Avaya Solution & Interoperability Test Lab

Configuring SIP IP Telephony Using Avaya SIP Enablement Services, Avaya Communication Manager, and Snom 190/220/360 SIP Telephones - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to connect Snom 190/220/360 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.

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 R3.0 (SES) adds new feature and scalability enhancements to the SIP functionality previously introduced as 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 Snom 190/220/360 SIP telephones with Avaya SIP Enablement Services, S8300 Media Server, and G700 Media Gateway. Only those configuration steps pertinent to interoperability of Snom 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.

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 Modular Stackable Switch, and are administered as a single subnet. Each Snom SIP telephone is configured to register to one of two SIP Enablement Services home servers and is administered as an OPS station on an 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 on the G700 Media

---

[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.

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

The main difference between the Snom SIP telephone models is the number of line appearances supported. The configuration steps described in these Application Notes apply to all models. **Table 1** profiles the network management capabilities of the phones.

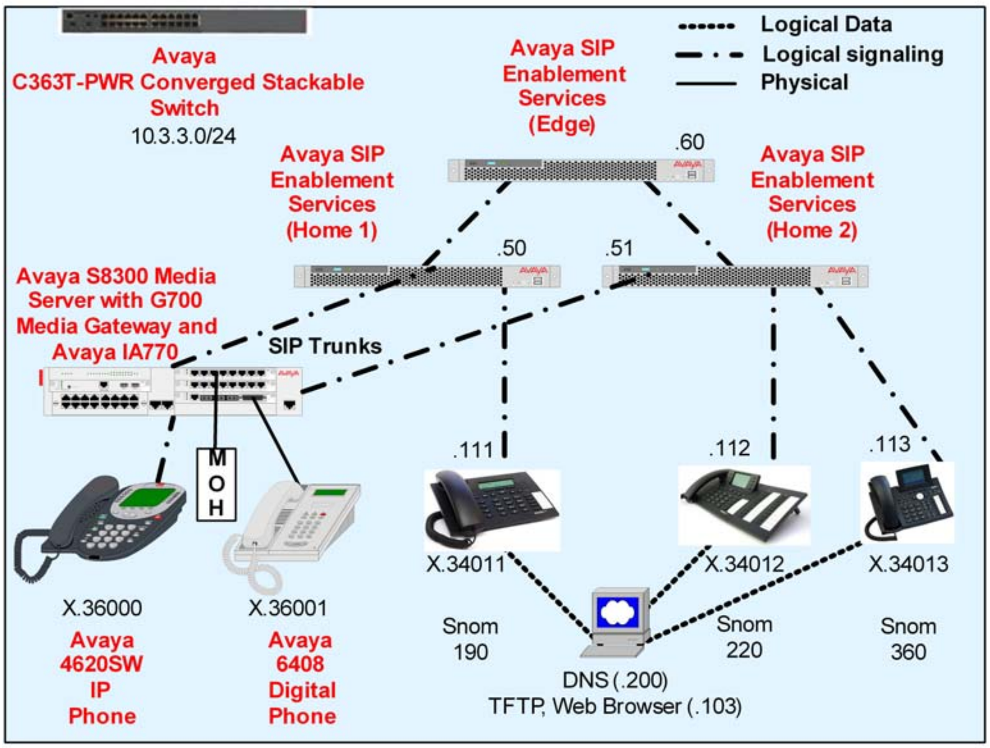


Figure 1: Avaya SIP Test Configuration with Snom SIP Phones

Administration mechanisms	Configuration files, web browser
Administration levels	Administrator, user
File transfer server	HTTP
Error logs	SIP message trace via phone web browser
802.3af Power over Ethernet Support	Yes
SNMP support	Snom 360 only

Table 1: Network Management Capabilities of Snom SIP Telephones

2. Equipment and Software Validated

The following equipment and software were used in the configuration shown in **Table 2**.

Equipment	Software
Avaya SIP Enablement Services (SES) Server	3.0 (Load 31)
Avaya C363T-PWR Modular Stackable Switch	4.3.10
Avaya S8300 Media Server with G700 Media Gateway	Avaya Communication Manager 3.0 (Load 340.3)
Avaya IA770 INTUITY®AUDIX™	N3.0-1.7
Snom SIP Telephones: Model 190 Model 220 Model 360	 3.56y 3.56y 3.60p
PC (DNS, 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 all models of the Snom 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 or Automatic Route Selection (AAR/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	
Basic call to legacy phones	NO	YES	
Intercept tones/displays	YES	YES	Busy 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 3.2.1
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 feature or OPS FNE (See Section 3.2.2)
Call Forward Busy	NO	YES	Changeable via OPS FNE
Call Forward No Answer	YES	YES	Local feature or OPS FNE (See Section 3.2.2)
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 on CM stations
Incoming Call Screening	YES	YES	Local feature or CM/OPS Class Of Restriction (COR)
Outgoing Call Screening	NO	YES	Via CM/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	YES	YES	
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 redial 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	YES	YES	Local DND feature or OPS FNE
Send All Calls Cancel	YES	YES	Local DND feature 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. Message Waiting Indicator (MWI)

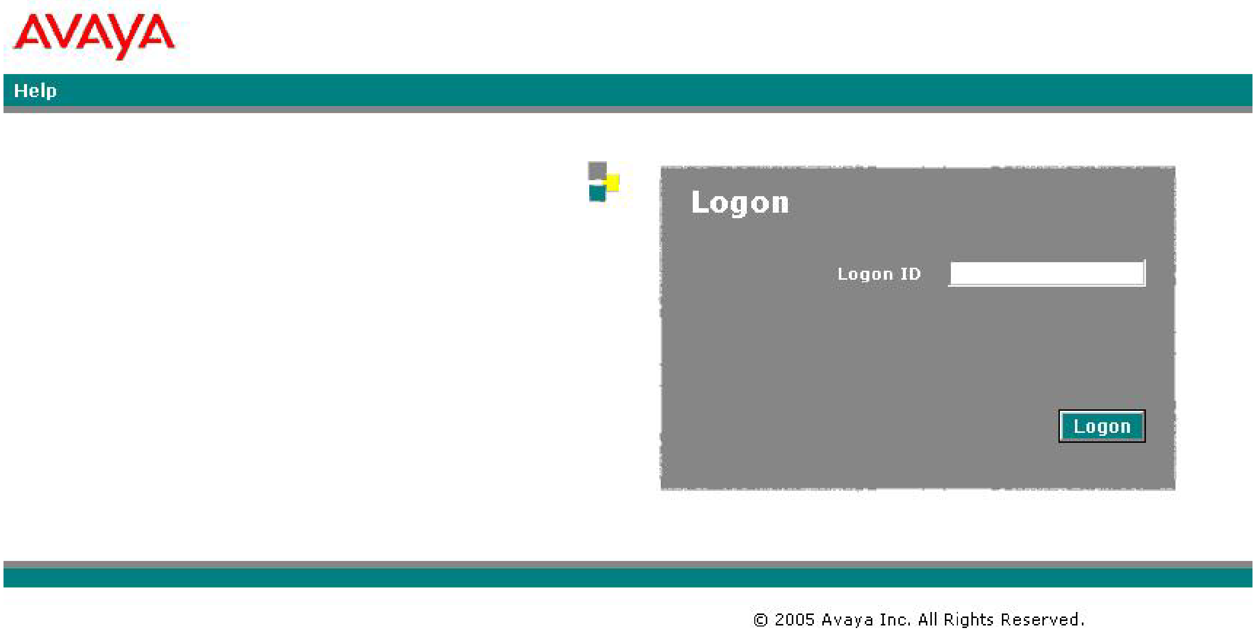
All models of Snom SIP telephones support IETF RFC 3842 (Subscribe/Notify method) and will illuminate/extinguish the MWI lamp when voice messages are left/read for that extension. However, pressing the available hard or soft key to access messages results in the telephone dialing its own extension. Use one of the line appearance keys as a speed dial to implement a "messages" button.

