Avaya Solution & Interoperability Test Lab

Application Notes for Avaya 1120E, 1140E, and 1165E IP Deskphones with Avaya Aura™ Communication Manager, Avaya Aura™ Session Manager, and Avaya Modular Messaging – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Aura™ Communication Manager, Avaya Aura™ Session Manager, Avaya Modular Messaging, and Avaya 1100-Series IP Deskphones with SIP software. During compliance testing, the IP Deskphones successfully registered with Session Manager, placed and received calls to and from SIP and non-SIP telephones, and executed other telephony features such as conference, transfer, hold, and Off-PBX-Station (OPS) related features such as Call Pickup, Call Park, Whisper Page, and Transfer to Voice Mail.

Information in these Application Notes has been obtained through interoperability testing and additional technical discussions, and was conducted at the Avaya Solution and Interoperability Test Lab at the request of Avaya 1100-Series IP Deskphone Product Management.

1. Introduction

These Application Notes describe a solution comprised of Avaya Aura™ Communication Manager, Avaya Aura™ Session Manager, Avaya Modular Messaging, and Avaya 1100-Series IP Deskphones with SIP software (formerly known as Nortel 1100 series SIP Phones). These telephones were originally developed under the Nortel brand, and as such do not currently support the Avaya Advanced SIP Telephony (AST) protocol implemented in Avaya 9600 Series IP Telephones (SIP). Nevertheless, Communication Manager and Session Manager have the capability to extend some advanced telephony features to non-AST telephones. The configuration steps described include how to set up these features as well as the standard calling features supported by the phones. See **Section 4** for a summary of the features supported.

2. Reference Configuration

In the test configuration shown below, The Avaya S8720 Servers with Avaya G650 Media Gateway are configured as an Access Element and support the Avaya 6408D+ Digital Telephone, Avaya 9630 IP Telephone (H.323), and Avaya One-X Communicator (H.323). The Avaya S8510 Server with Avaya G450 Media Gateway is configured as a Feature Server and supports all of the Avaya SIP telephones shown. The Avaya 1100-Series IP Deskphone models tested were the 1120E (4 line monochrome), 1140E (6 line monochrome), and the 1165E (8 line color). Communication between the Communication Managers and Avaya Modular Messaging is via Session Manager. Modular Messaging supports all telephones for voice messaging coverage.

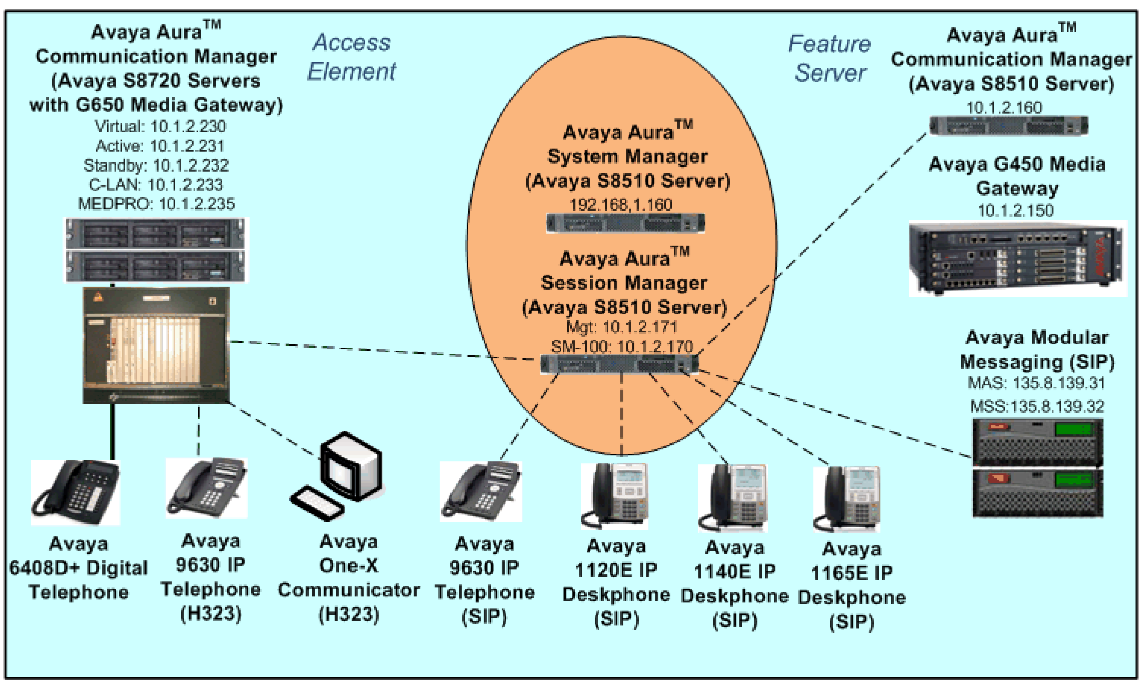


Figure 1: Sample Configuration

In general, a SIP Telephone originates a call by sending a call request (SIP INVITE message) to Session Manager, which then routes the call over a SIP trunk to Communication Manager (Feature Server) for origination services. If the call is destined for another local SIP telephone, then Communication Manager routes the call back over the SIP trunk to Session Manager for delivery to the destination SIP telephone. If the call is destined for an H.323 or Digital Telephone, then Communication manager routes the call back to Session Manager for delivery to the Communication Manager (Access Element) supporting H.323 and Digital endpoints.

These application notes assume that Communication Manager and Session Manager are already installed and basic configuration steps have been performed. Only steps relevant to SIP telephone calling features will be described in this document. For further details on configuration steps not covered in this document, consult the appropriate document in **Section 10**.

3. 3. Equipment and Software Validated

The following equipment and software/firmware were used for the sample configuration provided.

Equipment	Software/Firmware
S8720 Server with G650 Media Gateway S8510 Server with G450 Media Gateway	Avaya Aura™ Communication Manager 5.2.1, Load 16.4, Update 17774
S8510 Server	Avaya Aura™ Session Manager 5.2 Service Pack 2, Load 5.2.2.0.522007
	Avaya Aura™ System Manager 5.2 Service Pack 2, Load 5.2.2.0.522007
Avaya 9630 IP Telephone (SIP)	2.5
Avaya 9630 IP Telephone (H.323)	3.1
Avaya One-X Communicator (H.323)	5.2
Avaya 6408D+ Digital Telephone	-
Modular Messaging Storage Server	5.2, Build 5.2-11.0
Modular Messaging Application Server	5.2, Build 9.2.150.13 (Patch 520008)
Avaya 1120E and 1140E IP Deskphones	03.00.56.01
Avaya 1165E IP Deskphone	03.02.05.20

Table 1: Equipment and Software/Firmware

4. Calling Features

4.1. Overview

Table 2 below shows the calling features successfully tested. Notes on specific feature operations are included in **Section 4.2**. In addition to basic calling capabilities, the Internet Engineering Task Force (IETF) has defined a supplementary set of calling features in RFC 5359 [9], previously referred to as the SIPPING features. This provides a useful framework to describe product capabilities and compare features supported by various equipment vendors. Communication Manager can support many of these features if the telephone can not locally support them. In addition, features beyond those specified in RFC 5359 can be extended to the telephone using Communication Manager configured as a Feature Server.

SUPPORTED FEATURES	COMMENTS
Basic Calling features	
Extension to extension call	
Basic call to non-SIP phones	
Intercept tones/displays	Reorder with message
Call Waiting	
Do Not Disturb	Section 4.2.1
Speed Dial buttons	
Compressed codecs	G.729A, G.729AB (Section 4.2.2)
Message Waiting Support	
SIPPING (RFC 5359) Features	
Call Hold	
Consultation Hold	
Music on Hold	
Unattended Transfer	
Attended Transfer	
Call Forward Unconditional	(Sections 4.2.3, 5.8)
Call Forward Busy	Via FNE (Section 5.8)
Call Forward No Answer	Via FNE (Section 5.8)
3-way conference - 3rd party added	See Section 4.2.4
3-way conference - 3rd party joins	
Find-Me	Modular Messaging "Find Me" feature
Incoming Call Screening	Via Class Of Restriction (Section 5.9)
Outgoing Call Screening	Via Class Of Restriction (Section 5.9)
Call Park/Unpark	Via FNE (Section 5.8)
Call Pickup	Via FNE (Section 5.8)
Additional Station-Side Features	
Calling Name/Number Block	Via FNE (Section , 4.2.5, 5.8)
Directed Call Pick-Up	Via FNE (Section 5.8)
Priority Call	Via FNE (Sections 4.2.6, 5.8, 5.9)
Transfer to Voice Mail	Via FNE (Section 5.8)
Whisper Page	Via FNE (Section 5.8)

Table 2: SIP Telephony Feature Support

Some supported features shown in **Table 2** can be invoked by dialing a Feature Name Extension (FNE). Or, a speed dial button on the telephone can be programmed to an FNE. 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 Communication Manager can be found in References [1-3, 5-6].

