Avaya Solution & Interoperability Test Lab

Configuring SIP IP Telephony Using Avaya SIP Enablement Services, Avaya Communication Manager, and Cisco 7940/7960 SIP Telephones - Issue 1.1

Abstract

These Application Notes describe the configuration steps required to connect Cisco 7940/7960 SIP telephones to a SIP infrastructure consisting of an Avaya SIP Enablement Services (SES) server and an Avaya S8300 Media Server with G700 Media Gateway running Avaya Communication Manager. Also described is how Avaya Outboard Proxy SIP (OPS) station features can be made available 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.

This issue of these Application Notes adds Message Waiting Indicator (MWI) support to the features documented in Issue 1.0.

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 the existing telephony infrastructure with Avaya servers and re-use existing telephones.

Avaya SIP Enablement Services R3.1 (SES) adds new feature and scalability enhancements to the SIP functionality previously introduced as Avaya Converged Communications Server Release 2.1. SIP Enablement Services combines the standard functions of a SIP proxy and registrar server with SIP trunking support and duplicated server features to create a highly scalable, highly reliable SIP communications network supporting telephony, instant messaging, conferencing, and collaboration solutions.

In addition, Avaya Communication Manager running on Avaya Media Servers and Gateways extends advanced telephony features to SIP telephones via Outboard Proxy SIP (OPS) support. This feature set can be offered on non-Avaya SIP telephones, providing enhanced calling features in advance of SIP protocol definitions and telephone implementations. See Section 3.1. In SIP terminology, Avaya Communication Manager can be viewed as a feature server.

These Application Notes describe the configuration steps for using Cisco 7940/7960 SIP telephones with Avaya SIP Enablement Services, 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 10. The configuration described should be applicable to other Linux-based Avaya Media Servers and Media Gateways running Avaya Communication Manager. This issue of these Application Notes adds Message Waiting Indicator (MWI) support to the features documented in Issue 1.0.

1.2. Configuration

The configuration used as an example in these Application Notes is shown in **Figure 1**. The diagram indicates logical signaling connections. With the exception of the Avaya 6408D Digital Telephone, all components are physically connected to a single Avaya C363T-PWR Converged Stackable Switch, and are administered as a single subnet. Each Cisco 7940/7960 SIP telephone is configured to register to one of two SIP Enablement Services home servers and is administered as an OPS station on an Avaya S8300 Media Server with G700 Media Gateway.[1] The Avaya IA770 INTUITY™ AUDIX® Messaging Application resides on the S8300 Media Server and is used to support voice messaging. An audio source is connected to an analog port

[1] The sample configuration uses multiple SIP Enablement Services servers for illustrative purposes. For installations under 6000 users, a single server configured as an Edge/Home combination would suffice.

on the G700 Media Gateway for Music On Hold (MOH). PCs support DNS, HTTP, and TFTP servers as well as a web browser for administration of the Avaya servers.

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.

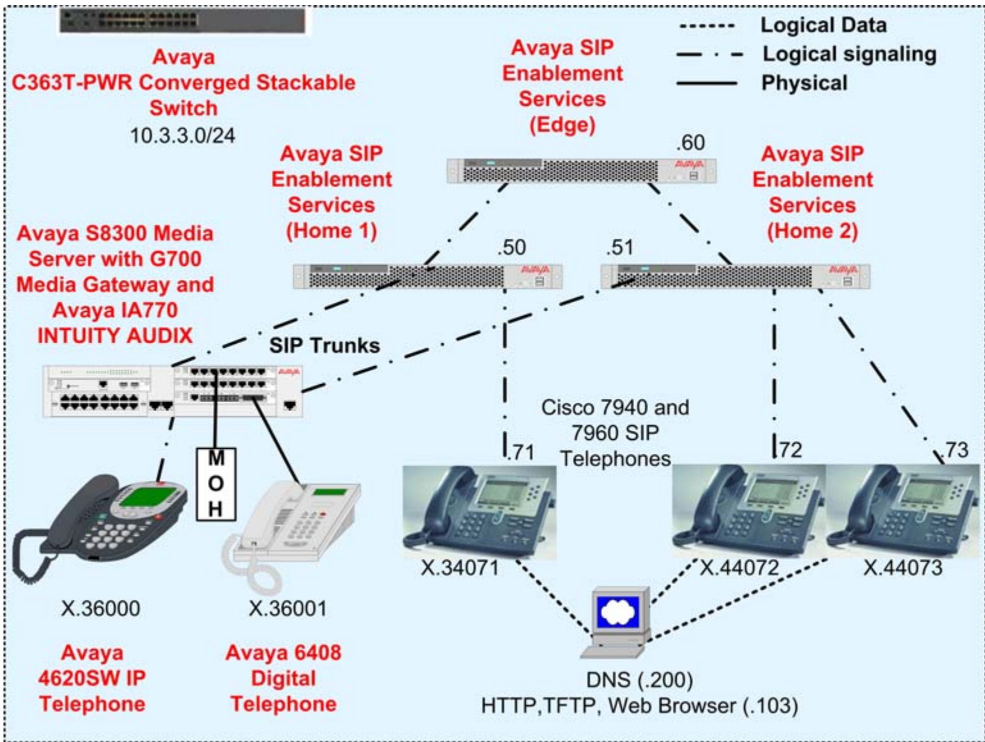


Figure 1: Avaya SIP Test Configuration with Cisco 7940/7960 SIP Phones

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 Cisco 7940/7960 SIP Telephones

2. Equipment and Software Validated

The following equipment and software were used in the configuration.

Equipment	Software
Avaya SIP Enablement Services (SES) Server	3.1 (Load 316)
Avaya C363T-PWR Converged Stackable Switch	4.3.10
Avaya S8300 Media Server with G700 Media Gateway	Avaya Communication Manager 3.1 (Load 627)
Avaya IA770 INTUITY®AUDIX™	N3.1-22.0
Cisco 7940/7960 SIP Telephones	POS3-07-4
PC (HTTP, TFTP servers)	Microsoft Windows 2000 Professional Workstation, 5.00.2195, SP 4

Table 2: Equipment and Software Versions Used

3. Supported Features

3.1. Overview

Table 3 gives a summary of the features available on Cisco 7940/7960 SIP telephones. Notes on specific feature operations are included in Section 3.2. Some features are supported locally at the telephone, while others are only available with Avaya SIP Enablement Services and Avaya Communication Manager with OPS. 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 features supported by various equipment vendors. OPS can support many of these features even though the telephone may not locally support them. Additional features beyond the SIPPING-19 can be extended to the telephone using OPS.

Some OPS features shown in **Table 3** can be invoked at the phone by dialing a Feature Name Extension (FNE) or pressing a speed dial button programmed to that extension. Avaya Communication Manager automatically handles many other standard features via OPS, such as call coverage, trunk selection using Automatic Alternate Routing (AAR) and Automatic Route Selection (ARS), Class Of Service/Class Of Restriction (COS/COR), and voice messaging. Details on operation and administration can be found in Reference [4]. The Avaya SIP solution requires all SIP telephones to be configured in Avaya Communication Manager as OPS.

FEATURE	Supported		COMMENTS
	Locally at the Phone	With Avaya SIP Offer	
Basic Calling features			
Extension to extension call	**YES**	**YES**	See Section 3.2.1
Basic call to legacy phones	**NO**	**YES**	See Section 3.2.1
Intercept tones/displays	**YES**	**YES**	Reorder for all (announcements available with OPS)
Call Waiting	**YES**	**YES**	
Do Not Disturb	**YES**	**YES**	
Speed Dial buttons	**YES**	**YES**	
Message Waiting Support	**YES**	**YES**	See Section 3.2.2
SIPPING-19 Features			
Call Hold	**YES**	**YES**	
Consultation Hold	**YES**	**YES**	
Music on Hold	**NO**	**YES**	
Unattended Transfer	**YES**	**YES**	
Attended Transfer	**YES**	**YES**	
Transfer - Instant Messaging	**NO**	**NO**	
Call Forward Unconditional	**YES**	**YES**	Local soft key or OPS FNE (See Section 3.2.3)
Call Forward Busy	**NO**	**YES**	Changeable via OPS FNE
Call Forward No Answer	**NO**	**YES**	Changeable via OPS FNE
3-way conference - 3rd party added	**YES**	**YES**	
3-way conference - 3rd party joins	**NO**	**NO**	
Single Line Extension	**NO**	**NO**	
Find-Me	**NO**	**YES**	Via bridged appearances
Incoming Call Screening	**NO**	**YES**	Via OPS Class Of Restriction (COR)
Outgoing Call Screening	**NO**	**YES**	Via OPS Class Of Restriction (COR)
Call Park/Unpark	**NO**	**YES**	Via OPS FNE
Call Pickup	**NO**	**YES**	Via OPS FNE
Automatic Redial	**NO**	**YES**	Via OPS FNE
Click to Dial	**NO**	**NO**	
OPS - Selected Additional Station-Side Features			
Conference on Answer	**NO**	**YES**	Via OPS FNE
Extended Group Call Pickup	**NO**	**YES**	Via OPS FNE
Directed Call Pick-Up	**NO**	**YES**	Via OPS FNE
Drop Last Added Party	**NO**	**YES**	Via OPS FNE
Last Number Dialed	**YES**	**YES**	Local soft key or OPS FNE
Malicious Call Trace	**NO**	**YES**	Via OPS FNE
Malicious Call Trace Cancel	**NO**	**YES**	Via OPS FNE
Priority Call	**NO**	**YES**	Via OPS FNE
Send All Calls	**NO**	**YES**	Local DND soft key or OPS FNE
Send All Calls Cancel	**NO**	**YES**	Local DND soft key or OPS FNE
Transfer to Voice Mail	**NO**	**YES**	Via OPS FNE
Whisper Page	**NO**	**YES**	Via OPS FNE

Table 3: SIP Telephony Feature Support

3.2. Operational Notes

The following sections correlate to references in **Table 3**, elaborating on the operational behavior of the feature.

3.2.1. Station to Station Calling

When called, the Cisco telephone will ring with the distinctive ring cadence. The selected ring type plays twice with a short pause in between. In call-waiting mode, two short tones are generated instead of one long tone.

