Avaya Solution & Interoperability Test Lab

Configuring SIP IP Telephony Using Avaya SIP Enablement Services, Avaya Communication Manager, and Cisco Unified IP Phones 7906, 7911, 7941, 7945, 7961, 7962, 7970, and 7975 - Issue 1.1

Abstract

These Application Notes describe the configuration steps required to connect Cisco Unified IP Phones to a SIP infrastructure consisting of an Avaya SIP Enablement Services (SES) server and an Avaya S8300 Server with G700 Media Gateway running Avaya Communication Manager. Also described is how Avaya Communication Manager 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 Servers and Media Gateways running Avaya Communication Manager.

This issue of these Application Notes reflects interoperability testing performed using Releases 4 and 5 of Avaya SIP Enablement Services and Avaya Communication Manager. It also includes procedures for upgrading telephone firmware from SCCP to SIP.

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 and gateways available to customers. There will be sales opportunities involving customers who wish to purchase Avaya SIP solutions, but already own SIP telephones other than those offered by Avaya. Customers may be interested in replacing their existing telephony infrastructure (e.g., Cisco Unified Communications Manager) with Avaya servers, but wish to re-use the existing telephones. In addition, a number of Avaya Communication Manager features can be extended to these SIP telephones, providing enhanced calling features in advance of SIP protocol definitions and implementation by telephone manufacturers.

These Application Notes describe the configuration steps for using Cisco Unified IP Phones 7906, 7911, 7941, 7945, 7961, 7962, 7970, and 7975 with Avaya SIP Enablement Services and Avaya Communication Manager. 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 is applicable to other Linux-based Avaya Servers and Media Gateways running Avaya Communication Manager.

This issue of these Application Notes reflects interoperability testing performed using Releases 4 and 5 of Avaya SIP Enablement Services and Avaya Communication Manager. It also includes procedures for upgrading telephone firmware from SCCP to SIP.

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 Unified IP Phone is configured to register to one of two SIP Enablement Services home servers and is administered as a station on an Avaya S8300 Server with G700 Media Gateway.[1] The Avaya IA770 INTUITY™ AUDIX® Messaging Application resides on the Avaya S8300 Server and is used to support voice messaging. An announcement is administered for Music on Hold. The PC supports a TFTP server as well as a web browser for administration of the Avaya servers.

The Cisco Unified IP Phone models listed above vary in the number of line appearances supported as well as the size, resolution, color, and touch sensitivity of the display. The configuration steps described in these Application Notes apply to all listed models. **Table 1** profiles the network management capabilities of the phones.

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

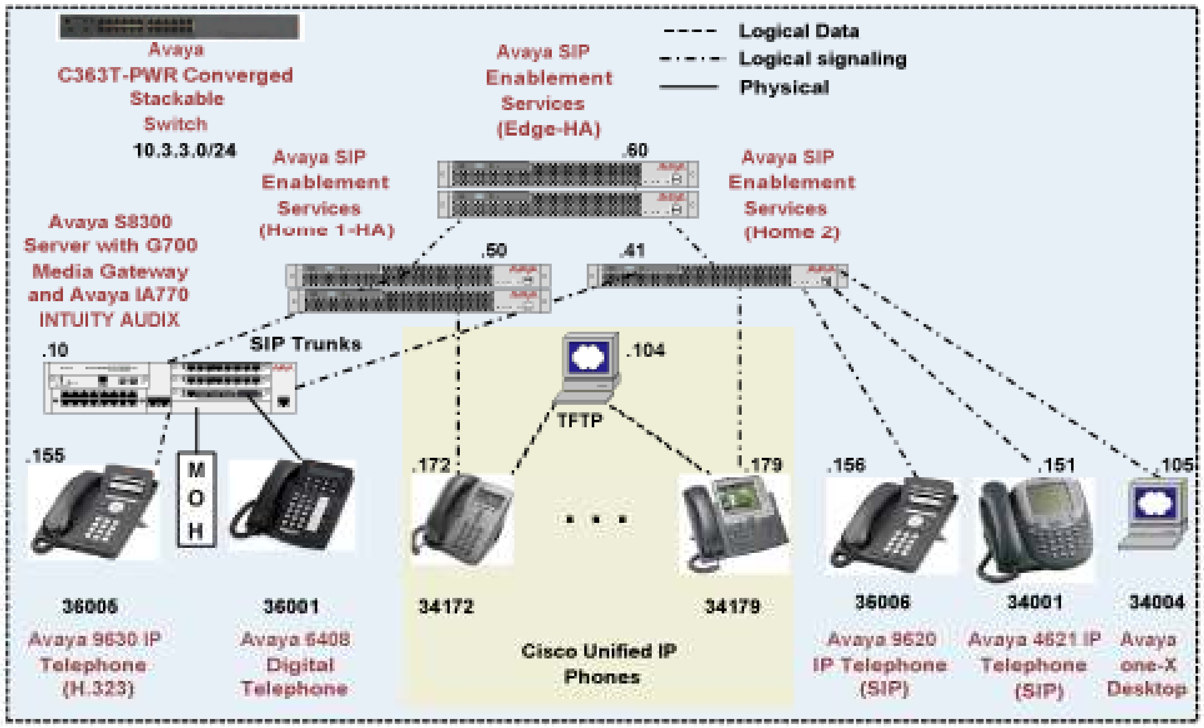


Figure 1: Avaya SIP Test Configuration with Cisco Unified IP Phones (HA = High Availability).

Administration mechanisms	Configuration files, Telnet, Read-only via Web
Administration levels	Administrator
File transfer server	TFTP
Error logs	Stored and viewed at phone or via Web
802.3af Power over Ethernet Support	Yes
SNMP support	No

Table 1: Network Management Capabilities of Cisco Unified IP Phones

2. Equipment and Software Validated

The following equipment and software were used in the configuration.

Equipment	Software
Avaya SIP Enablement Services (SES) Server	4.0 (Load 33.6), Service Pack 1
	5.0 (Load 825.31) Service Pack 825.4-SP2d
Avaya S8300 Server with G700 Media Gateway	Avaya Communication Manager 4.0.1 (Load 731.2) Update 14878
	Avaya Communication Manager 5.0 (Load 825.4), Update 15175
Avaya C363T-PWR Converged Stackable Switch	4.3.10
Avaya 6408 Digital Telephone	-
Avaya 4621 IP Telephone (SIP)	2.2.2
Avaya 9620 IP Telephone (SIP)	2.0.1.34 (5)
Avaya 9630 IP Telephone (H.323)	S1.5
Avaya one-X Desktop (SIP Softphone)	2.1 (Build 78)
Cisco Unified IP Phones 7906, 7911, 7941, 7945, 7961, 7962, 7970, 7975	8-3-2SR1
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 Unified IP Phones. 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. 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 [1]. This provides a useful framework to describe product capabilities and compare features supported by various equipment vendors. Avaya Communication Manager 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 Avaya Communication Manager.

Some Avaya Communication Manager features shown in **Table 3** can be invoked by dialing a Feature Name Extension (FNE). Or, a speed dial button on the telephone can be programmed to an FNE. Avaya Communication Manager automatically handles many other standard features 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 for Avaya Communication Manager can be found in References [2-4].

