Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura™ Communication Manager, Avaya Aura™ SIP Enablement Services, Avaya Modular Messaging and Intuity™ AUDIX® LX to support IPC Alliance MX - Issue 1.0

Abstract

These Application Notes describe the procedure to configure Avaya Aura™ Communication Manager, Avaya Aura™ SIP Enablement Services, Avaya Modular Messaging and Intuity™ AUDIX® LX to support IPC Alliance MX using QSIG (Q Signaling Protocol) Connectivity.

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 solution provided by IPC can interoperate with Avaya when connected by QSIG.
The Avaya solution will consist of the following:

- Avaya Aura™ Communication Manager
- Avaya Aura™ SIP Enablement Services
- Avaya Modular Messaging
- Intuity™ AUDIX® LX

The IPC solution will consist of the following:

- IPC Alliance MX
- IPC System Center
- IPC turrets

The Avaya Aura™ Communication Manager will be connected via an E1-QSIG trunk to the Alliance MX. The Alliance MX is a voice technology product designed to provide a high resiliency platform for the provisioning of telephony and other associated services to financial traders. The Alliance MX provides its users with connectivity to various telephone transport services. Included in the transport services is E1 connectivity for connection within the private telephony network where the signaling protocol is QSIG. Based on IPC support policy, there is no IPC configuration documented in these Application Notes. IPC engineers will be responsible for the installation and maintenance of Alliance MX products. These Application Notes describe the required configuration steps for Avaya solution components

1.1. Interoperability Compliance Testing

The interoperability compliance test focused on the ability for the IPC solution to interoperate with the Avaya solution. The following is a summary of the feature and serviceability testing that was undertaken.

- Basic Calls, including calling/connected party name/number display and restriction
- Hold
- Conference
- Call Transfer including calling/connected party name/number display and restriction at the primary and secondary party of the transfer
- Call forward with tests for call forward unconditional, call forward busy and call forward no reply
- Multiple call forward including calling/connected party name/number display at the calling and the diverted to party of the call forward.
- Call forward, loop avoidance
- Mail box access and message retrieval
- Message waiting indication activation and deactivation

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.

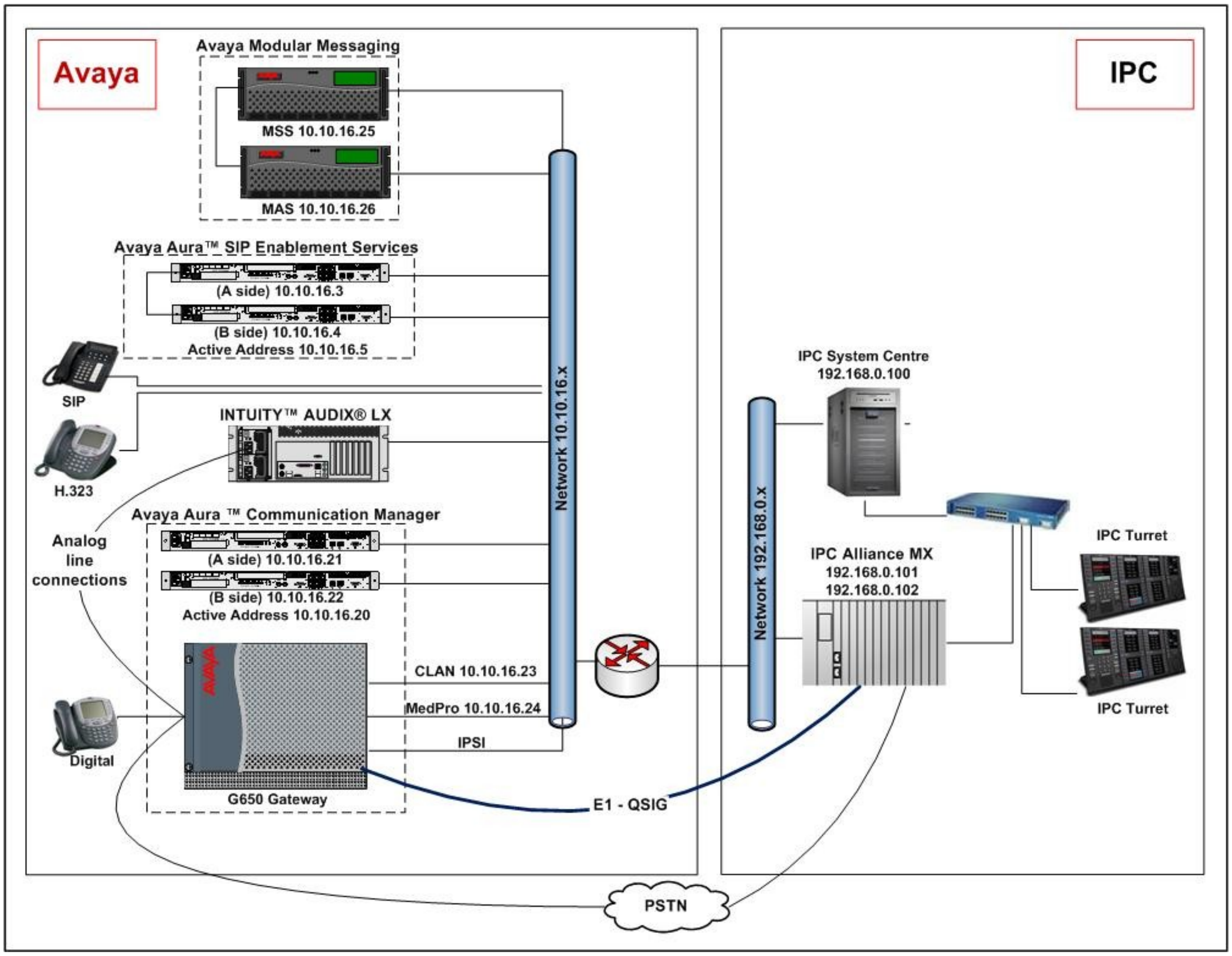
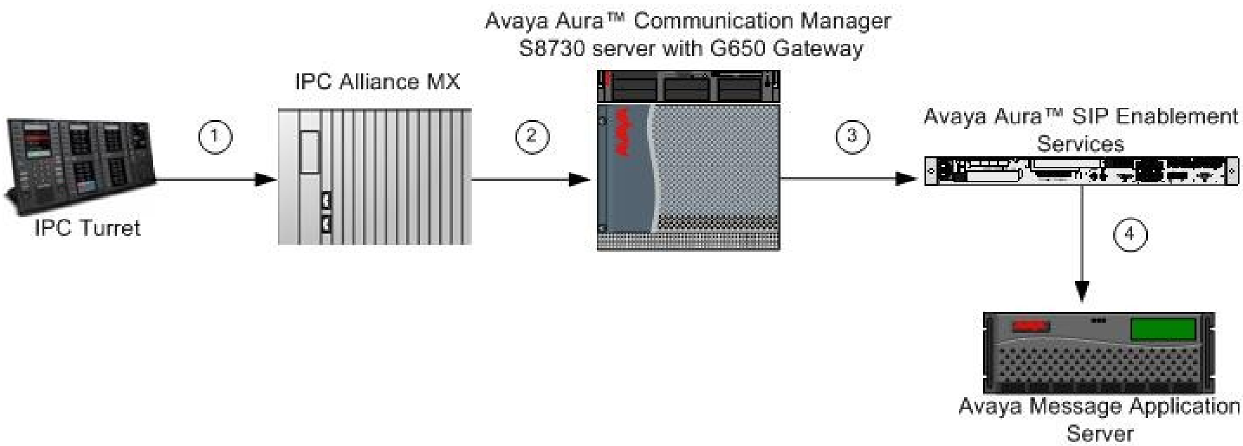


Figure 1: Test Environment Network Topology

Note: Although the Avaya and IPC IP networks are connected, all voice traffic between Avaya and IPC components use the E1-QSIG connection represented by the blue line toward the bottom of **Figure 1.** The PSTN connection was used for test purposes only and the configuration of this connection is not covered in these Application Notes.

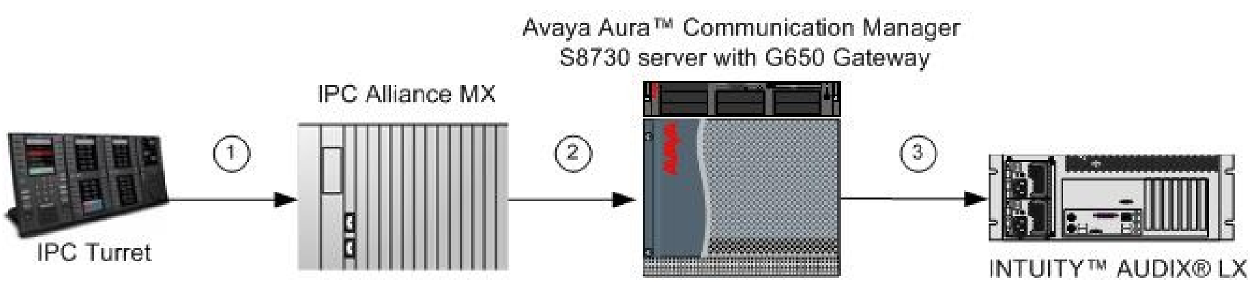
To better understand the logical connections between the two solutions shown in **Figure 1.** Two call flows are described in this section. The first call scenario is an incoming call from an IPC turret to Modular Messaging

1. An IPC turret user dials the Modular Messaging pilot number
2. IPC Alliance MX routes the call via the QSIG trunk to Communication Manager
3. Communication Manager uses its dial plan configuration to route the call to SES (SIP Enablement Services) via a SIP trunk
4. SES routes the call to Modular Messaging via another SIP trunk configured to the MAS (Message Application Server) where the call is answered.



The second call scenario is an incoming call from an IPC user to Intuity AUDIX LX

1. An IPC turret user dials the AUDIX hunt group number assigned in the Communication Manager
2. IPC Alliance MX routes the call via the QSIG trunk to Communication Manager
3. Communication Manager uses its hunt group with analog lines to route the call to Intuity AUDIX LX where the call is answered.



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 (Access Element)
Avaya G650 Media Gateway - CLAN - TN799DP - MedPro - TN 2602AP	 HW16 FW032 HW08 FW048
Avaya S3210 Server	INTUITY™ AUDIX® LX 2.0. Service Pack 2. Patch 07034rf+b
Avaya S3500 Server	Avaya Modular Messaging, Message Application Server 5.2. Patch 8
Avaya S3500 Server	Avaya Modular Messaging, Message Storage Server 5.2. Patch 8
Avaya 9630 IP Telephones	SIP: 2.5.0.0 H.323: R3.0
- IPC System Center (Dell R710) - IPC Information Systems Alliance MX - IPC IQ/MAX Turrets	16.00.00 Patch 2

4. Configure Avaya Aura^{TM} Communication Manager

This section describes the steps for configuring the Communication Manager. All configurations in the section are administered using the System Access Terminal (SAT). These Application Notes assume that the basic Communication Manager configuration has already been administered. The procedures covered include the following:

- Confirm Necessary Features
- Confirm Special Applications
- Confirm Call forwarding Configuration
- Administer Feature Access Codes
- Configure QSIG Trunk to Alliance MX
- Administer Private Numbering
- Administer IP Network Region
- Administer IP Codec Set
- Configure SIP Trunk to SES
- Administer Public Numbering
- Administer Route patterns
- Administer Dialplan Analysis
- Administer Uniform Dialplan
- Administer AAR
- Administer Hunt Group for Modular Messaging
- Administer Coverage Path for Modular Messaging
- Administer LAN Integration to AUDIX
- Administer Hunt Group for AUDIX
- Administer Coverage Path for AUDIX

4.1. Confirm Necessary Features

The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity. Log into the Communication Manager SAT interface and use the **display system-parameters customer-options** command to determine these values. On **Page 2** verify that the available **Maximum Administered SIP Trunks** is equal to or greater than the desired number of simultaneous SIP trunk connections.

```
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: 1800   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: 200    78
  Maximum Administered Ad-hoc Video Conferencing Ports: 0      0
```

On **Page 3** verify the fields **ARS** and **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)? n
          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 the fields **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
```

On **Page 8**, verify that **Basic Call Setup**, **Basic Supplementary Services**, **Centralized Attendant**, **Supplementary Services with Rerouting** and **Transfer into QSIG Voice Mail** are all 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 **display system-parameters features** command to verify the following. An attendant console was used during the compliance test to intercept calls to unassigned numbers. On **Page 1** verify **DID/Tie/ISDN/SIP Intercept Treatment** is set to **attd** to make sure these calls are routed to the attendant console.

```
display system-parameters features                              Page   1 of  18
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? y
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y
                             Music/Tone on Hold: none
              Music (or Silence) on Transferred Trunk Calls? no
                       DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n
```

On **Page 8** confirm **QSIG/ETSI TSC Extension** and **QSIG Path Replacement Extension** fields are configured with valid extensions and that the **MWI – Number of Digits Per Voice Mail Subscriber** is configured with the appropriate extension length.

```
display system-parameters features                              Page   8 of  18
                        FEATURE-RELATED SYSTEM PARAMETERS
ISDN PARAMETERS
                                                         PARAMETERS FOR CREATING
 Send Non-ISDN Trunk Group Name as Connected Name? y     QSIG SELECTION NUMBERS
 Display Connected Name/Number for ISDN DCS Calls? y       Network Level:
       Send ISDN Trunk Group Name on Tandem Calls? y        Level 2 Code:
                Send Custom Messages Through QSIG? y        Level 1 Code:

                          QSIG/ETSI TSC Extension: 6666
 MWI - Number of Digits Per Voice Mail Subscriber: 4
                                 Feature Plus Ext:
                              National CPN Prefix:
                         International CPN Prefix:
                                Pass Prefixed CPN: ASAI? n   VDN/Vector? n
    Unknown Numbers Considered Internal for AUDIX? y        Maximum Length: 5
             USNI Calling Name for Outgoing Calls? n
               Path Replacement with Measurements? y
                   QSIG Path Replacement Extension: 6667
   Send QSIG Path Replacement Conf. Event to ASAI? y
```

On **Page 9** confirm that **CPN/ANI/ICLID PARAMETERS** have a relevant text string configured.

