



## **Avaya Solution & Interoperability Test Lab**

---

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

## **Abstract**

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

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

# Table of Content

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

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

# 1. Introduction

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

## 1.1. Interoperability Compliance Testing

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

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

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

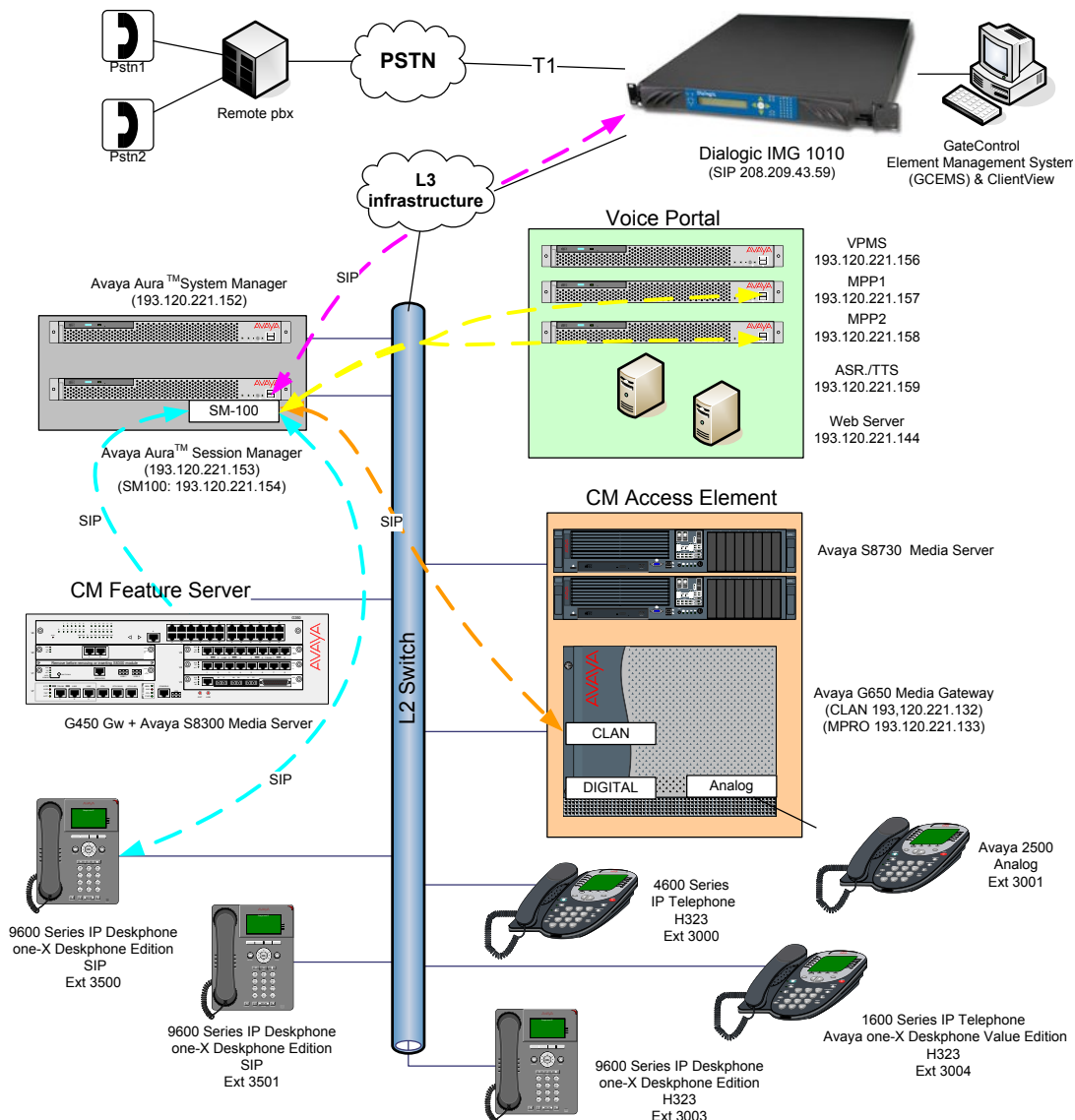
## 1.2. Support

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

- Phone: +1 781 433 9600
- E-mail: [americas.support@dialogic.com](mailto:americas.support@dialogic.com)

## 2. Reference Configuration

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



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

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

### 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"> <li>• IPSI (TN2312BP)</li> <li>• C-LAN (TN799DP)</li> <li>• IP Media Resource 320 (TN2602AP)</li> <li>• Analog (TN2793B)</li> <li>• Digital line (TN2214CP)</li> </ul>	<ul style="list-style-type: none"> <li>• TN2312BP HW28 FW050</li> <li>• TN799DP HW01 FW037</li> <li>• TN2602AP HW08 FW053</li> <li>• TN2793B 000005</li> <li>• TN2214CP HW10 FW015</li> </ul>
Avaya Voice Portal VPMS on Avaya S8510 Server	Avaya Voice Portal 5.0 Service Pack 2 (5.0.0.2.0104)
Avaya Voice Portal MPP on Avaya S8510 Server	Avaya Voice Portal 5.0 Service Pack 2 (5.0.0.2.01045.0 SP2 (5.0.0.2.0104)
Automatic Speech Recognition and Text To Speech	Nuance RealSpeak 4.5
Avaya IP Telephones: <ul style="list-style-type: none"> <li>• 9630 &amp; 9620 (SIP)</li> <li>• 9620 (H323)</li> <li>• 1616 (H323)</li> <li>• 4621 (H323)</li> <li>• Avaya Digital Telephones (2420)</li> <li>• Avaya Analog (2500)</li> </ul>	<ul style="list-style-type: none"> <li>• Avaya one-X™ Deskphone SIP 2.5.0</li> <li>• Avaya one-X™ Deskphone S3.1</li> <li>• Release 1.2.2</li> <li>• Release R2.9 SP1</li> <li>• N/A</li> <li>• N/A</li> </ul>
<b>Dialogic Inc.</b>	
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 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 extension **30xx** for stations, SIP endpoints administrated by Session Manager **35xx** reachable with **aar**. Avaya Voice Portal applications are accessible as **36xx** 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							
36	4	aar							
8	3	dac							
9	1	fac							

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



## 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	
<b>Maximum Administered SIP Trunks:</b>		<b>1000</b>	<b>300</b>	

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

## 4.2. Configure IP Node Names

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

change node-names ip		Page	1 of	2
IP NODE NAMES				
Name	IP Address			
Gateway001	193.120.221.129			
<b>SM100</b>	<b>193.120.221.154</b>			
<b>clan</b>	<b>193.120.221.132</b>			
default	0.0.0.0			
<b>mpro</b>	<b>193.120.221.133</b>			
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. 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							
<b>Audio</b>	Silence	Frames	Packet				
<b>Codec</b>	Suppression	Per Pkt	Size (ms)				
1: <b>G.711A</b>	n	2	20				
2:							
3:							

## 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.

<b>change ip-network-region 1</b>		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.

<b>add signaling-group 3</b>		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			
UI 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

<b>add signaling-group 2</b>		Page 1 of 1
SIGNALING GROUP		
Group Number: 2	<b>Group Type: sip</b>	
	<b>Transport Method: tls</b>	
IMS Enabled? n		
IP Video? n		
<b>Near-end Node Name: clan</b>	<b>Far-end Node Name: SM100</b>	
<b>Near-end Listen Port: 5061</b>	<b>Far-end Listen Port: 5061</b>	
<b>Far-end Domain:</b>	Far-end Network Region: 1	
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
<b>DTMF over IP: rtp-payload</b>	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.

<b>add trunk-group 4</b>		Page 1 of 21	
TRUNK GROUP			
Group Number: 4	<b>Group Type:</b> sip	CDR Reports: y	
<b>Group Name:</b> GWInbound	COR: 1	TN: 1	<b>TAC: 804</b>
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
<b>Service Type:</b> tie	Auth Code? n		
		<b>Signaling Group: 4</b>	
		<b>Number of Members: 10</b>	

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

<b>add trunk-group 4</b>		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
<b>Numbering Format: public</b>		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
LAR									PARM	No. Numbering	
		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 and with **36xx** corresponding to interactive voice application residing on Voice Portal. Use the **change aar analysis 0** command and add an entry to specify how to route calls to **35xx** and **36xx**. Enter the following values for the specified fields and retain the default values for the remaining fields. Submit these changes.

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

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

change aar analysis 0						Page	1	of	2
AAR DIGIT ANALYSIS TABLE									
Location: all						Percent Full:			1
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd			
35	4	4	3	aar					
36	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.

<b>display system-parameters customer-options</b>		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	
<b>Maximum Administered SIP Trunks: 100</b>		<b>50</b>	

## 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	
<b>procr</b>	<b>193.120.221.180</b>	
<b>sm100</b>	<b>193.120.221.154</b>	

### 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	<b>Authoritative Domain: avaya.com</b>	
<b>Name: Enterprise</b>		
MEDIA PARAMETERS	<b>Intra-region IP-IP Direct Audio: yes</b>	
<b>Codec Set: 1</b>	<b>Inter-region IP-IP Direct Audio: yes</b>	
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 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

<b>add signaling-group 1</b>		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**)

<b>add trunk-group 1</b>		<b>Page 1 of 21</b>	
TRUNK GROUP			
Group Number: 1	<b>Group Type: sip</b>	CDR Reports: y	
<b>Group Name: with-SessionManager</b>	COR: 1	TN: 1	TAC: 101
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
<b>Service Type: tie</b>	Auth Code? n		
		<b>Signaling Group: 1</b>	
		<b>Number of Members: 20</b>	

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

<b>add trunk-group 1</b>		<b>Page 3 of 21</b>	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
<b>Numbering Format: private</b>			
		UII 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

<b>add signaling-group 3</b>		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. **102**)
- **Service Type:** **tie**
- **Signaling Group:** The number of the signaling group associated (i.e. **2**)
- **Number of Members:** The number of SIP trunks to be allocated to calls routed to **Session Manager** (must be within the limits of the total **trunks available** from licensed verified in **Section 5.1**)

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

<b>add trunk-group 3</b>		Page 1 of 21	
TRUNK GROUP			
Group Number: 3	<b>Group Type: sip</b>	CDR Reports: y	
<b>Group Name: OUTSIDE CALL</b>	COR: 1	TN: 1	<b>TAC: 103</b>
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
<b>Service Type: tie</b>	Auth Code? n		
<b>Signaling Group: 3</b>			
<b>Number of Members: 30</b>			

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

<b>add trunk-group 3</b>		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
<b>Numbering Format: private</b>			
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 and 4-digit extensions beginning with **36** accessible as application on Voice Portal. Use the **change dialplan analysis** command to define **Dialed String 350** as an **aar Call Type**.

<b>change dialplan analysis</b>										Page 1 of 12
DIAL PLAN ANALYSIS TABLE										
Location: all										Percent Full: 2
	<b>Dialed String</b>	<b>Total Length</b>	<b>Call Type</b>	<b>Dialed String</b>	<b>Total Length</b>	<b>Call Type</b>	<b>Dialed String</b>	<b>Total Length</b>	<b>Call Type</b>	
1		3	dac							
	<b>30</b>	<b>4</b>	<b>aar</b>							
	<b>35</b>	<b>4</b>	<b>ext</b>							
	<b>36</b>	<b>4</b>	<b>aar</b>							
	9	1	fac							
	*	1	fac							

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

<b>change aar analysis 0</b>										Page 1 of 2
AAR DIGIT ANALYSIS TABLE										
Location: all										Percent Full: 2
	<b>Dialed String</b>	<b>Total</b>		<b>Route</b>	<b>Call</b>	<b>Node</b>	<b>ANI</b>			
		<b>Min</b>	<b>Max</b>	<b>Pattern</b>	<b>Type</b>	<b>Num</b>	<b>Reqd</b>			
	<b>30</b>	<b>4</b>	<b>4</b>	<b>1</b>	<b>aar</b>		<b>n</b>			
	<b>35</b>	<b>4</b>	<b>4</b>	<b>1</b>	<b>aar</b>		<b>n</b>			
	<b>36</b>	<b>4</b>	<b>4</b>	<b>1</b>	<b>aar</b>		<b>n</b>			

## 5.9. Administer ARS Analysis

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

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

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

change ars analysis 0						Page	1 of	2
ARS DIGIT ANALYSIS TABLE								
Location: all						Percent Full:		1
	Dialed	Total		Route	Call	Node	ANI	
	String	Min	Max	Pattern	Type	Num	Reqd	
0		3	25	1	pubu		n	

## 5.10. Administer Feature Access Code

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

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

## 5.11. Save Changes

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

save translation		
SAVE TRANSLATION		
Command Completion Status		Error Code
Success		0

## 6. Configure Avaya Aura™ Session Manager

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

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

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

**AVAYA** Avaya Aura™ System Manager 5.2 Welcome, admin Last logged on at: Mar. 26, 2010 12:25 AM Help | Log off

Home / Network Routing Policy

**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)

**AVAYA** Avaya Aura™ System Manager 5.2 Welcome, **admin** Last Logged on at Mar. 26, 2010 12:25 AM

[Home](#) / [Network Routing Policy](#) / [SIP Domains](#) [Help](#) | [Log off](#)

**Domain Management** [Commit](#) [Cancel](#)

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

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

\* Input Required [Commit](#) [Cancel](#)

## 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:

Notes:

Managed Bandwidth:

\* Average Bandwidth per Call:  Kbit/sec

\* Time to Live (secs):

**Location Pattern**

Add Remove

2 Items | Refresh Filter: Enable

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

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 8.2.3**. To add an adaptation, under the **Network Routing Policy select Adaptations** on the left and click on the **New** button on the right. The following screen will then be shown. Fill in the following:

Under **General**:

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

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

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

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

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

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

**AVAYA** Avaya Aura™ System Manager 5.2 Welcome, **admin** Last Logged on at Apr. 14, 2010 10:00 AM [Help](#) [Log off](#)

Home / Network Routing Policy / Adaptations / Adaptation Details

**Adaptation Details** [Commit](#) [Cancel](#)

**General**

\* Adaptation name:

Module name:

Module parameter:

Egress URI Parameters:

Notes:

**Digit Conversion for Incoming Calls to SM**

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

1 Item | Refresh Filter: Enable

<input type="checkbox"/>	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

<input type="checkbox"/>	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 )

**Left Sidebar:**

- Asset Management
- Communication System Management
- User Management
- Monitoring
- Network Routing Policy
  - Adaptations**
  - Dial Patterns
  - Entity Links
  - Locations
  - Regular Expressions
  - Routing Policies
  - SIP Domains
  - SIP Entities
  - Time Ranges
  - Personal Settings
- Security
- Applications
- Settings
- Session Manager

**Shortcuts:**

- Change Password
- Help for Adaptation Details
- Fields
- Help for Committing configuration changes

## 6.4. Add Host Entries for Voice Portal MPPs

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

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

Shown below is the updated screen for the sample configuration.