3.2.2. Message Waiting Indicator (MWI)

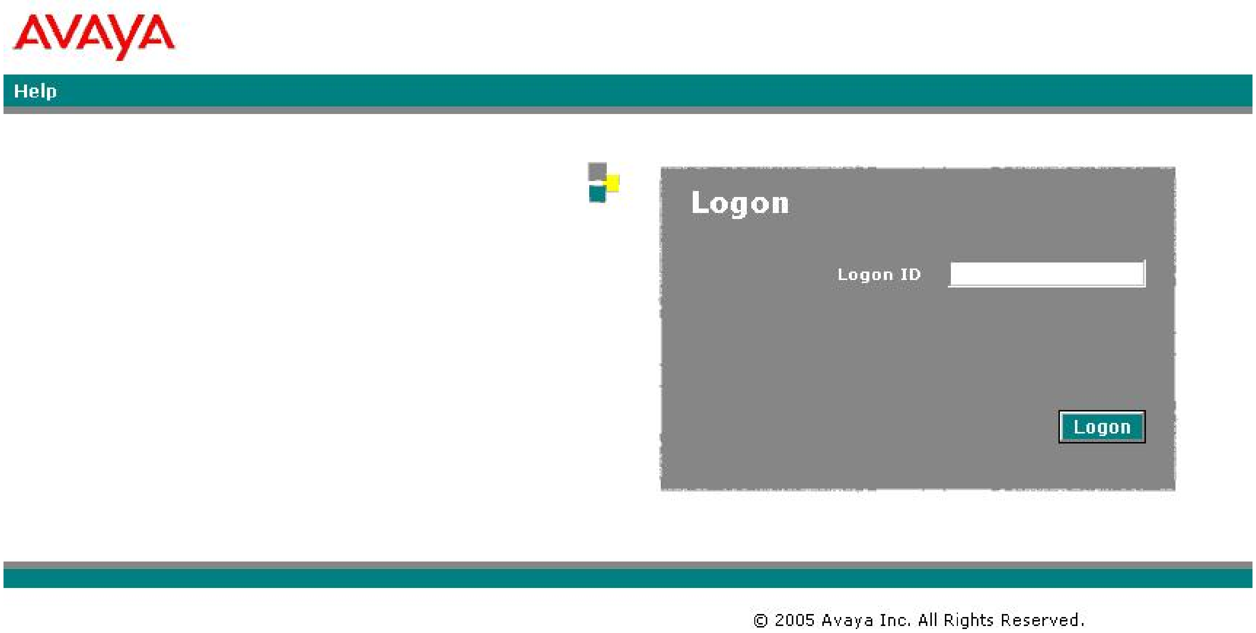
SIP telephones that support IETF RFC 3842 (Subscribe/Notify method) will illuminate/extinguish the MWI lamp when voice messages are left/read for that extension. Cisco 7940/7960 SIP phones do not support this standard, but support an unsolicited Notify method for MWI. To support this important feature for Cisco SIP phones, the Avaya SIP 3.1 offer now supports both methods.

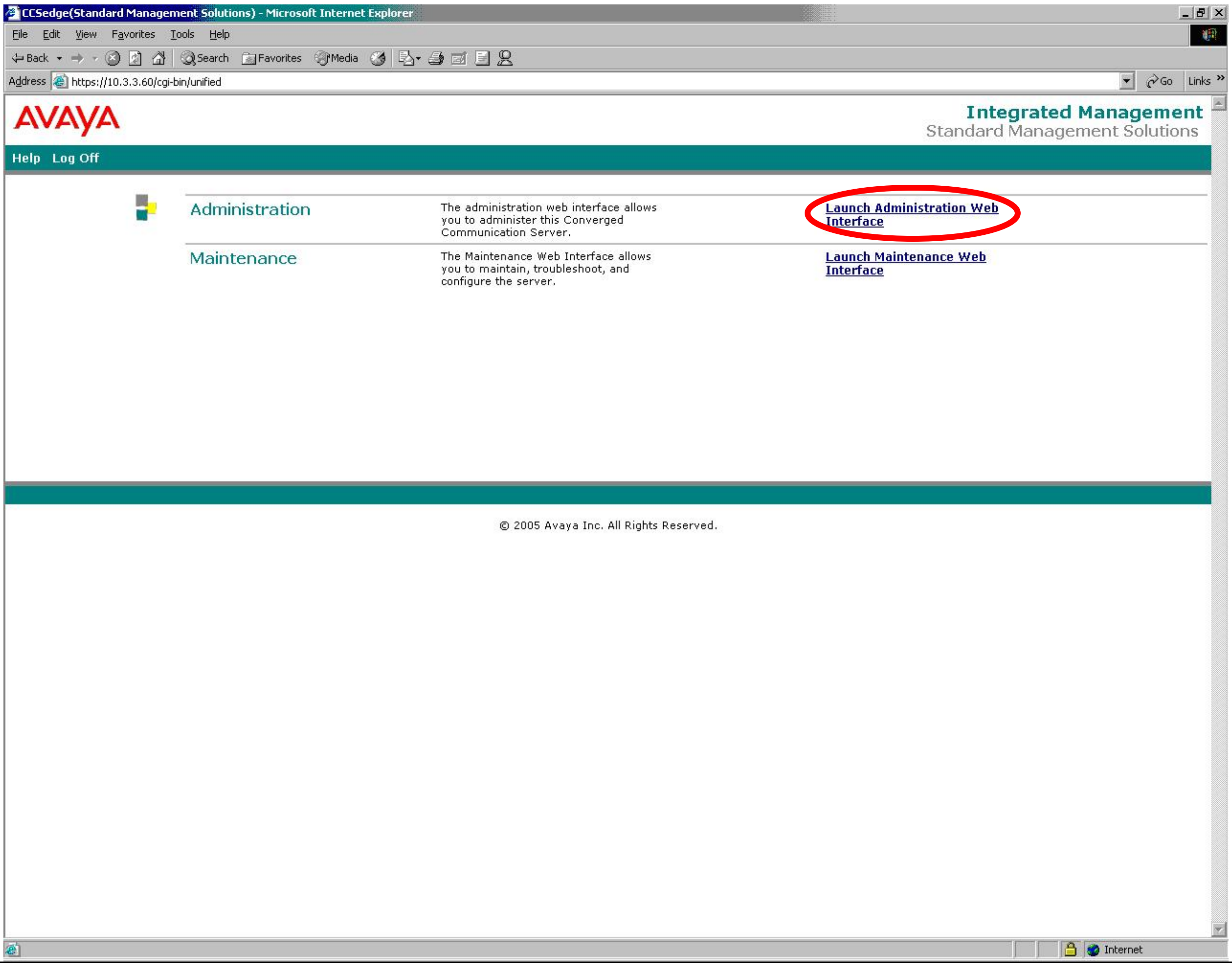
3.2.3. Call Forward Unconditional

It is recommended that this feature be administered as an Avaya OPS FNE rather than using the local call forward feature visibly displayed as a soft key on the telephone. The user of local call forward will not benefit from any of the call coverage features available in Avaya Communication Manager with OPS, including coverage to voice messaging. Administration for local call forwarding is included in these Application Notes for completeness, but the customer should be aware of these limitations.