4.2. Operational Notes

4.2.1. Do Not Disturb

When Do Not Disturb is activated, the call is not presented to or displayed on the phone. The call follows the coverage path configured for the extension in Communication Manager. This feature is locally supported by the telephone, and is recommended instead of the Send All Calls and Send All Calls Cancel Communication Manager FNEs.

4.2.2. Compressed Codecs

G.722-64k will be supported with Communication Manager 5.2.1.Service Pack 3.

4.2.3. Call Forward Unconditional

It is recommended that this feature be administered as a 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 Communication Manager, including coverage to voice messaging.

4.2.4. 3-way Conference – 3rd Party Added

In one specific scenario - Avaya 9630 IP telephone (SIP) calls Avaya 1100-Series IP Deskphone, which conferences a second 1100-Series IP Deskphone - at times when the conferenced party hangs up, the remaining call may be dropped. This issue will be resolved in Communication Manager 5.2.1 Service Pack 4.

4.2.5. Calling Name/Number Block

The Avaya 1100-Series IP Deskphones support privacy by means of the Remote-Party-ID SIP header in the INVITE message. Since Communication Manager supports the newer Privacy header along with P-Asserted-Identity header, this local feature is not supported. It is recommended that the Calling Number Block FNE in Communication Manager be used instead. This can be configured as speed dial button on the telephone (see **Sections 5.8** and **7.4**).

4.2.6. Priority Call

The telephone may originate priority calls based on the class of service administered for it (see **Section 5.9**) 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) and Avaya 6408D+ Digital Telephones will properly indicate them via distinctive ringing and calling party display.

5. Configure Avaya Aura™ Communication Manager

This section describes a procedure for setting up a SIP trunk between Communication Manager serving as a Feature Server and Session Manager. This includes steps for setting up a list of IP codecs, an IP network region, a signaling group and its interface. Before a trunk can be configured, it is necessary to verify if there is enough capacity to setup an additional trunk. Also, a procedure is described here to configure SIP telephones and features available with OPS in Communication Manager. Configuration in the following sections is only for the fields where a value needs to be entered or modified. Default values are used for all other fields.

These steps are performed from the Communication Manager System Access Terminal (SAT) interface. Avaya and third party SIP telephones are configured as Off-PBX Stations (OPS) in Communication Manager. Communication Manager does not directly control an OPS endpoint, but its features and calling privileges can be applied to it by associating a local extension with the OPS endpoint. Similarly, a SIP telephone in Session Manager is associated with an extension on Communication Manager. SIP telephones register with Session Manager and use Communication Manager for call origination and termination services, including Feature Name Extension (FNE) support. Enter the **save translation** command after completing this section.

5.1. Capacity Verification

Step	Description
1.	Enter the **display system-parameters customer-options** command. Verify that there are sufficient **Maximum Off-PBX Telephones – OPS** licenses. If not, contact an authorized Avaya account representative to obtain additional licenses.

```
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: 12                            RFA Module ID (MID): 1

                                                               USED
                                Platform Maximum Ports: 44000 200
                                      Maximum Stations: 450   60
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 10    0
                    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
```

2.	Proceed to **Page 2** of **OPTIONAL FEATURES** form. Verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient for the number of SIP trunks needed. If not, contact an authorized Avaya account representative to obtain additional licenses. **Note:** Each SIP call between two SIP endpoints requires four SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted.

```
display system-parameters customer-options                      Page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 450   100
           Maximum Concurrently Registered IP Stations: 18000 1
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 5     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 5     0
                       Maximum Administered SIP Trunks: 300    40
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 5     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
```

5.2. IP Codec Set

This section describes the steps for administering an IP codec set in Communication Manager. This codec set is used in the IP network region for communications between Communication Manager and Session Manager.

Step	Description
1.	Enter the **change ip-codec-set n** command, where **n** is a number between **1** and **7**, inclusive. IP codec sets are used in **Section 5.3** for configuring an IP network region to specify which codec sets may be used within and between network regions. For the compliance testing, **G.722-64K**, **G.711MU**, and **G.729AB** were used and **Media Encryption** was set to **none**. If only one codec should be used, then only specify the one that is to be used. Note that for G.729 interoperability between Avaya 1100-Series IP Deskphones, Avaya 9600 Series SIP Telephones, and Avaya one-X Communicator in Road-Warrior mode, the G.729A codec should be used, and the configuration file settings for the 9600 SIP Telephone should include the line: SET ENABLE_G729 "1".

```
change ip-codec-set 7                                           Page   1 of   2

                          IP Codec Set

    Codec Set: 7

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.722-64K                   2        20
 2: G.711MU          n          2        20
 3: G.729AB          n          2        20
 4:
 5:
 6:
 7:


     Media Encryption
 1: none
 2:
 3:
```

5.3. IP Network Region

This section describes the steps for administering an IP network region in Communication Manager for communication between Communication Manager and Session Manager.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">1.</td><td>Enter the change ip-network-region n command, where n is a number between 1 and 250 inclusive and configure the following as shown in the display screen below:

Authoritative Domain – Set to avaya.com in this example. This should match the SIP Domain value configured in Session Manager.
Intra-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or Session Manager in the same IP network region.
Inter-region IP-IP Direct Audio – Set to yes to allow direct IP-to-IP audio connectivity between endpoints registered to Communication Manager or Session Manager in different IP network regions.
Codec Set – Set the codec set number as provisioned in Section 5.2.
</td></tr>
<tr><td><pre>
display ip-network-region 1   Page   1 of 19
                               IP NETWORK REGION
  Region: 1
Location: 1   Authoritative Domain: avaya.com
    Name: Company X
MEDIA PARAMETERS   Intra-region IP-IP Direct Audio: yes
      Codec Set: 7   Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048   IP Audio Hairpinning? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS   RTCP Reporting Enabled? y
 Call Control PHB Value: 46   RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46   Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5   AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS   RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
</pre></td></tr>
</table>

2.	Proceed to **Page 3** of IP network region configuration and enable inter-region connectivity between regions as per below. For this compliance testing, **codec set** is automatically set to the IP codec set entered in Step 1.

```
display ip-network-region 1                                     Page   3 of  19

 Source Region: 1     Inter Network Region Connection Management     I       M
                                                                     G  A    e
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G    a
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L    s
 1    7                                                             all
 2
 3
 4
 5
 6
 7
 8
 9
 10
 11
 12
 13
 14
 15
```

5.4. IP Node Names

This section describes the steps for administering a node name in Communication Manager for Session Manager to be used in the configuration of the SIP signaling group.

