Avaya Solution & Interoperability Test Lab

Configuring SIP IP Telephony Using Avaya SIP Enablement Services, Avaya Communication Manager, and Samsung SMT-i3010/3015 SIP Telephones – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to connect Samsung SMT-i3010/3015 SIP telephones to a SIP infrastructure consisting of Avaya SIP Enablement Services (SES) and Avaya Communication Manager running on an Avaya S8720 Media Server with an Avaya G650 Media Gateway. Also described is how Avaya Outboard Proxy SIP (OPS) station features can be made available to Samsung SIP telephones in addition to the standard features supported in the telephone. The configuration steps described are also applicable to other Linux-based Avaya Media Servers and Media Gateways running Avaya Communication Manager.

Information in these Application Notes has been obtained through Developer*Connection* compliance testing and additional technical discussions. Testing was conducted via the Developer*Connection* Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

1.1. Background

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

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

These Application Notes describe the configuration steps for using the Samsung SMT-i3010/3015 SIP telephones with the Avaya SES, S8720 Media Server and G650 Media Gateway. Only those configuration steps pertinent to interoperability of Samsung 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**. Several Samsung SMT-i3010/3015 SIP telephones are configured in a single subnet with an Avaya SES and S8720 Media Server with G650 Media Gateway. A PC provides web browser support. The telephones are registered to the Avaya SES and are also administered as OPS stations in Avaya Communication Manager, so that in addition to the SIP telephony features supported by the phones, OPS features are available from Avaya Communication Manager. The Avaya Modular Messaging Servers with SIP integration is providing Messaging Application for voice messaging support. These Application Notes do not address configuration of the Avaya 4620 SIP telephones, which were successfully tested using the standard product configuration steps.

The main difference between the Samsung SMT-i3010 and SMT-i3015 SIP telephones is the power supply supported by each type of phone. SMT-i3010 uses the external power adapter whereas the SMT-i3015 works on standard PoE. The configuration steps described in these Application Notes apply to both models. **Table 1** profiles the network management capabilities of the phones.

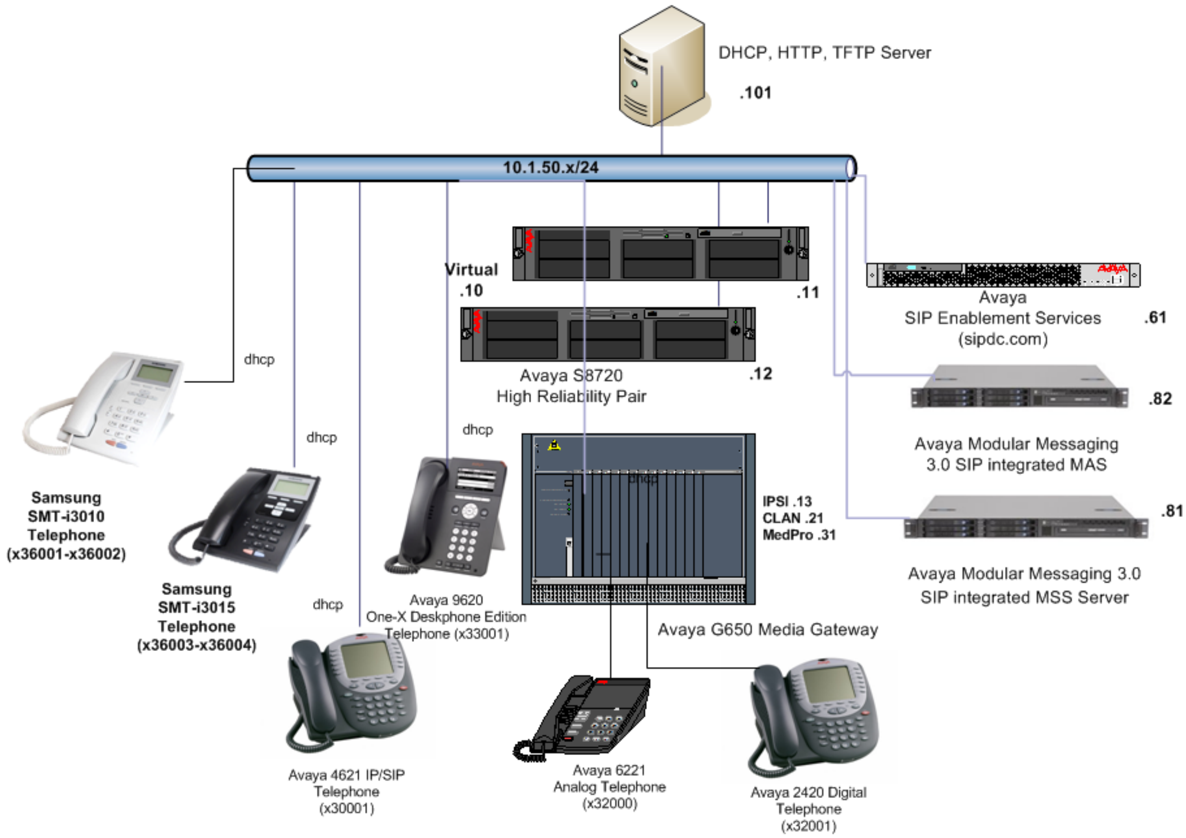


Figure 1: Avaya SIP Test Configurations with Samsung SIP Phones

Administration mechanisms	Web browser or soft menu on Phone
Administration levels	admin
File transfer server	HTTP, TFTP
Error logs	None
802.3af Power over Ethernet Support	Only SMT-i3015 supports PoE
SNMP support	None

Table 1: Network Management Capabilities of the Samsung SMT-i3010/3015

2. Equipment and Software Validated

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

Equipment	Software
Avaya SIP Enablement Services Server (SES)	3.1.1
Avaya P333T Modular Stackable Switch	4.5.14
Avaya S8720 Media Server with G650 Media Gateway	Avaya Communication Manager (3.1.2) S8720-013-01.2.632.1 SP 13149
Avaya Modular Messaging	3.0 SP2
Avaya 4620 SIP Phone	Firmware 2.2.2
Avaya one-X Deskphone Edition (9620 IP phone)	Firmware 1.2
Samsung SMT-i3010 and SMT-i3015	Version 1.00 dated 4th Apr 07

Table 2: Equipment and Software Versions Used

3. Supported Calling Features

3.1. The SIPPING-19

In addition to basic calling capabilities, the Internet Engineering Task Force (IETF) has defined a supplementary set of calling features, often referred to as the SIPPING-19 [2]. This provides a useful framework to describe product capabilities and compare calling features supported by various equipment vendors. **Table 3** gives a summary of calling features supported on the Samsung SMT-i3010/3015 SIP telephones within the Avaya SIP infrastructure. Some features are provided by the Samsung telephones, while others are provided by Avaya Communication Manager and OPS feature set.

Avaya OPS provides advanced calling features beyond the SIPPING-19 that can be extended to the telephone. These features are summarized in **Table 4**. Since the Samsung SMT-i3010/3015 SIP telephones are compatible with OPS, these features can be made available to the user.

