



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.

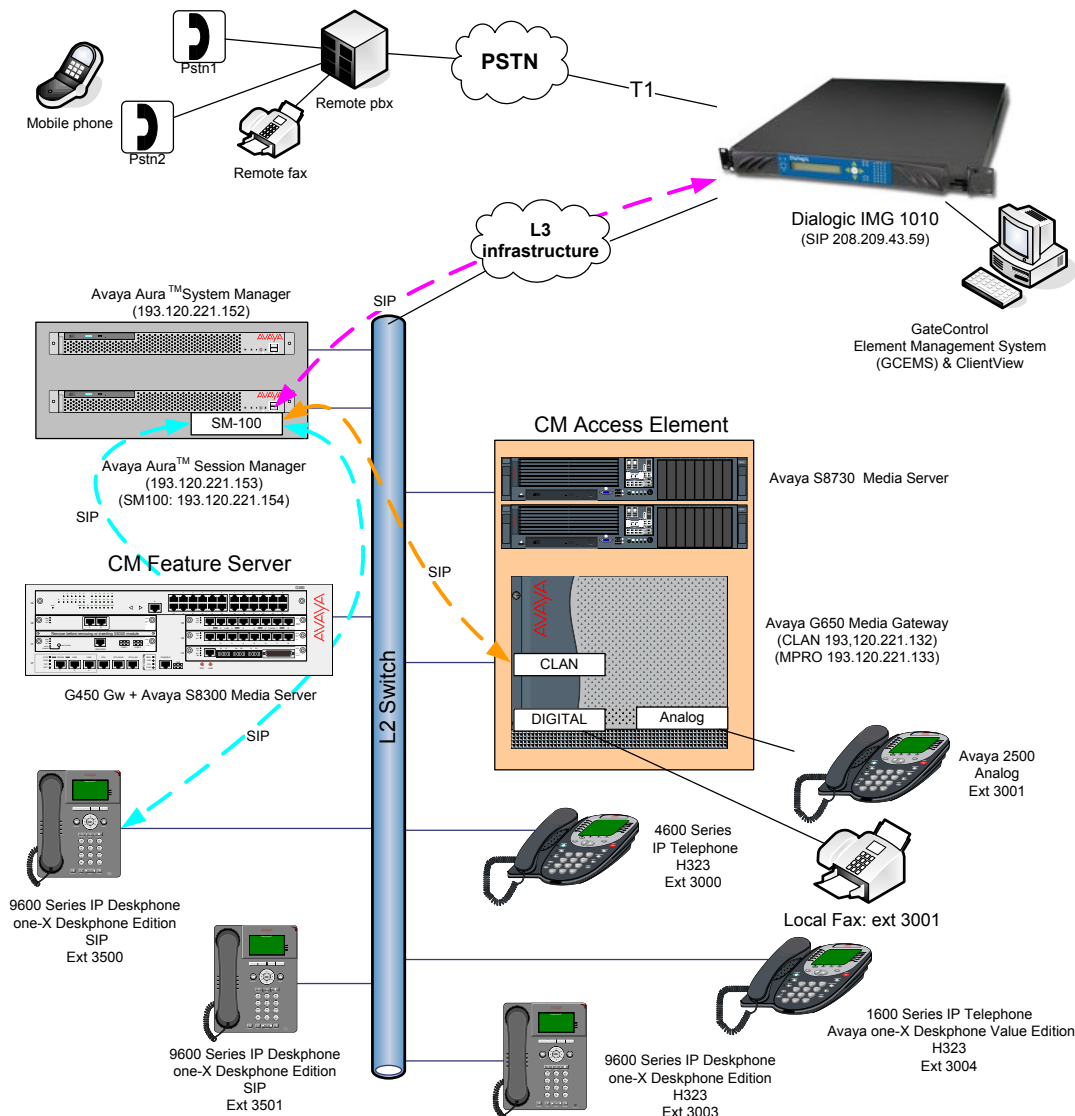


Figure 1 – Sample configuration for Avaya Aura™ Communication Manager and Avaya Aura™ Session Manager with Dialogic IMG 1010 using Sip Trunking

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 <ul style="list-style-type: none"> • IPSI (TN2312BP) • C-LAN (TN799DP) • IP Media Resource 320 (TN2602AP) • Analog (TN2793B) • Digital line (TN2214CP) 	<ul style="list-style-type: none"> • TN2312BP HW28 FW050 • TN799DP HW01 FW037 • TN2602AP HW08 FW053 • TN2793B 000005 • TN2214CP HW10 FW015
Avaya IP Telephones: <ul style="list-style-type: none"> • 9630 & 9620 (SIP) • 9620 (H323) • 1616 (H323) • 4621 (H323) • Avaya Digital Telephones (2420) • Avaya Analog (2500) 	<ul style="list-style-type: none"> • 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** reachable 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 String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call 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 it is required to have the SM100 IP interface 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									
ON	Type	Slot	Code/Sfx	Node Name/ IP-Address	Mask	Gateway	Node	Net Rgn	VLAN
y	C-LAN	01A02	TN799 D	clan 193.120.221.132	/25	Gateway001		1	n
y	MEDPRO	01A03	TN2602	mpro 193.120.221.133	/25	Gateway001		1	n

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					
FAX	Mode	Redundancy			
Modem	t.38-standard	0			
TDD/TTY	off	0			
Clear-channel	US	3			
	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		
Codec Set: 1	Intra-region IP-IP Direct Audio: yes	
	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 Manager, 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		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? n	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Direct IP-IP Early 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			
UII 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 Domain:	Far-end Network Region: 1	
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? n	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Direct IP-IP Early 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 simultaneous 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	Night Service:	
Dial Access? n			
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			
UII 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 String	Total Min Max		Route Pattern	Call Type	Node Num	ANI Req'd				
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 String	Total Min Max		Route Pattern	Call Type	Node Num	ANI 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™ Communication Manager Feature Server

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

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

5.1. Verify Avaya Aura™ Communication Manager License

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

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

display system-parameters customer-options		Page	2 of 10
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
Maximum Administered H.323 Trunks:	100	0	
Maximum Concurrently Registered IP Stations:	450	0	
Maximum Administered Remote Office Trunks:	0	0	
Maximum Concurrently Registered Remote Office Stations:	0	0	
Maximum Concurrently Registered IP eCons:	0	0	
Max Concur Registered Unauthenticated H.323 Stations:	100	0	
Maximum Video Capable Stations:	100	0	
Maximum Video Capable IP Softphones:	100	0	
Maximum Administered SIP Trunks:	100	50	

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 Manager, 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		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? n	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Direct IP-IP Early 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	Night Service:	
Dial Access? n			
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			
		UI 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:		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
	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 simultaneous 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	Night Service:	
Dial Access? n			
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			
UII 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									
DIAL PLAN ANALYSIS TABLE									
Location: all									
Percent Full: 2									
	Dialed String	Total Length	Call Type		Dialed String	Total Length	Call Type	Dialed String	Total Length
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									
AAR DIGIT ANALYSIS TABLE									
Location: all									
Percent Full: 2									
	Dialed String	Total		Route	Call	Node	ANI		
		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 String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd	
	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.

The screenshot displays the Avaya Aura™ System Manager 5.2 web interface. The top header shows the Avaya logo, the system name, and user information. The left sidebar contains a navigation menu with 'Network Routing Policy' highlighted. The main content area provides an introduction to the NRP workflow, listing steps from creating domains to regular expressions, and includes a 'Dial Pattern driven approach to define routing policies' section.

AVAYA Avaya Aura™ System Manager 5.2 Welcome, admin Last logged on at Mar. 30, 2010 12:25 AM Help | Log off

Home / Network Routing Policy

Introduction to Network Routing Policy (NRP)

Network Routing Policy consists of several NRP applications like "Domains", "Locations", "SIP Entities", etc.

The recommended order to use the NRP applications (that means the overall NRP workflow) to configure your network configuration is as follows:

- Step 1: Create "Domains" of type SIP (other NRP applications are referring domains of type SIP).
- Step 2: Create "Locations"
- Step 3: Create "Adaptations"
- Step 4: Create "SIP Entities"
 - SIP Entities that are used as "Outbound Proxies" e.g. a certain "Gateway" or "SIP Trunk"
 - Create all "other SIP Entities" (Session Manager, CM, SIP/PSTN Gateways, SIP Trunks)
 - Assign the appropriate "Locations", "Adaptations" and "Outbound Proxies"
- Step 5: Create the "Entity Links"
 - Between Session Managers
 - Between Session Managers and "other SIP Entities"
- Step 6: Create "Time Ranges"
 - Align with the tariff information received from the Service Providers
- Step 7: Create "Routing Policies"
 - Assign the appropriate "Routing Destination" and "Time Of Day"
 - (Time Of Day = assign the appropriate "Time Range" and define the "Ranking")
- Step 8: Create "Dial Pattern"
 - Assign the appropriate "Locations" and "Routing Policies" to the "Dial Pattern"
- Step 9: Create "Regular Expressions"
 - Assign the appropriate "Routing Policies" to the "Regular Expressions"

Each "Routing Policy" defines the "Routing Destination" (which is a "SIP Entity") as well as the "Time of Day" and its associated "Ranking".

IMPORTANT: the appropriate dial patterns are defined and assigned afterwards with the help of NRP application "Dial pattern". That's why this overall NRP workflow can be interpreted as

"Dial Pattern driven approach to define routing policies"

That means (with regard to steps listed above):

- Step 7: "Routing Policies" are defined
- Step 8: "Dial Pattern" are defined and assigned to "Routing Policies" and "Locations" (one step)
- Step 9: "Regular Expressions" are defined and assigned to "Routing Policies" (one step)

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)

The screenshot shows the Avaya Aura System Manager 5.2 interface. The top header includes the Avaya logo, the product name 'Avaya Aura™ System Manager 5.2', and a welcome message for user 'admin' last logged on at Mar. 26, 2010 12:25 AM. A red navigation bar contains 'Home / Network Routing Policy / SIP Domains'. On the left, a sidebar lists various management categories, with 'SIP Domains' highlighted under 'Network Routing Policy'. The main content area is titled 'Domain Management' and features a table with one item. The table has columns for Name, Type, Default, and Notes. The 'Name' column contains 'avaya.com', the 'Type' column contains 'sip', and the 'Default' column has an unchecked checkbox. A red asterisk and the text '* Input Required' are visible below the table. At the bottom right of the main area, there are 'Commit' and 'Cancel' buttons.

Name	Type	Default	Notes
*avaya.com	sip	<input type="checkbox"/>	

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.

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 / Locations / Location Details

Location Details [Commit] [Cancel]

General

* Name: Enterprise

Notes:

Managed Bandwidth:

* Average Bandwidth per Call: 80 Kbit/sec

* Time to Live (secs): 3600

Location Pattern

Add Remove

2 Items | Refresh Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	208.209.43.*	
<input type="checkbox"/>	193.120.221.*	

Select : All, None (0 of 2 Selected)

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

Home / Network Routing Policy / Adaptations / Adaptation Details

Help | Log off

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

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
<input type="checkbox"/>	* * * *					destination	modifies To: on Inbound

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
<input type="checkbox"/>	* * * *					origination	modifies P-AI: on Outbound

Select : All, None (0 of 1 Selected)

Shortcuts

Change Password

Help for Adaptation Details fields

Help for Committing configuration changes

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™ 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.

