



Avaya Solution & Interoperability Test Lab

Application Notes for Bell Canada SIP Trunking Service with Avaya Business Communication Manager 450 Release 6.0 – Issue 1.0

Abstract

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Bell Canada SIP Trunking Service and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Business Communication Manager 450 Release 6.0 and various Avaya endpoints.

Bell Canada SIP Trunking Service provides PSTN access via a SIP trunk between the enterprise and Bell Canada network as an alternative to traditional PSTN trunks such as analog and/ or ISDN-PRI. This approach generally results in lower cost for the enterprise.

Bell Canada is a member of the Avaya DevConnect Service Provider Program. 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.

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1. Introduction

These Application Notes describe the steps to configure Session Initiation Protocol (SIP) Trunking between Bell Canada SIP Trunking Service (Bell Canada) and an Avaya SIP-enabled enterprise solution. The Avaya solution consists of Avaya Business Communication Manager 450 (BCM450) Release 6.0 and various Avaya endpoints.

Bell Canada SIP Trunking Service referenced within these Application Notes is designed for enterprise business customers. Customers using Bell Canada SIP Trunking Service with the Avaya SIP-enabled enterprise solution are able to place and receive PSTN calls via a broadband WAN connection using SIP protocol. This converged network solution is an alternative to traditional PSTN trunks such as analog and/ or ISDN-PRI.

Bell Canada applies Digest Authentication for outgoing calls from the enterprise. It uses challenge-response authentication with a “401 Unauthorized” responding to each outgoing initial INVITE to Bell Canada. The subsequent INVITE from the enterprise provides the “Authorization” header with a configured user name and password. This credential is provided by Bell Canada and configured on BCM450. This call authentication scheme as specified in RFC 3261 provides authentication for the SIP signaling.

2. General Test Approach and Test Results

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.

Bell Canada is a member of the Avaya DevConnect Service Provider Program. The general test approach is to connect a simulated enterprise to the Vendor Validation Circuit through the Internet and exercise the features and functionalities listed in **Section** Error! Reference source not found..

2.1. Interoperability Compliance Testing

To verify Bell Canada SIP Trunking interoperability, the following features and functionalities were covered during the compliance testing:

- Incoming PSTN calls to various phone types including UNISTim, digital, and analog telephones at the enterprise. All incoming calls from PSTN are routed to the enterprise across the SIP Trunk from the service provider.
- Outgoing PSTN calls from various phone types including UNISTim, digital, and analog telephones at the enterprise. All outgoing calls to PSTN are routed from the enterprise across the SIP Trunk to the service provider.
- Incoming and outgoing PSTN calls to/ from PC 2050 softphones.
- Dialing plans including local, long distance, international, outgoing toll-free, operator assisted calls, local directory assistance (411) calls... etc.

- Calling Party Name presentation and Calling Party Name restriction.
- Proper codec negotiation with G.711MU codec.
- Proper media transmission using G.711MU codec.
- Proper early media transmission using G.711MU codec.
- Incoming and outgoing fax over IP using G.711MU codec.
- DTMF tone transmission as out-of-band RTP events as per RFC 2833.
- Voicemail navigation for incoming and outgoing calls.
- CallPilot voicemail hosted on BCM450.
- User features such as hold and resume, call waiting, call park, transfer, forward, and conference.
- Off-net call transfer using subsequent INVITE method.
- Off-net call forward using Diversion method.
- Find Me Follow Me (FMFM) twining to incoming calls to cellular.
- Meet-me conferencing hosted on BCM450.
- Response to OPTIONS heartbeat.
- Response to incomplete call attempts and trunk errors.
- Session Timers implementation.

Items are not supported by Bell Canada on the test environment or not tested as part of the compliance testing, are listed as following:

- Inbound toll-free and outgoing emergency calls (E911) are supported but were not tested as part of the compliance testing because Bell Canada has not provided the necessary configuration.
- G.729 codec is not supported.
- Fax over IP with T.38 codec is not supported.
- Off-net calls transfer using REFER method is not supported.
- Off-net call forward using History-Info method is not supported.

