Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Voice Portal, Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager Access Element and Avaya Aura™ Communication Manager Feature Server with Dialogic® IMG 1010 Gateway - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to enable calls between Avaya Voice Portal and the PSTN through a SIP infrastructure consisting of Avaya Aura™ Session Manager, SIP Endpoints registered with Aura™ Communication Manager Feature Server, H.323 Endpoints registered with Avaya Aura™ Communication Manager and the Dialogic IMG1010 Gateway using SIP trunking. Avaya Voice Portal is a Web services based, speech enabled interactive voice response system that can accept both traditional DTMF touch tone inputs and prerecorded audio files for output, as well as VoiceXML2.0 compliant speech applications to guide callers through call flows. This solution allows Avaya Voice Portal to receive calls from the PSTN and transfer calls to Avaya SIP telephones or the PSTN. The Dialogic IMG1010 Gateway combines signaling and media gateway functions providing PSTN access to SIP-based telephony networks.

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.

©2010 Avaya Inc. All Rights Reserved.

Table of Content

1. Introduction 4
   1.1. Interoperability Compliance Testing 4
   1.2. Support 4
2. Reference Configuration 5
3. Equipment and Software Validated 7
4. Configure Avaya Aura™ Communication Manager Access Element 8
   4.1. Verify Avaya Aura™ Communication Manager License 9
   4.2. Configure IP Node Names 9
   4.3. Verify/List IP Interfaces 10
   4.4. Configure IP Codec Set 10
   4.5. Configure IP network Region 11
   4.6. Administer SIP Trunks with Avaya AuraTM Session Manager 12
      4.6.1. Add SIP Signaling Group for Calls within the Enterprise 12
      4.6.2. Configure a SIP Trunk Group for Calls within the Enterprise 13
      4.6.3. Add SIP Signaling Group for Dialogic IMG1010 Gateway 14
      4.6.4. Configure a SIP Trunk Group for Dialogic IMG1010 Gateway 15
   4.7. Configure Route Patterns 16
   4.8. Configure Public Unknown Numbering 16
   4.9. Administer AAR Analysis 17
   4.10. Administer ARS Analysis 18
   4.11. Save Translations 18
5. Configure Avaya AuraTM Communication Manager Feature Server 19
   5.1. Verify Avaya AuraTM Communication Manager License 19
   5.2. Administer System Parameters Features 20
   5.3. Administer IP Node Names 21
   5.4. Administer IP Network Region and Codec Set 21
   5.5. Administer SIP Trunks with Avaya AuraTM Session Manager 23
      5.5.1. Add SIP Signaling Group for Calls within the Enterprise 23
      5.5.2. Configure a SIP Trunk Group for Calls within the Enterprise 24
      5.5.3. Add SIP Signaling Group for IMG 1010 25
      5.5.4. Configure a SIP Trunk Group for IMG1010 26
   5.6. Configure Route Patterns 27
      5.6.1. Route Pattern for Enterprise Calls 27
   5.7. Administer Private Numbering 27
   5.8. Administer Dial Plan and AAR analysis 28
   5.9. Administer ARS Analysis 29
   5.10. Administer Feature Access Code 29
   5.11. Save Changes 29
6. Configure Avaya AuraTM Session Manager 30
   6.1. Specify SIP Domain 31
   6.2. Add Locations 32
   6.3. Add Adaptations 33
   6.4. Add Host Entries for Voice Portal MPPs 34
   6.5. Add SIP Entities 35
      6.5.1. Adding Avaya AuraTM Communication Manager Access Element SIP Entity 35
      6.5.2. Adding Avaya AuraTM Communication Manager Feature Server SIP Entity 36
      6.5.3. Adding Voice Portal SIP Entity 37

©2010 Avaya Inc. All Rights Reserved.

6.5.4. Adding Dialogic IMG 1010 Gateway SIP Entity 38
6.5.5. Adding Avaya AuraTM Session Manager SIP Entity 39
6.6. Add Entity Links 40
6.7. Add Routing Policies 42
6.8. Add Dial Patterns 44
6.9. Add Avaya AuraTM Session Manager 47
6.10. Add Avaya AuraTM Communication Manager as a Feature Server 48
6.10.1. Create an Application Entry 48
6.10.2. Create a Feature Server Application 50
6.10.3. Create a Feature Server Application Sequence 51
6.10.4. Synchronize Avaya AuraTM Communication Manager Data 52
6.11. Add Users for SIP Phones 52
7. Configure Avaya Aura™ Voice Portal 56
7.1. Configuring a SIP Connection for Avaya AuraTM Session Manager 57
7.2. Add the MPP Servers 58
7.3. Configuring the VoIP Audio Format 59
7.4. Add an ASR Server 60
7.5. Add a TTS Server 60
7.6. Add an Application 61
7.7. Start the MPP Servers 62
8. Dialogic IMG 1010 Configuration 63
8.1. IMG Configuration Name 64
8.2. Profiles 65
8.2.1. DS1 Profiles 65
8.2.2. IP Bearer Profiles 66
8.2.3. SIP Profiles 68
8.2.4. Logical IMG 69
8.2.5. External Network Elements 79
8.3. Routing configuration 81
8.3.1. Channel Groups 81
8.4. Routing 86
8.4.1. Route Entry 87
8.5. Additional configuration 87
9. Verification Steps 88
9.1. Verify Avaya AuraTM Communication Manager Access Element Trunk Status 88
9.2. Verify Avaya AuraTM Communication Manager Feature Server Trunk Status 89
9.3. SIP Monitoring on Avaya AuraTM Session Manager 89
9.4. Voice Portal Monitoring – System Monitor 90
9.5. Voice Portal Monitoring – Port Distribution 91
9.6. Functional Verification 91
10. General Test Approach 92
10.1. Test Results and Remarks 92
11. Conclusion 92
12. Additional References 93
13. APPENDIX 94
13.1. Web application server intro.vxml file 94
13.2. Web application server testblindtransfer.vxml file 96
13.3. Web application server testbridgetransfer.vxml file 98
13.4. Web application server testconsulttransfer.vxml file 100

©2010 Avaya Inc. All Rights Reserved.

1. Introduction

These Application Notes present a sample configuration for a network that uses Avaya Voice Portal and the PSTN through a SIP infrastructure consisting of Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager Access Element and Avaya Aura™ Communication Manager Feature Server, Avaya Voice Portal and the Dialogic® IMG 1010 Gateway. The IMG 1010 Integrated Media Gateway is a carrier-grade VoIP gateway that supports both media and signaling in a single chassis. It provides any-to-any voice network connectivity and can deliver SIP services into legacy PRI, CAS, and SS7 networks, as well as IP-to-IP transcoding and multimedia border element functions, such as SIP mediation for network edge applications. Its compact 1U high-density design features integrated SS7 termination across multiple gateways, GUI-based management, and software licensing for in-service capacity expansion. In this configuration, the IMG 1010 connects to Voice Portal through SIP trunks on Session Manager.

1.1. Interoperability Compliance Testing

The primary focus of testing is to verify SIP trunking interoperability between Voice Portal and Dialogic® IMG 1010 Gateway using SIP. Test cases are selected to exercise a sufficiently broad segment of functionality to have a reasonable expectation of interoperability in production configurations. The feature testing focused on verifying the following:

- Placing calls from the Avaya SIP-based network to the PSTN
- Placing calls from the PSTN to Avaya Voice Portal
- Call transfers from Voice Portal to users on the PSTN and the Avaya SIP-based network
- Performing Blind, Supervised, and Bridged call transfers from Voice Portal to users on the PSTN and Avaya SIP-based network
- Sending UUI during call transfers from Voice Portal to the PSTN
- Receiving UUI from the PSTN to Voice Portal

The serviceability testing focused on verifying the ability of the IMG 1010 to recover from adverse conditions, such as power failures and disconnecting cables from the IP network.

1.2. Support

Technical Support on Dialogic IMG 1010 Gateway can be obtained through the following contacts:

- Phone: +1 781 433 9600
- E-mail: americas.support@dialogic.com

©2010 Avaya Inc. All Rights Reserved.

2. Reference Configuration

As shown in **Figure 1**, the Avaya Voice Portal and enterprise network uses SIP trunking for call signaling internally and with the Dialogic IMG 1010 Gateway in order to access the PSTN. The IMG 1010 is managed by using the Dialogic Inc. GateControl Element Management System (GCEMS) and ClientView running on a Linux server. Session Manager with its SM-100 (Security Module) network interface routes calls between the different entities using SIP Trunks. All inter-system calls are carried over these SIP trunks. Session Manager supports flexible inter-system call routing based on the dialed number, the calling number and the system location; it can also provide protocol adaptation to allow multi-vendor systems to interoperate. Session Manager is managed by System Manager via the management network interface.

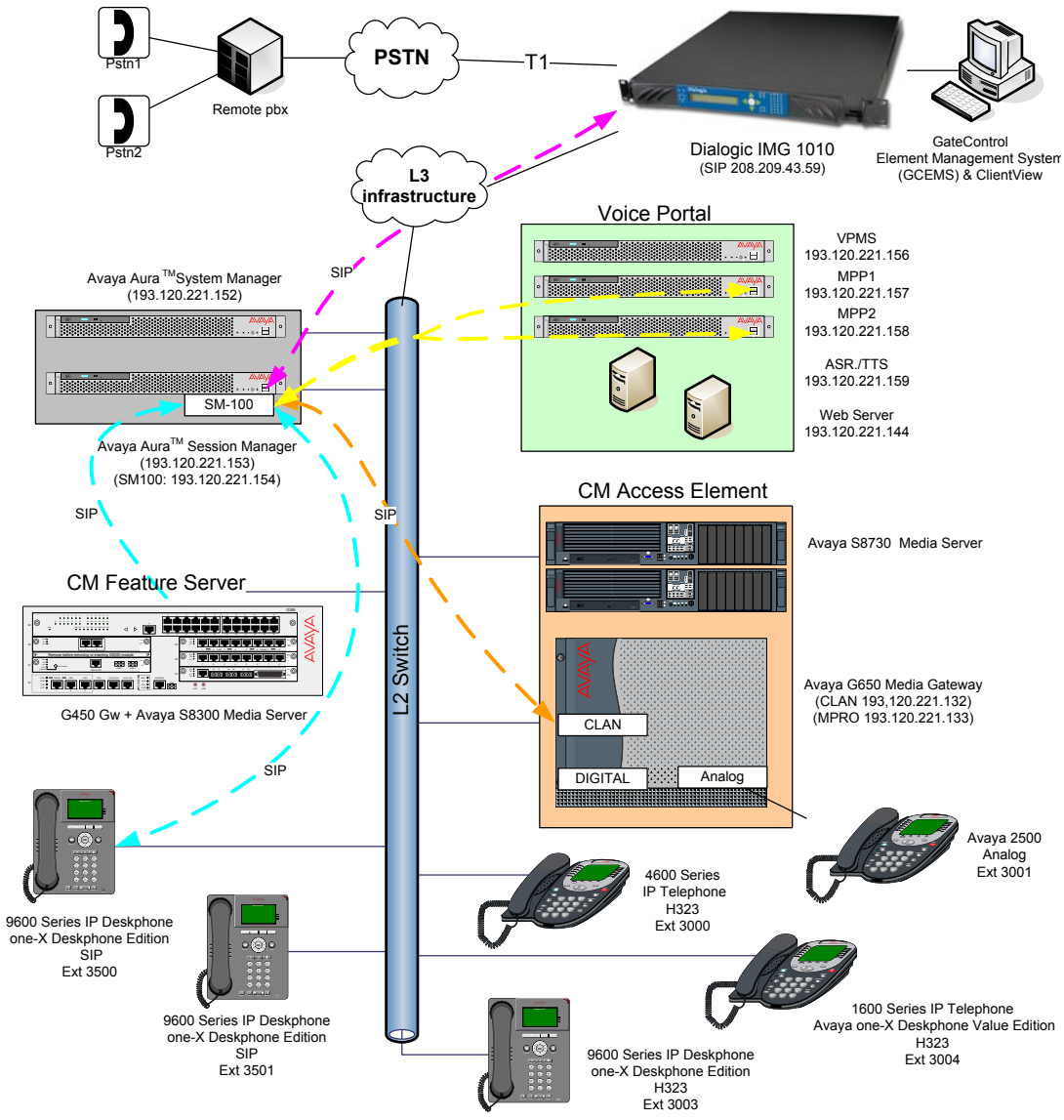


Figure 1 – Sample configuration for Avaya Aura™ Session Manager and Avaya Voice Portal with Dialogic IMG 1010 using Sip Trunking

©2010 Avaya Inc. All Rights Reserved.

For the sample configuration shown in **Figure 1**, Session Manager runs on an Avaya S8510 Server, Communication Manager Access Element runs on an Avaya S8730 Server with an Avaya G650 Media Gateway, and Communication Manager Feature Server runs on an Avaya S8300D inside an Avaya G450 Media Gateway. For the Communication Manager Access Element, the results in these Application Notes are applicable to other Communication Manager Server and Media Gateway combinations. Avaya Voice Portal is comprised of a Voice Portal Management System (VPMS) server, a Media Processing Platform (MPP) server, a Web Application Server, and a Speech Processing server. Voice Portal Management System (VPMS) manages the MPPs and provides a web interface for administering VP. Media Processing Platform (MPP) provides the main processing for self service applications. The Web Application Server utilizes a workflow defined in Dialog Designer to provide the self service application. The MPP calls the application server and coordinates media resources available for processing the call. The Speech Server provides Automatic Speech Recognition (ASR) and Text To Speech (TTS) capabilities. These Application Notes will focus on the configuration of the SIP trunks and call routing. Detailed administration of the endpoint telephones will not be described. Refer to the appropriate documentation in **Section 12**.

©2010 Avaya Inc. All Rights Reserved.

3. Equipment and Software Validated

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

Avaya Product / Hardware Platform	Software Version
Avaya Aura ™ Session Manager on Avaya S8510 Server	Avaya Aura ™ Session Manager 5.2 5.2.1.1.521012 – 5.2.1 SP1
Avaya Aura ™ System Manager Template running on Avaya System Platform	Avaya Aura ™ System Manager 5.2 5.2.1.0.521001 - 05_02_GA_01_Dec10
Avaya Aura ™ System Platform on Avaya S8510 Server	Avaya Aura ™ System Platform Version 1.1.1.0.2
Avaya Aura ™ Communication Manager - Access Element – Avaya Media Server S8730	Avaya Aura ™ Communication Manager 5.2.1 R015x.02.1.016.4 – patch 17959
Avaya Aura ™ Communication Manager – Feature Server – Avaya Media Server S8300C	Avaya Aura ™ Communication Manager 5.2.1 R015x.02.1.016.4 – patch 17959
Avaya Media Gateway G450	Firmware 30 .11 .3
Avaya G650 Media Gateway • IPSI (TN2312BP) • C-LAN (TN799DP) • IP Media Resource 320 (TN2602AP) • Analog (TN2793B) • Digital line (TN2214CP)	 • TN2312BP HW28 FW050 • TN799DP HW01 FW037 • TN2602AP HW08 FW053 • TN2793B 000005 • TN2214CP HW10 FW015
Avaya Voice Portal VPMS on Avaya S8510 Server	Avaya Voice Portal 5.0 Service Pack 2 (5.0.0.2.0104)
Avaya Voice Portal MPP on Avaya S8510 Server	Avaya Voice Portal 5.0 Service Pack 2 (5.0.0.2.01045.0 SP2 (5.0.0.2.0104)
Automatic Speech Recognition and Text To Speech	Nuance RealSpeak 4.5
Avaya IP Telephones: • 9630 & 9620 (SIP) • 9620 (H323) • 1616 (H323) • 4621 (H323) • Avaya Digital Telephones (2420) • Avaya Analog (2500)	 • Avaya one-X™ Deskphone SIP 2.5.0 • Avaya one-X™ Deskphone S3.1 • Release 1.2.2 • Release R2.9 SP1 • N/A • N/A
Dialogic Inc.	
Product /Hardware Platform	**Software Version**
Dialogic® IMG 1010 Integrated Media Gateway	Dialogic® IMG System Software 10.5.3 BUILD 119
Dialogic® Gate Control Element Management System	GCEMS 10.5.3 BUILD 94

©2010 Avaya Inc. All Rights Reserved.

4. Configure Avaya Aura™ Communication Manager Access Element

This section provides the procedures for configuring Communication Manager as an Access Element. The procedures include the following areas:

- Verify Avaya Aura™ Communication Manager License
- Configure IP Node Names
- Verify/List IP Interfaces
- Configure IP Codec Set
- Configure IP Network Region
- Administer SIP Trunks with Session Manager
- Configure Route Pattern
- Configure Public Unknown Numbering
- Administer AAR Analysis
- Administer ARS Analysis
- Save Translations

Throughout this section the administration of Communication Manager is performed using a System Access Terminal (SAT). The following commands are entered on the system with the appropriate administrative permissions. Some administration screens have been abbreviated for clarity. These instructions assume that Communication Manager has been installed, configured, licensed and provided with a functional dial plan. Refer to the appropriate documentation as described in **Reference [1]** and **[2]** for more details**.** In these Application Notes, Communication Manager was configured with 4 digit extention **30xx** for stations, SIP endpoints administrated by Session Manager **35xx** reacheble with **aar**. Avaya Voice Portal applications are accessible as **36xx** reacheable with **aar**. Diaplan analysis can be verified with the **display dialplan analysis** command.

```
display 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
    30           4    ext
    35           4    aar
    36           4    aar
    8            3    dac
    9            1    fac
```

Other numbers on PSTN (accessible from the IMG 1010 Gateway) are reachable via **ars** table with the use of **feature access code 9**.

©2010 Avaya Inc. All Rights Reserved.

4.1. Verify Avaya Aura™ Communication Manager License

Use the **display system-parameters customer-options** command. Navigate to **Page 2** and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections. Verify highlighted value, as shown below.

```
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: 18000 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: 100   0
                       Maximum Video Capable Stations: 100   0
                   Maximum Video Capable IP Softphones: 100   9
                       Maximum Administered SIP Trunks: 1000  300
```

If there is insufficient capacity of SIP Trunks or a required feature is not enabled, contact an authorized Avaya sales representative to make the appropriate changes.

4.2. Configure IP Node Names

All calls from and to Communication Manager are signalled over SIP trunks with Session Manager. The signalling interface on Session Manager is provided by the SM100 security module, therefore in configuring SIP trunks its required having the SM100 IP interfacace in the **node-names** table. Use the **change node-names ip** command to add the **Name** and **IP Address** for the Session Manager. **SM100** and **193.120.221.154** was used in this example.

```
change node-names ip                                          Page   1 of   2
                                  IP NODE NAMES
    Name                  IP Address
Gateway001              193.120.221.129
SM100                   193.120.221.154
clan                    193.120.221.132
default                 0.0.0.0
mpro                    193.120.221.133
procr                   0.0.0.0
```

Note: In the example, some other values (CLAN, MedPro) have been already created as per installation and configuration of Communication Manager.

©2010 Avaya Inc. All Rights Reserved.

4.3. Verify/List IP Interfaces

Use the **list ip-interface all** command and note the **C-LAN** to be used for SIP trunks between Communication Manager and Session Manager.

```
list ip-interface all
                                IP INTERFACES
                                                                    Net
ON Type   Slot  Code/Sfx    Node Name/       Mask  Gateway Node     Rgn VLAN
                           IP-Address
-- ------ ----- --------    ---------------  ----  ---------------  ---  ----
 y C-LAN  01A02 TN799  D     clan             /25   Gateway001       1    n
                            193.120.221.132
 y MEDPRO 01A03 TN2602       mpro             /25   Gateway001       1    n
                            193.120.221.133
```

4.4. Configure IP Codec Set

Use the **change ip-codec-set n** command where **n** is the codec set used in the configuration. The ITU G.711A-law is described here. Configure the IP Codec Set as it follows:

- **Audio Codec** Set **G.711A**

Retain the default values for the remaining fields.

```
change ip-codec-set 1                                        Page   1 of   2
                          IP Codec Set

    Codec Set: 1

    Audio         Silence      Frames   Packet
    Codec         Suppression  Per Pkt  Size(ms)
 1: G.711A            n          2        20
 2:
 3:
```


©2010 Avaya Inc. All Rights Reserved.

4.5. Configure IP network Region

Use the **change ip-network-region n** command where **n** is the number of the network region used. Set the **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** fields to **yes**. For the **Codec Set,** enter the corresponding audio codec set configured in **Section 4.4**. Set the **Authoritative Domain** to the SIP domain. Retain the default values for the remaining fields, and submit these changes.

Note: In the test configuration, **network region 1** was used. If a new network region is needed or an existing one is modified, ensure to configure it with the correct parameters.

```
change ip-network-region 1                                    Page   1 of  19
                              IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name: Enterprise
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
```


©2010 Avaya Inc. All Rights Reserved.

4.6. Administer SIP Trunks with Avaya Aura™ Session Manager

Two SIP trunks are needed for the configuration presented in these notes: one for calls within the Enterprise and another one for calls with Dialogic IMG Gateway. To administer a SIP Trunk on Communication Manger, two intermediate steps are required, creation of a signaling group and trunk group.

4.6.1. Add SIP Signaling Group for Calls within the Enterprise

Use the **add signaling-group n** command, where **n** is an available signaling group number, for one of the SIP trunks to the Session Manager, and fill in the indicated fields. Default values can be used for the remaining fields:

- **Group Type:** **sip**
- **Transport Method:** **tls**
- **Near-end Node Name:** C-LAN node name from **Section 4.2** (i.e., **clan**).
- **Far-end Node Name:** Session Manager node name from **Section 4.2** (i.e. **SM100**).
- **Near-end Listen Port:** **5061**
- **Far-end Listen Port:** **5061**
- **Far-end Domain:** **avaya.com**
- **DTMF over IP:** **rtp-payload**

Submit these changes.

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

 Group Number: 3              Group Type: sip
                         Transport Method: tls
  IMS Enabled? n
     IP Video? n

   Near-end Node Name: clan                 Far-end Node Name: SM100
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1
Far-end Domain: 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? n
         Enable Layer 3 Test? n                   Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 6
```


©2010 Avaya Inc. All Rights Reserved.

4.6.2. Configure a SIP Trunk Group for Calls within the Enterprise

Add the corresponding trunk group controlled by this signaling group via the **add trunk-group n** command, where **n** is an available trunk group number and fill in the indicated fields.

- **Group Type:** **sip**
- **Group Name:** A descriptive name (i.e**. to AuraSM**)
- **TAC:** An available trunk access code (i.e. **803**)
- **Service Type:** **tie**
- **Signaling Group:** The number of the signaling for outbound calls (i.e. **3**)
- **Number of Members:** The number of SIP trunks to be allocated to calls routed to **Session Manager (**must be within the limits of the total **trunks available** from licensed verified in **Section 4.1**)

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

Group Number: 3                     Group Type: sip           CDR Reports: y
  Group Name: To AuraSM                      COR: 1       TN: 1      TAC: 803
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                  Signaling Group: 3
                                                Number of Members: 30
```

Navigate to **Page 3** and change **Numbering Format** to **public.** Use default values for all other fields.

```
add trunk-group 3                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y
                      Numbering Format: public
                                               UUI Treatment: service-provider
                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
```


©2010 Avaya Inc. All Rights Reserved.

4.6.3. Add SIP Signaling Group for Dialogic IMG1010 Gateway

To accept inbound calls from the IMG 1010 Gateway, it is necessary to configure a sip signalling group. Use the **add signaling-group n** command, where **n** is an available signaling group number, for one of the SIP trunks to the Session Manager, and fill in the indicated fields. Default values can be used for the remaining fields:

- **Group Type:** **sip**
- **Transport Method:** **tls**
- **Near-end Node Name:** C-LAN node name from **Section 4.2** (i.e. **clan**)
- **Far-end Node Name:** Session Manager node name from **Section 4.2** (i.e. **SM100**)
- **Near-end Listen Port:** **5061**
- **Far-end Listen Port:** **5061**
- **Far-end Domain:** Leave it blank
- **DTMF over IP:** **rtp-payload**

```
add signaling-group 2                                          Page   1 of   1
                                SIGNALING GROUP
Group Number: 2                Group Type: sip
                           Transport Method: tls
  IMS Enabled? n
     IP Video? n

   Near-end Node Name: clan                 Far-end Node Name: SM100
 Near-end Listen Port: 5061               Far-end Listen Port: 5061
                                       Far-end Network Region: 1
Far-end Domain:
                                      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? n
         Enable Layer 3 Test? n                 Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n        Alternate Route Timer(sec): 15
```


©2010 Avaya Inc. All Rights Reserved.

4.6.4. Configure a SIP Trunk Group for Dialogic IMG1010 Gateway

Add the corresponding trunk group controlled by this signaling group via the **add trunk-group n** command, where **n** is an available trunk group number and fill in the indicated fields.

- **Group Type:** **sip**
- **Group Name:** A descriptive name (i.e**. GWInbound)**
- **TAC:** An available trunk access code (i.e. **804**)
- **Service Type:** **tie**
- **Signaling Group:** Number of the signaling group added in **Section 4.6.3** (i.e. **4**)
- **Number of Members:** The number of SIP trunks to be allocated to calls routed to **Session Manager (**must be within the limits of the total trunks available from licensed verified in **Section 4.1**)

Note: The number of members determines how many simulataneous calls can be processed by the trunk through Session Manager.

```
add trunk-group 4                                          Page   1 of  21
                                TRUNK GROUP

Group Number: 4                     Group Type: sip           CDR Reports: y
  Group Name: GWInbound                 COR: 1        TN: 1        TAC: 804
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n

                                                        Signaling Group: 4
                                                      Number of Members: 10
```

Navigate to **Page 3** and change **Numbering Format** to **public.** Use default values for all other fields. Submit these changes.

```
add trunk-group 4                                            Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y

                    Numbering Format: public
                                              UUI Treatment: service-provider

                                               Replace Restricted Numbers? n
                                              Replace Unavailable Numbers? n
```


©2010 Avaya Inc. All Rights Reserved.

4.7. Configure Route Patterns

Configure two route patterns to correspond to the newly added SIP trunk groups. Use **change route pattern n** command, where **n** is an available route pattern. When changing the route pattern, enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Pattern Name:** A descriptive name (i.e., **toSessionManager**)
- **Grp No:** The trunk group number from **Section 4.6.2**
- **FRL:** Enter a level that allows access to this trunk, with **0** being least restrictive

```
change route-pattern 3                                          Page   1 of   3
                    Pattern Number: 3   Pattern Name: toSessionManager
                                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

     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            unre                                    none
 2: y y y y y n  n            rest                                    none
```

4.8. Configure Public Unknown Numbering

Use the **change public-unknown-numbering 0** command to assign number presented by Communication Manager for calls leaving Session Manager. Add an entry for the Extensions configured in the dialplan. Enter the following values for the specified fields, and retain default values for the remaining fields. Submit these changes.

- **Ext Len:** Number of digits of the Extention i.e. **4**
- **Ext. Code:** Digits beginning the Extention number, i.e. **30**
- **Trk Group:** Leave it blank (meaning any trunk)
- **CPN Prefix:** Leave it blank
- **Total CPN Len** Number of digits i.e. **4**

```
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  30                                     4          Maximum Entries: 9999
```


©2010 Avaya Inc. All Rights Reserved.

4.9. Administer AAR Analysis

This section provides sample Automatic Alternate Routing (AAR) used for routing calls with dialed digits **35xx** corresponding to SIP endpoint registered on Session Manager and with **36xx** corresponding to interactive voice application residing on Voice Portal. Use the **change aar analysis 0** command and add an entry to specify how to route calls to **35xx** and **36xx**. Enter the following values for the specified fields and retain the default values for the remaining fields. Submit these changes.

- **Dialed String:** Dialed prefix digits to match on, in this case **35**
- **Total Min:** Minimum number of digits, in this case **4**
- **Total Max:** Maximum number of digits, in this case **4**
- **Route Pattern:** The route pattern number from **Section 4.7** i.e. **3**
- **Call Type:** **aar**

Create another entry for routing calls to Voice Portal i.e. beginning with **36** with **4**-digits length.

```
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

      35              4    4     3       aar
      36              4    4     3       aar
```


©2010 Avaya Inc. All Rights Reserved.

4.10. Administer ARS Analysis

This section provides sample Auto Route Selection (ARS) used for routing calls with dialed digits beginning with **0** corresponding to national numbers accessible via the IMG 1010. Use the **change ars analysis 0** command and add an entry to specify how to route calls. Enter the following values for the specified fields and retain the default values for the remaining fields. Submit these changes.

- **Dialed String:** Dialed prefix digits to match on, in this case **0**
- **Total Min:** Minimum number of digits, in this case **3**
- **Total Max:** Maximum number of digits, in this case **25**
- **Route Pattern:** The route pattern number from **Section 4.7** i.e. **3**
- **Call Type:** **pubu**

Note that additional entries may be added for different number destinations.

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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
             0            3    25    3        pubu         n
```

4.11. Save Translations

Configuration of Communication Manager is complete. Use the **save translations** command to save these changes.

```
save translation
                              SAVE TRANSLATION
           Command Completion Status                              Error Code
           Success                                                  0
```


©2010 Avaya Inc. All Rights Reserved.

5. Configure Avaya Aura™ Communication Manager Feature Server

This section shows the configuration in Communication Manager. All configurations in this section are administered using the System Access Terminal (SAT). These Application Notes assumed that the basic configuration has already been administered. For further information on Communication Manager, please consult with **References [10]** and **[13].** The procedures include the following areas:

- Verify Avaya Aura™ Communication Manager License
- Administer System Parameters Features
- Administer IP Node Names
- Administer IP Network Region and Codec set
- Administer SIP Signalling Group and Trunk Group
- Administer Route Pattern
- Administer Private Numbering
- Administer Dial Plan and AAR analysis
- Administer ARS analysis
- Administer Feature Access Codes
- Save Changes

5.1. Verify Avaya Aura™ Communication Manager License

Use the **display system-parameter customer options** command to verify whether the **Maximum Administered SIP Trunks** field value with the corresponding value in the **used** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections.

Note: The license file installed on the system controls the maximum features permitted. If there is insufficient capacity or a required feature is not enabled, contact an authorized Avaya sales representative to make the appropriate changes.

```
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: 450   0
            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: 100   0
                       Maximum Video Capable Stations: 100   0
                  Maximum Video Capable IP Softphones: 100   0
                      Maximum Administered SIP Trunks: 100   50
```


©2010 Avaya Inc. All Rights Reserved.

5.2. Administer System Parameters Features

Use the **change system-parameters features** command to allow for trunk-to-trunk transfers. This feature is needed to allow for transferring an incoming/outgoing call from/to a remote switch back out to the same or different switch. For simplicity, the **Trunk-to-Trunk Transfer** field was set to **all** to enable all trunk-to-trunk transfers on a system wide basis.

Note: This feature poses significant security risk and must be used with caution. As an alternative, the trunk-to-trunk feature can be implemented using Class Of Restriction or Class Of Service levels.

```
change 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
```


©2010 Avaya Inc. All Rights Reserved.

5.3. Administer IP Node Names

Use the **change node-names ip** command to add entries for Communication Manager and Session Manager that will be used for connectivity. In the sample network, the processor Ethernet interface **procr** and **193.120.221.180** are entered as **name** and **IP Address** for the signalling in Communication Manager running on the Avaya S8300 Server. In addition, **SM100** and **193.120.221.154** are entered for Session Manager.

```
change node-names ip                                        Page   1 of   2
                                IP NODE NAMES
    Name               IP Address

default              0.0.0.0
procr                193.120.221.180
sm100                193.120.221.154
```

5.4. Administer IP Network Region and Codec Set

Use the **change ip-network-region n** command, where **n** is the network region number to configure the network region being used. In the sample network ip-network-region 1 is used. For the **Authoritative Domain** field, enter the SIP domain name configured for this enterprise and a descriptive **Name** for this ip-network-region. Set **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** to **yes** to allow for direct media between endpoints. Set the **Codec Set** to **1** to use ip-codec-set 1.

```
change ip-network-region 1                                   Page   1 of  19
                             IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: avaya.com
    Name: Enterprise
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
```


©2010 Avaya Inc. All Rights Reserved.

Use the **change ip-codec-set n** command where **n** is codec set used in the configuration The ITU G.711A-law is described here. Configure the IP Codec Set as it follows:

- **Audio Codec:** Set **G.711A**

Retain the default values for the remaining fields.