The screenshot displays the Avaya Aura System Manager 5.2 web interface. The top navigation bar includes the Avaya logo, the product name 'Avaya Aura™ System Manager 5.2', and a user status message: 'Welcome, admin Last Logged on at Mar. 26, 2010 12:25 AM'. A 'Help | Log off' link is also present. Below the navigation bar, a breadcrumb trail reads 'Home / Network Routing Policy / SIP Entities / SIP Entity Details'. The left sidebar contains a tree view of the system's configuration areas, with 'Network Routing Policy' and 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' and features a 'General' tab. The form fields are as follows: 'Name' (text box with 'CM-AE'), 'FQDN or IP Address' (text box with '193.120.221.132'), 'Type' (dropdown menu with 'CM' selected), 'Notes' (text box), 'Adaptation' (dropdown menu), 'Location' (dropdown menu with 'Enterprise' selected), and 'Time Zone' (dropdown menu with 'Europe/Dublin' selected). There is an unchecked checkbox for 'Override Port & Transport with DNS SRV'. Below this, 'SIP Timer B/F (in seconds)' is set to '4', and 'Credential name' is an empty text box. 'Call Detail Recording' is set to 'none'. At the bottom, the 'SIP Link Monitoring' section has a dropdown menu set to 'Use Session Manager Configuration'. 'Commit' and 'Cancel' buttons are located at the top right of the form area.

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.

The screenshot displays the Avaya Aura System Manager 5.2 web interface. The top header includes the Avaya logo, the product name 'Avaya Aura™ System Manager 5.2', and a user status bar indicating 'Welcome, admin' and 'Last Logged on at Mar. 26, 2010 12:25 AM'. A navigation breadcrumb trail shows 'Home / Network Routing Policy / SIP Entities / SIP Entity Details'. On the left, a sidebar menu lists various management categories, with 'Network Routing Policy' and 'SIP Entities' highlighted. The main content area is titled 'SIP Entity Details' and contains a 'General' tab. The form fields are as follows: 'Name' is 'CM-FS', 'FQDN or IP Address' is '193.120.221.180', 'Type' is 'CM', 'Location' is 'Enterprise', and 'Time Zone' is 'Europe/Dublin'. There are also checkboxes for 'Override Port & Transport with DNS SRV' and 'SIP Timer B/F (in seconds)' set to 4. A 'Credential name' field and a 'Call Detail Recording' dropdown set to 'none' are also present. At the bottom, the 'SIP Link Monitoring' section is set to 'Use Session Manager Configuration'. 'Commit' and 'Cancel' buttons are located at the top right of the form area.

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.

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 / SIP Entities / SIP Entity Details

SIP Entity Details **Commit** **Cancel**

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:

Time Zone:

Override Port & Transport with DNS SRV: ☐

* SIP Timer B/F (in seconds):

Credential name:

Call Detail Recording:

SIP Link Monitoring

SIP Link Monitoring:

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.

AVAYA Avaya Aura™ System Manager 5.2 Welcome, **admin** Last Logged on at Apr. 08, 2010 4:35 AM [Help](#) | [Log off](#)

Home / Network Routing Policy / SIP Entities / SIP Entity Details

SIP Entity Details [Commit](#) [Cancel](#)

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Location:

Outbound Proxy:

Time Zone:

Credential name:

SIP Link Monitoring

SIP Link Monitoring:

Entity Links

Entity Links can be modified after SIP Entity is committed.

Port

[Add](#) [Remove](#)

2 Items | [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	UDP	avaya.com	<input type="text"/>
<input type="checkbox"/>	5061	TLS	avaya.com	<input type="text"/>

Select : All, None (0 of 2 Selected)

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** of 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 screenshot shows the Avaya Aura System Manager 5.2 interface. The left sidebar has a red box around 'Entity Links' under 'Network Routing Policy'. The main area shows a table with one row: Name: SM-CMAE, SIP Entity 1: SessionManager, Protocol: TLS, Port: 5061, SIP Entity 2: CM-AE, Port: 5061, Trusted: checked. There are 'Commit' and 'Cancel' buttons at the top right and bottom right.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* SM-CMAE	* SessionManager	TLS	* 5061	* CM-AE	* 5061	<input checked="" type="checkbox"/>	

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

The screenshot shows the Avaya Aura System Manager 5.2 interface. The left sidebar has a red box around 'Entity Links' under 'Network Routing Policy'. The main area shows a table with one row: Name: SM-CMFS, SIP Entity 1: SessionManager, Protocol: TLS, Port: 5061, SIP Entity 2: CM-FS, Port: 5061, Trusted: checked. There are 'Commit' and 'Cancel' buttons at the top right and bottom right.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
* SM-CMFS	* SessionManager	TLS	* 5061	* CM-FS	* 5061	<input checked="" type="checkbox"/>	

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

Avaya Aura™ System Manager 5.2

Welcome, admin Last Logged on at Apr. 14, 2010 10:00 AM

Home / Network Routing Policy / Entity Links

Entity Links

1 Item | Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
*SM-IMG1010	*SessionManager	UDP	*5060	*Gateway	*5060	<input checked="" type="checkbox"/>	

* Input Required

Commit Cancel

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

Avaya Aura™ System Manager 5.2

Welcome, admin Last Logged on at Apr. 14, 2010 10:00 AM

Home / Network Routing Policy / Entity Links

Entity Links

Edit New Duplicate Delete More Actions Commit

3 Items | Refresh Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
<input type="checkbox"/>	SM-CMAE	SessionManager	TCP	5060	CM-AE	5060	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	SM-CMFS	SessionManager	TLS	5061	CM-FS	5061	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	SM-IMG1010	SessionManager	UDP	5060	Gateway	5060	<input checked="" type="checkbox"/>	

Select : All, None (0 of 3 Selected)

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 screenshot displays the Avaya Aura System Manager 5.2 interface. The left sidebar shows the navigation menu with 'Network Routing Policy' selected. The main content area is titled 'Routing Policy Details' and contains three sections: 'General', 'SIP Entity as Destination', and 'Time of Day'. In the 'General' section, the 'Name' field is set to 'RP-to-CM-AE' and the 'Notes' field contains 'Routes to CM'. In the 'SIP Entity as Destination' section, the 'Select' button is highlighted, and a table lists the selected entity 'CM-AE' with its FQDN or IP address '193.120.221.132' and type 'CM'. In the 'Time of Day' section, the 'Add' button is highlighted, and a table shows the selected time range '24/7' with a start time of '00:00' and an end time of '23:59'. The 'Commit' button is visible in the top right corner.

Avaya Aura™ System Manager 5.2

Welcome, admin Last Logged on at Apr. 07, 2010 9:51 PM

Help | Log off

Home / Network Routing Policy / Routing Policies / Routing Policy Details

Routing Policy Details

Commit Cancel

General

* Name: RP-to-CM-AE

Disabled: ☐

Notes: Routes to CM

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CM-AE	193.120.221.132	CM	

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Always Active

Select : All, None (0 of 1 Selected)

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

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 / Routing Policies / Routing Policy 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

Routing Policy Details

Commit
Cancel

General

* Name:
Disabled: ☐
Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Gateway	208.209.43.59	Gateway	

Time of Day

Add
Remove
View Gaps/Overlaps

1 Item | Refresh
Filter: Enable

	Ranking 1 ▲	Name 2 ▲	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Always Active

Select : All, None (0 of 1 Selected)

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.

AVAYA Avaya Aura™ System Manager 5.2 Welcome, admin Last Logged on at Apr. 07, 2010 9:51 PM [Help](#) | [Log off](#)

Home / Network Routing Policy / Dial Patterns / **Dial Pattern Details**

Dial Pattern Details [Commit](#) [Cancel](#)

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call: ☐

SIP Domain:

Notes:

Originating Locations and Routing Policies

[Add](#) [Remove](#)

1 Item | [Refresh](#) Filter: Enable

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	RP-to-CM-AE	0	<input type="checkbox"/>	CM-AE	Routes to CM

Select : All, None (0 of 1 Selected)

Denied Originating Locations

[Add](#) [Remove](#)

0 Items | [Refresh](#) Filter: Enable

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

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.

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 / Dial Patterns / Dial Pattern Details

Dial Pattern Details Commit Cancel

General

* Pattern:

* Min:

* Max:

Emergency Call: ☐

SIP Domain:

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item | Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	RP-to-Gateway	0	<input type="checkbox"/>	Gateway	

Select : All, None (0 of 1 Selected)

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**

The screenshot displays the Avaya Aura™ System Manager 5.2 web interface. The top header includes the Avaya logo, the product name, and a welcome message for user 'admin' with the last login time. A red navigation bar shows the path: Home / Applications / Application Management / Applications Details. On the left, a sidebar menu lists various management categories, with 'Applications' and 'Entities' highlighted. The main content area is titled 'New CM Instance' and contains a form with the following fields: 'Application' (a dropdown menu), 'Name' (text input with 'CM-featureServer'), 'Type' (dropdown menu with 'CM'), 'Description' (a large text area), and 'Node' (dropdown menu with '193.120.221.180'). 'Commit' and 'Cancel' buttons are located at the top right of the form area.

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.

The screenshot shows a web form titled 'Attributes' with a dropdown menu. The form contains several input fields and checkboxes. The following fields are highlighted with red circles: the 'Attributes' tab, the 'Login' field (containing 'init'), the 'Password' field (containing '*****'), the 'Confirm Password' field (containing '*****'), and the 'Commit' button. The form also includes a checkbox for 'Is SSH Connection' (checked), a required field for 'Port' (containing '5022'), and several other fields: 'Alternate IP Address', 'RSA SSH Fingerprint (Primary IP)', 'RSA SSH Fingerprint (Alternate IP)', 'Is ASG Enabled' (unchecked), 'ASG Key', 'Confirm ASG Key', and 'Location'. A legend at the bottom left indicates that an asterisk (*) denotes a required field.

Field	Value
Login	init
Password	*****
Confirm Password	*****
Is SSH Connection	<input checked="" type="checkbox"/>
* Port	5022
Alternate IP Address	
RSA SSH Fingerprint (Primary IP)	
RSA SSH Fingerprint (Alternate IP)	
Is ASG Enabled	<input type="checkbox"/>
ASG Key	
Confirm ASG Key	
Location	

*Required

Commit Cancel

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.

AVAYA Avaya Aura™ System Manager 5.2 Welcome, **admin** Last Logged on at Apr. 09, 2010 4:38 AM [Help](#) [Log off](#)

Home / Session Manager / Application Configuration / Application Editor

Application Editor **Commit** Cancel

Application Editor

* **Name**

* **SIP Entity**

Description

Application Attributes (optional)

Name	Value
Application Handle	<input type="text"/>
URI Parameters	<input type="text"/>

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.

Avaya Aura™ System Manager 5.2

Welcome, admin Last Logged on at Apr. 09, 2010 4:38 AM

Help Log off

Home / Session Manager / Application Configuration / Application Sequence Editor

Application Sequence Editor Commit Cancel

Sequence Name

* Name

Description

Applications in this Sequence

Move First Move Last Remove

1 Item

<input type="checkbox"/>	Sequence Order (first to last)	Name	SIP Entity	Mandatory	Description
<input type="checkbox"/>	*	App-FeatureServer	CM-FS	<input checked="" type="checkbox"/>	

Select : All, None (0 of 1 Selected)

Available Applications

1 Item Refresh Filter: Enable

	Name	SIP Entity	Description
<input checked="" type="checkbox"/>	App-FeatureServer	CM-FS	

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.