The next few sections of these Application Notes describe the steps for configuring the Samsung telephone, Avaya SES and Avaya Communication Manager to support the extended feature (those indicated by a "YES" in the "With Avaya SIP Offer" column of **Table 3 and Table 4**).

Feature	SMT-i3010/3015		Comment
	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	
Call Waiting	YES	YES	
Do Not Disturb	YES	YES	
Speed Dial Buttons	YES	YES	
Message Waiting Support	YES	YES	
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	NO	YES	Via OPS FNE
Call Forward Busy	NO	YES	Via OPS FNE
Call Forward No Answer	NO	YES	Via OPS FNE
3 way conference – 3rd Party added	NO	NO	
3 way conference – 3rd Party joins	NO	NO	
Single Line Extension	NO	NO	
Find Me	NO	YES	Via Bridged Appearances
Incoming Call Screening	YES	YES	Via Local Setting or OPS COR
Outgoing Call Screening	YES	YES	Via Local Setting or OPS COR
Call Park/Unpark	NO	YES	Via OPS FNE
Call Pickup	NO	YES	Via OPS FNE
Automatic Redial	NO	YES	Via OPS FNE
Click to Dial	NO	NO	

Table 3: SIPPING-19 Telephony Feature Support

Feature	SMT-i3010/3015		Comment
	Locally at the Phone	With Avaya SIP Offer	
Advanced SIP Telephony Features via OPS			
Active Appearance Select	NO	NO	Not for OPS SIP users
Automatic Call-Back	NO	YES	Via OPS FNE
Automatic Call-Back Cancel	NO	YES	Via OPS FNE
Call Forwarding All	NO	YES	Via OPS FNE
Call Forwarding Busy/No Answer	NO	YES	Via OPS FNE
Call Forwarding Cancel	NO	YES	Via OPS FNE

Call Park	NO	YES	Via OPS FNE
Call Park Answer Back	NO	YES	Via OPS FNE
Call Pick-up	NO	YES	Via OPS FNE
Conference on Answer	NO	YES	Via OPS FNE
Calling Number Block	NO	YES	Via OPS FNE
Calling Number Unblock	NO	YES	Via OPS FNE
Directed Call Pick-Up	NO	YES	Via OPS FNE
Drop Last Added Party	NO	YES	Via OPS FNE
Exclusion (Toggle On/Off)	NO	NO	Not for OPS SIP users
Extended Group Call Pickup	NO	YES	Via OPS FNE
Held Appearance Select	NO	NO	Not for OPS SIP users
Idle Appearance Select	NO	YES	Via OPS FNE
Last Number Dialed	YES	YES	Via OPS FNE
Malicious Call Trace	NO	YES	Via OPS FNE
Malicious Call Trace Cancel	NO	YES	Via OPS FNE
Off-PBX Call Enable	NO	YES	Via OPS FNE
Off-PBX Call Disable	NO	YES	Via OPS FNE
Priority Call	NO	YES	Via OPS FNE
Send All Calls	NO	YES	Via OPS FNE
Send All Calls Cancel	NO	YES	Via OPS FNE
Transfer on Hang-Up	NO	YES	Via OPS FNE
Transfer to Voice Mail	NO	YES	Via OPS FNE
Whisper Page Activation	NO	YES	Via OPS FNE

Table 4: OPS Telephony Features Beyond SIPPING-19

3.2. Message Waiting Indicator (MWI)

With the OPS extended feature set, a SIP telephone that supports IETF RFC 3265 and MWI Draft 4 (Subscribe/Notify method) will illuminate/extinguish its MWI lamp when voice messages are left/read for that extension. Samsung SIP phones support unsolicited Notify method for MWI and this feature is also supported in Avaya Communication Manager.

3.3 Codec and Shuffling

Samsung SMT-i3010/3015 supports the following codecs on Avaya Communication Manager:

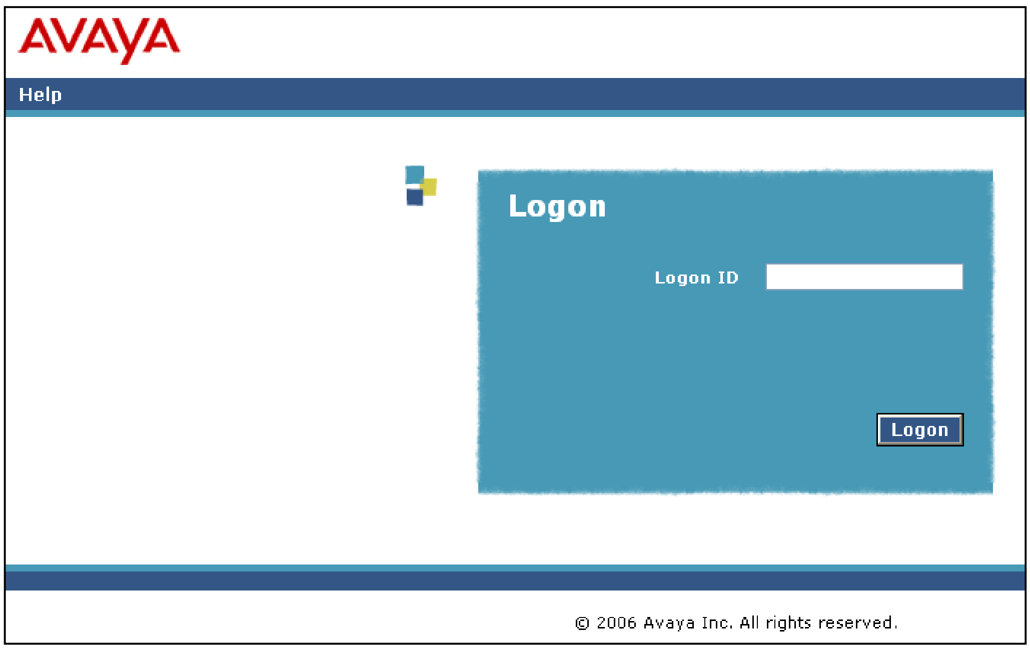
a. G.711Alaw
b. G.711Mulaw
c. G.729/A/B/AB

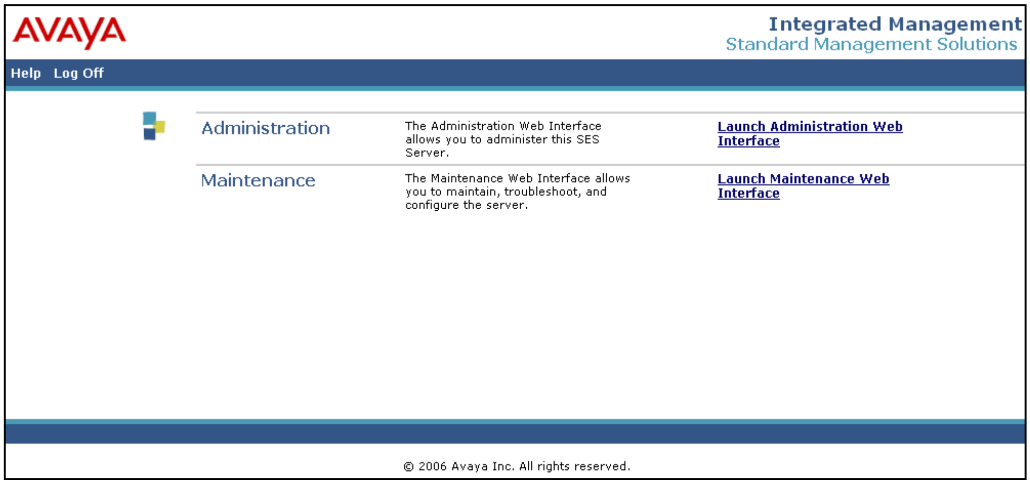
Samsung SMT-i3010/3015 also supports shuffling of their endpoints with Avaya 4600 series SIP and Avaya one-X Deskphone Edition Telephones. Note that shuffling between SIP and H323 endpoints is only supported from Avaya Communication Manager 3.1.2 onwards.

