Avaya Solution & Interoperability Test Lab

Application Notes for Configuring 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 procedure to configure an Enterprise network built on Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager Access Element and Avaya Aura™ Communication Manager Feature Server to interoperate with Dialogic IMG1010 Gateway using SIP trunking.

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

1. Introduction

These Application Notes present a sample configuration for an Enterprise network that Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager Access Element and Avaya Aura™ Communication Manager Feature Server as SIP infrastructure to accessing the PSTN with Dialogic® IMG 1010 Gateway using SIP. 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.

1.1. Interoperability Compliance Testing

The primary focus of testing is to verify SIP trunking interoperability between an Avaya Aura™ SIP-based network 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.

Basic Interoperability:

- PSTN calls delivered via the IMG 1010 to an Enterprise endpoint
- PSTN calls sent via the IMG1010 from an Enterprise endpoint
- Calling with various Avaya telephone models including IP/SIP models as well as traditional analog and digital TDM phones
- Verify ITU-T codecs: G.711A G.711MU G.729A G.729B support
- Various PTSN dialing plans including national and international calling, toll-free, operator, directory assistance and direct inward dialed calling
- SIP transport using UDP

Advanced Interoperability:

- Codec negotiation
- Telephony supplementary features, such as Hold, Call transfer, Conference Calling and Call Forwarding
- DTMF Tone Support
- T.38 Fax support
- Voicemail Coverage and Retrieval
- Direct IP-to-IP Media (also known as “Shuffling”) over SIP Trunk. Direct IP-to-IP media allows compatible phones to reconfigure the RTP path after call establishment directly between the Avaya phones and the IMG 1010 Gateway and release media processing resources on the Avaya Media Gateway
- EC500 for Avaya Aura™ Communication Manager

1.2. Support

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

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

2. Reference Configuration

As shown in **Figure 1**, the Avaya 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 the 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.

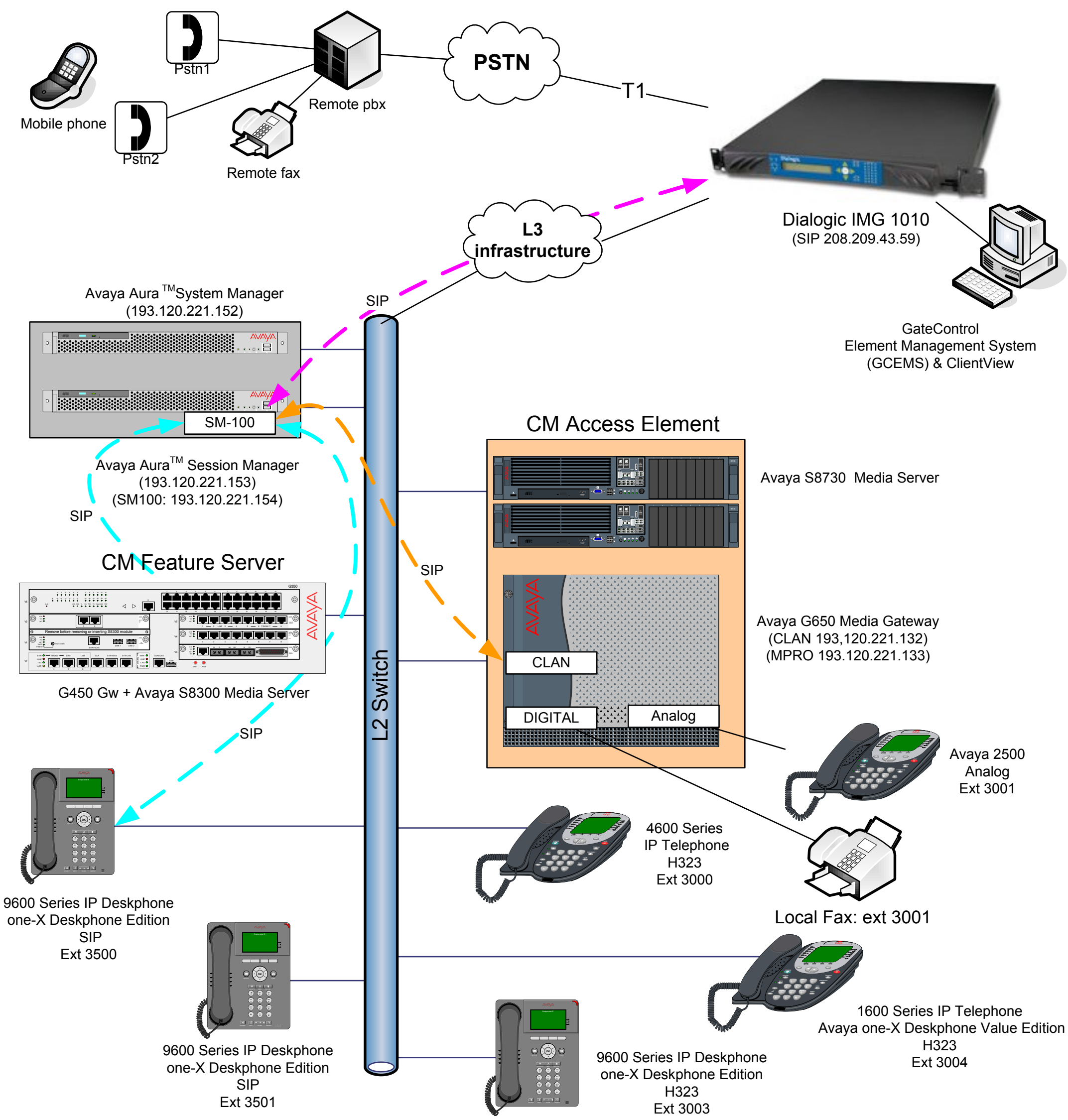


Figure 1 – Sample configuration for Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager with Dialogic IMG 1010 using Sip Trunking

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. 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 10**.

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 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
Fax Machine Canon FAX JX500	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

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 the 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**. 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
     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**.

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 is required to have 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.

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. A list of supported interoperability compliance tests is presented in **Section 1.1**.The ITU G.711A-law is described here. Configure the IP Codec Set as 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:
```

To configure fax support, navigate to **Page 2** and change **FAX** to **t.38-standard.** Use default values for all other fields. Submit these changes.

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

                              Allow Direct-IP Multimedia? n



                    Mode                 Redundancy
    FAX             t.38-standard          0
    Modem           off                    0
    TDD/TTY         US                     3
    Clear-channel   n                      0
```

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
```

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
```

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
```

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
```

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
```

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
```

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. Use the **change aar analysis 0** command and add an entry to specify how to route calls to **35xx**. 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**

```
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
```

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
```

5. Configure Avaya Aura^{TM} 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^{TM} 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^{TM} 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
```

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
```

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
```

Use the **change ip-codec-set n** command where **n** is codec set used in the configuration. A list of supported by the interoperability compliance testing is presented in **Section 1.1** 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:
```

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
```

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
```

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
```

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. **103**)
- **Service Type:** **tie**
- **Signaling Group:** The number of the signaling group associated (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 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
```

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
```

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 . 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
     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**).

```
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
```

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.

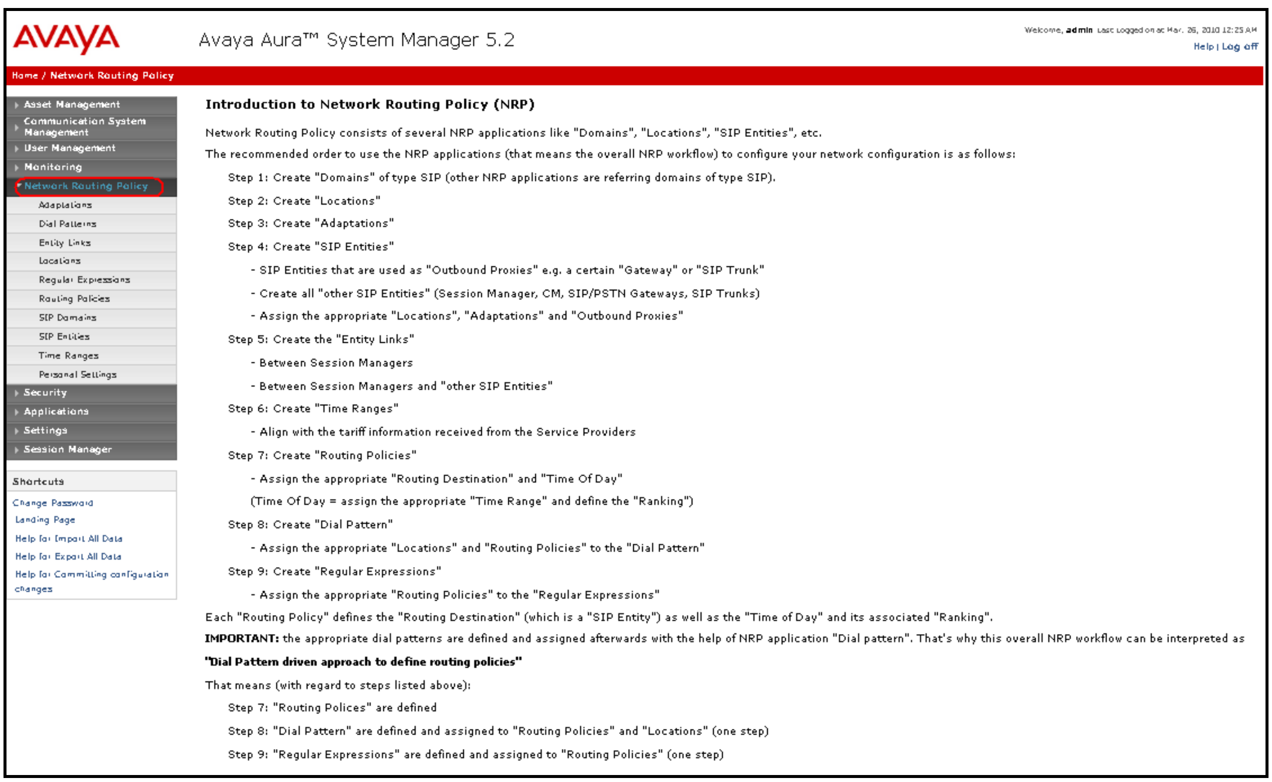
```
save translation
                               SAVE TRANSLATION
           Command Completion Status                                 Error Code
           Success                                                     0
```

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 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

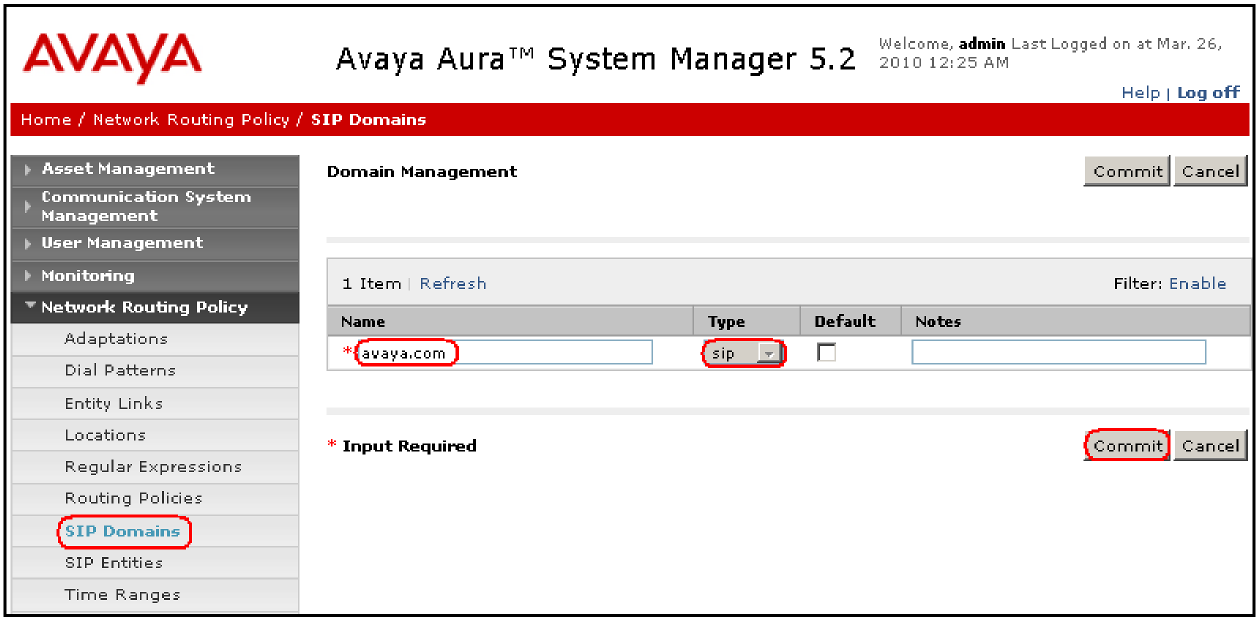
Configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL **http://<ip-address>/SMGR**, where **<ip-address>** is the IP address of System Manager. Log in with the appropriate credentials and accept the Copyright Notice. The menu shown below is displayed. Expand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



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)



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:

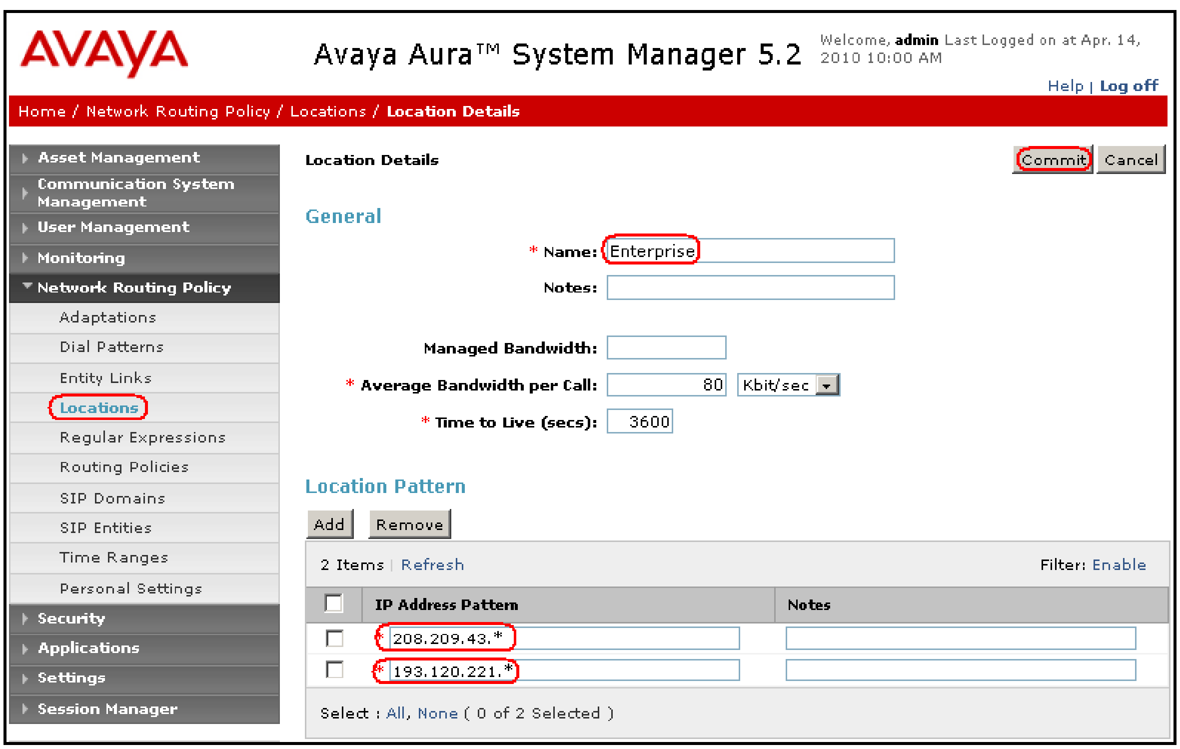
Under **General**:

- **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.



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 7.2.3**. 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

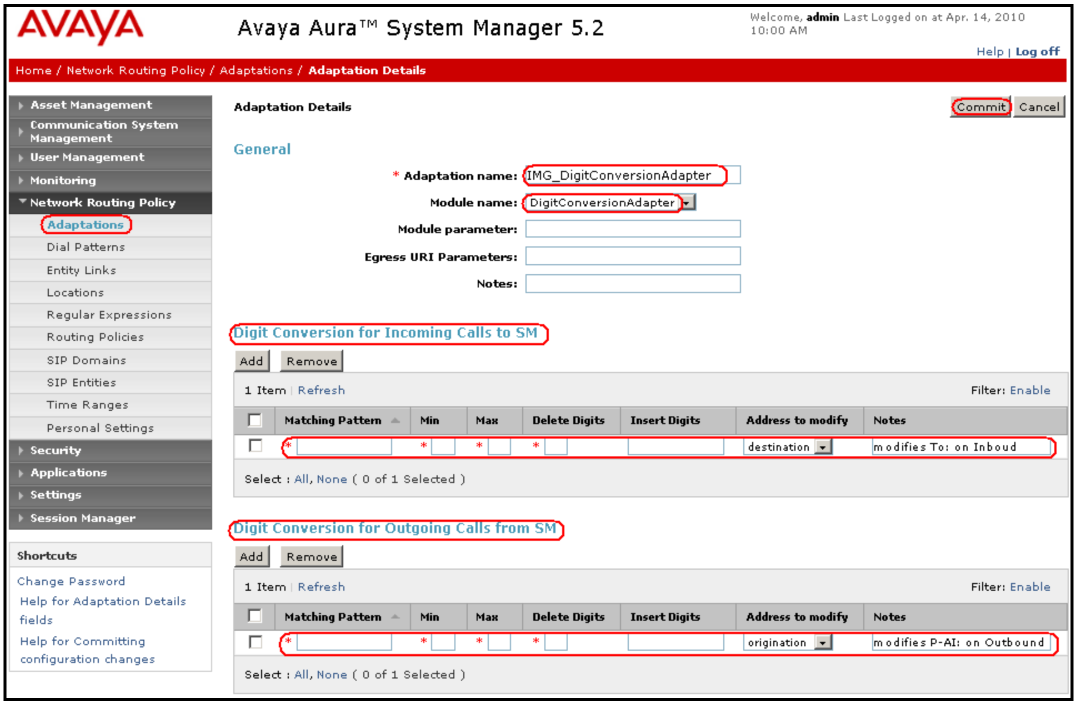
Under **Digit Conversion for Incoming Calls to SM**:

- **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.