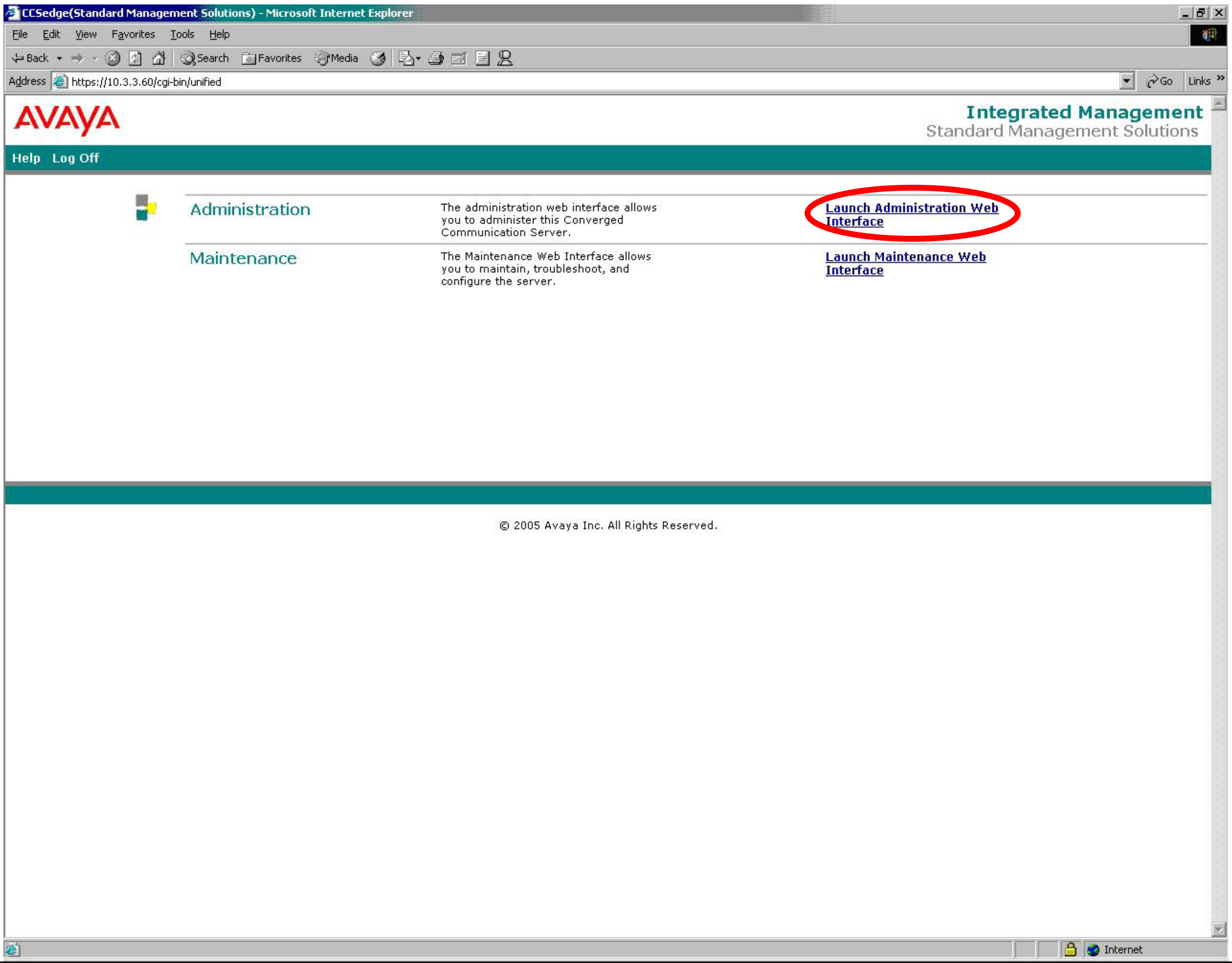
3.2.2. Call Forward

It is recommended that this feature be administered as an Avaya OPS FNE rather than using the local call forward feature administered at the telephone. This is because 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.

