Avaya Solution & Interoperability Test Lab

Configuring SIP IP Telephony Using Avaya Converged Communications Server, Avaya Communication Manager, and Cisco 7940/7960 SIP Telephones – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to connect Cisco 7940/7960 SIP telephones to a SIP infrastructure consisting of the Avaya Converged Communications Server and Avaya S8300 Media Server with G700 Media Gateway. Also described is how Avaya Outboard Proxy SIP station features can be made available to Cisco telephones in addition to the standard features supported in the telephone. The configuration steps described are also applicable to other Linux-based Avaya Media Servers and Media Gateways running Avaya Communication Manager.

1. Introduction

1.1. Background

With the introduction of the SIP protocol standard that supports telephony as well as a wide range of other communication modes, there is a much broader range of SIP telephones available to customers. This allows customers to replace their existing telephony infrastructure with Avaya servers and re-use their existing telephones.

In addition, Avaya Communication Manager running on Avaya Media Servers and Gateways has the capability to extend advanced telephony features to Outboard Proxy SIP stations. This feature set can be extended to non-Avaya SIP phones, providing enhanced calling features in advance of SIP protocol definitions and telephone implementations. See Section 3.1.

These Application Notes describe the configuration steps for using the Cisco 7940/7960 SIP telephones with the Avaya Converged Communications Server, S8300 Media Server, and G700 Media Gateway. Only those configuration steps pertinent to interoperability of Cisco and Avaya equipment are covered. General administration information can be found in the product documentation as well as the specific references listed in Section 8. The configuration described should be applicable to other Linux-based Avaya Media Servers and Media Gateways running Avaya Communication Manager.

1.2. Configuration

The configuration used as an example in these Application Notes is shown in **Figure 1**. Several Cisco 7940/7960 SIP telephones are configured in a single subnet with the Avaya Converged Communications Server and S8300 Media Server with G700 Media Gateway. A PC provides web browser and TFTP server support. The telephones are registered to the Converged Communications Server and are also administered as Outboard Proxy SIP Stations in the S8300, so that in addition to the SIP telephony features supported by the phones, Outboard Proxy SIP features are available from Communication Manager. The media server also contains the Avaya IA770 INTUITY™ AUDIX® Messaging Application for voice messaging support. These Application Notes do not address configuration of the Avaya 4602 SIP telephones, which were successfully tested using the standard product configuration steps.

The main difference between the Cisco 7940 and 7960 SIP telephones is the number of line appearances supported by each phone (two and six, respectively). The configuration steps described in these Application Notes apply to both models. **Table 1** profiles the network management capabilities of the phones.

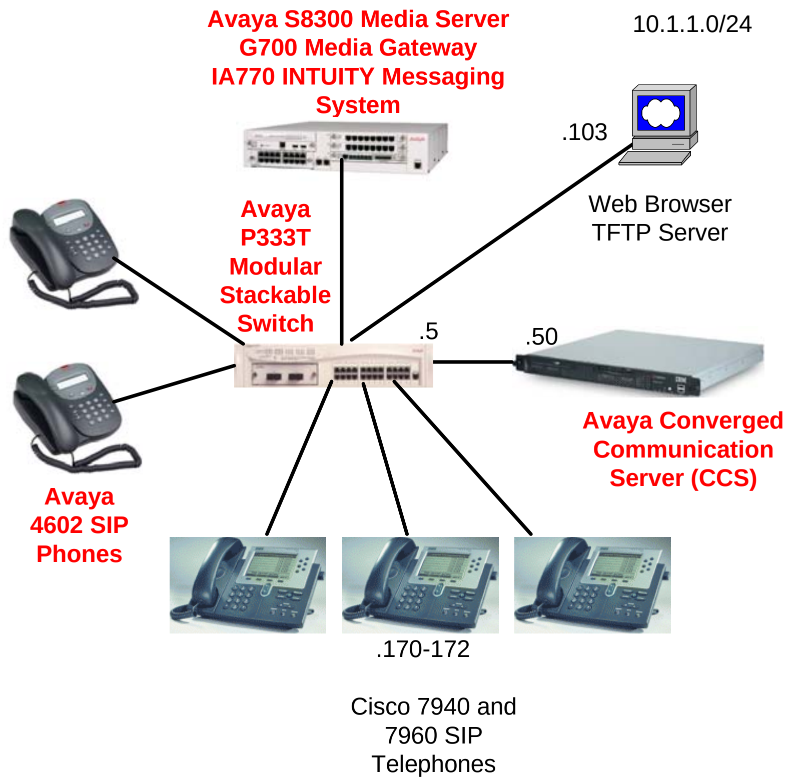


Figure 1: Avaya SIP Test Configuration with Cisco 7940/7960 SIP Phones

Administration mechanisms	Configuration files, Telnet
Administration levels	Administrator
File transfer server	TFTP
Error logs	Stored and viewed at phone
802.3af Power over Ethernet Support	No
SNMP support	None

Table 1: Network Management Capabilities of the 7940/7960

2. Equipment and Software Validated

The following equipment and software were used in the configuration shown in **Table 2**. Be sure to use the software version combination shown when following these Application Notes.

Equipment	Software
Avaya Converged Communication Server	2.0
Avaya P333T Modular Stackable Switch	3.12.1
Avaya S8300 Media Server with G700 Media Gateway	Communication Manager 2.0.1
Avaya IA770 INTUITY®AUDIX™ Messaging System	N1.2-5.2
Cisco 7940/7960 SIP Telephones	POS3-06-3

Table 2: Equipment and Software Versions Used

3. Supported Calling Features

3.1. The SIPPING-19

In addition to basic calling capabilities, the Internet Engineering Task Force (IETF) has defined a supplementary set of calling features, often referred to as the SIPPING-19 [2]. This provides a useful framework to describe product capabilities and compare calling features supported by various equipment vendors. **Table 3** gives a summary of calling features supported on the Cisco 7940/7960 SIP telephones within the Avaya SIP infrastructure. Some features require only the Converged Communications Server and Cisco telephones, while others additionally require Avaya Communication Manager and the Outboard Proxy SIP feature set.

Avaya Outboard Proxy SIP provides advanced calling features beyond the SIPPING-19 that can be extended to the telephone. These features are summarized in **Table 4**. Since the Cisco 7940/7960 SIP telephones are compatible with Outboard Proxy SIP, these features can be made available to the user. Details on their operation and administration can be found in Reference [4].

Section 4 of these Application Notes describes the steps for configuring the Cisco telephone and Avaya Converged Communications Server to support the basic feature set (those indicated by a "yes" in the *Phone* + Converged Communications Server column of **Table 3**). To take advantage of the extended feature set, including additional SIPPING-19 features (indicated in the *Telephone + Converged Communications Server + Communication Manager/Outboard Proxy SIP* column) and others listed in **Table 4**, the Cisco telephones must be configured for Outboard Proxy SIP support. The additional configuration steps required for this are described in Section 5.

No.	FEATURE	Telephone + CCS	Telephone + CCS + Communication Manager/OPS	COMMENTS
1	Call Hold	YES	YES	
2	Consultation Hold	YES	YES	
3	Music On Hold	NO	NO	Available in OPS (future release)
4	Unattended Transfer	YES	YES	
5	Attended Transfer	YES	YES	
6	Transfer - IM	NO	NO	
7	Call Forward Unconditional	YES	YES	
8	Call Forward Busy	NO	YES	OPS
9	Call Forward No Answer	NO	YES	OPS
10	3-way conference - 3rd party added by user	YES	YES	
11	3-way conference - 3rd party calls and is joined by user	NO	NO	
12	Single Line Extension (forking)	NO	YES	OPS + bridged appearance
13	Find-me	NO	YES	OPS + bridged appearance
14	Incoming Call Screening	NO	NO	
15	Outgoing Call Screening	NO	YES	OPS + Class of Restriction
16	Call Park	NO	YES	OPS call park/answer back
17	Call Pick-up	NO	YES	OPS call pick-up & directed call pick-up
18	Automatic Redial	NO	NO	Available in OPS (future release)
19	Click to Dial	NO	NO	

Table 3: SIPPING-19 Telephony Feature Support

No.	FEATURE	Telephone + CCS	Telephone + CCS + Communication Manager/OPS
1	Active Appearance Select	NO	YES
2	Conference on Answer	NO	YES
3	Calling Number Block	NO	YES
4	Calling Number Unblock	NO	YES
5	Drop Last Added Party	NO	YES
6	Held Appearance Select	NO	YES
7	Idle Appearance Select	NO	YES
8	Last Number Dialed	NO	YES
9	Malicious Call Trace	NO	YES
10	Malicious Call Trace Cancel	NO	YES
11	Priority Call	NO	YES
12	Send All Calls	NO	YES
13	Send All Calls Cancel	NO	YES
14	Transfer On Hang-Up	NO	YES
15	Transfer to Voice Mail	NO	YES

