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.

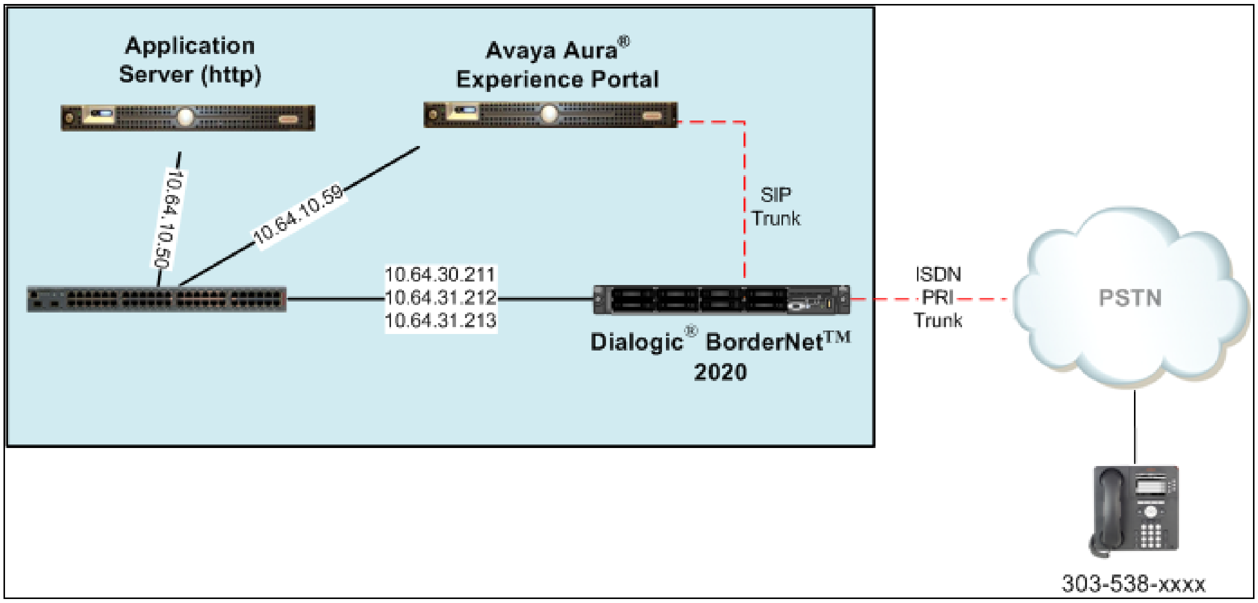


Figure 1 – Sample configuration for Avaya Aura® Experience Portal with Dialogic® BorderNet™ 2020 using Sip Trunking

4. Equipment and Software Validated

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

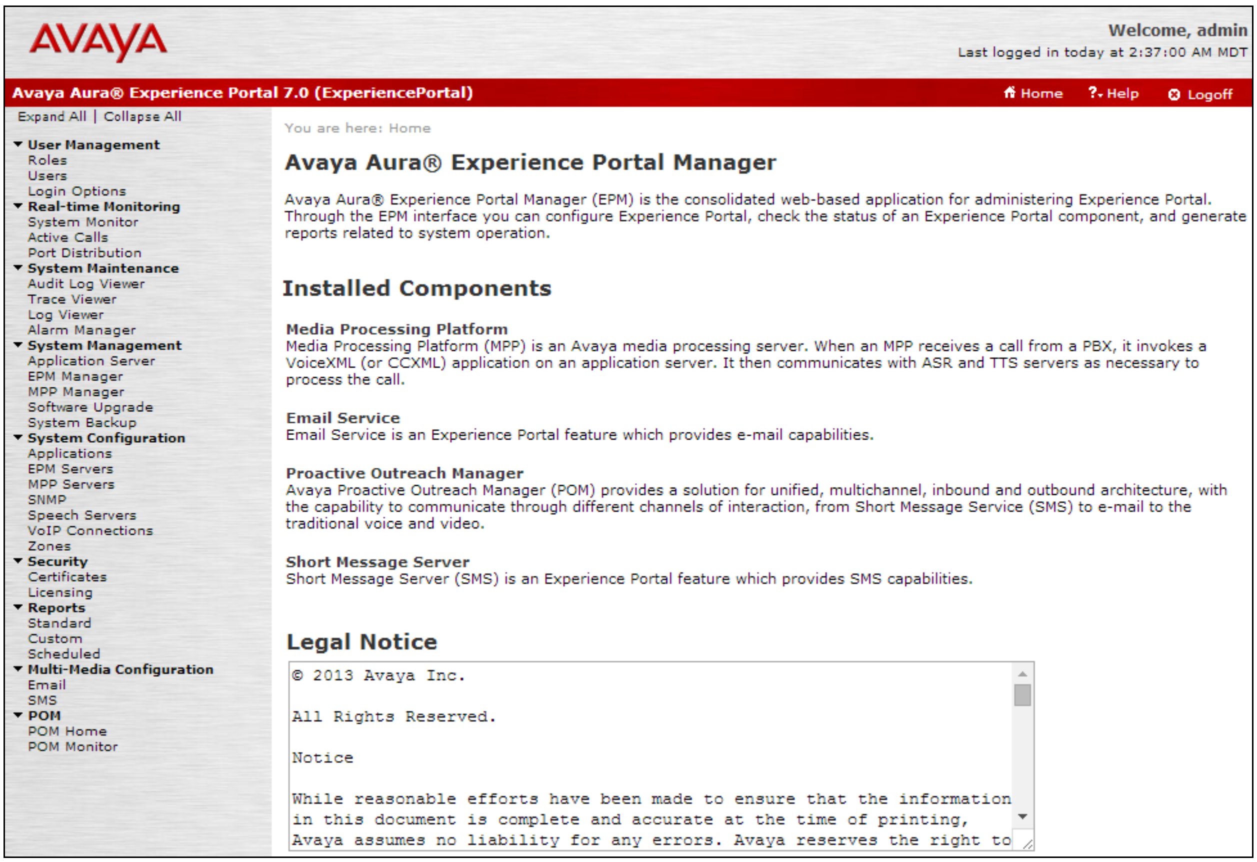
Equipment/Software	Version
Avaya Aura® Experience Portal: • Experience Portal Manager (EPM) • Media Processing Platform (MPP)	7.0
Dialogic® BorderNet™ 2020 • Integrated Media Gateway • Dialogic® WebUI	2.2 SP2 b1561 2.2 SP2

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:

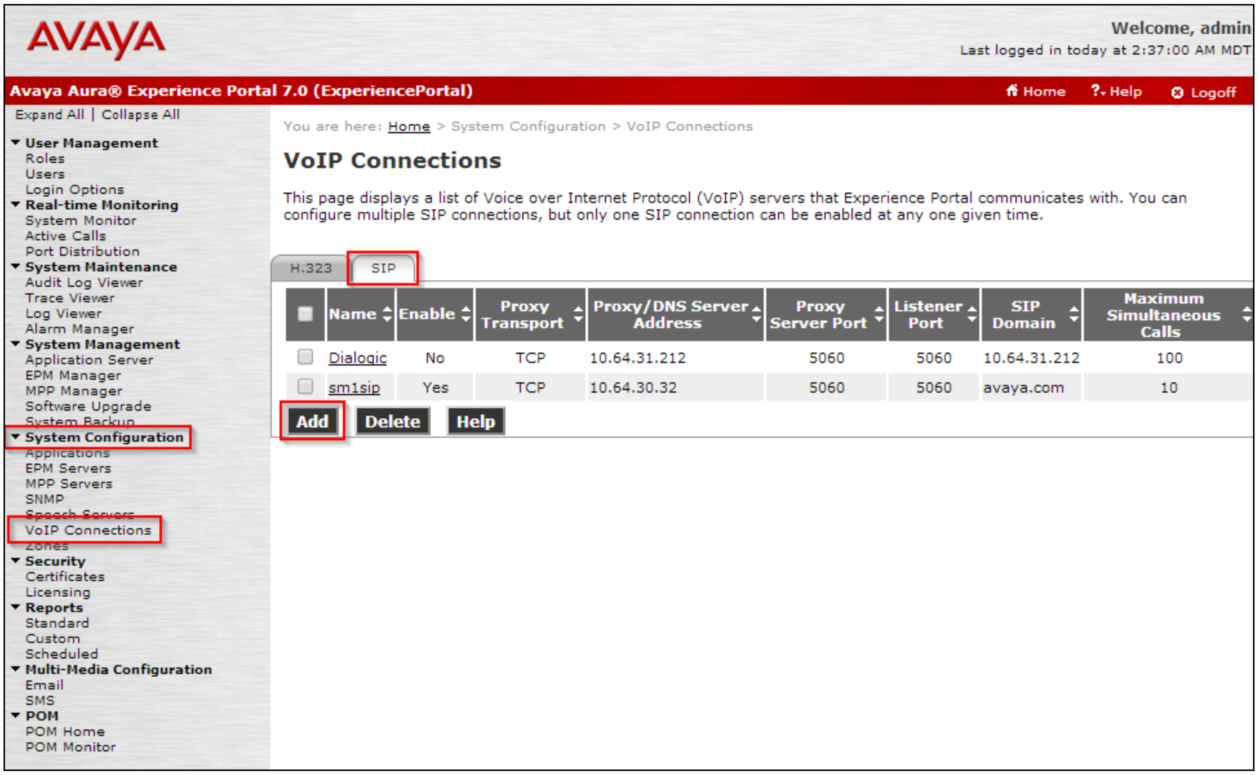
- Configure SIP connection
- Add MPP server
- Add speech server
- Add voice application
- Start MPP server

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.



