Avaya Solution & Interoperability Test Lab

Configuration Notes for Configuring Avaya Communication Manager and Avaya SIP Enablement Services to Support Enterprise Reach for IPC – Issue 1.0

Abstract

These Application Notes describe the procedures for configuring Avaya Communication Manager and Avaya SIP Enablement Services to support Enterprise Reach for IPC.

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

1. Introduction

The IPC/Avaya Enterprise Reach (ER) solution allows IPC Alliance lines to be shared with Avaya phones. The IPC PBXUA (PBX User Agent) and B2BUA (Back to Back User Agent) are software components within this architecture. The IPC PBXUA and B2BUA are responsible for enforcing line states in Avaya Communication Manager such that they mirror the line states in IPC Alliance.

The IPC PBXUA is a SIP B2BUA software automaton that runs on the IPC Enterprise SIP Server (ESS). The IPC PBXUA uses one SIP UA (User Agent) context to communicate with the IPC Alliance/ESS (IPC Enterprise SIP Server) and the other SIP UA context to communicate with Avaya Communication Manager. The IPC PBXUA is configured as an Avaya 4620 SIP phone device. The IPC PBXUA registers with Avaya SIP Enablement Services and subscribes for dialog-package events for bridged line appearance (BLA) groups. As a part of enforcing line states, the IPC PBXUA places outbound calls and answers inbound calls.

The IPC B2BUA is a SIP B2BUA that runs on the IPC ESS (IPC Enterprise SIP Server) and controls creating the call leg to the Alliance line and enforcing line states. The IPC B2BUA is responsible for placing a call into Avaya Communication Manager to show incoming ring on an Alliance line as well as working in conjunction with the PBXUA to show busy, hold and idle states.

1.1. Interoperability Compliance Testing

IPC Alliance Enterprise Reach (ER) allows the sharing of its telephony lines with Avaya Communication Manager which is enabled with a capability that allows it to integrate to the IPC Alliance ER implementation. Line type of the Alliance that be shared, vary between the most simplistic telephony presentation, Private Wire Manual Ring Down (PW MRD), Private wire Circuits, Analog PSTN dial tone, Channel Associated Signaling System, T1 line side and ISDN services to the most complex, Q point Signaling System (QSIG).

The interoperability compliance testing is based on the Private Wire Circuits and QSIG line types and following functions. Lines that are enabled at the Alliance for Enterprise Reach should be presented to user extensions of the associated Avaya Communication Manager:

- Inbound calls will be alerted at both the Alliance turrets and Avaya Communication Manager user extensions:
    - Call is answered at either party (IPC Alliance or Avaya extension); the busy status will be indicated at the corresponding party.
    - Answered calls, may then be placed on hold by the party that is in 'conversation state', and then 'hold' status is indicated at the corresponding party; and dependant on the call state, either:
        - Intrude on a call (Alliance may invoke privacy to prevent Avaya Communication Manager user intrusion), multi party conversation can be made
        - Retrieve held calls (even they were placed on hold by another).
    - Only the party in 'conversation state' can release the call.

During these changes of call state, the current state would be reflected and indicated to all those parties sharing that line.

- Similarly outward bound calls can be made from Avaya Communication Manager extension on the Alliance enabled ER lines, in the following scenarios:
  - Dial tone type lines, Avaya Communication Manager extensions can dial to establish a routed call path.
  - Private wire lines, Avaya Communication Manager extensions can activate 'ring down'.

  After the call has been established, or is being established, similar functions as to 'inbound calls' can be invoked and reflected/indicated at the sharing parties of that line.

ER (Enterprise Reach) function can therefore afford:

- Co-sharing of lines presented to Alliance, to users of Avaya Communication Manager, on a peer equal basis, where the presentation of this line may also be to many Avaya Communication Manager users/extensions (on the same Avaya Communication Manager), and where status of the line is indicated to all.
- Co-sharing of the lines without geographical limitation (assuming IP connectivity availability) where the sharing Avaya Communication Manager maybe in a different geographical location to the Alliance.
- The ability for Alliance ER enabled lines to be shared with two Avaya Communication Managers (of the same type and software version) so that a call can be shared simultaneously between Alliance and users of both these Avaya Communication Manager.

1.2. Support

Technical support for the Avaya products can be obtained from Avaya. See the support link at support.avaya.com for contact information.

Technical support for the IPC products can be obtained from IPC. See the support link at www.ipc.com for contact information.

2. Reference Configuration

Table 1 shows the IPC Line Type, IPC lines on the IPC turrets, Speed Dial Numbers and the bridged line appearances (BLA) associated with the IPC lines using in this example. PBXUA is an extension which has to be configured both on IPC and Avaya side. Also it has to be registered on Avaya SIP Enablement Services. In this sample application notes, extension 6605 is used.

Note: The PBXUA extension (6605) and the BLAs used in the Avaya configuration must match what is configured on the IPC for Enterprise Reach to work correctly. LAC/VLAC is virtual logical address codes, which are assigned and associated with keys of the IPC turrets. Speed Dial Number is configured as button assignments on Avaya extensions and associated with the BLAs, these numbers is used to originate outbound calls over the ER PW or ER QSIG.

IPC Line Type	LAC/VLAC/DDI	Speed Dial Number	Bridged Line Appearance (BLA)
ER PW	1138/Private 1	301501138	6605
ER PW	1140/Private 2	301501140	6601
ER QSIG	63574/3102	301563574	6602
ER QSIG	63576/3104	301563576	6603

Table 1 – Bridged Line Appearances Corresponding to IPC lines

Private Line Interface Card (PLIC) extensions (e.g., Private 1 and Private 2) were configured on the IPC turrets. Bridged Line Appearances (e.g., 6605 and 6601) were configured on Avaya Communication Manager to reflect the status (i.e., idle, busy, or hold) of the PLICs.
QSIG extensions (e.g., 3102 and 3104) was configured on the IPC turrets, Bridged Line Appearances (e.g., 6602 and 6603) was configured on Avaya Communication Manger to reflect the status (i.e., idle, busy, or hold) of the QSIG extensions.

The PBXUA (6605) is defined as an OPTIM (Off-PBX Telephone Integration and Mobility) station on Avaya Communication Manager and as a user with a media server extension on the Avaya SIP Enablement Services.

The speed dial numbers are configured on the Avaya 4620SW H.323 and Avaya 2420 DCP phones, associated with BLAs.

The PLIC speed dial is used to originate outbound calls over the PLIC when the corresponding BLA (i.e., 6601 or 6605) was selected. The QSIG speed dial is used to originate outbound calls over QSIG when the corresponding BLA (Bridged Line Appearance) (i.e., 6602 or 6603) was selected.

There are few scenarios to show how to verify using the sample configuration. ***Note:*** For Avaya user to make a call on an IPC Enterprise Reach line, the user has to undertake a number of button presses to achieve connection with the distant party.

Scenario 1: Avaya user places an outbound call from BLA (6601) to verify IPC's PLIC extension (Private 2) status. Avaya user must select the appropriate BLA line key (6601), then dial the Enterprise reach line access number (301501140) by pressing the speed dial button. Verify Private 2 on the IPC turrets indicates busy.

Scenario 2: Avaya user places an outbound call to IPC's QSIG extension (3104) from BLA (6602), and verifies IPC's QSIG extension (3104) status. Avaya user select the appropriate BLA line key (6602), then dial the Enterprise reach line access number (301563574) by pressing the speed dial button. After hearing dial tone, call QSIG extension 3104. Verify 3104 on the IPC turrets indicates an incoming call.

Scenario 3: IPC user places an outbound call from Private 1 to verify Avaya BLA (6605) status. Place an outbound call at Private 1 from the IPC Turrets by selecting Private 1 on an IPC Turret. Verify that BLA (6605) on Avaya stations indicate busy.

Scenario 4: IPC user places an inbound call to QSIG extension (3104) from an IPC turret, and verifies Avaya BLA (6603) status. Select QSIG extension (3102) and dial 3104 on an IPC turret, verify that BLA (6603) on Avaya stations ring.

Figure 1 illustrates the configuration that was used to verify these Application Notes.

