



Avaya Solution & Interoperability Test Lab

Application Notes for Dialogic IMG 2020 Release 2.2 with Avaya Aura® Communication Manager 7.0 using Avaya Aura® Session Manager 7.0 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Dialogic IMG 2020 Gateway with Avaya Aura® Communication Manager using Avaya Aura® Session Manager. IMG 2020 is a SIP to ISDN gateway that connects to Avaya Aura® Session Manager as a SIP entity.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

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

1. Introduction

These Application Notes describe the configuration steps required to successfully integrate Dialogic IMG 2020 Gateway with Avaya Aura® Communication Manager using Avaya Aura® Session Manager. IMG 2020 is a SIP to ISDN gateway that connects to Avaya Aura® Session Manager as a SIP entity. IMG 2020 allows SIP calls to be routed out to service providers over ISDN trunks.

2. General Test Approach and Test Results

The general test approach was to configure the IMG 2020 server to communicate with Communication Manager via a SIP Trunk connected to Session Manager. Endpoints present on Communication Manager were used to make and receive calls via the IMG 2020 gateway.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test included both feature functionality and serviceability testing. The feature functionality testing focused on call scenarios using supported features and codecs with good quality audio. The tests included:

- IMG 2020 SIP trunk is connected and in service.
- IMG 2020 can route calls to SIP, Digital and H.323 endpoints.
- IMG 2020 can route calls from SIP, Digital and H.323 endpoints.
- IMG 2020 can use the G.711, G.722 and G.729 codecs for call routing.
- IMG 2020 can route fax calls using T.38
- Failover/Service – Tests the behaviour of IMG 2020 during certain failure conditions.

2.2. Test Results

All test cases were passed.

2.3. Support

Technical support for customers with a Dialogic support contract:

2.3.1. Telephone Support:

Americas: 781-433-9770 (8am - 8pm EST).
Asia Pacific: +91 20 66211611
Europe, Middle East and Africa: +44 1628 641 799 (9am - 5pm UK time)
Note: Extended hours may be available depending on your support contract.

2.3.2. Online Support

Helpweb: [Click here for technical notes and other support information.](#)
Online support: [Visit the DSC Online Service page.](#)
Discussion Forums: [To login or join, click here.](#)

2.3.3. Customers who do not have a direct support agreement with Dialogic

Helpweb: [Click here for technical notes and other support information.](#)
Contact your Dialogic reseller or local Dialogic office for information on how to obtain support.

3. Reference Configuration

The configuration shown in **Figure 1** was used during the compliance test of Dialogic IMG 2020 SIP to ISDN gateway with Communication Manager via Session Manager. IMG 2020 utilizes a SIP trunk to communicate with the Communication Manager handsets.

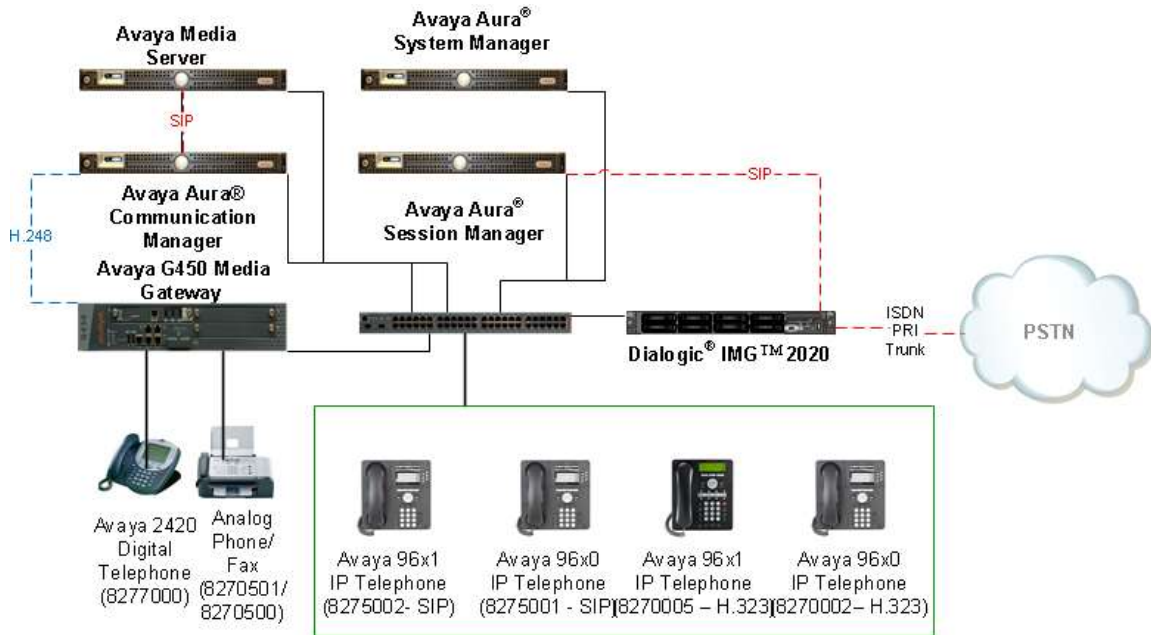


Figure 1: Connection of IMG 2020 with Avaya Aura® Communication Manager via Session Manager

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on VMware Virtual Machine	R7 SP1 R017.00.0.441.22438
Avaya Aura® Session Manager running on VMware Virtual Machine	R7.0 7.0.0.0.700007
Avaya Aura® System Manager	R7.0 7.0.0.0.1626-7.0.9.912
Avaya Media Server	R7.7 7.7.0.226 A11
Avaya G430 Media Gateway	FW 37.19.0
Avaya 96x1 Series IP Deskphones H.323	6.6.0.29
Avaya 96x1 Series IP Deskphones SIP	6.5.0
Dialogic IMG 2020	Release 2.2 SP4

5. Configure Avaya Aura® Communication Manager

This section describes the steps required to allow Communication Manager to communicate with IMG 2020. It is assumed that Communication Manager is installed and configured before implementing the configuration steps below. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 11**. The configuration illustrated in this section was performed using Communication Manager System Administration Terminal (SAT).

Configuration steps include:

- Check SIP Trunk Licensing.
- Add entries in the Dial Plan for use with SIP Trunk and routing via IMG 2020.
- SIP Trunk Administration (to Session Manager).
- Adding Route Pattern.

Using the *display system-parameters customer-options* command go to **page 2** and check that the system is sufficiently licensed for SIP Trunks.

<code>display system-parameters customer-options</code>		Page 2 of 11
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
	Maximum Administered H.323 Trunks:	12000 0
	Maximum Concurrently Registered IP Stations:	18000 3
	Maximum Administered Remote Office Trunks:	12000 0
	Maximum Concurrently Registered Remote Office Stations:	18000 0
	Maximum Concurrently Registered IP eCons:	414 0
	Max Concur Registered Unauthenticated H.323 Stations:	100 0
	Maximum Video Capable Stations:	41000 0
	Maximum Video Capable IP Softphones:	18000 0
	Maximum Administered SIP Trunks:	24000 10

Use the *change node-names ip* command to add the Session Manager. An entry for **SM1677** was added.

```
change node-names ip                                     Page 1 of 2
                                     IP NODE NAMES
Name                               IP Address
SM1677                          10.10.16.77
procr                             10.10.16.27
procr6                             :-
```

Use *change dialplan analysis* to add a **3** digit dial access code(**dac**) for use in the SIP Trunk, a uniform dial plan(**udp**) entry for calling out over the SIP Trunk and check that there is an entry for feature access codes(**fac**).

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

Dialed   Total Call   Dialed   Total Call   Dialed   Total Call
String   Length Type     String   Length Type     String   Length Type
2        7    udp      7        3    dac
8        5    udp    8        7    udp
8        7    udp    827     7    ext
9        1    fac    *        3    fac
#        3    fac
```

Use *add-signaling-group x* where x is the number of the group required. Set **Transport Method** to **tcp**, **Near-end Node Name** to **procr** and **Far-end Node Name** to the entry added in **node-names**. Set the **Far-end Network Region** to **1** and **Direct IP-IP Audio Connections?** to **n**.

```
add signaling-group 76                               Page 1 of 2
                                     SIGNALING GROUP

Group Number: 76                               Group Type: sip
IMS Enabled? n                               Transport Method: tcp
Q-SIP? n
IP Video? n                               Enforce SIPS URI for SRTP? y
Peer Detection Enabled? y Peer Server: SM
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
Near-end Node Name: procr                   Far-end Node Name: SM1677
Near-end Listen Port: 5060                   Far-end Listen Port: 5060
Far-end Network Region: 1

Far-end Domain:
Incoming Dialog Loopbacks: eliminate         Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload                   RFC 3389 Comfort Noise? n
Direct IP-IP Audio Connections? n
Session Establishment Timer(min): 3         IP Audio Hairpinning? n
Enable Layer 3 Test? y
Alternate Route Timer(sec): 6
```

Use **add trunk-group x** where x is the number administered for the signaling group. on **Page 1** set the **Group Type** to **sip**. Set the **TAC** to suitable entry based on the dial plan **dac** administered above. Set the **Service Type** to **tie**, **Signaling group** to the one administered above and **Number of Members** to a number satisfactory for call routing required (**255** shown is the max for this type of trunk group).

```

add trunk-group 76                                     Page 1 of 21
                                     TRUNK GROUP
Group Number: 76                                     Group Type: sip          CDR Reports: y
Group Name: ToSM7                                   COR: 1                  TN: 1          TAC: 776
Direction: two-way                                 Outgoing Display? n
Dial Access? n                                     Night Service:
Queue Length: 0
Service Type: tie                                   Auth Code? n
                                                Member Assignment Method: auto
                                                Signaling Group: 76
                                                Number of Members: 255

```

On **page 3** set the **Numbering Format**. For this test the **private** numbering table were used to set the calling party number format.

