Avaya Solution & Interoperability Test Lab

Configuring the Samsung Ubigate^{TM} iBG-3026 with Avaya SIP Enablement Services and Avaya Communication Manager - Issue 1.0

Abstract

These Application Notes describe the procedures for configuring the Samsung Ubigate^{TM} iBG-3026 to communicate via a SIP interface with Avaya SIP Enablement Services and Avaya Communication Manager. The Samsung Ubigate^{TM} iBG-3026 functions as a Multi-service IP Switch/Router with integrated SIP gateway functionality that serves as a SIP gateway between IP-based PBX systems and analog endpoints or trunks. When connected to the Samsung Ubigate^{TM} iBG-3026, analog endpoints at customer enterprise sites are able to register as SIP endpoints with Avaya SIP Enablement Services and function as an Off-PBX Station extension of the Avaya Communication Manager.

1. Introduction

These Application Notes describe the procedures for configuring the Samsung Ubigate^{TM} iBG-3026 to communicate via a SIP interface with Avaya SIP Enablement Services and Avaya Communication Manager. The Samsung Ubigate^{TM} iBG-3026 functions as a Multi-service IP Switch/Router with integrated SIP gateway functionality that serves as a SIP gateway between IP-based PBX systems and analog endpoints. When connected to the Samsung Ubigate^{TM} iBG-3026 in the test configuration, analog endpoints at the customer enterprise site are able to register as SIP endpoints with Avaya SIP Enablement Services and function as an Off-PBX Station (OPS) extension of the Avaya Communication Manager. Central Office trunks can be utilized as part of Avaya Communication Manager accessible trunks.

2. Test Configuration

Figure 1 shows the physical connection of the setup for Avaya 9600 Series IP Telephones, Avaya 4600 Series SIP Telephones and the Samsung Ubigate^{TM} iBG-3026.

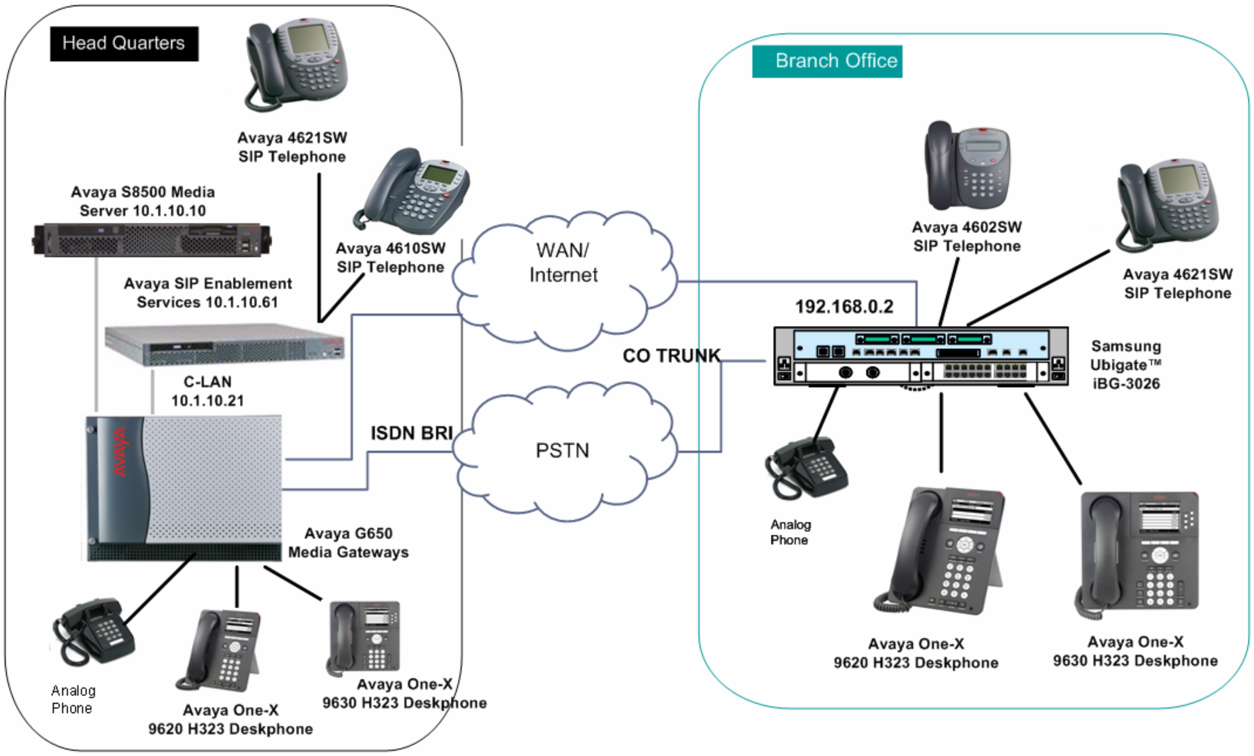


Figure 1 – Samsung Ubigate™ iBG-3026 Configurations

The configuration covered in these Application Notes allows:

- Intra-Network Region codec set: G.711Alaw, G.711Mulaw
- Inter-Network Region codec set: G.729a, G.729b, G.711Alaw, G.711Mulaw

Note that the variation of G.729 codec was set for Inter-Network Region codec. Samsung Ubigate^{TM} iBG-3026 support G.729a and Avaya SIP 4600 Series Telephone supports G.729b.

The codec selected for calls between analog endpoints of Samsung Ubigate^{TM} iBG-3026 depends on the codec preference set on the Samsung Ubigate^{TM} iBG-3026. The preferred order used in the sample configuration is: G.729a (1st preference), G.711Alaw, G.711Mulaw.

3. Equipment and Software Validated

The following equipment and software/firmware were used for the reference configuration provided:

Network Component	Software Versions
Avaya Communication Manager	3.1.2 with SP 12372
Avaya G650 Media Gateway TN2312BP TN799D C-LAN TN2302AP TN2602AP	 HW07 FW31 HW01 FW17 HW20 FW113 HW02 FW24
Avaya SIP Enablement Services (SES)	SES03.1.1-03.1.114.0
Avaya 9620 One-X Deskphone	1.1 (H323)
Avaya 9630 One-X Deskphone	1.1 (H323)
Avaya 4602SW Telephone	2.2.2 (SIP)
Avaya 4610SW Telephone	2.2.2.3 (SIP)
Avaya 4621SW Telephone	2.2.2.3 (SIP)
Analog Phone	NA
Samsung Ubigate^{TM} iBG-3026	SNOS 1.0.5.7 Advanced dsp 1.0.2 firmware

4. Configure Avaya Communication Manager

This section details the administration on Avaya Communication Manager to integrate with Avaya SIP Enablement Services and to enable the analog telephones connected to the Samsung Ubigate^{TM} iBG-3026 to register as SIP endpoints. The commands listed in the following sections were issued at the Avaya System Access Terminal (SAT) screen to configure Avaya Communication Manager.

4.1. Configure Integration with Avaya SIP Enablement Services

A SIP network interface must be configured between Avaya Communication Manager and Avaya SIP Enablement Services. This interface is a trunk group that handles all SIP signaling between Avaya SIP Enablement Services (which interfaces with the Samsung Ubigate^{TM} iBG-3026 as a SIP proxy) and Avaya Communication Manager. The steps described below enable the features and create the administrative objects necessary to support this interface.

Step	Description
1.	Enter **display system-parameters customer-options** and advance to Page 2. Under the **IP PORT CAPACITIES** section, confirm that the **Maximum Administered SIP Trunks** is enough to support the expected traffic to and from the Samsung Ubigate^{TM} iBG-3026. Any call involving a SIP endpoint (e.g., an analog telephone connected to an Samsung Ubigate^{TM} iBG-3026) will use a SIP trunk per SIP endpoint. If the capacity indicated is deemed insufficient, an authorized Avaya support technician will need to install an appropriately enabled license file.

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

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 800   0
           Maximum Concurrently Registered IP Stations: 2400  4
             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: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 800   200

   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 1     0
                     Maximum G250/G350/G700 VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 0     0
          Maximum TN2602 Boards with 320 VoIP Channels: 0     0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

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

Step	Description
2.	Enter **change node-names ip** to add the **ses1** and **10.1.10.61** for Avaya SIP Enablement Services. Note also the administered C-Lan IP interface in the Avaya G650 Media Gateway (**C-Lan**, **10 .1 .10 .21**); this will be used in **Section 5, Step 5** to create the Media Server Interface in Avaya SIP Enablement Services.

```
change node-names ip                                               Page   1 of
                                  IP NODE NAMES
    Name              IP Address          Name              IP Address
aes1                10 .1  .10 .71                            .   .   .
coecms              135.27 .4  .253                           .   .   .
coeir1              192.168.8  .14                            .   .   .
default             0  .0  .0  .0                             .   .   .
mypc                135.27 .13 .26                            .   .   .
procr               10 .1  .10 .10                            .   .   .
s8300-g250          10 .1  .40 .10                            .   .   .
s8300-siteb         10 .1  .20 .10                            .   .   .
s8500               10 .1  .10 .10                            .   .   .
s8500-clan1         10 .1  .10 .21                            .   .   .
s8500-clan2         10 .1  .10 .22                            .   .   .
s8500-medpro1       10 .1  .10 .31                            .   .   .
s8500-medpro2       10 .1  .10 .32                            .   .   .
s8500-val1          10 .1  .10 .41                            .   .   .
ses1                10 .1  .10 .61                            .   .   .
                       .   .   .                              .   .   .
( 15 of 15   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

Step	Description
3.	Enter **change ip-codec-set *n*** (where ***n*** is the number of the codec set specified in **Step 4** below) to specify the audio codecs to be used. The order of the codecs listed will determine the negotiating preference for each call established. A prime consideration here is bandwidth utilization where more calls can be placed given a fixed bandwidth. The codecs must be among those supported by the Samsung Ubigate^{TM} iBG-3026. In the configurations below **ip-codec-set 1** is set as G.711 for intra-region calls where there is more bandwidth available. As for **ip-codec-set 2**, G.729a is set as the preferred codec for inter-region calls as G.729 requires less bandwidth.

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

                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711A           n         2        20
 2: G.711Mu          n         2        20
 3:
 4:
 5:
 6:
 7:


     Media Encryption
 1: none
 2:

 3:
```

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

                          IP Codec Set

    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.729A           n         2        20
 2: G.729B           n         2        20
 3: G.711A           n         2        20
 4: G.711Mu          n         2        20
 5:
 6:
 7:


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

Step	Description
4.	Enter **change ip-network-region *n***, where ***n*** is the IP network region where the Avaya SIP Enablement Services server will reside, to define the connectivity settings for all VoIP resources and IP endpoints within that region. In this example, region 1, the default region for the Media Server running Avaya Communication Manager, was used. The following fields should be considered: • **Authoritative Domain:** Enter a value that matches the SIP Domain of the Avaya SIP Enablement Services server (in this example, **sglab.com***)*. • **Intra-region IP-IP Direct Audio, Inter-region IP-IP Direct Audio:** Keep the default value of **yes** for each of these fields to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G650 Media Gateway (a feature known as “shuffling”). • **Codec Set (Page 1):** Enter the IP codec set **1** as specified in **Step 3 for intra-region call**. This determines the set of codecs to be used for calls within this IP network region. In Page 3: • **Codec set (Src Rgn-1, Dst Rgn-5):** This codec set is set as **2** (which is created in **Step 3**) for inter-region call. The Avaya S8500 Media Server is placed in a separate region from the Samsung Ubigate™ iBG-3026. Calls between analog telephones connected to Samsung Ubigate™ iBG-3026 and other IP Phones or analog phones in the main site would be subject to this codec set specifications.

```
change ip-network-region 1                                      Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: sglab.com
    Name: Site A - Main
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              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? n
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

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

                   Inter Network Region Connection Management

 src dst codec direct   Total           Video                          Dyn
 rgn rgn  set   WAN  WAN-BW-limits  WAN-BW-limits  Intervening-regions  CAC IGAR
 1   1    1
 1   2
 1   3
 1   4    7     y        :NoLimit       :NoLimit                              n
 1   5    2     y        :NoLimit       :NoLimit                              n
```

Step	Description
5.	Enter **add signaling-group *n***, where ***n*** is the signaling group number, to create a new SIP signaling group (to be used by the SIP trunk group to be created in **Step 7)**. In this example, signaling group 5 was created. The following fields should be considered: • **Group Type:** Enter **sip**. • **Near-end Node Name:** Enter the node name for the C-Lan supporting the Avaya S8500 Media Server (in this example, **s8500-clan1**). For Media Server platforms that do not use C-Lan boards, **procr** would be specified here. • **Far-end Node Name:** Enter the node name for the Avaya SIP Enablement Services server (in this example, **ses1**). • **Far-end Listen Port:** Enter 5061 (the recommended TLS port value). • **Far-end Network Region:** This determines which IP network region contains the Samsung Ubigate^{TM} iBG-3026. • **Far-end Domain:** Enter the domain name of the Avaya SIP Enablement Services server (in this example, **sglab.com**). • **DTMF over IP:** Enter **rtp-payload**. This allows Avaya Communication Manager to send DTMF tones using RFC 2833. See **Section 6 Step 12** for configuring the Samsung Ubigate^{TM} iBG-3026. • **Direct IP-IP Audio Connections:** Enter **y** to disable shuffling between the near-end and far-end IP endpoints.

```
add signaling-group 5                                           Page   1 of   1
                                SIGNALING GROUP

 Group Number: 5             Group Type: sip
                       Transport Method: tls





  Near-end Node Name: s8500-clan1          Far-end Node Name: ses1
 Near-end Listen Port: 5061              Far-end Listen Port: 5061
                                      Far-end Network Region: 5
       Far-end Domain: sglab.com

                                         Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload        Direct IP-IP Audio Connections? y
                                                    IP Audio Hairpinning? n
 Session Establishment Timer(min): 120
```

Step	Description
6.	Enter another signaling group through **add signaling-group *n*** to create a new SIP signaling group (to be used by the SIP trunk group to be created in **Step 8**) for directing calls through the FXO. In this example, signaling group 6 was created. The **Far-end Domain** in this case is using a different label i.e. "trkA.sglab.com". This domain is associated with the Samsung Ubigate iBG-3026 VoIP gateway.

```
add signaling-group 6                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 6             Group Type: sip
                       Transport Method: tls




   Near-end Node Name: s8500-clan1          Far-end Node Name: ses1
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 5
       Far-end Domain: trkA.sglab.com

                                            Bypass If IP Threshold Exceeded? n

         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
                                                       IP Audio Hairpinning? n
 Session Establishment Timer(min): 120
```

Step	Description
7.	Enter **add trunk-group *n***, where ***n*** is the signaling group number, to create a new SIP trunk group for calls to the Samsung Ubigate^{TM} iBG-3026 analog endpoints. In this example, trunk group 5 was created.

On Page 1, enter the following values:

- **Group Type:** Enter **sip**.
- **Group Name:** Enter a descriptive name.
- **TAC**: Enter a valid trunk access code.
- **Service Type:** Enter **tie**.
- **Signaling Group:** Enter the number of the signaling group created in **Step 5**.
- **Number of Members:** Enter an appropriate number of SIP trunks, not exceeding the maximum number of available SIP trunks as indicated in **Step 1**.

```
add trunk-group 5                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 5                    Group Type: sip           CDR Reports: y
  Group Name: SIP - ses1                  COR: 1        TN: 1        TAC: 705
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                          Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 5
                                                     Number of Members: 60
```

On Page 3, specify the following value:

- **Numbering Format:** Enter **public**. This determines the outgoing calling party number format.

```
add trunk-group 5                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



                     Numbering Format: public
                                              Prepend '+' to Calling Number? n


                                                Replace Unavailable Numbers? n
```

Step	Description
8.	Enter another **add trunk-group *n*** to create a new SIP trunk group for directing calls to the FXO. In this example, trunk group 6 was created.

```
add trunk-group 6                                           Page   1 of  21
                                TRUNK GROUP

Group Number: 6                    Group Type: sip          CDR Reports: y
  Group Name: Sarak CO Trunk TrkA         COR: 1        TN: 1        TAC: 706
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                          Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 6
                                                     Number of Members: 4
```

On Page 3, specify the following value:

- **Numbering Format:** Enter **public**. This determines the outgoing calling party number format.

```
add trunk-group 6                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y


                     Numbering Format: public
                                              Prepend '+' to Calling Number? n

                                                Replace Unavailable Numbers? n
```

Step	Description
9.	Enter **change route-pattern *n*** to administer the route pattern that will be used to direct outgoing SIP calls to Avaya SIP Enablement Services. In this example, route pattern 5 was used. Enter the following values: • **Pattern Name**: Enter a descriptive name. • **Grp No**: Enter the number of the SIP trunk group created in **Step 7** • **FRL**: Enter a Facility Restriction Level for this entry in the route pattern, from 0 (least restrictive, i.e., all originating SIP endpoints can use this entry) to 7 (most restrictive).

```
change route-pattern 5                                          Page   1 of   3
                    Pattern Number: 5   Pattern Name: SIP-devlabses1
                             SCCAN? n     Secure SIP? y
    Grp FRL NPA Pfx Hop Toll No.  Inserted                             DCS/ IXC
    No          Mrk Lmt List Del  Digits                               QSIG
                             Dgts                                      Intw
 1: 5    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
```

Step | **Description**

10.

Enter **change route-pattern *n*** to administer the route pattern that will be used to direct outgoing SIP calls to Samsung Ubigate^{TM} iBG-3026. In this example, route pattern 6 was used.

```
change route-pattern 6                                          Page   1 of   3
                    Pattern Number: 6   Pattern Name: SIP Sarak CO
                             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: 6    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
```

11.

Enter **change locations** to assign the above-configured route pattern to a location. This assignment is necessary to enable SIP endpoints to use certain features, such as Transfer. The reference configuration as a whole is in the default location (Main). Enter the following value:

- **Proxy Sel. Rte. Pat.**: Enter the number of the route pattern configured in **Step 9.**

```
change locations                                                Page   1 of  16
                                   LOCATIONS

                ARS Prefix 1 Required For 10-Digit NANP Calls? y

 Loc.  Name           Timezone Rule  NPA  ARS   Attd         Pre-   Proxy Sel.
 No.                   Offset             FAC   FAC          fix    Rte. Pat.
 1:   Main            + 00:00   0                                    5
 2:                       :
 3:                       :
 4:                       :
 5:                       :
 6:                       :
 7:                       :
 8:                       :
```

4.2. Configure Analog/SIP Endpoints

This section provides the steps to enable analog endpoints connected to the Samsung Ubigate^{TM} iBG-3026 to be treated as SIP stations by Avaya Communication Manager. These endpoints are administered as Off-PBX Station (OPS) extensions that are accessed via a SIP trunk group. For more details, see [6] and [7].

Step	Description
1.	Enter **display system-parameters customer-options** and examine Page 1 to confirm that the license file has allocated enough OPS extensions (**Maximum Off-PBX Telephones – OPS**) to support all enterprise sites. If not, an authorized Avaya support technician will need to install an appropriately enabled license file.

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

     G3 Version: V13
       Location: 2                             RFA System ID (SID): 1
       Platform: 12                            RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 3200  591
                                      Maximum Stations: 2400  342
                              Maximum XMOBILE Stations: 0     0
                    Maximum Off-PBX Telephones - EC500: 100   1
                    Maximum Off-PBX Telephones -   OPS: 100   14
                    Maximum Off-PBX Telephones - SCCAN: 0     0







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

Step	Description
2.	Enter **change off-pbx-telephone configuration-set *n*** to assign call treatment options to Off-PBX telephones. In this example, configuration set **5** was administered.

```
change off-pbx-telephone configuration-set 5                    Page   1 of   1



                               CONFIGURATION SET: 5

      Configuration Set Description: SIP Phones
               Calling Number Style: network
                CDR for Origination: phone-number
 CDR for Calls to EC500 Destination? y
        Fast Connect on Origination? n
       Post Connect Dialing Options: dtmf
      Cellular Voice Mail Detection: none
                      Barge-in Tone? n
        Calling Number Verification? y
             Identity When Bridging: principal
```

Step	Description
3.	Enter **add station *x***, where ***x*** is an available valid extension in the dial plan, to create a station extension for an analog/SIP endpoint. Enter the following values on Page 1: • **Type**: Set to **6408D+** (the default). • **Port**: Enter **X**. This indicates that the station is Administered Without Hardware (AWOH), i.e., not assigned to a specific port on Avaya Communication Manager. • **Name**: Enter a descriptive name.

```
add station 10058                                               Page   1 of   4
                                     STATION

Extension: 10058                           Lock Messages? n          BCC: 0
     Type: 6408D+                          Security Code: *           TN: 1
     Port: x                             Coverage Path 1:            COR: 1
     Name: Sarak FXS 01                  Coverage Path 2:            COS: 1
                                         Hunt-to Station:

STATION OPTIONS
              Loss Group: 2              Personalized Ringing Pattern: 1
             Data Module? n                          Message Lamp Ext: 10058
            Speakerphone: 2-way                   Mute Button Enabled? y
        Display Language: english

                                                    Media Complex Ext:
                                                         IP SoftPhone? n

                                                  Remote Office Phone? n
```

Step	Description
4.	On Page 2 of this **Station** form, enter the following value: • **Direct IP-IP Audio Connections**: Enter **y** to enable shuffling calls involving this station.

```
add station 10058                                               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? n                       Restrict Last Appearance? y
  Active Station Ringing: single          Conf/Trans on Primary Appearance? n

        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: basic
                                                Display Client Redirection? n
              AUDIX Name:                      Select Last Used Appearance? n
                                                 Coverage After Forwarding? s

                                               Direct IP-IP Audio Connections? y
 Emergency Location Ext: 10058                        IP Audio Hairpinning? n
```