2.2. Test Results

Interoperability testing of Bell Canada with the Avaya SIP-enabled enterprise solution was successfully completed with the exception of the observations/ limitations described below.

1. **The BCM450 station resumes a held outgoing PSTN call causes the Calling Party Number to change.** After retrieving the held call, the BCM450 station displayed the “VoIP Line00X” instead of the original Calling Party Number of PSTN party. This is a known behavior of BCM450 with no available resolution at this time. This issue has low user impact, it is listed here simply as an observation.
2. **The BCM450 station which forwards an internal call to PSTN, displays the improper Calling Party Number.** Before the forwarded call being answered by PSTN party, the BCM450 station displayed “Priv. OLI” > “PSTN Calling Party Number”. After PSTN party answered the call, the display on the BCM450 station changed to “VoIP Line00X”. It should display the Calling Party Number of PSTN party. This is a known

behavior of BCM450 with no available resolution at this time. This issue has low user impact, it is listed here simply as an observation.

3. **The BCM450 station which off-net conferences an incoming call from PSTN_1 back to PSTN_2, displays the improper Calling Party Number.** Before the PSTN_2 answered the call, the BCM450 station displayed “Target LineXXX” > “PSTN_2 Calling Party Number”. After PSTN_2 party answered the call, the display on the BCM450 station changed to “Target LineXXX” - “VoIP Line00X”. It should display the Calling Party Number of PSTN parties. This is a known behavior of BCM450 with no available resolution at this time. This issue has low user impact, it is listed here simply as an observation.
4. **The BCM450 station off-net transfers an incoming/ outgoing from/ to PSTN_1 call back to PSTN_2, the Calling Party Number is not updated.** Even the call was successfully transferred, but BCM450 did not send subsequent INVITE or UPDATE to Bell Canada on both call legs with the “Contact” or “P-Asserted-Identity” headers containing the Calling Party Number of the true connected PSTN parties. Both PSTN parties still displayed the “Pub. OLI” of the BCM450 station. This is a known behavior of BCM450 with no available resolution at this time. This issue has low user impact, it is listed here simply as an observation.
5. **The 3-way conference call which was initiated by the BCM450 station drops if all local stations leave the conference.** This is a known behavior of BCM450 with no available resolution at this time. This issue has low user impact, it is listed here simply as an observation.
6. **The Find Me Follow Me (FMFM) users display the “Pub. OLI” of the Virtual Application DN instead of displaying the original Calling Party Number of PSTN party.** Bell Canada examined the Calling Party Number of outgoing twinned calls so the “Pub. OLI” of the Virtual Application DN has to be set to one of the subscribed DID numbers. Otherwise, outgoing calls to FMFM users will be rejected. This workaround results the original Calling Party Number of PSTN party will not be displayed on the FMFM users. The FMFM users will display the “Pub. OLI” of the Virtual Application DN.
7. **The off-net call forward unconditional or busy scenarios fail if the forwarded PSTN party takes longer than 8 seconds to respond.** The issue has been seen on the cellular PSTN phone. BCM450 relies on the respond time of the cellular PSTN phone on the 2nd leg to transmit the ringback back to the original calling PSTN party on the 1st leg. If the respond time is greater than 8 seconds, Bell Canada terminates the call with a CANCEL request on the 1st call leg. This fails the scenarios. The issue does not happen if the cellular PSTN phone in good wireless coverage that can significantly reduce the respond time under 8 seconds limit. The same call scenarios are successful on the regular PSTN phone (instead of the cellular PSTN party). Bell Canada is recommended to increase the Session Provisioning Timer to correct the issue. This is acknowledged as a known

behavior of Bell Canada SIP Trunking Service with no resolution available at this time, it is listed here as a limitation.

2.3. Support

For technical support on the Avaya products described in these Application Notes visit <http://support.avaya.com>.

For technical support on Bell Canada SIP Trunking Service, contact Bell Canada at http://www.bell.ca/enterprise/EntPrd_SIP_Trunking.page.