```

add trunk-group 76                                     Page 3 of 21
TRUNK FEATURES
ACA Assignment? n                                   Measured: none
                                                Maintenance Tests? y

                                     Numbering Format: private
                                                UUI Treatment: service-provider
                                                Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                Hold/Unhold Notifications? y
Modify Tandem Calling Number: no

Show ANSWERED BY on Display? y

```


Next a route pattern needs to be added so that call can be routed out of Communication Manager to Session Manager. use *change route-pattern x* where x is the number of the SIP trunk created. Enter the Trunk group created above beside the first **Grp No**, an **FRL** of **0**.

```

change route-pattern 76                                     Page 1 of 3
      Pattern Number: 76      Pattern Name: ToSM7
  SCCAN? n      Secure SIP? n      Used for SIP stations? n

  Grp FRL NPA Pfx Hop Toll No.  Inserted      DCS/ IXC
  No          Mrk Lmt List Del  Digits      QSIG
                                     Dgts      Intw
1: 76  0
2:
3:
4:
5:
6:

      BCC VALUE  TSC CA-TSC      ITC BCIE Service/Feature PARM Sub  Numbering LAR
      0 1 2 M 4 W      Request      Dgts Format
1: y y y y y n  n      rest      none
  
```

An Alternate Route Selection (ars) entry must be made for dialing the external numbers that are to be routed via the IMG 2020 gateway. Use *change ars analysis x* where x is the first number in the dialed string. Set **Dialed String** to **x**, **Total Min/Max** to the length of the number to be dialed, **Route Pattern** to the one administered above and **Call Type** to **pubu**.

```

change aar analysis 8                                     Page 1 of 2
      ARS DIGIT ANALYSIS TABLE
      Location: all      Percent Full: 2