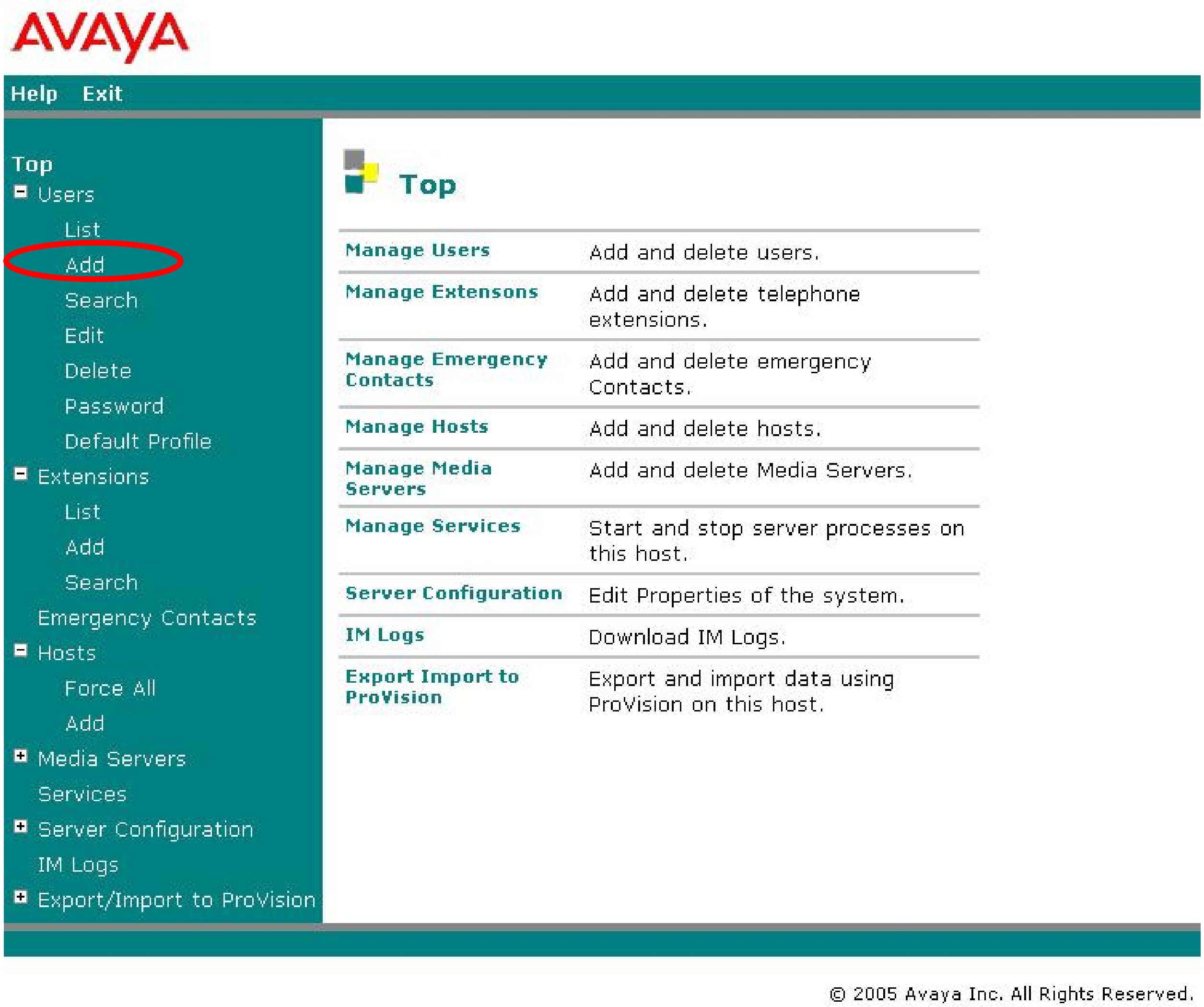
4. Administer SIP Enablement Services

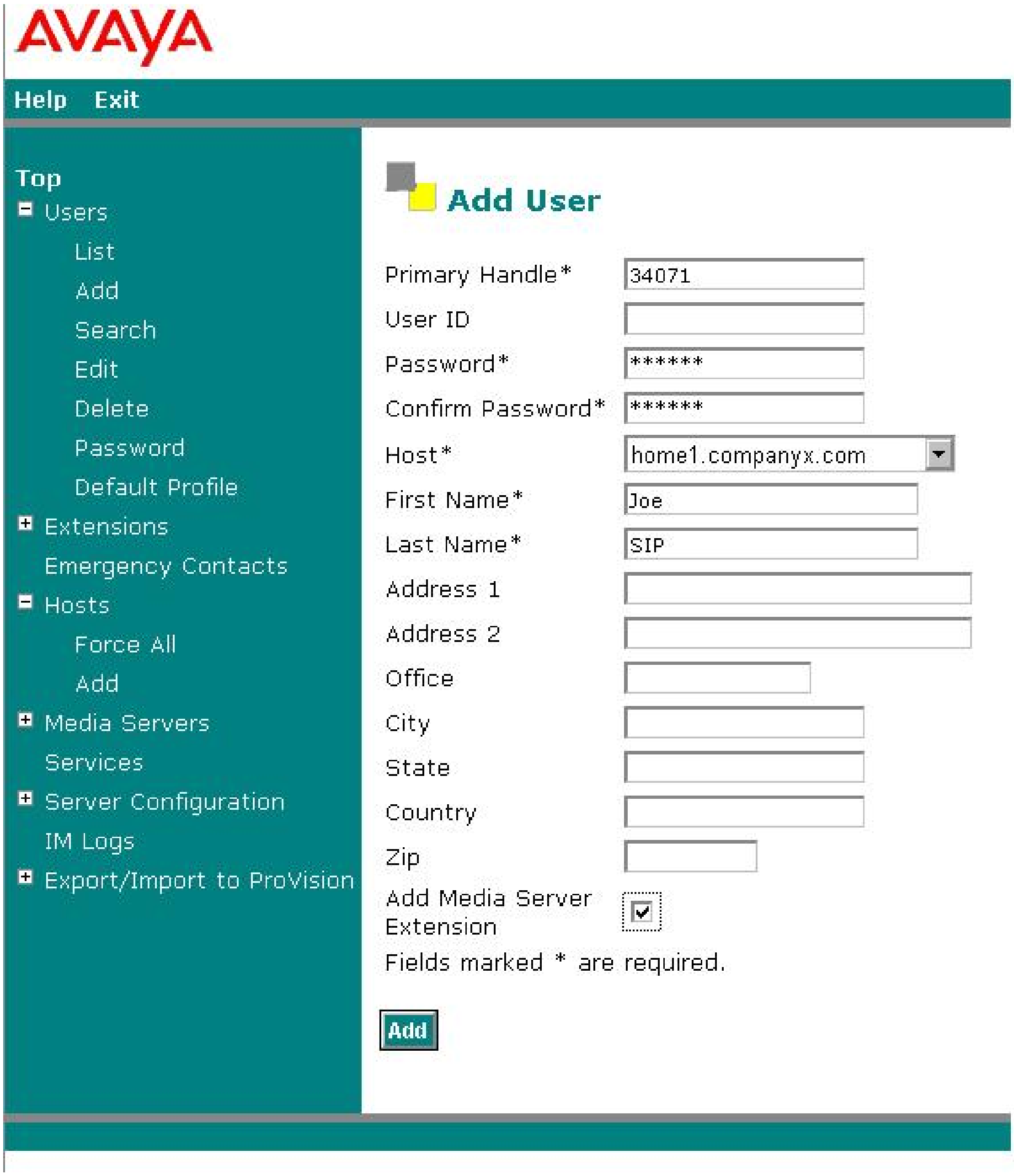
The following steps describe configuration of Avaya SIP Enablement Services for use with Cisco 7940/7960 SIP telephones. Other standard administration functions are covered in Reference [1].

Steps	Description
1.	Avaya SIP Enablement Services is configured using a web browser. Set the URL of the browser to http://IP-address/admin, where *IP-address* is the IP address of the Avaya SIP Enablement Services Edge or Edge/Home Server, and log in as "admin". When prompted, enter the password. 

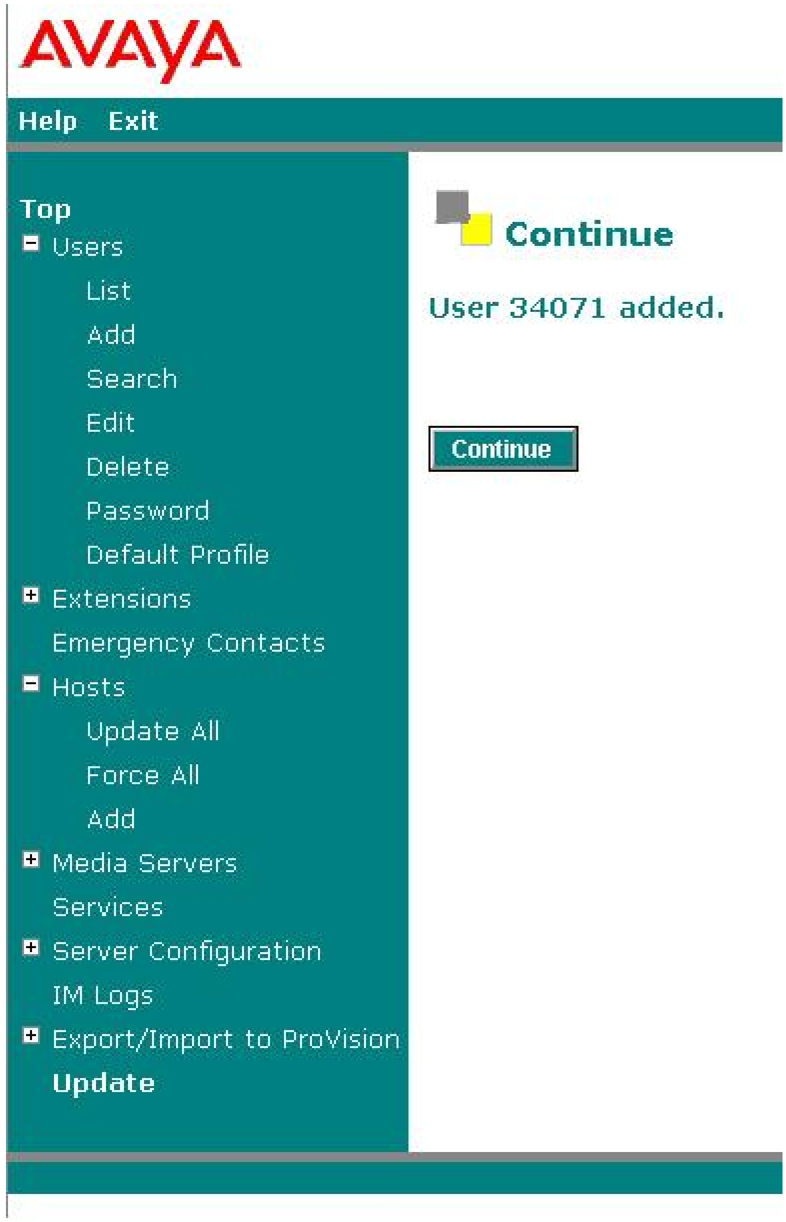
Steps	Description
2.	The main administration screen will be displayed after logging in. Click on **Launch Administration Web Interface.** 

Steps	Description
3.	The SIP Enablement Services administration web interface will be displayed. Click on **Add** under the **Users** heading on the left side of the page.

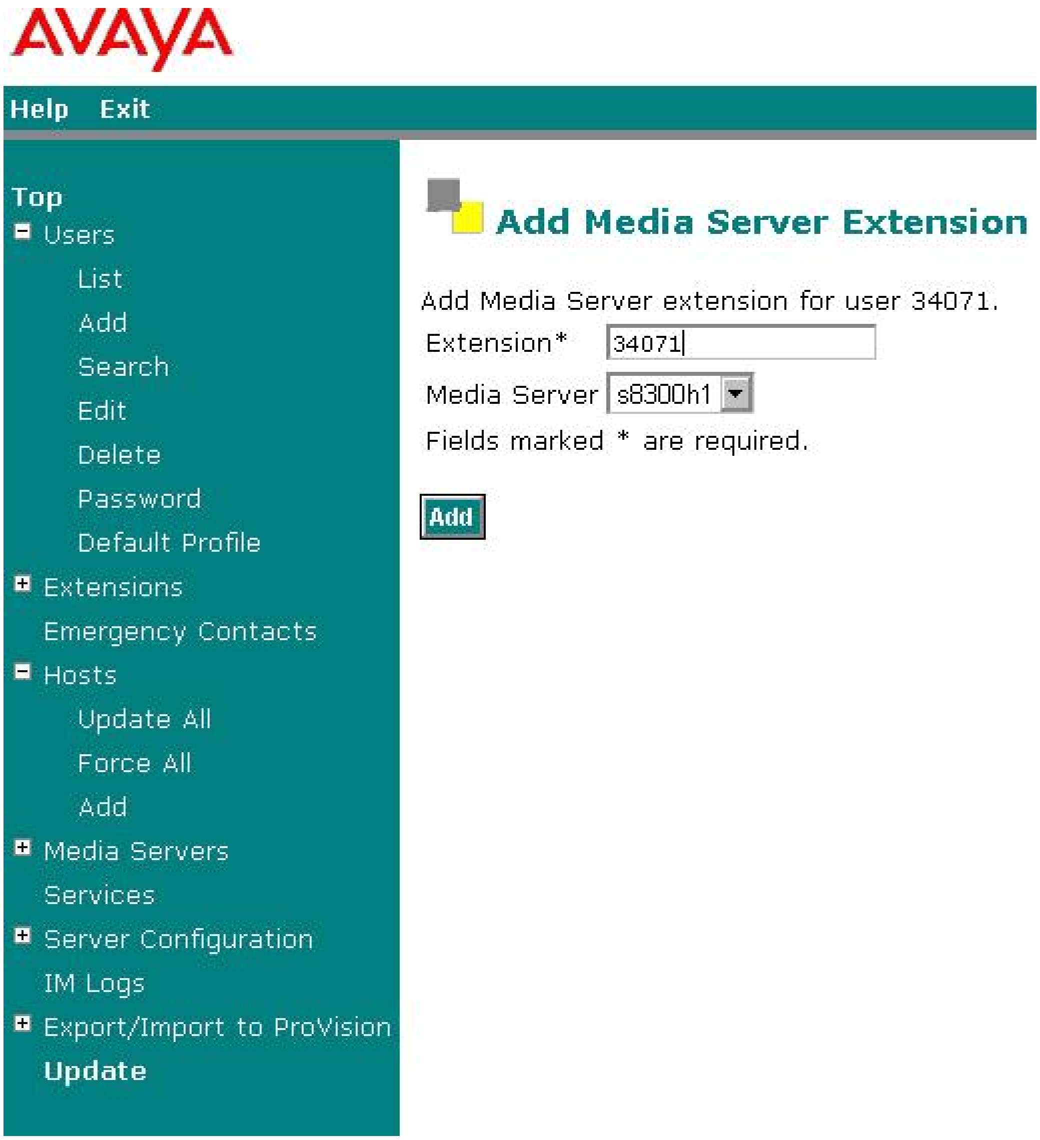


<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>4.</td><td>The Add User page will be displayed. Fill in the required fields (indicated by *). In the screen below, the user corresponding to a SIP telephone is being added. Enter the extension number in the Primary Handle field. The Host field should be set to the DNS host name of the Avaya SIP Enablement Services Home or Home/Edge server to which the user’s phone will register. In this configuration, the telephone will register to Home 1. Check the Add Media Server Extension checkbox. Click on Add.


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

Steps	Description
5.	The confirmation page will be displayed. Click on **Continue**. 

<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>6.</td><td>The Add Media Server Extension page will be displayed. Although not required, it is recommended that the same extension entered in Step 4 be entered in the Extension field. Click on Add. Since the user is being added to Home 1, the Media Server corresponding to the SIP trunk between the S8300 Media Server and Home 1 is selected automatically.



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

Steps	Description
7.	The confirmation page will be displayed. Click on **Continue**. 
8.	Repeat Steps 3-7 for each user to be added to the system.
9.	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.

5. Configure Avaya Communication Manager

This section highlights the important commands for defining SIP telephones as Off-PBX Stations (OPS)[2] on Avaya Communication Manager and administering support for the OPS features indicated in **Table 3**. As mentioned in Section 3.1, many other standard Avaya Communication Manager call features are available to these stations. For complete documentation on administration, see References [4,6,7]. Use the System Access Terminal (SAT) interface to perform the following steps. Log in with the appropriate permissions.

5.1. Verify OPS Capacity

Use the **display system-parameters customer-options** command to verify that **Maximum Off-PBX Telephones – OPS** has been set to the value that has been licensed, and that this value will accommodate the number of phones to be used.