AVAYA Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Apr. 09, 2010 6:17 AM

Home / Communication System Management / Telephony

Synchronize CM Data and Configure Options

Synchronize CM Data/Launch Element Cut Through | Configuration Options | Expand All | Collapse All

Synchronize CM Data/Launch Element Cut Through

1 Item	Refresh	Filter: Enable					
<input checked="" type="checkbox"/>	Element Name	FQDN/IP Address	Last Sync Time	Sync Type	Sync Status	Location	Software Version
<input checked="" type="checkbox"/>	CM-featureServer	193.120.221.180	April 9, 2010 4:00:21 AM +01:00	Incremental	Completed		R015x.02.1.016.4

Select: All, None (1 of 1 Selected)

☒ Initialize data for selected devices
☐ Incremental Sync data for selected devices

Now Schedule Cancel Launch Element Cut Through

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**.

AVAYA Avaya Aura™ System Manager 5.2

Welcome, **admin** Last Logged on at Apr. 09, 2010 6:17 AM

Home / User Management / User Management / New User

New User Profile

General | Identity | Communication Profile | Roles | Override Permissions | Group Membership | Attribute Sets | Default Contact List | Private Contacts | Expand All | Collapse All

General

* Last Name: Joe

* First Name: Bloggs

Middle Name:

Description:

User Type:

- ☐ administrator
- ☐ communication_user
- ☐ agent
- ☐ supervisor
- ☐ resident_expert
- ☐ service_technician
- ☐ lobby_phone

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

Identity ▼

* Login Name: 3500

* Authentication Type: Basic ▼

SMGR Login Password:

* Password: *****

* Confirm Password: *****

Shared Communication Profile Password: *****

Confirm Password: *****

Localized Display Name:

Endpoint Display Name:

Honoric :

Language Preference: ▼

Time Zone: ▼

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**.

Communication Profile

New Delete Done Cancel

Name
Primary

Select : None

* Name: Primary

Default : ☒

Communication Address

New Edit Delete

Type	SubType	Handle	Domain
No Records found			

Type: sip

SubType: username

* Fully Qualified Address: 3500 @ avaya.com

Add Cancel

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**

The screenshot displays a configuration interface with two main sections: **Session Manager** and **Station Profile**. Both sections are expanded, indicated by checkmarks and dropdown arrows. In the **Session Manager** section, the **Session Manager Instance** is set to **SessionManager**, the **Origination Application Sequence** is **AppSeq-FeatureServer**, and the **Termination Application Sequence** is **AppSeq-FeatureServer**. In the **Station Profile** section, the **System** is **CM-featureServer**, **Use Existing Stations** is unchecked, the **Extension** is **3500**, the **Template** is **DEFAULT_9630SIP**, the **Set Type** is **9630SIP**, the **Security Code** is empty, the **Port** is **IP**, and **Delete Station on Unassign of Station from User** is unchecked.

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.

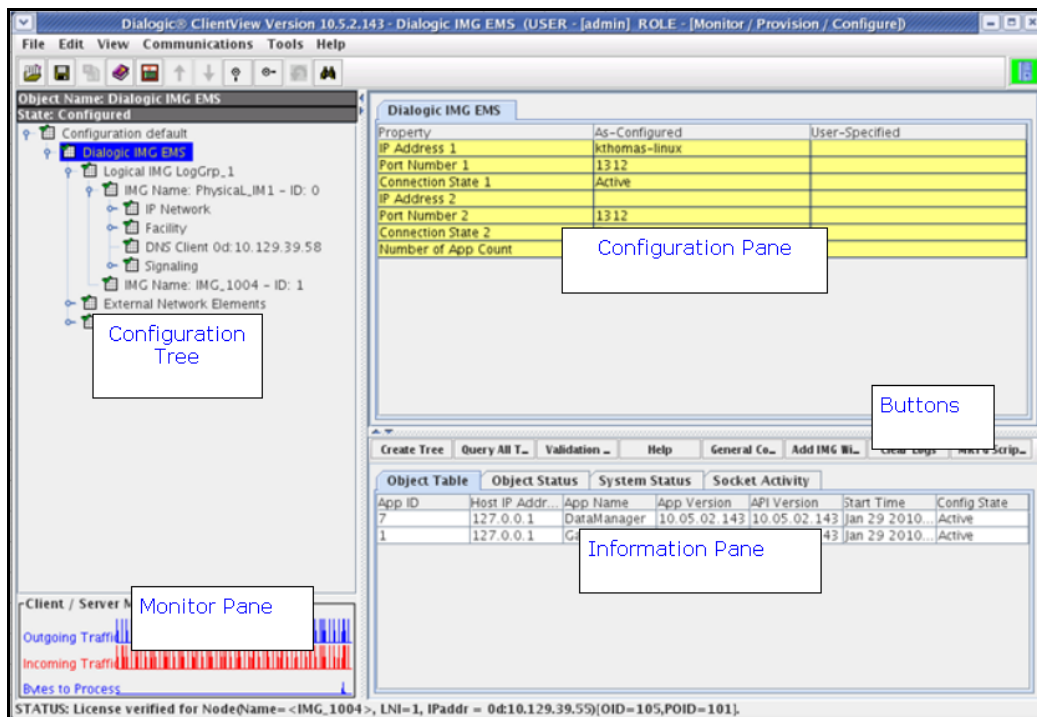
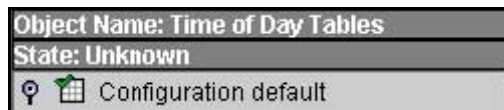


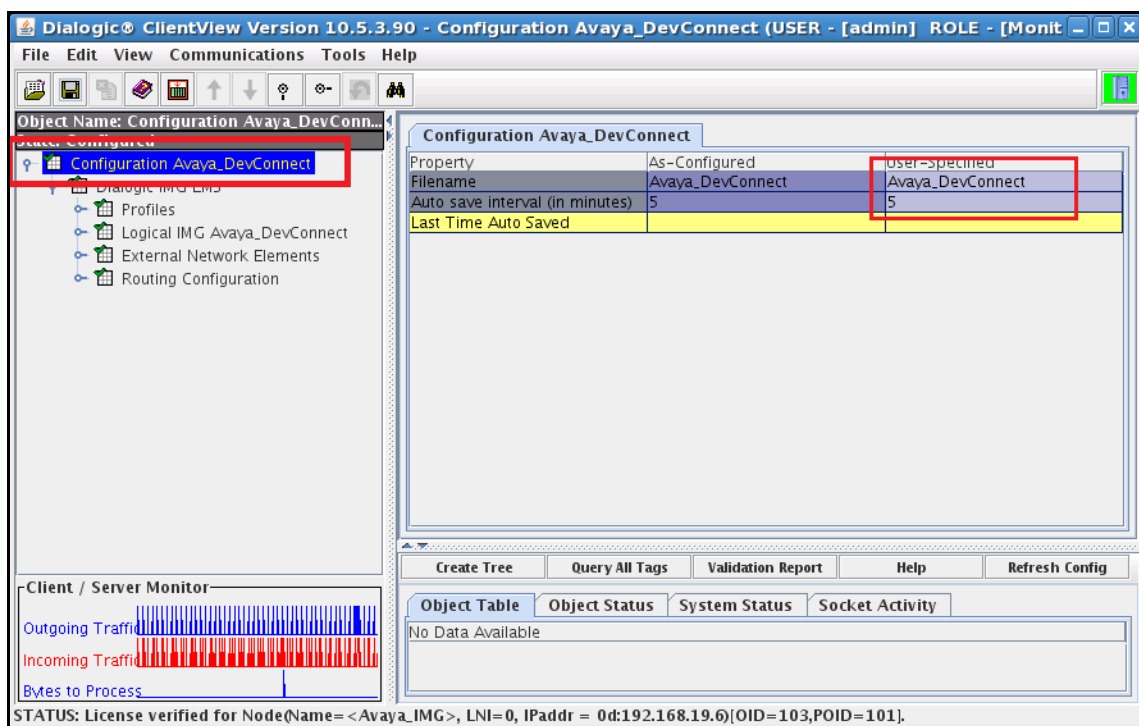
Figure 2 - Dialogic Inc. ClientView Main Window

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.

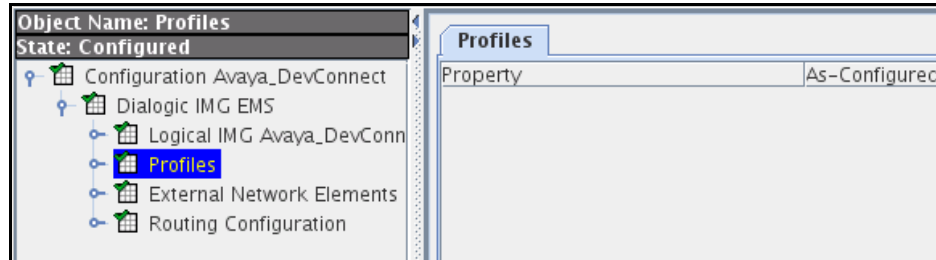


Enter a descriptive name in the **Filename** field in the Configuration Pane. To save the changes, right-click **Configuration Avaya_DevConnect**, and select **Commit**. The picture below shows the actions performed on IMG 1010



7.2. Profiles

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

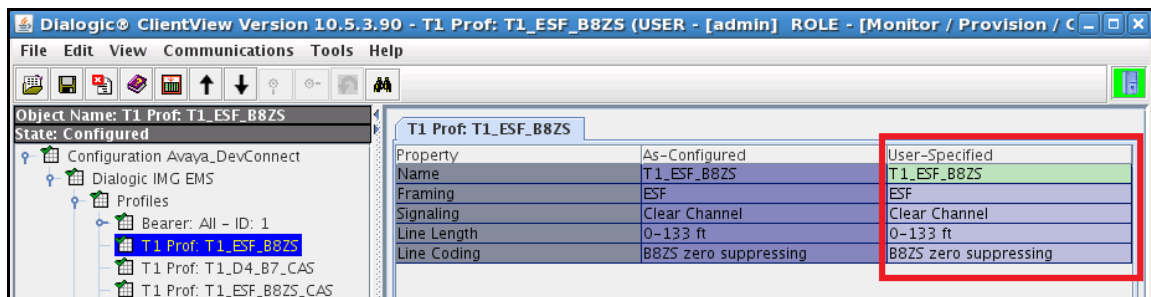


7.2.1. DS1 Profiles

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

Dialogic ClientView Version 10.5.3.90 - Bearer: All - ID: 1 (USER - [admin] ROLE - [Monitor / Provision])

File Edit View Communications Tools Help

Object Name: Bearer: All - ID: 1
State: Configured

Configuration Tree:

- Configuration Avaya_DevConnect
 - Dialogic IMG EMS
 - Profiles
 - Bearer: All - ID: 1**
 - Profile: 1 - Entry: 0
 - Profile: 1 - Entry: 4
 - Profile: 1 - Entry: 6
 - Profile: 1 - Entry: 1
 - Profile: 1 - Entry: 2
 - Profile: 1 - Entry: 3
 - Profile: 1 - Entry: 5
 - T1 Prof: T1_ESF_B8ZS
 - T1 Prof: T1_D4_B7_CAS
 - T1 Prof: T1_ESF_B8ZS_CAS
 - SIP: default - ID: 0
 - SIP: SIP_Q_850 - ID: 6
 - SIP: Avaya_SM - ID: 1
 - Logical IMG Avaya_DevConnect
 - External Network Elements
 - Routing Configuration

Client / Server Monitor

Outgoing Traffic
Incoming Traffic
Bytes to Process

STATUS: License verified for Node(Name= <Avaya_IMG>, LNI=0, IPAddr = 0d:192.168.19.6)[OID=103,POID=101].