FEATURE	Supported		COMMENTS
	Locally at the Phone	With Avaya SIP Offer	
Basic Calling features			
Extension to extension call	YES	YES	See Section 3.2.5
Basic call to non-SIP phones	NO	YES	
Intercept tones/displays	YES	YES	Reorder (for announcements, see Section 5.2)
Call Waiting	YES	YES	
Do Not Disturb	YES	YES	See Section 3.2.1
Auto-answer (intercom)	YES	YES	
Speed Dial buttons	YES	YES	
Compressed codecs	YES	YES	G.729, G.729A
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	See Section 3.2.5
Attended Transfer	YES	YES	See Section 3.2.5
Call Forward Unconditional	YES	YES	Local admin or FNE (Sections 3.2.3, 3.2.5, 5.5)
Call Forward Busy	NO	YES	Via FNE (Sections 3.2.5, 5.5)
Call Forward No Answer	NO	YES	Via FNE (Sections 3.2.5, 5.5)
3-way conference - 3rd party added	YES	YES	
Find-Me	NO	YES	Via coverage paths (Section 5.8)
Incoming Call Screening	NO	YES	Via Class Of Restriction (Section 5.7)
Outgoing Call Screening	NO	YES	Via Class Of Restriction (Section 5.7)
Call Park/Unpark	NO	YES	Via FNE (Section 5.5)
Call Pickup	NO	YES	Via FNE (Section 5.5)
Automatic Redial	NO	YES	Via FNE (Section 5.5)
OPS - Selected Additional Station-Side Features			
Conference on Answer	NO	YES	
Extended Group Call Pickup	NO	YES	Via FNE (Section 5.5)
Directed Call Pick-Up	NO	YES	Via FNE (Section 5.5)
Drop Last Added Party	NO	YES	Via FNE (Section 5.5)
Last Number Dialed	YES	YES	Local soft key or FNE (Section 5.5)
Malicious Call Trace	NO	YES	Via FNE (Section 5.5)
Malicious Call Trace Cancel	NO	YES	Via FNE (Section 5.5)
Priority Call	NO	YES	Via FNE (Sections 3.2.4, 5.5, 5.6)
Send All Calls	NO	YES	Via FNE (Section 5.5)
Send All Calls Cancel	NO	YES	Via FNE (Section 5.5)
Transfer to Voice Mail	NO	YES	Via FNE (Section 5.5)
Whisper Page	NO	YES	Via FNE (Section 5.5)

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. Do Not Disturb

When Do Not Disturb is activated, the call is presented to and displayed by the phone, but the ringer is not activated. The call can be answered. If it is not answered, the call follows the coverage path configured for the extension in Avaya Communication Manager.

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 Unified IP Phones do not support this standard, but support an unsolicited Notify method for MWI. Avaya Communication Manager and SIP Enablement Services support both methods.

3.2.3. Call Forward Unconditional

It is recommended that this feature be administered as an Avaya Communication Manager FNE rather than using the local call forward of the telephone. The user of local call forward will not benefit from any of the call coverage features available in Avaya Communication Manager, including coverage to voice messaging.

3.2.4. Priority Call

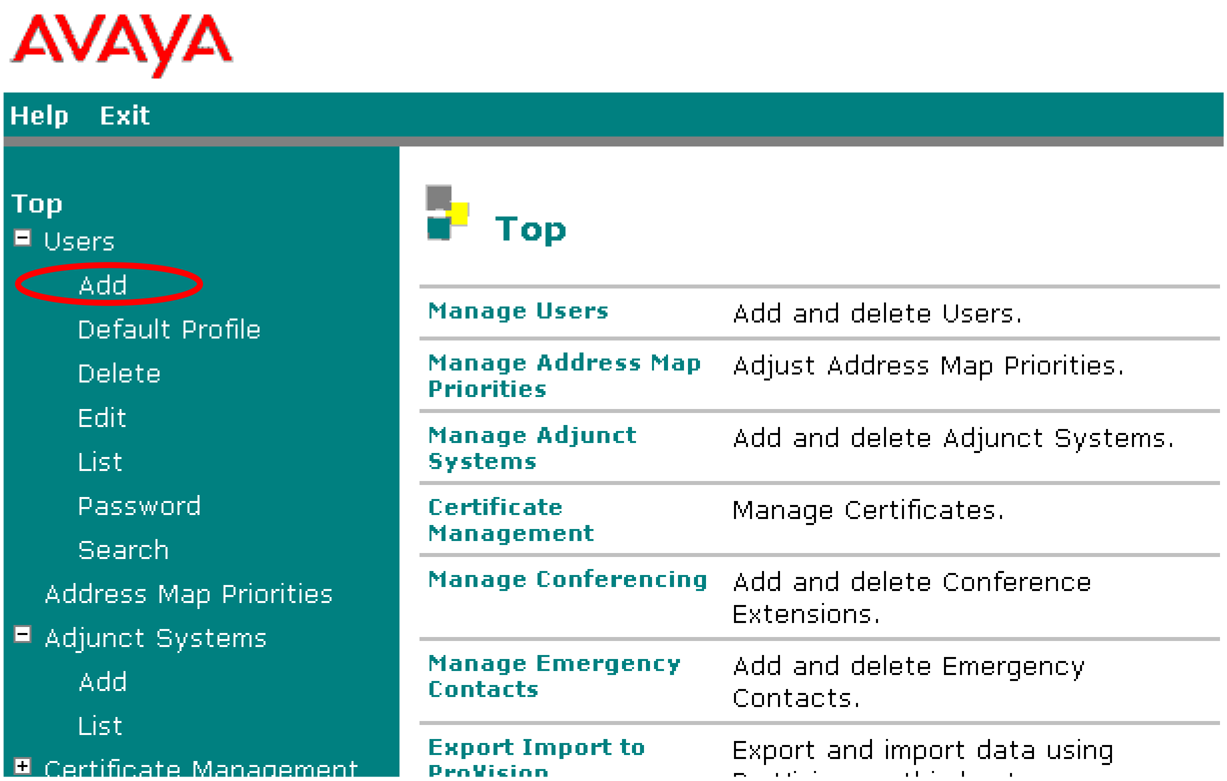
The telephone may originate priority calls based on the class of service administered for it (see Section 5.6) or if the user dials the appropriate FNE. Note however, that it will not indicate a received priority call. Avaya 4600 and 9600 Series IP Telephones (SIP) will properly indicate them via distinctive ringing and calling party display.

3.2.5. Station to Station Calling

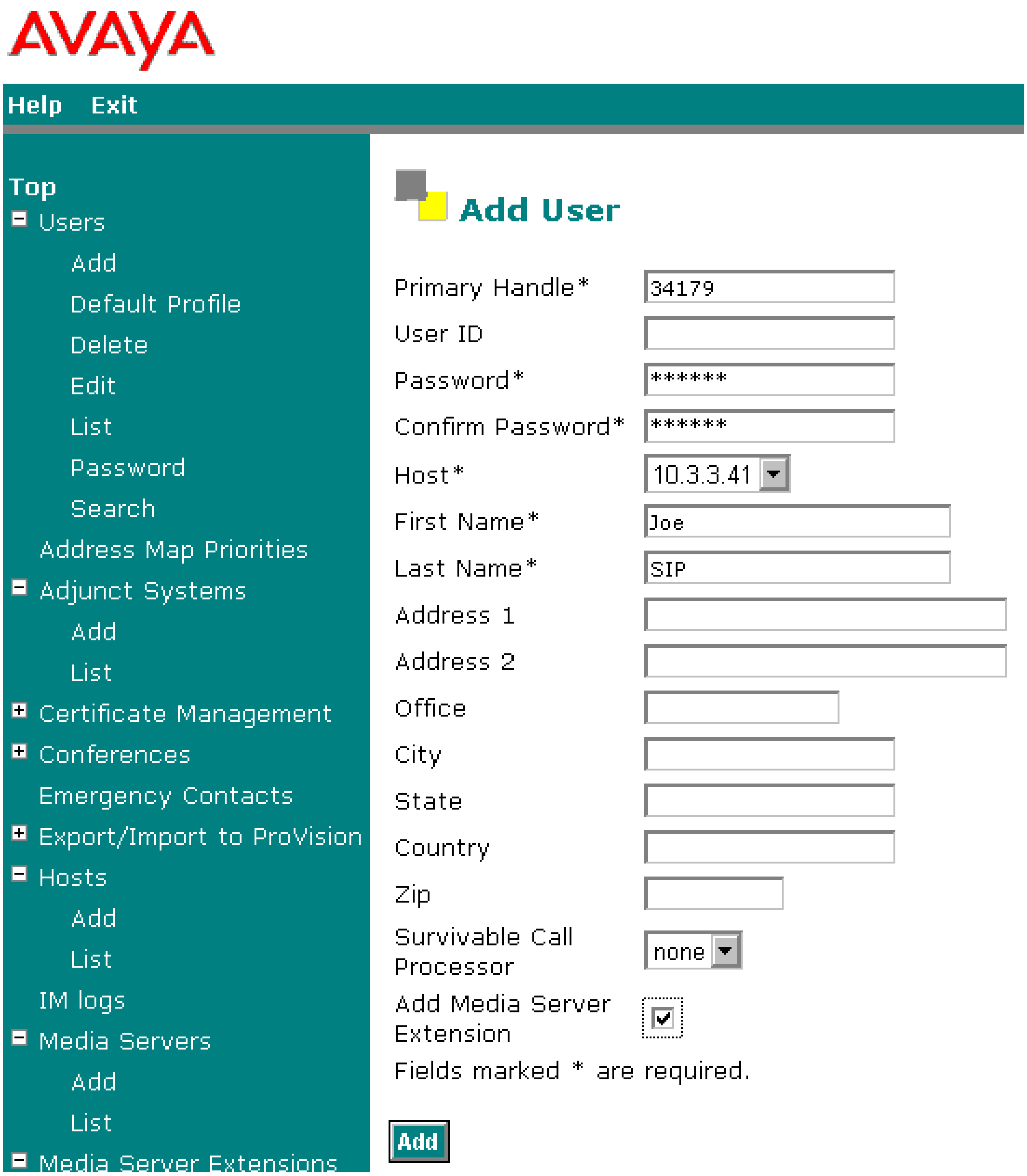
Note that the calling Cisco telephone display will not show the called/connected party name, but only the dialed number. Similarly, a transferred party's Cisco telephone display will not be updated to the transfer-to party's name/number.

