Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Communication Manager, Avaya Modular Messaging and Avaya SIP Enablement Services to Support System Interconnect for IPC – Issue 1.0

Abstract

These Application Notes describe how to configure Avaya Communication Manager, Avaya Modular Messaging and Avaya SIP Enablement Services to support System Interconnect for IPC.

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

1. Introduction

These Application Notes describe how to configure Avaya Communication Manager, Avaya Modular Messaging (MM) and Avaya SIP Enablement Services (SES) to support System Interconnect for IPC.

The System Interconnect is a SIP solution, which consists of the following components:

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

The Alliance MX is a voice technology product designed to provide a high resiliency platform for provision of telephony and other associated services to financial traders. The Alliance MX provides its users with connectivity to various telephone transport services. IPC ESS (Enterprise SIP Server) is a SIP proxy server, IPC System Center is a UNIX based Sun workstation, which is the administration terminal for the Alliance MX. IPC turrets are SIP-based VoIP trading phones.

These Application Notes describe detail configuration steps for

- Configure Avaya Communication Manager (**Section 4**)
  - Configure SIP Trunk
  - Configure Call Routing
- Configure Avaya SIP Enablement Services (**Section 5**)
- Configure Avaya Modular Messaging (**Section 6**)

There is no IPC configuration documented in this application notes based on their support policy. IPC engineers will be responsible to install and for the maintenance of Alliance MX products.

1.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The feature testing focused on verifying the Alliance MX's ability to request and respond to Avaya Communication Manager, Avaya Modular Messaging features through SIP.
The serviceability testing focused on verifying the IPC ESS (Enterprise SIP Server) and Avaya SIP Enablement Services' redundancy/failover ability to recover from an outage condition.
In addition to simplistic routed voice communication (basic call), the interoperability compliance testing covers the following supplementary services and features:

- Provision and display of both calling and connected party name and number.
- Restriction of provision and display of both calling and connected party name and number.
- Hold
- Call transfer, with informational phases.
- Call forward (busy, unconditional and no reply), with informational phases, by either forward switch methodology.
- Message Waiting Indicator (MWI).

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 configuration that was used to verify these Application Notes.

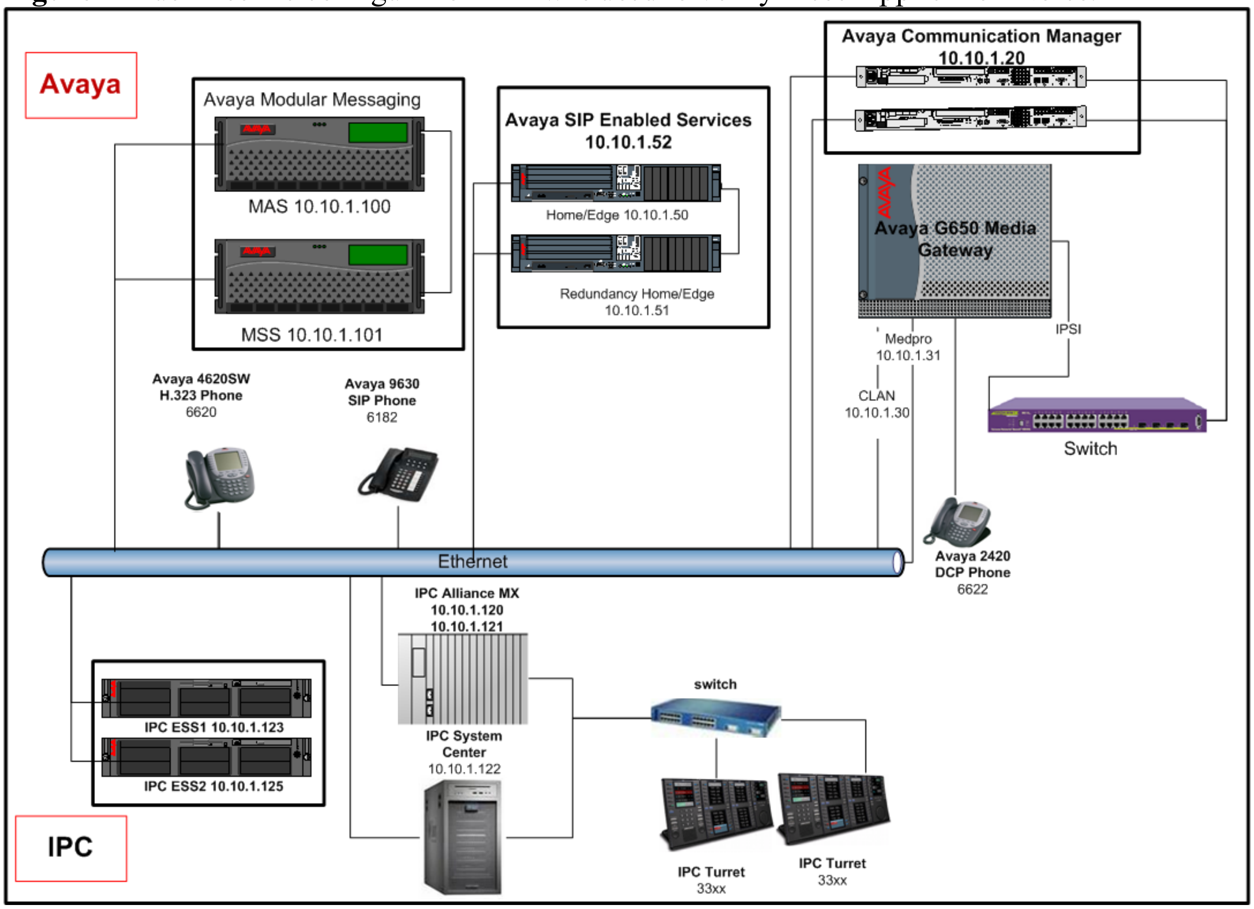


Figure 1: Network Diagram of the Compliance Tested Configuration

3. Equipment and Software Validated

The following hardware and software versions were used for this configuration is outlined in **Table 1** below.

Equipment	Software
Avaya S8730 Servers (2)	Avaya Communication Manager 5.1.1 sp3 Build 415.1-17105
Avaya G650 Media Gateway : • IPSI (TN2312AP) • C-LAN (TN799DP) • MEDPRO (TN2302AP) • Digital Line (TN2214CP)	 HW02 FW044 HW01 FW026 HW11 FW118 HW05 FW015
Avaya S8510 Server	Avaya SIP Enablement Services 5.1.2 Build SES-01.2.416.4b-sp0
Avaya Modular Messaging Servers (S3500) : • Messaging Application Server (MAS) • Messaging Storage Server (MSS)	Avaya Modular Messaging 4.0 sp3 Build 7.2.767.3001
Avaya 4620SW IP Telephones	2.9 (H.323)
Avaya 2420 Digital Telephones	---
Avaya 9630 IP Telephones	2.0.5.0 SIP
IPC Information Systems Alliance MX IPC System Center (Sun ULTRA 25) IPC IQ/MAX Turrets	Alliance Release 15.02.00.10
IPC ESS (SIP Proxy Server)	2.00.01-8

Table 1 – Equipment and Software Version Validated

4. Configure Avaya Communication Manager

This section describes the configuration on Avaya Communication Manager to interoperate with the SIP interface of the Avaya SIP Enablement Services and Avaya Modular Messaging to support IPC System Interconnect solution.

4.1. Configure SIP Trunk

This section describes the steps for configuring SIP trunk between Avaya Communication Manager and Avaya SIP Enablement Services. It details the administration on Avaya Communication Manager to enable the IPC turrets to register as SIP endpoints and utilize certain Avaya Communication Manager features. Also the trunk provides support for Modular Messaging SIP Integration. The commands listed in the following sections were issued at the Avaya System Access Terminal (SAT).

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

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

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

                                                              USED
                              Platform Maximum Ports: 48000 239
                                    Maximum Stations: 36000 21
                             Maximum XMOBILE Stations: 10    1
                   Maximum Off-PBX Telephones - EC500: 10    1
                   Maximum Off-PBX Telephones -   OPS: 100   7
                   Maximum Off-PBX Telephones - PBFMC: 0     0
                   Maximum Off-PBX Telephones - PVFMC: 0     0
                   Maximum Off-PBX Telephones - SCCAN: 0     0




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

Step	Description
	On **Page 2**, verify the **Maximum Administered SIP Trunks** is enough to support the expected total traffic to and from all Avaya and IPC extensions. Any call involving a SIP endpoint will use one SIP trunk per SIP endpoint. If the capacity indicated is deemed insufficient, an authorized Avaya support technician will need to install an appropriately enabled license file.

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

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 100   0
           Maximum Concurrently Registered IP Stations: 12000 2
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                        Maximum Video Capable Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 100   100
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 0     0

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

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

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

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



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

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

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

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

           IP Attendant Consoles? n
        (NOTE: You must logoff & login to effect the permission changes.)
```

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

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

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


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

Step	Description

2.	Enter **display system-parameters features** command. On **Page 9**, verify **CPN/ANI/ICLID PARAMETERS** are set to **restricted** for CPN (calling party number) restriction testing with IPC.
	
	`display system-parameters features                              Page   9 of  17`
	`FEATURE-RELATED SYSTEM PARAMETERS`
	`CPN/ANI/ICLID PARAMETERS`
	`CPN/ANI/ICLID Replacement for Restricted Calls: restricted`
	`CPN/ANI/ICLID Replacement for Unavailable Calls: restricted`
	`DISPLAY TEXT`
	`Identity When Bridging: principal`
	`INTERNATIONAL CALL ROUTING PARAMETERS`
	`Local Country Code:`
	`International Access Code:`
	`ENBLOC DIALING PARAMETERS`
	`Enable Enbloc Dialing without ARS FAC? n`
	`CALLER ID ON CALL WAITING PARAMETERS`
	`Caller ID on Call Waiting Delay Timer (msec): 200`
	
	On **Page 16**, turn on shuffling by setting **Direct IP-IP Audio Connections** to **y**.
	
	`change system-parameters features                               Page  16 of  17`
	`FEATURE-RELATED SYSTEM PARAMETERS`
	`AUTOMATIC EXCLUSION PARAMETERS`
	`Automatic Exclusion by COS? n`
	`Recall Rotary Digit: 2`
	`Duration of Call Timer Display (seconds): 3`
	`WIRELESS PARAMETERS`
	`Radio Controllers with Download Server Permission (enter board location)`
	`1:          2:          3:          4:          5:`
	`IP PARAMETERS`
	`Direct IP-IP Audio Connections? y`
	`IP Audio Hairpinning? y`
	`RUSSIAN MULTI-FREQUENCY PACKET SIGNALING`
	`Re-try? n`
	`T2 (Backward Signal) Activation Timer (secs): 20`

Step	Description
3.	Enter **display feature-access-codes** command. On **Page 1**, verify **Auto Alternate Routing (AAR) Access Code** is set to a defined fac (feature access code). In this example, **Auto Alternate Routing (AAR) Access Code** is set to **1**.

```
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: 655
                      Announcement Access Code:
                       Answer Back Access Code: #3
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 1
    Auto Route Selection (ARS) - Access Code 1: *7    Access Code 2:
                 Automatic Callback Activation: *4     Deactivation: #4
Call Forwarding Activation Busy/DA: *2     All: *3     Deactivation: #2
   Call Forwarding Enhanced Status:        Act: 622    Deactivation: 623
                         Call Park Access Code: #5
                       Call Pickup Access Code: *6
CAS Remote Hold/Answer Hold-Unhold Access Code: #6
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
                   Contact Closure   Open Code:          Close Code:
```

On **Page 3**, verify **Per Call CPN Blocking Code Access Code** is set to a defined fac (feature access code). In this example, **Per Call CPN Blocking Code Access Code** is set to **606**.

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

 PASTE (Display PBX data on Phone) Access Code:
  Personal Station Access (PSA) Associate Code:       Dissociate Code:
        Per Call CPN Blocking Code Access Code: 606
      Per Call CPN Unblocking Code Access Code: 607
                 Posted Messages Activation:          Deactivation:
                  Priority Calling Access Code: 653
                           Program Access Code:

      Refresh Terminal Parameters Access Code:
             Remote Send All Calls Activation:        Deactivation:
               Self Station Display Activation: 624
                     Send All Calls Activation: *1     Deactivation: #1
         Station Firmware Download Access Code:
```

Step	Description
4.	Enter **display dialplan parameters** command to assign **Local Node Number**. Ensure PBX has an assigned **Local Node Number**, if there is no assigned number, enter **1**.
5.	Enter **display node-names ip** command, assign IP Node Name and IP addresses to C-LAN (**clan**), IP Media Processor (**medpro**) and SES (**ses**), in this sample application notes, the values are assigned as below.