**AVAYA** Avaya Aura™ System Manager 5.2 Welcome, **admin** Last Logged on at Apr. 15, 2010 1:19 PM  
[Help](#) [Log off](#)

Home / Session Manager / Network Configuration / Local Host Name Resolution

**Local Host Name Resolution**  
This page allows you to add, edit, or remove local host name entries. Host name entries on this page will override information provided by DNS.

[Local Host Name Entries](#)

[New](#) [Edit](#) [Delete](#) [More Actions](#)

2 Items [Refresh](#) Filter: Enable

<input type="checkbox"/>	Host Name (FQDN)	IP Address	Port	Priority	Weight	Transport
<input type="checkbox"/>	VoicePortal.avaya.com	193.120.221.157	5060	100	100	TCP
<input type="checkbox"/>	VoicePortal.avaya.com	193.120.221.158	5060	100	100	TCP

Select : All, None ( 0 of 2 Selected )

## 6.5. Add SIP Entities

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

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

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

Under **General**:

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

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

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 system management categories: Asset Management, Communication System Management, User Management, Monitoring, Network Routing Policy (highlighted), Adaptations, Dial Patterns, Entity Links, Locations, Regular Expressions, Routing Policies, SIP Domains, SIP Entities (highlighted), Time Ranges, Personal Settings, Security, Applications, Settings, and Session Manager. The main content area is titled 'SIP Entity Details' and features a 'Commit' button and a 'Cancel' button. The 'General' tab is active, showing the following fields: 'Name' (CM-AE), 'FQDN or IP Address' (193.120.221.132), 'Type' (CM), 'Notes' (empty), 'Adaptation' (empty), 'Location' (Enterprise), 'Time Zone' (Europe/Dublin), 'Override Port & Transport with DNS' (unchecked), 'SRV' (unchecked), 'SIP Timer B/F (in seconds)' (4), 'Credential name' (empty), 'Call Detail Recording' (none), and 'SIP Link Monitoring' (Use Session Manager Configuration).

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

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

Under **General**:

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

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

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 welcome message: 'Welcome, admin Last Logged on at Mar. 26, 2010 12:25 AM'. A navigation menu on the left lists various system management categories, with 'Network Routing Policy' and its sub-item '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 set to 'CM-FS'; 'FQDN or IP Address' is set to '193.120.221.180'; 'Type' is set to 'CM'; 'Location' is set to 'Enterprise'; and 'Time Zone' is set to 'Europe/Dublin'. There are also checkboxes for 'Override Port & Transport with DNS SRV' (unchecked), 'SIP Timer B/F (in seconds)' (set to 4), and 'Call Detail Recording' (set to none). At the bottom, the 'SIP Link Monitoring' section is set to 'Use Session Manager Configuration'. 'Commit' and 'Cancel' buttons are located in the top right corner of the form area.

**AVAYA** Avaya Aura™ System Manager 5.2 Welcome, admin Last Logged on at Mar. 26, 2010 12:25 AM Help | Log off

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

**SIP Entity Details** [Commit] [Cancel]

**General**

\* Name: CM-FS

\* FQDN or IP Address: 193.120.221.180

Type: CM

Notes:

Adaptation:

Location: Enterprise

Time Zone: Europe/Dublin

Override Port & Transport with DNS SRV: ☐

\* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: none

**SIP Link Monitoring**

SIP Link Monitoring: Use Session Manager Configuration

### 6.5.3. Adding Voice Portal SIP Entity

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

Under **General**:

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

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

**AVAYA** Avaya Aura™ System Manager 5.2 Welcome, admin Last Logged on at Apr. 15, 2010 1:19 PM Help | Log off

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

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

**General**

\* Name: VoicePortal

\* FQDN or IP Address: VoicePortal.avaya.com

Type: Voice Portal

Notes:

Adaptation:

Location: Enterprise

Time Zone: Etc/GMT

Override Port & Transport with DNS SRV: ☐

\* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: none

**SIP Link Monitoring**

SIP Link Monitoring: Use Session Manager Configuration

### 6.5.4. Adding Dialogic IMG 1010 Gateway SIP Entity

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

Under **General**:

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

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

The screenshot displays the Avaya Aura System Manager 5.2 interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura™ System Manager 5.2', and a welcome message for the 'admin' user. The breadcrumb trail indicates the current path: Home / Network Routing Policy / SIP Entities / SIP Entity Details. The left sidebar contains a tree view of system components, with 'SIP Entities' selected under 'Network Routing Policy'. The main content area is titled 'SIP Entity Details' and features a 'Commit' button. The 'General' tab is active, showing the following configuration fields: Name (Gateway), FQDN or IP Address (208.209.43.59), Type (Gateway), Adaptation (IMG\_DigitConversionAdapter), Location (Enterprise), Time Zone (Europe/Dublin), SIP Timer B/F (4 seconds), and Call Detail Recording (none). The 'SIP Link Monitoring' section at the bottom shows 'Use Session Manager Configuration'.

### 6.5.5. Adding Avaya Aura™ Session Manager SIP Entity

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

Under **General**:

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

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

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

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

**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)

## 6.6. Add Entity Links

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

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

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

The screenshot shows the Avaya Aura System Manager 5.2 interface. The left sidebar has a menu with 'Network Routing Policy' selected, and 'Entity Links' is highlighted. The main area shows a table with one item, 'SM-CMAE', configured with 'SessionManager' as SIP Entity 1, 'TLS' as Protocol, '5061' as Port, 'CM-AE' as SIP Entity 2, '5061' as Port, and 'Trusted' checked. The 'Commit' button is circled in red.

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 menu with 'Network Routing Policy' selected, and 'Entity Links' is highlighted. The main area shows a table with one item, 'SM-CMFS', configured with 'SessionManager' as SIP Entity 1, 'TLS' as Protocol, '5061' as Port, 'CM-FS' as SIP Entity 2, '5061' as Port, and 'Trusted' checked. The 'Commit' button is circled in red.

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 for Voice Portal.

The screenshot shows the 'Entity Links' configuration page for 'SM-VP'. The left sidebar lists navigation options: Asset Management, Communication System Management, User Management, Monitoring, and Network Routing Policy. Under 'Network Routing Policy', 'Entity Links' is selected. The main area shows a table with one item. The table columns are Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Trusted, and Notes. The values are: Name: SM-VP, SIP Entity 1: SessionManager, Protocol: TCP, Port: 5060, SIP Entity 2: VoicePortal, Port: 5060, Trusted: checked. There are 'Commit' and 'Cancel' buttons at the top right and bottom right. A '\* Input Required' message is at the bottom left.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
*SM-VP	*SessionManager	TCP	*5060	*VoicePortal	*5060	<input checked="" type="checkbox"/>	

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

The screenshot shows the 'Entity Links' configuration page for 'SM-IMG1010'. The left sidebar is the same as the previous screenshot, with 'Entity Links' selected. The main area shows a table with one item. The table columns are Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Trusted, and Notes. The values are: Name: SM-IMG1010, SIP Entity 1: SessionManager, Protocol: UDP, Port: 5060, SIP Entity 2: Gateway, Port: 5060, Trusted: checked. There are 'Commit' and 'Cancel' buttons at the top right and bottom right. A '\* Input Required' message is at the bottom left.

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

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

The screenshot shows the 'Entity Links' summary view. The left sidebar is the same. The main area shows a table with four items. The table columns are Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Trusted, and Notes. The values are: Name: SM-CMAE, SIP Entity 1: SessionManager, Protocol: TCP, Port: 5060, SIP Entity 2: CM-AE, Port: 5060, Trusted: checked; Name: SM-CMFS, SIP Entity 1: SessionManager, Protocol: TLS, Port: 5061, SIP Entity 2: CM-FS, Port: 5061, Trusted: checked; Name: SM-IMG1010, SIP Entity 1: SessionManager, Protocol: UDP, Port: 5060, SIP Entity 2: Gateway, Port: 5060, Trusted: checked; Name: SM-VP, SIP Entity 1: SessionManager, Protocol: TCP, Port: 5060, SIP Entity 2: VoicePortal, Port: 5060, Trusted: checked. There are 'Edit', 'New', 'Duplicate', 'Delete', 'More Actions', and 'Commit' buttons at the top. A 'Select: All, None ( 0 of 4 Selected )' message is at the bottom left.

	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"/>	
<input type="checkbox"/>	SM-VP	SessionManager	TCP	5060	VoicePortal	5060	<input checked="" type="checkbox"/>	

## 6.7. Add Routing Policies

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

Under **General**:

- Enter a descriptive name in **Name**

Under **SIP Entity as Destination**:

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

Under **Time of Day**:

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

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

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 includes a 'Commit' button. The 'General' section shows the policy name 'RP-to-CM-AE' and a note 'Routes to CM'. The 'SIP Entity as Destination' section shows a 'Select' button and a table with one entry: 'CM-AE' with FQDN or IP Address '193.120.221.132' and Type 'CM'. The 'Time of Day' section shows an 'Add' button and a table with one entry: '24/7' with Start Time '00:00', End Time '23:59', and Notes 'Always Active'.

**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
<b>CM-AE</b>	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
<b>0</b>	<b>24/7</b>	<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 picture shows the Routing Policy for Voice Portal.

**AVAYA** Avaya Aura™ System Manager 5.2 Welcome, admin Last Logged on at Apr. 15, 2010 1:19 PM  
[Help](#) | [Log off](#)

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

**Routing Policy Details** Commit Cancel

**General**

\* Name:

Disabled: ☐

Notes:

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type	Notes
VoicePortal	VoicePortal.avaya.com	Voice Portal	

**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
<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 )

The following screen shows the Routing Policy for 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 / Routing Policies / Routing Policy Details

**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	Name	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 )

## 6.8. Add Dial Patterns

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

Under **General**:

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

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

**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: 30

\* Min: 4

\* Max: 4

Emergency Call: ☐

SIP Domain: ALL

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
--------------------------	----------------------	-------

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

**Avaya Aura™ System Manager 5.2**

Welcome, **admin** Last Logged on at Apr. 15, 2010 1:19 PM [Help](#) | [Log off](#)

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

**Dial Pattern Details** Commit Cancel

**General**

\* Pattern: 06

\* Min: 3

\* Max: 24

Emergency Call: ☐

SIP Domain: -ALL-

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-VoicePortal	0	<input type="checkbox"/>	VoicePortal	

Select : All, None ( 0 of 1 Selected )

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

Under **General**:

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

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

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 / Dial Patterns / Dial Pattern Details

Asset Management

Communication System Management

User Management

Monitoring

Network Routing Policy

Dial Patterns

Entity Links

Locations

Regular Expressions

Routing Policies

SIP Domains

SIP Entities

Time Ranges

Personal Settings

Security

Applications

Settings

Session Manager

Dial Pattern Details

Commit

Cancel

General

\* Pattern: 0

\* Min: 3

\* Max: 24

Emergency Call: ☐

SIP Domain: -ALL-

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-Gateway	0	<input type="checkbox"/>	Gateway	

Select : All, None ( 0 of 1 Selected )

## 6.9. Add Avaya Aura™ Session Manager

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

Under **General**:

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

Under **Security Module**:

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

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

**AVAYA** Avaya Aura™ System Manager 5.2 Welcome, **admin** Last Logged on at Feb. 09, 2010 5:30 PM [Help](#) [Log off](#)

[Home](#) / [Session Manager](#) / [Session Manager Administration](#) / [Edit Session Manager](#)

**Add Session Manager** **Commit** **Cancel**

[General](#) | [Security Module](#) | [Monitoring](#) | [CDR](#) | [Personal Profile Manager \(PPM\)](#) - [Connection Settings](#) | [Event Server](#) | [Expand All](#) | [Collapse All](#)

**General**

SIP Entity Name

Description

\*Management Access Point Host Name/IP

\*Direct Routing to Endpoints

**Security Module**

SIP Entity IP Address

\*Network Mask

\*Default Gateway

\*Call Control PHB

\*QOS Priority

\*Speed & Duplex

VLAN ID

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

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

### 6.10.1. Create an Application Entry

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

Under **Application**:

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

The screenshot shows the Avaya Aura™ System Manager 5.2 web interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura™ System Manager 5.2', and a welcome message for user 'admin' last logged on at Apr. 09, 2010 4:38 AM. A red breadcrumb trail reads '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 configuration form titled "Attributes" in a dropdown menu. The form contains several input fields and checkboxes. Red circles highlight the "Login" field (containing "init"), the "Password" field (containing "\*\*\*\*\*"), the "Confirm Password" field (containing "\*\*\*\*\*"), and the "Commit" button at the bottom right. Other fields include "Is SSH Connection" (checked), "\* Port" (5022), "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 "\*Required".

Attributes
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

### 6.10.2. Create a Feature Server Application

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

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

Click on **Commit** to save.