Step	Description
5.	On Page 3 of this **Station** form, enter the following **BUTTON ASSIGNMENTS**: • Add a no-hold-conference (**no-hld-cnf**) button to allow the station to activate the Conference on Answer feature. • Add an auto-callback (**auto-cback**) button to allow the station to activate the Automatic Callback feature.

```
add station 10058                                               Page   3 of   4
                                     STATION
 SITE DATA
       Room:                                            Headset? n
       Jack:                                            Speaker? n
      Cable:                                           Mounting: d
      Floor:                                        Cord Length: 0
   Building:                                          Set Color:



ABBREVIATED DIALING
     List1:                   List2:                    List3:

BUTTON ASSIGNMENTS
 1: call-appr                           5: auto-cback
 2: call-appr                           6:
 3: call-appr                           7:
 4: no-hld-cnf                          8:
```

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>

Enter **add off-pbx-telephone station-mapping** to map the new station extension to an OPS station. Enter the following values in the first available row:

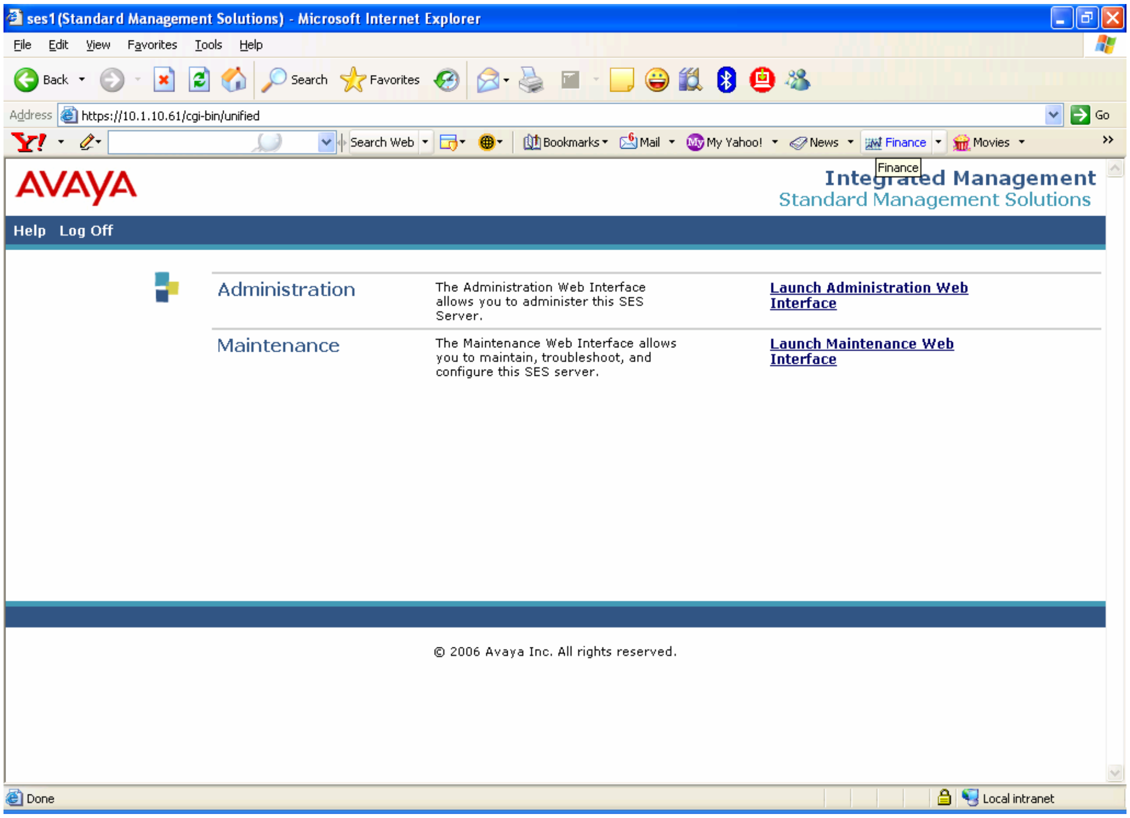
- **Station Extension:** Enter the extension of the station created in **Step 3**.
- **Application:** Enter **OPS**.
- **Phone Number:** Enter the phone number of the associated Off-PBX Telephone, i.e. the analog telephone connected to the Samsung Ubigate^{TM} iBG-3026.
- **Trunk Selection:** Enter the number of the SIP trunk group created in **Section 4.1, Step 7** (in this example, **5**).
- **Configuration Set:** Enter the number of the Configuration set administered in **Step 2** (in this example, **5**)

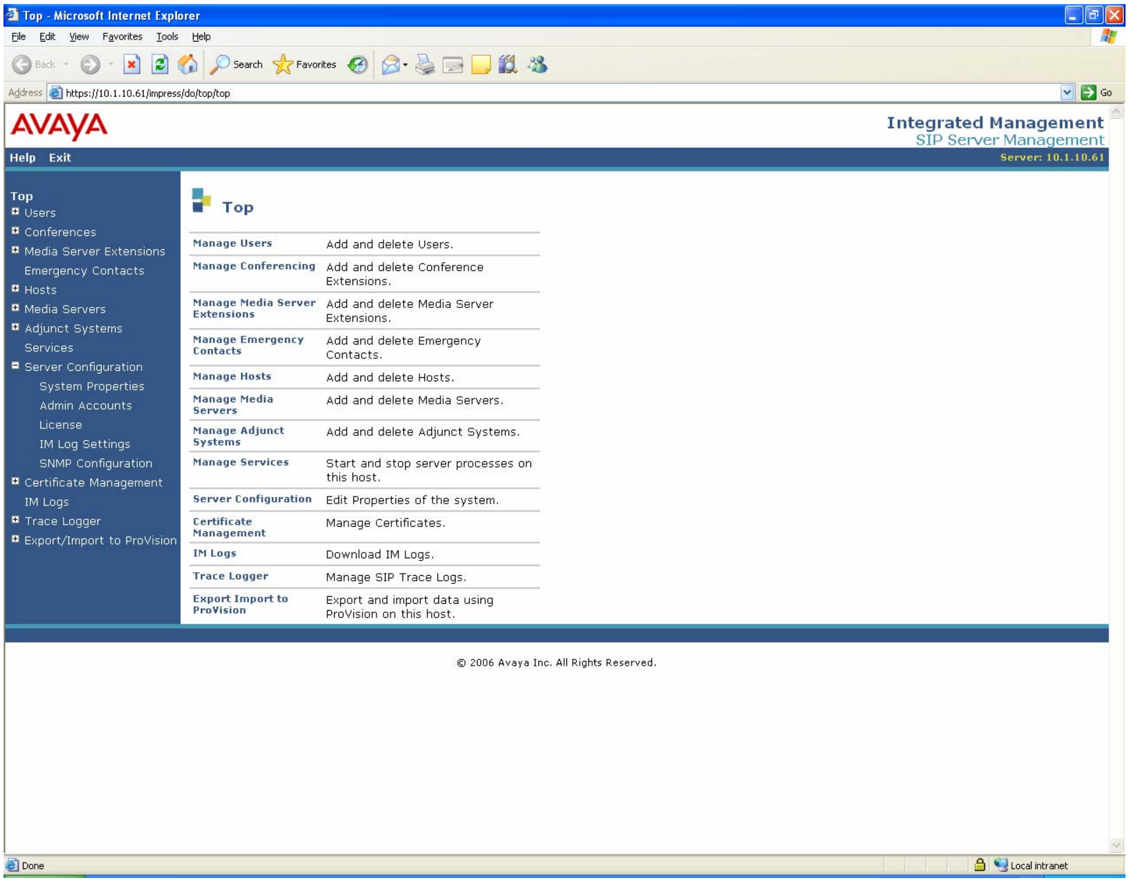
```
add off-pbx-telephone station-mapping                          Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station     Application   Dial   Phone Number      Trunk       Configuration
 Extension                 Prefix                   Selection   Set
 10058         OPS              - 10058              5          5
                                -
                                -
                                -
                                -
                                -
                                -
                                -
                                -
                                -
                                -
                                -
                                -
                                -
                                -
```

</td></tr>
<tr><td>7.</td><td>Repeat Steps 3-6 for each additional analog/SIP endpoint connected to the Samsung Ubigate^{TM} iBG-3026.</td></tr>
</table>

5. Configure Avaya SIP Enablement Services

This section addresses the administrative steps to be performed on Avaya SIP Enablement Services. The installation of the Avaya SIP Enablement Services software and license file, as well as the initial configuration of the server, is beyond the scope of this document. Please see [3] for the details of these procedures.

Step	Description
1.	To administer Avaya SIP Enablement Services, navigate to http://<*ip-addr*>/admin (where <*ip-addr*> is the IP address of the Avaya SIP Enablement Services server) from a Web browser. After logging in with an appropriate login and password, the main page appears.  ses1 (Standard Management Solutions) - Microsoft Internet Explorer Address https://10.1.10.61/cgi-bin/unified Integrated Management Standard Management Solutions Help Log Off Administration — The Administration Web Interface allows you to administer this SES Server. — Launch Administration Web Interface Maintenance — The Maintenance Web Interface allows you to maintain, troubleshoot, and configure this SES server. — Launch Maintenance Web Interface © 2006 Avaya Inc. All rights reserved. Done Local intranet

Step	Description
2.	From the main page, click on the **Launch Administration Web Interface** link. The Administration Home Page appears. **NOTE**: After making each of the changes described in this section, use the **Update** link, found at the bottom of the blue navigation pane, or the **Update** button, at the end of the completed form, to commit the changes to the Avaya SIP Enablement Services database. 

Top

Manage Users	Add and delete Users.
Manage Conferencing	Add and delete Conference Extensions.
Manage Media Server Extensions	Add and delete Media Server Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Manage Hosts	Add and delete Hosts.
Manage Media Servers	Add and delete Media Servers.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Services	Start and stop server processes on this host.
Server Configuration	Edit Properties of the system.
Certificate Management	Manage Certificates.
IM Logs	Download IM Logs.
Trace Logger	Manage SIP Trace Logs.
Export Import to ProVision	Export and import data using ProVision on this host.