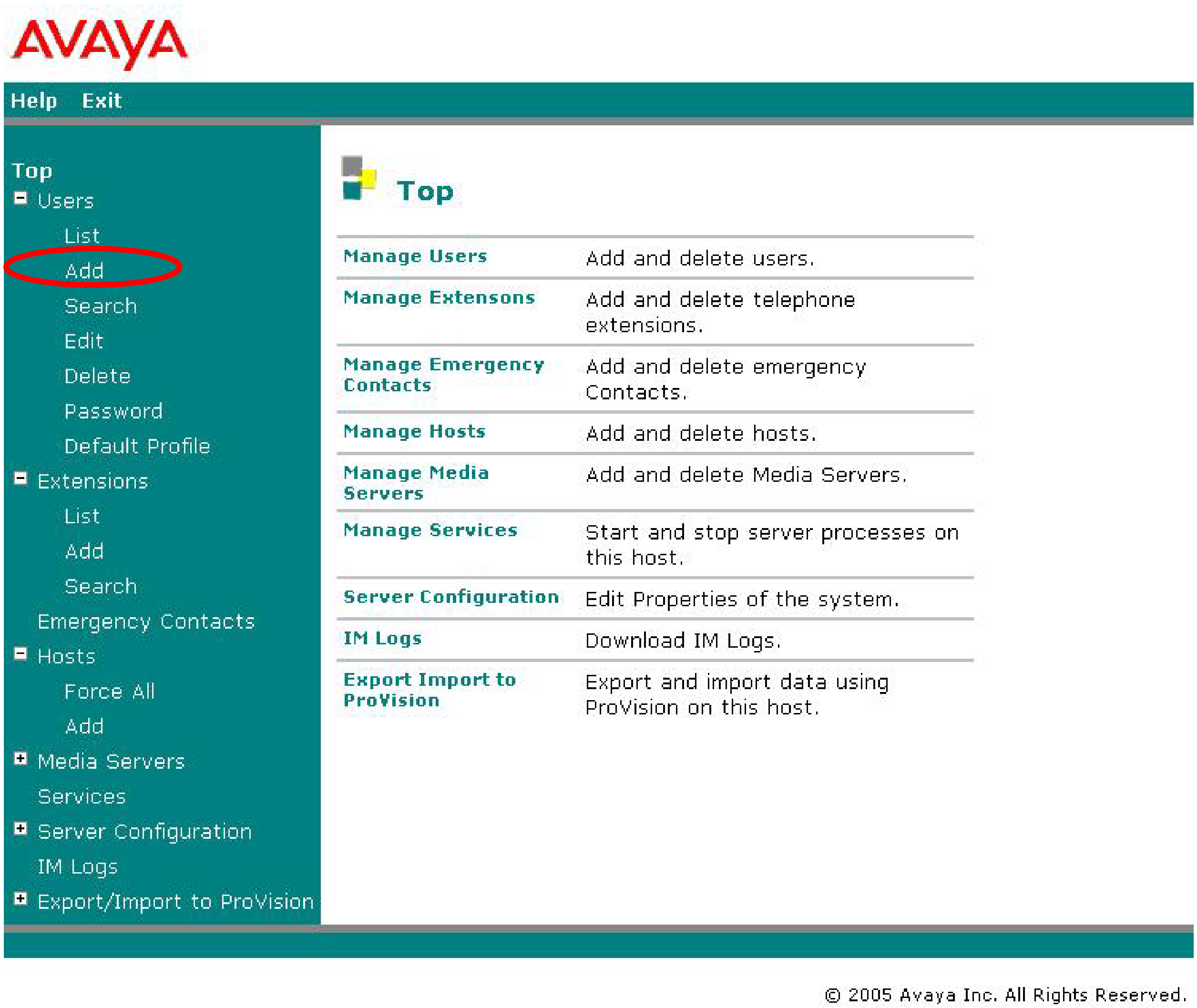
4. Administer SIP Enablement Services

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

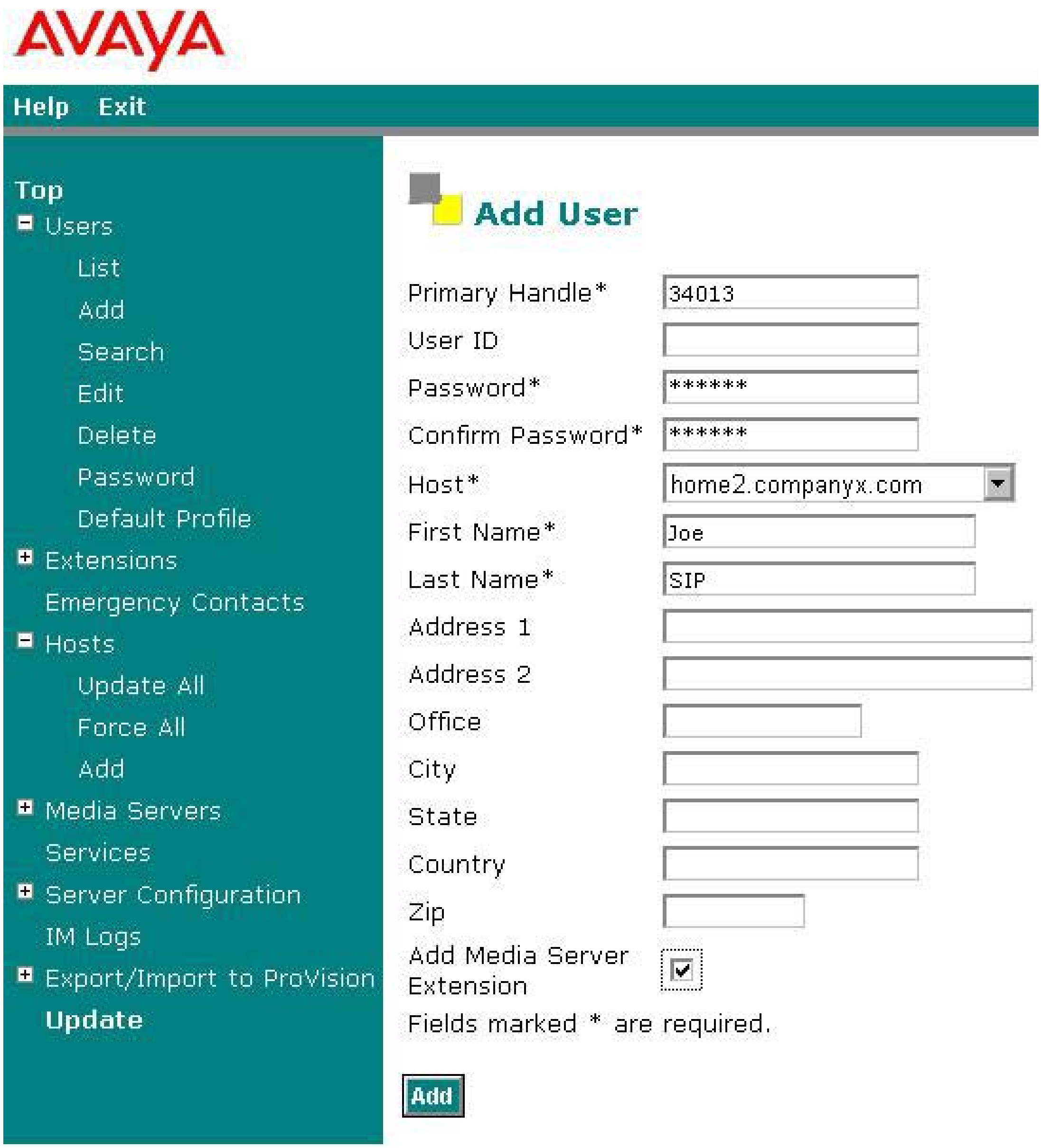
Steps	Description
1.	Avaya SIP Enablement Services is configured using a web browser. Set the URL of the browser to http://IP-address/admin, where *IP-address* is the IP address of the Avaya SIP Enablement Services Edge or Edge/Home Server, and log in as "admin". When prompted, enter the password. 

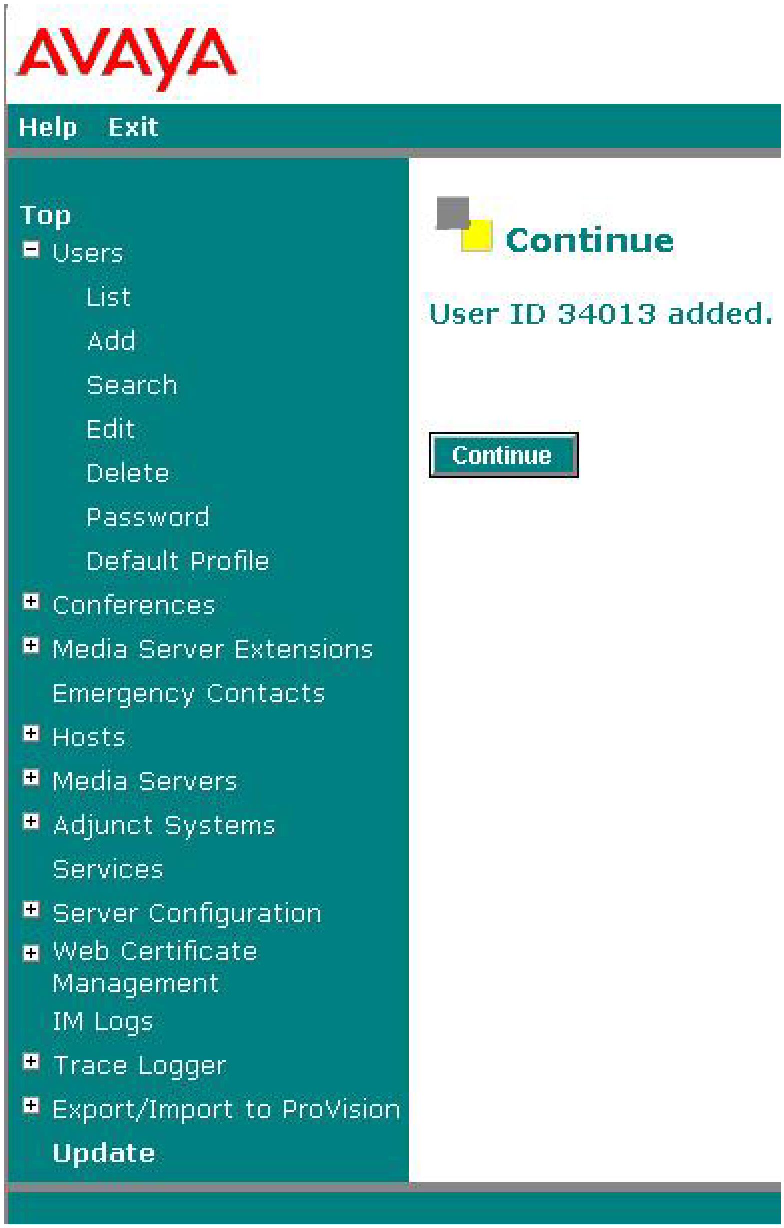
Steps	Description
2.	The main administration screen will be displayed after logging in. Click on **Launch Administration Web Interface.** 

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.