Step 4 screen:

```
display dialplan parameters
                             DIAL PLAN PARAMETERS

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

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

Step 5 screen:

```
display node-names ip                                          Page   1 of   2
                                IP NODE NAMES
    Name              IP Address
clan                10.10.1.30
default             0.0.0.0
gateway             10.10.1.5
medpro              10.10.1.31
procr               10.10.1.18
ses                 10.10.1.52
```

Step	Description
6.	Enter **change ip-codec-set 1** to specify the audio codec to be used for calls routed to and from IPC extensions via Avaya SIP Enablement Services. The codec supported by the IPC are shown in the diagram below. In addition, **Media Encryption** preference must be set to **none**, since IPC doesn't support media encryption. **Frames Per Pkt** field depends on customer's requirements. By default it is set to **2**.

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

                          IP Codec Set
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711A           n          3        30
 2: G.711MU          n          3        30
 3: G.729B           n          3        30
 4:
 5:
 6:
 7:
     Media Encryption
 1: none
 2:
 3:
```

Step	Description
7.	Enter **display ip-network region *n***, where ***n*** is the IP network region where the Avaya SIP Enablement Services server resides. **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** are set to **yes** to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya Media Gateway (a feature known as shuffling). **Authoritative Domain** (e.g., **du.rnd.avaya.com**) must match what is used on the Signaling group or a call from the Avaya Modular Messaging to the Avaya Communication Manager will not authenticate. Assign a **Codec Set** (e.g., **1**) to use for this network region.

```
display ip-network-region 1                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: du.rnd.avaya.com
    Name: ses_local
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                      RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

Step	Description

8.	Use the **add signaling-group *n*** command to add a signaling group for the SIP connection between Avaya Communication Manager and Avaya SIP Enablement Services. **Near-end Node Name/Far-end Node Name** is set to the name assigned to the IP address of the Avaya CLAN/SIP Enablement Services in **Step 5** of this section. Set the **Far-end Network Region** to the IP Network Region (e.g., **1**) defined in **Step 7**. **Far-end Domain** is set to blank for the SIP trunk between Avaya Communication Manager and Avaya SIP Enablement Services. This will allow INVITE messages to be sent by IPC using the IP address of the IPC ESS (IPC Enterprise SIP Server) instead of a domain name. Also verify that the others **bold** items are set.
	``add signaling-group 1 Page 1 of 1
	                                SIGNALING GROUP
	 Group Number: 1 Group Type: sip
	                       Transport Method: tls
	          IP Video? n
	   Near-end Node Name: clan Far-end Node Name: ses
	 Near-end Listen Port: 5061 Far-end Listen Port: 5061
	                                      Far-end Network Region: 1
	       Far-end Domain:
	                                       Bypass If IP Threshold Exceeded? n
	         DTMF over IP: rtp-payload Direct IP-IP Audio Connections? y
	                                                   IP Audio Hairpinning? y
	         Enable Layer 3 Test? n
	 Session Establishment Timer(min): 3 Alternate Route Timer(sec): 6```

9.	Use the **add trunk-group *n*** command to add a trunk group between Avaya Communication Manager and Avaya SIP Enablement Services. On **Page 1**, verify that the **Number of Members** field is appropriate to support the anticipated traffic involving the turrets, but not exceeding the maximum number of available SIP trunks as indicated in **Step 1** of this section. Also verify that the others **bold** items are set. **Page 1** of the trunk group form is shown below.
	``add trunk-group 1 Page 1 of 21
	                                TRUNK GROUP
	
	Group Number: 1 Group Type: sip CDR Reports: y
	  Group Name: ses_local COR: 1 TN: 1 TAC: 511
	   Direction: two-way Outgoing Display? y
	 Dial Access? n Night Service:
	Queue Length: 0
	Service Type: tie Auth Code? n
	                                                   Signaling Group: 1
	                                                 Number of Members: 50```

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

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

TRUNK PARAMETERS

     Unicode Name? y

                                            Redirect On OPTIM Failure: 5000

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

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

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


                     Numbering Format: public
                                                UUI Treatment: service-provider

                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? n


Show ANSWERED BY on Display? n
```

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

```
display trunk-group 1                                            Page   5 of  21
                                 TRUNK GROUP
                                      Administered Members (min/max):   1/50
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  50
       Port            Name
  1: T00030            ses_local
  2: T00031            ses_local
  3: T00032            ses_local
  4: T00033            ses_local
  5: T00034            ses_local
  6: T00035            ses_local
  7: T00036            ses_local
  8: T00037            ses_local
  9: T00038            ses_local
 10: T00039            ses_local
 11: T00040            ses_local
 12: T00041            ses_local
 13: T00042            ses_local
 14: T00043            ses_local
 15: T00044            ses_local
```

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>10.</td><td>Use the add signaling-group n command to add another signaling group for the SIP connection between Avaya Communication Manager and Avaya SIP Enablement Services. Verify that the following bold items are set. Note: Far-end Domain must match Authoritative Domain field (e.g., du.rnd.avaya.com) on the Network Region in Step 7 of this section. Otherwise, inbound calls or SIP Messages to Avaya Communication Manager from the Avaya Modular Messaging may not work.

```
add signaling-group 6                                          Page   1 of   1
                               SIGNALING GROUP
 Group Number: 6             Group Type: sip
                       Transport Method: tls

          IP Video? n
   Near-end Node Name: clan                    Far-end Node Name: ses
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region: 1
      Far-end Domain: du.rnd.avaya.com
                                               Bypass If IP Threshold Exceeded? n

        DTMF over IP: rtp-payload             Direct IP-IP Audio Connections? y
                                                        IP Audio Hairpinning? y
        Enable Layer 3 Test? n
 Session Establishment Timer(min): 3              Alternate Route Timer(sec): 6
```
</td></tr>
<tr><td>11.</td><td>Use the add trunk-group n command to add a trunk group between Avaya Communication Manager and Avaya SIP Enablement Services. On Page 1 verify Number of Members field is appropriate to support the anticipated traffic involving the turrets, but not exceeding the maximum number of available SIP trunks as indicated in Step 1 of this section. Also verify that the other bold items are set.

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

Group Number: 6                    Group Type: sip           CDR Reports: y
  Group Name: MM SIP                      COR: 1        TN: 1        TAC: 506
   Direction: two-way       Outgoing Display? y
 Dial Access? n                                      Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n

                                                    Signaling Group: 6
                                                  Number of Members: 50
```

Page 2 of the trunk group form is shown below. Preferred Minimum Session Refresh Interval (sec) is set depending on customers requirements. In this sample configuration, 600 was used.

```
display trunk-group 6                                          Page   2 of  21
      Group Type: sip

TRUNK PARAMETERS

     Unicode Name? y
                                             Redirect On OPTIM Failure: 5000

            SCCAN? n                                 Digital Loss Group: 18
                        Preferred Minimum Session Refresh Interval(sec): 600
```
</td></tr>
</table>

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

```
display 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? n


Show ANSWERED BY on Display? y
```

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

```
display trunk-group 6                                           Page   5 of  21
                                  TRUNK GROUP
                                       Administered Members (min/max):   1/50
GROUP MEMBER ASSIGNMENTS                   Total Administered Members:  50

       Port            Name
  1: T00169            MM SIP
  2: T00170            MM SIP
  3: T00171            MM SIP
  4: T00172            MM SIP
  5: T00173            MM SIP
  6: T00174            MM SIP
  7: T00175            MM SIP
  8: T00176            MM SIP
  9: T00177            MM SIP
 10: T00178            MM SIP
 11: T00179            MM SIP
 12: T00180            MM SIP
 13: T00181            MM SIP
 14: T00182            MM SIP
 15: T00183            MM SIP
```

Step	Description
12.	Use the **add hunt-group *n*** command (not shown) to configure a hunt group to be used as the Call Coverage Point for the Call Coverage Path assigned to the Avaya Modular Messaging Application Server (MAS) subscribers. **Change hunt-group *n*** command is used to show the configuration after the hunt group was added. **Page 1** of the hunt-group form is shown below.

```
change hunt-group 3                                             Page   1 of  60
                                   HUNT GROUP

            Group Number: 3                                ACD? n
              Group Name: sipMAS                         Queue? n
         Group Extension: 7776                          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: mbr-name
```

Step	Description
	On **Page 2**, **Voice Mail Number** is going to be used as the Avaya Modular Messaging Access Number. This hunt group is configured with no members assigned to it. **Voice Mail Handle** will be used by the proxy in a later step, use the generic identifier that administer on Avaya SIP Enablement Services system not the actual pilot number. Also, in the Routing Digit field of this form, enter **Auto Alternate Routing (AAR) Access Code** as defined in the Feature Access Code from **Section 4.1 Step 3**.

```
change hunt-group 3                                           Page   2 of  60
                                 HUNT GROUP



                      Message Center: sip-adjunct

     Voice Mail Number         Voice Mail Handle          Routing Digits
                                                   (e.g., AAR/ARS Access Code)
     7777                      silmm                      1
```

4.2. Configure Call Routing

Step	Description
1.	Use the **change dialplan analysis** command to define the number range for the IPC turrets and Avaya extensions. In this sample Application Notes, 33xx are used as IPC SIP extensions, 61xx and 66xx are Avaya extensions, 77xx is used as Avaya Modular Messaging Voice Mail Number. Therefore, add an entry in the Dial Plan Analysis Table for 4-digit numbers beginning with **33** to use the Uniform Dial Plan (**udp**). Add another 3 entries for 4-digit numbers beginning with **61**, **66** and **77** as extensions.

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

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

Step	Description
2.	Use the **change uniform-dialplan *n*** command to add entries to route 4-digit numbers beginning with **33** and **7777** using Alternate Automatic Routing (**AAR**).

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

  Matching                   Insert             Node
  Pattern       Len Del      Digits     Net Conv Num
 22              4   0                  aar  n
 30              9   0                  aar  n
 3005            8   0                  aar  n
 31              4   0                  aar  n
 33              4   0                  aar  n
 7777            4   0                  aar  n
 8889            4   0                  aar  n
 972             5   0                  aar  n
```

Step	Description
3.	Use the **change public-unknown-numbering *n*** command to add entries so that calls placed from stations with a 4-digit extension beginning with a **61** or **66** and routed over all trunk groups will send a 4-digit calling party number to the far end.

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

Step	Description
4.	Use the **change locations** command to verify that the following **bold** items are set.

```
change locations                                            Page   1 of  16
                                 LOCATIONS

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

 Loc  Name           Timezone Rule  NPA  ARS  Atd Loc  Disp   Prefix   Proxy Sel
 No                   Offset             FAC  FAC Parm Parm              Rte Pat
 1:  Main            + 00:00   0                   1    1                 1
```

Step	Description
5.	Use the **change route-pattern *n*** command to route calls for pattern **1** using trunk group **1** which is the SIP trunk between Avaya Communication Manager and SIP Enablement Services.
6.	Use the **change route-pattern *n*** command to route calls for pattern **6** using trunk group **6** which is a SIP trunk between Avaya Communication Manager and the Avaya SIP Enablement Services which will support Avaya Modular Messaging SIP integration.