```
display system-parameters features                              Page   9 of  18
                        FEATURE-RELATED SYSTEM PARAMETERS

CPN/ANI/ICLID PARAMETERS
   CPN/ANI/ICLID Replacement for Restricted Calls: restricted
  CPN/ANI/ICLID Replacement for Unavailable Calls: restricted
```

On **Page 15** confirm that **Chained Call-forwarding** is set to **y.** This feature enables the ability to alter the number of allowed QSIG re-routes covered in **Section 4.3**.

```
display system-parameters features                              Page  15 of  18
                        FEATURE-RELATED SYSTEM PARAMETERS
SPECIAL TONE
                                  Special Dial Tone? n
          Special Dial Tone for Digital/IP Stations: none

REDIRECTION NOTIFICATION
                           Display Notification for Do Not Disturb? n
                           Display Notification for Send All Calls? n
                             Display Notification for Call Forward? n
                    Display Notification for Enhanced Call Forward? n
                         Display Notification for a locked Station? n
         Display Notification for Limit Number of Concurrent Calls? n
                          Display Notification for Posted Messages? n
                                Scroll Status messages Timer(sec.):
Chained Call Forwarding? y
```

4.2. Special Applications

Use the **display system-parameters special-applications** command. On **Page 3**, verify that **(SA8440) - Unmodified QSIG Reroute Number?** is set to **y**. When a call that arrives on a QSIG trunk is then diverted off net, a facility message is sent back toward the switch that originated the call to allow the originating switch to pick a better route to reach the diverted-to party. The facility message contains the number of the diverted-to party. This number is normally processed by Communication Manager so that the digits in the facility message are not the same digits as those entered when the call forwarding feature was activated. When SA8440 feature is active, the number in the facility message will not be processed by Communication Manager so it will exactly match the number entered when call forwarding was activated. If this option is not set, please contact Avaya sales team or business partner for the appropriate license file.

```
display system-parameters special-applications                  Page   3 of   9
                             SPECIAL APPLICATIONS

                   (SA8141) - LDN Attendant Queue Priority? n
       (SA8143) - Omit Designated Extensions From Displays? n
            (SA8146) - Display Update for Redirected Calls? n
              (SA8156) - Attendant Priority Queuing by COR? n
               (SA8157) - Toll Free Vectoring until Answer? n
  (SA8201) - Start Time and 4-Digit Year CDR Custom Fields? n
                        (SA8202) - Intra-switch CDR by COS? n
                    (SA8211) - Prime Appearance Preference? n
                      (SA8240) - Station User Admin of FBI? n
                                 (SA8312) - Meet-Me Paging? n
                   (SA8323) - Idle Call Preference Display? n
                         (SA8339) - PHS X-Station Mobility? n
                  (SA8348) - Map NCID to Universal Call ID? n
               (SA8428) - Station User Button Ring Control? n
             (SA8434) - Delay PSTN Connect on Agent Answer? n
                          (SA8439) - Forward Held-Call CPN? n
                   (SA8440) - Unmodified QSIG Reroute Number? y

                                            (SA8475) - SOSM? n
```

4.3. Confirm Call Forwarding Configuration

Use the **display system-parameters coverage-forwarding** command to verify on **Page 2** that the **Maximum Number Of Call Forwarding Hops** is set to a value mutually agreed with IPC**.** This feature determines the number of QSIG re-route requests the Communication Manager will accept. If this value is lower than the value used by IPC then the Communication Manager will reject any QSIG re-route requests from the Alliance MX once the specified value has been reached. This will force the Alliance MX to trombone calls by forward switching any further diversions.

```
display system-parameters coverage-forwarding                     Page   2 of   2
                SYSTEM PARAMETERS CALL COVERAGE / CALL FORWARDING

COVERAGE OF CALLS REDIRECTED OFF-NET (CCRON)

                       Coverage Of Calls Redirected Off-Net Enabled? y
 Activate Answer Detection (Preserves SBA) On Final CCRON Cvg Point? y
                                  Ignore Network Answer Supervision? n
                 Disable call classifier for CCRON over ISDN trunks? n
                  Disable call classifier for CCRON over SIP trunks? n

CHAINED CALL FORWARDING
                             Maximum Number Of Call Forwarding Hops: 6
                Station Coverage Path For Coverage After Forwarding: principal
```

4.4. 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
```

On **Page 3**, Verify a **Per Call CPN Blocking Code Access Code** is assigned.

```
display feature-access-codes                                      Page   3 of   8
                               FEATURE ACCESS CODE (FAC)
             Leave Word Calling Send A Message:
           Leave Word Calling Cancel A Message:
   Limit Number of Concurrent Calls Activation:        Deactivation:
               Malicious Call Trace Activation:        Deactivation:
         Meet-me Conference Access Code Change:
          Message Sequence Trace (MST) Disable:

 PASTE (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:        Dissociate Code:
        Per Call CPN Blocking Code Access Code: *34
      Per Call CPN Unblocking Code Access Code: *35
                   Posted Messages Activation:         Deactivation:
                 Priority Calling Access Code: *30
                          Program Access Code:
```

4.5. Configure QSIG Trunk to Alliance MX

This section describes the steps needed to configure a QSIG trunk to Alliance MX on the Communication Manager. In the sample configuration this trunk will be used to transit calls between the Avaya and IPC solutions.

4.5.1. Administer DS1

Use the **add ds1 n** command where **n** is the board location of the DS1 Circuit Pack that will be used for the QSIG connection between Communication Manager and the Alliance MX. The values used should be agreed with IPC prior to configuration. The screen output below shows the values used during this compliance test. Modified fields are shown in bold, and all other fields were left as default.

```
add ds1 01a06                                                 Page   1 of   1
                                DS1 CIRCUIT PACK

            Location: 01A06                          Name: QSIG-IPC
            Bit Rate: 2.048                   Line Coding: hdb3

      Signaling Mode: isdn-pri
             Connect: pbx                       Interface: peer-master
   TN-C7 Long Timers? n                     Peer Protocol: Q-SIG
Interworking Message: PROGress                       Side: a
Interface Companding: alaw                           CRC? y
           Idle Code: 11111111          Channel Numbering: timeslot
                                DCP/Analog Bearer Capability: 3.1kHz

                                         T303 Timer(sec): 4
                                        Disable Restarts? n

      Slip Detection? n                 Near-end CSU Type: other

   Echo Cancellation? n
```

4.5.2. Administer QSIG Signaling Group

Use the **add signaling-group n** command; where **n** is the number of the signaling-group to create.

- Set the **Group Type** field to be **isdn-pri**
- The **Primary D-Channel** is set to channel 16 of the DS1 circuit pack configured in **Section 4.5.1**
- The **TSC Supplementary Service Protocol** is set to **b**

The **Max number of NCA TSC, Trunk Group for NCA TSC** and **Trunk Group for Channel Selection** must all be set after the trunk group has been added by running the command **change signaling-group 3.** The **Max number of NCA TSC** must be at least 2, one for Communication Manager and one for Alliance MX.

```
add signaling-group 3                                        Page   1 of   1
                                SIGNALING GROUP

 Group Number: 3               Group Type: isdn-pri
                      Associated Signaling? y          Max number of NCA TSC: 5
                         Primary D-Channel: 01A0616     Max number of CA TSC: 5
                                                     Trunk Group for NCA TSC: 3
       Trunk Group for Channel Selection: 3
      TSC Supplementary Service Protocol: b          Network Call Transfer? n
```

4.5.3. Administer QSIG Trunk Group

Use the command **add trunk-group n** where **n** is the number of the QSIG trunk group to create. This trunk will be used to connect Communication Manager to Alliance MX.

- Set the **Group Type** field to be **isdn**
- Add a descriptive name into the **Group Name** field
- Set the **TAC** field to a valid dial access code (dac) according to the dial plan configuration
- Set the **Carrier Medium** field to **PRI/BRI**
- Set the **Service Type** field to **tie**

```
add trunk-group 3                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 3                     Group Type: isdn          CDR Reports: y
  Group Name: IPC QSIG                     COR: 1        TN: 1        TAC: 503
   Direction: two-way        Outgoing Display? n         Carrier Medium: PRI/BRI
 Dial Access? y              Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n             TestCall ITC: rest
                         Far End Test Line No:
TestCall BCC: 4
```

On **Page 2** of the trunk group form set the **Supplementary Service Protocol** to **b.** The **Digit Handling (in/out)** field should be set to a value mutually agreed with IPC; in the sample configuration **overlap/enbloc** is used.

```
add trunk-group 3                                                Page   2 of  21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
        Max Message Size to Send: 260   Charge Advice: none
  Supplementary Service Protocol: b     Digit Handling (in/out): overlap/enbloc
       Digit Treatment:                                       Digits:
            Trunk Hunt: cyclical
                                                    Digital Loss Group: 13
Incoming Calling Number - Delete:     Insert:                Format:
              Bit Rate: 1200         Synchronization: async    Duplex: full
 Disconnect Supervision - In? y  Out? n
 Answer Supervision Timeout: 0
          Administer Timers? n        CONNECT Reliable When Call Leaves ISDN? n
```

On **Page 3** of the trunk group form set **Send Name** and **Send Calling Number** to **y.** Set the **Format** field to **private** so that calls will reference the private numbering table. Set the **Replace Restricted Numbers?, Replace Unavailable Numbers?** and **Send Connected Number** to **y**. **Modify Reroute Number** is the administrative control for special application SA8440 (covered in **Section 4.2**) and should be set to **n.**

```
add trunk-group 3                                                Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member: 1
                                      Send Name: y      Send Calling Number: y
            Used for DCS? n             Hop Dgt? n     Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: private
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y
                                                      Send Connected Number: y
                                                  Hold/Unhold Notifications? y
             Send UUI IE? y                    Modify Tandem Calling Number? n
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                          Ds1 Echo Cancellation? n
                                                      Modify Reroute Number? n
    Apply Local Ringback? n
 Show ANSWERED BY on Display? y
                             Network (Japan) Needs Connect Before Disconnect? n
 DSN Term? n
```

On **Page 4** of the trunk group form set **Diversion by Reroute, Path Replacement** and **Display Forwarding Party Name** to **y.**

```
add trunk-group 3                                             Page   4 of  21
                       QSIG TRUNK GROUP OPTIONS




   TSC Method for Auto Callback: drop-if-possible
           Diversion by Reroute? y
               Path Replacement? y
Path Replacement with Retention? n
        Path Replacement Method: better-route
                           SBS? n
  Display Forwarding Party Name? y
    Character Set for QSIG Name: eurofont
               QSIG Value-Added? n
```

4.6. Administer Private Numbering

To ensure that the caller number is correctly presented, the QSIG trunk group set up in **Section 4.5.3** references the private numbering table. Enter the command **change private-numbering n** and set the following values:

- Set **Ext Len** field to **4,** this is the length of the extensions that will be using the table.
- Set **Ext Code** to match the leading digits of extension ranges to be used
- Set **Trk Grp(s)** to **3,** this is the number of the trunk group that will use this entry.
- Set **Total Len** to **4,** this is the total length of the calling number that will be presented by the trunk group.

```
change private-numbering 0                                       Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk       Private           Total
Len Code             Grp(s)    Prefix            Len
 4  31               3                           4     Total Administered: 4
 4  37                                           4        Maximum Entries: 540
 4  66               3                           4
```

4.7. Administer IP Network Region

Use the **change ip-network-region n** command, where **n** is the network region number to configure. For the **Authoritative Domain** field, enter the SIP domain name for this enterprise, a descriptive **Name** for the 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 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
```

4.8. 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. In the sample configuration Modular Messaging uses the G.711A codec, this codec must be included.

```
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:
```

4.9. Configure SIP Trunk to SES

This section describes the steps needed to configure a SIP trunk to the SES on the Communication Manager. In the sample configuration this trunk will be used to route calls to Modular Messaging that is connected via another SIP trunk to the SES.

4.9.1. Administer IP Node Names

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

```
 change node-names ip
                                 IP NODE NAMES
    Name              IP Address
CLAN1               10.10.16.23
Gateway             10.10.16.1
sesactive           10.10.16.5
default             0.0.0.0
```

4.9.2. Administer SIP Signaling Group

To create a SIP Signaling group use the **add signaling-group n** command.

- Set the **Group Type** field to be **sip**
- Set the **Transport Method** to the desired transport method; tcp (Transport Control Protocol) or tls (Transport Layer Security). For transparency **tcp** was used during this compliance test but the recommended method is tls
- 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 that was entered into the IP Node-names form
- The **Far-end network Region** to the region configured in **Section 4.7**
- The **Far-end Domain** is set to the domain name that is used by SES and Modular Messaging

```
add 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
```

4.9.3. Administer SIP Trunk Group

Use the **add trunk-group n** command where **n** is the number of the SIP trunk group to create. This trunk will be used to connect Communication Manager to SES

- Set the **Group Type** field to be **sip**
- Add a descriptive name into the **Group Name** field
- Set the **TAC** field to a valid dial access code (dac) according to the dial plan configuration
- Set the **Service Type** field to **tie**
- Set the **Signaling Group** field to the signaling group set up in **Section 4.9.2**
- Set the **Number of Members** field to the number of channels required on the trunk group

```
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
```

On **Page 3** of the trunk-group form set the **Numbering Format** field to **public**.

```
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
```

On **Page 4** of the trunk-group form ensure the **Support Request History** field is set to **y** as Modular Messaging relies on the History Info headers to select an appropriate mail box.

```
add trunk-group 6                                            Page   4 of  21
                              PROTOCOL VARIATIONS
                      Mark Users as Phone? n
            Prepend '+' to Calling Number? n
      Send Transferring Party Information? n
                 Network Call Redirection? n
                    Send Diversion Header? n
                  Support Request History? y
             Telephone Event Payload Type:
```

4.10. Administer Public Numbering

To ensure that the caller number is correctly presented the SIP trunk group set up in **Section 4.9.3** references the public numbering table, use the command **change public-unknown-numbering n.** The following values should be set:

- Set **Ext Len** field to **4,** this is the length of the extensions that will be using the table.
- Set **Ext Code** to match the leading digits of extension ranges to be used
- Set **Trk Grp(s)** to **6,** this is the number of the trunk group that will use this entry.
- Set **Total Len** to **4,** this is the total length of the calling number that will be presented by the trunk group.