4. Configuring for the Avaya SES

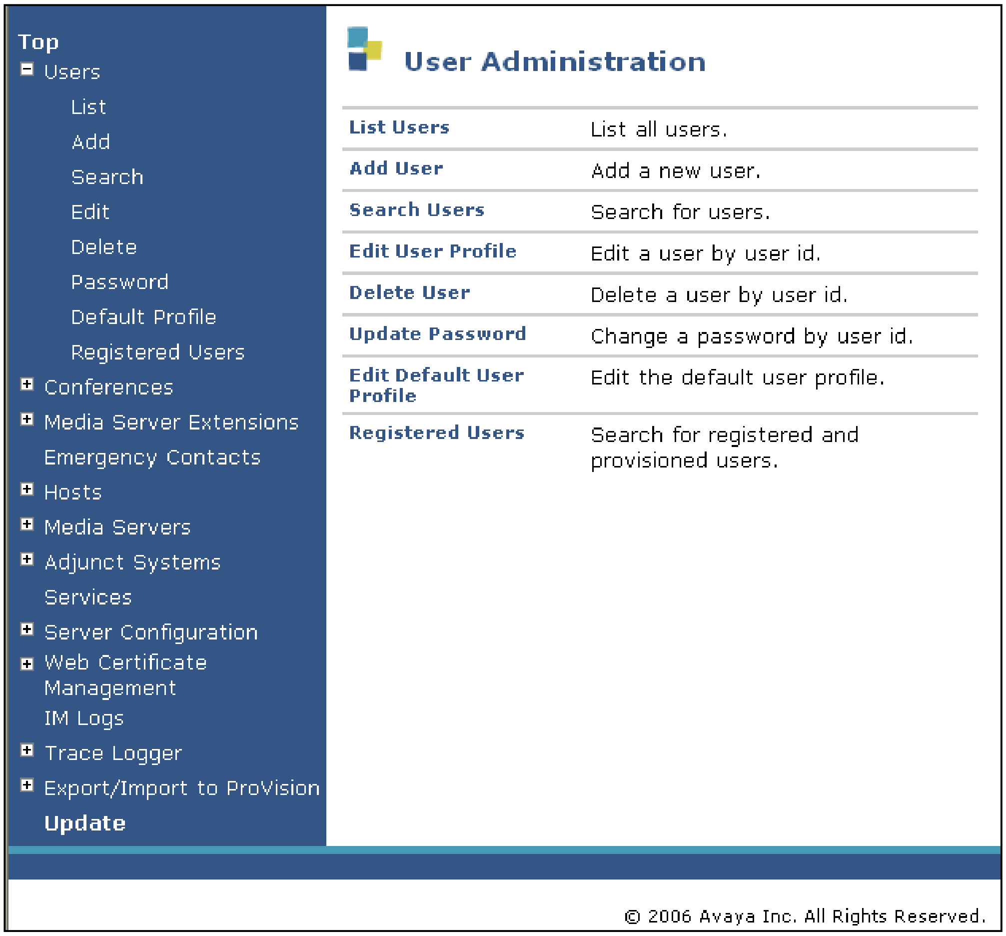
4.1. Administer Users on the Avaya SES

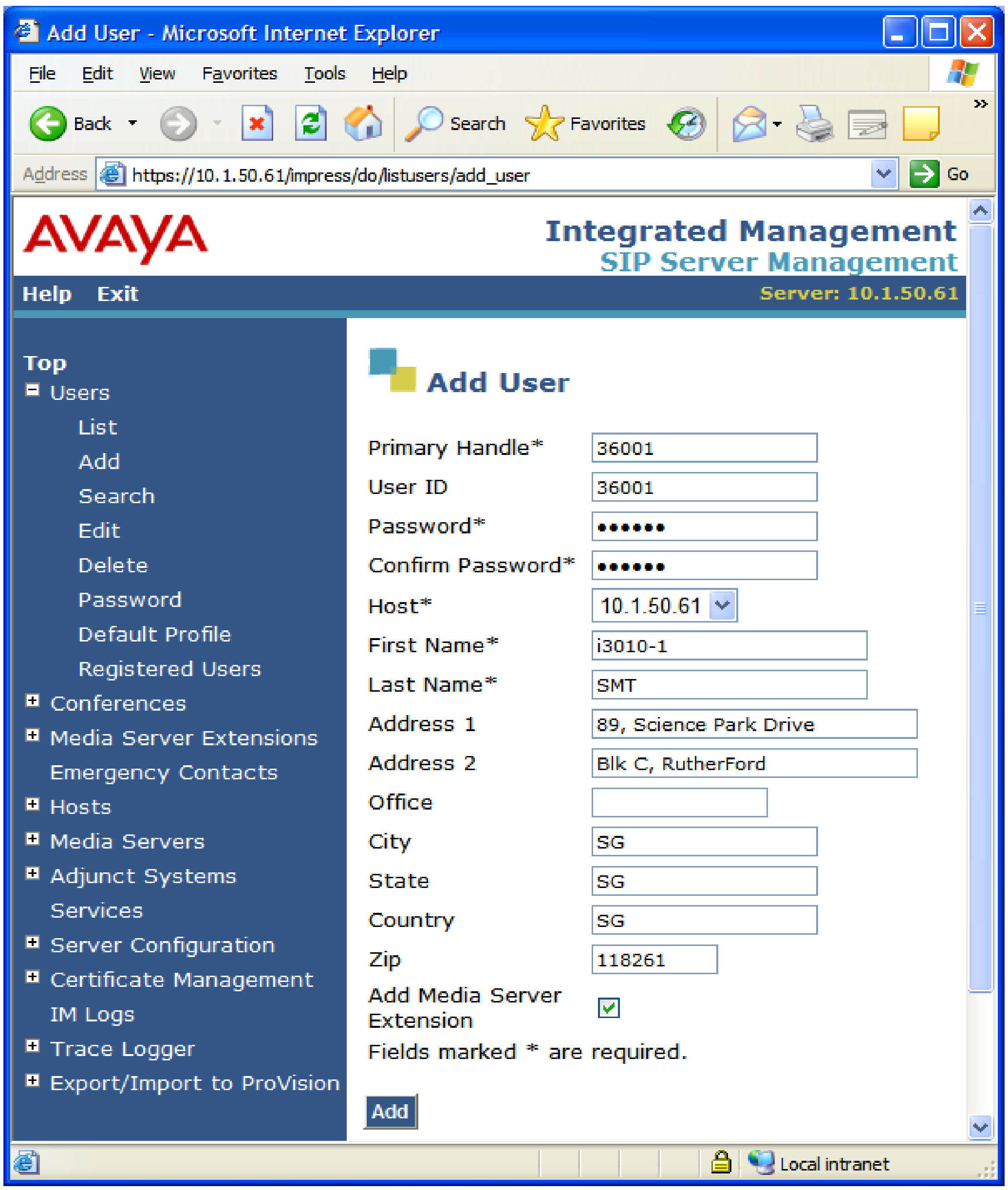
The following steps describe configuration of the Avaya SES to for use with Samsung SMT-i3010/3015 SIP telephones. Other standard administration functions are covered in Reference [1].