Step 5 screen:

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

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

Step 6 screen:

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

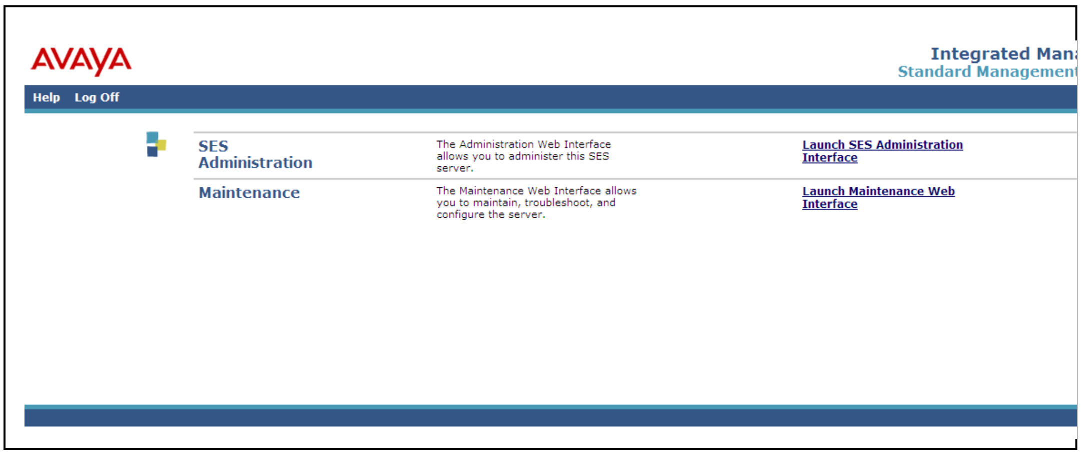
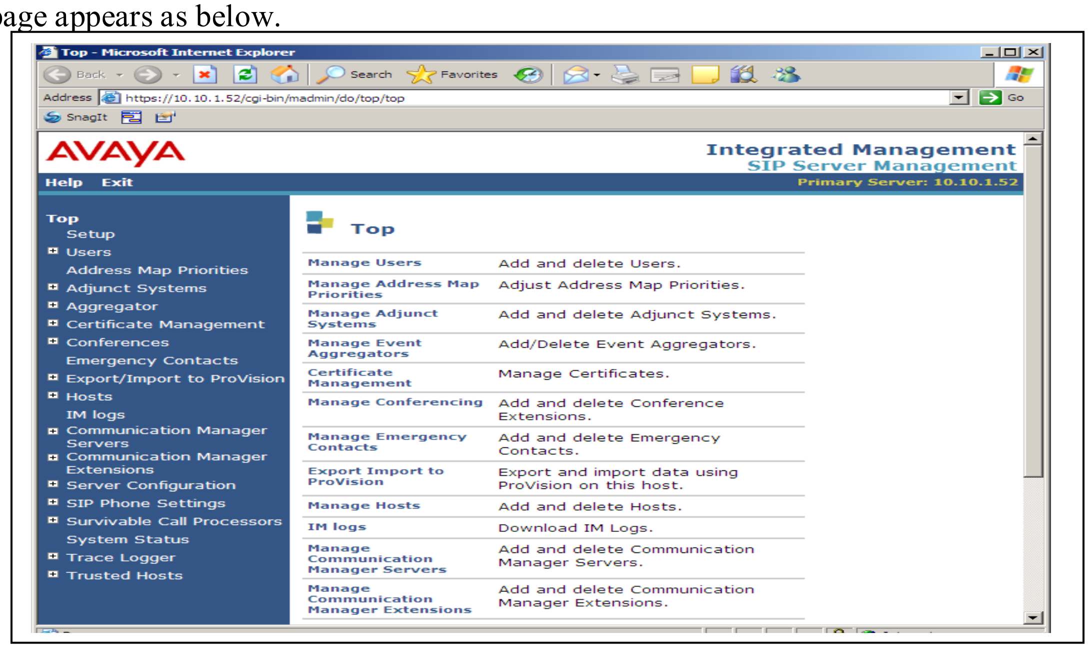
Step	Description
7.	Use the **change aar analysis *n*** command to specify which route pattern to use based upon the number dialed. In this example, **Route Pattern 1** is used for IPC extension 33xx, **Route Pattern 6** is used for voice mail number **7777**. **change aar analysis 0** will display all **Dialed Strings** defined in this Table.

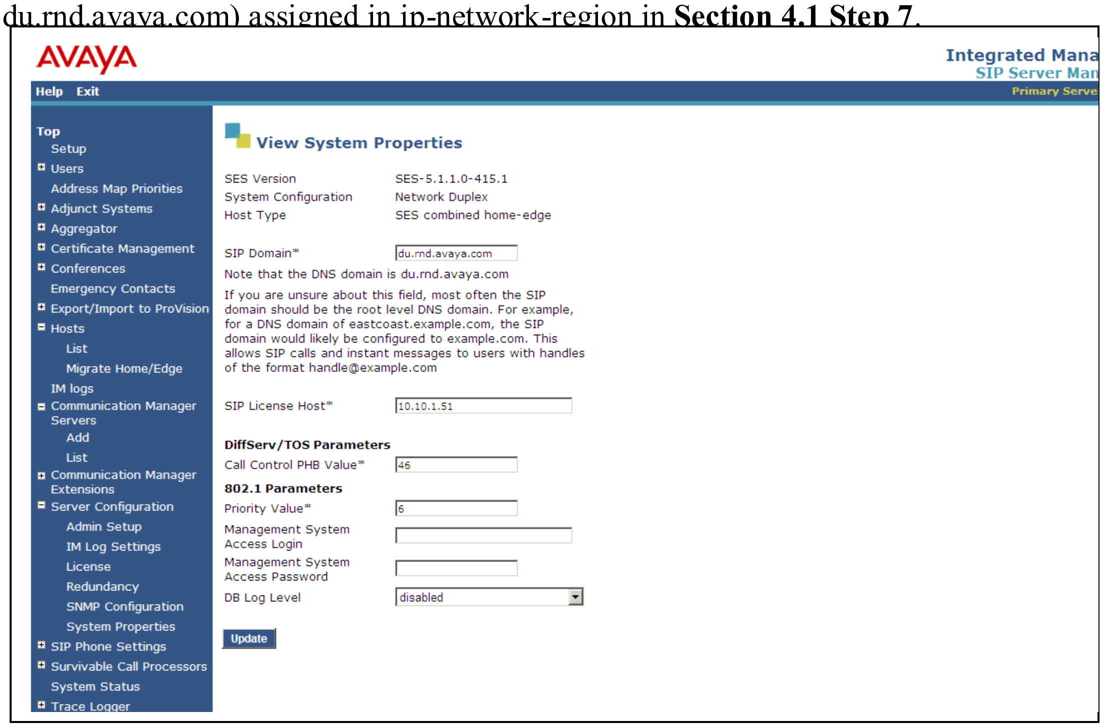
```
change aar analysis 0                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    1

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    2                     4    4     999      aar          n
    30                    9    9     1        aar          n
    3005                  8    8     1        aar          n
    31                    4    4     4        aar          n
    33                    4    4     1        aar          n
    38                    5    5     3        aar          n
    4                     4    4     3        aar          n
    5                     7    7     999      aar          n
    7777                  4    4     6        aar          n
    8                     7    7     999      aar          n
    8889                  4    4     2        aar          n
    9                     7    7     999      aar          n
    972                   5    5     5        aar          n
```

5. Configure Avaya SIP Enablement Services

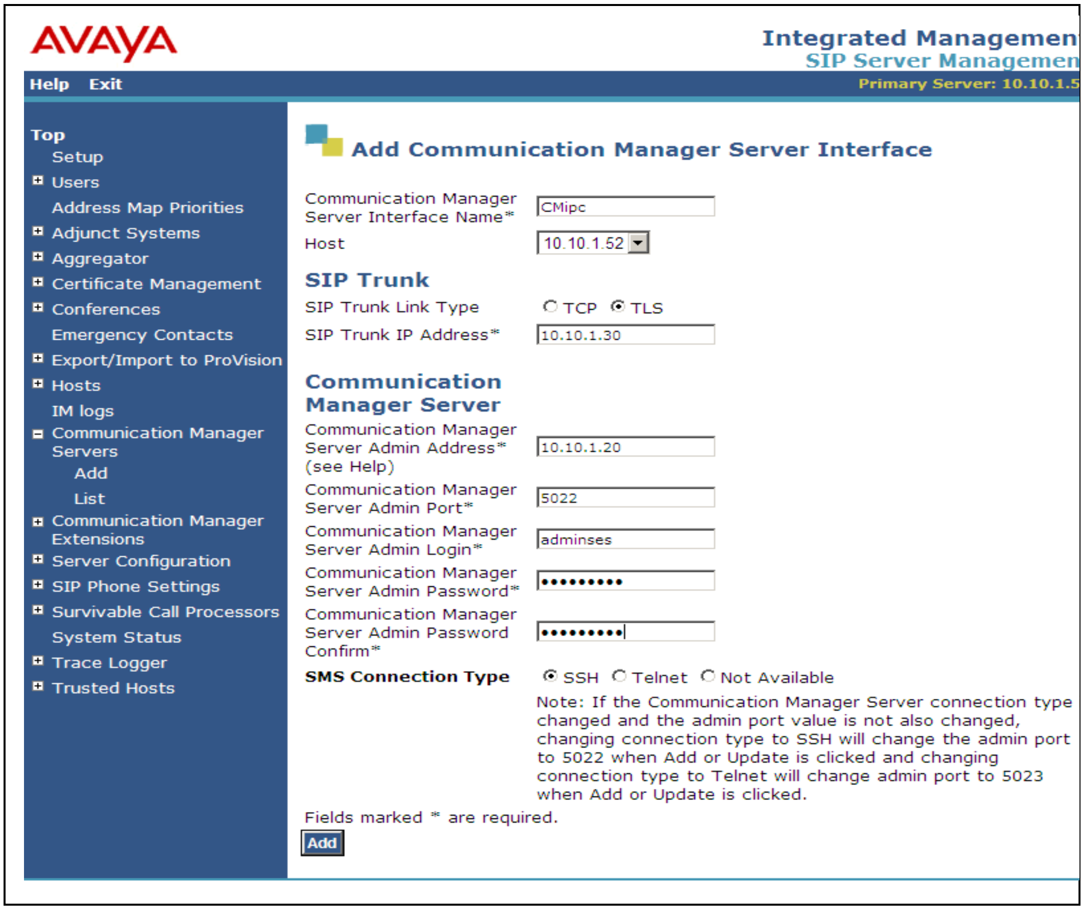
This section addresses the administrative steps to be performed on the Avaya SIP Enablement Services. The installation of Avaya SIP Enablement Services software and license file, as well as the initial configuration of the server and its basic integration with Avaya Communication Manager, is beyond the scope of this document.

Step	Description
1.	From a Web browser, navigate to http://<*ip-addr*>/admin (where <*ip-addr*> is the IP address of the Avaya SIP Enablement Services Edge Server). After logging in with an appropriate login and password, the main page appears. 
2.	Click on the **Launch SES Administration Interface** link, the administration home page appears as below. 

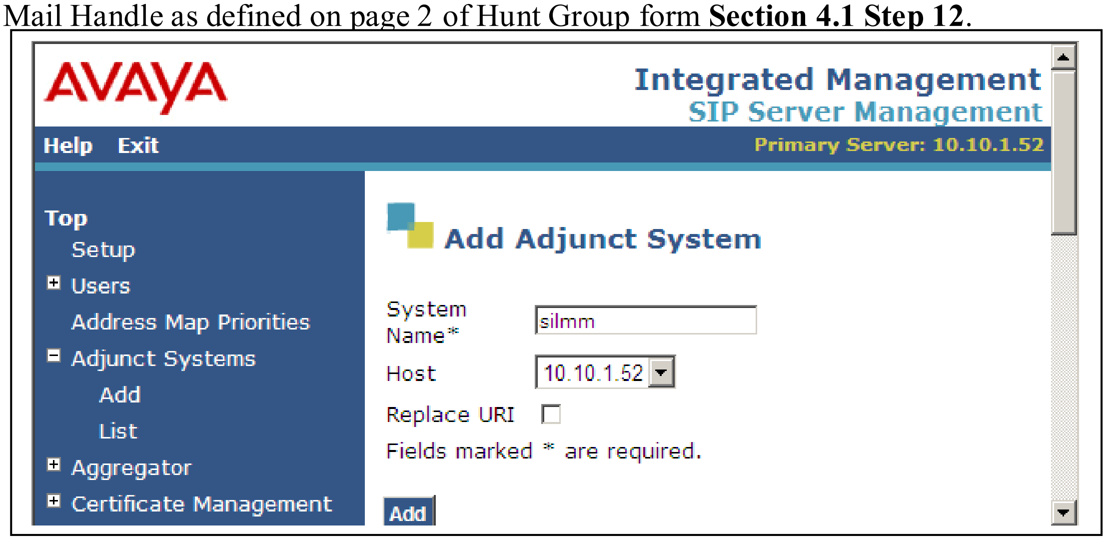
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>Navigate to Server Configuration → System Properties. Ensure that the DNS server can resolve the SIP Domain to the IP address of the Avaya SIP Enablement Services so that requests from the IPC Enterprise SIP Server (ESS) can be forwarded correctly. SIP Domain value must match the Authoritative Domain (e.g., du.rnd.avaya.com) assigned in ip-network-region in Section 4.1 Step 7.