AVAYA
Avaya Aura™ System Manager 5.2
Welcome, admin Last Logged on at Apr. 14, 2010 10:00 AM
Help | Log off
Home / Network Routing Policy / Adaptations / Adaptation Details
Asset Management
Communication System Management
User Management
Monitoring
Network Routing Policy
Adaptations
Dial Patterns
Entity Links
Locations
Regular Expressions
Routing Policies
SIP Domains
SIP Entities
Time Ranges
Personal Settings
Security
Applications
Settings
Session Manager
Shortcuts
Change Password
Help for Adaptation Details fields
Help for Committing configuration changes
Adaptation Details
Commit
Cancel
General
* Adaptation name: IMG_DigitConversionAdapter
Module name: DigitConversionAdapter
Module parameter:
Egress URI Parameters:
Notes:
Digit Conversion for Incoming Calls to SM
Add
Remove
1 Item | Refresh
Filter: Enable
Matching Pattern
Min
Max
Delete Digits
Insert Digits
Address to modify
Notes
destination
modifies To: on Inboud
Select : All, None (0 of 1 Selected)
Digit Conversion for Outgoing Calls from SM
Add
Remove
1 Item | Refresh
Filter: Enable
Matching Pattern
Min
Max
Delete Digits
Insert Digits
Address to modify
Notes
origination
modifies P-AI: on Outbound
Select : All, None (0 of 1 Selected)

6.4. 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 and the Dialogic IMG 1010 Gateway on the Service Provider.

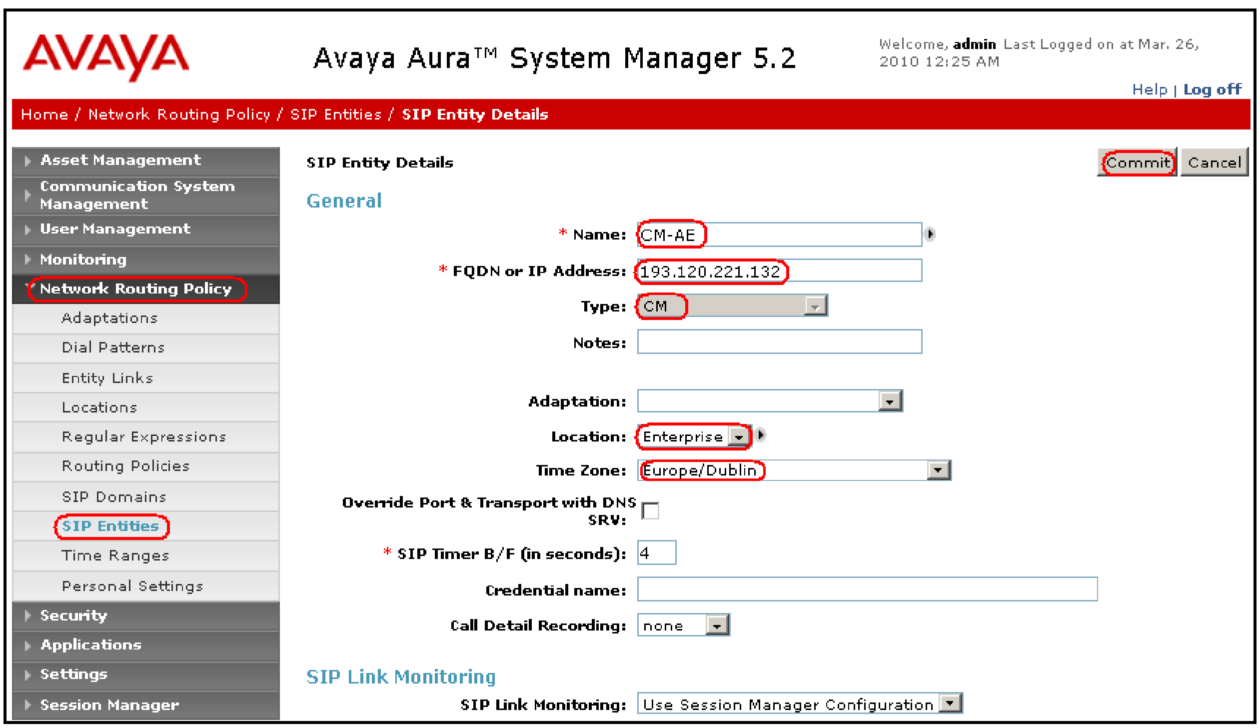
6.4.1. Adding Avaya Aura^{TM} Communication Manager Access Element SIP Entity

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.



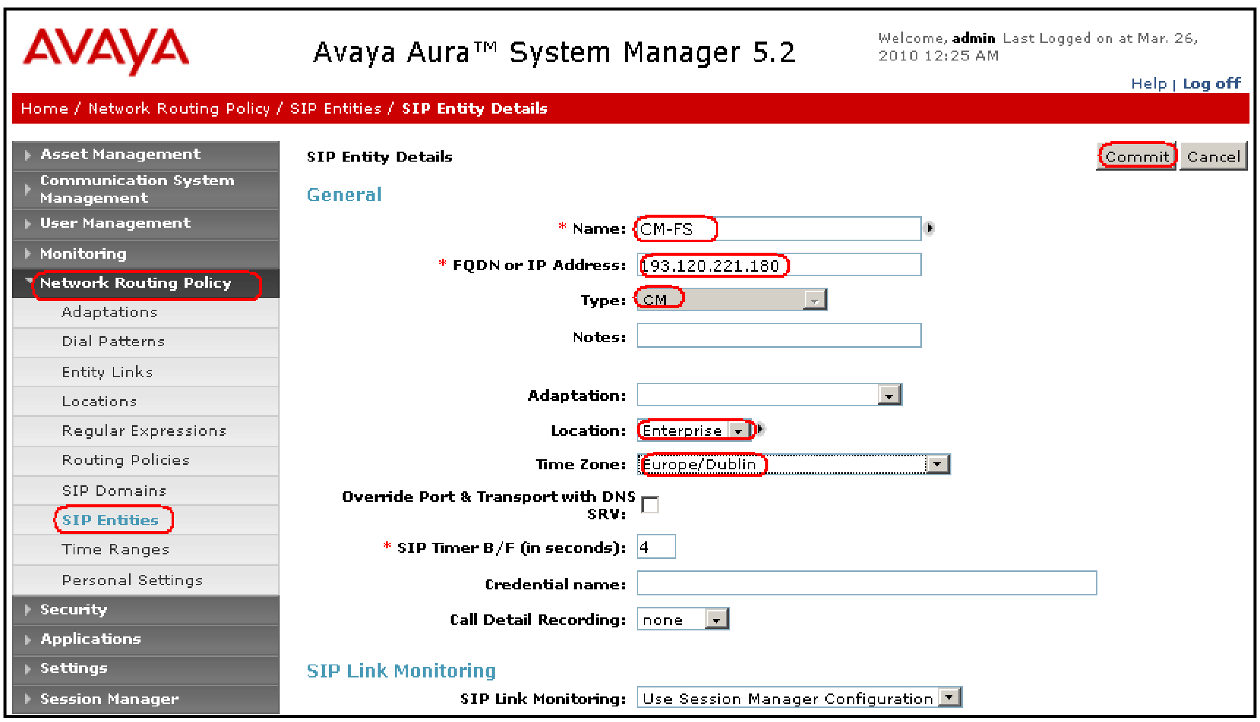
6.4.2. Adding Avaya Aura™ Communication Manager Feature Server SIP Entity

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.



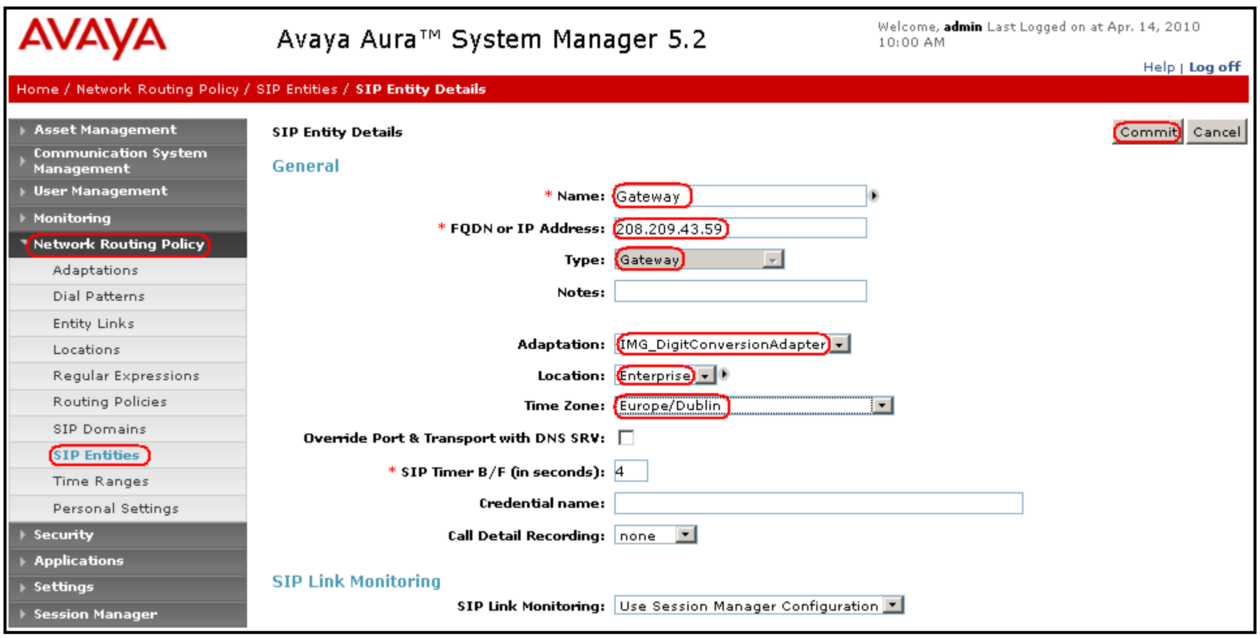
6.4.3. Adding Dialogic IMG 1010 Gateway SIP Entity

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 screen below shows the configuration of the SIP Entity related to Dialogic IMG 1010.



6.4.4. 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.

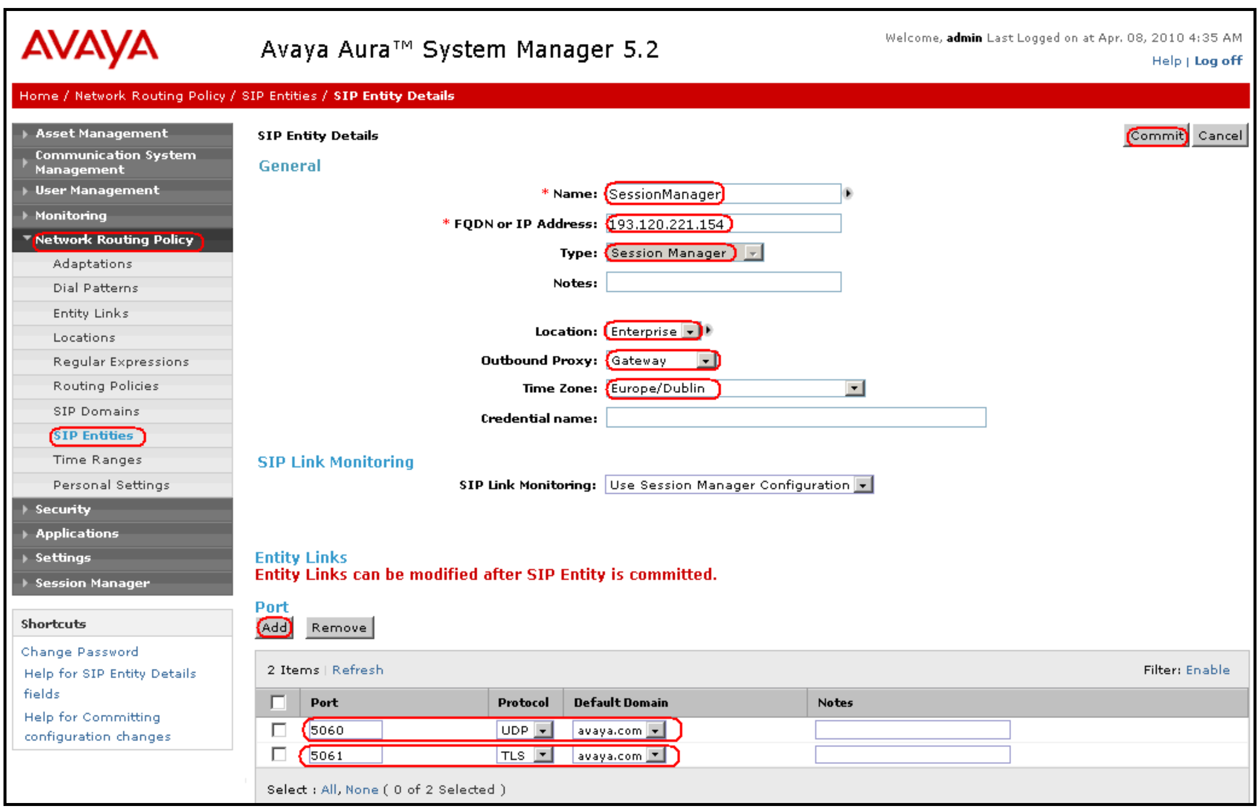
Under **General**:

- **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.

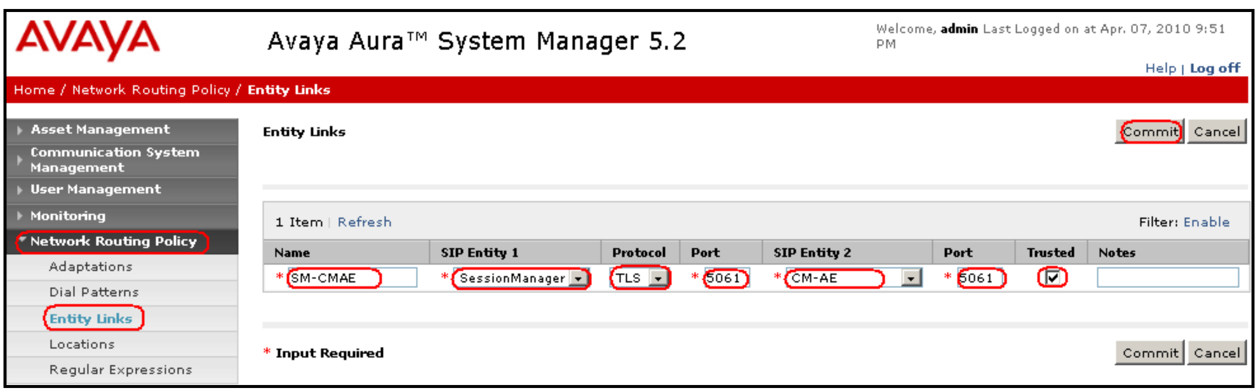


6.5. Add Entity Links

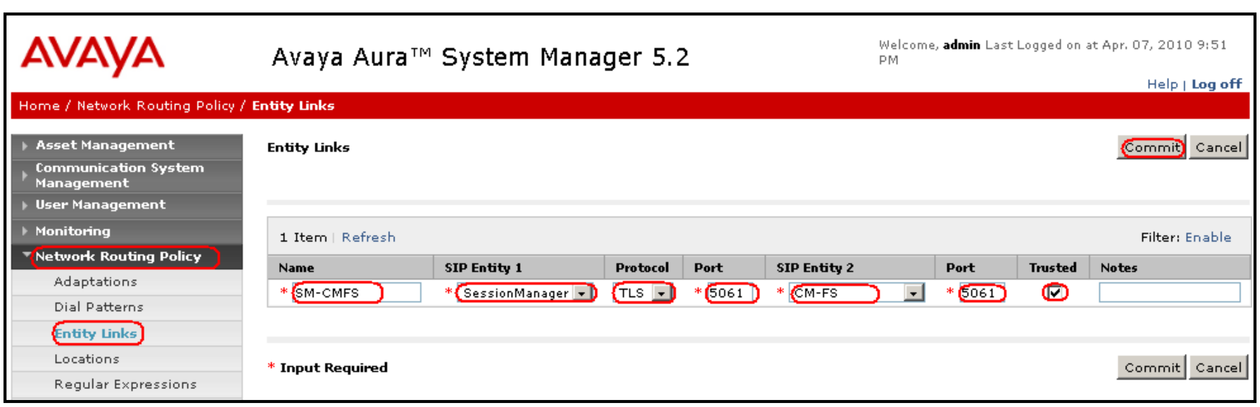
A SIP trunk between Session Manager and a telephony system is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button on the right. Fill in the following fields in the new row that is displayed:

- **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** while **UDP** for **Dialogic IMG 1010.**

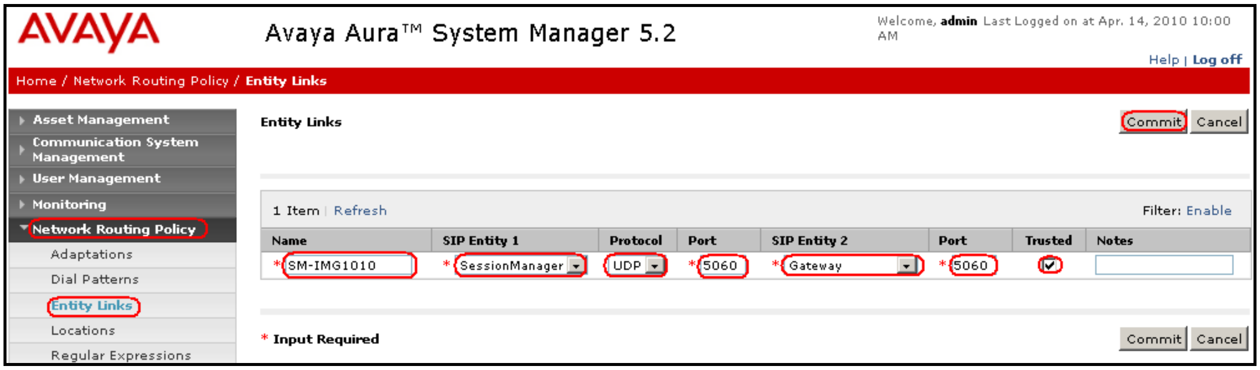
Click **Commit** to save each Entity Link definition**.** The following screen illustrates adding the Entity Link for Communication Manager Access Element.



