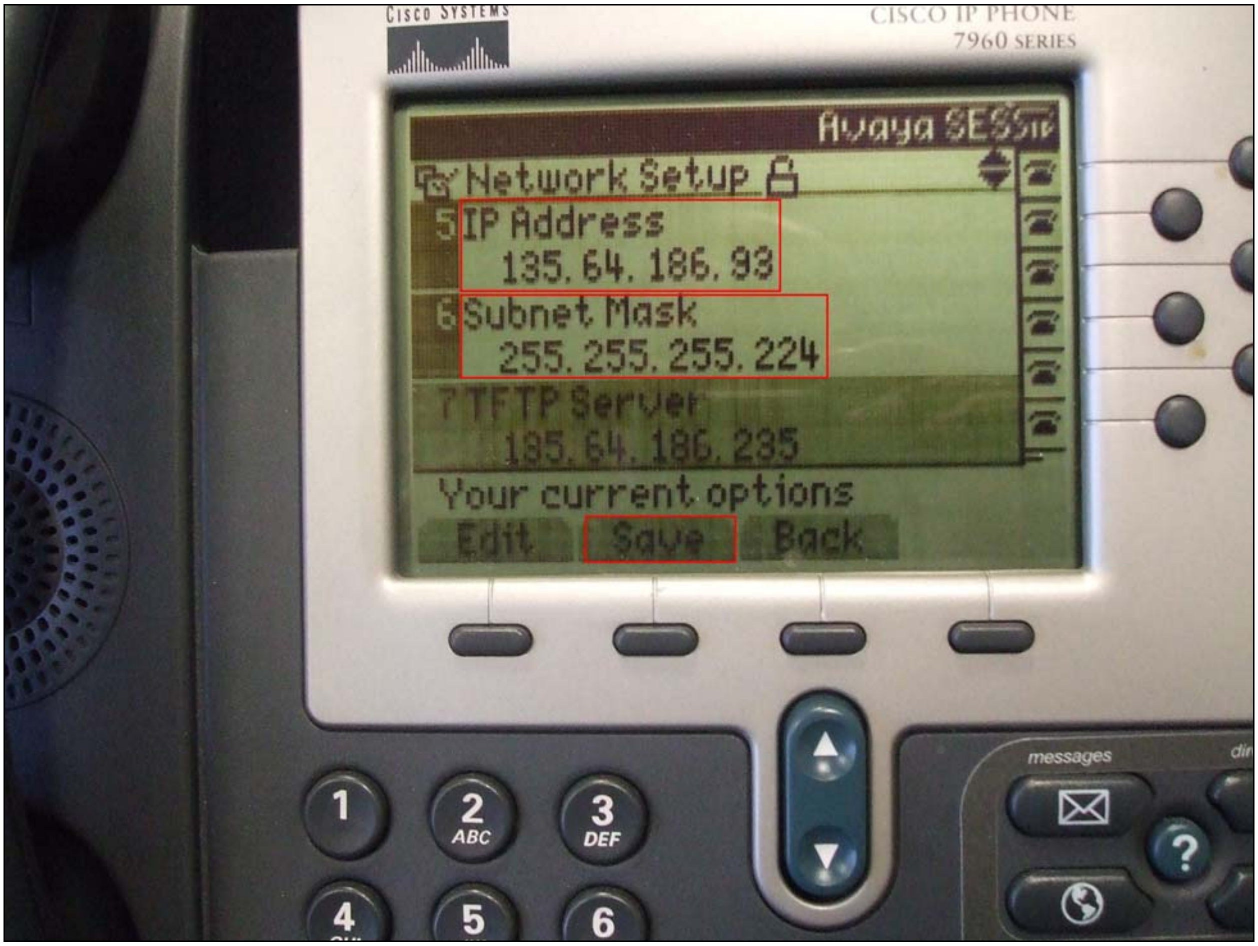

Figure 8: Cisco Telephone

Reboot the Cisco telephone by removing and reconnecting the network cable at the back of the Cisco telephone. **Figure 9** shows the **SIP Firmware boot process**. It shows the SIP firmware files being successfully uploaded to the Cisco telephone as part of the boot process. The configuration files **SIPDefault.cnf** and **SIP001200B4743E.cnf**. The SIP firmware image version **P0S30-8-12-00.loads** was uploaded to the Cisco telephone. The **dialplan.xml** file and the **RINGLIST.DAT** file were uploaded from the TFTP Server.