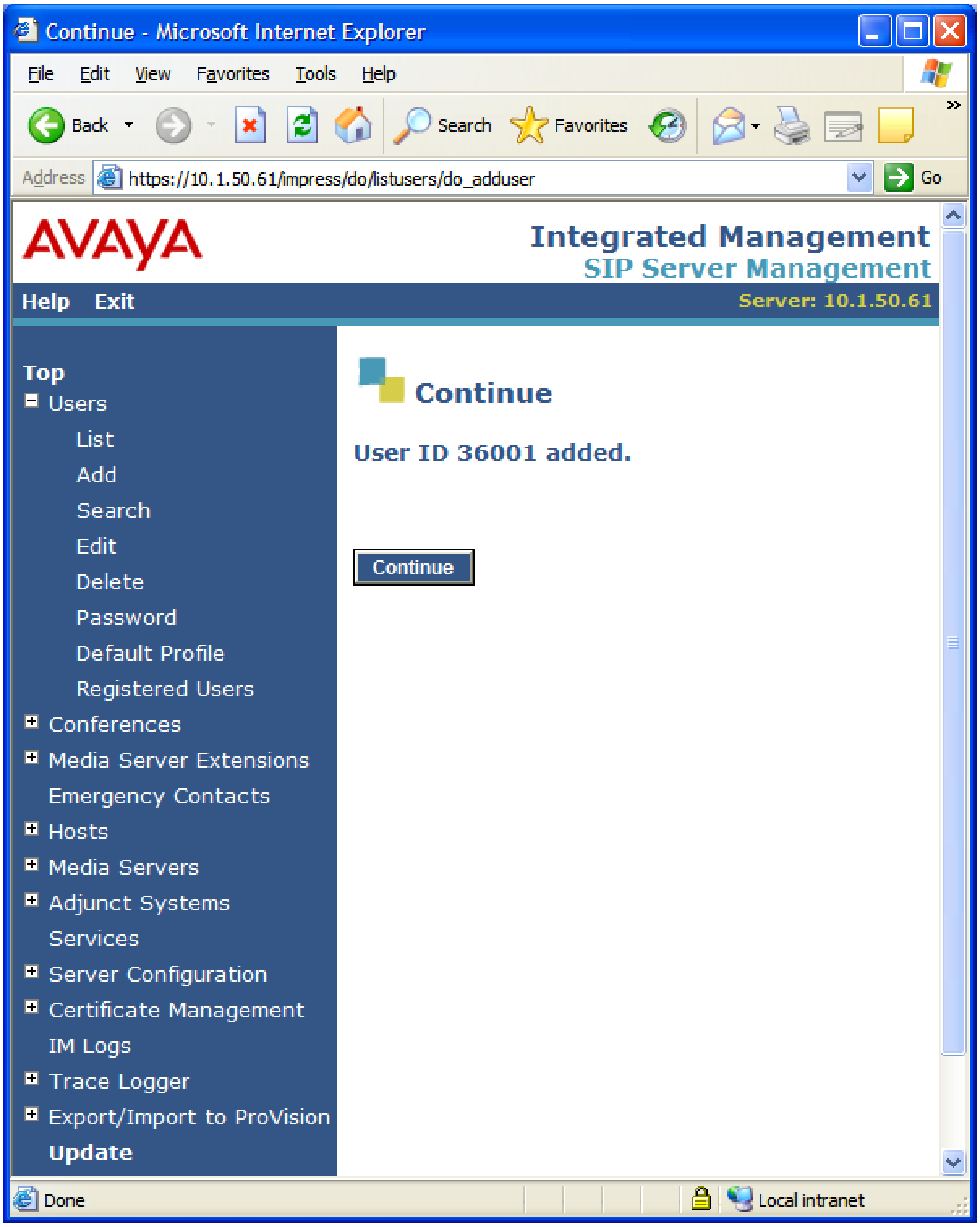
Steps	Description
1.	Avaya SIP Enablement Services is configured using a web browser. Set the URL of the browser to http://IP-address/admin, where IP-address is the IP address of the Avaya SIP Enablement Services Edge or Edge/Home Server, and log in with the appropriate permission. 

<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>2.</td><td>The main administration screen will be displayed after logging in. Click on Launch Administration Web Interface.

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

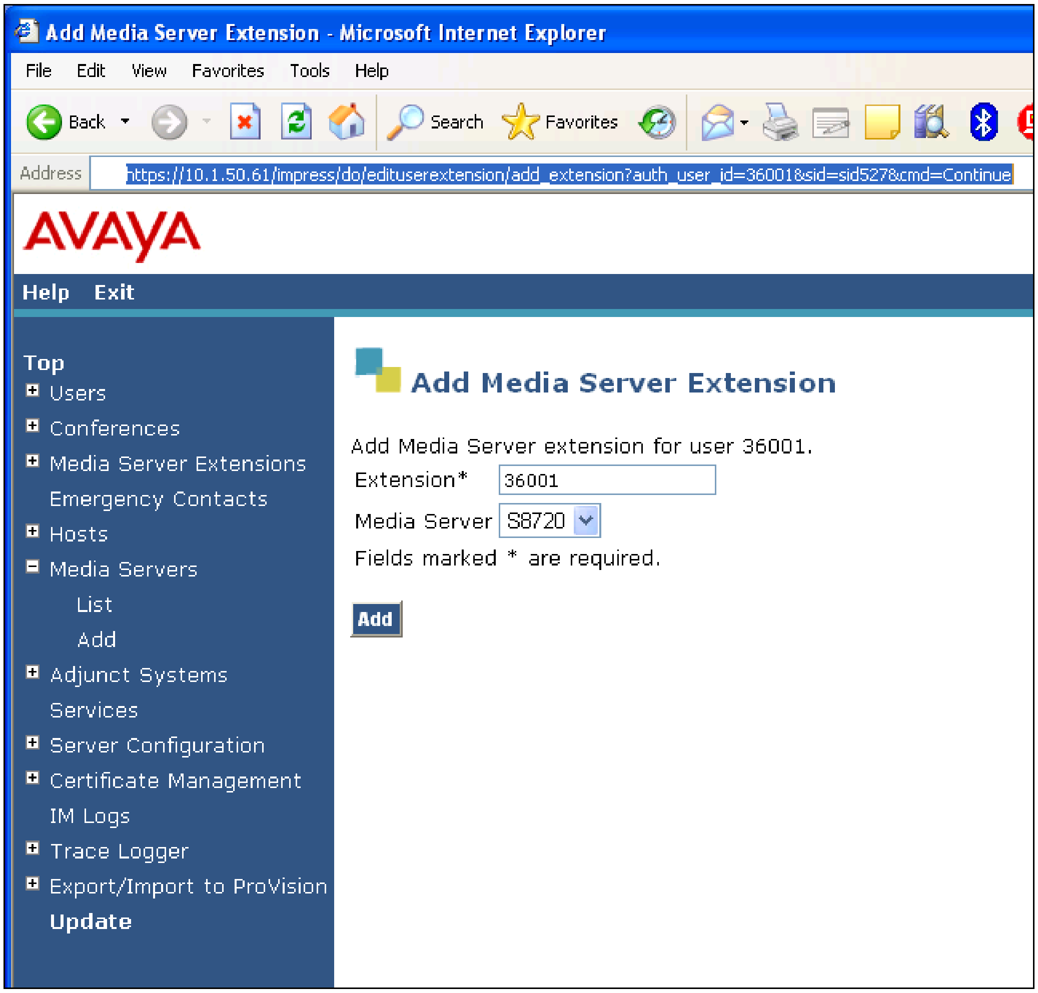
Steps	Description
3.	The SIP Enablement Services administration web interface will be displayed. Click on **Add** under the Users heading on the left side of the page.



