Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services to Support IPC Enterprise Reach - Issue 1.0

Abstract

These application notes describe the procedures to configure Avaya Aura ™ Communication Manager and Avaya Aura ™ SIP Enablement Services to support IPC Alliance MX and IPC ESS (Enterprise SIP Servers) using an Enterprise Reach configuration

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 objective of this compliance test is to verify the Enterprise Reach (ER) solution provided by IPC can interoperate with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services (SES). Enterprise Reach is a solution that consists of the following IPC components:

- IPC Alliance MX
- IPC ESS (Enterprise SIP Server)
- IPC System Center
- IPC IQ/MAX turrets

The Enterprise Reach solution allows IPC Alliance MX to share lines with Avaya Aura™ Communication Manager. IPC use software components within their architecture to enforce lines states in Avaya Aura™ Communication Manager such that they mirror the line states in the IPC Alliance MX, the software components used are known as PBXUA (Private Branch Exchange User Agent) and B2BUA (Back to Back User Agent). The PBXUA and B2BUA run on the IPC ESS (Enterprise SIP Server) each PBXUA can share multiple lines with Communication Manager; this is achieved by adding BLAs (bridged line appearances) for the shared Communication Manager extensions to the Communication Manager User extension that will serve as the PBXUA. The PBXUA registers with Avaya Aura™ SIP Enablement Services and subscribes for BLA event notifications.

The IPC PBXUA uses one SIP UA (User Agent) to communicate with the IPC Alliance/ESS and another SIP UA to communicate with Avaya Aura™ Communication Manager; these SIP UAs comprise the B2BUA. As a part of enforcing line states, the IPC PBXUA places outbound calls and answers inbound calls.

The IPC B2BUA controls the creating of call legs to the Alliance line and enforcing line states as well as being responsible for placing a call into Avaya Aura ™ Communication Manager to show incoming ring on an Alliance MX line, working in conjunction with the PBXUA to show busy, hold and idle states.

These Application Notes describe the required configuration steps for the Avaya solution components. In accordance with the IPC support policy, IPC configuration procedures are not included in these Application Notes. IPC engineers will be responsible for the installation and maintenance of IPC products.

1.1. Interoperability Compliance Testing

As IPC Enterprise Reach allows the sharing of its telephony lines with Communication Manager; lines that are enabled at the Alliance for Enterprise Reach should be presented to user extensions of the associated Communication Manager extension. IPC support line sharing for various line types such as Private Wire Manual Ring Down (PW MRD), Private wire Circuits, Analog PSTN dial tone, Channel Associated Signaling System, T1 line side, ISDN and Q point Signaling System (QSIG). The interoperability compliance testing focuses on the Private Wire Circuits and QSIG line types.

A simulated enterprise site using an Avaya IP enabled telephony solution was connected to the IPC solution via SIP. The SIP connection is provisioned between the SES and IPCs ESSs. The compliance test included the following:

- Inbound calls to IPC line confirming line status is reflected at the appropriate Communication Manager extensions
- Outbound calls from IPC turrets using ER lines, confirming line status is reflected at the appropriate Communication Manager extensions
- Outbound calls from Communication manager extensions using ER lines, confirming line status is reflected at the appropriate IPC Turrets.
- Call hold and retrieval using IPC turrets and Communication manager extensions
- Call transfer from and too various endpoints on both Avaya and IPC environments.
- Conference and exclusion using various endpoints on both Avaya and IPC environments

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

Figure 1 illustrates the network topology of the lab environment used for compliance testing. **Note:** The E1-QSIG link and PSTN links between Communication Manager and IPC Alliance MX are used for test purpose only. The configurations for these links are not included in these Application Notes

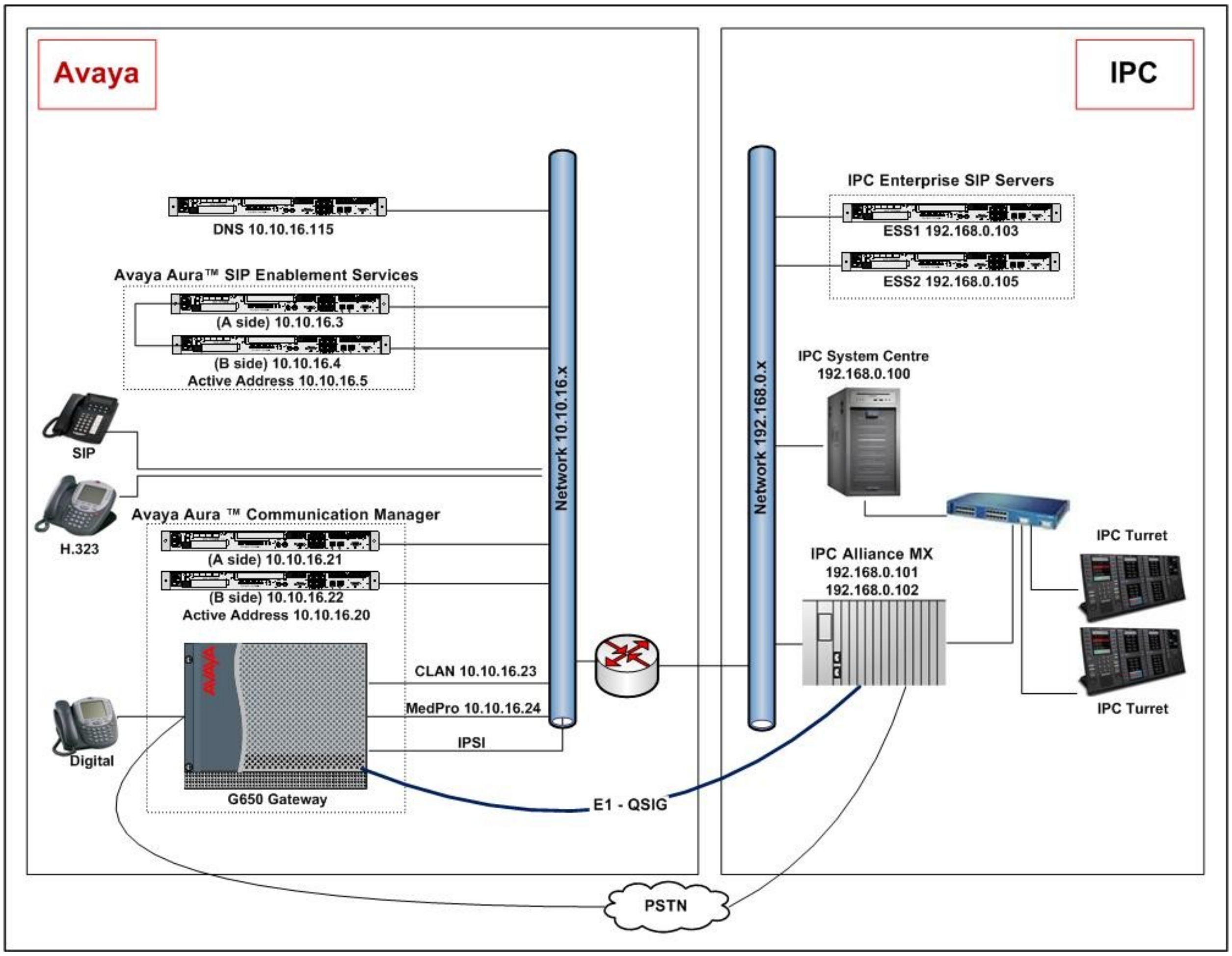


Figure 1: Test Environment Lab Topology

In these Application Notes four IPC lines will be shared with four Communication Manager User extensions via ER, each Communication Manager extension has an associated LAC (Logical Address Code) on IPC Alliance MX. Each LAC is assigned and associated with keys of the IPC Turrets, the association between the IPC LAC and Communication Manager Extension used in this configuration are shown in **Figure 2** below. None of the Communication Manager extensions in **Figure 2** are assigned to a physical endpoint; instead BLAs are used to access the Communication Manager extensions and the associated ER line. To seize the associated ER Line an Avaya user must dial an access code; these are indicated in the Access Code column of **Figure 2**. For an Avaya user to make a call on an ER line, the user has to undertake a number of button presses to achieve connection with the distant party. To help simplify this process the access codes are set up as speed dials where ever a BLA for the associated Communication Manager Extension is used. To better understand how an outbound call is achieved from a Communication Manager Extension a step by step description of a typical outbound call is given:

- Lift phone handset and select the desired BLA line key corresponding to one of the Communication manager extension in **Figure 2** (e.g. 6625).
- Dial the associated access code (e.g. 301501140) by pressing the key pre configured as a speed dial button
- After dial tone is heard the ER line (e.g. PLIC 1140) has been seized
- An outbound call can now be made by dialing the number of the distant party.