```
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              6                        4         Maximum Entries: 9999
 4  31              6                        4
```

4.11. Administer Route Patterns

Use the **change route-pattern n** command to add the route pattern that will direct calls to the QSIG trunk group. AAR will select this route pattern for calls to IPC. In this configuration trunk group **3** is added under the **Grp No** field. Set **TSC** to **y**, **CA TSC Request** to **none** and the **Numbering Format** field to **unk-unk**

```
 change route-pattern 3                                          Page   1 of   3
                    Pattern Number: 3   Pattern Name: IPC_QSIG
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 3    0                                                             n   user
 2:                                                                    n   user
 3:                                                                    n   user
 4:                                                                    n   user
 5:                                                                    n   user
 6:                                                                    n   user

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

Use the **change route-pattern n** command to add the route pattern that will direct calls to the SIP trunk group. AAR will select this route pattern for calls to Modular Messaging. In this configuration trunk group **6** is added under the **Grp No** field.

```
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
```

4.12. Administer Dialplan Analysis

Use the **change dialplan analysis** command to administer the dialplan. In this configuration extensions in the range 31xx are assigned to IPC turrets and are configured as **udp** to send calls via the UDP (uniform dial plan). Extensions ranges 66xx, 89xx, 88xx and 79xx are Communication Manager extensions and are configured as **ext.**

```
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     79         4    ext
    2            4    udp     88         4    ext
    30           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
    4            4    aar
    4            5    ext
    5            3    dac
    6            3    fac
    61           4    ext
    66           4    ext
```

4.13. Administer Uniform Dialplan

Use the **change uniform-dialplan** command to administer the UDP routing. It is possible to use the UDP to manipulate the dialed digits but in this configuration UDP is used to direct the matching calls to AAR. In addition to 31xx calls for IPC turrets, extension 8889 is directed to the AAR as it is the modular messaging pilot number

```
change uniform-dialplan
                      UNIFORM DIAL PLAN TABLE

 Matching Pattern   Len   Del   Insert Digits   Net    Conv   Node Num
 31                 4     0                     aar     n
 33                 4     0                     aar     n
 37                 4     0                     aar     n
 663                4     0                     aar     n
 8889               4     0                     aar     n
 972                5     0                     aar     n
```

4.14. Administer AAR

Use the **change aar analysis n** command to specify which route pattern to use based upon the number dialed. In this example, **Route Pattern 3** is used for IPC extensions beginning **31** and **Route Pattern 6** is used for the Modular Messaging pilot number **8889**.

```
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
    31                      4    4     3        aar           n
    33                      4    4     2        aar           n
    37                      4    4     7        aar           n
    663                     4    4     2        aar           n
    8889                    4    4     6        aar           n
    972                     5    5     4        aar           n
```

4.15. Administer Hunt Group for Modular Messaging

Use the **add hunt-group n** command where **n** is the number of the hunt-group to add. Give the hunt group a descriptive name and set **ISDN/SIP Caller Display** to **grp-name.** Set the **Group Extension** to a valid extension according to the dial plan.

Note: the hunt group **Group Extension** must be different from the extension used for the Modular Messaging pilot number.

```
add hunt-group 2                                             Page   1 of  60
                                  HUNT GROUP
            Group Number: 2                                ACD? n
              Group Name: Modular Messaging              Queue? n
         Group Extension: 8999                          Vector? n
              Group Type: ucd-mia                 Coverage Path:
                      TN: 1         Night Service Destination:
                     COR: 1                   MM Early Answer? n
           Security Code:               Local Agent Preference? n
 ISDN/SIP Caller Display: grp-name
```

On **Page 2** of the hunt group form set the **Message Center** to be **sip-adjunct** and enter a **Voice Mail Number** and **Voice Mail Handle**, in this configuration they are set to **8889** and **modmessaging** respectively. Enter the AAR access code as defined in the feature access codes form (**Section 4.4)** for **Routing Digits.**

```
add hunt-group 2                                             Page   2 of  60
                                  HUNT GROUP

                      Message Center: sip-adjunct

     Voice Mail Number        Voice Mail Handle        Routing Digits
                                                 (e.g., AAR/ARS Access Code)
     8889                     modmessaging             1
```

4.16. Administer Coverage Path for Modular Messaging

Use command **change coverage path n** where **n** is the number of the coverage path to administer. Set **Point 1** to **h2** to send covered calls using this coverage path to hunt group 2.

```
change coverage path 2                                          Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 2
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?             n               n
               Busy?             y               y
        Don't Answer?            y               y        Number of Rings: 2
                All?             n               n
 DND/SAC/Goto Cover?             y               y
   Holiday Coverage?             n               n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h2             Rng:    Point2:
  Point3:                        Point4:
```

Use the **change station n** command to add the coverage path to a station where **n** is the extension number of the station to administer. Enter the coverage path number in the **Coverage Path 1** field.

```
change station 6621                                             Page   1 of   5
                                     STATION

Extension: 6621                          Lock Messages? n              BCC: 0
     Type: 9630                          Security Code: ****            TN: 1
     Port: S00002                      Coverage Path 1: 2              COR: 1
     Name: IP2nd                       Coverage Path 2:                COS: 1
                                       Hunt-to Station:
```

4.17. Administer LAN Integration to AUDIX

Communication Manager is connected to AUDIX by a combination of analog stations and links configured against a CLAN.

4.17.1. Verify Local Node Number

Enter **display diaplan parameters** and verify a **Local Node Number** has been assigned if no node number has been assigned enter **1**. This number will be used as the machine ID in AUDIX administration and communication-interface processor-channels.

```
display dialplan parameters
                             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
```

4.17.2. Verify CLAN link number

To verify the link number that will be used when configuring the communication-interface processor-channels. Enter the command **display ip-interface n**, where **n** is the board location of the CLAN used to interface with AUDIX. The link number can be seen in the **Ethernet Link** field

```
display ip-interface 1a02                                          Page   1 of   3
                                  IP INTERFACES
                  Type: C-LAN
                  Slot: 01A02       Target socket load and Warning level: 400
           Code/Suffix: TN799  D          Receive Buffer TCP Window Size: 8320
      Enable Interface? y                          Allow H.323 Endpoints? y
                  VLAN: n                           Allow H.248 Gateways? y
        Network Region: 1                             Gatekeeper Priority: 5

                                 IPV4 PARAMETERS
             Node Name: CLAN1
           Subnet Mask: /24
     Gateway Node Name: Gateway

         Ethernet Link: 1
         Network uses 1's for Broadcast Addresses? y
```

4.17.3. Administer AUDIX Node Names

Use the **change node-names audix** command to define a name and **IP Address** for AUDIX.

```
change node-names audix                                            Page   1 of   1
                                  AUDIX NODE NAMES

      Audix Names   IP Address
      intuity       10 .10 .16 .35
                       .   .   .
```

4.17.4. Administer Communication Processor Channels

Use the **change communication-interface processor-channels** command to administer the required processor channels. In total three processor channels are required for this sample configuration. The following values should be used:

- **Enable** should be set to **y** to activate the channels once the entries have been saved.
- For **Appl** the first channel should be set to **audix** for the link between Communication Manager and AUDIX. The second channel should be set to **qsig-mwi** to support MWI interrogation and the third channel should be set to **gateway** for remote-AUDIX integration.
- **Mode** is always set to **s** .
- **Interface Link** should be set to the CLAN link number verified in **Section 4.17.2**.
- Set **Interface Chan** to **5002** for the **audix** application, for **qsig-mwi** and **gateway** applications a port beginning with **6xxx** must be used.
- **Destination Node** must match the name assigned on the node-name AUDIX screen in **Section 4.17.3**.
- **Destination Port** is always set **0** for direct connection with AUDIX.
- Set **Session Local** to **1** for the **audix** application, this is in order to match the node number assigned on the dialplan parameters screen. Set to **2** for the **qsig-mwi** application which will match the **machine ID** in the isnd mwi-prefixes screen to (be configured in the next step). Set to **3** for the **gateway** application. These session numbers will be matched to the configuration of AUDIX system
- **Session Remote** must match the AUDIX number assigned in the configuration of the AUDIX system.
- **Mach ID** should be set to **1** for the **audix** application and **2** for the **qsig-mwi** application

```
change communication-interface processor-channels              Page   1 of  24
                        PROCESSOR CHANNEL ASSIGNMENT
Proc                    Gtwy      Interface        Destination      Session   Mach
Chan Enable   Appl.      To   Mode Link/Chan       Node        Port  Local/Remote ID
  1:   y    audix            s    1   5002  intuity         0       1     1      1
  2:   y    qsig-mwi         s    1   6003  intuity         0       2     1      2
  3:   y    gateway          s    1   6001  intuity         0       3     1
  4:   n                                                    0
```

4.17.5. Administer MWI

Use command **change isdn mwi-prefixes** to configure support for MWI interrogation. In the **Machine ID** row matching the session local and mach ID assigned in **Section 4.17.4** for the **qsig-mwi** application, add an **AUDIX Mach ID. The AUDIX Mach ID** should match the session local and mach ID assigned in **Section 4.17.4** for the **audix** application

```
change isdn mwi-prefixes                                        Page   1 of   7
             MESSAGE WAITING INDICATION SUBSCRIBER NUMBER PREFIXES

 Send QSIG Message Center ID? n
 Machine Inserted      Routing  AUDIX
   ID    Digits        Digits   Mach ID
    1:
    2:                          1
    3:
```

4.17.6. Administer Analog stations for AUDIX

Analog stations must be configured on the Communication Manager to provide voice connectivity to the AUDIX system. These stations are then used in the hunt group configured to route calls to AUDIX voicemail. To add an analog station use the **add station n** command, where **n** is the extension number of the station to add. On **Page 1** of the station form set **Type** to **2500**, in the **Port** field enter the port and board location of the analog card that will be connected to AUDIX. Enter a descriptive **Name** for the station and define an appropriate **COR** and **COS.**

```
add station 7991                                                Page   1 of   4
                                     STATION

Extension: 7991                         Lock Messages? n              BCC: 0
     Type: 2500                         Security Code: 1234            TN: 1
     Port: 01A1101                    Coverage Path 1:                COR: 11
     Name: Audix Port 1               Coverage Path 2:                COS: 11
                                      Hunt-to Station:              Tests? y
STATION OPTIONS
      XOIP Endpoint type: auto            Time of Day Lock Table:
              Loss Group: 1          Message Waiting Indicator: led
    Off Premises Station? n                   Message Lamp Ext: 7991

          Survivable COR: internal
   Survivable Trunk Dest? y
```

On **Page 2** set **Switchhook Flash** and **Adjunct Supervision** to **y.** The **Multimedia Mode** is set to **basic** and the **AUDIX Name** should be set to the AUDIX node name defined in **Section 4.17.3**

```
add station 7991                                                Page   2 of   4
                                     STATION
FEATURE OPTIONS
           LWC Reception: AUDIX
          LWC Activation? y                          Coverage Msg Retrieval? y
  LWC Log External Calls? n                                    Auto Answer: none
             CDR Privacy? n                               Data Restriction? n
   Redirect Notification? y                         Call Waiting Indication: y
 Per Button Ring Control? n                    Att. Call Waiting Indication: y
   Bridged Call Alerting? n                      Distinctive Audible Alert? y
        Switchhook Flash? y                            Adjunct Supervision? y
    Ignore Rotary Digits? n
        H.320 Conversion? n         Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed
         Multimedia Mode: basic
    MWI Served User Type:
              AUDIX Name: intuity
                                                 Coverage After Forwarding? s
                                                   Multimedia Early Answer? n
                                               Direct IP-IP Audio Connections? y
  Emergency Location Ext: 7991                         IP Audio Hairpinning? n
     Precedence Call Waiting? y
```

Repeat these steps to configure all required analog stations that will be connected to the AUDIX system and used in the hunt group for AUDIX. In the sample configuration analog stations 7991, 7992, 7993 and 7994 were added.

4.18. Administer Hunt Group for AUDIX

Use the **add hunt-group n** command where **n** is the number of the hunt-group to add. Give the hunt group a descriptive name and a valid extension according to the dial plan. The **group type** must be **ucd-mia.** Set **ISDN/SIP Caller Display** to **grp-name.**

```
add hunt-group 79                                               Page   1 of  60
                                  HUNT GROUP

            Group Number: 79                            ACD? n
              Group Name: AUDIX                       Queue? n
         Group Extension: 7999                       Vector? n
              Group Type: ucd-mia             Coverage Path:
                      TN: 1         Night Service Destination:
                     COR: 1                  MM Early Answer? n
           Security Code:                Local Agent Preference? n
 ISDN/SIP Caller Display: grp-name
```

On **Page 2** set the **LWC Reception** and **Message Center** fields to **audix.** Set the **AUDIX Name** and **Message Center AUDIX Name** field to the AUDIX node name entered in **Section 4.17.3.** The **Calling Party Number to INTUITY AUDIX** field should be set to **y**

```
add hunt-group 79                                              Page   2 of  60
                                  HUNT GROUP

                      LWC Reception: audix          AUDIX Name: intuity

                     Message Center: audix

               Message Center AUDIX Name: intuity
                                  Primary? n
    Calling Party Number to INTUITY AUDIX? y
```

On **Page 3** enter the analog stations that are connected to the AUDIX system configured in **Section 4.17.6.**

```
add hunt-group 79                                              Page   3 of  60
                                  HUNT GROUP
         Group Number: 79   Group Extension: 7999         Group Type: ucd-mia
  Member Range Allowed: 1 - 1500       Administered Members (min/max): 1   /4
                                           Total Administered Members: 4
GROUP MEMBER ASSIGNMENTS
      Ext           Name(19 characters)       Ext           Name(19 characters)
   1: 7991                                14:
   2: 7992                                15:
   3: 7993                                16:
   4: 7994                                17:
```