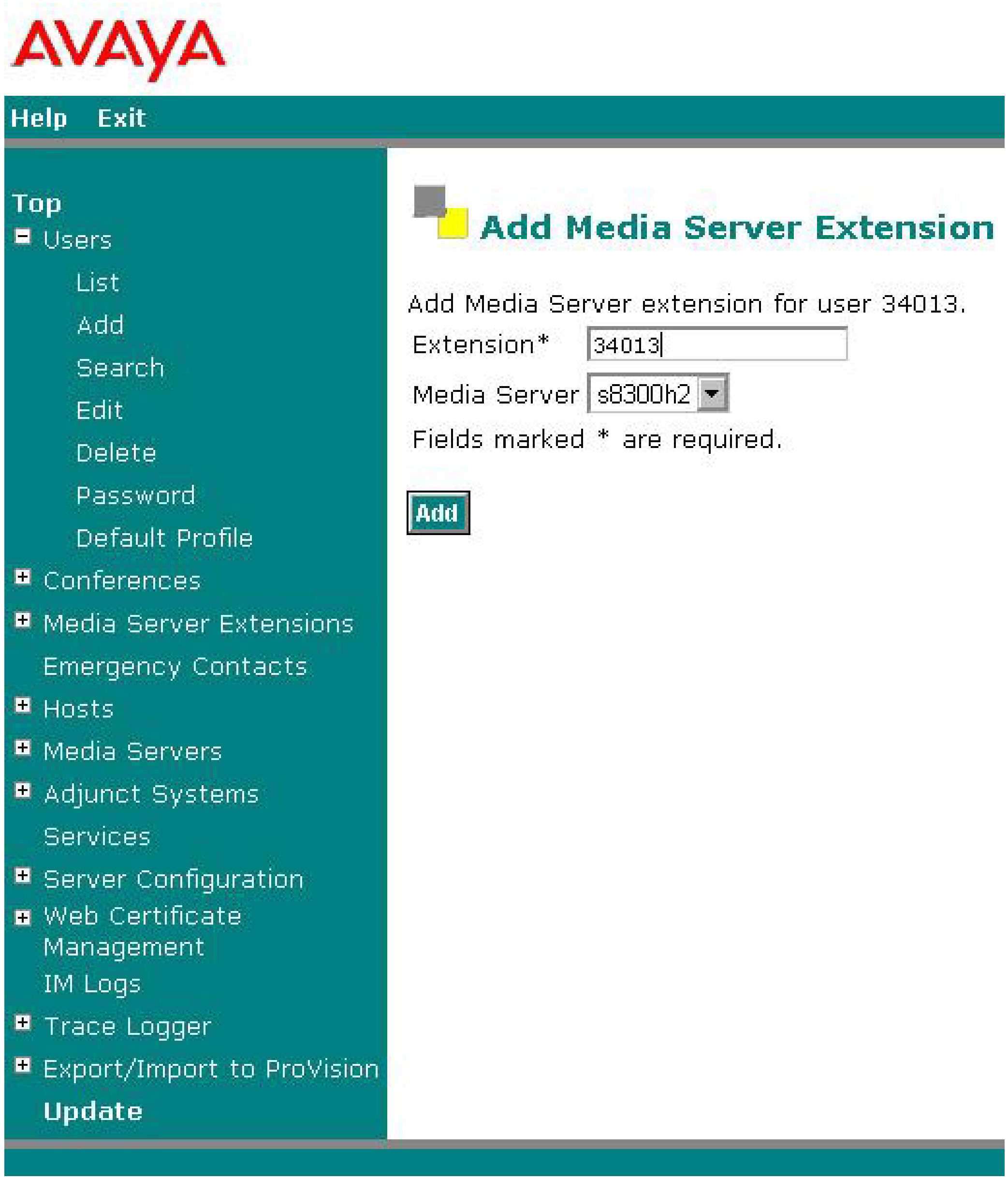


Steps	Description
4.	The *Add User* page will be displayed. Fill in the required fields (indicated by *). In the screen below, the user corresponding to a SIP telephone is being added. Enter the extension number in the **Primary Handle** field. The **Host** field should be set to the DNS host name of 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 2. Check the **Add Media Server Extension** checkbox. Click on **Add**.



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

Steps	Description
6.	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 2, the **Media Server** corresponding to the SIP trunk between the S8300 Media Server and Home 2 is selected automatically.



<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>7.</td><td>The confirmation page will be displayed. Click on Continue.


</td></tr>
<tr><td>8.</td><td>Repeat Steps 3-7 for each user to be added to the system.</td></tr>
<tr><td>9.</td><td>To apply the administration in the above steps, click on Update on the left side of the page. This link appears on the current page whenever updates are outstanding, and can be used at any time to save the administration performed to that point.</td></tr>
</table>

5. 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

Set up 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 Snom telephones, the *Display Name* ("Joe SIP") and *URI* (34013@companyx.com) administered at the calling phone is displayed at the called phone.

```
add station 34013                                              Page   1 of   4
                                      STATION

Extension: 34013                           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: 34013
            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 34013                                              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: 34013                          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. To support certain transfer and conference scenarios, the number of "call-appr" buttons should be at least 3, and should be one more than the number of line appearances configured at the telephone.