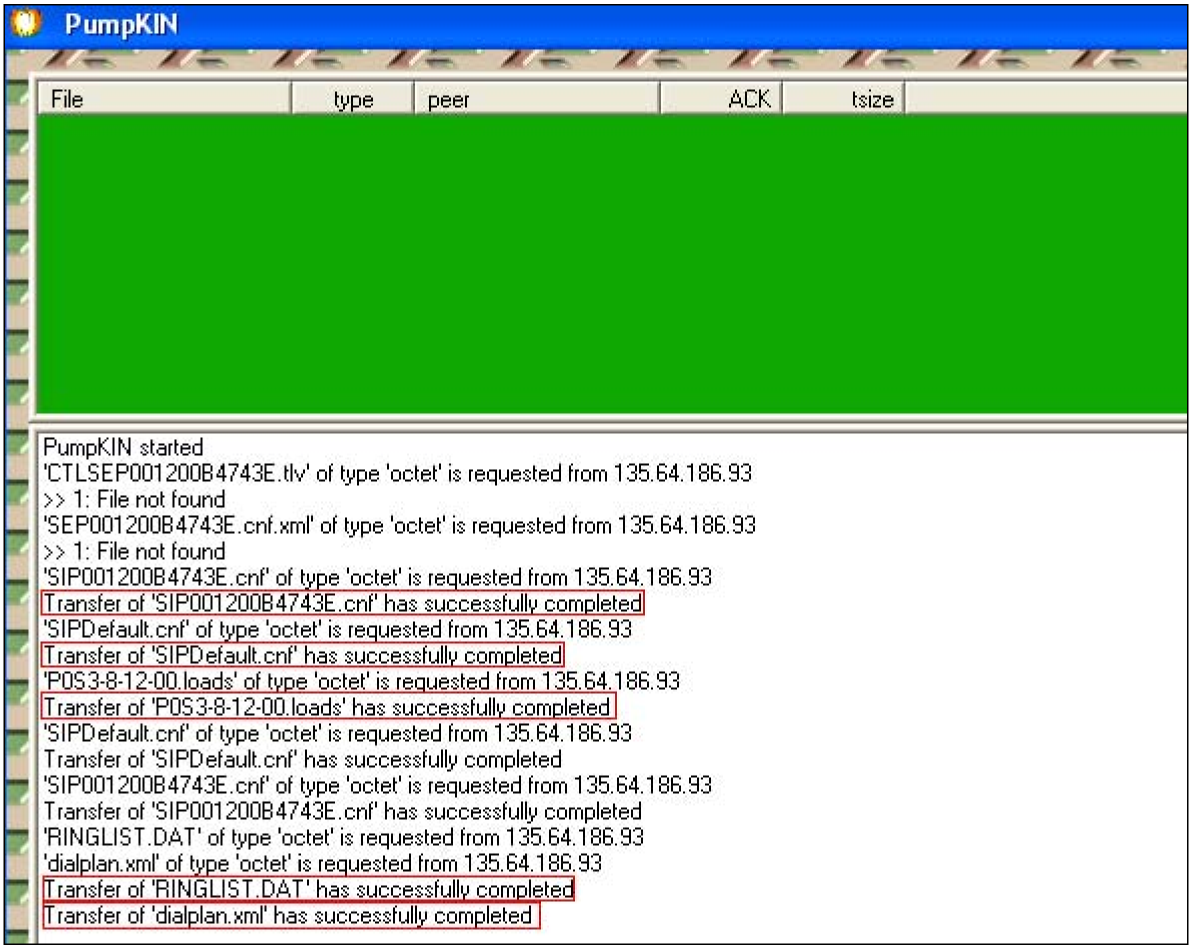

Figure 9: SIP Firmware Boot Process

To confirm the latest SIP firmware is now currently installed on the Cisco telephone, press the **Settings** button on the Cisco telephone. Scroll down to the **Status** Heading in **Figure 10**.

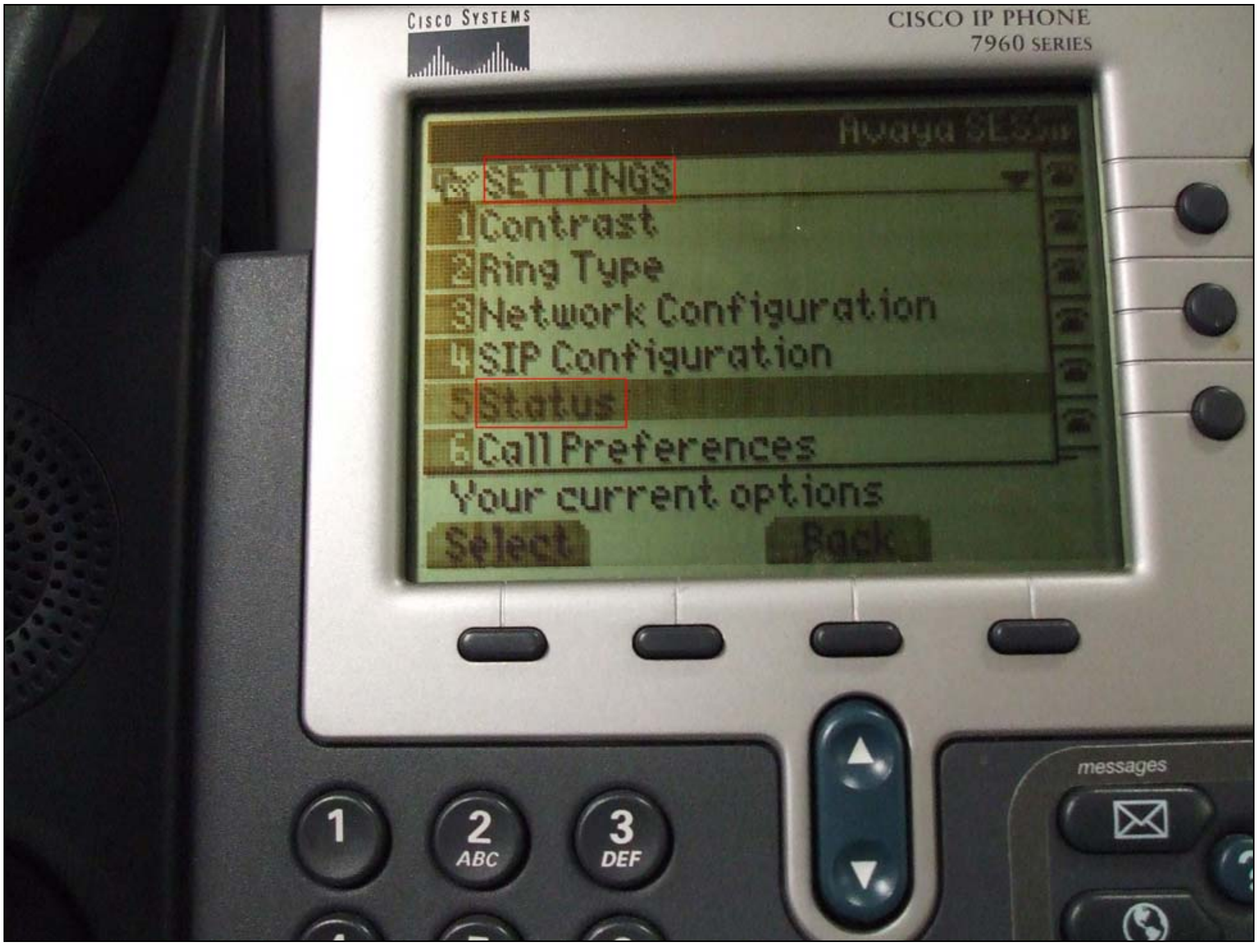

Figure 10: Cisco Telephone

After pressing the **Setting Status** heading and scroll down to the **Firmware Version** heading as shown in **Figure 11**.

