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Avaya Solution & Interoperability Test Lab

Configuring SIP IP Telephony Using Avaya SIP Enablement Services, Avaya Communication Manager, and Samsung SMT-W6100 WiFi SIP Telephones – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to connect Samsung SMT-W6100 WiFi 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 WiFi 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-W6100 WiFi 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**. Two Samsung SMT-W6100 WiFi SIP telephones are configured in a single subnet with an Avaya SES and S8720 Media Server with Avaya 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, Avaya AP-8 WiFi Access Point and SMT-R2000 Access Point which were successfully tested using the standard product configuration steps.

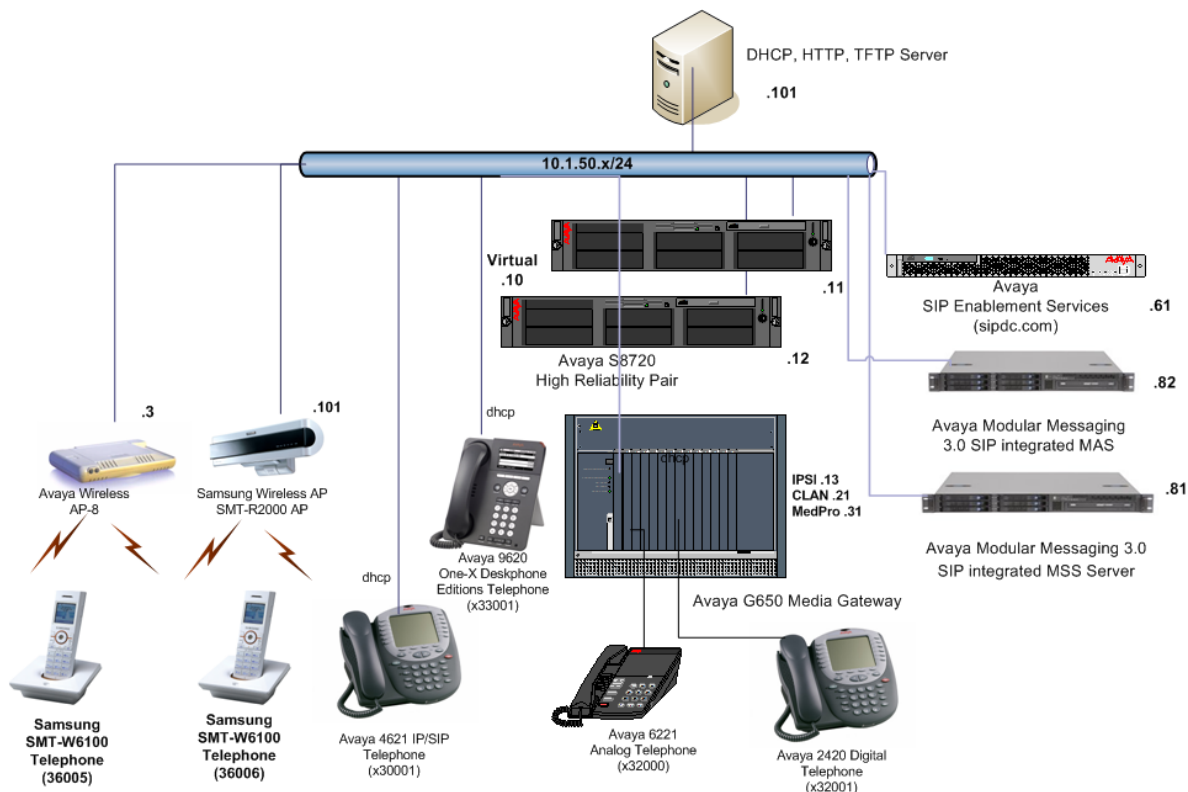**Table 1** profiles the network management capabilities of the phones.

Figure 1: Avaya SIP Test Configurations with Samsung WiFi SIP Phones

Administration mechanisms	Web browser or soft menu on Phone
Administration levels	Admin
File transfer server	HTTP, FTP
Error logs	Stored and viewed at phone
802.3af Power over Ethernet Support	No
SNMP support	None

Table 1: Network Management Capabilities of the Samsung SMT-W6100

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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 Avaya G650 Media Gateway	Avaya Communication Manager (3.1.2) S8720-013-01.2.632.1 SP 13149
Avaya Modular Messaging	3.0 SP2
Avaya Wireless AP-8	V3.1.0
Avaya 4620 SIP Phone	Firmware 2.2.2
Avaya one-X Deskphone Edition (9620 IP phone)	Firmware 1.2
Samsung SMT-W6100	Version 1.2 dated 29th Mar 07
Samsung SMT-R2000 Access Point	V2.03.00 dated 28th Dec 06

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-W6100 WiFi SIP telephones within the Avaya SIP infrastructure. Some features are provided by Samsung telephones, while others are provided by Avaya Communication Manager and the 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-W6100 WiFi 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

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extended feature (those indicated by a "YES" in the "With Avaya SIP Offer" column of **Table 3 and Table 4**).