```
add station 34013                                            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:34012
 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 34013. A bridged appearance was defined on station 34012 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 (34013) 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 34013                    Page   1 of   2
                    STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station      Application    Dial   Phone Number       Trunk       Configuration
 Extension                   Prefix                    Selection   Set
 34013          OPS               - 34013               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 34013                    Page   2 of   2
                    STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Call       Mapping      Calls       Bridged
 Extension      Limit      Mode         Allowed     Calls
 34013        4          both           all         both
```

6. Configure the Snom SIP Telephone

6.1. Registration and Basic Dialing

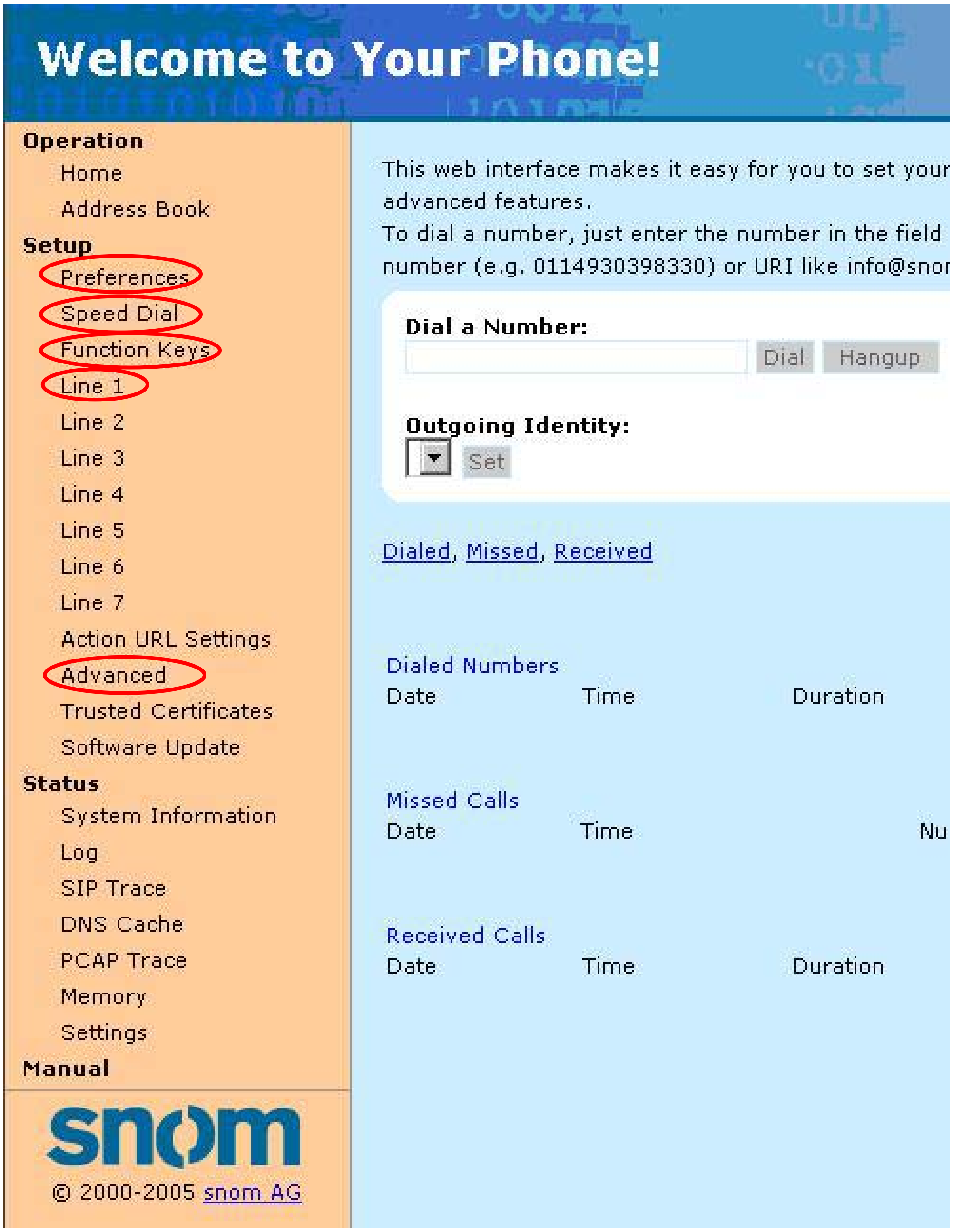
Snom SIP telephones can be configured using two methods:

1. Configuration files downloaded from an HTTP server specified via DHCP at boot time. Two such files can be installed on the server: a default configuration file containing parameter settings that apply to all phones (`http://`*`hostname`*`/settings.htm`), and a telephone-specific configuration file containing settings applicable only to that telephone (`http://`*`hostname`*`/settings.php?mac={`*`mac`*`}`, where *`mac`* is the MAC address of the phone).

2. Manual configuration of the phone using its web browser interface.

Use of a configuration file 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 when the phone is re-booted. See Reference [5] for details on installing and maintaining Snom SIP telephones using configuration files. For the sample configuration, the IP address was manually entered at the phone. The remaining configuration was done via the web browser interface. The parameters displayed on the browser interface can also be set using the configuration files. The following steps are shown for the Snom 360, but are similar for the 190 and 220 models. A picture of the Snom 360 SIP telephone configured for this sample configuration is shown in **Figure 2**, and can be used for reference in the following configurations steps.

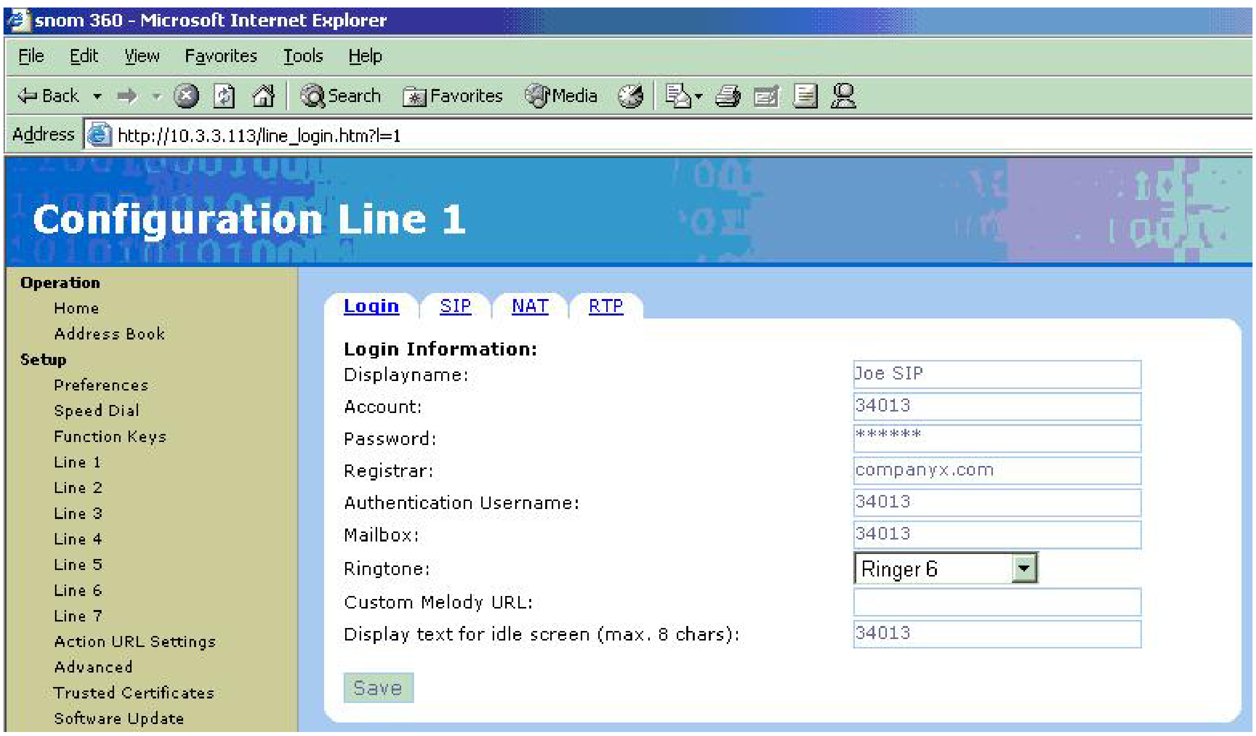
Figure 2: Snom 360 SIP Telephone

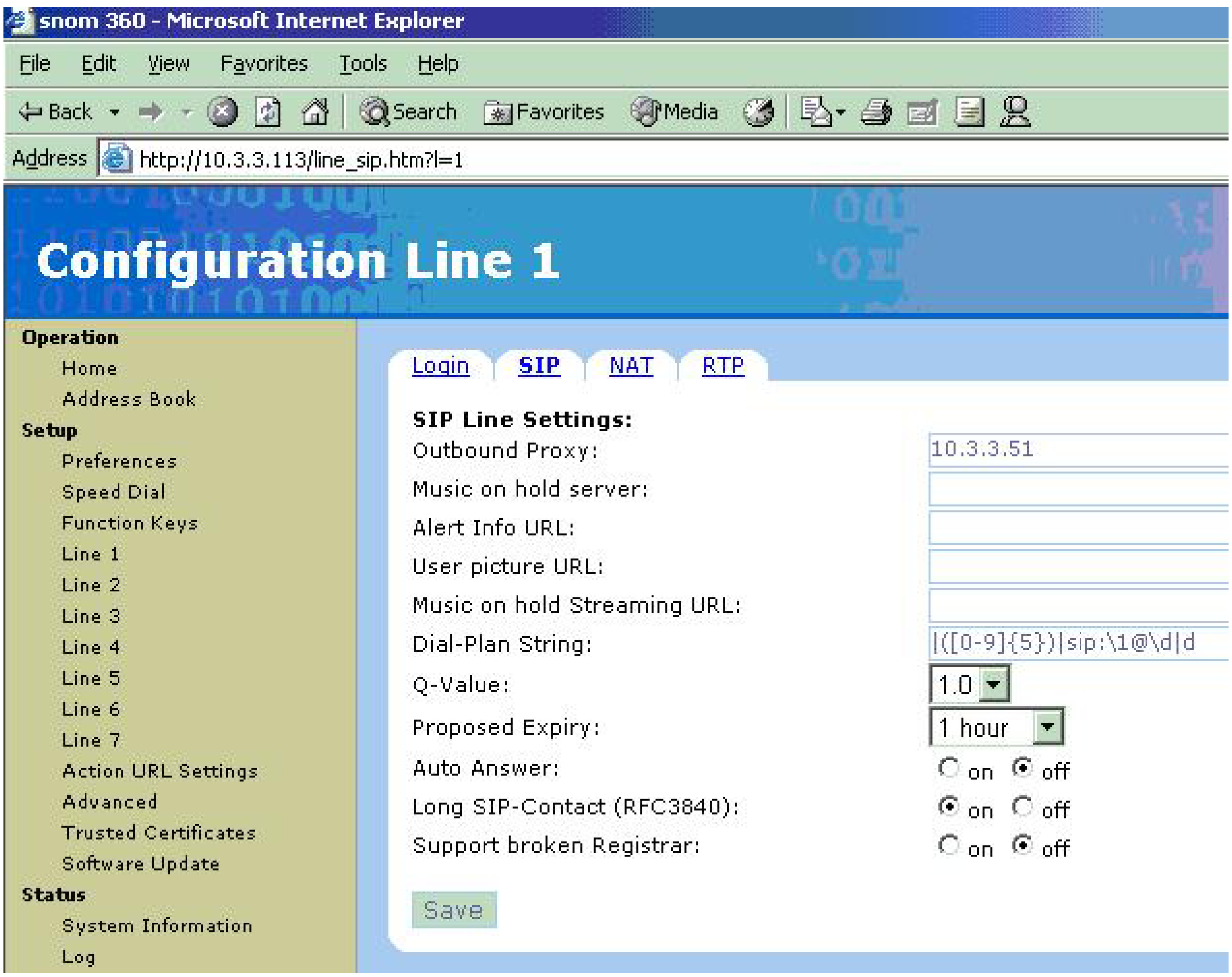
Steps	Description
1.	Using a web browser, navigate to the IP address of the telephone. The following screen is displayed. The circled links on the left will be used in the following steps to configure the telephone. 

<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>2.</td><td>Click on Line 1 under the Setup menu to specify the parameters required to register the telephone with SIP Enablement Services. Under the Login tab, fill in the following fields as indicated:

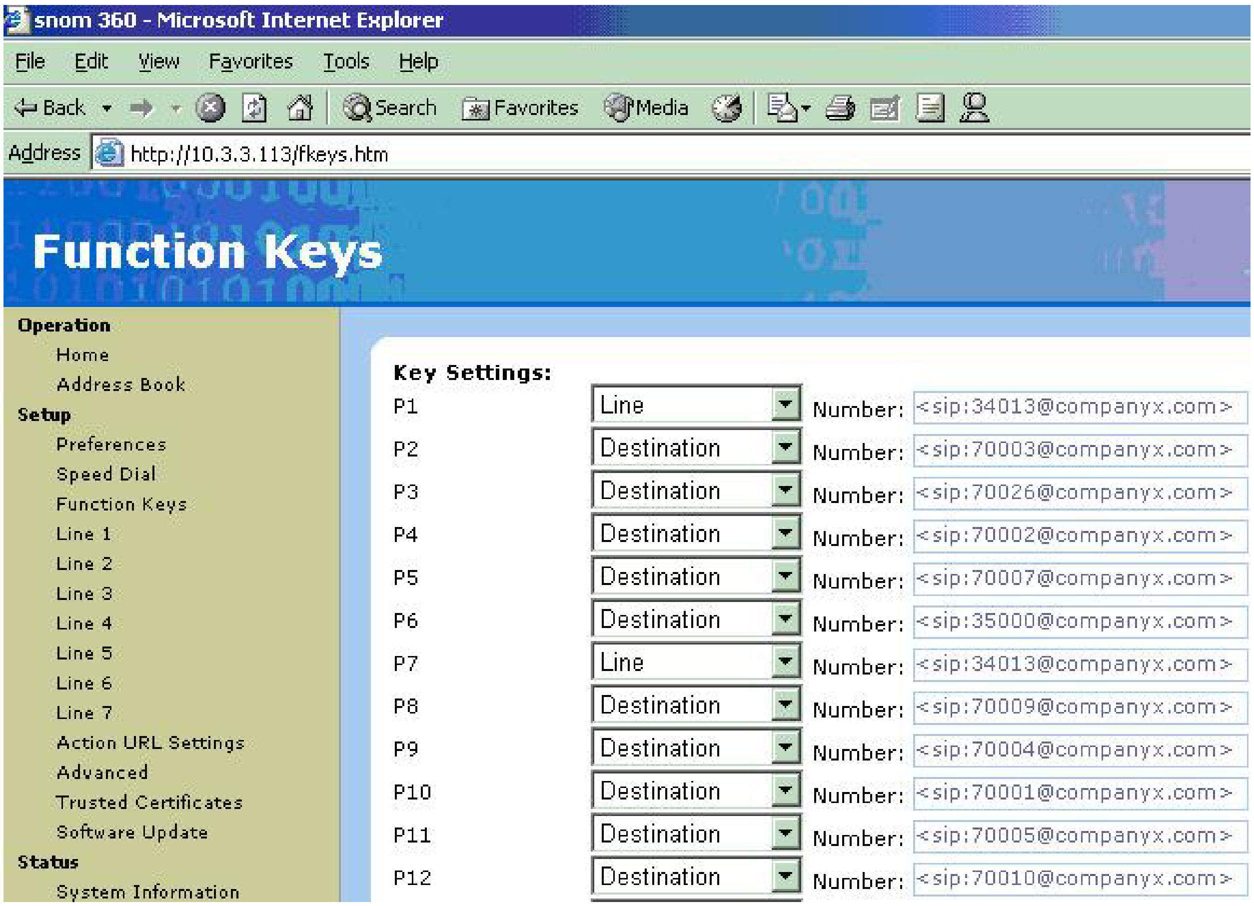
Displayname User name (will be displayed on called party's telephone)
Account The telephone extension
Password The authentication password
Registrar The SIP domain of SIP Enablement Services
Authentication Username The telephone extension
Mailbox The telephone extension
Display text for idle screen The telephone extension

Default values can be used for the remaining parameters. Click on Save when done.


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