Table 4: Outboard Proxy SIP Telephony Features Beyond SIPPING-19

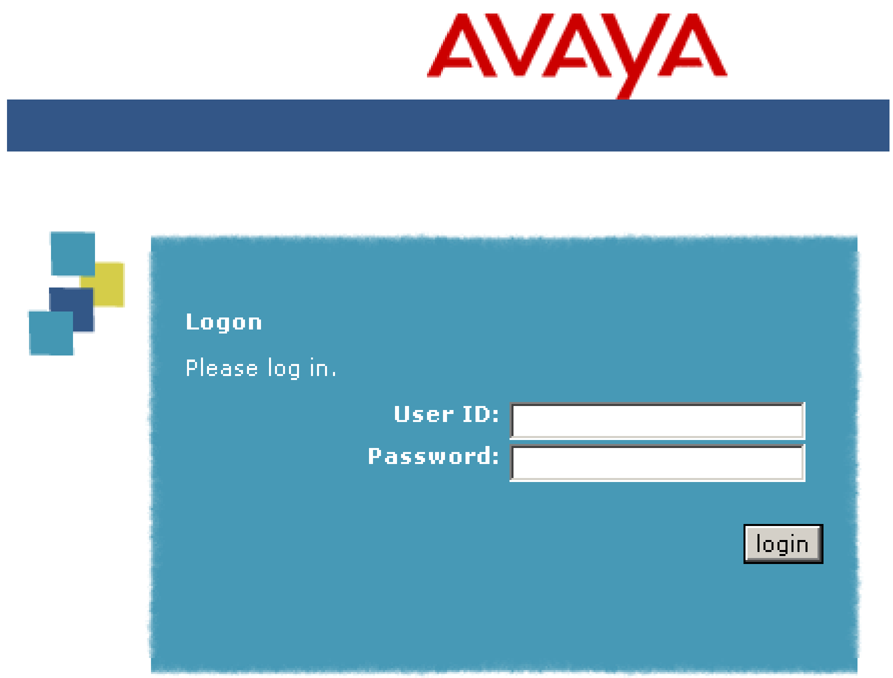
3.2. Message Waiting Indicator (MWI)

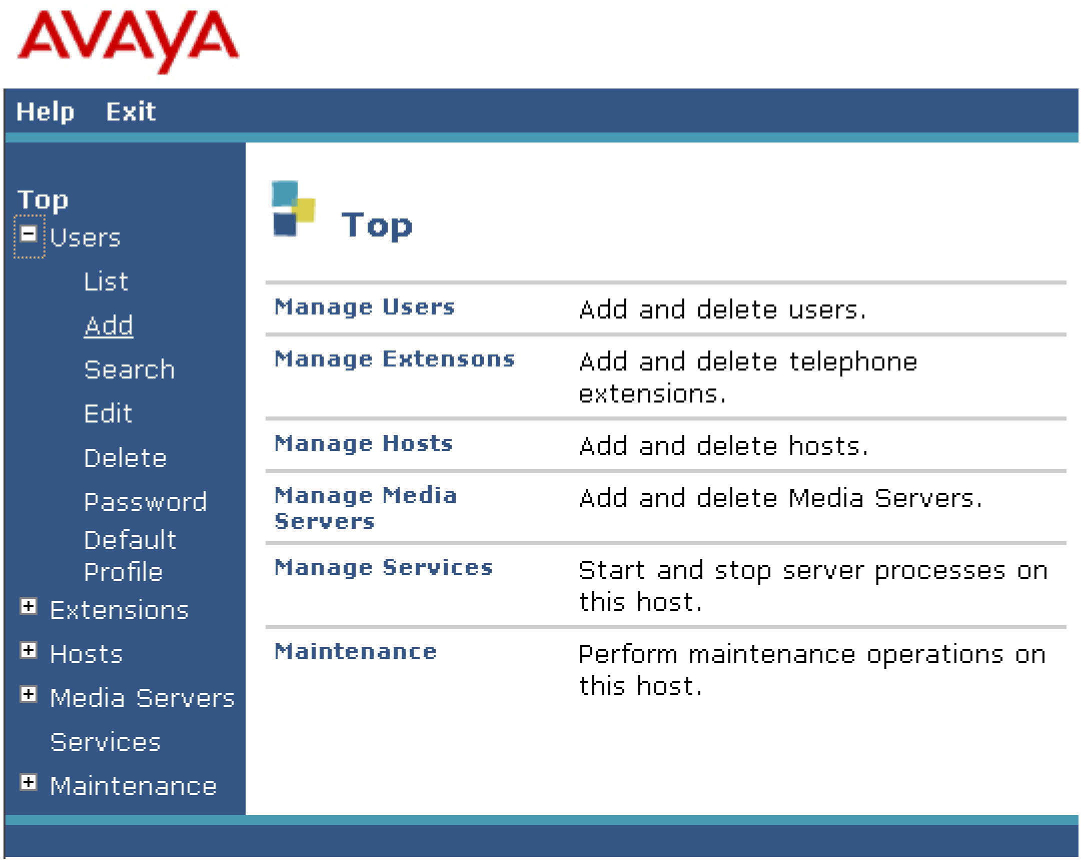
With the Outboard Proxy SIP extended feature set, a SIP telephone that supports IETF RFC 3265 and MWI Draft 4 (Subscribe/Notify method) will illuminate/extinguish its MWI lamp when voice messages are left/read for that extension. Since the Cisco 7940/7960 SIP phones support only the unsolicited Notify method for MWI, this feature is not currently supported in Communication Manager.

4. Configuring for the Basic Feature Set

4.1. Administer Users on the Avaya Converged Communications Server

The following steps describe configuration of the Avaya Converged Communications Server to for use with Cisco 7940/7960 SIP telephones. Other standard administration functions are covered in Reference [1].

Steps	Description
1.	The Avaya Converged Communications Server is configured using a web browser. Set the URL of the browser to the IP address of the Converged Communications Server, and log in as *admin* using the appropriate administrator password. 

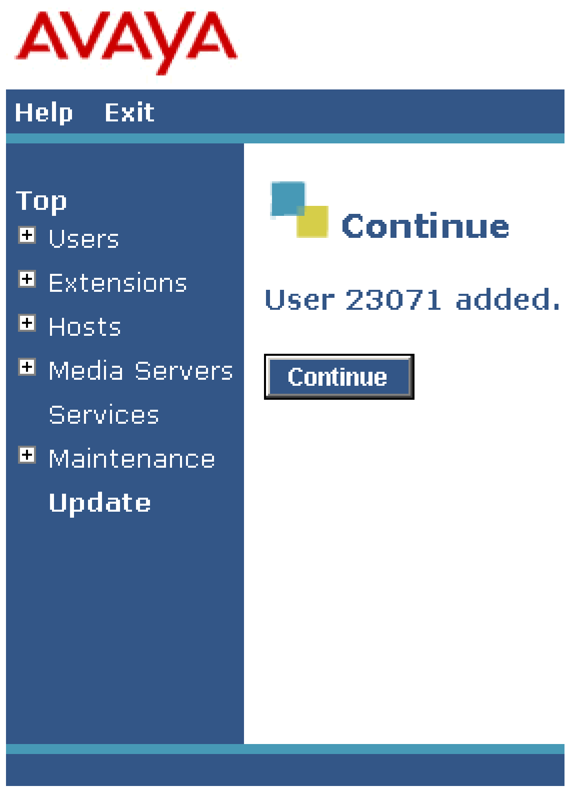
Steps	Description
2.	The Converged Communications Server administration web interface will be displayed. Expand the **Users** link on the left side of the page and click on **Add**. 

<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>3.</td><td>

The *Add User* page will be displayed. Fill in the required fields (indicated by *). In the screen below, the user corresponding to a Cisco 7960 SIP telephone is being added. Enter the extension number in the **Handle** and **User ID** fields. The **Host** field should be set to the name of the Converged Communications Server Home or Home/Edge server to which the user's phone will register. In this configuration, there is only one Home/Edge Converged Communications Server, so the default value is shown ("impress").

Click on **Add**.

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

Steps	Description
4.	The confirmation page will be displayed. Click **Continue**.  Repeat Steps 2-4 for each user to be supported.
5.	To apply the administration in the above steps, click on **Update** on the left side of the page. This link appears on the current page whenever updates are outstanding, and can be used at any time to save the administration performed to that point.