Step	Description
1.	Use the **change node-names ip** command to add a new node name for Session Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name                IP Address
SM1                   10.1.2.170
default               0.0.0.0
procr                 10.1.2.160
```

5.5. SIP Signaling

This section describes the steps for administering a signaling group in Communication Manager for communication between Communication Manager and Session Manager.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">1.</td><td>Enter the command add signaling-group n, where n is an available signaling group and configure the following as shown in the display screen below:

Group Type – Set to sip.
Transport Method – Set to tls.
IMS Enabled – Set to y.
Near-end Node Name - Set to procr.
Near-end Listen Port - Defaults to 5061 for TLS.
Far-end Node Name - Set to the node name configured in Section 5.4.
Far-end Listen Port - Defaults to 5061 for TLS.
Far-end Network Region - Set to the Region configured in Section 5.3.
Far-end Domain - Set to avaya.com in this example. This should match the SIP Domain value configured in Session Manager.
Direct IP-IP Audio Connection – Set to y.
</td></tr>
<tr><td><pre>
display signaling-group 60
                                SIGNALING GROUP

 Group Number: 60   Group Type: sip
                        Transport Method: tls
  IMS Enabled? y

   Near-end Node Name: procr   Far-end Node Name: SM1
 Near-end Listen Port: 5061   Far-end Listen Port: 5061
                                        Far-end Network Region: 1
Far-end Domain: avaya.com

                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate   RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload   Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3   IP Audio Hairpinning? n
         Enable Layer 3 Test? n   Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n   Alternate Route Timer(sec): 6
</pre></td></tr>
</table>

5.6. SIP Trunking

This section describes the steps for administering a trunk group in Communication Manager for communication between Communication Manager and Session Manager.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">1.</td><td>Issue the command add trunk-group n, where n is an unallocated trunk group and configure the following as shown in the display screen below:

Group Type – Set to the Group Type field to sip.
Group Name – Enter any descriptive name.
TAC (Trunk Access Code) – Set to any available trunk access code.
Signaling Group – Set to the Group Number field value configured in Section 5.5. (i.e., 60)
Number of Members – Allowed values are between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

Note: Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunk members for the duration of the call. The license file installed on the system controls the maximum permitted.</td></tr>
<tr><td><pre>
display trunk-group 60   Page   1 of 21
                                TRUNK GROUP

Group Number: 60   Group Type: sip   CDR Reports: y
  Group Name: SM1   COR: 1   TN: 1   TAC: 160
   Direction: two-way   Outgoing Display? n
 Dial Access? n   Night Service:
Queue Length: 0
Service Type: tie   Auth Code? n

                                                         Signaling Group: 60
                                                       Number of Members: 20
</pre></td></tr>
</table>

5.7. Define System Features

This section describes the steps for administering system wide call features and options related to OPS in Communication Manager.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Use the change system-parameters features command to administer system wide features for the SIP telephones. Those related to features listed in Table 2 are shown in bold. These are all standard Communication Manager features.
<pre>
change system-parameters features   Page 17 of 18
                        FEATURE-RELATED SYSTEM PARAMETERS

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

WHISPER PAGE
   Whisper Page Tone Given To: paged

change system-parameters features   Page 18 of 18
                        FEATURE-RELATED SYSTEM PARAMETERS

IP PARAMETERS

                   Direct IP-IP Audio Connections? U
                             IP Audio Hairpinning? y

              SDP Capability Negotiation for SRTP? n
CALL PICKUP
  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
                   Enhanced Call Pickup Alerting? n
</pre></td></tr>
</table>

5.8. Define the Dial Plan

This section describes the steps for administering the dial plan in Communication Manager, including overall dial plan format, Feature Access Codes (FACs), and Feature Name Extensions (FNEs).

Step	Description
1.	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 2**, a Feature Access Code (FAC) must also be specified for the corresponding feature. In the sample configuration, telephone extensions are five digits long and begin with 3, FNEs are five digits beginning with 7, and the FACs have formats as indicated with **Call Type** "fac". Note that a FAC of "8" was used for AAR routing by a voice mail hunt group, the configuration for which is not included in these Application Notes. See Reference [10] for more information.

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

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     0              3   fac
     1              3   dac
     2              5   ext
     3              5   ext
     4              4   ext
     5              5   ext
     6              3   fac
     7              5   ext
     732           10   udp
     8              1   fac
     9              1   fac
     *              2   fac
     #              2   fac
```

2.	Use **change feature-access-codes** to define the access codes for the FNEs highlighted in red. The following screens have been abbreviated to highlight those FACs involved in supporting the FNEs and the AAR FAC.

```
change feature-access-codes                                 Page   1 of   9
                               FEATURE ACCESS CODE (FAC)
                         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


change feature-access-codes                                 Page   2 of   9
                               FEATURE ACCESS CODE (FAC)
                   Contact Closure  Pulse 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


change feature-access-codes                                 Page   3 of   9
                               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: 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


change feature-access-codes                                 Page   4 of   9
                               FEATURE ACCESS CODE (FAC)


                  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
```

3.	FNEs are 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 highlighted fields correspond to those features listed as supported in **Table 2**. 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: 70030
             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
  Conditional Call Extend Enable:
 Conditional Call Extend Disable:
             Conference Complete:
            Conference on Answer: 70011
           Directed Call Pick-Up: 70014
           Drop Last Added Party: 70015


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

       Exclusion (Toggle On/Off): 70016
      Extended Group Call Pickup: 70025
          Held Appearance Select: 70017
          Idle Appearance Select: 4000
             Last Number Dialed: 70019
            Malicious Call Trace: 70029
     Malicious Call Trace Cancel: 70021
             Off-Pbx Call Enable: 70027
            Off-Pbx Call Disable: 70028
                   Priority Call: 70000
                          Recall:
                  Send All Calls: 70001
           Send All Calls Cancel: 70002
               Transfer Complete:
             Transfer On Hang-Up: 70022
          Transfer to Voice Mail: 70023
        Whisper Page Activation: 70026
```

5.9. Specify Class of Service (COS) and Class Of Restriction (COR)

This section describes the steps for administering the COS and COR in Communication Manager, which affects what calling features and feature options are permitted for defined groups of telephone users.

Step	Description
1.	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 9600 Series 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  y  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  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding B/DA       n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Trk-to-Trk Transfer Override   n  y  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
```

2.	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 1. Note that Page 4 can be used to implement a form of centralized call screening for groups of stations and trunks.

```
change cor 1                                                  Page   1 of   23
                              CLASS OF RESTRICTION

               COR Number: 1
          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? y
     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             Add/Remove Agent Skills? 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 1                                                  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? y      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? n      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
```

5.10. SIP Stations

This section describes the steps for administering OPS stations in Communication Manager and associating the OPS station extensions with the telephone numbers of the Avaya Avaya 1100-Series IP Deskphones. The configuration is the same for all phones except for the desired number of call appearances as detailed in Step 3. Note that the corresponding users must be configured in Session Manager. There are two methods to sequence these steps:

1. Configure the station and off-PBX-station forms for each user in Communication Manager. Then configure the corresponding user in Session Manager, being sure to **check** the "Use Existing Stations" box (see **Section 6**).
2. Configure the user in Session Manager, being sure to leave the "Use Existing Stations" box **unchecked** (see Section 6. Session Manager will automatically generate the corresponding station and off-PBX-station information in Communication Manager. Then use the **change station** command in Communication Manager to add other configuration data, such as **Coverage Path**, **MWI Served User Type**, and additional call appearances, if needed.

Method 2 was used in the sample configuration. For method 1, perform the following steps for each user; then follow the steps in **Section 6**. For method 2, follow the steps in **Section 6** first; then use **change station n** to modify any station parameters as described below using the station form in this section as a guide.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Enter the add station n command, where n is an available extension in the dial plan, to administer an OPS station. On Page 1 of the form configure the following fields as shown in the display screen below:

Type – Set to 9630SIP.
Port – Leave blank. (Once the form is submitted, a virtual port is assigned, e.g., S00022)
Name – Enter any descriptive name.
Coverage Path – Enter the coverage path number defined for this telephone (e.g., for coverage to voice mail).

<pre>
add station 30043   Page   1 of   6
                                     STATION

Extension: 30043   Lock Messages? n   BCC: 0
     Type: 9630SIP   Security Code: TN: 1
     Port: Coverage Path 1: 60   COR: 1
     Name: Avaya 1165E   Coverage Path 2:   COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19
                                                Message Lamp Ext: 30043

        Display Language: english   Button Modules: 0

          Survivable COR: internal
   Survivable Trunk Dest? y   IP SoftPhone? n
</pre></td></tr>
</table>

2.	Proceed to **Page 2** of the form. Set **MWI Served User Type** to **sip-adjunct**.

```
add station 30043                                               Page   2 of   6
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe
          LWC Activation? y                        Coverage Msg Retrieval? y
                                                              Auto Answer: non
             CDR Privacy? n                             Data Restriction? n
                                               Idle Appearance Preference? n
                                             Bridged Idle Line Preference? n
   Bridged Call Alerting? n
  Active Station Ringing: single

        H.320 Conversion? n         Per Station CPN - Send Calling Number?
                                                     EC500 State: enabled

    MWI Served User Type: sip-adjunct

                                                Coverage After Forwarding? s

                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 30043        Always Use? n IP Audio Hairpinning? n
     Precedence Call Waiting? y
```

3.	Proceed to **Page 4** of the form and add the desired number of **call-appr** entries in the **BUTTON ASSIGNMENTS** section. This governs how many concurrent calls can be supported. Avaya 1100 Series IP Deskphones have the capability of handling 11 call appearances, but display only one local call appearance button when idle (see display in **Section 7.4** Step 3), so the number of entries shown below are not required to match that displayed on the telephone. Three are configured here to support conferencing scenarios.

```
add station 30043                                               Page   4 of   6
                                     STATION
 SITE DATA
       Room:                                                Headset? n
       Jack:                                                Speaker? n
      Cable:                                               Mounting: d
      Floor:                                            Cord Length: 0
   Building:                                              Set Color:

ABBREVIATED DIALING
     List1:                    List2:                    List3:



BUTTON ASSIGNMENTS
 1: call-appr                          5:
 2: call-appr                          6:
 3: call-appr                          7:
 4:                                    8:
```

4.	Enter the **change off-pbx-telephone station-mapping** command and configure the following as shown in the change screen below: • **Station Extension** – Set the extension of the OPS station as configured above. • **Application** – Set to **OPS**. • **Phone Number** – Enter the number that the SIP telephone will use for registration and call termination. In the example below, the **Phone Number** is the same as the **Station Extension**, though it is not required to be the same. • **Trunk Selection –** Set to **aar**.