**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"/>

### 6.10.3. Create a Feature Server Application Sequence

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

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 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	

## 6.10.4. Synchronize Avaya Aura™ Communication Manager Data

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

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.

## 6.11. Add Users for SIP Phones

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

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 6.10.3**. Click on **Station Profile** to expand that section. Enter the following fields and use defaults for the remaining fields:

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

☒ **Session Manager**

\* Session Manager Instance: SessionManager

Origination Application Sequence: AppSeq-FeatureServer

Termination Application Sequence: AppSeq-FeatureServer

☒ **Station Profile**

\* System: CM-featureServer

Use Existing Stations: ☐

\* Extension: 3500

\* Template: DEFAULT\_9630SIP

Set Type: 9630SIP

Security Code:

\* Port: IP

Delete Station on Unassign of Station from User: ☐

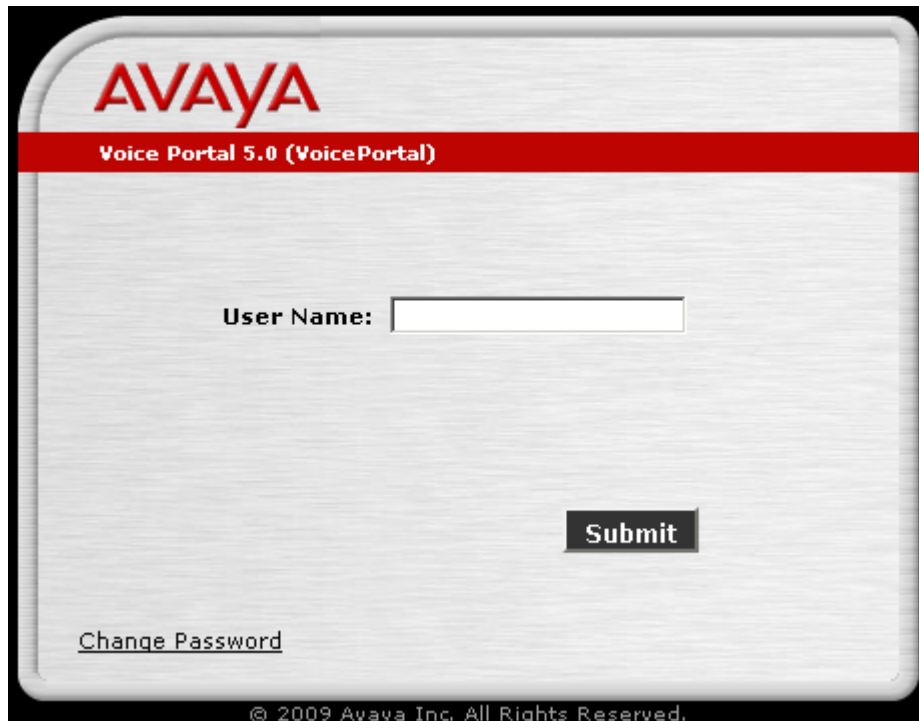
Click on **Commit** to save (not shown).

## 7. Configure Avaya Aura™ Voice Portal

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

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

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

The image shows the Avaya Voice Portal 5.0 login interface. At the top, the Avaya logo is displayed in red. Below it, a red banner contains the text "Voice Portal 5.0 (VoicePortal)". The main area is white and contains a "User Name:" label followed by a text input field. Below the input field is a "Submit" button. At the bottom left, there is a link labeled "Change Password". At the bottom center, the copyright notice "© 2009 Avaya Inc. All Rights Reserved." is visible.



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

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

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

**AVAYA** Welcome, vpadmin  
Last logged in 3/24/10 at 11:28:55 AM GMT

**Voice Portal 5.0 (VoicePortal)** Home ? Help Logoff

Expand All | Collapse All

- ▼ **User Management**
  - Roles
  - Users
  - Login Options
- ▼ **Real-Time Monitoring**
  - System Monitor
  - Active Calls
  - Port Distribution
- ▼ **System Maintenance**
  - Audit Log Viewer
  - Trace Viewer
  - Log Viewer
  - Alarm Manager
- ▼ **System Management**
  - MPP Manager
  - Software Upgrade
  - System Backup
- ▼ **System Configuration**
  - Alarm Codes
  - Alarm/Log Options
  - Applications
  - MPP Servers
  - Report Data
  - SNMP
  - Speech Servers
  - VoIP Connections**
  - VPMS Servers
- ▼ **Security**
  - Certificates
  - Licensing
- ▼ **Reports**
  - Standard
  - Custom
  - Scheduled

You are here: [Home](#) > [System Configuration](#) > [VoIP Connections](#) > [Change SIP Connection](#)

### Change SIP Connection

Use this page to change the configuration of a SIP connection.

Name: toSM

Enable: ☒ Yes ☐ No

Proxy Transport: TCP

Address	Port	Administration
193.120.221.154	5060	Administration Remove

[Additional Proxy Server](#)

Listener Port: 5060

SIP Domain: avaya.com

P-Asserted-Identity:

**Call Capacity**

Maximum Simultaneous Calls: 20

☒ All Calls can be either inbound or outbound  
☐ Configure number of inbound and outbound calls allowed

**Save Apply Cancel Help**

## 7.2. Add the MPP Servers

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

**AVAYA** Welcome, vpadmin  
Last logged in 3/24/10 at 11:28:55 AM GMT

**Voice Portal 5.0 (VoicePortal)** Home Help Logoff

Expand All | Collapse All

You are here: [Home](#) > [System Configuration](#) > [MPP Servers](#) > Change MPP Server

### Change MPP Server

Use this page to change the configuration of an MPP. Take care when changing the MPP Trace Logging Thresholds. Do not set Trace Levels to Finest if your Voice Portal system has heavy call traffic. The system might experience performance issues if Trace Levels are set to Finest. Set Trace Levels to Finest only when you are troubleshooting the system.

Name:

Host Address:

Network Address (VoIP):

Network Address (MRCP):

Network Address (AppSvr):

Maximum Simultaneous Calls:

Restart Automatically: ☐ Yes ☒ No

#### MPP Certificate

Owner: CN=mmp1.avaya.com,O=Avaya,OU=MPP  
Issuer: CN=mmp1.avaya.com,O=Avaya,OU=MPP  
Serial Number: 9a619062173219e7  
Valid from: Fri Mar 05 12:12:08 GMT 2010 until: Mon Mar 02 12:12:08 GMT 2020  
Certificate fingerprints  
MD5: 45:be:bb:31:0f:c8:f3:f9:89:23:2b:9b:c9:55:ad:ee  
SHA: c8:f1:f7:06:02:df:ed:52:e1:ac:8b:6d:66:54:29:48:09:07:ff:3c

Categories and Trace Levels >

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

**AVAYA** Welcome, vpadmin  
Last logged in 3/24/10 at 11:28:55 AM GMT

**Voice Portal 5.0 (VoicePortal)** Home ? Help Logoff

Expand All | Collapse All

You are here: [Home](#) > System Configuration > MPP Servers

### MPP Servers

This page displays the list of Media Processing Platform (MPP) servers in the Voice Portal system. When an MPP receives a call from a PBX, it invokes a VoiceXML application on an application server and communicates with ASR and TTS servers as necessary to process the call.

<input type="checkbox"/>	Name	Host Address	Network Address (VoIP)	Network Address (MRCP)	Network Address (AppSvr)	Maximum Simultaneous Calls	Trace Level
<input type="checkbox"/>	mpp1	193.120.221.157	<Default>	<Default>	<Default>	10	Use MPP Settings
<input type="checkbox"/>	mpp2	193.120.221.158	<Default>	<Default>	<Default>	10	Use MPP Settings

[Add](#) [Delete](#)

[MPP Settings](#) [AVB Settings](#) [Event Handlers](#) [Video Settings](#) [VoIP Settings](#) [Help](#)

**Left Navigation Menu:**

- ▼ User Management
  - Roles
  - Users
  - Login Options
- ▼ Real-Time Monitoring
  - System Monitor
  - Active Calls
  - Port Distribution
- ▼ System Maintenance
  - Audit Log Viewer
  - Trace Viewer
  - Log Viewer
  - Alarm Manager
- ▼ System Management
  - MPP Manager
  - Software Upgrade
  - System Backup
- ▼ System Configuration
  - Alarm Codes
  - Alarm/Log Options
  - Applications
    - MPP Servers**
    - Report Data
    - SNMP
    - Speech Servers
    - VoIP Connections
    - VPMS Servers
- ▼ Security
  - Certificates
  - Licensing
- ▼ Reports
  - Standard
  - Custom
  - Scheduled

### 7.3. Configuring the VoIP Audio Format

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

**AVAYA** Welcome, vpadmin  
Last logged in 3/24/10 at 11:28:55 AM GMT

**Voice Portal 5.0 (VoicePortal)** Home ? Help Logoff

Expand All | Collapse All

You are here: [Home](#) > System Configuration > [MPP Servers](#) > VoIP Settings

### VoIP Settings

Voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols such as H.323 and Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the network. Note that if you make any changes to this page, you must restart all MPPs.

#### Port Ranges

	Low	High
UDP:	23000	30999
TCP:	31000	31999
MRCP:	32000	32999
H.323 Station:	35000	50000

#### RTCP Monitor Settings

Host Address:

Port:

#### VoIP Audio Formats

MPP Native Format: **audio/x-alaw-basic**

**Left Navigation Menu:**

- ▼ User Management
  - Roles
  - Users
  - Login Options
- ▼ Real-Time Monitoring
  - System Monitor
  - Active Calls
  - Port Distribution
- ▼ System Maintenance
  - Audit Log Viewer
  - Trace Viewer
  - Log Viewer
  - Alarm Manager
- ▼ System Management
  - MPP Manager
  - Software Upgrade
  - System Backup
- ▼ System Configuration
  - Alarm Codes
  - Alarm/Log Options
  - Applications
    - MPP Servers**
    - Report Data
    - SNMP
    - Speech Servers
    - VoIP Connections
    - VPMS Servers
- ▼ Security
  - Certificates
  - Licensing
- ▼ Reports
  - Standard
  - Custom
  - Scheduled

## 7.4. Add an ASR Server

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

The screenshot shows the Avaya Voice Portal 5.0 (VoicePortal) interface. The left navigation pane has 'Speech Servers' selected under 'System Configuration'. The main area shows the 'ASR' tab for 'Speech Servers'. A table lists the configured ASR server:

	Name	Enable	Network Address	Engine Type	MRCP	Base Port	Total Number of Licensed ASR Resources	Languages
<input type="checkbox"/>	nuance-asr	Yes	193.120.221.160	Nuance	MRCP V1	4900	10	English(USA) en-us

Buttons for 'Add', 'Delete', 'Customize', and 'Help' are visible below the table.

## 7.5. Add a TTS Server

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

The screenshot shows the Avaya Voice Portal 5.0 (VoicePortal) interface. The left navigation pane has 'Speech Servers' selected under 'System Configuration'. The main area shows the 'TTS' tab for 'Speech Servers'. A table lists the configured TTS server:

	Name	Enable	Network Address	Engine Type	MRCP	Base Port	Total Number of Licensed TTS Resources	Voices
<input type="checkbox"/>	nuance-tts	Yes	193.120.221.160	Nuance	MRCP V1	4900	10	English(USA) en-US Jennifer F

Buttons for 'Add', 'Delete', 'Customize', and 'Help' are visible below the table.

## 7.6. Add an Application

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

**AVAYA** Welcome, vpadmin  
Last logged in 3/24/10 at 11:28:55 AM GMT

**Voice Portal 5.0 (VoicePortal)** Home ? Help Logoff

Expand All | Collapse All

You are here: [Home](#) > [System Configuration](#) > [Applications](#) > [Change Application](#)

### Change Application

Use this page to change the configuration of a VoiceXML or CCXML application.

**Name:** SampleApp

**Enable:** ☒ Yes ☐ No

**MIME Type:** VoiceXML

**VoiceXML URL:** http://193.120.221.144/misc/avptestapp/intro.vxml **Verify**

#### Speech Servers

**ASR:** Nuance **TTS:** Nuance

**Languages:** English(USA) en-us **Voices:** English(USA) en-US Jennifer F

#### Application Launch

**Type:** ☒ Inbound ☐ Inbound Default ☐ Outbound

☒ Number ☐ Number Range ☐ URI

**Called Number:**  **Add**

3600 **Remove**

**Speech Parameters** ▶  
**Reporting Parameters** ▶  
**Advanced Parameters** ▶

**Save** **Apply** **Cancel** **Help**

## 7.7. Start the MPP Servers

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

**AVAYA** Welcome, vpadmin  
Last logged in 3/24/10 at 11:28:55 AM GMT

**Voice Portal 5.0 (VoicePortal)** Home Help Logoff

Expand All | Collapse All

You are here: [Home](#) > System Management > MPP Manager

**MPP Manager (4/6/10 5:31:02 PM IST)** Refresh

This page displays the current state of each MPP in the Voice Portal system. To enable the state and mode commands, select one or more MPPs. To enable the mode commands, the selected MPPs must also be stopped.