4.19. Administer Coverage Path for AUDIX

Use command **change coverage path n** where **n** is the number of the coverage path to administer. Set **Point 1** to **h79** to send covered calls using this coverage path to hunt group 79.

```
change coverage path 79                                         Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 79
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA

    Station/Group Status    Inside Call     Outside Call
             Active?             n               n
               Busy?             y               y
       Don't Answer?             y               y        Number of Rings: 2
                All?             n               n
 DND/SAC/Goto Cover?             y               y
   Holiday Coverage?             n               n

COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h79            Rng:    Point2:
  Point3:                        Point4:
```

Use the **change station n** command to add the coverage path to a station where **n** is the extension number of the station to administer. Enter the coverage path number in the **Coverage Path 1** field.

```
change station 6622                                             Page   1 of   5
                                     STATION

Extension: 6622                         Lock Messages? n               BCC: 0
     Type: 9630                         Security Code: ****             TN: 1
     Port: S00003                     Coverage Path 1: 79              COR: 1
     Name: IP3rd                      Coverage Path 2:                 COS: 1
                                      Hunt-to Station:
```

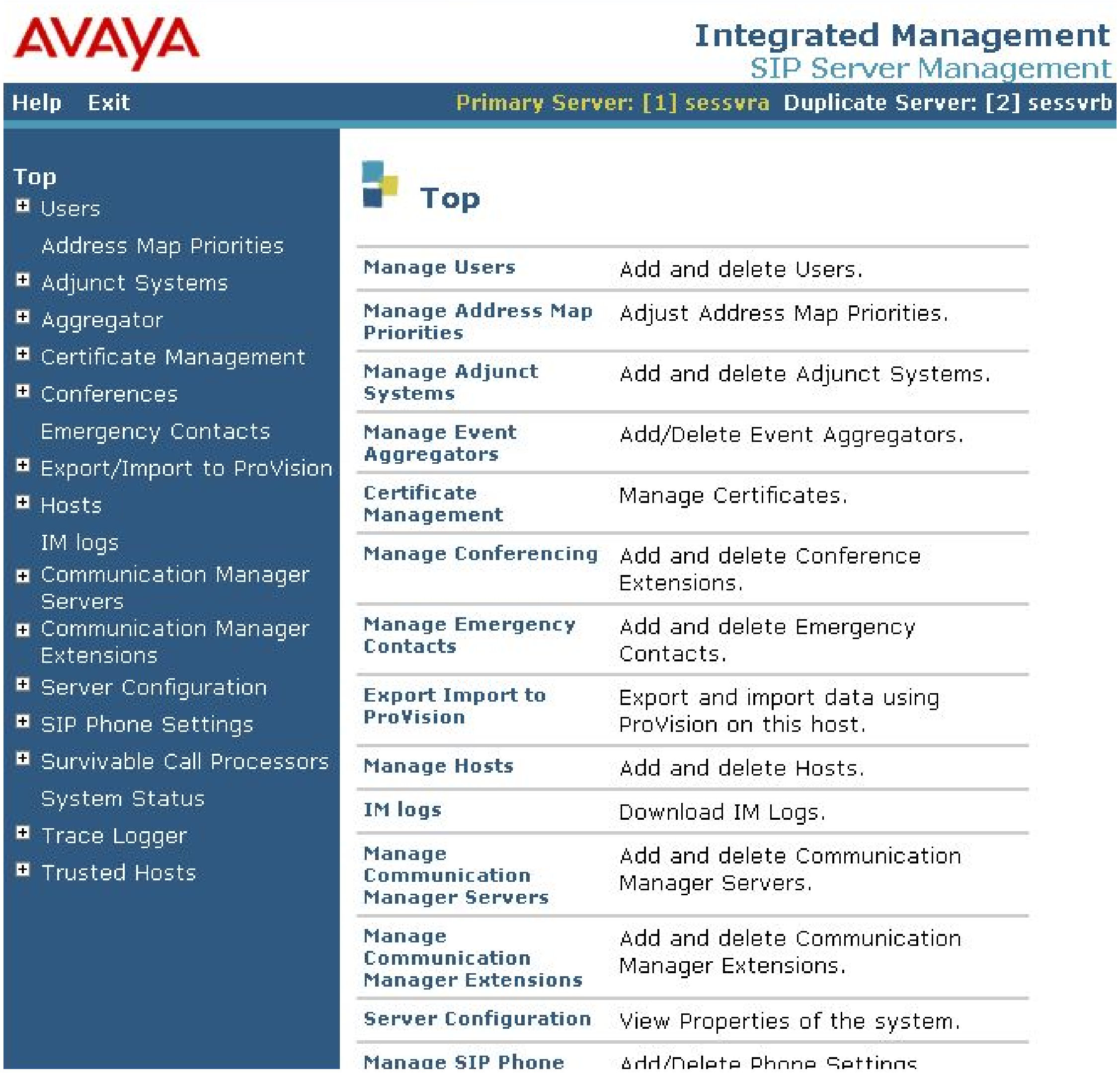
5. Configure Avaya Aura™ SIP Enablement Services

This section covers the administration of SES to support Modular Messaging. 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^{TM} SIP Enablement Services
- Verifying System Properties
- Administer Avaya Aura^{TM} SIP Enablement Services Host properties
- Add Avaya Modular Messaging as an Adjunct
- Add Avaya Aura^{TM} Communication Manager Server
- Avaya Aura^{TM} Communication Manager Address Maps

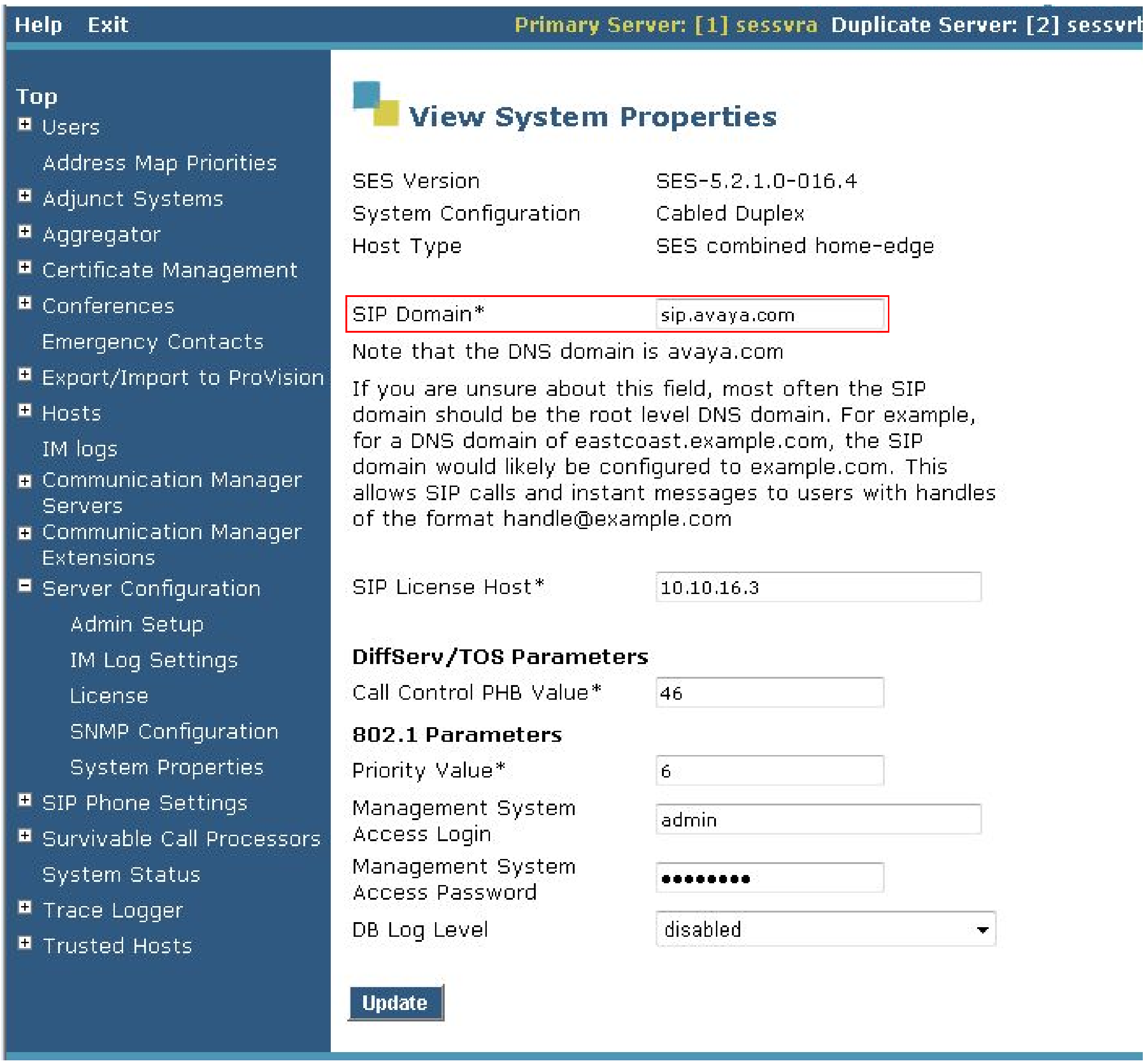
5.1. Logging onto Avaya Aura™ SIP Enablement Servcies

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.



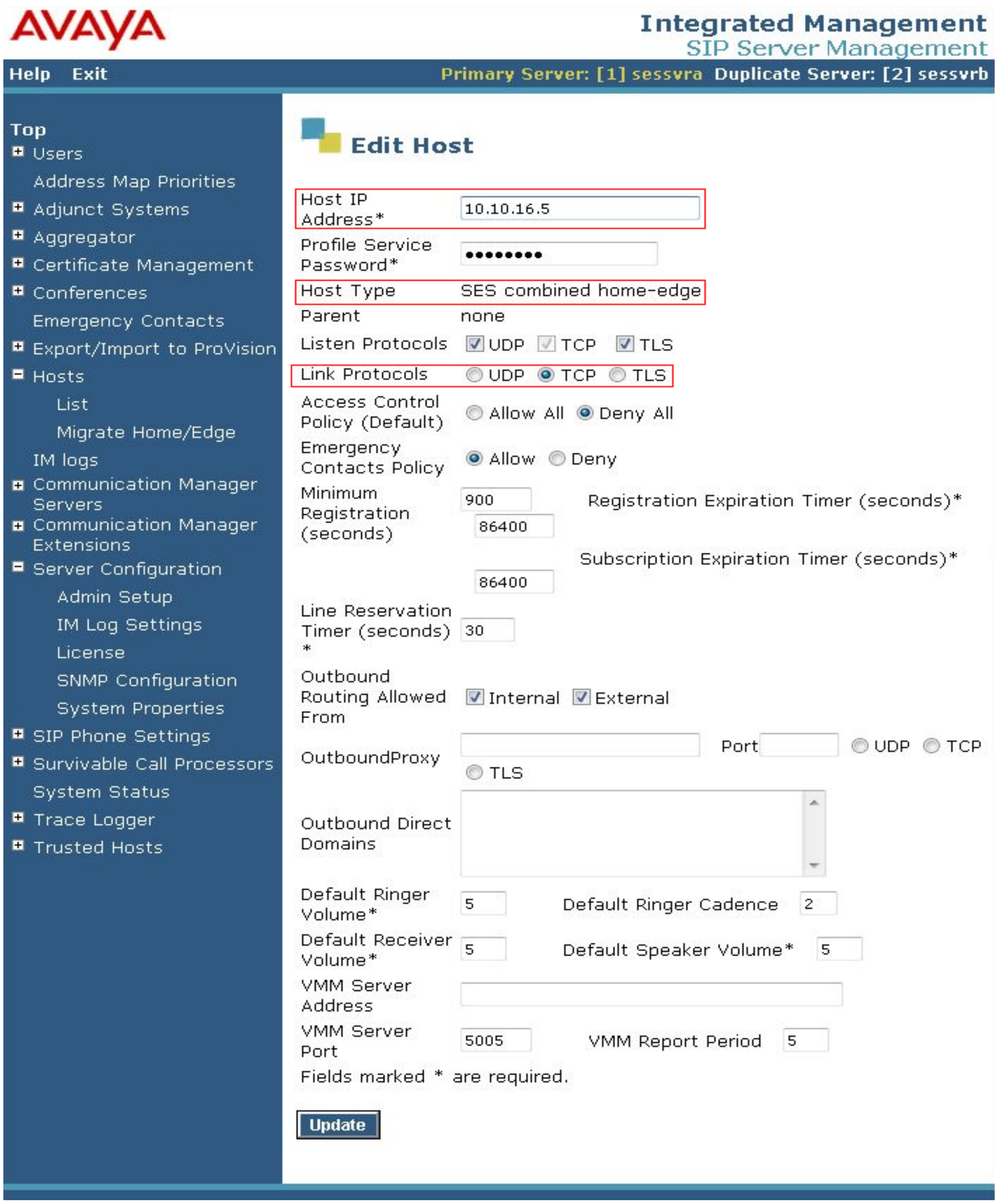
5.2. Verifying System Properties

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 network region and the SIP signaling group to SES.



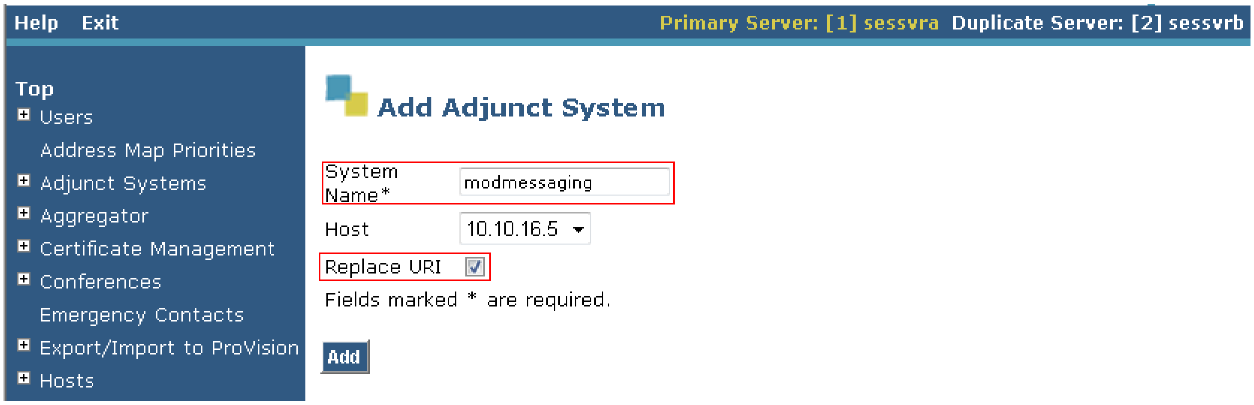
5.3. Administer Avaya Aura™ SIP Enablement Services Host Properties

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 the example configuration **TCP** was used. The default values for the other fields may be used as shown below.