```
change ip-codec-set 1                                          Page   1 of   2
                           IP Codec Set

    Codec Set: 1

    Audio          Silence      Frames   Packet
    Codec          Suppression  Per Pkt  Size(ms)
 1: G.711A             n          2        20
 2:
 3:
```


©2010 Avaya Inc. All Rights Reserved.

5.5. Administer SIP Trunks with Avaya Aura™ Session Manager

In the test configuration, Communication Manager acts as a Feature Server in this case, trunks with Session Manager must be IMS enabled. Two SIP trunks are needed for the configuration presented in these notes: one for calls with IMG1010 and another one for calls within the Enterprise. To administer a SIP Trunk on Communication Manger, two intermediate steps are required, creation of a signaling group and trunk group

5.5.1. Add SIP Signaling Group for Calls within the Enterprise

Use the **add signaling-group n** command, where **n** is an available signaling group number, for one of the SIP trunks to the Session Manager, and fill in the indicated fields. Default values can be used for the remaining fields:

- **Group Type:** **sip**
- **Transport Method:** **tls**
- **IMS Enabled:** **y**
- **Near-end Node Name:** **procr**
- **Far-end Node Name:** Session Manager node name from **Section 5.3** (i.e. **sm100**)
- **Near-end Listen Port:** **5061**
- **Far-end Listen Port:** **5061**
- **Far-end Domain:** **avaya.com**
- **DTMF over IP:** **rtp-payload**

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

 Group Number: 1              Group Type: sip
                         Transport Method: tls
  IMS Enabled? y
     IP Video? n


   Near-end Node Name: procr                 Far-end Node Name: sm100
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                         Far-end Network Region: 1
Far-end Domain: 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? n
         Enable Layer 3 Test? n                 Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n       Alternate Route Timer(sec): 30
```


©2010 Avaya Inc. All Rights Reserved.

5.5.2. Configure a SIP Trunk Group for Calls within the Enterprise

Add the corresponding trunk group controlled by this signaling group via the **add trunk-group n** command, where **n** is an available trunk group number and fill in the indicated fields.

- **Group Type:** **sip**
- **Group Name:** A descriptive name (i.e**. with-SessionManager**)
- **TAC:** An available trunk access code (i.e. **101**)
- **Service Type:** **tie**
- **Signaling Group:** The number of the signaling group associated (i.e. **1**)
- **Number of Members:** The number of SIP trunks to be allocated to calls routed to **Session Manager (**must be within the limits of the total **trunks available** from licensed verified in **Section 5.1**)

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

Group Number: 1                     Group Type: sip           CDR Reports: y
  Group Name: with-SessionManager        COR: 1        TN: 1        TAC: 101
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                        Signaling Group: 1
                                                      Number of Members: 20
```

Navigate to **Page 3** and change **Numbering Format** to **private.** Use default values for all other fields.

```
add trunk-group 1                                              Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                        Maintenance Tests? y
                      Numbering Format: private
                                              UUI Treatment: service-provider
                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n
```


©2010 Avaya Inc. All Rights Reserved.

5.5.3. Add SIP Signaling Group for IMG 1010

Use the **add signaling-group n** command, where **n** is an available signaling group number, for one of the SIP trunks to the Session Manager, and fill in the indicated fields. Default values can be used for the remaining fields:

- **Group Type:** **sip**
- **Transport Method:** **tls**
- **IMS Enabled:** **y**
- **Near-end Node Name:** **procr**
- **Far-end Node Name:** Session Manager node name from **Section 5.3** (i.e. **sm100**)
- **Near-end Listen Port:** **5061**
- **Far-end Listen Port:** **5061**
- **Far-end Domain:** Leave it blank
- **DTMF over IP:** **rtp-payload**

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

 Group Number: 3                Group Type: sip
                          Transport Method: tls
  IMS Enabled? y
     IP Video? n


   Near-end Node Name: procr                 Far-end Node Name: sm100
 Near-end Listen Port: 5061                Far-end Listen Port: 5061
                                         Far-end Network Region: 1
Far-end Domain:

                                           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? n
         Enable Layer 3 Test? n                 Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n       Alternate Route Timer(sec): 30
```


©2010 Avaya Inc. All Rights Reserved.

5.5.4. Configure a SIP Trunk Group for IMG1010

Add the corresponding trunk group controlled by this signaling group via the **add trunk-group n** command, where **n** is an available trunk group number and fill in the indicated fields.

- **Group Type:** **sip**
- **Group Name:** A descriptive name (i.e**. OUTSIDE CALL**)
- **TAC:** An available trunk access code (i.e. **102**)
- **Service Type:** **tie**
- **Signaling Group:** The number of the signaling group associated (i.e. **2**)
- **Number of Members:** The number of SIP trunks to be allocated to calls routed to **Session Manager (**must be within the limits of the total **trunks available** from licensed verified in **Section 5.1**)

Note: The number of members determines how many simulataneous calls can be processed by the trunk through Session Manager.

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

Group Number: 3                    Group Type: sip           CDR Reports: y
  Group Name: OUTSIDE CALL                    COR: 1       TN: 1      TAC: 103
   Direction: two-way        Outgoing Display? n
 Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                  Signaling Group: 3
                                                Number of Members: 30
```

Navigate to **Page 3** and change **Numbering Format** to **private.** Use default values for all other fields. Submit these changes.

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

                     Numbering Format: private
                                              UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n
```


©2010 Avaya Inc. All Rights Reserved.

5.6. Configure Route Patterns

Configure two route patterns to correspond to the newly added SIP trunk groups Use **change route pattern n** command, where **n** is an available route pattern.

5.6.1. Route Pattern for Enterprise Calls

When changing the route pattern, enter the following values for the specified fields, and retain the default values for the remaining fields. Submit these changes.

- **Pattern Name:** A descriptive name (i.e. **toSessionManager**)
- **Grp No:** The trunk group number from **Section 5.5.2**
- **FRL:** Enter a level that allows access to this trunk, with **0** being least restrictive

```
change route-pattern 1                                          Page   1 of   3
                   Pattern Number: 1   Pattern Name: toSessionManager
                            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

     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            unre                                    none
 2: y y y y y n  n            rest                                    none
none
```

5.7. Administer Private Numbering

Use the **change private-numbering** command to define the calling party number to be sent out through the SIP trunk. In the sample network configuration below, all calls originating from a **4**-digit extension (**Ext Len**) beginning with **35** (**Ext Code**) will result in a **4**-digit calling number (**Total Len**). The calling party number will be in the SIP "From" header.

```
change private-numbering 0
                           NUMBERING - PRIVATE FORMAT

Ext Ext            Trk        Private          Total
Len Code           Grp(s)     Prefix           Len
 4  35                                        4     Total Administered: 1
                                                     Maximum Entries: 540
```


©2010 Avaya Inc. All Rights Reserved.

5.8. Administer Dial Plan and AAR analysis

Configure the dial plan for dialing **4**-digit extensions beginning with **30** to stations registered with Communication Manager Access Element and **4**-digit extensions beginning with **36** accessible as application on Voice Portal. Use the **change dialplan analysis** command to define **Dialed String 350** as an **aar Call Type**.

```
change dialplan analysis                                      Page   1 of  12
                            DIAL PLAN ANALYSIS TABLE
                                 Location:  all           Percent Full:    2

       Dialed   Total  Call    Dialed   Total  Call     Dialed   Total  Call
       String   Length Type    String   Length Type     String   Length Type
     1            3    dac
     30           4    aar
     35           4    ext
     36           4    aar
     9            1    fac
     *            1    fac
```

Use the **change aar analysis n** command where **n** is the dial string pattern to configure an **aar** entry for **Dialed String 30** (Extensions on Communication Manager Access Element**)** to use **Route Pattern 1** (defined in **Section 5.6.1**), configure another entry for Dialed String 36 for applications on Voice Portal.

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

          Dialed           Total      Route    Call   Node  ANI
          String          Min  Max   Pattern   Type   Num   Reqd
    30                    4    4     1         aar          n
    35                    4    4     1         aar          n
    36                    4    4     1         aar          n
```


©2010 Avaya Inc. All Rights Reserved.

5.9. Administer ARS Analysis

This section provides sample Auto Route Selection (ARS) used for routing calls with dialed digits beginning with **0** corresponding to national numbers accessible via the IMG 1010. Use the **change ars analysis 0** command and add an entry to specify how to route calls. Enter the following values for the specified fields and retain the default values for the remaining fields. Submit these changes.

- **Dialed String:** Dialed prefix digits to match on, in this case **0**
- **Total Min:** Minimum number of digits, in this case **3**
- **Total Max:** Maximum number of digits, in this case **25**
- **Route Pattern:** The route pattern number from **Section 5.6.1** i.e. **1**
- **Call Type:** **pubu**

Note that additional entries may be added for different number destinations.

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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
    0                     3    25    1        pubu         n
```

5.10. Administer Feature Access Code

Configure a feature access code to use for AAR routing. Use the **change feature access code** command to define an **Auto Alternate Routing (AAR) Access Code** and for **Auto Route Selection (ARS).** In these notes, **9** and ***** were used.

```
change feature-access-codes                                  Page   1 of   8
                             FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: 9
    Auto Route Selection (ARS) - Access Code 1: *     Access Code 2:
                 Automatic Callback Activation:        Deactivation:
```

5.11. Save Changes

Use the **save translation** command to save all changes.

```
save translation
                               SAVE TRANSLATION
           Command Completion Status                              Error Code
           Success                                                  0
```

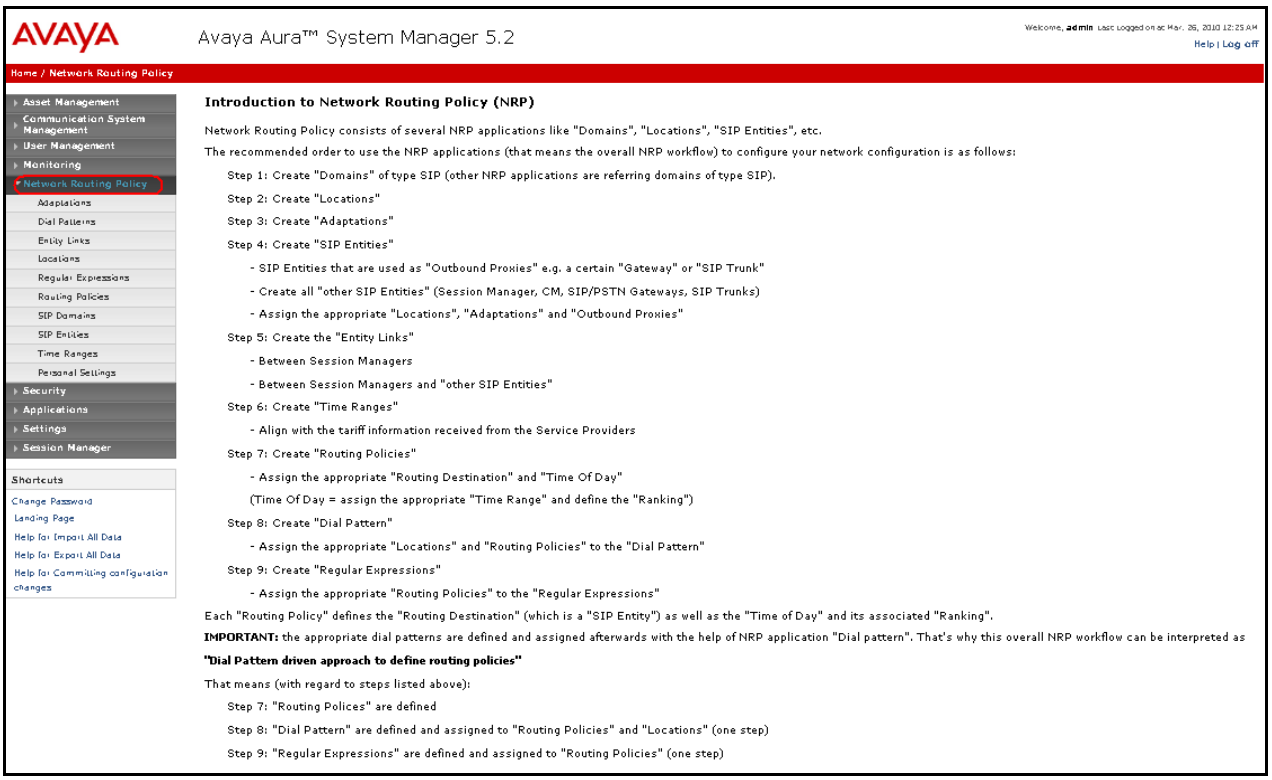

©2010 Avaya Inc. All Rights Reserved.

6. Configure Avaya Aura™ Session Manager

This section provides the procedures for configuring Session Manager, assuming it has been installed and licensed as described in **Reference [3]**. The procedures include adding the following items:

- Specify SIP Domain
- Add Locations
- Add Adaptations
- Add Local Host Entries for Voice Portal MPPs
- Add SIP Entities
- Add Entity Links
- Add Routing Policies
- Add Dial Patterns
- Add Session Manager
- Add Communication Manager as Feature Server
- Add Users for Sip Phones

Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR**, where **<ip-address>** is the IP address of System Manager. Log in with the appropriate credentials and accept the Copyright Notice. The menu shown below is displayed. Expand the **Network Routing Policy** Link on the left side as shown.

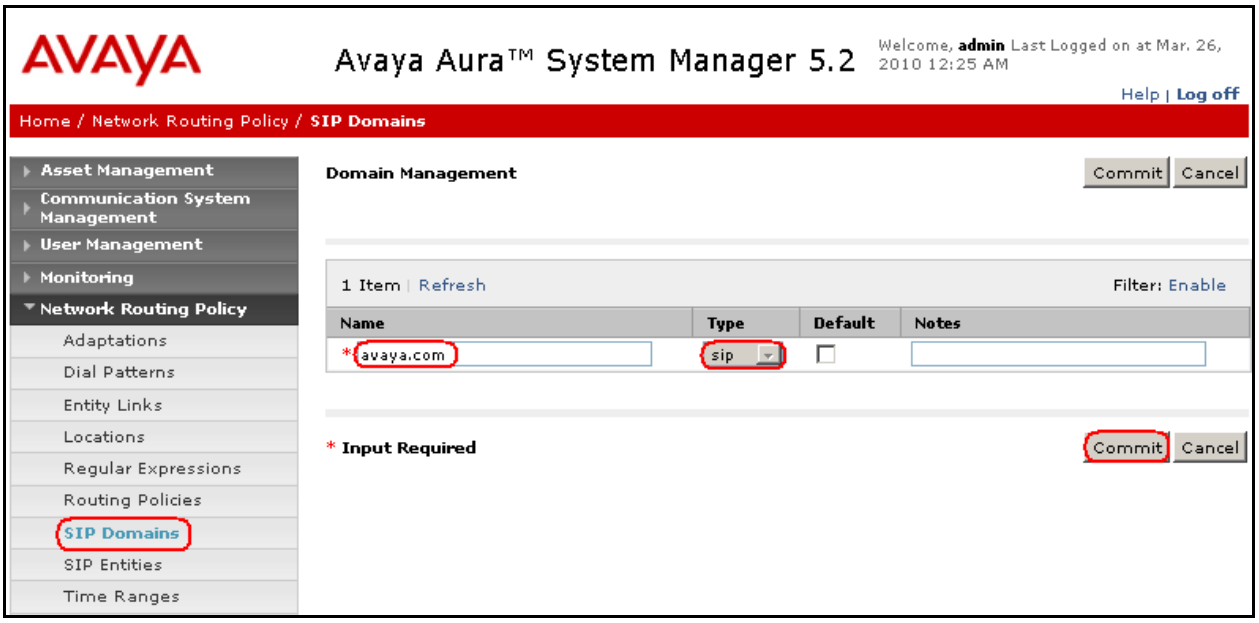


©2010 Avaya Inc. All Rights Reserved.

6.1. Specify SIP Domain

Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **SIP Domains** on the left and clicking the **New** button on the right. The following screen will then be shown. Fill in the following fields and click **Commit**.

- **Name:** The authoritative domain name (e.g. **avaya.com**)
- **Type** Select **sip**
- **Notes:** Descriptive text (optional)



©2010 Avaya Inc. All Rights Reserved.

6.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management. A single location is added to the configuration for Communication Manager Access Element, Feature Server and Dialogic IMG 1010 Gateway. To add a location, select **Locations** on the left and click on the **New** button on the right. The following screen will then be shown. Fill in the following:

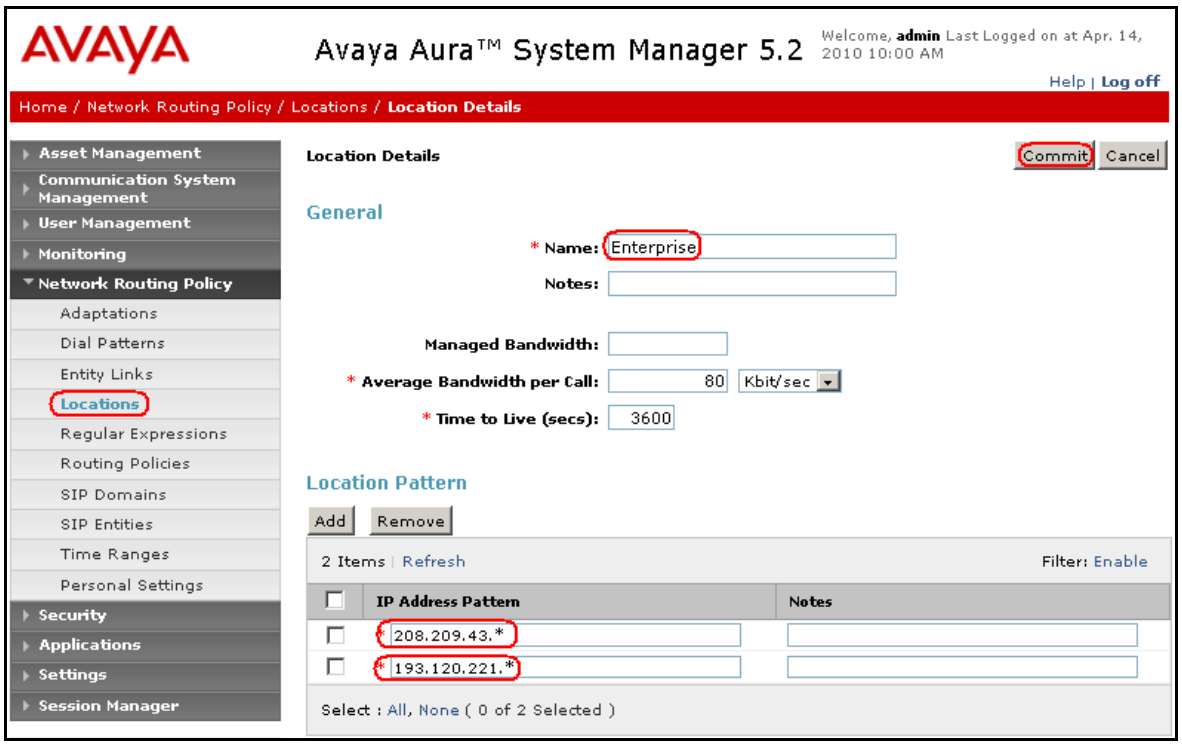
Under **General**:

- **Name:** A descriptive name
- **Notes:** Descriptive text (optional)
- **Managed Bandwidth:** Leave the default or customize as described in **[5]**

Under **Location Pattern**:

- **IP Address Pattern:** A pattern used to logically identify the location. In these Application Notes, the pattern selected defined the networks involved e.g. **193.120.221.*** for referring the Enterprise network and **208.209.43.*** for IP network where the IMG 1010 Gateway resides.
- **Notes:** Descriptive text (optional)

The screen below shows addition of the **Enterprise** location, which includes all the components of the compliance environment. Click **Commit** to save.



©2010 Avaya Inc. All Rights Reserved.

6.3. Add Adaptations

In order to maintain digit manipulation centrally on Session Manager, an adaptation module can be configured with numbering plan offered from the PSTN Service Provider. Alternatively the numbering plan translation can be implemented in the Dialogic IMG 1010 Gateway. Note that the **Digit Conversion for Outgoing Calls from SM** will modify the P-AI field in the SIP invite, requiring the IMG 1010 privacy setting as described in **Section 8.2.3**. To add an adaptation, under the **Network Routing Policy select Adaptations** on the left and click on the **New** button on the right. The following screen will then be shown. Fill in the following:

Under **General**:

- **Name:** A descriptive name i.e. **IMG_DigitConversionAdapter**
- **Module Name:** From the dropdown list select **DigitConversionAdapter**
- **Module Parameter:** Leave it blank

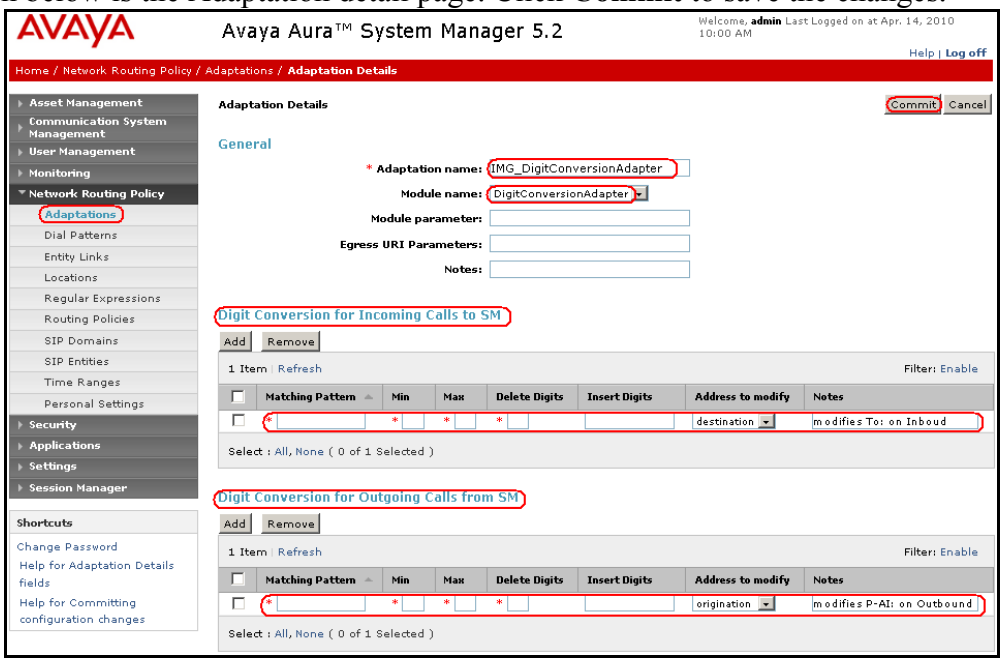
Under **Digit Conversion for Incoming Calls to SM**:

- **Matching Pattern**: The dialed number from the PSTN
- **Min/Max**: Minimum/Maximum number of digits
- **Delete**: Digits to be deleted
- **Insert Digits**: Digit to be added
- **Address to modify**: Select **destination**

Under **Digit Conversion for Outgoing Calls from SM**:

- **Matching Pattern**: The dialed number from enterprise network
- **Min:/ Max**: Minimum/ Maximum number of digits
- **Delete**: Digits to be deleted
- **Insert Digits**: Digit to be added
- **Address to modify**: Select **origination**

The screen below is the Adaptation detail page. Click **Commit** to save the changes.



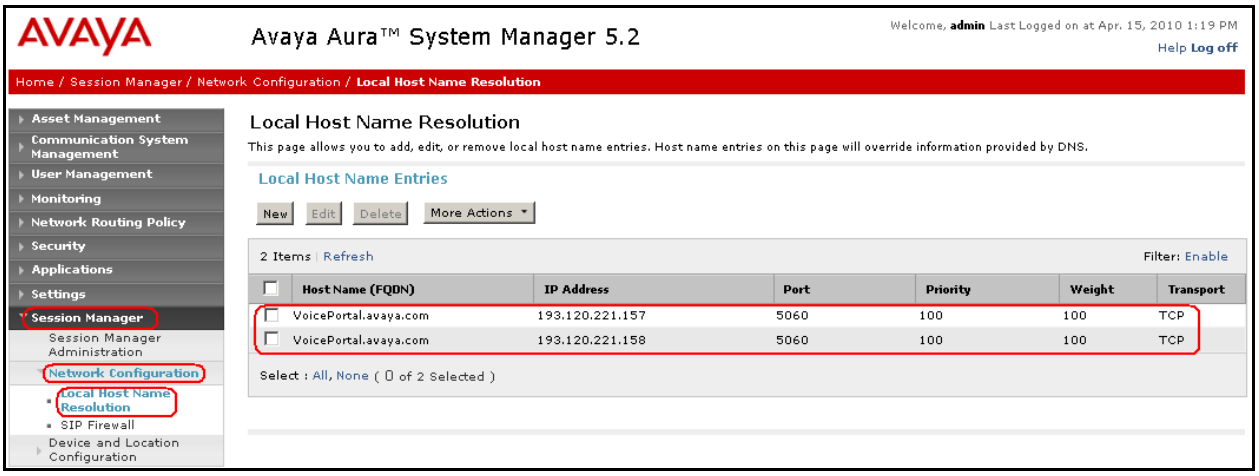
©2010 Avaya Inc. All Rights Reserved.

6.4. Add Host Entries for Voice Portal MPPs

Session Manager can connect calls to an entity with multiple IP interfaces to perform failover/load sharing. In order to configure load sharing among different MPP servers, a Host Name must be defined. Expand **Session Manager → Network Configuration** in the web interface

- Select **Local Host Name Resolution** to enter the details of the MPPs with their respective IP addresses.
  - Click **New**
  - Under **Host Name,** add an identifier followed by the SIP domain (added in **Section 6.1**). This Host Name is going to be common for the 2 of MPPs being added.
  - Under **IP Address,** enter the IP address of the respective MPPs.
  - Under **Port,** enter **5060**.
  - Enter the appropriate **Priority** and **Weight** as required.
  - Under **Transport,** select **TCP**.
  - Click **Commit** to save.

Shown below is the updated screen for the sample configuration.



©2010 Avaya Inc. All Rights Reserved.

6.5. Add SIP Entities

A SIP Entity must be added for Session Manager and for each SIP-based telephony system supported by it using SIP trunks. In the sample configuration, a SIP Entity is added for the Session Manager, the C-LAN board in the Avaya G650 Media Gateway for the Communication Manager Access Element, the Proc interface for the Communication Manager Feature Server, Voice Portal and the Dialogic IMG 1010 Gateway on the Service Provider.

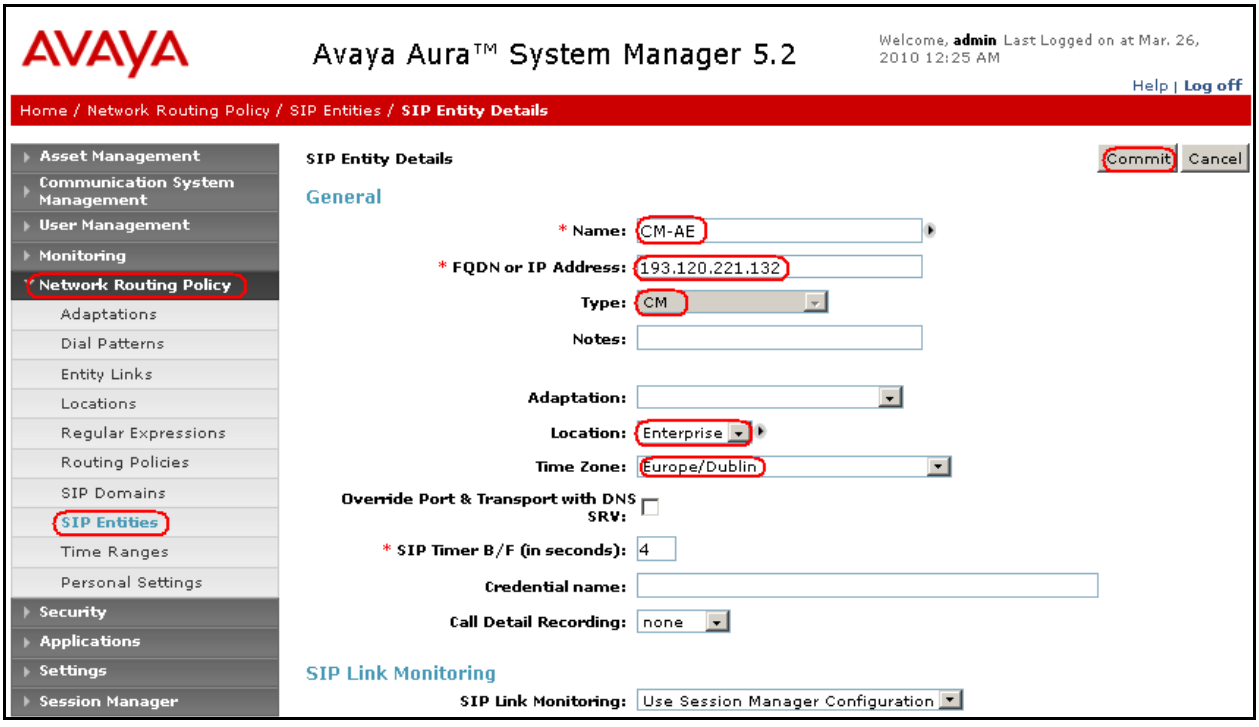
6.5.1. Adding Avaya Aura™ Communication Manager Access Element SIP Entity

To add a SIP Entity, navigate **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right.

Under **General**:

- **Name:** A descriptive name (i.e. **CM-AE**)
- **FQDN or IP Address:** IP address of the signaling interface of CLAN board in the G650 Media gateway, i.e. **193.120.221.132**
- **Type:** Select **CM**
- **Location:** Select one of the locations defined previously i.e. **Enterprise**
- **Time Zone:** Time zone for this entity

Defaults can be used for the remaining fields. Click **Commit** to save SIP Entity definition. The following screen shows addition of Communication Manager Access Element.



©2010 Avaya Inc. All Rights Reserved.

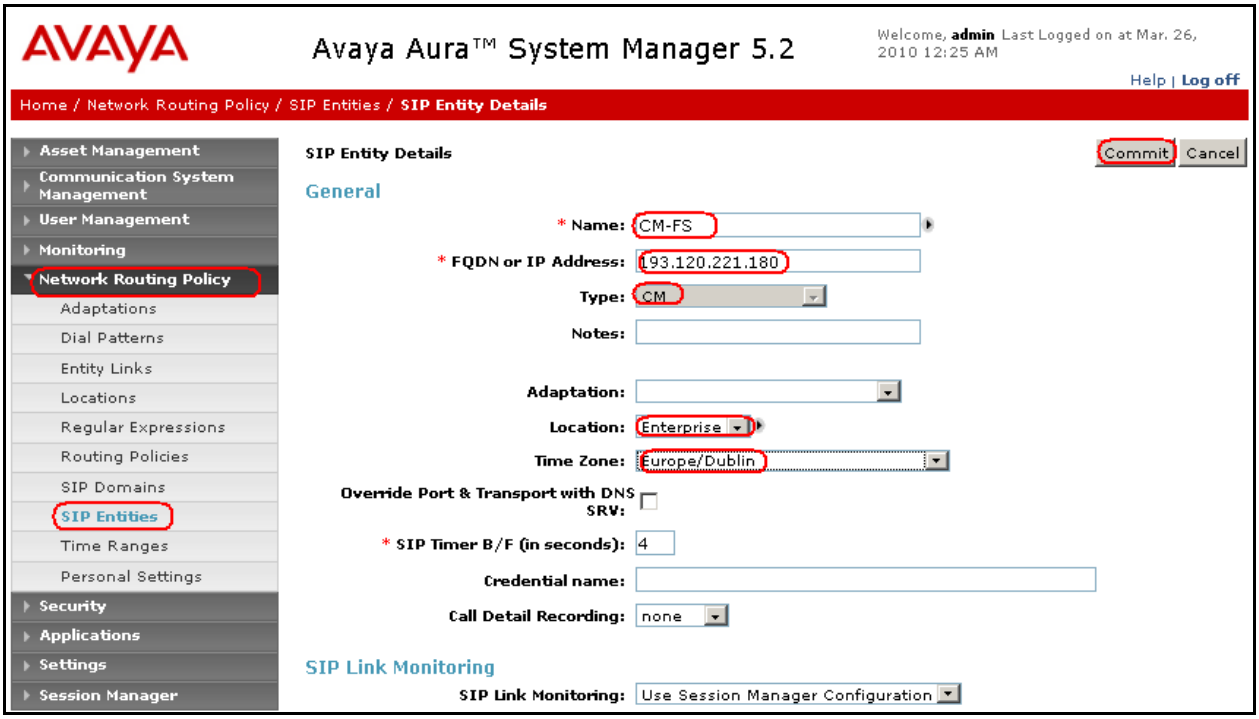
6.5.2. Adding Avaya Aura™ Communication Manager Feature Server SIP Entity

To add a SIP Entity, navigate **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right.

Under **General**:

- **Name:** A descriptive name (i.e. **CM-FS**)
- **FQDN or IP Address:** IP address of the Proc interface of S8300 Server, i.e. **193.120.221.180**
- **Type:** Select **CM**
- **Location:** Select one of the locations defined previously i.e. **Enterprise**
- **Time Zone:** Time zone for this entity

Defaults can be used for the remaining fields. Click **Commit** to save SIP Entity definition. The following screen shows addition of Communication Manager Feature Server.



©2010 Avaya Inc. All Rights Reserved.

