



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Dialogic® BorderNet™ 2020 Integrated Media Gateway with Avaya Aura® Experience Portal using SIP Trunks - Issue 1.0

Abstract

These Application Notes describe the procedure to configure Dialogic® BorderNet™ 2020 to interoperate with Avaya Aura® Experience Portal as an ISDN PRI/SIP gateway using SIP trunking.

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

1. Introduction

These Application Notes describe the procedure to configure Dialogic® BorderNet™ 2020 to interoperate with Avaya Aura® Experience Portal as an ISDN PRI/SIP gateway using SIP trunking.

Dialogic® BorderNet™ 2020 Integrated Media Gateway combines integrated media and signaling IP and TDM gateway capabilities with session border controller functionality in a compact 1U form factor appliance.

The compliance testing of the Dialogic® BorderNet™ 2020 Integrated Media Gateway focused on its ISDN PRI/SIP gateway functions.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability. During the test, various call scenarios were exercised to verify call and feature interoperability of BorderNet 2020 and Experience Portal. Network and server outage conditions were used to verify serviceability of the joint solution.

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 primary focus of the feature testing was to verify SIP trunking interoperability between Experience Portal and BorderNet 2020. Test cases were selected to verify the following areas. All tests were performed using sample applications that are part of Experience Portal.

Basic Interoperability:

- Basic Calls from PSTN to Experience Portal.
- Call Transfers by Experience Portal to PSTN; blind, consultative and bridged transfers.
- DTMF Tones; RFC2833 support.
- SIP transport using TCP
- G.711 mu-Law codec support

The serviceability testing focused on verifying the ability of the solution to recover from adverse conditions, such as network failures and BorderNet 2020 reboot.

2.2. Test Results

All test cases were executed and verified.

2.3. Support

Technical Support on Dialogic BorderNet 2020 can be obtained through the following phone contacts:

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

3. Reference Configuration

The reference configuration consists of Experience Portal, Application Server and BorderNet 2020. BorderNet 2020 is used as a SIP/ISDN gateway for PSTN access. The Experience Portal routes the calls to the BorderNet 2020 using SIP Trunks. The management interface of BorderNet 2020 has to be on a different subnet from the signaling and media interfaces.

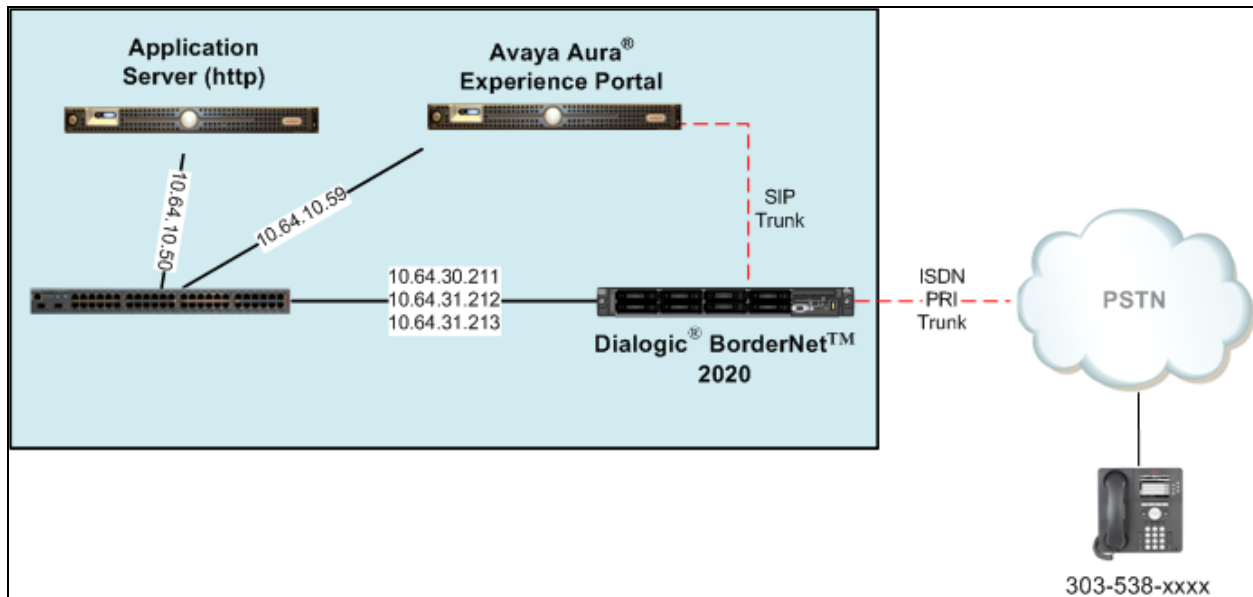


Figure 1 – Sample configuration for Avaya Aura[®] Experience Portal with Dialogic[®] BorderNet[™] 2020 using Sip Trunking

4. Equipment and Software Validated

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

Equipment/Software	Version
Avaya Aura [®] Experience Portal: <ul style="list-style-type: none">• Experience Portal Manager (EPM)• Media Processing Platform (MPP)	7.0
Dialogic [®] BorderNet [™] 2020 <ul style="list-style-type: none">• Integrated Media Gateway• Dialogic[®] WebUI	2.2 SP2 b1561 2.2 SP2

5. Configure Avaya Aura® Experience Portal

This section covers the administration of Experience Portal. The Experience Portal configuration required for interoperating with the BorderNet 2020 includes following areas:

- Configure SIP connection
- Add MPP server
- Add speech server
- Add voice application
- Start MPP server

Experience Portal is configured via the Experience Portal Manager (EPM) web interface. To access the web interface, enter `http://<ip-addr>/` as the URL in a web browser, where `<ip-addr>` is the IP address assigned to the EPM server. Log in using appropriate credentials. The initial Experience Portal screen after login is shown below.

The screenshot displays the Avaya Aura® Experience Portal 7.0 (ExperiencePortal) web interface. The top header features the Avaya logo on the left and a welcome message "Welcome, admin" with the last login time "Last logged in today at 2:37:00 AM MDT" on the right. Below the header, a red navigation bar contains links for "Home", "Help", and "Logoff". The main content area is divided into a left sidebar and a right main panel. The sidebar lists various management categories such as User Management, Real-time Monitoring, System Maintenance, System Management, System Configuration, Security, Reports, and Multi-Media Configuration. The main panel displays the "Avaya Aura® Experience Portal Manager" title and a brief description of the EPM interface. It also lists "Installed Components" including Media Processing Platform, Email Service, Proactive Outreach Manager, and Short Message Server. At the bottom, there is a "Legal Notice" section with a scrollable text area containing copyright information and a disclaimer.

5.1. Configure SIP Connection

To configure a SIP connection to the BorderNet 2020, navigate to **System Configuration** → **VoIP Connections**, and click on the **SIP** tab. Click the **Add** button to add a new connection.

Avaya Aura® Experience Portal 7.0 (ExperiencePortal)

Welcome, admin
Last logged in today at 2:37:00 AM MDT

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

VoIP Connections

This page displays a list of Voice over Internet Protocol (VoIP) servers that Experience Portal communicates with. You can configure multiple SIP connections, but only one SIP connection can be enabled at any one given time.