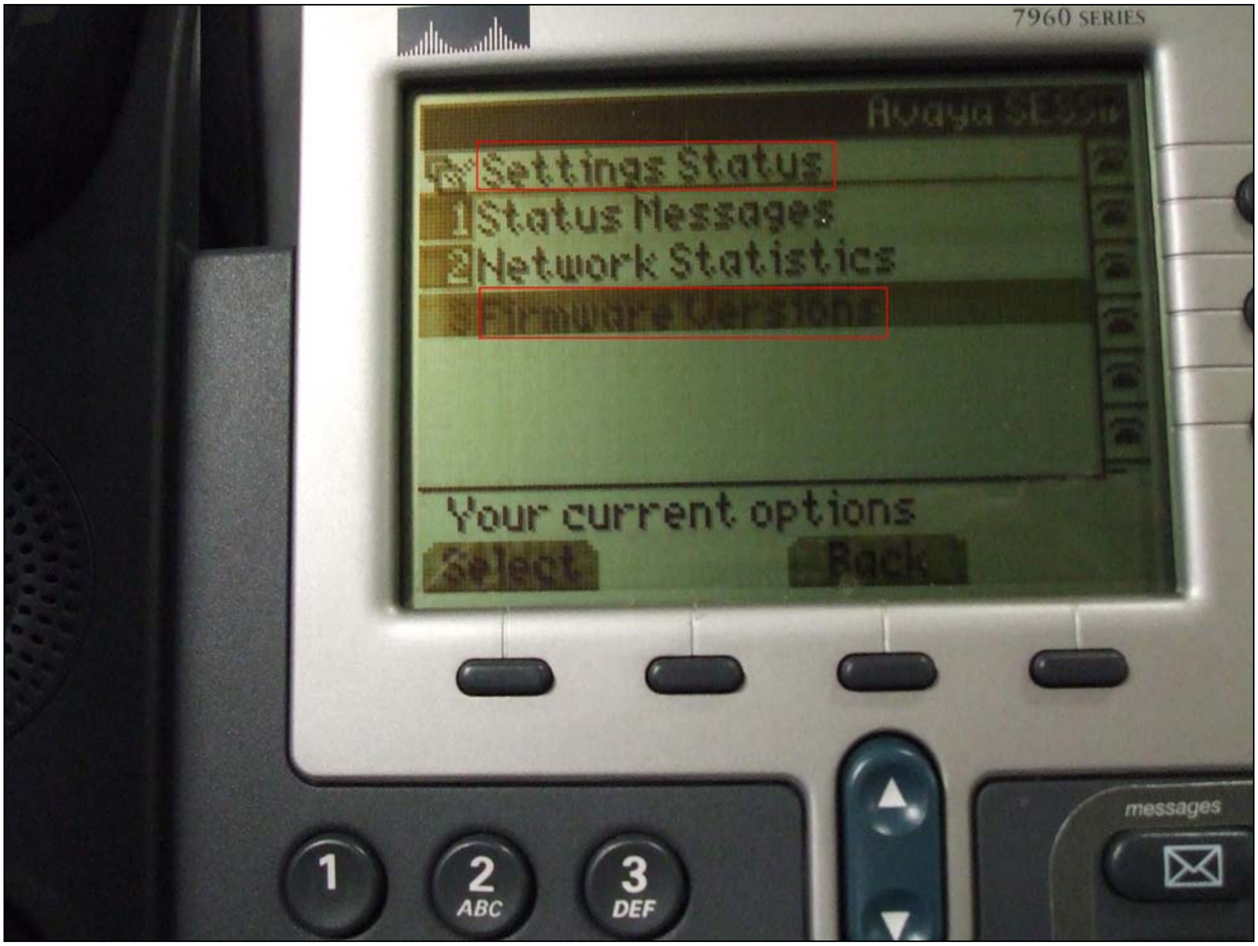

Figure 11: Cisco Telephone

The screenshot in **Figure 12** confirms **Application Load ID P0S3-8-12-00** is currently installed on the Cisco telephone. On the top right corner of the display it states **Avaya SES SIP** to confirm the change from a Cisco IP telephone to a Cisco SIP telephone. The default and telephone specific configuration files where downloaded correctly and the Cisco SIP telephone registered to the SIP Enablement Server. The absence on an X in the phone icon in the top right hand corner of the display confirms this.