4.2. Configure the Cisco 7940/7960 SIP Telephone

Cisco 7940/7960 SIP telephones can be configured using two methods:

1. Configuration files downloaded from a TFTP server specified via DHCP at boot time. Two such files are installed on the TFTP server: a default configuration file containing parameter settings that apply to all phones (`SIPDefault.cnf`), and a phone-specific configuration file containing settings applicable only to that phone (`SIP`<MAC-address>`.cnf`, where <MAC-address> is the MAC address of the phone).

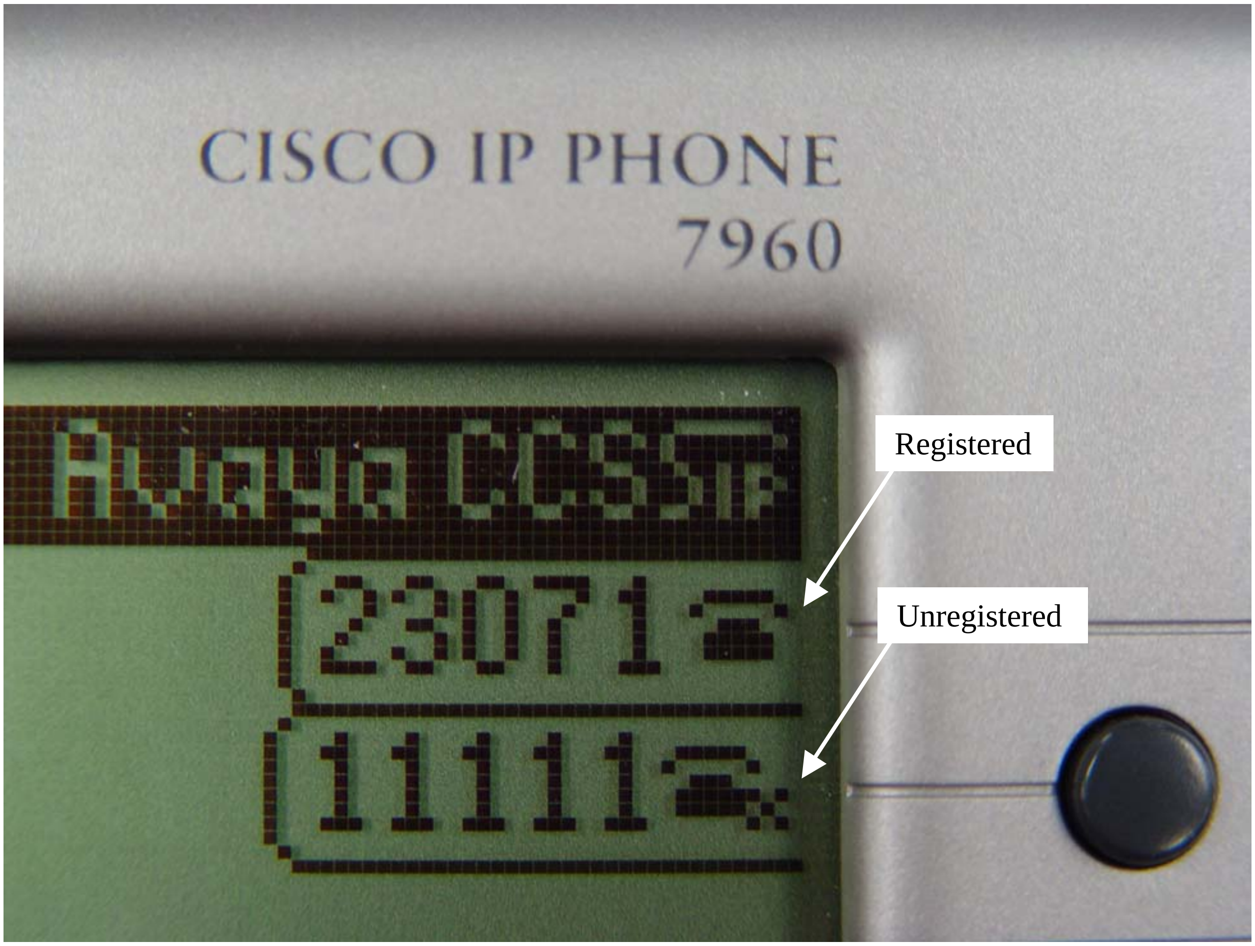
2. Manual configuration of the phone using its screen interface and keypad buttons.

With a few exceptions (one of which will be noted in Section 5.3.2), most parameters can be specified in the configuration file(s), and this is the preferred method for maintaining a large number of phones. Parameters that are manually changed at the phone will revert back to the values in the configuration file(s) when the phone is re-booted, unless the DHCP and TFTP parameters have been manually changed. See Reference [3] for details on installing and maintaining Cisco SIP telephones using configuration files. For the sample configuration, the IP address of the phone and its TFTP server were manually entered at the phone. The remaining configuration was done via the configuration files where possible.

Steps	Description

Steps	Description
1.	Edit the default and phone-specific configuration file(s). The table below shows the relationship between the parameters that must be configured for the phone and those administered in the Converged Communications Server for a telephony user. A sample value is shown for the configuration in **Figure 1**. Parameter names have the form `ObjectNameXParamName`, where "X" refers to the line appearance number to which the parameter applies (1-2 for the 7940 and 1-6 for the 7960 phone). The table shows the parameters for the first line appearance. Although other parameters may be configured, those listed are the minimum required for successful registration of the phone with the Converged Communications Server. Normally, the proxy parameters would reside in the default configuration file, and the name and password would reside in the phone-specific file. In the example, specification of the bitmap to be displayed as a logo on the phone ("*Powered by* Avaya") is in the default configuration file. See **Figures 2-3** for sample files.

Avaya Converged Communications Server	**Cisco 7940/7960**	
	Parameter Name	Example Value
User Administration		
User ID	line1_name	23071
Password	line1_password	hello123
Proxy Administration		
Proxy IP address	proxy1_address	10.1.1.50
Proxy Port	proxy1_port	5060
	Proxy_register	1

Steps	Description
2.	Reboot the phone. If TFTP support has been properly configured, the phone will download the default and specific configuration files, and register with the Avaya Converged Communications Server. Registration can be verified by the absence of an "X" near the phone icon for the line appearance, as shown below. 

For basic feature set operation, the dial plan can be specified using:

1. Address maps in the Avaya Converged Communications Server (see Section 5.2.1, and Reference [1]). They control how Avaya Converged Communications Server routes a call, based on the number dialed.
2. Dial plan file (`dialplan.xml`) downloaded to the phone from the TFTP server (see Reference [3]). The phone uses this file to determine when enough digits have been pressed to complete dialing, so that the user need not press an additional key to "send" the call. The dial plan file 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.

```
# SIP Default Generic Configuration File

# Image Version
image_version: P0S3-06-3-00

# Proxy Server
proxy1_address: "10.1.1.50"          ; Can be dotted IP or FQDN
proxy2_address: ""            ; Can be dotted IP or FQDN
proxy3_address: ""            ; Can be dotted IP or FQDN
proxy4_address: ""            ; Can be dotted IP or FQDN
proxy5_address: ""            ; Can be dotted IP or FQDN
proxy6_address: ""            ; 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

####### New Parameters added in Release 4.0
# URL for branding logo to be used on phone display
logo_url: "http://10.1.1.103/AvayaPhoneLogo.bmp"
```

Figure 2: Sample Default Configuration File for Cisco 7940/7960 SIP Phones (Abbreviated)

```
# SIP Configuration Generic File

# Line 1 appearance
line1_name: 23071

# Line 1 Registration Authentication
line1_authname: "23071"

# Line 1 Registration Password
line1_password: "hello123"

# Line 2 appearance
line2_name: ""

# Line 2 Registration Authentication
line2_authname: ""

# Line 2 Registration Password
line2_password: ""


####### 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 CCS"      ; Has no effect on SIP messaging

# Line 1 Display Name (Display name to use for SIP messaging)
line1_displayname: "Cisco 1"

# Line 2 Display Name (Display name to use for SIP messaging)
line2_displayname: ""


####### 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 classifcation used when Registering [ none(default), phone, ip ]
user_info: none
```

Figure 3: Sample Per-Phone Configuration File for Cisco 7940/7960 SIP Phones

5. Configuring for the Extended Feature Set

In addition to the steps outlined in Section 4, the following additional administration steps are required to support the extended feature set on the Cisco 7940/7960 SIP telephones:

1. On Communication Manager, define stations corresponding to those specified on the Avaya Converged Communications Server, off-PBX station mappings to route call requests involving those stations to the Avaya Converged Communications Server, and off-PBX Feature Name Extensions (FNEs) for invoking the extended features.
2. Define the appropriate address map in the Avaya Converged Communications Server so that FNEs and other Outboard Proxy SIP station extensions dialed at the phone will be routed to Communication Manager.
3. Add parameters to the default phone configuration file to support voice message access and Communication Manager off-PBX station mapping.
4. Administer speed dial buttons on the phone for frequently used FNEs.