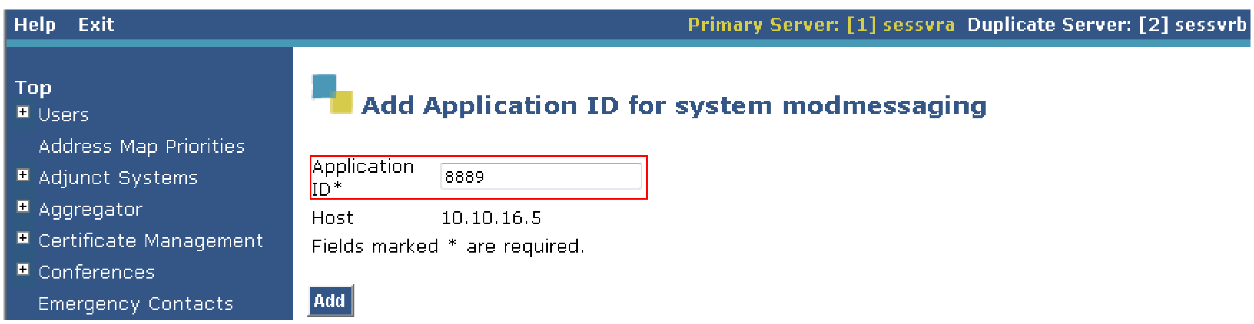


5.4. Add Avaya Modular Messaging as an Adjunct

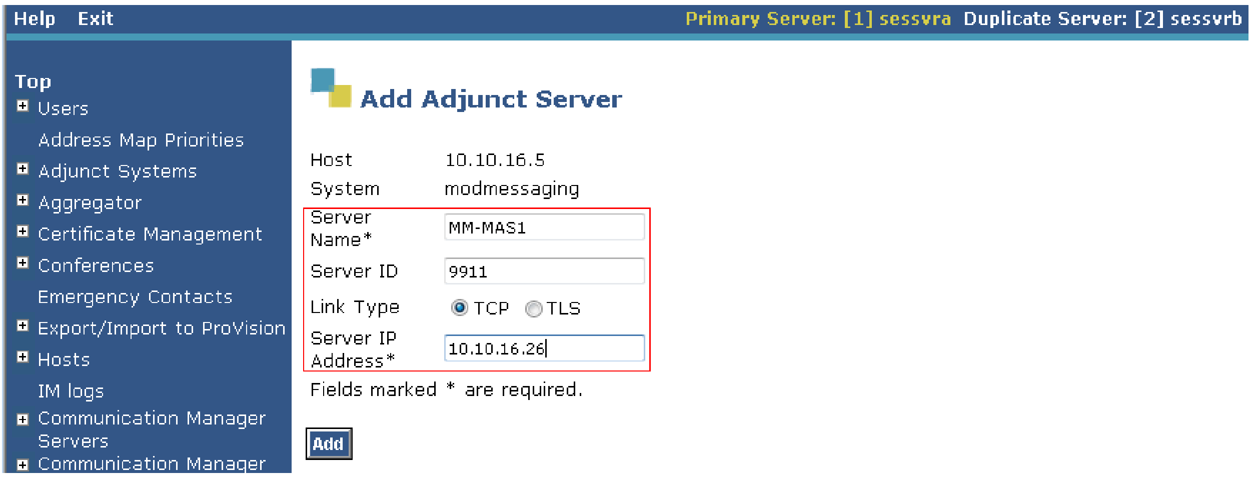
Under the **Adjunct Systems** option in the Administration web interface, select **Add**. In the resulting screen enter the voice mail handle specified on the Modular Messaging hunt group in **Section 4.15**. Select the **Replace URI** check box and click **Add**



In the resulting screen click on **List Application ID** then **Add an Application ID** (not shown). In the **Add Application ID for system modmessaging** screen, enter the Modular Messaging pilot number in the **Application ID** field and click **Add.**

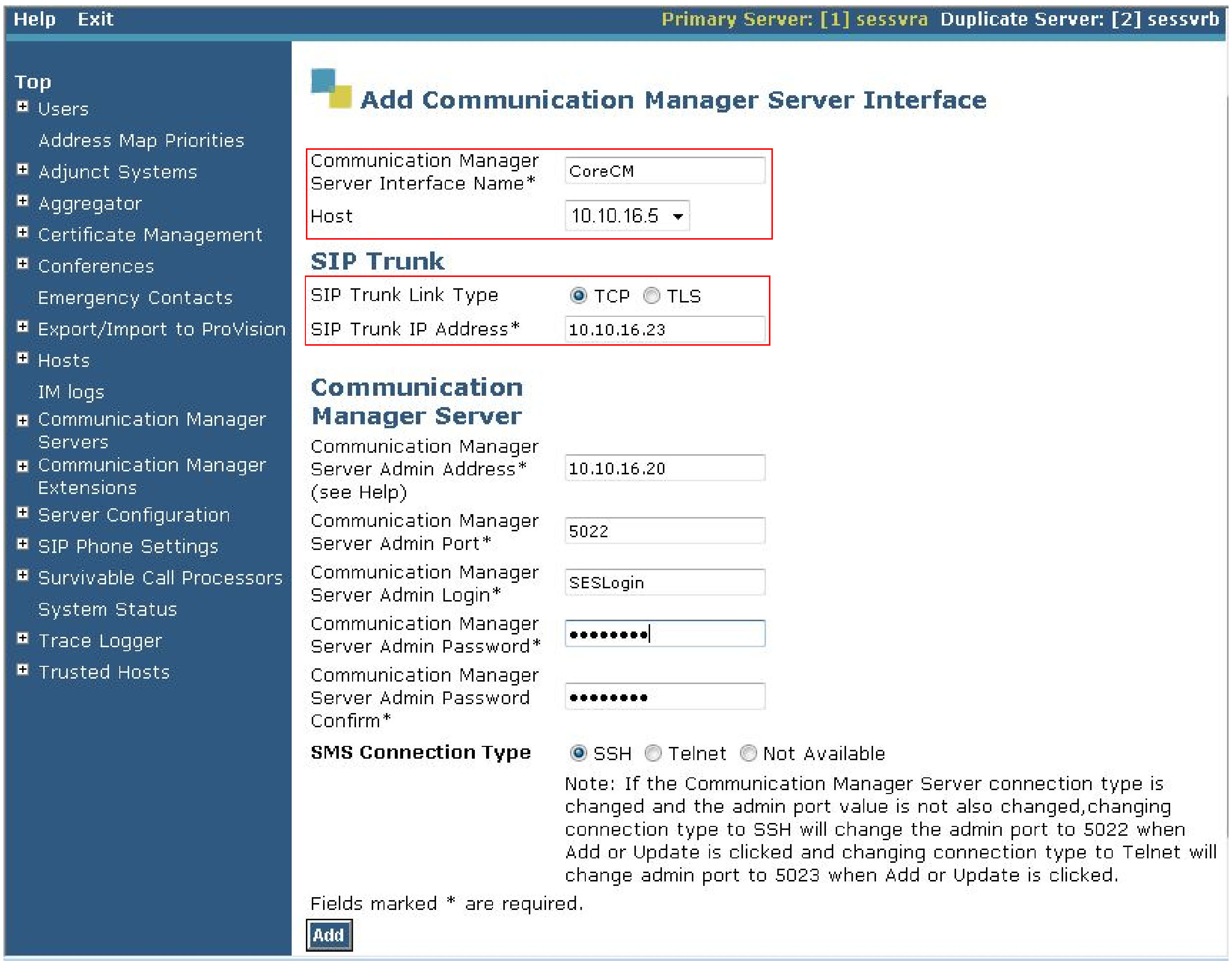


Under the **Adjunct Systems** option in the Administration web interface, select **List.** Click the **List Adjunct Servers** link and then the **add another adjunct server to system modmessaging** link (not shown). In the resulting screen enter a descriptive name for **Server Name**. Enter an extension that is not used on the Communication Manager in the **Server ID** field. Select the **link type** used to Modular Messaging. For transparency, in the sample configuration **TCP** is used however it is recommended that TLS is used in production. The **server IP Address** field should be set to the IP address of the Message Application Server. Click **Add.**



5.5. Add Avaya Aura™ Communication Manager Server

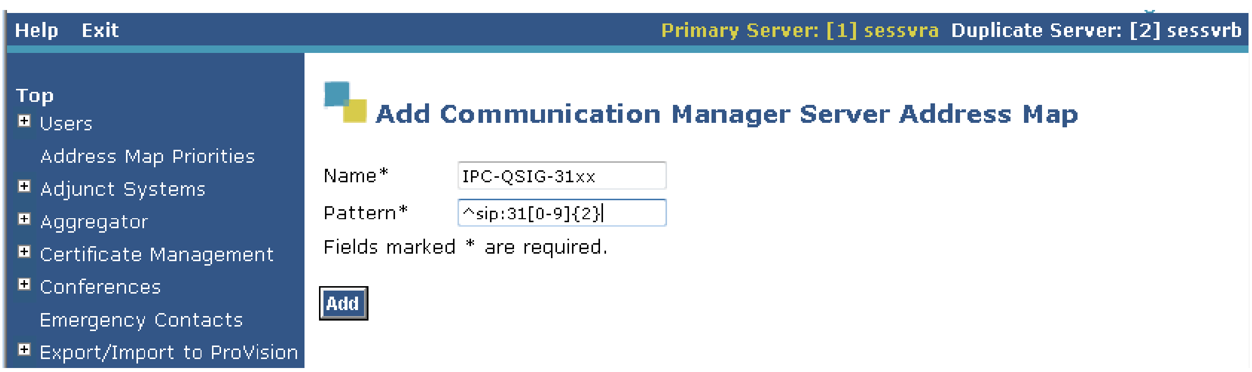
Under the **Communication Manager Servers** option in the Administration web interface, select **Add** to add the Avaya Media Server since a SIP trunk is required between Communication Manager and SES. In this screen, enter a descriptive name in the **Communication Manager Server Interface Name** field and select the home server from the drop down menu in the **Host** field. Select **TCP** for the **SIP Trunk 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**.



5.6. Avaya Aura™ Communication Manager Address Maps

As Modular Messaging is connected via SIP it will send SIP Notify messages to the SES to enable and disable the message waiting indicator on the subscriber endpoints. The SES must be able to route the Notify messages to the appropriate location, so an address map must be configured for each extension range that will subscribe to Modular Messaging, this includes the extensions used by IPC as they are connected via QSIG to the Communication Manager. To configure a **Communication Manager Server Address Map** select **Communication Manager Servers** in the left pane of the Administration web interface. Click **List Communication Manager Servers** and click on the **Map** link associated with the appropriate server. Click on the **Add Map In New Group** link. In the resulting screen**:**

- Enter a descriptive name in the **Name** field
- Enter the regular expression to be used for the pattern matching in the **Pattern** field. In this configuration, the IPC extension range is 31xx. A pattern specification for these IPC extensions is: **^sip:31[0-9]{2}**. URIs beginning with **sip:31** followed by 2 digits from 0 to 9.
- Click the **Add** button once the form is completed



After adding the address map, the **List Communication Manager Server Address Map** screen will appear. When the **Communication Manager Server Address Map** is added, a **Contact** is created automatically. The following contact was created:

sip:$(user)@10.10.16.23:5060;transport=tcp

The contact specifies the IP address of the Communication manager CLAN interface and the transport protocol used to send SIP signaling messages. The IPC extension sent in the user part of the original request URI is substituted for $(user)

The above steps should be repeated for each extension range that will subscribe to Modular Messaging.