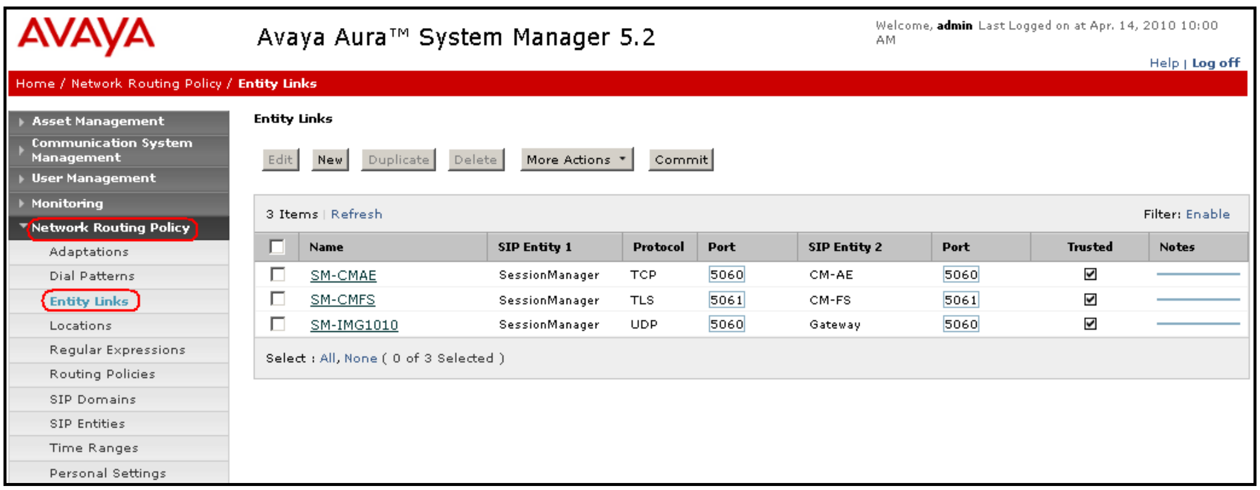
The screen below illustrates adding the Entity Link for Communication Manager Feature Server.



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



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



6.6. Add Routing Policies

Routing policies describe the condition under which calls will be routed to the SIP Entities specified in **Section 5.3**. Two routing policies must be added: one for Communication Manager Access Element 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**

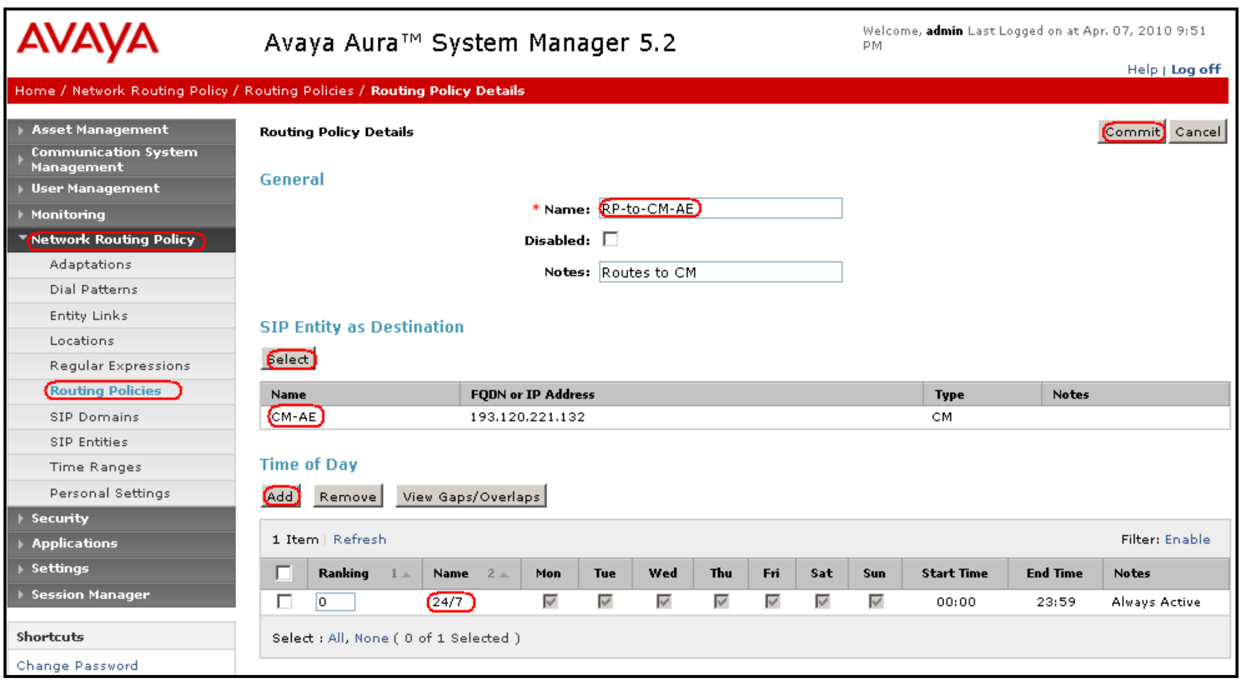
Under **SIP Entity as Destination:**

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

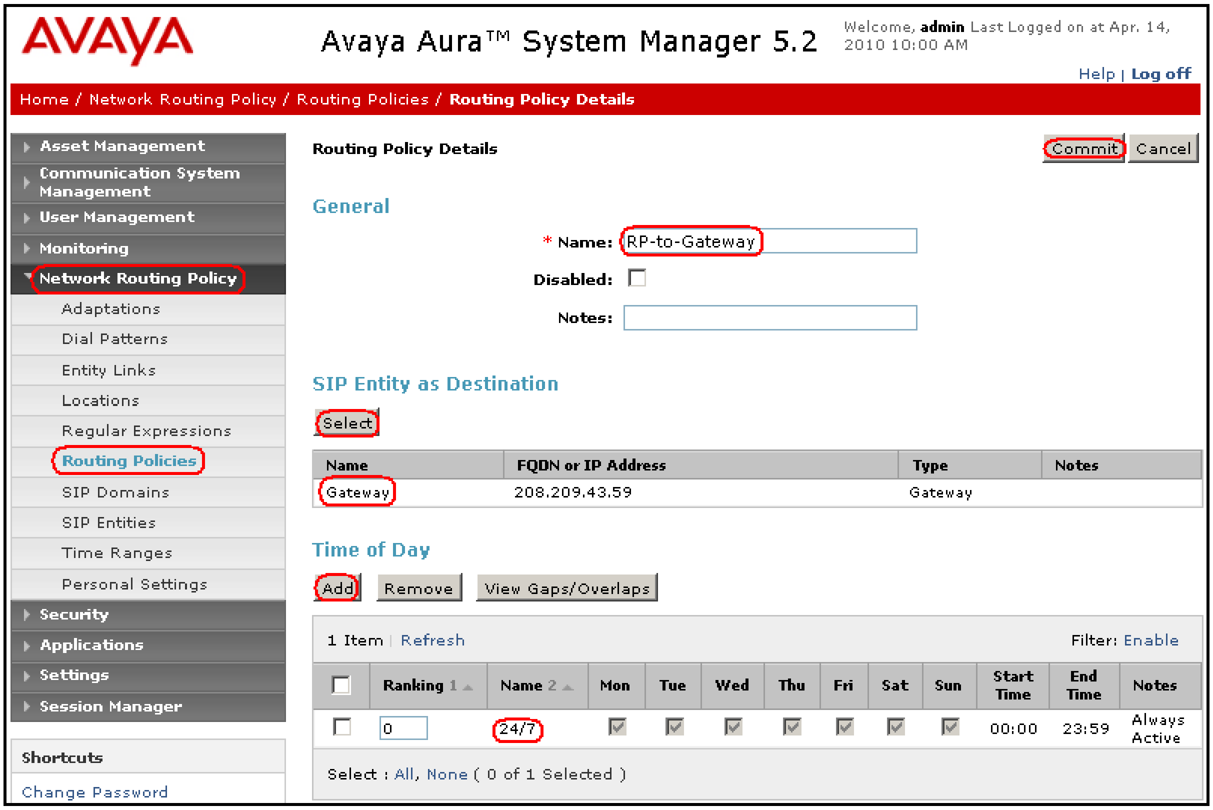
Under **Time of Day:**

- 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.



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



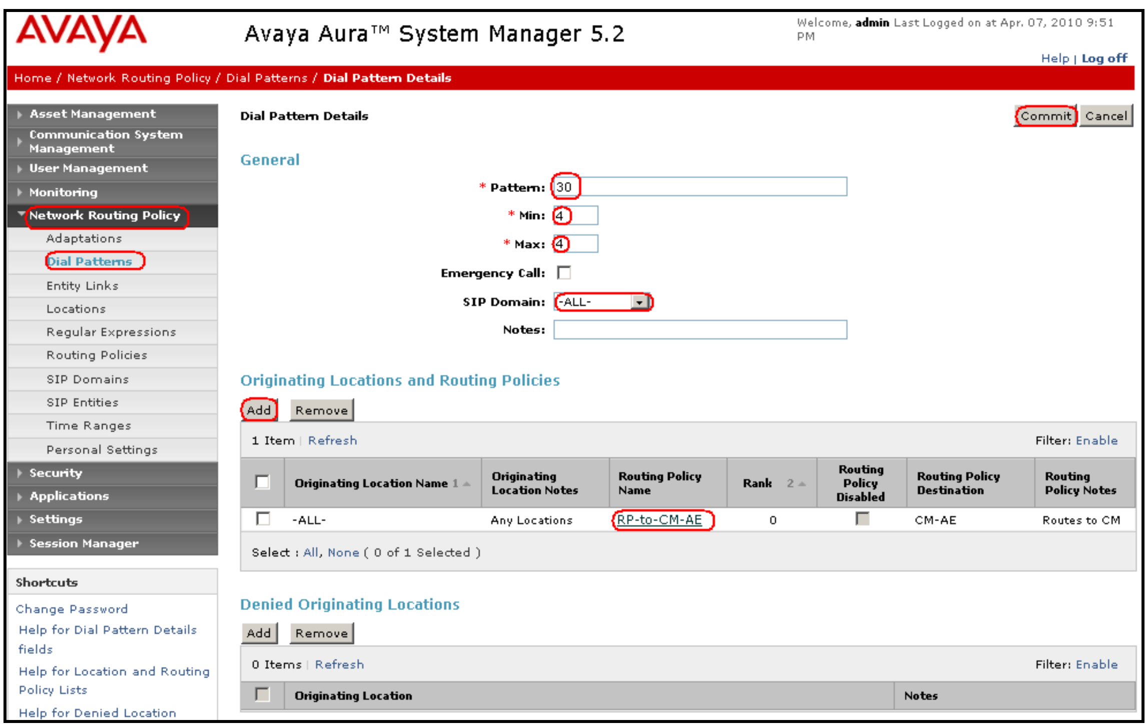
6.7. Add Dial Patterns

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, 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.

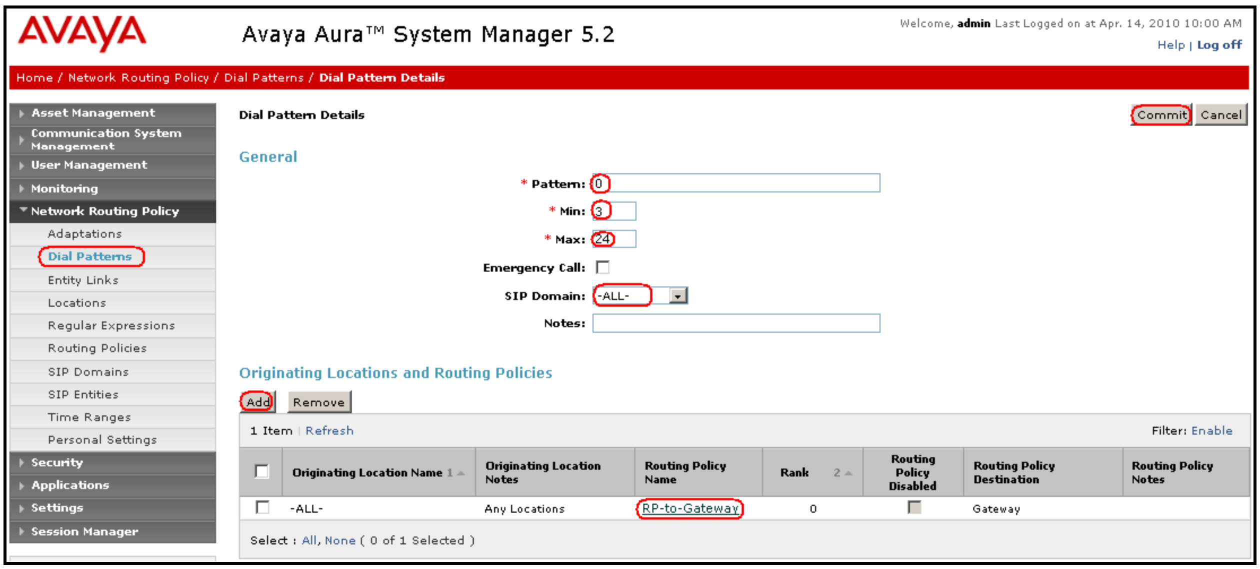


Repeat the process adding one or more dial patterns for the PSTN numbers that should be reached from the Dialogic IMG 1010. Fill in the following, as shown in the screen below, which corresponds to the dial pattern for routing calls to Dialogic IMG 1010:

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 the dial pattern definition for PSTN reachable with Dialogic IMG 1010.



6.8. 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:

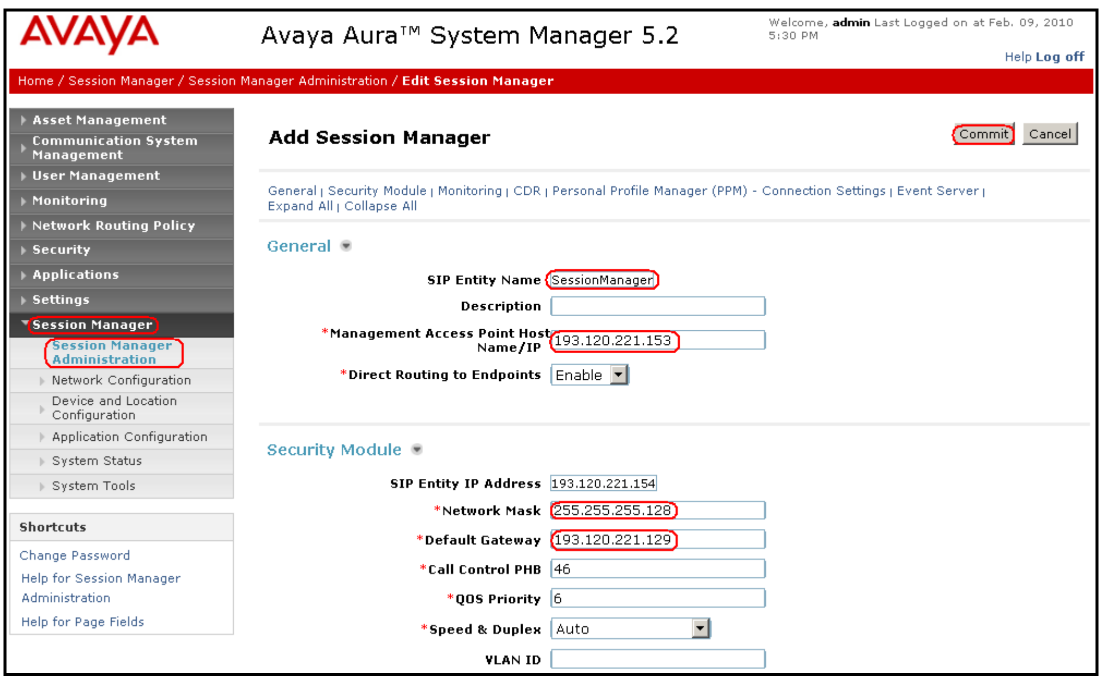
Under **General**:

- **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 **Commit** to add this configuration to Session Manager.



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

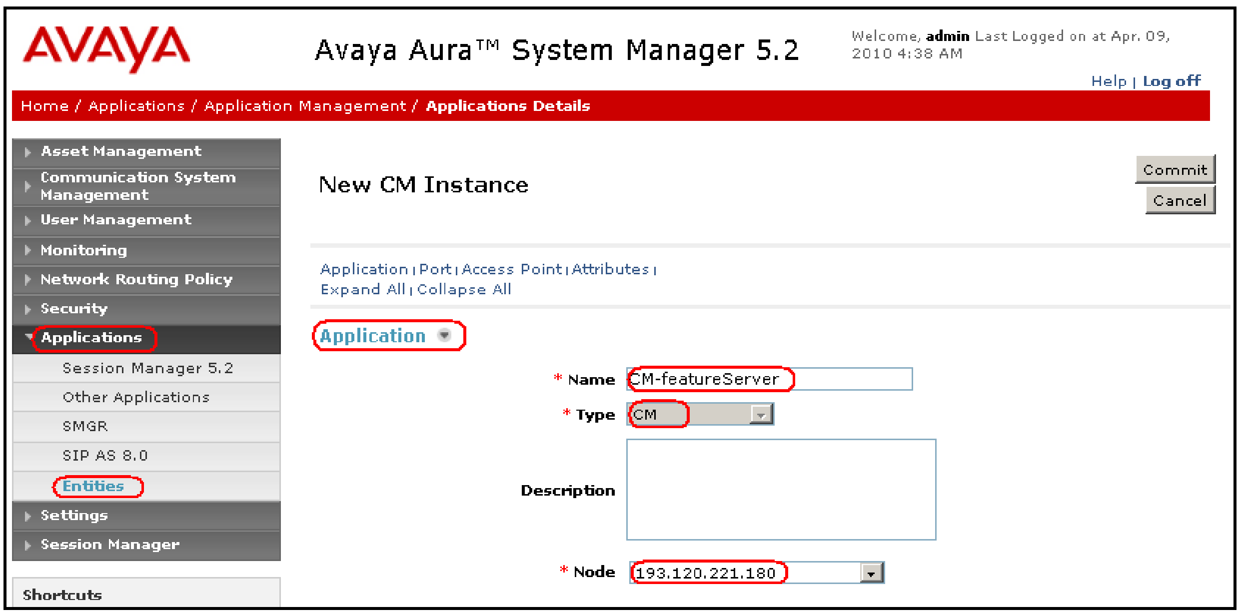
In order for Communication Manager to provide configuration and Feature Server support to SIP phones when they register to Session Manager, Communication Manager must be added as an application.