The following sections describe the administration details for these steps.

5.1. Configure Avaya Communication Manager

This section highlights the important commands for defining SIP telephones as Outboard Proxy SIP stations on Communication Manager. For complete documentation, see Reference [4]. Use the System Access Terminal (SAT) interface to perform these steps. Log in with the appropriate permissions.

5.1.1. Verify Outboard Proxy SIP Capacity

Use the **display system-parameters customer-options** command to verify that **Maximum Off-PBX Telephones – OPS** has been set to a value that will accommodate the number of phones to be used. Avaya Services has provisioned this during installation according to the system configuration purchased.

```
Display system-parameters customer-options                      Page   1 of  10
                                OPTIONAL FEATURES

     G3 Version: V12                           RFA System ID (SID): 1
       Location: 1                             RFA Module ID (MID): 1
       Platform: 7

                                                              USED
                                         Maximum Ports: 100   50
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 5     0
                    Maximum Off-PBX Telephones - OPS:   20    0
                    Maximum Off-PBX Telephones - SCCAN: 10    0
```

```
        (NOTE: You must logoff & login to effect the permission changes.)
```

5.1.2. Define Outboard Proxy SIP Feature Access Codes (FACs)

In order to define the FNEs for the Outboard Proxy SIP features listed in **Table 3**, a FAC must also be specified for each feature. Use the **change dialplan analysis** command to specify the format of the FAC, and then the **change feature-access-codes** command to define the codes themselves.

```
change dialplan analysis                                          Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                                              Percent Full:    3

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

```
change feature-access-codes                                       Page   1 of   6
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code: 612
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: 600    Deactivation: 601
Call Forwarding Activation Busy/DA: 602    All: 603    Deactivation: 604
                         Call Park Access Code: 605
                       Call Pickup Access Code: 606
CAS Remote Hold/Answer Hold-Unhold Access Code: 630
                  CDR Account Code Access Code:
                       Change COR Access Code:
                  Change Coverage Access Code:
                  Contact Closure   Open Code:          Close Code:
                  Contact Closure  Pulse Code:
```

```
change feature-access-codes                                    Page   2 of   6
                              FEATURE ACCESS CODE (FAC)
                  Data Origination Access Code:
                      Data Privacy Access Code:
              Directed Call Pickup Access Code: 607
     Emergency Access to Attendant Access Code:
                     Enhanced EC500 Activation:        Deactivation:
  Extended Call Fwd Activate Busy D/A      All:        Deactivation:
        Extended Group Call Pickup Access Code:
               Facility Test Calls Access Code:
                             Flash Access Code:
             Group Control Restrict Activation:        Deactivation:
                    Hunt Group Busy Activation:        Deactivation:
                              ISDN Access Code:
                Last Number Dialed Access Code: 608
     Leave Word Calling Message Retrieval Lock:
   Leave Word Calling Message Retrieval Unlock:
             Leave Word Calling Send A Message:
           Leave Word Calling Cancel A Message:
               Malicious Call Trace Activation: 620    Deactivation: 621
         Meet-me Conference Access Code Change:
```

```
change feature-access-codes                                    Page   3 of   6
                              FEATURE ACCESS CODE (FAC)

 PASTE (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:        Dissociate Code:
        Per Call CPN Blocking Code Access Code: 613
      Per Call CPN Unblocking Code Access Code: 614

                  Priority Calling Access Code: 609
                           Program Access Code:
       Refresh Terminal Parameters Access Code:
              Remote Send All Calls Activation:        Deactivation:
               Self Station Display Activation:
                     Send All Calls Activation: 610    Deactivation: 611
         Station Firmware Download Access Code:
                       Station Lock Activation:        Deactivation:
      Station Security Code Change Access Code:
              Station User Admin of FBI Assign:                   Remove:
  Station User Button Ring Control Access Code:
             Terminal Dial-Up Test Access Code:
```

```
change feature-access-codes                                    Page   4 of   6
                              FEATURE ACCESS CODE (FAC)
Terminal Translation Initialization Merge Code:          Separation Code:
            Transfer to Voice Mail Access Code: 623
          Trunk Answer Any Station Access Code:
              User Control Restrict Activation:        Deactivation:
  Voice Coverage Message Retrieval Access Code:
 Voice Principal Message Retrieval Access Code:
           Whisper Page Activation Access Code:
```

5.1.3. Define Feature Name Extensions (FNEs)

Now the FNEs can be defined using the **change off-pbx-telephone feature-name-extensions** command. It is recommended that a uniform format for these extensions be used as shown (e.g., all beginning with 70xxx), so that the definition of the address map in the Avaya Converged Communications Server later (Section 5.2) will be as simple as possible.

```
change off-pbx-telephone feature-name-extensions                Page   1 of   1
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

     Active Appearance Select: 70003          Idle Appearance Select: 70018
   Automatic Call-Back Cancel: 70004              Last Number Dialed: 70019
              Call Forward All: 70005           Malicious Call Trace: 70020
  Call Forward Busy/No Answer: 70006     Malicious Call Trace Cancel: 70021
           Call Forward Cancel: 70007                  Priority Call: 70000
                     Call Park: 70008                 Send All Calls: 70001
        Call Park Answer Back: 70009           Send All Calls Cancel: 70002
                 Call Pick-Up: 70010             Transfer On Hang-Up: 70022
         Conference on Answer: 70011          Transfer to Voice Mail: 70023
         Calling Number Block: 70012
       Calling Number Unblock: 70013
        Directed Call Pick-Up: 70014
        Drop Last Added Party: 70015
    Exclusion (Toggle On/Off): 70016
      Held Appearance Select: 70017
```

5.1.4. Specify Class of Service (COS)

Use the **change class-of-service** command to set the appropriate service permissions to support the off-PBX features (shown in bold). For the example, COS 1 was used.

```
change cos                                                      Page   1 of   1
                                CLASS OF SERVICE

                                0  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15
 Auto Callback                  n  n  y  n  y  n  y  n  y  n  y  n  y  n  y  n
 Call Fwd-All Calls             n  y  n  y  y  n  n  y  y  n  n  y  y  n  n  y
 Data Privacy                   n  y  n  n  n  y  y  y  y  n  n  n  n  y  y  y
 Priority Calling               n  y  n  n  n  n  n  n  n  y  y  y  y  y  y  y
 Console Permissions            n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Off-hook Alert                 n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Client Room                    n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Restrict Call Fwd-Off Net      y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
 Call Forwarding Busy/DA        n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Personal Station Access (PSA)  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding All        n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding B/DA       n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Trk-to-Trk Transfer Override   n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 QSIG Call Offer Originations   n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Contact Closure Activation     n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
```

5.1.5. Add stations

Use the **add-station** command to add a station for each SIP phone to be supported. Use *4620* for the **Station Type** and be sure to include the **Coverage Path** for voice messaging if it is available. Use the **COS** value specified in the previous section. The **Name** field is optional and is shown on the display of the destination phone when making calls. Use default values for the other fields on Page 1.

Note: For Outboard Proxy SIP configuration, the station extension must be different than that administered for the Avaya Converged Communications Server. The Communication Manager extension will be the "public" extension that users associate with their phones. The Avaya Converged Communications Server extension will be a "private" extension that is used in the system configuration, but is not dialed by users. To make it easy to relate the two, make the extensions the same except for one high order digit. For example, in the following Outboard Proxy SIP configuration, the public Communication Manager extension is 24071 and the private Converged Communications Server extension is 23071.

```
add station 24071                                               Page   1 of   4
                                     STATION

Extension: 24071                          Lock Messages? n          BCC: 0
     Type: 4620                           Security Code:             TN: 1
     Port: S00024                       Coverage Path 1: 1          COR: 1
     Name: Cisco 1                      Coverage Path 2:            COS: 1
                                        Hunt-to Station:

STATION OPTIONS
              Loss Group: 19            Personalized Ringing Pattern: 1
                                                    Message Lamp Ext: 24071
            Speakerphone: 2-way                  Mute Button Enabled? y
        Display Language: english                    Expansion Module? n

 Survivable GK Node Name:                           Media Complex Ext:
                                                         IP SoftPhone? n
                                                    IP Audio Hairpinning? y
                                             Interworking Message: PROGress
```

For most applications, the default values can be used for Page 2.