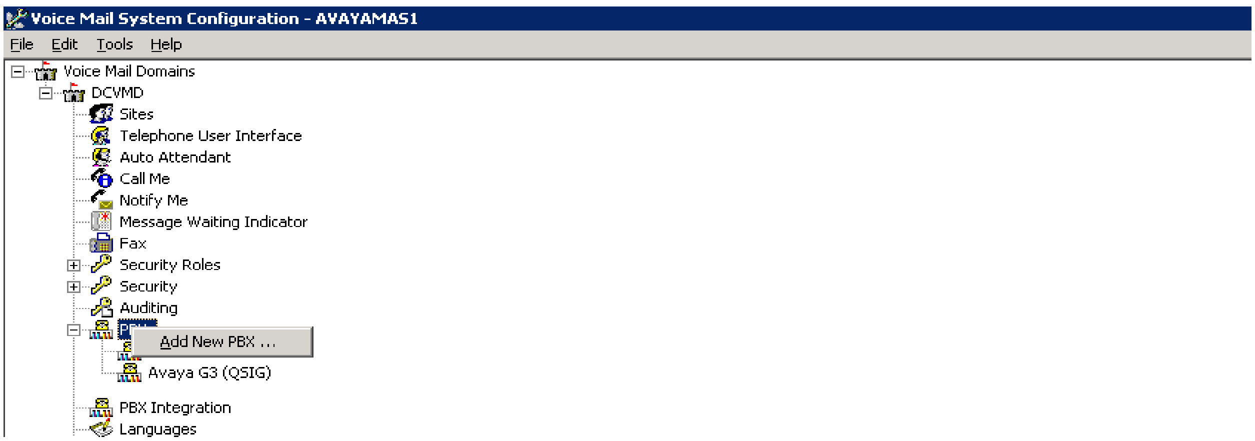
6. Configure Avaya Modular Messaging

This section provides the procedures for configuring Modular Messaging. The procedures include the following areas:

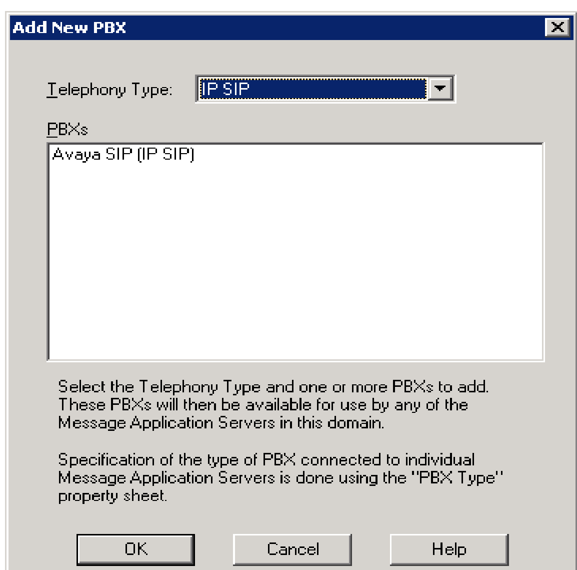
- Configure Avaya Message Application Server
- Configure Avaya Message Storage Server

6.1. Configure Avaya Message Application Server

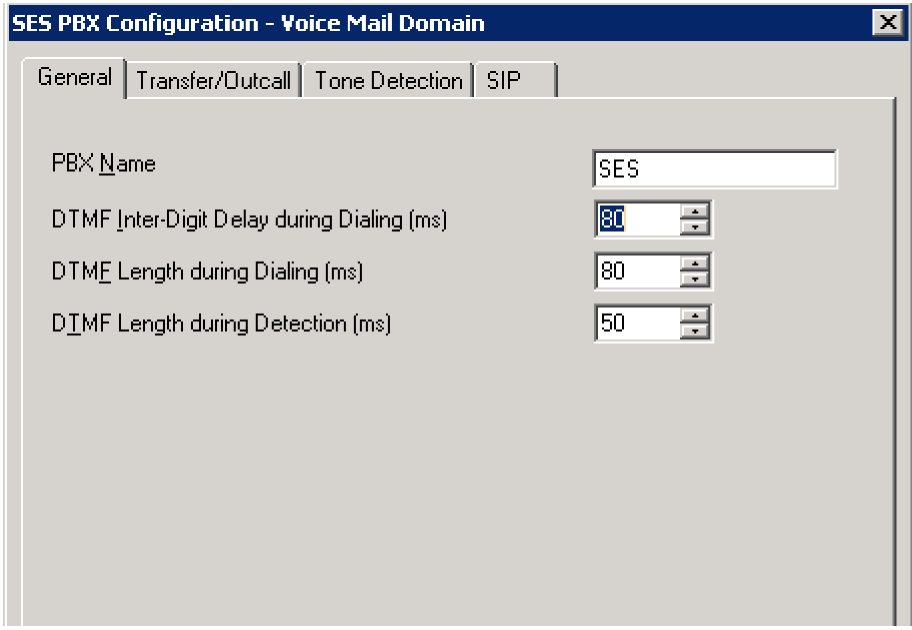
Select **Start** → **Programs** → **Avaya Modular Messaging** →**Voice Mail System Configuration – AVAYAMAS1**. Expand **Voice Mail Domains** and the administered domain name (**DCVMD** in the screenshot below). Right-click on **PBXs** and select **Add New PBX …**



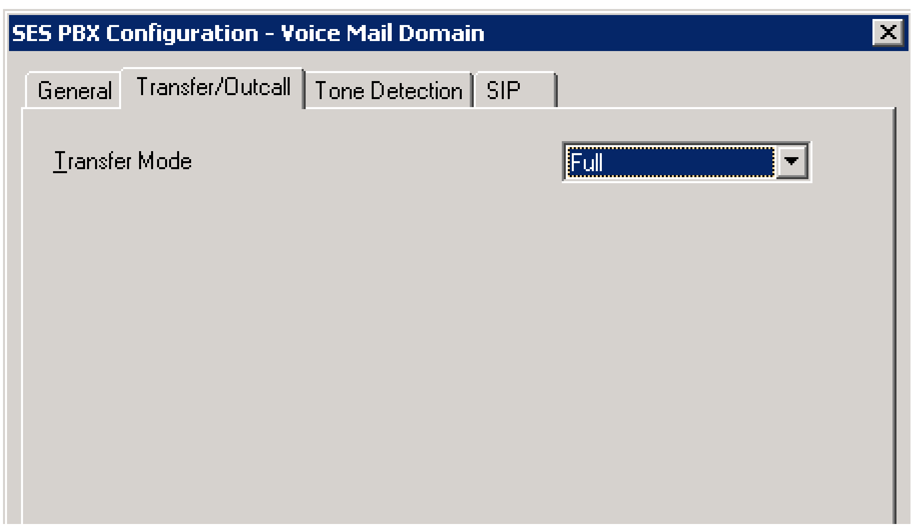
On the **Add New PBX** screen, select **IP SIP** from the **Telephony Type** drop down box, then select **Avaya SIP (IP SIP)** from the **PBXs** box. Select **OK** when completed.



On the **Voice Mail System Configuration – AVAYAMAS1** screen double-click on **PBXs**. On the **SES PBX Configuration** screen enter a descriptive **PBX Name**

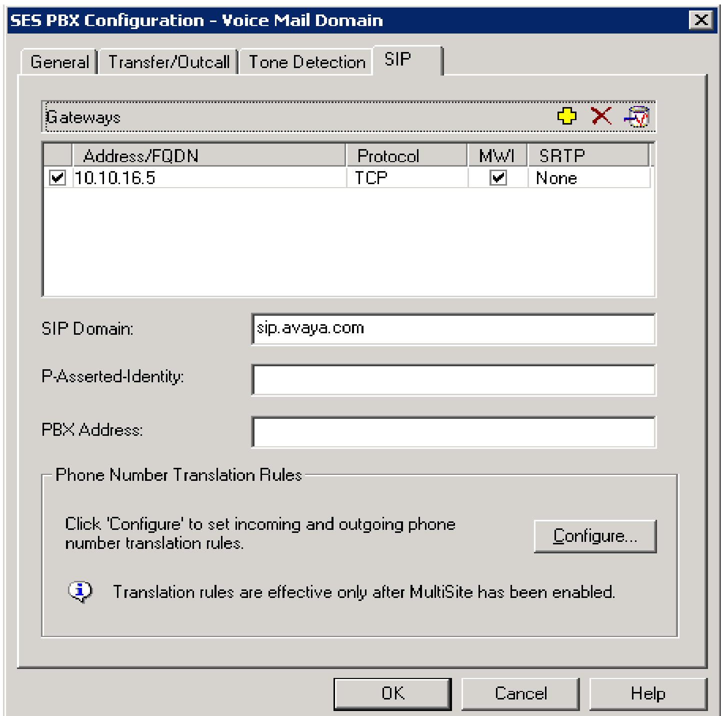


Select the **Transfer/Outcall** tab, in the **Transfer Mode** field select **Full** from the drop down menu.

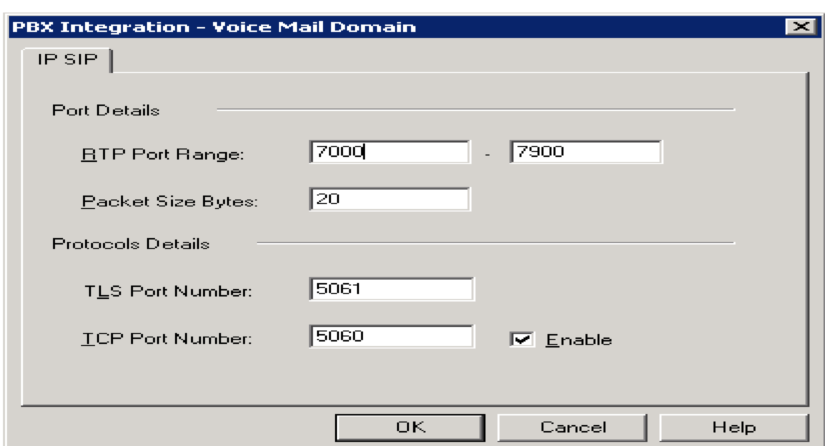


Select the **SIP** tab and enter the following fields.

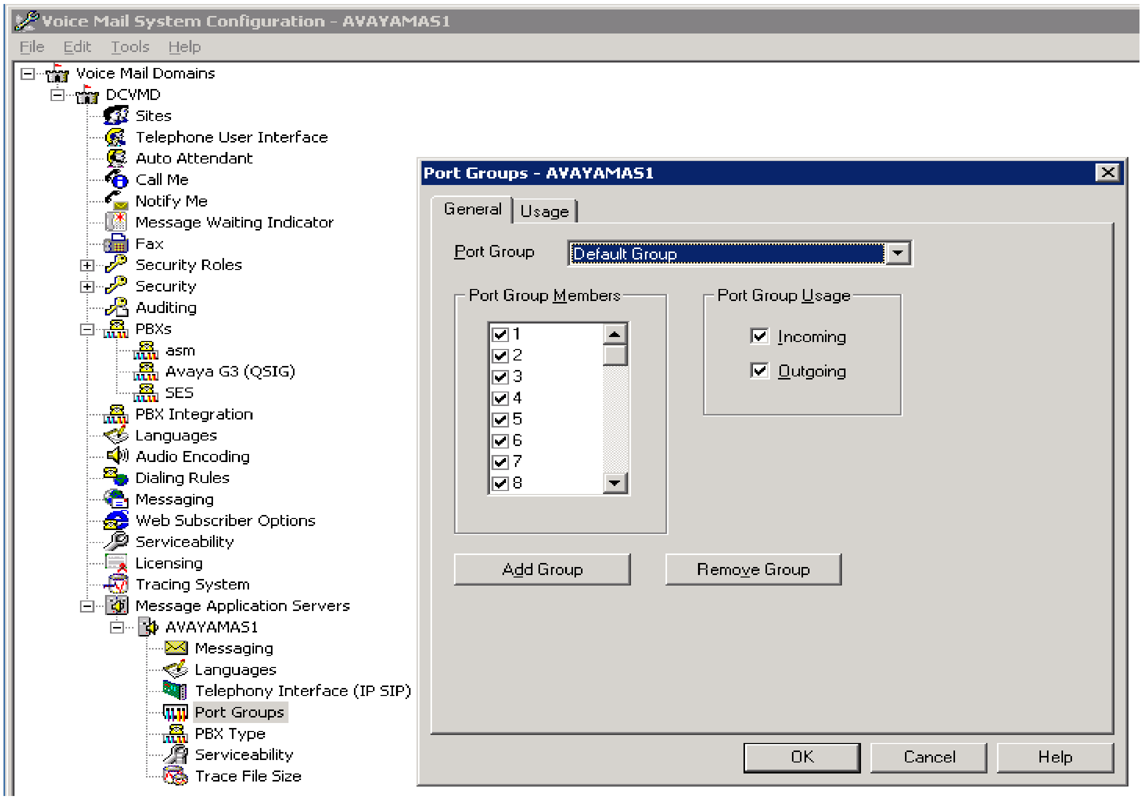
- In the **Address/FQDN** field enter the active IP address of the SES
- In the **Protocol** field select the protocol Modular Messaging will use for communication to the SES
- Select the **MWI** check box
- In the **SIP Domain** field enter the sip domain that is being used by SES and that Modular Messaging will become part of.
- Click **OK** when completed



On the **Voice Mail System Configuration – AVAYAMAS1** screen, double-click on **PBX Integration**. Confirm the default settings below and check the **Enable** check box if TCP is to be used. Click **OK** when completed.

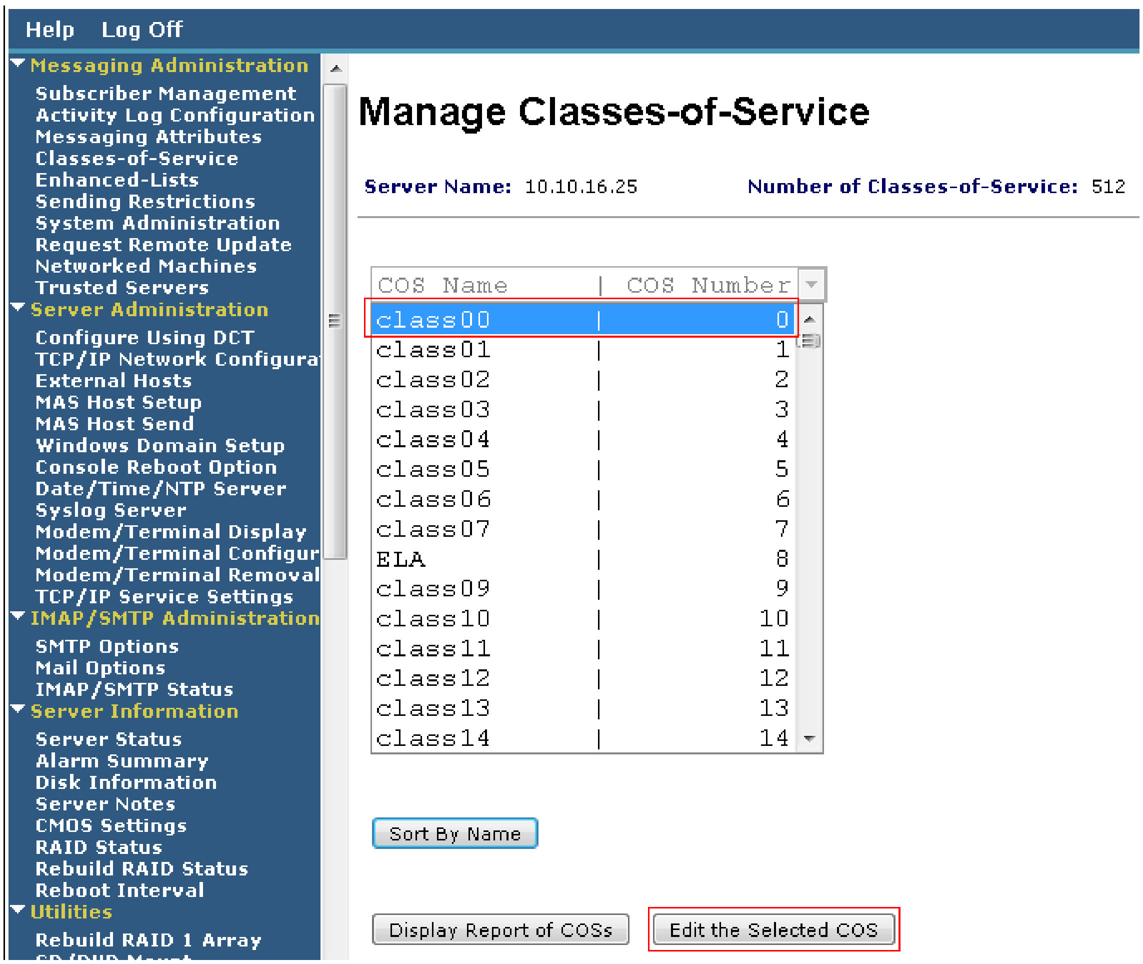


On the **Voice Mail System Configuration – AVAYAMAS1** screen, expand **Message Application Servers** and expand the appropriate MAS server. Double click **Port Groups** and confirm all the **Port Group Members** and both the **Incoming** and **Outgoing** check boxes are selected.