6.9.1. 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**:

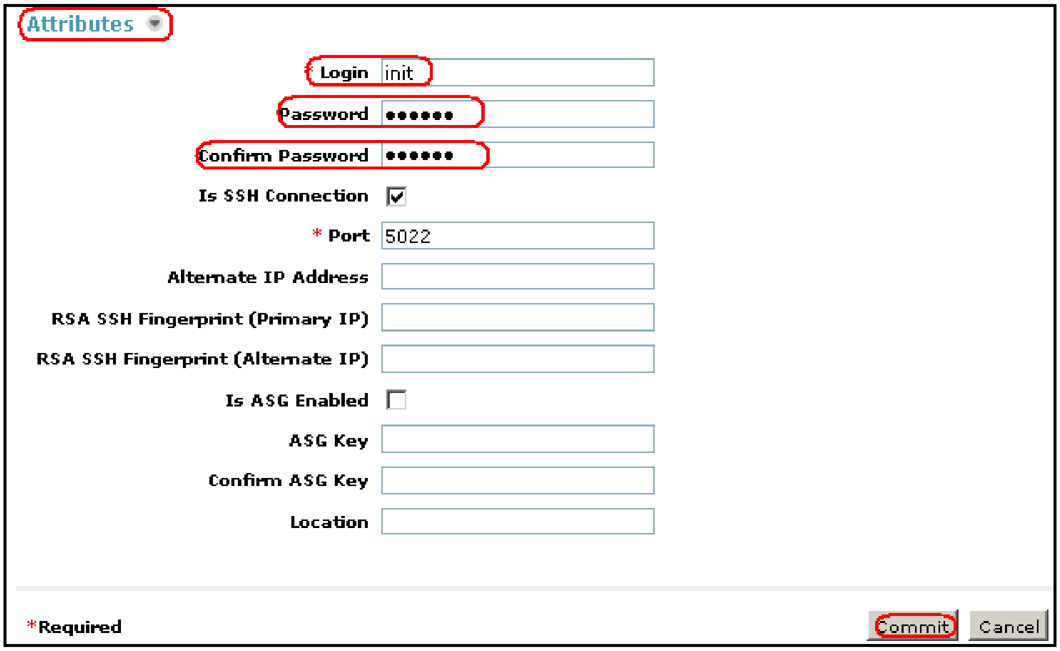
- **Name**: Enter a descriptive name i.e. **CM-featureServer**
- **Type**: Select **CM**
- **Node**: Select **Other..** and enter the IP address for CM SAT access i.e. **193.120.221.180**



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.

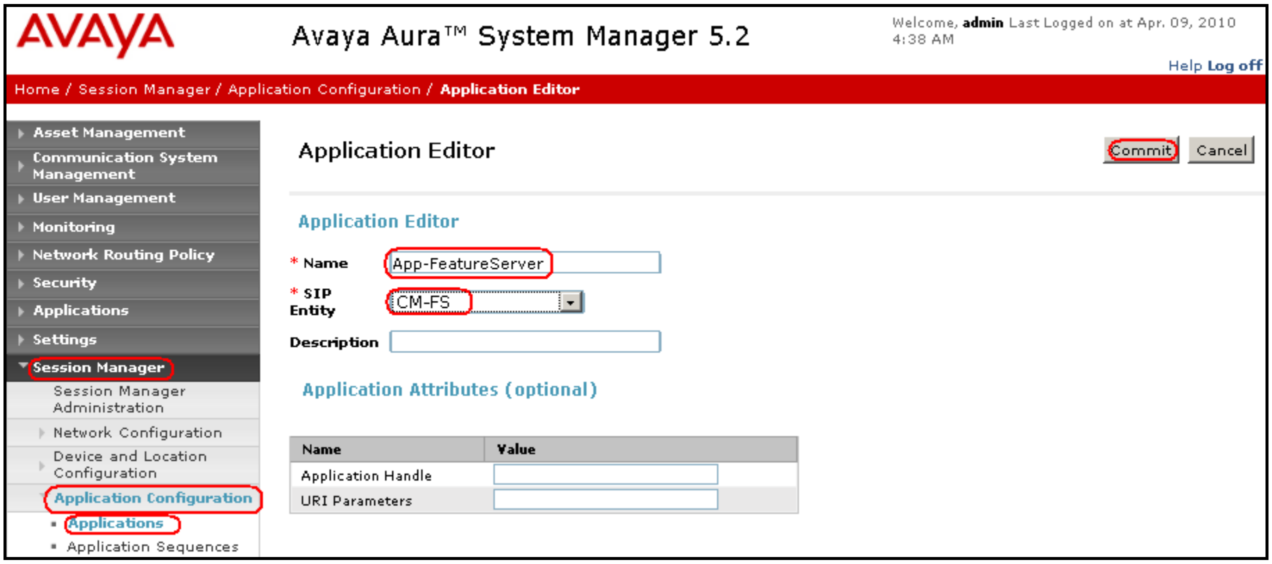


6.9.2. Create a Feature Server Application

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:

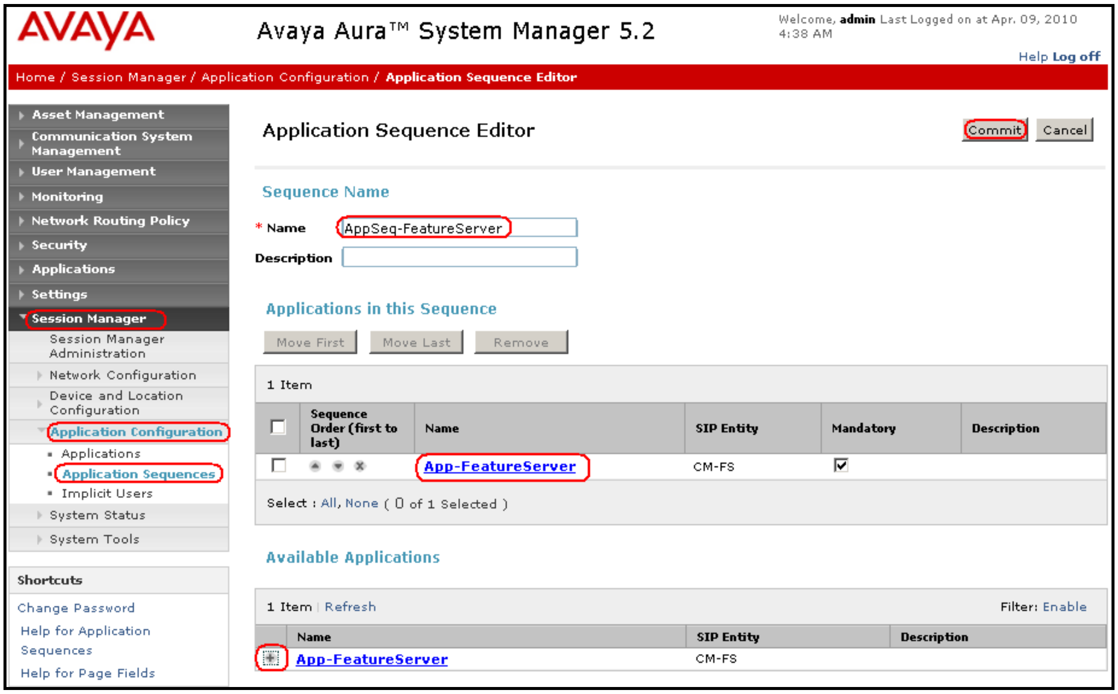
- **Name** A descriptive name
- **SIP Entity** Select the CM SIP Entity defined in **Section 6.4.2**

Click on **Commit** to save.



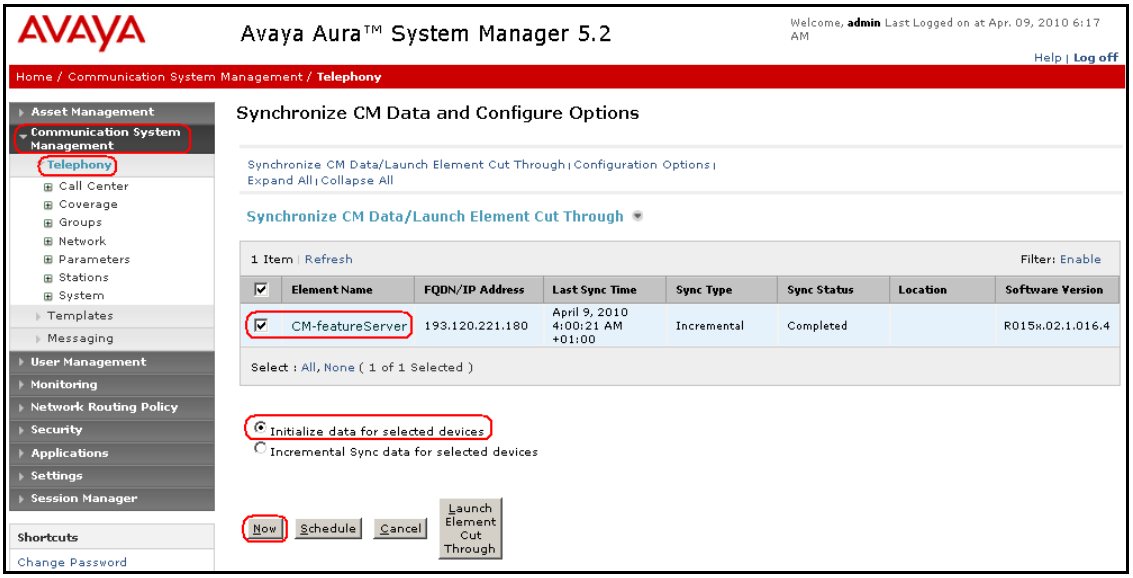
6.9.3. Create a Feature Server Application Sequence

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.



6.9.4. Synchronize Avaya Aura™ Communication Manager Data

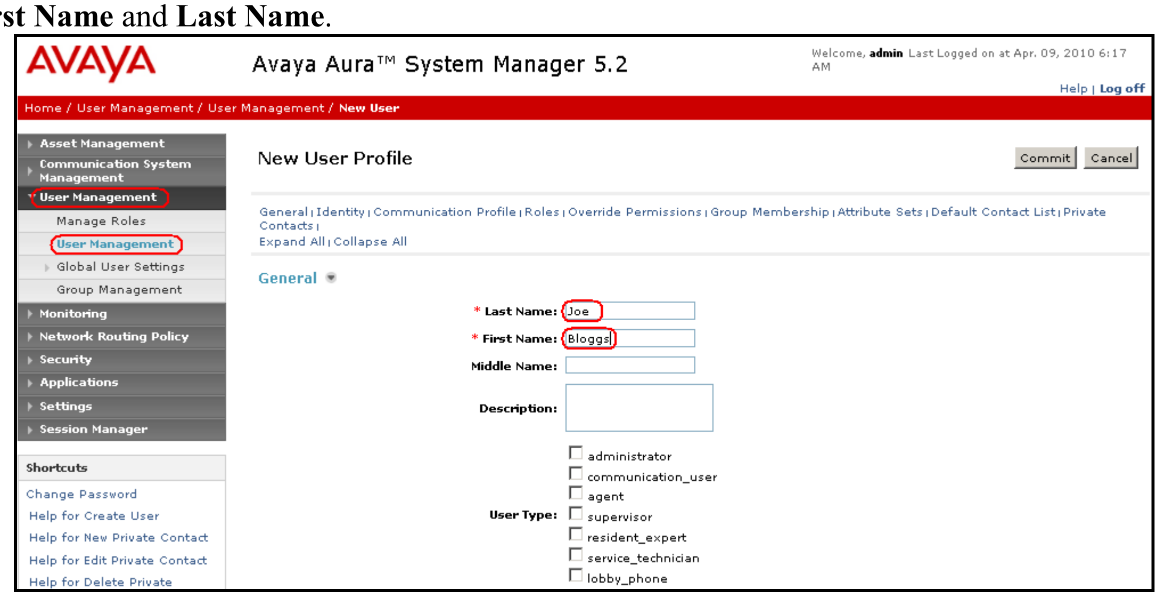
Select **Communications System Management → Telephony** on the left. Select the appropriate **Element Name**. Select **Initialize data for selected devices**, then click on **Now**. This may take some time.



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

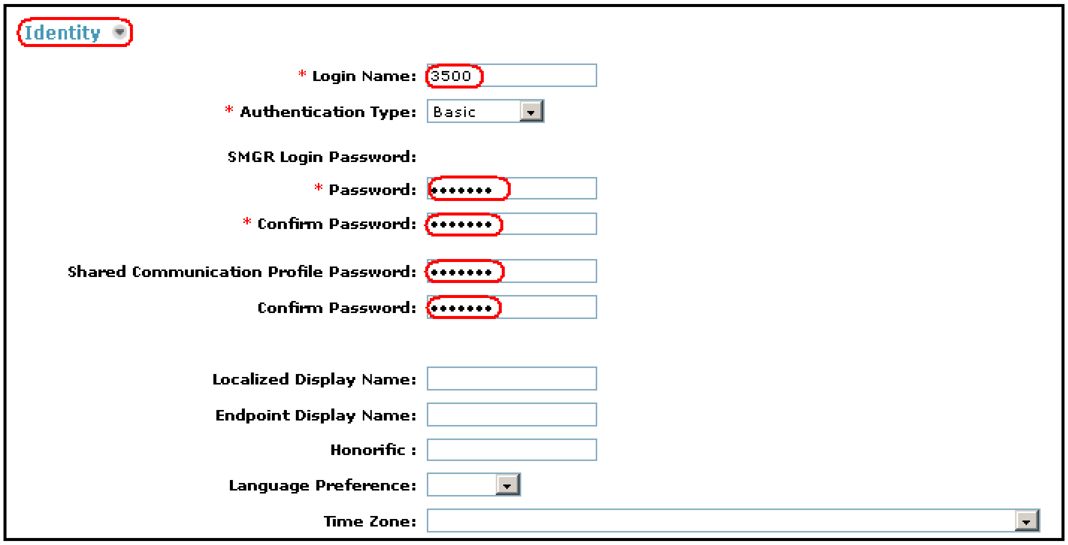
6.10. Add Users for SIP Phones

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**.



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

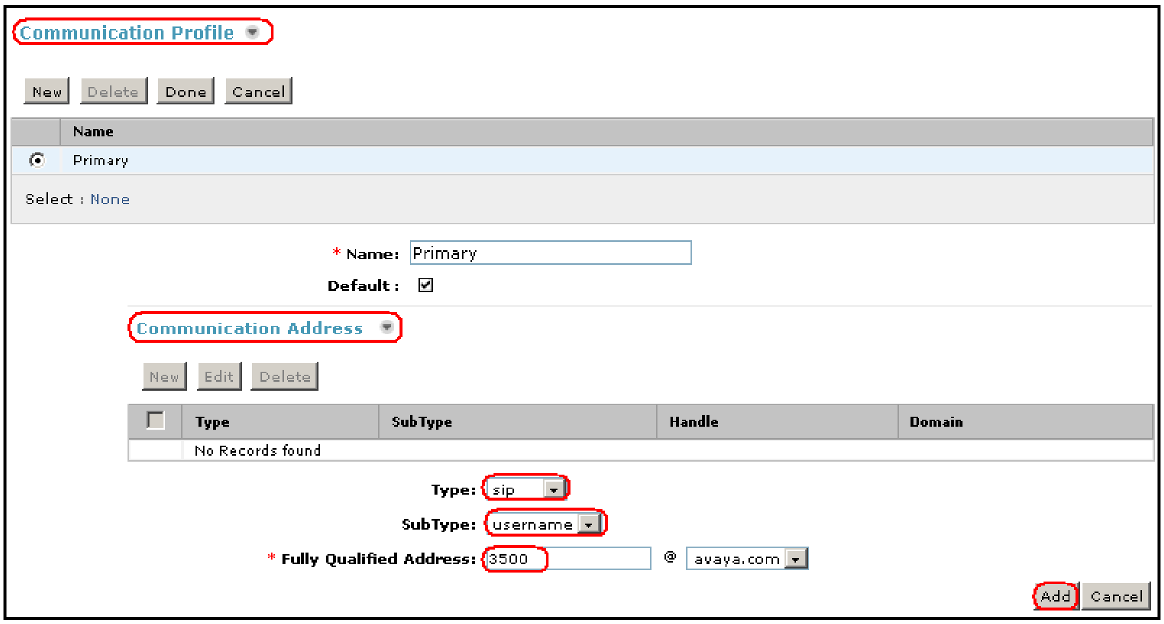
- **Login Name** The desired phone extension number @domain.com where domain was defined in **Section 6.1**
- **Password** Password for user to log into SMGR
- **Shared Communication Profile Password**
  Password to be entered by the user when logging into the phone



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:

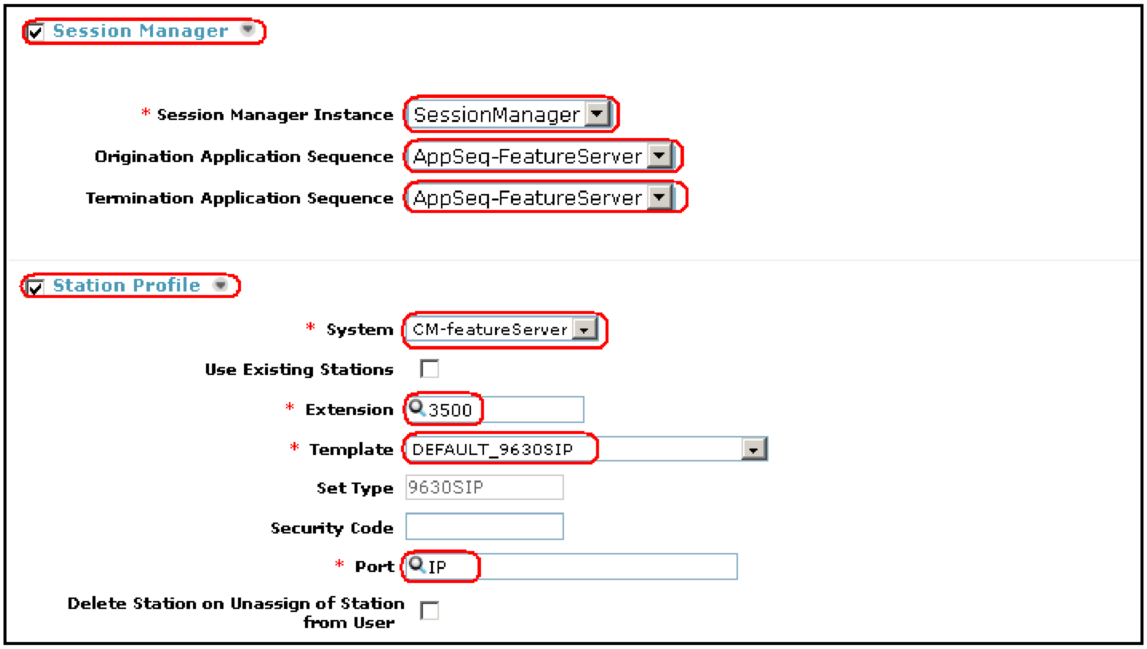
- **Type** Select **SIP**
- **SubType** Select **username**
- **Fully Qualified Address** Enter the extension number i.e. **3500**

Click on **Add**.