```
add station 24071                                               Page   2 of   4
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe             Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
             CDR Privacy? n                              Data Restriction? n
   Redirect Notification? y                  Idle Appearance Preference? n
 Per Button Ring Control? n
   Bridged Call Alerting? n                    Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n         Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
    MWI Served User Type:                    Display Client Redirection? n
              AUDIX Name:                   Select Last Used Appearance? n
                                              Coverage After Forwarding? s
                                                Multimedia Early Answer? n
                                            Direct IP-IP Audio Connections? y
 Emergency Location Ext: 24071                         IP Audio Hairpinning? y
```

On Page 3, fill in function button names, if required, for off-PBX FNEs that will be used at the phone. **Table 5** correlates the FNE to required function button names. For the example, the Idle Appearance Select requires *call-appr*. The maximum number of *call-appr* buttons should match that of the Cisco phone type (two for 7940, six for 7960).

```
add station 24071                                              Page   3 of   4
                                     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: call-appr                          7:
 4:                                    8:
```

Off-PBX Station Feature Name Extension (FNE)	Station Button Required
Active Appearance Select	call-appr, brdg-appr, or abrdg-appr
Automatic Call-Back Cancel	(not yet supported)
Call Forward All	(none)
Call Forward Busy/No Answer	(none)
Call Forward Cancel	(none)
Call Park	(none)
Call Park Answer Back	(none)
Call Pick-Up	(none)
Conference on Answer	no-hold-conf
Calling Number Block	(none)
Calling Number Unblock	(none)
Directed Call Pick-Up	(none)
Drop Last Added Party	drop
Exclusion (Toggle On/Off)	exclusion
Held Appearance Select	call-appr, brdg-appr, or abrdg-appr
Idle Appearance Select	(none)
Last Number Dialed	(none)
Malicious Call Trace	(none)
Malicious Call Trace Cancel	(none)
Priority Call	(none)
Send All Calls	(none)
Send All Calls Cancel	(none)
Transfer On Hang-Up	transfer
Transfer to Voice Mail	(none)

Table 5: Station Buttons Required for off-PBX Features

Use the **change off-pbx-telephone station-mapping** command to map the Communication Manager extension (24071) to the Converged Communications Server extension (23071). Enter the field values shown. For the sample configuration, the **Trunk Selection** value indicates that Automatic Alternate Routing (AAR) will be used, and that the routing pattern to reach this Converged Communications Server extension will point to the SIP trunk to the Converged Communications Server. The **Configuration Set** value can reference a set that has the default settings in Communication Manager.

```
change off-pbx-telephone station-mapping 24071                  Page   1 of   2
                    STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station      Application    Dial   Phone Number       Trunk        Configuration
 Extension                   Prefix                    Selection    Set
 24071          OPS              - 23071                aar         1
                                 -
```

5.2. Configure the Avaya Converged Communications Server

5.2.1. Adding an Address Map

Address maps are used in the Converged Communications Server to specify how incoming SIP calls are to be routed, based on the dialed number. They are grouped by the SIP contact to which they will be routed. For *host* maps, specified in the **Hosts** link, the SIP contact must be specified completely. To configure support for the Outboard Proxy SIP features supported by Communication Manager on an Avaya Media Server, *Media Server* maps are defined using the **Media Servers** link. In this case, the contact information is automatically generated and dialed numbers matching the map specification cause the call to be routed to the associated Media Server. In this configuration, any number the user may dial, including the FNEs defined in Section 5.1, must be covered by the map specification, so that these calls (i.e. SIP INVITE messages) will be routed to the S8300/G700, where Communication Manager will route the call based on its dial plan. This means that the map specification(s) must agree with the dial plan administered in Communication Manager.