4. Administer SIP Enablement Services

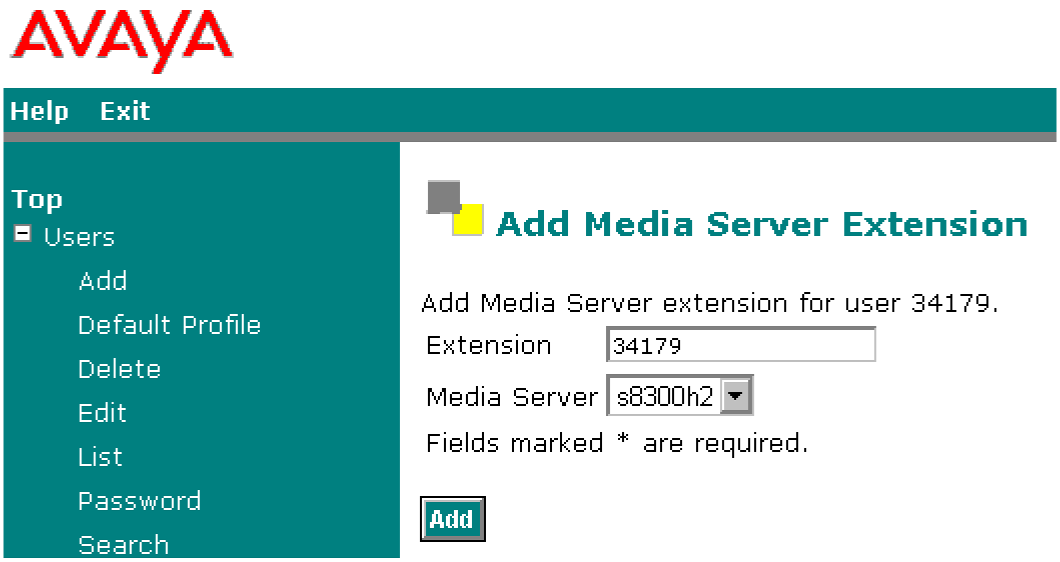
The following steps describe configuration of Avaya SIP Enablement Services for use with Cisco Unified IP Phones. Other standard administration functions are covered in Reference [5]. The steps apply to both Releases 4 and 5 unless otherwise noted. The screens shown are for Release 5 unless otherwise noted.

<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>1.</td><td>Log into the SIP Enablement Services administration web interface using the appropriate credentials. Expand the Users heading on the left side of the page and click on Add.


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

<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>2.</td><td>The Add User page will be displayed. Fill in the required fields (indicated by *). Enter the extension number or the user’s handle in the Primary Handle field. The Host field should be set to the Avaya SIP Enablement Services Home or Home/Edge server to which the telephone will register. In this configuration, the telephone will register to Home 2. Check the Add Media Server Extension checkbox. Click on Add, and then Continue on the subsequent confirmation page.


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

Steps	Description
3.	The *Add Media Server Extension* page will be displayed. Enter the user's telephone extension in the **Extension** field. Since the user is being added to Home 2, the **Media Server** corresponding to the SIP trunk between the Avaya S8300 Server and Home 2 is selected automatically. Click on **Add**, and then **Continue** on the subsequent confirmation page. 
4.	Repeat **Steps 1-3** for each user to be added to the system.
5.	*If using Avaya SES Release 4* - To apply the administration in the above steps, click on **Update** at the top of the left side of the page, as shown below. 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. This step is not required for Release 5. 

5. Configure Avaya Communication Manager

This section highlights the important commands for defining the telephone as a SIP station in Avaya Communication Manager, and administering support for the 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 SIP administration, see References [2-3]. The steps apply to both Release 4 and 5 unless otherwise noted. The screens shown are for Release 5. Log in to the System Access Terminal (SAT) interface with the appropriate permissions.

5.1. Verify SIP Telephone 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 addition of the SIP telephones.

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

     G3 Version: V15                           Software Package: Standard
       Location: 1                          RFA System ID (SID): 1
       Platform: 7                          RFA Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 900   372
                                      Maximum Stations: 450   72
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 10    3
                    Maximum Off-PBX Telephones -   OPS: 200   55
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
```

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.

```
change system-parameters features                              Page   4 of  17
                        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
 Maximum Number of Digits for Directed Group Call Pickup:4
               Call Pickup on Intercom Calls? y        Call Pickup Alerting? n
 Temporary Bridged Appearance on Call Pickup? y        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
               Controlled Toll Restriction Replaces: none
```

```
change system-parameters features                              Page  17 of  17
                        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

6400/8400/2420J 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

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 used in the system. This includes all telephone extensions, Feature Name Extensions (FNEs), and Feature Access Codes (FACs). To define the FNEs for the features listed in **Table 3**, a Feature Access Code (FAC) must also be specified for the corresponding feature[2]. In the sample configuration, telephone extensions are five digits long and begin with 2 , 3, or 5, FNEs are five digits beginning with 7, and the FACs have formats as indicated with **Call Type** "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    dac
        2       5    ext
        3       5    ext
        5       5    ext
        6       3    fac
        7       5    ext
        8       1    fac
        9       1    fac
        *       2    fac
        #       2    fac
```

5.4. Feature Access Codes (FACs)

Use **change feature-access-codes** to define the access codes for the features listed in **Table 3**.

```
change feature-access-codes                                     Page   1 of   8
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: 601
         Abbreviated Dialing List2 Access Code: 602
         Abbreviated Dialing List3 Access Code: 603
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code: 606
                       Answer Back Access Code: 605
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9     Access Code 2:
                 Automatic Callback Activation: *5     Deactivation: #5
Call Forwarding Activation Busy/DA: *2     All: 612    Deactivation: #2
   Call Forwarding Enhanced Status:         Act:       Deactivation:
                         Call Park Access Code: 604
                       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:
```

[2] 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 Reference [3] for more details.