	Name	Enable	Proxy Transport	Proxy/DNS Server Address	Proxy Server Port	Listener Port	SIP Domain	Maximum Simultaneous Calls
<input type="checkbox"/>	Dialogic	No	TCP	10.64.31.212	5060	5060	10.64.31.212	100
<input checked="" type="checkbox"/>	sm1sip	Yes	TCP	10.64.30.32	5060	5060	avaya.com	10

Add **Delete** **Help**

On the resulting screen, configure the parameters as follows:

- Enter a descriptive text for **Name**.
- Select the **Yes** radio button for **Enable**.
- Select **TCP** as the **Proxy Transport**.
- Specify the signaling IP address assigned to BorderNet 2020 for **Proxy Server Address** and specify **5060** for **Proxy Server Port**.
- Set the **Listener Port** field to **5060** for TCP.
- Specify the signaling IP address assigned to BorderNet 2020 for the **SIP Domain**.
- Select **REFER** radio button for **Consultative Transfer**.
- Set the **Maximum Simultaneous Calls**. In this example, a maximum of 10 calls is specified.
- Accept the default values for the other fields and select **Save** to save this configuration.

AVAYA

Welcome, admin
Last logged in today at 2:37:00 AM MDT

Avaya Aura® Experience Portal 7.0 (ExperiencePortal)

HomeHelpLogoff

Expand All | Collapse All

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

Add SIP Connection

Use this page to add a new SIP connection.

Name:

Enable: ☒ Yes ☐ No

Proxy Transport:

☒ Proxy Servers ☐ DNS SRV Domain

Address	Port	Priority	Weight	
<input type="text" value="10.64.31.212"/>	<input type="text" value="5060"/>	<input type="text" value="0"/>	<input type="text" value="0"/>	Remove

Additional Proxy Server

Listener Port:

SIP Domain:

P-Asserted-Identity:

Maximum Redirection Attempts:

Consultative Transfer: ☐ INVITE with REPLACES ☒ REFER

SIP Reject Response Code: ☒ ASM (503) ☐ SES (480) ☐ Custom

SIP Timers

T1: milliseconds

T2: milliseconds

B and F: milliseconds

Call Capacity

Maximum Simultaneous Calls:

☒ All Calls can be either inbound or outbound

☐ Configure number of inbound and outbound calls allowed

Save

Cancel

Help

KJA; Reviewed:
SPOC 7/15/2014

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DialogicAAEP7

5.2. Add MPP Server

Add a Media Processing Platform (MPP) server by navigating to **System Configuration** → **MPP Servers**. Click the **Add** button to add a new MPP Server (not shown). In the MPP Server configuration page, specify a descriptive name and the host address of the MPP server. Also, specify the **Maximum Simultaneous Calls** supported on this MPP server. The screen below shows the configuration for the first MPP server used in the reference configuration.

AVAYA Welcome, admin
Last logged in today at 2:37:00 AM MDT

Avaya Aura® Experience Portal 7.0 (ExperiencePortal) Home ? Help Logoff

Expand All | Collapse All

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▼ POM
POM Home
POM Monitor

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 Experience 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: MPPLocal
Host Address: 10.64.10.59
Network Address (VoIP): <Default>
Network Address (MRCP): <Default>
Network Address (AppSvr): <Default>
Maximum Simultaneous Calls: 100
Restart Automatically: ☒ Yes ☐ No

MPP Certificate

Owner: CN=aaep7.avaya.com,O=Avaya,OU=EPM
Issuer: CN=aaep7.avaya.com,O=Avaya,OU=EPM
Serial Number: a69b4ccc662b591c
Valid from: March 4, 2014 9:23:50 AM MST until March 1, 2024 9:23:50 AM MST
Certificate fingerprints
MD5: da:4f:1a:71:62:73:60:d8:15:7b:46:01:2f:b0:d4:16
SHA: 62:c1:8f:1b:7b:b8:d2:0b:24:b0:80:f9:5e:8e:4a:57:07:be:7b:8d

Categories and Trace Levels ▶

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

5.3. Add Speech Server

Adding a speech server for providing ASR (Automatic Speech Recognition) and/or TTS (Text To Speech) services is part of the standard configuration for Experience Portal. This configuration is not directly related to achieving interoperability between the BorderNet 2020 and Experience Portal. It is included here for completeness.

To configure the ASR server, navigate to **System Configuration → Speech Servers**, select the **ASR** tab (not shown), and then click **Add**. The screen below shows the configuration for the ASR server used during compliance testing. Set the **Engine Type** to the appropriate value. In the reference configuration, a Nuance ASR server was used so the engine type was set to **Nuance**. Set the **Network Address** field to the IP address assigned to the speech server and select the desired **Languages** to be supported. The other fields were set to their default values.

The screenshot shows the Avaya Aura Experience Portal 7.0 (ExperiencePortal) interface. The top navigation bar includes the Avaya logo, a welcome message for 'admin', and links for Home, Help, and Logoff. The left sidebar contains a tree view of system components, with 'System Configuration' expanded to show 'Speech Servers'. The main content area is titled 'Change ASR Server' and provides a form for configuring an ASR server. The form includes fields for Name (VM_Nuance), Enable (Yes/No), Engine Type (Nuance), Network Address (10.64.101.83), Base Port (554), Total Number of Licensed ASR Resources (100), New Connection per Session (Yes/No), Languages (a list box with various English language codes), MRCP settings (Ping Interval, Response Timeout, Protocol, and RTSP URL), and buttons for Save, Apply, Cancel, and Help.

AVAYA Welcome, admin
Last logged in today at 2:37:00 AM MDT

Avaya Aura® Experience Portal 7.0 (ExperiencePortal) Home Help Logoff

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- ▼ POM
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 - POM Monitor

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

Change ASR Server

Use this page to change the configuration of an ASR server.

Name: VM_Nuance

Enable: ☒ Yes ☐ No

Engine Type: Nuance

Network Address: 10.64.101.83

Base Port: 554

Total Number of Licensed ASR Resources: 100

New Connection per Session: ☐ Yes ☒ No

Languages: Dutch(Netherlands) nl-NL, English(Australia) en-AU, English(UK) en-GB, English(India) en-IN, English(Singapore) en-SG, English(USA) en-US

MRCP

Ping Interval: 15 seconds

Response Timeout: 4 seconds

Protocol: MRCP V1

RTSP URL: 10.64.101.83/media/speechrecognizer

Save Apply Cancel Help

To configure the TTS server, navigate to **System Configuration → Speech Servers**, select the **TTS** tab (not shown), and then click **Add**. The screen below shows the configuration for the TTS server used during compliance testing. 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 assigned to the speech server and select the desired **Languages** to be supported. The other fields were set to their default values.

AVAYA Welcome, admin
Last logged in today at 2:37:00 AM MDT

Avaya Aura® Experience Portal 7.0 (ExperiencePortal) Home Help Logoff

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

Change TTS Server

Use this page to change the configuration of a TTS server.

Name: VM_Nuance

Enable: ☒ Yes ☐ No

Engine Type: Nuance ▼

Network Address: 10.64.101.83

Base Port: 554

Total Number of Licensed TTS Resources: 100

New Connection per Session: ☐ Yes ☒ No

Voices: English(Irish) en-IE Moira F
English(South_African) af-ZA Tessa F
English(Scottish) en-SC Fiona F
English(USA) en-US Donna F
English(USA) en-US Erica F
English(USA) en-US Jennifer F

MRCP

Ping Interval: 15 seconds

Response Timeout: 4 seconds

Protocol: MRCP V1 ▼

RTSP URL: 10.64.101.83/media/speechsynthesizer

Save Apply Cancel Help

5.4. Add Voice Application

Adding a voice application for Experience Portal is part of Experience Portal's standard administration. This configuration is not directly related to achieving interoperability between the BorderNet 2020 and Experience Portal. It is included here for completeness.