Feature	Supported		Comments
	Locally at the Phone	With Avaya SIP Offer	
Basic Calling Features			
Extension to Extension call	YES	YES	
Basic call to Legacy Phones	NO	YES	
Intercept Tones/displays	YES	YES	
Call Waiting	YES	YES	
Do Not Disturb	YES	YES	Hard Button
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	NO	YES	Via OPS COR
Outgoing Call Screening	NO	YES	Via 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	Supported		Comments
	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 supported in Avaya Communication Manager.

3.3 Codec and Shuffling

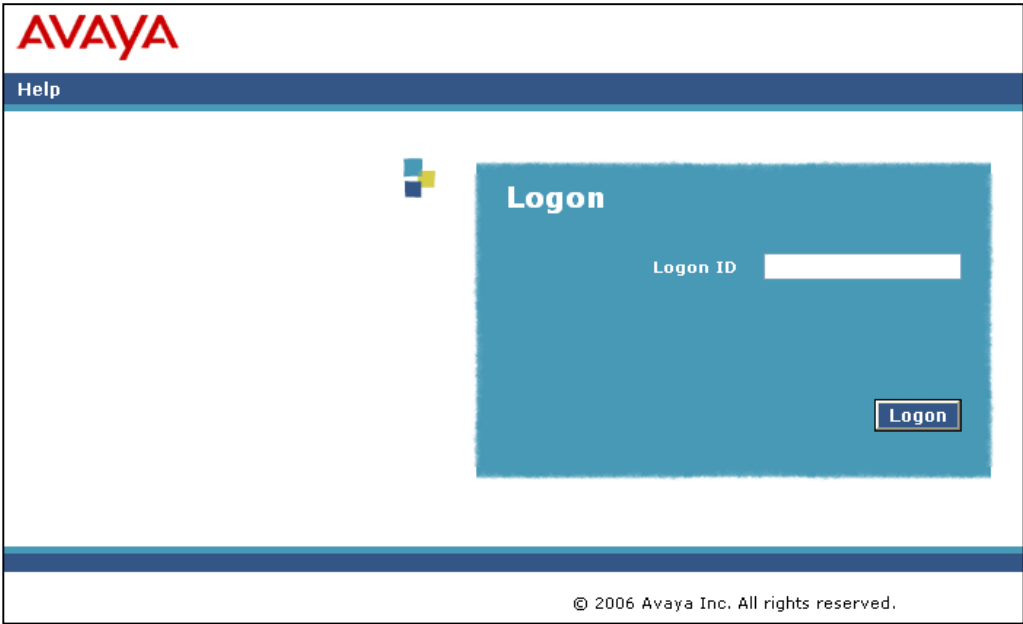Samsung SMT-W6100 supports the following codecs on Avaya Communication Manager:
   a. G.711Alaw
   b. G.711Mulaw
   c. G.729B/AB