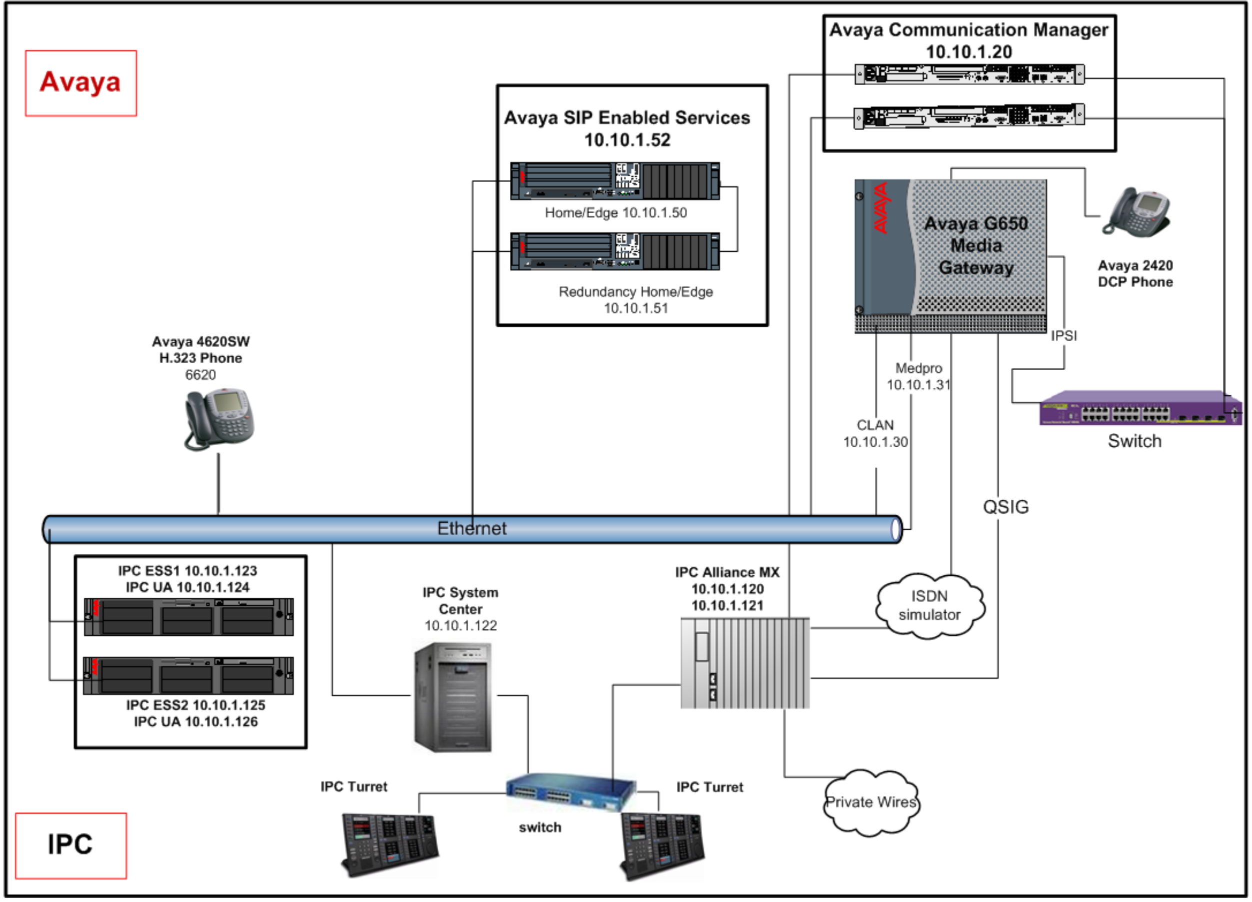


Figure 1: Network Diagram of the Compliance Tested Configuration

Note: The QSIG link and ISDN link between Avaya Communication Manager and IPC Alliance MX are used for test purpose only. The configurations for these links are not included in these Application Notes. These Application Notes consists of:

- Configure Avaya Communication Manager (**Section 4**)
  - Configure Avaya Communication Manager System Parameters
  - Configure SIP trunk between Avaya Communication Manager and Avaya SIP Enablement Services
  - Configure the Line Sharing IP Stations
  - Configure Avaya IP Station
  - Configure Avaya DCP Station
  - Configure IPC PBXUA Station
- Configure Avaya SIP Enablement Services (**Section 5**)

There is no IPC configuration documented in these Application Notes based on IPC support policy. IPC engineer will be responsible to install and for the maintenance of IPC products.

3. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment	Software
Avaya S8730 Servers (2)	Avaya Communication Manager 5.1.1 sp3 Build 415.1-17105
Avaya G650 Media Gateway : • IPSI (TN2312AP) • C-LAN (TN799DP) • MEDPRO (TN2302AP) • Digital Line (TN2214CP)	 HW02 FW044 HW01 FW026 HW11 FW118 HW05 FW015
Avaya S8510 Server (2)	Avaya SIP Enablement Services 5.1.2 Build SIP Enablement Services-01.2.416.4b-SP0
Avaya 4620SW IP Telephones	2.9(H.323)
Avaya 2420 Digital Telephones	---
IPC Information Systems Alliance MX IPC System Center (Sun ULTRA 25) IPC IQ/MAX Turrets	Alliance Release 15.02.00.10
IPC ESS (SIP Proxy Server)	2.00.01-8

Table 1– Equipment and Software Version Validated

4. Configure Avaya Communication Manager

The steps in this section describe the configuration for Avaya Communication Manager to support IPC Enterprise Reach solution.

4.1. Configure Avaya Communication Manager System Parameters

This section describes how to configure the Avaya Communication Manager system parameters and features. The commands listed in the following sections were issued at the Avaya System Access Terminal (SAT).

Step	Description
1.	Enter **display system-parameters customer-options** command. On **Page 1** verify that the license file has allocated enough OPS extensions to support all SIP endpoints. If not, an authorized Avaya support technician will need to install an appropriately enabled license file.

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

     G3 Version: V15                              Software Package: Standard
       Location: 2                             RFA System ID (SID): 1
       Platform: 6                             RFA Module ID (MID): 1

                                                              USED
                                Platform Maximum Ports: 48000 248
                                      Maximum Stations: 36000 30
                              Maximum XMOBILE Stations: 10    1
                    Maximum Off-PBX Telephones - EC500: 10    1
                    Maximum Off-PBX Telephones -   OPS: 100   7
                    Maximum Off-PBX Telephones - PBFMC: 0     0
                    Maximum Off-PBX Telephones - PVFMC: 0     0
```

On **Page 2**, verify that the **Maximum Administered SIP Trunks** is enough to support the expected total traffic to and from all Avaya and IPC extensions. Any call involving a SIP endpoint will use one 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: 100   0
           Maximum Concurrently Registered IP Stations: 12000 2
             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 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 100   100
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

Step	Description
	On **Page 3**, verify that **ARS** and **ARS/AAR Partitioning** are set to **y**.

```
display system-parameters customer-options                      Page   3 of  10
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? n           Audible Message Waiting? n
        Access Security Gateway (ASG)? n               Authorization Codes? n
        Analog Trunk Incoming Call ID? n                        CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? n                          CAS Main? n
Answer Supervision by Call Classifier? n                 Change COR by FAC? n
                                  ARS? y  Computer Telephony Adjunct Links? n
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? y                       DCS (Basic)? y
          ASAI Link Core Capabilities? n                 DCS Call Coverage? y
          ASAI Link Plus Capabilities? n                DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? n
              ATM WAN Spare Processor? n                           DS1 MSP? n
                                 ATMS? n             DS1 Echo Cancellation? n
                  Attendant Vectoring? n
```

On **Page 4**, verify that the following bold items are set to **y**.

```
display system-parameters customer-options                      Page   4 of  10
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                              IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                        ISDN Feature Plus? y
                  Enhanced EC500? y        ISDN/SIP Network Call Redirection? y
    Enterprise Survivable Server? n                          ISDN-BRI Trunks? y
       Enterprise Wide Licensing? n                                 ISDN-PRI? y
              ESS Administration? n               Local Survivable Processor? n
          Extended Cvg/Fwd Admin? y                     Malicious Call Trace? y
     External Device Alarm Admin? n                 Media Encryption Over IP? y
  Five Port Networks Max Per MCC? n     Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? n                 Multifrequency Signaling? y
      Global Call Classification? n         Multimedia Call Handling (Basic)? y
             Hospitality (Basic)? y      Multimedia Call Handling (Enhanced)? y
 Hospitality (G3V3 Enhancements)? n               Multimedia IP SIP Trunking? y
                       IP Trunks? y
           IP Attendant Consoles? n
```

On **Page 5**, verify that **Private Networking** and **Uniform Dialing Plan** are set to **y**.

```
display system-parameters customer-options                      Page   5 of  10
                                OPTIONAL FEATURES

                Multinational Locations? y             Station and Trunk MSP? n
 Multiple Level Precedence & Preemption? y      Station as Virtual Extension? n
                     Multiple Locations? y
                                             System Management Data Transfer? n
          Personal Station Access (PSA)? y               Tenant Partitioning? n
                        PNC Duplication? y       Terminal Trans. Init. (TTI)? y
                   Port Network Support? y               Time of Day Routing? n
                        Posted Messages? y       TN2501 VAL Maximum Capacity? y
                                                        Uniform Dialing Plan? y
                     Private Networking? y     Usage Allocation Enhancements? y
               Processor and System MSP? n
                     Processor Ethernet? y                Wideband Switching? n
                                                                    Wireless? n
                          Remote Office? n
           Restrict Call Forward Off Net? y
                  Secondary Data Module? y
```

Step	**Description**
	On **Page 8**, verify that the following bold items are set to **y**.