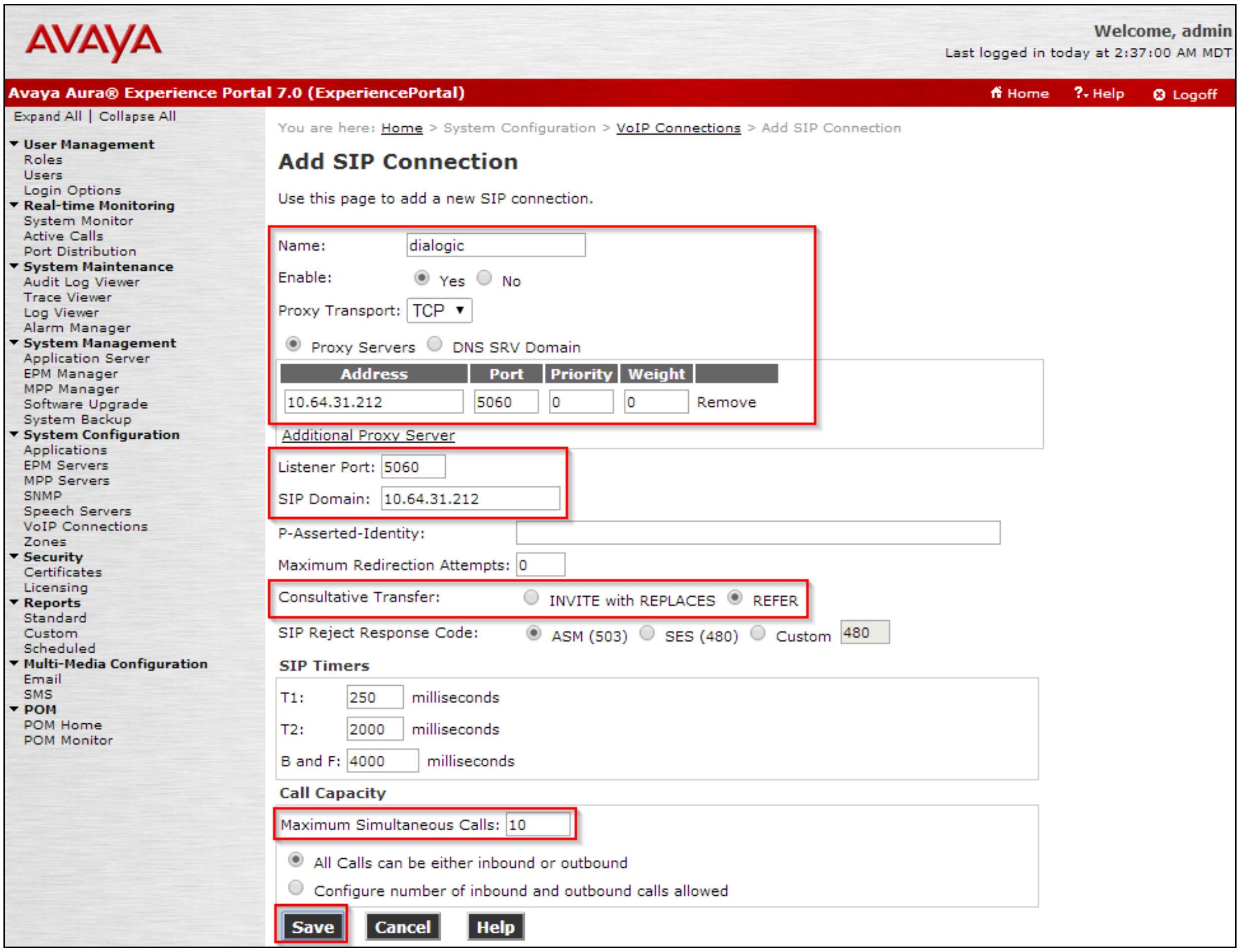
5.1. Configure SIP Connection

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.



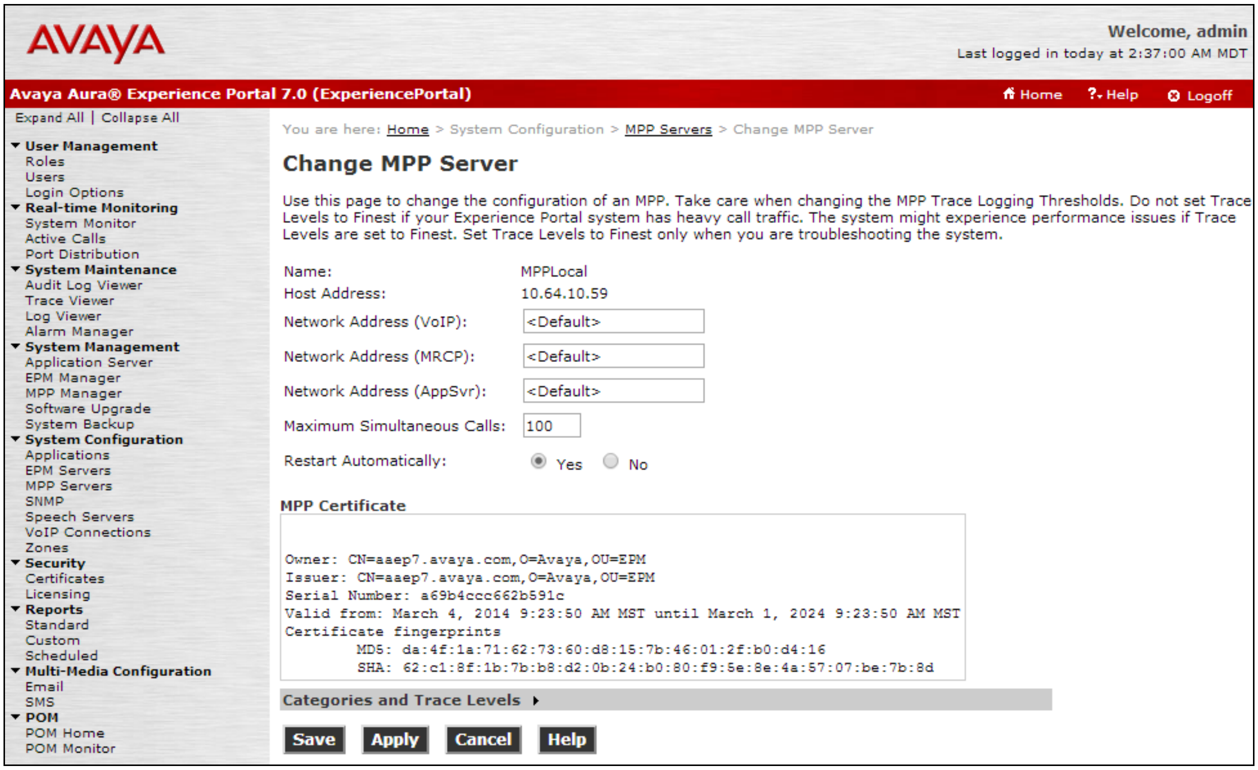
On the resulting screen, configure the parameters as follows:

- 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)
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
Application Server
EPM Manager
MPP Manager
Software Upgrade
System Backup
System Configuration
Applications
EPM Servers
MPP Servers
SNMP
Speech Servers
VoIP Connections
Zones
Security
Certificates
Licensing
Reports
Standard
Custom
Scheduled
Multi-Media Configuration
Email
SMS
POM
POM Home
POM Monitor
You are here: Home > System Configuration > VoIP Connections > Add SIP Connection
Add SIP Connection
Use this page to add a new SIP connection.
Name: dialogic
Enable: Yes No
Proxy Transport: TCP
Proxy Servers DNS SRV Domain
Address Port Priority Weight
10.64.31.212 5060 0 0 Remove
Additional Proxy Server
Listener Port: 5060
SIP Domain: 10.64.31.212
P-Asserted-Identity:
Maximum Redirection Attempts: 0
Consultative Transfer: INVITE with REPLACES REFER
SIP Reject Response Code: ASM (503) SES (480) Custom 480
SIP Timers
T1: 250 milliseconds
T2: 2000 milliseconds
B and F: 4000 milliseconds
Call Capacity
Maximum Simultaneous Calls: 10
All Calls can be either inbound or outbound
Configure number of inbound and outbound calls allowed
Save
Cancel
Help

5.2. Add MPP Server

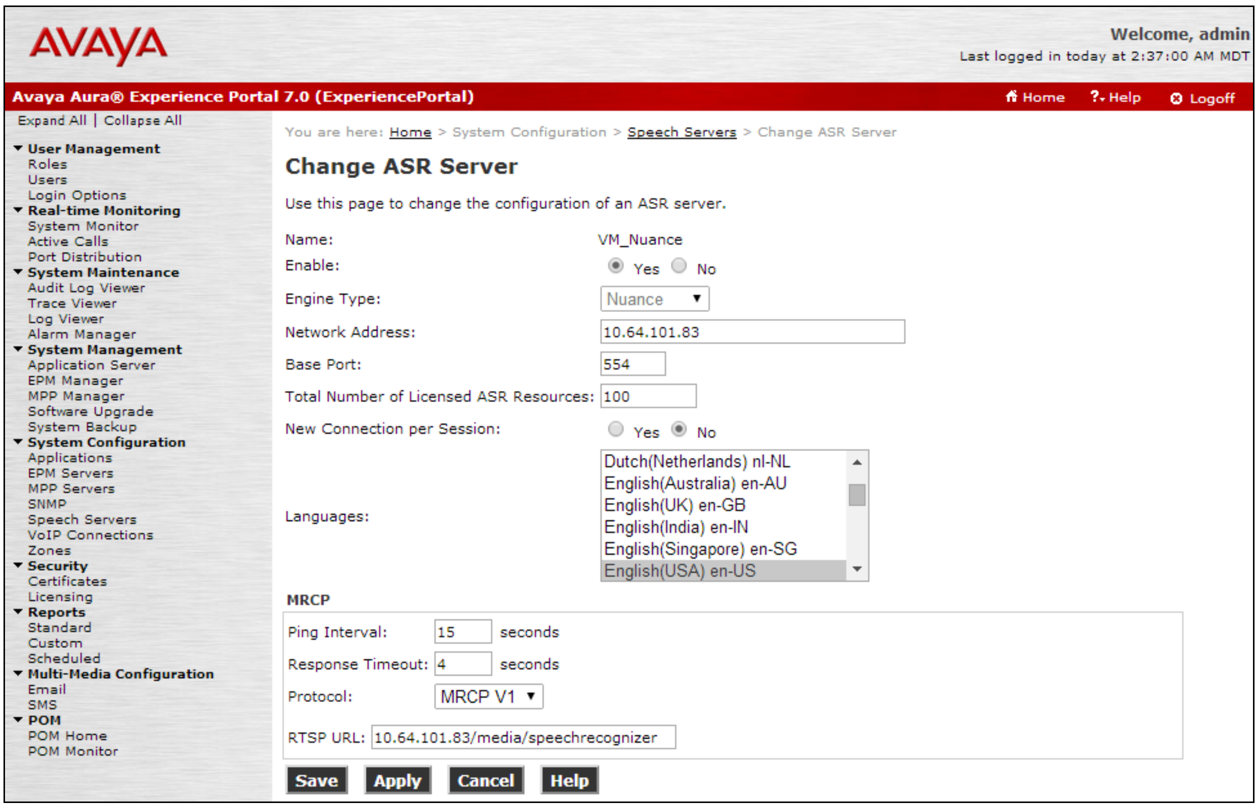
Add a Media Processing Platform (MPP) server by navigating to **System Configuration → MPP Servers**. Click the **Add** button to add a new MPP Server (not shown). In the MPP Server configuration page, specify a descriptive name and the host address of the MPP server. Also, specify the **Maximum Simultaneous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