Last Poll: 4/6/10 5:31:02 PM IST

	Server Name	Mode	State	Config	Auto Restart	Restart Schedule Today	Recurring	Active Calls In	Out
<input type="checkbox"/>	mmp1	Online	Running	OK	No	No	None	0	0
<input type="checkbox"/>	mmp2	Online	Running	OK	No	No	None	0	0

**State Commands**

Start Stop Restart Reboot Halt Cancel

**Mode Commands**

Offline Test Online

**Restart/Reboot Options**

☐ One server at a time

☐ All selected servers at the same time

Help

## 8. Dialogic IMG 1010 Configuration

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

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

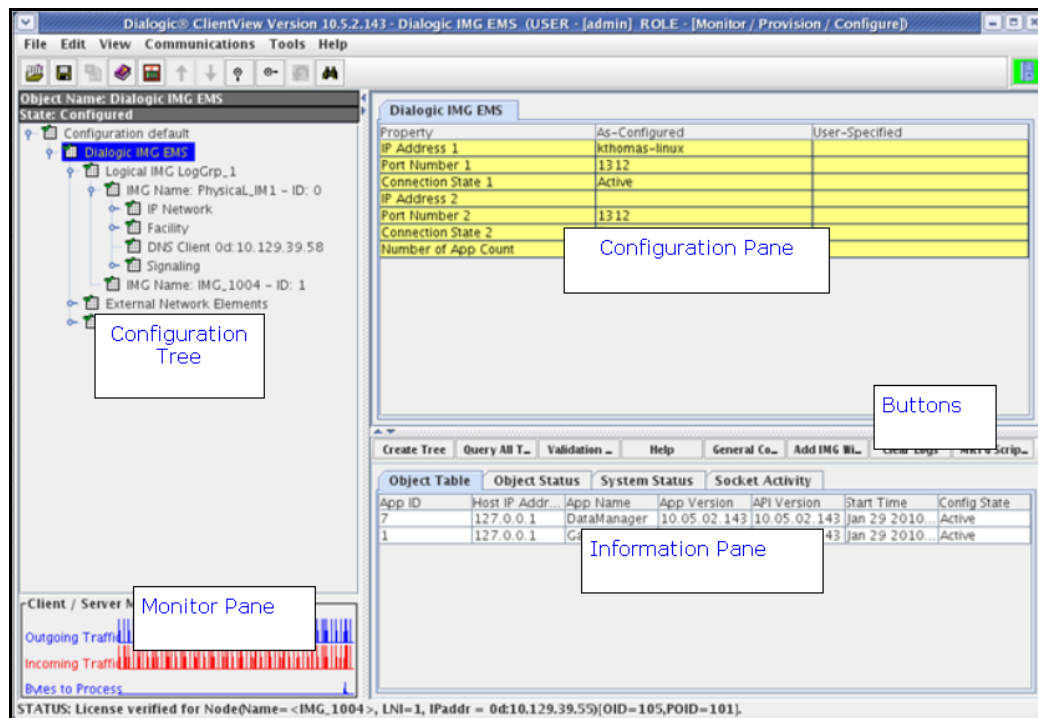
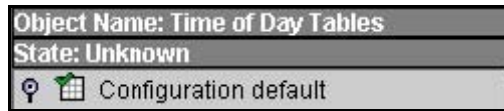


Figure 2 - Dialogic Inc. ClientView Main Window

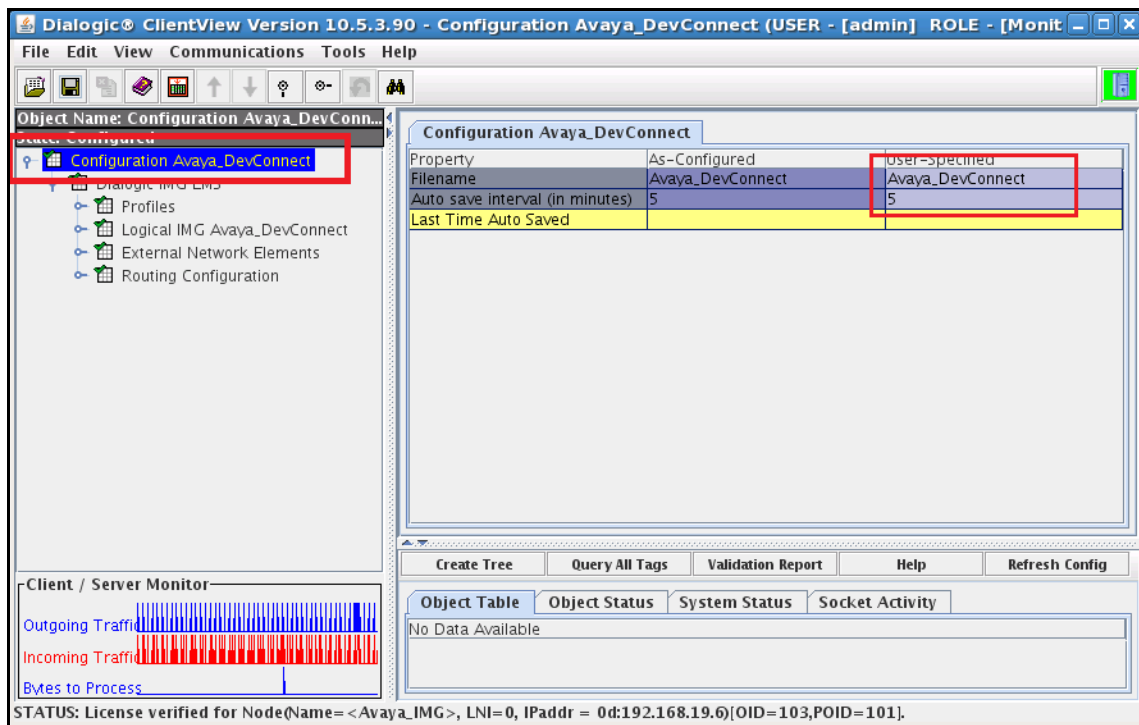


## 8.1. IMG Configuration Name

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



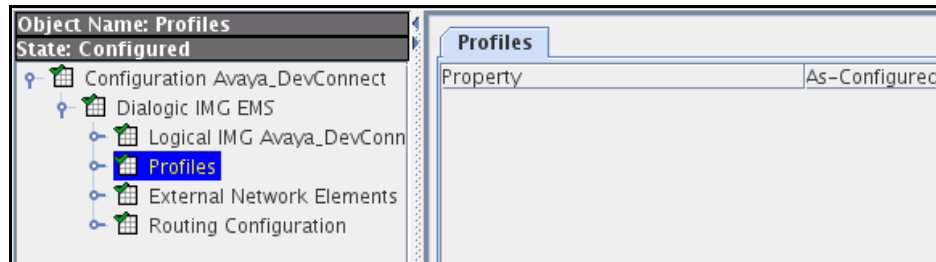
Enter a descriptive name in the **Filename** field in the Configuration Pane. To save the changes, right-click **Configuration Avaya\_DevConnect**, and select **Commit**. The picture below shows the actions performed on IMG 1010





## 8.2. Profiles

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

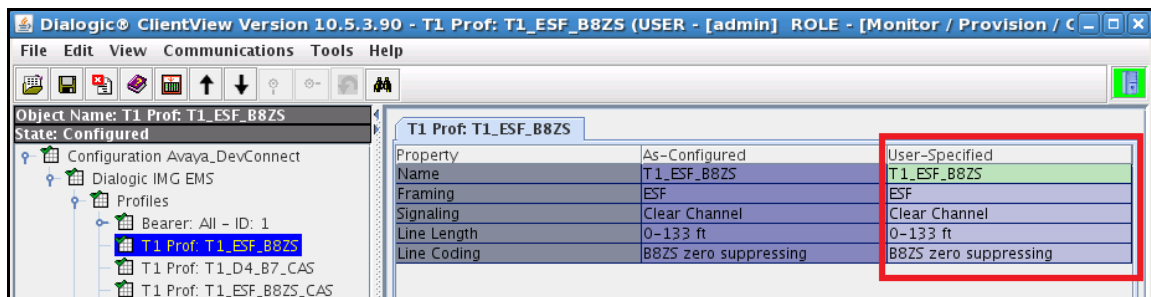


### 8.2.1. DS1 Profiles

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

- Right-click the Profile in the Configuration Tree, and select **New T1 or E1 Profile**.  
(A single profile can be used for many DS1 spans that all use the same configuration.)
- Enter a name for the profile i.e. **T1\_ESF\_B8ZS**
- Select **Clear Channel** from the drop down list for the **Signaling** field in the Configuration Pane.  
**Note:** **Clear Channel** is used for ISDN-PRI or SS7 spans.
- Configure remaining settings to match network configuration.
- Create additional profiles as needed.

The picture below shows the actions performed on IMG 1010



## 8.2.2. IP Bearer Profiles

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

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

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

**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

### 8.2.2.1 New IP codec in Bearer Profile

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

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

To save the changes, right-click **Profile: 1- Entry:0**, and select **Commit**.

Note: if using G.729A on Communication Manager, set **Annex B** to **No** in Bearer Profile.

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	10

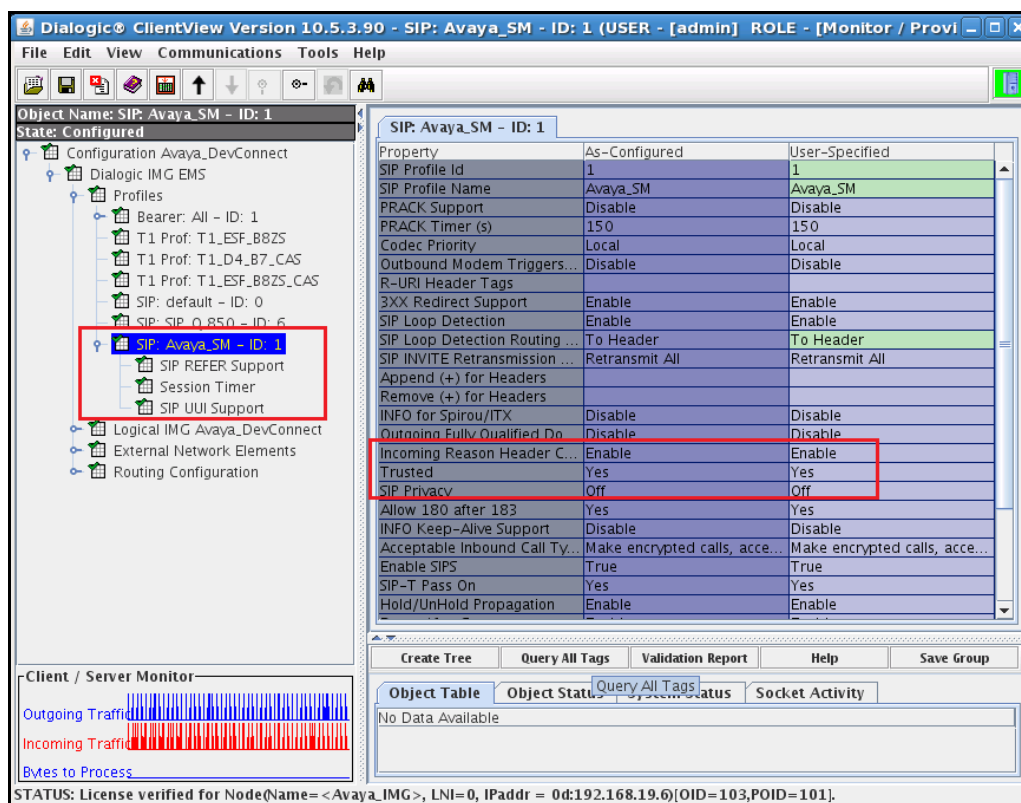
STATUS: License verified for Node(Name=<Avaya\_IMG>, LNI=0, IPAddr = 0d:192.168.19.6)[OID=103,POID=101].

### 8.2.3. SIP Profiles

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

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

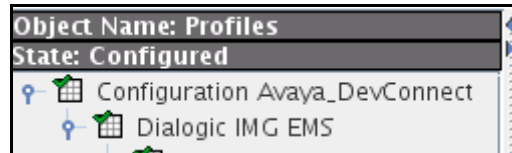
**Note:** If needed P-AI provide from Session Manager, set **SIP Privacy** to **On**



## 8.2.4. Logical IMG

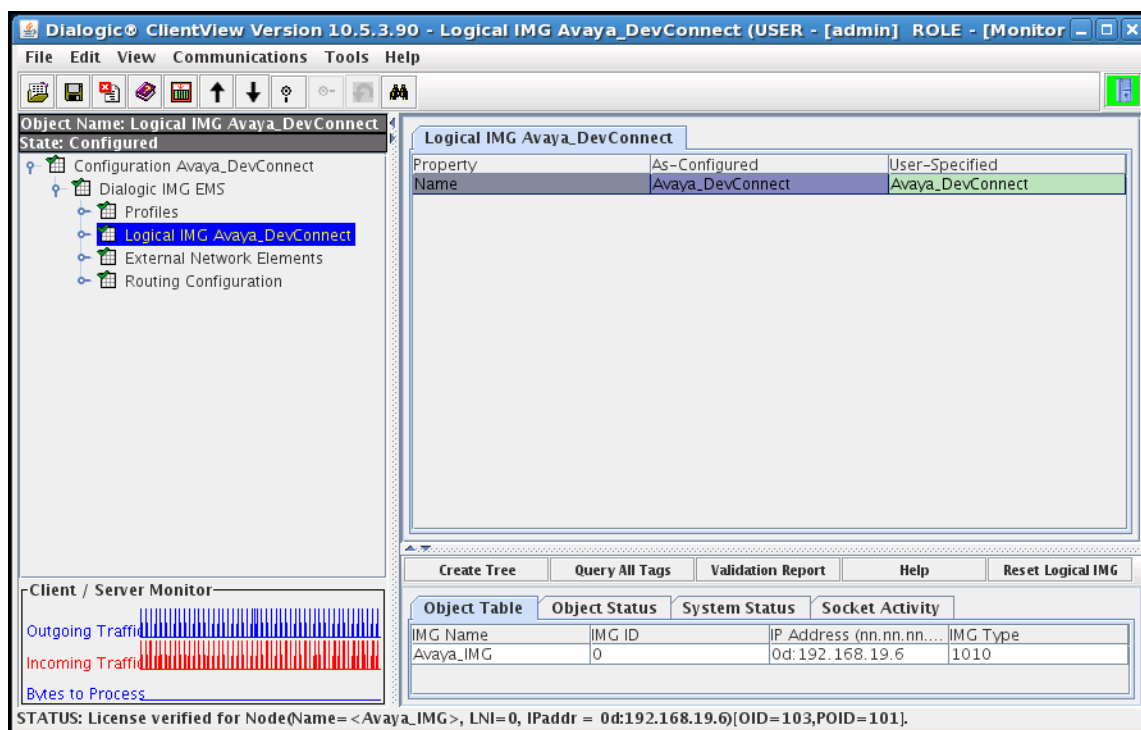
Create a logical IMG as follows:

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



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

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

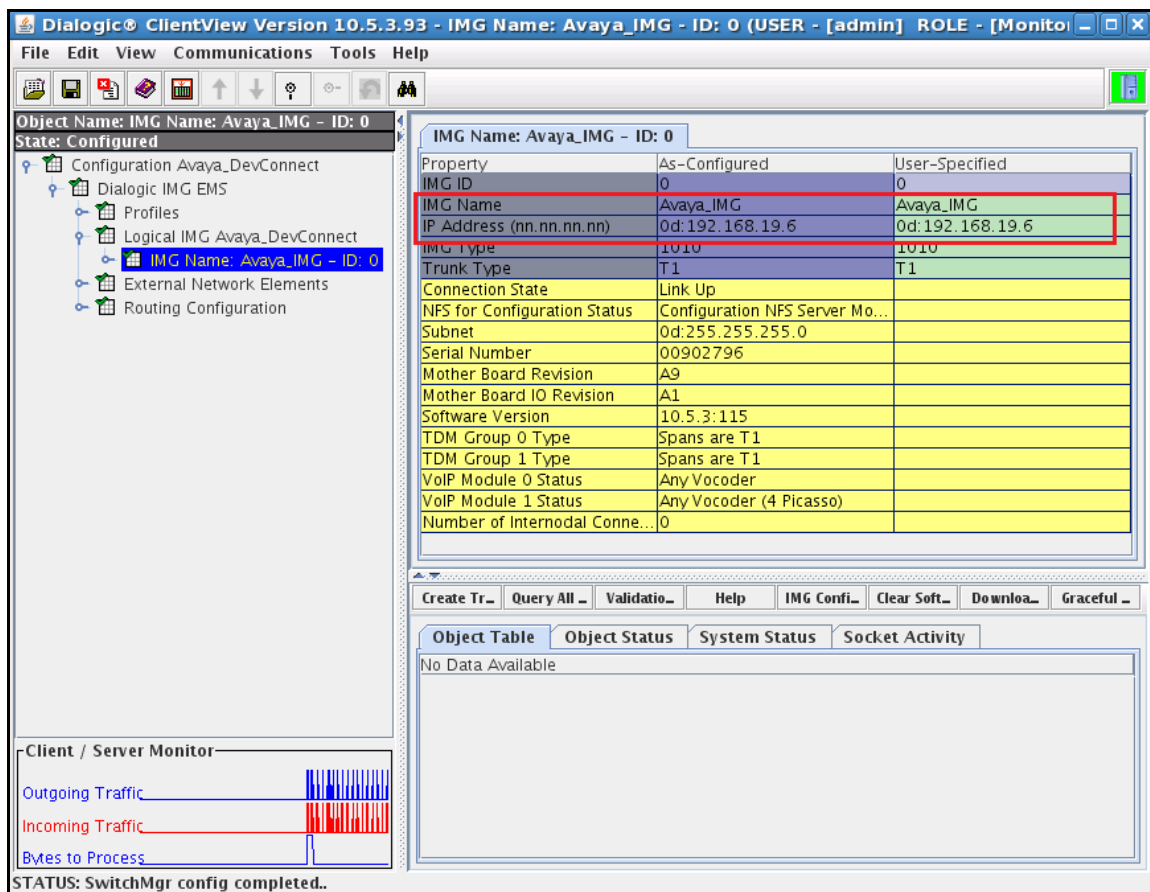


### 8.2.4.1 Physical IMG

Create a physical IMG as follows:

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

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

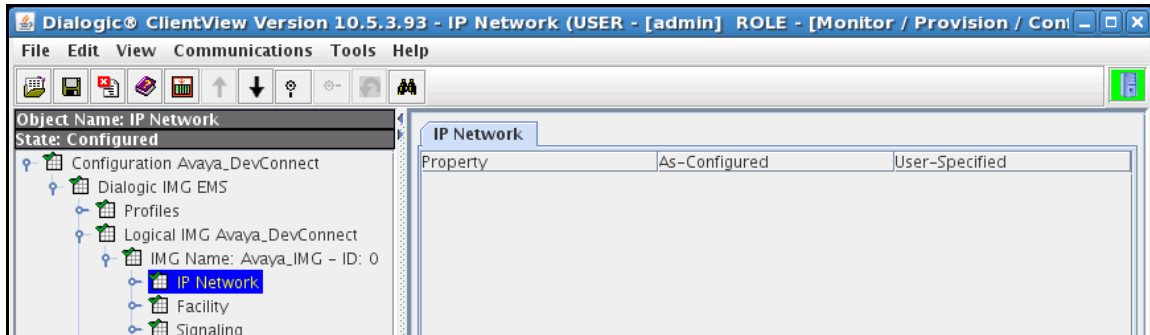


### 8.2.4.1.1 Network Interfaces

Create an object for Network Interfaces as follows:

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

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



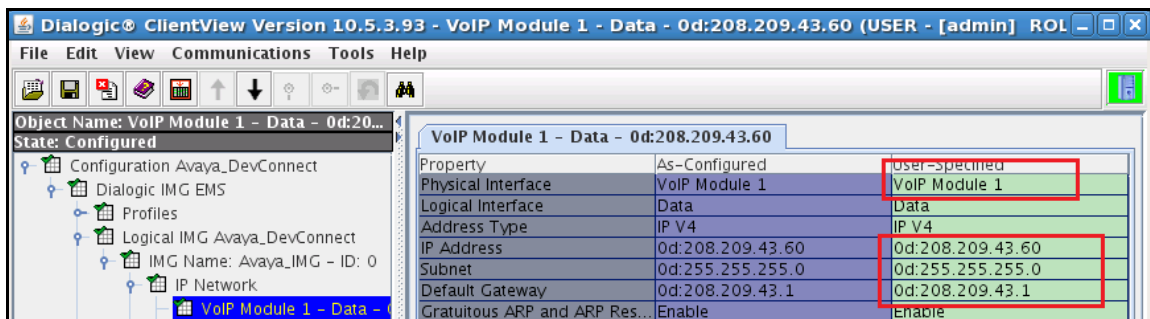
### 8.2.4.1.2 Network Interface VoIP module

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

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

To save the changes, right-click **VoIP Module 0: Port 0**, and select **Commit**. Repeat this step for VoIP module 1 if needed.

Note: this IP Address will be used for the media.



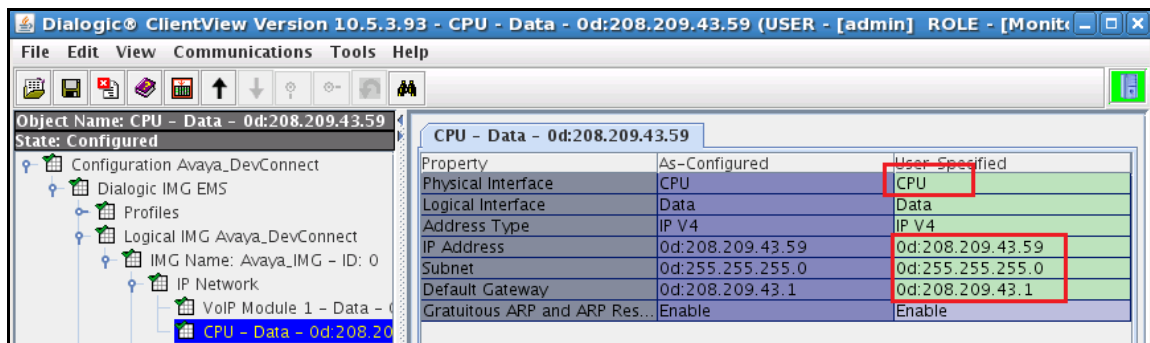
### 8.2.4.1.3 Network Interface CPU

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

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

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

Note: this IP Address will be used for SIP signalling with Session Manager.

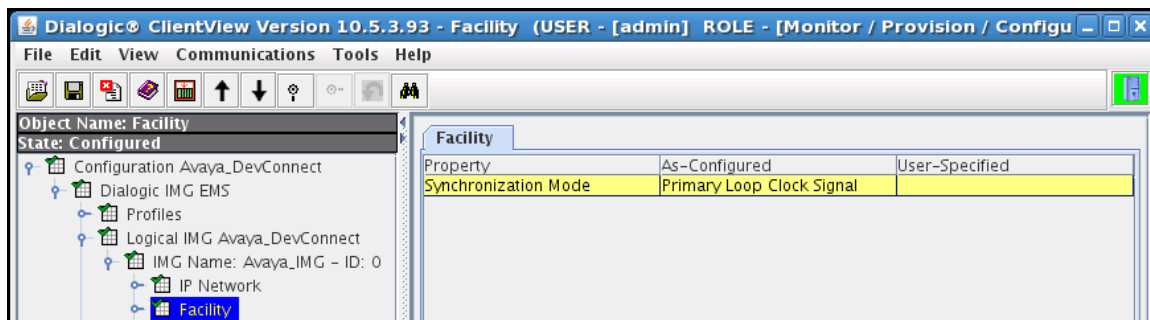


### 8.2.4.1.4 Facilities (DS1 and VoIP)

Create an object for a Facility as follows:

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

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





#### 8.2.4.1.5 VoIP Facilities

Configure VoIP Facilities as follows:

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

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

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

**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	

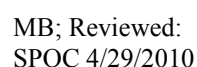
**Client / Server Monitor**

Outgoing Traffic  
Incoming Traffic  
Bytes to Process

STATUS: SwitchMgr config completed..

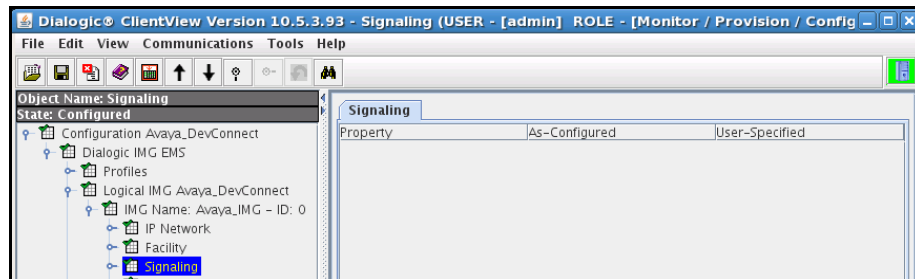
Configure a TDM DS1 T1 or E1 as follows:

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



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

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

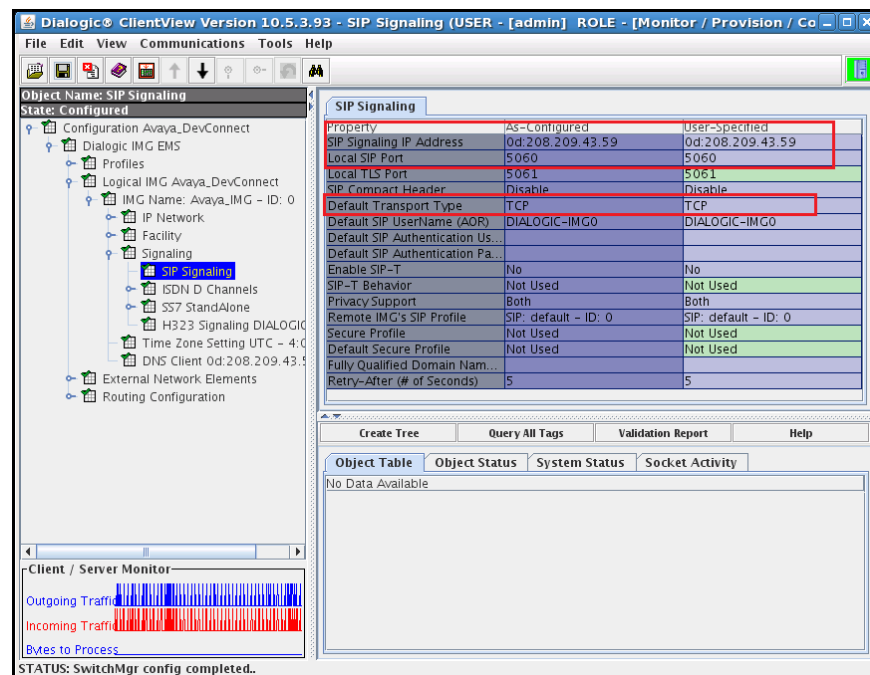


#### 8.2.4.1.8 SIP Signaling

Configure SIP Signaling as follows

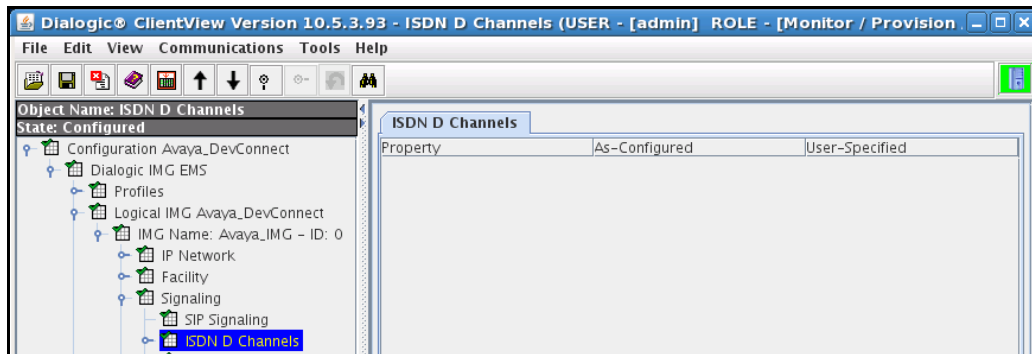
- Right-click **Signaling** in the Configuration Tree, and select **New SIP**.
- Administer settings in the Configuration Pane that enable SIP connectivity between the IMG and other SIP User Agents as follows:
  - Enter the IP address assigned to the IMG in the **SIP Signaling IP Address** field.
  -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