Note that the WiFi handset is compatible with G.729B/AB on Avaya Communication Manager. Samsung SMT-W6100 also support shuffling of their endpoints with Avaya 4600 series SIP and Avaya one-X Deskphone Editions Telephones. Note that shuffling between SIP and H323 endpoints is only supported from Avaya Communication Manager 3.1.2 onwards.
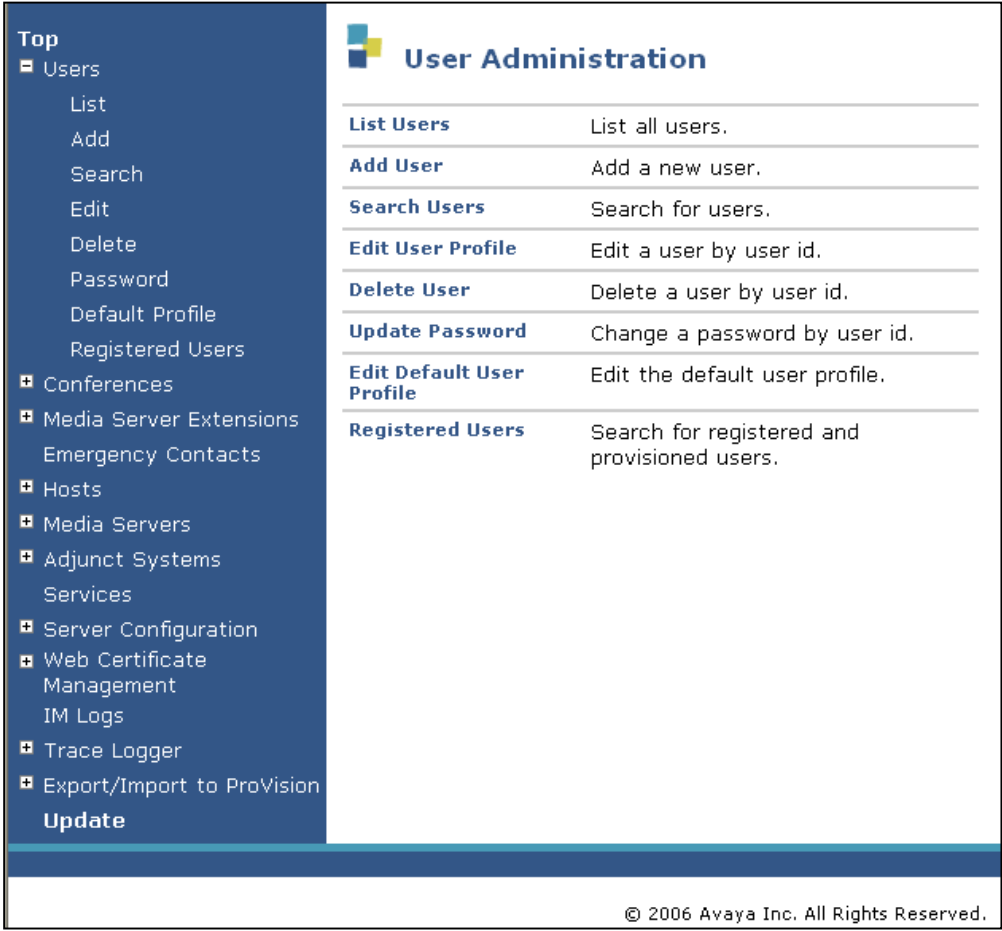
4. Configuring for the Avaya SES
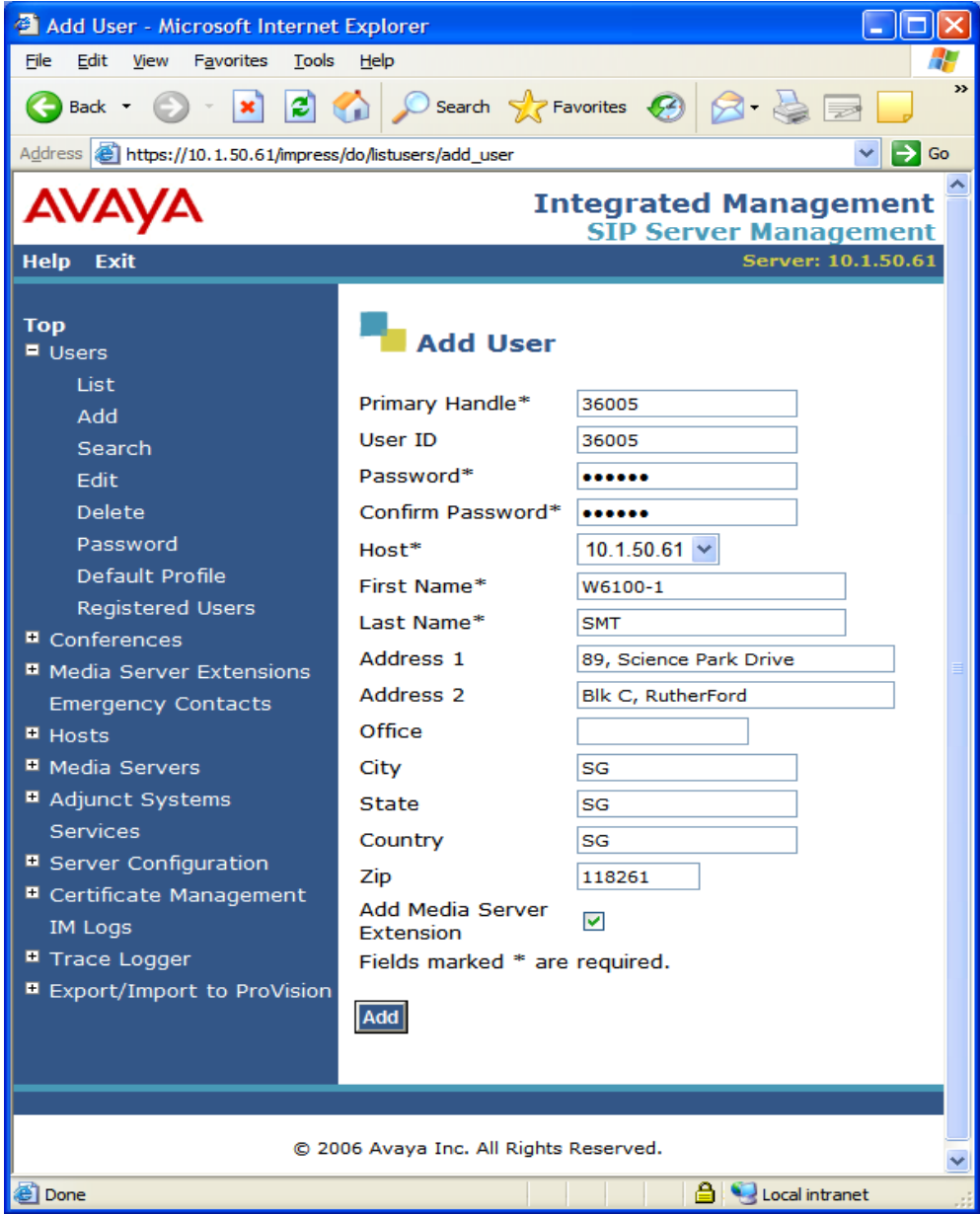
4.1. Administer Users on the Avaya SES
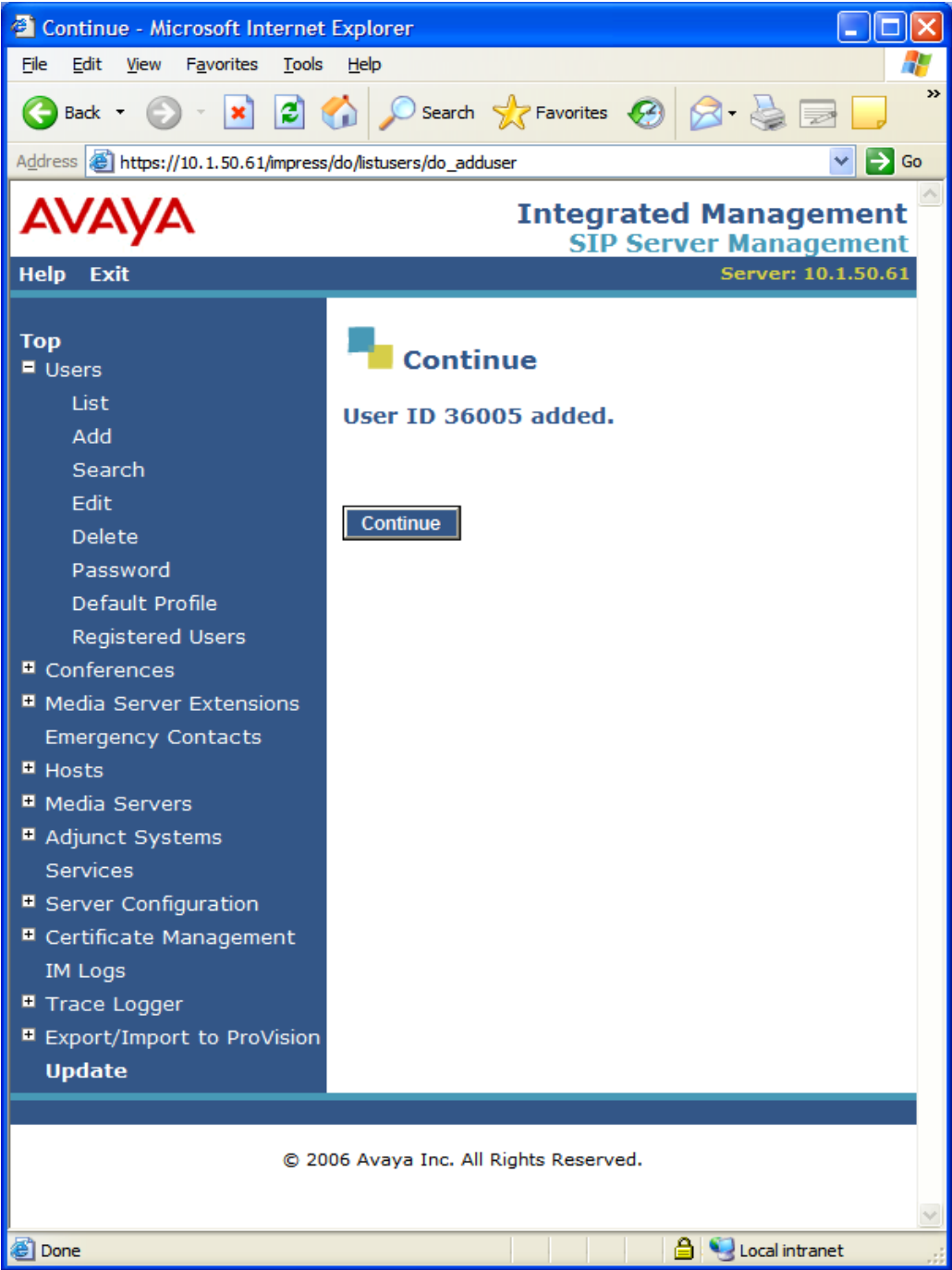
The following steps describe configuration of the Avaya SES to for use with