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

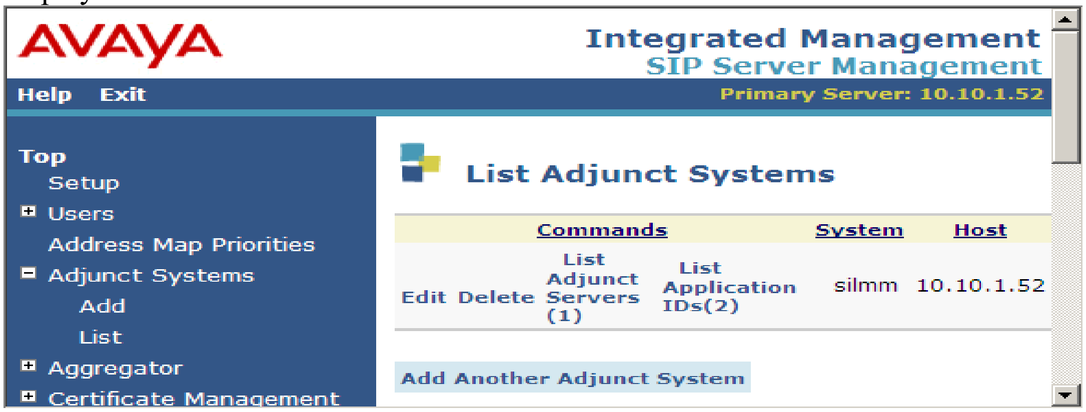
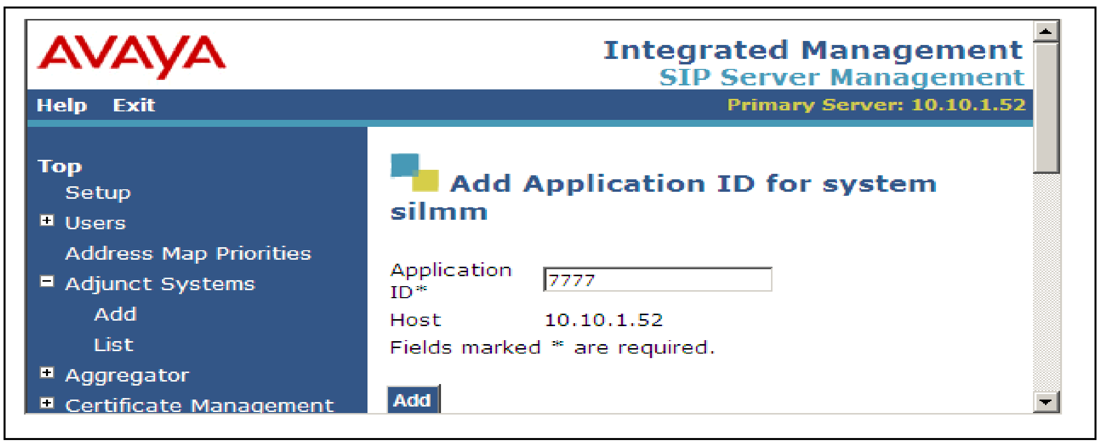
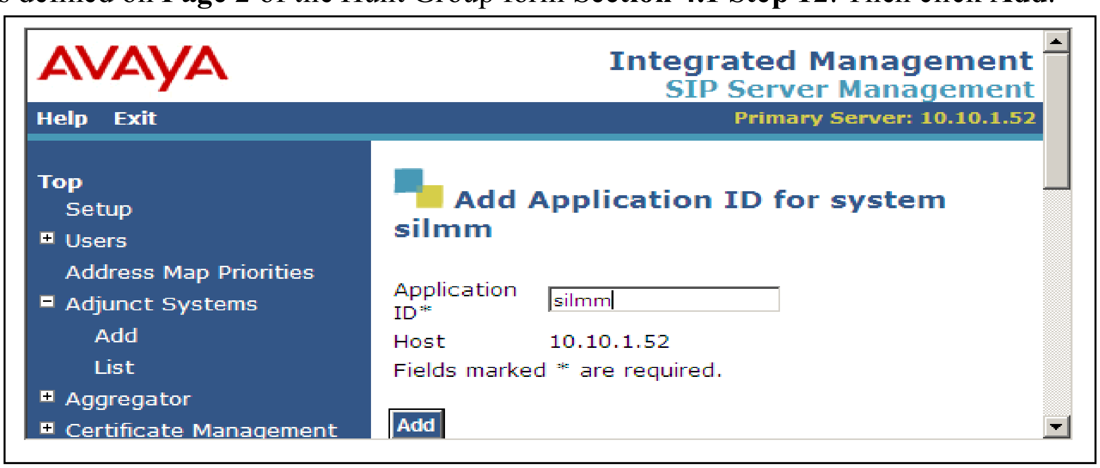
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>Navigate to Communication Manager Servers → Add. Verify SIP Trunk IP Address is set to CLAN IP shown on the node-names ip form in Section 4.1 Step 5.

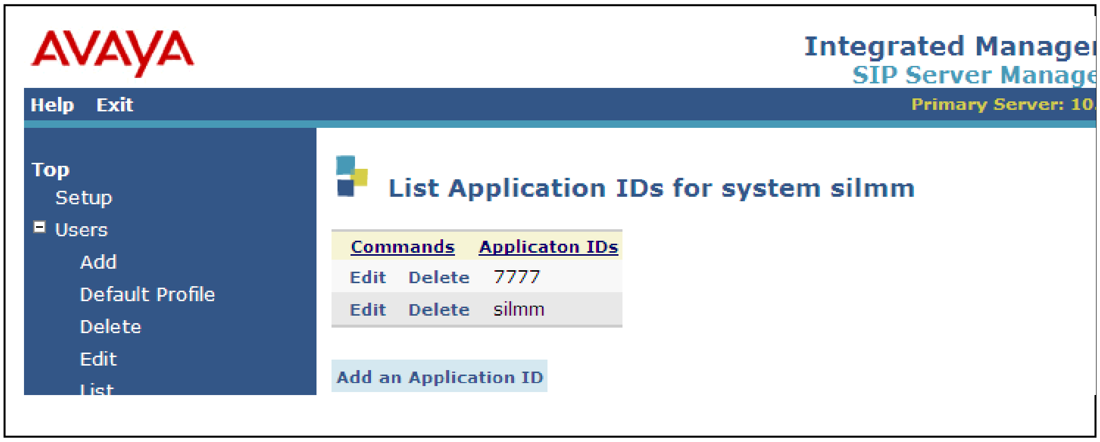
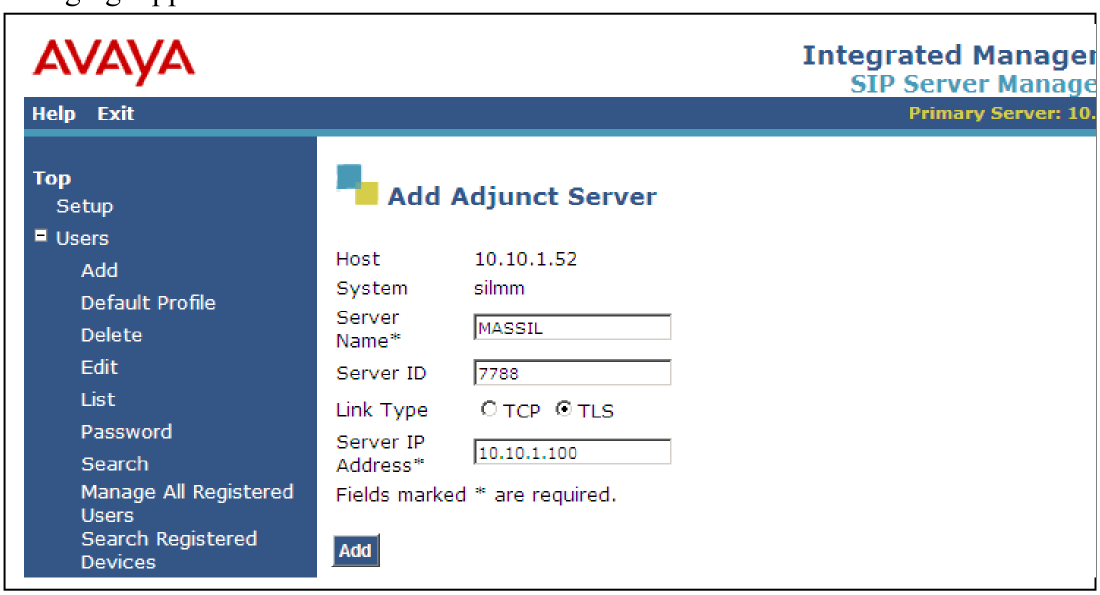


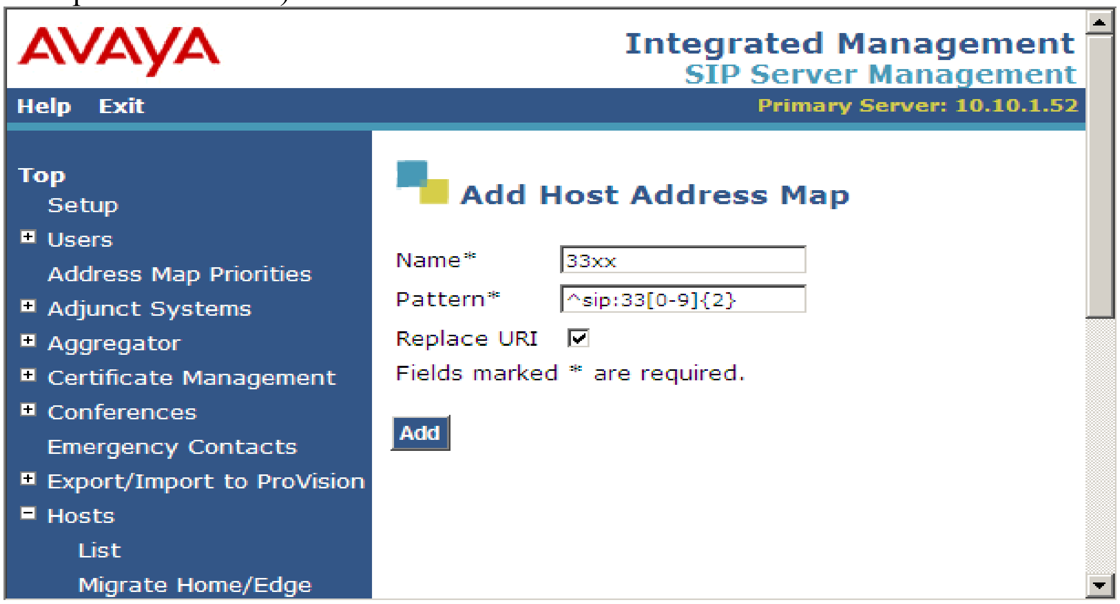
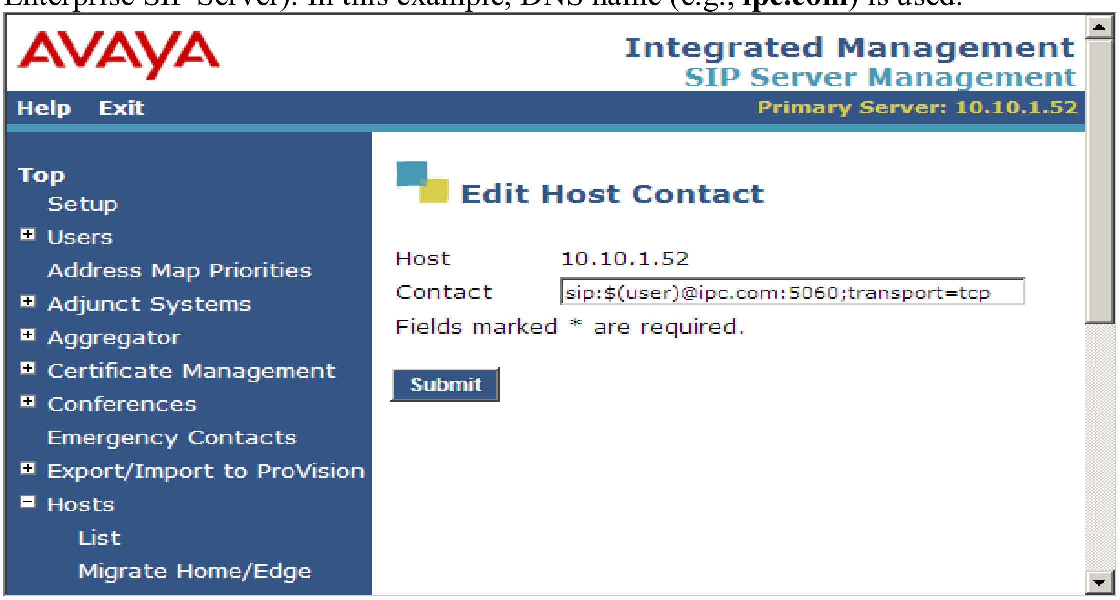
</td></tr>
<tr><td>5.</td><td>Navigate to Adjunct Systems → Add. In the System Name field, enter the Voice Mail Handle as defined on page 2 of Hunt Group form Section 4.1 Step 12.