```
display system-parameters customer-options                      Page   8 of  10
                             QSIG OPTIONAL FEATURES

                                         Basic Call Setup? y
                             Basic Supplementary Services? y
                                    Centralized Attendant? y
                                    Interworking with DCS? n
                    Supplementary Services with Rerouting? y
                            Transfer into QSIG Voice Mail? y
                                       Value-Added (VALU)? y
       (NOTE: You must logoff & login to effect the permission changes.)
```

2.	Enter **change system-parameters features** command On **Page 16**, turn on shuffling by setting **Direct IP-IP Audio Connections** to **y**.

```
change system-parameters features                               Page  16 of  17
                        FEATURE-RELATED SYSTEM PARAMETERS

AUTOMATIC EXCLUSION PARAMETERS
                       Automatic Exclusion by COS? n




                              Recall Rotary Digit: 2

         Duration of Call Timer Display (seconds): 3
WIRELESS PARAMETERS
  Radio Controllers with Download Server Permission (enter board location)
    1:         2:         3:         4:         5:

IP PARAMETERS
                   Direct IP-IP Audio Connections? y
                             IP Audio Hairpinning? y
RUSSIAN MULTI-FREQUENCY PACKET SIGNALING
                                           Re-try? n
     T2 (Backward Signal) Activation Timer (secs): 20
```

Step	Description
3.	Enter **change feature-access-codes** command to assign private network access code. On **Page 1**, verify **Auto Alternate Routing (AAR) Access Code** is set. In this example, AAR Access Code is set to **1**.

```
change feature-access-codes                                    Page   1 of   8
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code: #3
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 1
    Auto Route Selection (ARS) - Access Code 1: *7    Access Code 2:
                 Automatic Callback Activation: *4     Deactivation: #4
Call Forwarding Activation Busy/DA: *2     All: *3     Deactivation: #2
   Call Forwarding Enhanced Status:        Act: 622    Deactivation: 623
                         Call Park Access Code: #5
                       Call Pickup Access Code: *6
CAS Remote Hold/Answer Hold-Unhold Access Code: #6
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
                   Contact Closure   Open Code:          Close Code:
```

Step	Description
4.	Enter **change dialplan parameters** command to assign **Local Node Number**. Ensure PBX has an assigned **Local Node Number**, if there is no assigned number, enter **1**.

```
change dialplan parameters
                             DIAL PLAN PARAMETERS

                 Local Node Number: 1
                   ETA Node Number:
               ETA Routing Pattern:
        UDP Extension Search Order: local-extensions-first
      AAR/ARS Internal Call Prefix:
AAR/ARS Internal Call Total Length:
Retry ARS/AAR Analysis If All-Location Entry Inaccessible? n

 EXTENSION DISPLAY FORMATS
                                    Inter-Location/SAT     Intra-Location
         6-Digit Extension:         xx.xx.xx               xx.xx.xx
         7-Digit Extension:         xxx-xxxx               xxx-xxxx
         8-Digit Extension:         xx.xx.xx.xx            xx.xx.xx.xx
         9-Digit Extension:         xxx-xxx-xxx            xxx-xxx-xxx
        10-Digit Extension:         xxx-xxx-xxxx           xxx-xxx-xxxx
        11-Digit Extension:         xxxx-xxx-xxxx          xxxx-xxx-xxxx
        12-Digit Extension:         xxxxxx-xxxxxx          xxxxxx-xxxxxx
        13-Digit Extension:         xxxxxxxxxxxxx          xxxxxxxxxxxxx
```

4.2. Configure SIP Trunk between Avaya Communication Manager and Avaya SIP Enablement Services

This section describes configuration of SIP trunk between Avaya Communication Manager and Avaya SIP Enablement Services. The commands listed in the following sections were issued at the Avaya System Access Terminal (SAT).

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Enter display node-names ip command, in this sample application, values below are assigned for C-LAN (clan), IP Media Processor (medpro) and SES (ses).
<pre>
display node-names ip   Page   1 of   2
                                             IP NODE NAMES
    Name   IP Address
clan   10.10.1.30
default   0.0.0.0
gateway   10.10.1.5
medpro   10.10.1.31
procr   10.10.1.18
ses   10.10.1.52
</pre></td></tr>
<tr><td>2.</td><td>Enter change ip-codec-set 1 to specify the audio codecs to be used for calls routed to and from IPC extensions via Avaya SIP Enablement Services. The codecs supported by the IPC are listed below. In addition, Media Encryption preference is set to none, since IPC doesn't support media encryption. Change Frames Per Pkt will affect Packet Size (ms). In this example, Frame Per Pkt is set to 3.
<pre>
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   3   30
 2: G.711MU   n   3   30
 3: G.729B   n   3   30
 4:
 5:
 6:
 7:

     Media Encryption
 1: none
 2:
 3:
</pre></td></tr>
</table>

Step	Description
3.	Enter **display ip-network region *n***, where ***n*** is the IP network region where the Avaya SIP Enablement Services server resides. **Intra-region IP-IP Direct Audio**, **Inter-region IP-IP Direct Audio** could be set to **yes** to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway (a feature known as shuffling). **Authoritative Domain** field is set to the SIP Domain of the Avaya SIP Enablement Services administered in **Section 5 Step 2**, (e.g., du.rnd.avaya.com). Assign the Codec Set (refers to **Step 2**) to use for this Network Region.

```
display ip-network-region 1                                    Page   1 of  19
                                IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: du.rnd.avaya.com
    Name: ses_local
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? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

4.	Use the **add signaling-group *n*** command to add a signaling group for the SIP connection between Avaya Communication Manager and Avaya SIP Enablement Services. **Near-end Node Name/Far-end Node Name** is set to the name assigned to the IP address of the Avaya CLAN/SIP Enablement Services in **Step 1** of this section. Set the **Far-end Network Region** to the IP Network Region (e.g., **1**) defined in **Step 3**. **Far-end Domain** is set to blank for the SIP trunk between Avaya Communication Manager and Avaya SIP Enablement Services. This will allow INVITE messages to be sent by IPC using the IP address of the IPC ESS (IPC Enterprise SIP Server) instead of a domain name. Also verify that the others **bold** items are set.

```
add signaling-group 1                                       Page   1 of   1
                                SIGNALING GROUP
 Group Number: 1             Group Type: sip
                       Transport Method: tls
          IP Video? n
   Near-end Node Name: clan                 Far-end Node Name: ses
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1
      Far-end Domain:
                                         Bypass If IP Threshold Exceeded? n
        DTMF over IP: rtp-payload         Direct IP-IP Audio Connections? y
                                                    IP Audio Hairpinning? y
        Enable Layer 3 Test? n
 Session Establishment Timer(min): 3          Alternate Route Timer(sec): 6
```

Step	Description
5.	Use the **add trunk-group *n*** command to add a trunk group between Avaya Communication Manager and Avaya SIP Enablement Services. On **Page 1**, verify that the **Number of Members** field is appropriate to support the anticipated traffic involving the turrets, but not exceeding the maximum number of available SIP trunks as indicated in **Section 4.1 Step 1**. Also verify that the others **bold** items are set. **Page 1** of the trunk group form is shown below.

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

Group Number: 1                    Group Type: sip           CDR Reports: y
  Group Name: ses_local                   COR: 1        TN: 1        TAC: 511
   Direction: two-way       Outgoing Display? y
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 1
                                                     Number of Members: 50
```

Page 2 of the trunk group form is shown below. **Preferred Minimum Session Refresh Interval (sec)** is set depending on customers requirements.

```
add trunk-group 1                                           Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name? y

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 120
```

Page 3 of the trunk group form is shown below. Verify **Numbering Format** is set to **public** and **Replace Restricted Number** is set to **y**.

```
add trunk-group 1                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                          Maintenance Tests? y



                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? n


Show ANSWERED BY on Display? n
```

Step	Description
	Page 5 of the trunk group form is shown below. Verify all ports are assigned.