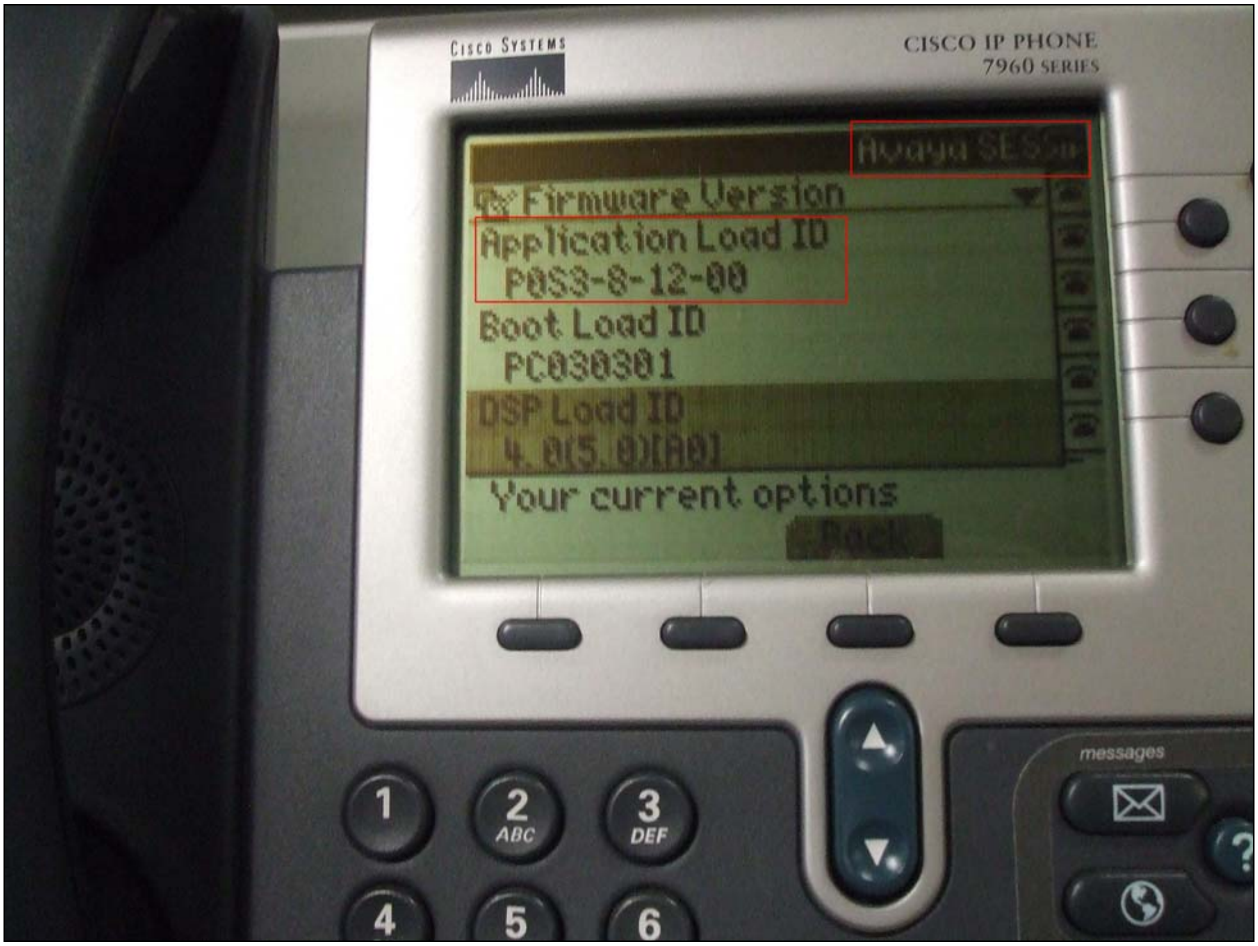

Figure 12: Firmware Version

4. Administer Avaya Aura™ SIP Enablement Services

The following steps describe configuration of SIP Enablement Services for use with Cisco 7960 and 7941 SIP telephones. .

4.1. 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 as shown **Figure 13**.

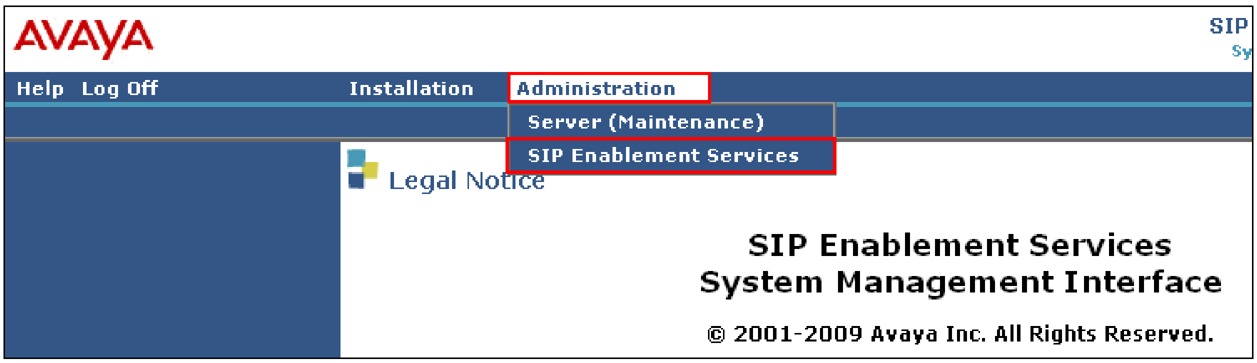

Figure 13: Access SIP Enablement Services

4.2. System Properties

The **View System Properties** screen defines the server's type and domain as shown in **Figure 14**. 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**.

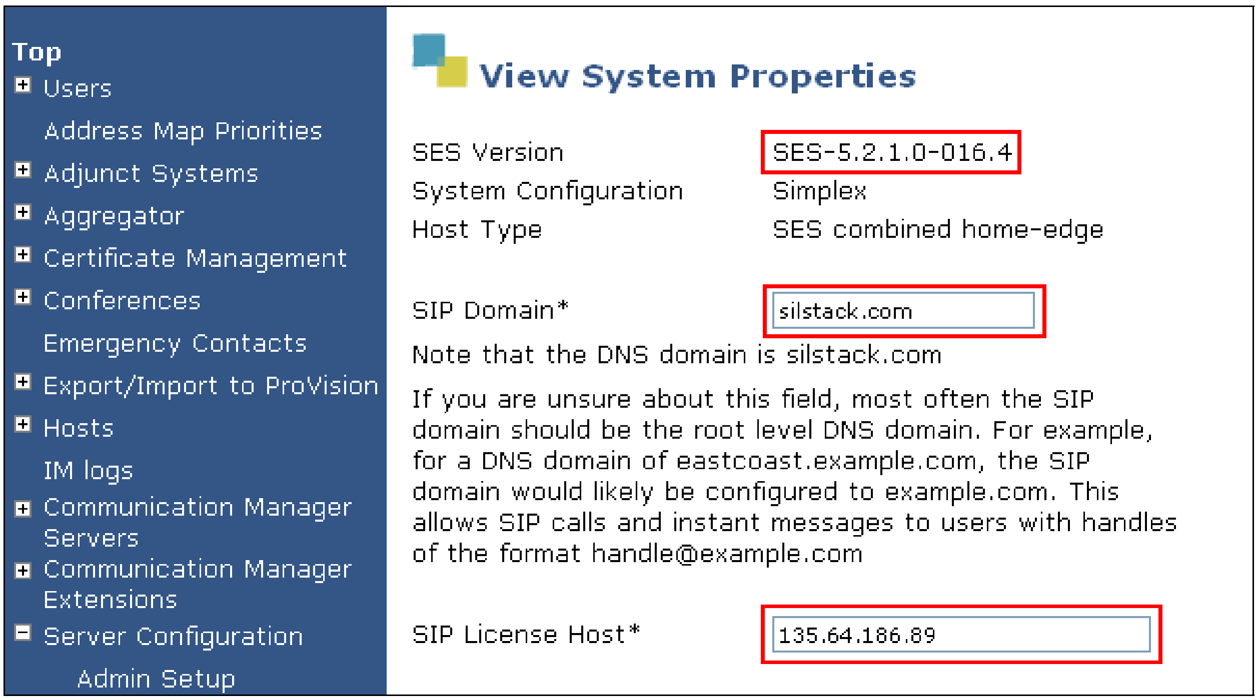

Figure: 14 System Properties

4.3. Add Host Screen

The **Host IP Address** field contains the IP address for this combined home/edge server as shown in **Figure 15**. 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 Cisco SIP telephone uses UDP port 5060 to register to the SIP Enablement Server. 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.