```
change feature-access-codes                                     Page   2 of   8
                               FEATURE ACCESS CODE (FAC)
                  Contact Closure  Pulse Code:
                 Data Origination Access Code:
                     Data Privacy Access Code:
             Directed Call Pickup Access Code: 654
       Directed Group Call Pickup  Access Code:
     Emergency Access to Attendant Access Code:
       EC500 Self-Administration Access Codes:
                    Enhanced EC500 Activation: 660     Deactivation: 661
           Enterprise Mobility User Activation:         Deactivation:
  Extended Call Fwd Activate Busy D/A      All:         Deactivation:
       Extended Group Call Pickup Access Code: 621
              Facility Test Calls Access Code:
                           Flash Access Code: 678
            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
```

```
change feature-access-codes                                     Page   3 of   8
                               FEATURE ACCESS CODE (FAC)
            Leave Word Calling Send A Message:
          Leave Word Calling Cancel A Message:
   Limit Number of Concurrent Calls Activation:         Deactivation:
              Malicious Call Trace Activation: 613     Deactivation: 614
        Meet-me Conference Access Code Change:

 PASTE (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:         Dissociate Code:
       Per Call CPN Blocking Code Access Code: 615
     Per Call CPN Unblocking Code Access Code: 616

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

      Refresh Terminal Parameters Access Code: 694
             Remote Send All Calls Activation:          Deactivation:
              Self Station Display Activation:
                    Send All Calls Activation: *3       Deactivation: #3
        Station Firmware Download Access Code:
```

```
change feature-access-codes                                     Page   4 of   8
                               FEATURE ACCESS CODE (FAC)
                      Station Lock Activation:          Deactivation:
     Station Security Code Change Access Code: 699
             Station User Admin of FBI Assign:                     Remove:
  Station User Button Ring Control Access Code:
            Terminal Dial-Up Test Access Code: 695
Terminal Translation Initialization Merge Code:           Separation Code:
            Transfer to Voice Mail Access Code: #9
          Trunk Answer Any Station Access Code:
             User Control Restrict Activation: 691     Deactivation: 692
  Voice Coverage Message Retrieval Access Code:
 Voice Principal Message Retrieval Access Code:
          Whisper Page Activation Access Code: 620
```

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 SIP telephones 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 set 1          Page   1 of   2
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME
                     Set Name:

     Active Appearance Select: 70024
          Automatic Call Back: 70003
   Automatic Call-Back Cancel: 70004
             Call Forward All: 70005
  Call Forward Busy/No Answer: 70006
          Call Forward Cancel: 70007
                    Call Park: 70008
        Call Park Answer Back: 70009
                 Call Pick-Up: 70010
         Calling Number Block: 70012
       Calling Number Unblock: 70013
         Conference on Answer: 70011
        Directed Call Pick-Up: 70014
        Drop Last Added Party: 70015
    Exclusion (Toggle On/Off):
   Extended Group Call Pickup: 70025
       Held Appearance Select:
```

```
change off-pbx-telephone feature-name-extensions set 1          Page   2 of   2
     EXTENSIONS TO CALL WHICH ACTIVATE FEATURES BY NAME

       Idle Appearance Select:
           Last Number Dialed: 70019
          Malicious Call Trace: 70029
  Malicious Call Trace Cancel: 70021
          Off-Pbx Call Enable:
         Off-Pbx Call Disable:
                Priority Call: 70000
               Send All Calls: 70001
        Send All Calls Cancel: 70002
          Transfer On Hang-Up:
       Transfer to Voice Mail: 70023
      Whisper Page Activation: 70026
```

5.6. Specify Class of Service (COS)

Use the **change class-of-service** command to set the appropriate service permissions to support the corresponding features (shown in bold). For the example, COS 1 was used. On Page 2, set the value of **VIP Caller** 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      n  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  n  n  n  n  n  n  n  n  n  n  n  n  n  n

 Masking CPN/Name Override      n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Call Forwarding Enhanced       y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
 Priority Ip Video              n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Ad-hoc Video Conferencing      n  n  n  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 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  23
                              CLASS OF RESTRICTION

               COR Number: 2
          COR Description: Stations

                      FRL: 0                                APLT? y
  Can Be Service Observed? y           Calling Party Restriction: none
Can Be A Service Observer? y            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   4 of  23
                              CLASS OF RESTRICTION

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

  0? y      15? y      30? y      44? y      58? y      72? y      86? y
  1? y      16? y      31? y      45? y      59? y      73? y      87? y
  2? y      17? y      32? y      46? y      60? y      74? y      88? y
  3? n      18? y      33? y      47? y      61? y      75? y      89? y
  4? y      19? y      34? y      48? y      62? y      76? y      90? y
  5? y      20? y      35? y      49? y      63? y      77? y      91? y
  6? y      21? y      36? y      50? y      64? y      78? y      92? y
  7? y      22? y      37? y      51? y      65? y      79? y      93? y
  8? y      23? y      38? y      52? y      66? y      80? y      94? y
  9? y      24? y      39? y      53? y      67? y      81? y      95? y
 10? y      25? y      40? y      54? y      68? y      82? y      96? y
 11? y      26? y      41? y      55? y      69? y      83? y      97? y
 12? y      27? y      42? y      56? y      70? y      84? y      98? y
 13? y      28? y      43? y      57? y      71? y      85? y      99? y
 14? y      29? 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: 1        Linkage   1    1

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: 3  Point2:
  Point3:                       Point4:
  Point5:                       Point6:
```

5.9. Add Stations

Use the **add station** command to add a station for each telephone to be supported. Assign the same extension as the media server extension administered in SIP Enablement Services. Use "4620" for the **Station Type** and be sure to include the **Coverage Path** for voice messaging or other hunt group if applicable. Use the **COS** and **COR** values administered in the previous sections. The **Name** field is optional and is shown on the display of the calling party's phone when receiving calls from this station. Use defaults for the other fields on Page 1.

```
change station 34179                                           Page   1 of   5
                                     STATION

Extension: 34179                         Lock Messages? n             BCC: 0
     Type: 4620                          Security Code: 123456         TN: 1
     Port: S00081                      Coverage Path 1: 1             COR: 2
     Name: Cisco 7975                  Coverage Path 2:               COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 34179
            Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english               Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal             Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? n




                                             Customizable Labels? y
```

On Page 2, note the following:

- If this 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 telephone 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 telephone. 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 34179                                              Page   2 of   5
                                     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 Idle Line Preference? n
   Bridged Call Alerting? y                     Restrict Last Appearance? n
  Active Station Ringing: single
                                                         EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
    MWI Served User Type: qsig-mwi              Display Client Redirection? n
                                               Select Last Used Appearance? n
                                                 Coverage After Forwarding? s
```

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, many calls can be active per line appearance, so the number of "call-appr" buttons should account for that.

```
add station 34179                                              Page   4 of   5
                                    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: brdg-appr  Btn:1  Ext:34176
 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 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 34179. Since call waiting is enabled at the telephone (see Section 6.3.2), additional call appearances have been provisioned. A bridged appearance was defined on station 34176 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 (34179) 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 34179                 Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station          Application Dial   CC   Phone Number      Trunk       Config
 Extension                    Prefix                        Selection   Set
 34179               OPS           -      34179              12         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 34179                 Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station         Call        Mapping       Calls        Bridged        Location
 Extension       Limit       Mode          Allowed      Calls
 34179            4        both            all          both
```

6. Configure the Cisco Unified IP Phone

The following sections describe the configuration steps for the Cisco Unified IP Phones. The particular phone used as an example is the 7975, since it can be used to illustrate the broadest feature set.

6.1. Load SIP Firmware

Cisco Unified IP phones can support either the SCCP or SIP protocols, and they require different application firmware for each. This section describes the steps required to convert a telephone from SCCP to SIP, or to upgrade the SIP firmware to a specific release. It is assumed that a PC running TFTP is available on the network.