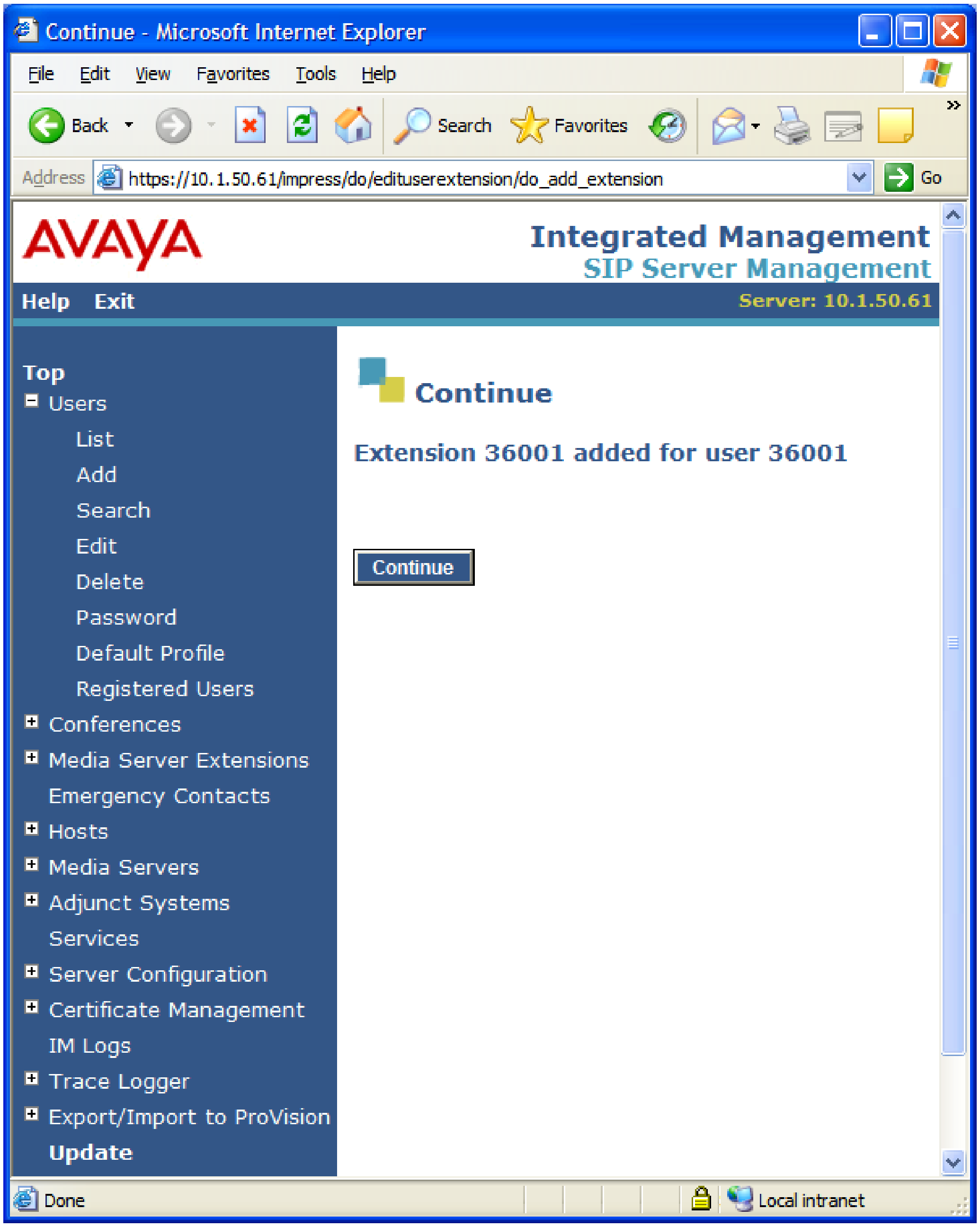
<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>4.</td><td>The Add User page will be displayed. Fill in the required fields (indicated by *). In the screen below, the user corresponding to a SIP telephone is being added. Enter the extension number in the Primary Handle field. The Host field should be set to the IP address 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 “10.1.50.61”. Check the Add Media Server Extension checkbox. Click on Add.

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

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

<table>
<tr><th>Steps</th><th>Description</th></tr>
<tr><td>6.</td><td>The Add Media Server Extension page will be displayed. Although not required, it is recommended that the same extension entered in Step 4 be entered in the Extension field. Select the Media Server as S8720. Click on Add.