In this example configuration, Communication Manager Extension **6632** (See **Figure 2)** is used as the IPC PBXUA. **6632** is configured as an OPTIM station on Communication Manager and as a user with Communication Manager Server extension on SES.

IPC Line Type	IPC LAC	Communication Manager Extension	Access Code
QSIG	3102	6632	301563574
QSIG	3104	6623	301563576
PLIC	1138	6624	301501138
PLIC	1140	6625	301501140

Figure 2: Shared Line Appearances

3. Equipment and Software Validated

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

Equipment	Software
Avaya™ S8510 Servers	Avaya Aura™ SIP Enablement Services 5.2.1 Service Pack 1
Avaya™ S8730 Servers	Avaya Aura™ Communication Manager 5.2.1 – S8730-15-02.1.016.4. Service Pack 0
Avaya™ G650 Media Gateway - CLAN - TN799DP - MedPro - TN 2602AP	 HW16 FW032 HW08 FW048
Avaya 9630 IP Telephones	SIP: 2.5.0.0 H.323: R3.0
Avaya 2420 Digital Telephones	---
IPC Information Systems Alliance MX IPC System Center (Dell R710) IPC IQ/MAX Turrets	Alliance Release 16.00.00.Patch 2
IPC ESS (SIP Proxy Server)	2.00.01-11

4. Configure Avaya Aura™ Communication Manager

The steps in this section describe the configuration for Communication Manager to support IPC ER solution. The commands listed in the following sections were issued at the Avaya System Access Terminal (SAT). The procedures covered in this section include:

- Verify Avaya Aura™ Communication Manager System Features
- Administer Dialplan Parameters
- Administer Feature Access Codes
- Configure SIP Trunk To SES
- Administer Dialplan Analysis
- Administer AAR
- Administer Public Numbering
- Administer Communication Manager User Extensions

4.1. Verify Avaya Aura™ Communication Manager System Features

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 281
                                      Maximum Stations: 36000 47
                              Maximum XMOBILE Stations: 0     0
                      Maximum Off-PBX Telephones - EC500: 200   0
                      Maximum Off-PBX Telephones -   OPS: 200   17
                      Maximum Off-PBX Telephones - PBFMC: 0     0
                      Maximum Off-PBX Telephones - PVFMC: 0     0
                      Maximum Off-PBX Telephones - SCCAN: 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 entities. 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: 200     0
           Maximum Concurrently Registered IP Stations: 18000 1
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                        Maximum Video Capable Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 300   138
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 100   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   1
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

On **Page 3,** verify the fields **ARS**, **ARS/AAR Partitioning** are set to **y**.

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

    Abbreviated Dialing Enhanced List? y         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? n
```

On **Page 4,** verify the fields **ISDN-PRI** and **IP Trunks** 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
```

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? y
 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? n       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
```

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
```

Use the **change system-parameters features** command and navigate to **Page 18**, confirm that **Direct IP-IP Audio Connections** is set to **y** to allow shuffling**.**

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

IP PARAMETERS

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

              SDP Capability Negotiation for SRTP? n
CALL PICKUP
  Maximum Number of Digits for Directed Group Call Pickup: 4
                   Call Pickup on Intercom Calls? y     Call Pickup Alerting? n
     Temporary Bridged Appearance on Call Pickup? y     Directed Call Pickup? y
                      Extended Group Call Pickup: none
                   Enhanced Call Pickup Alerting? n
```

4.2. Administer Dialplan Parameters

Use the **change dialplan parameters** command to assign **Local Node Number**. If there is no assigned number, enter **1.**

```
change dialplan parameters                                       Page   1 of   1
                             DIAL PLAN PARAMETERS

                 Local Node Number: 1                   ETA Node Number:
   UDP-ARS Calls Considered Offnet? n               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? y

 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
```

4.3. Administer Feature Access Codes

Use the **display feature-access-codes** command to verify the following. On **Page 1** confirm that **Auto Alternate Routing (AAR) Access Code** is set to a valid feature access code according to the dial plan.

```
display 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
```

4.4. Configure SIP Trunk to Avaya Aura™ SIP Enablement Services

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

4.4.1. Administer IP Node Names

Use the **change node-names ip** command to add the active IP address for the SES, also make note of the CLAN name as this will be used to configure the SIP signaling group.

```
display node-names ip
                                  IP NODE NAMES
    Name              IP Address
CLAN1               10.10.16.23
Gateway             10.10.16.1
MedPro1             10.10.16.24
SM100               10.10.16.11
default             0.0.0.0
procr               10.10.16.20
sesactive           10.10.16.5
```

4.4.2. Administer IP Network Region

Enter the **change ip-network-region n** command. For the **Authoritative Domain** field, enter the SIP domain name configured for this enterprise, a descriptive **Name** for this ip-network-region and set the **Codec Set** to the number of the codec set that will be used. **Intra-region IP-IP Direct Audio** and **Intra-region IP-IP Direct Audio** should be set to **yes** to enable IP Media shuffling. Although not highlighted, note also that the **IP Network Region** form is used to set the QoS packet parameters that provide priority treatment for signaling and audio packets over other data traffic. These parameters may need to be aligned with the specific values expected by the IP network.

```
change ip-network-region 1                                        Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: sip.avaya.com
    Name: Default Region
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? n
   UDP Port Max: 3329
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
```

4.4.3. Administer IP Codec Sets

Use the **change ip-codec-set n** command, where **n** is the codec set specified in the **IP Network Region** form. Enter the codecs eligible to be used; at least one of the codecs defined here must be supported by the far end device.

```
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.711MU           n         2        20
 2: G.711A            n         2        20
 3: G.729             n         2        20
 4:
 5:
```

4.4.4. Administer SIP Signaling Group

This signaling group will be used with the trunk group providing connection between Communication Manager and SES. Use the **add signaling-group *n*** command, where **n** is the signaling-group number to add. The **Near-end Node Name** is set to the name of the CLAN that will be used to process the signaling. The **CLAN1** name is assigned in the IP Node-names form. The **Far-end Node Name** is set to the name of the SES assigned in the IP Node-names form. Set the **Far-end Network Region** to the IP Network Region defined in **Section 4.4.2**. **Far-end Domain** is set to the name of the domain name that is used by SES.

Note: That if Communication Manager will receive contact from any SIP entity connected to the SIP Enablement Services not in the **sip.avaya.com** domain, another signaling group should be setup where the **Far-end Domain** is left blank.

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

 Group Number: 6             Group Type: sip
                        Transport Method: tcp
  IMS Enabled? n
     IP Video? n

   Near-end Node Name: CLAN1                Far-end Node Name: sesactive
 Near-end Listen Port: 5060               Far-end Listen Port: 5060
                                       Far-end Network Region: 1
Far-end Domain: sip.avaya.com

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

4.4.5. Administer SIP Trunk Group

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

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

Group Number: 6                    Group Type: sip           CDR Reports: y
  Group Name: SES                         COR: 1        TN: 1        TAC: 506
   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: 30
```

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 6                                                   Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                        Preferred Minimum Session Refresh Interval(sec): 300
```

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

```
Add    trunk-group 6                                                   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? y



 Show ANSWERED BY on Display? y
```

4.5. Administer Dialplan

Use the **change dialplan analysis** command to administer the dialplan. In this example 3015xxxxx are used as ER access codes (see **Figure 2**). Therefore, an entry is added for 9-digit numbers beginning with **3015** to use **udp**.

```
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
     0            1    ext    663         4    udp
     1            1    fac    7           4    ext
     2            4    udp    88          4    ext
     3015         9    udp    89          4    ext
     3005         8    udp    972         5    udp
     31           4    udp    99          4    ext
     33           4    udp    *           2    fac
     37           4    udp    #           2    fac
     38           5    aar
```

Use the **change uniform-dialplan n** command to add an entry to route 9-digit numbers beginning with 3015 to the AAR (Alternate Automatic Routing).

```
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
 3015            9   0                  aar  n
 31              4   0                  aar  n
 33              4   0                  aar  n
 37              4   0                  aar  n
 662             4   0                  ext  n
 663             4   0                  aar  n
 8889            4   0                  aar  n
 972             5   0                  aar  n
                                             n
```

4.6. Administer Route Pattern

Use the **change route-pattern n** command to administer a route pattern that will direct calls to trunk group **6** which is the SIP trunk between Communication Manager and SES.