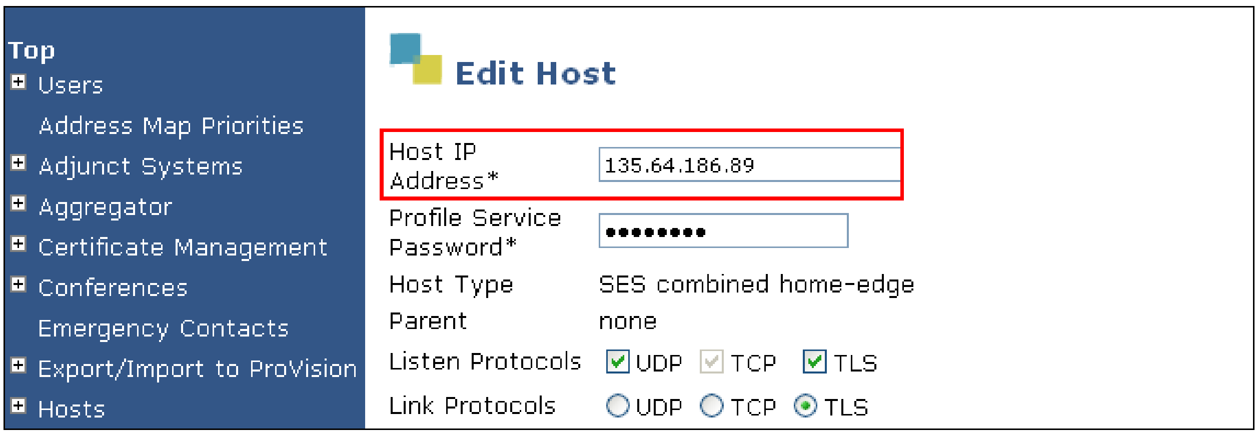

Figure 15: Host Screen

4.4. Administer Cisco SIP Telephones on SES

This screen allows Cisco SIP telephone users to be added to the SES as shown in **Figure 16**. 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 **40030**. The **Password** needs to be six characters long and was set to **123456**. This password is needed when the Cisco 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 Cisco SIP telephone was **Cisco 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 40030 must be added to Communication Manager also as shown in **Figure 17**.

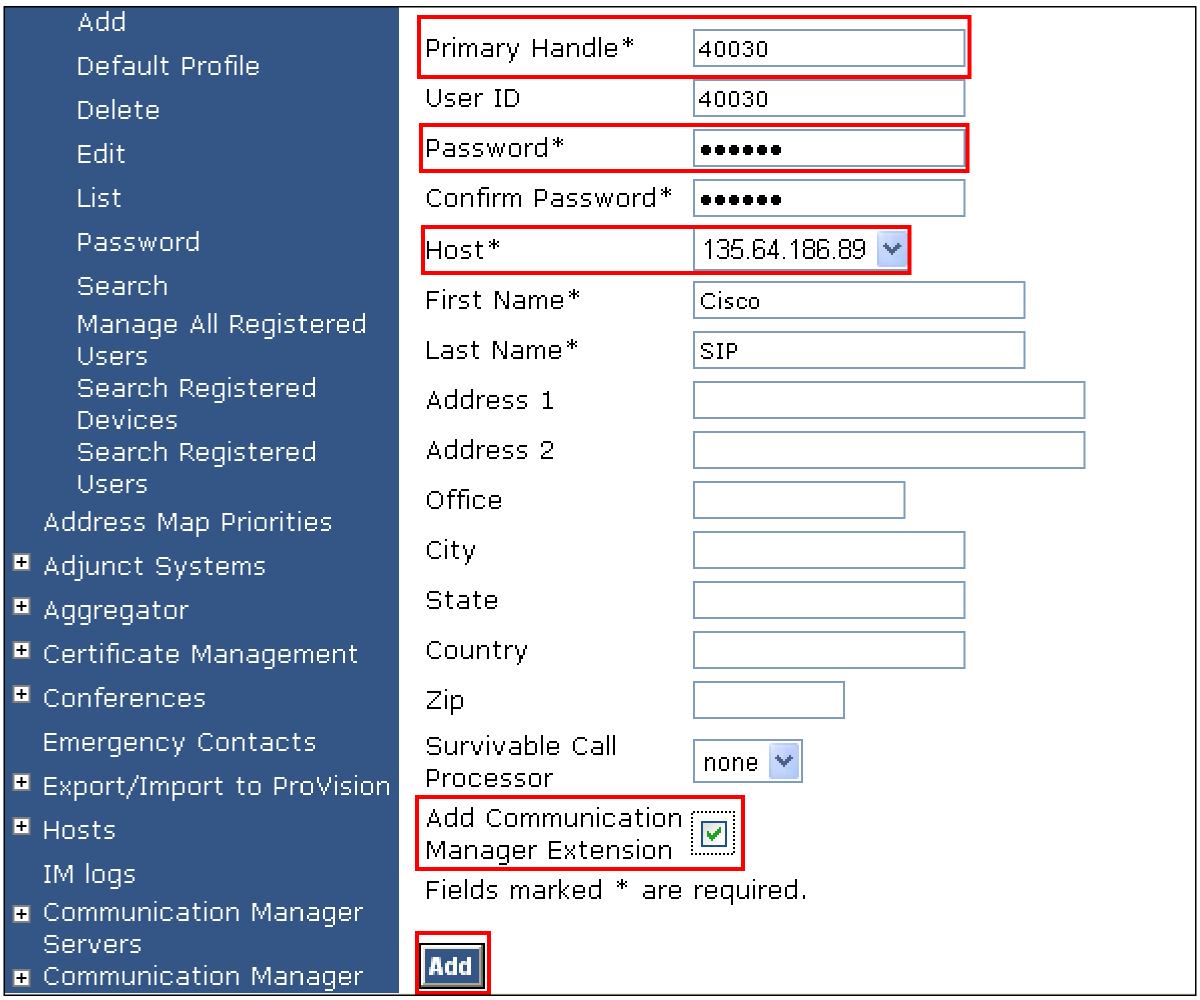

Figure 16 Cisco SIP Phone User

Once the **Add Communication Manager Extension** field is ticked the screen in **Figure 17** appears. Confirm that extension **40030** is the **Communication Manager Extension** and press **Add**.