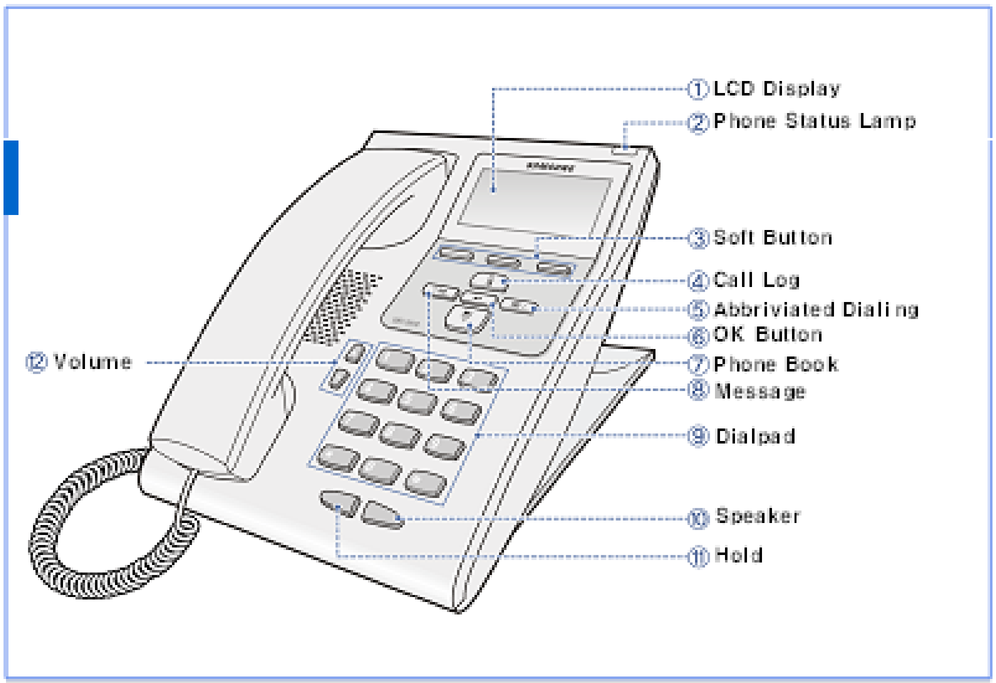
</td></tr>
</table>

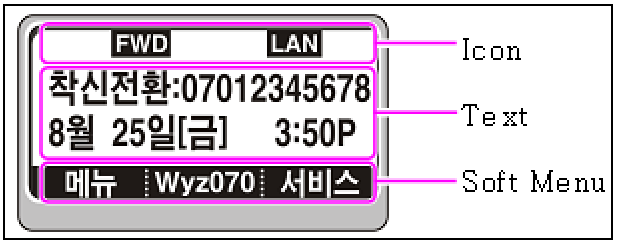
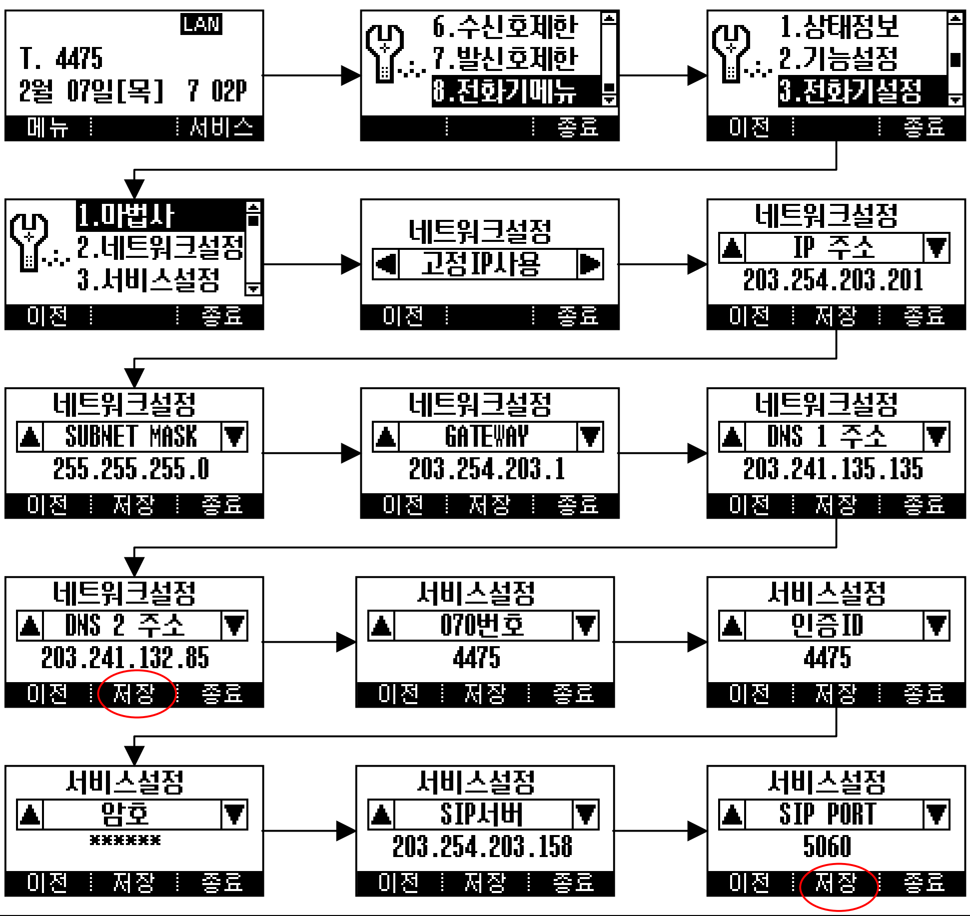
Steps	Description
7.	The confirmation page will be displayed. Click on **Continue.** 
8.	Repeat Steps 3-7 for each user to be added to the system.
9.	To apply the administration in the above steps, click on **Update** on the left side of the page. This link appears on the current page whenever updates are outstanding, and can be used at any time to save the administration performed

Steps	Description
	to that point.

5. Configure the Samsung SMT-i3010/3015 SIP Telephone

The Samsung SMT-i3010/3015 SIP Telephones only support Korean language at this moment. Therefore the language available for settings is only in Korean. The phone settings are done through the phone's soft menu as illustrated below. Further administration can be done through the web interface using the URL http://ipaddress:8000 by the default login "admin" and password "000000(six zeroes)". The feature name extension can be set via the soft menu on the phone or via the web interface.

Steps	Description
1.	**Button Assignment** 

2.	**Display Description** 
3.	**Installation Wizard** Menu(Left Soft Button) → 8. Phone Menu → 3. Phone Settings → 1. Wizard → Static IP → OK Button → IP Address → OK Button → Subnet Mask → OK Button → Gateway → OK Button → DNS 1 Address → DNS 2 Address → Save(Center Soft Button) → Extension Number → OK Button → User ID → OK Button → Password → OK Button → SIP Server Address → Save(Center Soft Button) → Left Navigation Button → OK Button → The phone will reboot. 

4.	**Web Administration Login** **http://ipaddress:8000, Default login: admin / password: 000000(six zeroes)** 

5. **Network Settings**

6. **Feature Name Extension Settings**

6. Configuring Avaya Communication Manager

The following administration steps are required on Avaya Communication Manager to support the Samsung SMT-i3010/3015 SIP telephones:

1. Verify system features and capacities required for SIP.
2. Define dial plan, feature access codes and feature name extensions for invoking extended features.
3. Define class of service, class of restriction, and a coverage path for the Samsung telephones.
4. Define stations corresponding to those specified on the Avaya SES and the corresponding off-PBX station mappings to route call requests involving those stations to the Avaya SES.

The following sections highlight the commands for defining SIP telephones as OPS stations on Avaya Communication Manager. For complete documentation, see Reference [1]. Use the System Access Terminal (SAT) interface to perform these steps. Log in with the appropriate permissions.