Steps	Description
1.	Obtain the zip file for the firmware release of the particular telephone model, which for the sample configuration is `cmterm-7975-sip.8-3-2SR1.zip`. Extract the files into the TFTP directory on the PC that will be accessed by the telephones during the boot process. Among the extracted files is a file having the ".loads" extension. In the sample configuration, this file is `SIP75.8-3-2SR1S.loads`. This file name (without extension) will be used in the next step.

Steps	Description
2.	Create a file whose name is of the form `SIP`<*MAC-address*>`.cnf.xml`, where <*MAC-address*> is the MAC address of the phone. The sample copy shown in Appendix A can be used as a starting point. Edit the file with a text or eXtensible Markup Language (XML) editor, and change the parameter line shown below to include the name of the file in **Step 1**, without the ".loads" extension: <loadInformation>**SIP75.8-3-2SR1S**</loadInformation>
3.	Connect the telephone to power and the network. Press the **settings** button, select **Network Configuration**, enable changes by pressing ****#** on the keypad, set **DHCP Enabled** to "no", and manually assign an IP address, network mask, and gateway. Set **TFTP server 1** to the IP address of the PC. Press the **Save** soft key, followed by the **Exit** soft key.
4.	The phone will download the configuration file and begin the upgrade process. When this process has completed, verify the firmware version by pressing the **settings** button followed by **63** on the keypad. The firmware versions displayed should match the ".loads" file name shown in **Step 1**. 

6.2. Configure Registration and Basic Dialing

Cisco Unified IP Phones must be configured using the XML configuration file created in **Step 2** of **Section 6.1**. The configuration settings take effect after the telephone is rebooted and it downloads the file. The following steps describe the parameter assignments required for basic registration and calling.

The basic parameter assignment syntax is:

<Parameter_Name>**Value**</Parameter_Name>

Since XML is hierarchical in format, the position of the parameter lines within the various *sections* in the file is important. A *section* is delineated as follows:

```
<sectionName>
  .
  .
  .
</sectionName >
```

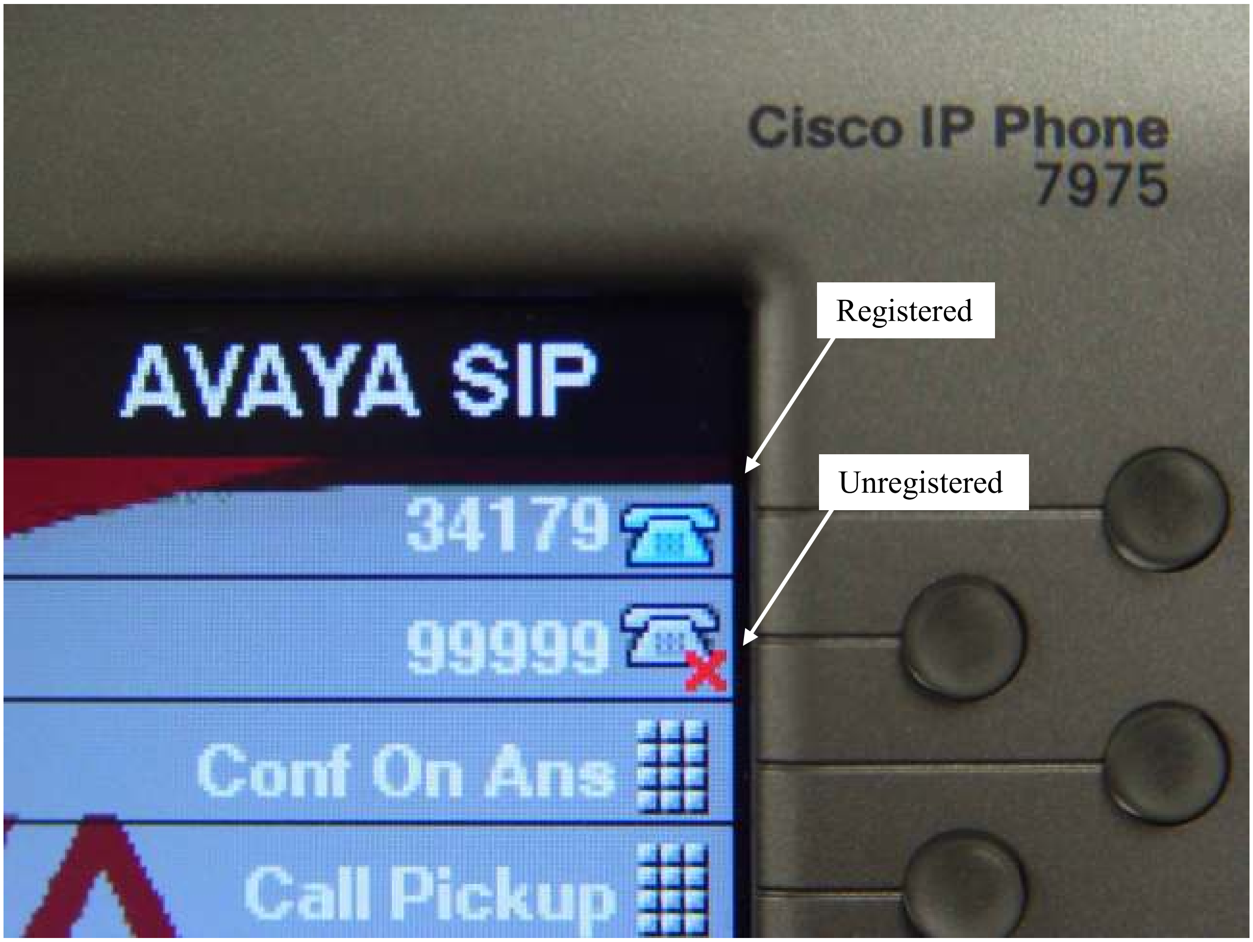
See Appendix A for a sample file showing key parameters with explanatory comments. Do not move the parameter lines from the locations shown in Appendix A.

Steps	Description
1.	Set the Network Time Protocol (NTP) server parameters so the phone will display the current date and time. In the *ntp* section set: <name>**10.3.3.10**</name> <ntpMode>**Unicast**</ntpMode> In the sample configuration, this is the Avaya S8300 Server.
2.	The SIP proxy parameters should be set to the appropriate Avaya SES home server (Home 2 in this example) in the following sections: *callManagerGroup* section: <processNodeName>**10.3.3.41**</processNodeName> *srstInfo* section: <ipAddr1>**10.3.3.41**</ipAddr1> <sipIpAddr1>**10.3.3.41**</sipIpAddr1> Verify or change the following parameter in the *sipProxies* section: <registerWithProxy>**true**</registerWithProxy>

Steps	Description
3.	Each *line* section corresponds to a physical button on the phone that can be used as a call appearance or as a speed dial. The model 7975 has 8 such buttons. In this example, the top two buttons will be used as call appearances. The *line* section for the top button (button 1) should be configured as shown below. Set **featureID** to "9" to indicate the call appearance function, **featureLabel**, **Name**, and **authName** to the phone extension administered in Avaya SES and Avaya Communication Manager, **proxy** to the IP address of Avaya SES, **port** to "5060", **displayName** to the phone user's name, and **authPassword** to the user password administered in Avaya SES. Set **messagesNumber** to the hunt group extension of the voice messaging system, so that the **messages** button can be pressed to access that system. If call waiting is desired, set **callWaiting** to "3"; a value of "2" disables it. All other parameters in the *line* section can be left as shown in Appendix A. <line button="**1**"> <featureID>**9**</featureID> <featureLabel>**34179**</featureLabel> <proxy>**10.3.3.41**</proxy> <port>**5060**</port> <name>**34179**</name> <displayName>**Cisco 7975**</displayName> . . . <callWaiting>**3**</callWaiting> <authName>**34179**</authName> <authPassword>**123456**</authPassword> . . . <messagesNumber>**35000**</messagesNumber> . . . </line>
4.	Repeat **Step 3** for each button to be used as a call appearance, setting the **button** parameter to the corresponding button number. The remaining parameters should be set to the same values as shown for button 1.
5.	The phone will use a local dial plan configuration 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 a call. The file is downloaded from the TFTP server at boot time. Near the end of the *sipProfile* section, set the **dialTemplate** parameter to the file name to be used: <dialTemplate>**dialplan.xml**</dialTemplate> The sample configuration used the file `dialplan.xml`.