Samsung SMT-W6100 WiFi SIP telephones. Other standard administration functions are covered in Reference [1].
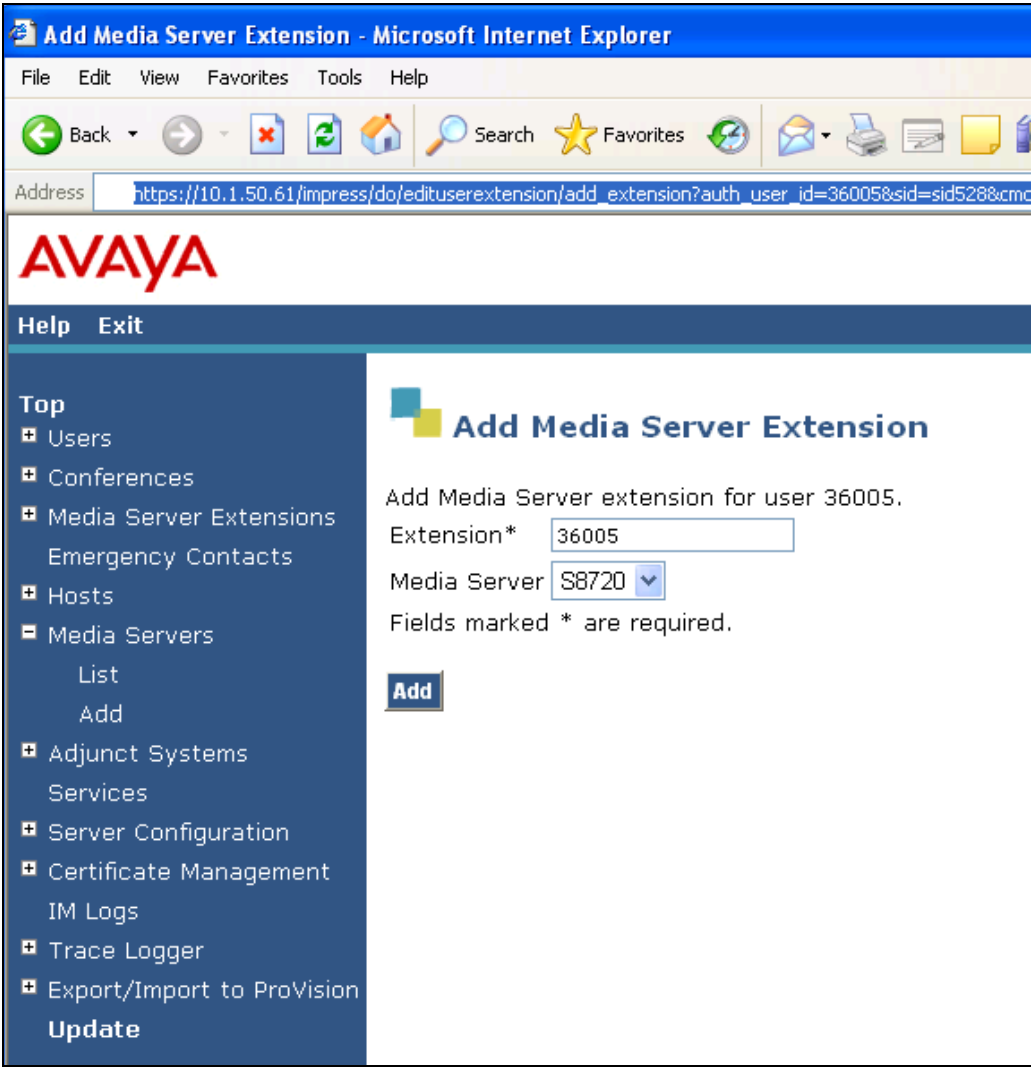
Steps	Description
1.	Avaya SIP Enablement Services is configured using a web browser. Set the URL of the browser to http://IP-address/admin, where IP-address