5.1.1 Verify OPS Capacity

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

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

     G3 Version: V13
       Location: 2                            RFA System ID (SID): 1
       Platform: 6                            RFA Module ID (MID): 1

                                                                USED
                                Platform Maximum Ports: 44000 219
                                      Maximum Stations: 36000 119
                              Maximum XMOBILE Stations: 100   0
                    Maximum Off-PBX Telephones - EC500: 100   3
                    Maximum Off-PBX Telephones -   OPS: 100   14
                    Maximum Off-PBX Telephones - SCCAN: 100   0



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

5.1.2 Define System Features

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

```
change system-parameters features                              Page   1 of  17
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
    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: ext   58001
              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  17
                        FEATURE-RELATED SYSTEM PARAMETERS
          Reserved Slots for Attendant Priority Queue: 5
                            Time before Off-hook Alert: 10
               Emergency Access Redirection Extension:
 Number of Emergency Calls Allowed in Attendant Queue: 5

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

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



                  Controlled Toll Restriction Replaces: none
```

```
change system-parameters features                               Page  17 of  17
                        FEATURE-RELATED SYSTEM PARAMETERS

INTERCEPT TREATMENT PARAMETERS
       Invalid Number Dialed Intercept Treatment: 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: all

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
```

5.1.3 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, a Feature Access Code (FAC) must also be specified for the corresponding feature. In the sample configuration, telephone extensions are five digits in length and begin with 3, FNEs are also five digits beginning with 31XXX, and the FACs have various formats as indicated with the **Call Type** of “dac”.

```
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
        1       5    ext
        3       5    ext
        5       5    ext
        8       1    fac
        9       1    fac
        *       3    dac
        #       3    dac
```

5.1.4 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   7
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code: *11
         Abbreviated Dialing List2 Access Code: *12
         Abbreviated Dialing List3 Access Code: *13
Abbreviated Dial - Prgm Group List Access Code: *14
                      Announcement Access Code: *15
                       Answer Back Access Code: *16
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 8
    Auto Route Selection (ARS) - Access Code 1: 9      Access Code 2:
                 Automatic Callback Activation: *17     Deactivation: #17
Call Forwarding Activation Busy/DA: *18    All: *19     Deactivation: #18
                         Call Park Access Code: *20
                       Call Pickup Access Code: *21
CAS Remote Hold/Answer Hold-Unhold Access Code: *22
                  CDR Account Code Access Code: *23
                        Change COR Access Code:
                   Change Coverage Access Code: *24
                   Contact Closure   Open Code:           Close Code:
                   Contact Closure  Pulse Code:
```

```
change feature-access-codes                                     Page   2 of   7
                               FEATURE ACCESS CODE (FAC)
                  Data Origination Access Code: *25
                      Data Privacy Access Code:
              Directed Call Pickup Access Code: *26
     Emergency Access to Attendant Access Code:
         EC500 Self-Administration Access Code: *27
                     Enhanced EC500 Activation: *28     Deactivation: #28
           Enterprise Mobility User Activation: *29     Deactivation: #29
  Extended Call Fwd Activate Busy D/A *30  All: *31     Deactivation: #30
        Extended Group Call Pickup Access Code: *32
               Facility Test Calls Access Code:
                             Flash Access Code:
             Group Control Restrict Activation:         Deactivation:
                    Hunt Group Busy Activation:         Deactivation:
                              ISDN Access Code:
             Last Number Dialed Access Code: *33
     Leave Word Calling Message Retrieval Lock: *34
   Leave Word Calling Message Retrieval Unlock: *35
             Leave Word Calling Send A Message: *36
           Leave Word Calling Cancel A Message: *37
               Malicious Call Trace Activation: *38     Deactivation: #38
```

```
change feature-access-codes                                     Page   3 of   7
                               FEATURE ACCESS CODE (FAC)
         Meet-me Conference Access Code Change:

 PASTE (Display PBX data on Phone) Access Code: *39
  Personal Station Access (PSA) Associate Code: *40    Dissociate Code: #40
        Per Call CPN Blocking Code Access Code: *41
      Per Call CPN Unblocking Code Access Code: #41
                  Posted Messages Activation: *42    Deactivation: #42
                  Priority Calling Access Code: *43
                         Program Access Code: #43

       Refresh Terminal Parameters Access Code: *44
              Remote Send All Calls Activation: *45    Deactivation: #45
               Self Station Display Activation:
                     Send All Calls Activation: *46    Deactivation: #46
         Station Firmware Download Access Code:
                       Station Lock Activation:        Deactivation:
      Station Security Code Change Access Code:
              Station User Admin of FBI Assign:                   Remove:
  Station User Button Ring Control Access Code:
             Terminal Dial-Up Test Access Code:
```

```
change feature-access-codes                                     Page   4 of   7
                               FEATURE ACCESS CODE (FAC)
Terminal Translation Initialization Merge Code:          Separation Code:
            Transfer to Voice Mail Access Code: *48
          Trunk Answer Any Station Access Code:
              User Control Restrict Activation:        Deactivation:
  Voice Coverage Message Retrieval Access Code:
 Voice Principal Message Retrieval Access Code:
           Whisper Page Activation Access Code: *47
```

5.1.5 Define Feature Name Extensions (FNEs)

The FNEs are 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: 31018
          Automatic Call Back: 31002              Last Number Dialed: 31019
   Automatic Call-Back Cancel: 31003            Malicious Call Trace: 31020
             Call Forward All: 31004     Malicious Call Trace Cancel: 31021
  Call Forward Busy/No Answer: 31005             Off-Pbx Call Enable: 31022
          Call Forward Cancel: 31006            Off-Pbx Call Disable: 31023
                    Call Park: 31007                   Priority Call: 31024
        Call Park Answer Back: 31008                  Send All Calls: 31025
                 Call Pick-Up: 31009           Send All Calls Cancel: 31026
         Conference on Answer: 31010             Transfer On Hang-Up: 31027
         Calling Number Block: 31011           Transfer to Voice Mail: 31028
       Calling Number Unblock: 31012         Whisper Page Activation: 31029
        Directed Call Pick-Up: 31013
        Drop Last Added Party: 31014
    Exclusion (Toggle On/Off):
   Extended Group Call Pickup: 31016
       Held Appearance Select:
```

5.1.6 Specify Class of Service (COS)

Use the **change class-of-service** command to set the appropriate service permissions to support features (shown in bold). In this 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 also supported on Samsung SIP phones.

```
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  y
 Data Privacy                   n  n  n  n  n  y  y  y  y  n  n  n  n  y  y  y
 Priority Calling               n  y  n  n  n  n  n  n  n  y  y  y  y  y  y  y
 Console Permissions            n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Off-hook Alert                 n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Client Room                    n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Restrict Call Fwd-Off Net      y  y  y  y  y  y  y  y  y  y  y  y  y  y  y  y
 Call Forwarding Busy/DA        n  y  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Personal Station Access (PSA)  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding All        n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Extended Forwarding B/DA       n  n  n  n  n  n  n  n  n  n  n  n  n  n  n  n
 Trk-to-Trk Transfer Override   n  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
```