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

     G3 Version: V13
       Location: 1                            RFA System ID (SID): 1
       Platform: 7                            RFA Module ID (MID): 1

                                                               USED
                                Platform Maximum Ports: 900   116
                                      Maximum Stations: 450   41
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 0     0
                    Maximum Off-PBX Telephones -   OPS: 50    26
                    Maximum Off-PBX Telephones - SCCAN: 0     0
```

[2] Depending on the Avaya server product, the acronym OPS stands for two different feature names that are functionally equivalent. For SIP Enablement Services, the extended features capability is referred to as Outboard Proxy SIP. This capability is provided by Avaya Communication Manager as part of a more general feature extension package known as Off-PBX Stations, which can be applied to other remote devices such as cell phones. For that reason, the administration screens in this section will refer to the latter name or "off-pbx-telephone." For the purposes of the Avaya SIP offer and these Application Notes, the terms can be used interchangeably.

5.2. Define System Features

Use the **change system-parameters features** command to administer system wide features for the SIP telephones. Those related to features listed in **Table 3** are shown in bold. These are all standard Avaya Communication Manager features that are also available to OPS stations.

```
change system-parameters features                               Page   1 of  16
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: none
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                  AAR/ARS Dial Tone Required? y
                              Music/Tone on Hold: music Type: port  001V208
              Music (or Silence) on Transferred Trunk Calls? no
                       DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n





             Abbreviated Dial Programming by Assigned Lists? n
       Auto Abbreviated/Delayed Transition Interval (rings): 2
                     Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? n
```

```
change system-parameters features                               Page   4 of  16
                        FEATURE-RELATED SYSTEM PARAMETERS
          Reserved Slots for Attendant Priority Queue: 5
                            Time before Off-hook Alert: 10
              Emergency Access Redirection Extension:
 Number of Emergency Calls Allowed in Attendant Queue: 5

              Call Pickup on Intercom Calls? y         Call Pickup Alerting? n
 Temporary Bridged Appearance on Call Pickup? n        Directed Call Pickup? y
                  Extended Group Call Pickup: simple

    Deluxe Paging and Call Park Timeout to Originator? n
   Controlled Outward Restriction Intercept Treatment: tone
  Controlled Termination Restriction (Do Not Disturb): tone
            Controlled Station to Station Restriction: tone
AUTHORIZATION CODE PARAMETERS        Authorization Codes Enabled? n
```

```
change system-parameters features                              Page  16 of  16
                        FEATURE-RELATED SYSTEM PARAMETERS

INTERCEPT TREATMENT PARAMETERS
       Invalid Number Dialed Intercept Treatment: announcement  35010
                   Invalid Number Dialed Display:
    Restricted Number Dialed Intercept Treatment: announcement  35011
                Restricted Number Dialed Display:
   Intercept Treatment On Failed Trunk Transfers? n

WHISPER PAGE
   Whisper Page Tone Given To: paged

DIGITAL STATION LINE APPEARANCE LED SETTINGS
                    Station Putting Call On Hold: green  wink
                     Station When Call is Active: steady
         Other Stations When Call Is Put On Hold: green  wink
              Other Stations When Call Is Active: green
                                         Ringing: green  flash
                                            Idle: steady
           Display Information With Bridged Call? n
                              Pickup On Transfer? y
```

5.3. Define the Dial Plan

Use the **change dialplan analysis** command to define the dial plan formats to be used in the system. This includes all telephone extensions and OPS Feature Name Extensions (FNEs). To define the FNEs for the OPS features listed in **Table 3**, a Feature Access Code (FAC) must also be specified for the corresponding feature[3]. In the sample configuration, telephone extensions are five digits in length and begin with 3 or 4, FNEs are five digits beginning with 7, and the FACs have various formats as indicated with the **Call Type** of “fac”.

```
change dialplan analysis                                       Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                                           Percent Full:    0

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

[3] Note that if SIP Universal Resource Identifiers (URIs) can be programmed into the telephone, then Feature Name URIs (FNUs) can be used instead, and neither FACs nor FNEs need to be defined for these OPS features. See [6] for more details.

5.4. Feature Access Codes (FACs)

Use the **change feature-access-codes** command to define the access codes corresponding to the OPS FNEs, shown in bold.

```
change feature-access-codes                                     Page   1 of   5
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: 101
         Abbreviated Dialing List2 Access Code: 102
         Abbreviated Dialing List3 Access Code: 103
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: 106
                       Answer Back Access Code: 105
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code:
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: *5     Deactivation: #5
Call Forwarding Activation Busy/DA: *2     All: 112    Deactivation: #2
                         Call Park Access Code: 104
                       Call Pickup Access Code: *6
CAS Remote Hold/Answer Hold-Unhold Access Code: #6
                  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   5
                               FEATURE ACCESS CODE (FAC)
                  Data Origination Access Code:
                      Data Privacy Access Code:
             Directed Call Pickup Access Code: 154
     Emergency Access to Attendant Access Code:
         EC500 Self-Administration Access Code:
                     Enhanced EC500 Activation:        Deactivation:
  Extended Call Fwd Activate Busy D/A      All:        Deactivation:
       Extended Group Call Pickup Access Code: 121
               Facility Test Calls Access Code:
                             Flash Access Code:
             Group Control Restrict Activation:        Deactivation:
                    Hunt Group Busy Activation: *8     Deactivation: #8
                              ISDN Access Code:
               Last Number Dialed Access Code: *9
     Leave Word Calling Message Retrieval Lock: *1
   Leave Word Calling Message Retrieval Unlock: #1
             Leave Word Calling Send A Message:
           Leave Word Calling Cancel A Message:
              Malicious Call Trace Activation: 113    Deactivation: 114
         Meet-me Conference Access Code Change:
```

```
change feature-access-codes                                     Page   3 of   5
                               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: 115
      Per Call CPN Unblocking Code Access Code: 116

                   Priority Calling Access Code: *7
                            Program Access Code: *0

       Refresh Terminal Parameters Access Code: 094
              Remote Send All Calls Activation:        Deactivation:
               Self Station Display Activation: 107
                     Send All Calls Activation: *3     Deactivation: #3
         Station Firmware Download Access Code:
                       Station Lock Activation:        Deactivation:
      Station Security Code Change Access Code: 099
              Station User Admin of FBI Assign:                   Remove:
  Station User Button Ring Control Access Code:
             Terminal Dial-Up Test Access Code: 095
```

```
change feature-access-codes                                     Page   4 of   5
                               FEATURE ACCESS CODE (FAC)
Terminal Translation Initialization Merge Code:          Separation Code:
            Transfer to Voice Mail Access Code: #9
          Trunk Answer Any Station Access Code:
              User Control Restrict Activation: 091    Deactivation: 092
  Voice Coverage Message Retrieval Access Code:
 Voice Principal Message Retrieval Access Code:
           Whisper Page Activation Access Code: 120
```

5.5. Define Feature Name Extensions (FNEs)

The FNEs can be defined using the **change off-pbx-telephone feature-name-extensions** command. This command is used to support both OPS and Extension to Cellular. The fields that have been left blank correspond to those more appropriate for Extension to Cellular.

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

     Active Appearance Select:                  Idle Appearance Select:
          Automatic Call Back: 70003               Last Number Dialed: 70019
   Automatic Call-Back Cancel: 70004             Malicious Call Trace: 70029
             Call Forward All: 70005      Malicious Call Trace Cancel: 70021
  Call Forward Busy/No Answer: 70006               Off-Pbx Call Enable:
          Call Forward Cancel: 70007              Off-Pbx Call Disable:
                    Call Park: 70008                     Priority Call: 70000
        Call Park Answer Back: 70009                    Send All Calls: 70001
                 Call Pick-Up: 70010             Send All Calls Cancel: 70002
         Conference on Answer: 70011               Transfer On Hang-Up:
         Calling Number Block: 70012            Transfer to Voice Mail: 70023
       Calling Number Unblock: 70013           Whisper Page Activation: 70026
        Directed Call Pick-Up: 70014
        Drop Last Added Party: 70015
    Exclusion (Toggle On/Off):
   Extended Group Call Pickup: 70025
       Held Appearance Select:
```

5.6. Specify Class of Service (COS)

Use the **change class-of-service** command to set the appropriate service permissions to support OPS features (shown in bold). For the example, COS 1 was used. In the case of **VIP Caller**, set the value to “y” only if all calls made by telephones with this COS should be priority calls. Priority call indication (e.g., distinctive ring and display of “Priority”) is only supported on Avaya Digital and IP telephones.

change cos   Page 1 of 2

CLASS OF SERVICE

	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
Auto Callback	n	**y**	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	n
Data Privacy	n	n	n	y	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	n
Console Permissions	y	n	y	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	n	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

change cos   Page 2 of 2

CLASS OF SERVICE

	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
VIP Caller	n	**n**	y	n	n	n	n	n	n	n	n	n	n	n	n	n

5.7. Specify Class of Restriction (COR)

Use the **change class-of-restriction** command to enable applicable calling features. To use the Directed Call Pickup feature, the **Can Use Directed Call Pickup** and **Can Be Picked Up By Directed Call Pickup** fields must be set to “y” for the affected stations. In the sample configuration, the SIP telephones were assigned to COR 2. Note that Page 3 can be used to implement a form of centralized call screening for groups of stations and trunks.

```
change cor 2                                                    Page   1 of   4
                              CLASS OF RESTRICTION

               COR Number: 2
          COR Description: Stations

                      FRL: 0                                    APLT? y
  Can Be Service Observed? n           Calling Party Restriction: none
Can Be A Service Observer? n            Called Party Restriction: none
 Partitioned Group Number: 1       Forced Entry of Account Codes? n
         Priority Queuing? n                 Direct Agent Calling? n
     Restriction Override: none       Facility Access Trunk Test? n
     Restricted Call List? n                  Can Change Coverage? n

            Access to MCT? y            Fully Restricted Service? n
Group II Category For MFC: 7
         Send ANI for MFE? n
            MF ANI Prefix:              Automatic Charge Display? n
Hear System Music on Hold? y   PASTE (Display PBX Data on Phone)? n
                        Can Be Picked Up By Directed Call Pickup? y
                                    Can Use Directed Call Pickup? y
                                    Group Controlled Restriction: inactive
```

```
change cor 2                                                    Page   3 of   4
                              CLASS OF RESTRICTION


CALLING PERMISSION (Enter "y" to grant permission to call specified COR)

  0? y     12? y     24? y     36? y     48? y     60? y    72? y     84? y
  1? y     13? y     25? y     37? y     49? y     61? y    73? y     85? y
  2? y     14? y     26? y     38? y     50? y     62? y    74? y     86? y
  3? y     15? y     27? y     39? y     51? y     63? y    75? y     87? y
  4? y     16? y     28? y     40? y     52? y     64? y    76? y     88? y
  5? y     17? y     29? y     41? y     53? y     65? y    77? y     89? y
  6? y     18? y     30? y     42? y     54? y     66? y    78? y     90? y
  7? y     19? y     31? y     43? y     55? y     67? y    79? y     91? y
  8? y     20? y     32? y     44? y     56? y     68? y    80? y     92? y
  9? y     21? y     33? y     45? y     57? y     69? y    81? y     93? y
 10? y     22? y     34? y     46? y     58? y     70? y    82? y     94? y
 11? y     23? y     35? y     47? y     59? y     71? y    83? y     95? y
```

5.8. Add Coverage Path

Configure the coverage path to be used for the voice messaging hunt group, which is group h1 in the sample configuration. The default values shown for **Busy?**, **Don't Answer?**, and **DND/SAC/Goto Cover?** can be used for the *Coverage Criteria*. In this case, the **Number of Rings** before the call goes to voice messaging has been extended from the default of 2 to 4 rings.