is the IP address of the Avaya SIP Enablement Services Edge or Edge/Home Server, and log in using the appropriate permissions.
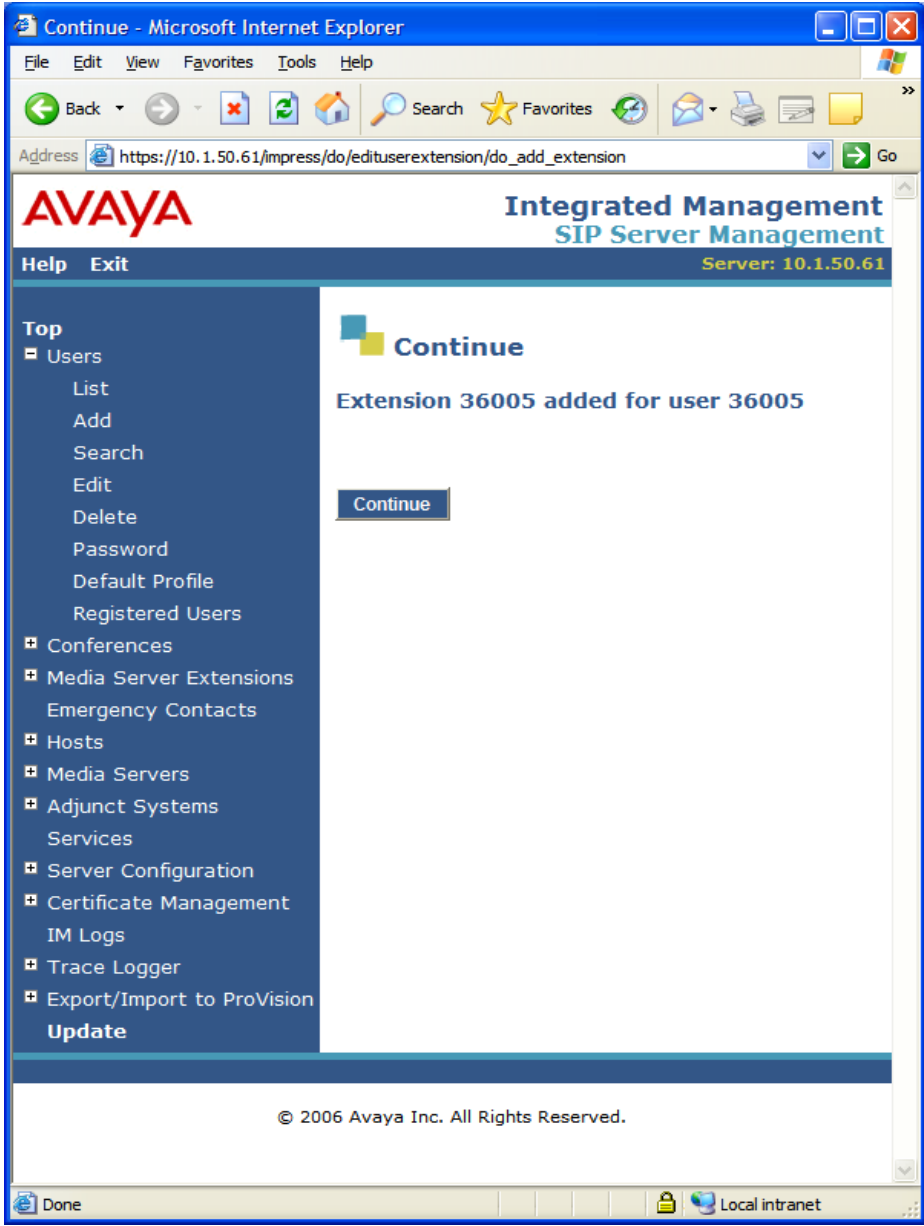
Steps	Description
2.	The main administration screen will be displayed after logging in. Click on **Launch Administration Web Interface**.