5.1.7 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 1. Note that Page 3 can be used to implement a form of centralized call permissions for groups of stations and trunks.

```
change cor 1                                                    Page   1 of  22
                              CLASS OF RESTRICTION

               COR Number: 1
          COR Description: Local

                      FRL: 1                                      APLT? y
  Can Be Service Observed? n           Calling Party Restriction: none
Can Be A Service Observer? n            Called Party Restriction: none
        Time of Day Chart: 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            Hear VDN of Origin Annc.? n
         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 1                                                    Page   3 of  22
                              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? n      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? y      25? y      40? y      54? y      68? y      82? y      96? y
 11? y      26? y      41? y      55? y      69? y      83? y      97? y
 12? y      27? y      42? y      56? y      70? y      84? y      98? y
 13? y      28? y      43? y      57? y      71? y      85? y      99? y
 14? y      29? y
```

5.1.8 Add Coverage Path

Configure the coverage path to be used for the voice messaging hunt group, which is group h1 in the sample configuration. The default values shown for **Busy?**, **Don’t Answer?**, and **DND/SAC/Goto Cover?** can be used for the *Coverage Criteria*. In this case, the **Number of Rings** before the call goes to voice messaging has been extended from the default of 2 to 3 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: 3
                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:
```

5.1.9 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 Server. Use the default value for the **Station Type,** "IP" 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 telephones when receiving calls from this station. Use default values for the other fields on Page 1.

```
add station 36001                                                 Page   1 of   4
                                     STATION

Extension: 36001                          Lock Messages? n           BCC: 0
     Type: 4620                           Security Code: 12345        TN: 1
     Port: IP                           Coverage Path 1: 1           COR: 1
     Name: SMT-i3010-1                  Coverage Path 2:             COS: 1
                                        Hunt-to Station:

STATION OPTIONS
              Loss Group: 19            Personalized Ringing Pattern: 1
                                                    Message Lamp Ext: 36001
            Speakerphone: 2-way                  Mute Button Enabled? y
        Display Language: english                   Expansion Module? n
 Survivable GK Node Name:
          Survivable COR: internal                 Media Complex Ext:
   Survivable Trunk Dest? y                             IP SoftPhone? n



                                                 Customizable Labels? y
```

On Page 2, note the following:

- If this 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 type of the voice messaging system administered for this system in MWI Server User Type. In this case, the Avaya Modular Messaging Servers with SIP Integration is used.

```
add station 36001                                             Page   2 of   4
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe             Auto Select Any Idle Appearance? n
          LWC Activation? y                        Coverage Msg Retrieval? y
  LWC Log External Calls? n                                   Auto Answer: none
             CDR Privacy? n                              Data Restriction? n
   Redirect Notification? y                    Idle Appearance Preference? n
 Per Button Ring Control? n                  Bridged Idle Line Preference? n
   Bridged Call Alerting? Y                      Restrict Last Appearance? n
  Active Station Ringing: single         Conf/Trans on Primary Appearance? n
                                                       EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number? y
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
    MWI Served User Type: sip-adjunct          Display Client Redirection? n
                                              Select Last Used Appearance? n
                                                Coverage After Forwarding? s

                                            Direct IP-IP Audio Connections? y
 Emergency Location Ext: 36001     Always Use? n      IP Audio Hairpinning? y
```

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

- To support certain transfer and conference scenarios, the minimum number of "call-appr" buttons should be 3.

```
add station 36001                                               Page   3 of   4
                                     STATION
 SITE DATA
       Room:                                           Headset? n
       Jack: 223                                       Speaker? n
      Cable:                                          Mounting: d
      Floor:                                      Cord Length: 0
   Building:                                        Set Color:




ABBREVIATED DIALING
     List1: group    1         List2:                   List3:

BUTTON ASSIGNMENTS
 1: call-appr                           5: call-appr
 2: call-appr                           6: no-hld-cnf
 3: call-appr                           7: auto-cback
 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.

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, 5 line appearances were administered at the telephone for extension 36001. An Automatic Callback and the Conference On Answer FNE was included as function buttons.

Use the **change off-pbx-telephone station-mapping** command to map Avaya Communication Manager extension (36001) to the same SIP Enablement Services media server extension. Enter the field values shown. For the sample configuration, the **Trunk Selection** value indicates the “aar” which determines the SIP trunk group via the route pattern. The “aar” configuration will be explained in the later part of this section. The **Configuration Set** value can reference a set that has the default settings in Avaya Communication Manager.

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

 Station      Application   Dial   Phone Number      Trunk        Configuration
 Extension                  Prefix                   Selection    Set
 36001          OPS              - 36001              aar         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).

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

 Station       Call       Mapping      Calls        Bridged
 Extension     Limit      Mode         Allowed      Calls
 36001        5         both           all          both
```

The aar analysis table below shows how the SIP call is routed via **Route Pattern 3** which points to the SIP **Trunk Group 3**.

```
change aar analysis 3                                           Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                                            Percent Full:    0

          Dialed           Total     Route    Call    Node  ANI
          String          Min  Max  Pattern   Type    Num   Reqd
    3                     5    5     3        aar           n
```

```
change route-pattern 3                                          Page   1 of   3
                    Pattern Number: 3   Pattern Name: MM-SIP
                             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: 3    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 3 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
```

7. Verification Steps

1. After rebooting, use the web administration to verify that the parameters are correctly saved.
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 the feature list.
3. Verify that the soft menu for FNEs is defined locally at the phone and are displayed on the screen.

4. Verify OPS features by selecting soft menu 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 COR/COS, and the proper station button assignment to support the feature.
5. Call a telephone that currently has no voice messages and leave a message. Verify that the message-waiting indicator illuminates on the called telephone. Use voice messaging menu to retrieve and delete the voice message, verifying that DTMF is interpreted correctly by the system, and that the message waiting indicator extinguishes.

8. General Test Approach

The general approach to the testing is to test each of those features listed in Table 3 and 4 after setting up and verifying the Samsung SIP endpoints is working. Codecs and shuffling of the endpoints is also tested. Avaya SES is also rebooted to confirm if the Samsung SMT-i3010/3015 SIP Telephone is properly registered.

9. Conclusion and Results

These Application Notes have described the administration steps required to use Samsung SMT-i3010 and SMT-i3015 SIP telephones with the Avaya SES and Avaya Communication Manager. Samsung SMT-i3010/3015 SIP telephone is able to work with the listed Avaya SIP Offer solution features. It is also compatible with Avaya G.711 and G.729 codecs.

10. Additional References

[1] *Installing and Administering SIP Enablement Services R3.1.1*, Doc # 03-600768, Issue 2, August, 2006, available at http://support.avaya.com.

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

[3] *Avaya Extension to Cellular and Off-PBX Station (OPS) Installation and Administration Guide Release 3.0*, Doc. # 210-100-500, Version 6, Issue 9, June, 2005, available at http://support.avaya.com.

[4] *SIP Support in Release 3.1 of Avaya Communication Manager*, Doc # 555-245-206, Issue 6, February, 2006, available at http://support.avaya.com.

[5] *Technical Information on Samsung Electronic Product can be obtained from:*
Internet: http://www.samsungdocs.co.kr

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 Developer*Connection* Program at devconnect@avaya.com.