	<pre>change off-pbx-telephone station-mapping 30043 Page 1 of 3
	                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION
	
	 Station Application Dial CC Phone Number Trunk Config Dual
	 Extension Prefix Selection Set Mode
	 30043 OPS - 30043 aar 1</pre>

5.	Repeat **Steps 1 - 4** as necessary to administer additional OPS stations and associations for the SIP telephones.

5.11. Routing

Step	Description
1.	Enter the **change aar analysis n** command, where **n** is the number to be routed; in this case 300 (matching any extensions starting with 300xx). On Page 1 of the form configure the following fields as shown in the screen below: • **Dialed String –** Set to **300**. • **Total Min/Max –** Set to **5** • **Route Patten -** Set to the appropriate route pattern, in this case **60**. • **Call Type –** Set to aar.

	<pre>change aar analysis 3 Page 1 of 2
	                            AAR DIGIT ANALYSIS TABLE
	                                  Location: all Percent Full: 0
	
	          Dialed Total Route Call Node ANI
	          String Min Max Pattern Type Num Reqd
	    300 5 5 60 aar n</pre>

2.	Enter the **change route-pattern n** command, where n is the route-pattern to be configured, in this case **60**. On Page 1 of the form configure the following fields as shown in the screen below: • **Pattern name** – Set to an appropriate name. • **Grp No –** Set to the trunk group being used, in this case **60** (see **Section 5.6**). • **FRL –** Set to **0** (lowest restriction, or a higher number if appropriate). • **No. Del Dgts -** Set to **0** (all digits are being sent).

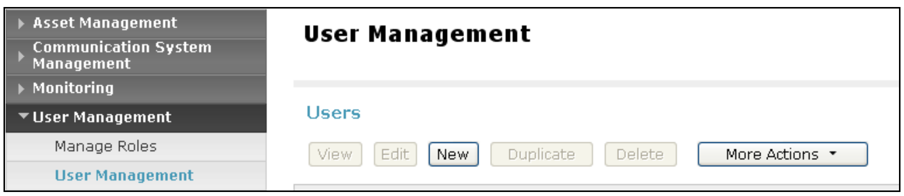
```
change route-pattern 60                                          Page   1 of   3
                    Pattern Number: 60  Pattern Name: SM FS
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 60   0                    0                                          n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                         none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
Precedence Call Waiting? y
```

6. Configure Avaya Aura™ Session Manager

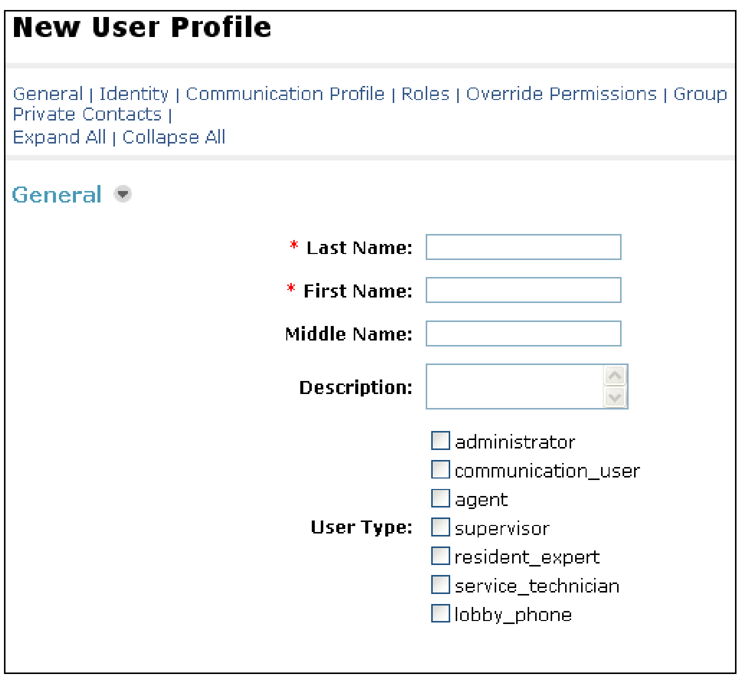
This section describes the administration of SIP telephones in Session Manager. It is assumed that a trunk has already been provisioned that matches the Communication Manager configuration in **Sections 5.5** and **5.6**. For additional references in configuring SIP trunking between Communication Manager and Session Manager see [4] and [6]. The following screens show a sample configuration for a SIP telephone whose extension is 30043. The same procedure should be followed for all such telephones.

Session Manager is configured via System Manager. Use a web browser and enter "https://<ip-address>/SMGR, where <ip-address> is the IP address of System Manager

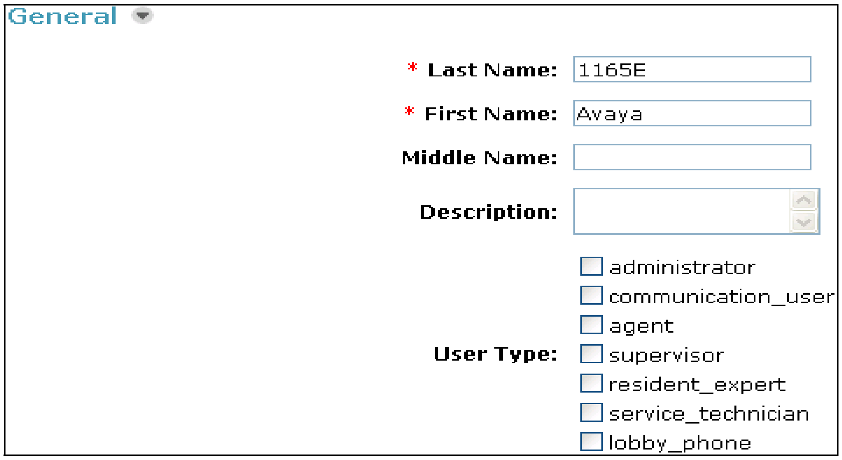
Log in using the appropriate credentials. On the main configuration page, select **User Management** under **User Management**, and click **New** to administer a new telephone user.



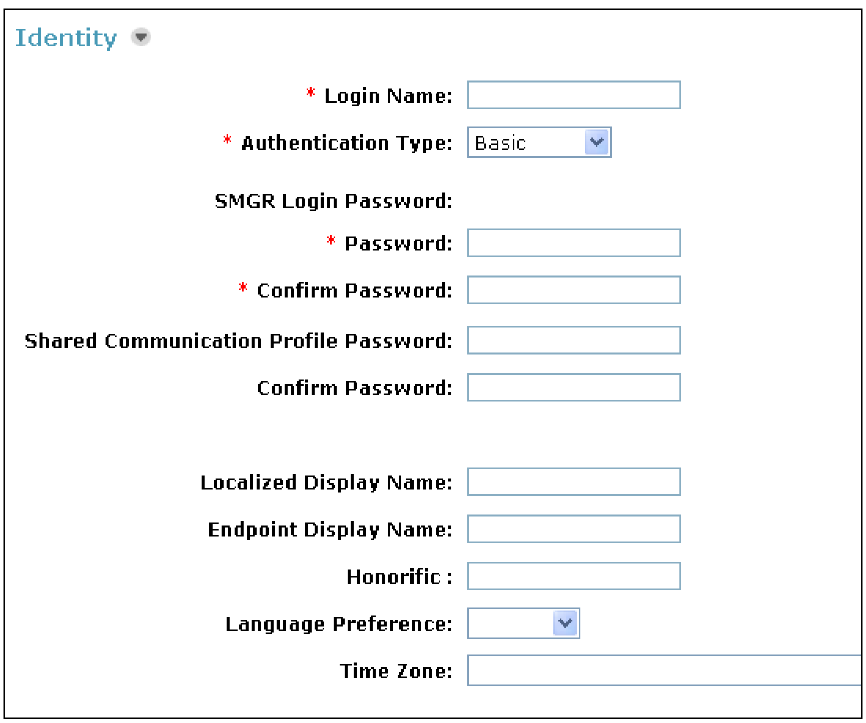
This will create a new User Profile. In the General section, enter a **Last Name** and **First Name**. Note that fields market with * are required to be filled in.



The following screen shows what was entered for extension 30043.