Bearer: All - ID: 1

Property	As-Configured	User-Specified
IP Bearer Profile Id	1	1
IP Bearer Profile Name	All	All
Silence Suppression	Disable	Disable
Echo Cancellation	Enable	Enable
RTP Redundancy	No Redundancy	No Redundancy
RTP Payload Type for Redundancy	Not Used	Not Used
Fax Mode	Enable Relay (T.38)	Enable Relay (T.38)
Fax Bypass Codec	G711 alaw	G711 alaw
Fax Packet Redundancy	No Redundancy	No Redundancy
Digit Relay	DTMF Packetized	DTMF Packetized
Digit Relay Packet Type	101	101
Modem Behavior	Bypass	Bypass
H245 Outbound Tunneling	Enable	Enable
Initial Media Inactivity Timer	Disable	Disable
Media Inactivity Timer	Disable	Disable
Comedia Mode	Disable	Disable
Source Port Validate	Disable	Disable

Create Tree Query All Tags Validation Report Help

Entry ID	Payload Ty...	Preferred ...	Minimum P...	Maximum ...	Default Pay...	Annex B Su...
0	G729	20	10	60	Not Used	Yes
4	G711 alaw	20	10	30	Not Used	Not Used
6	G711 ulaw	20	10	30	Not Used	Not Used
1	G723 6.3 ...	30	30	60	Not Used	Not Used
2	iLBC 20ms	20	20	20	96	Not Used
3	GSM-FR St...	20	20	60	Not Used	Not Used
5	G729E	20	10	30	108	Yes

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.

Dialogic ClientView Version 10.5.3.90 - Profile: 1 - Entry:0 (USER - [admin] ROLE - [Monitor / Provisioning])

File Edit View Communications Tools Help

Object Name: Profile: 1 - Entry:0
State: Configured

Configuration Avaya_DevConnect
Dialogic IMG EMS
Profiles
Bearer: All - ID: 1
Profile: 1 - Entry:0
Profile: 1 - Entry:4
Profile: 1 - Entry:6
Profile: 1 - Entry:1
Profile: 1 - Entry:2
Profile: 1 - Entry:3
Profile: 1 - Entry:5
T1 Prof: T1_ESF_B8ZS
T1 Prof: T1_D4_B7_CAS
T1 Prof: T1_ESF_B8ZS_CAS
SIP: default - ID: 0
SIP: SIP_Q.850 - ID: 6
SIP: Avaya_SM - ID: 1
Logical IMG Avaya_DevConnect
External Network Elements
Routing Configuration

Client / Server Monitor
Outgoing Traffic
Incoming Traffic
Bytes to Process

Profile: 1 - Entry:0

Property	As-Configured	User-Specified
Entry ID	0	0
Payload Type	G729	G729
Preferred Payload Size (ms)	20	20
Minimum Payload Size (ms)	10	10
Maximum Payload Size (ms)	60	60
Default Payload Type	Not Used	Not Used
Annex B Support	Yes	Yes

Create Tree Query All Tags Validation Report Help

Object Table	Object Status	System Status	Socket Activity
Entry ID	Payload Ty...	Preferred ...	Minimum P...
0	G729	20	10
4	G711 alaw	20	10
6	G711 ulaw	20	10
1	G723 6.3 ...	30	30
2	iLBC 20ms	20	20
3	CSM-FR St...	20	60
5	G729E	20	30

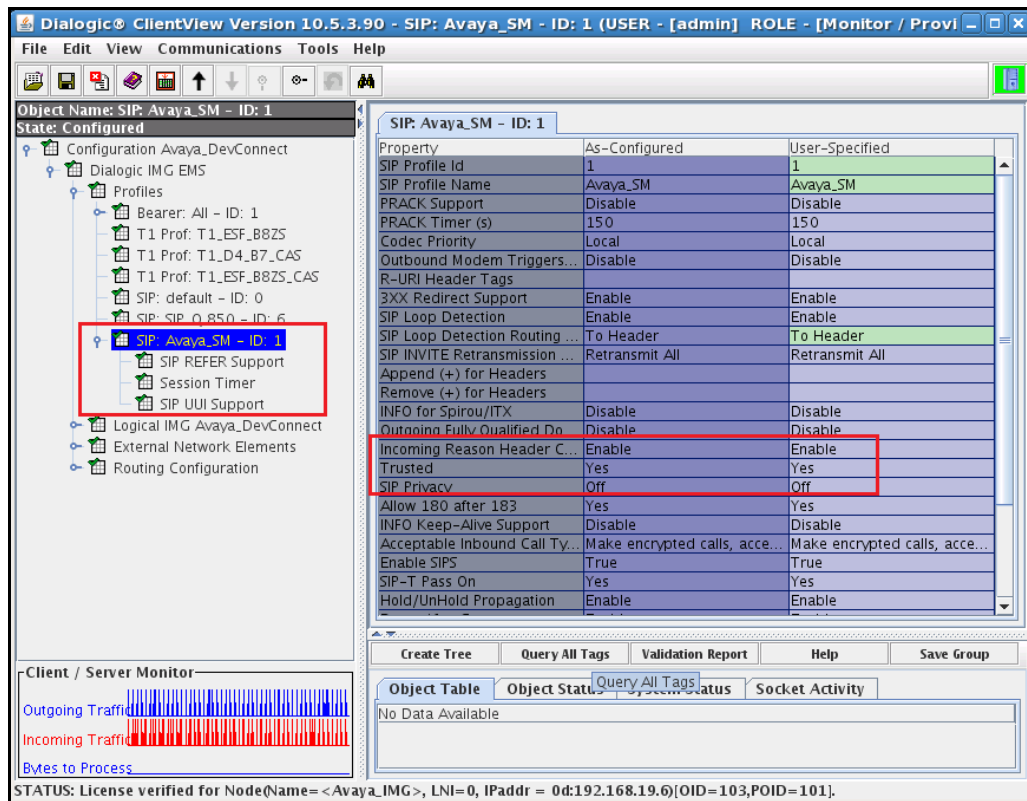
STATUS: License verified for Node(Name=<Avaya_IMG>, LNI=0, IPAddr = 0d:192.168.19.6)[OID=103,POID=101].

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.

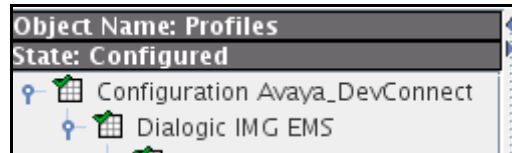
Note: if needed P-AI provide from Session Manager, set **SIP Privacy** to **On**



7.2.4. Logical IMG

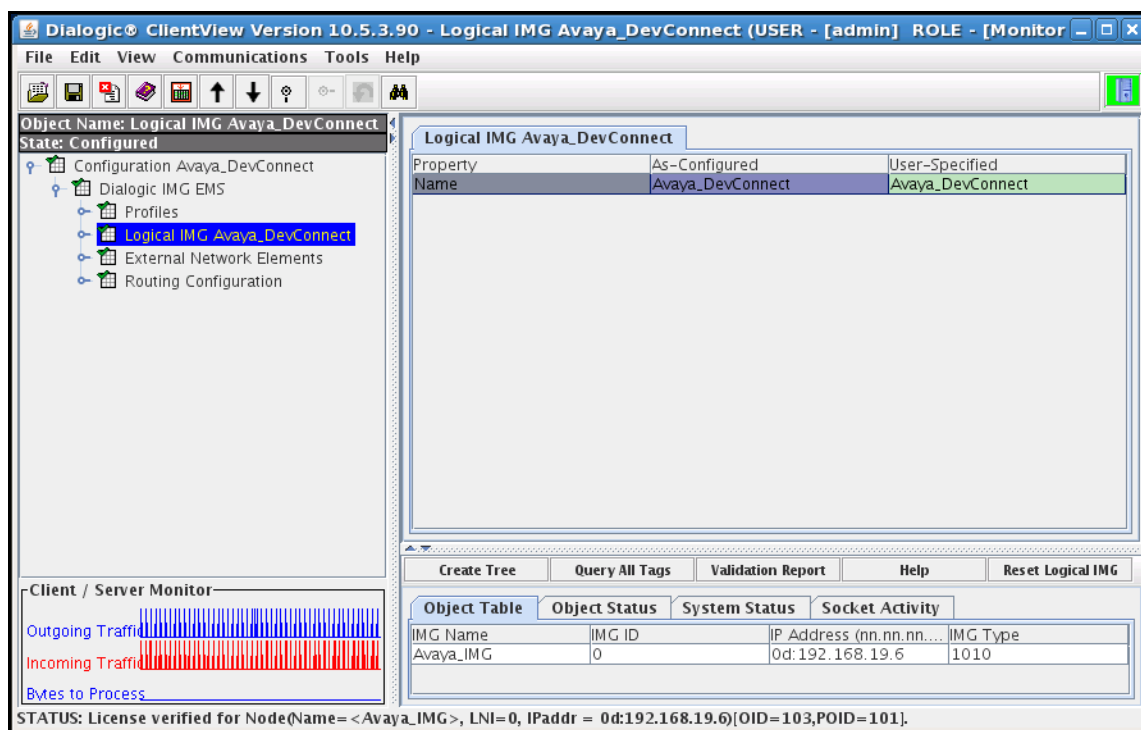
Create a logical IMG as follows:

- 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**.