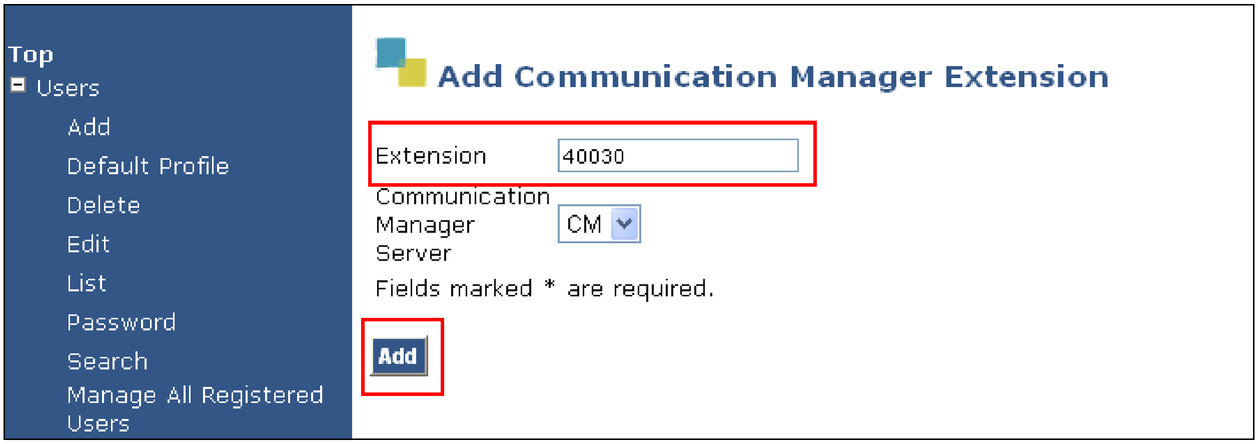

Figure 17: Communication Manager Extension

4.5. Cisco SIP 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. The following screenshot shows the Cisco SIP telephone registered to SIP Enablement Services in **Figure 18**. The Cisco SIP telephone extension 40030 is displayed in the format of **40030@silstack.com SIP, Cisco.** This is the handle, 40030@silstack.com, the first name, Cisco and last name, SIP for user 40030. Silstack.com is the domain name of the SIP Enablement Server. The sip uri address for user 40030 is also displayed as **sip:40030@135.64.186.93.** The IP Address 135.64.186.93 is the physical IP Address of the Cisco SIP telephone. The transport protocol used is **udp** and with udp port number **5060.**

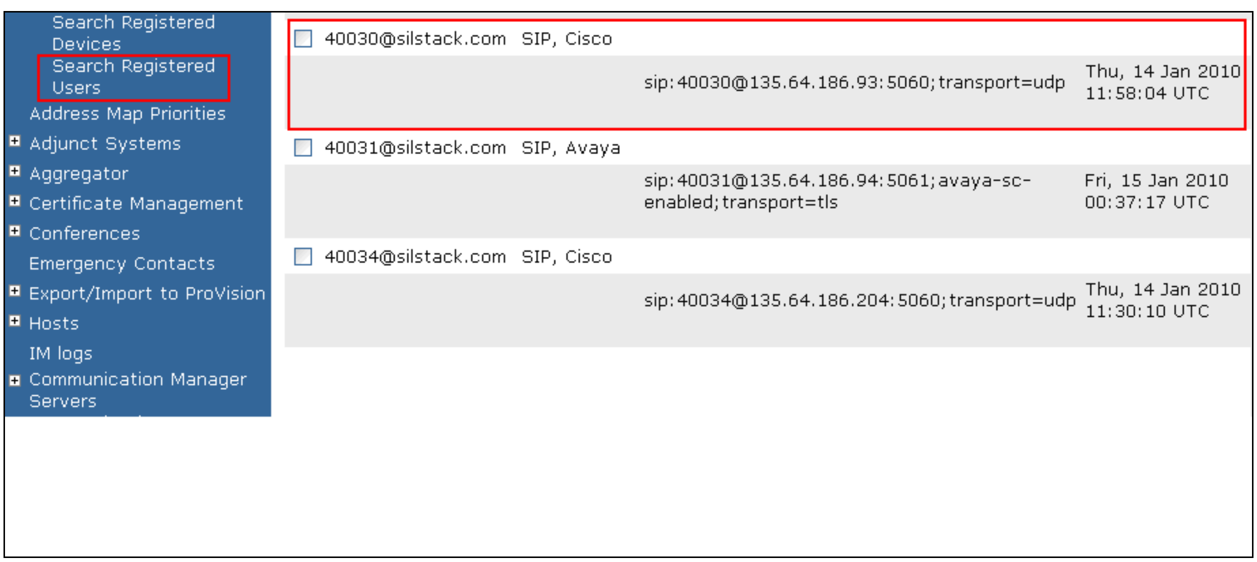

Figure 18: Cisco SIP Telephone Registration

5. Administer Avaya Aura™ Communication Manager

This section highlights the important commands for defining Cisco SIP telephone as an Off-PBX Station (OPS) and administering a SIP Trunk and Signaling Group to carry calls between Cisco SIP endpoints and Avaya SIP and IP endpoints.

5.1. Verify OPS Capacity

Use the **display system-parameters customer-options** command to verify that **Maximum Off-PBX Telephones – OPS** in **Figure 19** 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
```

Figure 19: System-Parameter Customer-Options Page 1

As shown in **Figure 20**, verify there are sufficient licenses to administer the SIP Trunk. This is the **Maximum Administered SIP Trunk** value on **Page 2** of System Parameter Customer-Options.