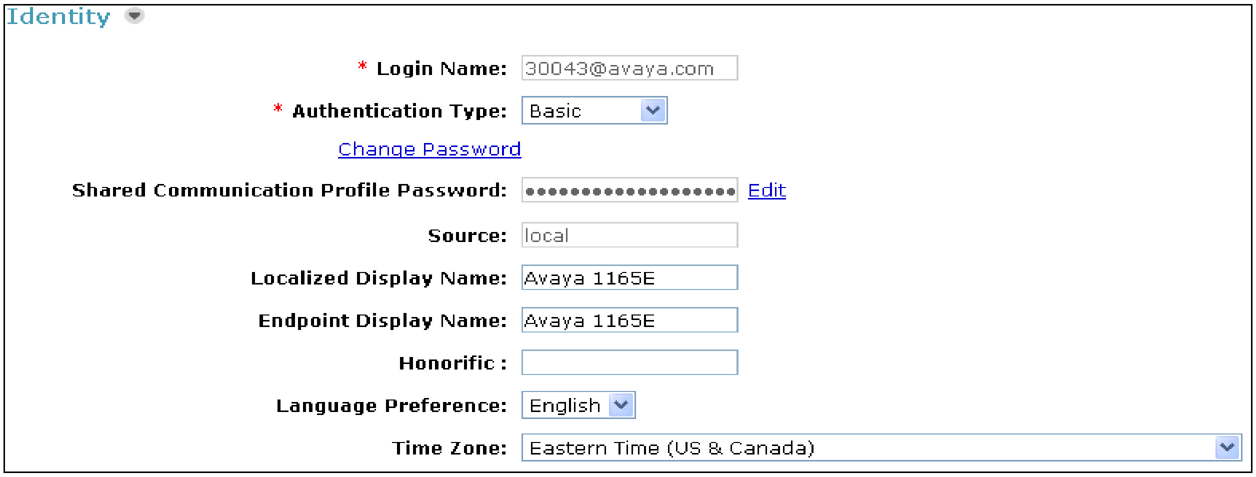


In the Identity section, enter a Login Name, for example 30043@avaya.com, and the required passwords, as shown on the next page. Note that the **Shared Communication Profile** password is the one the telephone is required to use when registering to Session Manager. It is also recommended to enter the display names. The L**ocalized Display Name** is what is displayed on a telephone when a call is made.[1] **SMGR Login Password**, while required, was not used in this sample configuration, and can be any value.

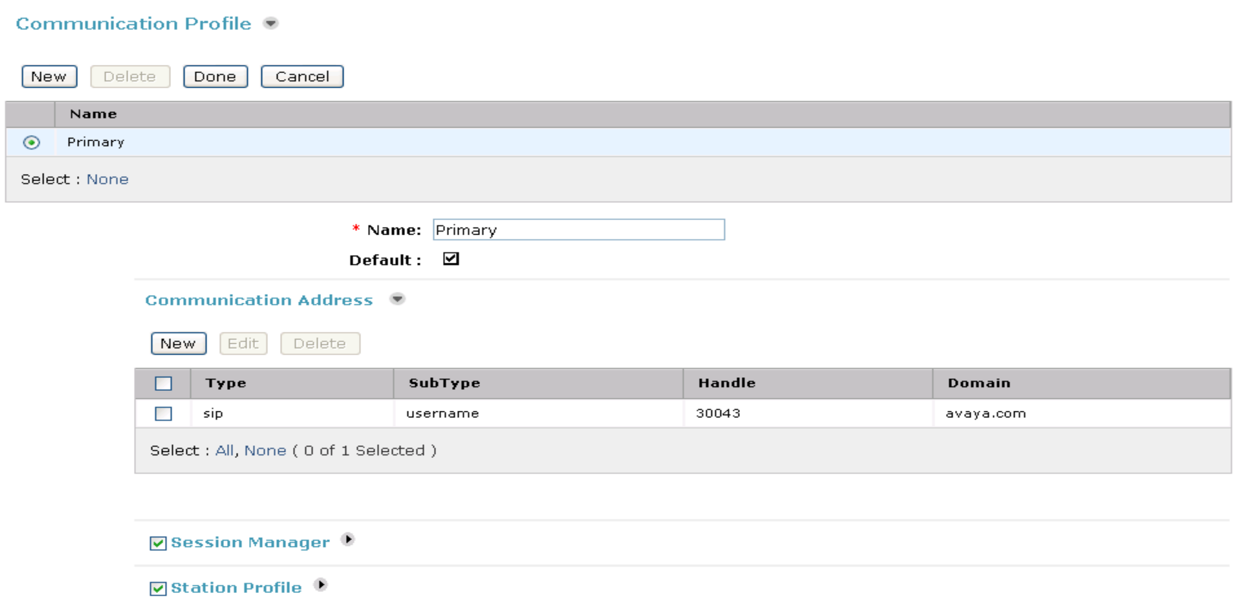


[1] When using Method 2 to configure telephone users (see **Section 5.10**), Session Manager uses this field to populate the **Name** field in the station form in Communication Manager.

The information below is what was entered for extension 30043. Note that the passwords are not displayed when viewing an endpoint's configuration.

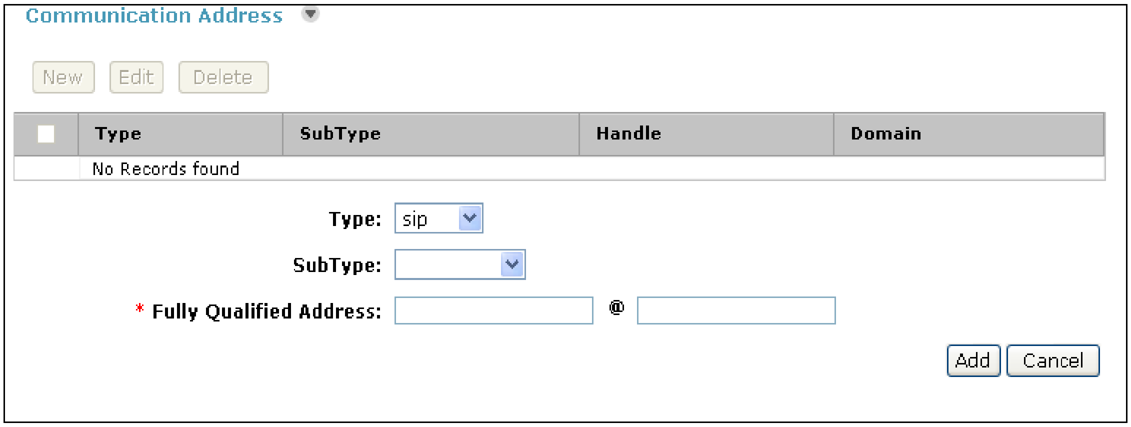


In the Communication Profile section, there are three sub-sections that need to be filled in: Communication Address, Session Manager, and Station Profile. Clicking on the arrow next to Communication Profile reveals the other sections.

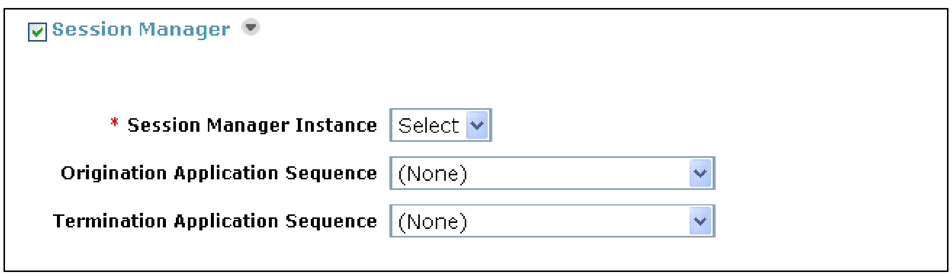


Click **New** under Communication Address.

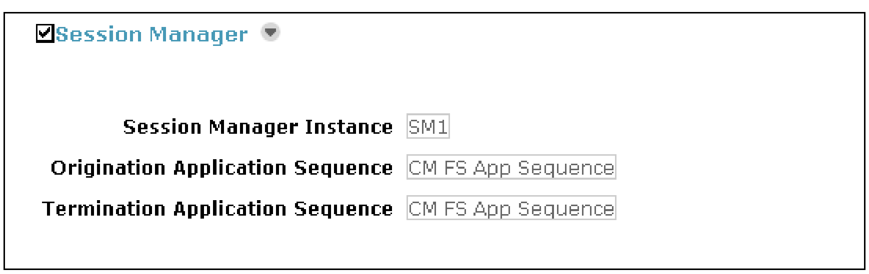
Set **Subtype** to “username”, and fill in the extension portion of the **Fully Qualified Address**, e.g., “30043”. “@avaya.com” will be automatically filled in. Then click **Add**. This will move the entry to the table as shown in the previous screen.