3. Reference Configuration

Figure 1 illustrates a sample Avaya SIP-enabled enterprise solution connected to Bell Canada Vendor Validation Circuit through the Internet.

For confidentiality and privacy purposes, the actual public IP addresses and PSTN routable phone numbers used in the certification testing have been replaced with fictitious parameters throughout the Application Notes.

For the compliance testing, Bell Canada provided the service provider public SIP domain as **sipxxxxxxxx.bell.ca** and the enterprise public SIP domain as **cust2xxxx.xxxx.bell.ca**. These public SIP domains will be used for the public SIP traffic between the enterprise and Bell Canada across the public networks with the transport protocol UDP.

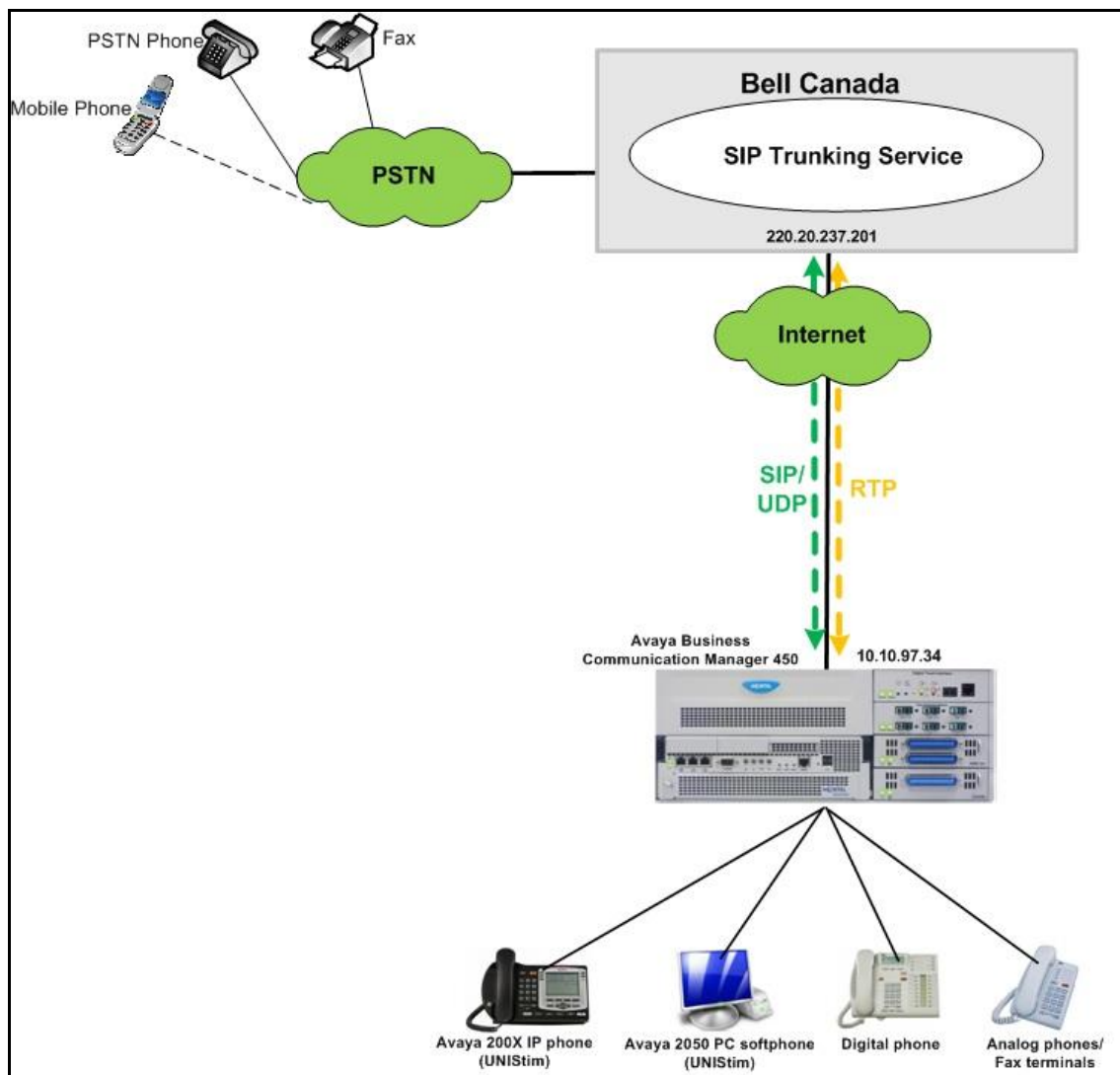


Figure 1: Avaya IP Telephony Network Connecting to Bell Canada SIP Trunking Service.

4. Equipment and Software Validated