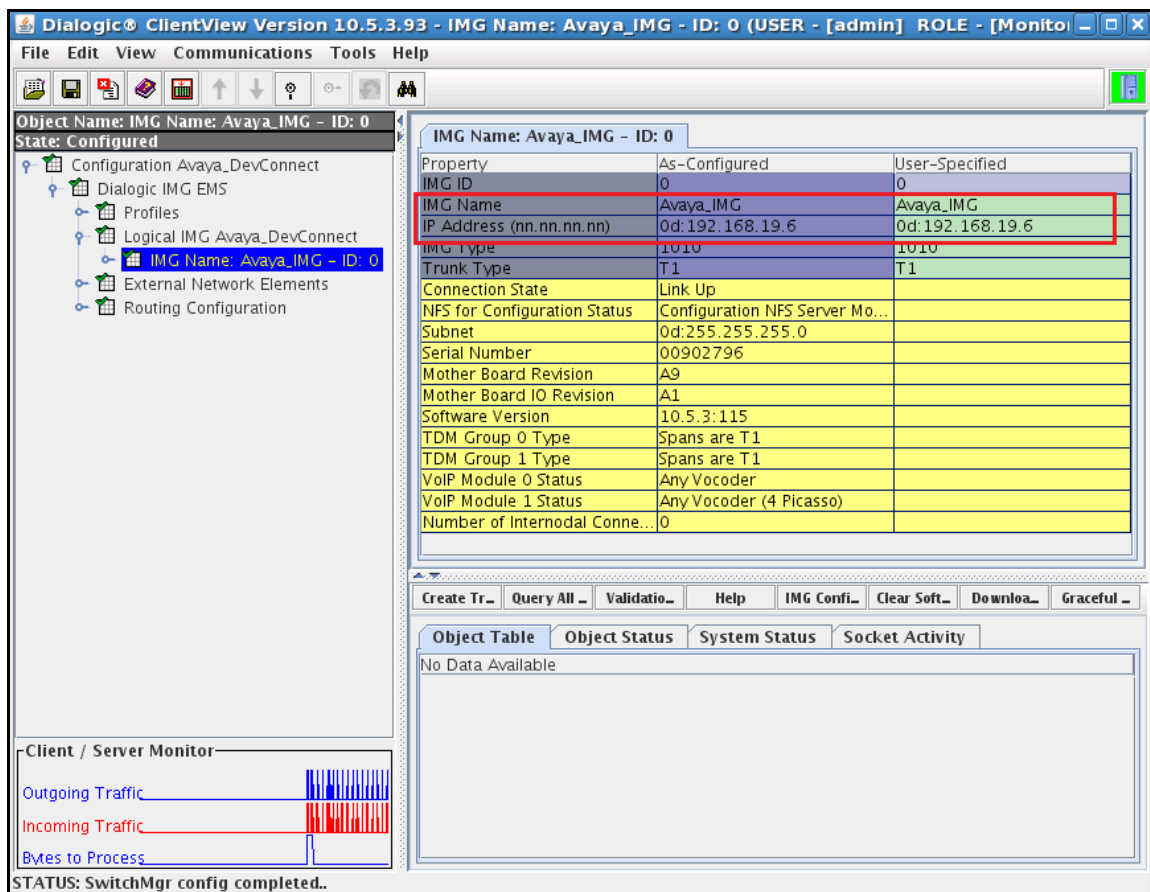


7.2.4.1 Physical IMG

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**.

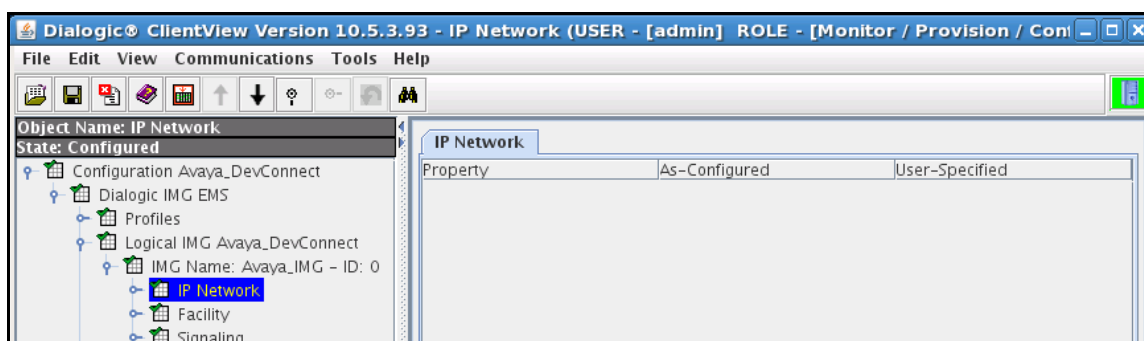


7.2.4.1.1 Network Interfaces

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.



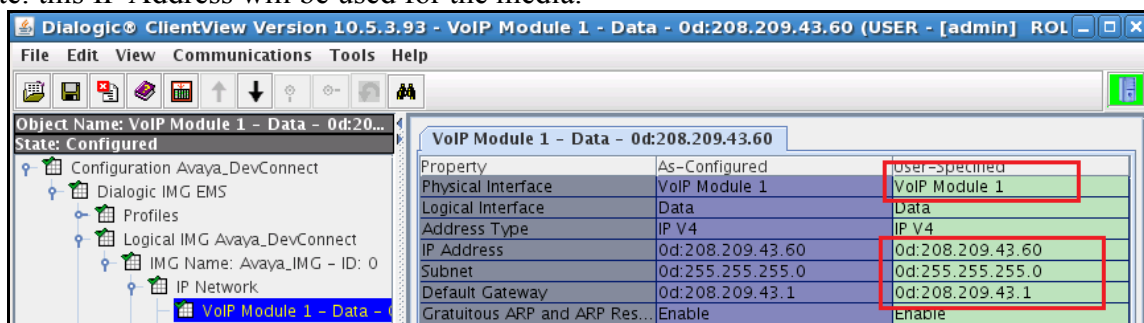
7.2.4.1.2 Network Interface VoIP module

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.



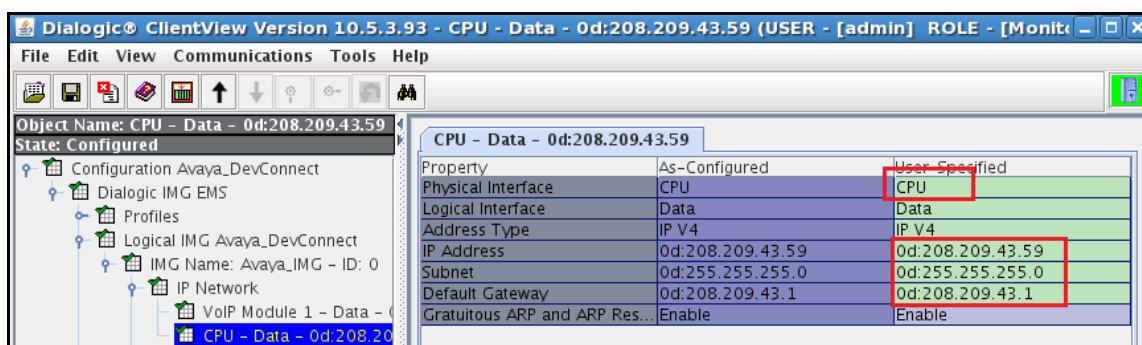
7.2.4.1.3 Network Interface CPU

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.

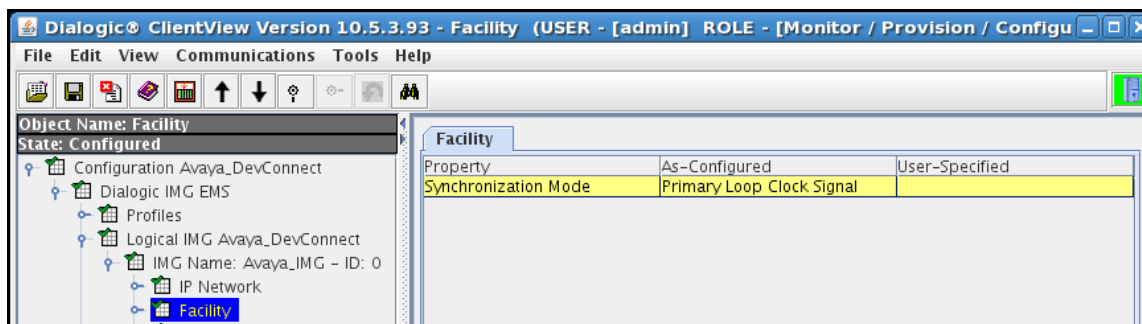


7.2.4.1.4 Facilities (DS1 and VoIP)

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**.

VoIP Resource 1

Property	As-Configured	User-Specified
Module ID	1	1
Network Interface	VoIP Module 1	VoIP Module 1
Network IP Address	0d:208.209.43.60	0d:208.209.43.60
Module Configuration Profile	iLBC Profile (4 Picasso)	iLBC Profile (4 Picasso)
Starting RTP Port	11072	11072
Fully Qualified Domain Nam...		
Number of Channels Config...	512	

Client / Server Monitor

Outgoing Traffic
Incoming Traffic
Bytes to Process

STATUS: SwitchMgr config completed..

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**.

The screenshot displays the Dialogic ClientView interface for configuring Bearer Spans. The left pane shows the Configuration Tree with the following structure:

- Configuration Avaya_DevConnect
 - Dialogic IMG EMS
 - Profiles
 - Logical IMG Avaya_DevConnect
 - IMG Name: Avaya_IMG - ID: 0
 - IP Network
 - Facility
 - VoIP Resource 1
 - Bearer Spans** (selected)
 - Signaling
 - Time Zone Setting UTC - 4:00
 - DNS Client 0d:208.209.43.5

The main pane shows the Bearer Spans configuration table:

Property	As-Configured	User-Specified
Interface	Bearer	Bearer
Offset - 0 Configuration	T1_ESF_B8ZS	T1_ESF_B8ZS
Offset - 1 Configuration	T1_ESF_B8ZS	T1_ESF_B8ZS
Offset - 2 Configuration	T1_ESF_B8ZS	T1_ESF_B8ZS
Offset - 3 Configuration	T1_ESF_B8ZS	T1_ESF_B8ZS
Offset - 4 Configuration	Not Used	Not Used
Offset - 5 Configuration	Not Used	Not Used
Offset - 6 Configuration	Not Used	Not Used
Offset - 7 Configuration	Not Used	Not Used
Offset - 8 Configuration	T1_ESF_B8ZS	T1_ESF_B8ZS
Offset - 9 Configuration	T1_ESF_B8ZS	T1_ESF_B8ZS
Offset - 10 Configuration	T1_ESF_B8ZS	T1_ESF_B8ZS
Offset - 11 Configuration	T1_ESF_B8ZS	T1_ESF_B8ZS
Offset - 12 Configuration	T1_ESF_B8ZS	T1_ESF_B8ZS
Offset - 13 Configuration	T1_ESF_B8ZS	T1_ESF_B8ZS
Offset - 14 Configuration	Not Used	Not Used
Offset - 15 Configuration	Not Used	Not Used

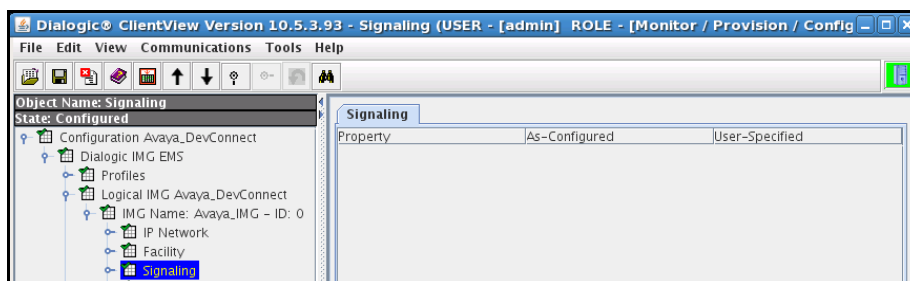
Below the table is the Object Table with the following data:

Object Table	Object Status	System Status	Socket Activity
Comment	DS1 Loopba...	Interface	Interfa...
ISDN PBX	No Loopback	Bearer	0 In Service
SS7 PBX	No Loopback	Bearer	1 In Service
	No Loopback	Bearer	2 In Service
	No Loopback	Bearer	3 Receiving Red, Rec...
	No Loopback	Bearer	4 Not Used
	No Loopback	Bearer	5 Not Used
	No Loopback	Bearer	6 Not Used
	No Loopback	Bearer	7 Not Used
	No Loopback	Bearer	8 Receiving Red, Rec...
	No Loopback	Bearer	9 In Service

The bottom pane shows the Client / Server Monitor with a graph of Outgoing Traffic, Incoming Traffic, and Bytes to Process. The status bar indicates: STATUS: SwitchMgr config completed..