Steps	Description
3.	Under the *SIP* tab, fill in the following fields as indicated: **Outbound Proxy** — IP address of the SIP Enablement Services home server **Dial-Plan String** — String specifying when a call is automatically launched based on the number dialed **Dial-Plan String** in the example below is set for 5-digit extension dialing. See [8] for additional information on specifying dial plan strings. Default values can be used for the remaining parameters. Click on **Save** when done. 

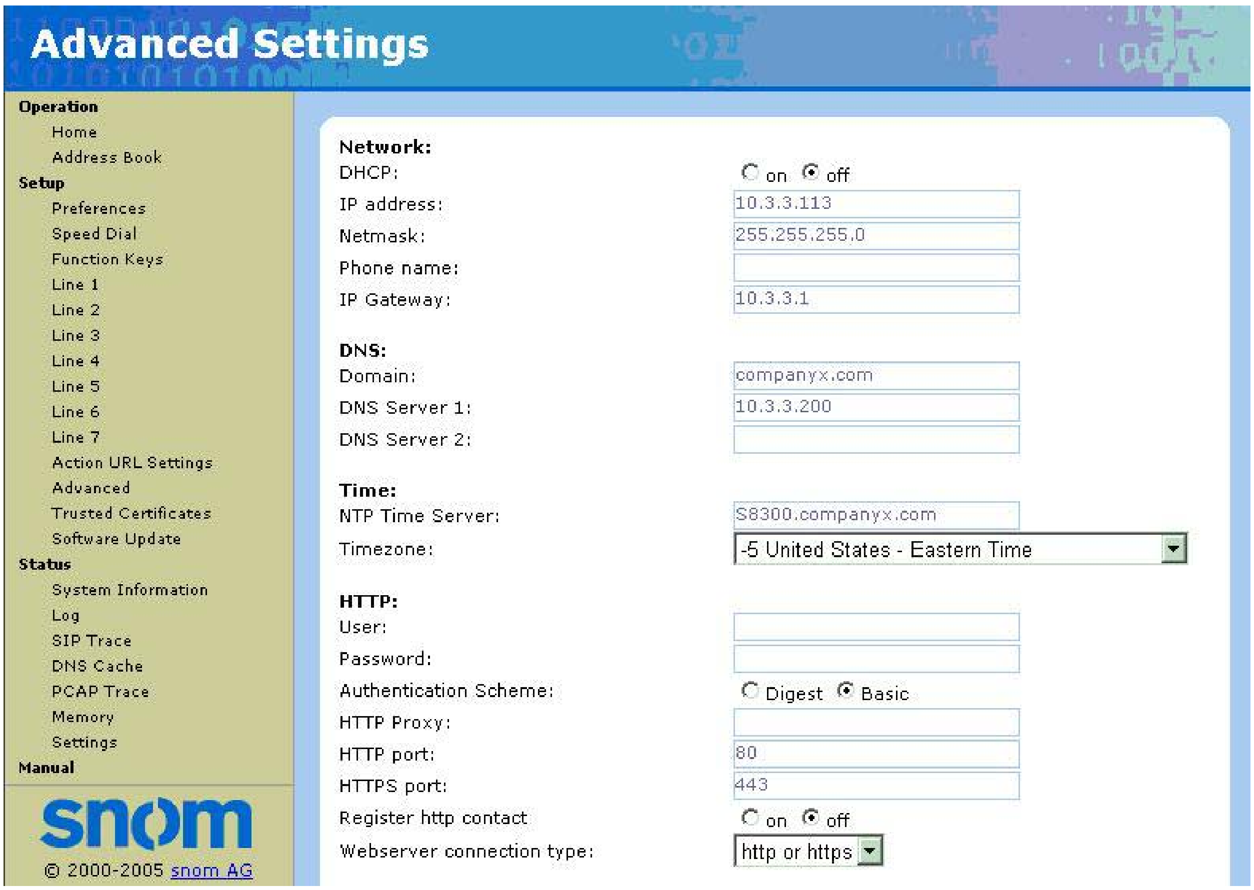
<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>4.</td><td>Click on Function Keys under the Setup menu to configure the function keys on the right side of the telephone. Each key can be designated as a line appearance (Line) or as a speed dial (Destination). In each case, enter the extension corresponding to a registered line (Line 1-Line 7) or a speed dial extension. In the sample configuration below, two buttons (P1 and P7) are line appearances[5] for the extension configured in Step 2, and the rest are speed dials for Avaya OPS extended features, corresponding to extensions administered on the off-pbx station-mapping form in Section 5.5. Click on Save when done (not shown here for brevity). The full URLs will be automatically filled in.


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

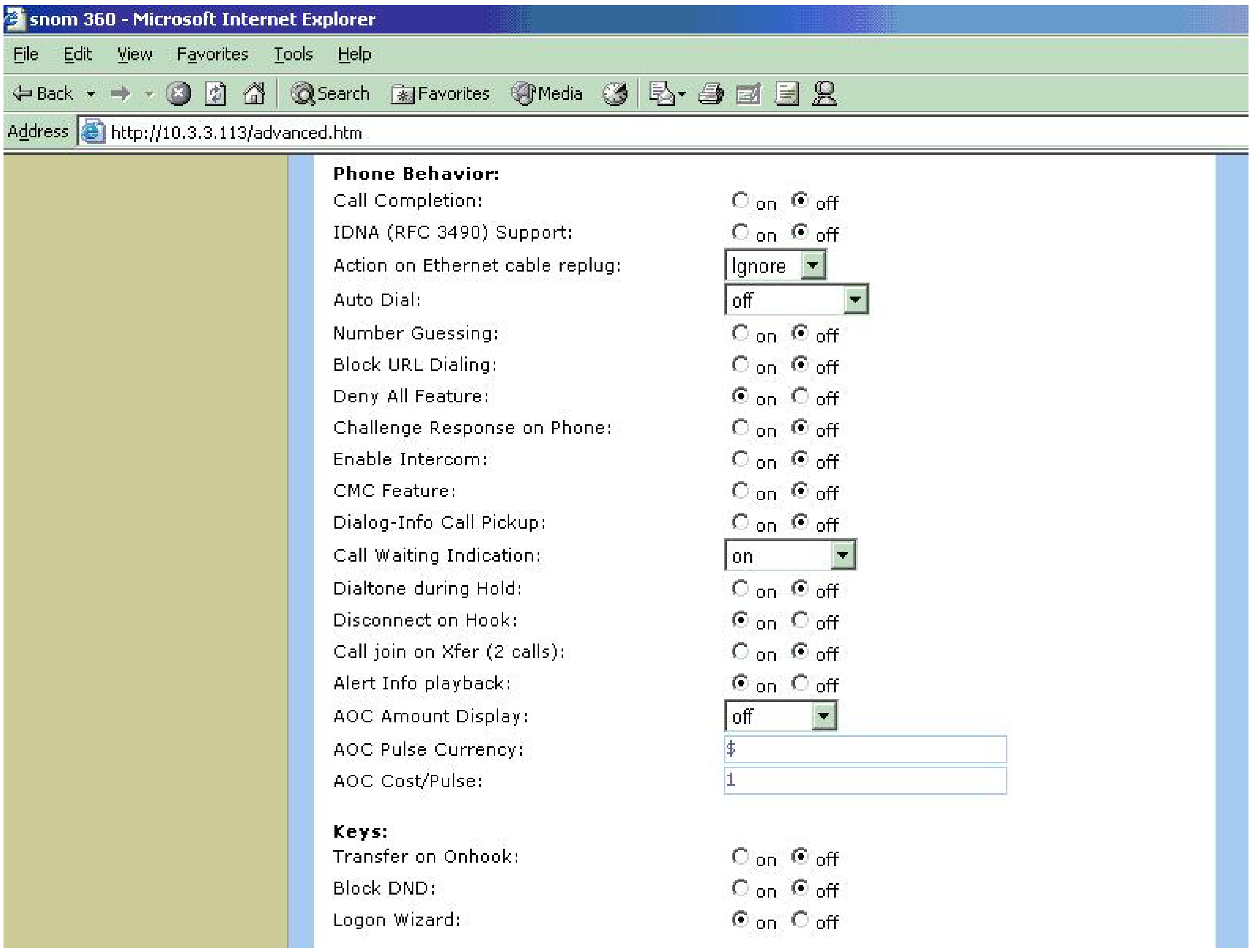
[5] On the Snom 360, buttons P1 and P7 are adjacent, since there are two vertical columns of 6 buttons (see below and **Figure 2**). The Snom 190 and 220 have a single row of 5 buttons.

1	7
2	8
3	9
4	10
5	11
6	12

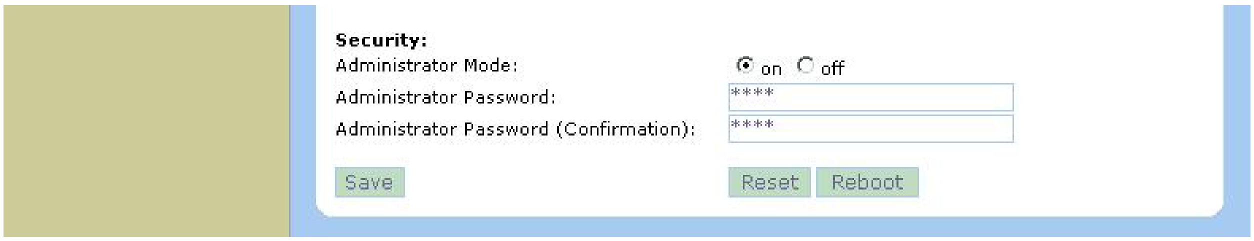
Steps	Description
5.	Click on **Advanced** under the *Setup* menu to configure additional network and phone behavior parameters. Some of these may be automatically configured via DHCP. In the sample configuration, the **IP address**, **Netmask**, and **IP Gateway** were manually configured at the telephone prior to using the web interface, and are displayed below under the *Network* section. Using the web interface, set the DNS **Domain** and IP address of the primary and secondary DNS servers under the *DNS* section. Under the *Time* section, set the **NTP Time Server**. In the sample configuration, it is set to the NTP server on the S8300 Media Server. Default values can be used for the parameters under the *HTTP* section.

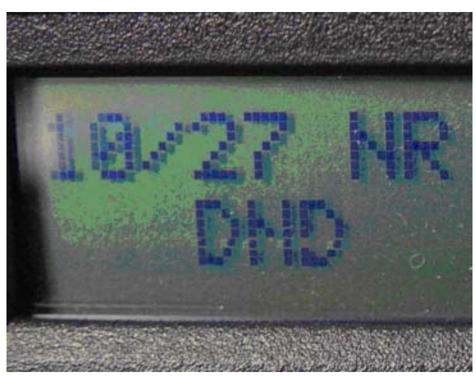


<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>6.</td><td>Continuing under the Advanced web page, match the parameter values set below. Default values can be used in most cases. It is recommended that Dialtone during Hold and Disconnect on Hook be set as shown. Block DND should be set to off if the Do Not Disturb button can be used to send all calls to the coverage path administered in Avaya Communication Manager.



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

Steps	Description
7.	Continuing under the *Advanced* web page, default values can be used for the remaining parameters on the page. This includes the *Advanced Network*, *Update*, *VLAN ID*, *Debug*, and *SNMP* sections (not shown here for brevity). Click **Save** when finished with the *Advanced* web page. If network settings have been changed, click on **Reboot** so the new values will take effect. *Caution: do not click on **RESET** unless it is desired to reset all telephone parameters to default values.* 

Steps	Description