```
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 3
             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  160
  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
```

Figure 20: System-Parameter Customer-Options Page 2

5.2. Administer Dial Plan Analysis

This section describes the **Dial Plan Analysis** screen as shown in **Figure 21.** 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 SIP 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
     3             5    aar
     4             5    ext
     5             5    ext
```

Figure 21: Dial Plan Analysis

5.3. Administer IP Node-Name

This section describes **IP Node-Name** as shown in **Figure 22.** 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
```

Figure 22: IP Node-Name

5.4. Administer Signaling Group

This section describes the **Signaling Group** screen as shown in **Figure 23**. 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
```

Figure 23: Signaling Group

5.5. Administer Trunk Group

This section describes **Trunk Group** used to carry calls between the Cisco 7960/7941 SIP telephones and the Avaya 9630 SIP and 1616 IP telephone shown in **Figure 24**. 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 the Cisco SIP telephones and Avaya SIP and IP 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
```

Figure 24: Trunk Group Page 1

5.6. Administer IP Network Region

This section describes **IP Network Region** screen shown in **Figure 25.** It was decided to place all 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 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
```

Figure 25: IP Network Region

5.7. Administer IP Codec Set

This section describes the **IP Codec Set** screen in **Figure 26**. It was decided to use IP Codec **G.711MU, G.711A** and **G.729** for testing purposes with the 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:
```

Figure 26: IP Codec Set

5.8. Administer Off PBX Telephone Station Mapping

This section show the **off-pbx-telephone station-mapping** in **Figure 27**. The Cisco SIP telephone extension **40030** 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
 40030             OPS          -      40030            3           1
 40031             OPS          -      40031            3           1
 40034             OPS          -      40034            3           1
```

Figure 27: Off PBX Telephone Station Mapping Page 1

The **Call Limit** is set to **6** as shown below. This is the maximum amount of simultaneous calls for extension 40030. 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
 40030          OPS       6          both          all          none
 40031          OPS       4          both          all          both
 40034          OPS       6          both          all          none
```

Figure 28: Off PBX Telephone Station Mapping Page 2

5.9. Administer Station Screen

This screen describes the **station** form setup for the Cisco SIP telephone on Communication Manager as shown in **Figure 29**. The **Extension** used was **40030** with phone **Type 4620**. Phone type 4620 was the recommended phone type to use for the Cisco SIP telephone. The **Name** of the phone was set to **Cisco SIP** and all other values on **Page 1** of the station form were left as default.

```
display station 40030                                          Page   1 of   5
                                     STATION

Extension: 40030                         Lock Messages? n              BCC: 0
     Type: 4620                          Security Code:                 TN: 1
     Port: S00010                      Coverage Path 1:                COR: 1
     Name: Cisco SIP                   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
```

Figure 29: Station Screen Page 1

6. Verification Steps

The following verification steps were tested using the sample configuration. The following steps can be used to verify installation in the field.

1. Verified the Cisco 7960/7941 SIP telephones were registered to the SIP Enablement Server. This was established when the phone icon located next to each line appearance does not have X next to it.
2. Verified a call could be made with clear audio between the Cisco 7960 SIP telephone and Cisco 7941 SIP telephone. Verified the call was seen to be active on the SIP Trunk within Communication Manager.
3. Verified a call could be made with clear audio from both the Cisco 7960/7941 SIP telephones to the Avaya 9630 SIP telephone. Verified the call was seen to be active on the SIP Trunk within Communication Manager. This was successful.
4. Verified a call could be made with clear audio from both the Cisco 7960/7941 SIP telephones to the Avaya 1616 IP telephone. Verified the call was seen to be active on the SIP Trunk within Communication Manager. This was successful.
5. Verified supplementary features such as Call Hold, Call Forward, Conference and Transfer could be completed between the Cisco endpoints and the Avaya endpoints. This was successful.

7. Conclusion

These Application Notes have described the administration steps required to register Cisco 7960 and 7941 SIP Telephones to Avaya Avaya™ SIP Enablement Services with Avaya Avaya™ Communication Manager. Administration of a SIP Trunk to carry calls between Cisco SIP endpoints and Avaya SIP and IP endpoints.

8. 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] SIP Support in Avaya Communication Aura™ Manager Running on Avaya Servers, May 2009 Document Number 555-245-206.

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

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

[4] Cisco Unified Communication Manager Documentation Guide for Release 7.0(2), available at www.cisco.com

[5] Cisco 7960 and 7941 IP Phone Firmware Upgrade Matrix, November 13th 2006, available at www.cisco.com

[6] Cisco Unified IP Phone 7960G and 7941G (SIP) Release Notes for Firmware Release 8.11, available at www.cisco.com

Appendices A

Sample default configuration file for Cisco 7960 SIP Telephone (SIPDefault.cnf)

SIP Default Generic Configuration File

Image Version
image_version: P0S3-8-12-00

Proxy Server
proxy1_address: "135.64.186.89" ; Can be dotted IP or FQDN
proxy2_address: "135.64.186.89" ; Can be dotted IP or FQDN
proxy3_address: "135.64.186.89" ; Can be dotted IP or FQDN
proxy4_address: "135.64.186.89" ; Can be dotted IP or FQDN
proxy5_address: "135.64.186.89" ; Can be dotted IP or FQDN
proxy6_address: "135.64.186.89" ; Can be dotted IP or FQDN

Proxy Server Port (default - 5060)
proxy1_port: 5060
proxy2_port: 5060
proxy3_port: 5060
proxy4_port: 5060
proxy5_port: 5060
proxy6_port: 5060

Proxy Registration (0-disable (default), 1-enable)
proxy_register: 1

Phone Registration Expiration [1-3932100 sec] (Default - 3600)
timer_register_expires: 3600

Codec for media stream (g711ulaw (default), g711alaw, g729a)
preferred_codec: g711ulaw

TOS bits in media stream [0-5] (Default - 5)
tos_media: 5

Inband DTMF Settings (0-disable, 1-enable (default))
dtmf_inband: 1

Out of band DTMF Settings (none-disable, avt-avt enable (default), avt_always - always avt)
dtmf_outofband: avt

DTMF dB Level Settings (1-6dB down, 2-3db down, 3-nominal (default), 4-3db up, 5-6dB up)
dtmf_db_level: 3

SIP Timers
timer_t1: 500 ; Default 500 msec
timer_t2: 4000 ; Default 4 sec
sip_retx: 10 ; Default 10
sip_invite_retx: 6 ; Default 6
timer_invite_expires: 180 ; Default 180 sec

####### New Parameters added in Release 2.0 #######

```
# Dialplan template (.xml format file relative to the TFTP root directory)
dial_template: dialplan