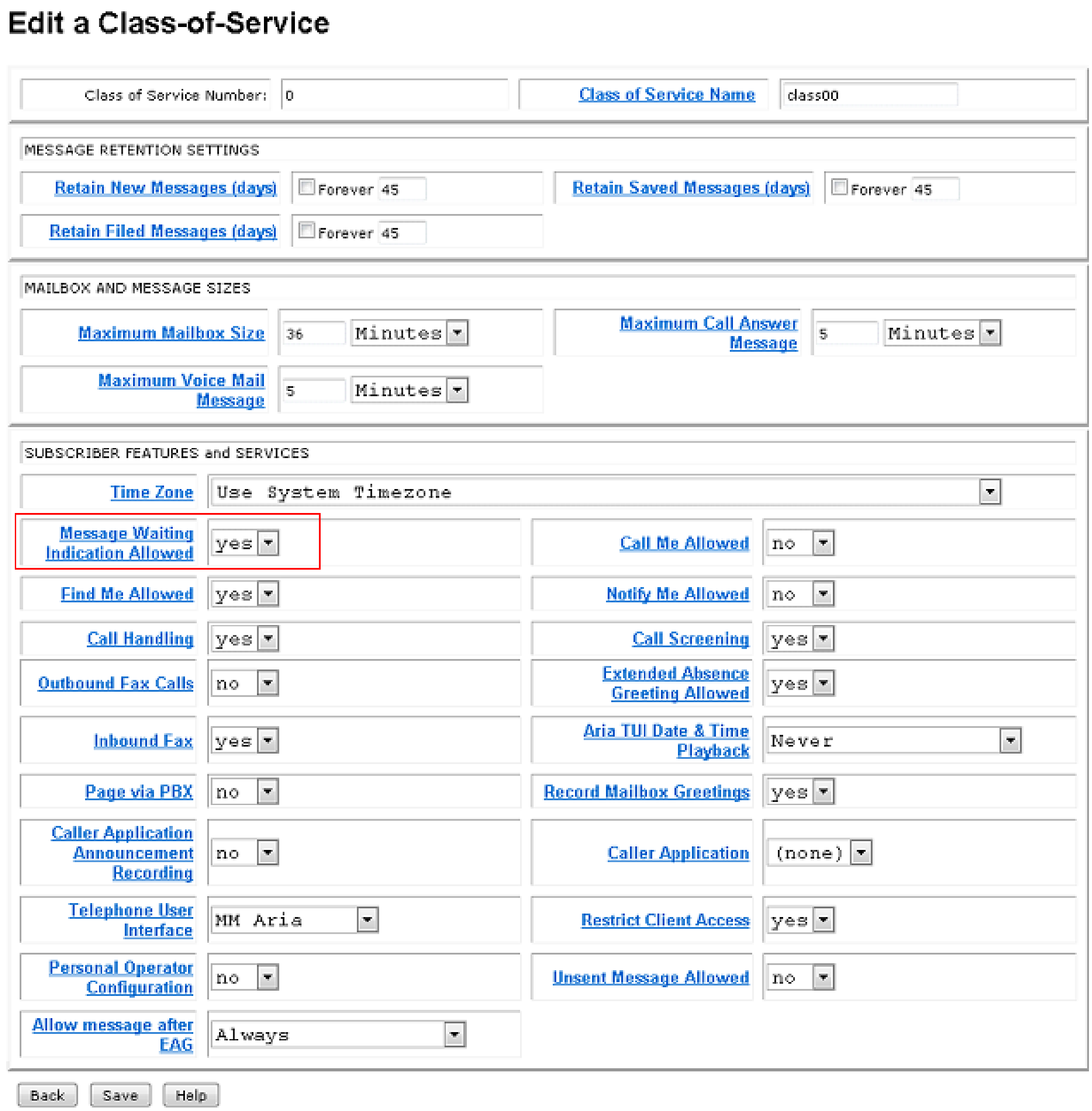


6.2. Configure Avaya Message Storage Server

From a Web browser, navigate to **http://<ip-addr>** where **<ip-addr>** is the IP address of the Avaya MSS. After logging in with an appropriate login and password, the main page appears. (not shown). Select **Messaging Administration → Classes-of-Service** from the left panel. From the **Manage Classes-of-Service** screen that is presented, select a Class of Service (COS) that will be used by subscribers using IPC turrets (in this example **class00** is selected). Click **Edit the Selected COS** button.



In the **Edit a Class-of-Service** screen that follows, select **yes** from the drop-down menu for the **Message Waiting Indication Allowed** field. Scroll down to the bottom of the screen and click the **Save** button.

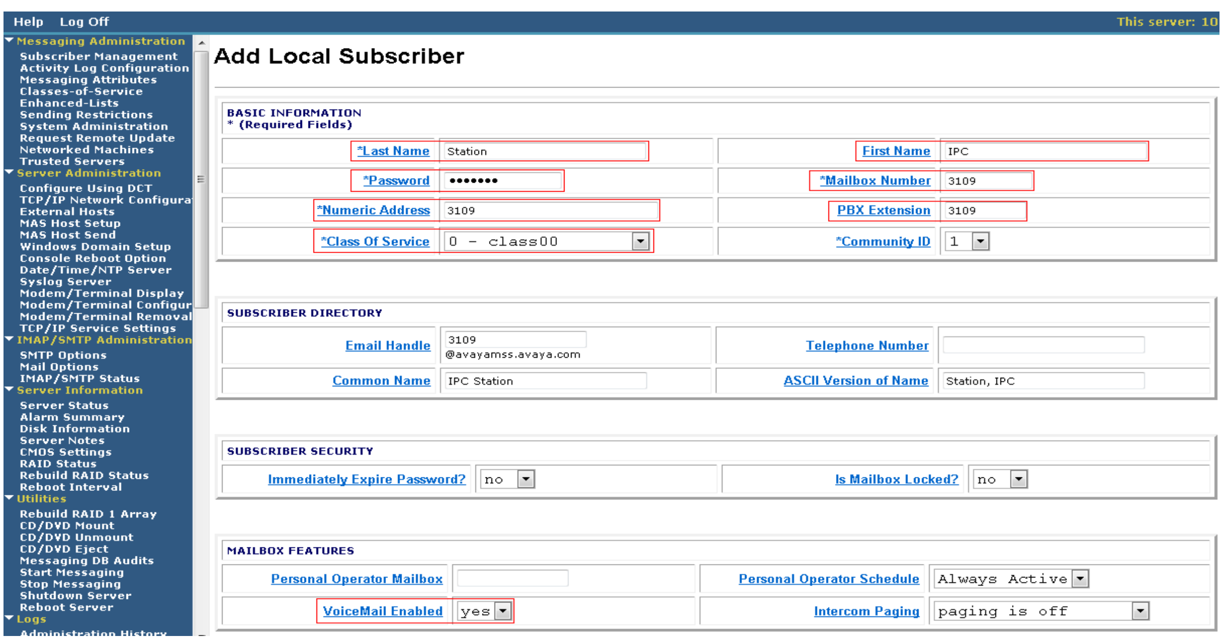


Select **Messaging Administration → Subscriber Management** in the left pane. The **Manage Subscribers** page appears, as shown below. In the **Local Subscriber Mailbox Number** field, enter the extension of the desired IPC turret and click the **Add or Edit** button.

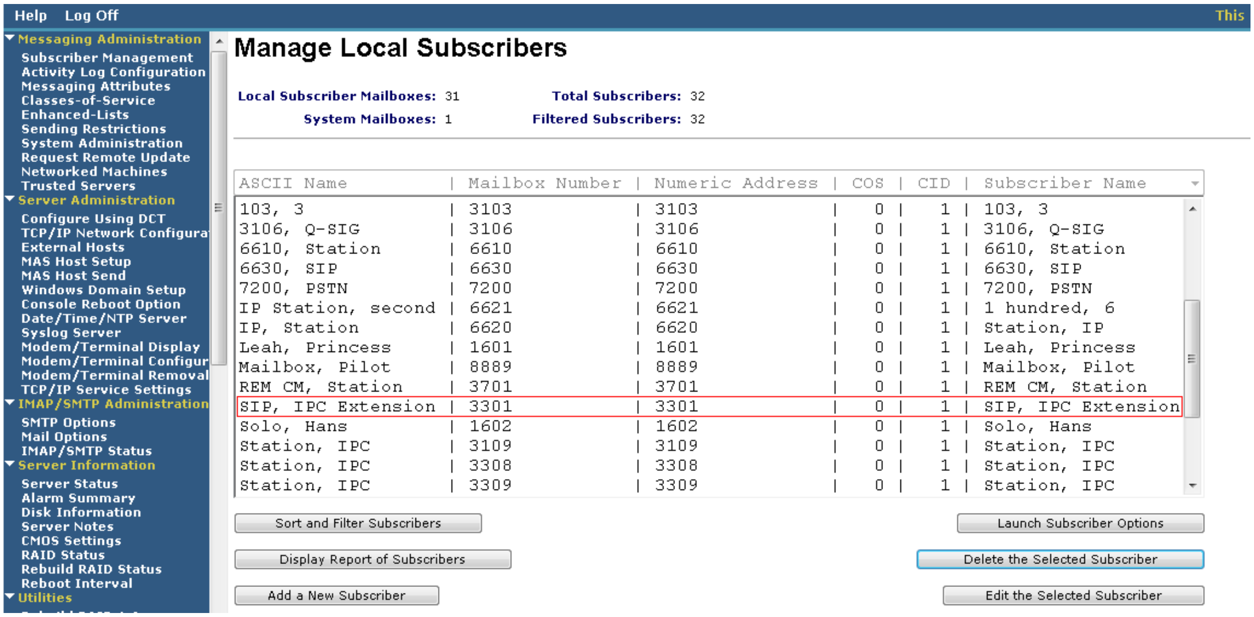
In the **Add Local Subscriber** screen, fill in the required fields. In this example, IPC extension 3109 is used:

- For **Last Name** and **First Name** fields enter values appropriate for the user
- For **Password** enter a default password for accessing the subscriber's mailbox, from one to 15 digits
- For **Mailbox Number** enter the subscribers extension number
- For **Numeric Address** re-enter the subscribers extension number
- For **Class of Service** select the Class of Service
- Verify **VoiceMail Enabled** is set to **yes**

Repeat this step for all IPC extensions.



To verify that mailboxes have been created, select **Messaging Administration → Subscriber Management**, click the **Manage** button to the right of the **Local Subscribers** entry. In the resulting **Manage Local Subscribers** screen that is presented (see below), verify that the mailboxes created appear in the list of subscribers.



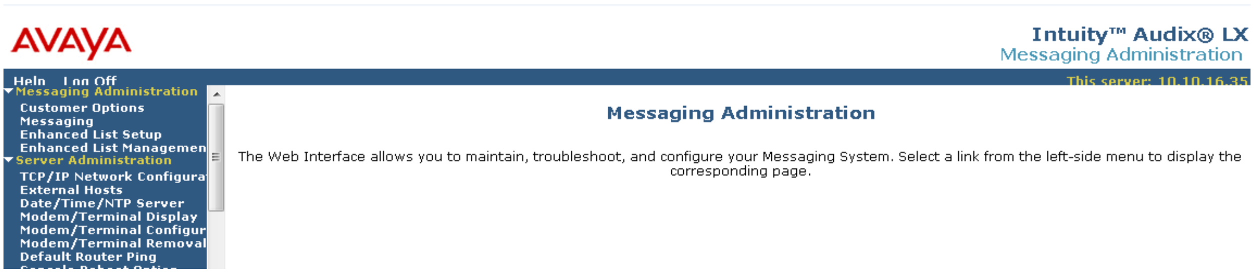
7. AUDIX configuration

This section provides the procedures for configuring Intuity AUDIX LX. It is assumed that the basic installation of AUDIX has already been completed. The procedures covered in this section include the following:

- Logging onto AUDIX
- Configure AUDIX System Links
- Add AUDIX Subscribers
- Configure AUDIX COS

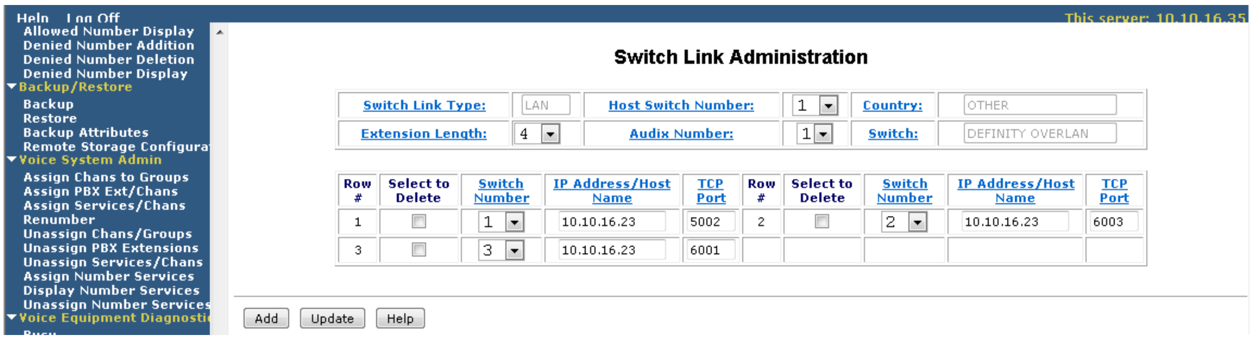
7.1. Logging onto AUDIX

From a Web browser, navigate to **http://<ip-addr>** where **<ip-addr>** is the IP address of the AUDIX. After logging in with an appropriate login and password, the main page appears.