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

Avaya IP Telephony Solution Components	
Component	Release
Avaya Business Communication Manager 450	Release 6.0 with following patches. <ul style="list-style-type: none">• BCM450.R600.CORE-TELEPHONY-12• BCM450.R600.SU.System-003.201101• BCM450.R600.SU-2• BCM450.R100.SU.System-012.201003• BCM450.R100.SU.Desktop-006.201006• BCM450.R100.DSP-FIRMWARE-75
Avaya phones	<ul style="list-style-type: none">• 2004 p2: 0604DCN (UNISim)• 1140: 0625C7M (UNISim)• 1120: 0624C7M (UNISim)• 2007: 0621C7G (UNISim)• 1220: 062AC7M (UNISim)• i2050 PC softphone (Release 3.2)• Avaya digital phones• Avaya analog phones
Bell Canada SIP Trunking Service Solution Components	
Component	Release
Bell Canada SIP Trunking Service	Version 1.3

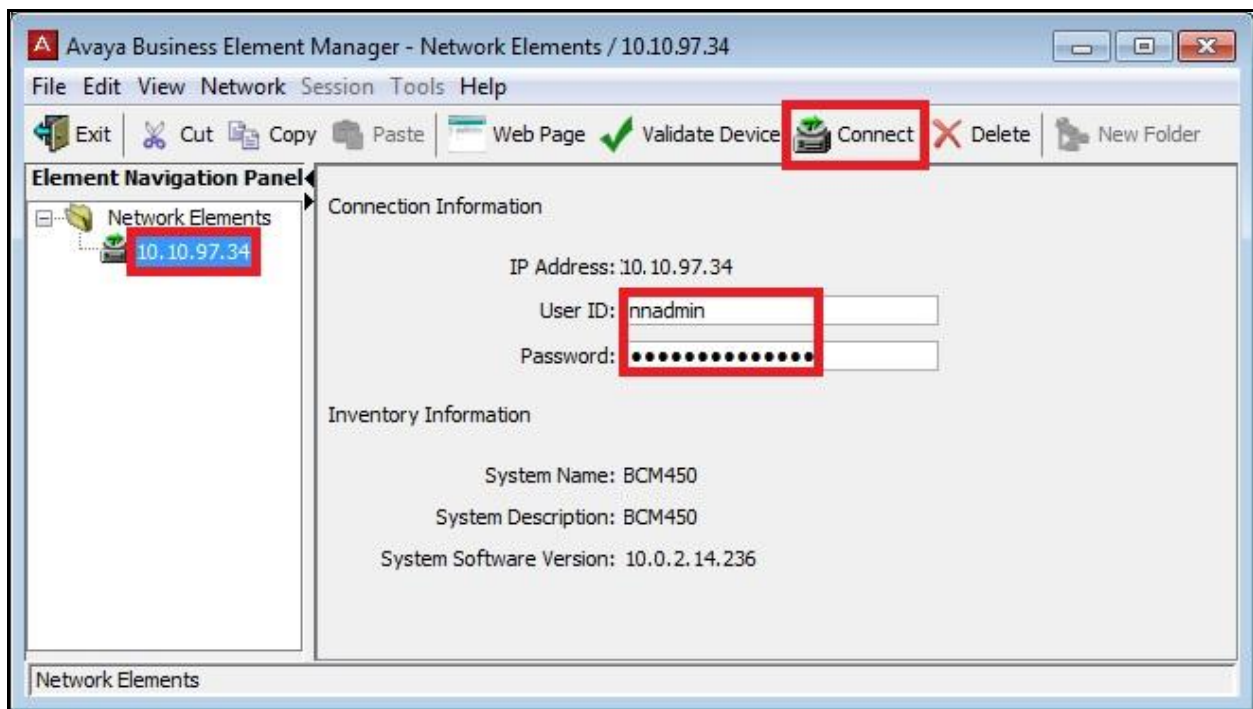
5. Avaya Business Communication Manager 450 Configuration