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 6.9.3**. 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).

7. 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 **[15], [16]** and **[17].** 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.

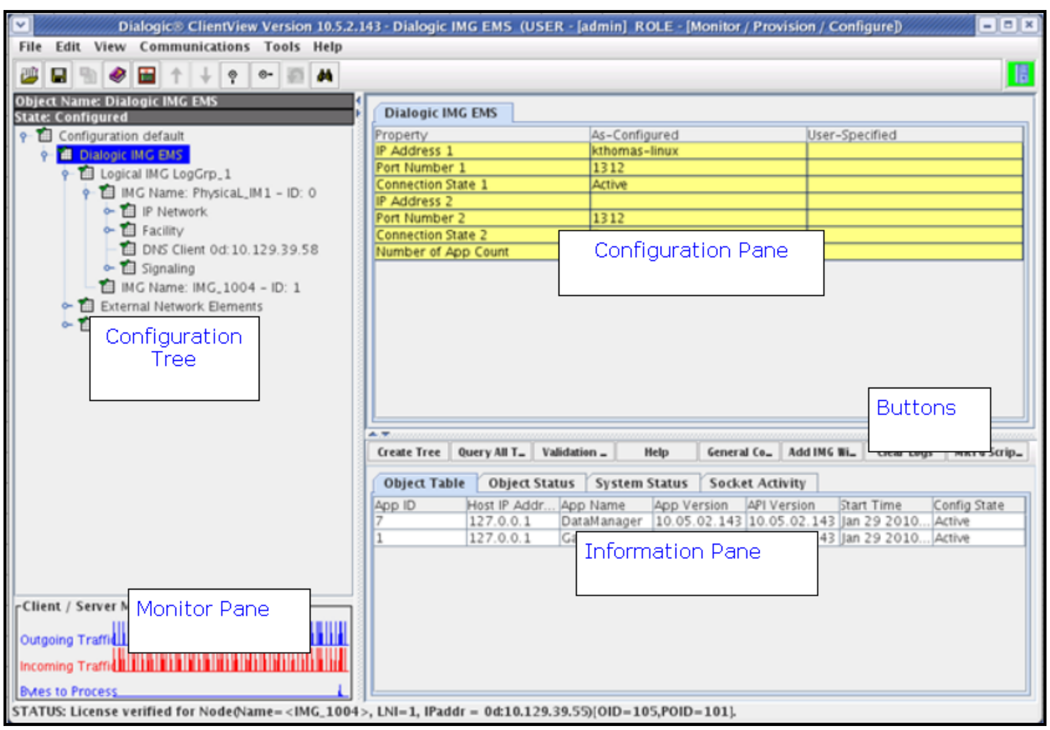
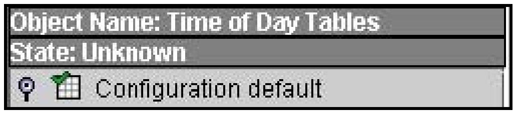


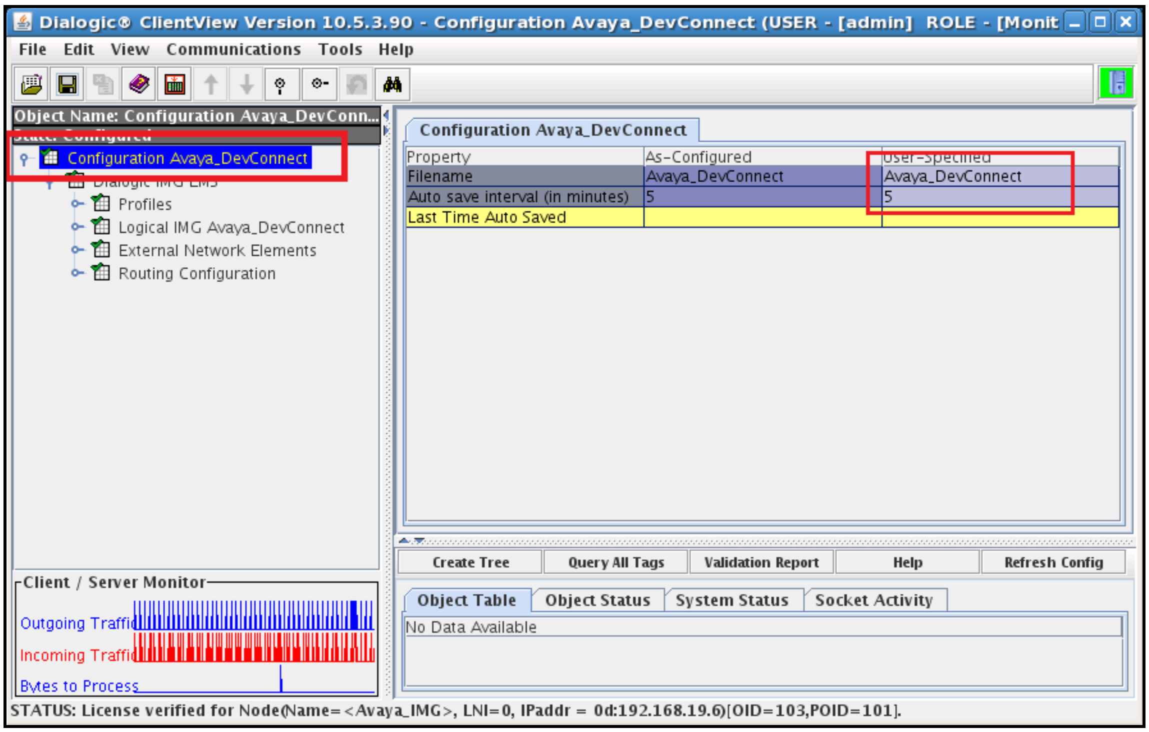
Figure 2 - Dialogic Inc. ClientView Main Window

7.1. 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.

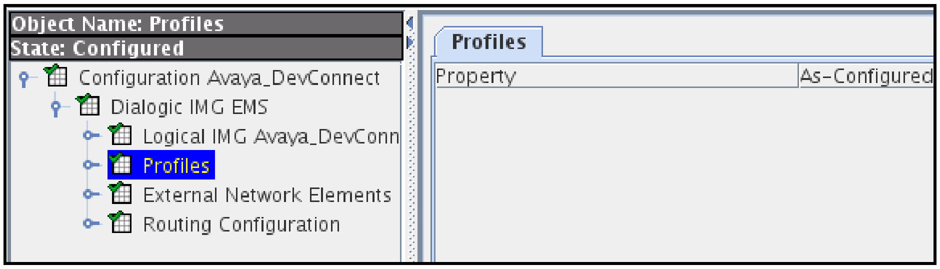


Enter a descriptive name in the **Filename** field in the Configuration Pane. To save the changes, right-click **Configuration Avaya_DevConnect**, and select **Commit**. The picture below shows the actions performed on IMG 1010



7.2. Profiles

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

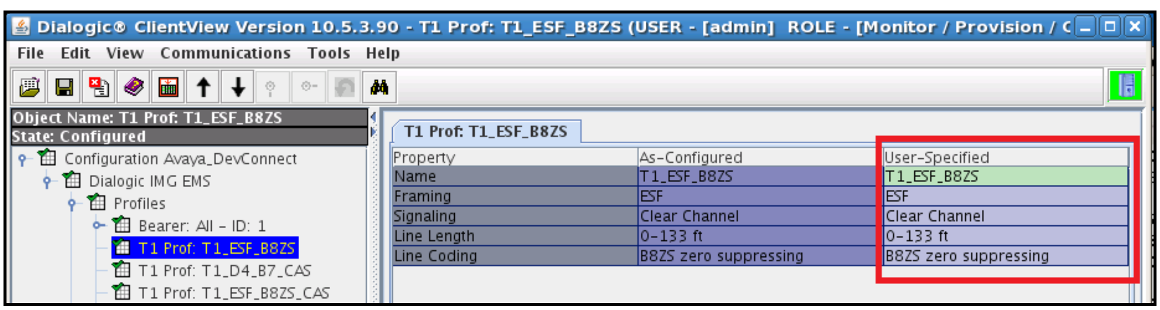


7.2.1. DS1 Profiles

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

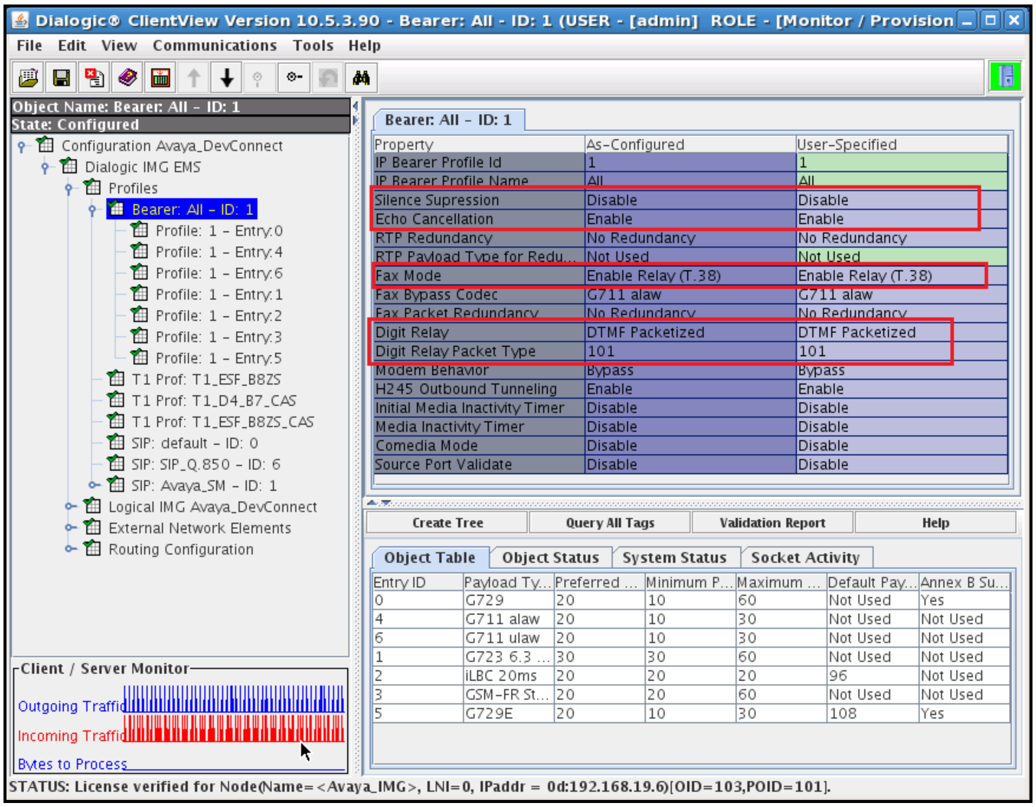


7.2.2. IP Bearer Profiles

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

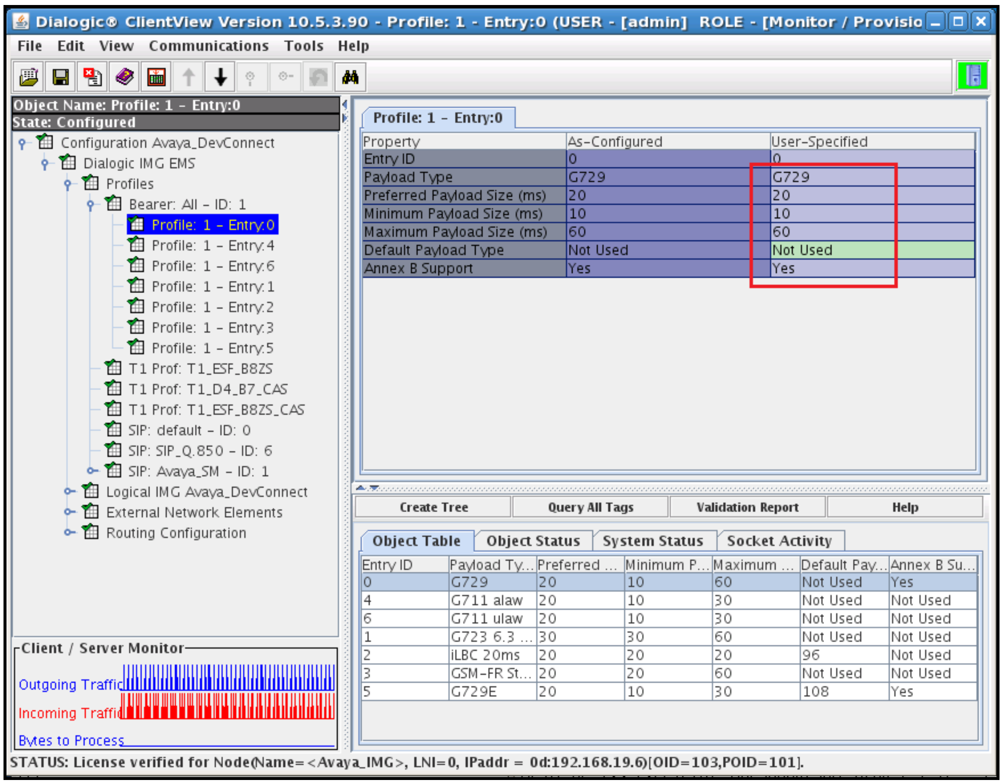


7.2.2.1 New IP codec in Bearer Profile

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.

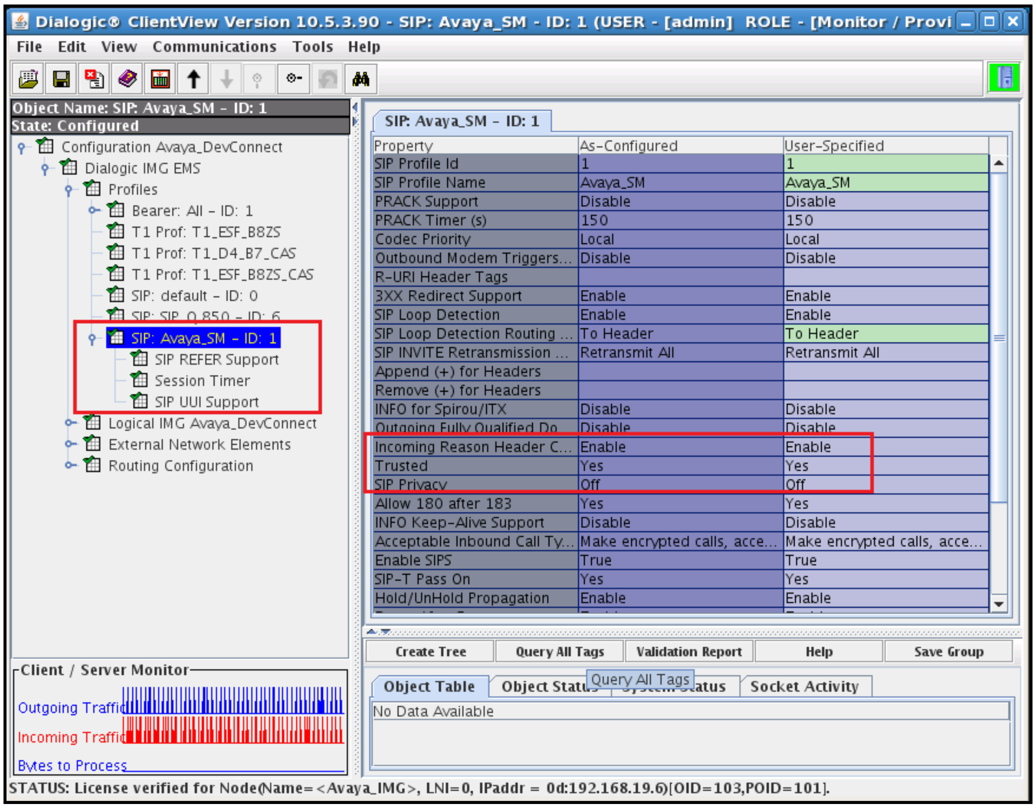


7.2.3. SIP Profiles

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.