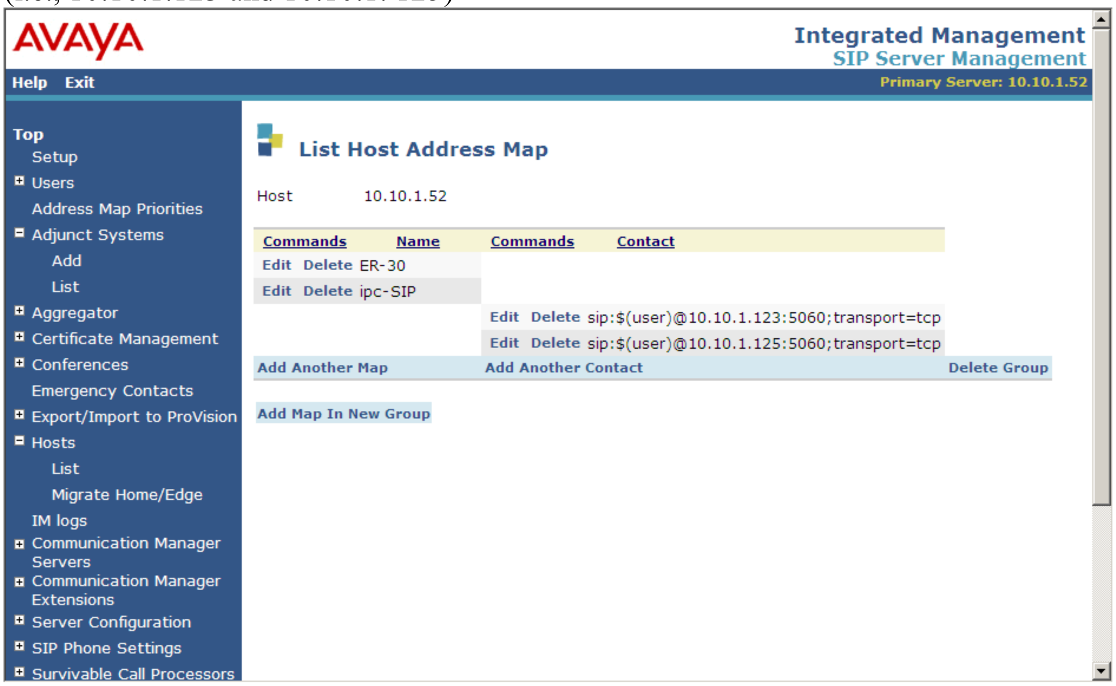


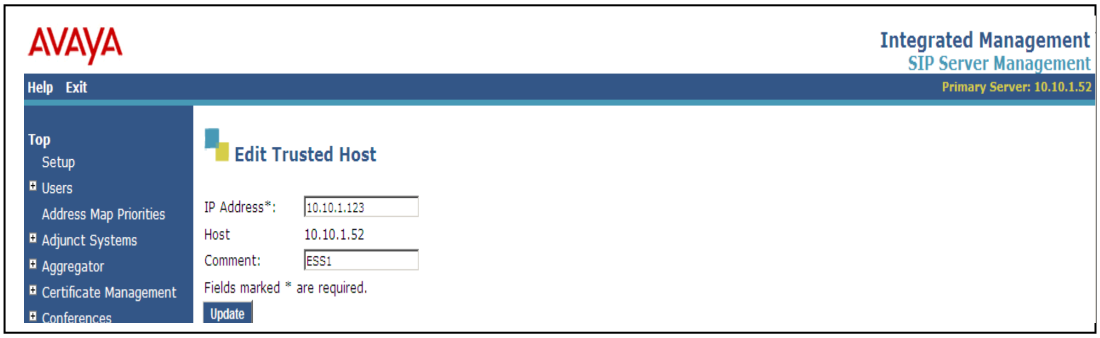
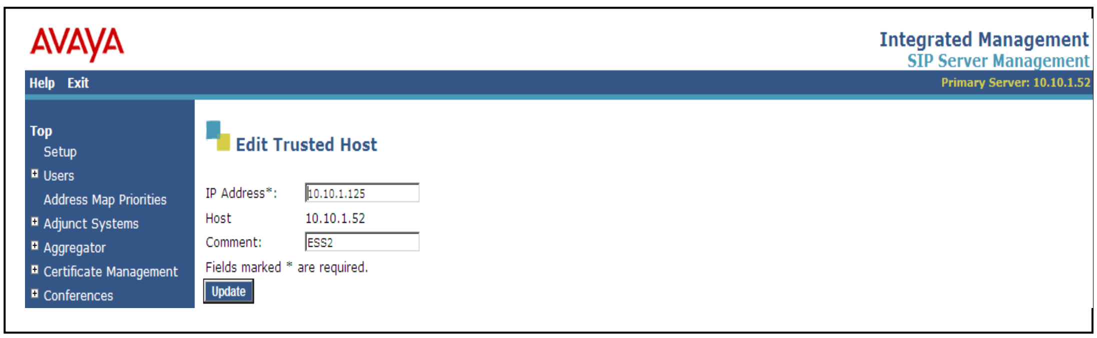
</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>Navigate to Adjunct Systems → List. The list of available Adjunct Systems is now displayed.

</td></tr>
<tr><td>7.</td><td>Click on List Application IDs → Add an Application ID. Enter Voice Mail Number as defined on Page 2 of the Hunt Group form in Section 4.1 Step 12. Then click Add.

</td></tr>
<tr><td>8.</td><td>Click on List Application IDs →Add an Application ID. Enter Voice Mail Handle as defined on Page 2 of the Hunt Group form Section 4.1 Step 12. Then click Add.

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

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>9.</td><td>The list of configured Application IDs is displayed.

</td></tr>
<tr><td>10.</td><td>Navigate to Adjunct Systems → List → List Adjunct System → Add an Adjunct Server to System silmm (silmm is the adjunct system created in Step 5 of this Section). The screen below is to add information for Avaya Modular Messaging Application Server. Server Name is the name of the Avaya Modular Messaging Application Server. Server ID is an extension defined in dial plan, which is different to voice mail number. Server IP Address is the IP Address of the Avaya Modular Messaging Application Server.

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

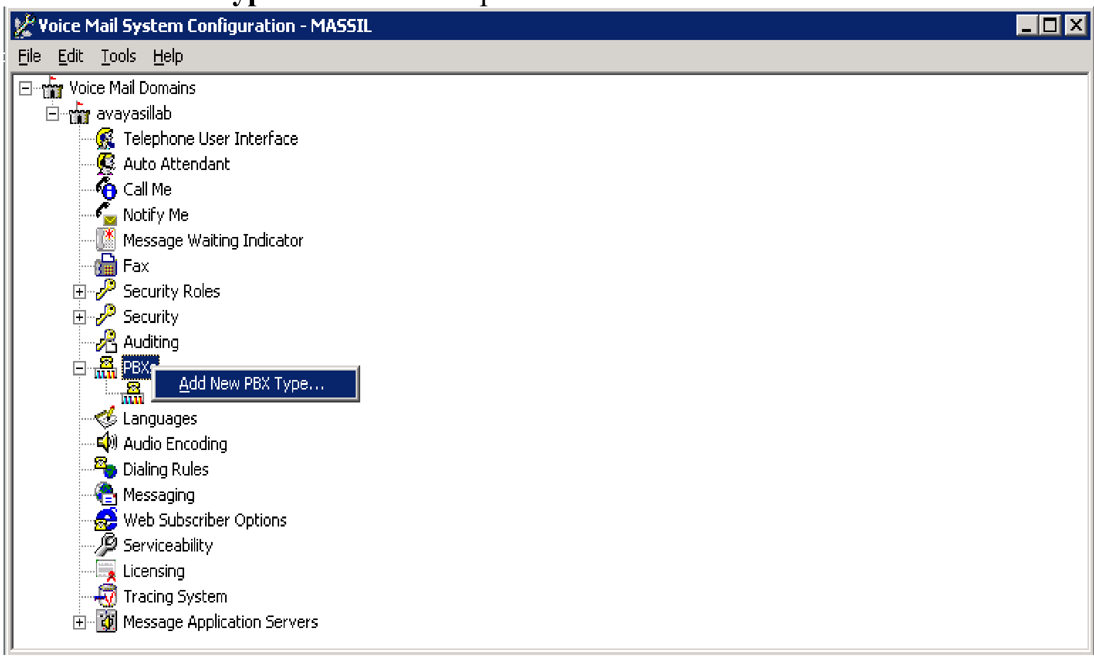
Step	Description
11.	Navigate to **Hosts → List → Map → Add Map in New Group**. Add a Host Address Map as shown below to route calls that begin with a leading **33** to the IPC ESS (IPC Enterprise SIP Server). 
12.	The contact associated with the Host Address Map is shown below to route the call to the IP address (e.g., 10.10.1.123) or DNS name (e.g., **ipc.com**) of the IPC ESS (IPC Enterprise SIP Server). In this example, DNS name (e.g., **ipc.com**) is used. 

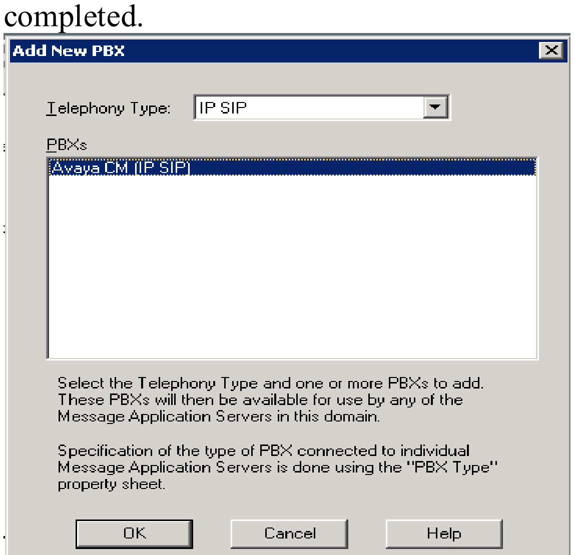
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>13.</td><td>For redundancy IPC ESS (IPC Enterprise SIP Server), add two Contact pointing to different IPC ESS (IPC Enterprise SIP Server). In this example, IP Address is used. (i.e., 10.10.1.123 and 10.10.1. 125)

</td></tr>
<tr><td>14.</td><td>To support redundancy IPC ESS (IPC Enterprise SIP Server), the following parameter settings are needed:
Change SIP parameters as shown below in the file /usr/impress/sip-server/etc/ccs.conf on the Avaya SIP Enablement Services server. Restart the server after the changes.
Default settings are :
PerContactWaitTime=30
MM_PerContactWaitTime=2
TimerB=32000
New settings are :
PerContactWaitTime=180
MM_PerContactWaitTime=0
TimerB=2000
PerContactWaitTime field is determined by customers’ needs, but it is recommended setting to at least 180 seconds.
MM_PerContactWaitTime is a fast sequential forking timer used for the SIP Enablement Services Adjunct Interface (e.g., Modular Messaging or Voice Portal).
TimerB is the INVITE transaction timeout timer as defined in RFC3261. If there is no response for the INVITE, Avaya SIP Enablement Services will try the next Contact or Adjunct Server after TimerB. In this example, TimerB is set to 2 seconds.</td></tr>
</table>

Step	Description
15.	Navigate to **Trusted Hosts→Add**. Add the IP address (e.g., **10.10.1.123**) of the IPC ESS (IPC Enterprise SIP Server) as a trusted host. The **Edit** option is used to show the entry after the trusted host was added.  If there is a redundant IPC ESS (IPC Enterprise SIP Server), added another entry for it (e.g., 10.10.1.125). 