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

Steps	Description
4.	The Add User page will be displayed. Fill in the required fields (indicated by *). In the screen below, the user corresponding to a SIP telephone is being added. Enter the extension number in the **Primary Handle** field. The Host field should be set to the 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**.

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

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

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

5. Configure the Samsung SMT-W6100 WiFi SIP Telephone

The Samsung SMT-W6100 SIP Telephone settings are available in both English and Korean on the handset. The phone settings are done through the phone as illustrated below. Further administration can be done through the web interface using the URL http://ipaddress:8000 using the login admin and default password "admin" but only Korean language is available.

Steps	Description
1.	**Button Assignment** Speaker LCD Display Volume User Service Button Left Soft Button Send Cancle/Back/Correctio i Manner Mode Microphone Headset Jack Do Not Disturb Navigation Right Soft Button End/Power Speaker Phone Dial Pad Auto Lock (: Reserved) SMT-W6100

2.	**Network selection (WLAN)** Menu(Left Soft Button) → 5 Settings(or Click Right Soft Button 4-times) → 6 Network Settings → 1 Network Selection → 1 Scanning Network → Select desired WLAN SSID → OK(Left Soft Button)
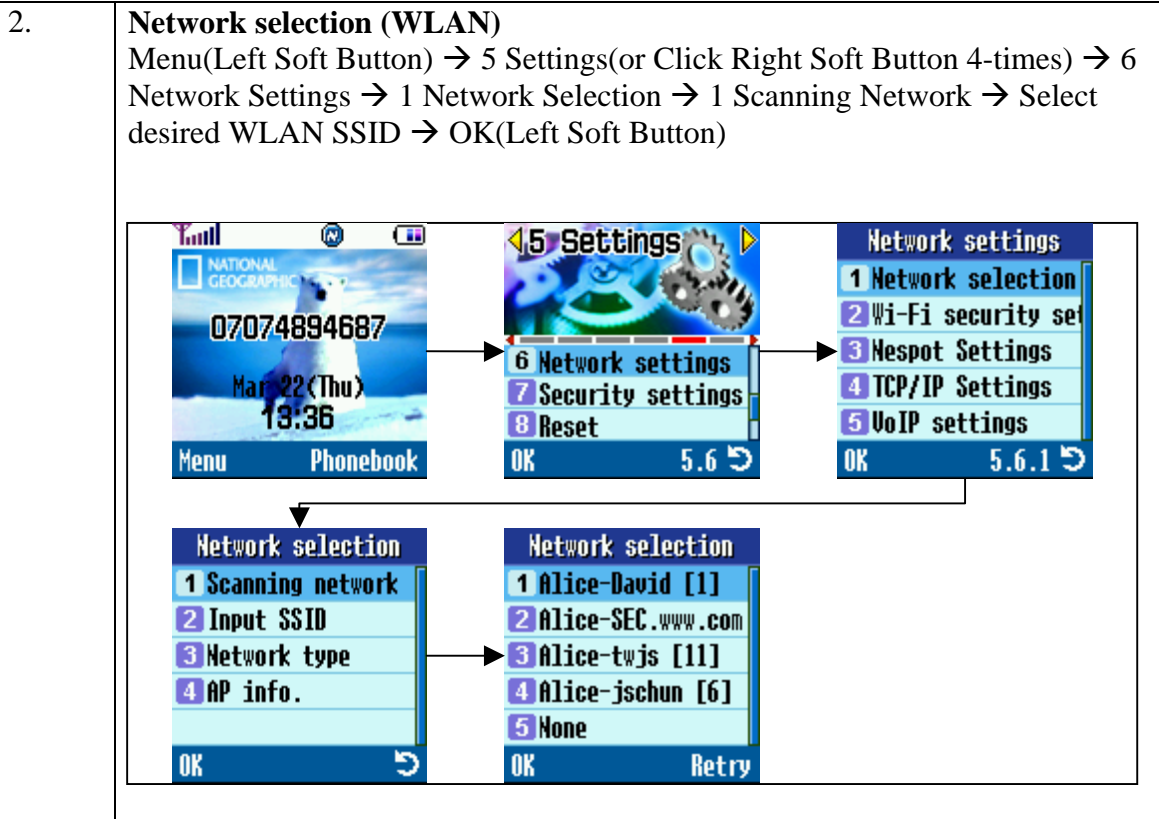
3.	**TCP/IP Settings** Menu(Left Soft Button) → 5 Settings(or Click Right Soft Button 4-times) → 6 Network Settings → 3 TCP/IP Settings → 1 IP Settings → Select DHCP Off → OK(Left Soft Button) → Input IP, Subnet Mask, Gateway → OK(Left Soft Button) → Phone will restart if there's any change.

4.	**VoIP Settings**
	Menu(Left Soft Button) → 5 Settings(or Click Right Soft Button 4-times) → 6 Network Settings → 4 VoIP Settings → Set each menu below when required
	    1. Service domain name
	    2. Service domain port
	    3. Expire time
	    4. Proxy
	    5. Proxy port
	    6. My SIP port
	    7. My RTP port
	    8. My RTP codec
	    9. Own number
	    10. Change name
	    11. SIP auth. user
	    12. SIP auth. Password

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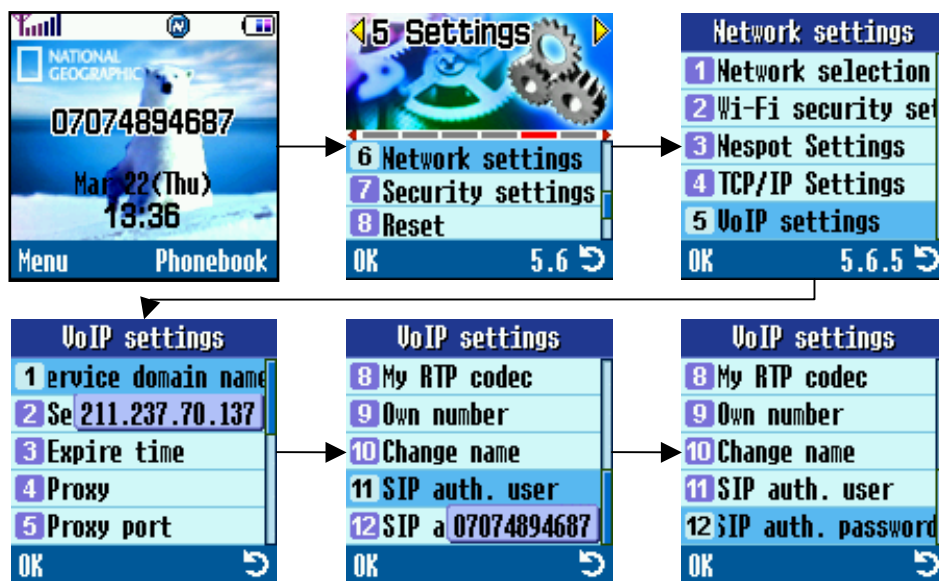
| 5. | **Web Administration**
After setting up, user can configure settings via web administration.
http://ipaddress:8000. Username: admin / Password: admin