Navigate to **System Configuration → Applications**, and then click **Add**. Specify a **Name** for the application, select the **Yes** radio button for **Enable**, 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. The configuration for the voice application used in the compliance test is shown in the screen below.

AVAYA Welcome, admin
Last logged in today at 2:37:00 AM MDT

Avaya Aura® Experience Portal 7.0 (ExperiencePortal) Home ? Help Logoff

Expand All | Collapse All

- ▼ User Management
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- ▼ POM
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You are here: [Home](#) > [System Configuration](#) > [Applications](#) > [Change Application](#)

Change Application

Use this page to change the configuration of an application.

Name: Sample_VoiceXML_intro

Enable: ☒ Yes ☐ No

Type:

Reserved SIP Calls: ☒ None ☐ Minimum ☐ Maximum

Requested:

URI

☒ Single ☐ Fail Over ☐ Load Balance

VoiceXML URL: [Verify](#)

Mutual Certificate Authentication: ☐ Yes ☒ No

Basic Authentication: ☐ Yes ☒ No

Speech Servers

ASR: TTS:

Languages: Voices:

Application Launch

☒ Inbound ☐ Inbound Default ☐ Outbound

☒ Number ☐ Number Range ☐ URI

Called Number: [Add](#)

[Remove](#)

[Save](#) [Apply](#) [Cancel](#) [Help](#)

5.5. Start MPP Server

Start the MPP server from **System Management → MPP Manager** as shown below. Select the MPP(s) for use and then click the **Start** button. The **Mode** of the started MPP should be **Online** and the **State** should be **Running**.

The screenshot shows the Avaya Aura Experience Portal 7.0 (ExperiencePortal) interface. The top navigation bar includes the Avaya logo, a welcome message for 'admin', and a timestamp 'Last logged in today at 2:37:00 AM MDT'. The main header displays 'Avaya Aura® Experience Portal 7.0 (ExperiencePortal)' and navigation links for Home, Help, and Logoff. A left sidebar contains a tree view of system management options, including User Management, Real-time Monitoring, System Maintenance, System Management, System Configuration, Security, Reports, and Multi-Media Configuration. The main content area is titled 'MPP Manager (May 1, 2014 3:10:06 AM MDT)' and includes a 'Refresh' button. Below the title, a message states: 'This page displays the current state of each MPP in the Experience 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.' A table lists MPPs with columns for selection, Server Name, Mode, State, Config, Auto Restart, Restart Schedule (Today, Recurring), and Active Calls (In, Out). The table shows one MPP, 'MPPLocal', with Mode 'Online', State 'Running', Config 'OK', Auto Restart 'Yes', and no active calls. Below the table, there are sections for 'State Commands' (Start, Stop, Restart, Reboot, Halt, Cancel), 'Mode Commands' (Offline, Test, Online), and 'Restart/Reboot Options' (One server at a time, All servers). A 'Help' button is located at the bottom left of the main content area.

AVAYA Welcome, admin
Last logged in today at 2:37:00 AM MDT

Avaya Aura® Experience Portal 7.0 (ExperiencePortal) Home Help Logoff

Expand All | Collapse All

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

MPP Manager (May 1, 2014 3:10:06 AM MDT) Refresh

This page displays the current state of each MPP in the Experience 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: May 1, 2014 3:09:39 AM MDT

	Server Name	Mode	State	Config	Auto Restart	Restart Schedule	Active Calls
						Today Recurring	In Out
<input type="checkbox"/>	MPPLocal	Online	Running	OK	Yes	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 servers

Help

6. Configure Dialogic® BorderNet™ 2020 Integrated Media Gateway

For the compliance test, two trunking interfaces were configured on BorderNet 2020. A SIP trunk interface was used to connect to Experience Portal and an ISDN PRI interface was used to connect to PSTN. This section focuses on the configuration at the SIP side which enables BorderNet 2020 to interoperate with Experience Portal.

It is assumed that basic administration such as IP addresses, Default Gateways, and VLAN IDs for the SIP signaling and media interfaces, Serial number, Security ID, and Packet Facility have been configured during installation.

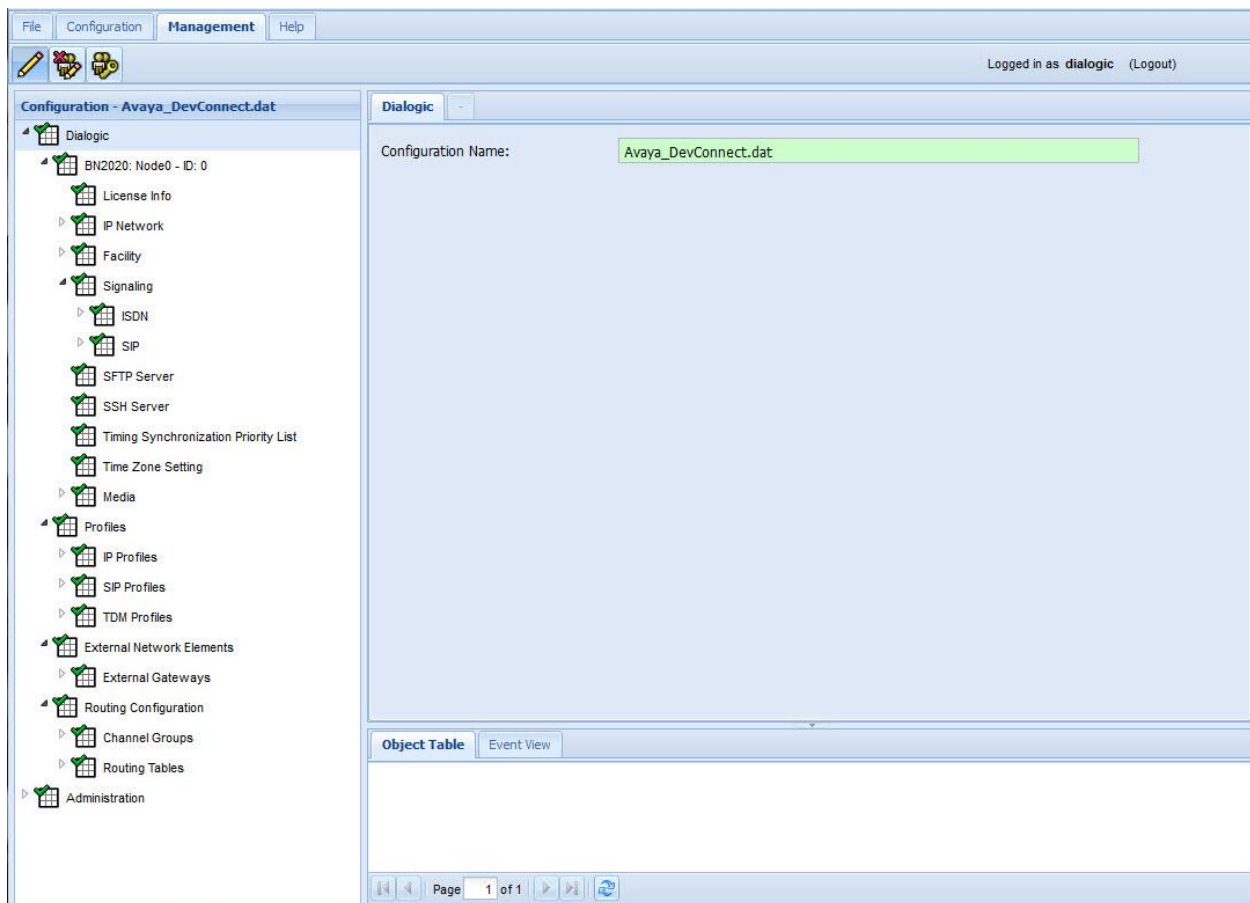
It is also assumed that the PSTN trunk has been properly configured, which includes the ISDN PRI interface, TDM Profile, associated Channel Group, and the underlining T1 interface.