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.

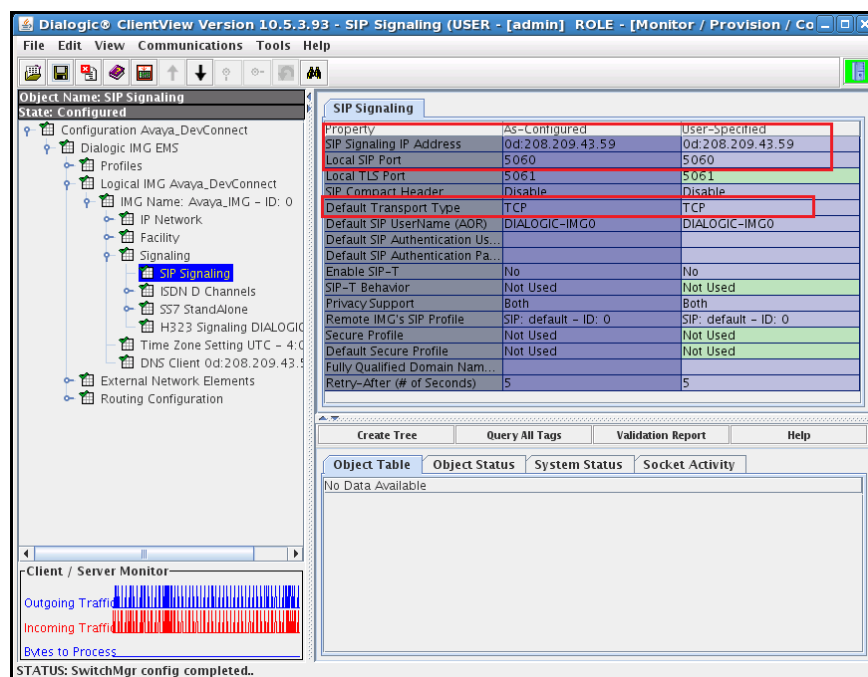


7.2.4.1.8 SIP Signaling

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:
 - Enter the IP address assigned to the IMG in the **SIP Signaling IP Address** field.
 - Enter values in the **Local SIP Port** and **Default Transport**.

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

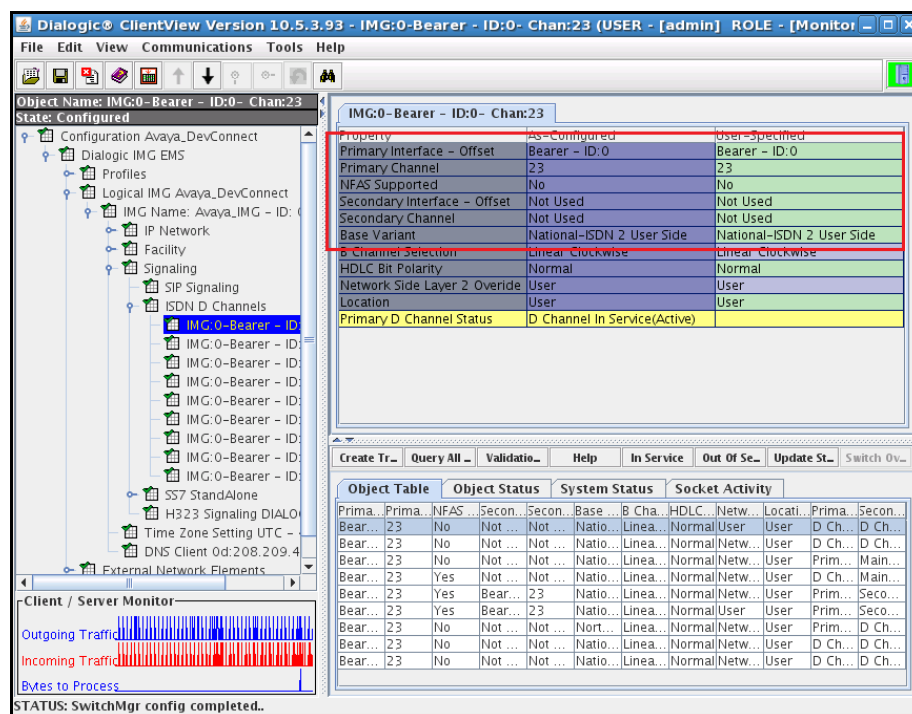


Create an object for ISDN as follows with right-click **Signaling** in the Configuration Tree, and select **New ISDN**. To save the changes, right-click **ISDN D Channels**, and select **Commit**.



- 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**.

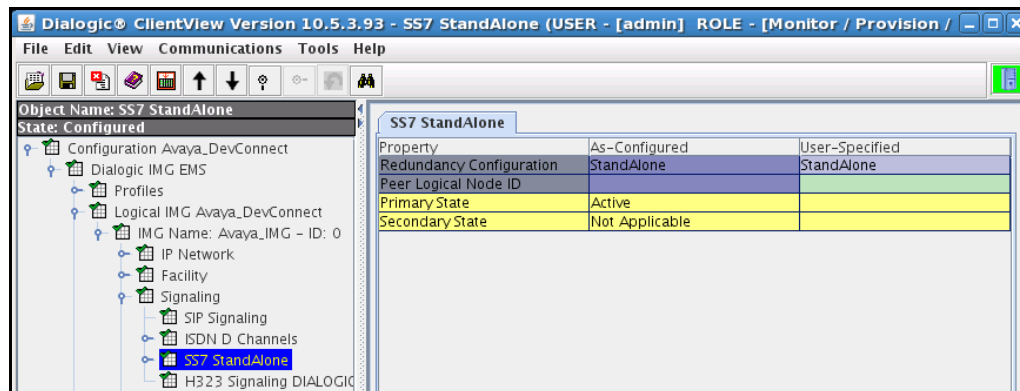


7.2.4.1.10 SS7 Signaling

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**.

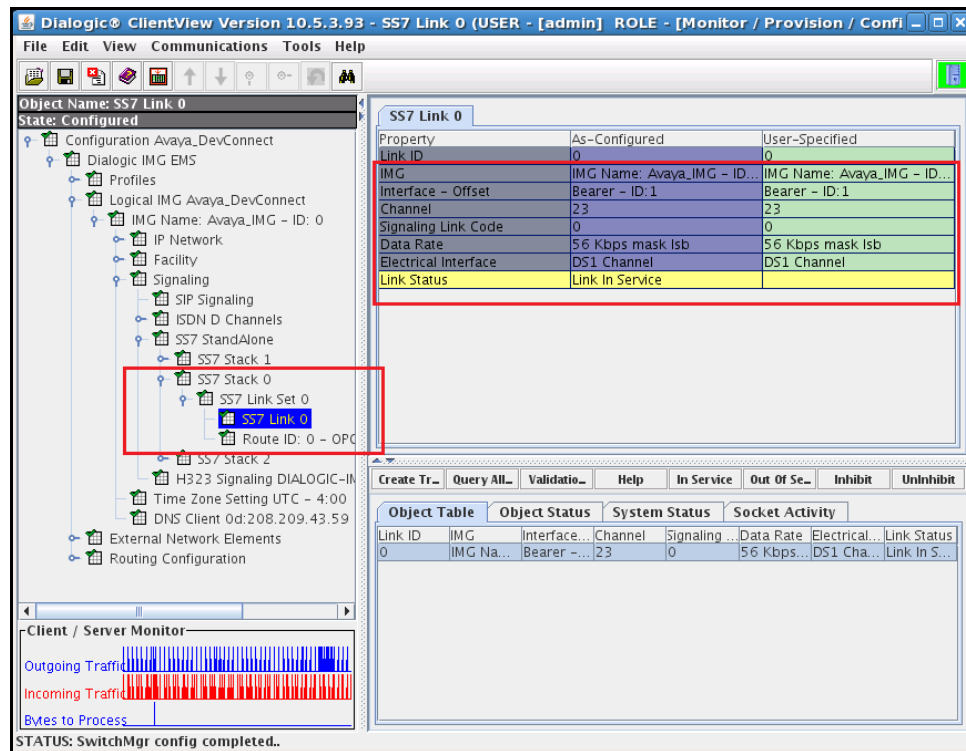


Configure SS7 Stack, Linkset, Links, & Routes as follows:

- 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**.



Dialogic ClientView Version 10.5.3.93 - SS7 Link 0 (USER - [admin] ROLE - [Monitor / Provision / Confi]

File Edit View Communications Tools Help

Object Name: SS7 Link 0
State: Configured

Configuration Avaya_DevConnect
Dialogic IMG EMS
Logical IMG Avaya_DevConnect
IMG Name: Avaya_IMG - ID: 0
IP Network
Facility
Signaling
SIP Signaling
ISDN D Channels
SS7 StandAlone
SS7 Stack 1
SS7 Stack 0
SS7 Link Set 0
SS7 Link 0
Route ID: 0 - OPC

SS7 Link 0

Property	As-Configured	User-Specified
Link ID	0	0
IMG	IMG Name: Avaya_IMG - ID: 0	IMG Name: Avaya_IMG - ID: 0
Interface - Offset	Bearer - ID: 1	Bearer - ID: 1
Channel	23	23
Signaling Link Code	0	0
Data Rate	56 Kbps mask Isb	56 Kbps mask Isb
Electrical Interface	DS1 Channel	DS1 Channel
Link Status	Link In Service	

Create Tr... Query All... Validation... Help In Service Out Of Se... Inhibit Uninhibit

Object Table	Object Status	System Status	Socket Activity
Link ID	IMG	Interface...	Channel
0	IMG Na...	Bearer -...	23
		Signaling ..	0
		Data Rate	56 Kbps...
		Electrical...	DS1 Cha...
		Link Status	Link In S...

Client / Server Monitor

Outgoing Traffic
Incoming Traffic
Bytes to Process

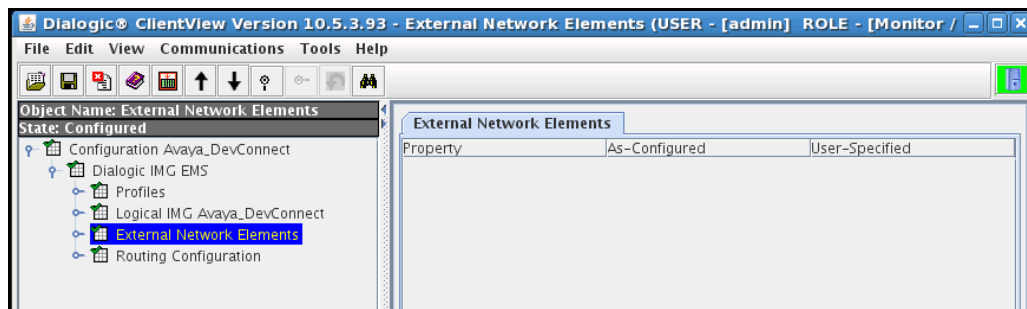
STATUS: SwitchMgr config completed..

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**.