This section describes the procedure for configuring BCM450 to inter-operating with Bell Canada SIP Trunking Service.

These Application Notes assume that the general installation and configuration of BCM450 have previously been administered. For further information, see the **References** in **Section 8**.

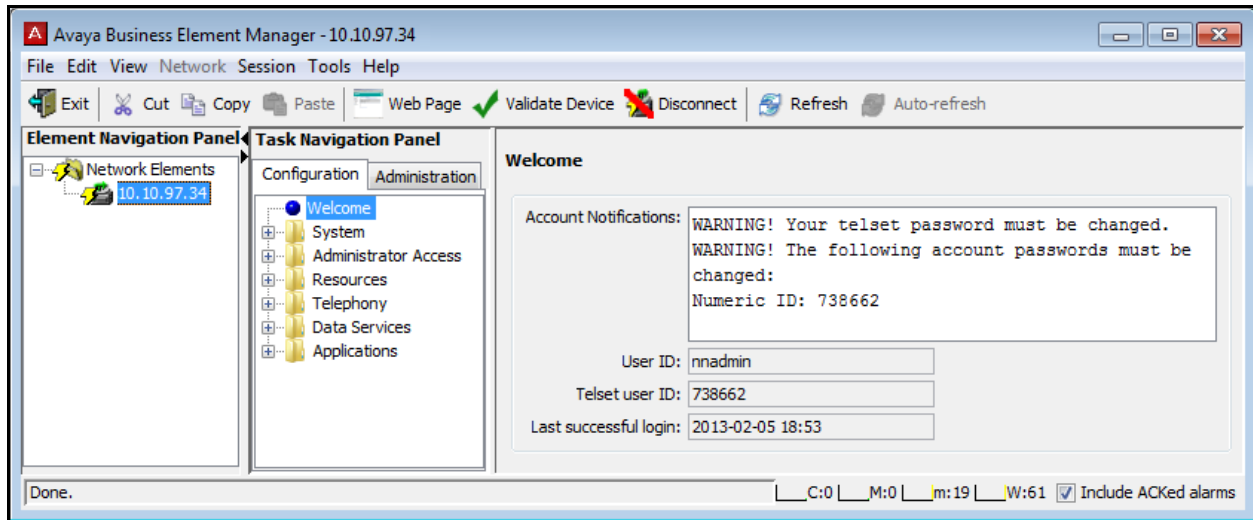
5.1. Login to the Avaya Business Element Manager

BCM450 configuration is accomplished by using the Avaya Business Element Manager installed on a PC. To log on to the Avaya Business Element Manager, select **Network Elements**. Enter the credential for **User ID** *nnadmin* then click **Connect** as shown in the following screenshot.



After the connection has been established, click **OK** in the **Confirm** dialog (not shown).

The screenshot below shows the Avaya Business Element Manager has been successfully logged on.