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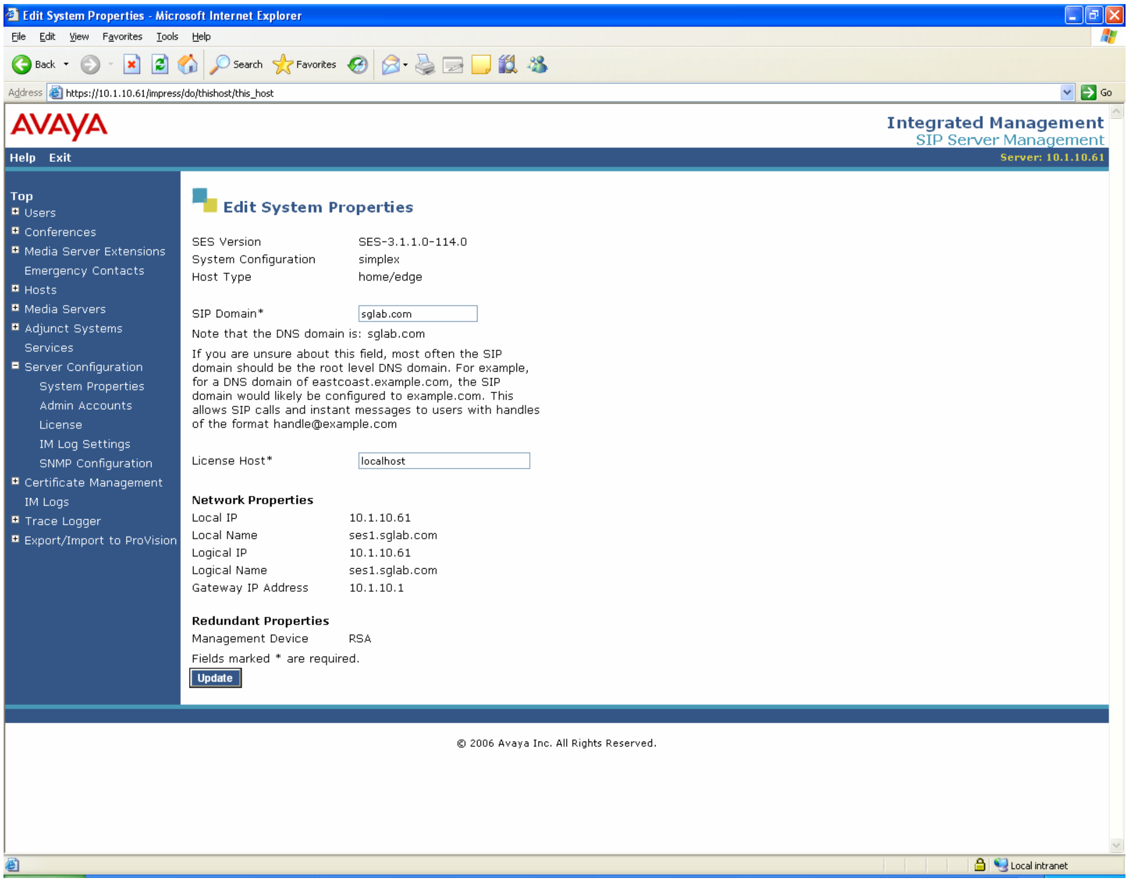
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>

From the blue navigation pane, select **Server Configuration** → **System Properties**. On the **Edit System Properties** page, enter the following values:

- **SIP Domain:** This must match the **Authoritative Domain** field configured on the **IP Network Region** form in Avaya Communication Manager shown in **Section 4.1, Step 4**. In this example, **sglab.com** is used.
- **License Host:** Enter the host name, the fully qualified domain name, or the IP address of the WebLM server where the license file for this Avaya SIP Enablement Services server is installed. In this example, the WebLM server is co-resident with the Avaya SIP Enablement Services server (IP address **10.1.10.61**).

The completed form appears as follows. Click **Update** to submit the form.



Edit System Properties

SES Version SES-3.1.1.0-114.0
System Configuration simplex
Host Type home/edge

SIP Domain* sglab.com
Note that the DNS domain is: sglab.com

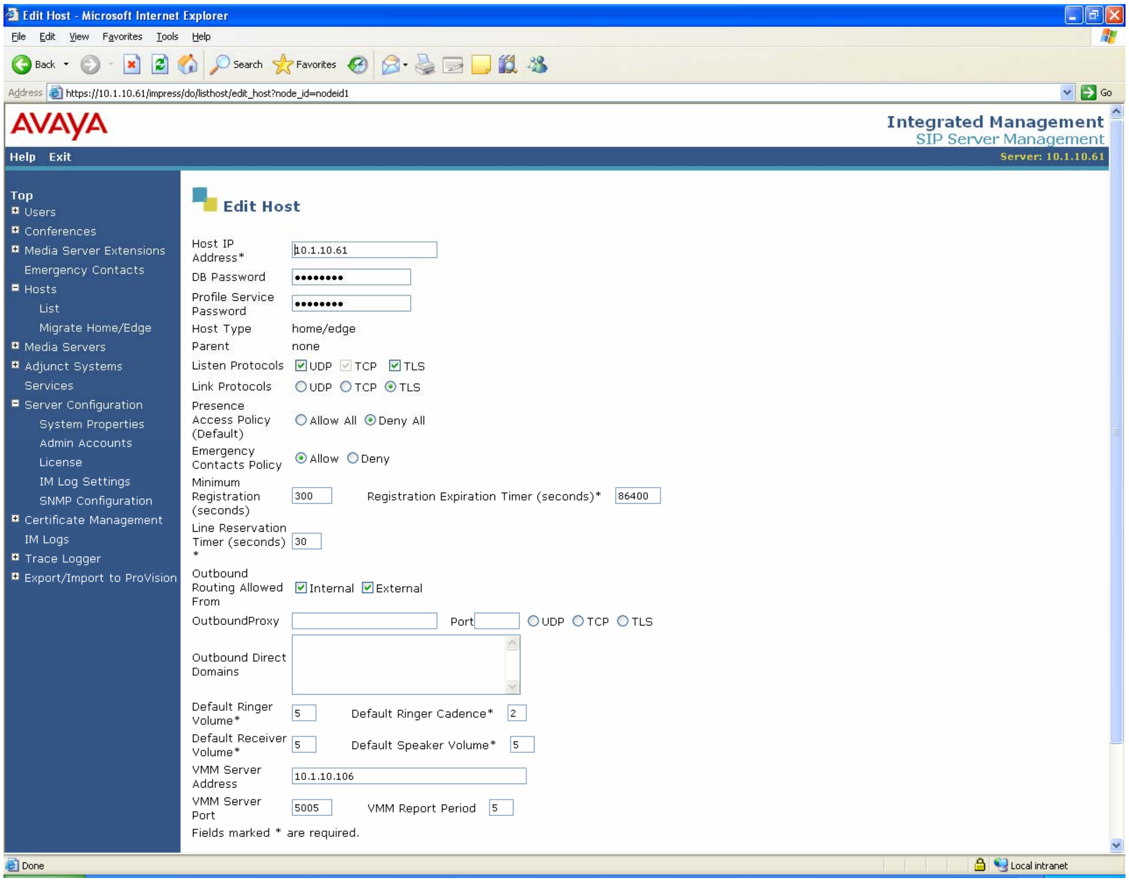
If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com