```
add coverage path 1                                                Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 1
                                                  Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?            n               n
               Busy?            y               y
       Don't Answer?            y               y        Number of Rings: 4
                All?            n               n
 DND/SAC/Goto Cover?            y               y
   Holiday Coverage?            n               n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n

  Point1: h1        Rng:    Point2:                    Point3:
  Point4:                   Point5:                    Point6:
```

5.9. Add stations

Use the **add station** command to add a station for each SIP phone to be supported. Assign the same extension as the media server extension administered in SIP Enablement Services. Use the default value for the **Station Type,** "x" for the **Port**, and be sure to include the **Coverage Path** for voice messaging or other hunt group if available. Use the **COS** and **COR** values administered in the previous sections. The **Name** field is optional and is shown on the display of Avaya non-SIP telephones when receiving calls from this station[4]. Use default values for the other fields on Page 1.

[4] For SIP-to-SIP calls using Cisco telephones, the *Display Name* ("Joe SIP") and *URI* (34071@companyx.com) administered at the calling phone is displayed at the called phone.

```
add station 34071                                               Page   1 of   4
                                     STATION

Extension: 34071                           Lock Messages? n          BCC: 0
     Type: 6408D+                          Security Code:             TN: 1
     Port: X                             Coverage Path 1: 1          COR: 2
     Name: Joe SIP                       Coverage Path 2:            COS: 1
                                         Hunt-to Station:

STATION OPTIONS
              Loss Group: 2              Personalized Ringing Pattern: 1
             Data Module? n                          Message Lamp Ext: 34071
            Speakerphone: 2-way                   Mute Button Enabled? y
        Display Language: english

                                                    Media Complex Ext:
                                                         IP SoftPhone? n
```

On Page 2, note the following:

- If this SIP telephone will have a bridged appearance for another telephone (see Page 3 for this station), then **Bridged Call Alerting** should be set to "y", so that this phone will ring when the other phone is called. Note that no other operational behaviors of the bridged appearance feature apply to SIP telephones (e.g. off-hook indication, bridge-on, etc.).
- By default, the last call appearance is reserved for outgoing calls from the phone. If it is desirable to allow an incoming call to use the last available call appearance when all others are occupied, set the **Restrict Last Appearance** field to "n". In this mode, all call appearances are available for making or receiving calls.
- Enter the name of the voice messaging system administered for this system in **AUDIX Name**.

```
add station 34071                                               Page   2 of   4
                                     STATION
FEATURE OPTIONS
           LWC Reception: audix            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 Idle Line Preference? n
   Bridged Call Alerting? y                       Restrict Last Appearance? n
  Active Station Ringing: single          Conf/Trans on Primary Appearance? n

        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: basic
     MWI Served User Type:                      Display Client Redirection? n
              AUDIX Name: audix                  Select Last Used Appearance? n
                                                   Coverage After Forwarding? s

                                                 Direct IP-IP Audio Connections? y
 Emergency Location Ext: 34071                          IP Audio Hairpinning? y
```

On Page 3 under the heading **BUTTON ASSIGNMENTS**, fill in the number of call appearances ("call-appr" buttons) that are to be supported for the telephone. Use the following guidelines for determining the correct number:

- To support certain transfer and conference scenarios, the minimum number of "call-appr" buttons should be 3.
- If call-waiting is activated at the phone, two calls can be active per line appearance, so the number of "call-appr" buttons should be double the number of line appearances configured at the telephone.

```
add station 34071                                              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: system            List2:                    List3:

BUTTON ASSIGNMENTS
 1: call-appr                        5: brdg-appr  Btn:1  Ext:44072
 2: call-appr                        6: no-hld-cnf
 3: call-appr                        7:
 4: call-appr                        8:
```

Under the same heading, enter the function button names, if required, for OPS FNEs that will be used at the phone. Only the FNEs shown in **Table 4** require the station to have a corresponding function button. Avaya Communication Manager features that do not require the user to dial an FNE, such as a bridged appearance, may require the appropriate function button, as shown above.

FNE Name	Function Button
Automatic Callback, Automatic Callback Cancel	auto-cback
Conference on Answer	no-hld-cnf

Table 4: Feature Name Extensions Requiring Station Buttons

In the sample configuration, two line appearances were administered at the telephone for extension 34071. A bridged appearance was defined on station 44072 and the Conference On Answer FNE was included in the speed dial button programming.

Use the **change off-pbx-telephone station-mapping** command to map the Avaya Communication Manager extension (34071) to the same SIP Enablement Services media server extension. Enter the field values shown. For the sample configuration, the **Trunk Selection** value indicates the SIP trunk group. The **Configuration Set** value can reference a set that has the default settings in Avaya Communication Manager.

```
change off-pbx-telephone station-mapping 34071                    Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station     Application   Dial   Phone Number       Trunk       Configuration
 Extension                 Prefix                    Selection   Set
 34071         OPS              - 34071               10         1
                                -
                                -
                                -
```

On Page 2, change the **Call Limit** to match the number of "call-appr" entries in the **add station** form. Also make sure that **Mapping Mode** is set to "both" (the default value for a newly added station).