Steps	Description

6.	Create the dial plan file (`dialplan.xml`). The sample configuration used the following dial plan file:
	
	`<DIALTEMPLATE>`
	`      <TEMPLATE MATCH="2...." Timeout="0" User="Phone" Rewrite="%s"/>`
	`      <TEMPLATE MATCH="3...." Timeout="0" User="Phone" Rewrite="%s"/>`
	`      <TEMPLATE MATCH="5...." Timeout="0" User="Phone" Rewrite="%s"/>`
	`      <TEMPLATE MATCH="7...." Timeout="0" User="Phone" Rewrite="%s"/>`
	`</DIALTEMPLATE>`
	
	The periods in the **MATCH** string stand for any digit. The templates cover 5 digit extensions beginning with 2, 3, and 5. The third covers cases where the user may dial an FNE rather than press a programmed speed dial button. 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 Reference [6].

Steps	Description
7.	Reboot the phone. If TFTP support has been properly configured, the phone will download the configuration files and register with Avaya SES. Registration can be verified by the absence of an "X" near the phone icon for the line appearance, as shown below. 
8.	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 6.3. Section 6.4 describes configuring the telephone to access additional Avaya Communication Manager features.

6.3. Configure Local Calling Features

The following sections describe how the telephone user can administer local telephone features that are compatible with Avaya Communication Manager and SIP Enablement Services.

6.3.1. Do Not Disturb (DND)

Steps	Description
1.	To enable display of the DND soft key, set the following parameter in the phone configuration file under the *sipCallFeatures* section. Reboot the phone so that this setting will take effect. <dndControl>**0**</dndControl>
2.	DND is enabled by pressing the **DND** soft key. The display shows that DND is active. When DND is activated, an incoming call is displayed on the phone, but the ringer is not activated. The call can be answered. If it not answered, it will be routed to the coverage path specified for this telephone in Avaya Communication Manager. This is typically a voice messaging system. To deactivate DND, press the **DND** soft key again. 

6.3.2. Call Waiting

Call waiting allows a second call to be answered on a call appearance while another call is in progress. Enable call waiting by setting the following parameter in the phone configuration file under the *line* section corresponding to a line button used as a call appearance:

```
<callWaiting>3</callWaiting>
```

If call waiting is not desired, then a value of "2" should be used. In this case the second call will immediately go to coverage. Reboot the phone so that this setting will take effect.

6.3.3. Auto-Answer

A telephone in auto-answer mode will automatically answer an incoming call, activating the speaker on the telephone. Since the Cisco Unified IP Phone models 7906 and 7911 do not have an integrated microphone capability, they would only be useful in paging applications.

In the phone configuration file, set the following parameter in the *autoAnswer* section under each *line* section corresponding to a call appearance button for which auto-answer is desired:

```
<autoAnswer>
  <autoAnswerEnabled>1</autoAnswerEnabled>
</autoAnswer>
```

The feature is enabled after the next phone reboot and remains enabled until this value is changed to "2" and the phone is rebooted once again.

6.4. Configure Speed Dial Buttons and the Avaya Extended Feature Set

Additional Avaya Communication Manager features can be accessed by dialing 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. 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 many cases facilitating one-button feature access.

The following steps describe how to configure Cisco Unified IP Phones with speed dial buttons. This technique is most useful with telephone models that have many line appearance buttons, such as the 7961, 7962, 7970, and 7975. Since the 7941 and 7945 have two line appearances, only one speed dial button is available in these cases. Although the 7906 and 7911 do not have call appearance buttons, the telephone menus can still be used to access up to 4 speed dial buttons.

Steps	Description
1.	In the phone configuration file, set the following parameter in the *sipCallFeatures* section: <disableLocalSpeedDialConfig>**true**</disableLocalSpeedDialConfig> This prevents the telephone user from changing the values of the speed dials, since rebooting the phone will over-write the user defined values with those defined in the phone configuration file.
2.	For each physical button on the telephone that is to be assigned a speed dial number, define a *line* section in the phone configuration file. Set the **button** parameter to the button number, **featureID** to “2”, **featureLabel** to the name of the feature, and **speedDialNumber** to the corresponding FNE. The definitions shown below assign FNEs for Avaya Communication Manager features to the bottom 6 buttons on a Cisco Unified IP Phone Model 7975. See Appendix A for where these definitions are located in the file. <line button="**3**"> <featureID>**2**</featureID> <featureLabel>**Conf On Ans**</featureLabel> <speedDialNumber>**70011**</speedDialNumber> </line> <line button="**4**"> <featureID>**2**</featureID> <featureLabel>**Call Pickup**</featureLabel> <speedDialNumber>**70010**</speedDialNumber> </line> <line button="**5**"> <featureID>**2**</featureID> <featureLabel>**Call Park**</featureLabel> <speedDialNumber>**70008**</speedDialNumber> </line> <line button="**6**"> <featureID>**2**</featureID> <featureLabel>**Call Unpark**</featureLabel> <speedDialNumber>**70009**</speedDialNumber> </line> <line button="**7**"> <featureID>**2**</featureID> <featureLabel>**SAC**</featureLabel> <speedDialNumber>**70001**</speedDialNumber> </line> <line button="**8**"> <featureID>**2**</featureID> <featureLabel>**SAC Cancel**</featureLabel> <speedDialNumber>**70002**</speedDialNumber> </line>

Steps	Description
3.	Access an Avaya Communication Manager feature via speed dial button on telephone models 7941, 7942, 7961, 7962 7970, and 7975 by pressing the appropriate line button. If the feature applies to an active call, put the call on hold first, and then press the feature button. 

Steps	Description
4.	Access an Avaya Communication Manager feature via speed dial button on telephone models 7906 and 7911 by pressing the **configuration settings** button (lighted in the figure below), followed by **24**, followed by the speed dial number (e.g., **3** for the Call Park feature configured in **Step 2**). 

6.5. Configure Background Image

The background image of Cisco Unified IP Phones can be customized using a bitmap image created by programs such as Microsoft Paint. The image size in pixels supported depends on the telephone model. **Table 5** shows the image sizes and color capabilities of the models covered by these Application Notes. Also shown is the directory name used by the phone to download background images. The following steps summarize the configuration required to use the images. For more details on setting up background images see the Cisco Administration Guide for the corresponding telephone model.

Cisco Unified IP Phone Model	Full Scale Image Size	Thumbnail Image Size	Image Directory Name
7906, 7911	95x34x1 (b & w)	23x8x1	95x34x1
7941, 7961	320x196x4 (gray scale)	80x49x4	320x196x4
7945	320x212x16 (color)	80x53x16	320x212x16
7962	320x196x4 (gray scale)	80x49x4	320x196x4
7970	320x212x12 (color)	80x53x12	320x212x12
7975	320x216x16 (color)	80x54x16	320x216x16

Table 5: Background Display Images Supported

Steps	Description
1.	Using a bitmap editor, create two "PNG" files containing the full scale and thumbnail versions of the desired background image. This pair of image files is required for each different phone model to be supported, as indicated in **Table 5**. For example, the thumbnail and full image files for the model 7975 could be named 7975logoTN.png and 7975logo.png, respectively.
2.	Under the TFTP directory supporting the telephones (the same directory containing the phone configuration file), create a directory named `Desktops`. Under that directory create the appropriate image directory with the name indicated in **Table 5**. Put the files created in **Step 1** under this new directory. Create a new configuration file list.xml containing the following lines: <CiscoIPPhoneImageList> <ImageItem Image="**TFTP:Desktops/320x216x16/7975logoTN.png**" URL="**TFTP:Desktops/320x216x16/7975logo.png**" /> </CiscoIPPhoneImageList> The above example shows the configuration file for the model 7975.
3.	In the phone configuration file, verify or set the following parameter in the *commonProfile* section: <backgroundImageAccess>**true**</backgroundImageAccess> Reboot the phone if this step required editing the file.