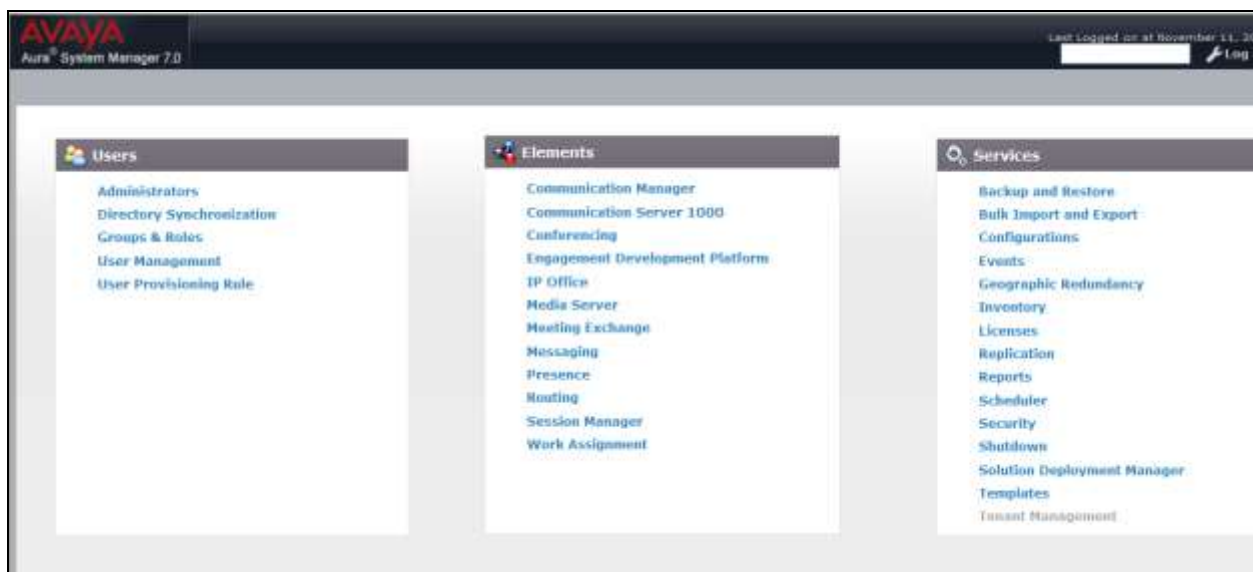
      Dialed      Total      Route      Call      Node      ANI
      String      Min Max  Pattern  Type     Num      Reqd
      01          10  11    76      pubu
      Customizable Labels? y
Extension      Prefix      Selection  Set      Mode
827-0999      EC500      -          88888    aar      3
  
```

6. Configure Avaya Aura® Session Manager

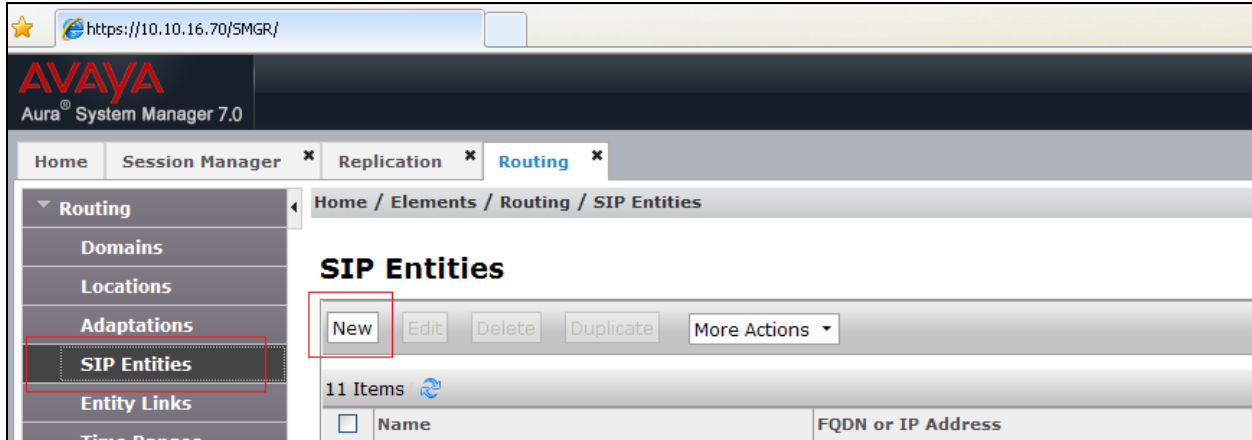
In this section the configuration steps required to connect IMG 2020 to Session Manager as a SIP entity are described. It is assumed that an existing Session Manager instance has already been installed and configured as this is out with the scope of this document. All Configuration steps were carried out using Avaya Aura® System Manager. Configuration steps will include:

- Adding an IMG 2020 SIP Entity.
- Adding an Entity Link.
- Adding a Routing Policy.
- Adding a Dial Pattern.

From the System Manager home screen select **Elements** → **Routing**.



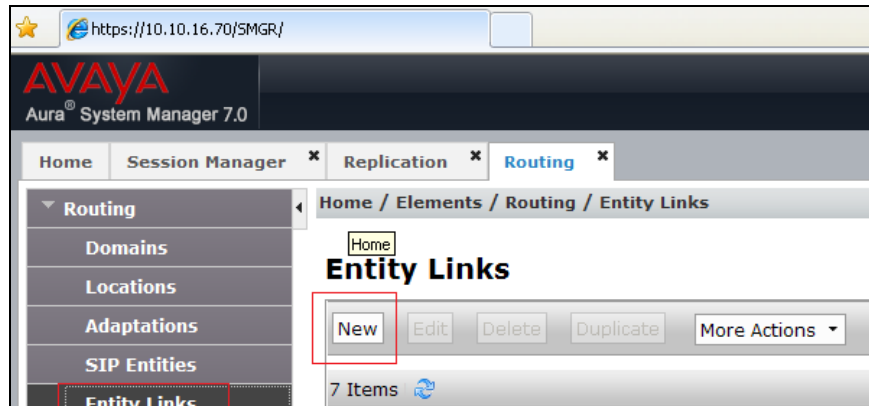
Select **SIP Entities** from the left hand menu and click on **New** to add the IMG 2020 entity.



Enter a descriptive **Name** and the **FQDN or IP Address** of the IMG 2020 gateway. Set **Type** as **SIP Trunk** and choose a **Time Zone** from the drop down menus. Click on **Commit** to save the changes.



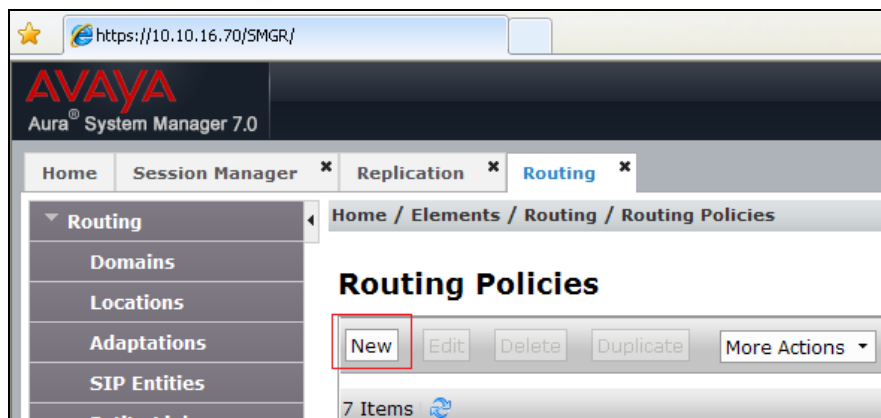
Next add an Entity link between the IMG 2020 and Session Manager entities. Select **Entity Links** from the left hand menu and click on **New**.



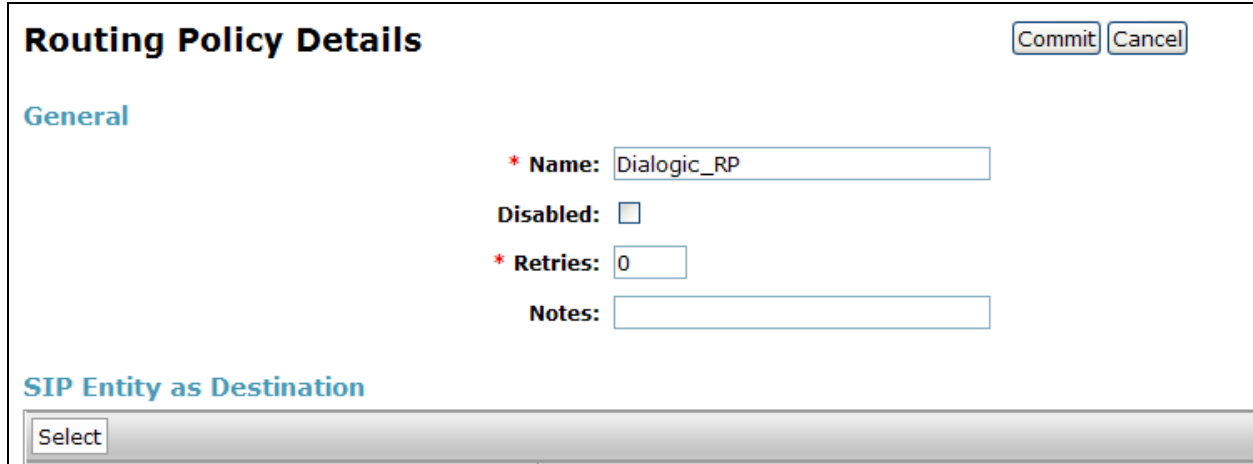
Enter a descriptive **Name** and then select the Session Manager as **SIP Entity 1** from the drop down. Select the **IMG 2020** entity as **SIP Entity 2**. Select the **Protocol** administered on the IMG 2020 server. **UDP** was used during testing. The ports will automatically change to the default **5060**. Click on Commit to save changes.



From the left hand menu select Routing Policies (not shown) and click on **New**.



Enter a descriptive **Name** and under **SIP Entity as Destination** click on **Select**.



Routing Policy Details Commit Cancel

General

* **Name:** Dialogic_RP

Disabled:

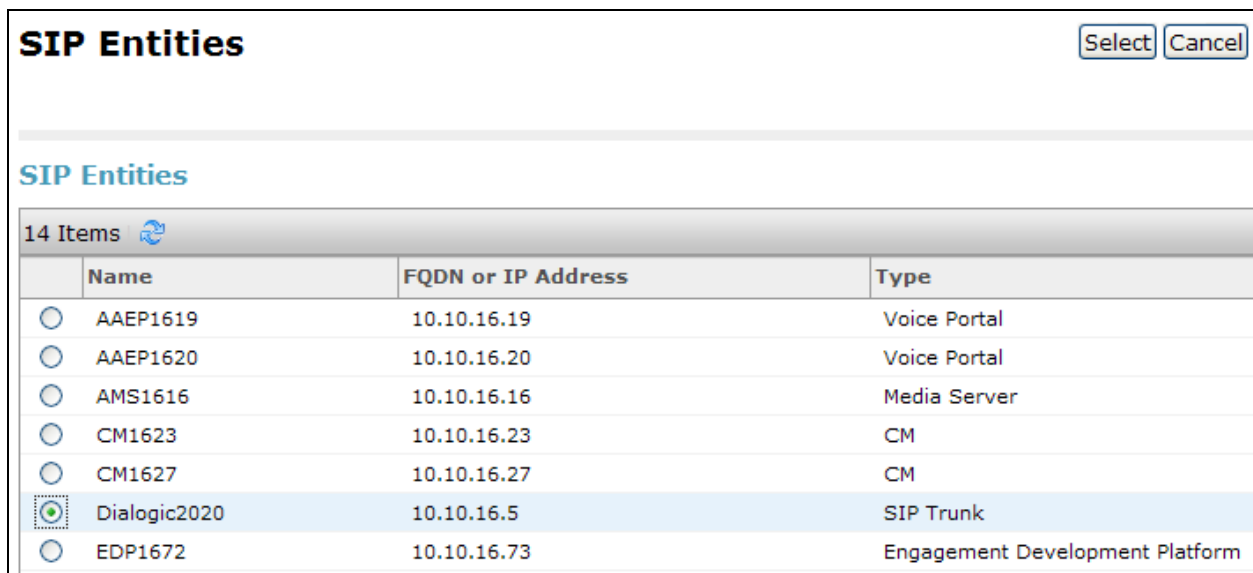
* **Retries:** 0

Notes:

SIP Entity as Destination

Select

From the list of **SIP Entities** select the **IMG 2020** entity and click on **Select** to save changes.



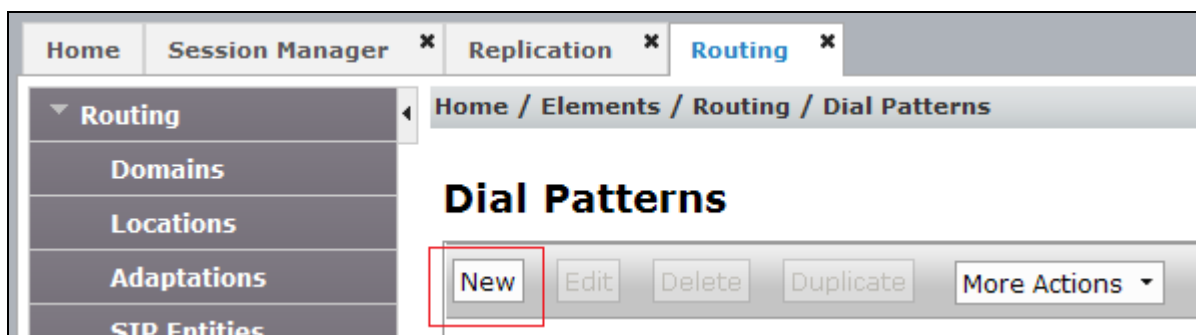
SIP Entities Select Cancel

SIP Entities

14 Items

	Name	FQDN or IP Address	Type
<input type="radio"/>	AAEP1619	10.10.16.19	Voice Portal
<input type="radio"/>	AAEP1620	10.10.16.20	Voice Portal
<input type="radio"/>	AMS1616	10.10.16.16	Media Server
<input type="radio"/>	CM1623	10.10.16.23	CM
<input type="radio"/>	CM1627	10.10.16.27	CM
<input checked="" type="radio"/>	Dialogic2020	10.10.16.5	SIP Trunk
<input type="radio"/>	EDP1672	10.10.16.73	Engagement Development Platform