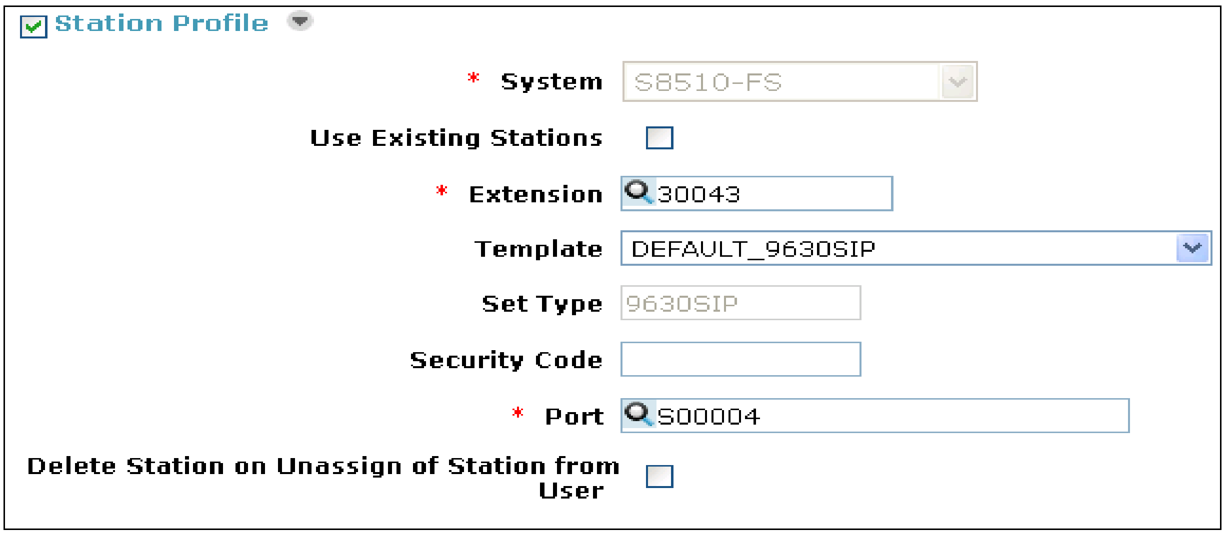
Click on the box next to **Session Manager**, and select the appropriate **Session Manager Instance** from the list. Select the appropriate **Origination** and **Termination Application Sequence**.



The screen below shows what was used for extension 30043.



Click on the box next to **Station Profile**, and enter the appropriate **System**, which is the Communication Manager Feature Server supporting the telephone. Check **Use Existing Stations** if using Method 1 (See **Section 5.10**), causing Session Manager to use the station previously entered in Communication Manager. Note that leaving this field un-checked will force System Manager to attempt to create the station in Communication Manager, and is used in Method 2. Enter an **Extension**, and select "DEFAULT_9630SIP" for the **Template**[2]. Leave the **Security Code** blank. Select "IP" for the **Port** field. The screen below shows what was used for extension 30043.



When done click 

at the bottom of the web page. Repeat the above steps for each telephone to be configured.

---

[2] This value for the **Template** applies for the 1120, 1140, and 1165 models.

7. Configure Avaya 1100-Series IP Deskphones

This section describes the basic configuration of the Avaya 1100-Series IP Deskphones. For additional details, see [8] available at http://www.avaya.com/.

Three models were tested: Avaya 1120E, 1140E, and 1165E. The configuration was done using configuration files and the local telephone screen interface, as shown in these Application Notes. The steps below show the configuration screens for the Avaya 1165E SIP Telephone. Configuration files can be used for most options to support mass deployments.

The configuration steps are similar for all three telephones, the main difference being the number of accounts or line appearances that each telephone supports. Make sure the number of lines used matches what is configured in Communication Manager.

7.1. Configure Initial Network Parameters

Network configuration of the telephone can be accomplished either manually at the telephone as shown below, or via DHCP. Once this is accomplished, configuration files can be used to configure the rest of the features. To manually configure the telephone, access the telephone screen interface by selecting **Prefs → Network,** starting with the **Prefs** soft key at the bottom of the screen. Enter the appropriate password to enter the network configuration submenus. Set appropriate values for IP address, mask, default gateway, file server address, and file server access type. In this case HTTP was selected as the configuration file server protocol. When the telephone boots, it will request the file "1165eSIP.cfg" from the root directory of the HTTP server, an abbreviated copy of which is shown below. This file instructs the telephone as to where to obtain its main configuration file (**`DEVICE_CONFIG`**), firmware (**`FW`**), and local dial plan file (**`DIALING_PLAN`**), used to determine end of dialing when making calls. Each section specifies the file name to be accessed and the protocol to be used with the file server. A value of "FORCED" as opposed to "AUTO" for the **`DOWNLOAD_MODE`** ensures explicit control over when files will be downloaded, and was used in the sample configuration.

```
[DEVICE_CONFIG]
DOWNLOAD_MODE    FORCED
VERSION          000100
PROTOCOL         HTTP
FILENAME         1165DeviceConfig.dat

[FW]
DOWNLOAD_MODE    FORCED
VERSION          SIP1165e03.02.05.20
PROTOCOL         HTTP
FILENAME         SIP1165e03.02.05.20.bin

[DIALING_PLAN]
DOWNLOAD_MODE    FORCED
VERSION          000020
PROTOCOL         HTTP
```

```
FILENAME        dialplan.txt
```