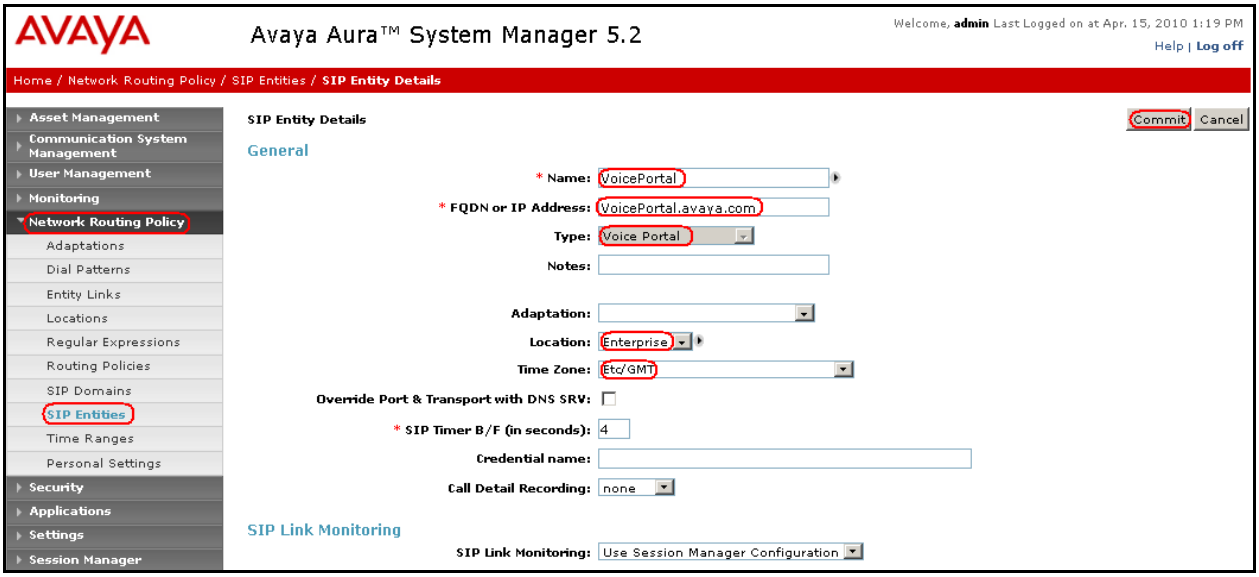
6.5.3. Adding Voice Portal SIP Entity

Navigate **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right.

Under **General**:

- **Name:** A descriptive name (i.e. **VoicePortal**)
- **FQDN or IP Address:** Use the FQDN that was defined on the Local Host Name Resolution defined in **Section 6.4**, i.e. **VoicePortal.avaya.com**
- **Type:** Select **Voice Portal**
- **Location:** Select one of the locations defined previously i.e. **Enterprise**
- **Time Zone:** Time zone for this entity

Defaults can be used for the remaining fields. Click **Commit** to save SIP Entity definition. The screen below shows the configuration of the SIP Entity related to Voice Portal.



©2010 Avaya Inc. All Rights Reserved.

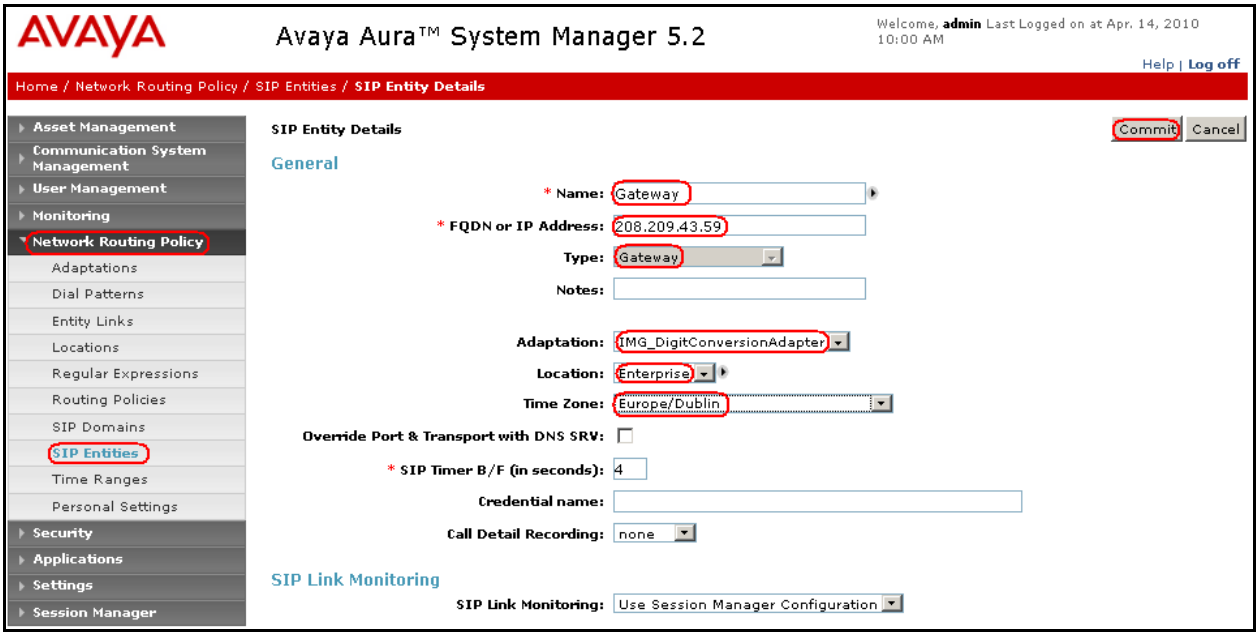
6.5.4. Adding Dialogic IMG 1010 Gateway SIP Entity

Navigate **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right.

Under **General**:

- **Name:** A descriptive name (i.e. **Gateway**)
- **FQDN or IP Address:** IP address of the signaling interface of IMG 1010 Gateway, i.e. **208.209.43.59**
- **Type:** Select **Gateway**
- **Adaptation:** Select the adaptation created in **Section 6.3** i.e. **IMG_DigitConversionAdapter**
- **Location:** Select one of the locations defined previously i.e. **Enterprise**
- **Time Zone:** Time zone for this entity

Defaults can be used for the remaining fields. Click **Commit** to save SIP Entity definition. The picture below shows the configuration of the SIP Entity related to Dialogic IMG 1010.



©2010 Avaya Inc. All Rights Reserved.

6.5.5. Adding Avaya Aura™ Session Manager SIP Entity

Navigate **Network Routing Policy → SIP Entities** on the left and click on the **New** button on the right.

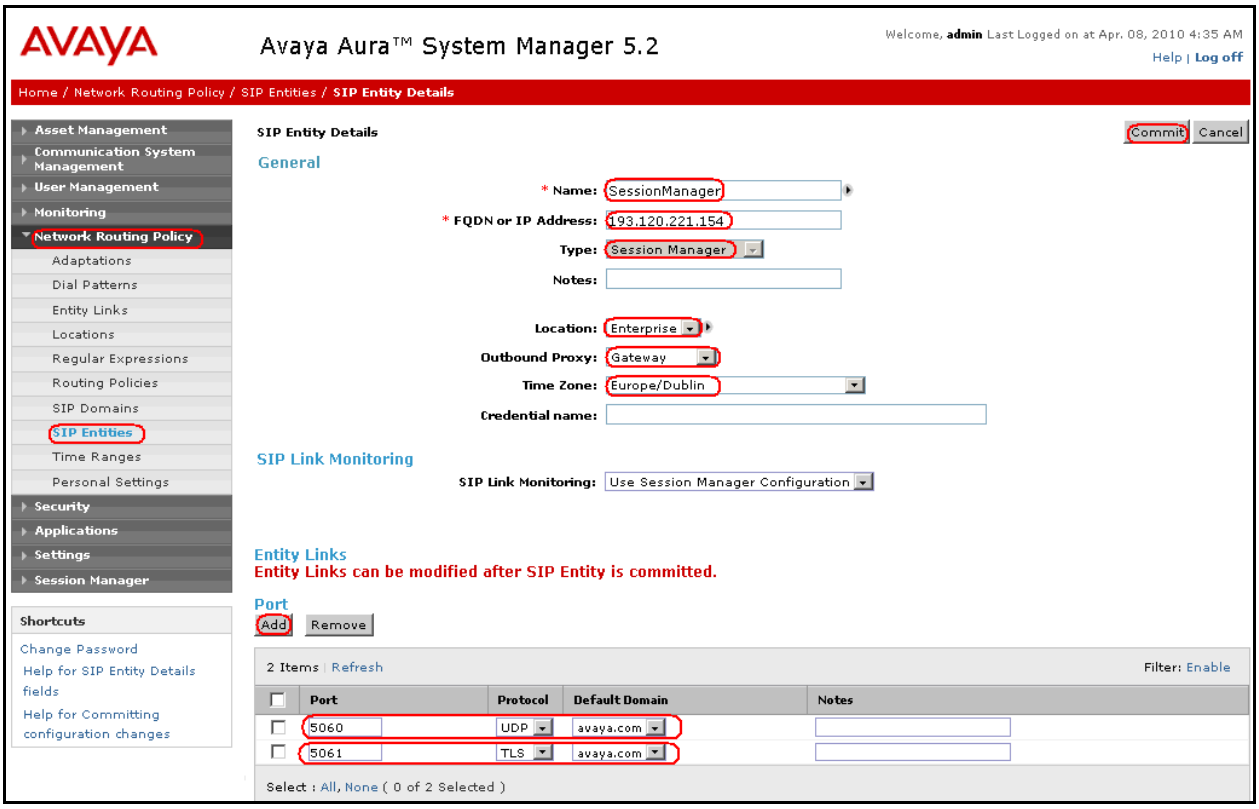
Under **General**:

- **Name:** A descriptive name, i.e. **SessionManager**
- **FQDN or IP Address:** IP address of the Session Manager i.e. **193.120.221.154**, the SM-100 Security Module
- **Type:** Select **Session Manager**
- **Location:** Select one of the locations defined previously
- **Outbound Proxy:** Select the SIP Entity defined previously for IMG, i.e. **Gateway**
- **Time Zone:** Time zone for this entity

Create two Port definitions, one for **TLS** and one for **UDP**. Under **Port**, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Port:** Port number on which the system listens for SIP requests
- **Protocol:** Transport protocol to be used to send SIP requests
- **Default Domain** The domain used (e.g., **avaya.com**)

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition. The following screen shows the addition of Session Manager.



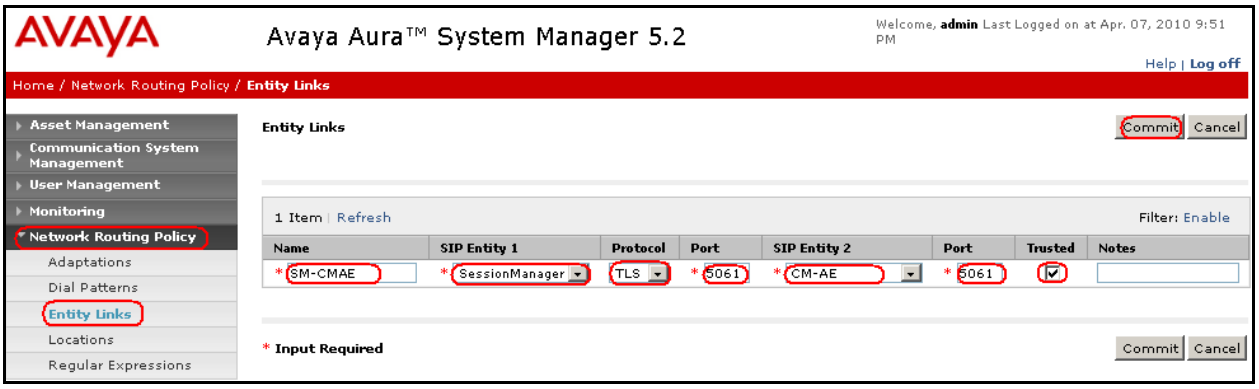
©2010 Avaya Inc. All Rights Reserved.

6.6. Add Entity Links

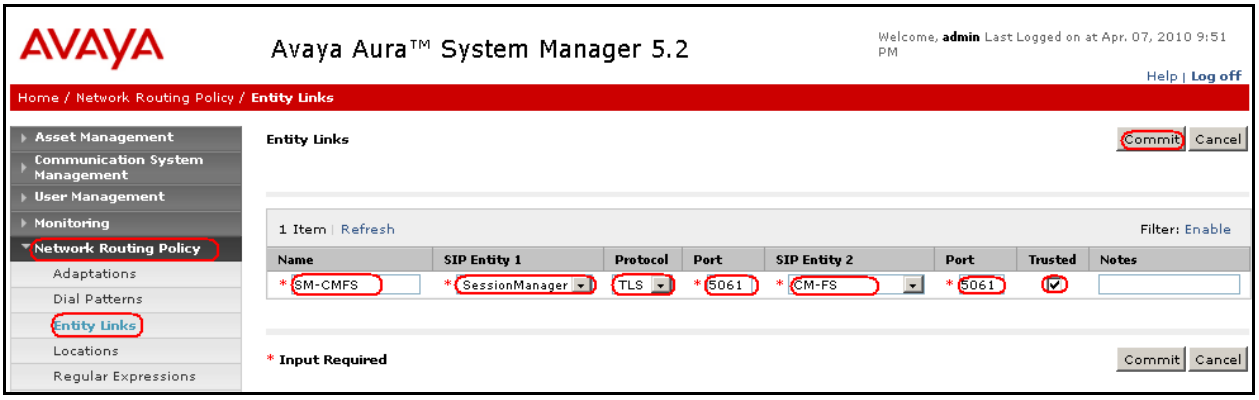
A SIP trunk between Session Manager and a telephony system is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name
- **SIP Entity 1:** Select the **SessionManager** entity
- **Port:** Port number to which the other system sends SIP requests
- **SIP Entity 2:** Select the name of the other system
- **Port:** Port number on which the other system receives SIP requests
- **Trusted:** Check this box, otherwise calls from the associated SIP Entity specified will be denied
- **Protocol:** Select the transport protocol between **UDP/TCP/TLS** to align with the definition on the **other end o**f the link. In these Application Notes **TLS** was used for **Communication Manager Access Element** and **Feature Server, TCP** for **Voice Portal** while **UDP** for **Dialogic IMG 1010.**

Click **Commit** to save each Entity Link definition**.** The following screen illustrates adding the Entity Link for Communication Manager Access Element.

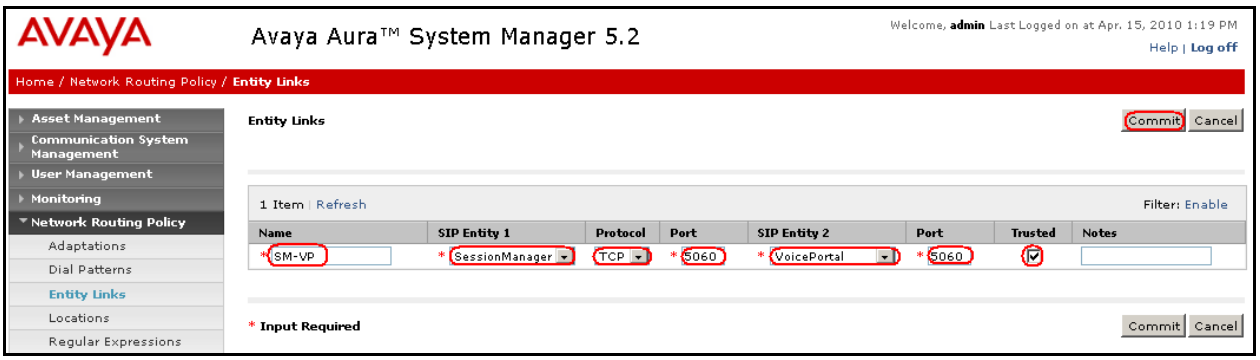


The screen below illustrates adding the Entity Link for Communication Manager Feature Server.

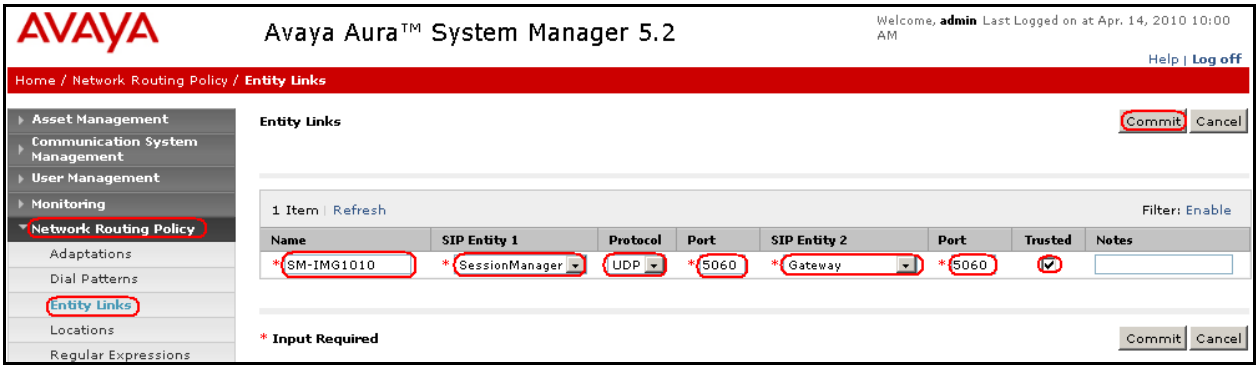


©2010 Avaya Inc. All Rights Reserved.

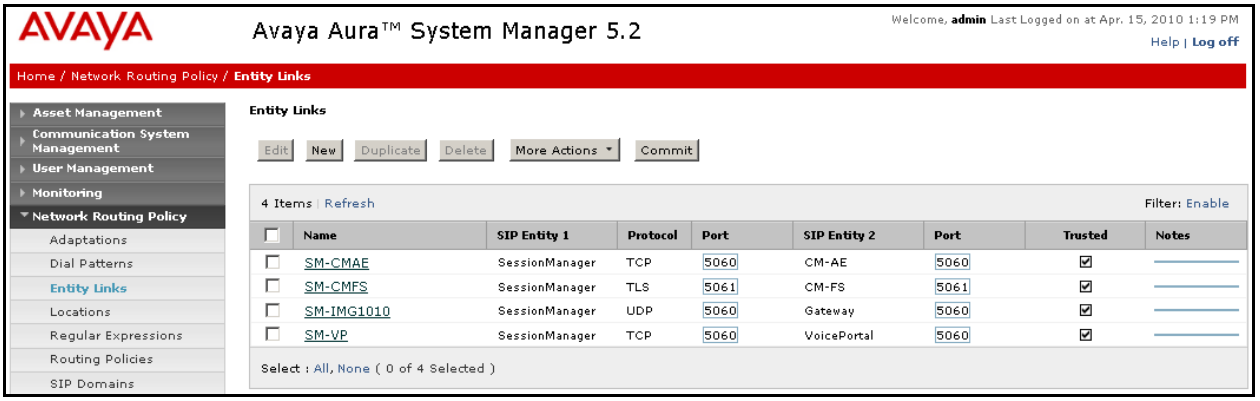
The screen below illustrates adding the Entity Link for Voice Portal.



The screen below illustrates adding the Entity Link for IMG 1010 Sip Entity.



The screen below summarizes the Entity Links view after the insertion of the four Entity Links.



©2010 Avaya Inc. All Rights Reserved.

6.7. Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.5**. Three routing policies must be added: one for Communication Manager Access Element, one for Voice Portal and one for the IMG 1010 Gateway. To add a routing policy, select **Routing Policies** on the left and click on the **New** button on the right. The following screen is displayed. Fill in the following:

Under **General:**

- Enter a descriptive name in **Name**

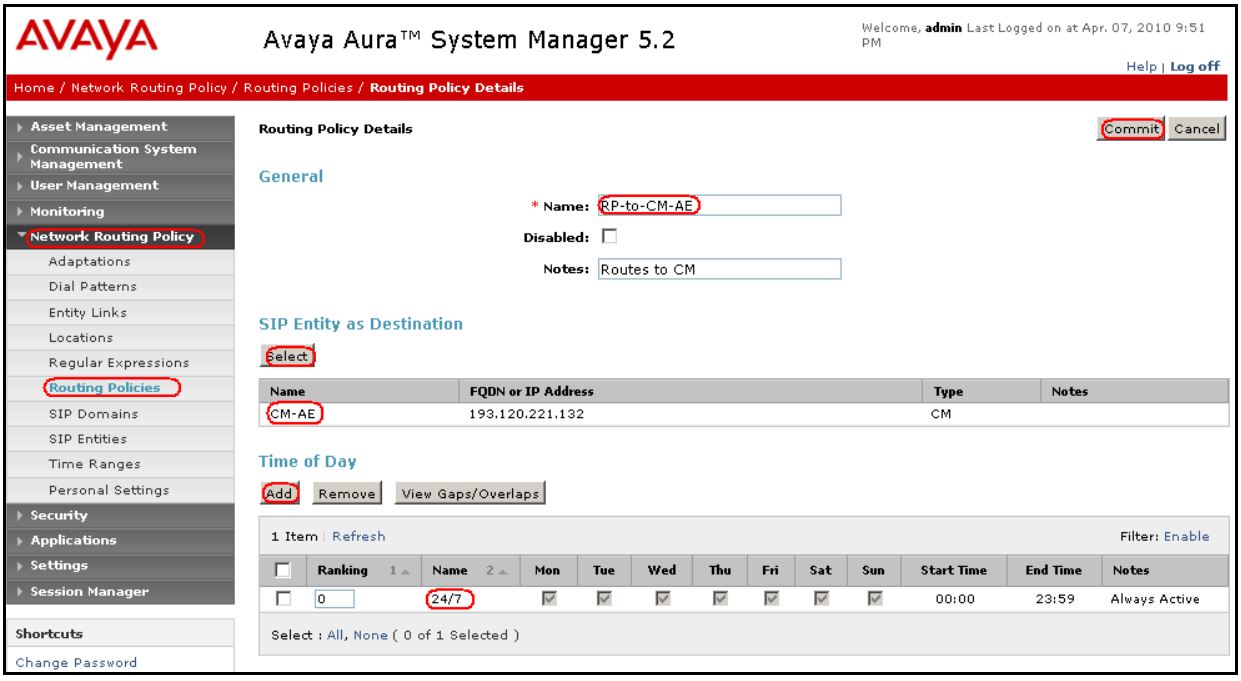
Under **SIP Entity as Destination:**

- Click **Select**, and then select the appropriate SIP entity to which this routing policy applies

Under **Time of Day:**

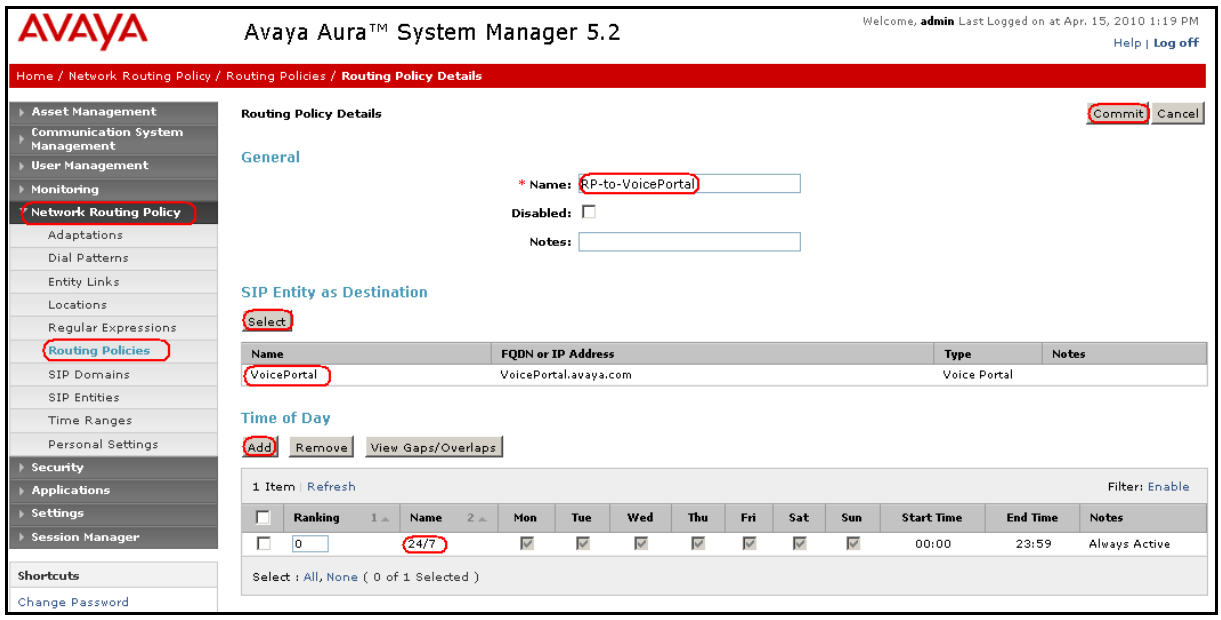
- Click **Add**, and select the time range configured. In these Application Notes, the predefined **24/7** Time Range is used

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following picture shows the Routing Policy for Communication Manager Access Element.

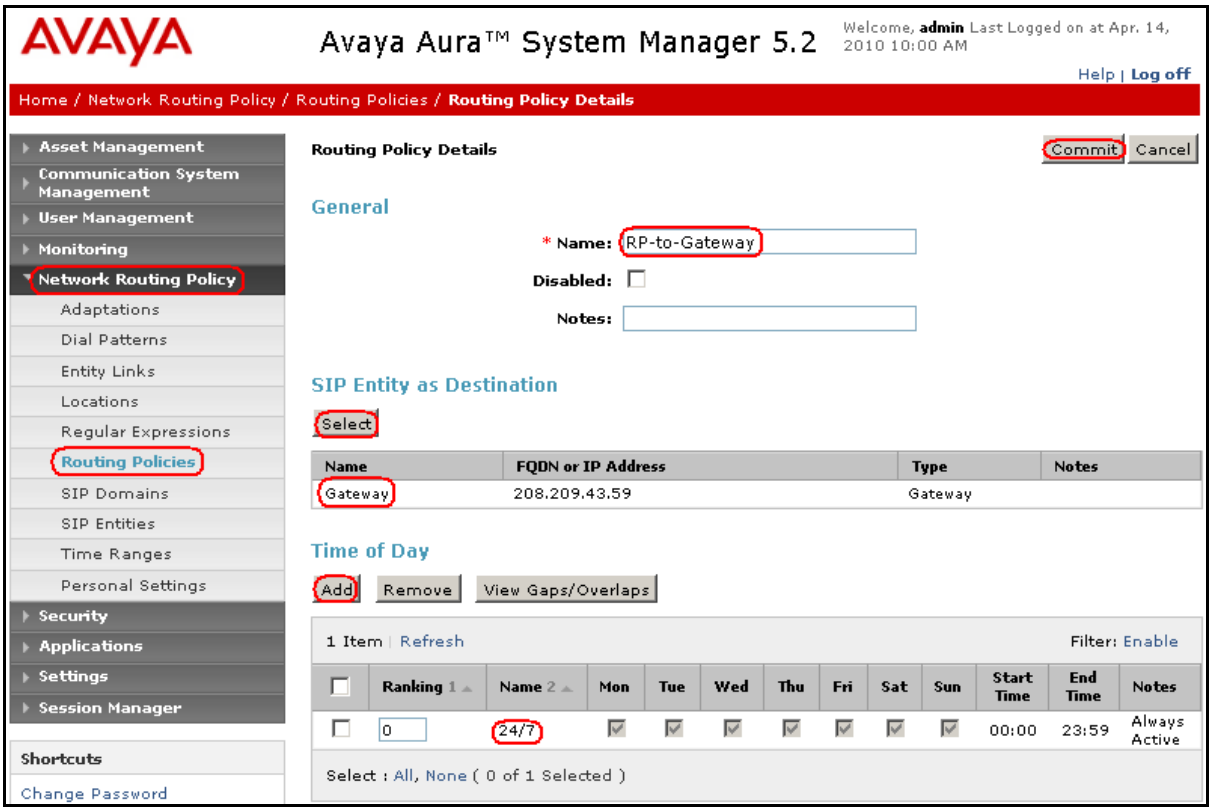


©2010 Avaya Inc. All Rights Reserved.

The following picture shows the Routing Policy for Voice Portal.



The following screen shows the Routing Policy for Dialogic IMG 1010.



©2010 Avaya Inc. All Rights Reserved.

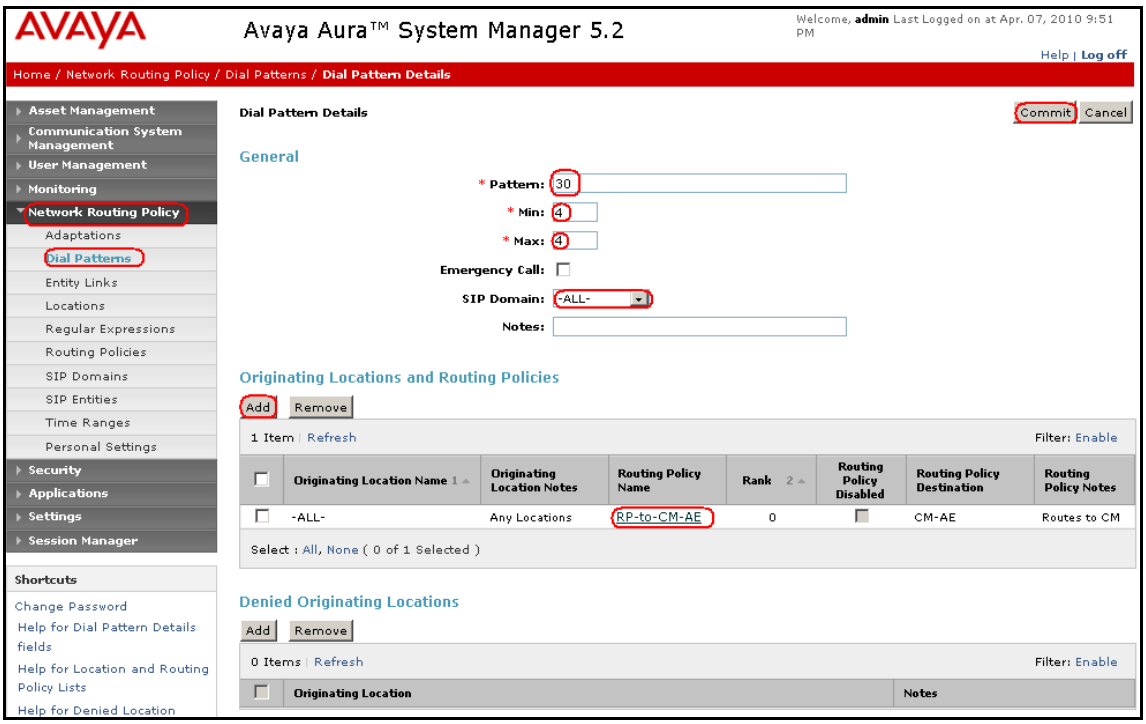
6.8. Add Dial Patterns

Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, **4**-digit extensions beginning with **30** reside on Communication Manager Access Element, **4**-digit extensions beginning with **36** reside on Voice Portal and numbers beginning with **0** with 3 to 25 digits reside on the Dialogic IMG 1010. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button on the right. Fill in the following, as shown in the screen below, which corresponds to the dial pattern for routing calls to Communication Manager Access Element:

Under **General**:

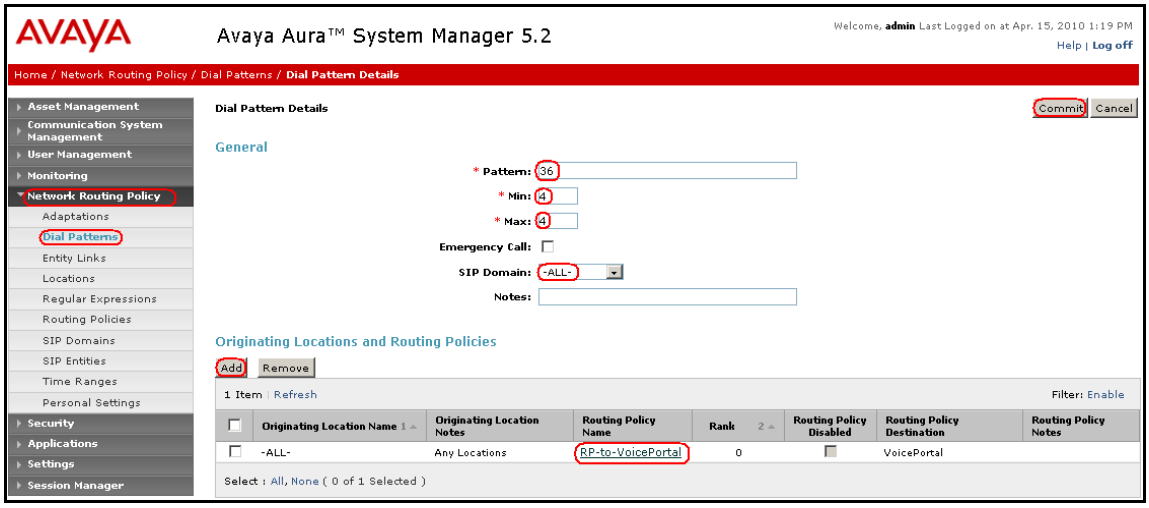
- **Pattern:** Dialed number or prefix i.e. **30**
- **Min:** Minimum length of dialed number i.e. **4**
- **Max:** Maximum length of dialed number i.e. **4**
- **SIP Domain:** Select **ALL**

Under **Originating Locations and Routing Policies**, click **Add**, and then select the appropriate location and routing policy from the list. Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows a sample dial pattern definition for Communication Manager Access Element.