Note: To route toll calls back through the Converged Communications Server and on to a SIP service provider, configure Communication Manager such that the number sent to Converged Communications Server on the SIP trunk is **not** identical to that dialed by the user. Otherwise the Converged Communications Server will route the call back to Communication Manager and a loop will result. For example, to dial outbound numbers, the user may have to dial "9" followed by the area code and number. A media server map in the Converged Communications Server would match on the initial 9 and route the call to Communication Manager. Communication Manager would in turn route the call back to the Converged Communications Server using a modified form of the number that would match a Converged Communications Server address map for outbound routing to the service provider.

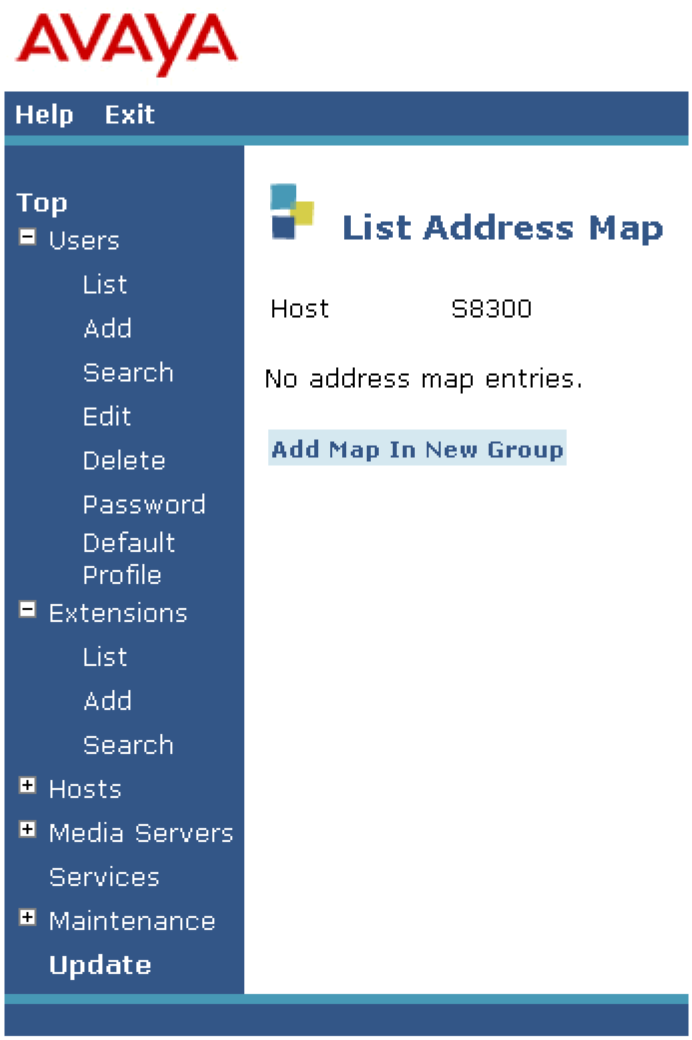
The following steps describe how to administer the Media Server Map. See Reference [1] for more information on the syntax used to specify address maps.

<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>1.</td><td>

Click on the **Media Servers** link on the left side of the main Converged Communications Server web page. The *List Media Servers* page is displayed.

Click on **Map**.

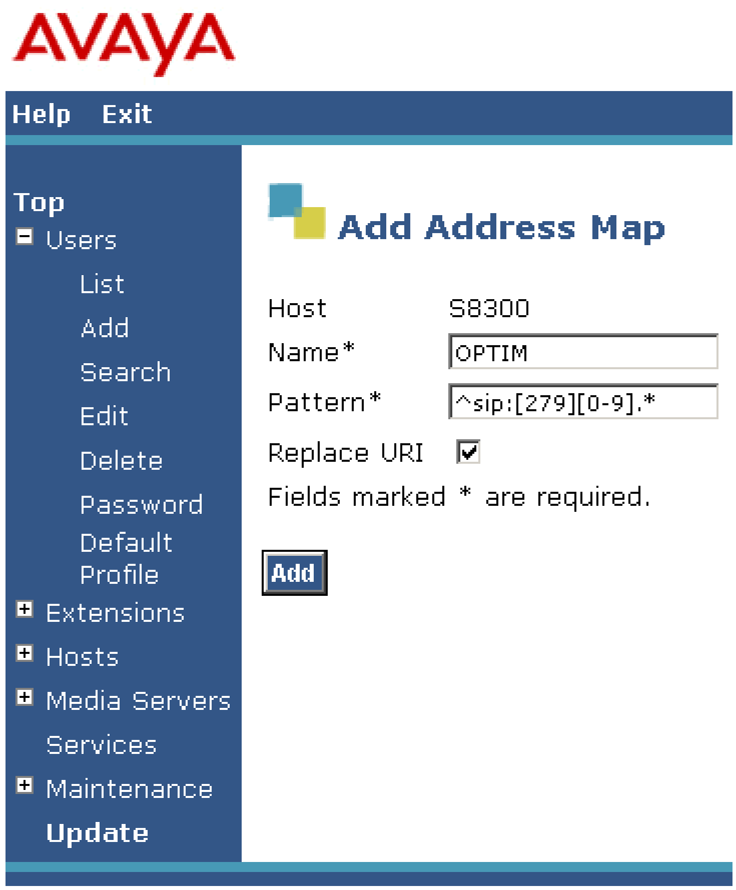
</td></tr>
</table>

Steps	Description
2.	The *List Address Map* page is displayed.  Select **Add Map in New Group**.

<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>3.</td><td>

The *Add Address Map* page will be displayed. Specify a **Name** for the address map, and the **Pattern** match specification. In this example, all FNEs begin with 70. The pattern match specification is applied to the Uniform Resource Identifier (URI) field of incoming INVITE messages. The URI usually takes the form sip:user@domain, where *domain* can be a domain name or an IP address. For example, the URI for the *Call Pickup* FNE could be sip:70010@avaya.com or sip:70010@10.1.1.50.

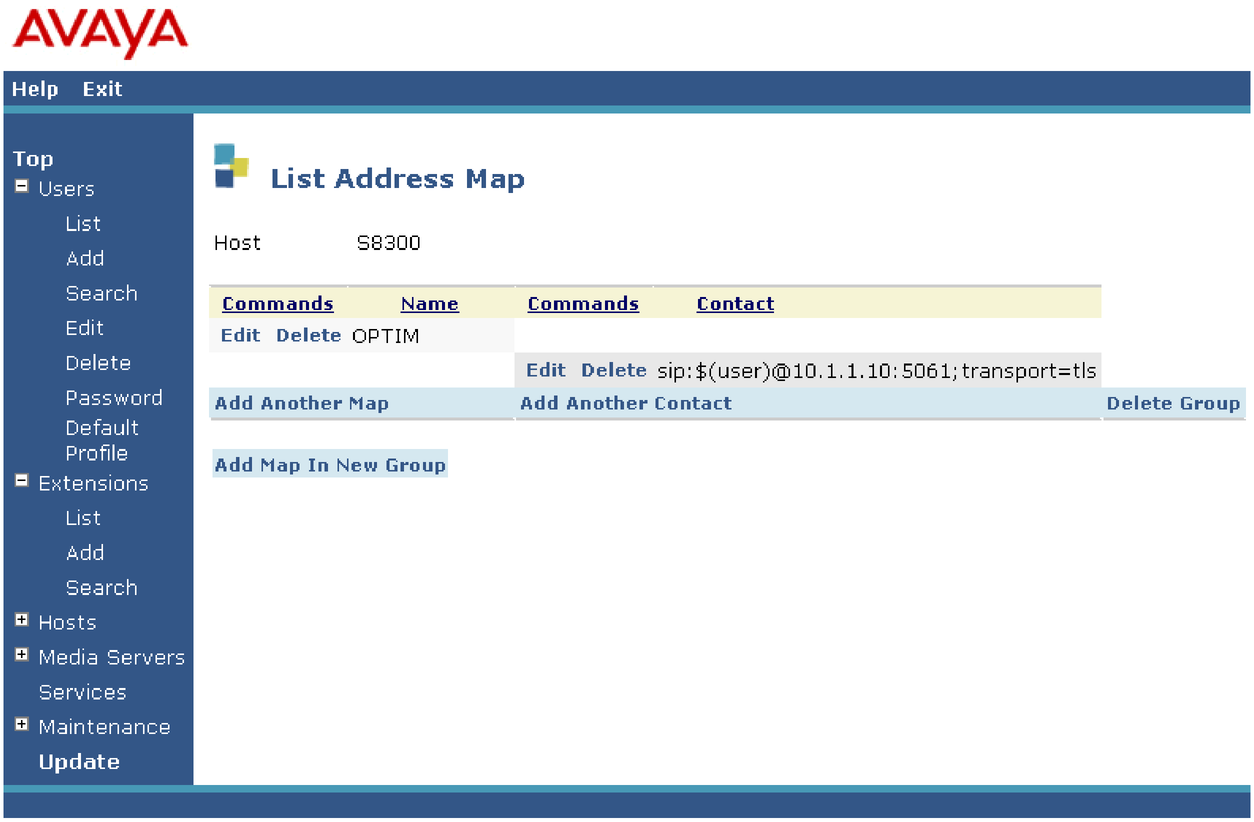
The specification means that if the URI begins with "sip:" followed by either a 2, 7, or 9, route the call to the Media Server. In the sample configuration, this would cover extension, Outboard Proxy SIP FNE, and outbound dialing, respectively. Check **Replace URI**. When routing the incoming INVITE from the phone, the Converged Communications Server will replace the request URI with the URI specified in the contact (see Step 4).



Click on **Add**; then click on **Continue** on the confirmation page.

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

<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>4.</td><td>The List Address Map page will be displayed again, this time with the updated map information. Note that the contact for the S8300/G700 has been automatically generated.



</td></tr>
<tr><td>5.</td><td>To apply the administration in the above steps, click on Update on the left side of the page. This link appears on the current page whenever updates are outstanding, and can be used at any time to save the administration performed to that point.</td></tr>
</table>

5.3. Configure the Cisco 7940/7960 SIP Telephone

Now that Communication Manager and Converged Communications Server have been administered for Outboard Proxy SIP support of the telephones, the extended feature set is available to 7940/7960 users. To dial a number, lift the receiver (or press **Speaker**) and dial any number using the dial plan centrally administered in Communication Manager on the Avaya Media Server. To access any of the Outboard Proxy SIP features, dial the corresponding FNE. For example, if the telephone has been defined in Communication Manager as part of a pickup group, then dial the Call Pickup FNE (in this case 70010) to answer a call to any member of that group. Outboard Proxy SIP features that involve an existing call (e.g., conference on answer) will require putting that call on hold, and placing a new call using the appropriate FNE.

5.3.1. Add Parameters to Configuration Files

The configuration files should be modified to support two functions relative to Communication Manager /Outboard Proxy SIP: Public extension display (see Section 5.1.5) and voice message access.

The Cisco 7940/7960 phone will normally display the "private" extension (User ID) used to register to the Converged Communications Server. The desired display is the "public" Outboard Proxy SIP extension. To do this, the **shortname** parameter must be added to the phone-specific configuration file for each phone and set to the public extension for that line appearance. In the sample configuration, the parameter for line appearance one would be:

```
line1_shortname: "24071"
```

This will cause 24071 to be displayed on the phone even though 23071 is used to register with Converged Communications Server.

Even though MWI is not supported, the **messages** button can be programmed to call the extension corresponding to Avaya voice messaging. This can be accomplished by setting a parameter in the default configuration file, and will be apply to all phones in the system.
The parameter to be added is:

```
messages_uri: 25000
```

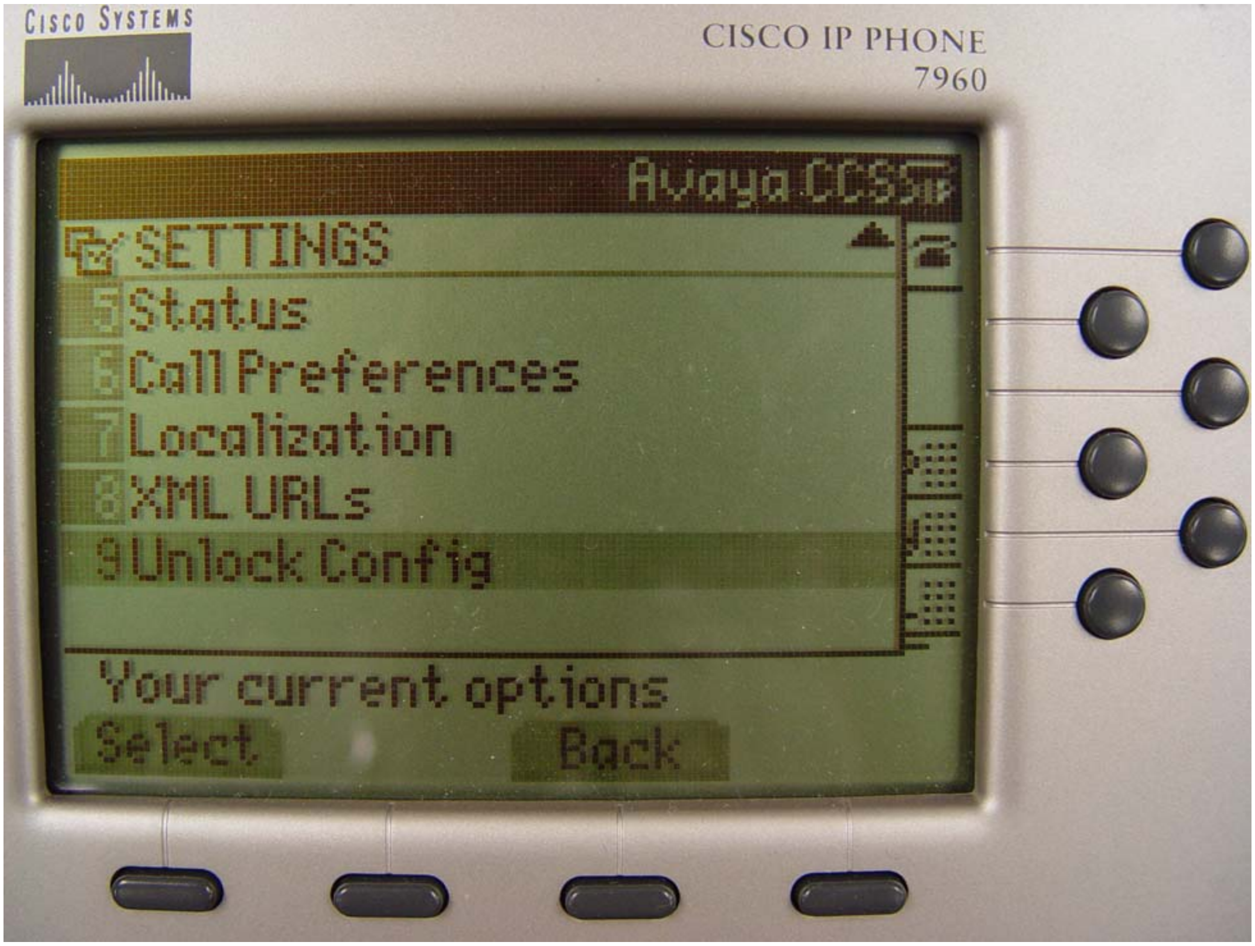
In the sample configuration, the AUDIX hunt group on the S8300/G700 has extension 25000.

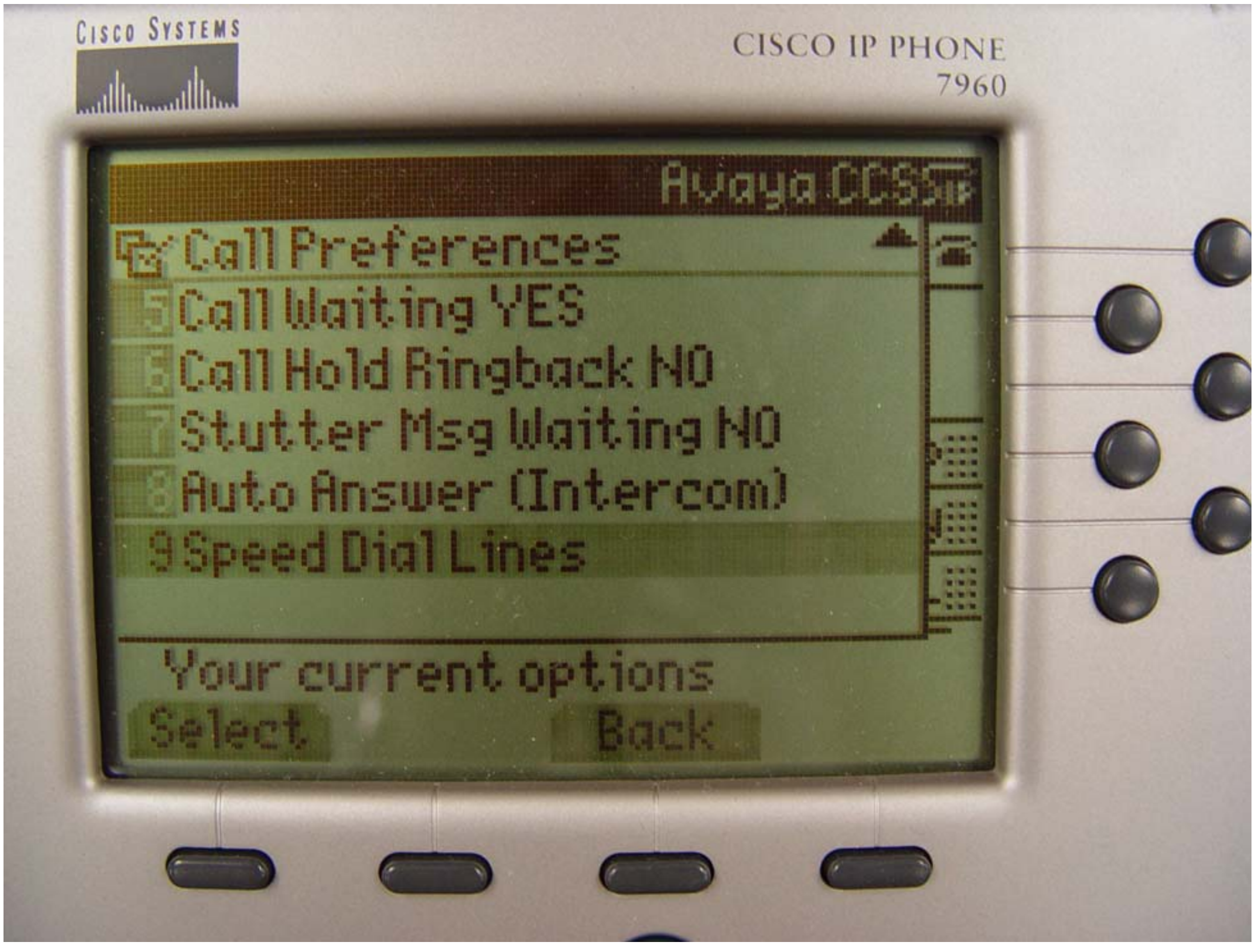
5.3.2. Define Outboard Proxy SIP Speed Dial Buttons

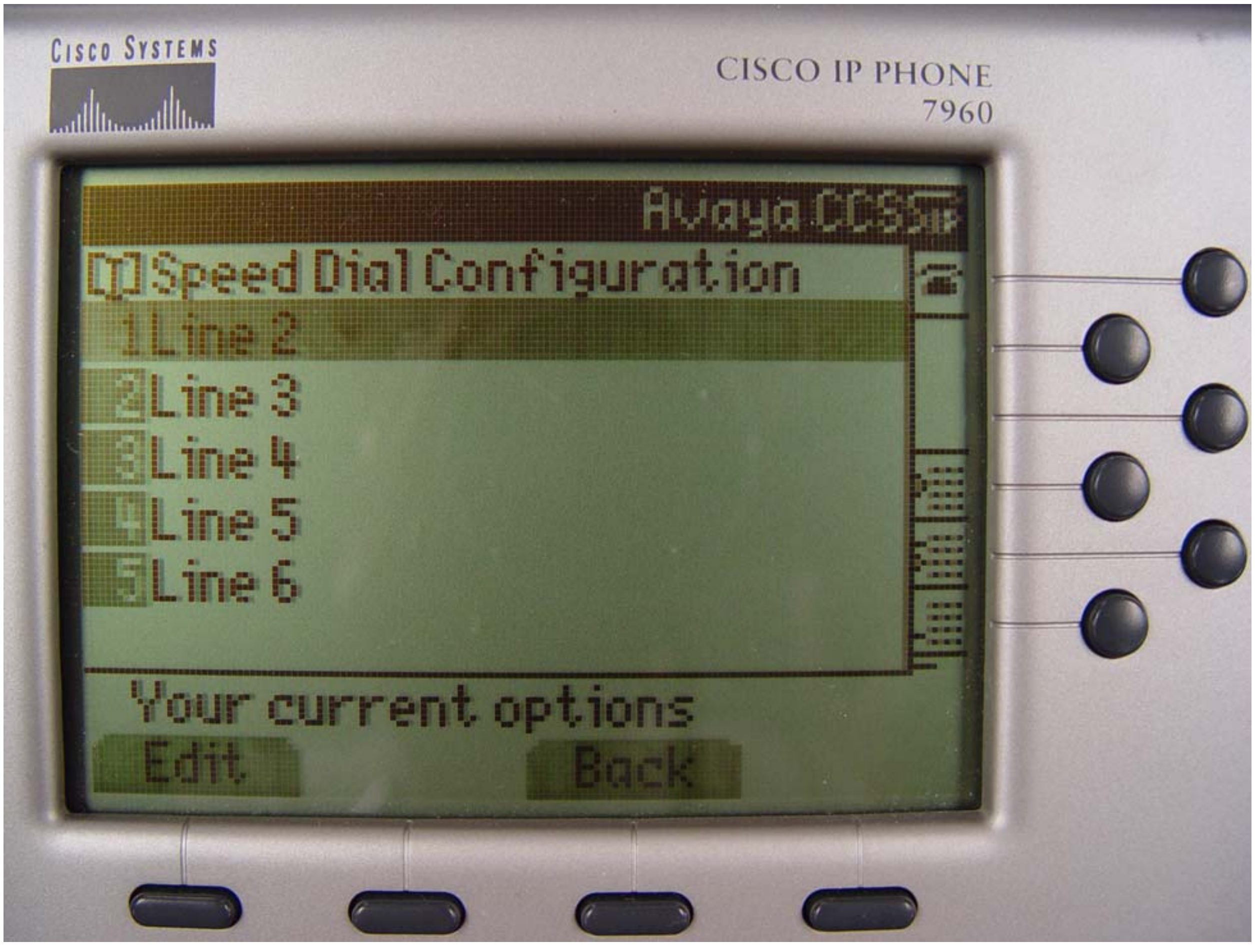
Access to Outboard Proxy SIP features can be streamlined by using free line appearance buttons on the telephone for speed dialing. Commonly used FNEs can be defined on these buttons, in many cases facilitating one-button dialing. Since the Cisco 7960 SIP telephone has six line appearances, it is preferred over the 7940, which has two. The following steps describe how to configure the 7960 set with Outboard Proxy SIP speed dial buttons. They also apply to the 7940, although only one speed dial button is available in this case. General configuration information can be found in Reference [3].

Note: The following configuration must be done at the telephone, and cannot be centrally defined in configuration files.

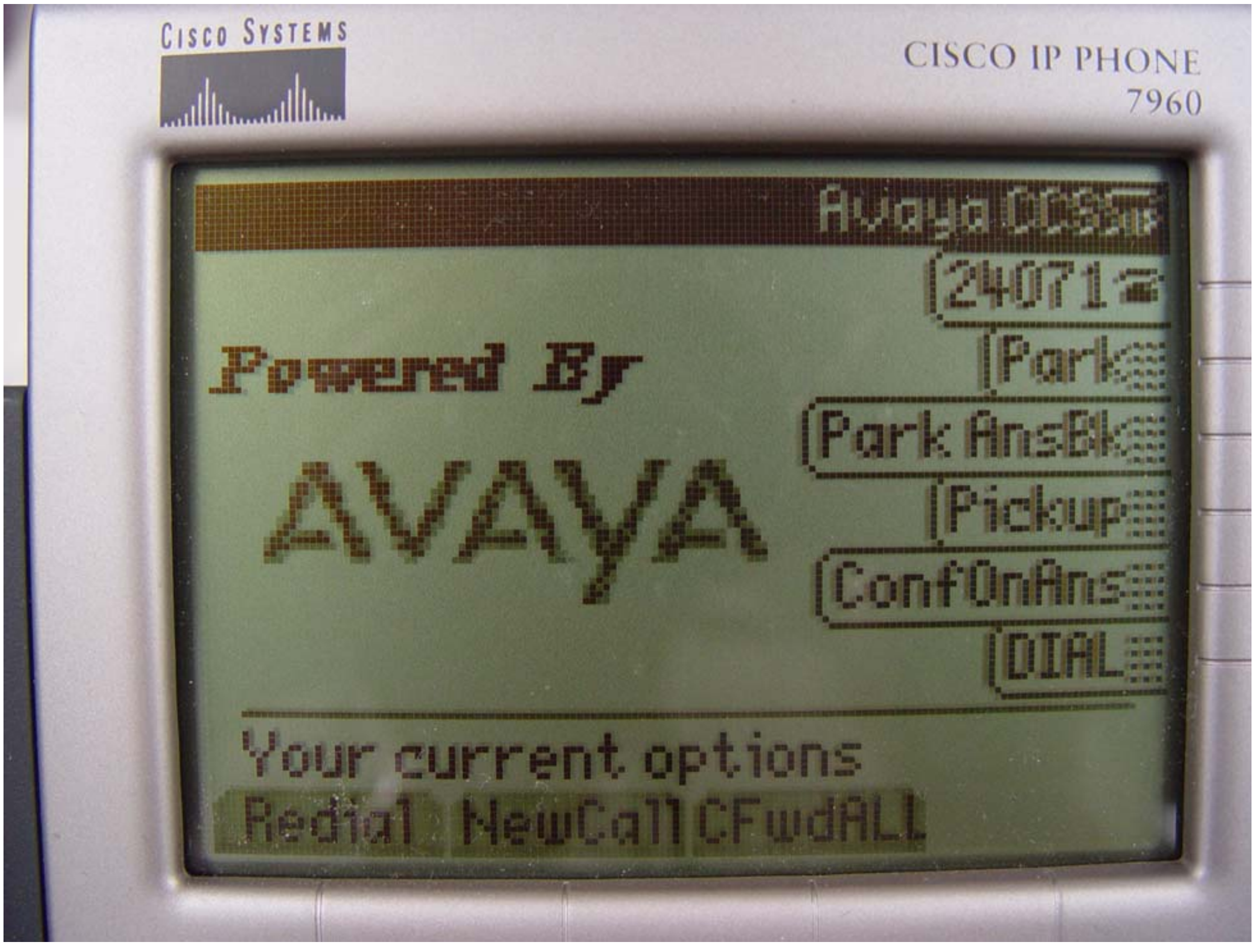
Steps	Description
1.	At the telephone, press the **settings** button to bring up the main configuration menu. 

Steps	Description
2.	Using the **up/down** button, move the highlighted selection to **Unlock Configuration**, and press the **Select** function button. Enter the appropriate password using the keypad.  The main configuration menu will be displayed again. Move the highlighted selection to **Call Preferences**, and press the **Select** function button.

Steps	Description
3.	The Call Preferences menu will be displayed. Move the highlighted selection to **Speed Dial Lines**, and press the **Select** function button. 

Steps	Description
4.	The Speed Dial Configuration menu will be displayed. Move the highlighted selection to the desired speed dial button and press the **Edit** function button. 

Steps	Description
5.	Enter the **Name** and **Number** for the speed dial button. For Outboard Proxy SIP features, this should be one of the FNEs.  Press the **Accept** function button when finished.

Steps	Description
6.	Repeat steps 4-5 for each button desired, selecting most frequently used Outboard Proxy SIP features for assignment to the buttons. A representative button arrangement is shown below. When finished, press the **settings** button to exit from phone configuration mode. 