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



### 8.2.4.1.9 ISDN Signaling

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

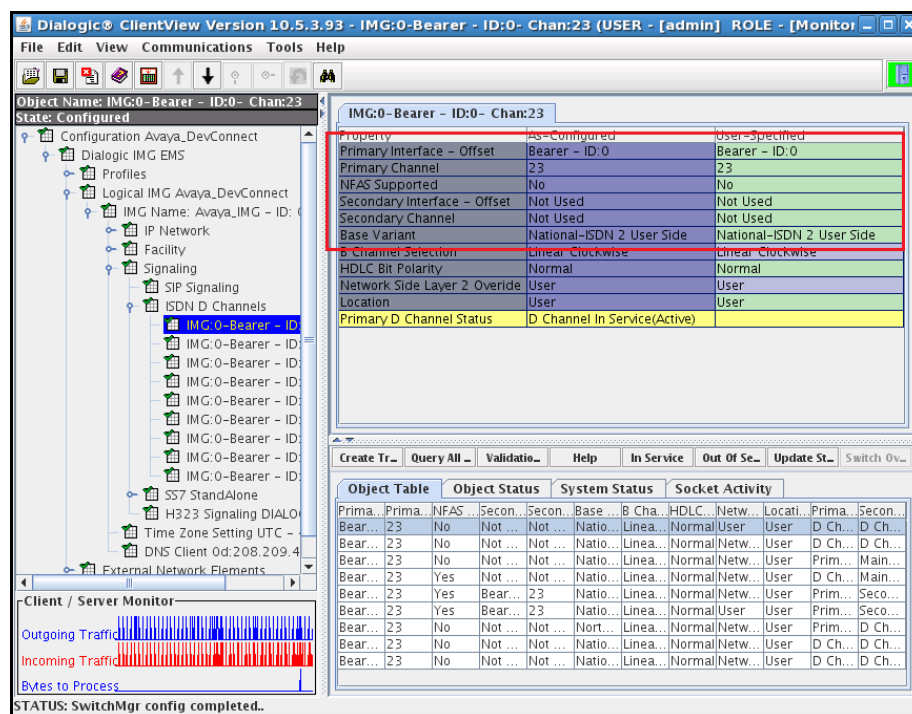


#### 8.2.4.1.9.1 ISDN D Channel

Configure an ISDN D Channel as follows:

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

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

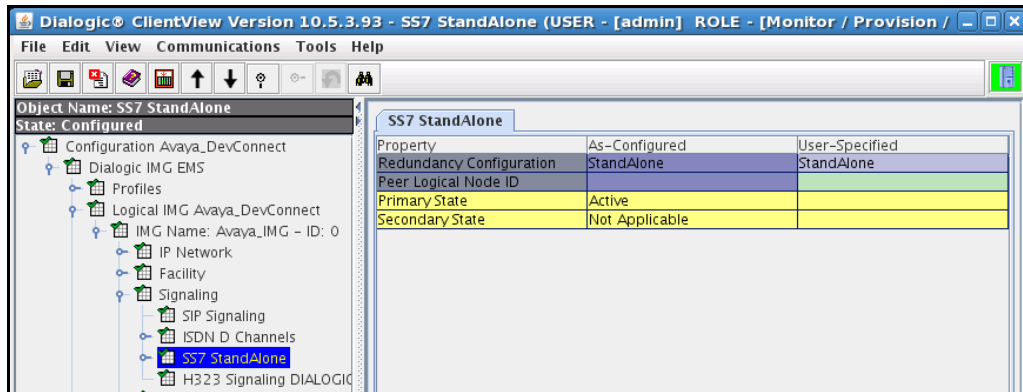


#### 8.2.4.1.10 SS7 Signaling

Configure SS7 Signaling as follows:

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

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

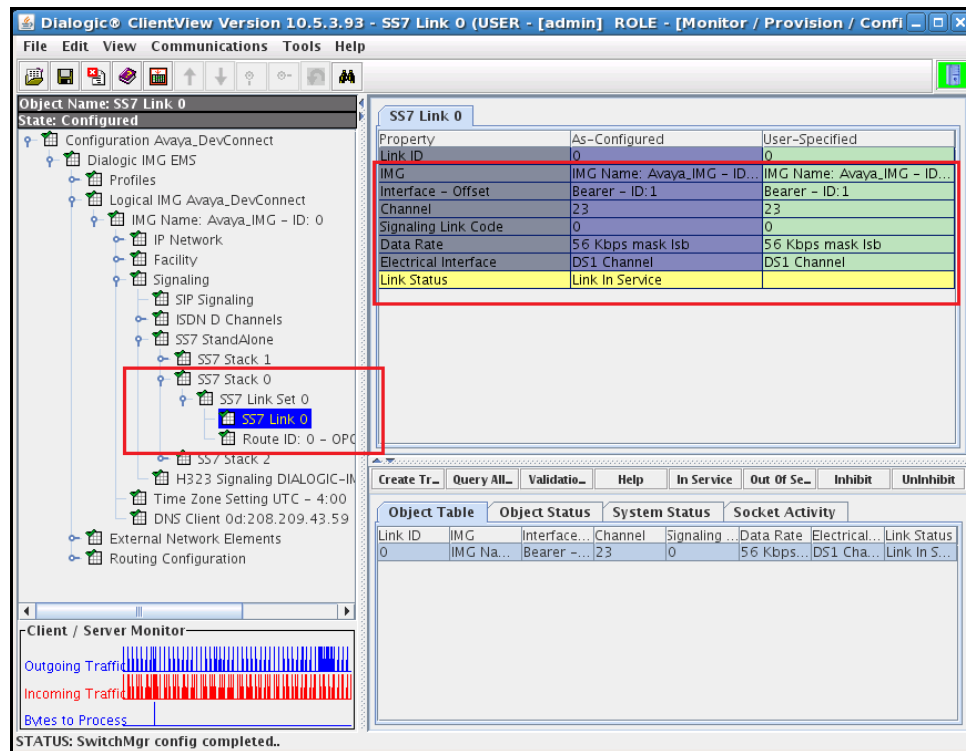


Configure SS7 Stack, Linkset, Links, & Routes as follows:

- 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**.



**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	

**Object Table**

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

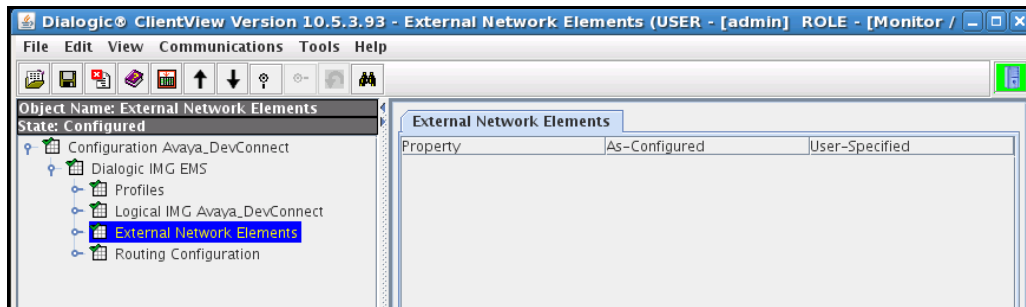
STATUS: SwitchMgr config completed..

### 8.2.5. External Network Elements

Create an object for External Network Elements as follows:

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

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

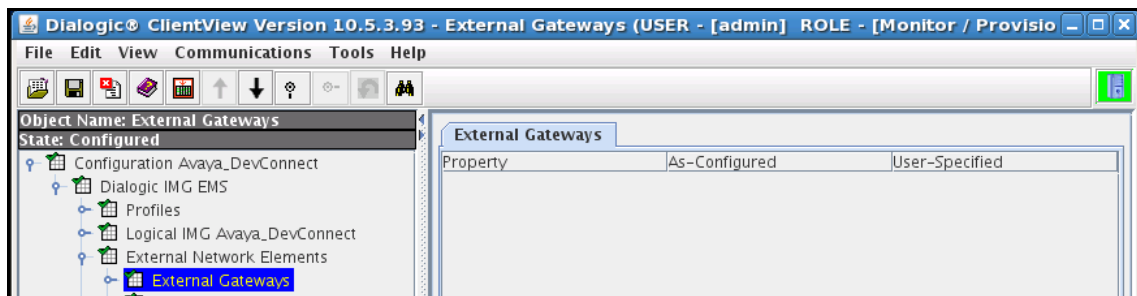


#### 8.2.5.1 External Gateways

Create an object for External Gateways as follows:

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

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

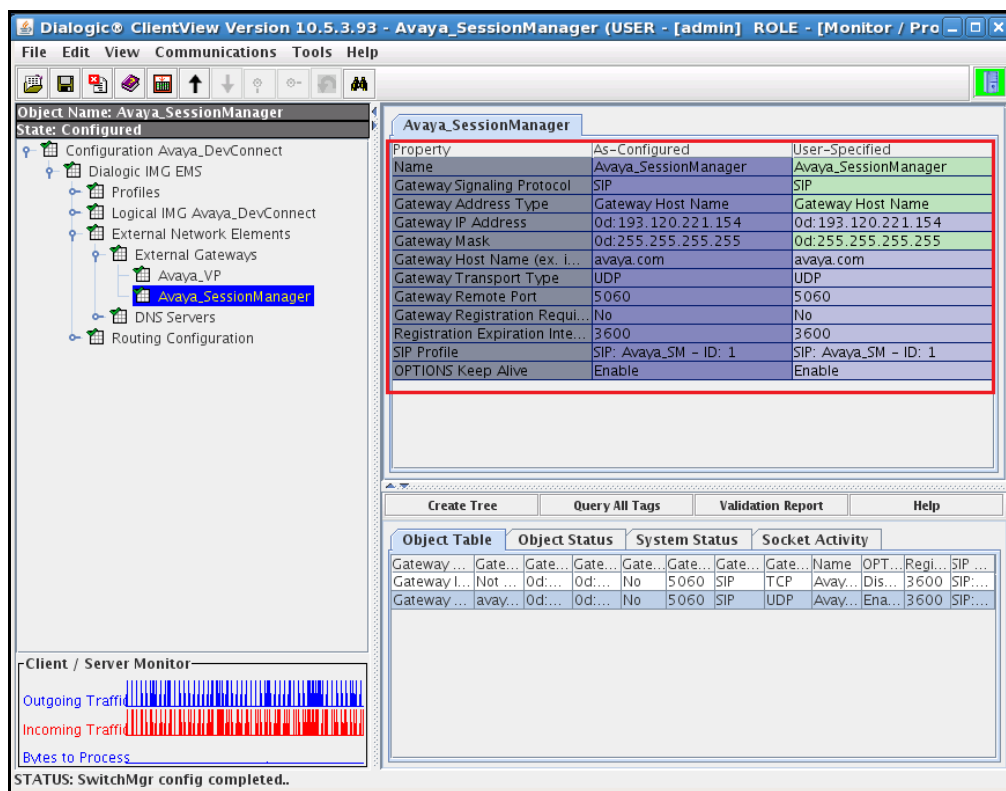


#### 8.2.5.1.1 External Gateway

## Configure an External Gateway:

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

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



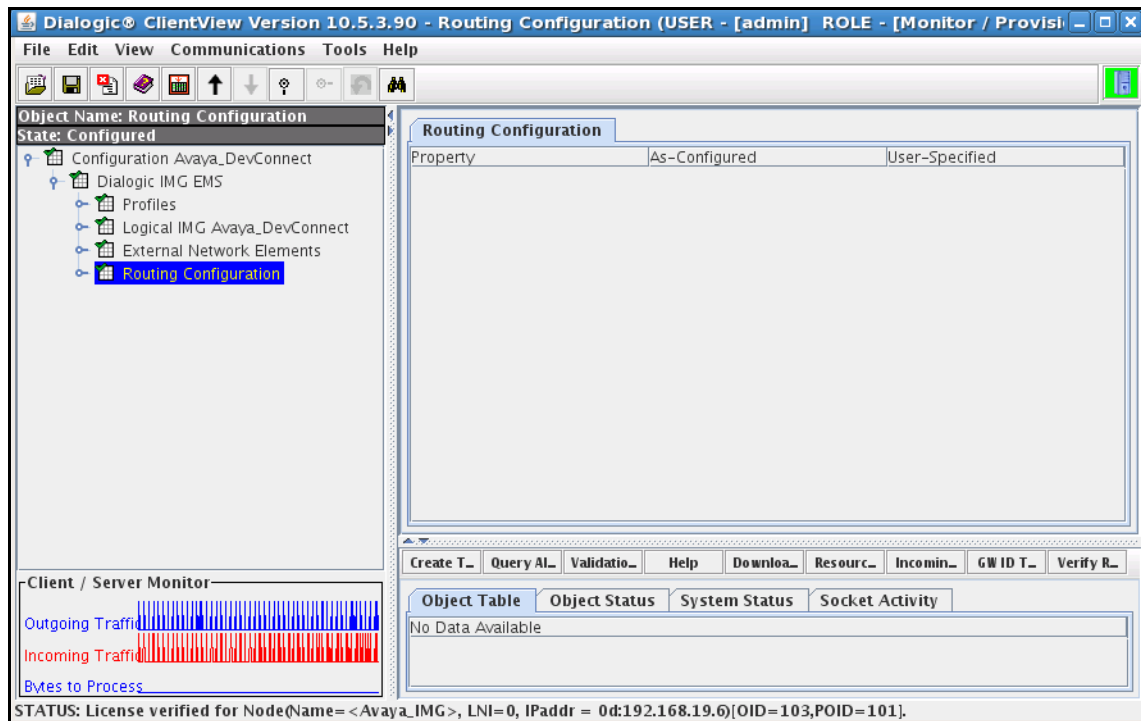


## 8.3. Routing configuration

Create an object for Routing Configuration as follows:

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

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

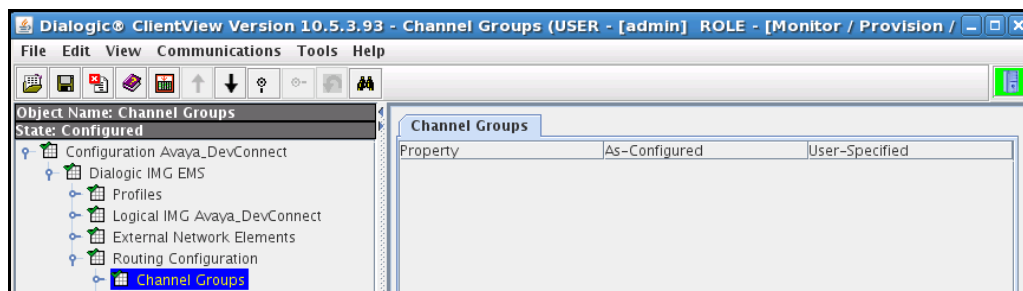


### 8.3.1. Channel Groups

Create an object for Channel Groups as follows:

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

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

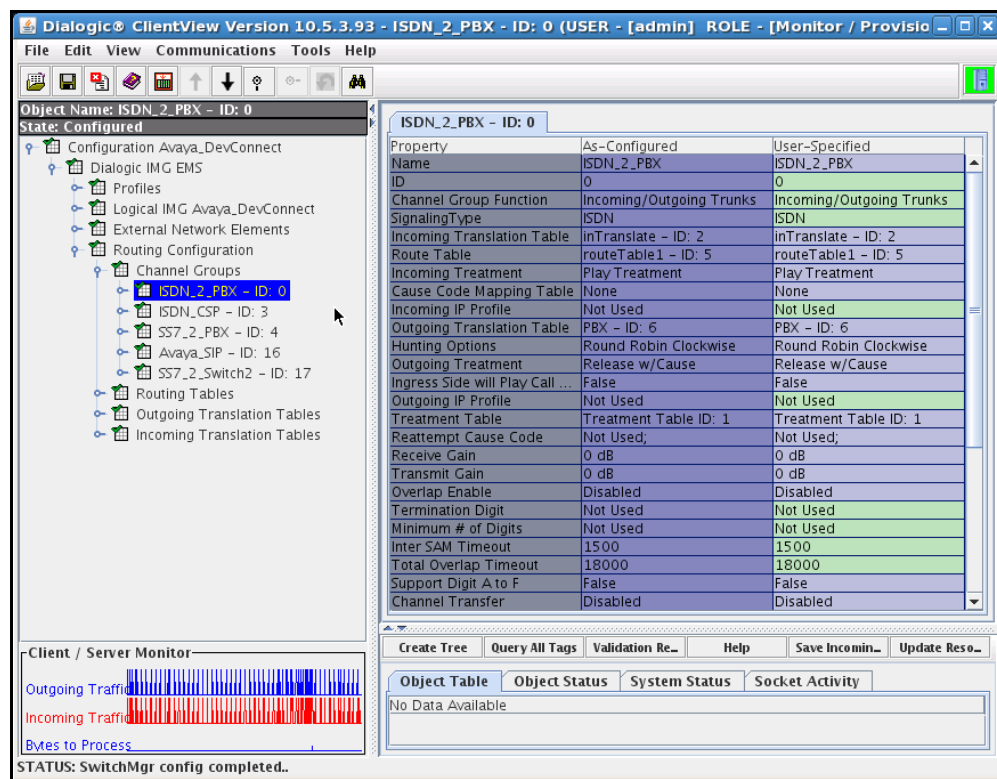


### 8.3.1.1 Channel Group (ISDN)

Configure an ISDN Channel Group:

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

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

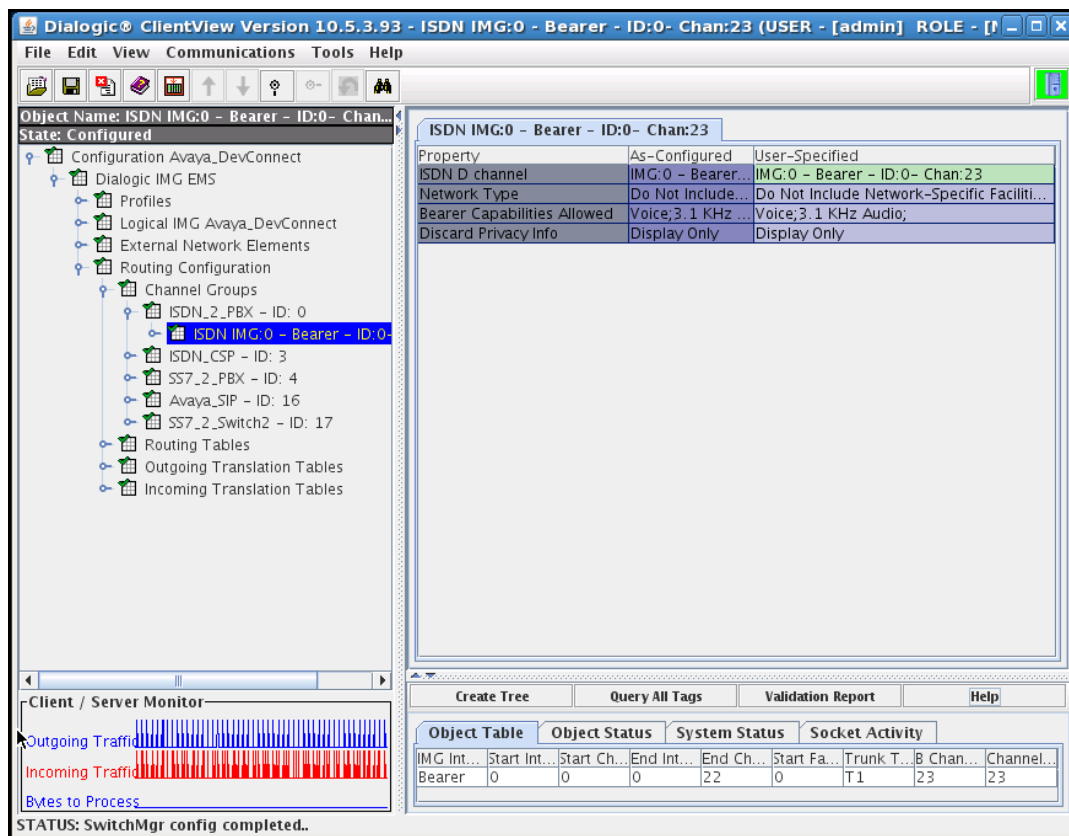


#### 8.3.1.1.1 Assign ISDN D channel to Channel group

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

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

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

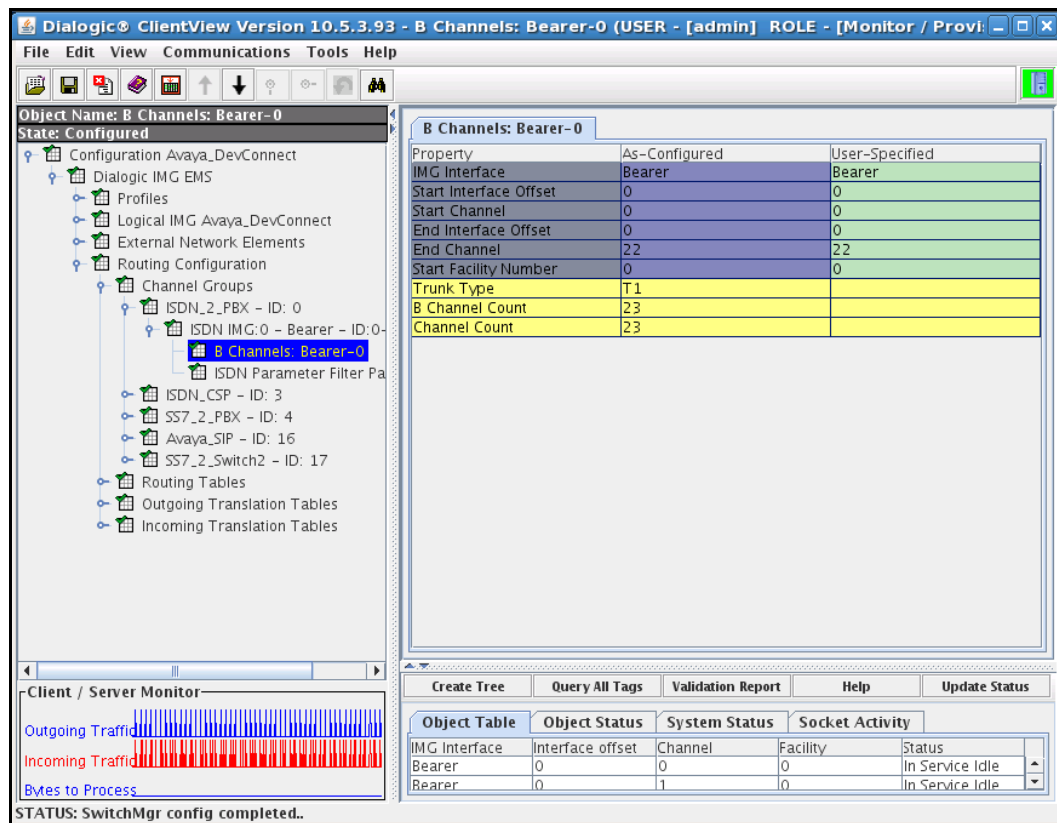


#### 8.3.1.1.2 Assign ISDN B channels to the ISDN Group

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

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

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

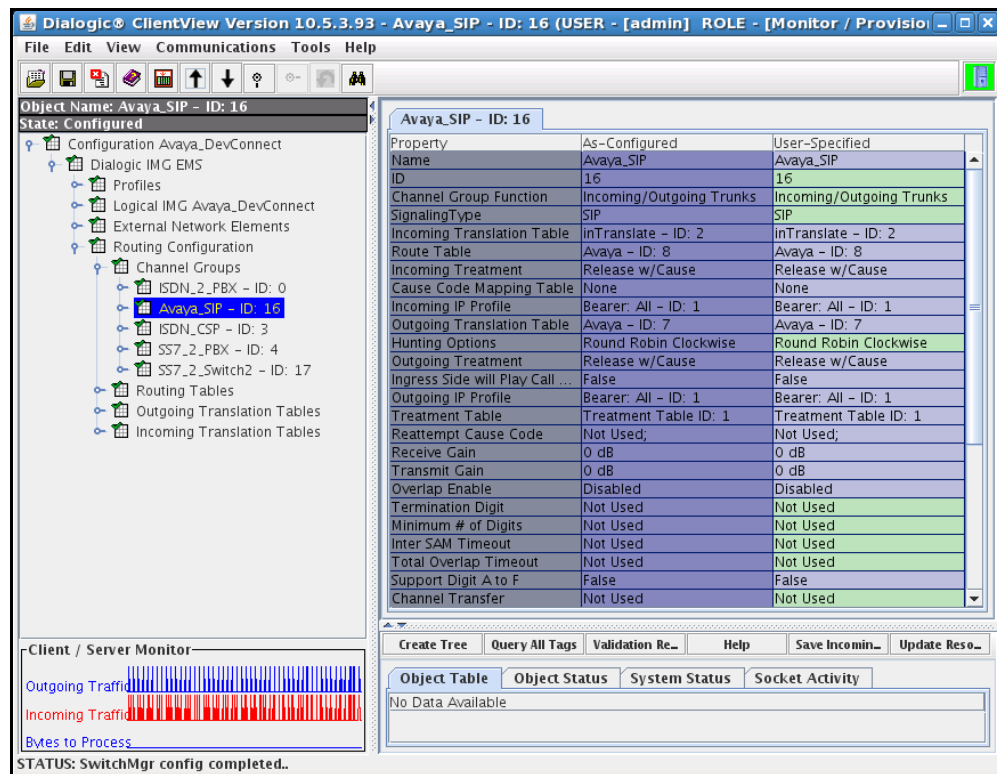


### 8.3.1.2 Channel Group (SIP)

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

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

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



### 8.3.1.3 Channel Group with SIP Gateway

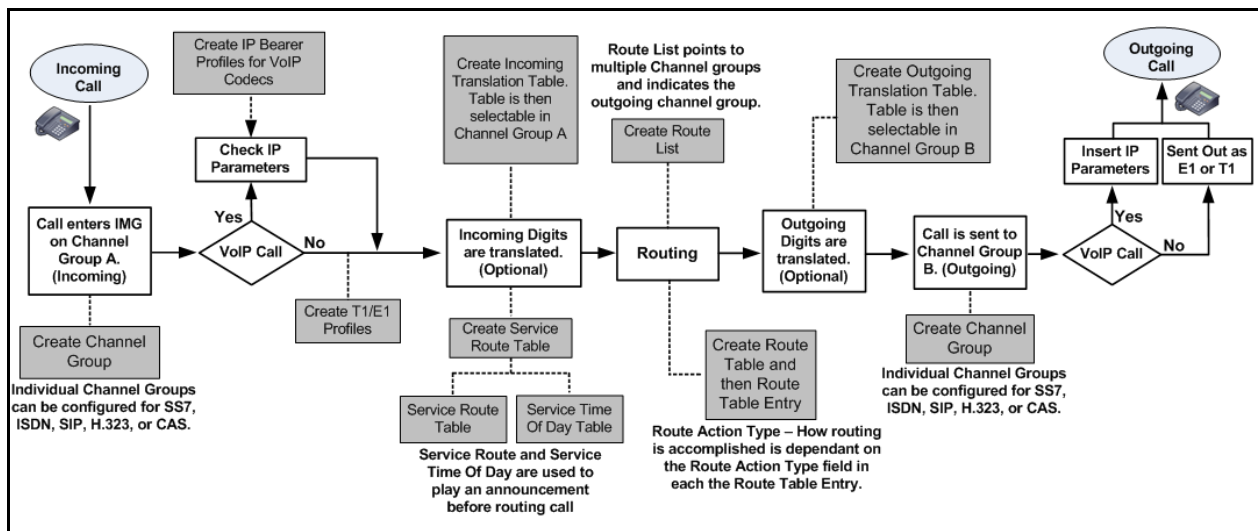
Assign a SIP Gateway to the Channel Group corresponding to each External Gateway previous:

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

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

## 8.4. Routing

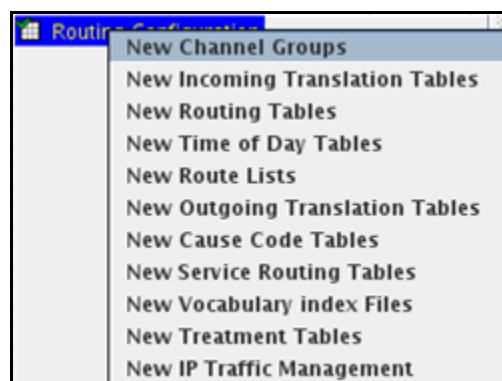
The following diagram shows the process the IMG goes through to route a call, and the ClientView GUI panes you use to configure various routing elements. Grey boxes represent ClientView configuration panes.