©2010 Avaya Inc. All Rights Reserved.

The screen below shows the definition of Dial Pattern for Voice Portal.



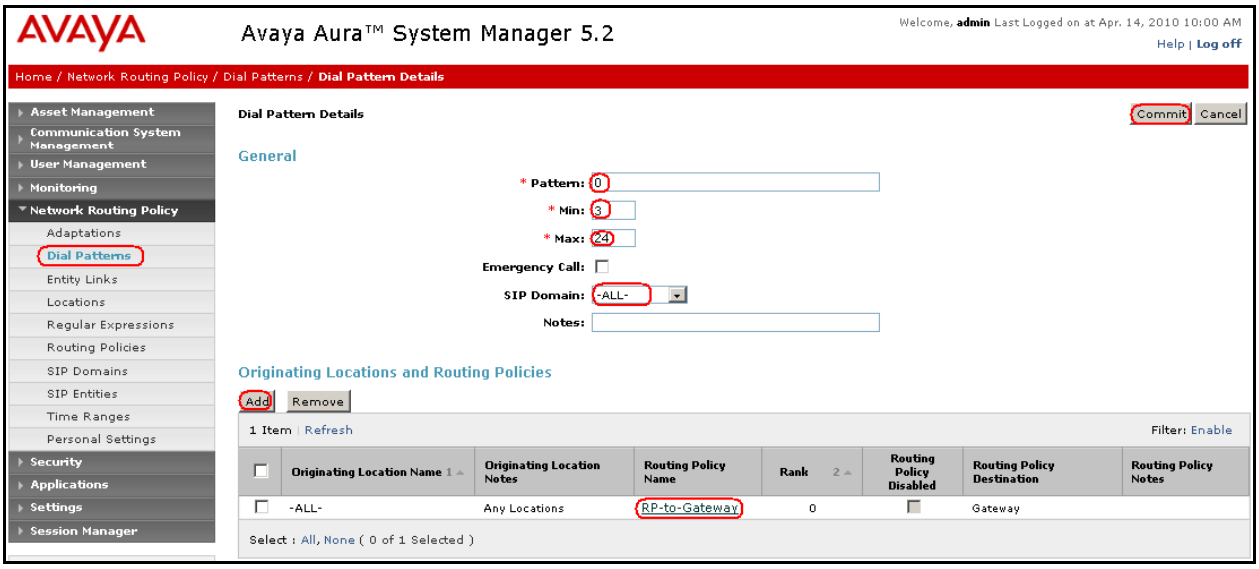
Repeat the process adding one or more dial patterns for the PSTN numbers that should be reached from the Dialogic IMG 1010. Fill in the following, as shown in the screen below, which corresponds to the dial pattern for routing calls to Dialogic IMG 1010:

Under **General**:

- **Pattern:** Dialed number or prefix i.e. **0**
- **Min:** Minimum length of dialed number i.e. **3**
- **Max:** Maximum length of dialed number i.e. **24**
- **SIP Domain:** Select **ALL**

Under **Originating Locations and Routing Policies**, click **Add**, and then select the appropriate location and routing policy from the list. Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows a sample dial pattern definition for PSTN reachable with Dialogic IMG 1010.

©2010 Avaya Inc. All Rights Reserved.



©2010 Avaya Inc. All Rights Reserved.

6.9. Add Avaya Aura™ Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Then click **Add**, and fill in the fields as described below and shown in the following screen:

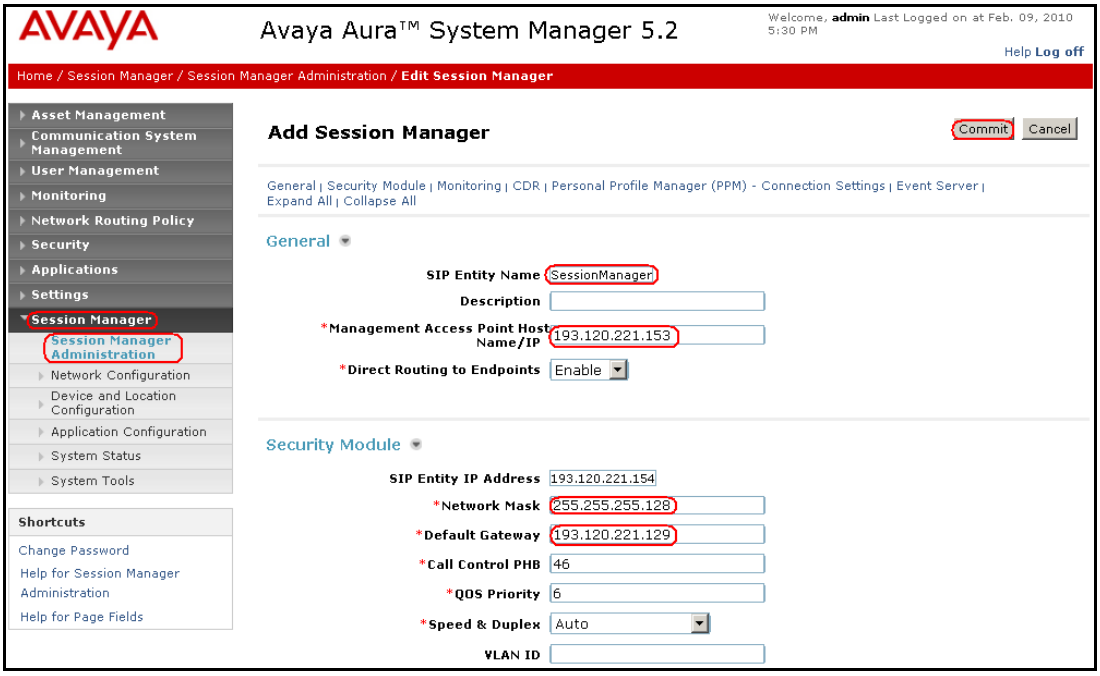
Under **General**:

- **SIP Entity Name**: Select the name of the SIP Entity added for Session Manager
- **Description**: Descriptive comment (optional)
- **Management Access Point Host Name/IP**: Enter the IP address of the Session Manager management interface

Under **Security Module**:

- **Network Mask**: Enter the network mask corresponding to the IP address of the SM100 interface (i.e., **255.255.255.128**)
- **Default Gateway**: Enter the IP address of the default gateway for SM100 interface (i.e., **193.120.221.129**)

Use default values for the remaining fields. Click **Save** to add this configuration to Session Manager.



©2010 Avaya Inc. All Rights Reserved.

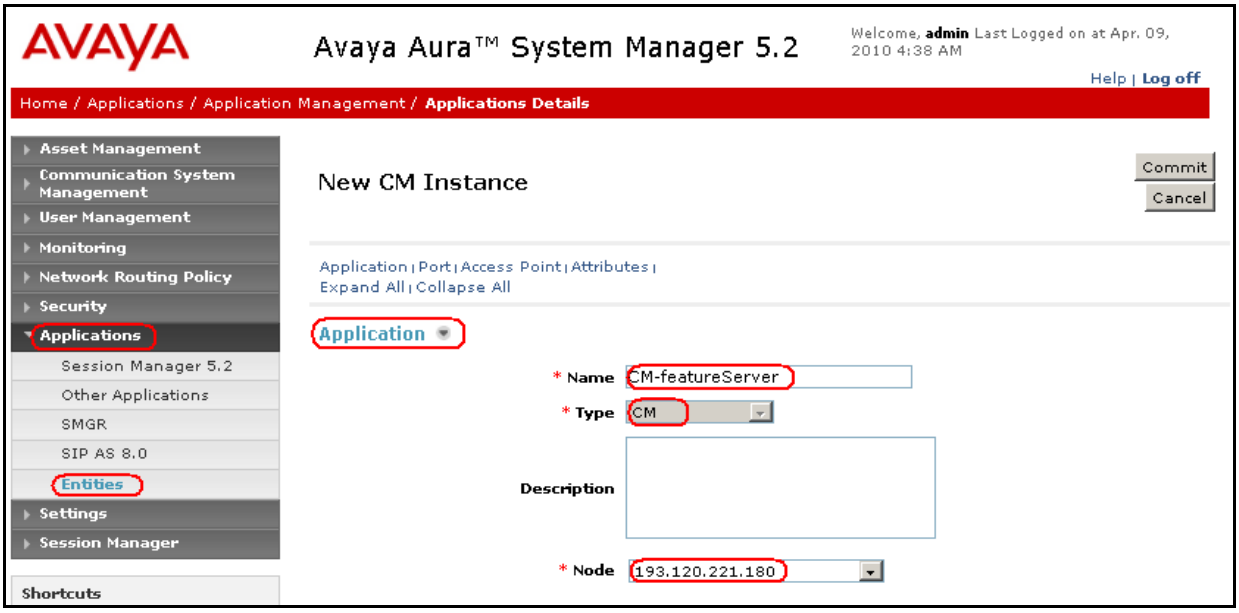
6.10. Add Avaya Aura™ Communication Manager as a Feature Server

In order for Communication Manager to provide configuration and Feature Server support to SIP phones when they register to Session Manager, Communication Manager must be added as an application.

6.10.1. Create an Application Entry

Expand **Application** menu, select **Entities** on left, click on **New** (not shown). Enter the following fields and retain defaults for the remaining fields.
Under **Application**:

- **Name**: Enter a descriptive name i.e. **CM-featureServer**
- **Type**: Select **CM**
- **Node**: Select **Other..** and enter the IP address for CM SAT access i.e. **193.120.221.180**

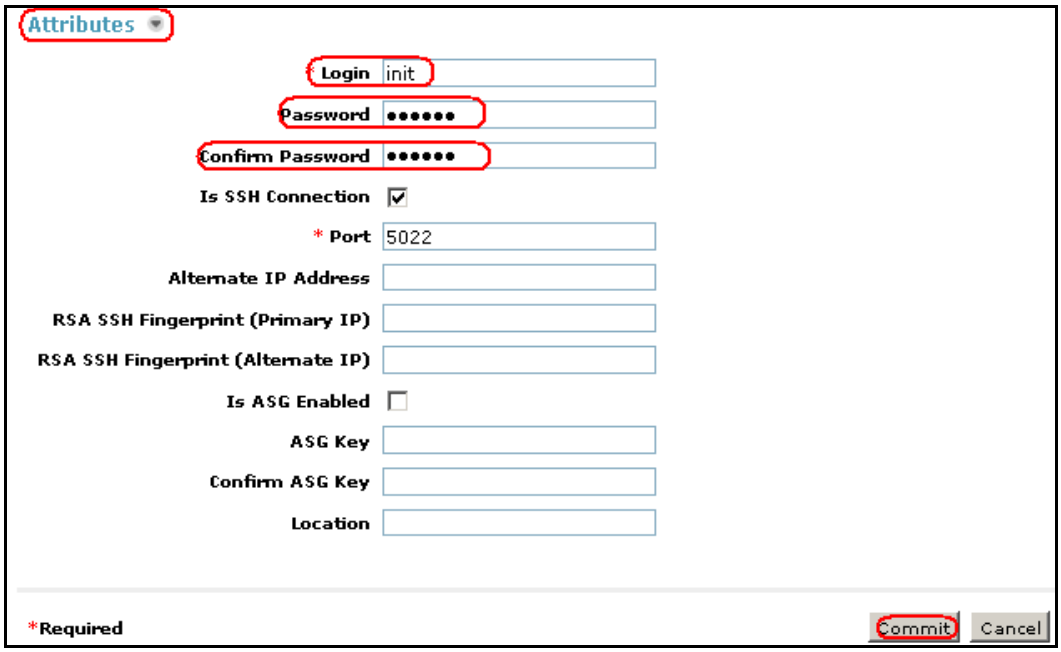


©2010 Avaya Inc. All Rights Reserved.

Navigate to the **Attributes** section and enter the following:

- **Login**: Login used for SAT access
- **Password**: Password used for SAT access
- **Confirm Password**: Password used for SAT access

Retain default values for the remaining fields. Click **Commit** to save.



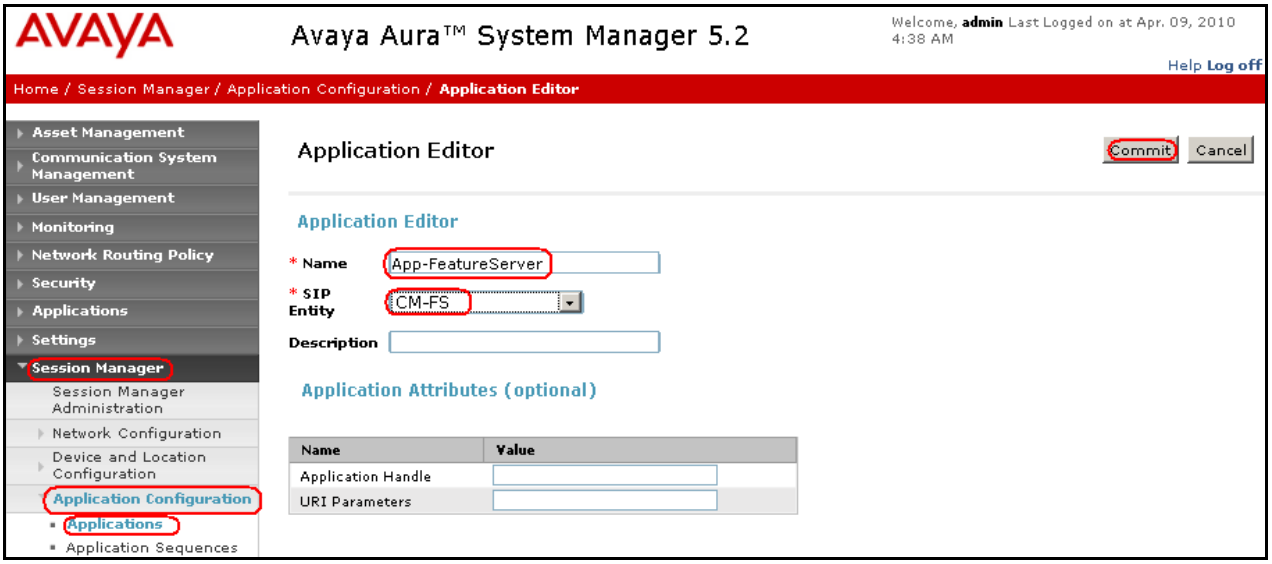
©2010 Avaya Inc. All Rights Reserved.

6.10.2. Create a Feature Server Application

Navigate to **Session Manger → Application Configuration → Applications** on the left menu. Click on **New** (not shown). Enter following fields and use defaults for the remaining fields:

- **Name** A descriptive name
- **SIP Entity** Select the CM SIP Entity defined in **Section 6.5.2**

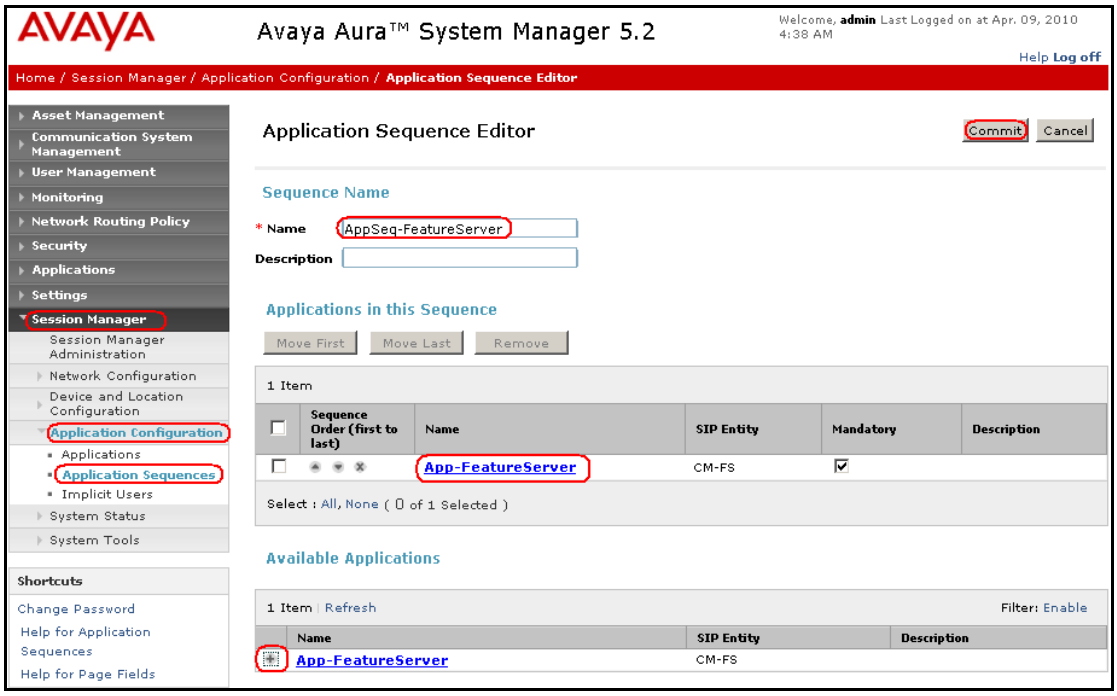
Click on **Commit** to save.



©2010 Avaya Inc. All Rights Reserved.

6.10.3. Create a Feature Server Application Sequence

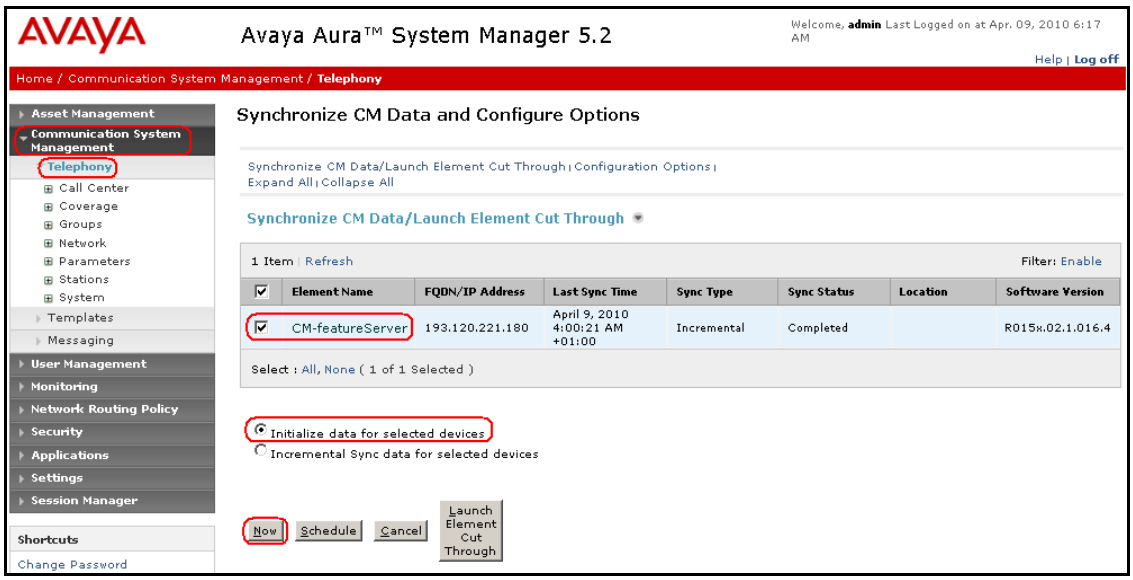
From the left menu, navigate to **Application Sequences** under **Session Manager Application Configuration**. Click on **New** (not shown). Enter a descriptive **Name**. Click on the + sign next to the appropriate **Available Applications** and they will move up to the **Applications in this Sequence** section. Click on **Commit** to save.



©2010 Avaya Inc. All Rights Reserved.

6.10.4. Synchronize Avaya Aura™ Communication Manager Data

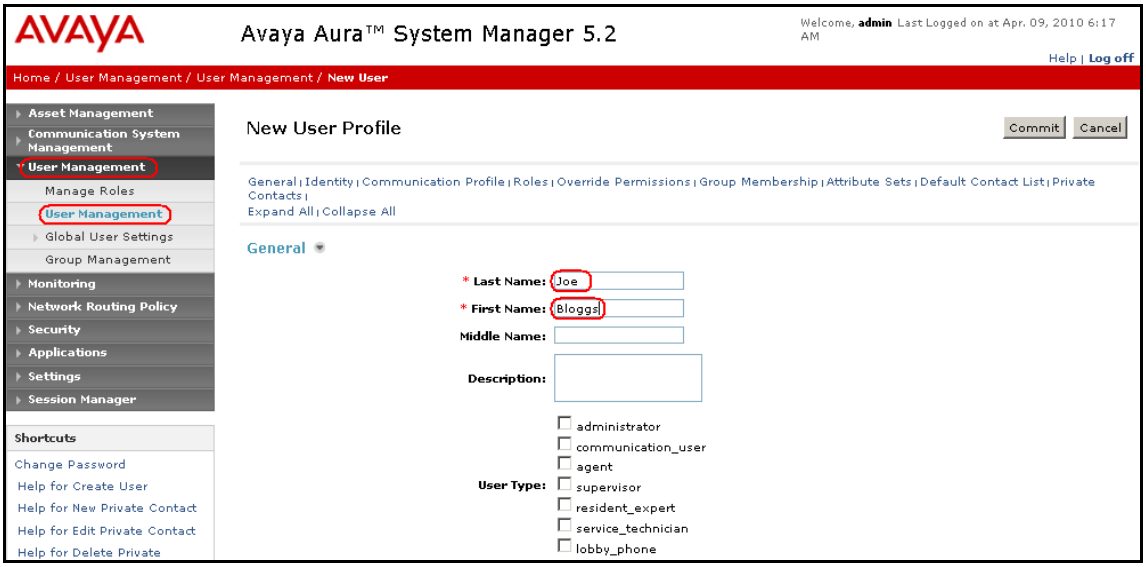
Select **Communications System Management → Telephony** on the left. Select the appropriate **Element Name**. Select **Initialize data for selected devices**, then click on **Now**. This may take some time.



Use the menus on the left under **Monitoring → Scheduler** to determine when the task is complete.

6.11. Add Users for SIP Phones

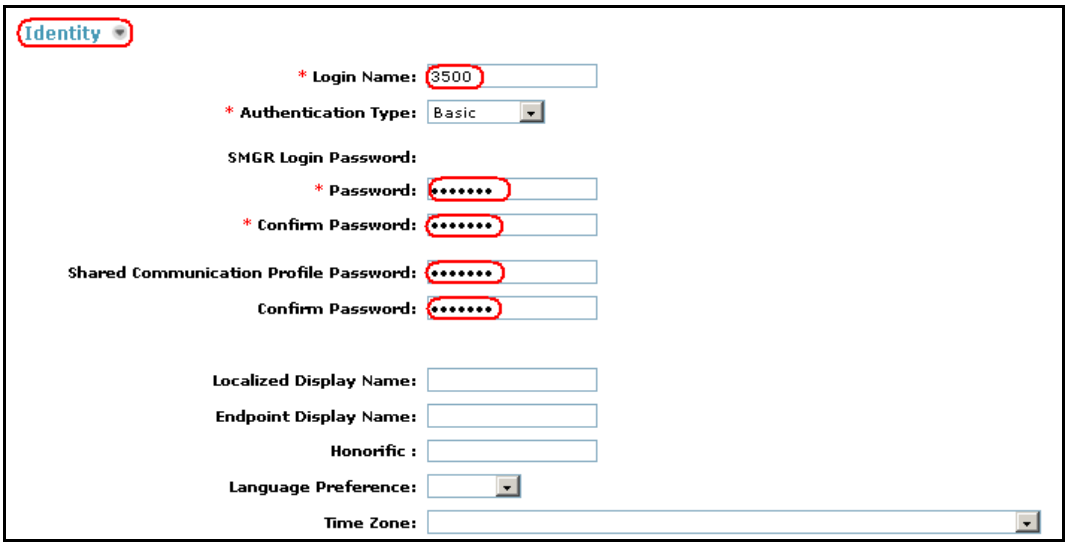
Users must be added via Session Manager and the details will be updated on the CM. Select **User Management → User Management** on the left. Then click on **New** (not shown). Enter a **First Name** and **Last Name**.



©2010 Avaya Inc. All Rights Reserved.

Navigate to the **Identity** section and enter the following and use defaults for other fields:

- **Login Name** The desired phone extension number @domain.com where domain was defined in **Section 6.1**
- **Password** Password for user to log into SMGR
- **Shared Communication Profile Password**
  Password to be entered by the user when logging into the phone

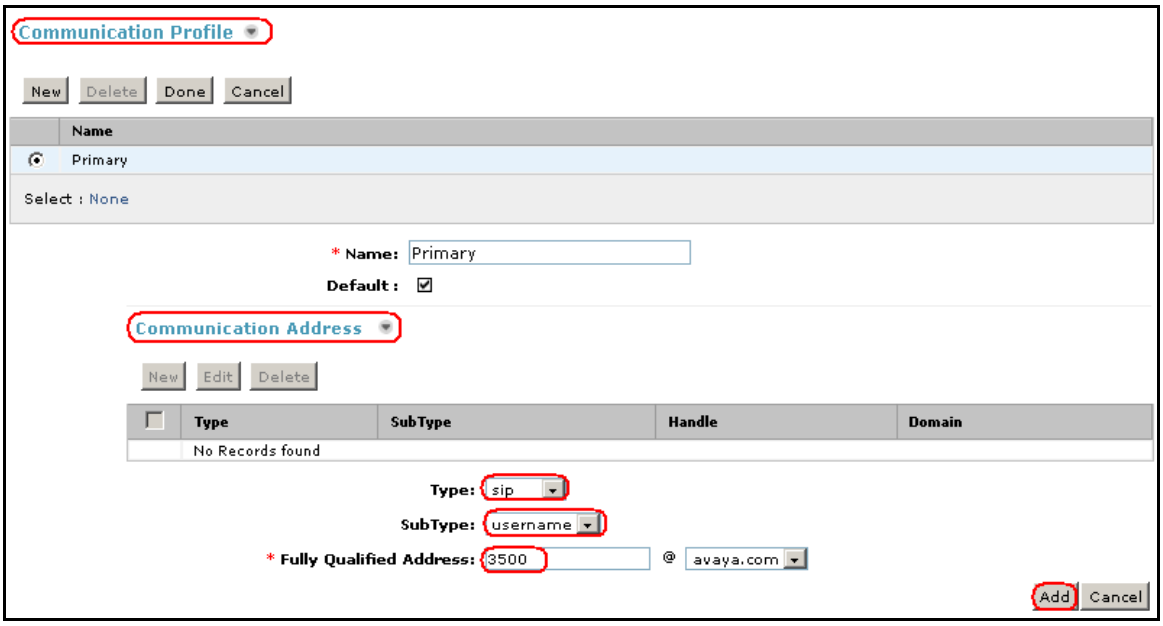


©2010 Avaya Inc. All Rights Reserved.

Navigate to and click on the **Communication Profile** section to expand. Then click on **Communication Address** to expand that section. Enter the following and defaults for the remaining fields:

- **Type** Select **SIP**
- **SubType** Select **username**
- **Fully Qualified Address** Enter the extension number i.e. **3500**

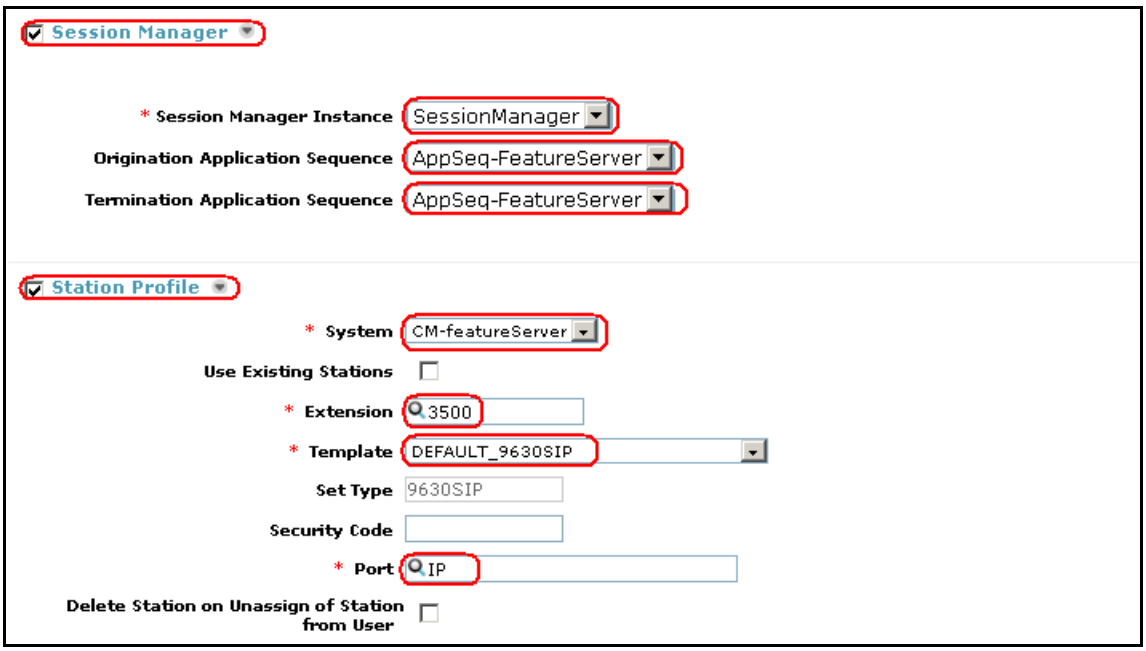
Click on **Add**.



©2010 Avaya Inc. All Rights Reserved.

Navigate to and click on the **Session Manager** section to expand. Select the appropriate Session Manager server for **Session Manager Instance**. For **Origination Application Sequence** and **Termination Application Sequence** select the application sequence created in **Section 6.10.3**. Click on **Station Profile** to expand that section. Enter the following fields and use defaults for the remaining fields:

- **System**: Select the CM Entity
- **Extension**: Enter a desired extension number i.e. **3500**
- **Template**: Select a telephone type template
- **Port**: Select **IP**



Click on **Commit** to save (not shown).

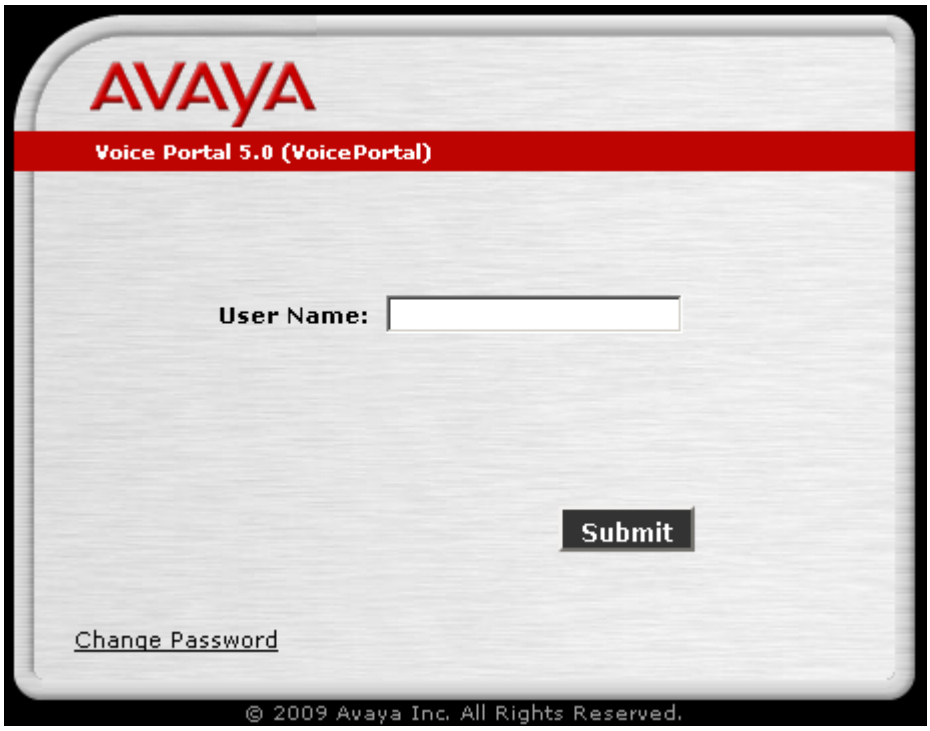
©2010 Avaya Inc. All Rights Reserved.