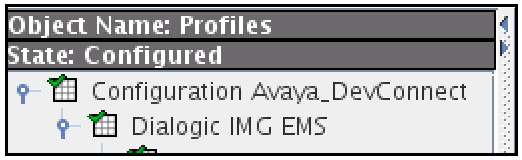
Note: if needed P-AI provide from Session Manager, set **SIP Privacy** to **On**



7.2.4. Logical IMG

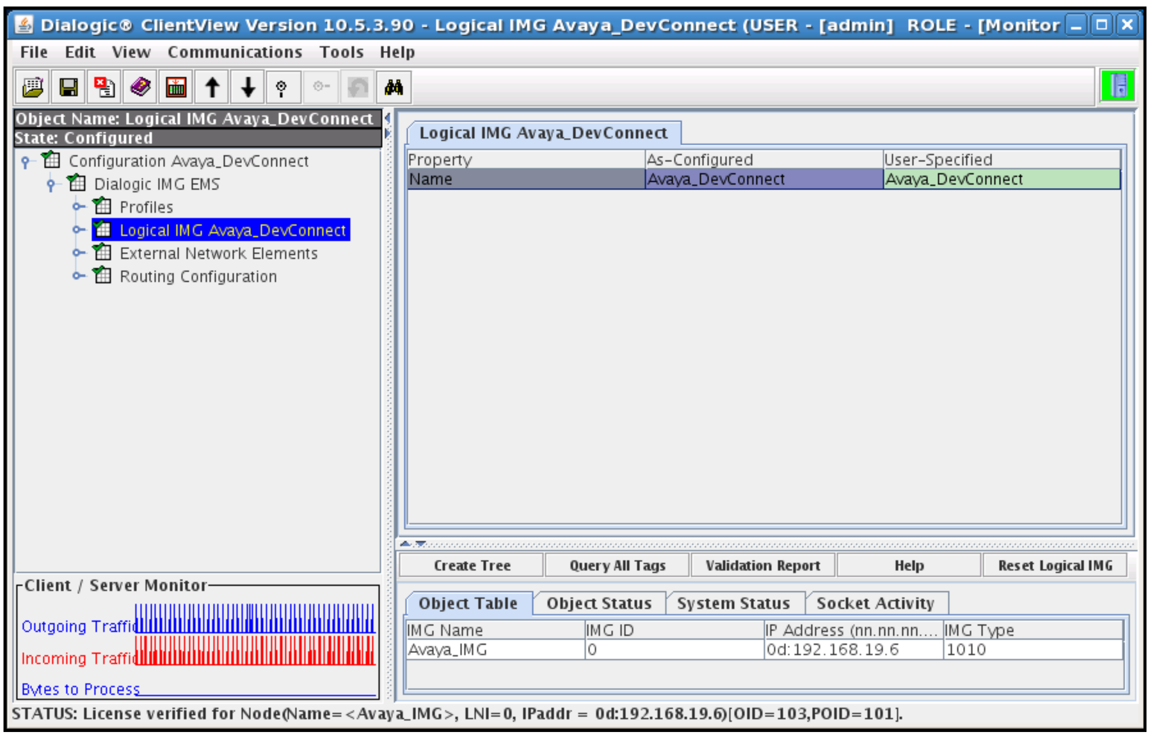
Create a logical IMG as follows:

- 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**.

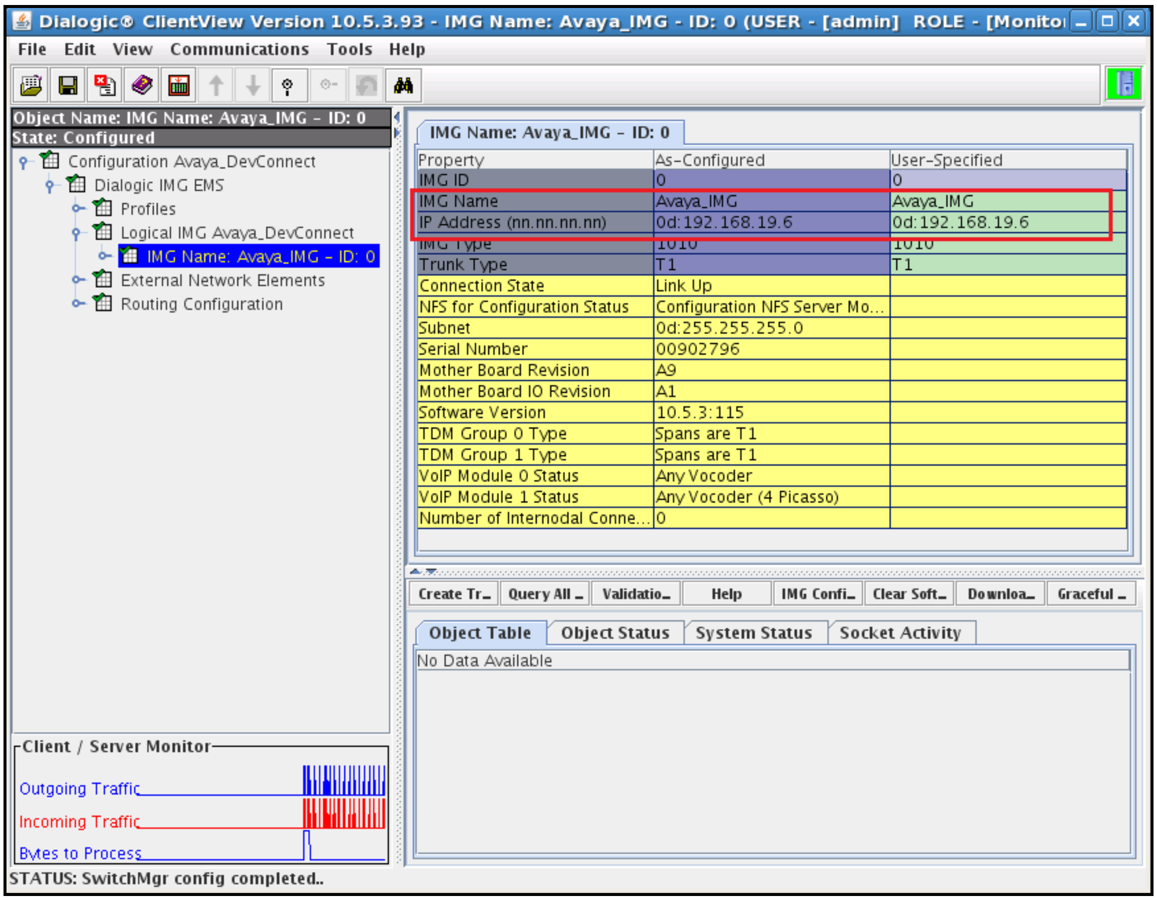


7.2.4.1 Physical IMG

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**.

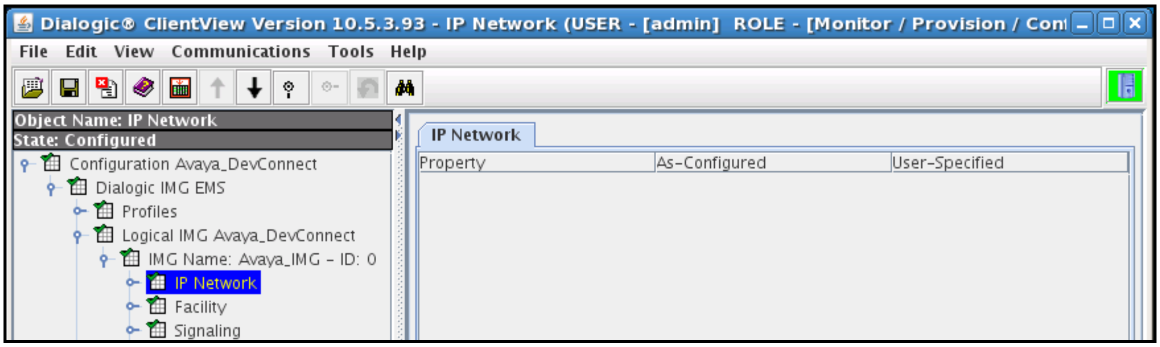


7.2.4.1.1 Network Interfaces

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.



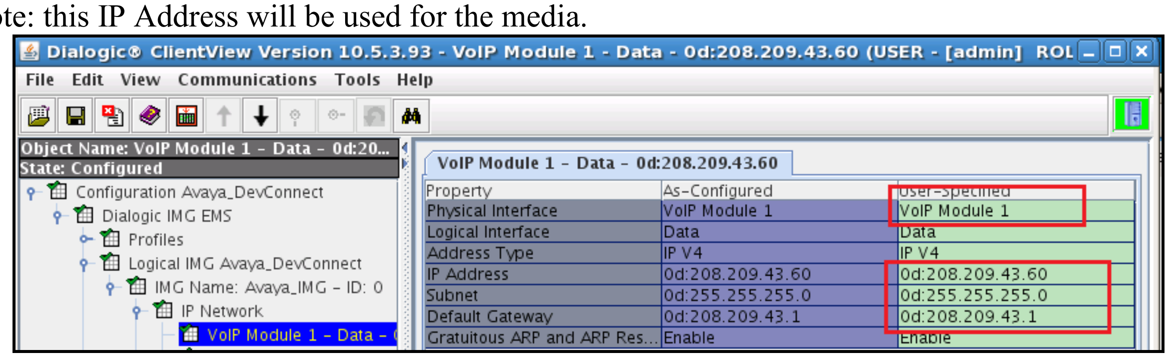
7.2.4.1.2 Network Interface VoIP module

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.

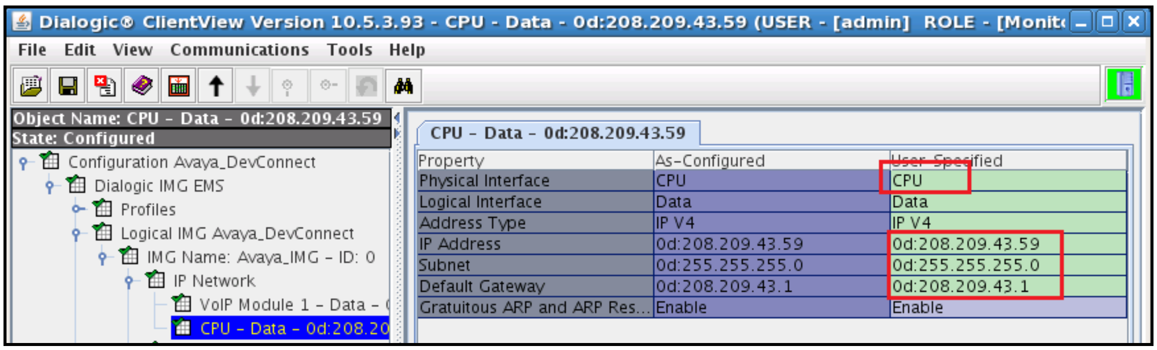


7.2.4.1.3 Network Interface CPU

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.

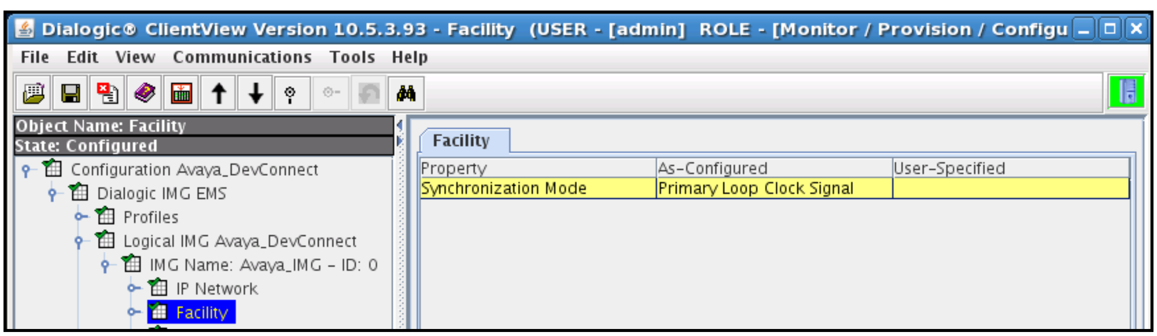


7.2.4.1.4 Facilities (DS1 and VoIP)

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**.



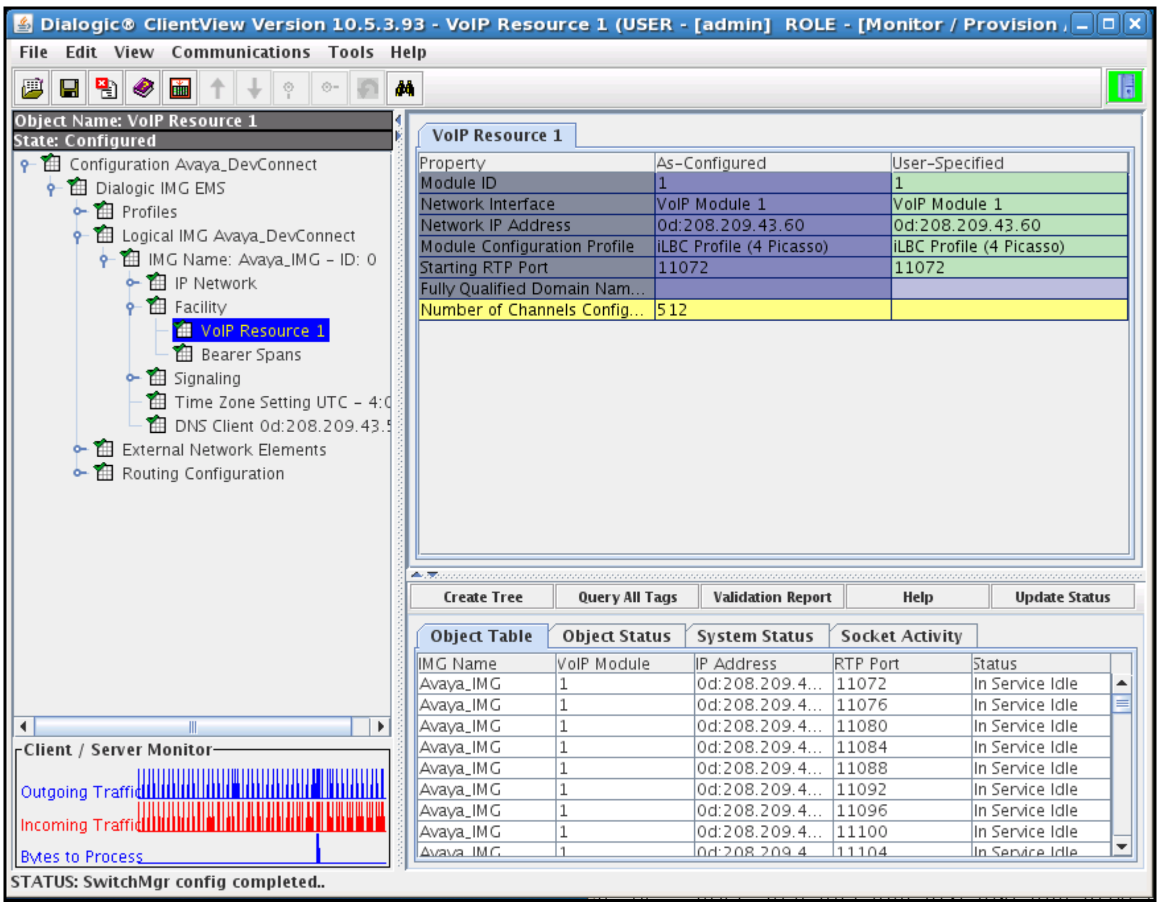
7.2.4.1.5 VoIP Facilities

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 7.3**.

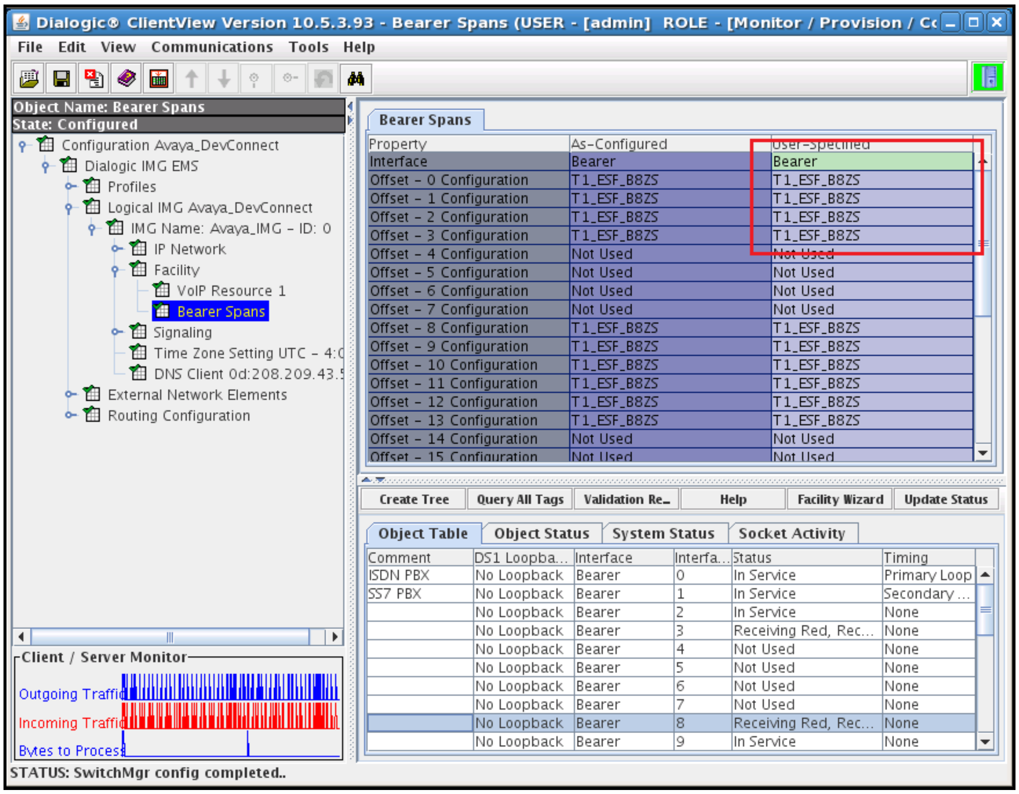


7.2.4.1.6 DS1 Facilities

Configure a TDM DS1 T1 or E1 as follows:

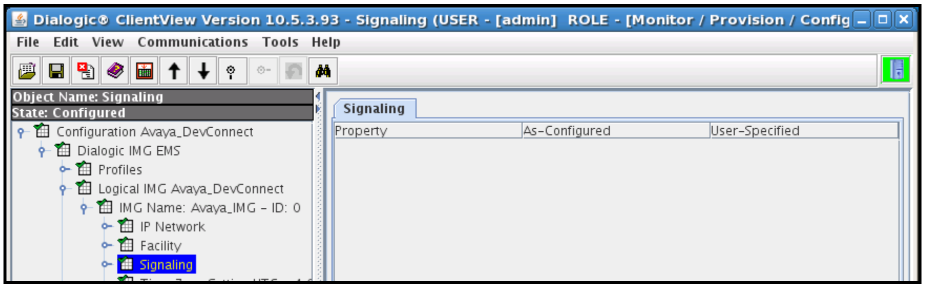
- Right-click **Facility** in the Configuration Tree, and select **New TDM Spans**.
- Select **Bearer** or **Signaling** spans.
- In the configuration pane select the DS1 span and select the profile for that span.

TDM spans will be brought in service and 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**.



7.2.4.1.7 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.