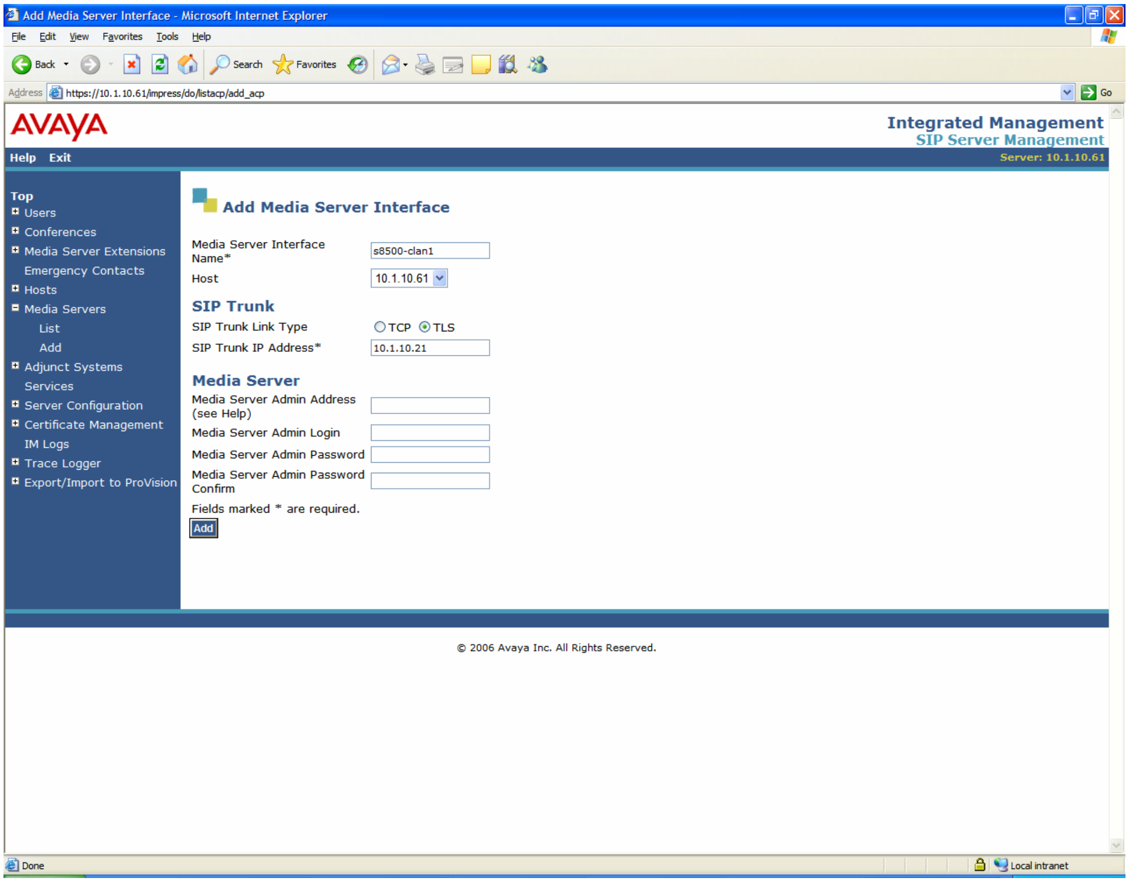
License Host* localhost

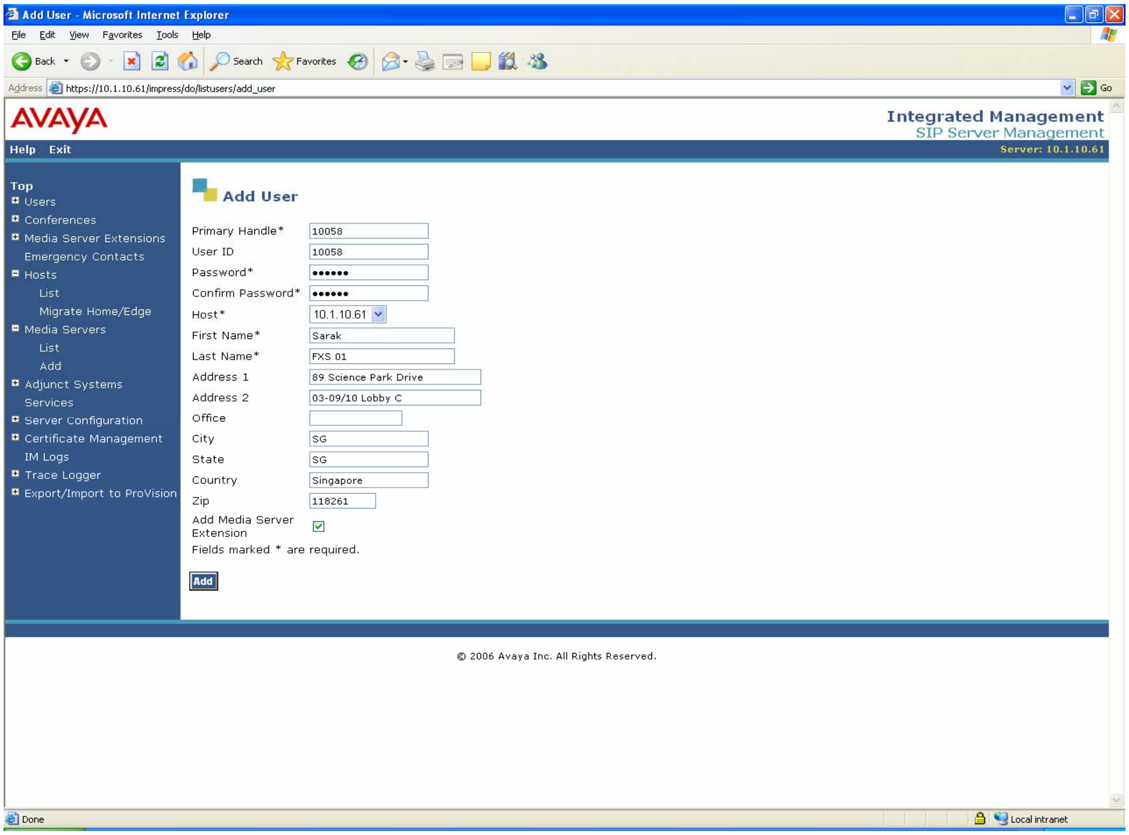
Network Properties
Local IP 10.1.10.61
Local Name ses1.sglab.com
Logical IP 10.1.10.61
Logical Name ses1.sglab.com
Gateway IP Address 10.1.10.1

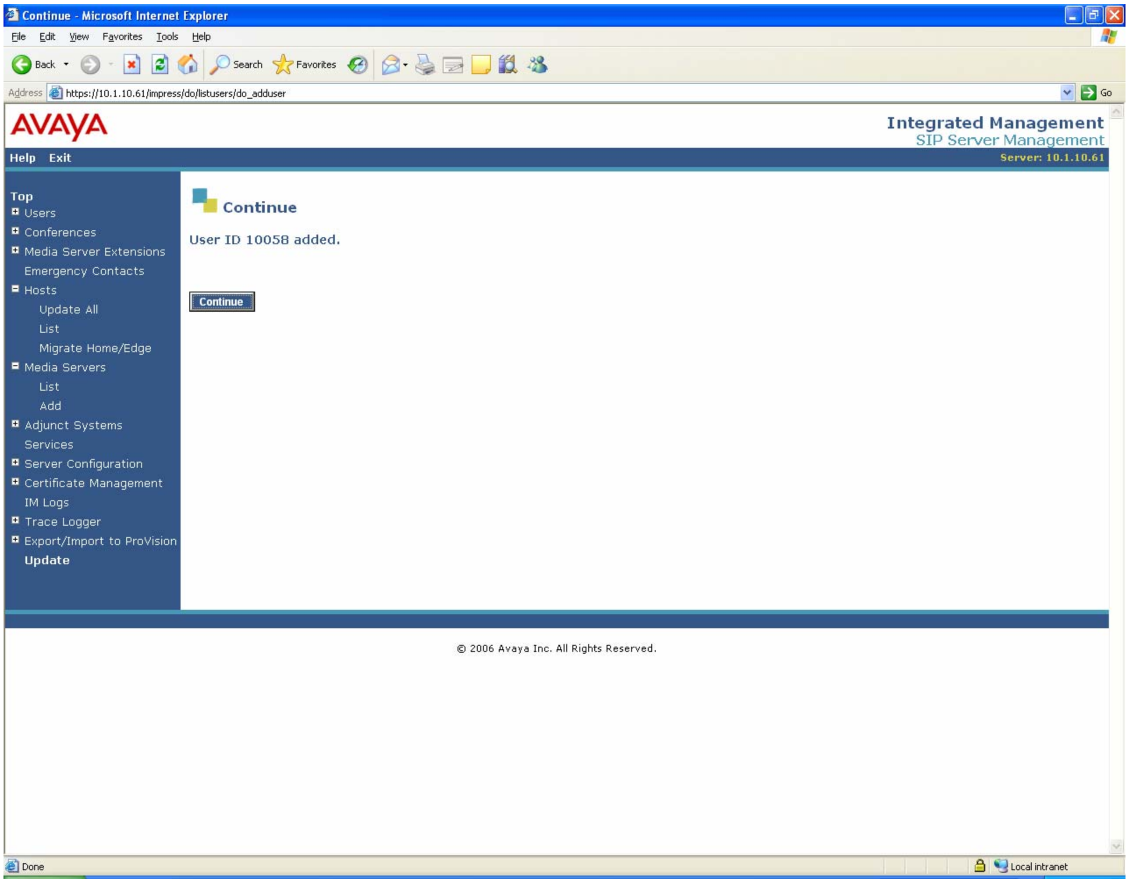
Redundant Properties
Management Device RSA
Fields marked * are required.

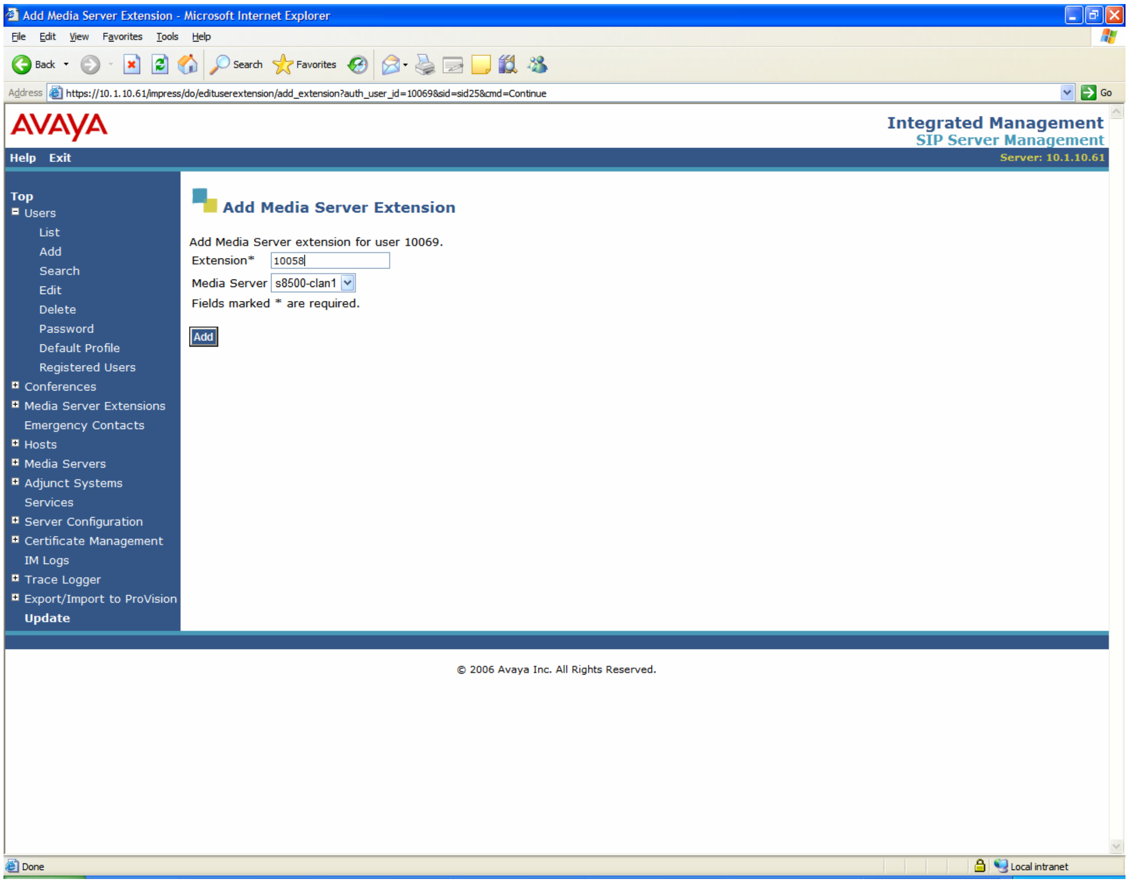
</td></tr>
</table>

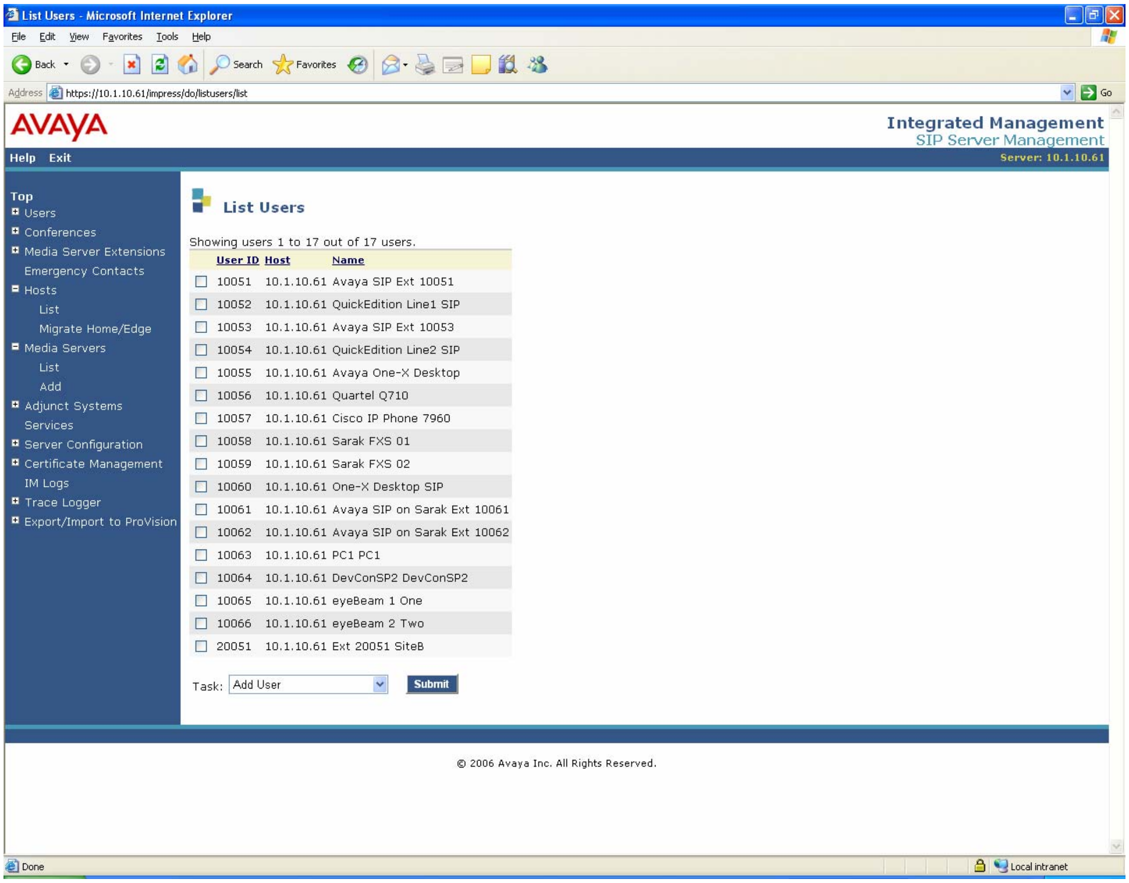
Step	Description
4.	From the blue navigation pane, select **Hosts**, then select **Edit** on the next page. The **Edit Host** page appears. Enter the following values: - **Host IP Address:** Enter the **Logical IP** or **Logical Name** (from the **Edit System Properties** form in **Step 3**). - **DB Password:** This is the password that was entered during execution of the system installation script. (See [3] for details.) - **Host Type:** This field indicates whether this Avaya SIP Enablement Services server functions as a “home” or “edge” server. Since, in this example, only a single Avaya SIP Enablement Services server is used, enter **home/edge**. - Keep all other default values. The completed form appears as follows. Click **Update** to submit the form.  Edit Host Host IP Address* 10.1.10.61 DB Password •••••••• Profile Service Password •••••••• Host Type home/edge Parent none Listen Protocols ☑ UDP ☑ TCP ☑ TLS Link Protocols ○ UDP ○ TCP ● TLS Presence Access Policy (Default) ○ Allow All ● Deny All Emergency Contacts Policy ● Allow ○ Deny Minimum Registration (seconds) 300 Registration Expiration Timer (seconds)* 86400 Line Reservation Timer (seconds)* 30 Outbound Routing Allowed From ☑ Internal ☑ External OutboundProxy Port ○ UDP ○ TCP ○ TLS Outbound Direct Domains Default Ringer Volume* 5 Default Ringer Cadence* 2 Default Receiver Volume* 5 Default Speaker Volume* 5 VMM Server Address 10.1.10.106 VMM Server Port 5005 VMM Report Period 5 Fields marked * are required.