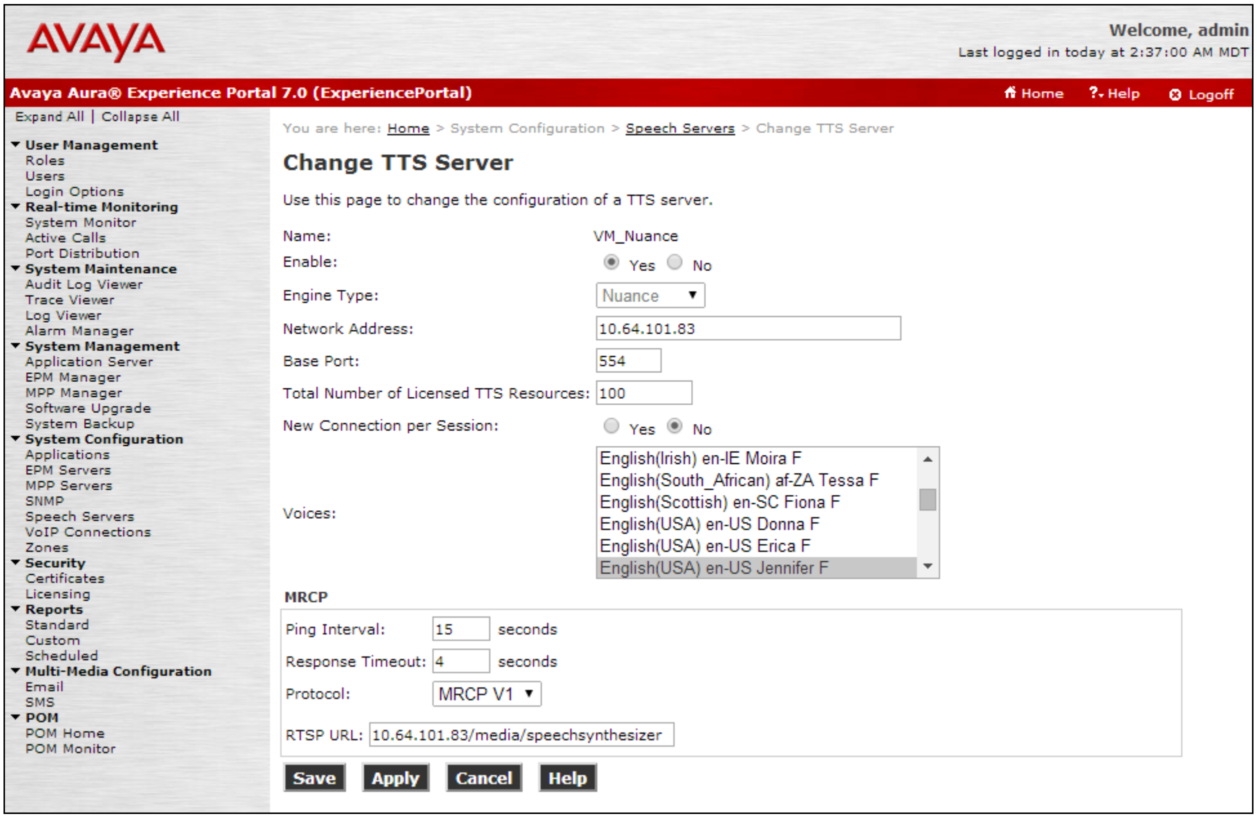
5.3. Add Speech Server

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.



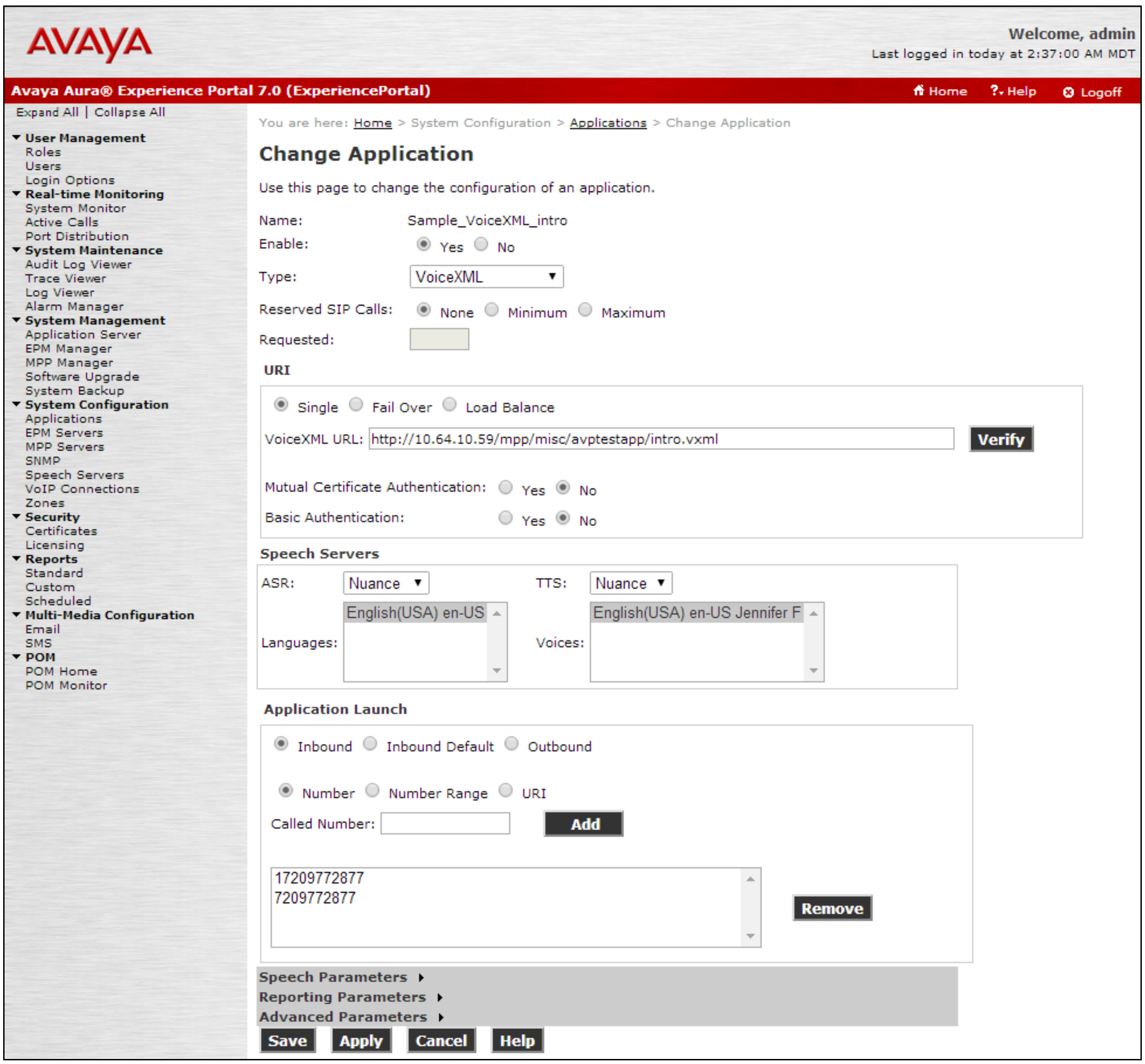
To configure the TTS server, navigate to **System Configuration → Speech Servers**, select the **TTS** tab (not shown), and then click **Add**. The screen below shows the configuration for the TTS server used during compliance testing. In this configuration, a Nuance TTS server was used so the engine 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.



5.4. Add Voice Application

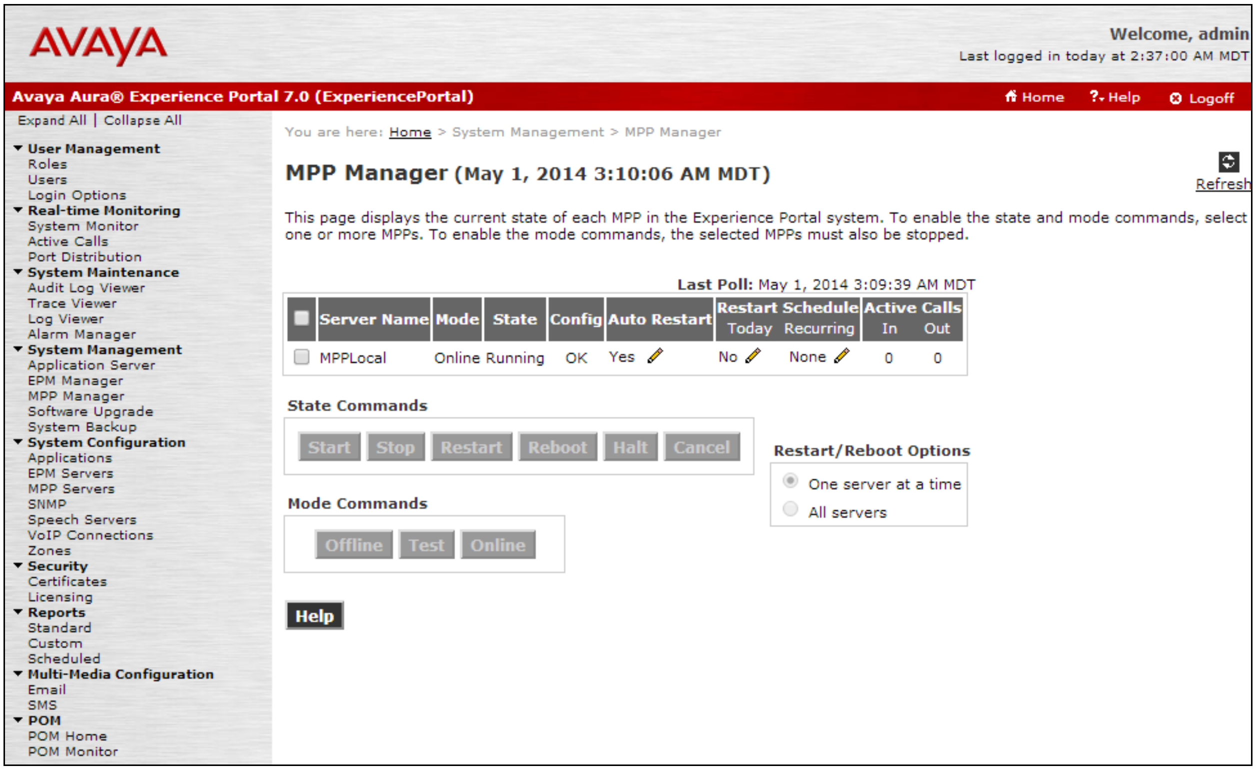
Adding a voice application for Experience Portal is part of Experience Portal's standard administration. This configuration is not directly related to achieving interoperability between the BorderNet 2020 and Experience Portal. It is included here for completeness.