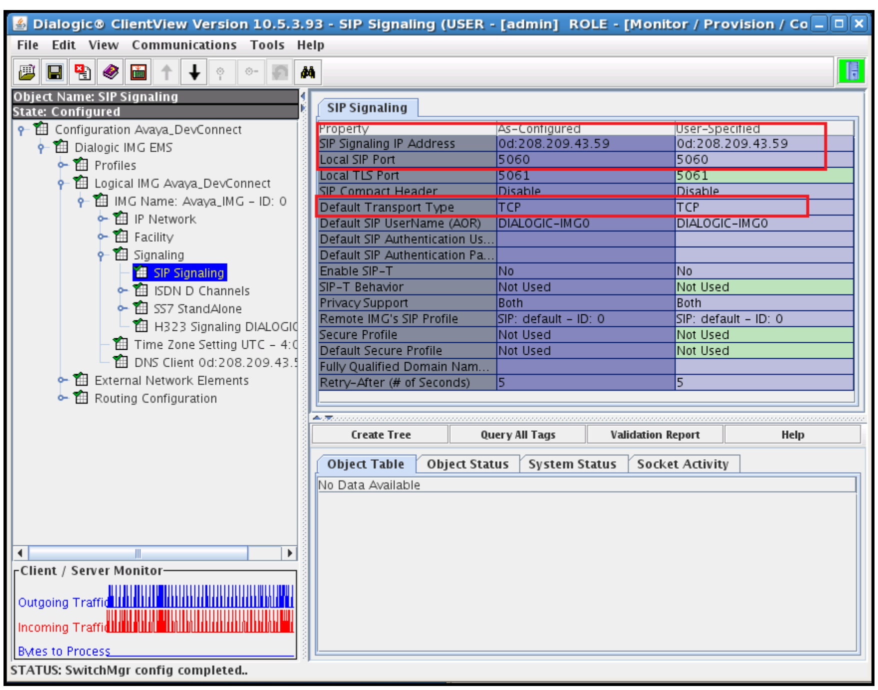


7.2.4.1.8 SIP Signaling

Configure SIP Signaling as follows

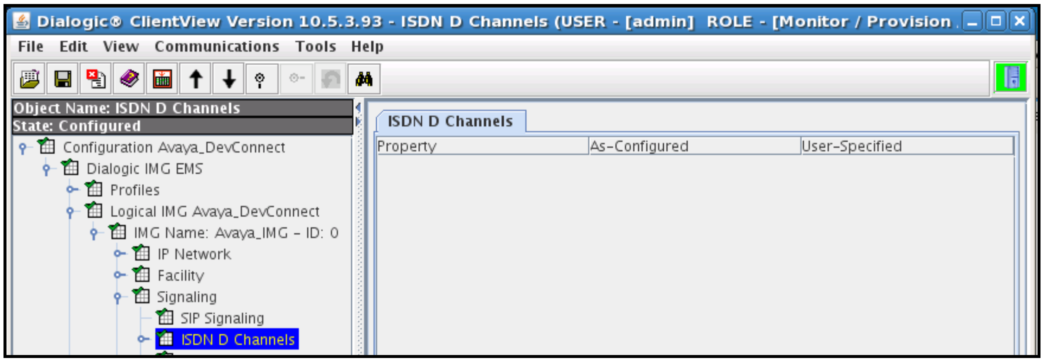
- Right-click **Signaling** in the Configuration Tree, and select **New SIP**.
- Administer settings in the Configuration Pane that enable SIP connectivity between the IMG and other SIP User Agents as follows:
  - o Enter the IP address assigned to the IMG in the **SIP Signaling IP Address** field.
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



7.2.4.1.9 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**.

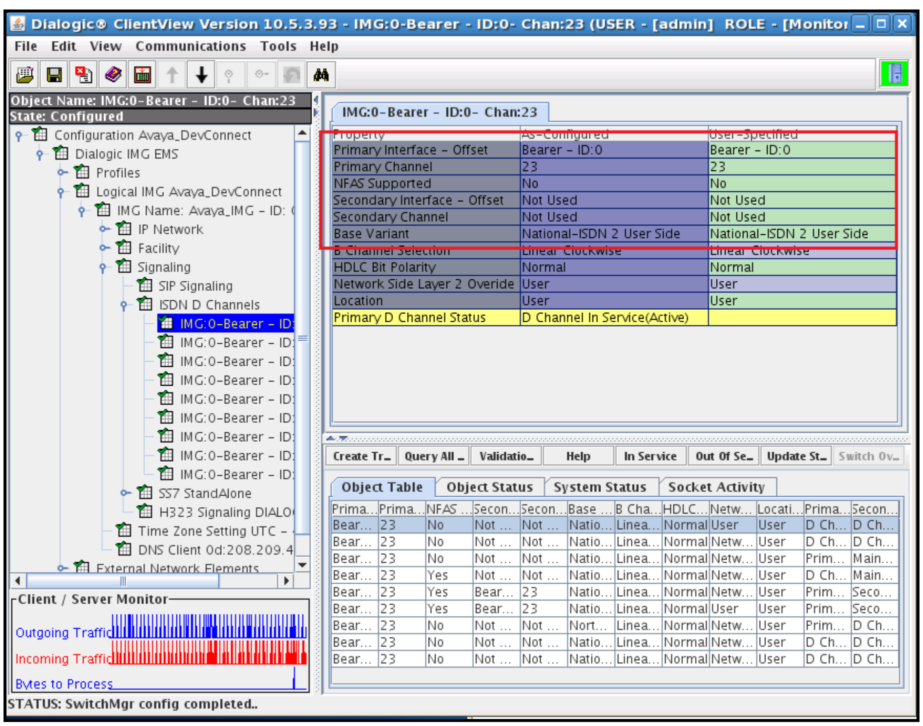


7.2.4.1.9.1 ISDN D Channel

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**.

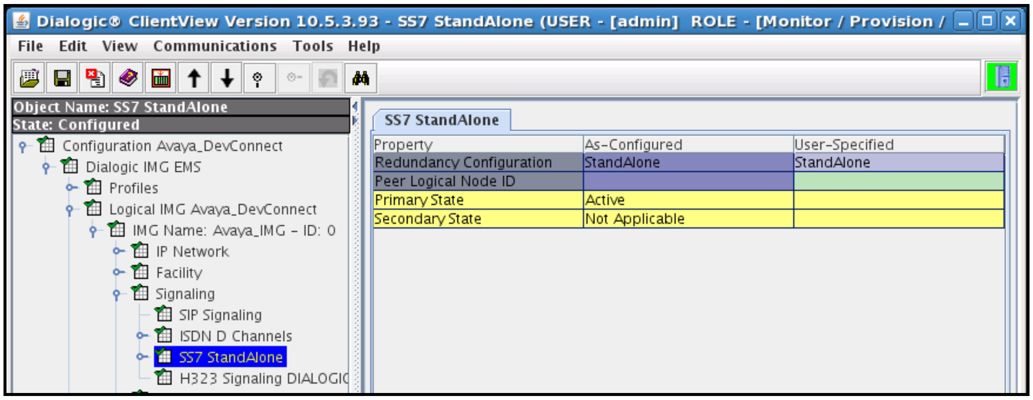


7.2.4.1.10 SS7 Signaling

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**.

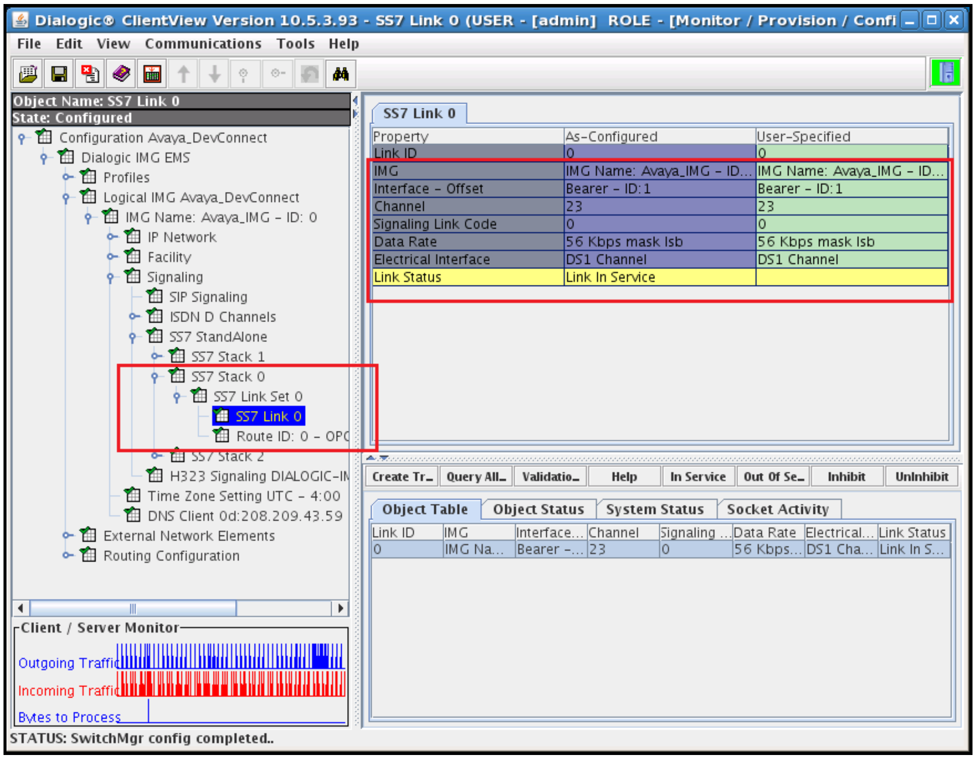


Configure SS7 Stack, Linkset, Links, & Routes as follows:

- 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**.

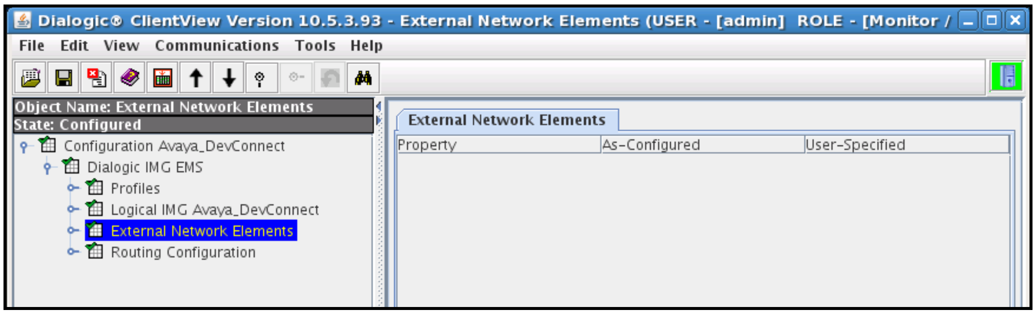


7.2.5. External Network Elements

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**.

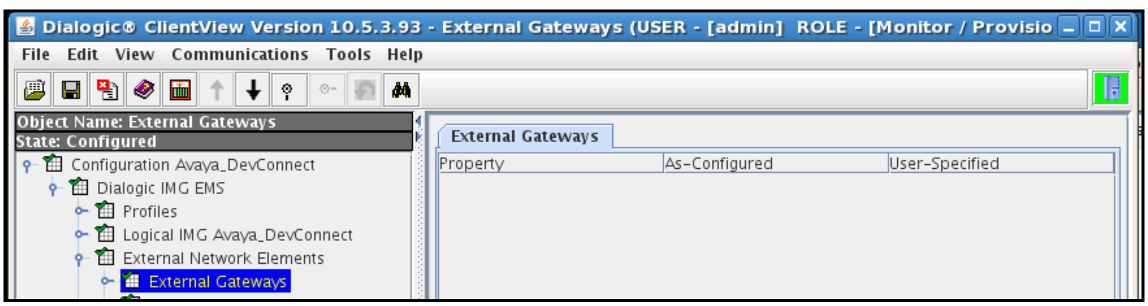


7.2.5.1 External Gateways

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**.

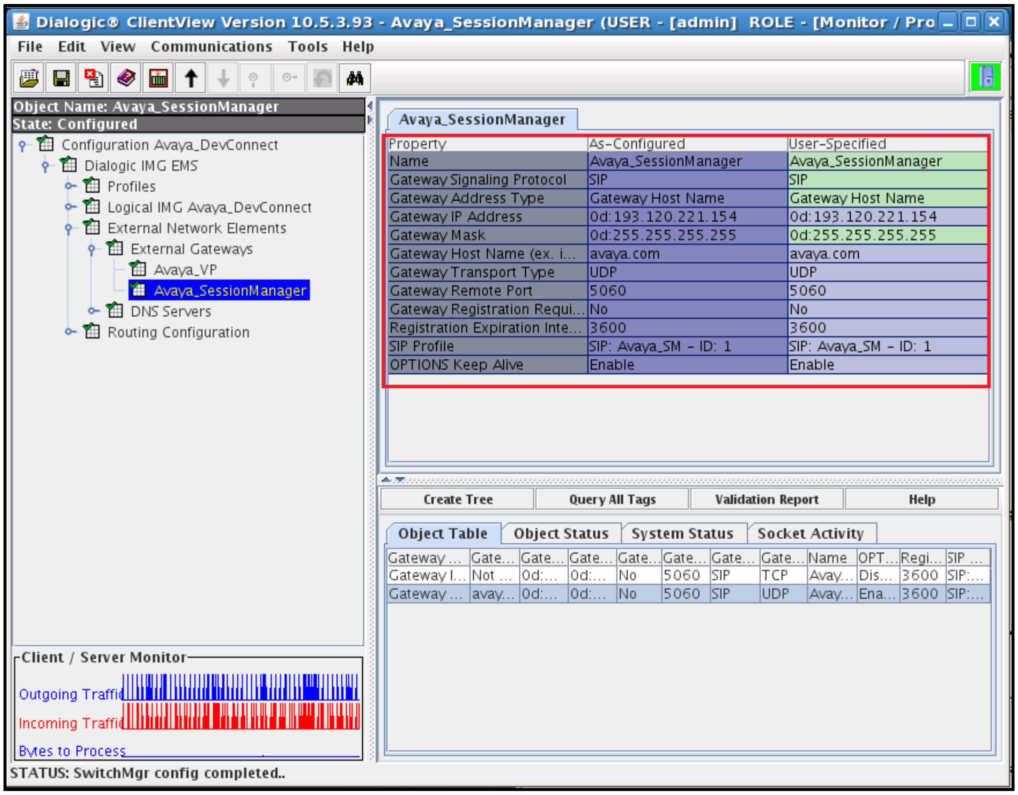


7.2.5.1.1 External Gateway

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**.

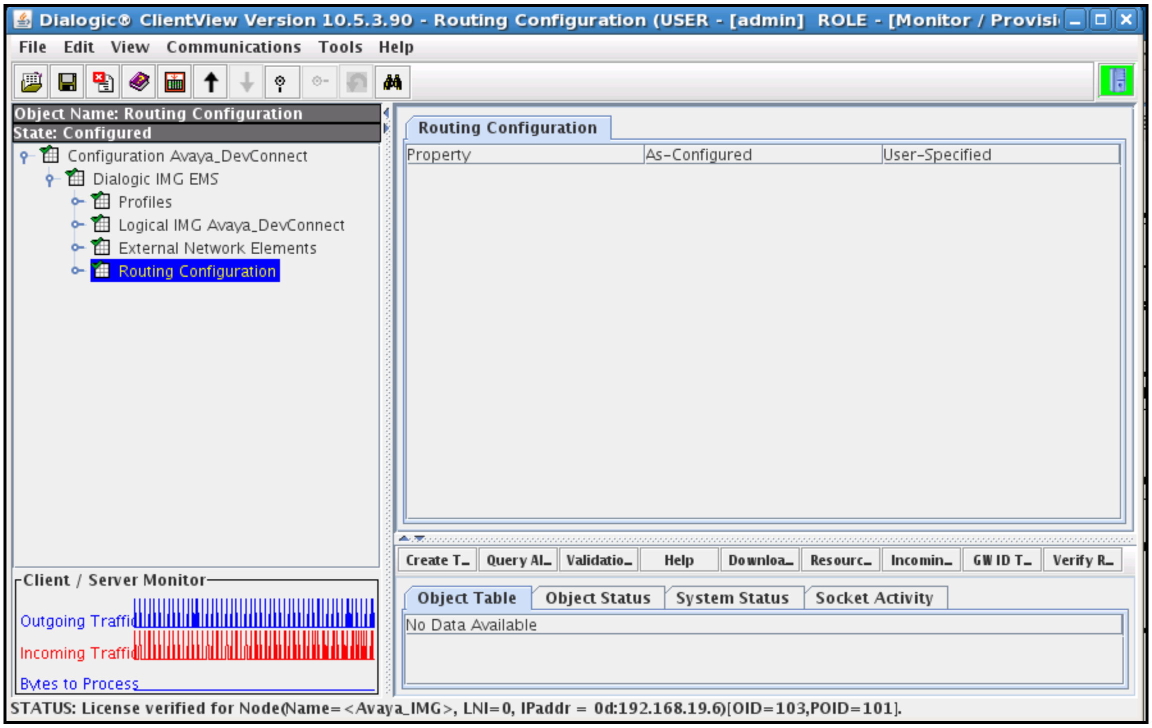


7.3. Routing configuration

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**.

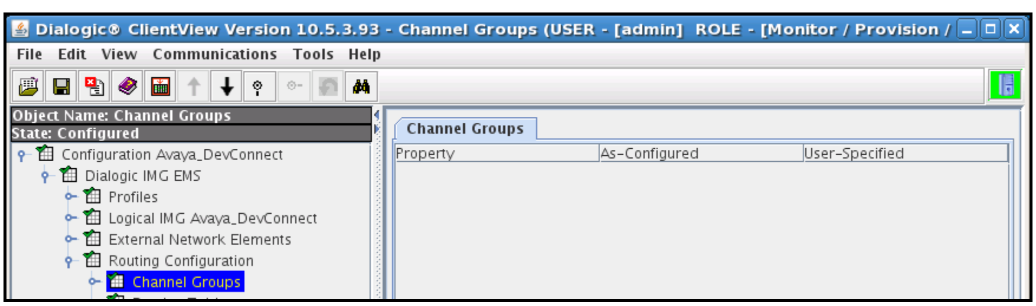


7.3.1. Channel Groups

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**.