```
display trunk-group 1                                           Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/50
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  50

       Port             Name
  1: T00030             ses_local
  2: T00031             ses_local
  3: T00032             ses_local
  4: T00033             ses_local
  5: T00034             ses_local
  6: T00035             ses_local
  7: T00036             ses_local
  8: T00037             ses_local
  9: T00038             ses_local
 10: T00039             ses_local
 11: T00040             ses_local
 12: T00041             ses_local
 13: T00042             ses_local
 14: T00043             ses_local
 15: T00044             ses_local
```

Step	Description
6.	Use the **change dialplan analysis** command to define numbers used by Enterprise Reach. In this example, 30xxxxxxx is defined as speed dial number in **Section 2**. Therefore, add an entry in the Dial Plan Analysis Table for 9-digit numbers beginning with **30** to use the Uniform Dial Plan (UDP) table.

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

       Dialed   Total  Call     Dialed   Total  Call     Dialed   Total  Call
       String   Length Type     String   Length Type     String   Length Type
     1            1    fac     88           4    ext
     2            4    udp     8889         4    udp
     3            4    ext     972          5    udp
     30           9    udp     99           4    ext
     3005         8    udp     *            2    fac
     31           4    udp     #            2    fac
     33           4    udp
     38           5    aar
     4            4    aar
     4            5    ext
     5            3    dac
     6            3    fac
     61           4    ext
     66           4    ext
     77           4    ext
```

Step **Description**

7. Use the **change uniform-dialplan *n*** command to add an entry to route 9-digit numbers beginning with 30 using Alternate Automatic Routing (AAR).

```
change uniform-dialplan 0                                       Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                              Percent Full: 0

  Matching                   Insert              Node
  Pattern       Len Del      Digits     Net Conv Num
 22              4   0                  aar  n
 30              9   0                  aar  n
 3005            8   0                  aar  n
 31              4   0                  aar  n
 33              4   0                  aar  n
 8889            4   0                  aar  n
 972             5   0                  aar  n
```

8. Use the **change route-pattern *n*** command to route calls for pattern **1** using trunk group **1** which is the SIP trunk between Avaya Communication Manager and SIP Enablement Services.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: IPC_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: 1    0                                                              n   user
 2:                                                                     n   user
 3:                                                                     n   user
 4:                                                                     n   user
 5:                                                                     n   user
 6:                                                                     n   user

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

Step	Description
9.	Use the **change aar analysis *n*** command to specify which route pattern to use based upon the number dialed. Add entries in the AAR Analysis Table to route 9-digit calls beginning with **30** using **Route Pattern 1** via the SIP trunk between Avaya Communication Manager and SIP Enablement Services.

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

          Dialed           Total     Route     Call   Node   ANI
          String          Min  Max  Pattern    Type   Num    Reqd
    2                      4    4    999       aar           n
    30                     9    9    1         aar           n
    3005                   8    8    1         aar           n
    31                     4    4    4         aar           n
    33                     4    4    1         aar           n
    38                     5    5    3         aar           n
    4                      4    4    3         aar           n
    5                      7    7    999       aar           n
    7777                   4    4    6         aar           n
    8                      7    7    999       aar           n
    888                    4    4    2         aar           n
    9                      7    7    999       aar           n
    972                    5    5    5         aar           n
```

10.	Use the **change public-unknown-numbering** command to add an entry so that calls placed from stations with a 4-digit extension beginning with **66** and routed over trunk groups will send a 4-digit calling party number to the far end.

```
change public-unknown-numbering 0                              Page   1 of   2
                        NUMBERING - PUBLIC/UNKNOWN FORMAT
                                          Total
Ext Ext           Trk       CPN            CPN
Len Code          Grp(s)    Prefix         Len
                                                   Total Administered: 3
 5  4                                      5          Maximum Entries: 9999
 4  61                                     4
 4  66                                     4
```

4.3. Configure the Line Sharing IP Stations

Configure the IP stations that will be used to indicate the status of the PLICs and QSIG extensions (refers to **Section 2**) on the IPC Turrets.

Step	Description
1.	Use the **add station *n*** command (not shown) to add extensions **6601**, **6602** and **6603** corresponding to Alliance Private 1, QSIG extension 3102 and 3104 on the IPC Turrets. The output of the **change station *n*** command is used to show the configuration after station 6601 was added. Repeat this step to add station **6602** and **6603**. These extensions must match the extensions that have been defined in the IPC configuration.

```
change station 6601                                              Page   1 of   5
                                     STATION

Extension: 6601                          Lock Messages? n                BCC: 0
     Type: 4610                          Security Code: 123456            TN: 1
     Port: S00013                      Coverage Path 1:                  COR: 1
     Name: 6601                        Coverage Path 2:                  COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 6601
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english
 Survivable GK Node Name:
          Survivable COR: internal                Media Complex Ext:
   Survivable Trunk Dest? y                             IP SoftPhone? n

                                                            IP Video? n

                                              Customizable Labels? y
```

Page 2 of the station form is shown below.

```
change station 6601                                              Page   2 of   5
                                     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
                                                         EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
    MWI Served User Type:                       Display Client Redirection? n
              AUDIX Name:                      Select Last Used Appearance? n
                                                 Coverage After Forwarding? s
                                                   Multimedia Early Answer? n
                                               Direct IP-IP Audio Connections? y
  Emergency Location Ext: 6601          Always Use? n IP Audio Hairpinning? n
     Precedence Call Waiting? y
```

Step	Description
	Page 3 of the station form is shown below.

```
change station 6601                                             Page   3 of   5
                                     STATION

          Conf/Trans on Primary Appearance? n
Bridged Appearance Origination Restriction? n


       Call Appearance Display Format: disp-param-default
IP Phone Group ID:

                              ENHANCED CALL FORWARDING
                                       Forwarded Destination        Active
 Unconditional For Internal Calls To:                                   n
                   External Calls To:                                   n
          Busy For Internal Calls To:                                   n
                   External Calls To:                                   n
      No Reply For Internal Calls To:                                   n
                   External Calls To:                                   n

            SAC/CF Override: No
```

Page 4 of the station form is shown below.

```
change station 6601                                             Page   4 of   5
                                     STATION
 SITE DATA
       Room:                                            Headset? n
       Jack:                                            Speaker? n
      Cable:                                           Mounting: d
      Floor:                                        Cord Length: 0
   Building:                                          Set Color:

ABBREVIATED DIALING
     List1:                    List2:                    List3:




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

4.4. Configure Avaya IP Station

Configure a H.323 station that will be used to share the PLIC and QSIG extensions (refers to **Section 2**) on the IPC Turrets.

Step	Description
1.	The **change station** command is used to show the configuration of the Avaya 4620 H.323 telephone after extension was added. In this example, Avaya extension **6620** is used.

```
change station 6620                                             Page   1 of   5
                                     STATION

Extension: 6620                          Lock Messages? n               BCC: 0
     Type: 4620                          Security Code: 1234             TN: 1
     Port: S00021                      Coverage Path 1: 3               COR: 2
     Name: H323 6620                   Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 6620
            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

                                                         IP Video? n


                                              Customizable Labels? y
```

Page 2 of the station form is shown below. The **Bridge Call Alerting** field should be set to **y** if ringing is desired for the BLA corresponding to the line on the IPC Turrets. Turn on shuffling by setting **Direct IP-IP Audio Connections?** to **y**.

```
change station 6620                                             Page   2 of   5
                                     STATION
FEATURE OPTIONS
          LWC Reception: spe               Auto Select Any Idle Appearance? n
         LWC Activation? n                          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
                                                         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
                                                     Multimedia Early Answer? n
                                              Direct IP-IP Audio Connections? y
  Emergency Location Ext: 6620         Always Use? n IP Audio Hairpinning? y
```

Step	Description
	Page 4 of the station form is shown below. Bridged Line Appearance (**brdg-appr**) **6601** will be used to show the status of Private 1 on the IPC Turrets. Bridged Line Appearance (**brdg-appr**) **6602** will be used to show the status of QSIG 3102 on the IPC Turrets. Corresponding speed dial numbers are configured through **autodial** button.

```
change station 6620                                            Page   4 of   5
                                     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: brdg-appr  B:1  E:6602
 2: call-appr                            6: autodial   Number: 301563574
 3: brdg-appr  B:1  E:6601               7: call-fwd   Ext:
 4: autodial   Number: 301501140         8: cfwd-bsyda Ext:
```

4.5. Configure Avaya DCP Station

Configure the DCP station that will be used to share the PLIC and QSIG extensions (refers to **Section 2**) on the IPC Turrets.