```
change route-pattern 6                                             Page   1 of   3
                    Pattern Number: 6   Pattern Name: to ses
                             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 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
```

4.7. Administer AAR

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 **3015** using **Route Pattern 6** via the SIP trunk between Communication Manager and SES.

```
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
    3015                  9    9      6        aar           n
    31                    4    4      3        aar           n
    33                    4    4      6        aar           n
    37                    4    4      5        aar           n
    663                   4    4      6        aar           n
    8889                  4    4      89       aar           n
    972                   5    5      4        aar           n
                                                             n
```

4.8. Administer Public Numbering

The SIP trunk group references the public numbering table, use the command **change public-unknown-numbering n**. Add an entry so that calls placed from stations with a 4-digit extension beginning with **66** and routed over a trunk group 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: 1
 4  66                                      4         Maximum Entries: 9999
```

4.9. Add Avaya Aura™ Communication Manager User Extensions

This section shows the configuration needed for the Communication Manager extensions that share lines with Alliance MX as well as the Communication Manager extensions that will use a BLA for the line sharing stations. The following procedures are covered in this section:

- Configure Line Sharing Station
- Configure PBXUA Station
- Configure BLA user Station

4.9.1. Add Line Sharing Station

To configure a station that will share an IPC line use the command **add station n** where **n** is the extension number of the station to be added. On **Page 1** of the station form use a 46xx or a 96xx station **Type** and enter a descriptive **Name**

```
add station 6623                                                Page   1 of   5
                                     STATION

Extension: 6623                          Lock Messages? n               BCC: 0
     Type: 4620                          Security Code: 6623             TN: 1
     Port: S00004                      Coverage Path 1:                 COR: 1
     Name: QSIG ER                     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: 6623
            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
```

On **Page 2** of the station form set **Restrict last Appearance** to **n.**

```
add station 6623                                                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? y
       Service Link Mode: as-needed                      EC500 State: enabled
         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
```

On **Page 4** of the station form add three call line appearances as highlighted in bold.

```
add station 6623                                                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:                                    8:
```

Repeat the above steps for all Communication Manager Extensions that share an ER line, excluding the PBXUA which is covered in the following section.

4.9.2. Add PBXUA Station

To configure a PBXUA station, enter the command **add station n** where **n**. On **Page 1** of the station form use a 46xx or 96xx station **Type** and enter a descriptive **Name**. For **Security Code** enter a 6-digit number.

```
add station 6632                                              Page   1 of   5
                                     STATION

Extension: 6632                         Lock Messages? n               BCC: 0
     Type: 9630                         Security Code: 123456           TN: 1
     Port: S00009                     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: 6632
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english                 Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal               Media Complex Ext:
   Survivable Trunk Dest? y                           IP SoftPhone? n
```

On **Page 2** of the station form set **Restrict last Appearance** to **n.**

```
add station 6632                                              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? y
       Service Link Mode: as-needed                      EC500 State: enabled
         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
```

On **Page 4** of the station form assign one call appearance to be used for its own extension and BLAs for the other Communication Managers extension sharing ER lines. Up to 8 BLAs can be configured for each PBXUA. In this example configuration the following B**UTTON ASSIGNMENTS** are used:

- Button assignment 1 is the primary call appearance for 6632 which will show the status of QSIG line 3102
- Button assignment 2 is used to bridge the line appearance for 6623 which will show the status of QSIG line 3104
- Button assignment 3 is used to bridge the line appearance for 6624 which will show the status of PLIC line 1138
- Button assignment 4 is used to bridge the line appearance for 6625 which will show the status of PLIC line 1140

```
add station 6632                                                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: personal 1         List2:                   List3:




BUTTON ASSIGNMENTS
 1: call-appr                          5:
 2: brdg-appr  B:1  E:6623             6:
 3: brdg-appr  B:1  E:6624             7:
 4: brdg-appr  B:1  E:6625             8:

    voice-mail Number:
```

4.9.3. Administer Off-PBX Station Mapping

Use the **change off-pbx-telephone station-mapping n** command where n is the number of the PBXUA extension to configure as an Off-PBX Station (OPS). Enter the following fields and accept the defaults for the other fields. For **Application** enter **OPS,** for the **Phone Number** enter the PBXUA extension number, for **Trunk Selection** enter the Trunk Group number (e.g. **6**) of the SIP trunk between Communication Manager and the SES. For **Config Set** enter the Configuration Set to assign to this OPS station. The default Configuration Set is **1**.

```
change off-pbx-telephone station-mapping 6632                   Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial    CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                      Selection   Set     Mode
 6632             OPS          -       6632             6          1
                               -
```

On **Page 2** of the form, set the **Call Limit** field to **10**, this is the maximum call limit allowed. Set the **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 6632                   Page   2 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Appl    Call        Mapping    Calls       Bridged       Location
 Extension     Name    Limit       Mode       Allowed     Calls
 6632          OPS      10       both         all         both
```

4.9.4. Add Station to use Bridged Line Appearances

Other stations on Communication Manager can use a BLA for each Communication Manager extension that shares a line to make calls via IPCs ER lines and receive status updates. The example below shows a digital station configured with a BLA for each Communication Manager extension that shares an IPC line. Enter the command **add station n.** On **Page 1** of the station form enter the appropriate digital station **Type** and **Port**. Enter a descriptive **Name** for the station.

```
add station 6610                                                Page   1 of   5
                                     STATION

Extension: 6610                          Lock Messages? n              BCC: 0
     Type: 2420                          Security Code:                 TN: 1
     Port: 01A1306                     Coverage Path 1:                COR: 1
     Name: Digi Station                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: 6610
            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
```

On **Page 2** of the station form set **Bridged Call Alerting** to **y**

```
add station 6610                                                Page   2 of   5
                                     STATION
FEATURE OPTIONS
           LWC Reception: audix           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? y
       Service Link Mode: as-needed                        EC500 State: enabled
         Multimedia Mode: basic
    MWI Served User Type:                      Display Client Redirection? n
              AUDIX Name: intuity             Select Last Used Appearance? n
                                                Coverage After Forwarding? s
                                                  Multimedia Early Answer? n
                                             Direct IP-IP Audio Connections? y
  Emergency Location Ext: 6610                         IP Audio Hairpinning? n
     Precedence Call Waiting? y
```

On **Page 4** of the station, configure the BLAs for the Communication Manager extensions that share ER lines. Recall from **Section 2** that multiple button presses are required to access an IPC Line configured for ER, after a BLA is selected the access code relating to that BLA must be dialed by selecting the relevant speed dial button (refer to **Figure 2**) the speed dial buttons are configured as **autodial.**

```
add    station 6610                                             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:6624
 2: call-appr                            6: brdg-appr  B:1  E:6625
 3: brdg-appr  B:1  E:6632               7: autodial   Number: 301563574
 4: brdg-appr  B:1  E:6623               8: autodial   Number: 301563576

    voice-mail Number:
```

Continue on **Page 5** if more button assignments are required as was the case in this example. An **exclusion** button is required to invoke privacy during an ER call

```
add station 6610                                              Page   5 of   5
                                     STATION