Step	Description
5.	From the blue navigation pane, select **Media Server** → **Add**. The **Add Media Server** page appears. Enter the following values: - **Media Server Interface Name:** Enter a descriptive name (in this example, **s8500-clan1**). - **Host:** Select the name or IP address of the Avaya SIP Enablement Services server from the drop-down menu (in this example, **10.1.10.61**). - **SIP Trunk Link Type:** Select **TLS** (Transport Link Security). - **SIP Trunk IP Address:** Enter the IP address of an Avaya Communication Manager's C-LAN board from **Section 4.1**, **Step 2**. (For Media Server platforms that do not use C-LAN boards, the IP address of the Avaya Media Server (i.e. the processor) would be specified here.) - Keep all other default values. The completed form appears as follows. Click **Add** to submit the form.  **Add Media Server Interface** Media Server Interface Name* s8500-clan1 Host 10.1.10.61 **SIP Trunk** SIP Trunk Link Type ○ TCP ● TLS SIP Trunk IP Address* 10.1.10.21 **Media Server** Media Server Admin Address (see Help) Media Server Admin Login Media Server Admin Password Media Server Admin Password Confirm Fields marked * are required. Add © 2006 Avaya Inc. All Rights Reserved.

Step	Description
6.	SIP users (identified by their corresponding telephone extensions) must be added. From the blue navigation pane, select **User → Add**. Enter the following values: ▪ **Primary Handle:** This specifies a user in Avaya SIP Enablement Services (in this example, **10058**). While not required, it is recommended that the Primary handle be the same as the **User ID**. ▪ **User ID:** The User ID (together with the **Password**) is used to authenticate a user in Avaya SIP Enablement Services. It must match the authentication configured on the Samsung Ubigate^{TM} iBG-3026 for an analog ports dial-peer settings in **Section 6**, **Step 9**. ▪ **Password:** This must match the password associated with the corresponding User ID configured on the Samsung Ubigate^{TM} iBG-3026 for an analog ports dial-peer settings in **Section 6**, **Step 9**. ▪ **Confirm Password:** Re-enter the above **Password**. ▪ **Host:** Select from the drop-down menu the host name or IP address of the Avaya SIP Enablement Services server (in this example, 10.1.10.61). ▪ **First Name, Last Name:** Enter descriptive values. ▪ **Add Media Server Extension:** Check this box to add an OPS extension for the user (see **Step 8**). The completed form appears as follows. Click **Add** to submit the form.  Address https://10.1.10.61/impress/do/listusers/add_user Add User Primary Handle* 10058 User ID 10058 Password* •••••• Confirm Password* •••••• Host* 10.1.10.61 First Name* Sarak Last Name* FXS 01 Address 1 89 Science Park Drive Address 2 03-09/10 Lobby C Office City SG State SG Country Singapore Zip 118261 Add Media Server Extension ☑ Fields marked * are required.

Step	Description
7.	On the page that follows (see below), click **Continue**.  User ID 10058 added.

Step	Description
8.	The **Add Media Server Extension** page will appear as follows. Enter the following values: - **Extension:** Enter the OPS extension corresponding to the station configured in Avaya Communication Manager. See **Section 4.2, Step 6**. - **Media Server**: Select the Media Server added in **Step 5** (in this example, **s8500-clan1**). The completed form appears as follows. Click **Add** to submit the form.  **Add Media Server Extension** Add Media Server extension for user 10069. Extension* 10058 Media Server s8500-clan1 Fields marked * are required.
9.	Repeat **Steps 6-8** to add additional users to support the remaining analog/SIP telephones at the enterprises.

Step	Description
10.	To view the configured users, select **Users → List** from the blue navigation pane. 

List Users

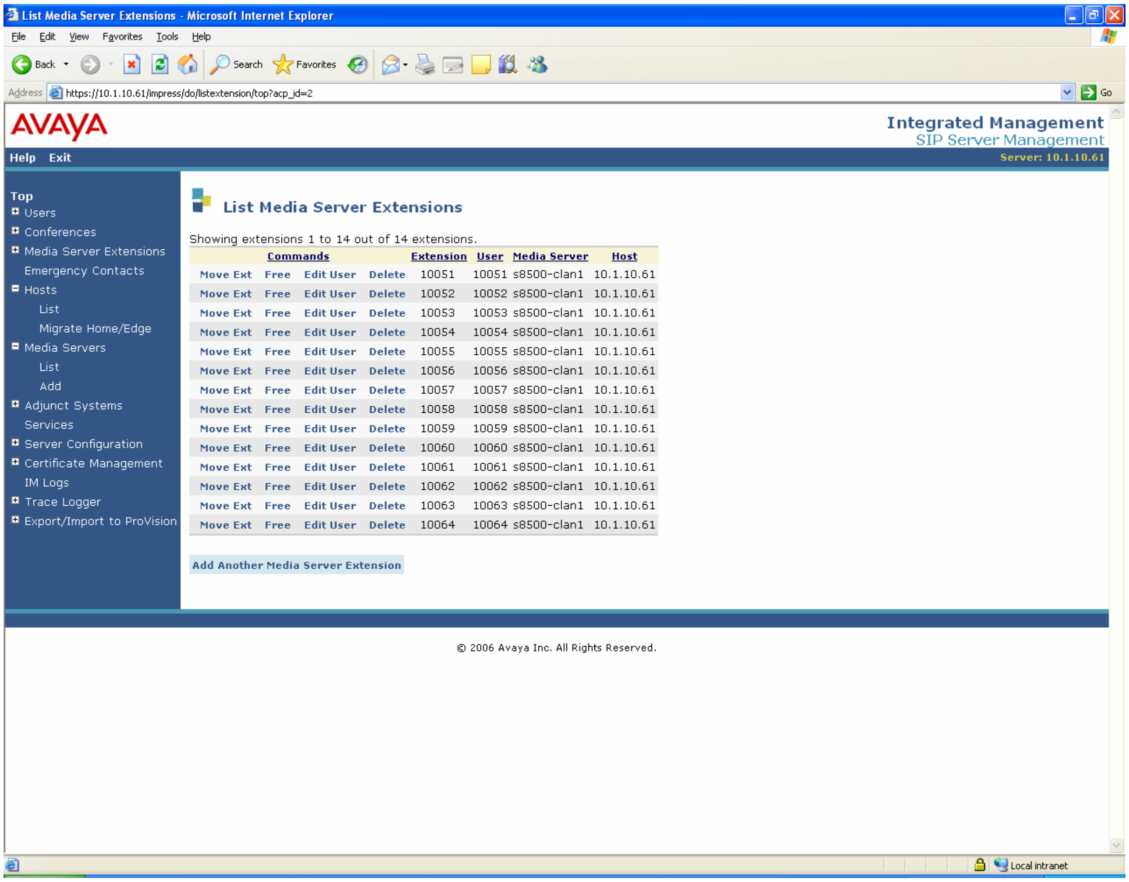
Showing users 1 to 17 out of 17 users.

	User ID	Host	Name
☐	10051	10.1.10.61	Avaya SIP Ext 10051
☐	10052	10.1.10.61	QuickEdition Line1 SIP
☐	10053	10.1.10.61	Avaya SIP Ext 10053
☐	10054	10.1.10.61	QuickEdition Line2 SIP
☐	10055	10.1.10.61	Avaya One-X Desktop
☐	10056	10.1.10.61	Quartel Q710
☐	10057	10.1.10.61	Cisco IP Phone 7960
☐	10058	10.1.10.61	Sarak FXS 01
☐	10059	10.1.10.61	Sarak FXS 02
☐	10060	10.1.10.61	One-X Desktop SIP
☐	10061	10.1.10.61	Avaya SIP on Sarak Ext 10061
☐	10062	10.1.10.61	Avaya SIP on Sarak Ext 10062
☐	10063	10.1.10.61	PC1 PC1
☐	10064	10.1.10.61	DevConSP2 DevConSP2
☐	10065	10.1.10.61	eyeBeam 1 One
☐	10066	10.1.10.61	eyeBeam 2 Two
☐	20051	10.1.10.61	Ext 20051 SiteB

Task: Add User Submit

Step **Description**

11. To view the configured extensions, select **Media Server Extensions** → **List** from the blue navigation pane.



List Media Server Extensions

Showing extensions 1 to 14 out of 14 extensions.

Commands				Extension	User	Media Server	Host
Move Ext	Free	Edit User	Delete	10051	10051	s8500-clan1	10.1.10.61
Move Ext	Free	Edit User	Delete	10052	10052	s8500-clan1	10.1.10.61
Move Ext	Free	Edit User	Delete	10053	10053	s8500-clan1	10.1.10.61
Move Ext	Free	Edit User	Delete	10054	10054	s8500-clan1	10.1.10.61
Move Ext	Free	Edit User	Delete	10055	10055	s8500-clan1	10.1.10.61
Move Ext	Free	Edit User	Delete	10056	10056	s8500-clan1	10.1.10.61
Move Ext	Free	Edit User	Delete	10057	10057	s8500-clan1	10.1.10.61
Move Ext	Free	Edit User	Delete	10058	10058	s8500-clan1	10.1.10.61
Move Ext	Free	Edit User	Delete	10059	10059	s8500-clan1	10.1.10.61
Move Ext	Free	Edit User	Delete	10060	10060	s8500-clan1	10.1.10.61
Move Ext	Free	Edit User	Delete	10061	10061	s8500-clan1	10.1.10.61
Move Ext	Free	Edit User	Delete	10062	10062	s8500-clan1	10.1.10.61
Move Ext	Free	Edit User	Delete	10063	10063	s8500-clan1	10.1.10.61
Move Ext	Free	Edit User	Delete	10064	10064	s8500-clan1	10.1.10.61

12. A Media Server Address Map must be administered to enable FAC and FNE entries to be routed to the appropriate Avaya Communication Manager.