Steps	Description
4.	Press the **settings** button on the phone. Then press **12** and use the navigation buttons to highlight the thumbnail for the newly created background image. 
5.	Press **Select**, followed by **Save** and then **Exit**. The new background image will be displayed as shown in Section 6.4 **Step 3**.

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 telephone, use the **settings** button at the phone to verify that the parameters set in the phone configuration file have been loaded. Verify registration with Avaya SES by verifying that the phone icon located next to each defined line appearance does *not* have an "X" next to it. If the "X" appears, check that the proxy server and port number are set correctly, and that the **registerWithProxy** parameter is set to *true*. 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, check the phone configuration file.
4. Verify additional Avaya Communication Manager 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 Cisco 7900 Unified IP Phones 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 Unified IP Phones 7906, 7911, 7941, 7945, 7961, 7962, 7970, and 7975 with Avaya SIP Enablement Services and Avaya Communication Manager. Both basic and extended feature sets were covered. The extended set includes features not yet implemented in SIP telephones using the current IETF standards.

10. Additional References

[1] *Session Initiation Protocol Service Examples - draft-ietf-sipping-service-examples-14*, SIPPING Working Group, Internet-Draft, 7/16/2007, available at http://tools.ietf.org/wg/sipping/draft-ietf-sipping-service-examples/draft-ietf-sipping-service-examples-14.txt.

[2] *Avaya Extension to Cellular and OPS Installation and Administration Guide*, Version 6.0 Issue 9, Doc ID 210-100-500, June 2005, available at http://support.avaya.com.

[3] *SIP Support in Avaya Communication Manager Running on Avaya S83xx Servers*, Issue 8, Doc ID 555-245-206, January, 2008, available at http://support.avaya.com.

[4] *Administrator Guide for Avaya Communication Manager*, Issue 4, Doc ID 03-300509, January 2008, available at http://support.avaya.com.

[5] *Installing and Administering SIP Enablement Services*, Issue 5, Doc ID 03-600768, January, 2008, available at http://support.avaya.com.

[6] *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.

Appendix A

Sample Configuration File for Cisco Unified IP Phone 7975

The following shows the configuration file as it would look when edited with a text editor. The parameters in bold should be set as desired. The remaining parameters need not be changed.

```
<device>
<fullConfig>true</fullConfig>
<deviceProtocol>SIP</deviceProtocol>
<sshUserId>user</sshUserId>
<sshPassword>pass</sshPassword>
<devicePool uuid="{a755aa55-089c-2b47-9603-c7d51b9ca4b5}">
<dateTimeSetting uuid="{9ec4850a-7748-11d3-bdf0-00108302ead1}">
<dateTemplate>M/D/Y</dateTemplate>
<timeZone>Eastern Standard/Daylight Time</timeZone>
<ntps>
<ntp>
<name>10.3.3.10</name>                          NTP server (Avaya S8300 Server)
<ntpMode>Unicast</ntpMode>
</ntp>
</ntps>
</dateTimeSetting>
<callManagerGroup>
<members>
<member priority="0">
<callManager>
<name>ccm-beta-5-1</name>
<description>CallManager 5.0 Beta Pub - 5.0.1.032</description>
<ports>
<ethernetPhonePort>2000</ethernetPhonePort>
<sipPort>5060</sipPort>
<securedSipPort>5061</securedSipPort>
</ports>
<processNodeName>10.3.3.41</processNodeName>    Avaya SES
</callManager>
</member>
</members>
</callManagerGroup>
<srstInfo uuid="{cd241e11-4a58-4d3d-9661-f06c912a18a3}">
<srstOption>Disable</srstOption>
<ipAddr1>10.3.3.41</ipAddr1>                    Avaya SES
<port1>2000</port1>
<ipAddr2></ipAddr2>
<port2>2000</port2>
```

```
<ipAddr3></ipAddr3>
<port3>2000</port3>
<sipIpAddr1>10.3.3.41</sipIpAddr1>                    Avaya SES
<sipPort1>5060</sipPort1>
<sipIpAddr2></sipIpAddr2>
<sipPort2>5060</sipPort2>
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<sipPort3>5060</sipPort3>
<isSecure>false</isSecure>
</srstInfo>
<connectionMonitorDuration>120</connectionMonitorDuration>
</devicePool>
<sipProfile>
<sipProxies>
<backupProxy></backupProxy>
<backupProxyPort></backupProxyPort>
<emergencyProxy></emergencyProxy>
<emergencyProxyPort></emergencyProxyPort>
<outboundProxy></outboundProxy>
<outboundProxyPort></outboundProxyPort>
<registerWithProxy>true</registerWithProxy>
</sipProxies>
<sipCallFeatures>
<cnfJoinEnabled>true</cnfJoinEnabled>
<callForwardURI>x-cisco-serviceuri-cfwdall</callForwardURI>
<callPickupURI>x-cisco-serviceuri-pickup</callPickupURI>
<callPickupListURI>x-cisco-serviceuri-opickup</callPickupListURI>
<callPickupGroupURI>x-cisco-serviceuri-gpickup</callPickupGroupURI>
<meetMeServiceURI>x-cisco-serviceuri-meetme</meetMeServiceURI>
<abbreviatedDialURI>x-cisco-serviceuri-abbrdial</abbreviatedDialURI>
<rfc2543Hold>false</rfc2543Hold>
<callHoldRingback>2</callHoldRingback>
<localCfwdEnable>false</localCfwdEnable>              Disable local call forward
<semiAttendedTransfer>true</semiAttendedTransfer>
<anonymousCallBlock>2</anonymousCallBlock>
<callerIdBlocking>2</callerIdBlocking>                Disabled - set to 1 to enable
<dndControl>0</dndControl>
<remoteCcEnable>true</remoteCcEnable>
</sipCallFeatures>
<sipStack>
<sipInviteRetx>6</sipInviteRetx>
<sipRetx>10</sipRetx>
<timerInviteExpires>180</timerInviteExpires>
<timerRegisterExpires>3600</timerRegisterExpires>
<timerRegisterDelta>5</timerRegisterDelta>
```

```
<timerKeepAliveExpires>120</timerKeepAliveExpires>
<timerSubscribeExpires>120</timerSubscribeExpires>
<timerSubscribeDelta>5</timerSubscribeDelta>
<timerT1>500</timerT1>
<timerT2>4000</timerT2>
<maxRedirects>70</maxRedirects>
<remotePartyID>true</remotePartyID>
<userInfo>None</userInfo>
</sipStack>
<autoAnswerTimer>1</autoAnswerTimer>
<autoAnswerAltBehavior>false</autoAnswerAltBehavior>
<autoAnswerOverride>true</autoAnswerOverride>
<transferOnhookEnabled>false</transferOnhookEnabled>
<enableVad>false</enableVad>
<preferredCodec>g711</preferredCodec>
<dtmfAvtPayload>101</dtmfAvtPayload>
<dtmfDbLevel>3</dtmfDbLevel>
<dtmfOutofBand>avt</dtmfOutofBand>
<alwaysUsePrimeLine>false</alwaysUsePrimeLine>
<alwaysUsePrimeLineVoiceMail>false</alwaysUsePrimeLineVoiceMail>
<kpml>3</kpml>
<phoneLabel>AVAYA SIP</phoneLabel>              Text displayed in upper right corner
<stutterMsgWaiting>2</stutterMsgWaiting>
<callStats>false</callStats>
<offhookToFirstDigitTimer>15000</offhookToFirstDigitTimer>
<silentPeriodBetweenCallWaitingBursts>10</silentPeriodBetweenCallWaitingBursts>
<disableLocalSpeedDialConfig>false</disableLocalSpeedDialConfig>
<startMediaPort>16384</startMediaPort>
<stopMediaPort>32766</stopMediaPort>
<sipLines>

<line button="1">                               Physical button 1 (1-8 top to bottom)
<featureID>9</featureID>                        9 = use for call appearance
<featureLabel>34179</featureLabel>              Extension # (User ID in Avaya SES)
<proxy>10.3.3.41</proxy>                        Avaya SES
<port>5060</port>
<name>34179</name>                              Extension # (User ID in Avaya SES)
<displayName>Cisco 7975</displayName>           User name
<autoAnswer>
<autoAnswerEnabled>2</autoAnswerEnabled>
</autoAnswer>
<callWaiting>3</callWaiting>                    Set to 2 to disable
<authName>34179</authName>                      Extension # (User ID in Avaya SES)
<authPassword>123456</authPassword>             User password in Avaya SES
<sharedLine>false</sharedLine>
```

```
<messageWaitingLampPolicy>3</messageWaitingLampPolicy>
<messagesNumber>35000</messagesNumber>              Voice mail extension
<ringSettingIdle>4</ringSettingIdle>
<ringSettingActive>5</ringSettingActive>
<contact>7b452e87-4496-4762-e11f-b26751a1884b</contact>
<forwardCallInfoDisplay>
<callerName>true</callerName>
<callerNumber>false</callerNumber>
<redirectedNumber>false</redirectedNumber>
<dialedNumber>true</dialedNumber>
</forwardCallInfoDisplay>
</line>

<line button="2">                                   Physical button 2 (1-8 top to bottom)
<featureID>9</featureID>                            9 = use for call appearance
<featureLabel>34179</featureLabel>                  Same settings as Button 1
<proxy>10.3.3.41</proxy>
<port>5060</port>
<name>34179</name>
<displayName>Cisco 7975</displayName>
<autoAnswer>
<autoAnswerEnabled>2</autoAnswerEnabled>
</autoAnswer>
<callWaiting>3</callWaiting>
<authName>34179</authName>
<authPassword>123456</authPassword>
<sharedLine>false</sharedLine>
<messageWaitingLampPolicy>3</messageWaitingLampPolicy>
<messagesNumber>35000</messagesNumber>
<ringSettingIdle>4</ringSettingIdle>
<ringSettingActive>5</ringSettingActive>
<contact>7b452e87-4496-4762-e11f-b26751a1884b</contact>
<forwardCallInfoDisplay>
<callerName>true</callerName>
<callerNumber>false</callerNumber>
<redirectedNumber>false</redirectedNumber>
<dialedNumber>true</dialedNumber>
</forwardCallInfoDisplay>
</line>

<line button="3">                                   Physical button 3 (1-8 top to bottom)
<featureID>2</featureID>                            2 = use for speed dial
<featureLabel>Bridge On</featureLabel>              Avaya Feature Name
<speedDialNumber>70024</speedDialNumber>            Feature Name Extension
</line>
```