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.



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**.



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.

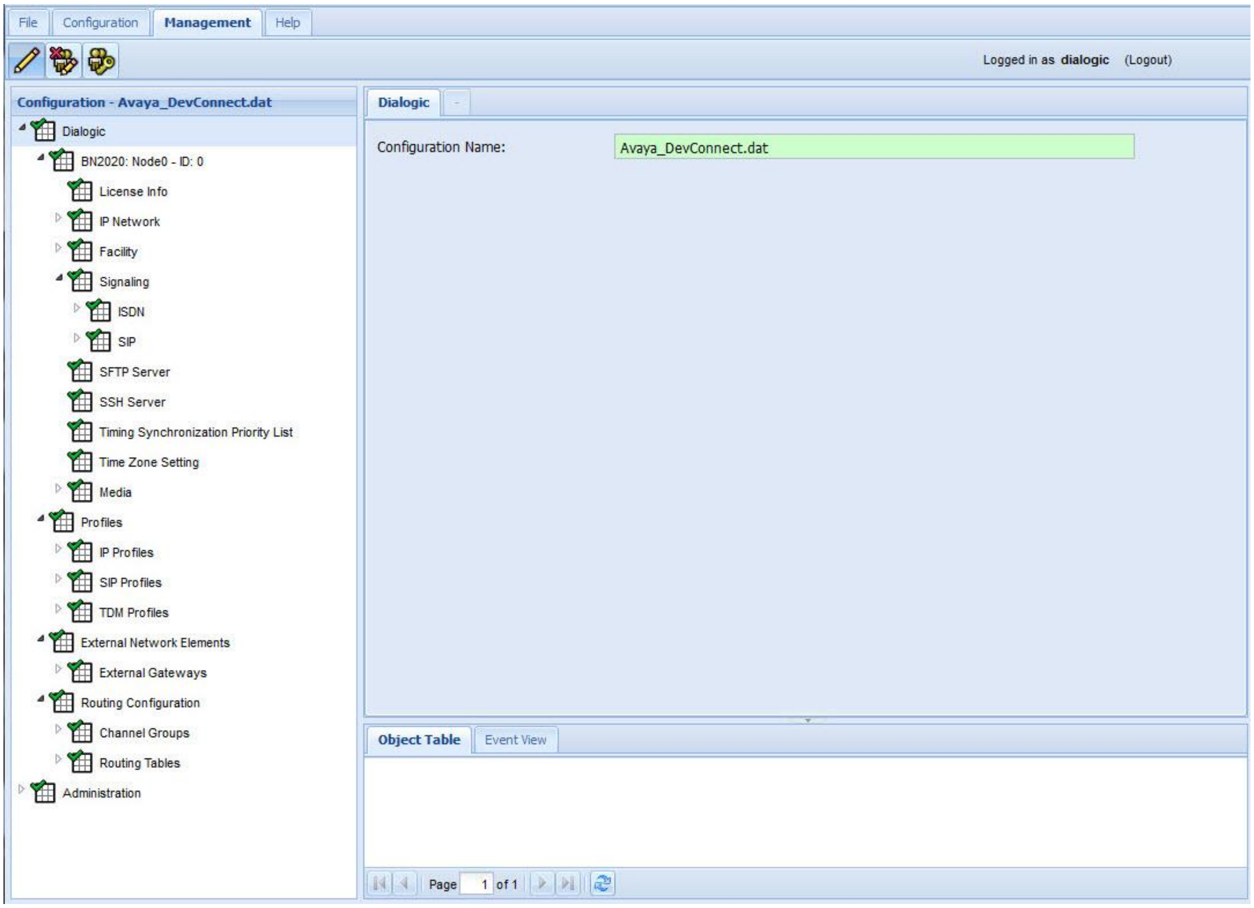
It is also assumed that the PSTN trunk has been properly configured, which includes the ISDN PRI interface, TDM Profile, associated Channel Group, and the underlining T1 interface.

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

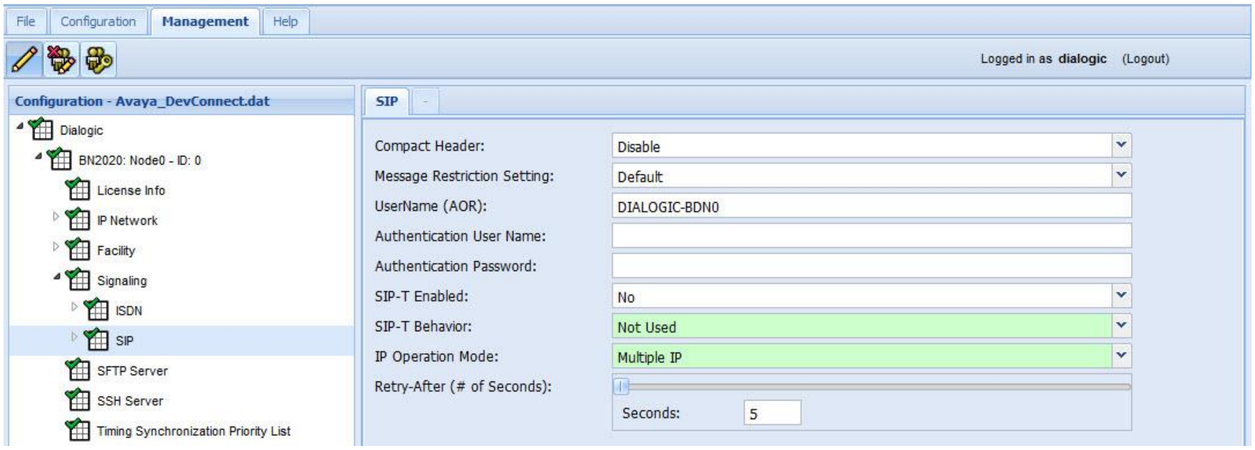
BorderNet 2020 is administered using a built-in web based management user interface. To access the interface, enter **http://<ip-addr>** as the URL in a Firefox web browser where **<ip-addr>** is the IP address of the Dialogic management port. Currently Firefox and Internet Explorer are the only officially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


File
Configuration
Management
Help
Logged in as dialogic (Logout)
Configuration - Avaya_DevConnect.dat
Dialogic
BN2020: Node0 - ID: 0
License Info
IP Network
Facility
Signaling
ISDN
SIP
SFTP Server
SSH Server
Timing Synchronization Priority List
Time Zone Setting
Media
Profiles
IP Profiles
SIP Profiles
TDM Profiles
External Network Elements
External Gateways
Routing Configuration
Channel Groups
Routing Tables
Administration
Dialogic
Configuration Name:
Avaya_DevConnect.dat
Object Table
Event View
Page 1 of 1

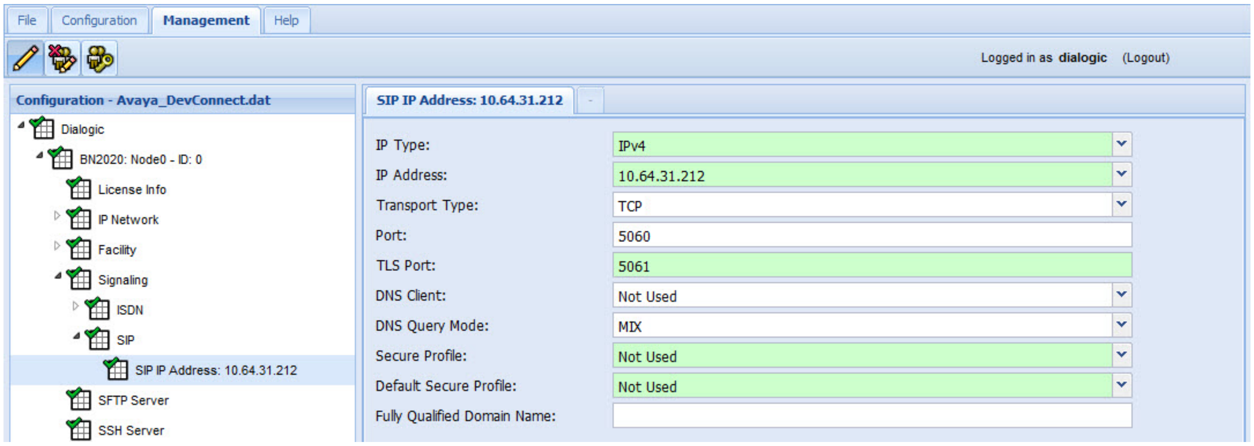
6.2. Configure BN2020 Node

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.