8.	After the telephone completes the reboot process, it should successfully register to SIP Enablement Services. This will be indicated on models 220 and 360 by a telephone symbol in the upper left corner of the display, as shown below.  Unsuccessful registration is indicated by an "X" in the same location.  Unsuccessful registration of the Snom 190 is indicated by the letters "NR" (not registered) on the left side of the 2-line display, as shown below. 

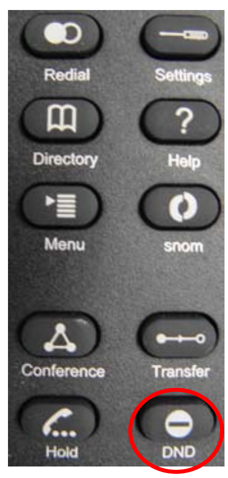
Steps	Description
9.	For basic calling, lift the receiver, press **Speaker**, or press a line appearance button, and dial any number using the dial plan centrally administered in Avaya Communication Manager on the Avaya Media Server. Those features listed in **Table 3** as being locally supported at 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. The steps refer to the Snom 360, but are similar for the other models.

6.2.1. Do Not Disturb (DND)

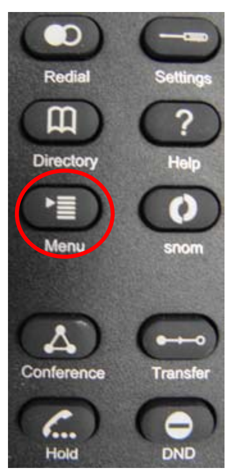
When enabled, DND will cause incoming calls to be routed to the coverage path specified for this telephone in Avaya Communication Manager. This is typically a voice messaging system.

Steps	Description
1.	Press the **DND** button, indicated below. 

Steps	Description
2.	The display will indicate that DND is active, as shown below.  Repeat Step 1 to deactivate the feature.

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. Two configuration methods are shown, using either the telephone or the web interface. The telephone interface is shown for the model 360; it is similar for the other models.

Steps	Description
1.	*At the telephone:* On the main telephone display, press the **Menu** key. 

<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>2.</td><td>

Select **Redirection** from the menu items listed and press the √ soft key.

Use the ← or → soft keys to select the redirection condition. For call forward all, select **Always**.

For call forward busy, select **Busy**. For Call forward no answer, select **Timeout**. Press the  key (see **Figure 2**).

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

Steps	Description
3.	For call forward all and no answer, select **Redirect Target** from the menu (for call forward busy, select **Busy Target**). Enter the extension to which calls are to be forwarded. Press the  key (see **Figure 2**).  

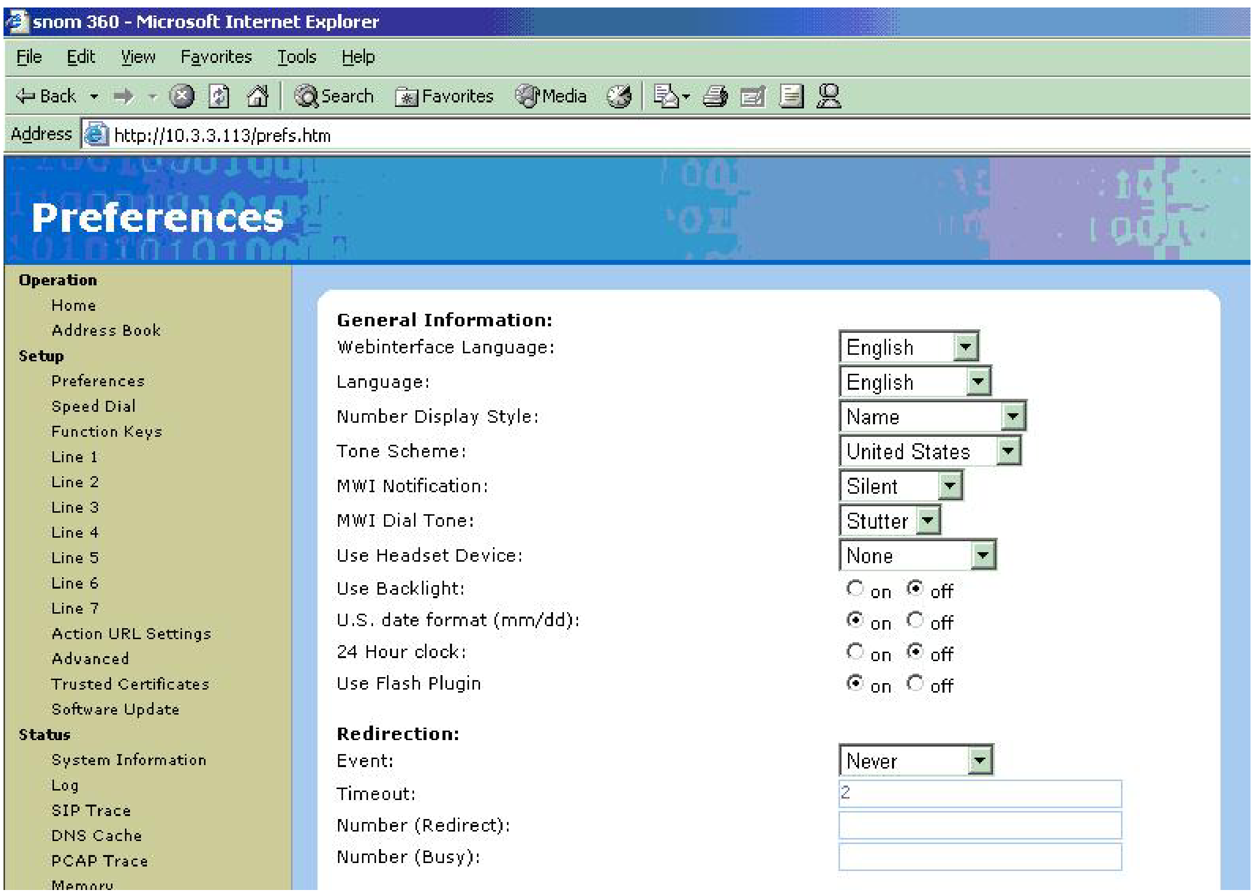
Steps	Description
4.	In addition to the previous step, for call forward no answer, select **Redirect Time** from the menu, and enter the number of seconds to elapse before the call is forwarded. Press the  key (see **Figure 2**). 

Alternatively, to configure local call forwarding using the web interface:

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

Click on **Preferences** under the *Setup* menu. Call forward all, no answer, and busy can be activated on the displayed page under the *Redirection* section. For call forward all, set **Event** to *Always*. Then enter the forward-to extension in **Number (Redirect)**. For call forward no answer, set **Event** to *Interval*, enter the forward-to extension in **Number (Redirect)**, and enter the number of seconds to elapse before the call is forwarded in **Timeout**. For call forward busy, set **Event** to *Busy*, and enter the forward-to extension in **Number (Busy)**.

Click on **Save** when finished.



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

After configuring call forwarding using either method, the main telephone display will show the call forward indication.