# TFTP Phone Specific Configuration File Directory
tftp_cfg_dir: ""                ; Example:  ./sip_phone/

# Time Server (There are multiple values and configurations refer to Admin Guide for Specifics)
sntp_server: ""                         ; SNTP Server IP Address
sntp_mode: unicast              ; unicast, multicast, anycast, or directedbroadcast (default)
time_zone: EST                          ; Time Zone Phone is in
dst_offset: 1                           ; Offset from Phone's time when DST is in effect
dst_start_month: April                  ; Month in which DST starts
dst_start_day: ""               ; Day of month in which DST starts
dst_start_day_of_week: Sun              ; Day of week in which DST starts
dst_start_week_of_month: 1              ; Week of month in which DST starts
dst_start_time: 02              ; Time of day in which DST starts
dst_stop_month: Oct                     ; Month in which DST stops
dst_stop_day: ""                ; Day of month in which DST stops
dst_stop_day_of_week: Sunday            ; Day of week in which DST stops
dst_stop_week_of_month: 8               ; Week of month in which DST stops 8=last week of month
dst_stop_time: 2                ; Time of day in which DST stops
dst_auto_adjust: 1                      ; Enable(1-Default)/Disable(0) DST automatic adjustment
time_format_24hr: 1                     ; Enable(1 - 24Hr Default)/Disable(0 - 12Hr)

# Do Not Disturb Control (0-off, 1-on, 2-off with no user control, 3-on with no user control)
dnd_control: 0                          ; Default 0 (Do Not Disturb feature is off)

# Caller ID Blocking (0-disbaled, 1-enabled, 2-disabled no user control, 3-enabled no user control)
callerid_blocking: 0                    ; Default 0 (Disable sending all calls as anonymous)

# Anonymous Call Blocking (0-disabled, 1-enabled, 2-disabled no user control, 3-enabled no user control)
anonymous_call_block: 0                 ; Default 0 (Disable blocking of anonymous calls)

# DTMF AVT Payload (Dynamic payload range for AVT tones - 96-127)
dtmf_avt_payload: 101                   ; Default 101

# Sync value of the phone used for remote reset
sync: 1                                 ; Default 1

####### New Parameters added in Release 2.1 #######

# Backup Proxy Support
proxy_backup: ""                ; Dotted IP of Backup Proxy
proxy_backup_port: 5060                 ; Backup Proxy port (default is 5060)

# Emergency Proxy Support
proxy_emergency: ""                     ; Dotted IP of Emergency Proxy
proxy_emergency_port: 5060              ; Emergency Proxy port (default is 5060)

# Configurable VAD option
enable_vad: 0                           ; VAD setting 0-disable (Default), 1-enable

####### New Parameters added in Release 2.2 ######

# NAT/Firewall Traversal
```

```
nat_enable: 0                   ; 0-Disabled (default), 1-Enabled
nat_address: ""                 ; WAN IP address of NAT box (dotted IP or DNS A record only)
voip_control_port: 5060         ; UDP port used for SIP messages (default - 5060)
start_media_port: 16384    ; Start RTP range for media (default - 16384)
end_media_port: 32766      ; End RTP range for media (default - 32766)
nat_received_processing: 0      ; 0-Disabled (default), 1-Enabled

# Outbound Proxy Support
outbound_proxy: ""              ; restricted to dotted IP or DNS A record only
outbound_proxy_port: 5060       ; default is 5060

####### New Parameter added in Release 3.0 #######

# Allow for the bridge on a 3way call to join remaining parties upon hangup
cnf_join_enable : 1             ; 0-Disabled, 1-Enabled (default)

####### New Parameters added in Release 3.1 #######

# Allow Transfer to be completed while target phone is still ringing
semi_attended_transfer: 1  ; 0-Disabled, 1-Enabled (default)

# Telnet Level (enable or disable the ability to telnet into the phone)
telnet_level: 1                 ; 0-Disabled (default), 1-Enabled, 2-Privileged

####### New Parameters added in Release 4.0 #######

# XML URLs
services_url: ""           ; URL for external Phone Services
directory_url: ""          ; URL for external Directory location
logo_url: ""                    ; URL for branding logo to be used on phone display

# HTTP Proxy Support
http_proxy_addr: ""             ; Address of HTTP Proxy server
http_proxy_port: 80             ; Port of HTTP Proxy Server (80-default)

# Dynamic DNS/TFTP Support
dyn_dns_addr_1: ""          ; restricted to dotted IP
dyn_dns_addr_2: ""          ; restricted to dotted IP
dyn_tftp_addr: ""          ; restricted to dotted IP

# Remote Party ID
remote_party_id: 0              ; 0-Disabled (default), 1-Enabled

####### New Parameters added in Release 4.4 #######

# Call Hold Ringback (0-off, 1-on, 2-off with no user control, 3-on with no user control)
call_hold_ringback: 0           ; Default 0 (Call Hold Ringback feature is off)

####### New Parameters added in Release 6.0 #######

# Dialtone Stutter for MWI
stutter_msg_waiting: 0          ; 0-Disabled (default), 1-Enabled

# RTP Call Statistics (SIP BYE/200 OK message exchange)
call_stats: 0                   ; 0-Disabled (default), 1-Enabled
```

Appendices B

Sample telephone specific configuration file for Cisco 7960 SIP Telephone (SIP001200B4743E.cnf)

SIP Configuration Phone-Specific File
Line 1 appearance
line1_name: 40030
Line 1 Registration Authentication
line1_authname: "40030"
Line 1 Registration Password
line1_password: "123456"
Line 2 appearance
line2_name: 40030
Line 2 Registration Authentication
line2_authname: "40030"
Line 2 Registration Password
line2_password: "123456"
####### New Parameters added in Release 2.0 #######
All user_parameters have been removed
Phone Label (Text desired to be displayed in upper right corner)
phone_label: "Avaya SES" ; Has no effect on SIP messaging
Line 1 Display Name (Display name to use for SIP messaging)
line1_displayname: "Cisco SIP"
Line 2 Display Name (Display name to use for SIP messaging)
line2_displayname: "Cisco SIP"
####### New Parameters added in Release 3.0 ######
Phone Prompt (The prompt that will be displayed on console and telnet)
phone_prompt: "SIP Phone" ; Limited to 15 characters (Default - SIP Phone)
Phone Password (Password to be used for console or telnet login)
phone_password: "cisco" ; Limited to 31 characters (Default - cisco)
User classification used when Registering [none(default), phone, ip]
user_info:none

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