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

- Launch Management Interface
- Configure BN2020 Node
- Configure Profiles
- Configure External Network Element
- Configure Routing Configuration

6.1. Launch Management Interface

BorderNet 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 Firefox web browser where <ip-addr> is the IP address of the Dialogic management port. Currently Firefox and Internet Explorer are the only officially supported web browsers for BorderNet 2020. Enter the appropriate credentials to log in. The following screen is displayed.



6.2. Configure BN2020 Node

From the configuration tree in the left pane, navigate to **Dialogic → BN2020 Node0 → Signaling**. If a SIP object is already present, skip the rest of this section and continue on **Section 6.3**. Otherwise, right click **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 address to be added in the future without having to perform a major reconfiguration.

The screenshot shows the 'SIP' configuration window. The left pane displays the configuration tree with 'Dialogic' expanded, showing 'BN2020: Node0 - ID: 0' and its sub-items: 'License Info', 'IP Network', 'Facility', 'Signaling', 'ISDN', 'SIP', 'SFTP Server', 'SSH Server', and 'Timing Synchronization Priority List'. The 'SIP' item is selected. The main pane shows the following fields:

Compact Header:	Disable
Message Restriction Setting:	Default
UserName (AOR):	DIALOGIC-BDN0
Authentication User Name:	
Authentication Password:	
SIP-T Enabled:	No
SIP-T Behavior:	Not Used
IP Operation Mode:	Multiple IP
Retry-After (# of Seconds):	5

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 signaling IP address from the dropdown menu. Set **Transport Type** to **TCP** and **Port** to **5060**. Please note that the port should match the port that was configured in **Section 5.1**. Keep the default values for the remaining fields. The following shows the completed **SIP IP Address** screen.

The screenshot shows the 'SIP IP Address' configuration window. The left pane displays the configuration tree with 'Dialogic' expanded, showing 'BN2020: Node0 - ID: 0' and its sub-items: 'License Info', 'IP Network', 'Facility', 'Signaling', 'ISDN', 'SIP', 'SFTP Server', 'SSH Server', and 'Timing Synchronization Priority List'. The 'SIP' item is selected, and 'SIP IP Address: 10.64.31.212' is highlighted. The main pane shows the following fields:

IP Type:	IPv4
IP Address:	10.64.31.212
Transport Type:	TCP
Port:	5060
TLS Port:	5061
DNS Client:	Not Used
DNS Query Mode:	MIX
Secure Profile:	Not Used
Default Secure Profile:	Not Used
Fully Qualified Domain Name:	

6.3. Configure Profiles

6.3.1. Configure IP Profiles

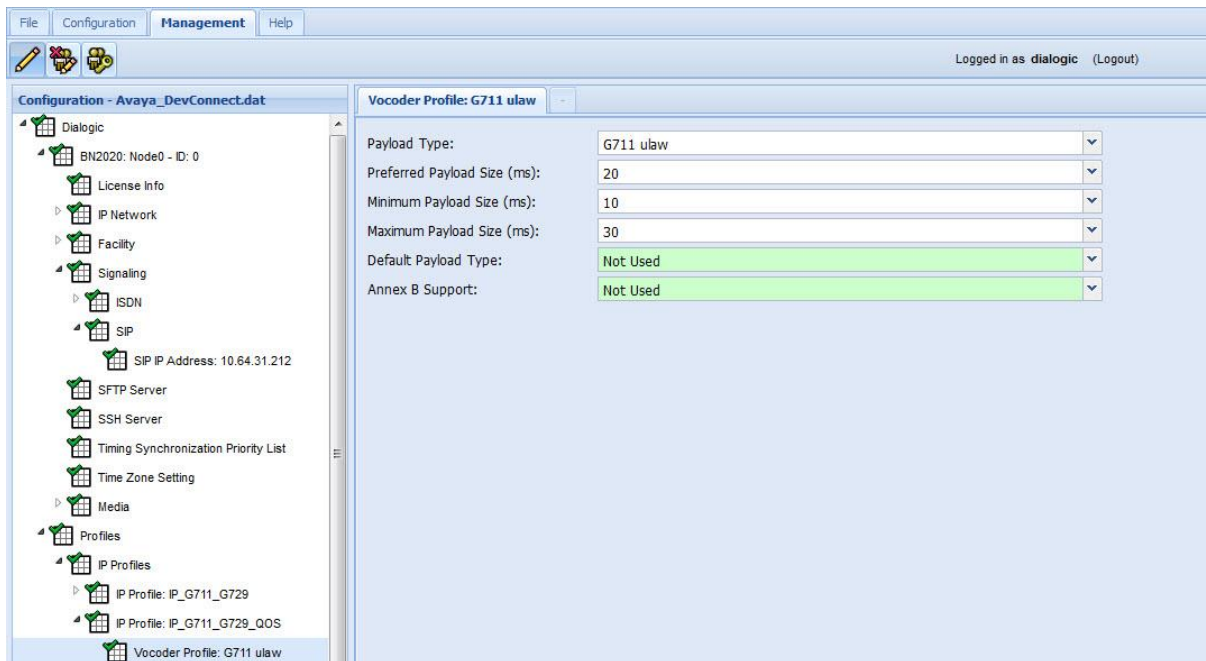
From the configuration tree in the left pane, right click **Dialogic** → **Profiles** → **IP Profiles** and select **New IP Profile**. The **IP Profile** screen is displayed. Enter a descriptive name in the **Name** field. For **Digit Relay**, select **DTMF Packetized** to use the RFC 2833 method. For **Fax Mode**, select **Enable Relay (T.38)**. For **Digit Relay Packet Type**, type in 127 as payload type. Keep the default values for the remaining fields. The following shows the completed **IP Profile** screen.

The screenshot shows the Avaya DevConnect Manager interface. The left pane displays the configuration tree with 'Dialogic' expanded, showing 'BN2020: Node0 - ID: 0' and 'IP Profiles'. The 'IP Profiles' folder is expanded, showing 'IP Profile: IP_G711_G729' and 'IP Profile: IP_G711_G729_QOS'. The right pane shows the configuration for 'IP Profile: IP_G711_G729_QOS'. The configuration fields are as follows:

Field	Value
Name	IP_G711_G729_QOS
Silence Suppression	Disable
Echo Cancellation	Enabled (NLP Enabled)
RTP Redundancy	No Redundancy
RTP Payload Type for Redundancy	Not Used
Digit Relay	DTMF Packetized
Fax Mode	Enable Relay (T.38)
Fax Bypass Codec	G711 ulaw
Fax Packet Redundancy	No Redundancy
Initial Media Inactivity Timer	Disable
Initial Media Inactivity Timer Value	Seconds: 181
Media Inactivity Timer	Disable
Media Inactivity Timer Value	Seconds: 30
Digit Relay Packet Type	127
Modem Behavior	Bypass
Source Port Validate	Enable
High Jitter	Disable