Step	Description
1.	The **change station *n*** command is used to show the configuration of the Avaya 2420 DCP telephone. In this example, Avaya extension 6622 is uesd.

```
change station 6622                                             Page   1 of   5
                                     STATION

Extension: 6622                         Lock Messages? n               BCC: 0
     Type: 2420                         Security Code:                  TN: 1
     Port: 01A1301                    Coverage Path 1:                 COR: 1
     Name: DCP 2420                   Coverage Path 2:                 COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 2         Personalized Ringing Pattern: 1
             Data Option: none                  Message Lamp Ext: 6622
            Speakerphone: 2-way              Mute Button Enabled? y
        Display Language: english               Expansion Module? n

          Survivable COR: internal             Media Complex Ext:
   Survivable Trunk Dest? y                         IP SoftPhone? n

                                                        IP Video? n


                                             Customizable Labels? y
```

Page 2 of the station form is shown below. The **Bridge Call Alerting** field should be set to **y** if ringing is desired for the Bridged Line Appreance corresponding to the line on the IPC Turrets. Turn on shuffling by setting **Direct IP-IP Audio Connections** to **y.**

```
change station 6622                                             Page   2 of   5
                                     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
                                                         EMU Login Allowed? n
        H.320 Conversion? n         Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: basic
    MWI Served User Type: qsig-mwi              Display Client Redirection? n
                                               Select Last Used Appearance? n
                                                 Coverage After Forwarding? s
                                                   Multimedia Early Answer? n
                                            Direct IP-IP Audio Connections? y
  Emergency Location Ext: 6622                         IP Audio Hairpinning? n
     Precedence Call Waiting? y
```

Step	Description
	Page 4 of the station form is shown below. Bridged Line Appearance (**brdg-appr**) **6605** will be used to show the status of Private Line 1 on the IPC Turrets. Bridged Line Appearance (**brdg-appr**) **6603** will be used to show the status of IPC QSIG extension 3104 (refers to **Section 2**).

```
change station 6622                                                 Page   4 of   5
                                     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: brdg-appr  B:1  E:6605
 2: call-appr                           6: autodial   Number: 301501138
 3: brdg-appr  B:1  E:6603              7:
 4: autodial   Number: 301563576        8:

    voice-mail Number:
```

4.6. Configure the IPC PBXUA Station

Configure the station that will be used by the IPC PBXUA.

Step	Description
1.	Use the **add station** command (not shown) to add an OPS station for the PBXUA. This extension must match the extension that has been defined in the IPC configuration. Enter **4620** in the **Type** field. Enter a name for the station (e.g., **PBXUA**) in the **Name** field.

```
change station 6605                                             Page   1 of   5
                                     STATION

Extension: 6605                          Lock Messages? n               BCC: 0
     Type: 4620                          Security Code: 123456           TN: 1
     Port: S00029                      Coverage Path 1:                 COR: 1
     Name: PBXUA                       Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 6605
            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

                                                         IP Video? n


                                              Customizable Labels? y
```

Page 2 of the station form is shown below. Verify **Restricted Last Appearance** is set to **n** and turn on shuffling by setting **Direct IP-IP Audio Conncetions?** to **y**.

```
change station 6605                                             Page   2 of   5
                                     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? n
  Active Station Ringing: single
                                                         EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: enhanced
    MWI Served User Type:                       Display Client Redirection? n
              AUDIX Name:                      Select Last Used Appearance? n
                                                 Coverage After Forwarding? s
                                                   Multimedia Early Answer? n
                                               Direct IP-IP Audio Connections? y
  Emergency Location Ext: 6605          Always Use? n IP Audio Hairpinning? n
     Precedence Call Waiting? y
```

Page 4 of the station form is shown below. Bridged Line Appearances **6601** will be

Step	Description
	used to show the status of Private 2 on the IPC Turrets. Bridged Line Appearance **6602** and **6603** will be used to show the status of IPC QSIG Extension 3102 and 3104 on the IPC Turrets. There must be a call appearance programmed for its own extension, and maximum 8 bridged line appearance could be configured (6605 is included). The **BUTTON ASSIGNMENTS** numerical values, 1 – 8 (1 in the case of 6605) must be matched by the Alliance MX ER line mappings BID number (Contact IPC engineer to verify IPC settings).

```
change station 6605                                         Page   4 of   5
                                     STATION
 SITE DATA
       Room:                                            Headset? n
       Jack:                                            Speaker? n
      Cable:                                           Mounting: d
      Floor:                                        Cord Length: 0
   Building:                                          Set Color:

ABBREVIATED DIALING
     List1:                    List2:                    List3:


BUTTON ASSIGNMENTS
 1: call-appr                          5:
 2: brdg-appr  B:1  E:6601             6:
 3: brdg-appr  B:1  E:6602             7:
 4: brdg-appr  B:1  E:6603             8:
```

Step	Description
2.	Use the **change off-pbx-telephone station-mapping** command to configure the station as an Off-PBX Station (OPS). Enter the following fields and accept the defaults for the other fields: • **Application**: Enter **OPS** for the Outboard Proxy SIP feature. • **Phone Number**: Enter the phone number to map this station to (e.g., **6605**). • **Trunk Selection**: Enter the Trunk Group number (e.g. **1**) of the SIP trunk between Avaya Communication Manager and the SIP Enablement Services. • **Config Set**: Enter the Configuration Set (e.g., **1**) to assign to this OPS station. The defaults were used for Configuration Set is **1**.

```
change off-pbx-telephone station-mapping 6605               Page   1 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number      Trunk       Config
 Extension                  Prefix                       Selection   Set
 6605              OPS           -     6605               1          1
```

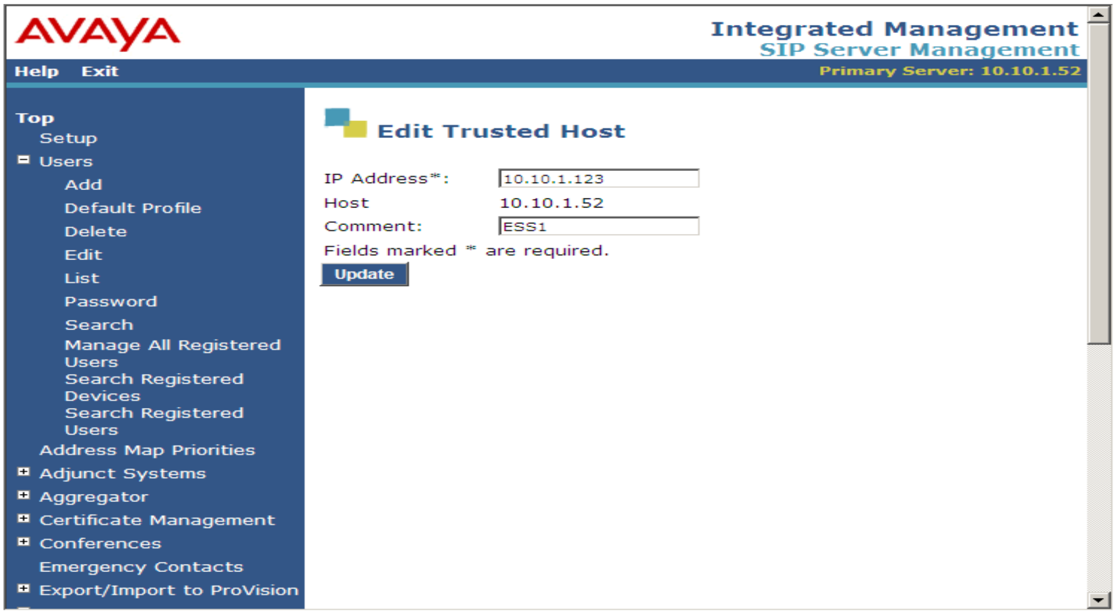
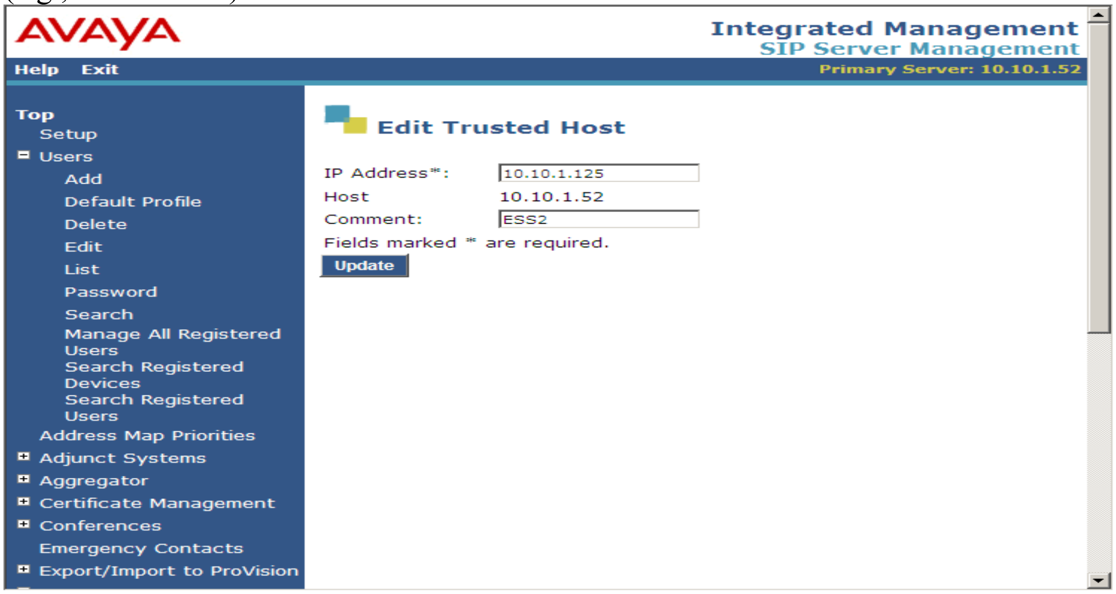
On **Page 2** of the form, set the **Call Limit** field to **10**, **Mapping Mode** field to **both**, **Calls Allowed** field to **all** and **Bridged Calls** to **both**. This will allow the Avaya SIP telephone to conference another party as well as originate and terminate calls.