7. Configure Avaya Aura™ Voice Portal

This section covers the administration of Voice Portal. Voice Portal solution is built on several components such as: Voice Portal Management System (VPMS), one or more Media Processing Platform (MPP) servers, one or more external speech servers. This section covers the administration of Voice Portal. The installation steps are defined in **Reference [15], [16] and [17].** In this configuration, Voice Portal connected to the IP network via a SIP interface. Voice Portal configuration required:

- Configuring SIP connection for Avaya Aura™ Session Manager
- Adding MPP servers
- Configuring the VoIP audio format (mu-law or a-law)
- Adding a speech server
- Adding applications
- Starting the MPP servers

Voice Portal is configured via the Voice Portal Management System (VPMS) web interface. To access the web interface, enter http://<ip-addr>/VoicePortal as the URL in an Internet browser, where <*ip-addr*> is the IP address of the VPMS. The screen shown below is displayed. Log in using the Administrator user role.

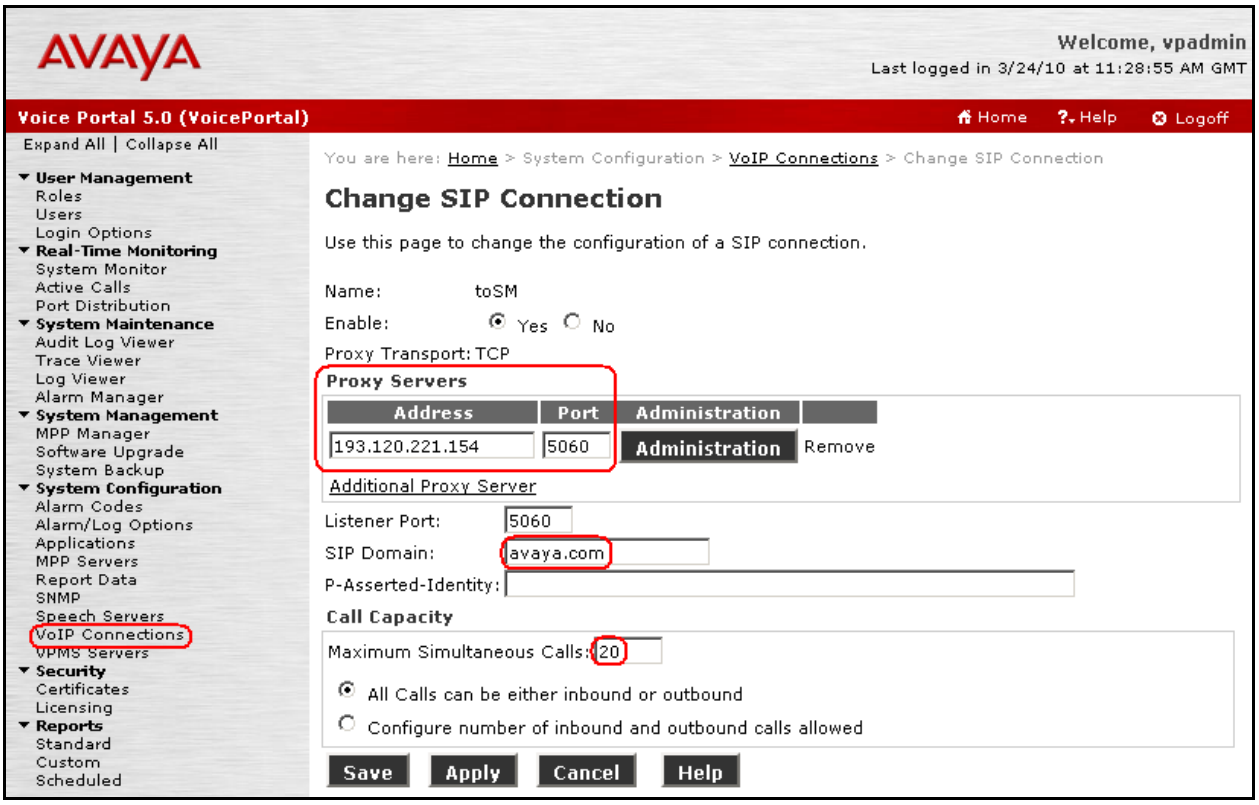


©2010 Avaya Inc. All Rights Reserved.

7.1. Configuring a SIP Connection for Avaya Aura™ Session Manager

To configure a SIP connection for Session Manager, navigate to the **VoIP Connections** page and then click on the **SIP** tab. In the SIP tab shown in the figure below, the following parameters must be configured:

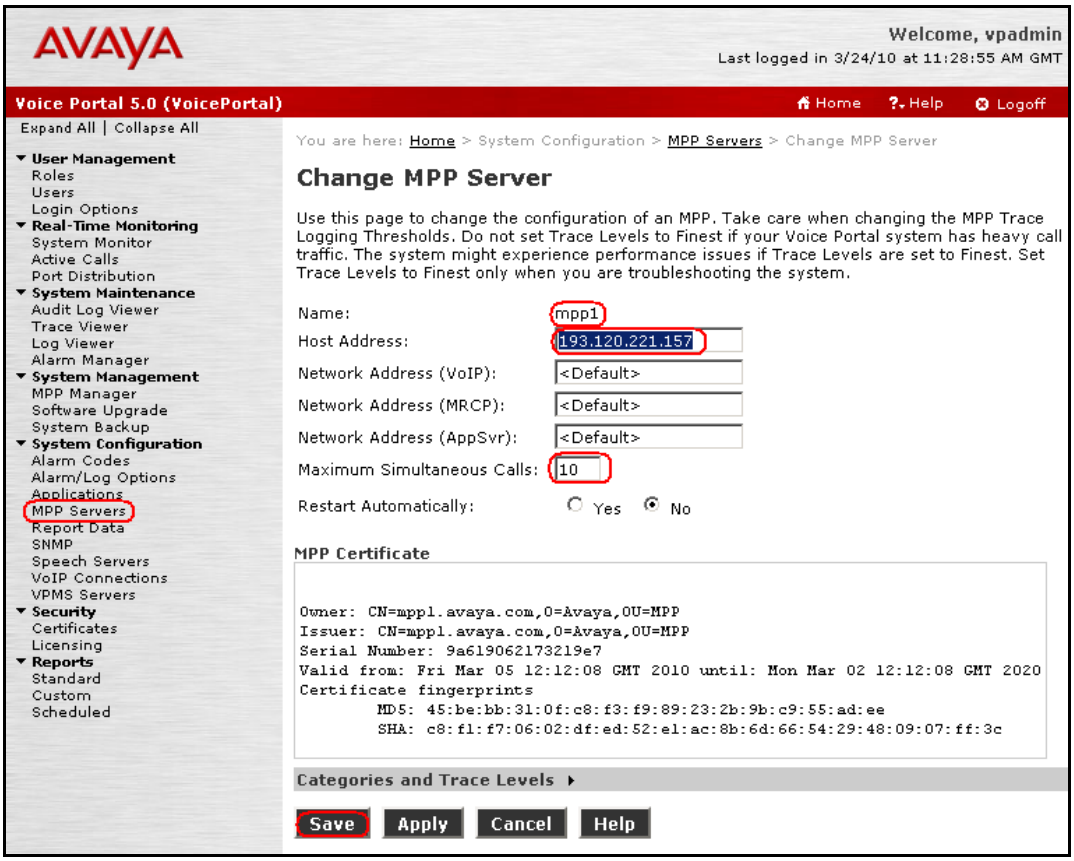
- Specify the IP address of Session Manager IP interface in the **Proxy Server Address** field (e.g. **193.120.221.154**)
- Set the **Proxy Server Port** and **Listener Port** fields to **5060** for TCP.
- Set the **SIP Domain** (e.g., **avaya.com**)
- Set the **Maximum Simultaneous Calls**. In this example, a maximum of **20** calls is supported between two MPP servers.
- Accept the default values for the other fields.



©2010 Avaya Inc. All Rights Reserved.

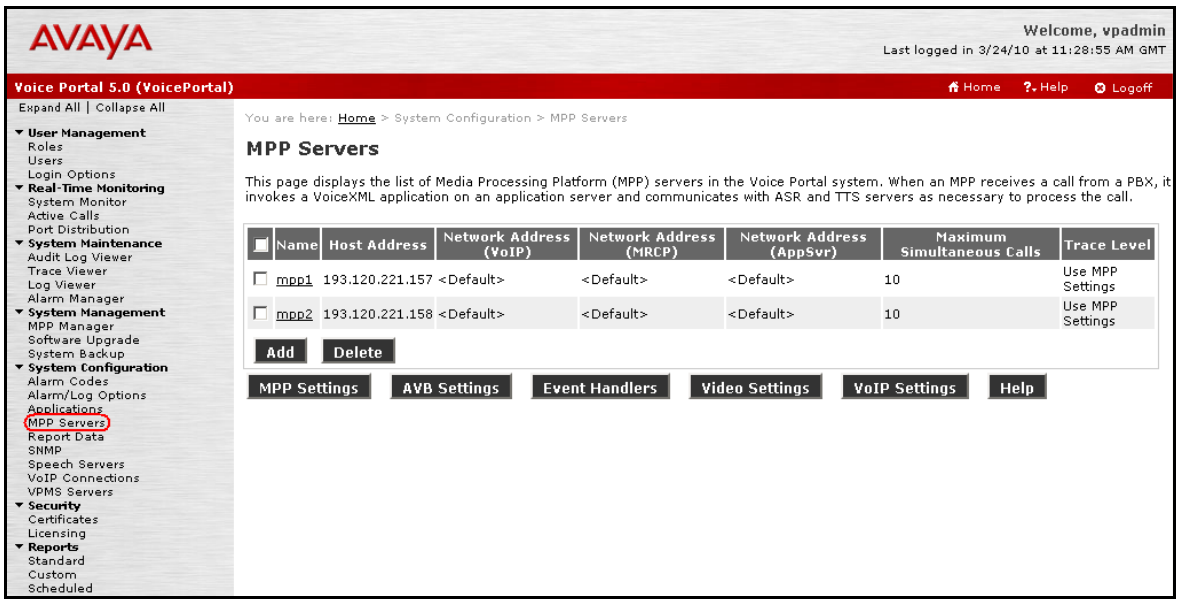
7.2. Add the MPP Servers

Add the required MPP servers (two in these Application Notes) by navigating to the **MPP Servers** screen. In the MPP Server configuration page, specify a descriptive name and the **Host Address** of each MPP server. Also, specify the **Maximum Simultaneous Calls** supported by each MPP server. The figure below shows the configuration for the first MPP server. Repeat these steps for the second MPP server.



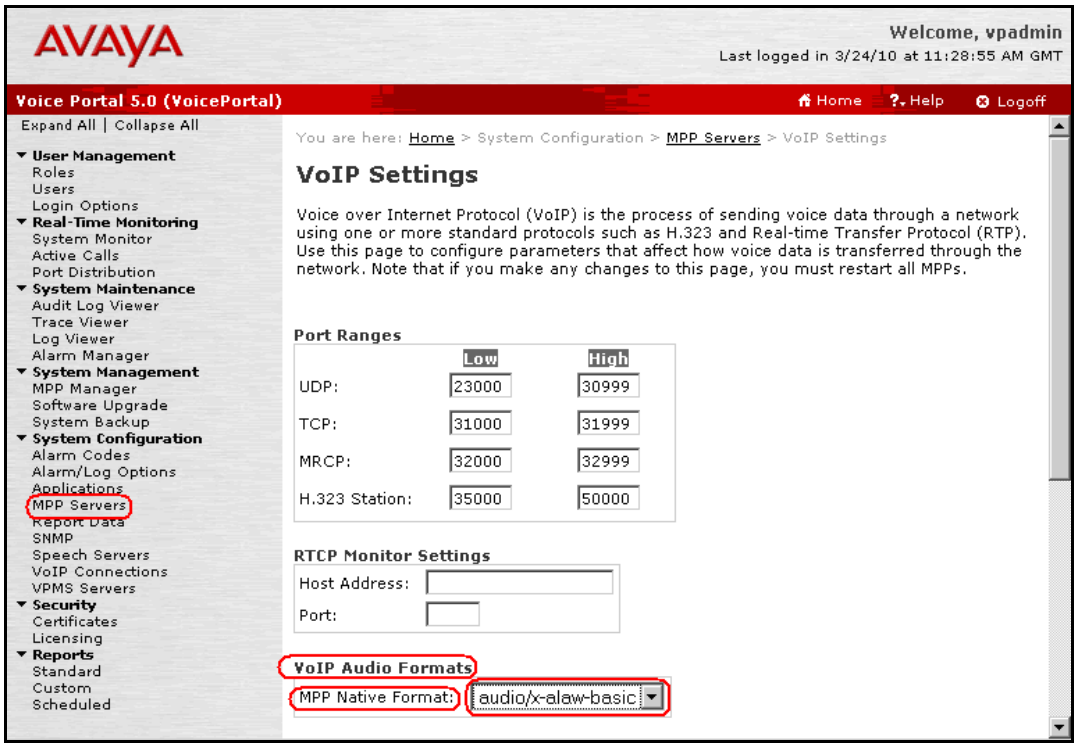
©2010 Avaya Inc. All Rights Reserved.

The following screen summarizes the process after the second MPP server is added into the system.



7.3. Configuring the VoIP Audio Format

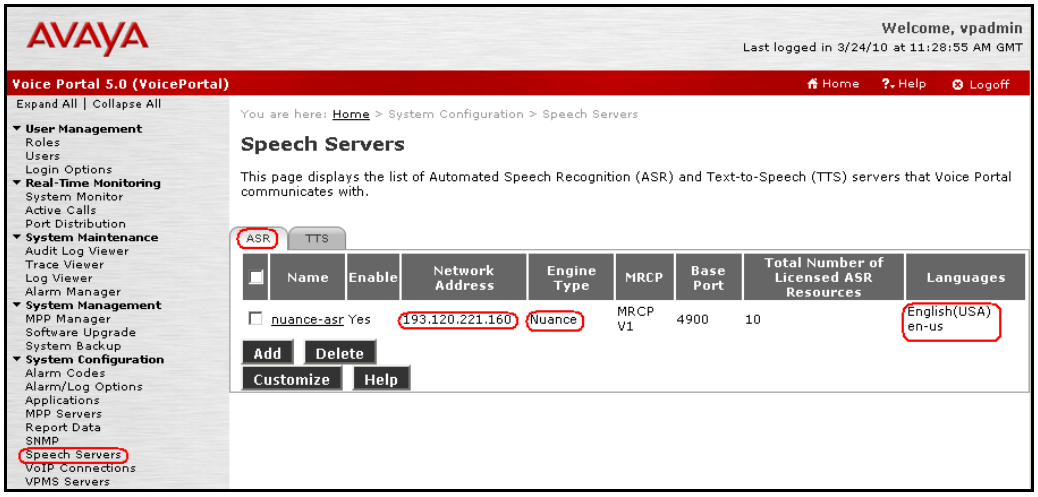
From the MPP Servers page shown in **Section 7.2**, click on the VoIP Settings button to activate VoIP Settings. Under the **VoIP Audio Format** section, set the **MPP Native Format** field to **audio/x-alaw-basic**.



©2010 Avaya Inc. All Rights Reserved.

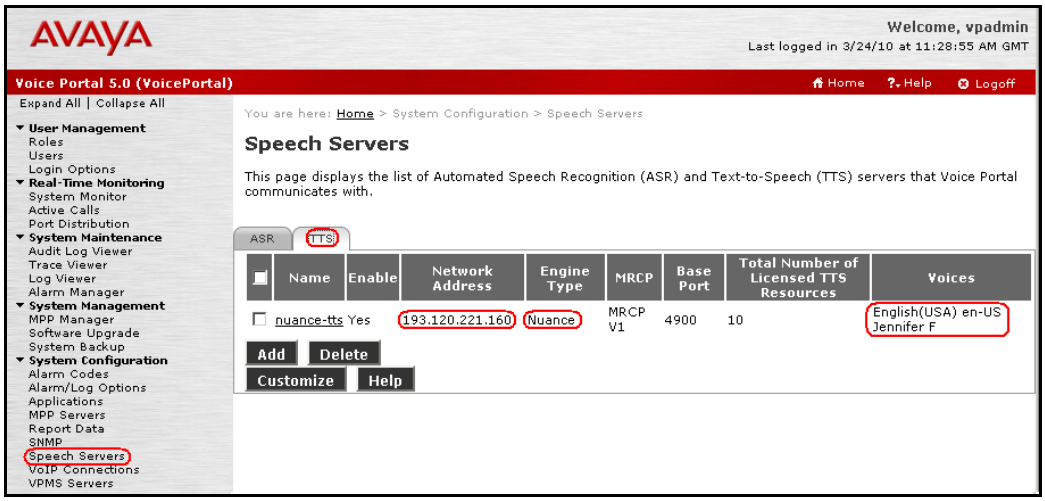
7.4. Add an ASR Server

To configure the ASR server, click on **Speech Servers** in the left pane, select the **ASR** tab, and then click **Add**. Set the **Engine Type** to the appropriate value. In this configuration, a Nuance ASR server was used so the engine type was set to **Nuance.** Set the **Network Address** field to the IP address of the speech server and select the desired **Languages**. The other fields were set to their default values. The following figure shows the screen after the ASR server has already been configured.



7.5. Add a TTS Server

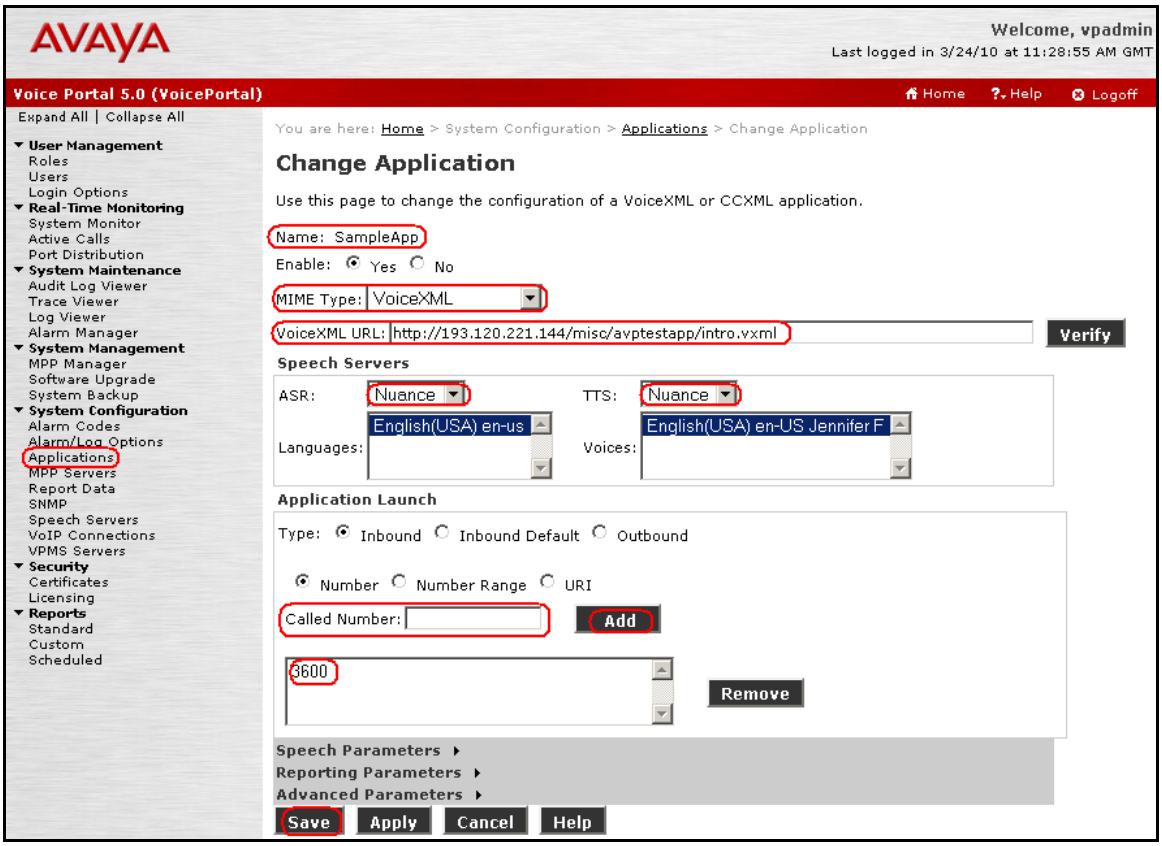
To configure the TTS server, click on **Speech Servers** in the left pane, select the **TTS** tab, and then click **Add**. Set the **Engine Type** to the appropriate value. In this configuration, a Nuance TTS server was used so the engine type was set to **Nuance**. Set the **Network Address** field to the IP address of the speech server and select the desired **Languages**. The other fields were set to their default values. The following figure shows the screen after the ASR server has already been configured.



©2010 Avaya Inc. All Rights Reserved.

7.6. Add an Application

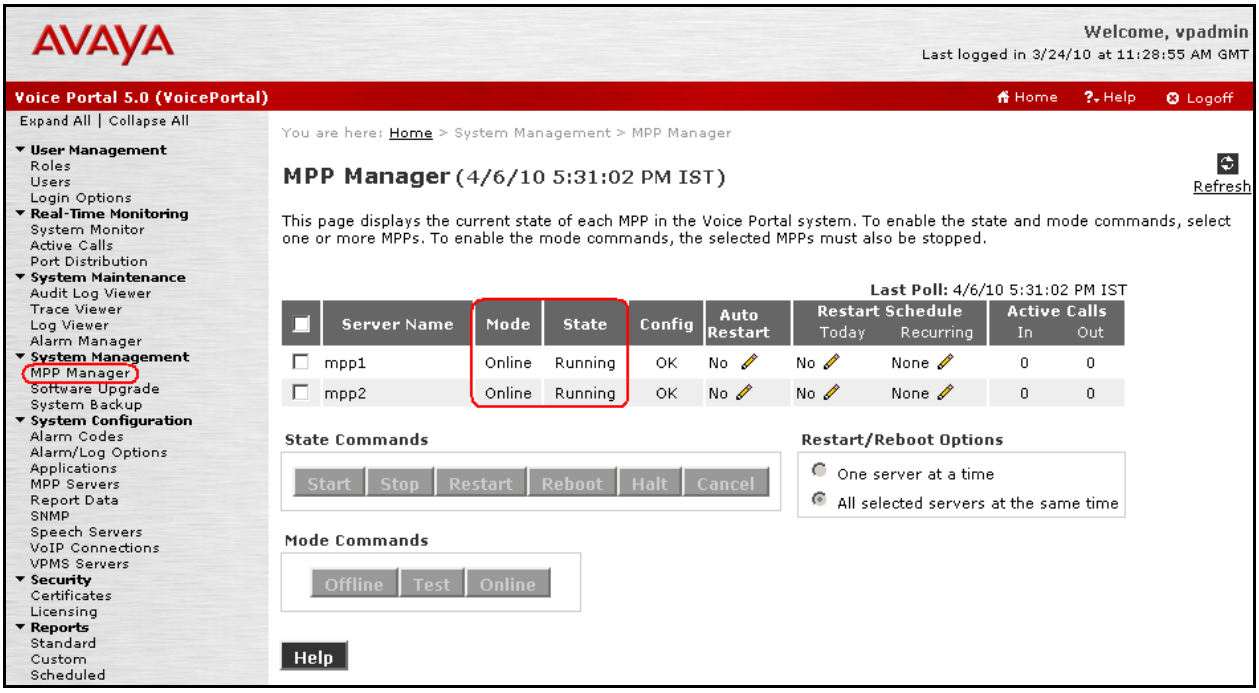
On the **Change Applications** page, add a Voice Portal application. Specify a **Name** for the application, set the **MIME Type** field to the appropriate value (e.g., **VoiceXML**), and set the **VoiceXML URL** field to point to a VoiceXML application on the application server. Next, specify the type of ASR and TTS servers to be used by the application and the called number that invokes the application (**Nuance** in these Application Notes). The called number is entered into the **Called Number** field and then the **Add** button is clicked. Click on **Save** once completed. The following screenshot summarizes the process.



©2010 Avaya Inc. All Rights Reserved.

7.7. Start the MPP Servers

Start the MPP servers from the **MPP Manager** page as shown in the figure below. Select each MPP and then click the **Start** button. The **Mode** of each MPP should be **Online** and the **State** should be **Running.**



©2010 Avaya Inc. All Rights Reserved.

8. Dialogic IMG 1010 Configuration

This section displays the configuration for enabling the IMG to interoperate with Session Manager. The IMG is administered using the Dialogic Gate Control Element Management System (GCEMS) and ClientView running on a Linux server. Refer to **[18], [19]** and **[20].** Note that this section displays the provisioning that was utilized for this sample configuration, and does not show exhaustive procedures for administering an initial configuration. For example, the screens for adding "new" elements to this sample configuration are not shown. However, the sequence of these procedures is relevant, as the configuration was administered in the order presented. Refer to the on-line help available on the Dialogic website regarding procedures/commands to administer an initial configuration. **Figure 2** illustrates the main window of the ClientView application that was utilized to provision the IMG. The following panes appear in the main window:

- The **Configuration Tree**, which is located in the top-left portion of the main window. This pane contains all of the items that can be configured. Right-click an item to access additional configuration items. Creating an entry in the Configuration Tree opens the corresponding Configuration Pane.
- The **Configuration Pane**, which is located in the top-right portion of the main window. This pane shows the properties of the selected object. This pane is used to view and edit the configuration.
    - The column titled **As-Configured**, shows the current configuration for parameters, as defined by the **Property** column. Enter or edit values in the **User-Specified** column.

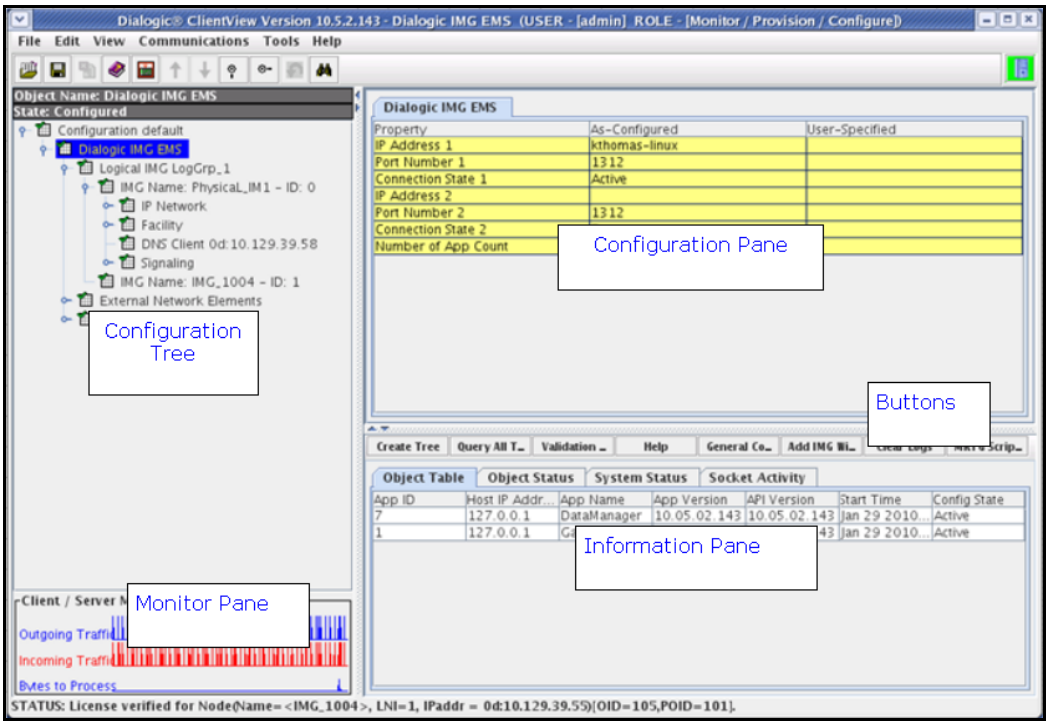
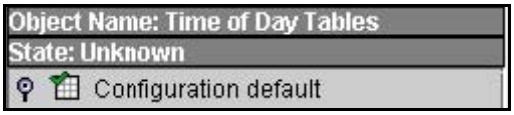


Figure 2 - Dialogic Inc. ClientView Main Window

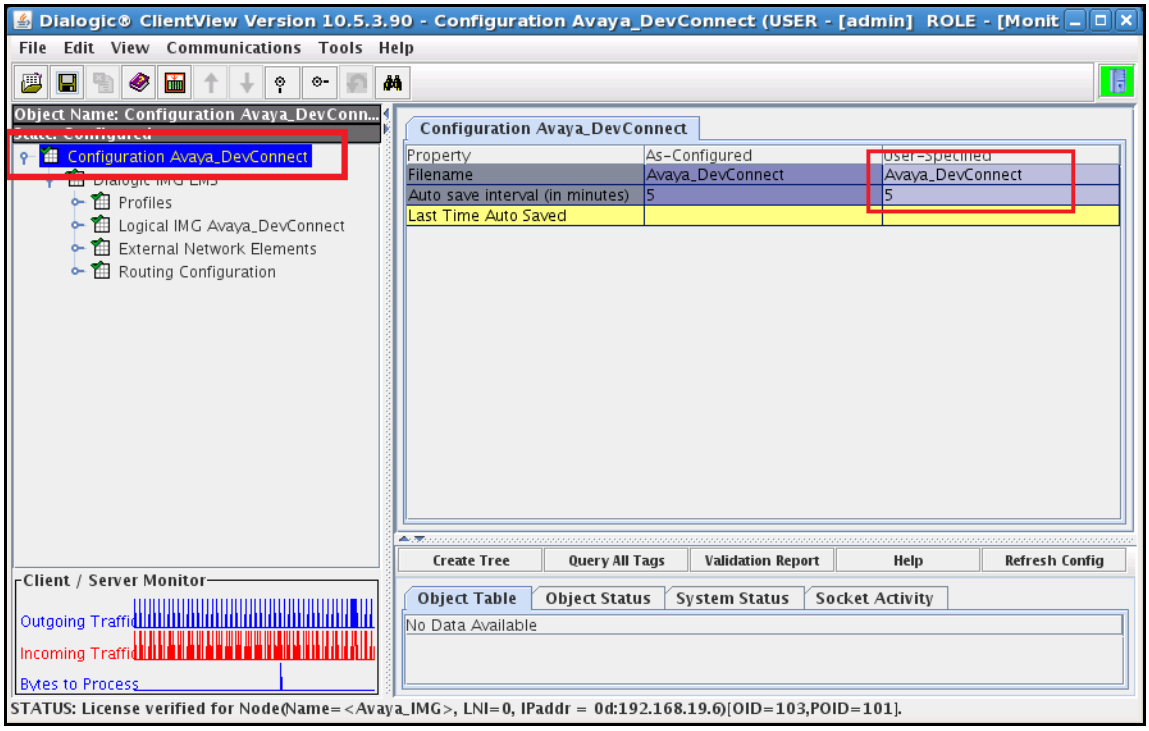
©2010 Avaya Inc. All Rights Reserved.

8.1. IMG Configuration Name

A default configuration file named “default” is created when ClientView connects to GCEMS. To save the configuration file with a new name, select the Filename property in the Configuration pane, and enter a new name.



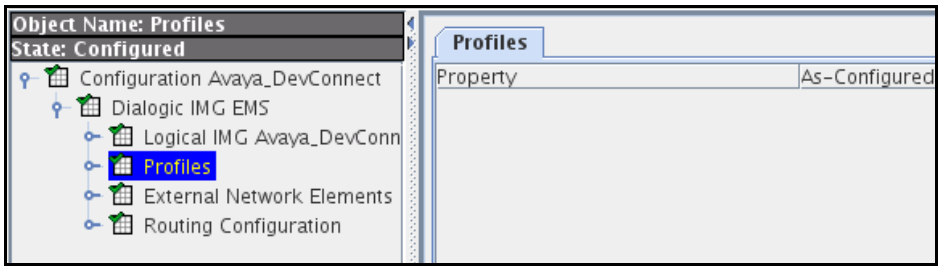
Enter a descriptive name in the **Filename** field in the Configuration Pane. To save the changes, right-click **Configuration Avaya_DevConnect**, and select **Commit**. The picture below shows the actions performed on IMG 1010



©2010 Avaya Inc. All Rights Reserved.

8.2. Profiles

Configure a Profile object by right-click **Dialogic IMG EMS** in the Configuration Tree, and select **Profile**.