FEATURE BUTTON ASSIGNMENTS

 9: autodial   Number: 301501138
10: autodial   Number: 301501140
11: auto-cback
12: send-calls Ext:
13: cpn-blk
14: cfwd-bsyda Ext:
15: call-fwd   Ext:
16: exclusion
17:
18:
19:
20:
21:
22:
23:
24:
```

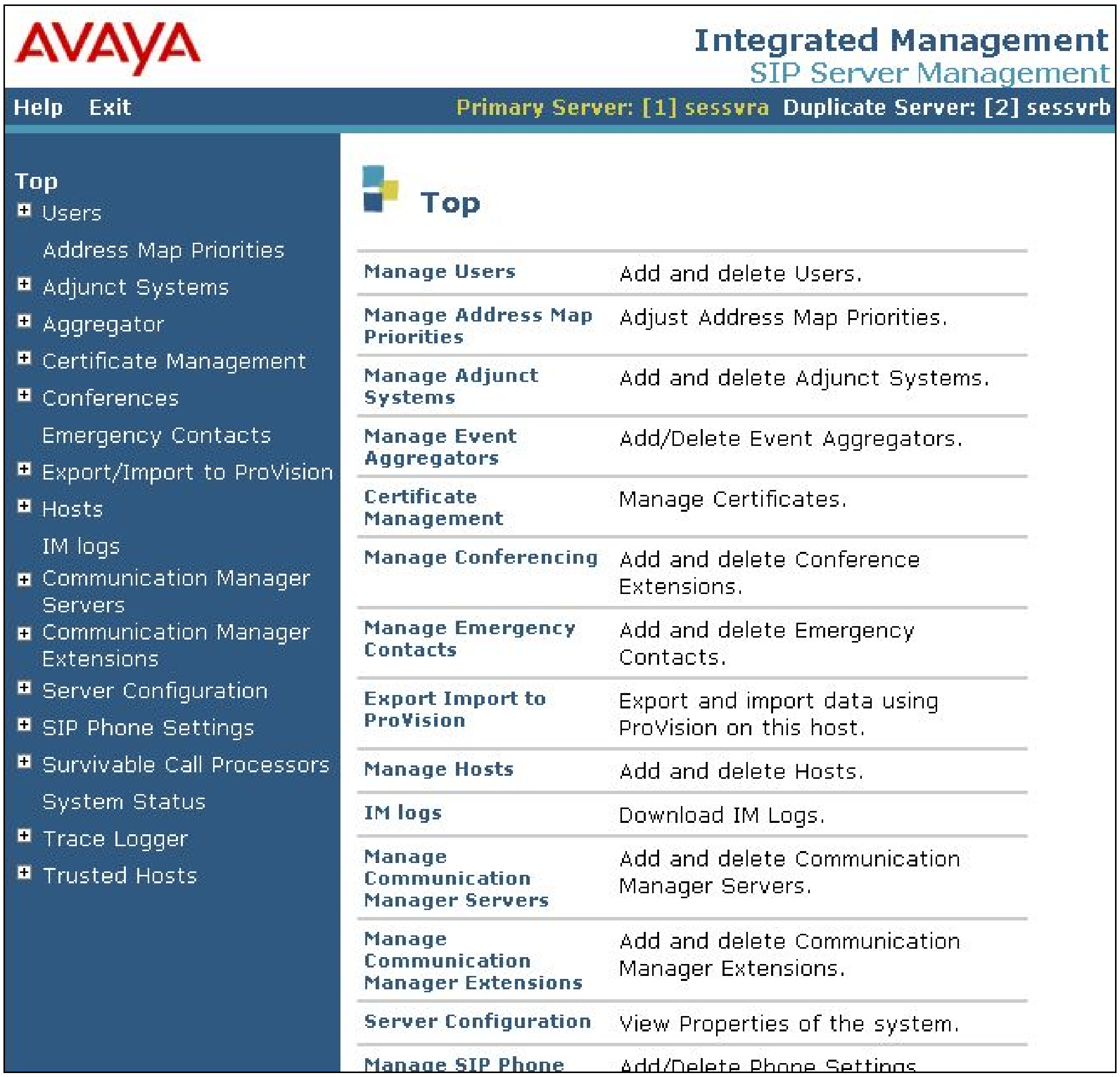
5. Configure Avaya Aura™ SIP Enablement Services

This section covers the administration of SES to support ER. SES is configured via an Internet browser using the Administration web interface. It is assumed that SES software and the license file have already been installed. The procedures covered in this section include:

- Logging onto Avaya Aura™ SIP Enablement Services
- Verifying System Properties
- Administer Avaya Aura™ SIP Enablement Services Host properties
- Add Avaya Aura™ Communication Manager Server
- Administer Trusted Hosts
- Administer Address Maps to IPC
- Administer Avaya Aura™ SIP Enablement Services PBXUA users

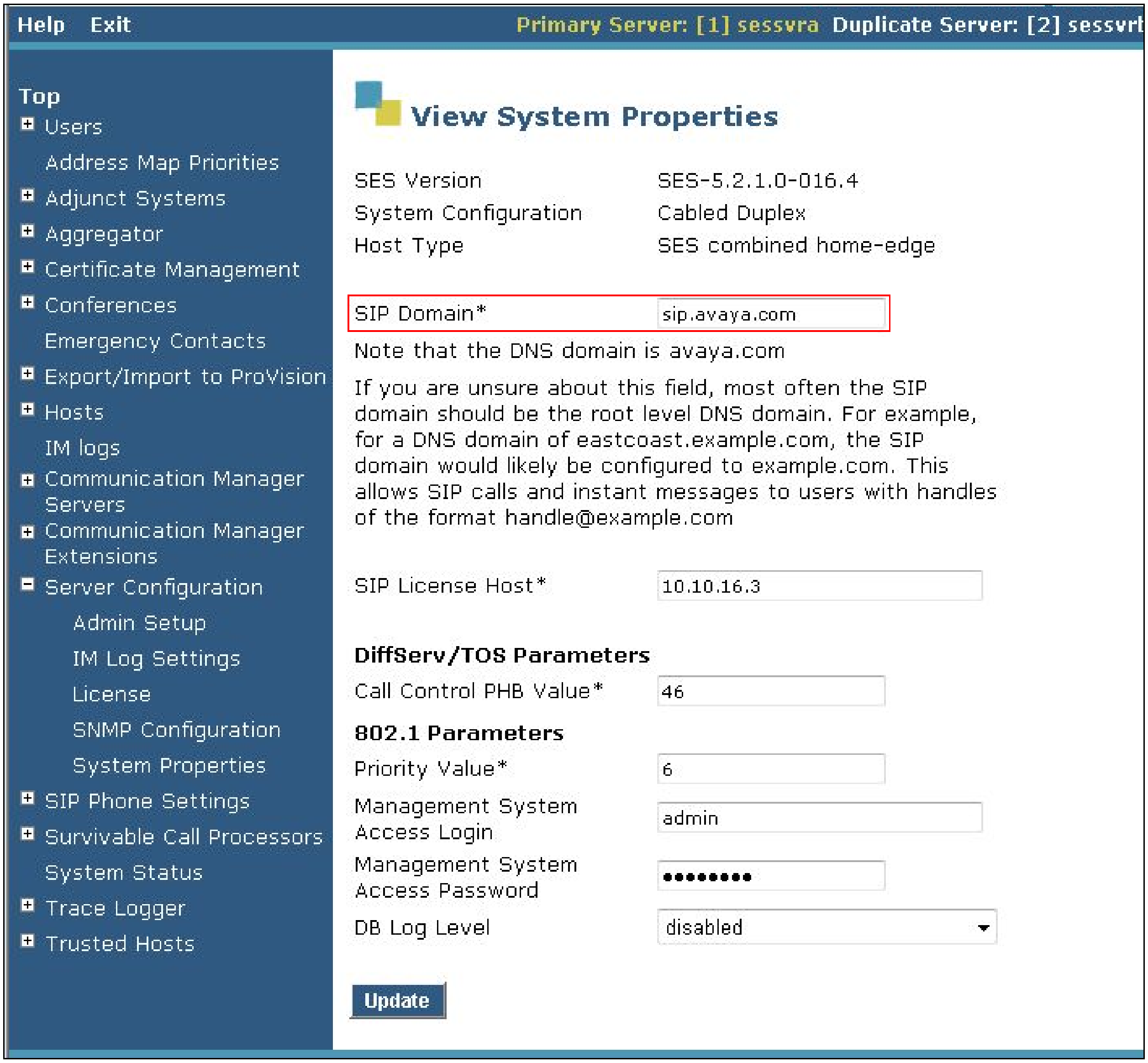
5.1. Logging onto Avaya Aura™ SIP Enablement Services

Access the SES Administration web interface, by entering **http://<ip-addr>/admin** as the URL in an internet browser, where **<ip-addr>** is the active IP address of the SES server. Log in with the appropriate credentials and select the **Administration** link and then **SIP Enablement Services** from the main screen (not shown). The SES administration home screen will be displayed.



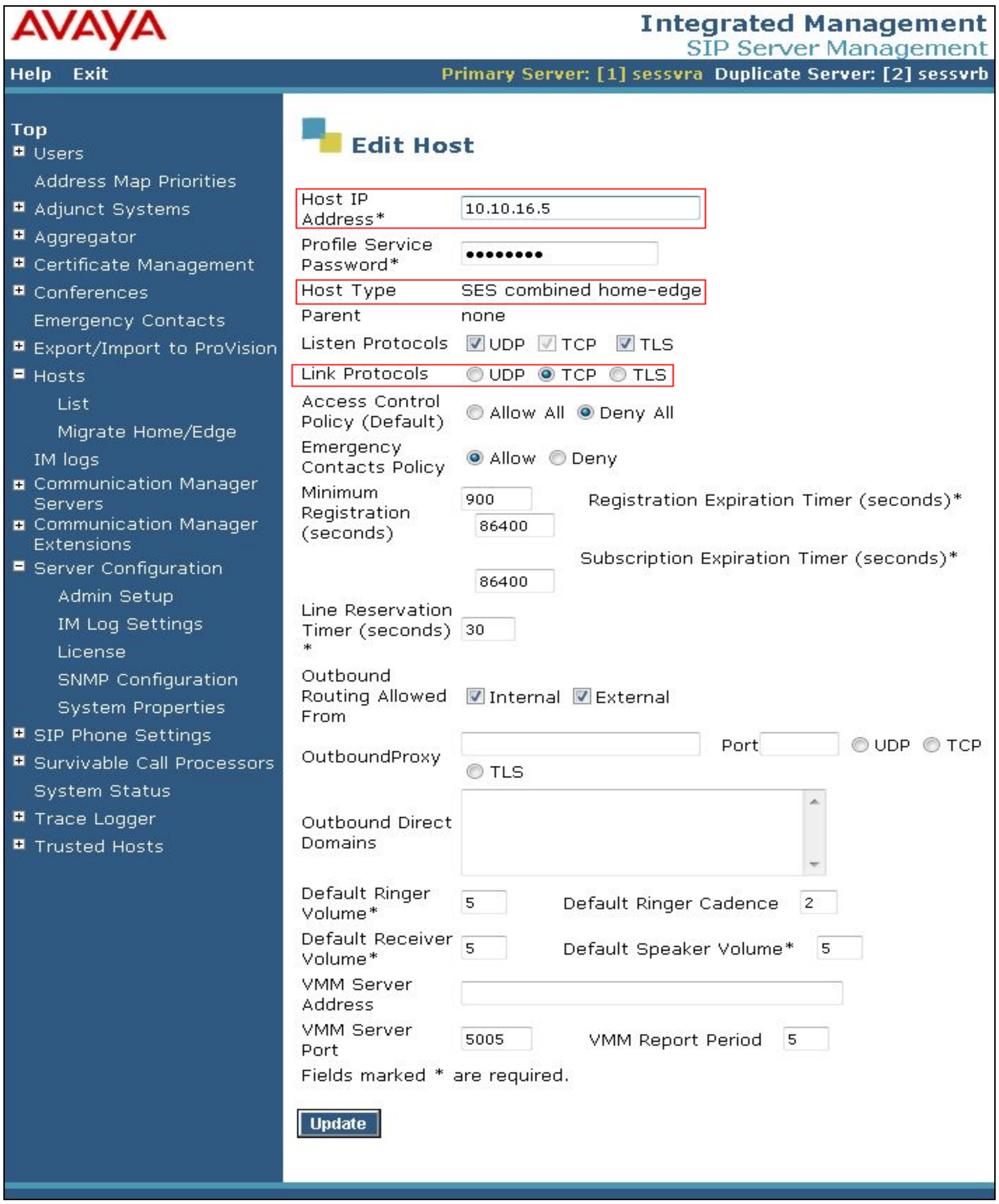
5.2. Verifying System Properties

From the left pane of the Administration web interface, expand the **Server Configuration** option and select **System Properties**. This screen displays the SES version and network properties configured during the installation process. In the **View System Properties** screen, verify the **SIP Domain** name assigned to SES. This domain should match the domain configured in Communication Manager for the IP network region covered in **Section 4.4.2** and the SIP signaling group to SES covered in **Section 4.4.4**.



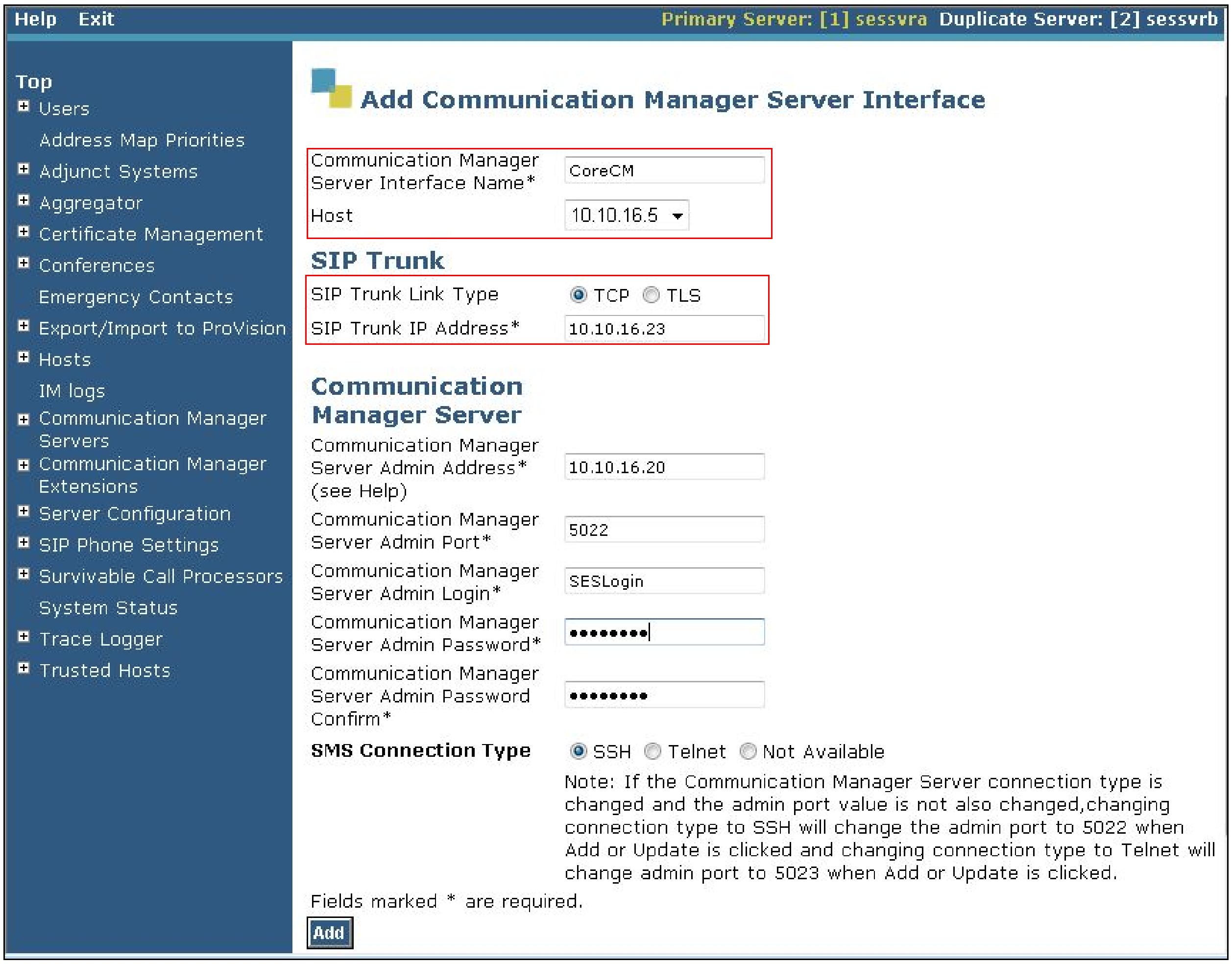
5.3. Administer Avaya Aura™ SIP Enablement Services Host Properties

After verifying the system properties create a host entry for SES. The following example shows the **Edit Host** screen since the host had already been configured. Enter the active IP address of SES in the **Host IP Address** field. The **Profile Service Password** was specified during the system installation. Next, verify the **Host Type** field. In this example, both servers in the redundant pair were configured as an **SES combined home/edge** during the initial setup. The **Link Protocols** selected defaults to TLS but in this example **TCP** was used. The default values for the other fields may be used as shown below.