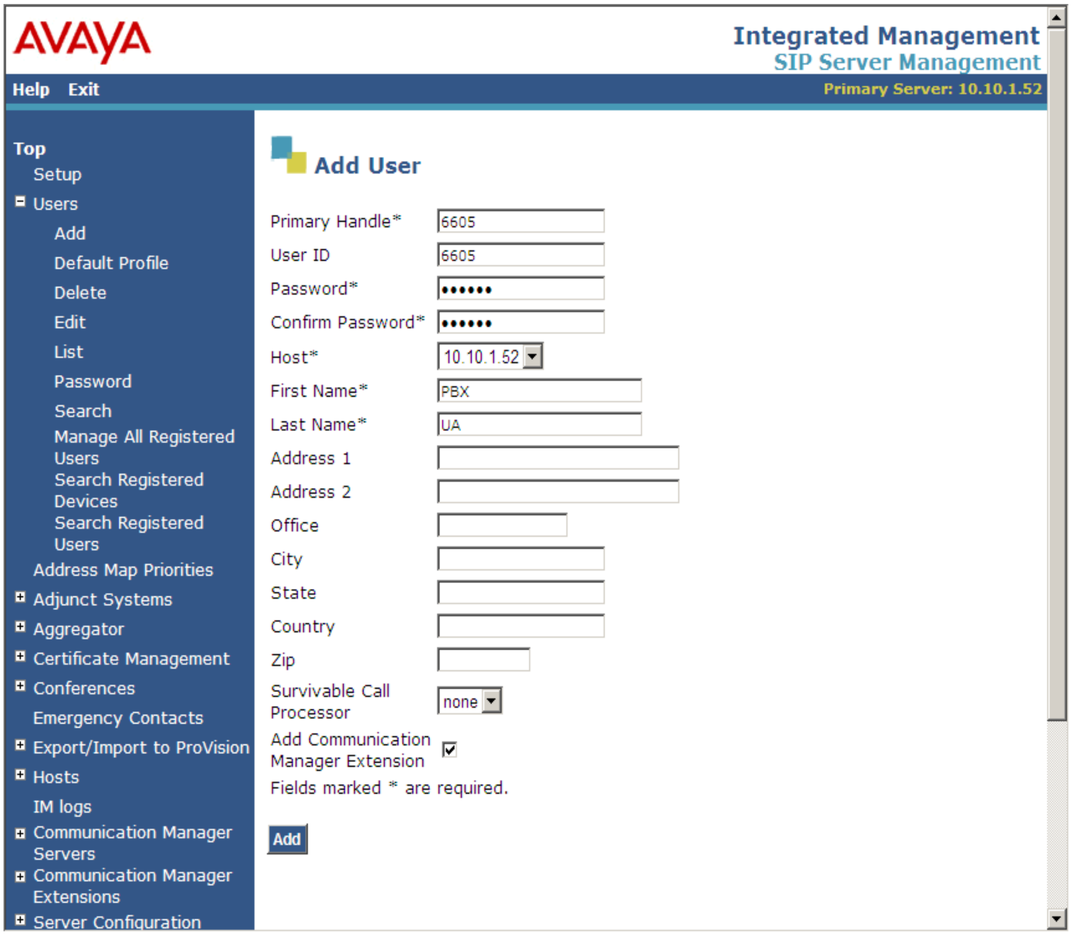
```
change off-pbx-telephone station-mapping 6605               Page   2 of   2
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Call       Mapping      Calls       Bridged        Location
 Extension     Limit      Mode         Allowed     Calls
 6605           10       both           all         both
```

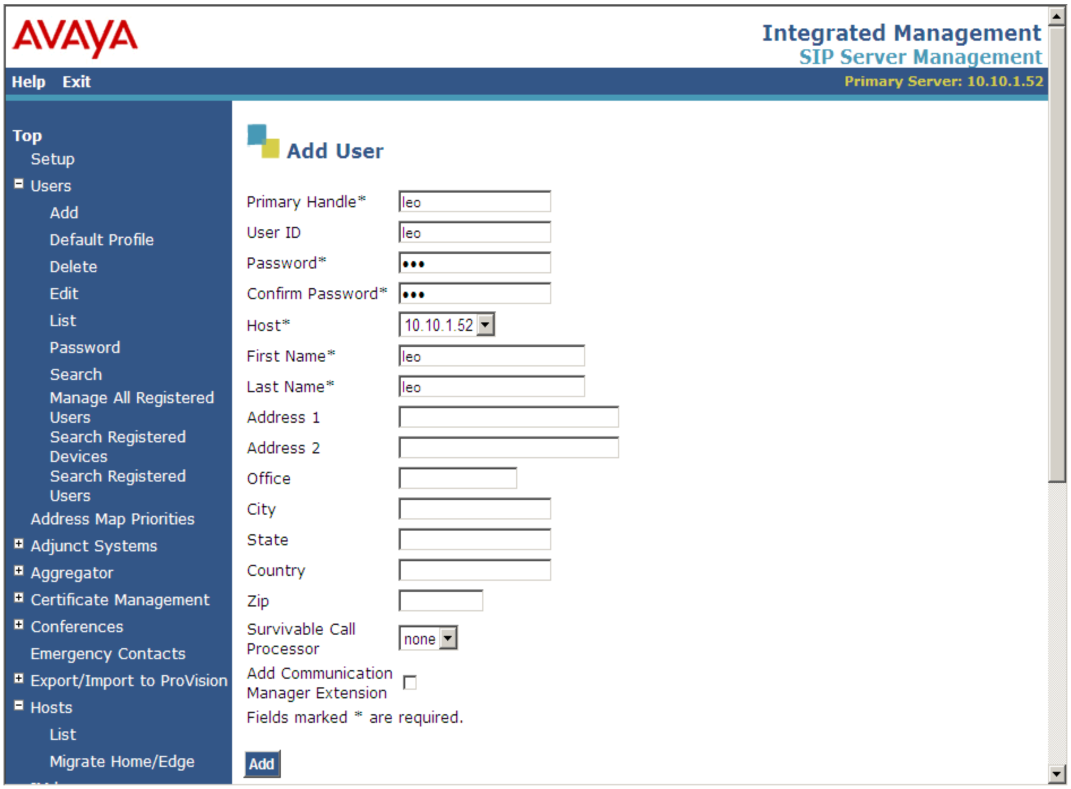
5. Configure Avaya SIP Enablement Services