SIP call requests are made via an INVITE message, which includes as its destination a Uniform Resource Identifier (URI). Generally, the URI is of the form

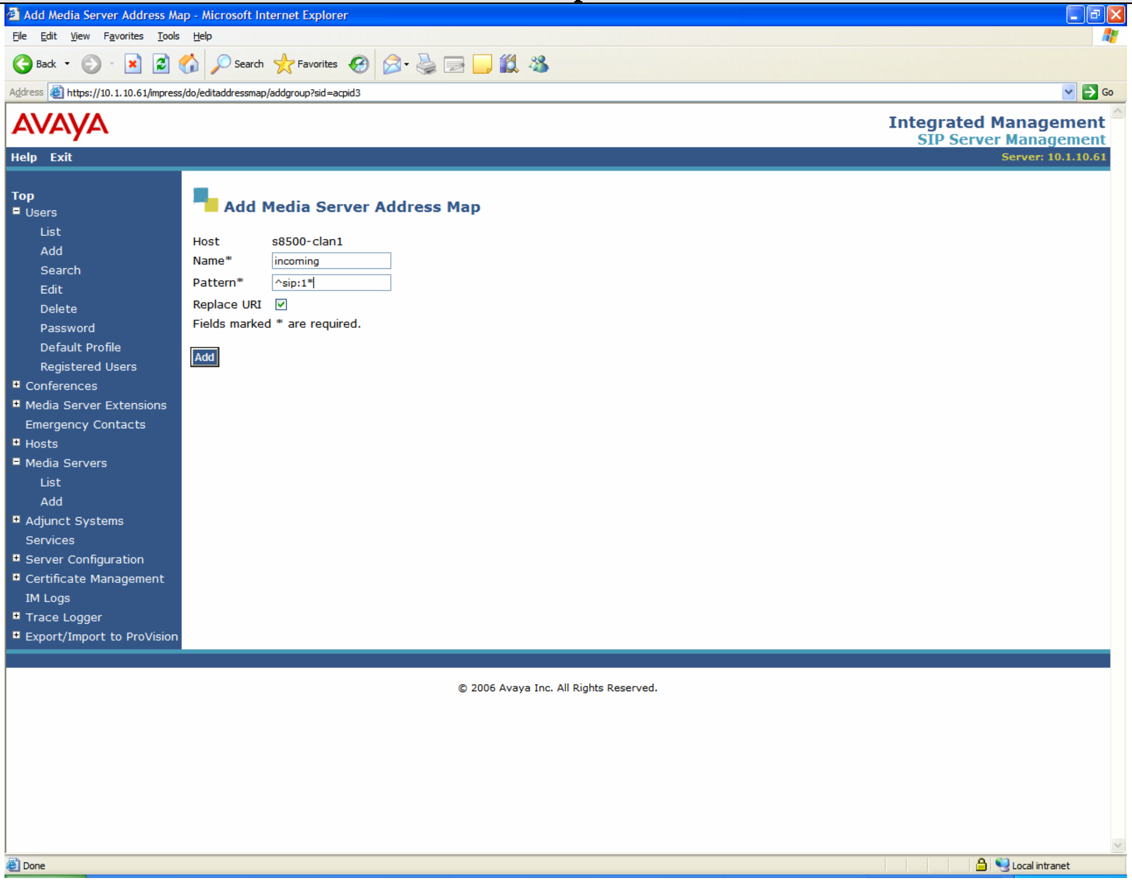
sip:<user>@<domain>

where <*domain*> is either a domain name or an IP address. For the reference configuration, a SIP URI in an INVITE message specifying the user created in **Step 6** would look like the following:

`sip: 10058@10.1.10.61`

Use the following steps to configure a Media Server Address Map:

Step	Description
	▪ From the blue navigation pane, select **Media Servers → List**. The **List Media Servers** page appears. ▪ Select the **Map** link associated with the appropriate Media Server. The **List Media Server Address Map** page appears. ▪ Select the **Add Map In New Group** link. The **Add Media Server Address Map** page appears. Enter the following values: o **Name**: Enter a descriptive value (in this example, **incoming**). o **Pattern:** Enter the regular expression to be used for pattern matching. In this example, pattern specification for station extensions is given: **^sip:1*.** This expression will match any URI that begins with the text string "sip:1*" (the ^ matches the beginning of the line) and ending with any other combination of digits. Any matching FACs or FNEs will then be routed to the Media Server associated with this Address Map (in this example, **s8500clan-1**). (See [4] for more details on address map syntax.) o **Replace URI:** Keep the default (√). The completed form appears as follows. Click **Add** to submit the form.

Step	Description
	

Add Media Server Address Map

Host s8500-clan1

Name* incoming

Pattern* ^sip:1*

Replace URI ☑

Fields marked * are required.

Add

6. Configure the Samsung Ubigate^{TM} iBG-3026

The Samsung Ubigate^{TM} iBG-3026 provides both browser-based and command-line-based (Telnet or COM port access) administrative interfaces; however, since the full range of necessary configuration features is supported only via the command line interface (CLI), the steps in this section use only the CLI.

Step	Description
1.	Connect to the Samsung Ubigate^{TM} iBG-3026 command line interface via a Telnet interface or a terminal emulation program (e.g., HyperTerminal) using the serial cable provided for the console port at the back of the machine. Enter the Username (samsung) and default Password (see [8]). A list of available commands is displayed as follows.

```
#-----------------------------------------------------------------------
# SAMSUNG ELECTRONICS CO., LTD. Login
#-----------------------------------------------------------------------

login: samsung
password:

                            SAMSUNG ELECTRONICS CO., LTD. CLI
sarak2# ?

      clear                        access clear commands
      configure                    configure from ( flash / network / terminal
                                   )
      debug                        accesses debug commands
      file                         access file commands
      password                     Change the user password
      ping                         invoke ping
      reboot                       reboot the system
      save                         save configuration to ( local / network )
      show                         access show commands
      telnet                       open a telnet connection
      test                         access test commands
      trace                        trace route to destination address or host
                                   name

sarak2#
```

Step	Description
2.	Enter **configure terminal** to configure the Samsung Ubigate™ iBG-3026. The following menu is displayed:

```
sarak2# configure terminal
sarak2/configure#  *Aug 20,2000,06:24:56: #PARSER-warning: samsung entered
confi
g mode ON SUN AUG 20 06:24:56 2000 FROM 10.1.10.163

sarak2/configure# ?

      SYS_REM                     add comments to system related configuration
                                  at the beginning
      SYS_REM_                    add comments to system related configuration
                                  at the end
      aaa                         To configure AAA parameters
      access-group                to apply an access list to an interface
      access-list                 Add An Access list Entry
      add_dscp_for_voip           configure DSCP for VoIP
      admin_name                  Change the administrator account name
      arp                         To add an arp entry on system
      arp_timeout                 Set global ARP cache timeout for dynamic
                                  entries
      aux                         configure aux port related parameters
      bgp                         Border Gateway Protocol(BGP)
      boot_params                 Configure boot parameters
      bridge                      bridge group commands
      call-admission              Configure Call Admission
      console_timeout             configure console timeout
      crypto                      Access Crypto configuration commands
      date                        Set date
      dial-peer                   for dial-peer related command
      dot1x                       IEEE 802.1X Port-Based Access Control
      event                       configure event log display options
      firewall                    Configure firewall related commands
Press any key to continue (q : quit | enter : next line) :
```

Step	Description
3.	You may configure the Fast Ethernet port 0/0 for telnet into the router for easy administration. The telnet session is enabled by default on the router. The values shown here were used for the Samsung Ubigate™ iBG-3026 in the reference configuration.

```
sarak2# configure terminal
sarak2/configure# interface ethernet 0/0
Configuring existing Ethernet interface
sarak2/configure/interface/ethernet (0/0)# ip address 10.1.10.52/24
```

Step	Description
4.	After configuring the router with an ip address, you can re-connect into the system by telnet session, this time using the IP address configured in **Step 3** (in this example, 10.1.10.52). The saving of running configurations to a local flash can be done using the following command:

```
sarak2# save local

WARNING :
Do not remove Compact Flash or reboot during this process
Saving current system configuration.
Please wait... (up to a minute)
10318
 *Aug 20,2000,06:33:17: #PARSER-warning: samsung saved configuration ON SUN
AUG
20 06:33:17 2000 FROM 10.1.10.163
Done.
sarak2#
```

Retrieving the configuration file can be done using the following command:

```
sarak2# configure flash
  fileName : system.cfg
```

Step	Description
5.	A VoIP Gateway needs to be setup for voice calls between PSTN and IP network. Samsung Ubigate^{TM} iBG-3026 supports SIP VoIP Call processing protocols. The VoIP bind command is used to specify the source IP address for SIP signaling as well as the media stream. In this case, the interface vlan 1010 is used with a source address of 192.168.0.2. The vlan interface must be created first before binding. Note that the voip-gateway must be shutdown first before any binding.

```
sarak2/configure# voip-gateway
sarak2/configure/voip-gateway# bind control interface vlan vlan1.1010
bind if=vlan vlan1.1010 ip=192.168.0.2
sarak2/configure/voip-gateway# bind media interface vlan vlan1.1010
sarak2/configure/voip-gateway# no shut
setup primary media ip 192.168.0.2
no shutdown
Starting  VoIP Gateway
sarak2/configure/voip-gateway#  *Aug 21,2000,03:00:02: #SECC-notification:
Start
 VoIP Gateway

sarak2/configure/voip-gateway# exit
sarak2/configure#
```

Alternatively, you can set the interface using ethernet ports as below:

```
sarak2/configure# voip-gateway
sarak2/configure/voip-gateway# bind control interface ethernet 0/0
sarak2/configure/voip-gateway# bind media interface ethernet 0/1
sarak2/configure/voip-gateway# no shut
sarak2/configure/voip-gateway# exit
sarak2/configure#
```

Step | **Description**

6.

SES is specified as the Call Server in the VoIP gateway configurations. The host domain-name “sglab.com” is used in this case for the Samsung Ubigate^{TM} iBG3026.

```
sarak2/configure# voip-gateway
sarak2/configure/voip-gateway# shut
Shutdown forced VoIP Gateway
sarak2/configure/voip-gateway#  *Aug 20,2000,10:04:23: #SECC-notification: Sh
own VoIP Gateway

sarak2/configure/voip-gateway# host domain-name sglab.com
sarak2/configure/voip-gateway# call-server
sarak2/configure/voip-gateway/call-server# ip-address ipv4:10.1.10.61
ok call server 10.1.10.61
sarak2/configure/voip-gateway/call-server# exit
sarak2/configure/voip-gateway# no shut
setup primary media ip 192.168.0.2
no shutdown
Starting  VoIP Gateway
sarak2/configure/voip-gateway#  *Aug 20,2000,10:05:29: #SECC-notification: St
 VoIP Gateway

sarak2/configure/voip-gateway#
```

7.