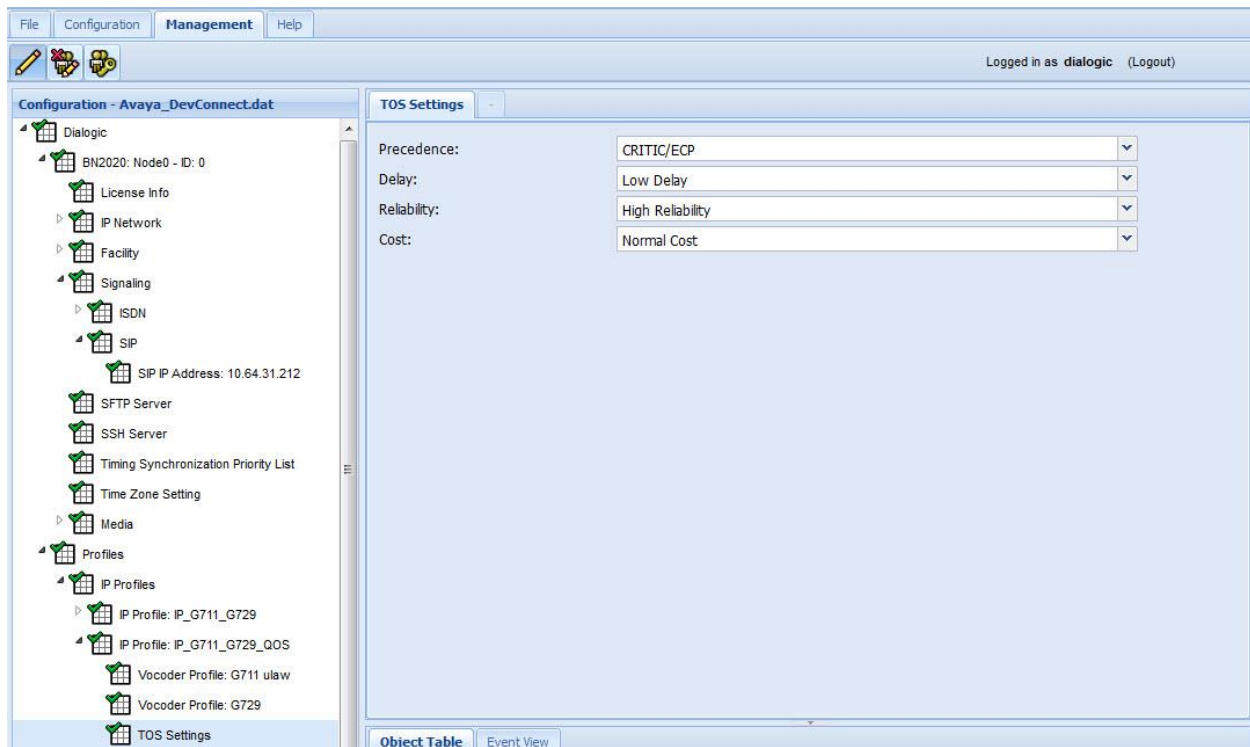
6.3.1.1 Configure IP Codecs in IP Profile

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.



6.3.1.2 Configure TOS Settings in IP Profile

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.



6.3.2. 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 such as **SIP_Remote_Codec** 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.

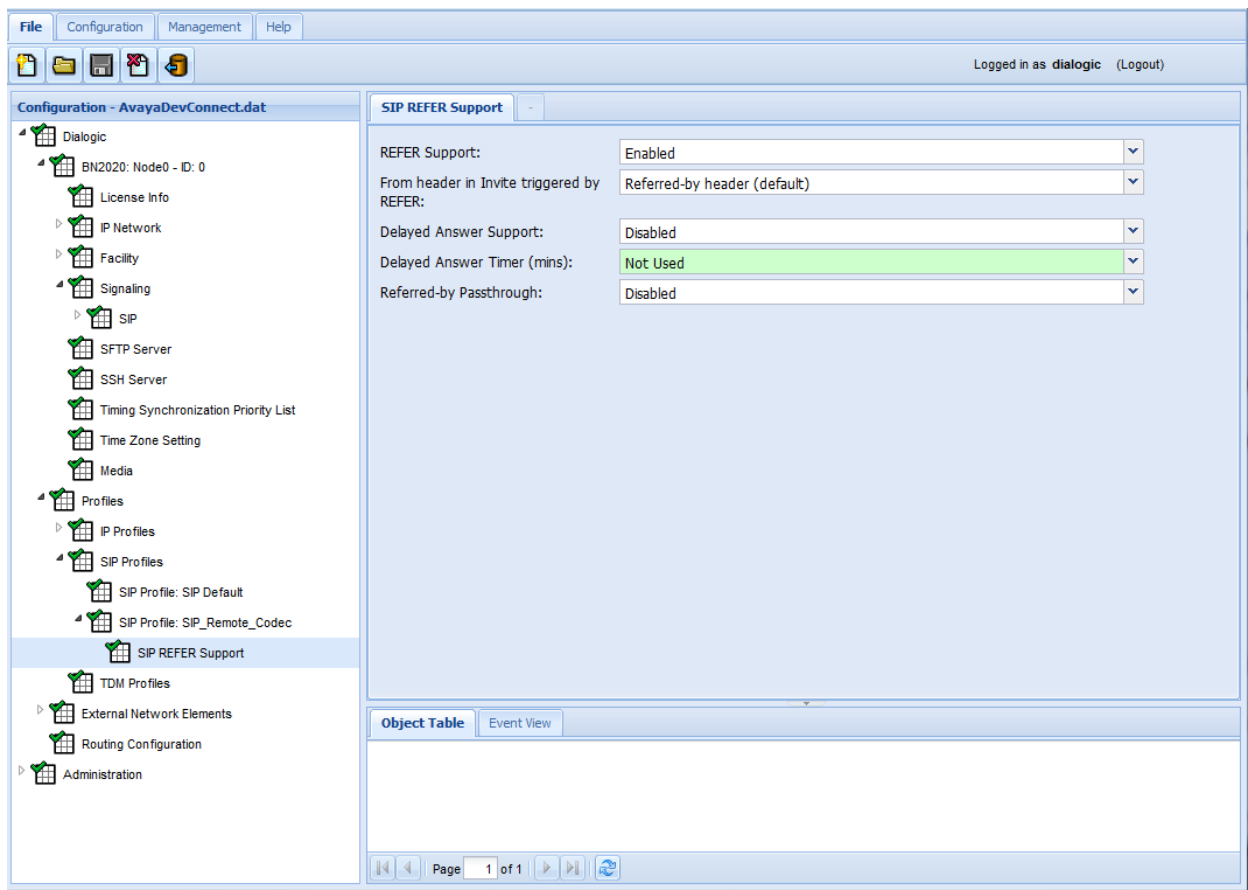
The screenshot shows the 'SIP Profile: SIP_Remote_Codec' configuration window. The left pane displays a configuration tree with 'Dialogic' expanded, showing various settings like License Info, IP Network, Facility, Signaling, ISDN, SIP, SFTP Server, SSH Server, Timing Synchronization Priority List, Time Zone Setting, Media, Profiles, IP Profiles, and SIP Profiles. The 'SIP Profiles' folder is selected, showing 'SIP Profile: SIP Default' and 'SIP Profile: SIP_Remote_Codec'. The main pane displays the configuration for 'SIP Profile: SIP_Remote_Codec' with the following fields:

Field	Value
Name	SIP_Remote_Codec
PRACK Support	Disabled
PRACK Timer (s)	150
Precondition Support	Disabled
Codec Priority	Remote
3XX Redirect Support	Enabled
Loop Detection	Enabled
Loop Detection Method	To Header
INVITE Retransmission Attempts	Retransmit All
Trusted	Enabled
Privacy	Disabled
PAID RPID Display Name	When none received send user part of URI
INFO Keep-Alive Support	Disabled
Outbound Delayed Media	Disabled
SRTP Mode	Disabled
180 Ringing Behavior	Send 183 Progress w/SDP

At the bottom of the window, there are tabs for 'Object Table' and 'Event View'.