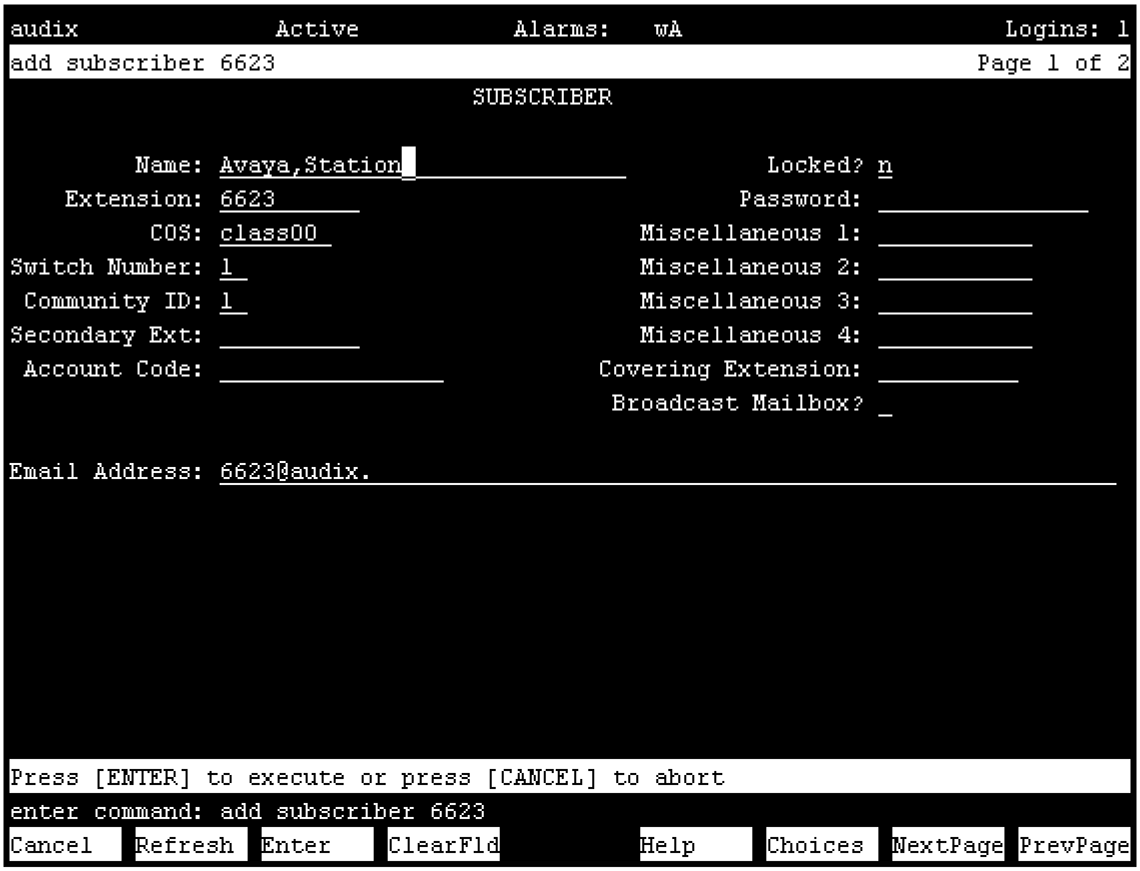
7.2. Configure AUDIX System Links

Select **Switch Administration → Switch Link Administration to** ensure that the **Extension Length** field is set to the correct length**.** Three switch links will need to be added, one for each communication processor channel application configured in **Section 4.17.4.** Click **Add** three times to add three switch link rows. The Switch number selected should match the session local field configured in **Section 4.17.4**. In each case the **IP Address/Host Name** fields should be set as the IP address of the connecting CLAN. The **TCP Port** configured should match the port assigned to the interface channel in **Section 4.17.4.**



7.3. Add Subscribers

From the administration web interface navigate to **Messaging Administration → Messaging**. In the resulting emulation window enter the appropriate login credentials and run the command **add subscriber n.** Enter a descriptive **Name a**nd assign the appropriate **COS (Class of Service).** The **Switch Number** used for an Avaya subscriber is **1.**



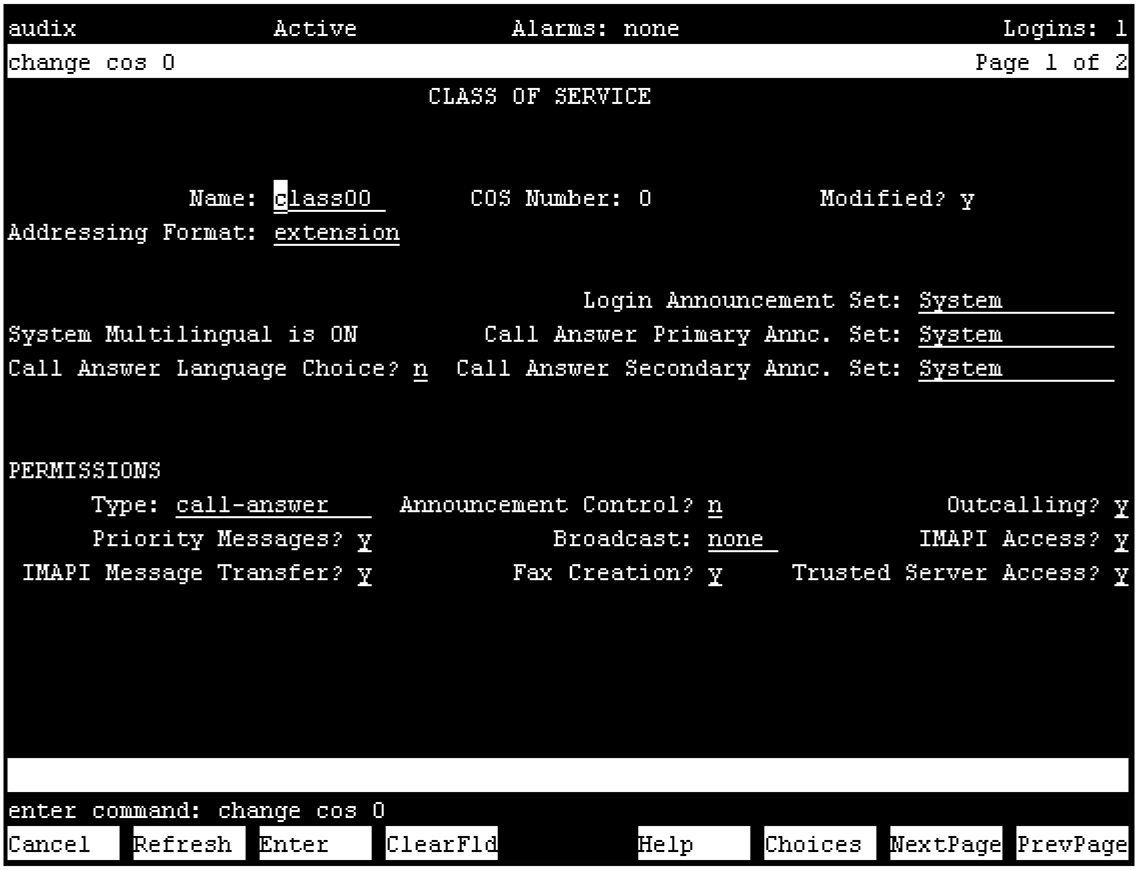
```
audix              Active           Alarms:   wA                    Logins: 1
add subscriber 6623                                                Page 1 of 2
                                 SUBSCRIBER

         Name: Avaya,Station                           Locked? n
    Extension: 6623                                  Password:
          COS: class00                        Miscellaneous 1:
Switch Number: 1                              Miscellaneous 2:
 Community ID: 1                              Miscellaneous 3:
Secondary Ext:                                Miscellaneous 4:
 Account Code:                             Covering Extension:
                                            Broadcast Mailbox? _

Email Address: 6623@audix.

Press [ENTER] to execute or press [CANCEL] to abort
enter command: add subscriber 6623
Cancel   Refresh  Enter    ClearFld           Help     Choices  NextPage PrevPage
```

To add a subscriber for an IPC user repeat the previous step. The **Switch Number** used for IPC subscribers is **2**

```
audix                  Active          Alarms:   wA                     Logins: 1
add subscriber 3109                                                   Page 1 of 2
                                    SUBSCRIBER

         Name: IPC,Station____________________          Locked? n
    Extension: 3109______                              Password: _______________
          COS: class00_                         Miscellaneous 1: ___________
Switch Number: 2_                               Miscellaneous 2: ___________
 Community ID: 1_                               Miscellaneous 3: ___________
Secondary Ext: __________                       Miscellaneous 4: ___________
 Account Code: ________________              Covering Extension: __________
                                              Broadcast Mailbox? _

Email Address: 3109@audix.________________________________________________________

Press [ENTER] to execute or press [CANCEL] to abort
enter command: add subscriber 3109
Cancel   Refresh  Enter    ClearFld           Help     Choices  NextPage PrevPage
```

7.4. Administer Class of Service

Run the command **change cos n** where **n** in the number of the class of service assigned to the previously added subscribers**.** Set **Outcalling** to **y.** all other fields can remain as default.


```
audix              Active           Alarms: none                Logins: 1
change cos 0                                                  Page 1 of 2
                              CLASS OF SERVICE


             Name: class00       COS Number: 0            Modified? y
Addressing Format: extension

                                          Login Announcement Set: System
System Multilingual is ON        Call Answer Primary Annc. Set: System
Call Answer Language Choice? n  Call Answer Secondary Annc. Set: System


PERMISSIONS
      Type: call-answer     Announcement Control? n            Outcalling? y
      Priority Messages? y           Broadcast: none         IMAPI Access? y
 IMAPI Message Transfer? y        Fax Creation? y     Trusted Server Access? y







enter command: change cos 0
Cancel   Refresh  Enter    ClearFld          Help     Choices  NextPage PrevPage
```

8. General Test Approach and Test Results

A simulated enterprise site using an Avaya IP telephony solution was connected to IPC via an E1-QSIG connection provisioned between Communication Manager and IPC's Alliance MX. The compliance test included the following:

- Incoming calls to the Avaya telephones, calls were made from IPC turrets to Avaya H.323, SIP, digital and analog telephones.
- Outgoing calls from the Avaya telephones, calls were made from Avaya H.323, SIP, digital and analog telephones to IPC turrets
- DTMF transmission with successful Voice Mail navigation
- User features such as hold and resume, transfer, conference, call forwarding, etc.
- Caller ID Presentation and Caller ID Restriction.
- Voicemail coverage and retrieval for endpoints at the enterprise sites.

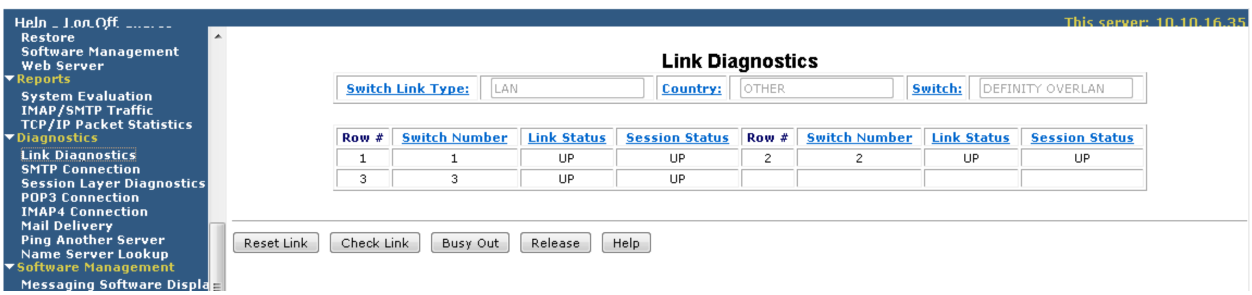
Testing of the sample configuration was completed with successful results for the IPC QSIG architecture.

9. Verification Steps

The following steps can be used to verify that the required configuration has been correctly administered to support IPC QSIG architecture. To verify that any of the trunk groups are up, from the Communication Manager SAT use the **status trunk n** command, where **n** is the number of the trunk group. (Refer to **Sections 4.5.3** and **4.9.3** for trunk details). Verify for each trunk, that the **Service State** shows **in-service/idle**.

```
                              TRUNK GROUP STATUS

Member    Port      Service State       Mtce Connected Ports
                                        Busy

0003/001  01A0601   in-service/idle     no
0003/002  01A0602   in-service/idle     no
0003/003  01A0603   in-service/idle     no
```

To ensure that the links to AUDIX are up and in service, from the administration web interface navigate to **Diagnostics → Link Diagnostics** and confirm that both **Session Status** and **Link Status** for each link configured shows as **UP**



To verify end-to-end connectivity and configuration, set one of the IPC turrets to call forward to voicemail, then call the IPC turret from an Avaya Communication Manager station. The call should route to the mailbox of the IPC turret. Leave a message and verify the MWI is activated. This test should be done using both AUDIX and Modular Messaging as the voicemail.

10. Conclusion

These Application Notes describe the steps required to configure the Avaya components to successfully interoperate with IPC QSIG architecture using E1-QSIG as the transport method between the Avaya and IPC environments. The configured and verified Avaya components include Avaya Aura^{TM} Communication Manager Avaya Aura^{TM} SIP Enablement Services, Avaya Modular Messaging, and Intuity™ AUDIX® LX

11. 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] *Avaya Aura™ Communication Manager Special Application Features, 10-Nov-2009*

[5] *Modular Messaging Admin Guide Release 5.2 with Avaya MSS, 29-Nov-2009*

[6] *INTUITY AUDIX LX Release 2.0 Documentation CD, 08-May-2007*

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.