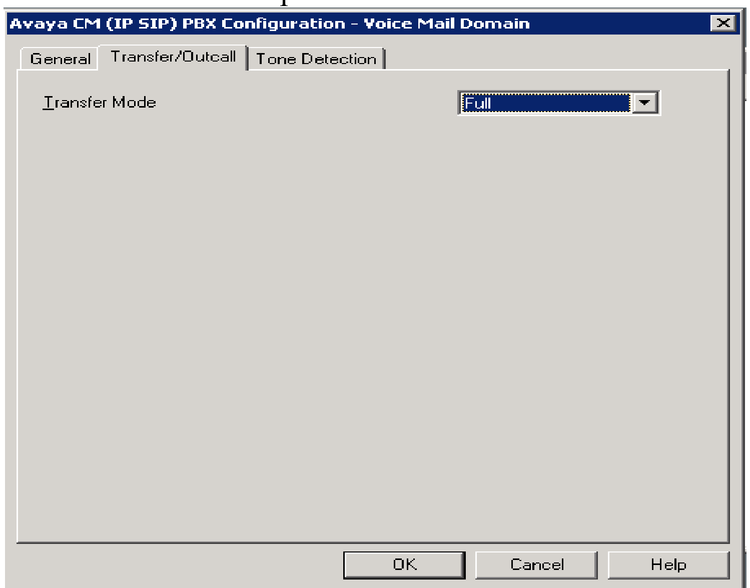
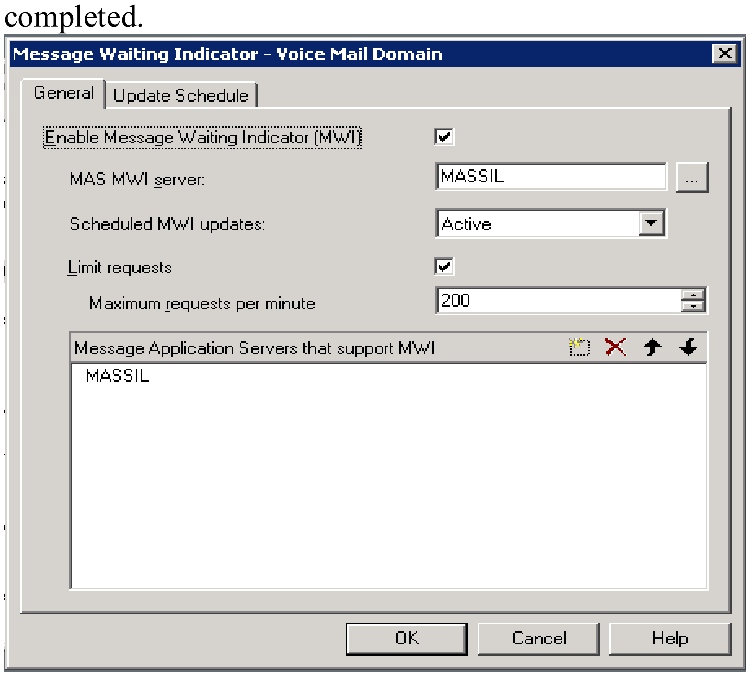
6. Configure Avaya Modular Messaging

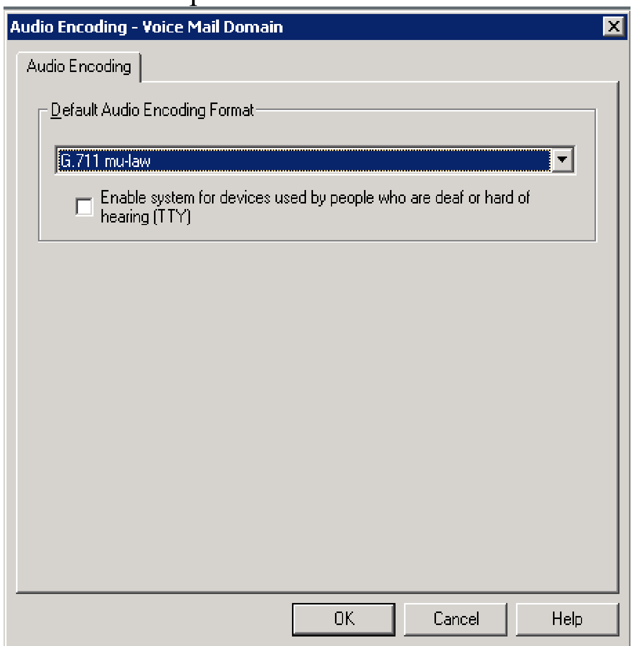
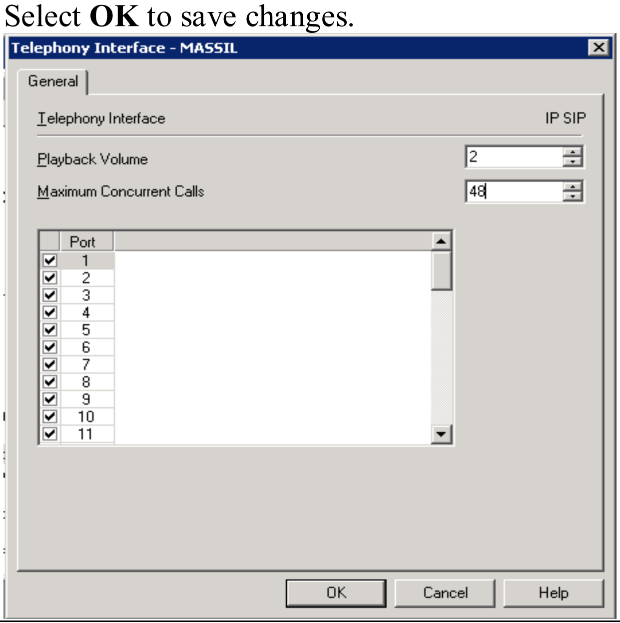
6.1. Configure Message Application Server

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>1.</td><td>Select Start → Programs → Avaya Modular Messaging → Voice Mail System Configuration - MASSIL. Expand Voice Mail Domains and the administered domain name (avayasillab in the screenshot below). Right-click on PBXs and select Add New PBX Type… from the drop down menu.


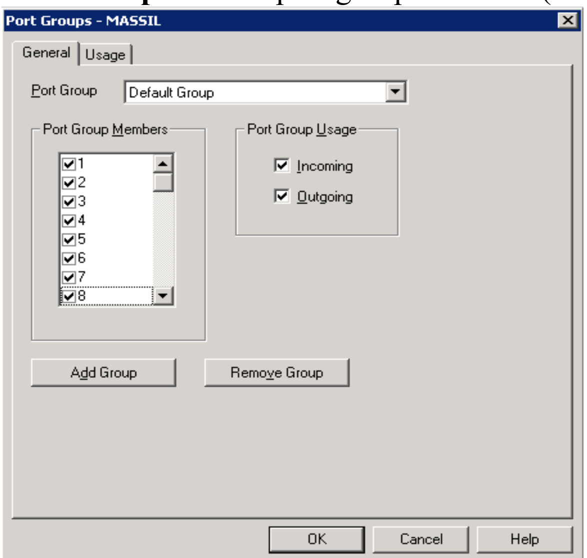
On the Add New PBX screen, select IP SIP from the Telephony Type drop down box, then select Avaya CM (IP SIP) from the PBXs box. Select OK when completed.


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

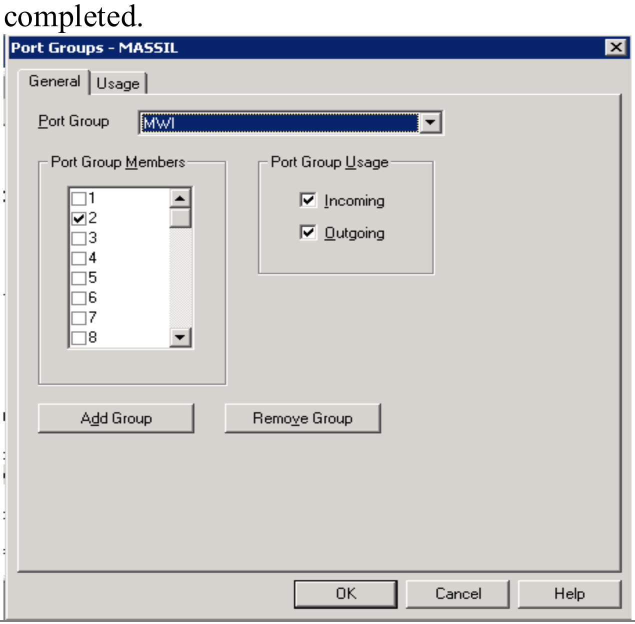
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.</td><td>On the Voice Mail System Configuration - MASSIL screen (see Step 1 of this section for screenshot), double-click on PBXs. On the Avaya CM (IP SIP) PBX Configuration screen, select the Transfer/Outcall tab. Set Transfer Mode to Full. Select OK when completed.

</td></tr>
<tr><td>3.</td><td>On the Voice Mail System Configuration - MASSIL screen, double-click on Message Waiting Indicator (MWI). Configure the fields as below. Click OK when completed.

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

Step	Description
4.	On the **Voice Mail System Configuration - MASSIL** screen, double-click on **Audio Encoding**. Chose **G.711 mu-law** or a-law for Default Audio Encoding Format. Click OK when completed. 
5.	On the **Voice Mail System Configuration - MASSIL** screen expand **Message Application Servers** and the host name of the Avaya Modular Messaging Application Server to be configured (**MASSIL** in this example). Double-click on **Telephony Interface (IP SIP)**. Set **Playback Volume** to **2** and set **Maximum Concurrent Calls** to **48** (if Avaya Modular Messaging Application Server is S3500) or 20 (if Avaya Modular Messaging Application Server is S3400). The ports are enabled by default. Select **OK** to save changes. 

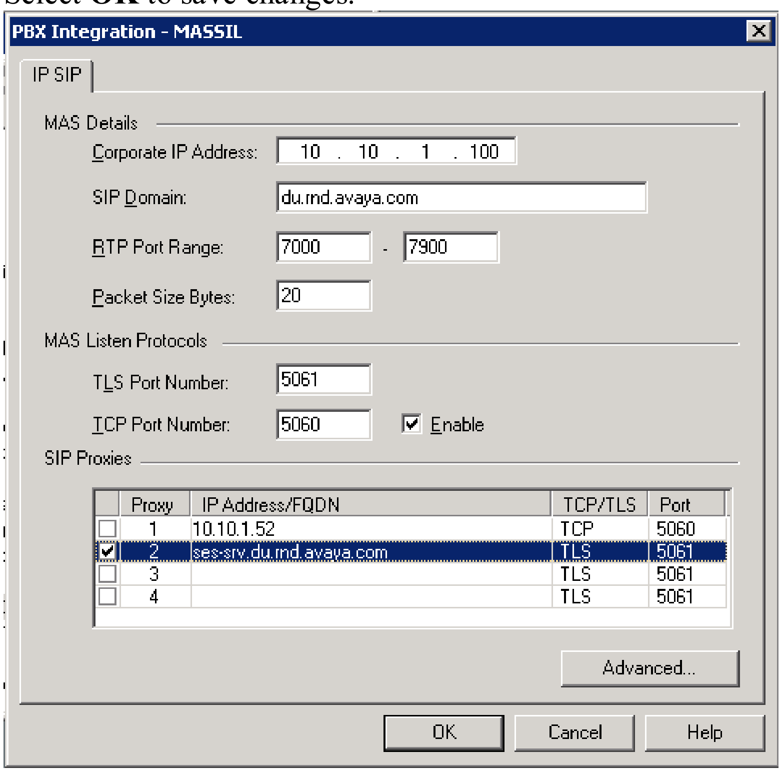
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>
On the Voice Mail System Configuration - MASSIL screen, expand Message Application Servers and the host name of the Avaya Modular Messaging Application Server to be configured (MASSIL in this example). Double-click on Port Groups. On the Port Groups screen, select Default Group from the Port Group drop down box and ensure all the checkboxes are checked in the Port Group Members box. Also ensure that both the Incoming and Outgoing checkboxes are checked. Select Add Group to add a port group for MWI (Message Waiting Indicator).