7.2. Configure Local Telephone Features

After the configuration file in the previous section has been downloaded, the telephone will attempt to download the files referenced. It will automatically upgrade to the firmware version specified if the firmware files are available at the file server. After that, the telephone will reboot and attempt to download the specified main device configuration and dial plan files. An annotated copy of the main device configuration file used in the sample configuration is shown below. The important parameters, their use, and the values used for the sample configuration are shown in **bold**.

```
# Device Config Version 00100
#------SIP domains
SIP_DOMAIN1 techtrial.com

# Multiple domains can be defined for login of the telephone
# The second domain corresponds to that used in the sample configuration
# and should match that configured in Communication Manager and Session
# Manager
SIP_DOMAIN2 avaya.com
SIP_DOMAIN3 abc.com
SIP_DOMAIN4 xyz.com
SIP_DOMAIN5 test.com

#------DNS domain
DNS_DOMAIN ca.avaya.com

#------Server IP addresses
SERVER_IP1_1 10.1.1.4
SERVER_IP1_2 10.1.1.4

# Specifies Session Manager as the SIP registrar for domain avaya.com
# A second address parameter specifies a backup registrar for failover (not
# tested)
SERVER_IP2_1 10.1.2.170
SERVER_IP2_2 10.1.2.170

SERVER_IP3_1 47.103.241.74
SERVER_IP3_2 47.103.241.74

SERVER_IP4_1 47.11.43.24
SERVER_IP4_2 47.11.43.24

SERVER_IP5_1 47.11.33.25
SERVER_IP5_2 47.11.33.25

#------UDP Port numbers
SERVER_PORT1_1 5060
SERVER_PORT1_2 5060
# UDP not used in the sample configuration
SERVER_PORT2_1 0
SERVER_PORT2_2 0
SERVER_PORT3_1 5060
```

```
SERVER_PORT3_2 5060
SERVER_PORT4_1 0
SERVER_PORT4_2 0
SERVER_PORT5_1 5060
SERVER_PORT5_2 5060

#------TCP Port numbers, 0 to disable
SERVER_TCP_PORT1_1 0
SERVER_TCP_PORT1_2 0
# TCP is used in the sample configuration
SERVER_TCP_PORT2_1 5060
SERVER_TCP_PORT2_2 5060
SERVER_TCP_PORT3_1 0
SERVER_TCP_PORT3_2 0
SERVER_TCP_PORT4_1 5060
SERVER_TCP_PORT4_2 5060
SERVER_TCP_PORT5_1 0
SERVER_TCP_PORT5_2 0

#------TLS Port numbers, 0 to disable, typically 5061 for TLS enabled.
SERVER_TLS_PORT1_1 0
SERVER_TLS_PORT1_2 0
# TLS not used in the sample configuration
SERVER_TLS_PORT2_1 0
SERVER_TLS_PORT2_2 0
SERVER_TLS_PORT3_1 0
SERVER_TLS_PORT3_2 0
SERVER_TLS_PORT4_1 0
SERVER_TLS_PORT4_2 0
SERVER_TLS_PORT5_1 0
SERVER_TLS_PORT5_2 0

#------Listening ports
SIP_UDP_PORT 5060
SIP_TCP_PORT 5060
SIP_TLS_PORT 0

#------Server retries
SERVER_RETRIES1 3
SERVER_RETRIES2 3
SERVER_RETRIES3 3

#--------- Device settings ---------------
# this command indicates which banner should be used-
# the one configured by the user or one from this file
#-----------------------------------------

#------Admin
ADMIN_PASSWORD 123456
ENABLE_LOCAL_ADMIN_UI YES
SECURE_UI_ENABLE NO
LOGOUT_WITHOUT_PASSWORD YES
SSH YES
SSHID 1234
SSHPWD 1234
```

```
SFTP Y
SFTP_READ_PATTERNS *.log, *.cfg
SFTP_WRITE_PATTERNS
PORT_MIRROR_ENABLE Yes
LOGSIP_ENABLE Yes


#------Recovery & Log levels
RECOVERY_LEVEL 2
LOG_LEVEL 255

#------Firmware update
AUTO_UPDATE YES
AUTO_UPDATE_TIME 0

#------Service Package
# Not supported in this configuration
ENABLE_SERVICE_PACKAGE NO

#------Service Package http or https
#SERVICE_PACKAGE_PROTOCOL HTTP

#------Banner
FORCE_BANNER YES
BANNER       Avaya

#------Autologin
AUTOLOGIN_ENABLE  YES

#------Enable/Disable SIP ping
SIP_PING YES

#------Time configuration
SNTP_ENABLE YES
SNTP_SERVER 10.1.1.21
TIMEZONE_OFFSET -18000
FORCE_TIME_ZONE No

#------VMAIL
VMAIL_DELAY 600

#------Expansion Module
EXP_MODULE_ENABLE YES

#------Address book mode - NETWORK, LOCAL, BOTH
ADDR_BOOK_MODE LOCAL

#------Mailbox entries
DEF_LANG English
MAX_INBOX_ENTRIES 100
MAX_OUTBOX_ENTRIES      100
MAX_REJECTREASONS 5
MAX_PRESENCENOTE  5
MAX_CALLSUBJECT         5
```

```
#------Instant Messaging
MAX_IM_ENTRIES 50
IM_MODE ENCRYPTED

#------- Enable IM blue LED
IM_NOTIFY YES

#------Bluetooth
ENABLE_BT YES

# Local Privacy feature disabled in favor of Calling Number Block FNE
# (see Section 4.2.3)
DISABLE_PRIVACY_UI Yes

#------VQMON configuration --------------
VQMON_PUBLISH NO
VQMON_PUBLISH_IP 10.1.1.120
#------------------------------------------
LISTENING_R_ENABLE No
LISTENING_R_WARN 80
LISTENING_R_EXCE 60
PACKET_LOSS_ENABLE Yes
PACKET_LOSS_WARN 222
PACKET_LOSS_EXCE 300
JITTER_ENABLE Yes
JITTER_WARN 700
JITTER_EXCE 900
DELAY_ENABLE Yes
DELAY_WARN 400
DELAY_EXCE 800
SESSION_RPT_EN  Yes
SESSION_RPT_INT 61

#------Transfer, Hold, and conference.
TRANSFER_TYPE    STANDARD
HOLD_TYPE        RFC3261
ENABLE_3WAY_CALL YES
REDIRECT_TYPE    RFC3261

#------Maximum number of Multi user logins
MAX_LOGINS 6

#------E911
E911_USERNAME    911
E911_PASSWORD    1234
E911_PROXY       techtrial.com
E911_TXLOC       INVITE

#------USB port
ENABLE_USB_PORT  YES
USB_MOUSE        UNLOCK
USB_KEYBOARD     UNLOCK
USB_HEADSET      UNLOCK
USB_MEMORY_STICK UNLOCK
```

```
#------Enable UPDATE method
ENABLE_UPDATE    YES
ENABLE_PRACK     YES


#------SRTP_MODE can be (BE-2MLines/SecureOnly/BE-Cap Neg)
SRTP_ENABLED      NO
SRTP_MODE   BE-2MLines
SRTP_CIPHER_1     AES_CM_128_HMAC_SHA1_80
SRTP_CIPHER_2     AES_CM_128_HMAC_SHA1_32

#------Audio Codecs
AUDIO_CODEC1 G722
AUDIO_CODEC2 PCMU
AUDIO_CODEC3 G729
AUDIO_CODEC4 PCMA
AUDIO_CODEC5
AUDIO_CODEC6
AUDIO_CODEC7
AUDIO_CODEC8
G729_ENABLE_ANNEXB YES
# G723_ENABLE_ANNEXA YES

#------PROXY Checking
PROXY_CHECKING YES

#------File Manager
FM_CONFIG_ENABLE YES
FM_CERTS_ENABLE Y

#------DOD
DOD_ENABLE NO

#------DSCP Settings
DSCP_OAM                  18
DSCP_CONTROL              40
DSCP_MEDIA_FLASHOVERRIDE  41
DSCP_MEDIA_FLASH          42
DSCP_MEDIA_IMMEDIATE      44
DSCP_MEDIA_PRIORITY       45
DSCP_MEDIA                46

#------Session Timer Setttings
SESSION_TIMER_ENABLE      NO
SESSION_TIMER_DEFAULT_SE  1800
SESSION_TIMER_MIN_SE      1800
SET_REQ_REFRESHER         0
SET_RESP_REFRESHER        2

#------Hotline Service Settings
HOTLINE_ENABLE            NO
HOTLINE_URL               hotline

#------Login banner
LOGIN_BANNER_ENABLE NO
```

```
#------IPV6
IPV6_ENABLE_GUI NO
PREFER_IPV6     NO
IPV6_ENABLE     NO

#------Connection Keep Alive
#CONN_KEEP_ALIVE          120
#KEEP_ALIVE_TYPE    CRLF

#------NAT signaling
NAT_SIGNALLING            SIP_PING

#------Login Notify - Notifies user of previous logins
LOGIN_NOTIFY              YES
LOGIN_NOTIFY_WITH_TIME    YES

#------Screen Saver & Background image
SCRNSVR_ENABLE            YES
SCRNSVR_UNPRTCTD_ENABLE   YES
SCRNSVR_UPASS_ENABLE      YES
SCRNSVR_MODE              NO_PASS
SCRNSVR_IMAGE               screensaver3.jpg

BG_IMAGE_ENABLE           YES
BG_IMG_SELECT_ENABLE      YES
USE_BG_IMAGE              screensaver2.jpg

#------Fonts
OUTLINEFONT_ENABLE YES
FONTSMOOTH_ENABLE YES

#------Login default to alpha or numeric SIP URI
LOGINALPHA_ENABLE: 0

#------Enable the caller image display
CALLINFO_IMAGE_ENABLE     No

#------BLF
BLF_ENABLE          No

#------Automatically clear the new call message when entering inbox
AUTOCLEAR_NEWCALL_MSG     Yes

#------pclient control of set
ENABLE_ANSWER_MODE NO

#------End
```

7.3. 7.3 Configure Local Telephone Dial Plan

The telephone 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 an additional key to launch the call. The file is downloaded from the file server at boot time, and was specified as "dialplan.txt" in **Section 7.1**. An annotated copy of the file used in the sample configuration is shown below. Note that entries in the file correspond to dialing of 3xxxx (telephone users) as well as 7xxxx (FNEs) extensions and corresponds to the dial plan configuration in Communication Manager. There is also an entry for long distance dialing using the FAC “9” for ARS routing. Note that each entry allows for the telephone user to also press the “#” key to indicate that dialing is complete.

```
/* ------------------------------------------------------------------- */
/*                                                                     */
/* Avaya 1100-series IP Deskphone Dial Plan                               */
/*                                                                     */
/* ------------------------------------------------------------------- */
/* Domain used in the dialed URL of the SIP INVITE message */
$n="avaya.com"
$t=300

%%

/* DIGITMAP: 12 digits starting with 9 followed by an initial 1 */
(9[^1]x{10})|(9[^1]x{10})#     && sip:$$@$n;user=phone          && t=300

/* DIGITMAP: Extensions beginning with 3 (Telephone Users)*/
(3x{4})|(3x{4})#     && sip:$$@$n;user=phone          && t=300

/* DIGITMAP: Extensions beginning with 7 (FNEs) */
(7x{4})|(7x{4})#     && sip:$$@$n;user=phone          && t=300

/* End of Dial Plan */
```

7.4. Configure Speed Dial Buttons for Avaya Extended Feature Set

Additional Communication Manager features can be accessed by dialing the corresponding FNE. For example, if the telephone has been defined in Communication Manager as part of a pickup group, then dial the Call Pickup FNE (in this case 70010) to answer a call to any member of that group. Features that involve an existing call (e.g., Call Park) will require putting that call on hold, and placing a new call using the appropriate FNE. Holding the existing call is done automatically by the telephone if another call is placed. 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.