```
change off-pbx-telephone station-mapping 34071                    Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged
 Extension     Limit      Mode         Allowed     Calls
 34071        4         both           all         both
```

6. Configure the Cisco 7940/7960 SIP Telephone

6.1. Registration and Basic Dialing

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 telephone-specific configuration file containing settings applicable only to that telephone (`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 6.3), 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
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 telephone and those administered in SIP Enablement Services. A sample value is shown for the configuration in **Figure 1**. Parameter names have the form `ObjectNameX_ParamName`, 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. **Proxy1_address** should be set to the SIP domain name administered in SIP Enablement Services. **Outbound_proxy** should be set to the IP address or host name of the SIP Enablement Services Server[5,6]. Normally, the proxy parameters would reside in the default configuration file, and the user name and password would reside in the phone-specific file. Other parameters that can be included in the default configuration file include: • The **image_version** parameter specifies the firmware image the telephone should be running. The telephone will upgrade to that version if the files are available under the root directory of the TFTP server. • The **dial_template** parameter specifies the name of the dial plan file to be loaded by the telephone (see Step 2). • Specification of the bitmap to be displayed as a logo on the phone ("*Powered by* Avaya" in the sample configuration). • The **messages_uri** parameter can be set to the hunt group extension of the voice messaging system, so that the Messages button can be pressed to access that system. • The **local_cfwd_enable** parameter controls whether the local call forward soft key is displayed on the phone (disabled in the sample configuration). See **Appendices A** and **B** for sample files showing key parameters with explanatory comments.

Avaya SIP Enablement Services	**Cisco 7940/7960**	
	Parameter Name	Example Value
User Administration		
User ID	line1_name	34071
Password	line1_password	123456
Proxy Administration		
SIP Domain	proxy1_address	companyx.com
Proxy Port	proxy1_port	5060
	proxy_register	1
	outbound_proxy	home1.companyx.com
	outbound_proxy_port	5060
	messages_uri	35000

[5] For multiple home configurations, one approach would be to have separate TFTP servers, each with a different default configuration file pointing to the corresponding home server. The DHCP server would specify the appropriate TFTP server based on the scope of the telephone making the request.
[6] Do not set **proxy_backup** unless special configurations such as Cisco Survivable Remote Site Telephony (SRST) are being used.

Steps	Description

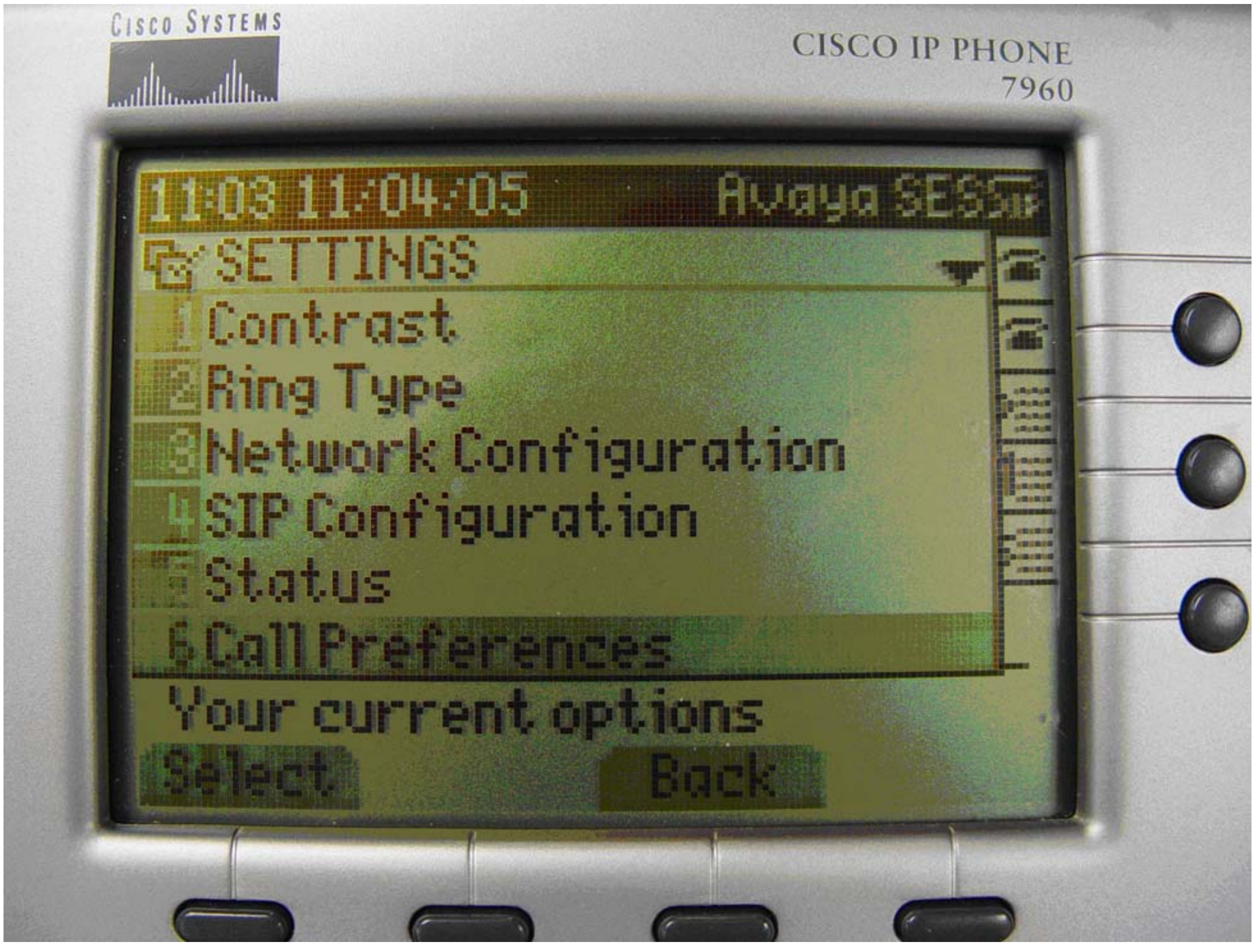
2.	Edit the dial plan file (`dialplan.xml`), which will be downloaded to the phone from the TFTP server (see Reference [3]). The phone uses this file to determine when enough digits have been pressed to complete dialing, so that the user need not press the “Dial” soft key to launch the call. The dial 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. The sample configuration used the following dial plan file:
	
	`<DIALTEMPLATE>`
	`    <TEMPLATE MATCH="3...." Timeout="0" User="Phone" Rewrite="%s"/>`
	`    <TEMPLATE MATCH="4...." Timeout="0" User="Phone" Rewrite="%s"/>`
	`    <TEMPLATE MATCH="7...." Timeout="0" User="Phone" Rewrite="%s"/>`
	`</DIALTEMPLATE>`
	
	This covers 5 digit extensions beginning with 3 and 4, as well as cases where the user may dial an OPS FNE rather than press a programmed speed dial button. The periods in the match string stand for any digit. The entries specified in this file should agree with the dial plan administered in Avaya Communication Manager.
	
	For further information on defining the dial plan, see “How to Create Dial Plans” in Chapter 3 of [3].

Steps	Description
3.	Reboot the phone. If TFTP support has been properly configured, the phone will download the default and specific configuration files, and register with SIP Enablement Services. Registration can be verified by the absence of an “X” near the phone icon for the line appearance, as shown below. 
4.	For basic calling, lift the receiver (or press **Speaker**) and dial any number using the dial plan centrally administered in Avaya Communication Manager. Those features listed in **Table 3** as being locally supported at the phone (e.g., hold, transfer, conference, etc.) can be used. Some of these features require activation at the phone, and are described in Section 6.2.1. Section 6.2.2 describes configuring the telephone to access the Avaya extended feature set available via OPS.