6. Verification Steps

The following steps can be used to verify and/or troubleshoot installations in the field.

1. After rebooting the 7940/7960 telephone, use the **settings** button at the phone to verify that the parameters set in the default (proxy server address and port number, register with proxy, etc.) and phone-specific (User ID, Password, etc.) configuration files have been loaded. Verify that the phone icon by each defined line appearance does *not* have an "X" next to it, indicating that registration has not occurred. If the "X" appears, check that the proxy server IP address and port number are correct and that the Proxy Register parameter is set to *Yes*. Verify that the line appearance shows the Communication Manager extension for that phone.

2. Verify basic feature set administration by lifting the handset (or pressing the **speaker** button), and making calls to other phones. Test those features for which *Yes* appears in the second column of **Table 2**.
3. Verify that speed dial buttons defined locally at the phone are displayed on the right hand side. If any are missing or are inoperative, use the local phone menus to re-check their configuration.
4. Verify extended Outboard Proxy SIP features by pressing the speed dial button for the feature, or lifting the handset and dialing the FNE. If busy or intercept tone is heard, check Communication Manager administration for the correct FNE, proper permissions under COS, and the proper station button assignment to support the feature.
5. Press the **messages** button and verify that the voice messaging system is called.

7. Conclusion

These Application Notes have described the administration steps required to use Cisco 7940 and 7960 SIP telephones with the Avaya Converged Communications Server and Communication Manager. Both basic and extended feature sets were covered. The extended set includes features not yet available to SIP telephones via the current IETF standards. IETF standards-based Message Waiting Indicator (MWI) support is not yet available on Cisco SIP Telephones, although one-touch access to Avaya voice messaging is available.

8. Additional References

[1] *Converged Communications Server Installation and Administration*, Doc # 555-245-705, February, 2004.

[2] *Session Initiation Protocol Service Examples - draft-ietf-sipping-service-examples-06*, SIPPING Working Group, Internet-Draft, 2/15/2004.

[3] *Cisco SIP IP Phone Administrator Guide*, *Release 6.0, 6.1, 7.0, 7.1*, May 2004, Cisco Systems, Inc.

[4] *Avaya Extension to Cellular and Off-PBX Station Installation and Administration Guide*, Doc. # 210-100-500, Issue 6, November, 2003.

[5] *Converting a Cisco 7940/7960 CallManager Phone to a SIP Phone and the Reverse Process*, Cisco Systems, Inc.

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