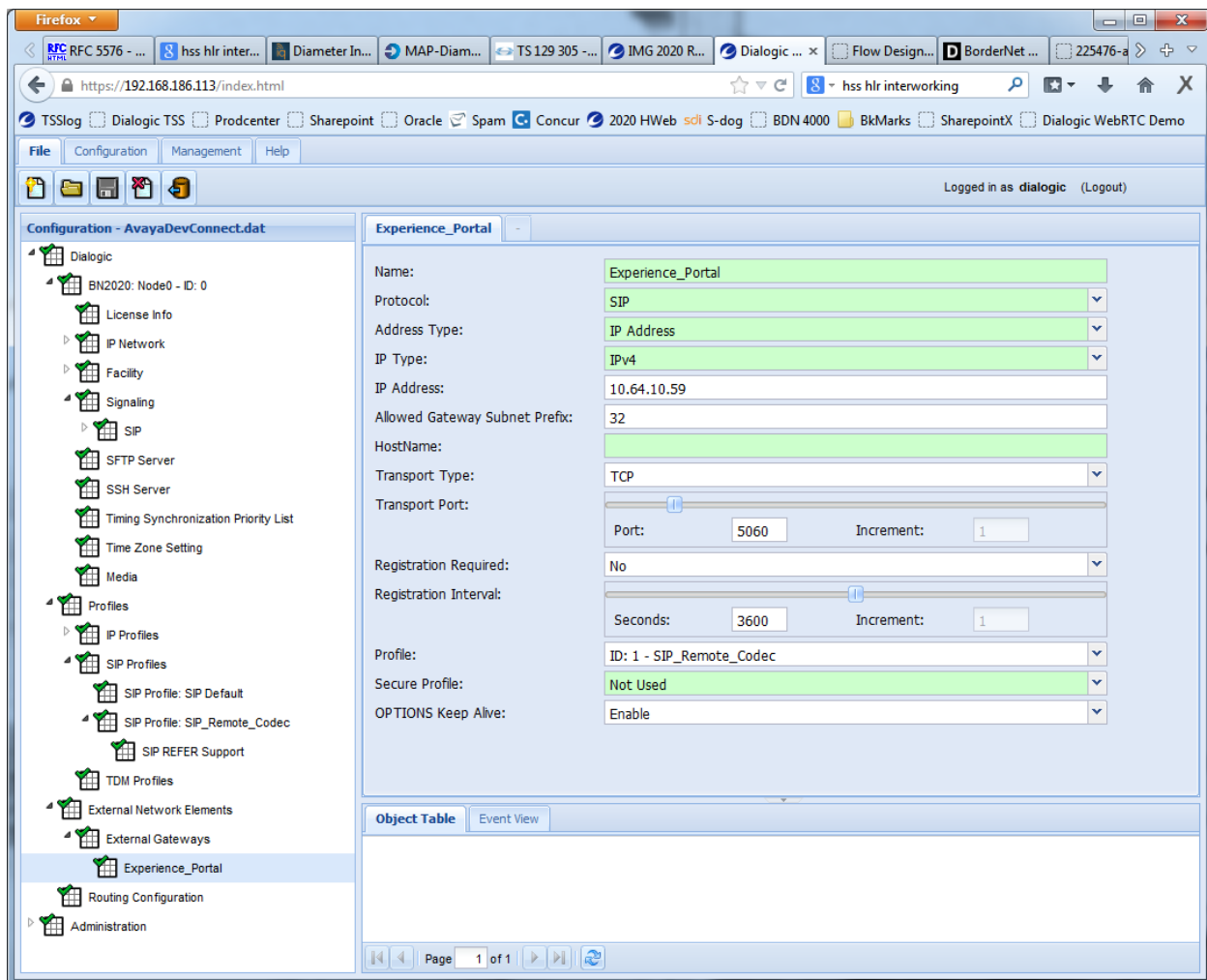
6.3.3. Configure SIP REFER Support

From the configuration tree in the left pane, right click the newly created SIP Profile **SIP_Remote_Codec** and select **New SIP REFER Support**. The **SIP REFER Support** screen is displayed. Change the **REFER Support** to **Enabled**. Keep the default values for the remaining fields. The following shows the completed **SIP REFER Support** screen.



6.4. 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 (not shown). Enter a descriptive name in the **Name** field such as **Experience_Portal**. For **Protocol**, select **SIP**. For **IP Address**, enter the IP address of the Experience Portal signaling interface. For **Transport Type** select **TCP**. Please note that the transport type should match the type configured in **Section 5.1**. For **Profile**, select the SIP Profile **SIP_Remote_Codec** configured in **Section 6.3.2**. 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 **Experience_Portal**.



6.5. Configure Routing Configuration

6.5.1. Configure Channel Group

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 (not shown). Enter a descriptive name in the **Name** field such as SIP_ExperiencePortal. For **Signaling Type**, select **SIP**. For **Incoming IP Profile** and **Outgoing IP Profile**, select the IP Profile configured in **Section 6.3.1**. Keep the default values for the remaining fields. The following shows the completed ChannelGroup screen for SIP_ExperiencePortal.

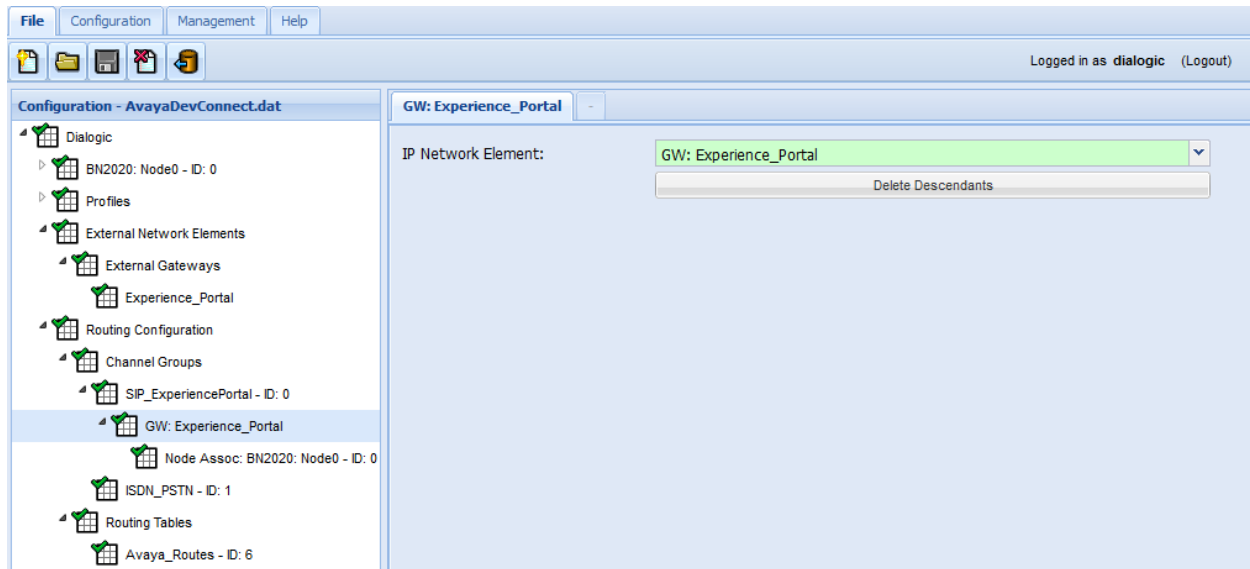
The screenshot displays the AvayaDevConnect configuration interface. The left pane shows the configuration tree with 'SIP_ExperiencePortal - ID: 0' selected under 'Channel Groups'. The right pane shows the configuration details for this channel group.