From the left hand menu select **Dial Patterns** (not shown) and click on **New**.



Home Session Manager * Replication * Routing *

Home / Elements / Routing / Dial Patterns

Dial Patterns

New Edit Delete Duplicate More Actions

Enter the **Pattern** that will route calls to the IMG 2020 server and set the **Min** and **Max** to the length of the number to be dialed. Under **Originating Location and Routing Policies**, click on **Add**.

Dial Pattern Details Commit Cancel

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call:

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

Add Remove

Select **Apply the Selected Routing Policy to All Originating Locations** and under **Routing Policies** select the IMG 2020 Routing Policy added above.

Originating Location Select Cancel

Originating Location

Apply The Selected Routing Policies to All Originating Locations

1 Item

	Name	Not
<input type="checkbox"/>	Devconnect	

Select : All, None

Routing Policies

10 Items

	Name	Disabled	Destination
<input type="checkbox"/>	AAEP1619_RP	<input type="checkbox"/>	AAEP1619
<input type="checkbox"/>	AAEP1620_RP	<input type="checkbox"/>	AAEP1620
<input type="checkbox"/>	AMS1616_RP	<input type="checkbox"/>	AMS1616
<input type="checkbox"/>	CM1623_RP	<input type="checkbox"/>	CM1623
<input type="checkbox"/>	CM1627_RP	<input type="checkbox"/>	CM1627
<input checked="" type="checkbox"/>	Dialogic_RP	<input type="checkbox"/>	Dialogic2020

7. Configure Dialogic IMG 2020

For the compliance test, two trunking interfaces were configured on the IMG 2020. A SIP trunk interface was used to connect to Session Manager and an ISDN PRI interface was used to connect to the PSTN.

It is assumed that basic configuration of the IMG 2020 such as IP addresses, default gateways, and VLAN IDs, SIP signaling and media interfaces, serial number, security ID, and packet facility have been configured during installation.

It is also assumed that the PSTN trunk has been properly configured; this could include the ISDN PRI interface, TDM profile, associated channel group, or the SS7 signaling configuration, link sets, routes and associated channel group.

This section provides the procedures for configuring the IMG 2020, assuming it has been installed and licensed. The procedures include the following items:

- Launch Management Interface.
- Configure Signaling.
- Configure Profiles.
- Configure External Network Element.
- Configure Routing Configuration.

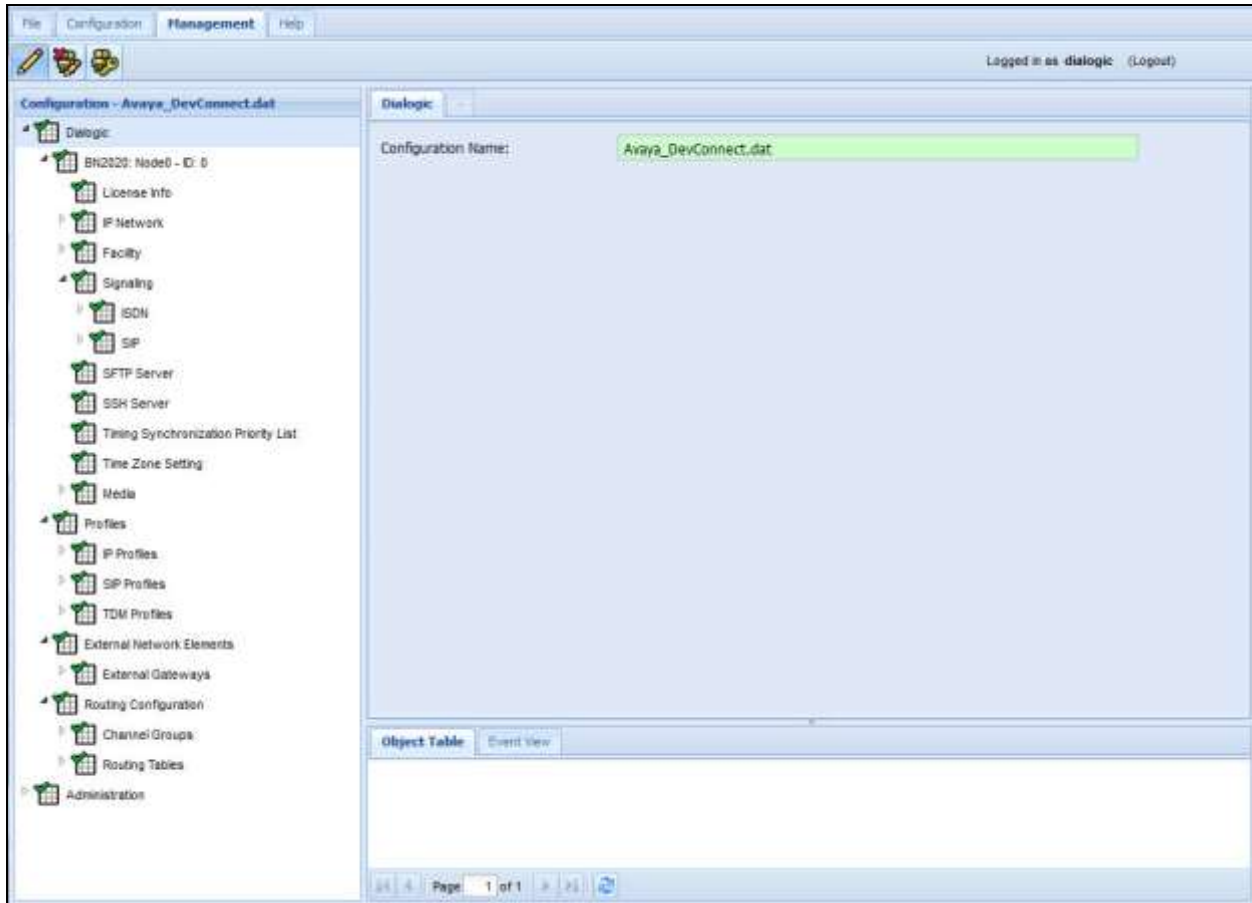
Full configuration information is available from here:

<http://www.dialogic.com/webhelp/BorderNet2020/2.2.0/WebHelp/default.htm>

Installation and configuration support is also available from your Dialogic Integrator or Dialogic Support. Contact details in **Section 2.3**.

7.1. Launch Management Interface

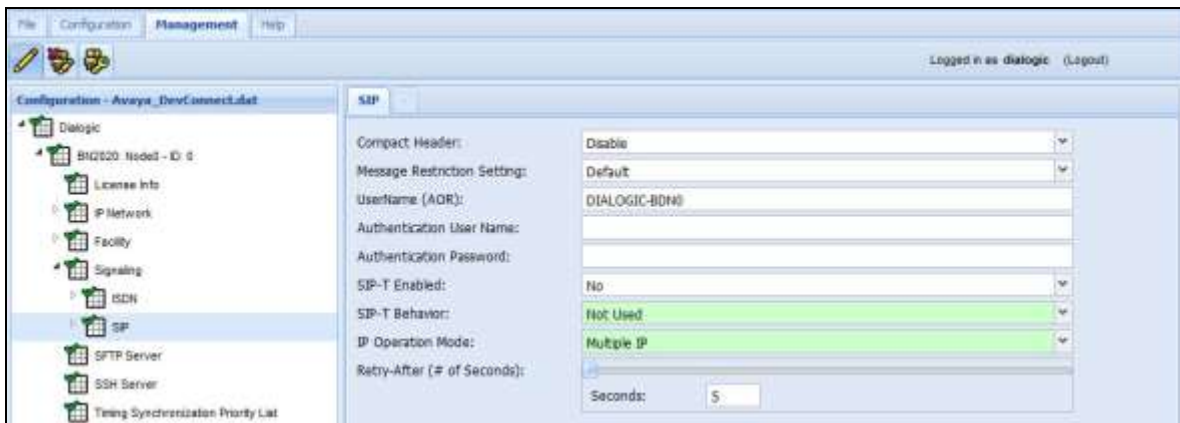