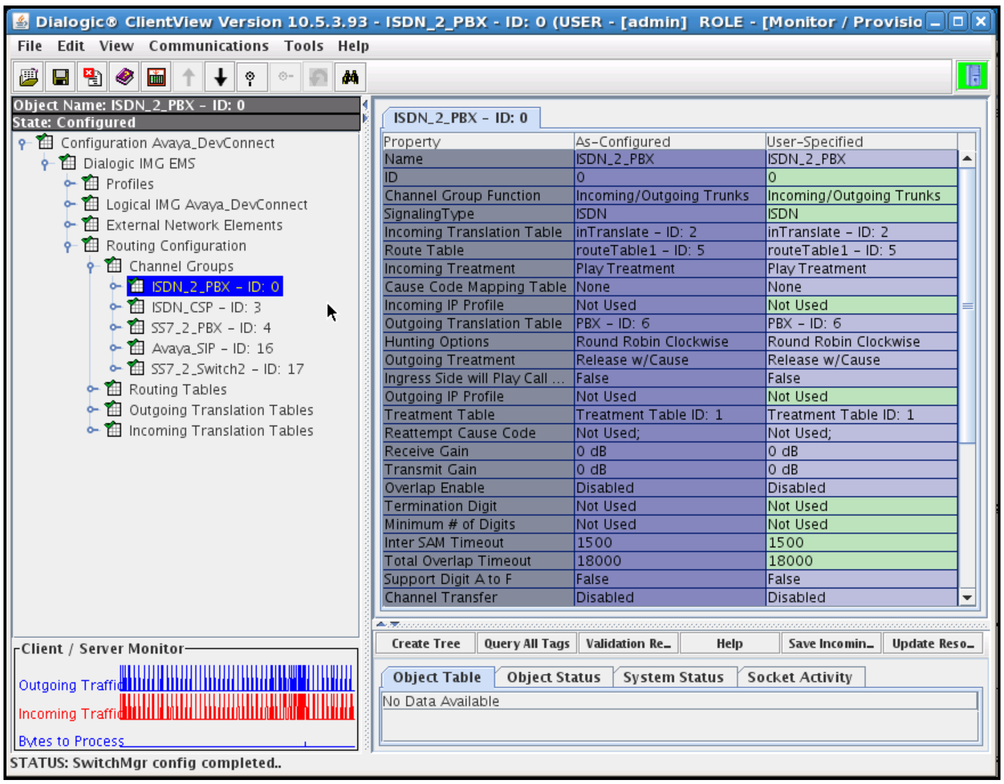


7.3.1.1 Channel Group (ISDN)

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**.

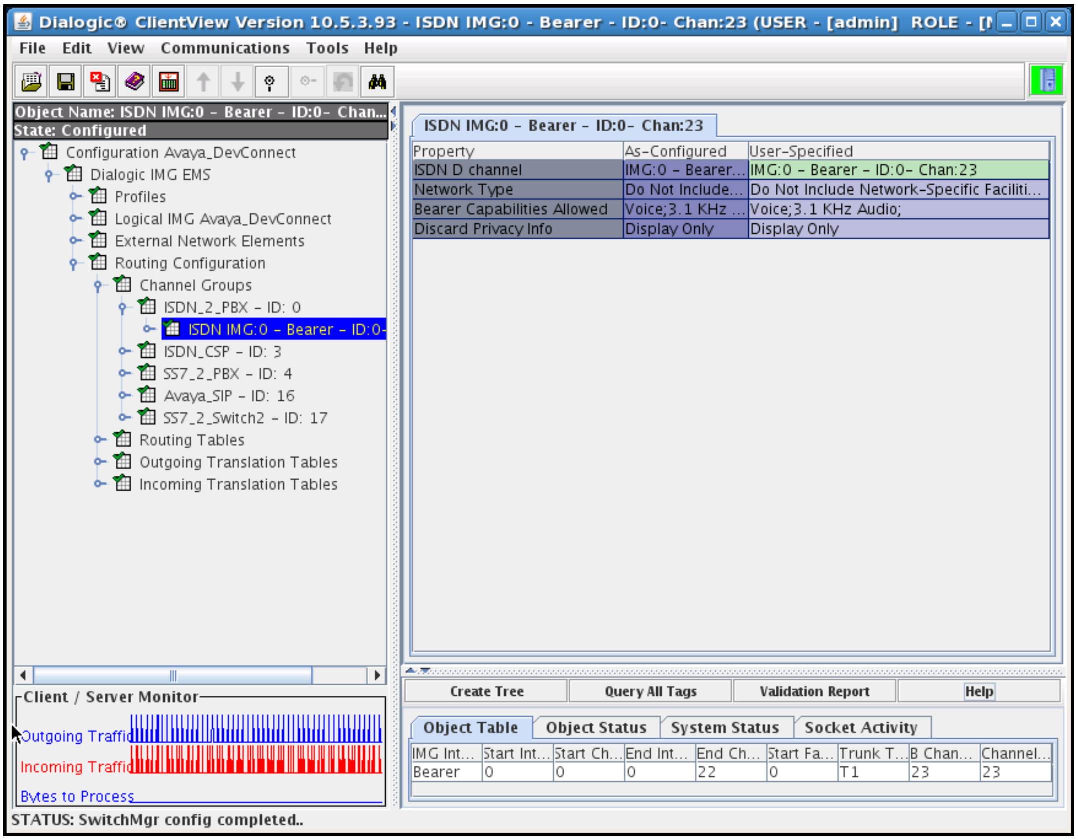


7.3.1.1.1 Assign ISDN D channel to Channel group

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**.

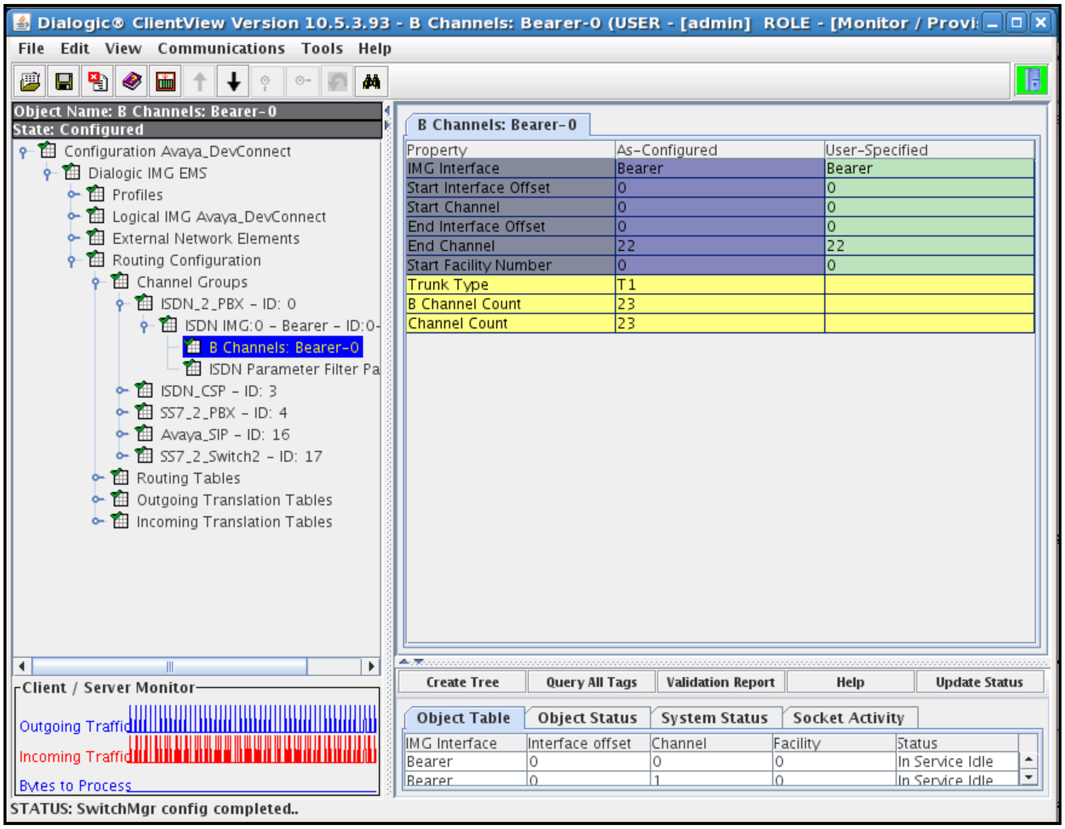


7.3.1.1.2 Assign ISDN B channels to the ISDN Group

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**.

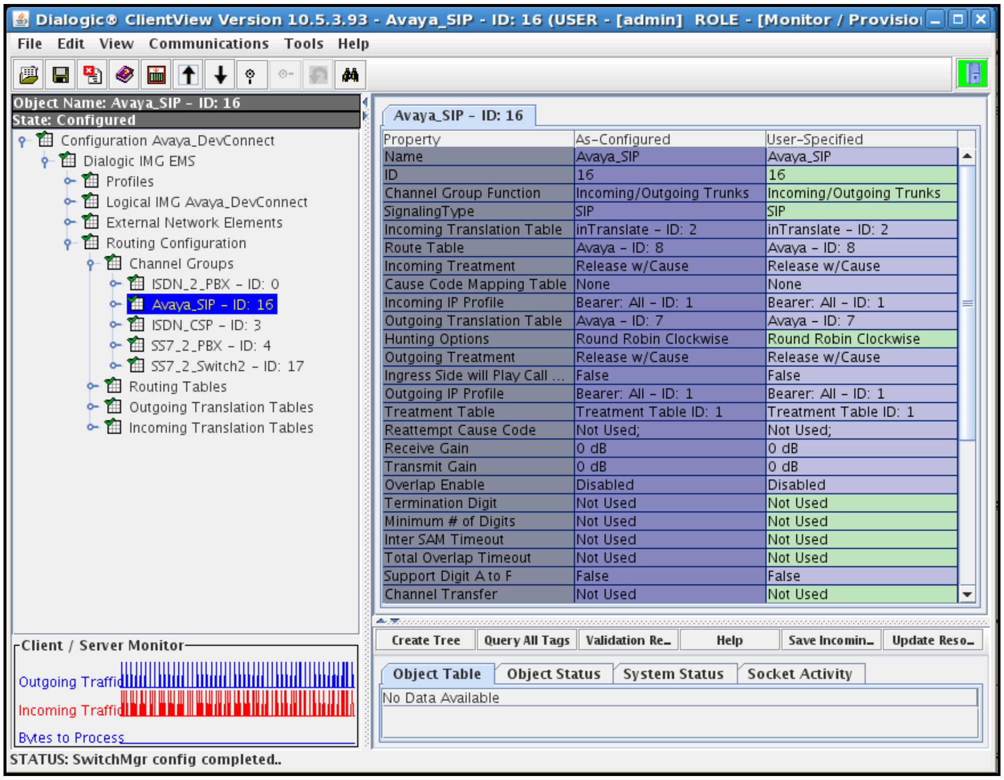


7.3.1.2 **Channel Group (SIP)**

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**.



7.3.1.3 Channel Group with SIP Gateway

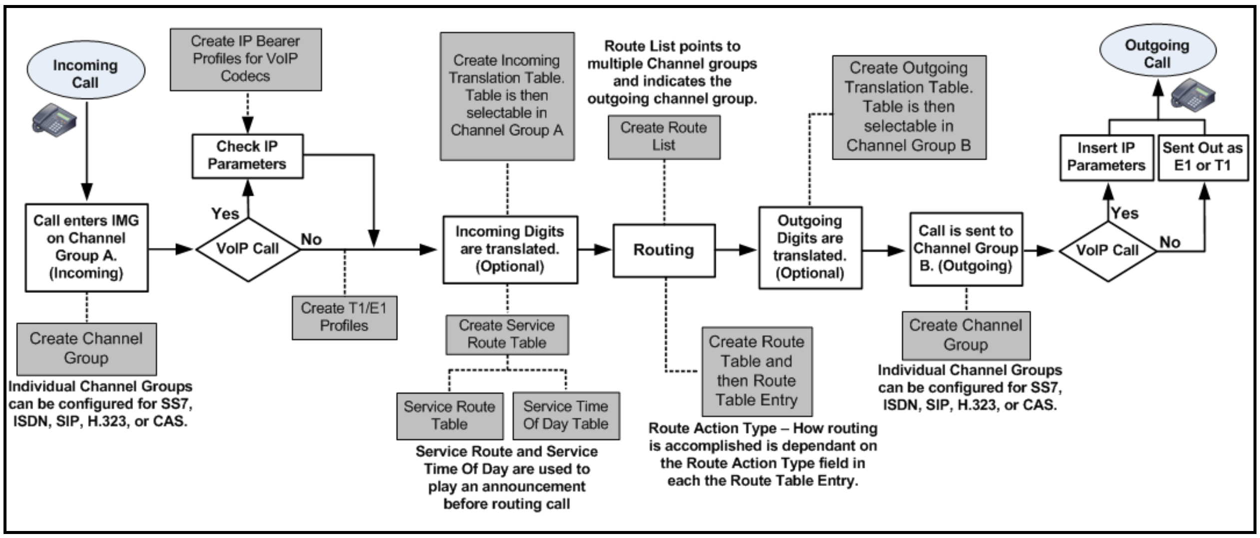
Assign a SIP Gateway to the Channel Group corresponding to each External Gateway previous:

- 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**.

7.4. Routing

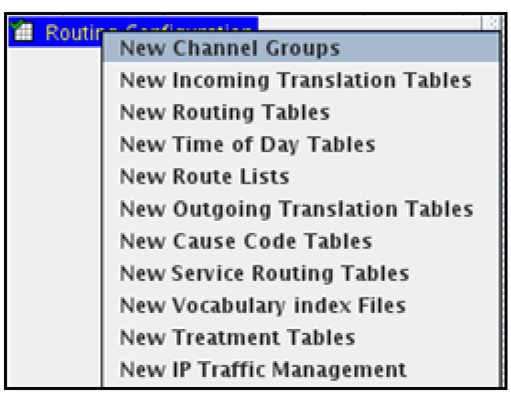
The following diagram shows the process the IMG goes through to route a call, and the ClientView GUI panes you use to configure various routing elements. Grey boxes represent ClientView configuration panes.



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**.



7.4.1. 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.

7.5. Additional configuration

Other optional configuration objects can be configured as needed.

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

8. Verification Steps

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

8.1. Verify Avaya Aura^{TM} 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
```

8.2. Verify Avaya Aura^{TM} Communication Manager Feature Server Trunk Status

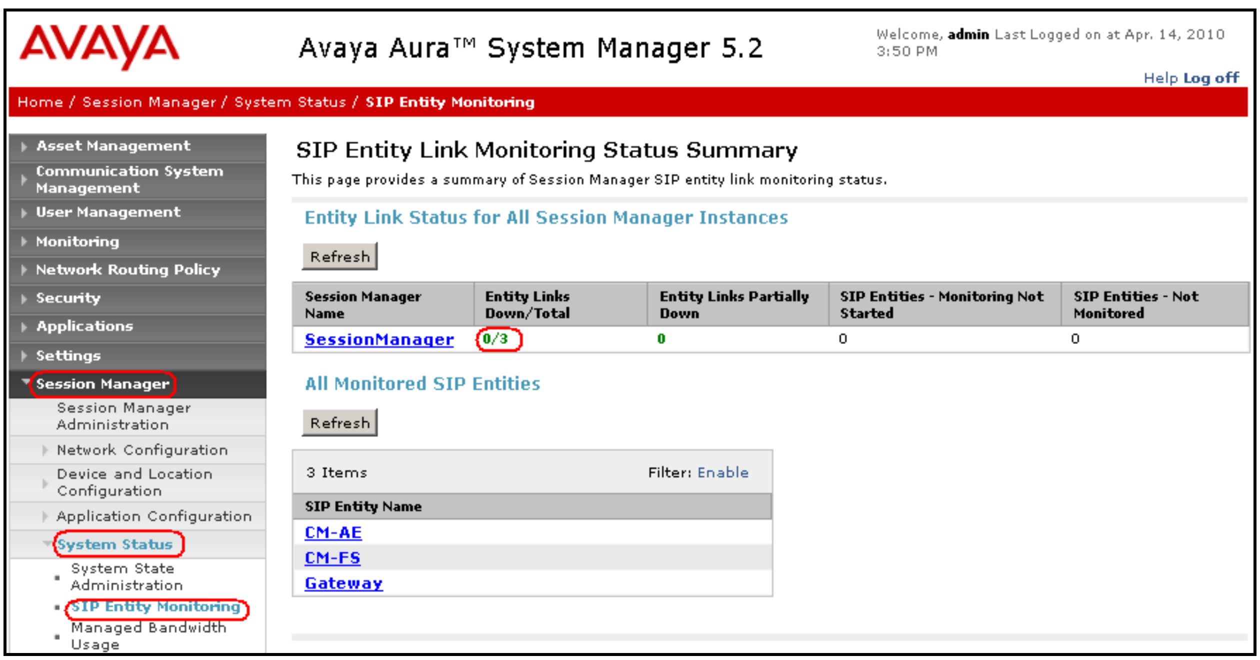
On Communication Manager Feature Server, ensure that all the signalling groups are in-service status by issuing the command status **signalling-group n** where **n** is the signalling group number.

```
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
```

8.3. SIP Monitoring on Avaya Aura^{TM} 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.



9. General Test Approach

The interoperability compliance test included feature and serviceability. The feature testing focused on verifying the following:
Basic Interoperability:

- PSTN calls from and to Avaya IP endpoint
- Calling with various Avaya telephone models including IP/SIP models as well as traditional analog and digital TDM phones
- Support G.711A/MU G.729A/B
- Various PTSN dialing plans including national and international calling, toll-free, operator, directory assistance and direct inward dialed calling
- SIP transport using UDP and TCP

Advanced Interoperability:

- Codec negotiation
- Telephony supplementary features, such as Hold, Call transfer, Conference Calling and Call Forwarding
- DTMF Tone Support
- Voicemail Coverage and Retrieval
- Direct IP-to-IP Media
- EC500 for Avaya Aura™ Communication Manager

The serviceability testing focused on verifying the ability of solution to recover from adverse conditions, such as network failures.

9.1. Test Results and Remarks

All test cases were executed. During the compliance testing, it has been noted that Dialogic IMG 1010 does not recognize rtp-events originated from H323 endpoints if shuffling is enabled on the signaling trunk group. A workaround is available by disabling shuffling on the trunk used.

10. Conclusion

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

11. 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

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

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

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

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

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.