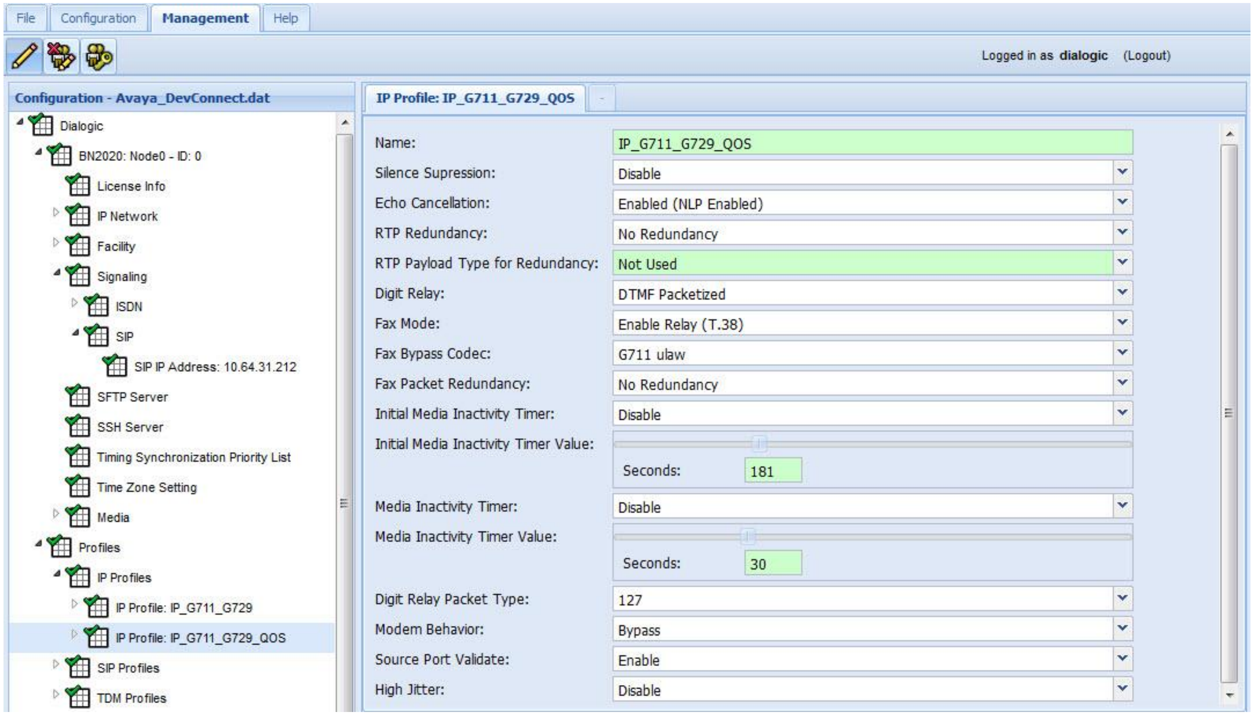
Right click **Dialogic → BN2020 Node0 → Signaling → SIP** and select **New SIP IP Address**. The **SIP IP Address** screen is displayed. For **IP Address**, select the signaling IP address from the dropdown menu. Set **Transport Type** to **TCP** and **Port** to **5060**. Please note that the port should 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.



6.3. Configure Profiles

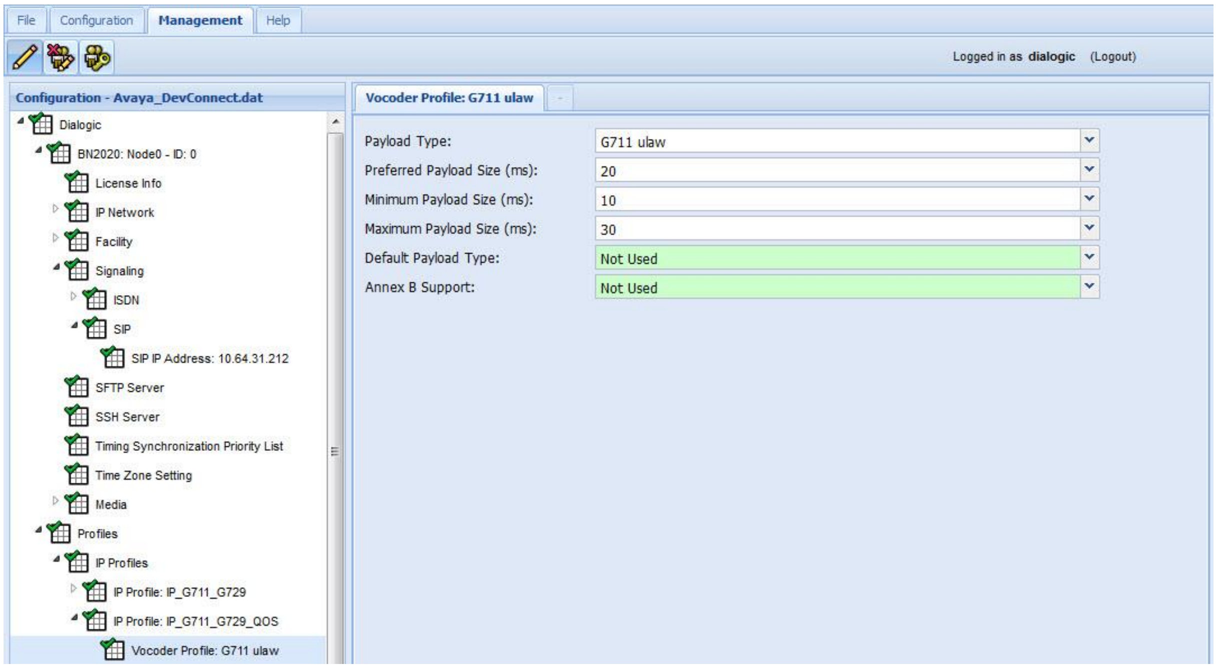
6.3.1. Configure IP Profiles

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.



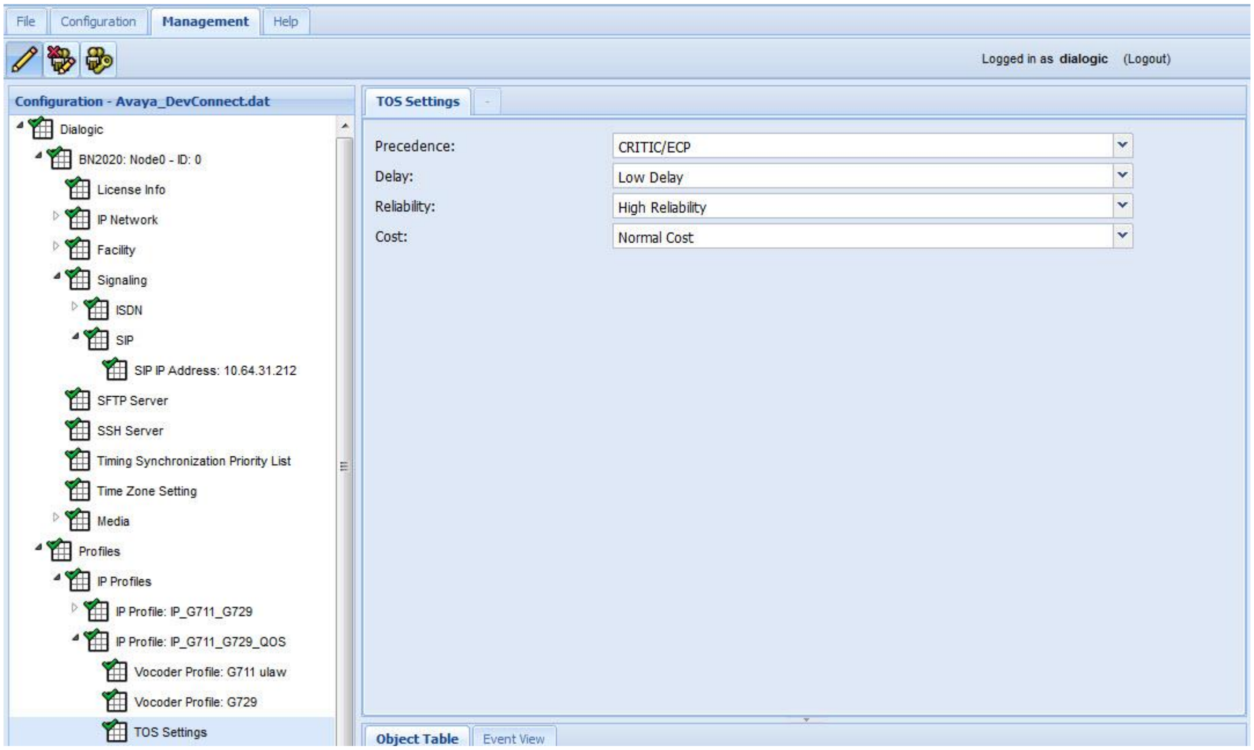
6.3.1.1 Configure IP Codecs in IP Profile

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.