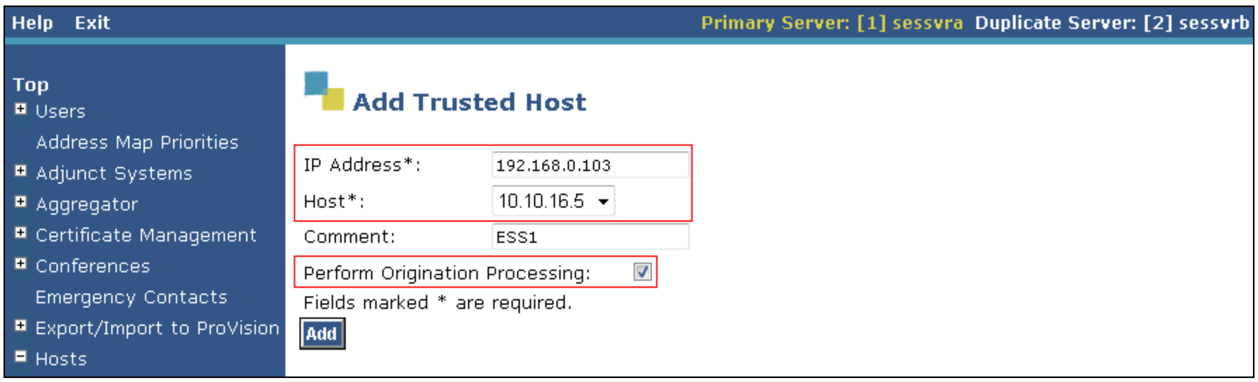
5.4. Add Avaya Aura™ Communication Manager Server

Under the **Communication Manager Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server since a SIP trunk is required between Communication Manager and SES. In this screen, enter a descriptive name in the **Communication Manager Server Interface Name** field and select the home server from the drop down menu in the **Host** field. Select **TCP** for the **Link Type** and enter the IP address of the C-LAN board in the Avaya G650 Media gateway in the **SIP Trunk IP Address** field. Scroll to the bottom, and click **Add**.



5.5. Administer Trusted Hosts

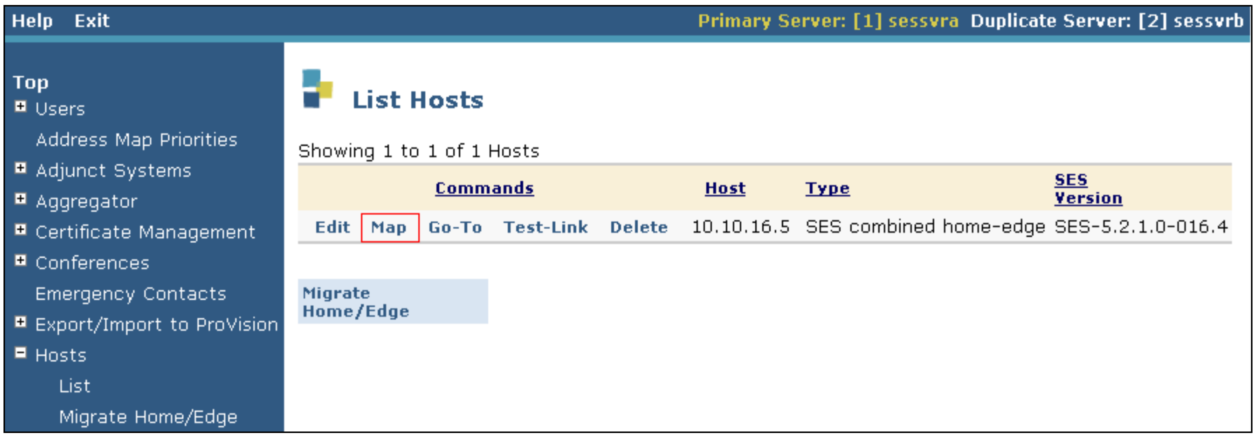
The IP addresses provided by IPC for the ESS servers and SIP network elements must be added as trusted hosts to the SES. For a trusted host, SES will not issue SIP authentication challenges for incoming requests from the designated IP address. If multiple SIP network elements are used in the IPC network the IP address of each must be added as a trusted host. From the left hand panel expand **Trusted Hosts** and click **Add**. In the **Add Trusted Host** screen enter the IP Address provided by IPC for the IPC network element in the **IP Address** field. As only one host has been configured the **Host*** field will default to the IP Address administered in **Section 5.3**. Enter a descriptive comment and select the **Perform Origination Processing** check box. Click **Add**.



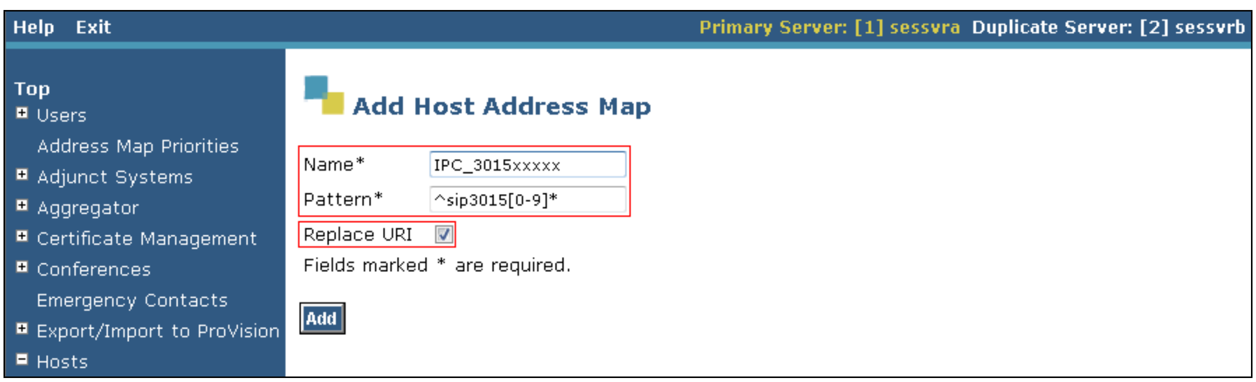
The resulting screen displays a message confirming the trusted host has been added. Click **Continue** (not shown) to view a list of the administered trusted hosts. The screen below shows the trusted hosts administered for the example configuration. Repeat the above step for each IPC network element that will issue SIP requests to SES.

5.6. Administer Address Maps to IPC

Host Address Maps use pattern matching on the SIP URI to direct messages to the corresponding contact address (e.g., IPC ESS). An example is the pattern of **^sip:301[0-9]*** which will match on all calls having digits beginning with 301. To configure a **Host Address Map** in the left panel select **Hosts → List Hosts** and in the right panel click on the **Map** link associated with the appropriate host.

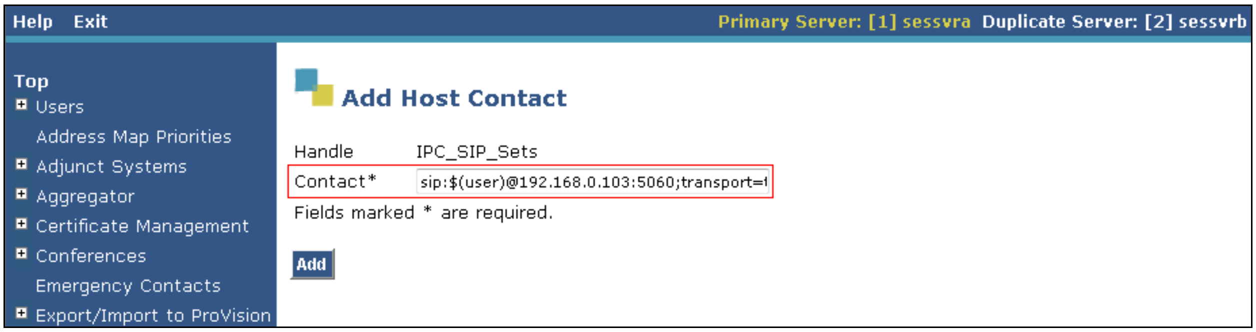


Click on the **Add Map In New Group** link (not shown). In the resulting screen enter a descriptive name in the **Name** field and specify an appropriate pattern for the call type. In this example, the pattern used is **^sip:3015[0-9]***.Leave the **Replace URI** checkbox selected and click the **Add** button once the form is completed.

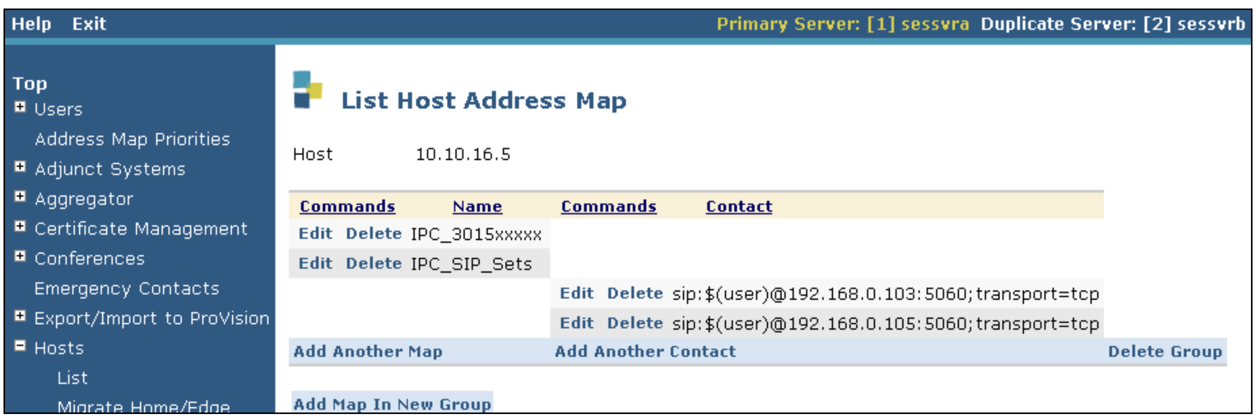