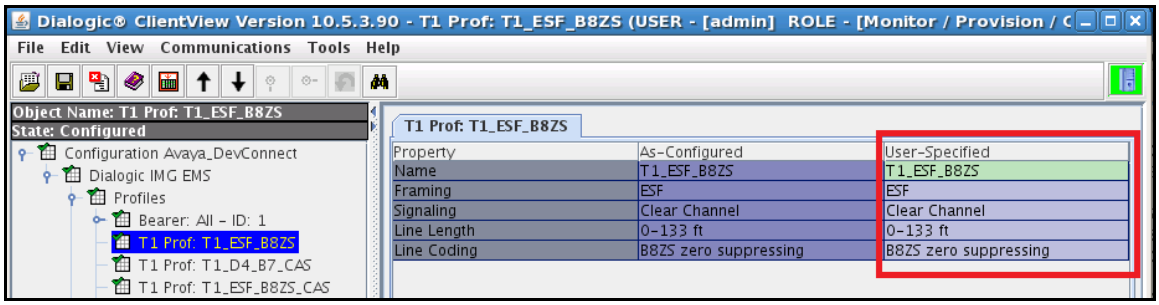


8.2.1. DS1 Profiles

Configure the T1 or E1 Physical Span for properties as follows;

- Right-click the Profile in the Configuration Tree, and select **New T1** or **E1 Profile**. (A single profile can be used for many DS1 spans that all use the same configuration.)
- Enter a name for the profile i.e. **T1_ESF_B8ZS**
- Select **Clear Channel** from the drop down list for the **Signaling** field in the Configuration Pane.
  Note: **Clear Channel** is used for ISDN-PRI or SS7 spans.
- Configure remaining settings to match network configuration.
- Create additional profiles as needed.

The picture below shows the actions performed on IMG 1010



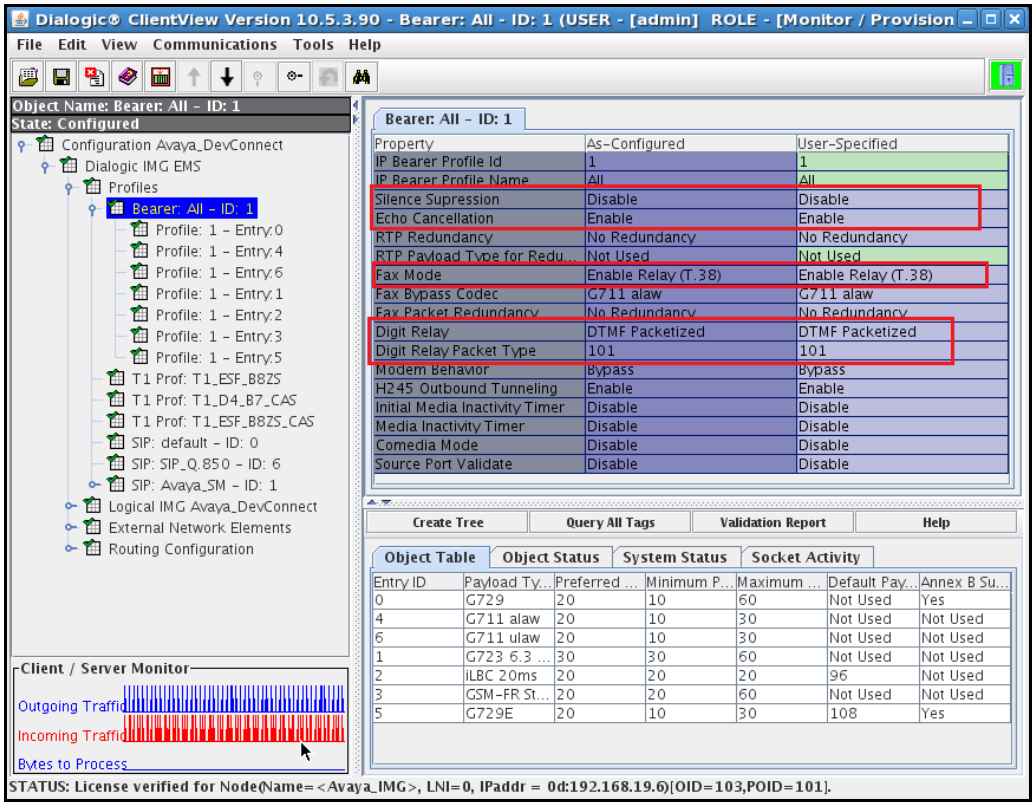
©2010 Avaya Inc. All Rights Reserved.

8.2.2. IP Bearer Profiles

Configure an IP Bearer Profile corresponding to Session Manager as follows:

- Right-click **Profiles** in the Configuration Tree, and select **New IP Bearer Profile**.
- Enter a descriptive name for the IP Bearer Profile in the **IP Bearer Profile Name** field in the Configuration Pane.
- Select the settings for remaining fields as appropriate.

The picture below shows the results after the configuration of IP Bearer Profile has executed on IMG 1010



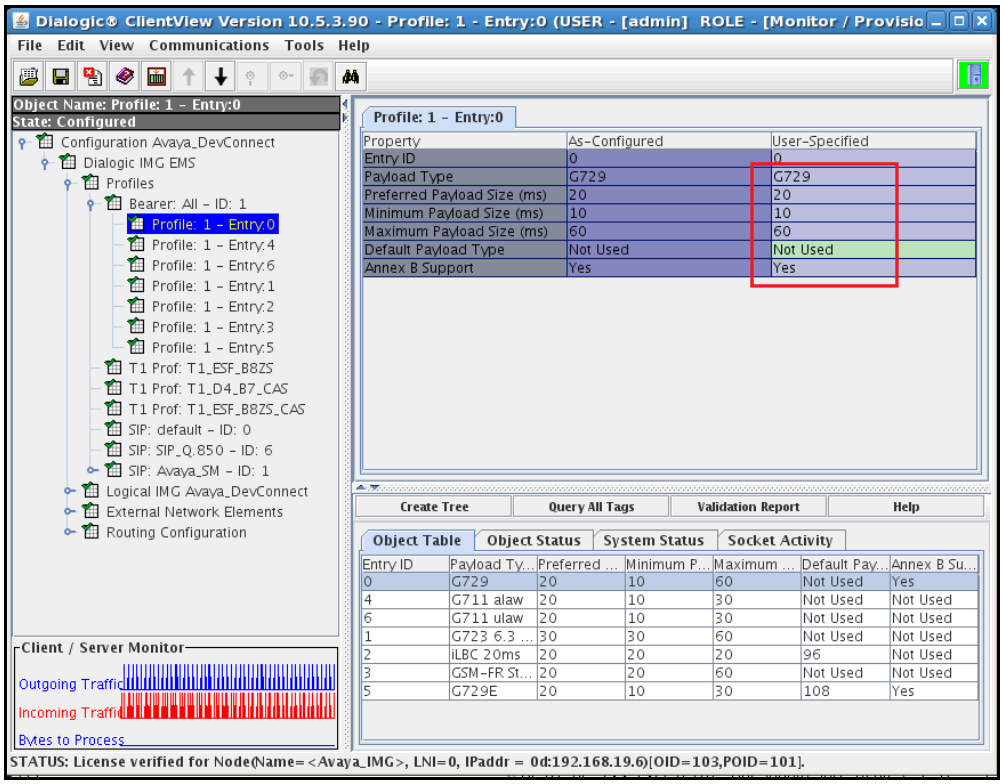
©2010 Avaya Inc. All Rights Reserved.

8.2.2.1 New IP codec in Bearer Profile

Assign one or more codec's to the IP Bearer Profile as follows:

- Right-click the IP Bearer Profile in the Configuration Tree, and select **New Supported Vocoders**.
- Select a codec from the drop down list for the **Payload Type** field in the Configuration Pane.

To save the changes, right-click **Profile: 1- Entry:0**, and select **Commit**.
Note: if using G.729A on Communication Manager, set **Annex B** to **No** in Bearer Profile.



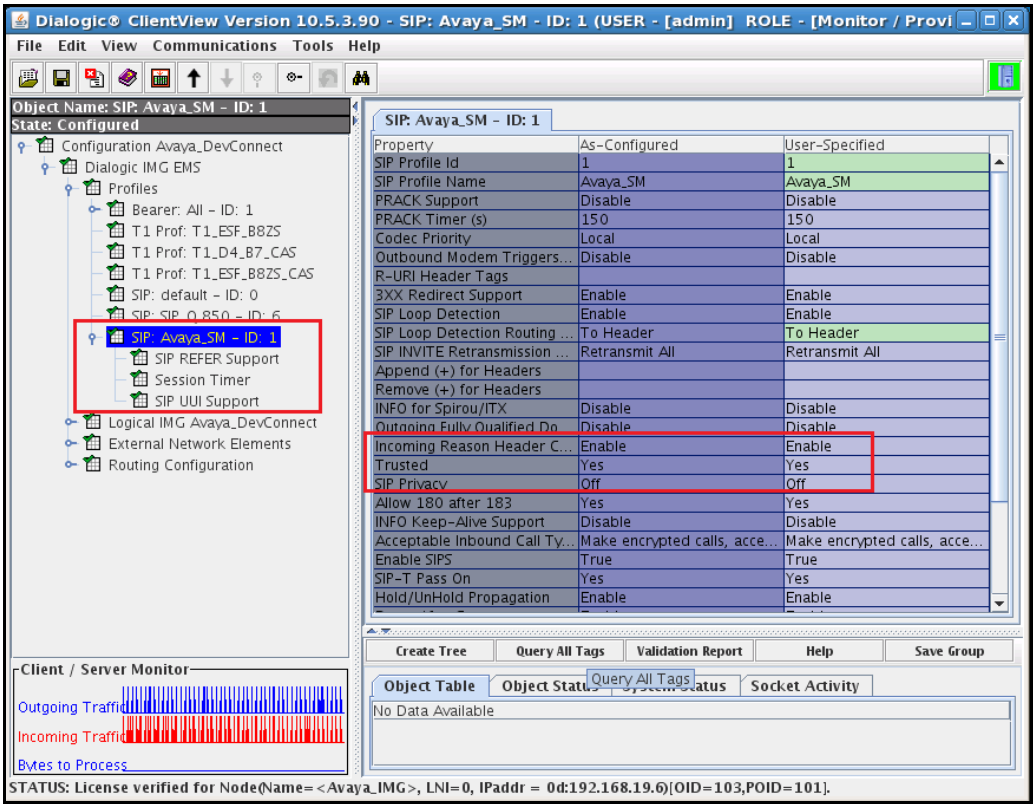
©2010 Avaya Inc. All Rights Reserved.

8.2.3. SIP Profiles

The SIP profile is optional and allows unique Signaling requirements for SIP gateways to be configured. Multiple gateways can use the same profile, or each gateway can have a unique profile as needed. To configure a SIP Profile:

- Right-click **Profiles** in the Configuration Tree, and select **New SIP SGP**.
- Enter a descriptive name for the SIP Profile in the Configuration Pane.
- Select the settings for remaining fields as appropriate.

Note: If needed P-AI provide from Session Manager, set **SIP Privacy** to **On**

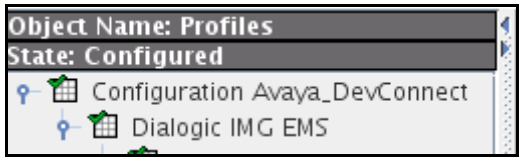


©2010 Avaya Inc. All Rights Reserved.

8.2.4. Logical IMG

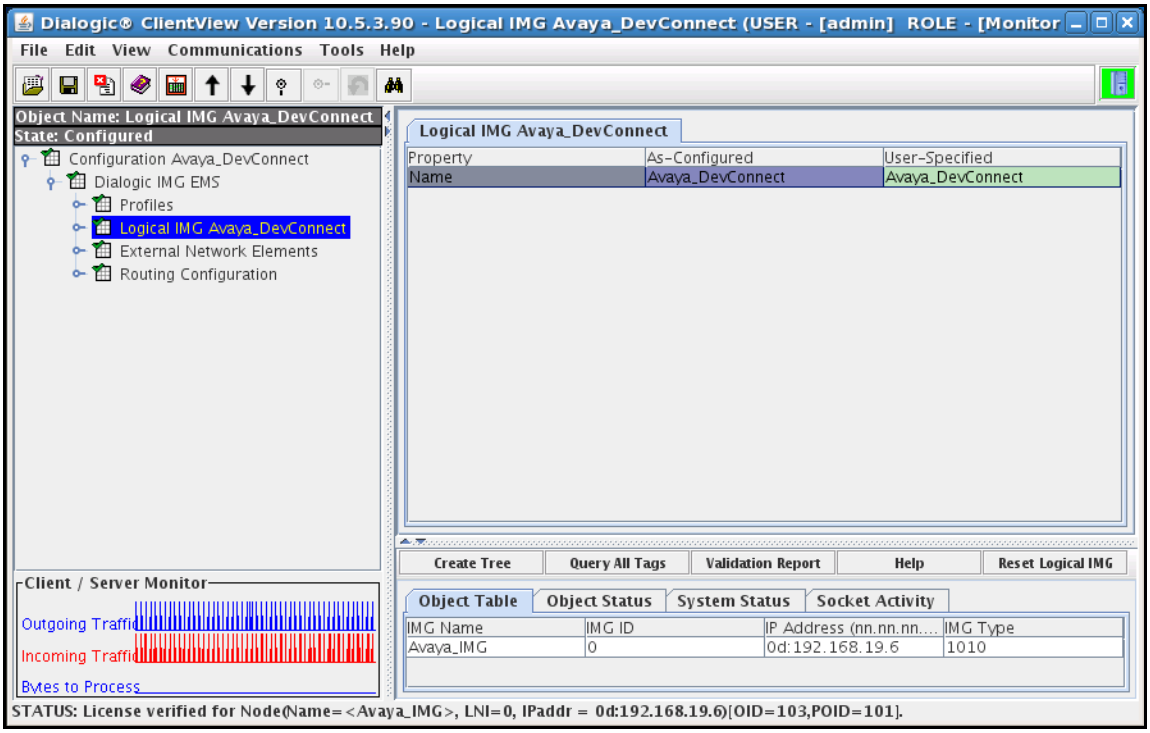
Create a logical IMG as follows:

- Right-click **Dialogic IMG EMS** in the Configuration Tree, and select **New Logical IMG**.



- Enter a descriptive name for the logical IMG in the **Name** field in the Configuration Pane.

To save the changes, right-click **Logical IMG Avaya-IMG**, and select **Commit**.



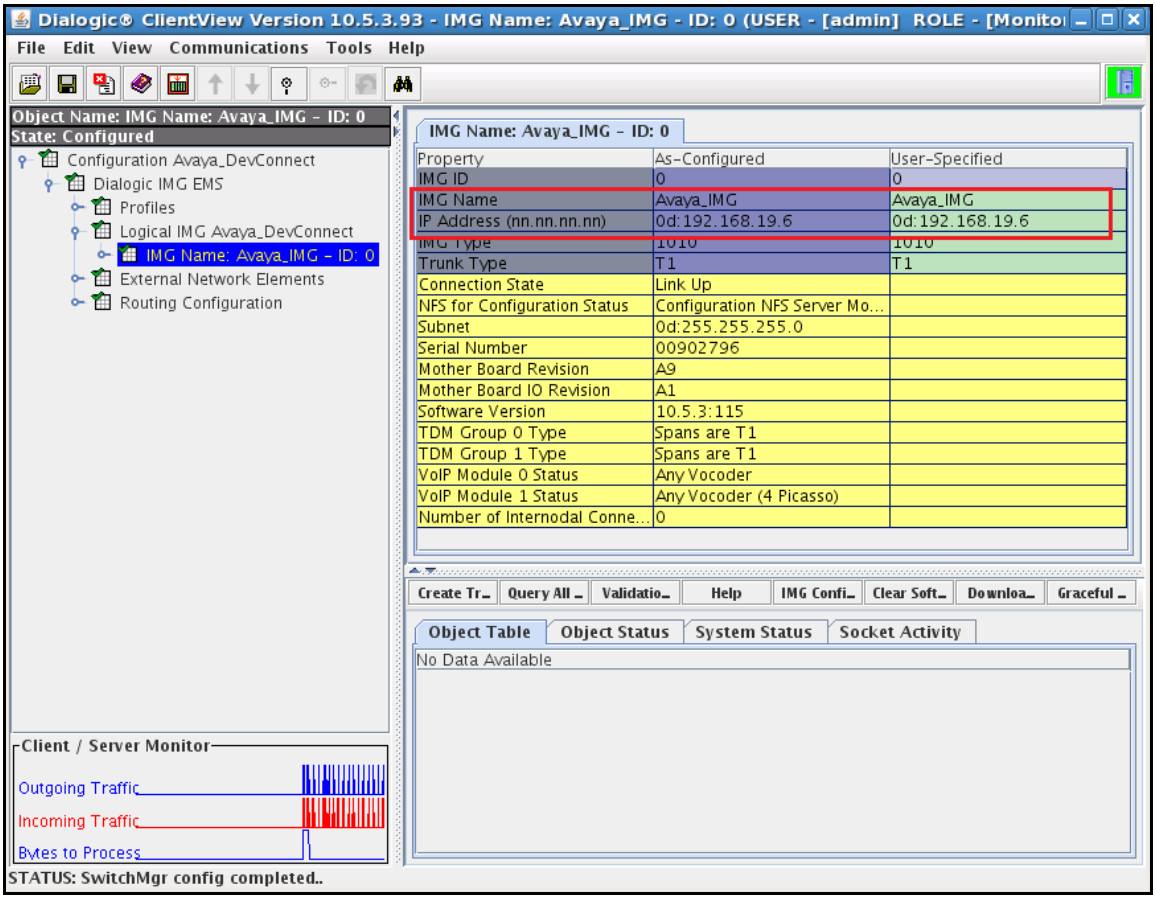
©2010 Avaya Inc. All Rights Reserved.

8.2.4.1 Physical IMG

Create a physical IMG as follows:

- Right-click the logical IMG in the Configuration Tree, and select **New Physical IMG**.
- Enter a descriptive name for the physical IMG in the **IMG Name** field in the Configuration Pane.
- Enter the IP address of the physical IMG in the **IP Address** field. This is the same IP address assigned via DHCP or the SD card to the CTRL 0 port on the back of the IMG.
- Use default settings for remaining fields.

To save the changes, right-click **IMG Name: Avaya-IMG - ID:0**, and select **Commit**.



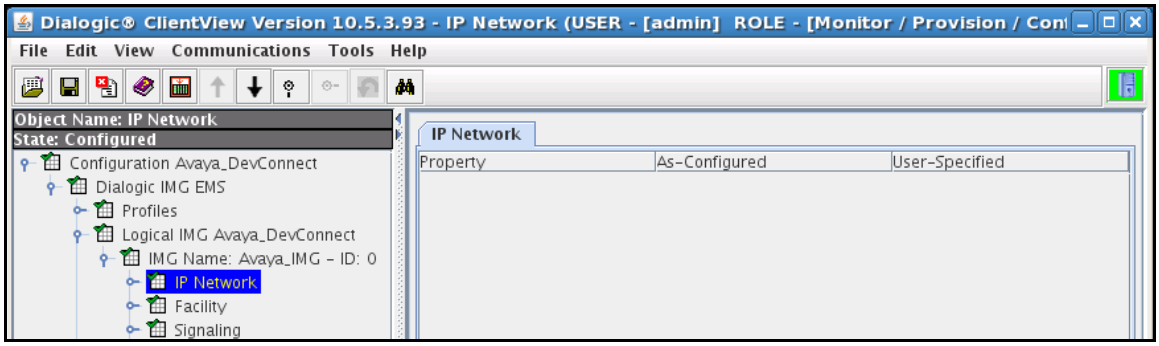
©2010 Avaya Inc. All Rights Reserved.

8.2.4.1.1 Network Interfaces

Create an object for Network Interfaces as follows:

- Right-click the physical IMG in the Configuration Tree, and select **New Network Interfaces**.

To save the changes, right-click **Network Interfaces**, and select **Commit**. The resultant provisioning is shown below.

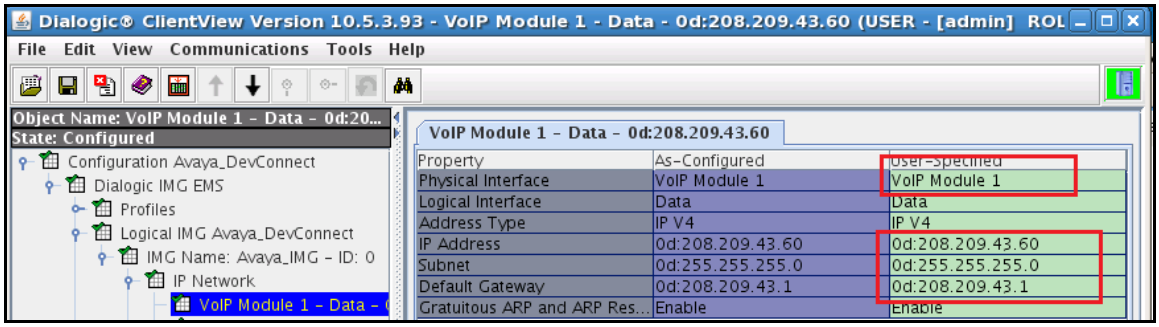


8.2.4.1.2 Network Interface VoIP module

Create a Network Interface corresponding to VoIP Module 0: Port 0 as follows:

- Right-click **IP Interfaces** in the Configuration Tree, and select **New IP Address**.
- Select **VoIP Module 0: Port 0** from the drop down list for the **Physical Interface** field in the Configuration Pane.
- Administer settings for module's IP network configuration in the **IP Address**, **Subnet** and **Default Gateway** fields respectively.
- Use default settings for remaining fields.

To save the changes, right-click **VoIP Module 0: Port 0**, and select **Commit**. Repeat this step for VoIP module 1 if needed.
Note: this IP Address will be used for the media.



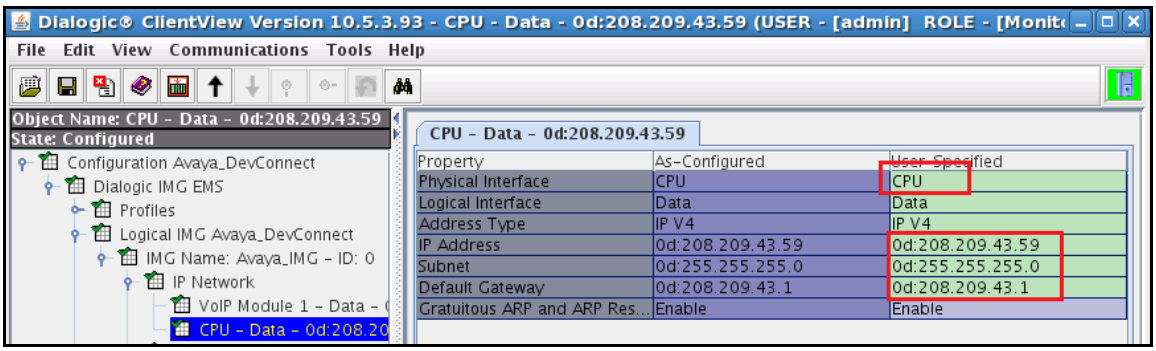
©2010 Avaya Inc. All Rights Reserved.

8.2.4.1.3 Network Interface CPU

Network Interface corresponding to the CPU is an optional IP address that can later be used for things such as SIP Signaling, H.323 Signaling, DNS, Radius, and to interface with other external network elements. To create a Network Interface corresponding to the CPU as follows:

- Right-click **Network Interfaces** in the Configuration Tree, and select **New Network Interface**.
- Select **CPU** from the drop down list for the **Physical Interface** field in the Configuration Pane.
- Administer settings for IP network configuration in the **IP Address**, **Subnet** and **Default Gateway** fields respectively.

To save the changes, right-click **CPU**, and select **Commit**.
Note: this IP Address will be used for SIP signalling with Session Manager.

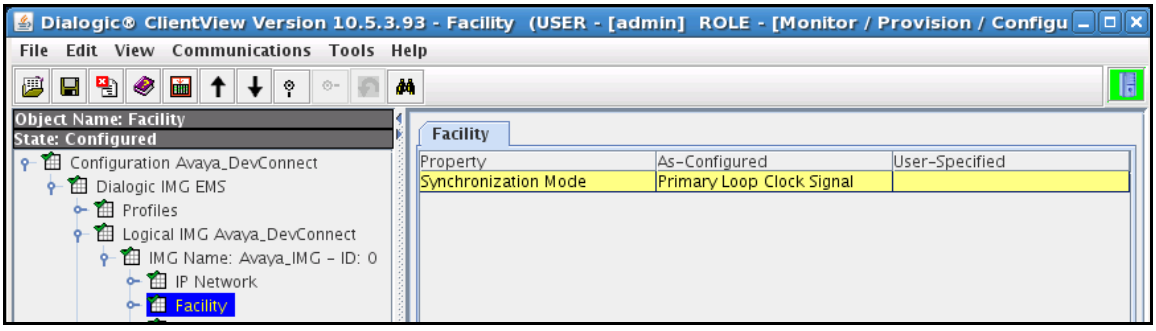


8.2.4.1.4 Facilities (DS1 and VoIP)

Create an object for a Facility as follows:

- Right-click the physical IMG in the Configuration Tree, and select **New Facility**.

To save the changes, right-click **Facility**, and select **Commit**.



©2010 Avaya Inc. All Rights Reserved.

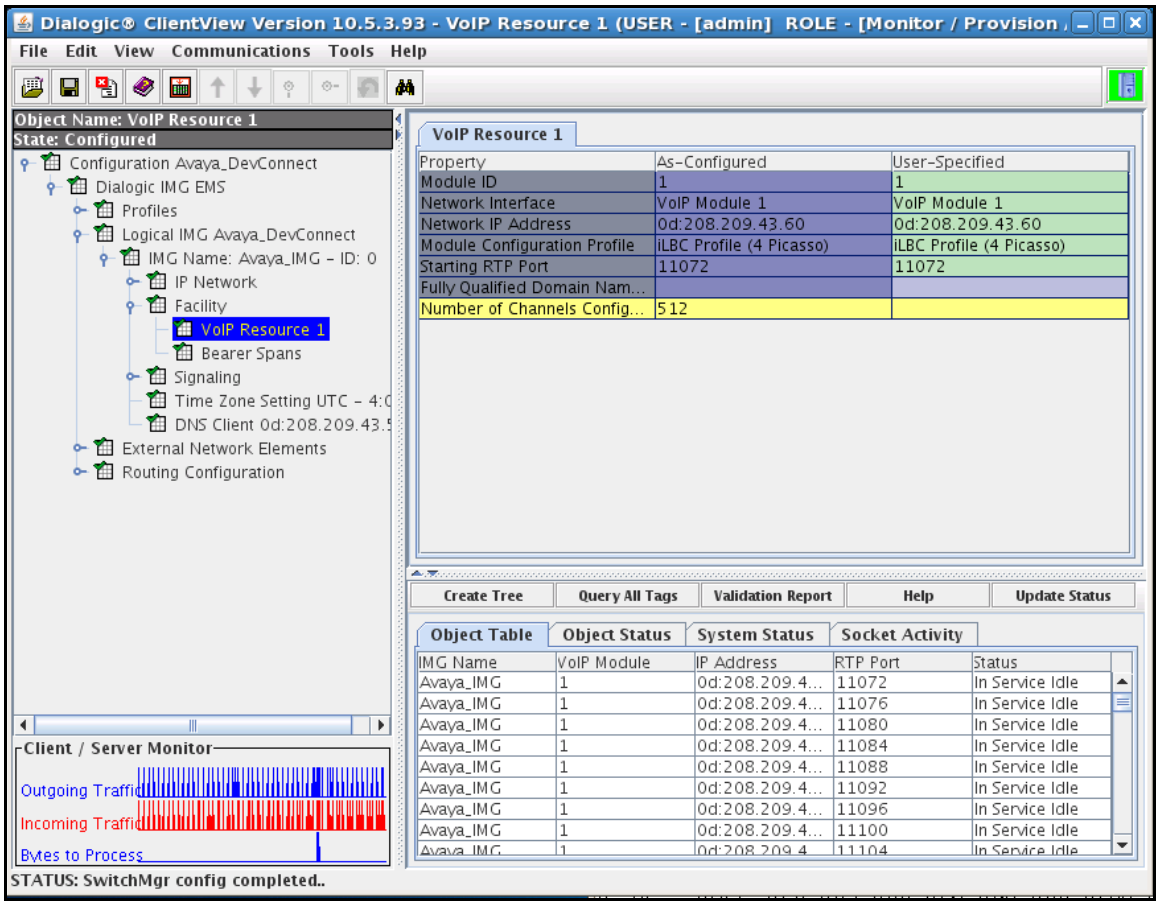
8.2.4.1.5 VoIP Facilities

Configure VoIP Facilities as follows:

- Right-click **Facility** in the Configuration Tree, and select **New Bearer - IP**.
- Use default settings for all fields.

To save the changes, right-click **VoIP Resource 1**, and select **Commit**. Repeat this step for VoIP module 1 if needed.

Note: The Network **IP Address** field is populated from the configuration provided for VoIP Module 0: Port 0 in **Section 8.3**.



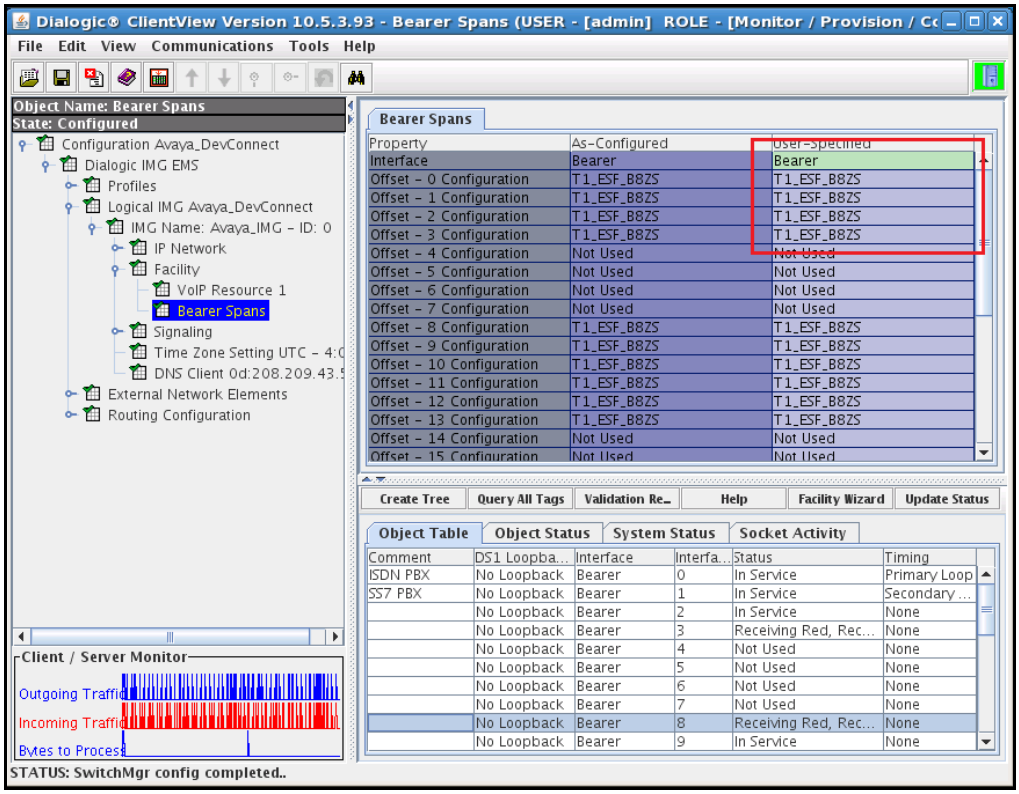
©2010 Avaya Inc. All Rights Reserved.

8.2.4.1.6 DS1 Facilities

Configure a TDM DS1 T1 or E1 as follows:

- Right-click **Facility** in the Configuration Tree, and select **New TDM Spans**.
- Select **Bearer** or **Signaling** spans.
- In the configuration pane select the DS1 span and select the profile for that span.

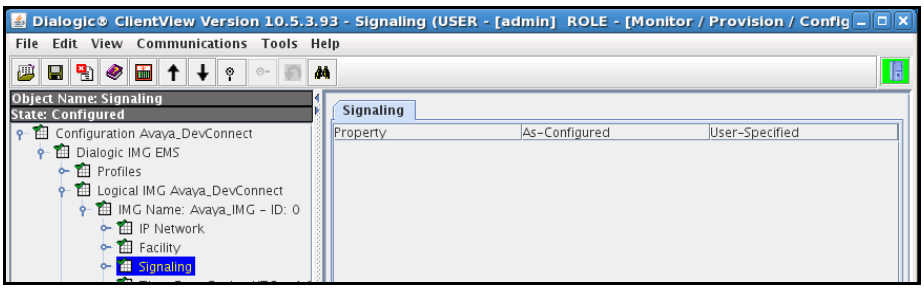
TDM spans will be brought in service & if the network is also in service then the span status will show in service. If the network is not in service the span status will show receiving remote alarm. To save the changes, right-click on the **Bearer** or **Signaling** span object, and select **Commit**.



©2010 Avaya Inc. All Rights Reserved.