On the Add New Group screen (not shown), enter a descriptive name (e.g., MWI) for the group and select OK to return to the Port Groups screen. Select the port group that was just created from the Port Group drop down box (MWI in the screenshot below). Check one port's checkbox in the Port Group Members box. Also ensure that both the Incoming and Outgoing checkboxes are checked. Select OK when completed.

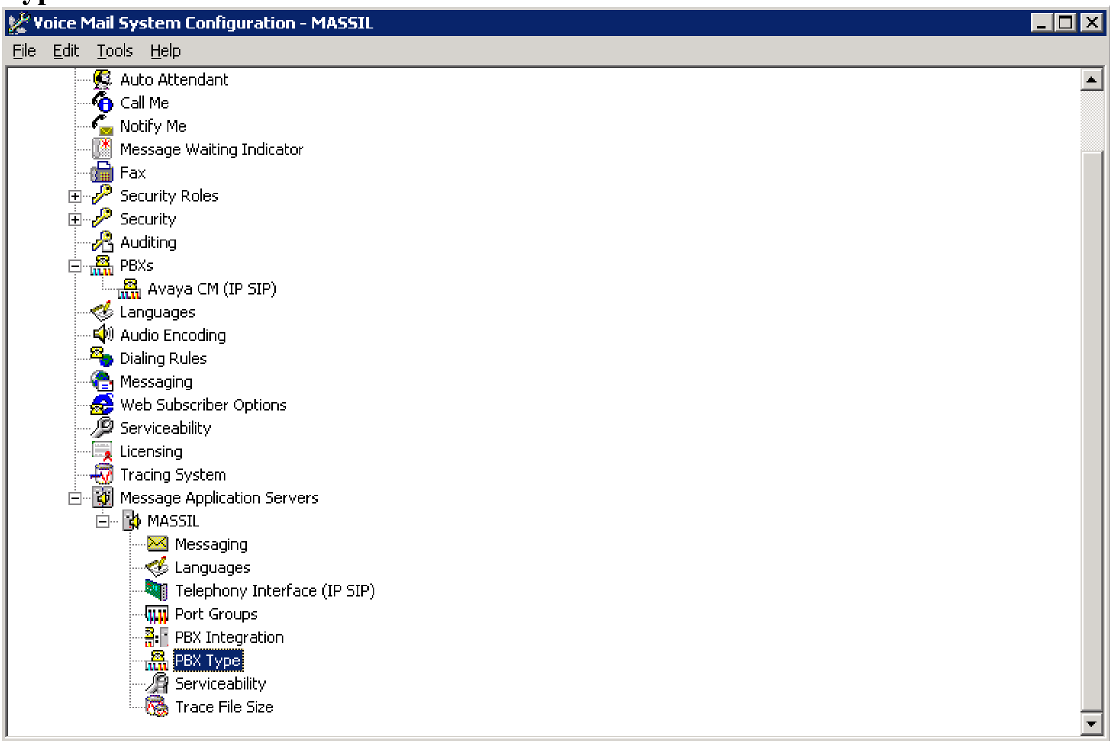


</td></tr>
</table>

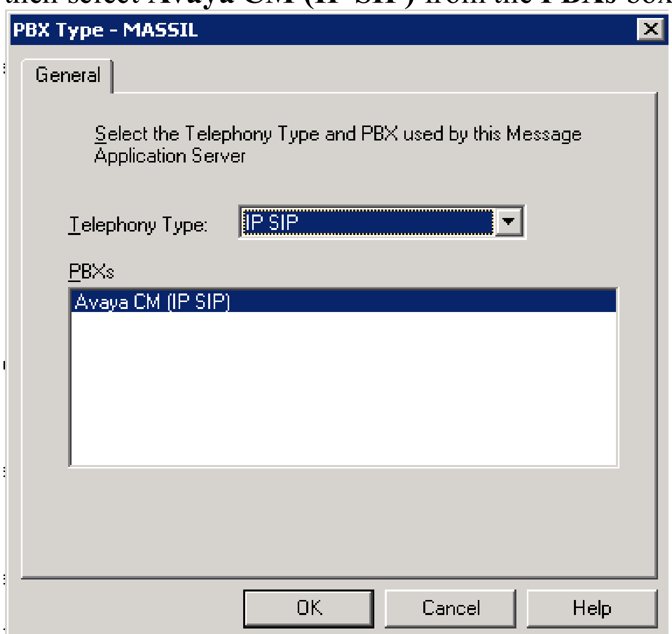
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>7.</td><td>
On the Voice Mail System Configuration - MASSIL screen expand Message Application Servers and the host name of the Avaya Modular Messaging Application Server to be configured (MASSIL in the screenshot below). Double-click on PBX Integration. The following sample configuration was used.
Corporate IP Address = IP Address assigned to the Avaya Modular Messaging Application Server.
SIP Domain = domain assigned in the IP Network Region on Avaya Communication Manager Section 4.1 Step 7.
Enable TCP port listening if desired (Note: most configurations will use TLS; TCP is for use by certified Avaya technicians only).
For each SIP Proxy select the checkbox and enter the IP Address or FQDN of the proxy server (e.g., ses-srv.du.rnd.avaya.com is the DNS name of Avaya SIP Enablement Services).
Select OK to save changes.


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

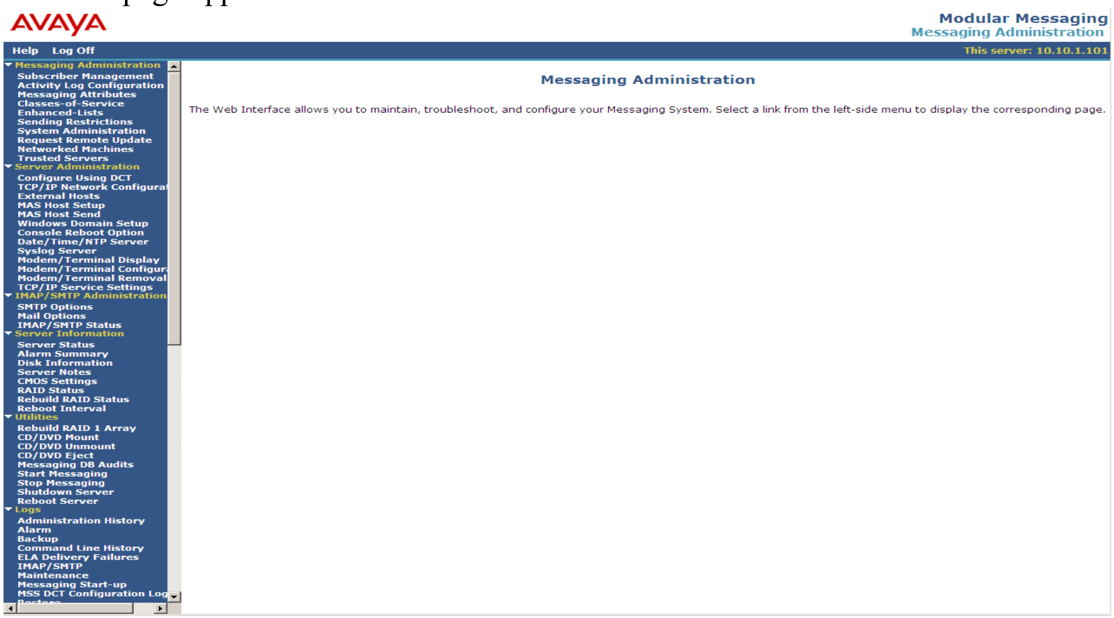
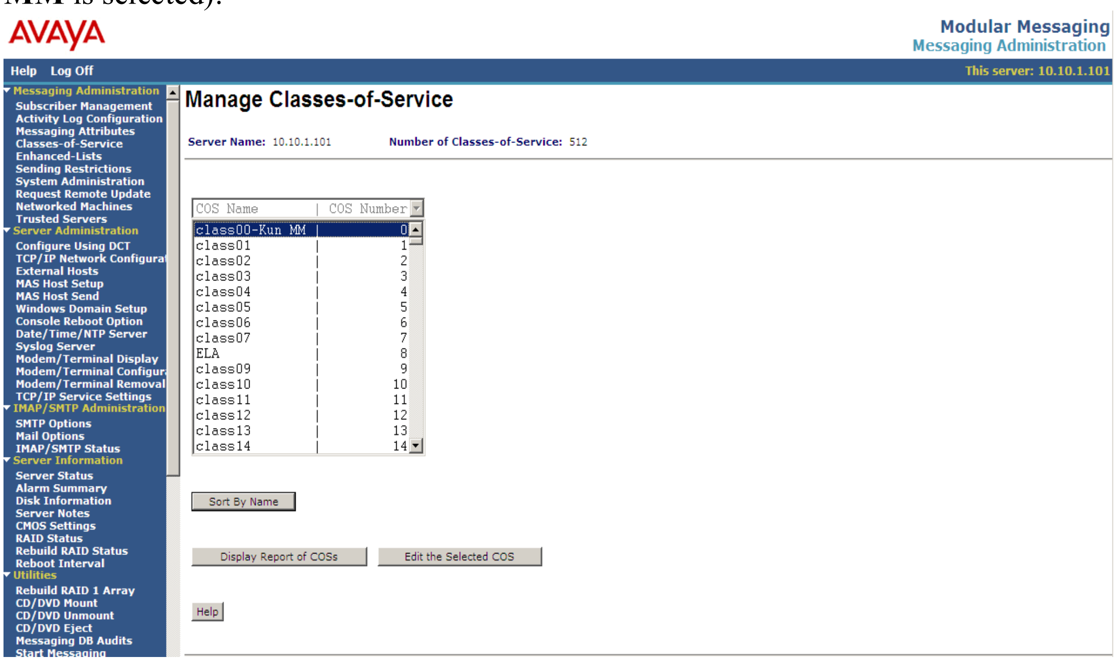
Step	Description
8.	On the **Voice Mail System Configuration - MASSIL** screen expand **Message Application Servers** and the host name of the Avaya Modular Messaging Application Server to be configured (MASSIL in the screenshot below). Double-click on **PBX Type**.