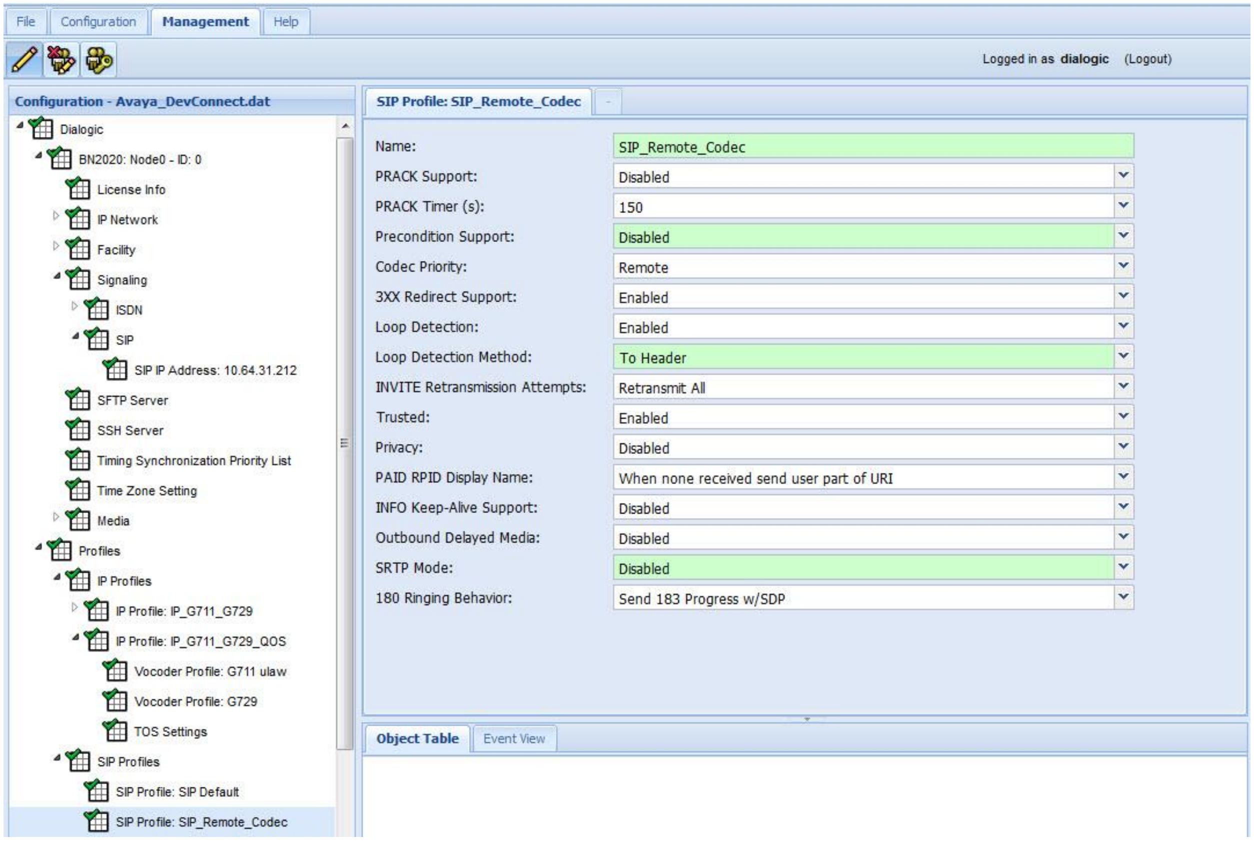
6.3.1.2 Configure TOS Settings in IP Profile

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**, s**elect **CRITIC/ECP**, **Low Delay**, **High Reliability**, and **Normal Cost** respectively. The following shows the completed **TOS Settings** screen.



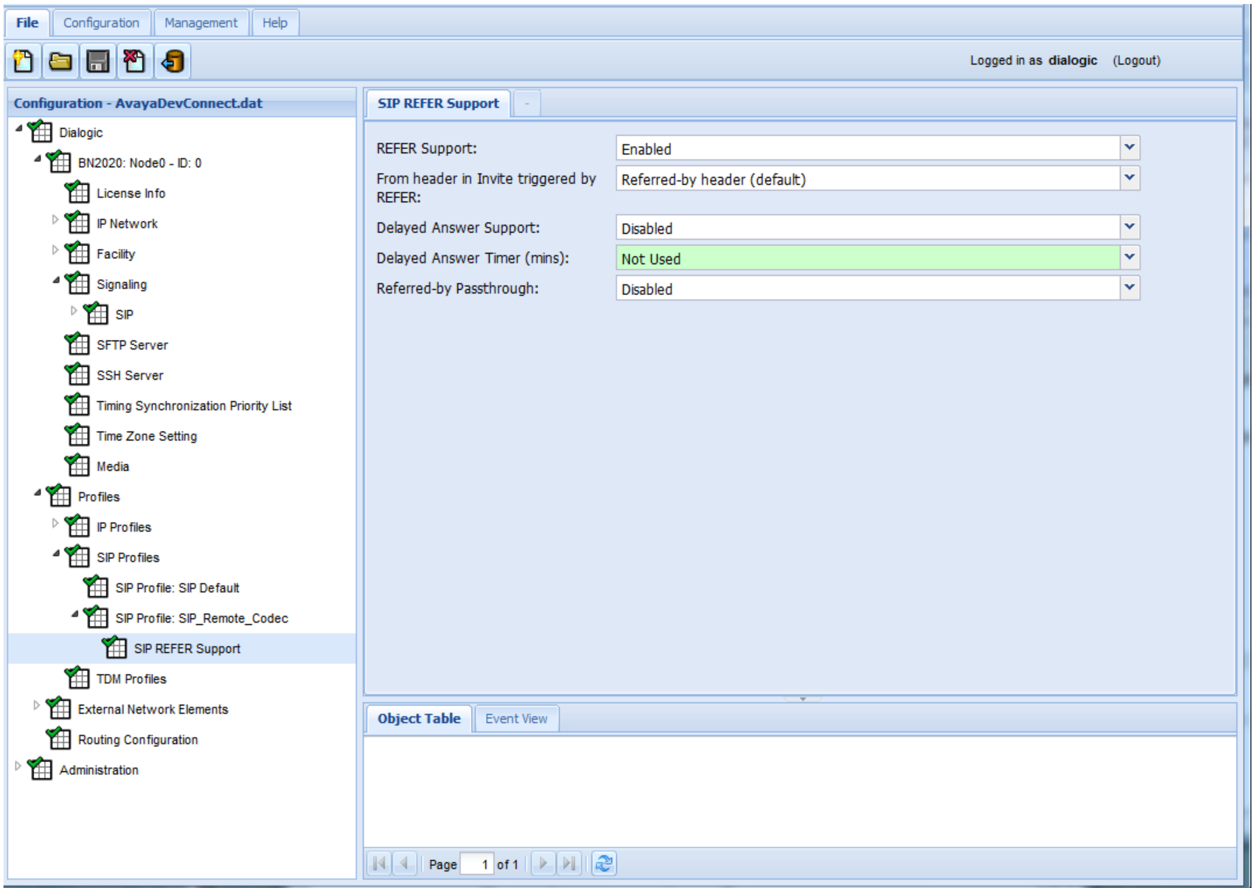
6.3.2. Configure SIP Profiles

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.



6.3.3. Configure SIP REFER Support

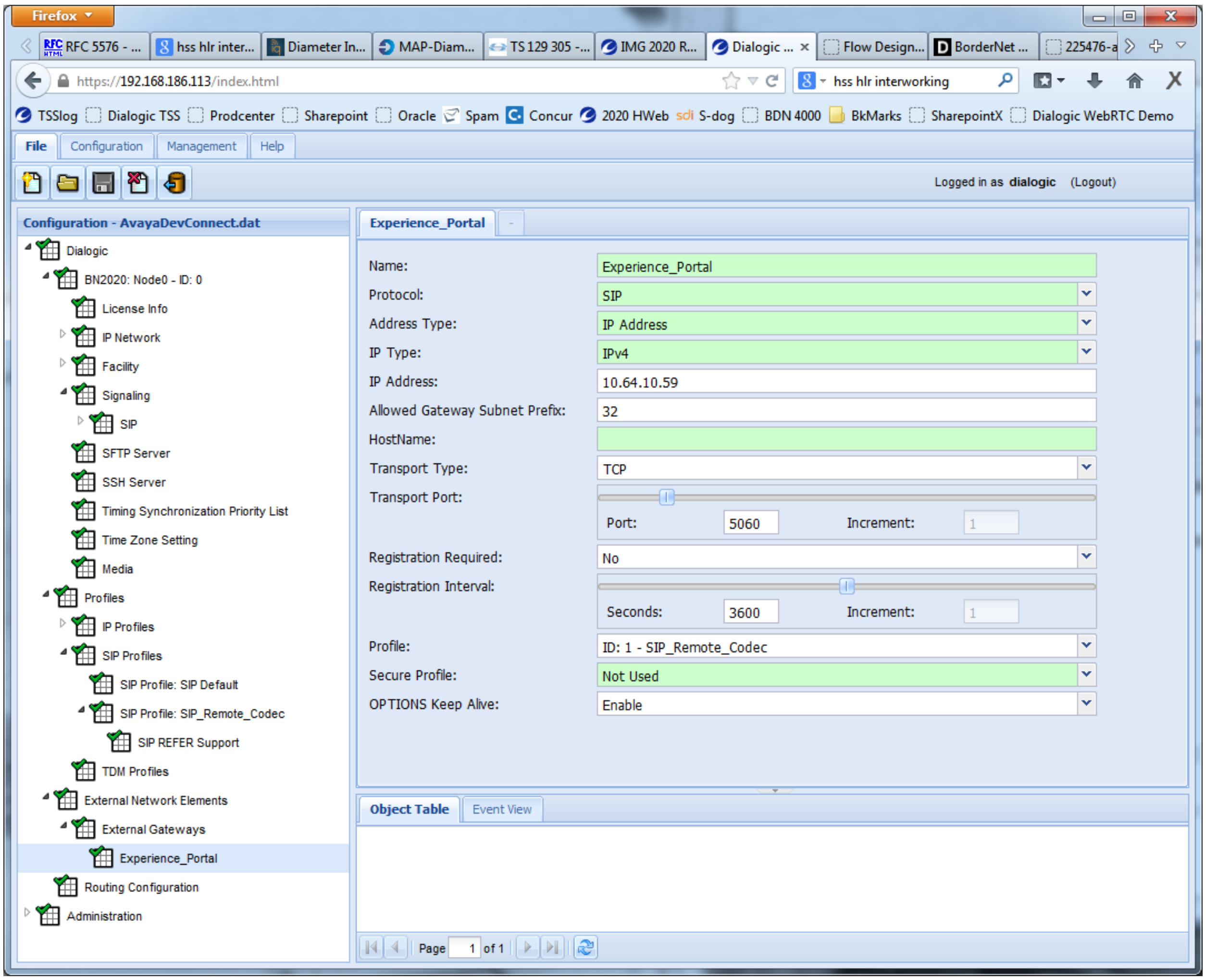
From the configuration tree in the left pane, right click the newly created SIP Profile **SIP_Remote_Codec** and select **New SIP REFER Support**. The **SIP REFER Support** screen is displayed. Change the **REFER Support** to **Enabled**. Keep the default values for the remaining fields. The following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


File
Configuration
Management
Help
Logged in as dialogic (Logout)
Configuration - AvayaDevConnect.dat
Dialogic
BN2020: Node0 - ID: 0
License Info
IP Network
Facility
Signaling
SIP
SFTP Server
SSH Server
Timing Synchronization Priority List
Time Zone Setting
Media
Profiles
IP Profiles
SIP Profiles
SIP Profile: SIP Default
SIP Profile: SIP_Remote_Codec
SIP REFER Support
TDM Profiles
External Network Elements
Routing Configuration
Administration
SIP REFER Support
REFER Support:
Enabled
From header in Invite triggered by REFER:
Referred-by header (default)
Delayed Answer Support:
Disabled
Delayed Answer Timer (mins):
Not Used
Referred-by Passthrough:
Disabled
Object Table
Event View
Page 1 of 1

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.