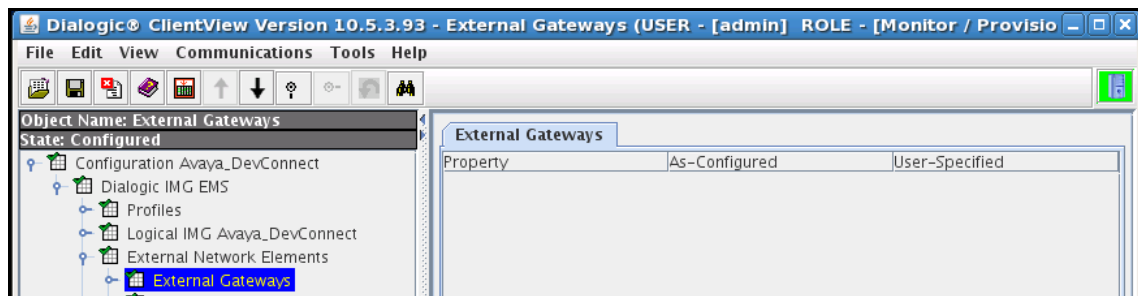


7.2.5.1 External Gateways

Create an object for External Gateways as follows:

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

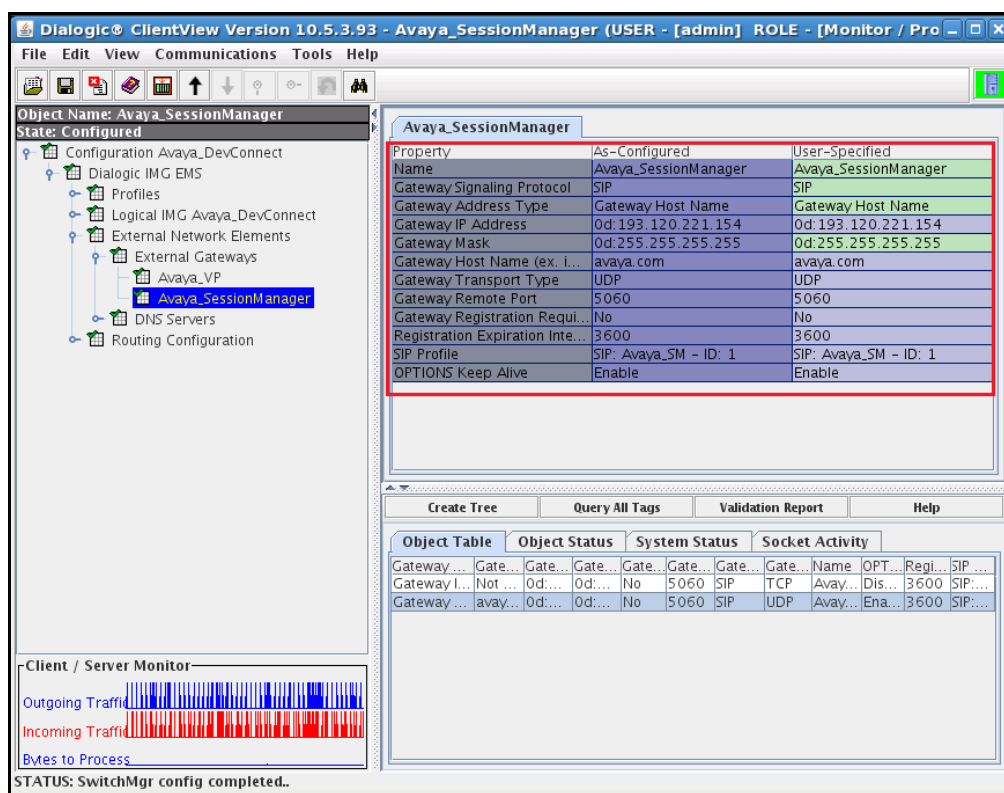
To save the changes, right-click **External Gateways**, and select **Commit**.



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**.

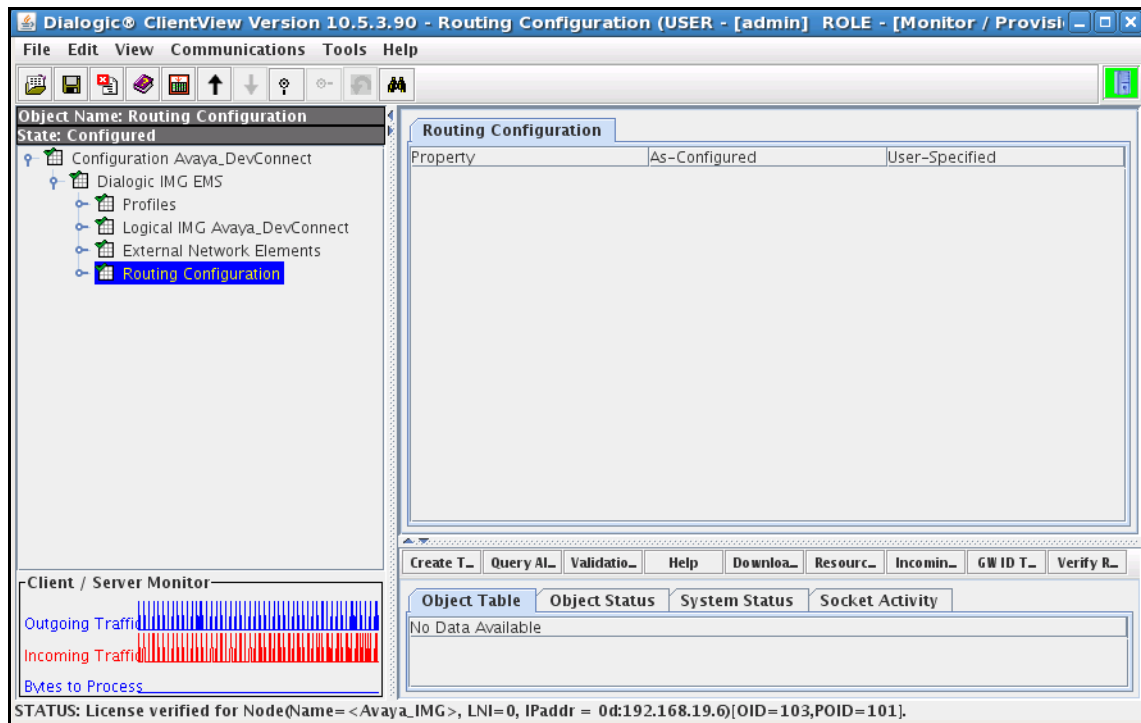


7.3. Routing configuration

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**.

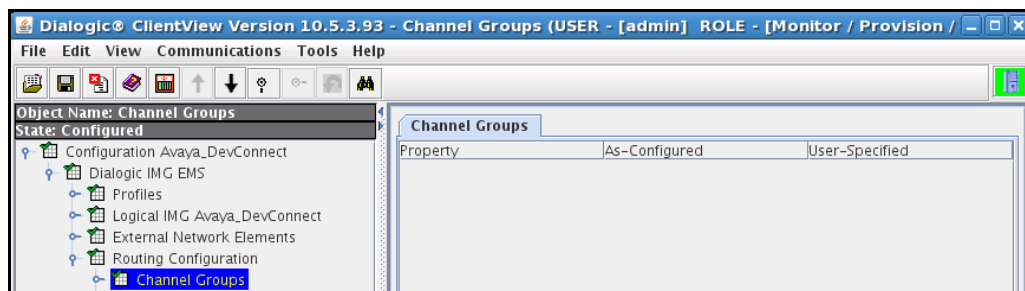


7.3.1. Channel Groups

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**.

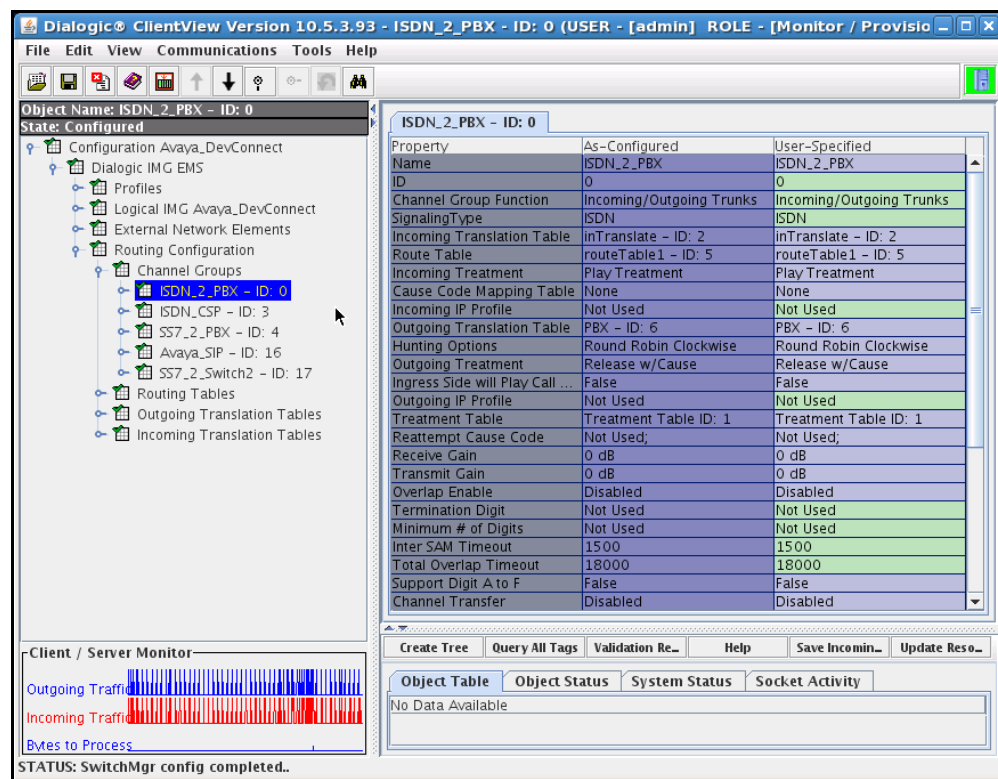


7.3.1.1 Channel Group (ISDN)

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.