6.2. Local Calling Features

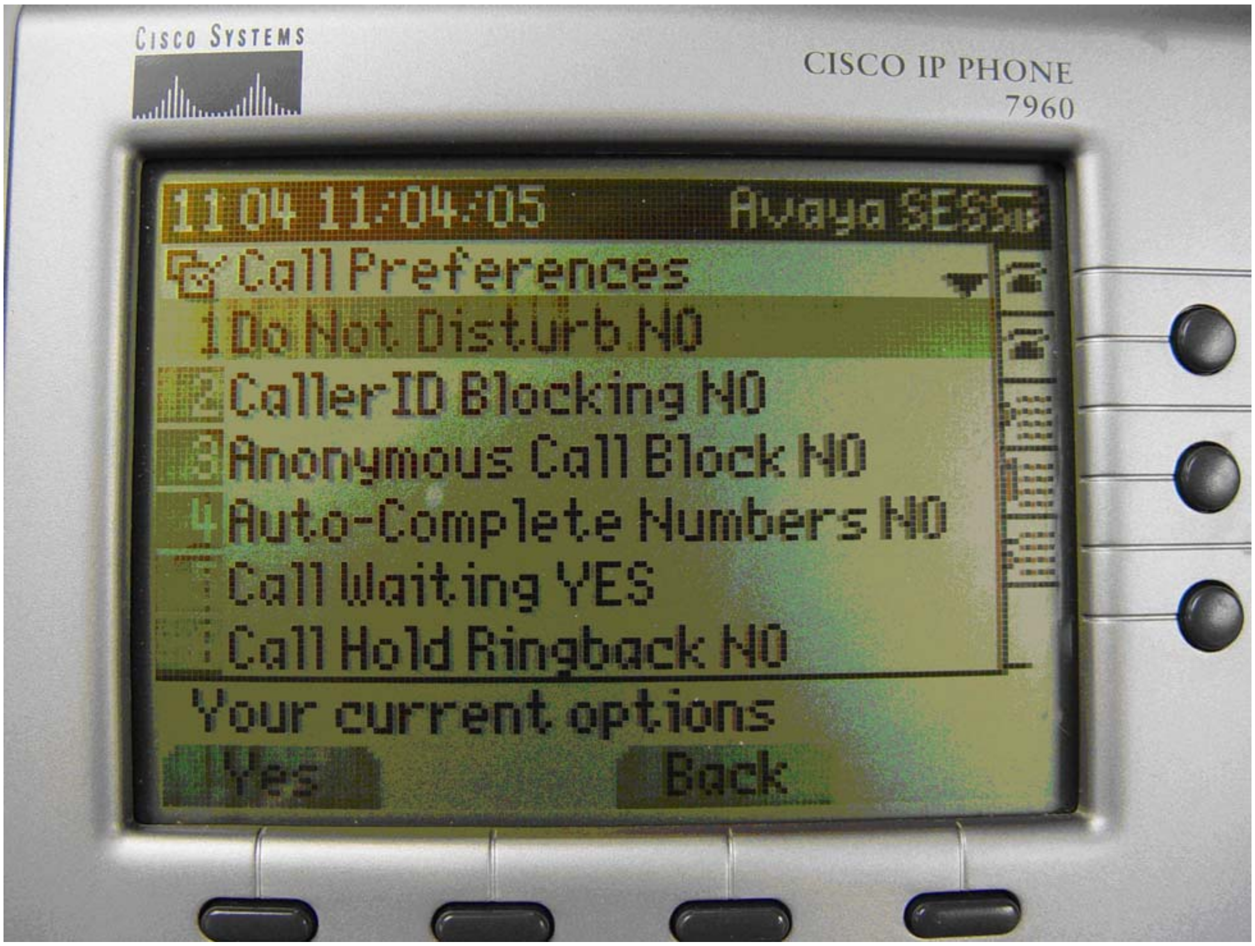
The following sections describe how the telephone user can administer local telephone features that are compatible with the Avaya SIP offer. To configure some of the features described in the following sections, the *Call Preferences* menu must be accessed as follows:

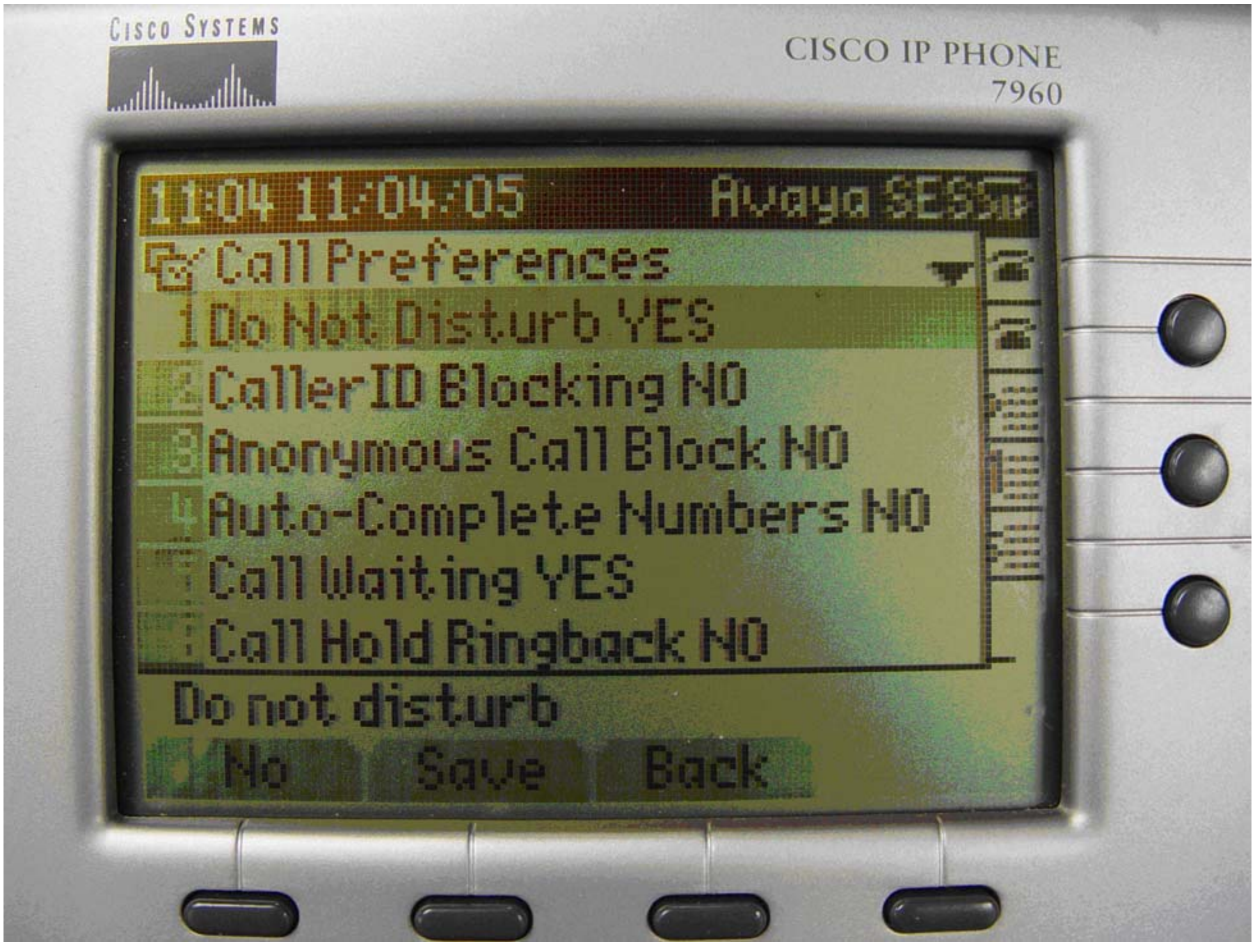
Steps	Description
1.	At the telephone, press the **settings** button to bring up the main configuration menu. 

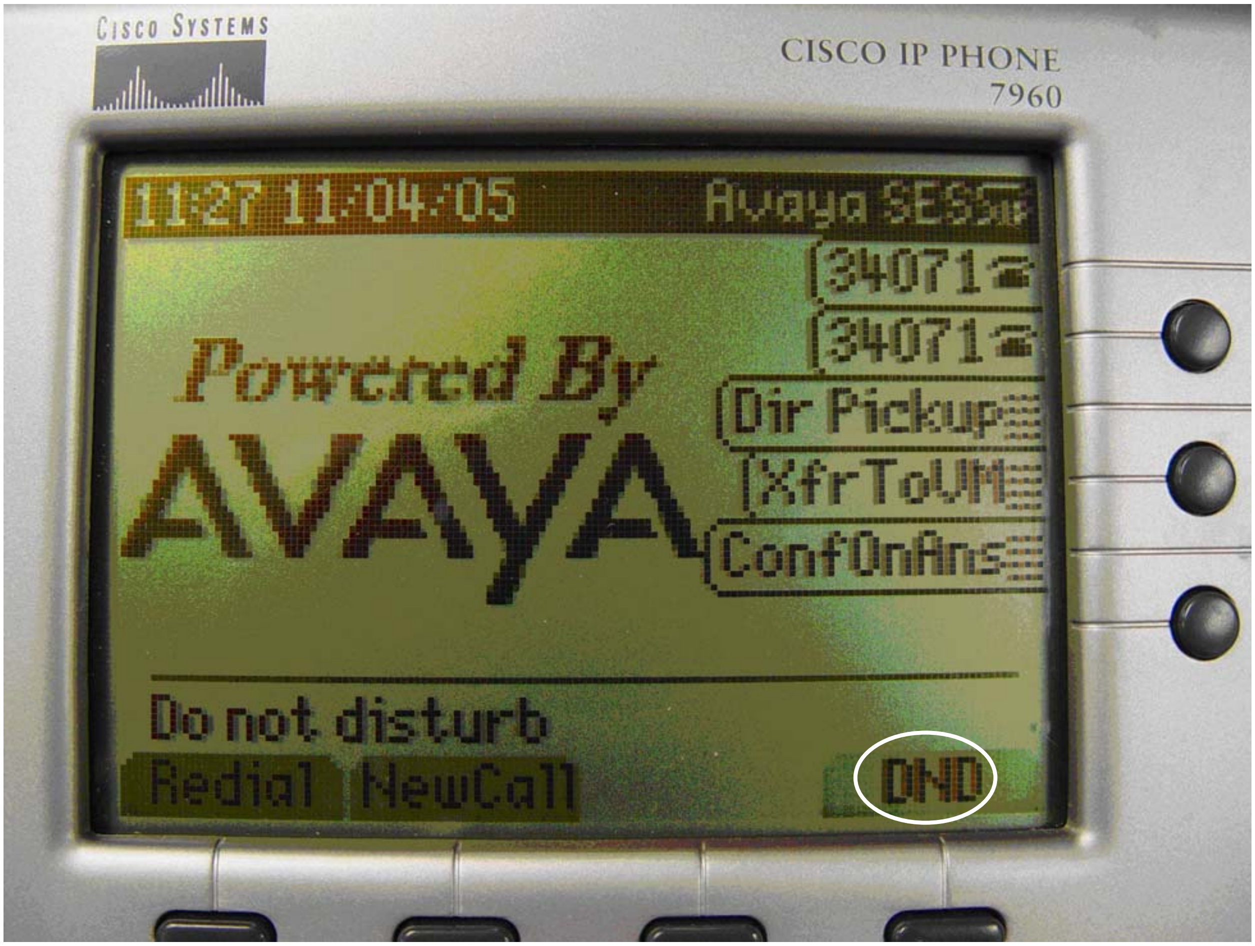
Steps	Description
2.	Using the **up/down** button, move the highlighted selection to **Call Preferences**, and press the **Select** soft key. 
3.	The Call Preferences menu will be displayed. Follow the steps in the following subsections to configure specific features.

6.2.1. Do Not Disturb (DND) and Call Waiting

When enabled, DND will cause incoming calls to be routed to the coverage path specified for this telephone in Avaya Communication Manager. This is typically a voice messaging system. Call waiting allows a second call to be answered while a call is in progress. The steps below address DND, but also apply to call waiting.

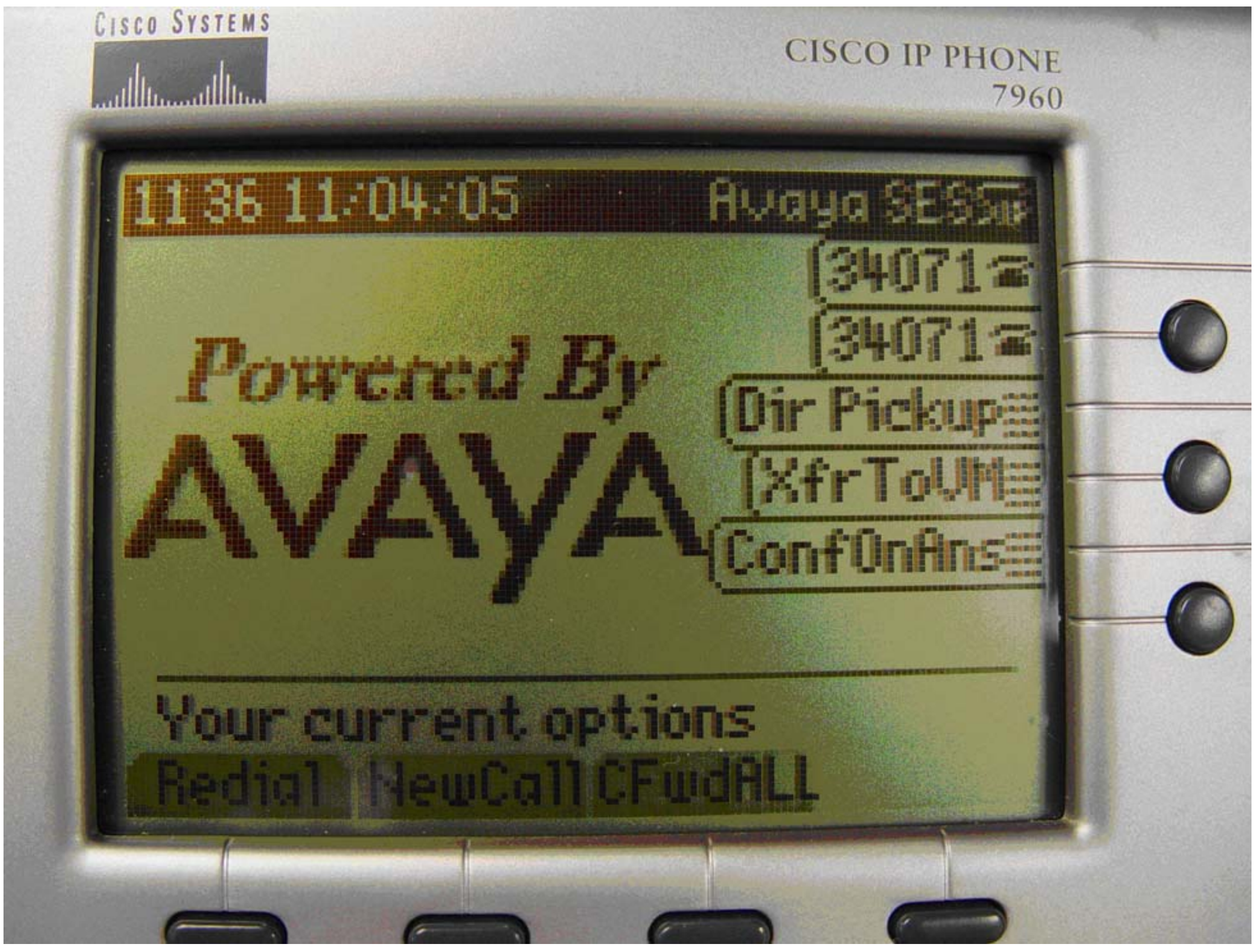
Steps	Description
1.	At the telephone, access the *Call Preferences* menu. Using the **up/down** button, move the highlighted selection to **Do Not Disturb**, and press the **Yes** soft key. 

Steps	Description
2.	Press the **Save** function button to save the setting. 

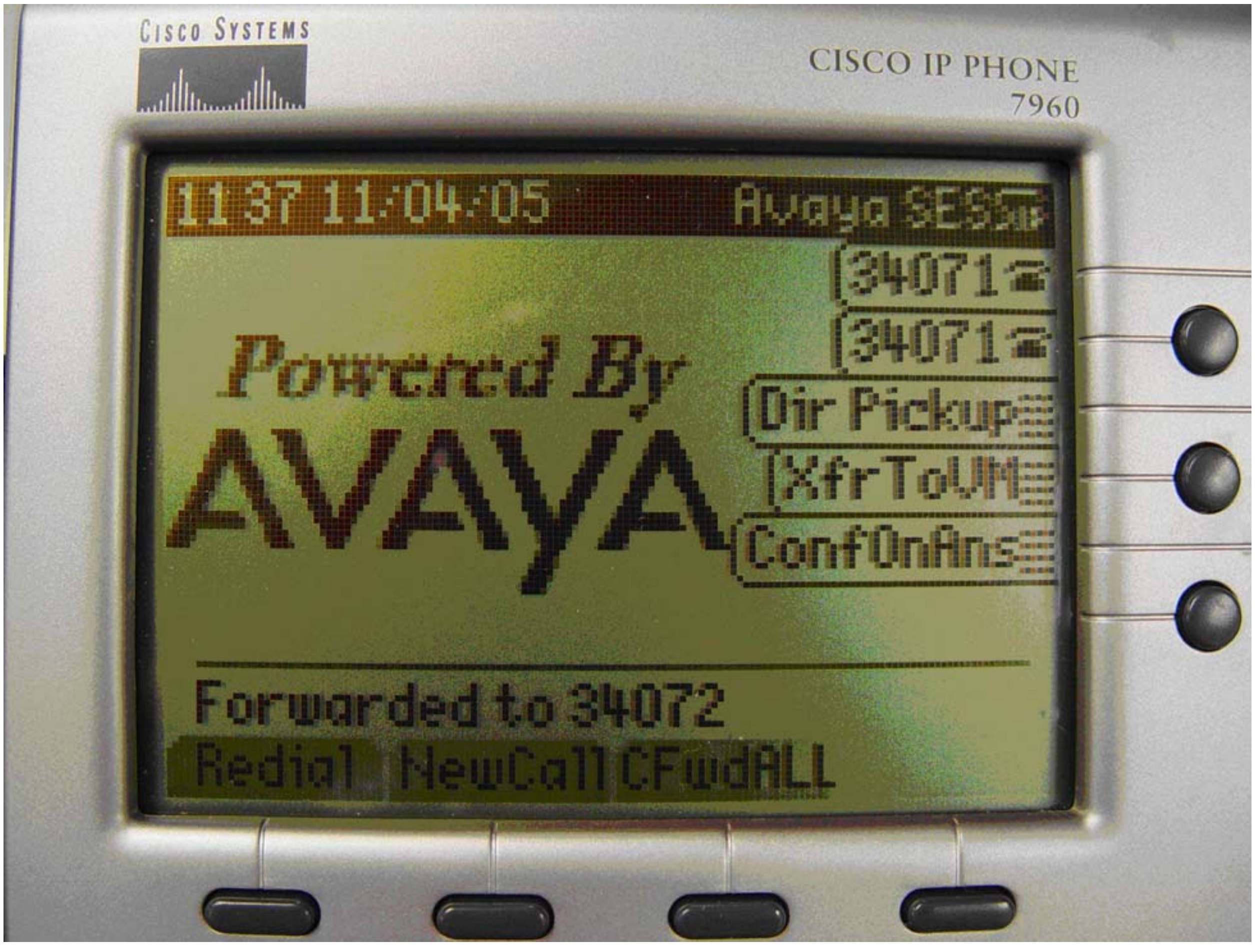
Steps	Description
3.	Press the **Settings** button to return to the main telephone display, which will now show the DND soft key. To deactivate the feature, press this soft key. In the case of Call Waiting, no indication is displayed, and steps 1-3 must be repeated to deactivate it. 

6.2.2. Call Forward

As described in Section 3.2.3, Avaya does not recommend using the phone's local call forward due to its limitations relative to call coverage and voice messaging features. However, the following steps show how to forward calls to a specific extension using local forwarding, should it be desirable under certain circumstances.

Steps	Description
1.	In the default configuration file, make sure that the **local_cfwd_enable** parameter is set to 1, so that the **CFwdALL** soft key will be displayed on the phone. If it is necessary to edit the file, reboot the phone.
2.	On the main telephone display, select the **CFwdALL** soft key. 

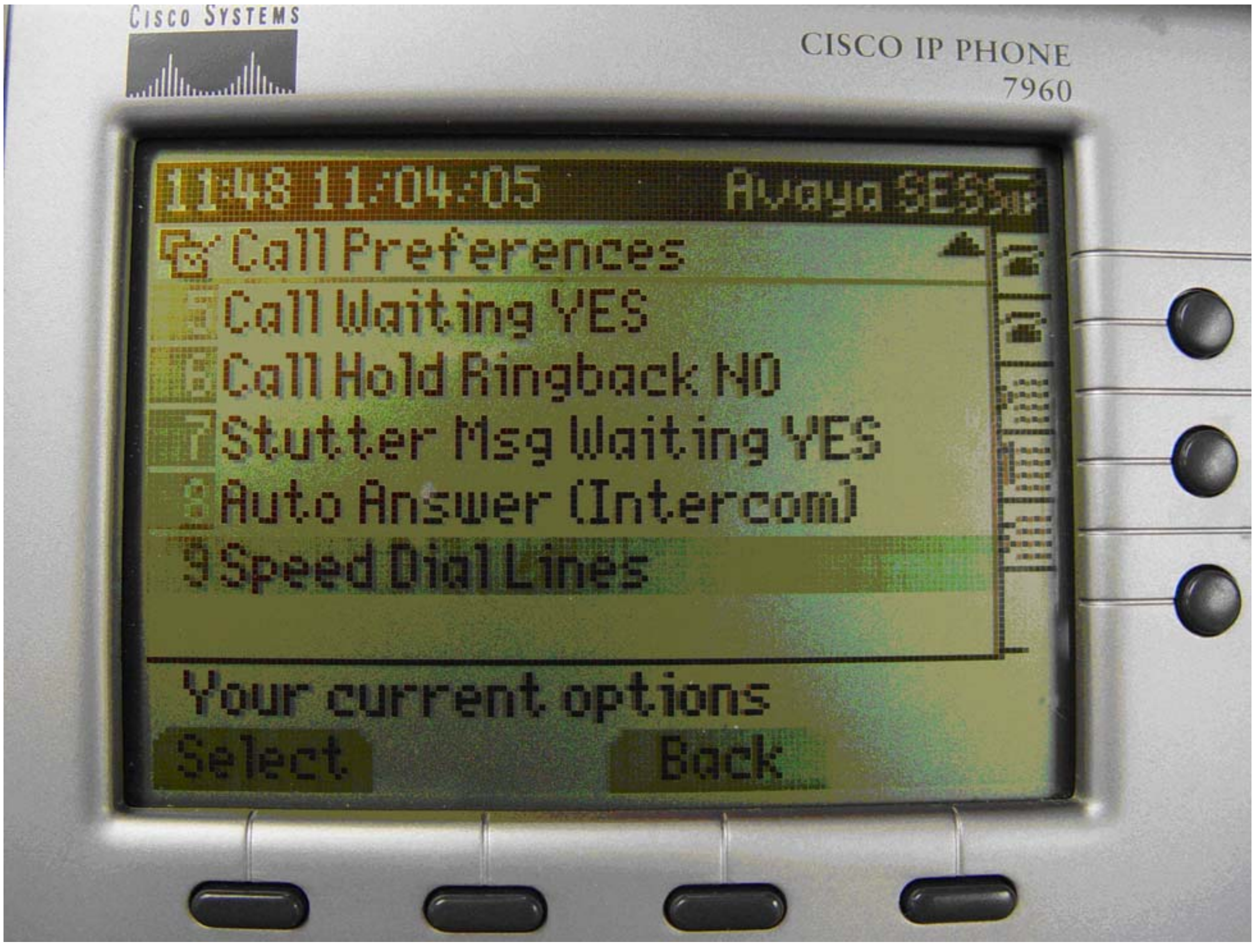
Steps	Description
3.	Enter the number to which all calls are to be forwarded, and press the **Accept** soft key. 

Steps	Description
4.	The main telephone display will return, showing the call forward indication. To deactivate the feature, press the **CFwdALL** soft key. 

6.3. The Avaya Extended Feature Set

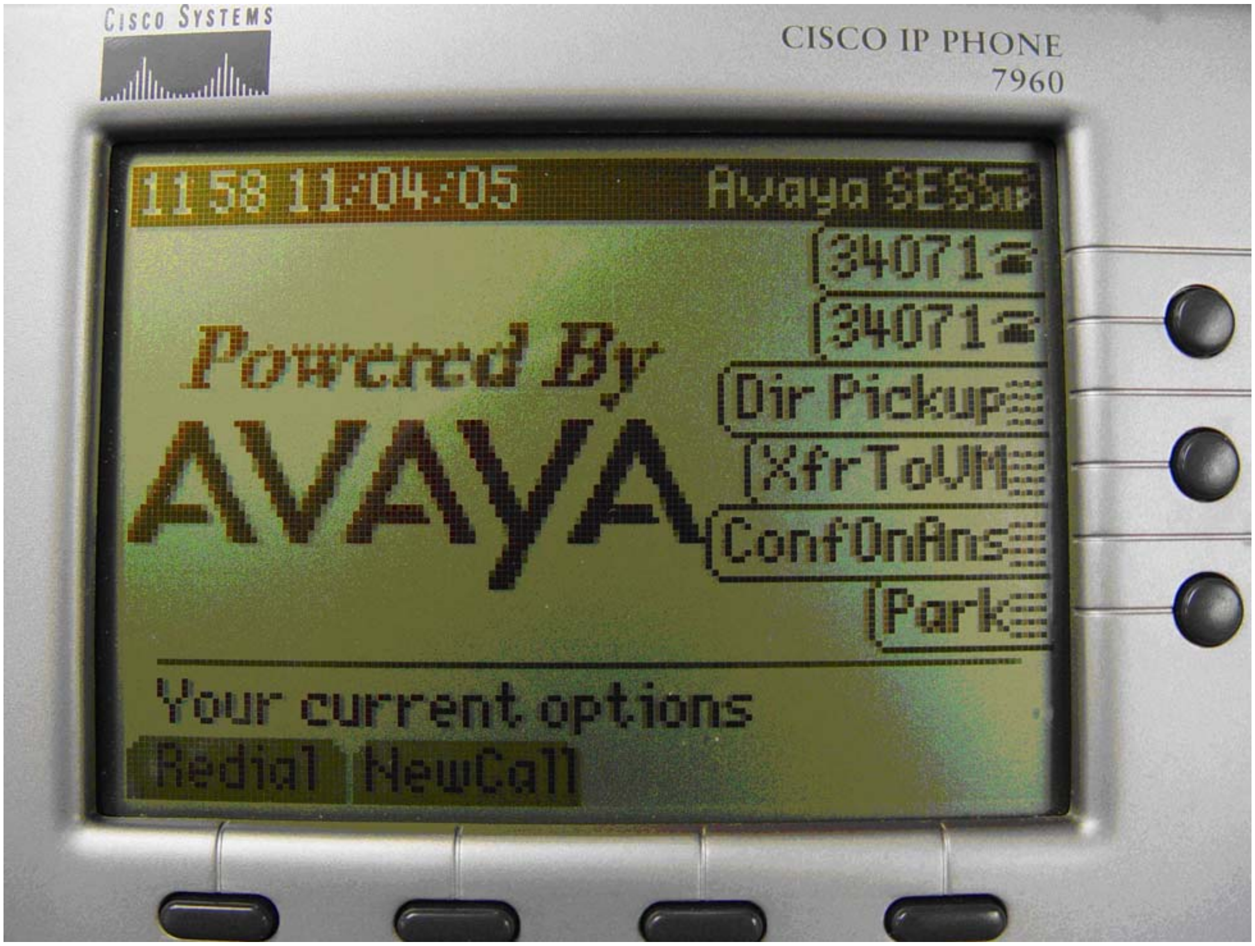
To access any of the OPS features, dial the corresponding FNE. For example, if the telephone has been defined in Avaya Communication Manager as part of a pickup group, then dial the Call Pickup FNE (in this case 70010) to answer a call to any member of that group. OPS features that involve an existing call (e.g., conference on answer) will require putting that call on hold, and placing a new call using the appropriate FNE. This procedure 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 most cases facilitating one-button feature access. 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 OPS speed dial buttons. The steps 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, access the *Call Preferences* menu. Using the **up/down** button, move the highlighted selection to **Speed Dial Lines**, and press the **Select** soft key. 

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

Steps	Description
3.	Enter the **Name** and **Number** for the speed dial button. For OPS features, **Number** should be one of the FNEs. Press the **Accept** soft key when finished. 

Steps	Description
4.	Repeat steps 3-4 for each button desired, selecting most frequently used OPS features for assignment to the buttons. An example button arrangement is shown below. When finished, press the **settings** button to exit from phone configuration mode. 

7. Verification Steps

All features shown in **Table 3** and Section 5.5 were tested using the sample configuration. 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 located next to each defined line appearance does *not* have an “X” next to it, indicating that registration has occurred. If the “X” appears, check that the proxy server address is set to the correct domain name, the outbound proxy IP address and port number are correct, and that the Proxy Register parameter is set to