The IMG 2020 is administered using a built-in web based management user interface. To access the interface, enter **http://<ip-addr>** as the URL in a web browser where **<ip-addr>** is the IP address of the Dialogic management port. Currently Firefox and Internet Explorer web browsers are the only officially supported web browsers for the IMG 2020. Enter the appropriate credentials to log in. The following screen is displayed.



7.2. Configure Signaling

From the configuration tree in the left pane, navigate to and right click **Dialogic** → **BN2020 Node0** → **Signaling** and select **New SIP**. The **SIP** screen is displayed. For **IP Operation Mode**, select **Multiple IP** from the dropdown menu. Keep the default values for the remaining fields. The following shows the completed **SIP** screen.

Note: For the compliance test there is only one IP address defined on this **SIP** screen. But it is a recommended practice to set the **IP Operation Mode** field to **Multiple IP** to allow another SIP IP address to be added in the future without having to perform a major reconfiguration.



Right click **Dialogic** → **BN2020 Node0** → **Signaling** → **SIP** and select **New SIP IP Address**. The **SIP IP Address** screen is displayed. For **IP Address**, select the proper address from the dropdown menu. Set **Transport Type** to **UDP** and **Port** to **5060**. Keep the default values for the remaining fields. The following shows the completed **SIP IP Address** screen.

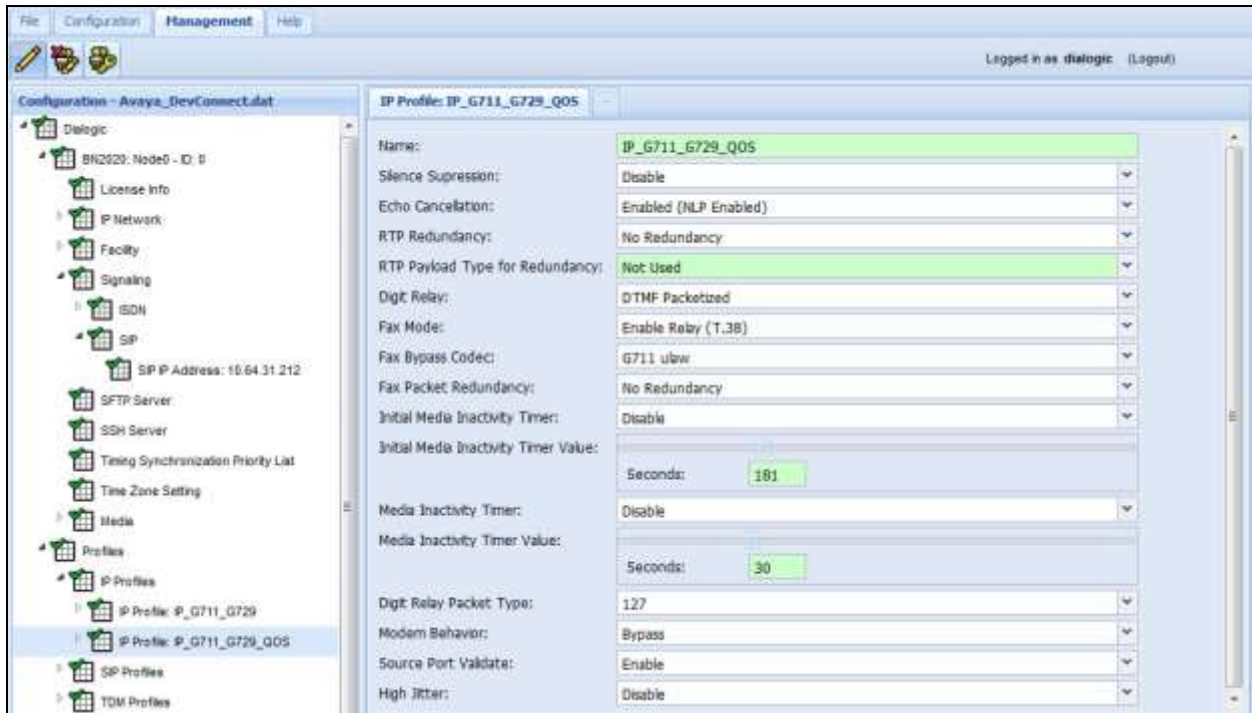


7.3. Configure Profiles

This section describes the configuration of the SIP and ISDN profiles.

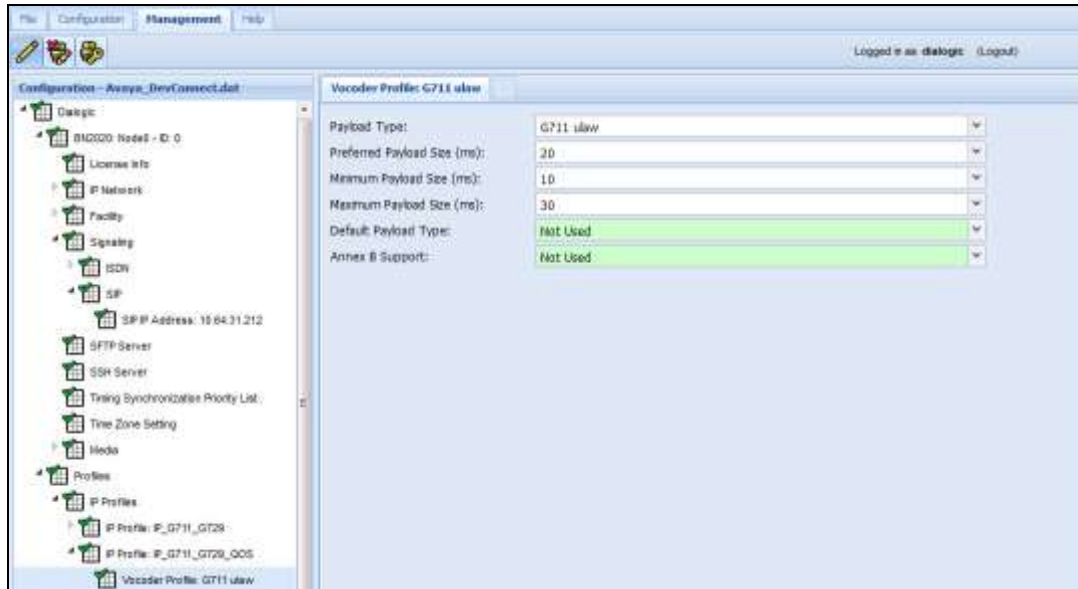
7.3.1. Configure IP Profiles

From the configuration tree in the left pane, right click **Dialogic** → **Profiles** → **IP Profiles** and select **New IP Profile**. The **IP Profile** screen is displayed. Enter a descriptive name in the **Name** field. For **Digit Relay**, select **DTMF Packetized** to use the RTC 2833 method. For **Fax Mode**, select **Enable Relay (T.38)**. For **Digit Relay Packet Type**, select a proper value. Keep the default values for the remaining fields. The following shows the completed **IP Profile** screen.

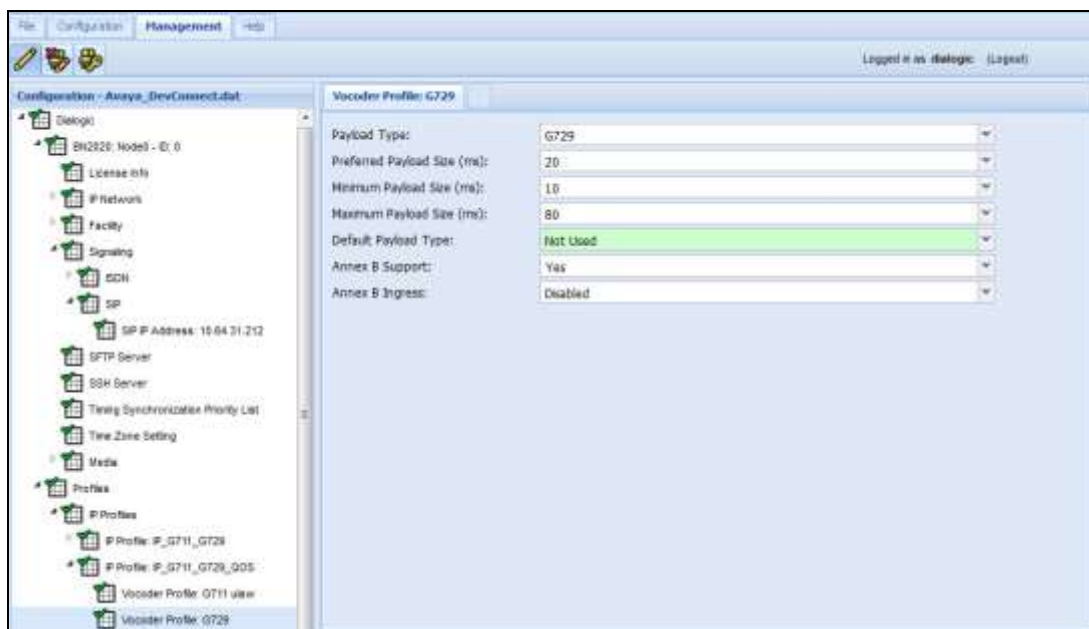


7.3.1.1 Configure IP Codecs in IP Profile

You will now choose the codecs that you wish to use for the connection between the IMG2020 and CM7, typically G.711A or U is used, and others tested were G.729 and G.722 Wideband. From the configuration tree in the left pane, right click the newly created IP Profile and select **New Vocoder Profile**. The **Vocoder Profile** screen is displayed. For **Payload Type**, select **G711 ulaw**. Keep the default values for the remaining fields. The following shows the completed **Vocoder Profile** screen.



Repeat the procedure for the **G.729** codec and set **Annex B Support** to **Yes**.

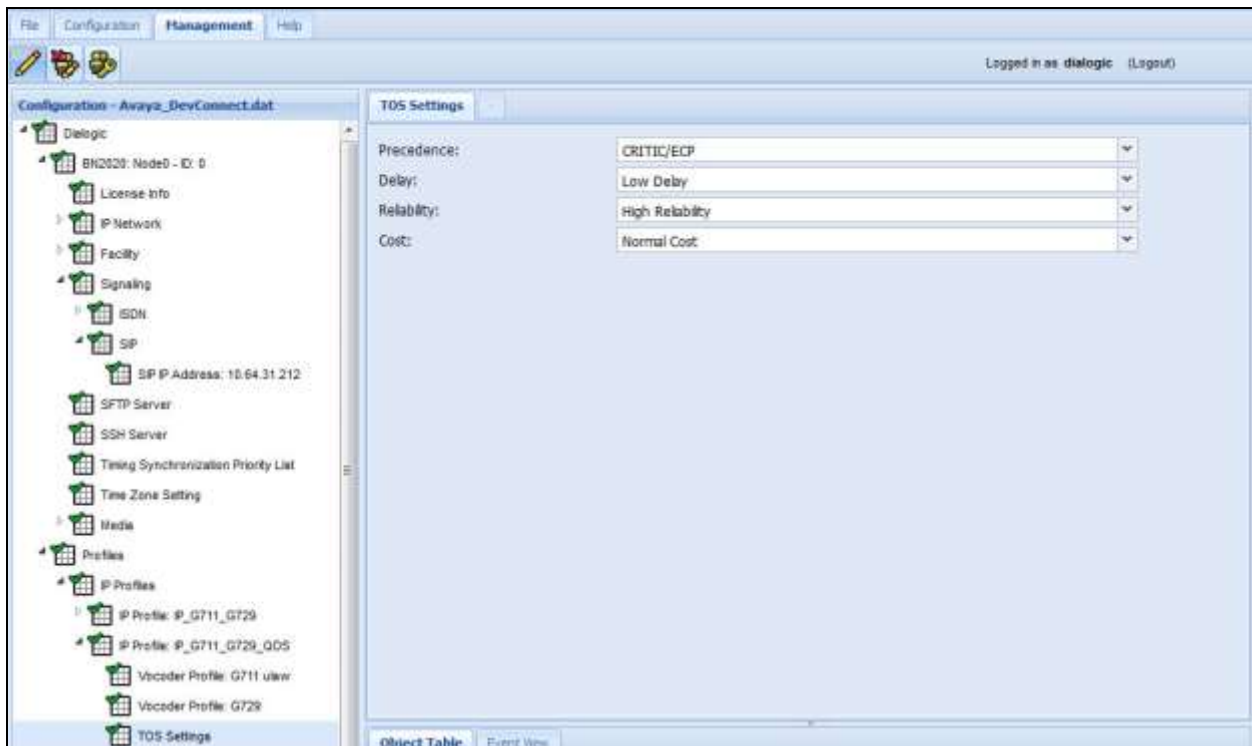