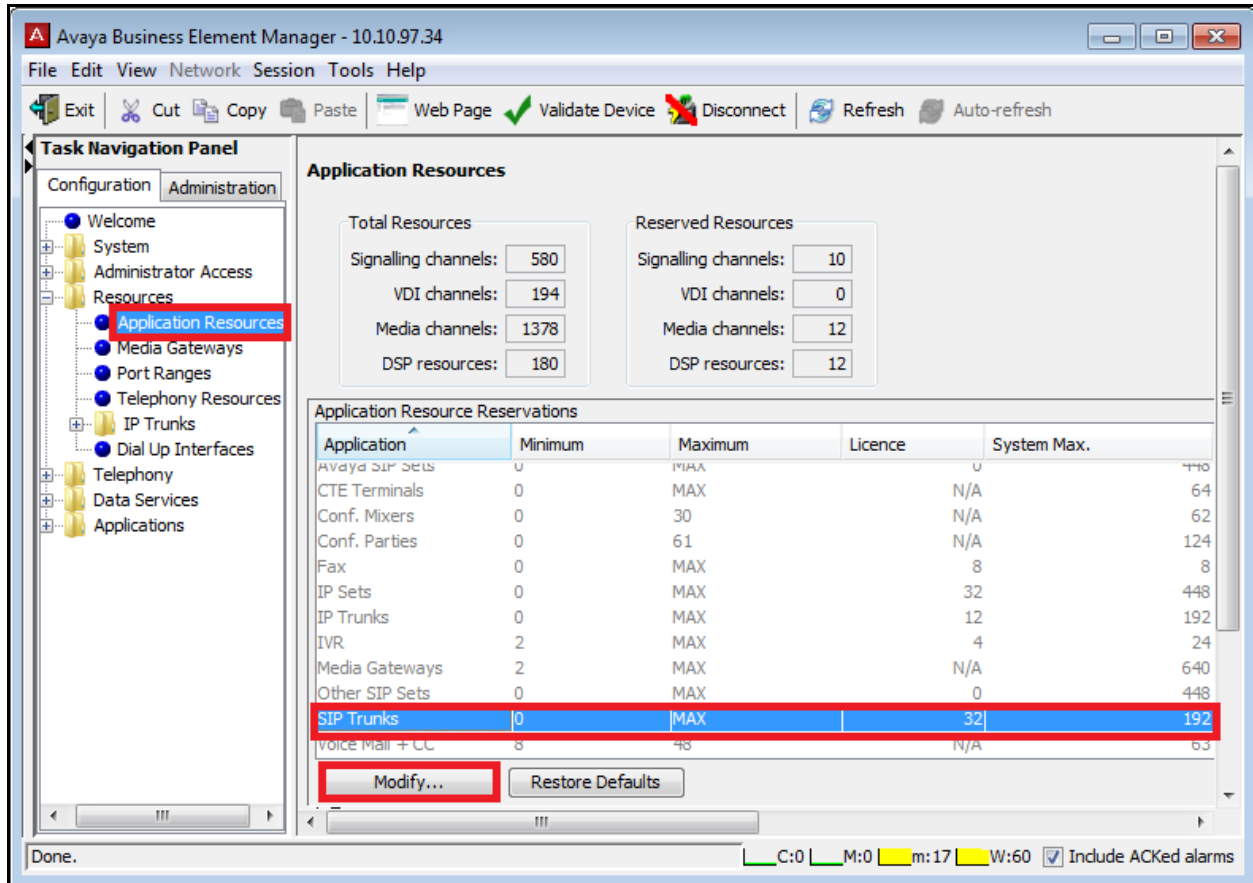


5.2. Administer Resources

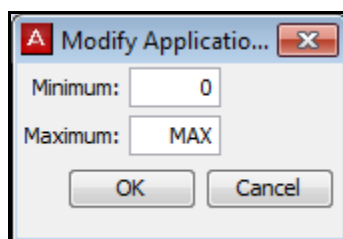
This section describes how to configure Application Resource, Media Gateway and Port Ranges that will be used to administer the SIP Trunk.

5.2.1. Administer Application Resources

To display the Application Resources, select **Configuration** → **Resources** → **Application Resources**. Verify the Application Resources for the SIP Trunks by selecting **SIP Trunks** then click **Modify** button as shown in the following screenshot.