8.2.4.1.7 IMG Signaling (ISDN, SS7, SIP, H.323)

Create an object for Signaling with right-click the physical IMG in the Configuration Tree, and select **New Signaling**. To save the changes, right-click **Signaling**, and select **Commit**. The resultant provisioning is shown below.

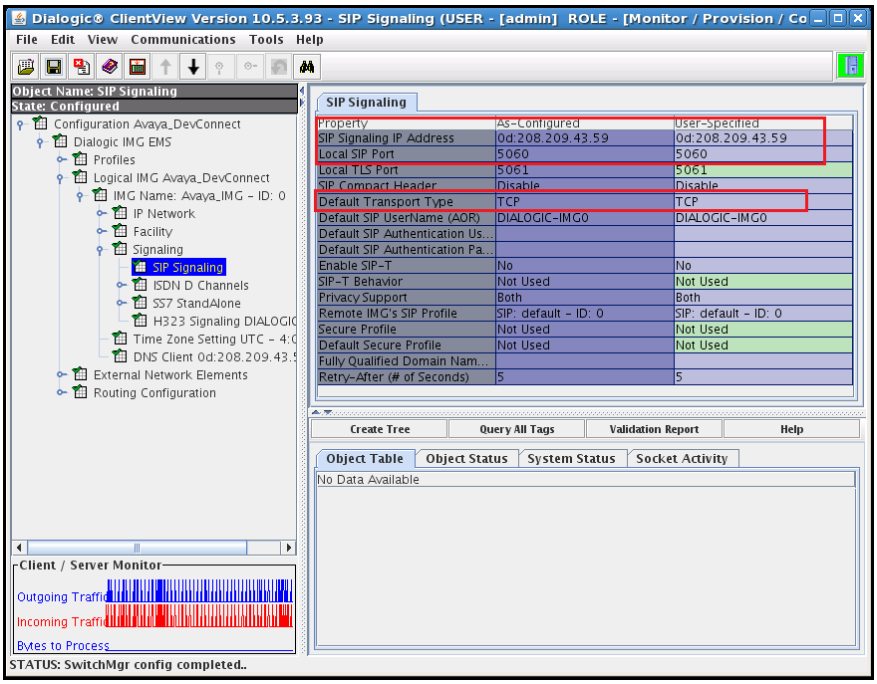


8.2.4.1.8 SIP Signaling

Configure SIP Signaling as follows

- Right-click **Signaling** in the Configuration Tree, and select **New SIP**.
- Administer settings in the Configuration Pane that enable SIP connectivity between the IMG and other SIP User Agents as follows:
  - Enter the IP address assigned to the IMG in the **SIP Signaling IP Address** field.
  - Enter values in the **Local SIP Port** and **Default Transport**.

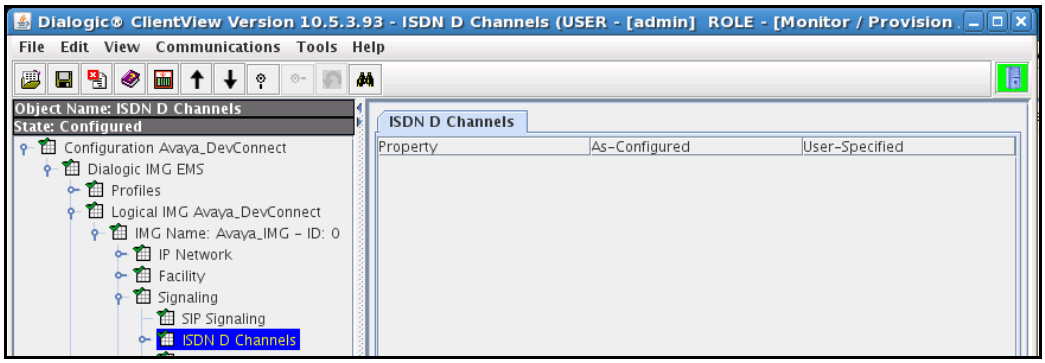
Use default settings for remaining fields. To save the changes, right-click **SIP Signaling**, and select **Commit**.



©2010 Avaya Inc. All Rights Reserved.

8.2.4.1.9 ISDN Signaling

Create an object for ISDN as follows with right-click **Signaling** in the Configuration Tree, and select **New ISDN**. To save the changes, right-click **ISDN D Channels**, and select **Commit**.

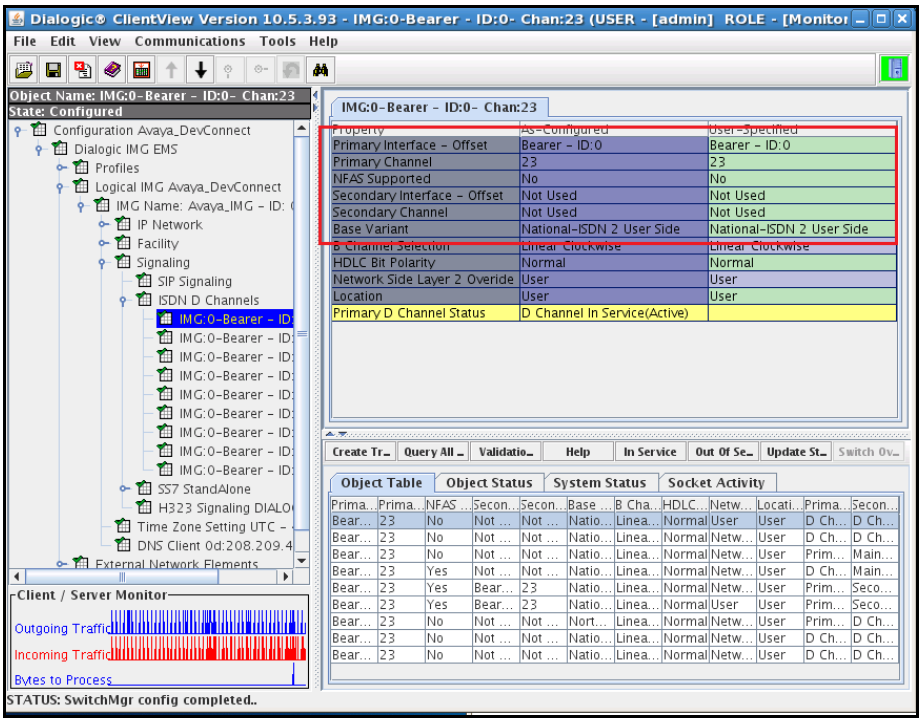


8.2.4.1.9.1 ISDN D Channel

Configure an ISDN D Channel as follows:

- Right-click **ISDN D Channels** in the Configuration Tree, and select **New ISDN D Channel**.
- Administer settings for the **Primary Interface, Channel**, and **Base Variant**.

Use default settings for remaining fields. To save the changes, right-click the **ISDN D channel**, and select **Commit**.



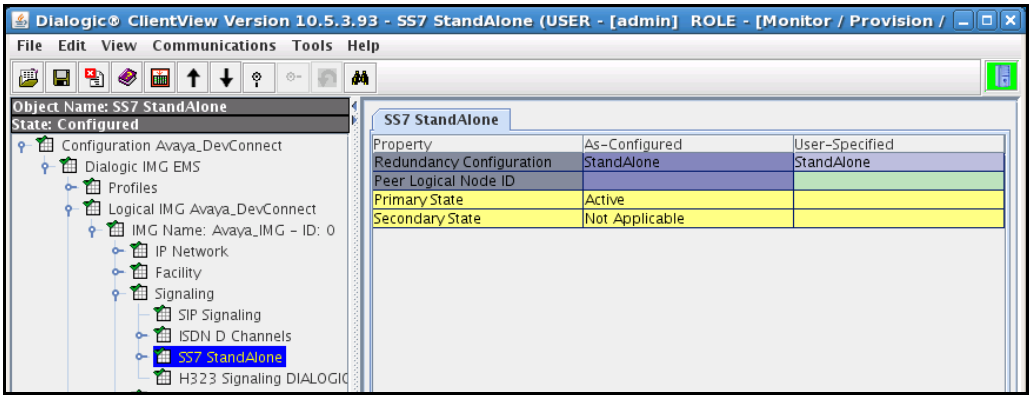
©2010 Avaya Inc. All Rights Reserved.

8.2.4.1.10 SS7 Signaling

Configure SS7 Signaling as follows:

- Right-click the Signaling in the Configuration Tree, and select **New SS7**.

To save the changes, right-click **New SS7**, and select **Commit**.



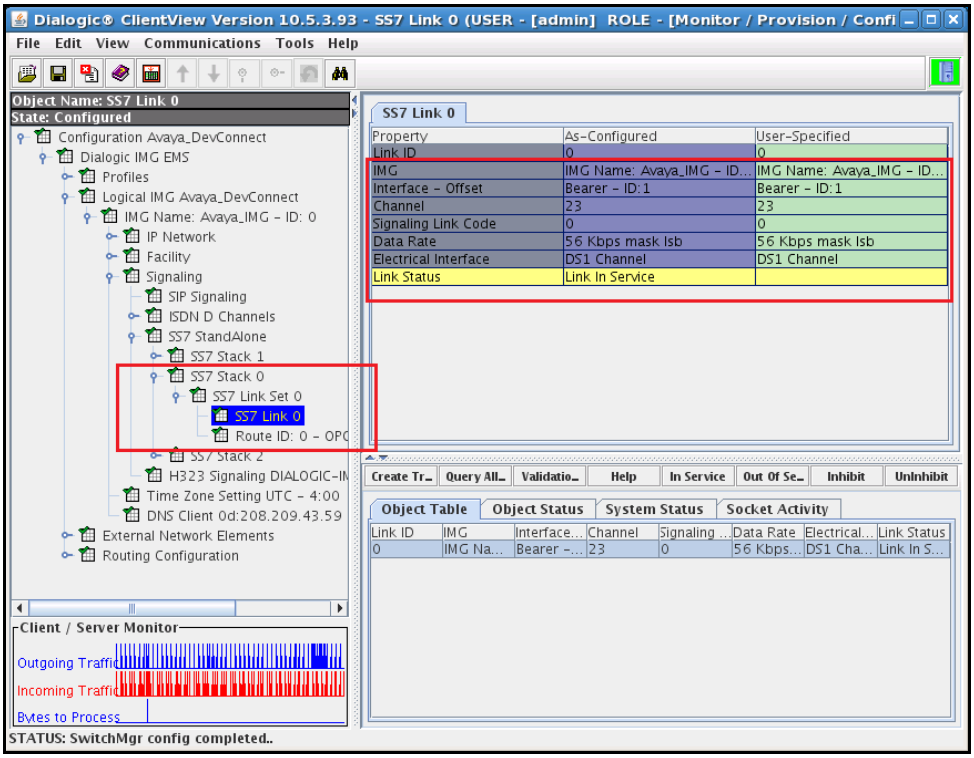
©2010 Avaya Inc. All Rights Reserved.

Configure SS7 Stack, Linkset, Links, & Routes as follows:

- Right-click **SS7** in the Configuration Tree, and select **New SS7 Stack**.

Configure fields to match SS7 Network:

- Right-click **SS7 Stack** and add **New SS7 Linkset**.
- Right-click **SS7 Linkset** and add **New SS7 Route**.
- Right-click **SS7 Linkset** and add **New SS7 Link**.



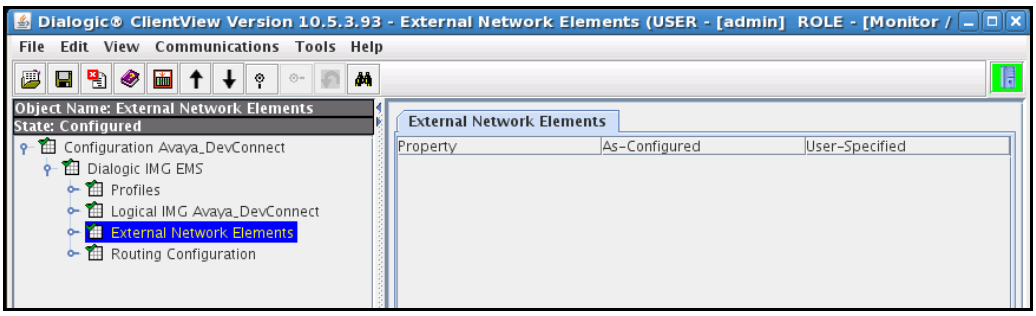
©2010 Avaya Inc. All Rights Reserved.

8.2.5. External Network Elements

Create an object for External Network Elements as follows:

- Right-click **Dialogic IMG EMS** in the Configuration Tree, and select **New External Network Elements**.

To save the changes, right-click **External Network Elements**, and select **Commit**.

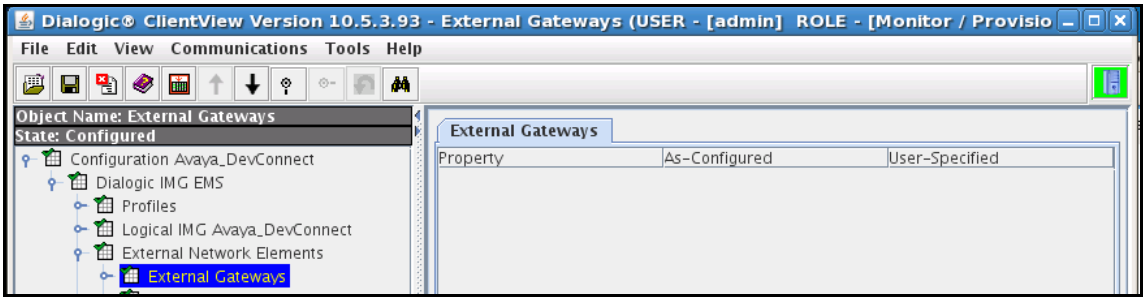


8.2.5.1 External Gateways

Create an object for External Gateways as follows:

- Right-click **External Network Elements** in the Configuration Tree, and select **New External Gateways**.

To save the changes, right-click **External Gateways**, and select **Commit**.



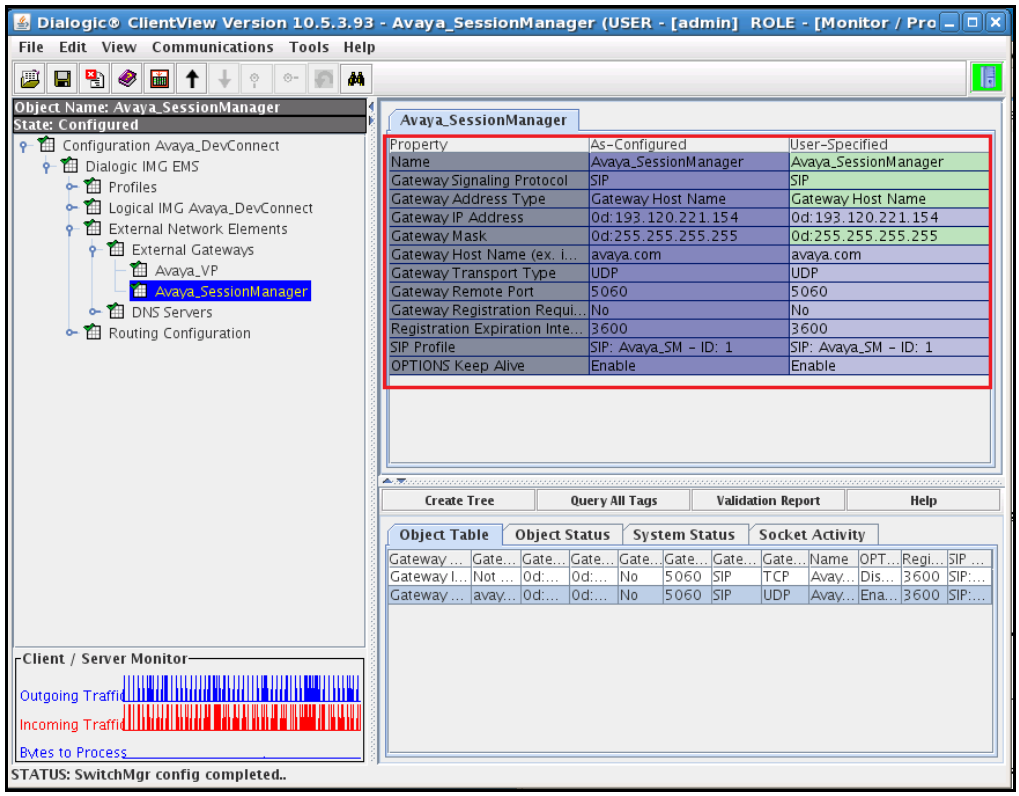
©2010 Avaya Inc. All Rights Reserved.

8.2.5.1.1 External Gateway

Configure an External Gateway:

- Right-click **External Gateways** in the Configuration Tree, and select **New External Gateway**.
- Enter a descriptive name for the Gateway in the **Name** field in the Configuration Pane.
- Select **SIP** from the drop down list for the **Gateway Signaling Protocol** field.
- Select **Address Type** and choose IP or Host Name from the drop down list.
- Enter the remaining fields to match the remote gateway.

To save the changes, right-click on the external gateway, and select **Commit**.



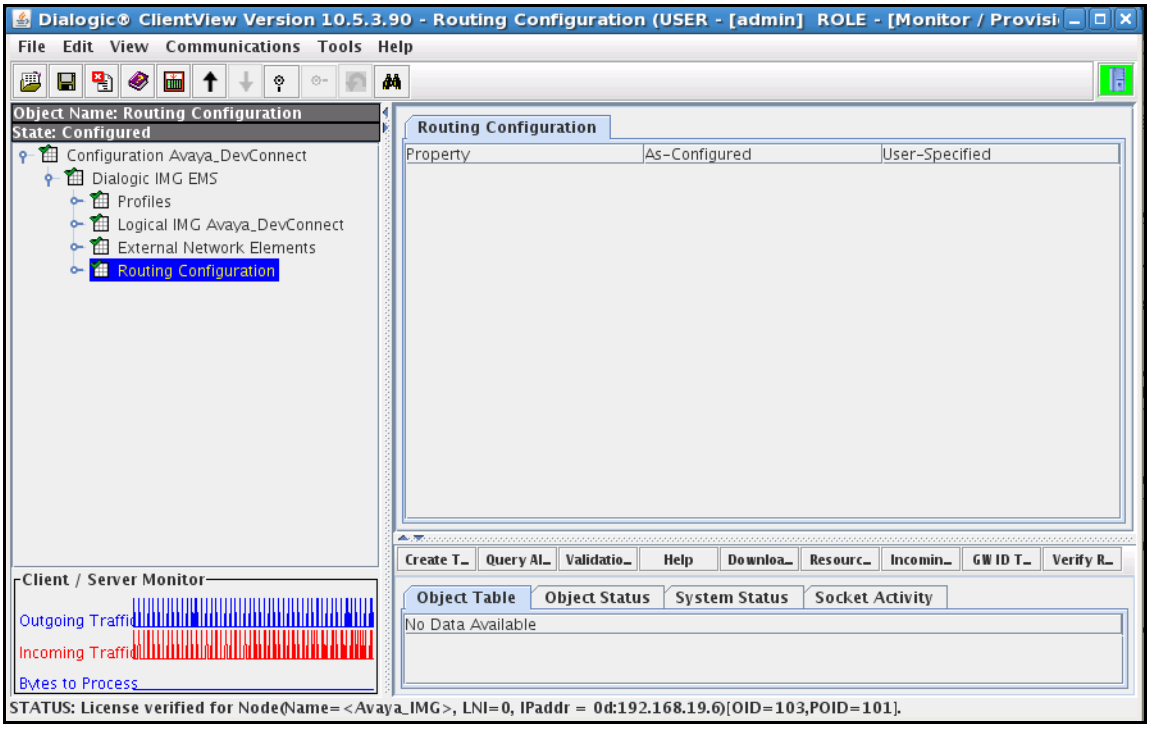
©2010 Avaya Inc. All Rights Reserved.

8.3. Routing configuration

Create an object for Routing Configuration as follows:

- Right-click **Dialogic IMG EMS** in the Configuration Tree, and select **New Routing Configuration**.

To save the changes, right-click **Routing Configuration**, and select **Commit**.

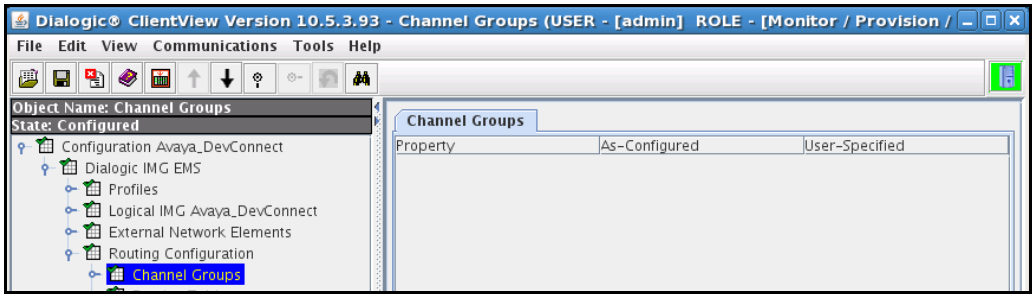


8.3.1. Channel Groups

Create an object for Channel Groups as follows:

- Right-click **Routing Configuration** in the Configuration Tree, and select **New Channel Groups**.

To save the changes, right-click **Channel Groups**, and select **Commit**.



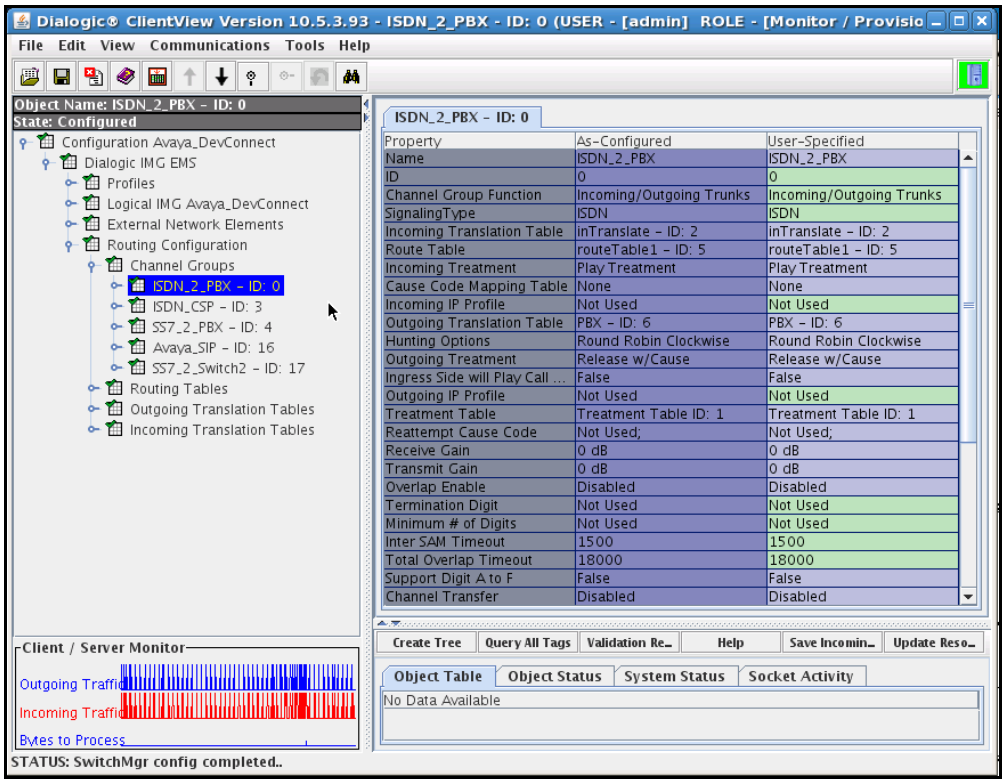
©2010 Avaya Inc. All Rights Reserved.

8.3.1.1 Channel Group (ISDN)

Configure an ISDN Channel Group:

- Right-click **Channel Groups** in the Configuration Tree, and select **New Channel Group**.
- Enter a descriptive name for the Channel Group in the **Name** field in the Configuration Pane.
- Select **ISDN** from the drop down list for the **Signaling Type** field.
- Select a hunt algorithm that selects B-channels inverse to the provisioning on the network from the drop down list for the **Hunting Options** field.

Use default settings for remaining fields. Note: The administration for the **Route Table** and **Translation table** fields are displayed in this screen, although the tables have not been created. When providing the IMG with an initial configuration, create a **Channel Group** first, then create a **Route Table** and optional **Translation Table**, then edit the **Channel Group** to include these tables. This note applies to all channel groups. To save the changes, right-click on the channel group, and select **Commit**.



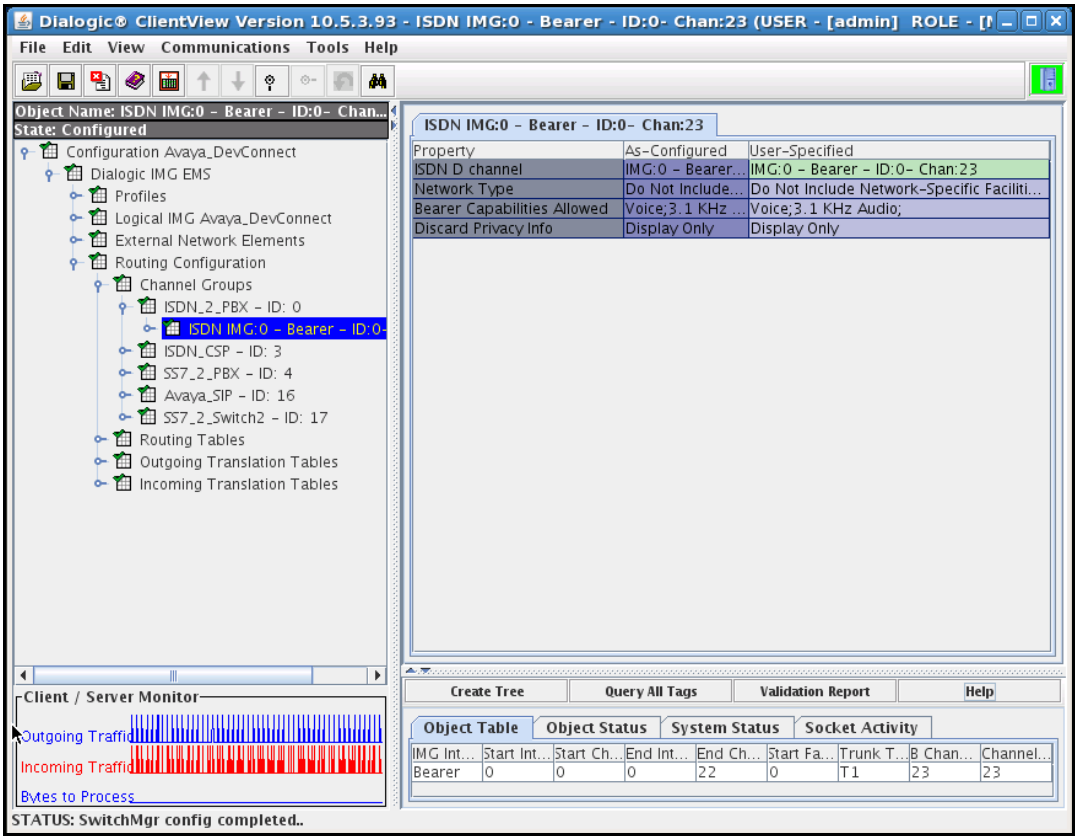
©2010 Avaya Inc. All Rights Reserved.

8.3.1.1.1 Assign ISDN D channel to Channel group

Assign a D-Channel configured under the Physical IMG to the Channel Group as follows:

- Right-click the Channel Group created in the Configuration Tree, and select **New ISDN Group**.
- Select the ISDN D channel. A single channel group can control 1 or more ISDN D Channels.
- Configure the other settings to match the network requirements.

To save the changes, right-click **ISDN Group** and select **Commit**.



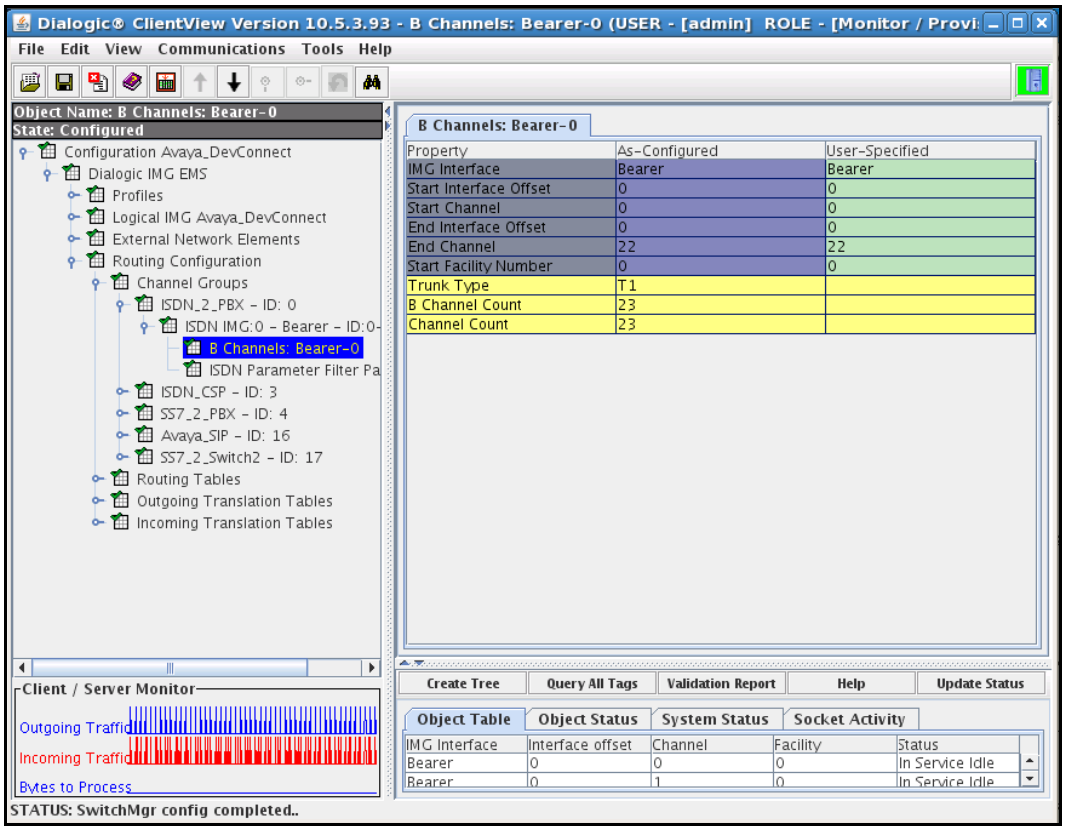
©2010 Avaya Inc. All Rights Reserved.

8.3.1.1.2 Assign ISDN B channels to the ISDN Group

Assign B-Channels to the ISDN Channel Group corresponding to PSTN provider as follows:

- Right-click the ISDN Group in the Configuration Tree, and select **New ISDN Circuits**.
- Select the Start span, Start channel, End span, and End channel.

To save the changes, right-click on the Channel group, and select **Commit**.



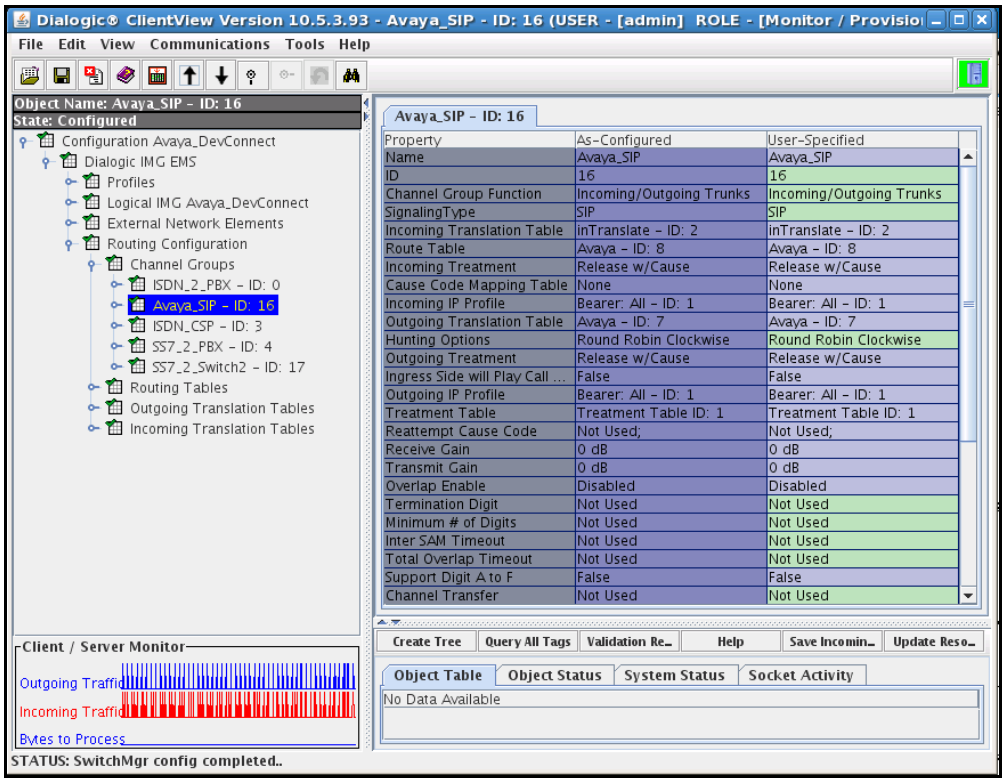
©2010 Avaya Inc. All Rights Reserved.