If required, repeat these steps to administer additional address maps matching other digit strings.

The next step is to enter the contact addresses for the IPC ESS. For redundancy two contact points are specified, one for each IPC ESS. In this example, an IP address is used to identify each IPC ESS. To add a contact address, click on the **Add Another Contact** link to open the **Add Host Contact** screen. In this screen, the **Contact** field specifies the destination for the call which is entered as: **sip:$(user)@192.168.0.103:5060;transport=tcp** The user part in the original request URI is inserted in place of the **$(user)** string before the message is sent to IPC. Click the **Add** button when completed. Repeat for the second ESS.



After configuring the host address maps and contact information, the **List Host Address Map** screen will appear as shown below.

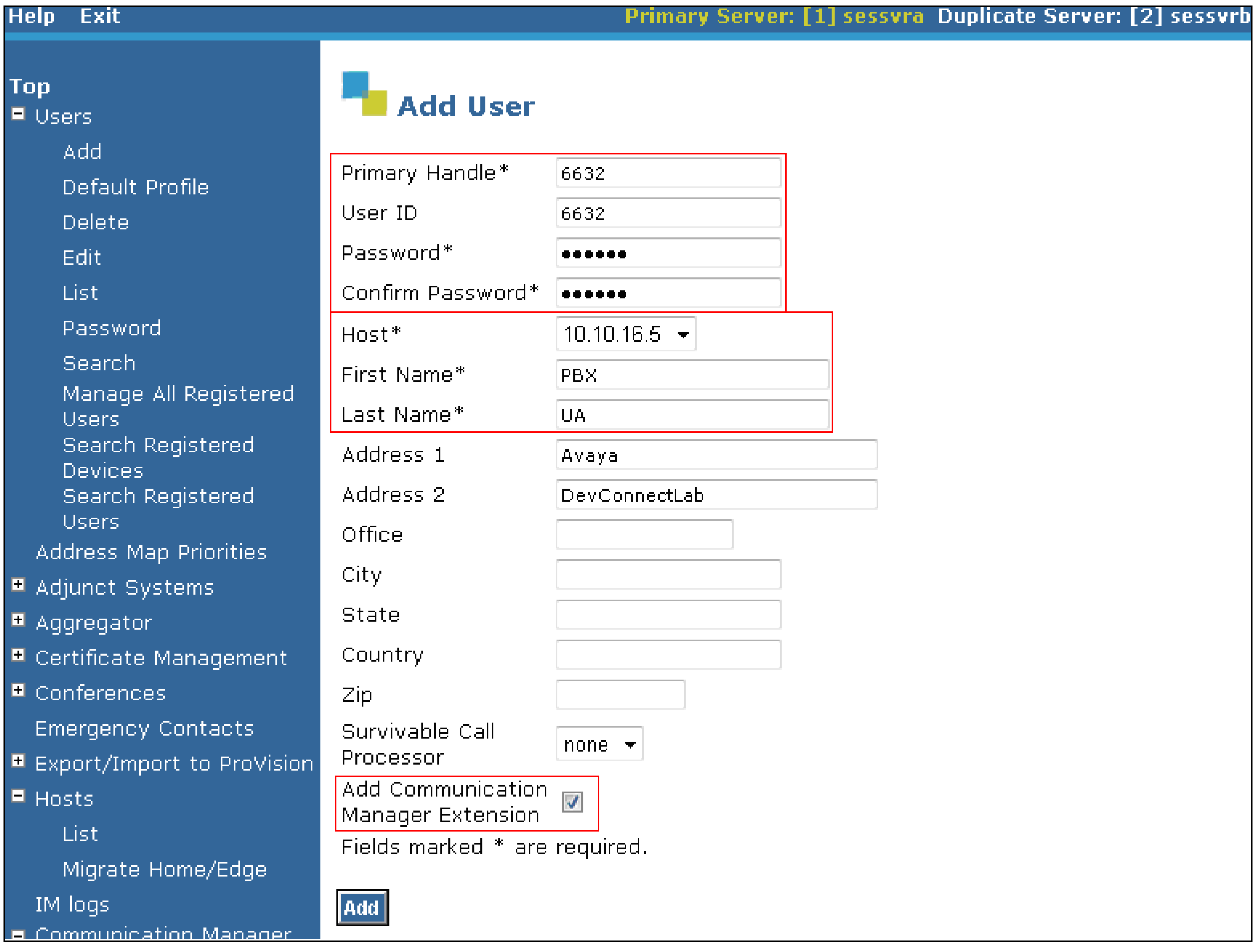


To support redundancy provided by multiple IPC ESS servers some specific configuration is required on SES. Change the SIP parameters shown in the table below in the file **/usr/impress/sip-server/etc/ccs.conf** on each of the SES servers. Restart each server after the changes have been saved

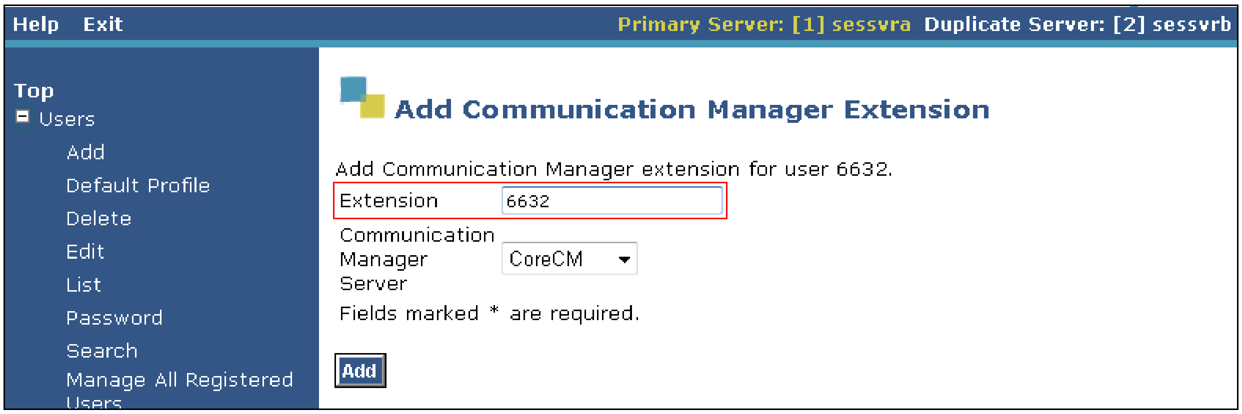
Default Settings	Required Settings
PerContactWaitTime=30	PerContactWaitTime=180
MM_PerContactWaitTime=2	MM_PerContactWaitTime=0
TimerB=32000	TimerB=2000

5.7. Administer Avaya Aura™ SIP Enablement Services PBXUA Users

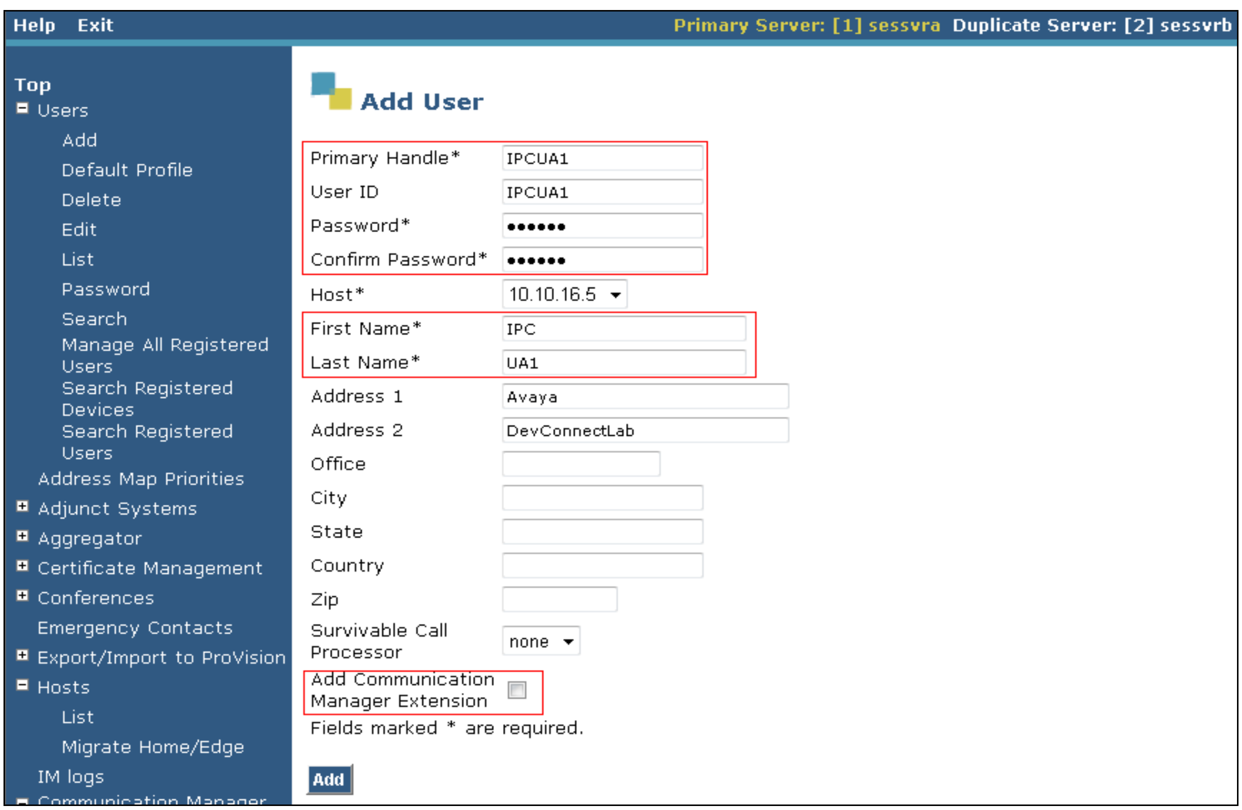
Each PBXUA registers with the SES as a user. In the left Panel navigate to **Users → Add**. Enter the Communication Manger extension that will be associated with this user for **Primary Handle** and **User ID**. Enter and confirm the user **Password** (the Primary Handle, User ID and Password must match what is defined in the IPC configuration). As only one host has been configured the **Host** field will default to the IP Address administered in **Section 5.3**. Enter a descriptive name for the **First Name** and **Last Name** fields. Select the **Add Communication Manger Extension** check box to assign a Communication Manager extension now. Click **Add**



In the resulting screen click **Continue** (not shown). In the **Add Communication Manager Extension** screen enter the **Extension** configured on the Communication Manager and ensure the correct **Communication Manager Server** associated with this extension is selected. Click **Add**