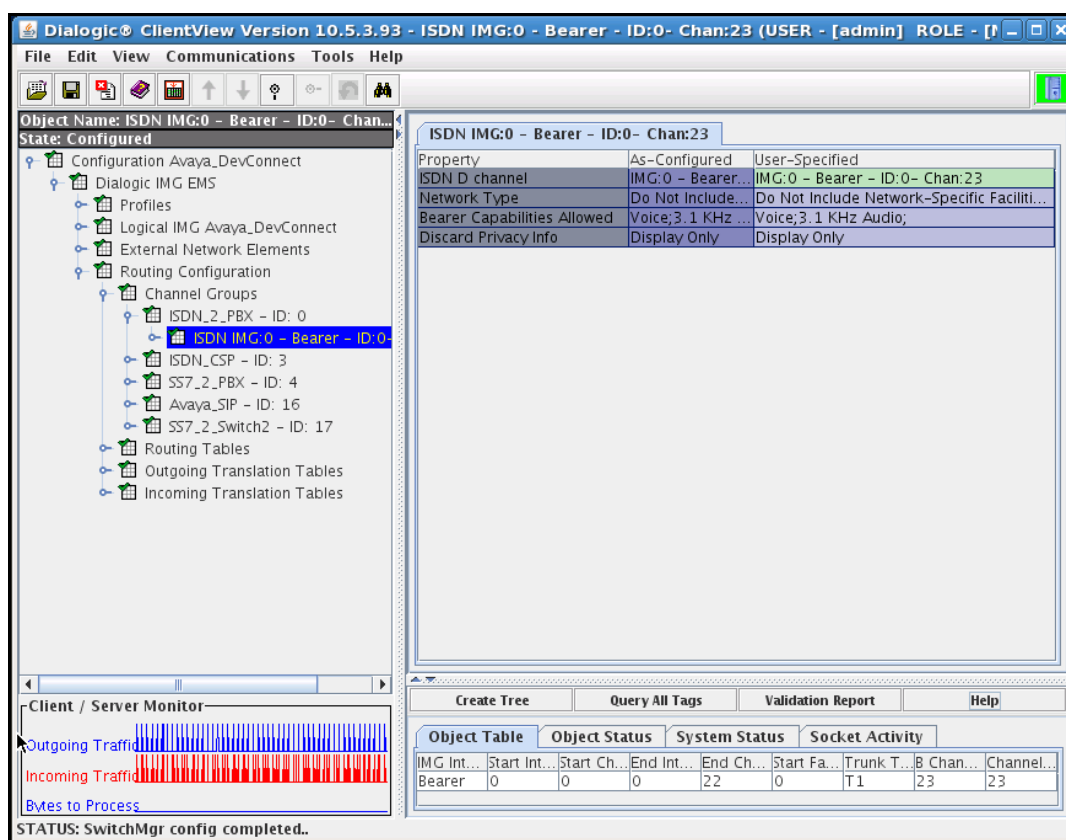
Use default settings for remaining fields. Note: The administration for the **Route Table** and **Translation table** fields are displayed in this screen, although the tables have not been created. When providing the IMG with an initial configuration, create a **Channel Group** first, then create a **Route 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**.



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**.

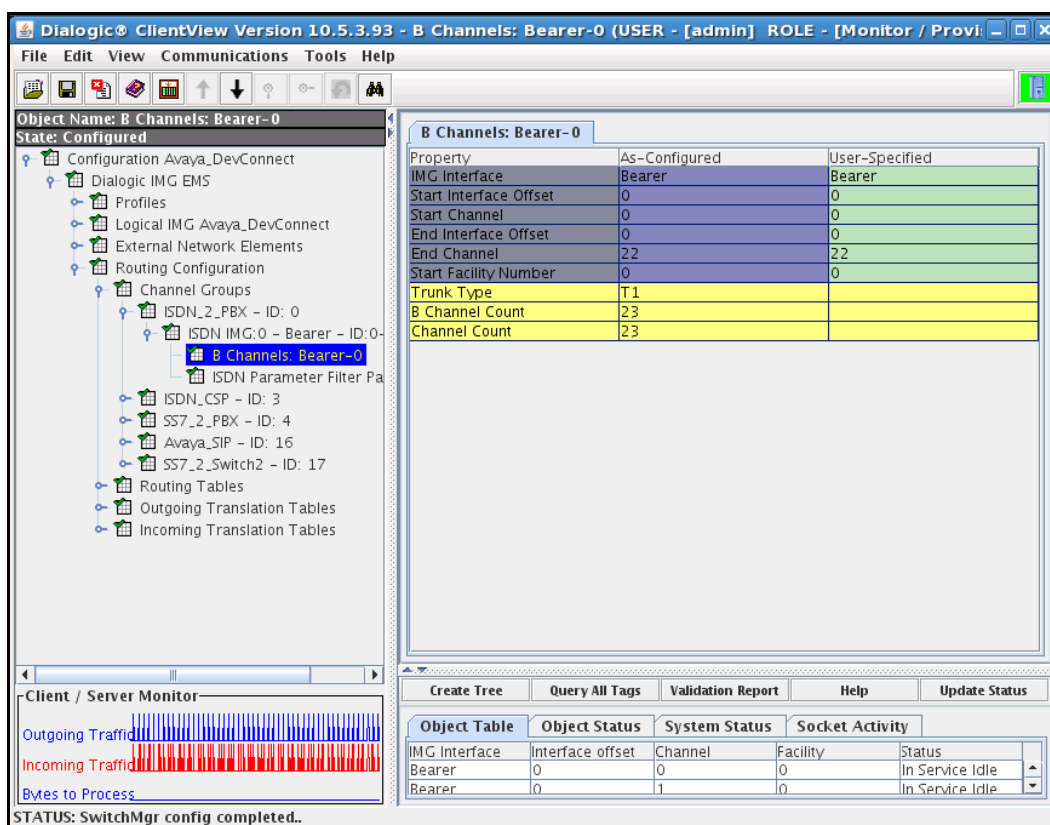


7.3.1.1.2 Assign ISDN B channels to the ISDN Group

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**.

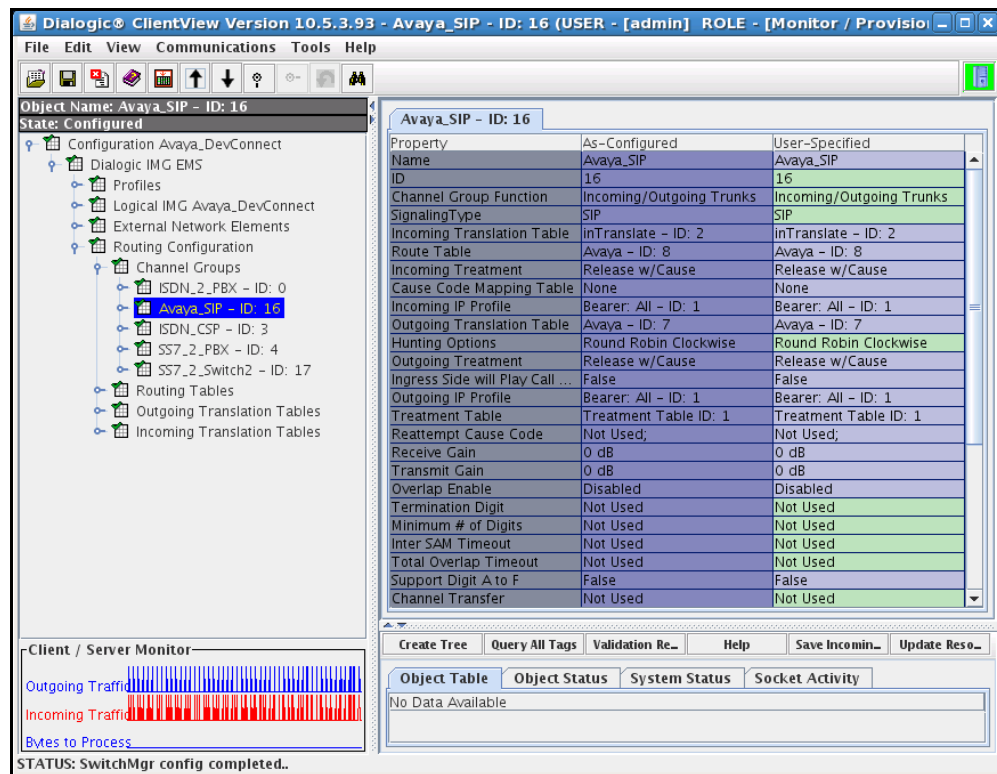


7.3.1.2 Channel Group (SIP)

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

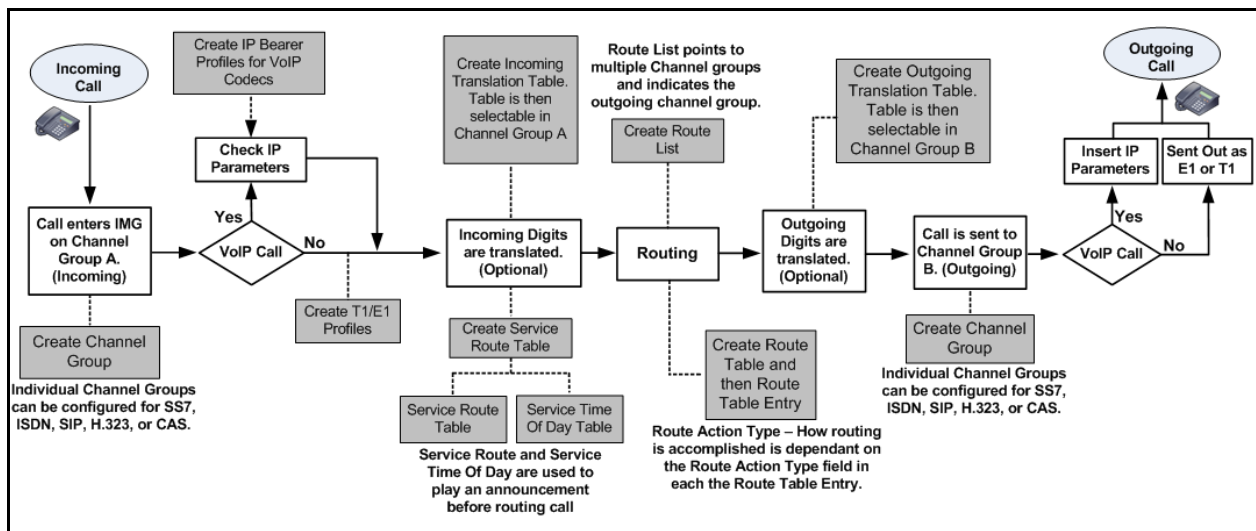
Assign a SIP Gateway to the Channel Group corresponding to each External Gateway previous:

- 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

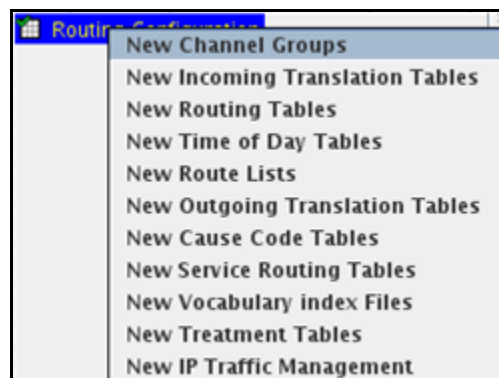
The following diagram shows the process the IMG goes through to route a call, and the ClientView GUI panes you use to configure various routing elements. Grey boxes represent ClientView configuration panes.



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™ enterprise network can establish and receive calls with Dialogic IMG 1010.

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

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

```
status signaling-group 3
```

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

```
status signaling-group 4
```

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

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

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

```
status signalling-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 signalling-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™ 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.

The screenshot displays the Avaya Aura System Manager 5.2 web interface. The left-hand navigation menu is expanded, showing the following hierarchy: **Session Manager** (highlighted with a red box) → **System Status** (highlighted with a red box) → **SIP Entity Monitoring** (highlighted with a red box). The main content area is titled "SIP Entity Link Monitoring Status Summary" and includes a "Refresh" button. Below this is a table showing the status of SIP entities:

Session Manager Name	Entity Links Down/Total	Entity Links Partially Down	SIP Entities - Monitoring Not Started	SIP Entities - Not Monitored
SessionManager	0/3	0	0	0

Below the table, there is a section titled "All Monitored SIP Entities" with another "Refresh" button. It shows 3 items with a filter set to "Enable". The listed entities are:

- [CM-AE](#)
- [CM-FS](#)
- [Gateway](#)

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 AuraTM 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 AuraTM Session Manager, Avaya AuraTM Communication Manager Access Element and Avaya AuraTM Communication Manager Feature Server. Avaya AuraTM.

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
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