8.3.1.2 Channel Group (SIP)

Configure a Channel Group corresponding to each **External Gateway** as follows:

- Right-click **Channel Groups** in the Configuration Tree, and select **New Channel Group**.
- Enter a descriptive name for the Channel Group in the **Name** field in the Configuration Pane.
- Select **SIP** from the drop down list for the **Signaling Type** field.

To save the changes, right-click on the Channel group, and select **Commit**.



©2010 Avaya Inc. All Rights Reserved.

8.3.1.3 Channel Group with SIP Gateway

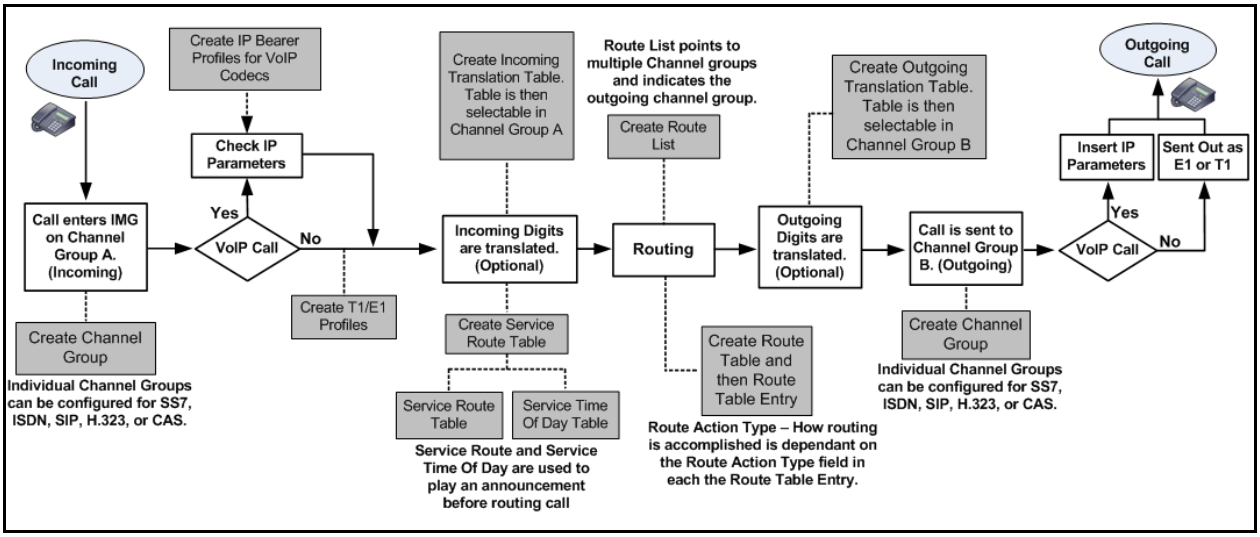
Assign a SIP Gateway to the Channel Group corresponding to each External Gateway previous:

- Right-click the Channel Group in the Configuration Tree, and select **New IP Network Element**.
- Select the External Gateway from the drop down list for the **IP Network Element** field.

To save the changes, right-click **IP Network Element**, and select **Commit**.

8.4. Routing

The following diagram shows the process the IMG goes through to route a call, and the ClientView GUI panes you use to configure various routing elements. Grey boxes represent ClientView configuration panes.



Create an object for Routing Tables as follows:

- Right-click **Routing Configuration** in the Configuration Tree, and select **New Routing Tables**.

To save the changes, right-click **Routing Tables**, and select **Commit**.

©2010 Avaya Inc. All Rights Reserved.

8.4.1. Route Entry

Add route entries to the Route Table as follows:

- Right-click the **Route Table** in the Configuration Tree and select **Add Route Entry**.
- Select the options for the different routing criteria and enter data to determine how the call is routed
- Select the Channel Group from the drop down list for the **Outgoing Channel Group** field.
  Note: This is displayed below under the **Route Action List** column.

Click **OK** in the **New Entry** dialog box.

8.5. Additional configuration

Other optional configuration objects can be configured as needed.

- Incoming Translation tables/Outgoing Translation tables
- Route Lists
- DNS
- Radius
- SNMP

©2010 Avaya Inc. All Rights Reserved.

9. Verification Steps

This section provides the verification steps that may be performed to verify that Avaya Aura™ enterprise network can establish and receive calls with Dialogic IMG 1010.

9.1. Verify Avaya Aura™ Communication Manager Access Element Trunk Status

On Communication Manager Access Element, ensure that all the signalling groups are in-service status by issuing the command status **signalling-group n** where **n** is the signalling group number.

```
status signaling-group 3
                         STATUS SIGNALING GROUP
        Group ID: 3                              Active NCA-TSC Count: 0
      Group Type: sip                             Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

```
status signaling-group 4
                         STATUS SIGNALING GROUP
        Group ID: 4                              Active NCA-TSC Count: 0
      Group Type: sip                             Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```


©2010 Avaya Inc. All Rights Reserved.

9.2. Verify Avaya Aura™ Communication Manager Feature Server Trunk Status

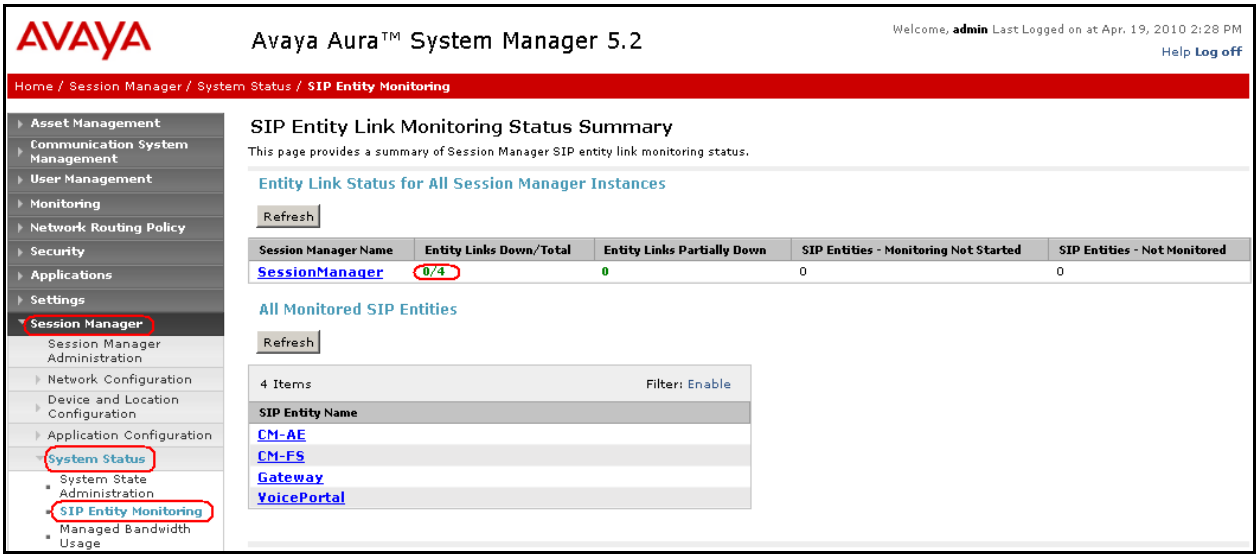
On Communication Manager Feature Server, ensure that all the signalling groups are in-service status by issuing the command status **signalling-group n** where **n** is the signalling group number.

```
status signaling-group 1
                         STATUS SIGNALING GROUP
        Group ID: 2                              Active NCA-TSC Count: 0
      Group Type: sip                             Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

```
status signaling-group 3
                         STATUS SIGNALING GROUP
        Group ID: 3                              Active NCA-TSC Count: 0
      Group Type: sip                             Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

9.3. SIP Monitoring on Avaya Aura™ Session Manager

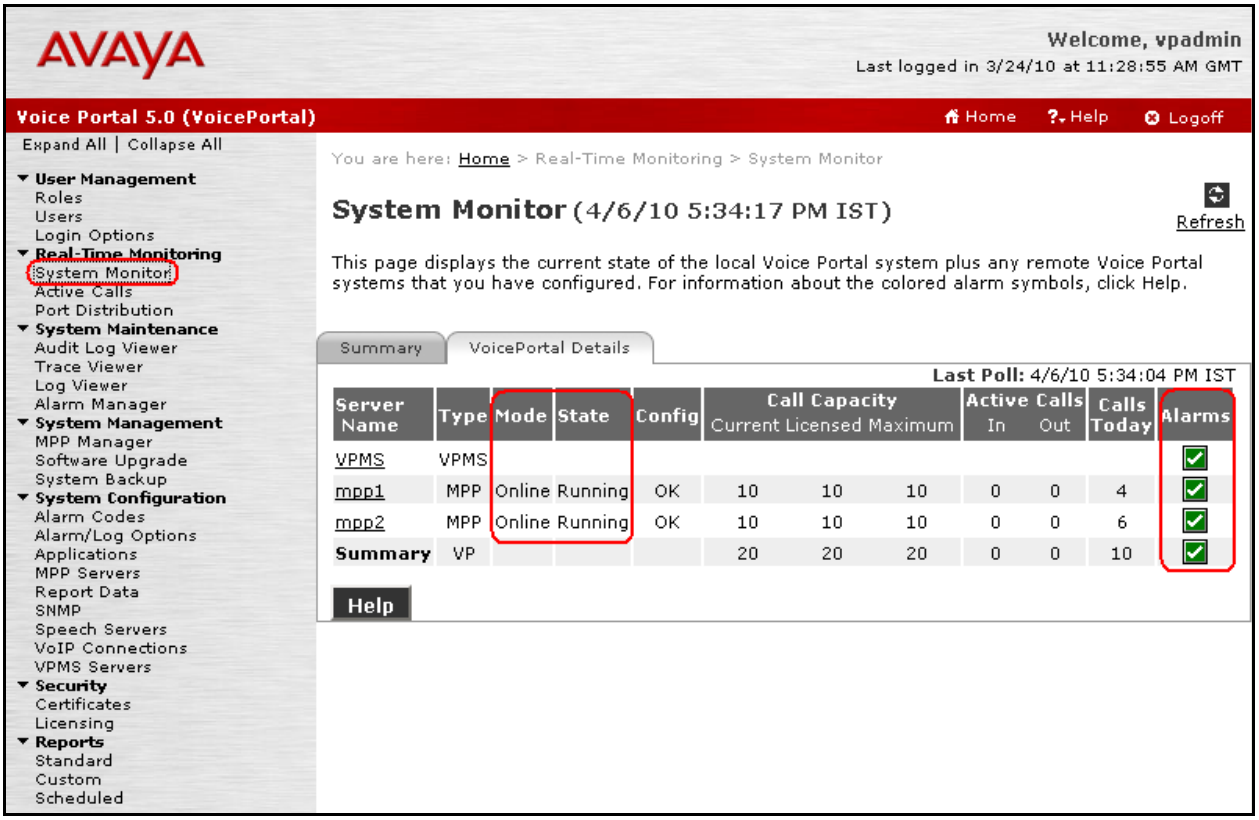
Expand the menu on the left and navigate **Session Manager→System Status →SIP Entity Monitoring**. Verify that none of the links to the defined SIP entities are down, indicating that they are all reachable for call routing.



©2010 Avaya Inc. All Rights Reserved.

9.4. Voice Portal Monitoring – System Monitor

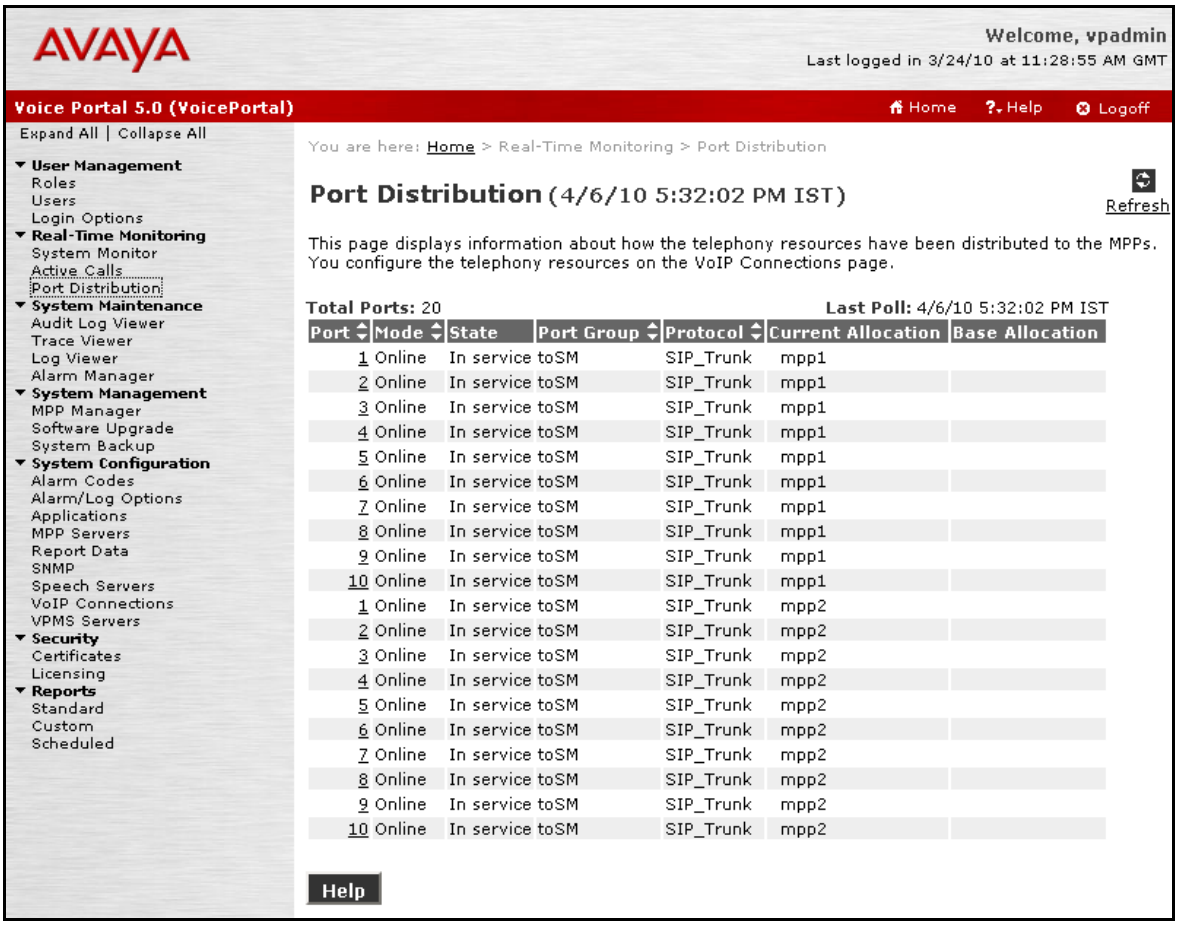
From the VPMS web interface, verify that for the MPP servers the **Mode** and **State** are **Online** and **Running,** and there are no **Alarms** associated with VP system in the **System Monitor** page shown **below**.



©2010 Avaya Inc. All Rights Reserved.

9.5. Voice Portal Monitoring – Port Distribution

From the VPMS web interface, verify that the ports on the MPP servers are in-service in the **Port Distribution** page shown below.



9.6. Functional Verification

From a phone on the PSTN, place a call to an application on Voice Portal. Verify that the call is established successfully and that the proper greeting is provided. Transfer the call to another user.

©2010 Avaya Inc. All Rights Reserved.

10. General Test Approach

This section describes the interoperability compliance testing used to verify calls between Avaya Voice Portal and the PSTN through the Dialogic IMG 1010, which served as a SIP-to-PSTN gateway. The interoperability compliance test included feature and serviceability. The feature testing focused on verifying the following:

- Placing calls from the Avaya SIP-based network to the PSTN
- Placing calls from the PSTN to Avaya Voice Portal
- Call transfers from Voice Portal to users on the PSTN and the Avaya SIP-based network
- Performing Blind, Supervised, and Bridged call transfers from Voice Portal to users on the PSTN and Avaya SIP-based network
- Sending UUI during call transfers from Voice Portal to the PSTN
- Receiving UUI from the PSTN to Voice Portal

The serviceability testing focused on verifying the ability of the IMG 1010 to recover from adverse conditions, such as power failures and disconnecting cables from the IP network.

10.1. Test Results and Remarks

All test cases were executed. During the compliance testing has been noted that Dialogic IMG 1010 when processing the SIP REFER messages needs to have the refer-to field set to <user>@<IP address>. Thanks to the flexibility of VXML in Voice Portal, it is possible to workaround this as shown in the sample programs in the Appendix.

11. Conclusion

As illustrated in these Application Notes, Dialogic® IMG 1010 Gateway can successfully offer access to PSTN to a Voice Portal solution in an enterprise telephony network built on Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager Access Element and Avaya Aura™ Communication Manager Feature Server.

©2010 Avaya Inc. All Rights Reserved.

12. Additional References

Avaya references, available at http://support.avaya.com

[1] "Avaya Aura™ Session Manager Overview", Document Number 03-603323, Issue 2, Release 5.2, November 2009

[2] "Installing and Upgrading Avaya Aura™ Session Manager", Document Number 03-603473, Issue 2, Release 5.2, November 2009

[3] "Administering Avaya Aura™ Session Manager", Document Number 03-603324, Issue 2, Release 5.2, November 2009

[4] "Avaya Aura™ Session Manager Case Studies", Document Number 03-603478, Issue 2, Release 5.2, November 2009

[5] "Maintaining and Troubleshooting Avaya Aura™ Session Manager, Document Number 03-603325, Issue 2, Release 5.2, November 2009

[6] "Installing and Configuring Avaya Aura™ System Platform", Release 1.1, November 2009

[7] "Installing and Upgrading Avaya Aura™ System Manager", Release 5.2, January 2010

[8] "Avaya Aura™ Communication Manager Overview", Document Number 03-300468, Issue 6, Release 5.2, May 2009

[9] "Administering Avaya Aura™ Communication Manager", Document Number 03-300509, Issue 5.0, Release 5.2, May 2009

[10] "Avaya Aura™ Communication Manager Feature Description and Implementation", Document Number 555-245-205, Issue 7.0, Release 5.2, May 2009

[11] "Administering Network Connectivity on Avaya Aura™ Communication Manager", Document Number 555-233-504, Issue 14, May 2009

[12] "SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers", Document Number 555-245-206, Issue 9, May 2009

[13] "Administering Avaya Aura™ Communication Manager as a Feature Server", Document Number 03-603479, Issue 1.2, Release 5.2, January 2010

[14] "Configuring 9600-Series SIP Phones with Avaya Aura™ Session Manager Release 5.2 – Issue 1.0", Application Note

[15] "Administering Voice Portal", Release 5.0, March 2009

[16] "Implementing Voice Portal on multiple servers", Release 5.0, March 2009

[17] "Configuring Avaya Aura™ Session Manager 5.2 with Avaya Aura™ Communication Manager Access Element, Avaya Voice Portal and Avaya Aura™ Communication Manager Feature Server – Issue 1.0". Application Note

Dialogic® Integrated Media Gateway (IMG) references are available on http://www.dialogic.com

[18] "IMG 1010 - Quick Start Guide" Doc ID 07-728-05, March 2010

[19] "IMG 1010/1004 Integrated Media Gateway Upgrading System Software ", November 2009 Application Note

[20] "Dialogic® IMG 1010 Integrated Media Gateway WebHelp" Release: 10.5.3 ER1

©2010 Avaya Inc. All Rights Reserved.

13. APPENDIX

In this section are presented the relevant configuration files for the devices used in the DevConnect compliance testing.

13.1. Web application server intro.vxml file

Here it is presented the **intro.vxml** file used in these Application Notes.

```
<?xml version="1.0" ?>
<vxml version="2.1" xmlns="http://www.w3.org/2001/vxml" xml:lang="en-US" >

<form id="form0">

        <field name="test_type">

              <prompt bargein="true" cond="session.connection.ccxml.values.test_page ==
'true'">
                    <audio src="prompts/introccxml.wav"/>
              </prompt>

              <prompt bargein="true" cond="session.connection.ccxml.values.test_page ==
undefined">
                    <audio src="prompts/introvxml.wav"/>
              </prompt>

                 <grammar src="builtin:dtmf/digits" />

                 <filled>
                     <if cond="test_type == 1">
                           <goto next="asrtest.vxml"/>
                     <elseif cond="test_type == 2"/>
                           <goto next="ttstest.vxml"/>
                     <elseif cond="test_type == 3"/>
                           <goto next="testbridgetransfer.vxml"/>
                     <elseif cond="test_type == 4"/>
                           <goto next="testblindtransfer.vxml"/>
                     <elseif cond="test_type == 5"/>
                           <goto next="testconsulttransfer.vxml"/>
                     <elseif cond="test_type == 6"/>
                           <goto next="playprompts.vxml"/>
                     <elseif cond="session.connection.ccxml.values.test_page ==
'true'"/>
                           <if cond="test_type > 9">
                                 <prompt bargein="false">
                                       <audio src="prompts/commonSorry.wav"/>
                                 </prompt>
                                 <clear namelist="test_type"/>
                           <elseif cond="test_type == 0"/>
                                 <prompt bargein="false">
                                       <audio src="prompts/Exit.wav"/>
                                 </prompt>
                           <else/>
                                 <exit namelist="test_type"/>
                           </if>
                     <else/>
                           <if cond="test_type == 7">
                                 <log expr="'Getting Ready To Exit'"/>
                                 <prompt bargein="false">
```


©2010 Avaya Inc. All Rights Reserved.

```
                                        <audio src="prompts/Exit.wav"/>
                                 </prompt>
                                 <exit/>
                           <else/>
                                 <prompt bargein="false">
                                        <audio src="prompts/commonSorry.wav"/>
                                 </prompt>
                                 <clear namelist="test_type"/>
                           </if>
                     </if>

                </filled>

              <noinput>
                     <prompt bargein="false">
                                 <audio src="prompts/commonSorry.wav"/>
                           </prompt>
                     <reprompt/>
              </noinput>

        </field>

</form>
</vxml>
```


©2010 Avaya Inc. All Rights Reserved.

13.2. Web application server testblindtransfer.vxml file

Here it is presented the **testblindtransfer.vxml** file used in these Application Notes

```
<?xml version="1.0" ?>
<vxml version="2.1" xmlns="http://www.w3.org/2001/vxml" xml:lang="en-US" >
        <var name="var1" expr="'sip:'"/>
        <var name="var2" expr="'@dialogic.com'"/>
        <var name="var3" expr="'@avaya.com'"/>
        <var name="domain" expr=""/>
        <form id="get_number">
                <field name="phone_number">
                        <prompt bargein="true">
                                <audio src="prompts/TransferGetNumber.wav"/>
                        </prompt>
                        <grammar src="builtin:dtmf/digits?minlength=1;maxlength=10" />
                        <noinput>
                                <prompt bargein="false">
                                        <audio
src="prompts/TransferNoNumberSorry.wav"/>
                                </prompt>
                                <reprompt/>
                        </noinput>
                        <filled>
                                <if cond="phone_number ==  '03347'">
                                        <assign name="var3" expr="'@208.209.43.59'"/>
                                </if>
                        </filled>
                </field>
                <transfer name="blindtransfer" destexpr="var1 + phone_number +var3 "
type="blind" aai="01234567890abcdef">
                        <prompt bargein="false">
                                <audio src="prompts/blindPerforming.wav"/>
                        </prompt>
                        <filled>
                                <if cond="blindtransfer == 'near_end_disconnect'">
                                        <audio src="prompts/nearEndDisc.wav"/>
                                        <log> near_end_disconnect </log>
                                <elseif cond="blindtransfer == 'unknown'"/>
                                        <audio src="prompts/failedUnknown.wav"/>
                                        <log> unknown </log>
                                </if>
                                <goto next="intro.vxml"/>
                        </filled>
                </transfer>
                <catch event="connection.disconnect.transfer">
                        <log> connection.disconnect.transfer </log>
                        <exit />
                </catch>
                <catch event="error.connection.noauthorization">
                        <log> error.connection.noauthorization </log>
                        <goto next="intro.vxml"/>
                </catch>
                <catch event="error.connection.baddestination">
                        <log> error.connection.baddestination </log>
                        <goto next="intro.vxml"/>
                </catch>
                <catch event="error.unsupported.uri">
                        <log> error.unsupported.uri </log>
```


©2010 Avaya Inc. All Rights Reserved.

```
                        <goto next="intro.vxml"/>
                </catch>
                <catch event="error.unsupported.transfer.blind">
                        <log> error.unsupported.transfer.blind </log>
                        <goto next="intro.vxml"/>
                </catch>
        </form>
</vxml>
```


©2010 Avaya Inc. All Rights Reserved.

13.3. Web application server testbridgetransfer.vxml file

Here it is presented the **testbridgetransfer.vxml** file used in these Application Notes

```
<?xml version="1.0" ?>
<vxml version="2.1" xmlns="http://www.w3.org/2001/vxml" xml:lang="en-US" >

        <var name="var1" expr="'sip:'"/>
        <var name="var2" expr="'@dialogic.com'"/>
        <var name="var3" expr="'@avaya.com'"/>
        <form id="get_number">

                <field name="phone_number">

                        <prompt bargein="true">
                                <audio src="prompts/TransferGetNumber.wav"/>
                        </prompt>

                        <grammar src="builtin:dtmf/digits?minlength=1;maxlength=10" />

                        <noinput>
                                <prompt bargein="false">
                                        <audio
src="prompts/TransferNoNumberSorry.wav"/>
                                </prompt>
                                <reprompt/>
                        </noinput>
                        <filled>
                                <if cond="phone_number ==  '03347'">
                                        <assign name="var3" expr="'@avaya.com'"/>
                                </if>
                        </filled>

                </field>

                <transfer name="bridgetransfer" destexpr="var1 + phone_number +var3"
type="bridge" aai="01234567890abcded">

                        <prompt bargein="true">
                                <audio src="prompts/bridgePerforming.wav"/>
                        </prompt>

                        <grammar src="builtin:dtmf/digits" />

                        <filled>
                                <if cond="bridgetransfer == 'busy'">
                                        <audio src="prompts/lineBusy.wav"/>
                                        <log> busy </log>
                                <elseif cond="bridgetransfer == 'noanswer'"/>
                                        <audio src="prompts/noAnswer.wav"/>
                                        <log> noanswer </log>
                                <elseif cond="bridgetransfer == 'network_busy'"/>
                                        <audio src="prompts/nwBusy.wav"/>
                                        <log> network_busy </log>
                                <elseif cond="bridgetransfer ==
'near_end_disconnect'"/>
                                        <audio src="prompts/nearEndDisc.wav"/>
                                        <log> near_end_disconnect </log>
                                <elseif cond="bridgetransfer == 'unknown'"/>
```


©2010 Avaya Inc. All Rights Reserved.

```
                                        <audio src="prompts/failedUnknown.wav"/>
                                        <log> unknown </log>
                                <elseif cond="bridgetransfer ==
'maxtime_disconnect'"/>
                                        <audio src="prompts/maxTimeDisc.wav"/>
                                        <log> maxtime_disconnect </log>
                                <elseif cond="bridgetransfer ==
'network_disconnect'"/>
                                        <audio src="prompts/nwDisc.wav"/>
                                        <log> network_disconnect </log>
                                <elseif cond="bridgetransfer ==
'far_end_disconnect'"/>
                                        <audio src="prompts/farEndDisconnect.wav"/>
                                        <log> far_end_disconnect </log>
                                </if>

                                <prompt bargein="false">
                                        <audio src="prompts/bridgeThanks.wav"/>
                                </prompt>

                                <goto next="intro.vxml"/>
                        </filled>

                </transfer>

                <catch event="connection.disconnect.hangup">
                        <log> connection.disconnect.hangup </log>
                        <exit />
                </catch>
                <catch event="error.connection.noauthorization">
                        <log> error.connection.noauthorization </log>
                        <goto next="intro.vxml"/>
                </catch>
                <catch event="error.connection.baddestination">
                        <log> error.connection.baddestination </log>
                        <goto next="intro.vxml"/>
                </catch>
                <catch event="error.unsupported.transfer.bridge">
                        <log> error.unsupported.transfer.blind </log>
                        <goto next="intro.vxml"/>
                </catch>
                <catch event="error.unsupported.uri">
                        <log> error.unsupported.uri </log>
                       <goto next="intro.vxml"/>
                </catch>
                <catch event="error.connection.noroute">
                        <log> error.connection.noroute </log>
                        <goto next="intro.vxml"/>
                </catch>
                <catch event="error.connection.noresource">
                        <log> error.connection.noresource </log>
                        <goto next="intro.vxml"/>
                </catch>
        </form>
</vxml>
```


©2010 Avaya Inc. All Rights Reserved.

13.4. Web application server testconsulttransfer.vxml file

Here it is presented the **testconsulttransfer.vxml** file used in these Application Notes.

```
<?xml version="1.0" ?>
<vxml version="2.1" xmlns="http://www.w3.org/2001/vxml" xml:lang="en-US" >
        <var name="var1" expr="'sip:'"/>
        <var name="var2" expr="'@dialogic.com'"/>
        <var name="var3" expr="'@avaya.com'"/>

        <form id="get_number">
                <field name="phone_number">
                <prompt bargein="true">
                        <audio src="prompts/TransferGetNumber.wav"/>
                </prompt>

        <grammar src="builtin:dtmf/digits?minlength=1;maxlength=12" />

        <noinput>
                <prompt bargein="false">
                        <audio src="prompts/TransferNoNumberSorry.wav"/>
                </prompt>
                <reprompt/>
        </noinput>
                 <filled>
                  <if cond="phone_number ==  '3347'">
                          <assign name="var3" expr="'@208.209.43.59'"/>
                    </if>
                </filled>
        </field>
                <transfer name="consultationtransfer" destexpr="var1 + phone_number +
var3" type="consultation" aai="0123456789abcef">
                        <prompt bargein="false">
                                <audio src="prompts/consultPerforming.wav"/>
                        </prompt>

                        <filled>
                                <if cond="consultationtransfer == 'busy'">
                                        <audio src="prompts/lineBusy.wav"/>
                                        <log> busy </log>
                                        <goto next="intro.vxml"/>
                                <elseif cond="consultationtransfer == 'noanswer'"/>
                                        <audio src="prompts/noAnswer.wav"/>
                                        <log> noanswer </log>
                                        <goto next="intro.vxml"/>
                                <elseif cond="consultationtransfer ==
'near_end_disconnect'"/>
                                        <audio src="prompts/nearEndDisc.wav"/>
                                        <log> near_end_disconnect </log>
                                        <goto next="intro.vxml"/>
                                <elseif cond="consultationtransfer ==
'network_busy'"/>
                                        <audio src="prompts/nwBusy.wav"/>
                                        <log> network_busy </log>
                                        <goto next="intro.vxml"/>
                                <elseif cond="consultationtransfer == 'unknown'"/>
                                        <audio src="prompts/failedUnknown.wav"/>
                                        <log> unknown </log>
                                        <goto next="intro.vxml"/>
```


©2010 Avaya Inc. All Rights Reserved.

```
                                </if>
                        </filled>

                </transfer>

                <catch event="connection.disconnect.hangup">
                        <log> connection.disconnect.hangup </log>
                        <goto next="intro.vxml"/>
                </catch>

                <catch event="error.connection.noauthorization">
                        <log> connection.disconnect.transfer </log>
                </catch>

                <catch event="error.connection.noauthorization">
                        <log> error.connection.noauthorization </log>
                        <goto next="intro.vxml"/>
                </catch>

                <catch event="error.connection.baddestination">
                        <log> error.connection.baddestination </log>
                        <goto next="intro.vxml"/>
                </catch>

                <catch event="error.connection.noroute">
                        <log> error.connection.noroute </log>
                        <goto next="intro.vxml"/>
                </catch>

                <catch event="error.connection.noresource">
                        <log> error.connection.noresource </log>
                        <goto next="intro.vxml"/>
                </catch>

                <catch event="error.unsupported.uri">
                        <log> error.unsupported.uri </log>
                        <goto next="intro.vxml"/>
                </catch>

                <catch event="error.unsupported.transfer.consultation">
                  <log> error.unsupported.transfer.consultation </log>
                        <goto next="intro.vxml"/>
                </catch>

        </form>
</vxml>
```


©2010 Avaya Inc. All Rights Reserved.

©2010 Avaya Inc. All Rights Reserved.
Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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.