Each PBXUA needs a second User defined on SES for authentication purposes. In the left panel navigate to **Users → Add**. Enter an alpha numeric value of at least 3 characters for **Primary Handle** and **User ID**, Enter and confirm the user **Password** (the **User ID** and **Password** must match what is defined in the IPC configuration). Enter a descriptive name for the **First Name** and **Last Name** fields. This user does not require a Communication Manager extension so leave the **Add Communication Manger Extension** check box unchecked. Click **Add.**



6. General Test Approach and Test Results

Enterprise Reach allows the Communication Manager to share the telephony lines on the Alliance MX as well as updating the Communication Manager extensions with call status (Active , Hold, etc) Communication Manager provides further updates to the stations that have BLAs configured for the shared lines. Calls can be made or received from any of the configured Avaya stations using the BLAs or IPC turrets via the telephony lines at the Alliance MX. The Compliance test focused on the interaction between the two systems when calls were made or received using the shared lines. Various call scenarios were run to confirm the correct call updates are exchanged between the Avaya and IPC systems including Basic calls, Hold, Transfer, Conference and Exclusion. Below is an example of the type of checks that were undertaken during an inbound call.

- Alerting at both the Alliance turrets and Communication Manager user extensions
- Call can be answered at either party (IPC Alliance or Avaya extension)
- Following an answer, busy status is indicated at the corresponding parties.
- Answered calls placed on hold by the party that is in conversation state
- Hold status is indicated at the corresponding parties
- Retrieve held calls (even if the user did not initiate call hold).
- Call intrusion, creating multi party conversation
- Activation of privacy with the use of exclusion.
- Only the party in conversation state can release the call.

Similarly outward bound calls can be made from Avaya stations using the BLAs or IPC turrets on the ER enabled lines. Similar checks to those listed above were undertaken for outward bound calls confirming that calls can be invoked and reflected/indicated at the sharing parties of that line. Testing of the sample configuration was completed with successful results for the IPC ER solution.

7. Verification Steps

The following steps can be used to verify that the required configuration has been correctly administered to support IPC ER. To verify that the trunk group to SES is up, from the Communication Manager SAT use the **status trunk n** command, where **n** is the number of the trunk group. (Refer to **Sections 4.4.5** for trunk details). Verify for each trunk, that the **Service State** shows in-service/idle.

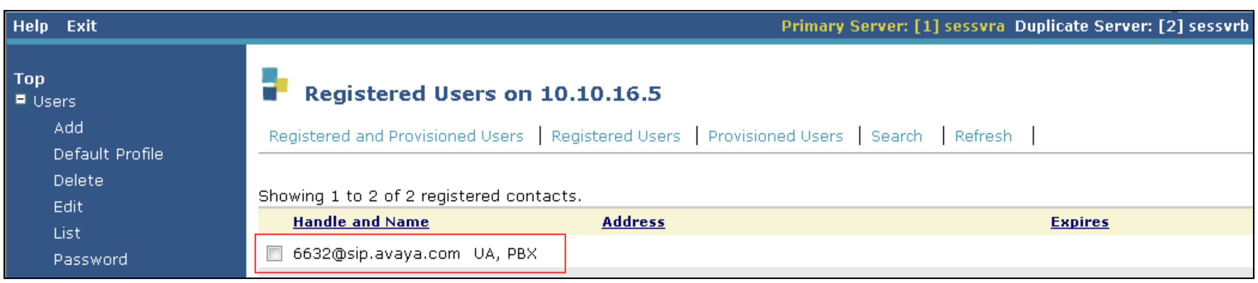
```
status trunk 6                                                                      Page   1