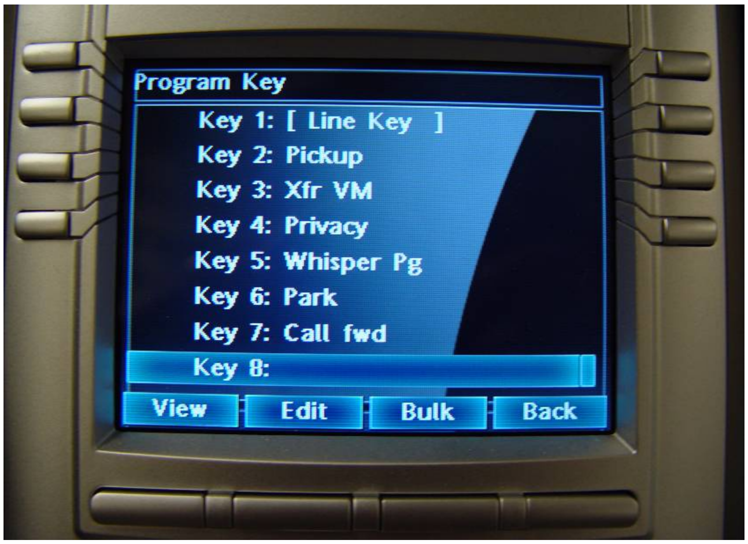
The following steps describe how to configure Avaya 1100-Series IP Deskphones with speed dial buttons. This technique is most useful with models that have many line appearance buttons, such as the 1140E and 1165E. **Section 7.4.1** shows how to manually configure speed dial buttons at each individual phone. For mass deployments, **Section 7.4.2** shows how the device configuration file and a speed dial list file can be used to support automatic configuration. Note that manually configured buttons will override automatically configured buttons at the same position. See Reference [8] for more details.

7.4.1. Manual Configuration

Steps	Description

<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>1.</td><td>Click on the More… soft key twice and click on Prefs. Navigate to Feature Options -> Feature Keys. Then select the desired line appearance key number (e.g., Key 8 as shown below) and click on Edit. The key numbers correspond to button positions on the left and right sides of the screen as follows (Key 1 is reserved for at least one line appearance).

<table>
<tr><th>Left</th><th>Right</th></tr>
<tr><td>8</td><td>4</td></tr>
<tr><td>7</td><td>3</td></tr>
<tr><td>6</td><td>2</td></tr>
<tr><td>5</td><td>1</td></tr>
</table>


</td></tr>
<tr><td>2.</td><td>A series of screen prompts will be presented. Respond with the following:

1. Select “1. Speed Dial” from the list button attributes.
2. Enter text for the button label at the “Enter a label:” prompt and select Next. In this example, “Fwd Cancel” was entered.
3. Enter the extension of the desired FNE at the “Enter address and press next” prompt and select Next. In this example “70007” was entered.
4. Select Next at the “Enter subject and press next” prompt.
5. Answer no to “Activate Auto-Retrieve of held call on hang up of speed dial call?”.
6. Select Back several times to show the main telephone screen. The new speed dial button should be displayed.</td></tr>
</table>

Steps	Description
3.	Access a Communication Manager feature via speed dial button by pressing the appropriate line button. 

7.4.2. Automatic (Mass) Configuration

Steps	Description
1.	Add the following line to the device configuration file for the corresponding phone type (e.g., 1165DeviceConfig.dat), where **SpeedDials.txt** will contain the speed dial button configuration data: **DEFAULT_CUSTOMKEYSFILE SpeedDials.txt**
2.	Create the file **SpeedDials.txt** with an entry for each speed dial button that is to be programmed. Set **index** to the key position number (see layout for the 1165E in Step 1 in **Section 7.4.1**), **label** to the desired text to be displayed at the button position, **target** to *FNE@domain*, where *FNE* is the extension of the FNE (see **Section 5.8** Step 3), and *domain* is the domain configured in Session Manager. The example below corresponds to the **Pickup** button configured for the 1165E, as displayed in Step 3 of **Section 7.4.1**). [key] index=2 label=Pickup target=70010@avaya.com type=spdial
3.	Reboot the phone, and it will automatically program the specified speed dial buttons.

8. Verification Steps

All features shown in **Table 2** were tested using the sample configuration. The following steps can be used to verify and/or troubleshoot installations in the field.

Step	Description
1.	After rebooting the telephone, use the **More** and **Prefs** soft keys at the phone to verify that the parameters set in the phone configuration file have been loaded. Verify registration with Session Manager by the appearance of the idle screen. If this is the first time registration is being attempted and multiple domains have been configured, enter the appropriate domain ("avaya.com" in the sample configuration). Verify that the line appearance shows the Communication Manager extension for that phone.
2.	Verify basic feature set administration by lifting the handset (or pressing the **speaker** button), and making calls to other phones. Test supported features according to **Table 2** and feature deployment plans at the site.
3.	Enter the **status trunk n** command, where **n** is the SIP trunk configured in **Section 5.6**. Note down the **Member** with **Service State** set to **in-service/active**. In this example, **0060/006** and **0060/007** are active and either member can be used to verify whether calls shuffled and which codec was used.

```
status trunk 60                                                             Page   1

                                TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0060/001 T00199   in-service/idle    no
0060/002 T00200   in-service/idle    no
0060/003 T00201   in-service/idle    no
0060/004 T00202   in-service/idle    no
0060/005 T00203   in-service/idle    no
0060/006 T00204   in-service/active  no   T00094
0060/007 T00205   in-service/active  no   T00063
```

4.	Enter **status trunk n,** where **n** is the member in active state as noted in the previous step for verification of codec used and shuffling status: • **Codec Type –** The codec used for Audio is **G.711MU** in this example. • **Shuffling** - If the **Near-end** and **Far-end** IP addresses for **Audio** belong to the Avaya 1100-Series IP Deskphones and the **Audio Connection Type** is **ip-direct**, it signifies that shuffling was successful. In this example, shuffling was successful.

```
status trunk 60/6                                              Page   2 of   3
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: 01A0017
  Signaling   IP Address                              Port
   Near-end:  10.1.2.160                            : 5060
    Far-end:  10.1.2.170                            : 5060
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:          H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-direct     Authentication Type: None
    Near-end Audio Loc:                       Codec Type: G.711MU
   Audio      IP Address                              Port
   Near-end:  10.1.2.143                            : 5058
    Far-end:  10.1.2.144                            : 5032

 Video Near:
  Video Far:
 Video Port:
      Video Near-end Codec:              Video Far-end Codec:
```

5.	Verify that speed dial buttons defined locally at the phone are displayed. If any are missing or are inoperative, check the local settings or the configuration file.
6.	Verify additional 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 Communication Manager for the correct FNE, proper permissions under COS/COR, and the proper station button assignment to support the feature.
7.	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.

9. Conclusion

These Application Notes have described the administration steps required to use Avaya 1100-Series IP Deskphones with SIP software with Session Manager, Communication Manager, and Modular Messaging. Basic, supplementary, and extended feature sets were covered. The extended set relies on Communication Manager Feature Server and Feature Name Extensions to support additional SIPPING features described in RFC 5359.

10. Additional References

Product documentation for Avaya products may be found at http://support.avaya.com/.

[1] *Administering Avaya Aura™ Communication Manager*, Release 5.2, Issue 5.0, May 2009, Document Number 03-300509.

[2] *Administering Network Connectivity on Avaya Aura™ Communication Manager*, Issue 14, May 2009, Document Number 555-233-504.

[3] *SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers*, Issue 9, May 2009, Document Number 555-245-206.

[4] *Administering Avaya Aura™ Session Manager*, Release 5.2, Issue 2.0, November 2009, Document Number 03-603324.

[5] *Avaya Aura™ Communication Manager Screen Reference*, Issue 1.0, May 2009, Document Number 03-602878.

[6] *Administering Avaya Aura™ Communication Manager as a Feature Server*, Release 5.2, Issue 1.2, January 2010, Document Number 03-603479.

[7] *Configuring 9600-Series SIP Phones with Avaya AuraTM Session Manager Release 5.2* — Issue 1.0, February 2010, Avaya Solution Interoperability Lab Application Notes.

[8] *SIP Software Release 3.0 for IP Phone 1165E Administration*.

[9] *Session Initiation Protocol Service Examples*, Internet Engineering Task Force, RFC 5259, available at http://www.ietf.org.

[10] *Modular Messaging Release 5.2 with Avaya MSS, Messaging Application Server (MAS) Administration Guide*, November 2009.

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.