Continue to add G.722 if required. Note that G.722 support requires the IMG 2020 to have been setup in Media Mode 'HD Voice' or 'Dynamic Audio Density'. If the IMG 2020 is setup in the default Audio LBR mode (see **Diallogic** → **BDN2020** → **Media Mode**) then G.722 can be configured but will fail to operate if requested in a call. To use G.722 the IMG must be setup to use the modes 'HD Voice' or 'Dynamic Audio Density' as mentioned above. This will require complete re-configuration of the IMG 2020 so is out of the scope of this document.

7.3.1.2 Configure TOS Settings in IP Profile

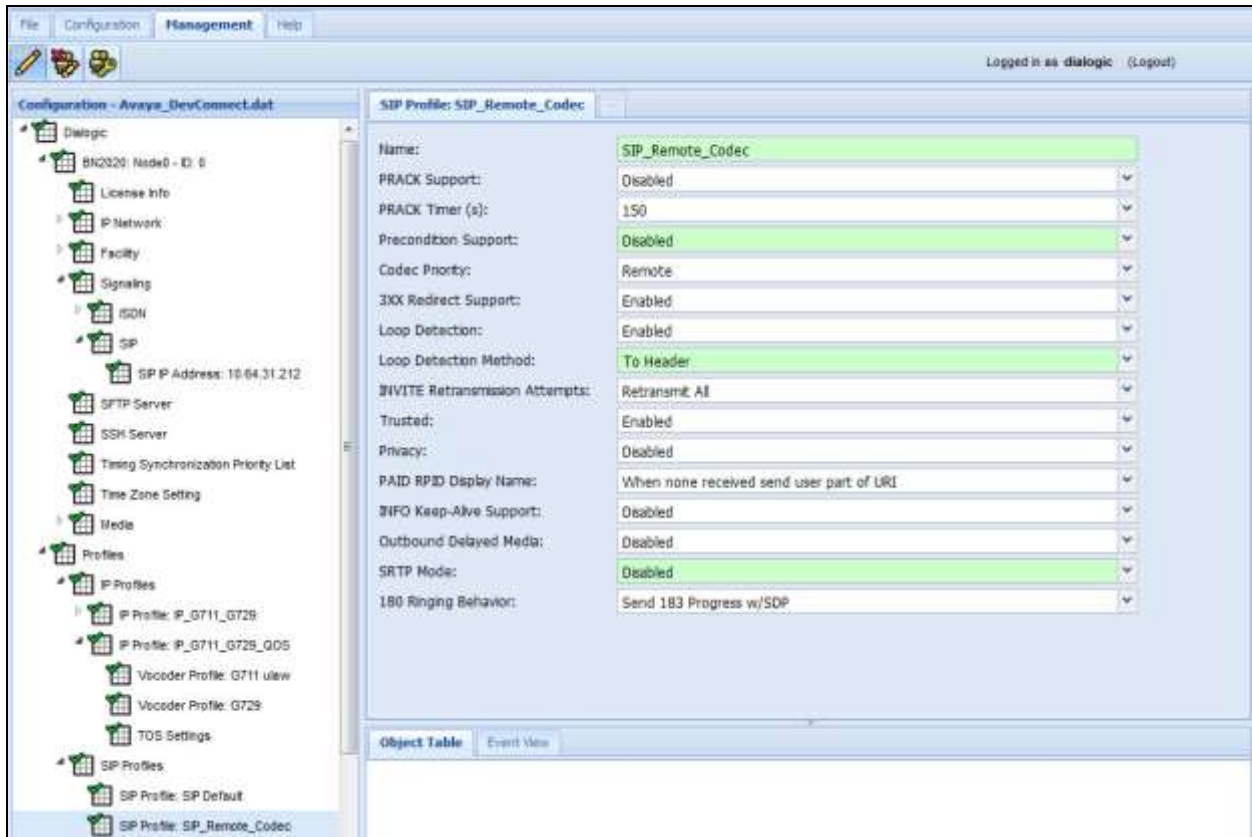
Quality of service settings can be made so that the IMG2020 will include the TOS requests in outgoing calls so requesting specific QOS from the network. A typical example that was tested in earlier DevConnect sessions is shown below.

From the configuration tree in the left pane, right click the newly created IP Profile and select **New TOS Settings**. The **TOS Settings** screen is displayed. For the **Precedence**, **Delay**, **Reliability**, and **Cost** fields, select **CRITIC/ECP**, **Low Delay**, **High Reliability**, and **Normal Cost** respectively. The following shows the completed **TOS Settings** screen.



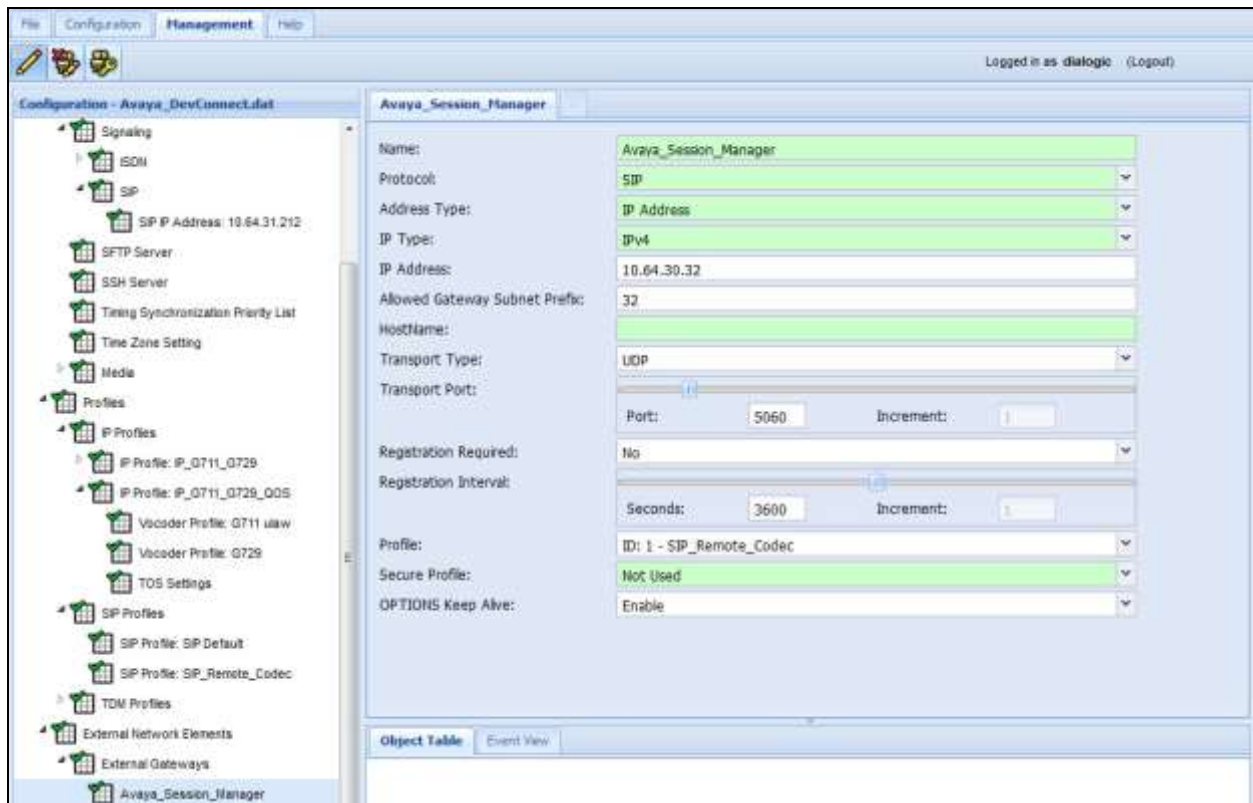
7.3.2. Configure SIP Profiles

From the configuration tree in the left pane, right click **Dialogic** → **Profiles** → **SIP Profiles** and select **New SIP Profile**. The **SIP Profile** screen is displayed. Enter a descriptive name in the **Name** field. For **Codec Priority**, select **Remote**. This gives the codecs in the far end higher priority during codec negotiation. Keep the default values for the remaining fields. The following shows the completed **SIP Profile** screen.



7.4. Configure External Network Element

From the configuration tree in the left pane, right click **Dialogic** → **External Network Elements** → **External Gateways** and select **New External Gateway**. The **ExternalGateway** screen is displayed. Enter a descriptive name in the **Name** field such as **Avaya_Session_Manager**. For **Protocol**, select **SIP**. For **IP Address**, enter the IP address of the Session Manager signaling interface. For Profile select the SIP Profile created in **Section 7.3.2**. For **OPTIONS Keep Alive**, select **Enable** to enable sending SIP Options messages. Keep the default values for the remaining fields. The following shows the completed **ExternalGateway** screen for **Avaya_Session_Manager**.



7.5. Configure Routing

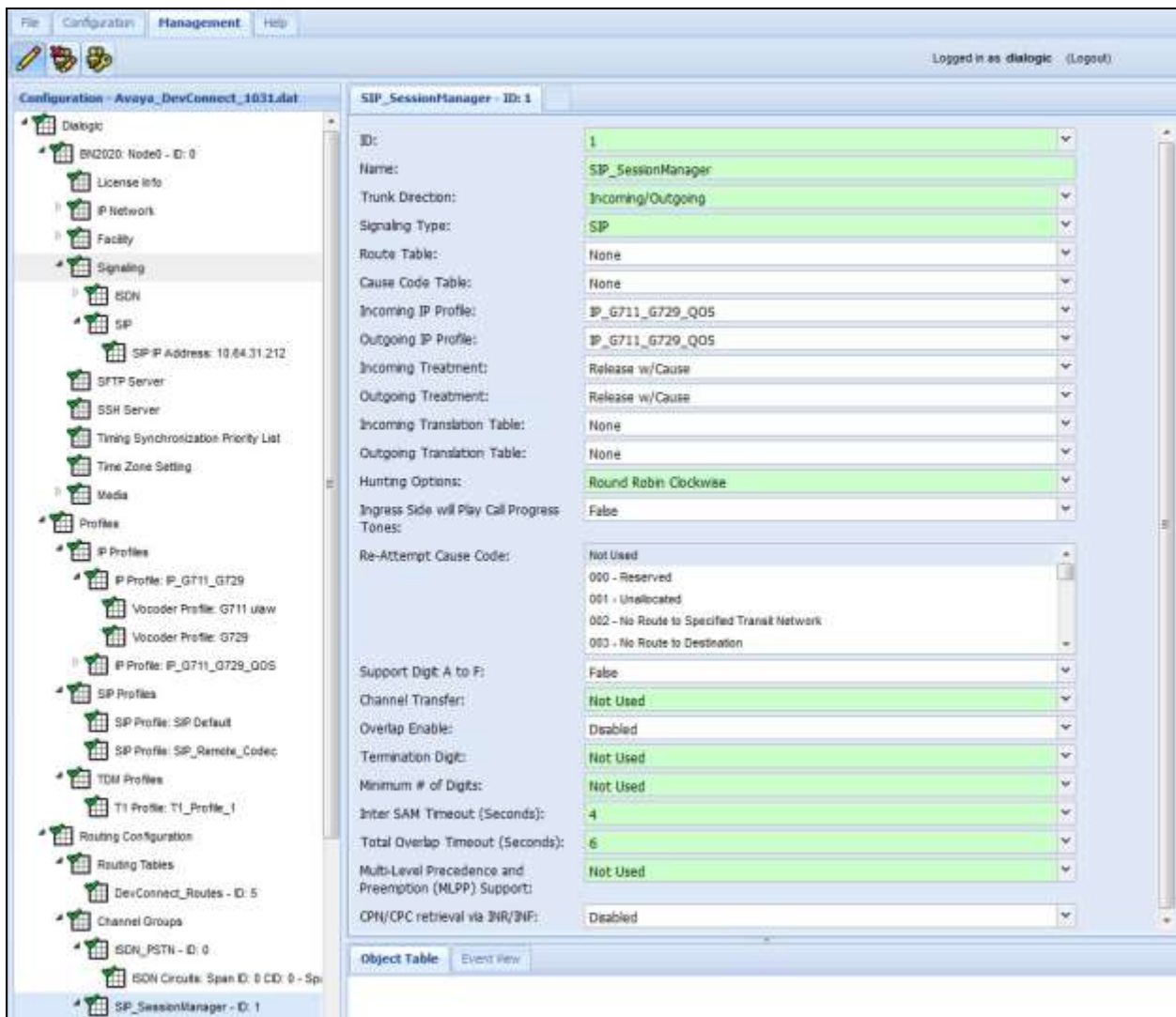
This section describes the configuration required to route calls

7.5.1. Configure Channel Group

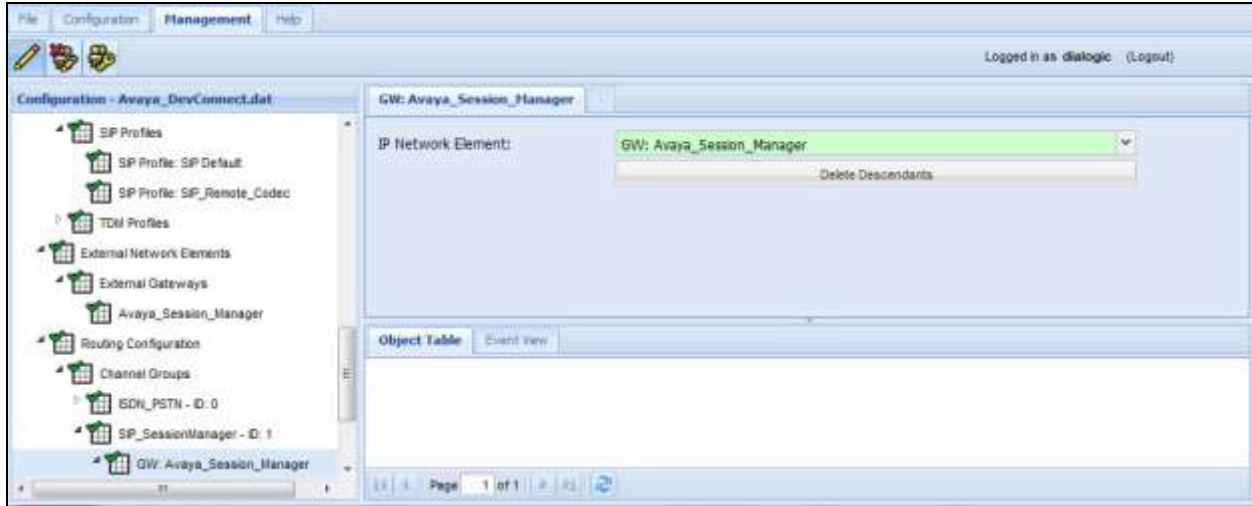
The Channel Group is a logical place holder for underlying network resources such as SIP Trunk or ISDN trunk.

From the configuration tree in the left pane, right click **Dialogic** → **Routing Configuration** → **Channel Groups** and select **New Channel Group**. The **ChannelGroup** screen is displayed.