Create an object for Routing Tables as follows:

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

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



### 8.4.1. Route Entry

Add route entries to the Route Table as follows:

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

**Note:** This is displayed below under the **Route Action List** column.

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

### 8.5. Additional configuration

Other optional configuration objects can be configured as needed.

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

## 9. Verification Steps

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

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

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

```
status signaling-group 3
```

```
STATUS SIGNALING GROUP
```

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

```
status signaling-group 4
```

```
STATUS SIGNALING GROUP
```

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



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

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

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

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

## 9.3. SIP Monitoring on Avaya Aura™ Session Manager

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

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) → **System Status** (highlighted) → **SIP Entity Monitoring** (highlighted). The main content area is titled "SIP Entity Link Monitoring Status Summary" and includes a "Refresh" button. Below this is a table showing the "Entity Link Status for All Session Manager Instances".

Session Manager Name	Entity Links Down/Total	Entity Links Partially Down	SIP Entities - Monitoring Not Started	SIP Entities - Not Monitored
<a href="#">SessionManager</a>	0/4	0	0	0

Below the table, there is a section titled "All Monitored SIP Entities" with another "Refresh" button and a list of 4 items. The list includes the following SIP Entity Names:

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

## 9.4. Voice Portal Monitoring – System Monitor

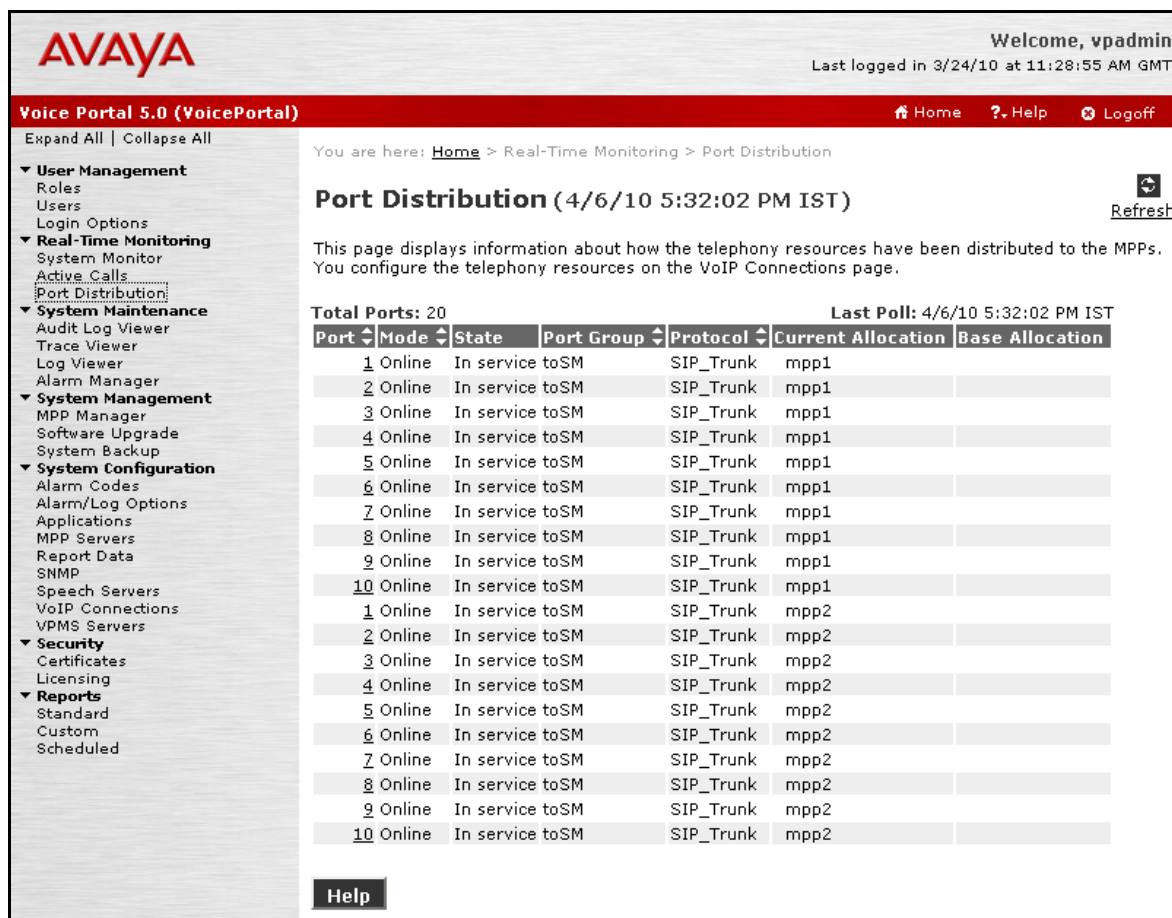
From the VPMS web interface, verify that for the MPP servers the **Mode** and **State** are **Online** and **Running**, and there are no **Alarms** associated with VP system in the **System Monitor** page shown below.

The screenshot displays the Avaya Voice Portal 5.0 (VoicePortal) interface. The top navigation bar includes the Avaya logo, a welcome message for 'vpadmin', and the last login time. The left sidebar contains a tree view of system management options, with 'System Monitor' highlighted under 'Real-Time Monitoring'. The main content area shows the 'System Monitor' page for 4/6/10 at 5:34:17 PM IST. A description states that the page displays the current state of the local Voice Portal system plus any remote systems. Below this, there are tabs for 'Summary' and 'VoicePortal Details'. The 'Summary' tab is active, showing a table of servers. The table has columns for Server Name, Type, Mode, State, Config, Call Capacity (Current, Licensed, Maximum), Active Calls (In, Out), Calls Today, and Alarms. The servers listed are VPMS, mpp1, mpp2, and a Summary row. The MPP servers are in 'Online Running' state with 'OK' configuration and no alarms. The 'Alarms' column shows green checkmarks for all servers, indicating no active alarms.

Server Name	Type	Mode	State	Config	Call Capacity	Active Calls	Calls Today	Alarms			
					Current	Licensed	Maximum	In	Out		
VPMS	VPMS										
mpp1	MPP	Online	Running	OK	10	10	10	0	0	4	
mpp2	MPP	Online	Running	OK	10	10	10	0	0	6	
Summary	VP				20	20	20	0	0	10	

## 9.5. Voice Portal Monitoring – Port Distribution

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



**AVAYA** Welcome, vpadmin  
Last logged in 3/24/10 at 11:28:55 AM GMT

**Voice Portal 5.0 (VoicePortal)** Home ? Help Logoff

Expand All | Collapse All

- ▼ **User Management**
  - Roles
  - Users
  - Login Options
- ▼ **Real-Time Monitoring**
  - System Monitor
  - Active Calls
  - Port Distribution
- ▼ **System Maintenance**
  - Audit Log Viewer
  - Trace Viewer
  - Log Viewer
  - Alarm Manager
- ▼ **System Management**
  - MPP Manager
  - Software Upgrade
  - System Backup
- ▼ **System Configuration**
  - Alarm Codes
  - Alarm/Log Options
  - Applications
  - MPP Servers
  - Report Data
  - SNMP
  - Speech Servers
  - VoIP Connections
  - VPMS Servers
- ▼ **Security**
  - Certificates
  - Licensing
- ▼ **Reports**
  - Standard
  - Custom
  - Scheduled

You are here: [Home](#) > Real-Time Monitoring > Port Distribution

### Port Distribution (4/6/10 5:32:02 PM IST)

This page displays information about how the telephony resources have been distributed to the MPPs. You configure the telephony resources on the VoIP Connections page.

Total Ports: 20 Last Poll: 4/6/10 5:32:02 PM IST

Port	Mode	State	Port Group	Protocol	Current Allocation	Base Allocation
1	Online	In service toSM	SIP_Trunk	mpp1		
2	Online	In service toSM	SIP_Trunk	mpp1		
3	Online	In service toSM	SIP_Trunk	mpp1		
4	Online	In service toSM	SIP_Trunk	mpp1		
5	Online	In service toSM	SIP_Trunk	mpp1		
6	Online	In service toSM	SIP_Trunk	mpp1		
7	Online	In service toSM	SIP_Trunk	mpp1		
8	Online	In service toSM	SIP_Trunk	mpp1		
9	Online	In service toSM	SIP_Trunk	mpp1		
10	Online	In service toSM	SIP_Trunk	mpp1		
1	Online	In service toSM	SIP_Trunk	mpp2		
2	Online	In service toSM	SIP_Trunk	mpp2		
3	Online	In service toSM	SIP_Trunk	mpp2		
4	Online	In service toSM	SIP_Trunk	mpp2		
5	Online	In service toSM	SIP_Trunk	mpp2		
6	Online	In service toSM	SIP_Trunk	mpp2		
7	Online	In service toSM	SIP_Trunk	mpp2		
8	Online	In service toSM	SIP_Trunk	mpp2		
9	Online	In service toSM	SIP_Trunk	mpp2		
10	Online	In service toSM	SIP_Trunk	mpp2		

Help

## 9.6. Functional Verification

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

## 10. General Test Approach

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

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

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

### 10.1. Test Results and Remarks

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

## 11. Conclusion

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

## 12. Additional References

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

- [1] “Avaya Aura™ Session Manager Overview”, Document Number 03-603323, Issue 2, Release 5.2, November 2009
- [2] “Installing and Upgrading Avaya Aura™ Session Manager”, Document Number 03-603473, Issue 2, Release 5.2, November 2009
- [3] “Administering Avaya Aura™ Session Manager”, Document Number 03-603324, Issue 2, Release 5.2, November 2009
- [4] “Avaya Aura™ Session Manager Case Studies”, Document Number 03-603478, Issue 2, Release 5.2, November 2009
- [5] “Maintaining and Troubleshooting Avaya Aura™ Session Manager, Document Number 03-603325, Issue 2, Release 5.2, November 2009
- [6] “Installing and Configuring Avaya Aura™ System Platform”, Release 1.1, November 2009
- [7] “Installing and Upgrading Avaya Aura™ System Manager”, Release 5.2, January 2010
- [8] “Avaya Aura™ Communication Manager Overview”, Document Number 03-300468, Issue 6, Release 5.2, May 2009
- [9] “Administering Avaya Aura™ Communication Manager”, Document Number 03-300509, Issue 5.0, Release 5.2, May 2009
- [10] “Avaya Aura™ Communication Manager Feature Description and Implementation”, Document Number 555-245-205, Issue 7.0, Release 5.2, May 2009
- [11] “Administering Network Connectivity on Avaya Aura™ Communication Manager”, Document Number 555-233-504, Issue 14, May 2009
- [12] “SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers”, Document Number 555-245-206, Issue 9, May 2009
- [13] “Administering Avaya Aura™ Communication Manager as a Feature Server”, Document Number 03-603479, Issue 1.2, Release 5.2, January 2010
- [14] “Configuring 9600-Series SIP Phones with Avaya Aura™ Session Manager Release 5.2 – Issue 1.0”, Application Note
- [15] “Administering Voice Portal”, Release 5.0, March 2009
- [16] “Implementing Voice Portal on multiple servers”, Release 5.0, March 2009
- [17] “Configuring Avaya Aura™ Session Manager 5.2 with Avaya Aura™ Communication Manager Access Element, Avaya Voice Portal and Avaya Aura™ Communication Manager Feature Server – Issue 1.0”. Application Note

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

- [18] “IMG 1010 - Quick Start Guide” Doc ID 07-728-05, March 2010
- [19] “IMG 1010/1004 Integrated Media Gateway Upgrading System Software”, November 2009 Application Note
- [20] “Dialogic® IMG 1010 Integrated Media Gateway WebHelp” Release: 10.5.3 ER1

## 13. APPENDIX

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

### 13.1. Web application server intro.vxml file

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

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

<form id="form0">

    <field name="test_type">

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

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

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

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

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

</filled>

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

</field>

</form>
</vxml>
```

## 13.2. Web application server testblindtransfer.vxml file

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

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



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

### 13.3. Web application server testbridgetransfer.vxml file

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

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

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

        <field name="phone_number">

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

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

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

        </field>

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

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

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

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

```

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

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

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

                        </transfer>

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

```

## 13.4. Web application server testconsulttransfer.vxml file

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

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

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

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

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

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

```

        </if>
    </filled>

</transfer>

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

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

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

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

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

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

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

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

</form>
</vxml>

```

---

**©2010 Avaya Inc. All Rights Reserved.**

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

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