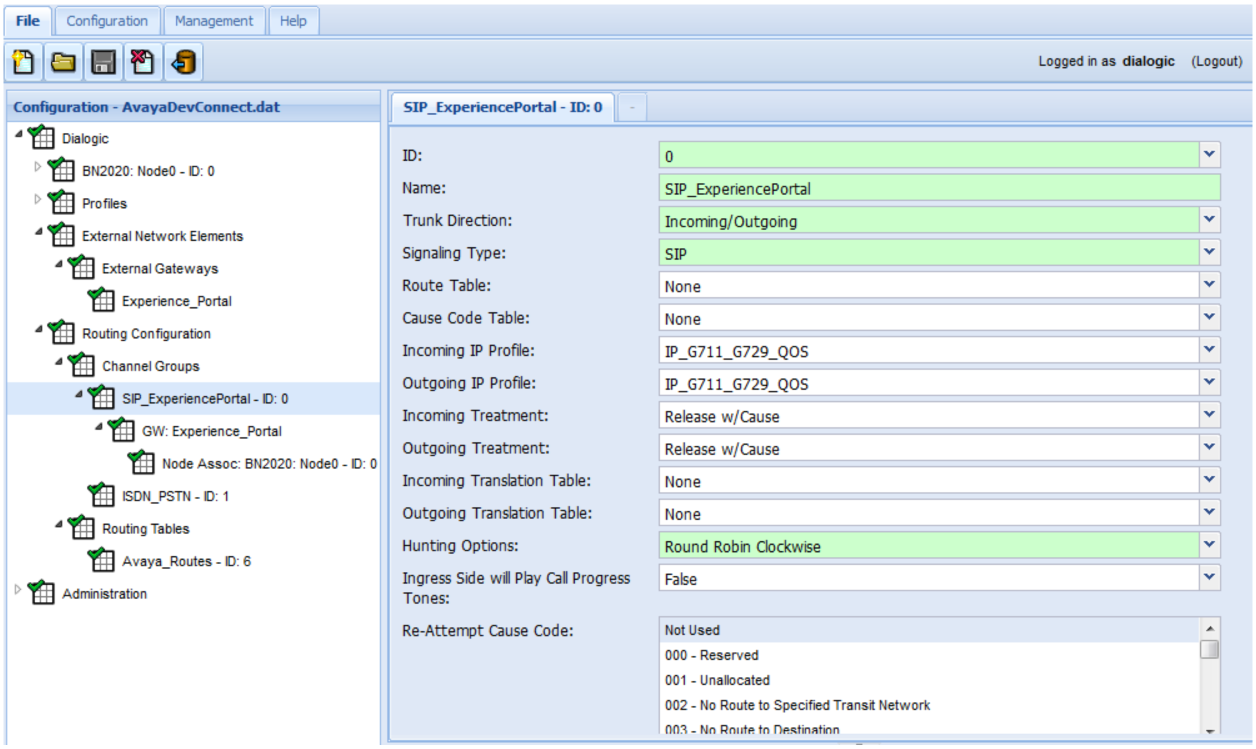
The following shows the completed **ExternalGateway** screen for **Experience_Portal**.



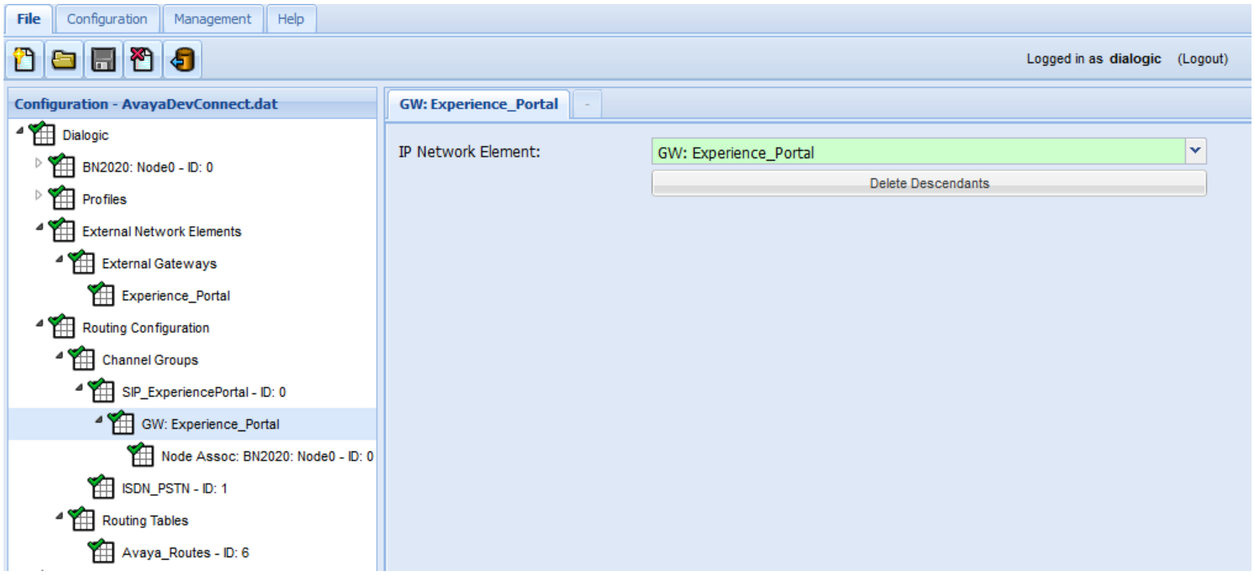
6.5. Configure Routing Configuration

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.

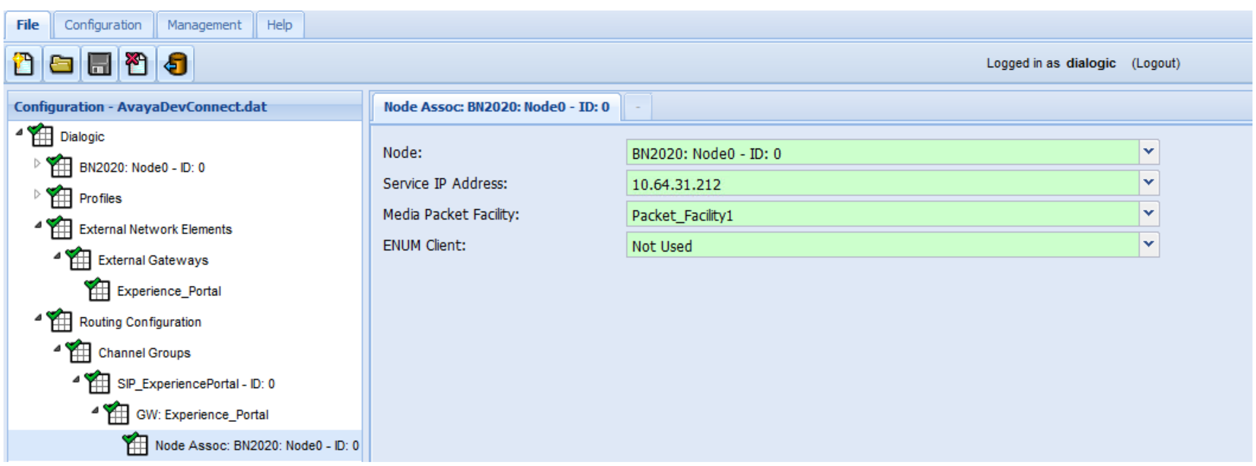


Right click the newly configured Channel Group in the left pane and select **New IP Network Element**. The **NetworkElement** screen is displayed (not shown). For **IP Network Element**, select the External Gateway configured in **Section 6.4**. The following shows the completed **NetworkElement** screen.



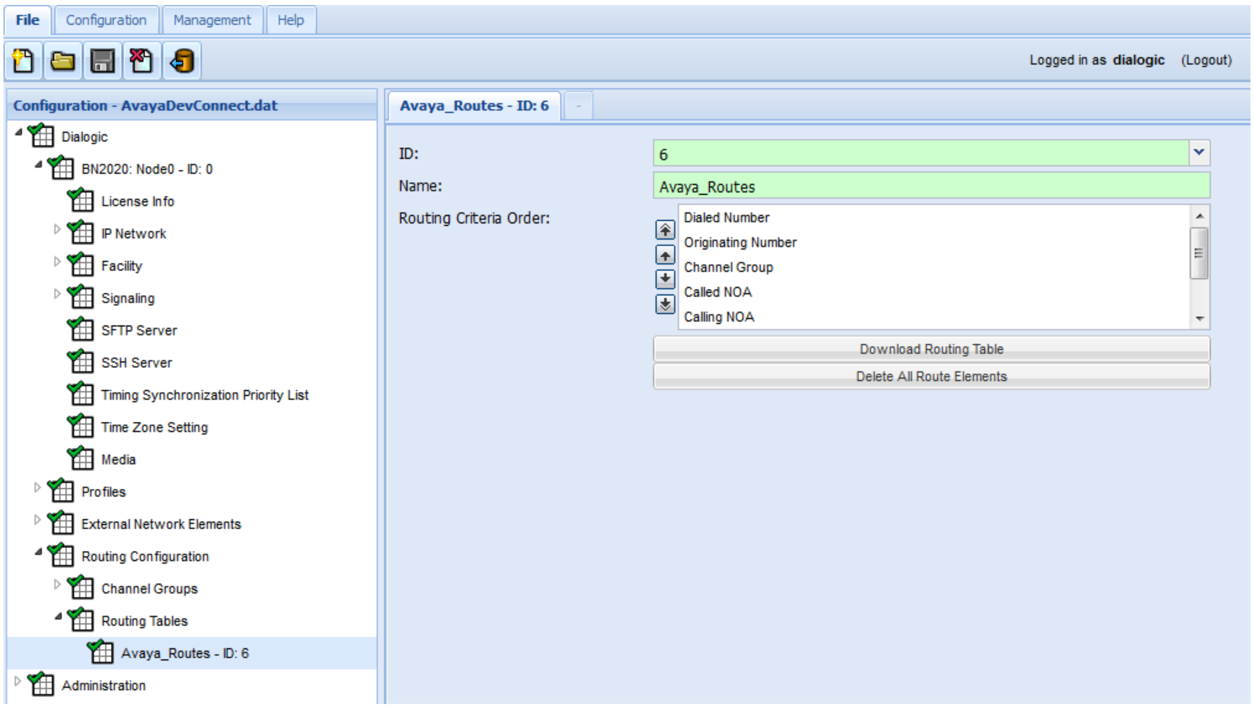
Right click the newly configured IP Network Element in the left pane and select **New Node Association**. The **Node Assoc** screen is displayed. For **Node**, **Service IP Address**, and **Media Packet Facility**, select proper values. Keep the default values for the remaining fields. The following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