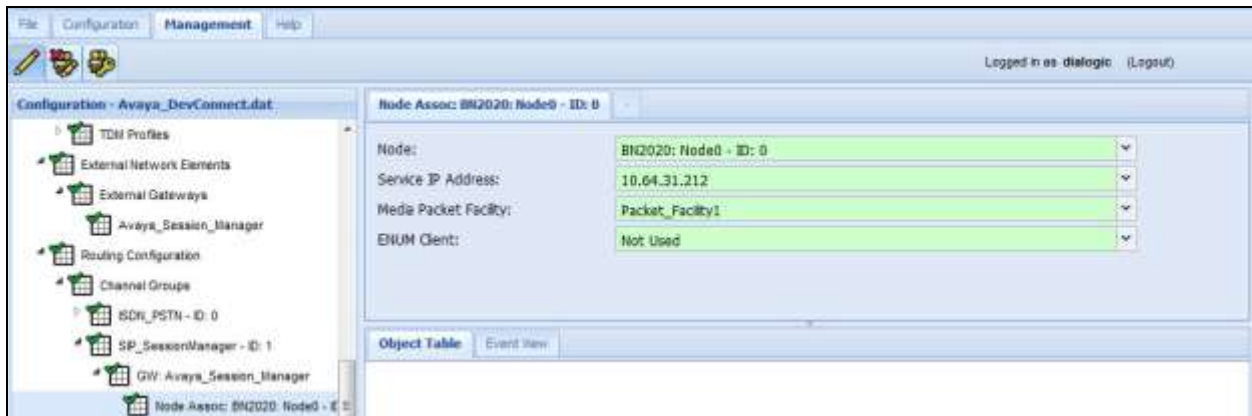
Enter a descriptive name in the **Name** field such as **SIP_SessionManager**. For **Signaling Type**, select **SIP**. For **Incoming IP Profile** and **Outgoing IP Profile**, select the IP Profile configured in **Section 7.3.1**. Keep the default values for the remaining fields. The following shows the completed **ChannelGroup** screen for **SIP_SessionManager**.



Right click the newly configured Channel Group in the left pane (not shown) and select **New IP Network Element**. The **NetworkElement** screen is displayed. For **IP Network Element**, select the External Gateway configured in **Section 7.4**. The following shows the completed **NetworkElement** screen.

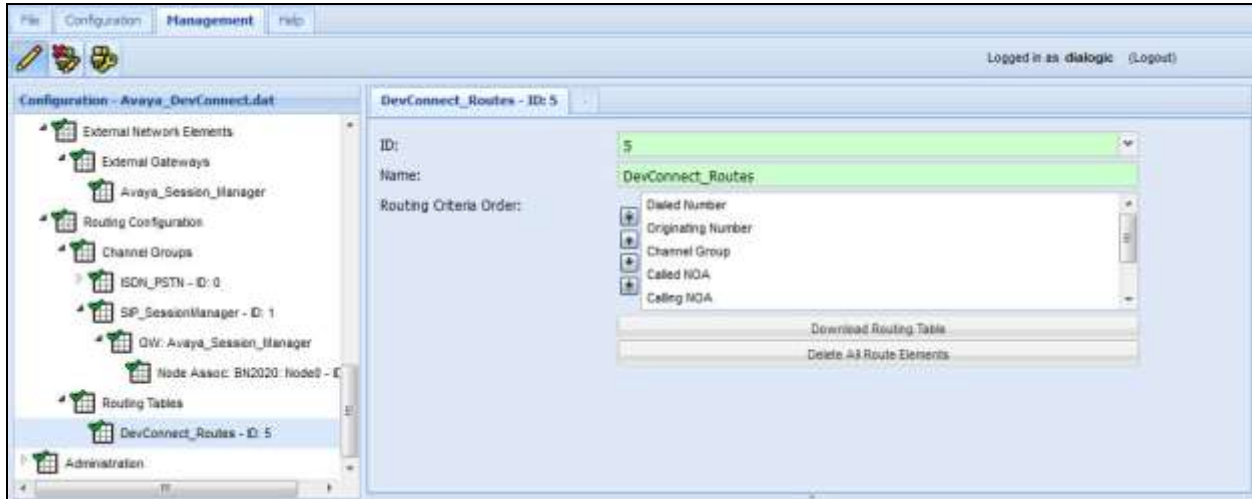


Right click the newly configured IP Network Element in the left pane and select **New Node Association**. The **Node Assoc** screen is displayed. For **Node**, **Service IP Address**, and **Media Packet Facility**, select proper values. Keep the default values for the remaining fields. The following shows the completed **Node Assoc** screen.

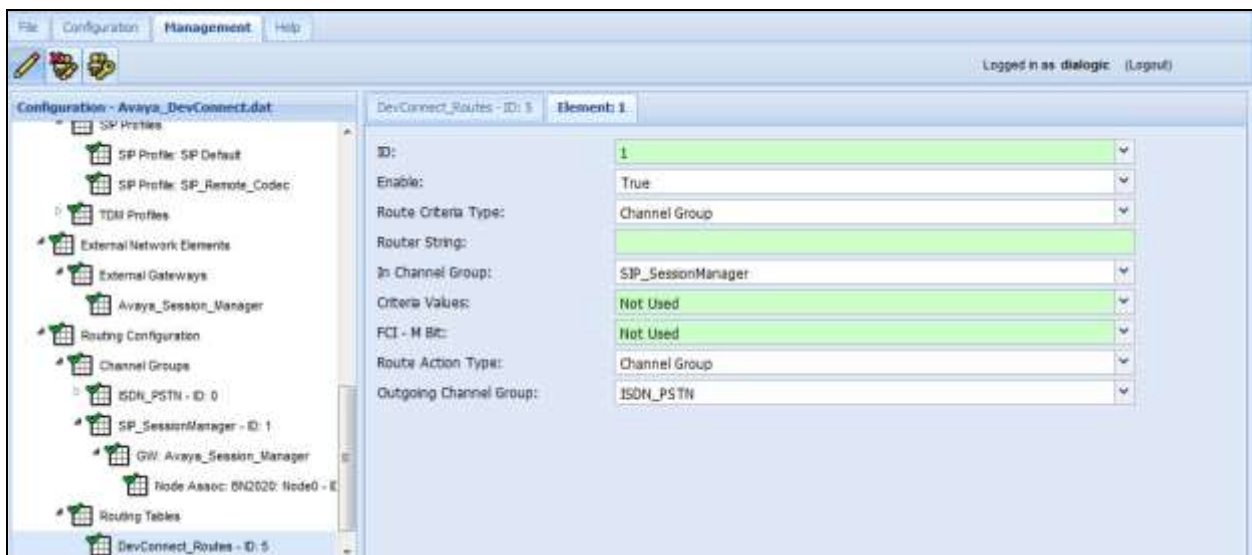


7.5.2. Configure Routing Tables

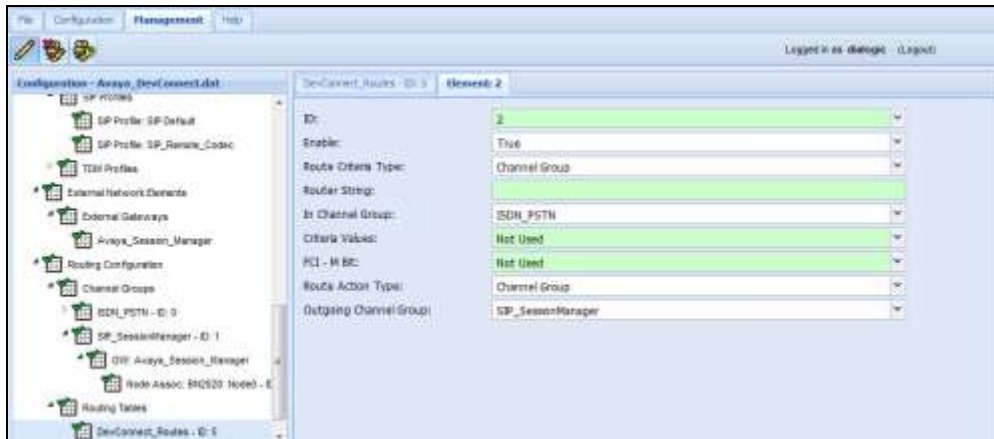
From the configuration tree in the left pane, right click **Dialogic** → **Routing Configuration** → **Routing Tables** and select **New Routing Table**. The **Table** screen is displayed. Enter a descriptive name in the **Name** field. Keep the default values for the remaining fields. The following shows the completed **Table** screen.



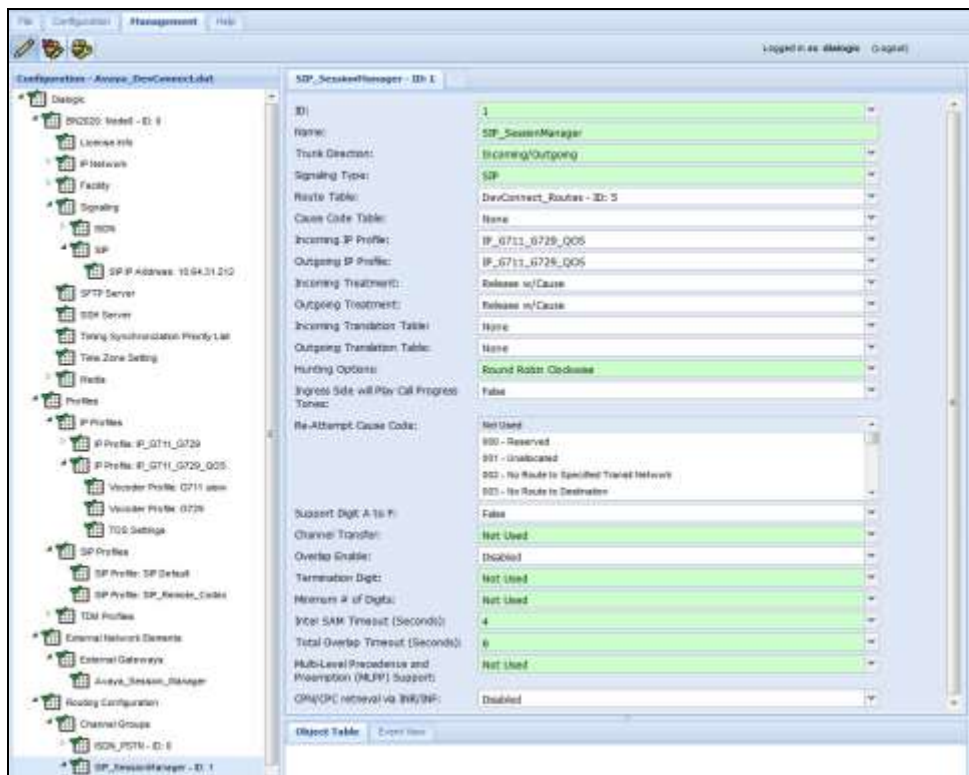
Right click the newly configured Routing Table in the left pane and select **New Route Element**. The **Element** tab is displayed. For **Route Criteria Type**, select **Channel Group**. For **In Channel Group**, select the channel group configured in **Section 7.5.1**. For **Outgoing Channel Group**, select the channel group pre-configured for the ISDN PRI interface. Keep the default values for the remaining fields. The following shows the completed **Element** tab.



Right click on the Route Table and select Commit to activate the route element. Repeat the above procedure for a second Route Element which routes calls from the ISDN PRI channel group to the Session Manager channel group.



Bring up the Channel Group configured in **Section 7.5.1** by right clicking **Dialogic** → **Routing Configuration** → **Channel Groups** → **SIP_SessionManager**. For **Route Table**, select the Route Table configured above. The following shows the updated **ChannelGroup** screen for **SIP_SessionManager**.



Repeat the above procedure for the Channel Group pre-configured for the ISDN PRI interface.

8. Verification Steps

This section describes the checks that can be carried out to verify the connection between IMG 2020, Communication Manager and Session Manager.