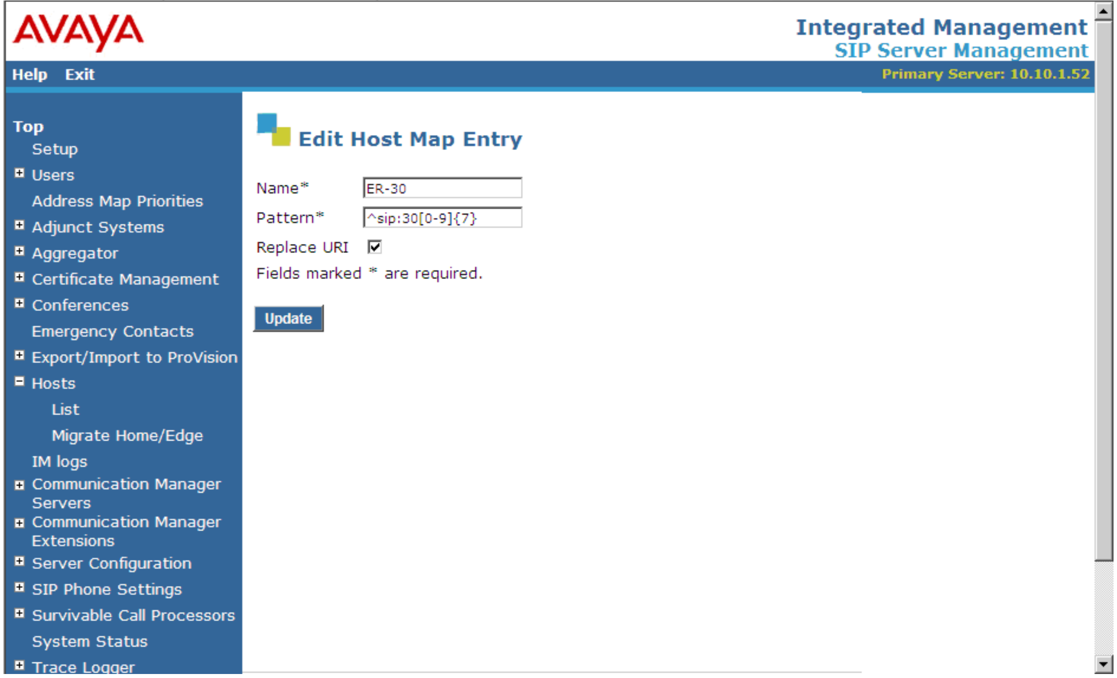
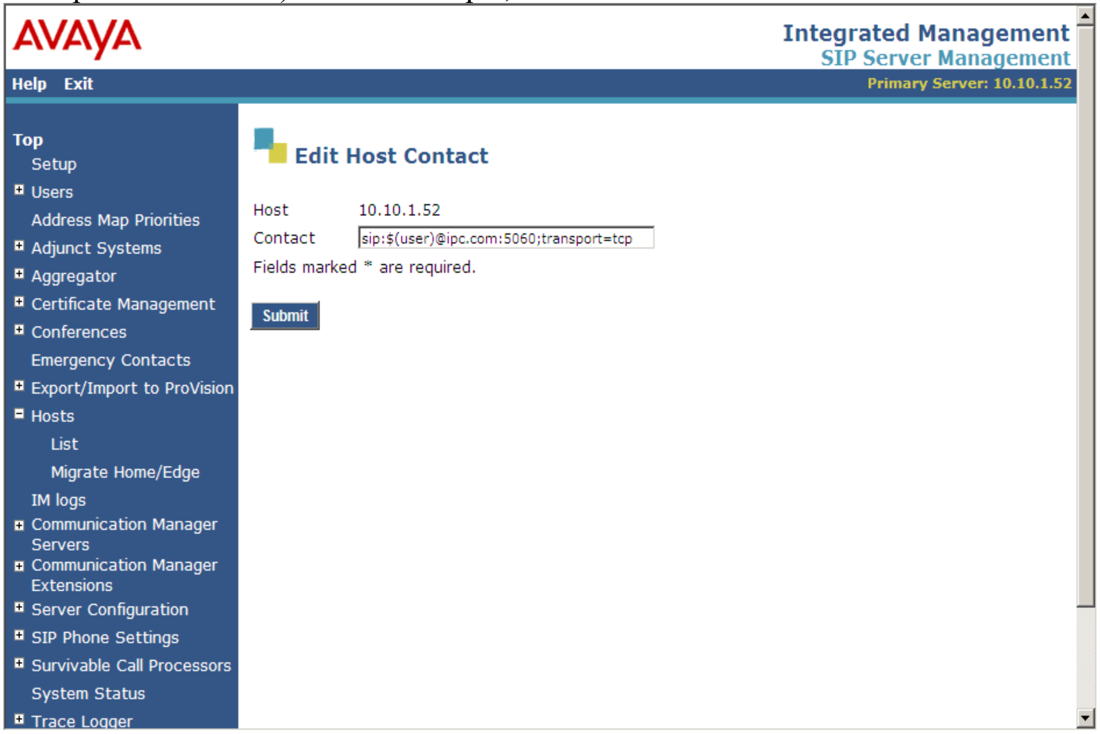
The following section describes how to configure the Avaya SIP Enablement Services to support Enterprise Reach for IPC using PLIC. It is assumed that the standard configuration of the Avaya SIP Enablement Services has already been completed.

Step	Description
1.	Log in as a user with administrator privileges to the Administration Web Interface of the Avaya SIP Enablement Services.
2.	Navigate to **Server Configuration → System Properties**. Ensure that the Domain Name Server (DNS) can resolve the *SIP Domain* (e.g., **du.rnd.avaya.com**) to the IP address of the Avaya SIP Enablement Services so that requests from the IPC Enterprise SIP Server (ESS) can be forwarded correctly. The *SIP Domain* value must match the Authoritative Domain assigned in **Section 4.2 Step 3**. In the sample configuration used in these Application Notes, a DNS service was running.  View System Properties SES Version SES-5.1.2.0-416.4b System Configuration Network Duplex Host Type SES combined home-edge SIP Domain* du.rnd.avaya.com Note that the DNS domain is du.rnd.avaya.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 SIP License Host* 10.10.1.51 **DiffServ/TOS Parameters** Call Control PHB Value* 46 **802.1 Parameters** Priority Value* 6 Management System Access Login Management System Access Password DB Log Level disabled

Step	Description
3.	Navigate to **Trusted Hosts→Add**. Add the IP address (e.g., **10.10.1.123**) of the IPC ESS (IPC Enterprise SIP Server) as a trusted host. The **Edit** option is used to show the entry after the trusted host was added. **Note:** Do not add IPC ESSPBXUA host as a trust host.  Edit Trusted Host IP Address*: 10.10.1.123 Host 10.10.1.52 Comment: ESS1 Fields marked * are required. If there is a redundant IPC ESS (IPC Enterprise SIP Server), added another entry for it (e.g., 10.10.1.125).  Edit Trusted Host IP Address*: 10.10.1.125 Host 10.10.1.52 Comment: ESS2 Fields marked * are required.

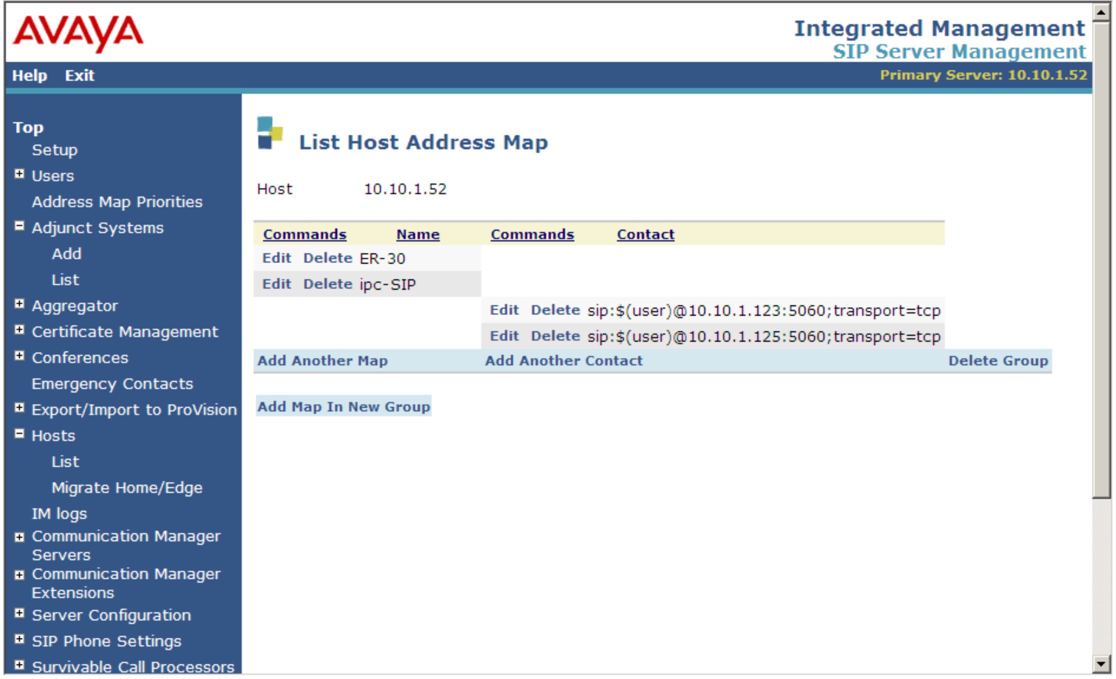
Step	Description
4.	Navigate to **Users→Add.** Add a new user on the Avaya SIP Enablement Services for the PBXUA. Select the **Add Communication Manager Extension** checkbox. Assign the same Communication Manager Extension (e.g., **6605**) for the user just created. This extension and password must match the extension and password that have been defined in the IPC cofiguration→ Edit Enterprise Reach PBX Config (Contact IPC engineer to verify IPC settings).  **Add User** Primary Handle* 6605 User ID 6605 Password* •••••• Confirm Password* •••••• Host* 10.10.1.52 First Name* PBX Last Name* UA Address 1 Address 2 Office City State Country Zip Survivable Call Processor none Add Communication Manager Extension ☑ Fields marked * are required. Add

Step	Description
5.	Navigate to **Users→Add.** Add another user (e.g., ***leo)*** on the Avaya SIP Enablement Services for the PBXUA authentication. ***Note***: there is no need to create an associated CM extension. **User ID** and **Password** must match the user and password that have been defined in the IPC configuration→Edit SIP Config (Contact IPC engineer to verify IPC settings).  Primary Handle* leo User ID leo Password* ••• Confirm Password* ••• Host* 10.10.1.52 First Name* leo Last Name* leo Address 1 Address 2 Office City State Country Zip Survivable Call Processor none Add Communication Manager Extension ☐ Fields marked * are required.