                                   TRUNK GROUP STATUS

Member    Port      Service State       Mtce Connected Ports
                                        Busy

0006/001 T00144   in-service/idle     no
0006/002 T00145   in-service/idle     no
0006/003 T00146   in-service/idle     no
0006/004 T00147   in-service/idle     no
0006/005 T00148   in-service/idle     no
0006/006 T00149   in-service/idle     no
```

To ensure that the PBXUA has successfully registered with SES from the SES administration web interface select **Administration → SIP Enablement Services** from the main screen (not shown). The SES administration home screen will be displayed. Navigate to **Users → Search Registered Users** and click **Search.** Check that the PBXUA is reported as being registered.



Place an in bound call to each of the IPC ER lines and verify that the ER line on the IPC turrets and the BLAs on the Communication Manager extensions indicate an incoming call and calls on each line ring and can be answered from both communication Manager and IPC turrets.

Place an outbound call over each Shared line using both IPC turrets and the BLA/speed dials configured on the Communication Manager extensions. Verify that the ER line on the IPC turrets and the other BLAs on the Communication Manager extensions indicate line busy status. Verify that the call can be put on hold and retrieved and sharing parties can barge in to the call.

8. Conclusion

These Application Notes describe the steps required to successfully configure the Avaya components to successfully interoperate with IPC ER line sharing solution using SIP as the transport method between the Avaya and IPC systems including Avaya Aura^{TM} Communication Manager and Avaya Aura^{TM} SIP Enablement services.

9. Additional References

This section references the Avaya documentation relevant to these Application Notes. Additional Avaya product documentation is available at http://support.avaya.com.

[1] *Administering Avaya Aura™ Communication Manager, 04-May-2009,* Document Number 03-300509

[2] *SIP Support in Avaya Aura™ Communication Manager Running on the Avaya S8xxx Servers, 04-May-2009*, Document Number 555-245-206

[3] *Avaya Aura™ SIP Enablement Services (SES) Implementation Guide, 04-May-2009,* Document Number 16-300140

[4] *RFC 3261 SIP: Session Initiation Protocol*, http://www.ietf.org/

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.