8.1. Session Manager

Select Elements → Session Manager from the home screen (not shown) and click on the value under Entity Monitoring

Verify that the IMG 2020 entry is UP.

10 Items Refresh		Filter: Enable							
	SIP Entity Name	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status	
<input type="radio"/>	CS1kPG	10.10.40.111	5060	TCP	FALSE	DOWN	408 Request Timeout	DOWN	
<input type="radio"/>	SM214	10.10.16.214	5060	TCP	FALSE	DOWN	500 Server Internal Error: Destination Unreachable	DOWN	
<input type="radio"/>	AAEP1620	10.10.16.20	5060	TCP	FALSE	DOWN	500 Server Internal Error: Destination Unreachable	DOWN	
<input type="radio"/>	Dialogic2020	10.10.16.5	5060	UDP	FALSE	UP	200 OK	UP	
<input type="radio"/>	MSG1689	10.10.16.89	5060	TCP	FALSE	UP	200 OK	UP	

8.2. IMG 2020 verification

This Section describes the verification steps needed to check the SIP and ISDN connections are active.

8.2.1. Event View

The Event View tab in the IMG2020 WEB UI displays any system alarms (not shown). If the Options Keep Alive has been configured as recommended in **Section 7.4** then the IMG will report any failure in the SIP trunk communications with Session Manager and raise an alarm which will be reported in Event View. Similarly a failure of the PSTN connection will generate an alarm.

8.2.2. Active Monitoring

The PSTN connection status can also be actively monitored by selecting **Dialogic → BN2020 → Signalling → ISDN → ISDN D channel** and checking the status is In Service (not shown).

8.2.3. SNMP

Full SNMP monitoring is available that integrates the IMG2020 into standard SNMP management applications.

9. Conclusion

These Application Notes describe the configuration steps required for Dialogic IMG 2020 to interoperate with Avaya Aura® Communication Manager using Avaya Aura® Session Manager. All feature functionality and serviceability test cases were completed successfully as outlined in **Section 2.2**.

10. Additional References

This section references the Avaya and Dialogic product documentation that are relevant to these Application Notes.

Product documentation for Avaya products may be found at <http://support.avaya.com>.

- [1] *Administering Avaya Aura® Communication Manager*, Document ID 03-300509
- [2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Document ID 555-245-205
- [3] *Administering Avaya Aura® Session Manager*, Release 7.0, 03-603324
- [4] *Quick Start Guide to Using the Avaya Aura® Media Server with Avaya Aura® Communication Manager*, August 2015

Product documentation for Dialogic IMG 2020 can be obtained by visiting the following website
Product details and specification:

<http://www.dialogic.com/en/products/gateways/img/img2020.aspx>

Installation and Configuration information:

<http://www.dialogic.com/webhelp/BorderNet2020/2.2.0/WebHelp/default.htm>

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