Step	Description
6.	Navigate to **Hosts→List→Map.** Add a Host Address Map as shown below to route calls that begin with a leading **30** to the IPC ESS (IPC Enterprise SIP Server).  Edit Host Map Entry Name* ER-30 Pattern* ^sip:30[0-9]{7} Replace URI ☑ Fields marked * are required.
7.	The contact associated with the Host Address Map is shown below to route the call to the IP address (e.g., **10.10.1.123**) or DNS name (e.g., **ipc.com**) of the IPC ESS (IPC Enterprise SIP Server). In this example, DNS name is used.  Edit Host Contact Host 10.10.1.52 Contact sip:$(user)@ipc.com:5060;transport=tcp Fields marked * are required.

Step | **Description**

8. For redundancy IPC ESS (IPC Enterprise SIP Server), add two **Contact** pointing to different IPC ESS (IPC Enterprise SIP Server). In this example, IP address is used. (i.e., 10.10.1.123 and 10.10.1. 125)



List Host Address Map

Host 10.10.1.52

Commands	Name	Commands	Contact
Edit Delete	ER-30		
Edit Delete	ipc-SIP		
		Edit Delete	sip:$(user)@10.10.1.123:5060;transport=tcp
		Edit Delete	sip:$(user)@10.10.1.125:5060;transport=tcp

9. To support redundancy IPC ESS (IPC Enterprise SIP Server), the following configuration is needed:

Change SIP parameters as shown below in the file /usr/impress/sip-server/etc/ccs.conf on the Avaya SIP Enablement Services server. Restart the server after the changes. Default settings are :

```
PerContactWaitTime=30
MM_PerContactWaitTime=2
TimerB=32000
```

New settings are :

```
PerContactWaitTime=180
MM_PerContactWaitTime=0
TimerB=2000
```

PerContactWaitTime field is determined by customers' needs, but it is recommended setting to at least 180 seconds.
MM_PerContactWaitTime is a fast sequential forking timer used for the SIP Enablement Services Adjunct Interface (e.g., Modular Messaging or Voice Portal).
TimerB is the INVITE transaction timeout timer as defined in RFC3261. If there is no response for the INVITE, Avaya SIP Enablement Services will try the next Contact or Adjunct Server after **TimerB**. In this example, **TimerB** is set to 2 seconds.

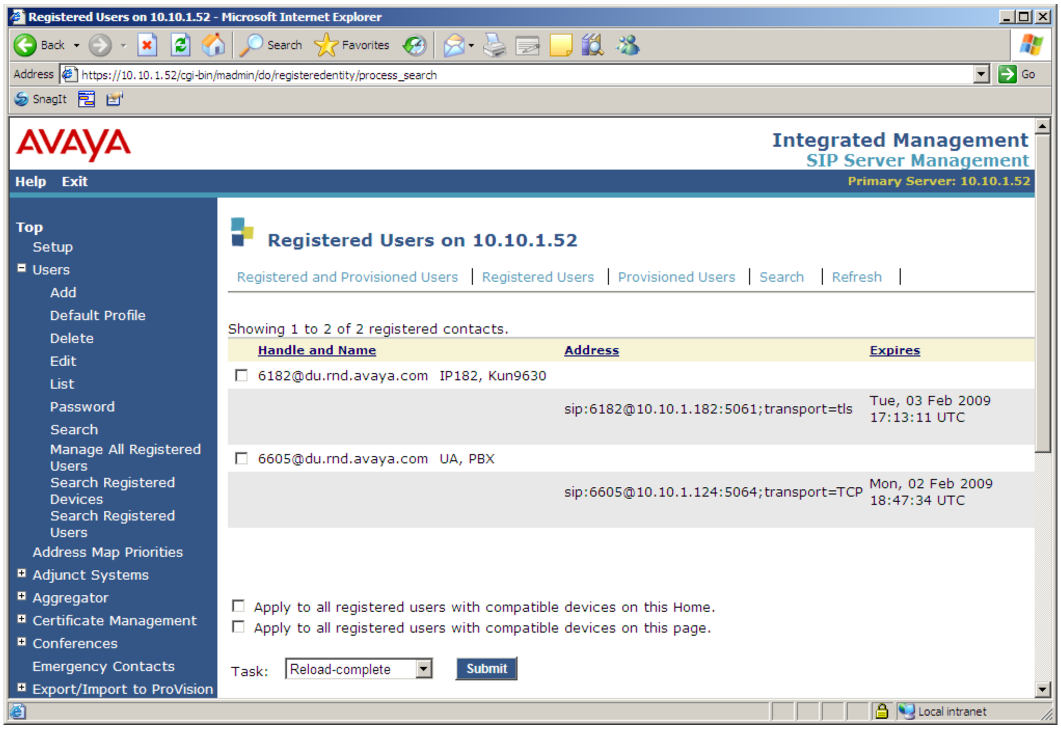
6. General Test Approach and Test Results

All feature and serviceability test cases were performed manually. The verification included function testing of Enterprise Reach solution, redundancy/failover testing on Avaya SIP Enablement Services and IPC ESS (IPC Enterprise SIP Server). The interoperability testing cases and results can be viewed in the IPC test schedule document in reference [2].

7. Verification Steps

The following steps can be used to verify that Avaya Communication Manager and SIP Enablement Services are configured correctly to support Enterprise Reach for IPC. In the following steps, specific DDI/phone numbers used are only for this example.

1. Verify that the PBXUA (i.e., user **6605)** is registered using the Administration Web Interface of the Avaya SIP Enablement Services.



Registered Users on 10.10.1.52

Showing 1 to 2 of 2 registered contacts.

Handle and Name	Address	Expires
☐ 6182@du.rnd.avaya.com IP182, Kun9630		
	sip:6182@10.10.1.182:5061;transport=tls	Tue, 03 Feb 2009 17:13:11 UTC
☐ 6605@du.rnd.avaya.com UA, PBX		
	sip:6605@10.10.1.124:5064;transport=TCP	Mon, 02 Feb 2009 18:47:34 UTC

☐ Apply to all registered users with compatible devices on this Home.
☐ Apply to all registered users with compatible devices on this page.

Task: Reload-complete

2. Place an outbound call at Private 1 from the IPC Turrets by selecting Private 1 on an IPC Turret. Verify that Private 1 on the other IPC Turrets and Avaya stations indicate busy on the corresponding BLA. Verify the Avaya phone can barge in with two-way audio by selecting the busy BLA.

3. Place a call over Private 2 from the Avaya telephones by selecting BLA 6601 on station 6620 and pressing the speed dial button for Private 2. Verify that Private 2 on the IPC Turrets and Avaya stations indicate busy. Verify Avaya can put the line on hold and resume the call. Verify that an IPC Turret can barge in with two-way audio.

4. Place an inbound call to QSIG DDI from the IPC Turrets by selecting QSIG extension 3101 and dialing 3102 on an IPC Turret. Verify that QSIG extension 3102 on the other IPC Turrets indicate an incoming call and BLA 6602 on Avaya stations 6620 ring and can be answered with two-way audio.
5. Place a call over QSIG from the Avaya telephones by selecting BLA 6602 on station 6620 and pressing the speed dial button for QSIG. After hearing dial tone, call another QSIG extension (e.g., 3104). Verify that QSIG extension 3104 on the IPC Turrets indicates an incoming call and that the call can be answered with two-way audio.

8. Conclusion

These Application Notes described how to configure Avaya Communication Manager and SIP Enablement Services to support Enterprise Reach for IPC.

9. Additional References

[1] *SIP Support in Avaya Communication Manager Running on Avaya S8xxx Servers, 555-245-206, Issue 8, January 2008* can be found at http://support.avaya.com.

[2] Administrator Guide for Avaya Communication, Issue 4, January 2008, Document Number 03-300509 can be found at http://support.avaya.com.

[3] *Enterprise Reach interworking & verification test schedule between IPC MX dealerboard,Rel 15.02 and AVAYA Communication Manager, Ver 5.1 SP3, with Avaya SIP Enablement Services, Ver 5.1.2 SP0*, available on request from IPC mark.rideout@ipc.com or Avaya devconnect@avaya.com.

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 DevConnect Program at devconnect@avaya.com.