 |
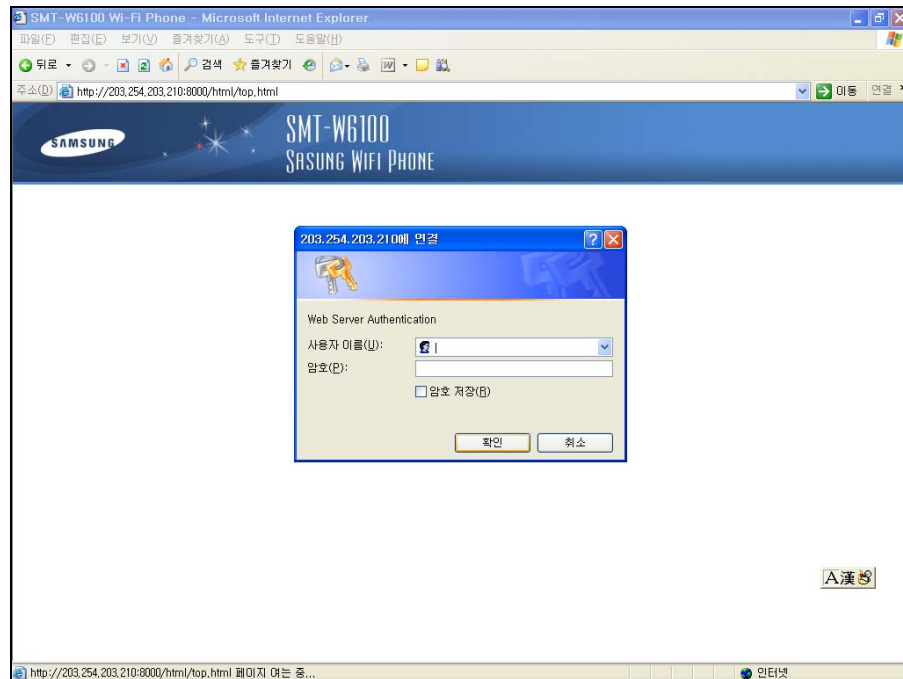
6.	**Details** (Only Korean supported at this moment)

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 (only display of "Priority") is also supported on Samsung WiFi SIP phone.

```
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  y  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 36005                                             Page   1 of    4
                                STATION

Extension: 36005                       Lock Messages? n         BCC: 0
    Type: 4620                         Security Code: 12345      TN: 1
    Port: IP                         Coverage Path 1: 1         COR: 1
    Name: SMT-W6100-1                 Coverage Path 2:          COS: 1
                                     Hunt-to Station:

STATION OPTIONS
           Loss Group: 19          Personalized Ringing Pattern: 1
                                           Message Lamp Ext: 36005
         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 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 36005                                      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.

```
change station 36005                                   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 36005. 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 (36005) 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 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
 36005          OPS             - 36005              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
 36005         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

<Description for Icons>

Icon	Description
📶	WLAN Signal.
☎	Busy
☎	Out of Service
▣	Message Waiting Indicator Icon
✉	New or Unread SMS/EMS/MMS

⌂	Alarm On
🕭	Vibration
🔇	Quiet Mode
Ⓦ	Connected to Access Point
Ⓝ	Registered to SIP Proxy
🔊	Speakerphone Mode.
🔋	Battery.

1. Verify that the phone detected WLAN(📶) successfully. After rebooting the telephone, use the 5 Settings menu at the phone to verify the values set in the default (proxy server address and port number, register with proxy, etc.) and phone-specific (User ID, Password, etc.) parameters. Verify that the phone is registered to the SIP Proxy by checking the Ⓝ icon on the display with its assigned number.

2. Verify the extended OPS features by dialing the FNE. If busy or intercept tone is heard, check Avaya Communication Manager administration for the correct FNE, proper permissions under COS/COR, and the proper station button assignment to support the feature.

3. Call a telephone that currently has no voice messages, and leave a message. Verify that the message-waiting icon(📧) appears on the display with alarming sound. 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 icon disappears.

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 Wireless endpoints is working. Codecs and shuffling of endpoints is also tested. Avaya SES is also rebooted to confirm if the Samsung SMT-W6100 WiFi SIP Telephone is properly registered.

9. Conclusion and Results

These Application Notes have described the administration steps required to use Samsung SMT-W6100 WiFi SIP telephones with the Avaya SES and Avaya Communication Manager. Samsung SMT-W6100 WiFi SIP telephone is able to work with the listed Avaya SIP Offer solution features. However, it is compatible with all Avaya G.729 codec except G.729a.

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