<line button="4">	*Physical button 4 (1-8 top to bottom)*
<featureID>2</featureID>	*2 = use for speed dial*
<featureLabel>Call Pickup</featureLabel>	*Avaya Feature Name*
<speedDialNumber>70010</speedDialNumber>	*Feature Name Extension*
</line>	
<line button="5">	*Physical button 5 (1-8 top to bottom)*
<featureID>2</featureID>	*2 = use for speed dial*
<featureLabel>Call Park</featureLabel>	*Avaya Feature Name*
<speedDialNumber>70008</speedDialNumber>	*Feature Name Extension*
</line>	
<line button="6">	*Physical button 6 (1-8 top to bottom)*
<featureID>2</featureID>	*2 = use for speed dial*
<featureLabel>Call Unpark</featureLabel>	*Avaya Feature Name*
<speedDialNumber>70009</speedDialNumber>	*Feature Name Extension*
</line>	
<line button="7">	*Physical button 7 (1-8 top to bottom)*
<featureID>2</featureID>	*2 = use for speed dial*
<featureLabel>SAC</featureLabel>	*Avaya Feature Name*
<speedDialNumber>70001</speedDialNumber>	*Feature Name Extension*
</line>	
<line button="8">	*Physical button 8 (1-8 top to bottom)*
<featureID>2</featureID>	*2 = use for speed dial*
<featureLabel>SAC Cancel</featureLabel>	*Avaya Feature Name*
<speedDialNumber>70002</speedDialNumber>	*Feature Name Extension*
</line>	
</sipLines>	
<voipControlPort>5060</voipControlPort>	
<dscpForAudio>184</dscpForAudio>	
<ringSettingBusyStationPolicy>0</ringSettingBusyStationPolicy>	
<dialTemplate>dialplan.xml</dialTemplate>	*Local dial plan file*
<softKeyFile>SK50719900-3bee-4594-bc3f-6400e1a33bf0.xml</softKeyFile>	
</sipProfile>	
<commonProfile>	
<phonePassword></phonePassword>	
<backgroundImageAccess>true</backgroundImageAccess>	
<callLogBlfEnabled>2</callLogBlfEnabled>	
</commonProfile>	

```
<loadInformation>SIP75.8-3-2SR1S</loadInformation>          Firmware version

<vendorConfig>
<disableSpeaker>false</disableSpeaker>
<disableSpeakerAndHeadset>false</disableSpeakerAndHeadset>
<pcPort>0</pcPort>
<settingsAccess>1</settingsAccess>
<garp>0</garp>
<voiceVlanAccess>0</voiceVlanAccess>
<videoCapability>0</videoCapability>
<autoSelectLineEnable>0</autoSelectLineEnable>
<webAccess>0</webAccess>
<daysDisplayNotActive>1,7</daysDisplayNotActive>
<displayOnTime>08:00</displayOnTime>
<displayOnDuration>10:30</displayOnDuration>
<displayIdleTimeout>01:00</displayIdleTimeout>
<spanToPCPort>1</spanToPCPort>
</vendorConfig>

<versionStamp>1136931633-57191cee-5ffc-4342-b286-4246b4991890</versionStamp>


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<uid>1</uid>
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<version>1.0.0.0-1</version>
<winCharSet>iso-8859-1</winCharSet>
</userLocale>

<networkLocale>United_States</networkLocale>
<networkLocaleInfo>
<name>United_States</name>
<uid>64</uid>
<version>1.0.0.0-1</version>
</networkLocaleInfo>
<deviceSecurityMode>1</deviceSecurityMode>
<idleTimeout>0</idleTimeout>
<authenticationURL>http://ccm-beta-5-1:8080/ccmcip/authenticate.jsp</authenticationURL>
<directoryURL>http://10.0.0.20/cisco_voip/PhoneDirectory.xml</directoryURL>
<idleURL></idleURL>
<informationURL>http://ccm-beta-5-
1:8080/ccmcip/GetTelecasterHelpText.jsp</informationURL>
<messagesURL></messagesURL>
```

```
<proxyServerURL>10.3.3.41</proxyServerURL>
<servicesURL>http://10.0.0.20/cisco_voip/services.xml</servicesURL>
<dscpForSCCPPhoneConfig>96</dscpForSCCPPhoneConfig>
<dscpForSCCPPhoneServices>0</dscpForSCCPPhoneServices>
<dscpForCm2Dvce>96</dscpForCm2Dvce>
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<capfAuthMode>0</capfAuthMode>

<capfList>
<capf>
<phonePort>3804</phonePort>
<processNodeName>ccm-beta-5-1</processNodeName>
</capf>
</capfList>

<certHash></certHash>
<encrConfig>false</encrConfig>

</device>
```

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