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



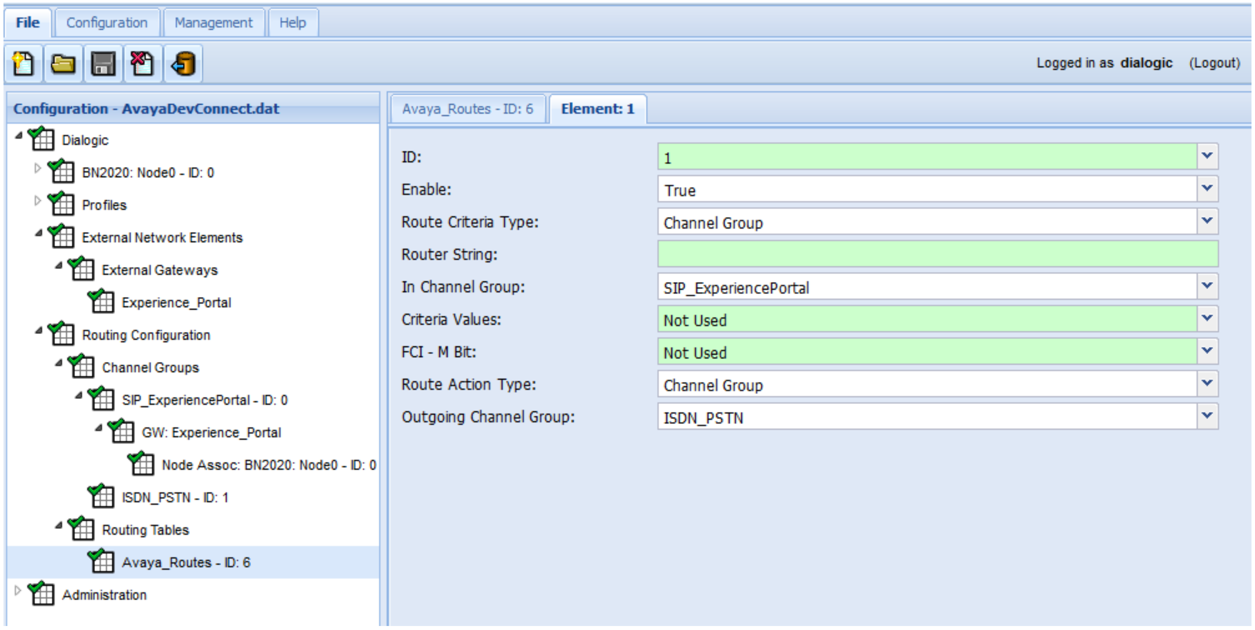
6.5.2. Add Routing Entries

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.

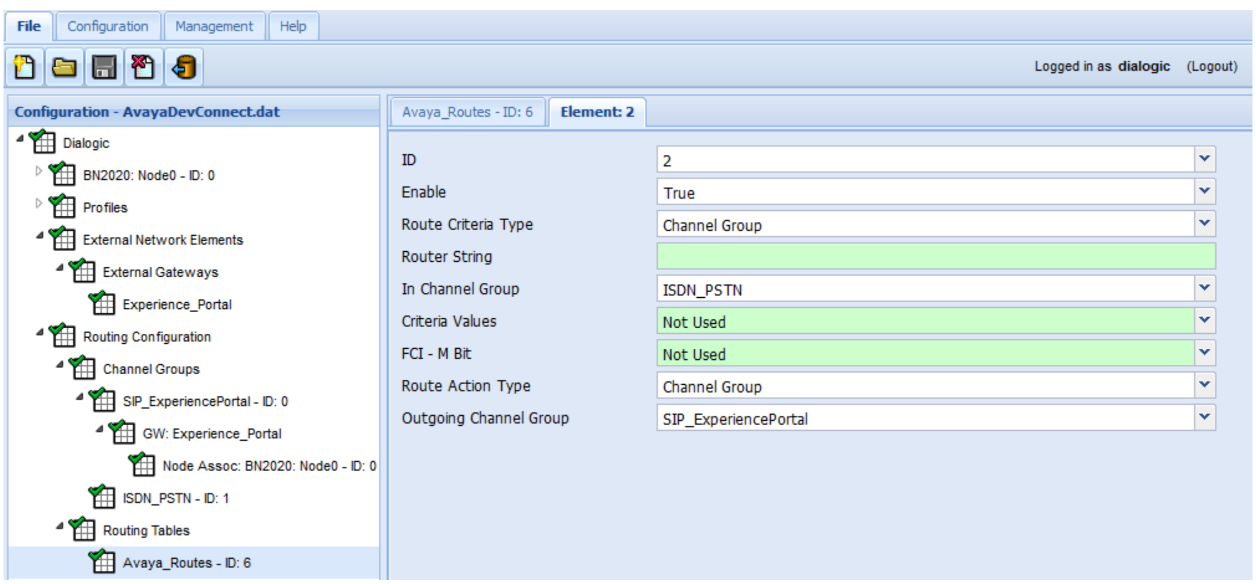


From the configuration tree in the left pane, right click **Dialogic → Routing Configuration → Routing Tables → Avaya_Routes** and select **New Element**.

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.

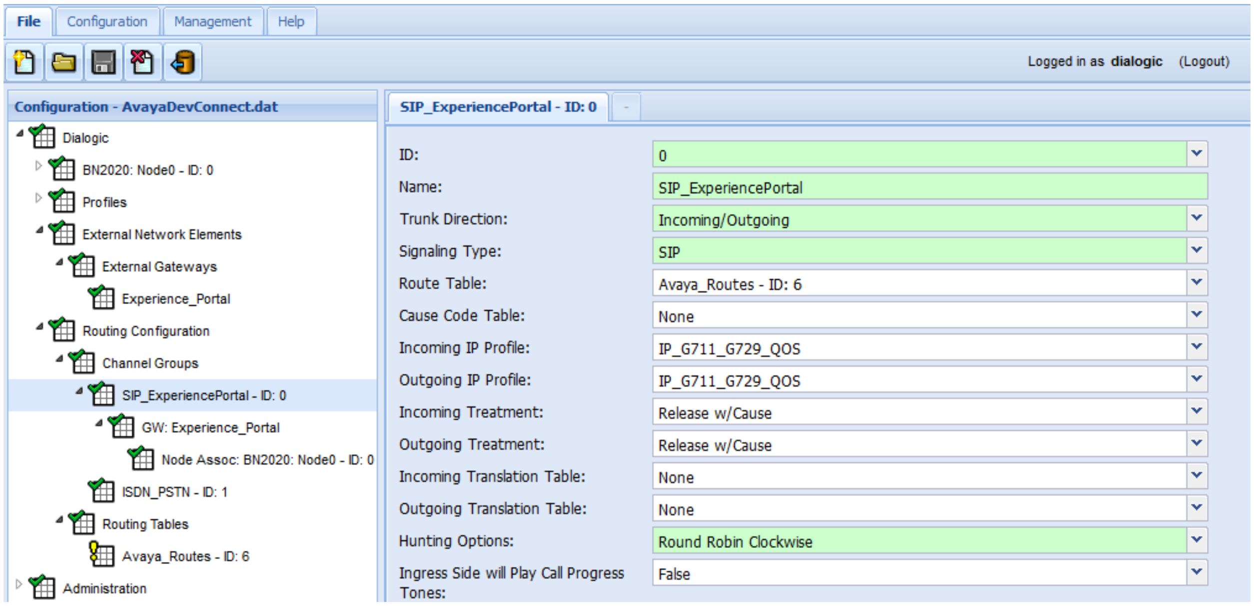


Repeat the above procedure for a second Route Element which routes calls from the ISDN PRI channel group to the **SIP_ExperiencePortal** channel group.



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**.



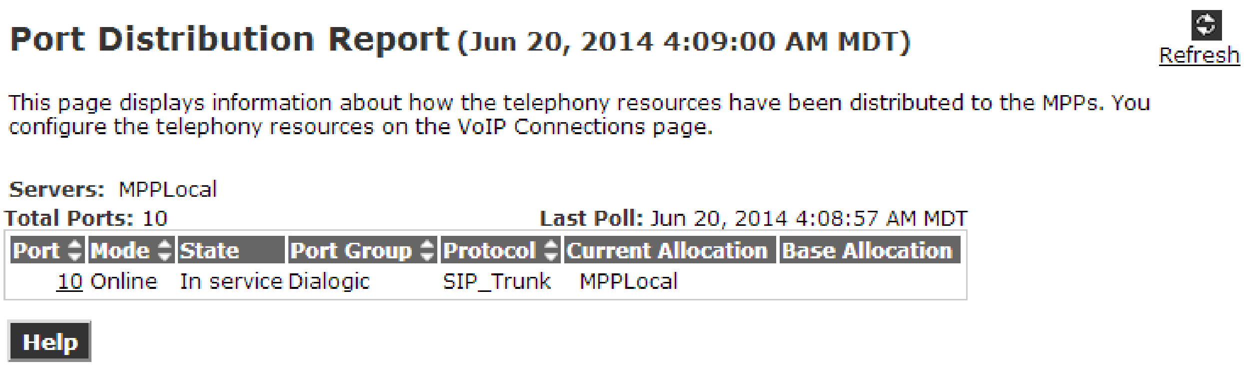
Repeat the above procedure for the channel group pre-configured for the ISDN PRI interface.

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.



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.

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.