Field	Value
ID:	0
Name:	SIP_ExperiencePortal
Trunk Direction:	Incoming/Outgoing
Signaling Type:	SIP
Route Table:	None
Cause Code Table:	None
Incoming IP Profile:	IP_G711_G729_QOS
Outgoing IP Profile:	IP_G711_G729_QOS
Incoming Treatment:	Release w/Cause
Outgoing Treatment:	Release w/Cause
Incoming Translation Table:	None
Outgoing Translation Table:	None
Hunting Options:	Round Robin Clockwise
Ingress Side will Play Call Progress Tones:	False
Re-Attempt Cause Code:	Not Used

The 'Re-Attempt Cause Code' dropdown menu is expanded, showing the following options:

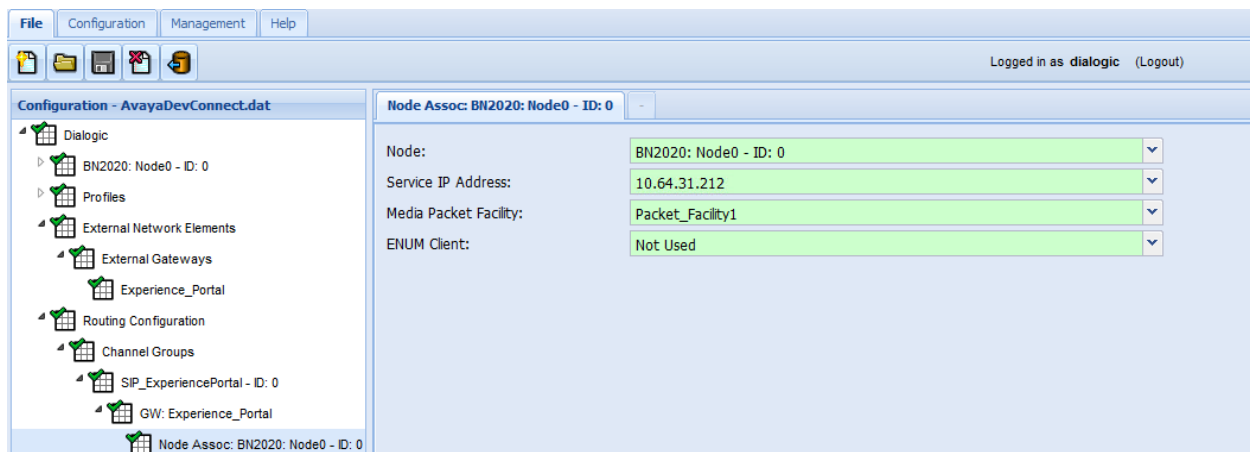
- 000 - Reserved
- 001 - Unallocated
- 002 - No Route to Specified Transit Network
- 003 - No Route to Destination

Right click the newly configured Channel Group in the left pane and select **New IP Network Element**. The **NetworkElement** screen is displayed (not shown). For **IP Network Element**, select the External Gateway configured in **Section 6.4**. The following shows the completed **NetworkElement** screen.



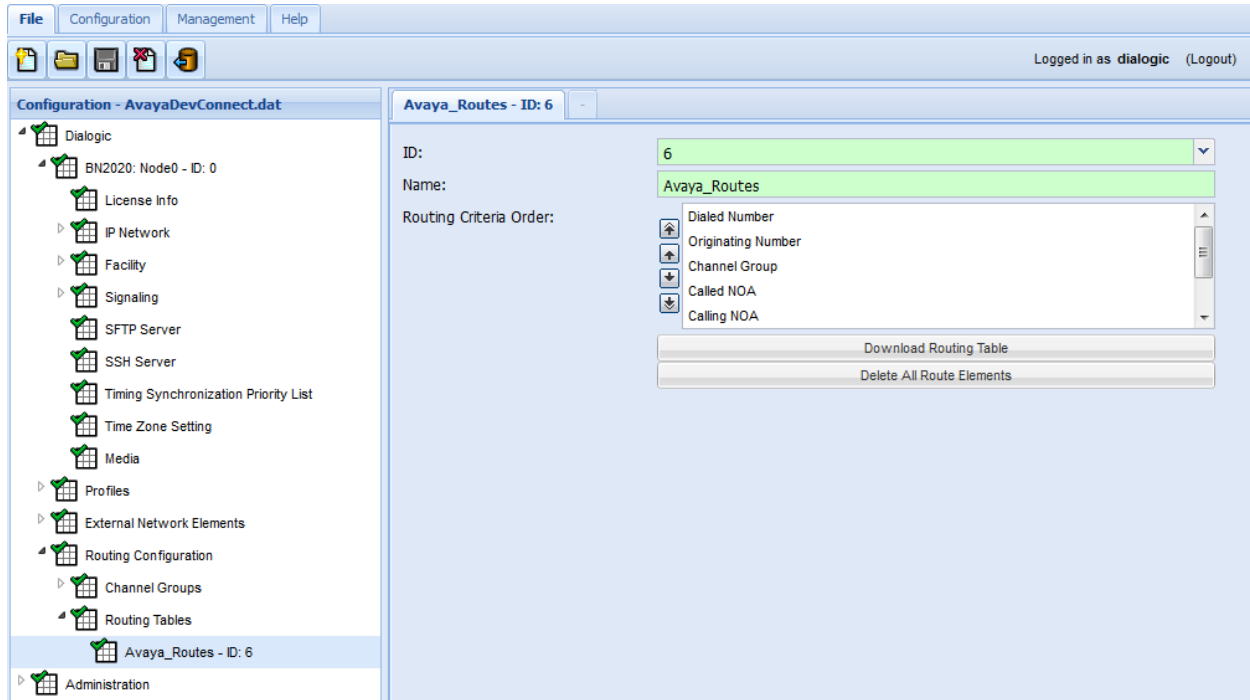
Right click the newly configured IP Network Element in the left pane and select **New Node Association**. The **Node Assoc** screen is displayed. For **Node**, **Service IP Address**, and **Media Packet Facility**, select proper values. Keep the default values for the remaining fields. The following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

Please note that Media Packet Facility was pre-configured and is not shown in this document.