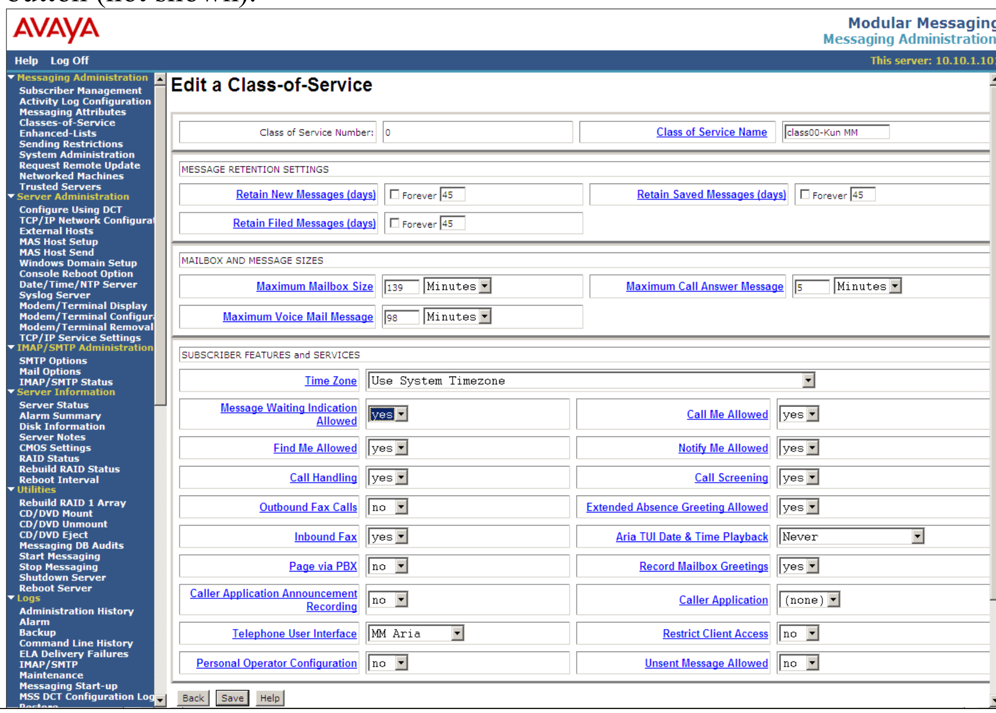
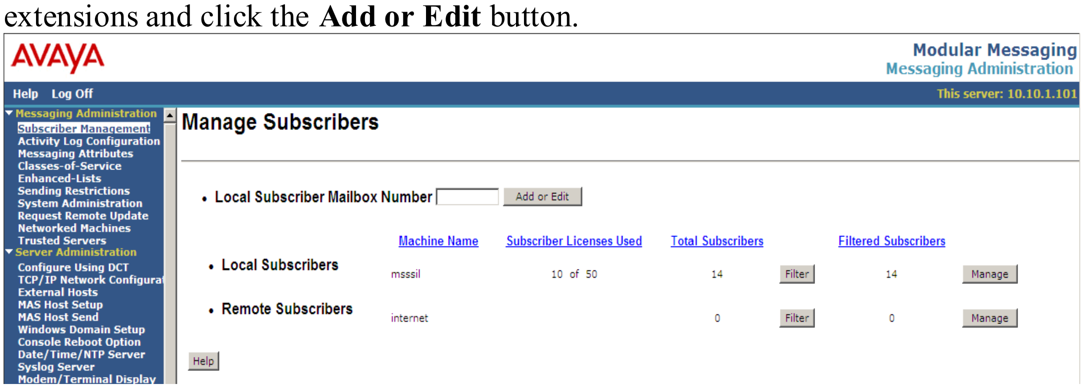


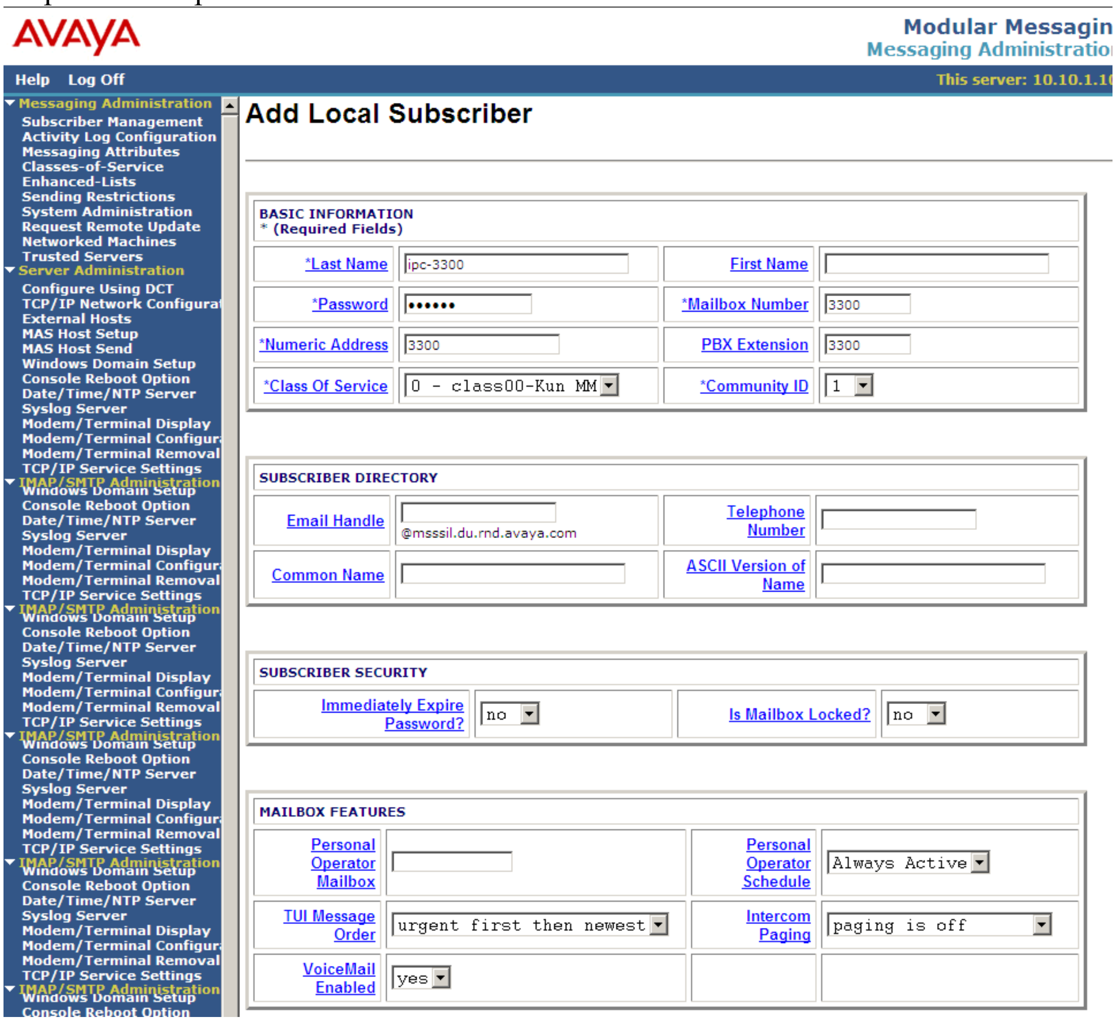
On the **PBX Type** screen select **IP SIP** from the **Telephony Type** drop down box then select **Avaya CM (IP SIP)** from the **PBXs** box. Select **OK** when completed.



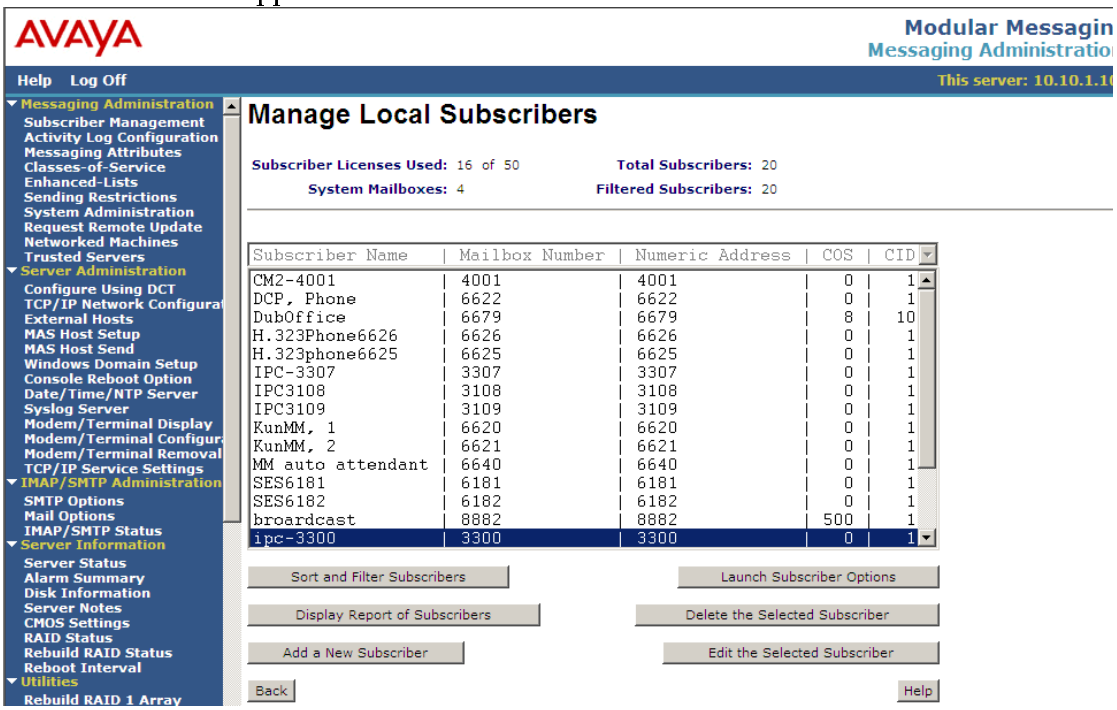
6.2. Configure Message Storage Server

Step	Description
1.	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. 
2.	Select **Messaging Administration → Classes-of-Service** from the left pane. 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-Kun MM** is selected). 

Step	Description
3.	Click **Edit the Selected COS** button on **Step 2**. 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 button of the screen and click the **Save** button (not shown). 
4.	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 turrets and Avaya extensions and click the **Add or Edit** button. 

Step	Description
5.	In the **Add Local Subscriber** screen (see below), fill in the required fields, in this example, IPC extension 3300 is used: • **Last Name**: Enter values appropriate for this user. • **Password**: Enter a default password for accessing the subscriber's mailbox, from one to 15 digits. • **Mailbox Number**: Enter a number, from 2 to 10 digits in length, which uniquely identifies the mailbox for the purpose of logging in or addressing messages. It must be within the range of Mailbox Numbers assigned to this system and be a valid length on the local machine. • **Numeric Address**: Enter a unique address in the voice mail network. • **Class of Service**: Select the Class of Service modified in **Step 3**. • **VoiceMail Enabled**: verify it is set to **yes**. Repeat this step for all desired IPC extensions. 

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



7. General Test Approach and Test Results

All feature and serviceability test cases were performed manually. All test cases were executed and a number of observations were made. These observations and the impact they have on the interoperability of the systems can be viewed in the test schedule document in reference [3].

8. Verification Steps

The following steps can be used to verify that Avaya Communication Manager and Avaya Modular Messaging are configured correctly to support the System Interconnect through SIP.

Step	Description

1.	In Avaya Communication Manager, to verify that either of the trunk groups is up, use the **status trunk *n*** command, where ***n*** is the number of the trunk group. (Refers to **Section 4.1** for trunk details). Verify for each trunk, that **Service State** shows in-service/idle on an idle system.
	`status trunk 6                                          Page   1`
	`TRUNK GROUP STATUS`
	`Member    Port      Service State      Mtce Connected Ports`
	`                                       Busy`
	`0006/001 T00169   in-service/idle    no`
	`0006/002 T00170   in-service/idle    no`
	`0006/003 T00171   in-service/idle    no`
	`0006/004 T00172   in-service/idle    no`
	`0006/005 T00173   in-service/idle    no`
	`0006/006 T00174   in-service/idle    no`
	`0006/007 T00175   in-service/idle    no`
	`0006/008 T00176   in-service/idle    no`
	`0006/009 T00177   in-service/idle    no`
	`0006/010 T00178   in-service/idle    no`
	`0006/011 T00179   in-service/idle    no`
	`0006/012 T00180   in-service/idle    no`
	`0006/013 T00181   in-service/idle    no`
	`0006/014 T00182   in-service/idle    no`

2.	To verify end-to-end connectivity and configuration, call forward one of the IPC turrets to voicemail, then call the IPC turret from an Avaya Communication Manager station. The call should route to the mailbox of the IPC turret. The turret should be able to call from its extension which is recognized as a local subscriber to retrieve the message successfully.

9. Conclusion

The Application Notes describe how to configure Avaya Communication Manager, Avaya Modular Messaging and Avaya SIP Enablement Services to support System Interconnect for IPC.

10. Additional References

This section references the product documentation relevant to these Application Notes.

[1] *Documentation for Avaya Communication Manager (5.1), Media Gateways and Servers,* January 2008, available at: http://support.avaya.com

[2] *Modular Messaging Release 4.0 with the Avaya MSS – Messaging Application Server (MAS) Administration Guide,* May 2008, available at http://support.avaya.com

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

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.