The Samsung Ubigate^{TM} iBG-3026 analog voice port hardware information is automatically detected by the system. When the system is started, the port number of analog voice port is shown automatically and is able to be entered into basic service without further configuration. The command below shows the voice port summary on the analog voice ports. However, you need to create the dial-peer for the analog voice port on the FXS and FXO to register to the SES. This is shown in **Step 9**.

```
sarak2/configure# show voice port sum
PORT     CH SIG-TYPE   ADMIN OPER IN STATUS       OUT STATUS      EC
======== == ========== ===== ==== ============== ============== ==
0/0/0    -- fxo-ls     up    up   idle           idle           y
0/0/1    -- fxo-ls     up    up   idle           idle           y
0/0/2    -- fxo-ls     up    up   idle           idle           y
0/0/3    -- fxo-ls     up    up   idle           idle           y
0/2/0    -- fxs-ls     up    up   on-hook        idle           y
0/2/1    -- fxs-ls     up    up   on-hook        idle           y
0/2/2    -- fxs-ls     up    up   on-hook        idle           y
0/2/3    -- fxs-ls     up    up   on-hook        idle           y
```

Step	Description
8.	Verify the port for the analog voice port FXS to be configured. You can configure specific requirement as done below such as loop-start and locale. The locale is meant to specify a regional analog voice-interface-related caller-id, tone, ring, and cadence setting. This command affects only the local interface. The following countries locale are available. See [11] for the details on the cadence.

```
sarak2/configure# voice-port 0/2/0
sarak2/configure/voice-port (0/2/0)# signal loop-start
[warning] 0/2/0 : Voice device driver will be changed after "no shutdown"
command
sarak2/configure/voice-port (0/2/0)# locale ?
  cn                             China
  de                             Germany
  in                             India
  in2                            India2
  kr                             Korea Republic
  us                             United States
sarak2/configure/voice-port (0/2/0)# locale us
sarak2/configure/voice-port (0/2/0)# compand-type a-law
voice port 0/2/0, companding-law set to 'a-law'
[warning] 0/2/0 : Voice device driver will be changed after "no shutdown"
command
sarak2/configure/voice-port (0/2/0)# no shut
sarak2/configure/voice-port (0/0/0)#
```

Similar changes can be applied to the FXS.

Step	Description

9.

Dial-peer has to be created for registration in SES SIP Registrar as below. The random number tag “1” is for identifying the dial-peer. The user registration has to be created on the SES before this. Refer to **Section 5 Step 6-8.** You might need to do a “no shut” as below to start the SIP registration. The sip-ua registration can be confirmed with the “show sip-ua registrations” command.

```
sarak2/configure# dial-peer voice pots 1
sarak2/configure/dial-peer/voice/pots 1# authentication 10058 123456
sarak2/configure/dial-peer/voice/pots 1# destination-pattern 10058
sarak2/configure/dial-peer/voice/pots 1# port 0/2/0
sarak2/configure/dial-peer/voice/pots 1# register e164
sarak2/configure/dial-peer/voice/pots 1# exit pots
sarak2/configure#
```

```
sarak2/configure# voip-gateway
sarak2/configure/voip-gateway# sip-ua
sarak2/configure/voip-gateway/sip-ua# shu
sarak2/configure/voip-gateway/sip-ua# no shut
sarak2/configure/voip-gateway/sip-ua#
sarak2#  *Aug 21,2000,02:44:31: #PARSER-warning: samsung exit config mode withou
t CHANGING configuration ON MON AUG 21 02:44:31 2000 FROM 135.27.8.160

sarak2#
sarak2# show sip-ua regist
ID    DEST-PATTERN EXPIRES STATUS PORT       AUTHENTICATION
----- ------------ ------- ------ ---------- --------------
1     10058        86400   yes    0/2/0      10058[****] #
2     10059        86400   yes    0/2/1      10059[****]
-----------------------------------------------------------
Number of 2/total(2) is registered

sarak2#
```

This indicates the primary registrations for the Samsung Ubigate^{TM} iBG-3026 UA.

10.

The trunk dial-peer is configured differently from stations. The dial-peer 100 is created below for an FXO port at slot #0, mini slot #0 and port #0. Note that the trunk label “trkA”. This label should match the Far-End Domain indicated on the trunk group 6 form in **Section 4.1, Step 6**.

```
sarak2/configure# dial-peer voice pots 100
sarak2/configure/dial-peer/voice/pots 100# trunkgroup-label trkA
sarak2/configure/dial-peer/voice/pots 100# port 0/0/0
sarak2/configure/dial-peer/voice/pots 100# exit pots
sarak2/configure#
```

Step	Description
11.	To specify the set of preferred codecs, the following command is added. The values shown here were set for the Samsung Ubigate^{TM} iBG-3026 in the reference configuration. Note that codec-list 2 was put into service.

```
sarak2/configure# voice class codec 1
sarak2/configure/voice/class/codec 1#codec-preference 1 g711alaw 20
sarak2/configure/voice/class/codec 1#codec-preference 2 g711ulaw 20
sarak2/configure/voice/class/codec 1#codec-preference 3 g729 20
sarak2/configure# voice class codec 2
sarak2/configure/voice/class/codec 2# codec 1 g729 20
sarak2/configure/voice/class/codec 2# codec 2 g711alaw 20
sarak2/configure/voice/class/codec 2# codec 3 g711ulaw 20
sarak2/configure#
sarak2/configure# voice service codec-list 2
sarak2/configure#
```

Step	Description
12.	To set the DTMF tones using RFC 2833, use the command below:

```
sarak2/configure# voice service sip

sarak2/configure/voice/service/sip# dtmf rtp-nte

sarak2/configure/voice/service/sip#
```

Step	Description
13.	To verify the sip-ua default settings on the Samsung Ubigate™ iBG-3026, use the command below:

```
sarak2/configure# show sip-ua parameters
SIP Configurations
  SIP-UA : up
  Operation mode : Call-server SES mode
  Handle Name    : 10058@sglab.com
SIP UA Timers
  SIP timer T1 : 500
  SIP timer T2 : 4000
  SIP timer T4 : 5000
  Keep alive (OPTIONS) duration time : 30
  Minimum Session Timer is Not Used
SIP UA Parameters
  SIP-UA default dtmf relay : RTP NTE
  SIP-UA default transport : udp
  SIP-UA default uri type : sip
  SIP-UA default UDP port : 5060
  SIP-UA default TCP port : 5060
  SIP-UA default TLS port : 5061
  SIP-UA default max forwards : 70
  SIP rel1xx is supported
  SIP Redirect ip to ip : no
  SIP Inband alerting : no
  SIP Redirection : no
  SIP Send SDP in 183
  SIP Suspend Resume : no
  SIP offer hold : direction attribute sendonly (RFC 3264)
  PSTN code for SIP Request CANCEL : 16
  SIP Early media at 180 : Enabled
  SIP Reason-header Override : no
  SIP no answer timer value for sip outbound call 120
  SIP calling-info PSTN-to-SIP unscreened discard : no
  SIP calling-info SIP-to-PSTN unscreened discard : no
Home Server Information
  System port : 5060
  UDP port : 5060
  TCP port : 5060
  TLS port : 5061
sarak2/configure#
```

7. Verification Steps

The following steps can be used to verify that the configuration steps documented in these Application Notes have been done correctly.

- From Avaya Communication Manager's SAT:
  - To verify that the SIP trunk group is in service, enter **status trunk *n*** (where ***n*** is the number of the trunk group to be verified).
  - To verify that the SIP signaling group is in-service, enter **status signaling-group *n*** (where ***n*** is the number of the signaling group to be verified).
- From Avaya SIP Enablement Services' Administration Web Interface:
  - To verify that an analog telephone behind the Samsung Ubigate^{TM} iBG-3026 can register with Avaya SIP Enablement Services, select **User → Registered Users**. Also, you can use the command **show sip-ua registrations** on the Samsung Ubigate^{TM} -3026 CLI to verify as well.
- Verify that a call can be placed between two analog telephones behind the Samsung Ubigate^{TM} iBG-3026. You can use the command **show sip-ua call-connections** to verify the call as well.
- Verify that a call can be placed between an analog telephone behind the Samsung Ubigate^{TM} iBG-3026 and a telephone in the PSTN through the Ubigate^{TM} iBG-3026 Analog trunk
- Verify that a call can be placed between an analog telephone behind the Samsung Ubigate^{TM} iBG-3026 and an Avaya H.323 IP telephone in the main or remote location.
- Verify that a call can be placed between an analog telephone behind the Samsung Ubigate^{TM} iBG-3026 and an Avaya 4600 Series SIP IP telephone in the main or remote location.
- Verify that a call can be placed between an analog telephone behind the Samsung Ubigate^{TM} iBG-3026 and an analog telephone behind the Avaya CM in the main location.

8. Conclusion

The Samsung Ubigate^{TM} iBG-3026 can successfully register to Avaya SIP Enablement Services and support the telephony features of Avaya Communication Manager.

9. Additional References

The following Avaya product documentation can be found at http://support.avaya.com.

[1] *Feature Description and Implementation For Avaya Communication Manager*, Issue 4.0, February 2006, Document Number 555-245-205.

[2] *Administrator Guide for Avaya Communication Manager*, Issue 2.1, May 2006, Document Number 03-300509.

[3] *Installing and Administering SIP Enablement Services R3.1*, Issue 1.4, February 2006, Document 03-600768.

[4] *SIP Support in Release 3.1 of Avaya Communication Manager Running on the Avaya S8300, S8500, S8500B, S8700, and S8710 Media Server*, February 2006, Issue 6, Document Number 555-245-206.

[5] *4600 Series IP Telephone Release 2.6 LAN Administrator Guide,* August 2006, Issue 4, Document Number 555-233-507.

[6] *Avaya Extension to Cellular User's Guide,* Issue 9, February 2006, Document Number 210-100-700.

[7] *Avaya Extension to Cellular and OPS Installation and Administration Guide*. January 2005, Issue 8, Document Number 210-100-500.

The following is Samsung Ubigate^{TM} iBG-3026 guide is available from Samsung. Visit http://www.samsungen.com for company and product information. However, you must be a registered partner of Samsung Electronics.

[8] *Ubigate iBG3026^{TM} Configuration Guide.*
[9] *Ubigate iBG3026^{TM} Command Reference.*
[10] *Handbook of the Ubigate Systems 3026.*
[11] *iBG3026_iBG-DM User Guide.*
[12] *iBG3026_Installation Manual.*
[13] *iBG3026_System Description.*
[14] *iBG3026_Message Reference Manual.*

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