Yes. Verify that the line appearance shows the Avaya 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 supported features according to **Table 3** and feature deployment plans at the site.
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 the configuration.
4. Verify extended OPS 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 Avaya Communication Manager administration for the correct FNE, proper permissions under COS/COR, and the proper station button assignment to support the feature.
5. Call a telephone that currently has no voice messages, and leave a message. Verify that the message-waiting indicator illuminates on the called telephone. Press the **messages** button on that telephone and verify that the voice messaging system is called. Use the voice messaging menus to retrieve and delete the voice message, verifying that DTMF is interpreted correctly by the system, and that the message waiting indicator extinguishes.

8. Support

For technical support of Cisco products:
Internet: http://www.cisco.com/en/US/support/index.html
Email: tac@cisco.com
Telephone: 1-800-553-2447

For technical information on the 7900 series telephones visit:
http://www.cisco.com/en/US/products/hw/phones/ps379/tsd_products_support_series_home.html

9. Conclusion

These Application Notes have described the administration steps required to use Cisco 7940 and 7960 SIP telephones with Avaya SIP Enablement Services and Avaya 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. The Cisco version of Message Waiting Indicator (MWI) is now supported on Cisco SIP Telephones with the Avaya 3.1 SIP offer.

10. Additional References

[1] *Installing and Administering SIP Enablement Services R3.1*, Issue 1.4, Doc ID 03-600768, February, 2006, available at http://support.avaya.com.

[2] *Session Initiation Protocol Service Examples - draft-ietf-sipping-service-examples-06*, SIPPING Working Group, Internet-Draft, 2/15/2004, available at http://www.ietf.org/proceedings/04mar/I-D/draft-ietf-sipping-service-examples-06.txt.

[3] *Cisco SIP IP Phone Administrator Guide, Release 6.0, 6.1, 7.0, 7.1*, May 2004, Cisco Systems, Inc. , available at http://www.cisco.com.

[4] *Avaya Extension to Cellular and OPS Installation and Administration Guide*, Version 6.0 Issue 9, DocID 210-100-500, June 2005, available at http://support.avaya.com.
[5] *Converting a Cisco 7940/7960 CallManager Phone to a SIP Phone and the Reverse Process*, Cisco Systems, Inc., available at http://www.cisco.com.
[6] *SIP Support in Release 3.1 of Communication Manager Running on the S8300, S8400, S8500, S8500B, S8700, and S8710 Media Server*, Issue 1.4, Doc ID 555-245-206, February, 2006, available at http://support.avaya.com.
[7] *Administrator Guide for Avaya Communication Manager*, Issue 2, Doc ID 03-300509, February 2006, available at http://support.avaya.com.

Appendix A

Sample Default Configuration File for Cisco 7940/7960 (SIPDefault.cnf)

```
# SIP Default Generic Configuration File

# Image Version
image_version: P0S3-07-4-00

# Proxy Server
proxy1_address: "companyx.com"            ; 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             ; Set to 1 so phone registers to proxy

# 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: never

# 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        ; Avoids having to press “DIAL” after number

# 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: "S8300.companyx.com  ; Can set to Media Server NTP server
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: "home1.companyx.com" ; 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: "http://10.3.3.104/AvayaPhoneLogo.bmp"                  ; 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: 1        ; 0-Disabled (default), 1-Enabled

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

# URI for Messages button
messages_uri: "35000"         ; Set to voice mail hunt group

# Enable (1) or Disable (0) Call Forward soft key
local_cfwd_enable: 0
```

Appendix B

Sample Telephone-Specific Configuration File for Cisco 7940/7960 (SIP*MacAddress*.cnf)

```
# SIP Configuration Phone-Specific File

# Line 1 appearance
line1_name: 34071

# Line 1 Registration Authentication
line1_authname: "34071"

# Line 1 Registration Password
line1_password: "123456"

# Line 2 appearance
line2_name: 34071

# Line 2 Registration Authentication
line2_authname: "34071"

# 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: "Joe SIP"

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