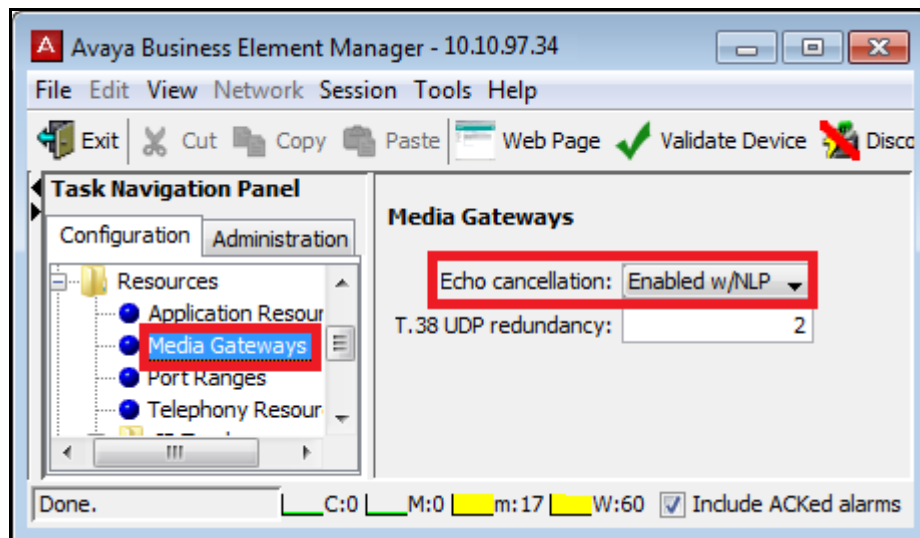


A new dialog appears with title **Modify Resource Application Reservations**, verify the maximum Application Resources for the SIP Trunks were allocated. The available Application Resources are based on the system capacity and the system license.



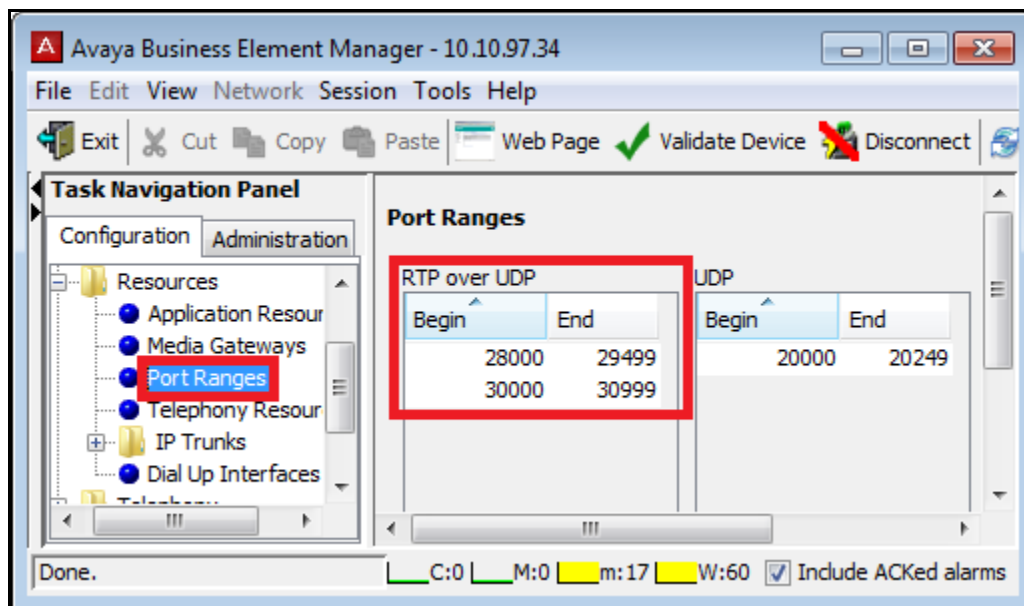
5.2.2. Administer Media Gateways

Enable **Echo cancellation** on the Media Gateways by selecting **Configuration → Resources → Media Gateways** then select **Enabled w/NLP** as shown in the following screenshot.