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. Configuration of speed dial buttons was described in Step 4 of Section 6.1. General configuration information can be found in Reference [3]. Speed dial buttons can also be centrally administered using configuration files, as described in [5].

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 **menu** button at the phone to verify that the parameters set in the global (outbound proxy address and port number, etc.) and phone-specific (Authentication Username, Password, etc.) configuration files have been loaded. Verify that the symbol in the upper left corner of the display is a telephone, indicating that registration has occurred. If an “X” appears (“NR” for the model 190), check that the proxy server address is set to the correct domain name, the outbound proxy IP address and port number are correct. Verify that the display shows the Avaya Communication Manager extension for that phone.
2. Verify basic feature set administration by lifting the handset, pressing the **speaker** button, or pressing a line appearance button, and making calls to other phones. Test supported features according to **Table 3** and feature deployment plans at the site.
3. If any line appearance or speed dial buttons are incorrect or inoperative, use the phone web interface 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. If administered, press the speed dial button for getting messages and verify that the voice messaging system is called.
6. Verify Message Waiting Indicator (MWI) operation by calling the phone, allowing the call to go to voice mail coverage, and leaving a message. After the call is hung up, the message light should light, and the display should indicate messages are waiting. Call the voice mail extension, verify receipt of the message, delete the message, and verify that the messages light extinguishes and the display no longer indicates messages waiting. If the MWI indicators fail to change, make sure that the *Mailbox* field (Section 6.1, Step 2) is properly set to the phone’s extension.

8. Support

For technical support of Snom products:

Internet: http://www.snom.com/support.html
Email: support@snom.com
Telephone:
Germany/Europe: +49 30 39833-0
India and SAARC: +91 80 51200227-8/9

9. Conclusion

These Application Notes have described the administration steps required to use Snom 190, 220, and 360 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.

10. Additional References

[1] *Converged Communications Server Installation and Administration*, *Release 3.0*, Issue 4, Doc ID 555-245-705, April, 2005, 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] *Snom 360 VoIP Business Phone Manua1*, Issue 1.00, 2005, *Snom 190 Version 3.00 Manual, 2004,* Snom Technology AG. All manuals available at www.snom.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] *Configuring Snom Phones for Mass Deployment,* document faq-04-03-26-v3.4-sf, September 17, 2004, V3.4, Snom Technology AG, available at www.snom.com/whitepapers.

[6] *SIP Support in Release 3.0 of Communication Manager Running on the S8300, S8500, S8700, and S8710 Media Server*, Issue 5.1, Doc ID 555-245-206, July, 2005, available at http://support.avaya.com.

[7] *Administrator Guide for Avaya Communication Manager*, Issue 1, Doc ID 03-300509, June 2005, available at http://support.avaya.com.

[8] Using the Dial Plan on the Snom Phones*,* document FAQ-04-03-27-cs.pdf, March 27, 2004, Snom Technology AG, available at www.snom.com/whitepapers.

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