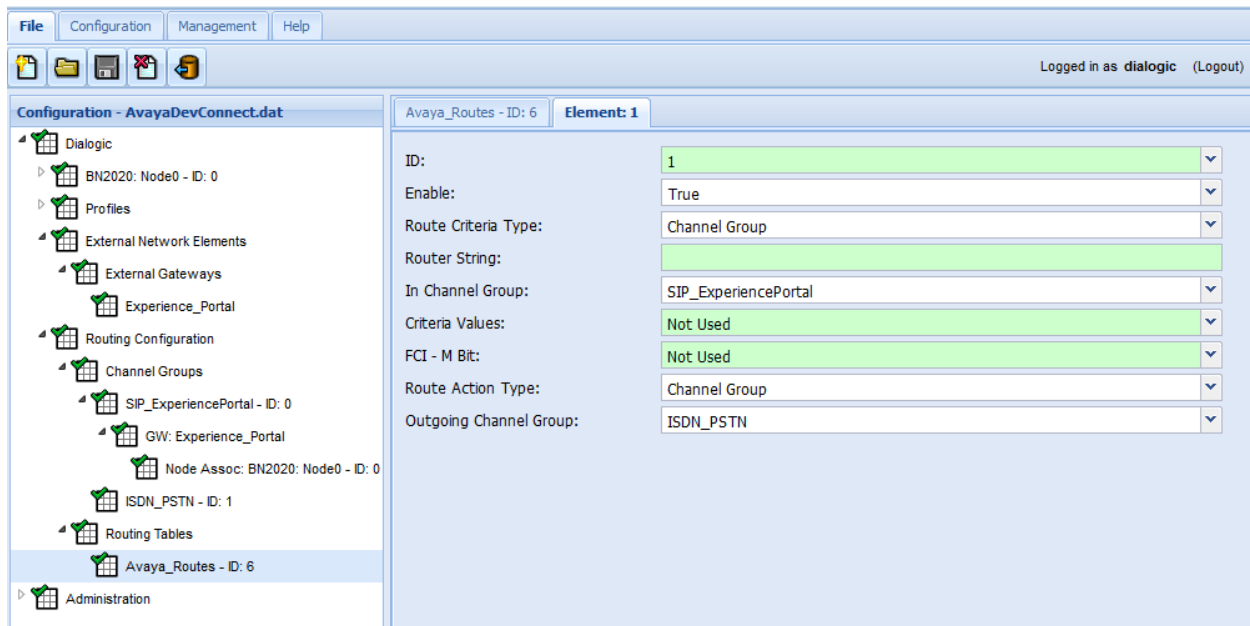
6.5.2. Add Routing Entries

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 (not shown). Enter a descriptive name in the **Name** field such as Avaya_Routes. Keep the default values for the remaining fields. The following shows the completed **Table** screen.

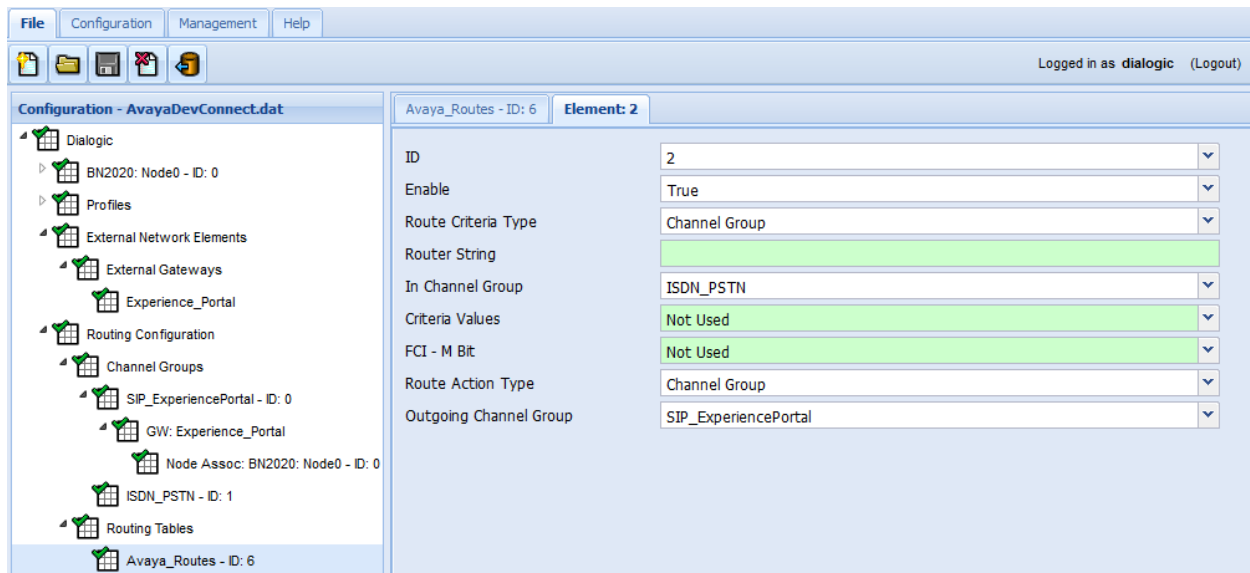


From the configuration tree in the left pane, right click **Dialogic** → **Routing Configuration** → **Routing Tables** → **Avaya_Routes** and select **New Element**.

The **Element** tab is displayed. For **Route Criteria Type**, select **Channel Group**. For **In Channel Group**, select the **SIP_ExperiencePortal** channel group configured in **Section 6.5.1**. 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.



Repeat the above procedure for a second Route Element which routes calls from the ISDN PRI channel group to the **SIP_ExperiencePortal** channel group.



6.5.3. Add Route Table to Channel Groups

Bring up the channel group configured in Section 6.5.1 by navigating to **Dialogic → Routing Configuration → Channel Groups → SIP_ExperiencePortal**. For **Route Table**, select the **Avaya_Routes** route table configured in Section 6.5.2. The following shows the updated **ChannelGroup** screen for **SIP_ExperiencePortal**.

The screenshot displays the Dialogic configuration application. The left-hand pane shows a tree view of the configuration hierarchy. The right-hand pane shows the configuration details for the selected 'SIP_ExperiencePortal - ID: 0'.

Field	Value
ID:	0
Name:	SIP_ExperiencePortal
Trunk Direction:	Incoming/Outgoing
Signaling Type:	SIP
Route Table:	Avaya_Routes - ID: 6
Cause Code Table:	None
Incoming IP Profile:	IP_G711_G729_QOS
Outgoing IP Profile:	IP_G711_G729_QOS
Incoming Treatment:	Release w/Cause
Outgoing Treatment:	Release w/Cause
Incoming Translation Table:	None
Outgoing Translation Table:	None
Hunting Options:	Round Robin Clockwise
Ingress Side will Play Call Progress Tones:	False

Repeat the above procedure for the channel group pre-configured for the ISDN PRI interface.

7. Verification Steps

This section provides the verification steps that may be performed to verify that Avaya Aura[®] Experience Portal can establish and receive calls from BorderNet 2020.

7.1. Verify SIP Connections on Avaya Aura[®] Experience Portal

On the EPM navigate to **Real-Time Monitoring → Port Distribution**. Select the MPP that was used for configuration in this document and select **OK** (not shown). Verify that the **Mode** is **Online** and **State** is **In service**. This ensures successful SIP connectivity between Experience Portal and BorderNet 2020.

Port Distribution Report (Jun 20, 2014 4:09:00 AM MDT)



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.

Servers: MPPLocal

Total Ports: 10

Last Poll: Jun 20, 2014 4:08:57 AM MDT

Port	Mode	State	Port Group	Protocol	Current Allocation	Base Allocation
10	Online	In service	Dialogic	SIP_Trunk	MPPLocal	

Help

8. Conclusion

These Application Notes describe the configuration steps required for Dialogic[®] BorderNet[™] 2020 Integrated Media Gateway to successfully interoperate with Avaya Aura[®] Experience Portal. All feature and serviceability test cases were completed.

9. Additional References

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

[1] Administering Avaya[®] Aura Experience Portal, Release 7.0.

Dialogic[®] BorderNet[™] 2020 Integrated Media Gateway references are available on <http://www.dialogic.com/en/products/session-border-controllers/bordernet-2020.aspx>.

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