5.2.3. Administer Port Ranges

To display the Port Ranges, select **Configuration → Resources → Port Ranges**. The default Port Ranges for the RTP traffic are shown in the following screenshot.



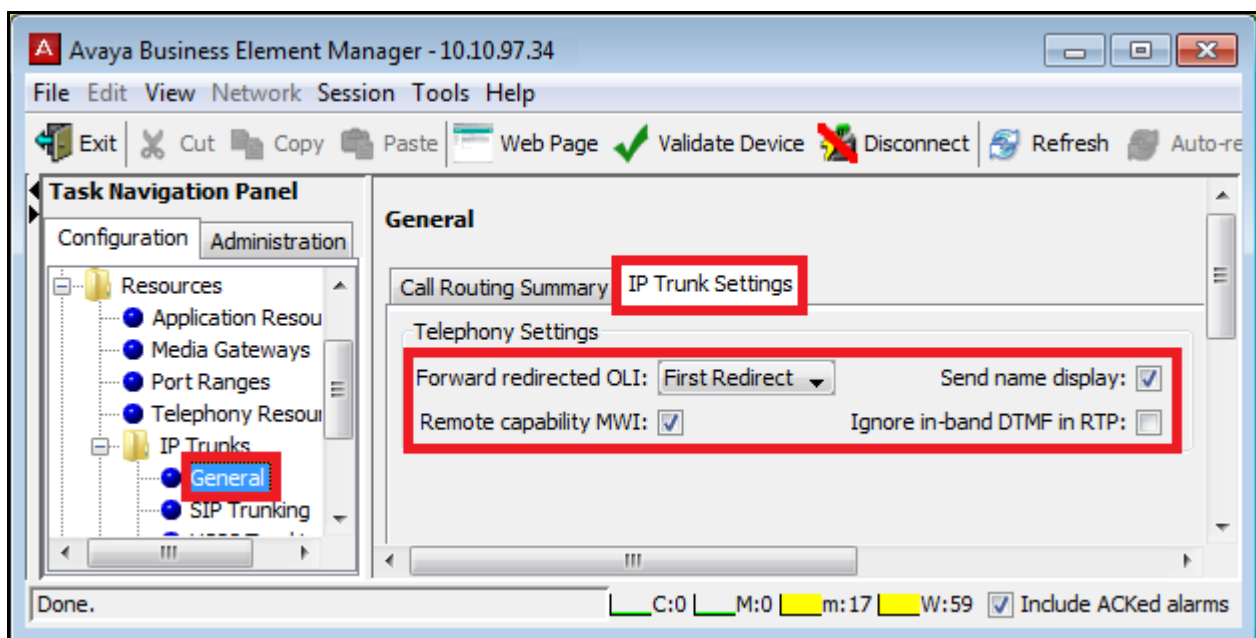
5.3. Administer the SIP Trunk

This section describes the steps for configuring the SIP Trunk on BCM450 connecting to Bell Canada.

5.3.1. General IP Trunk settings

The system settings for IP Trunk can be configured by selecting **Configuration → Resources → IP Trunks → General**. Under **IP Trunking Settings** tab, configure the **Telephony Settings** as shown in the screenshot below.

- **Forward redirected OLI:** First Redirect. In case of multiple call forwards, this setting allows the Calling Party Name and Calling Party Number of the call party that firstly redirects the calls.
- **Send name display:** Checked. This setting enables the Calling Party Name and Calling Party Number on the SIP Trunk.
- **Remote capacity MWI:** Checked.
- **Ignore in-band DTMF in RTP:** Unchecked. In the compliance testing, Bell Canada supported the out-of-band DTMF. This setting, however, also allow in-band DTMF on the SIP Trunk.



5.3.2. Administer Global

The global settings can be configured by selecting **Configuration → Resources → IP Trunks → SIP Trunking**. Under **SIP Trunking** tab, configure the **Global Settings** as shown in the screenshot below.

SIP Settings:

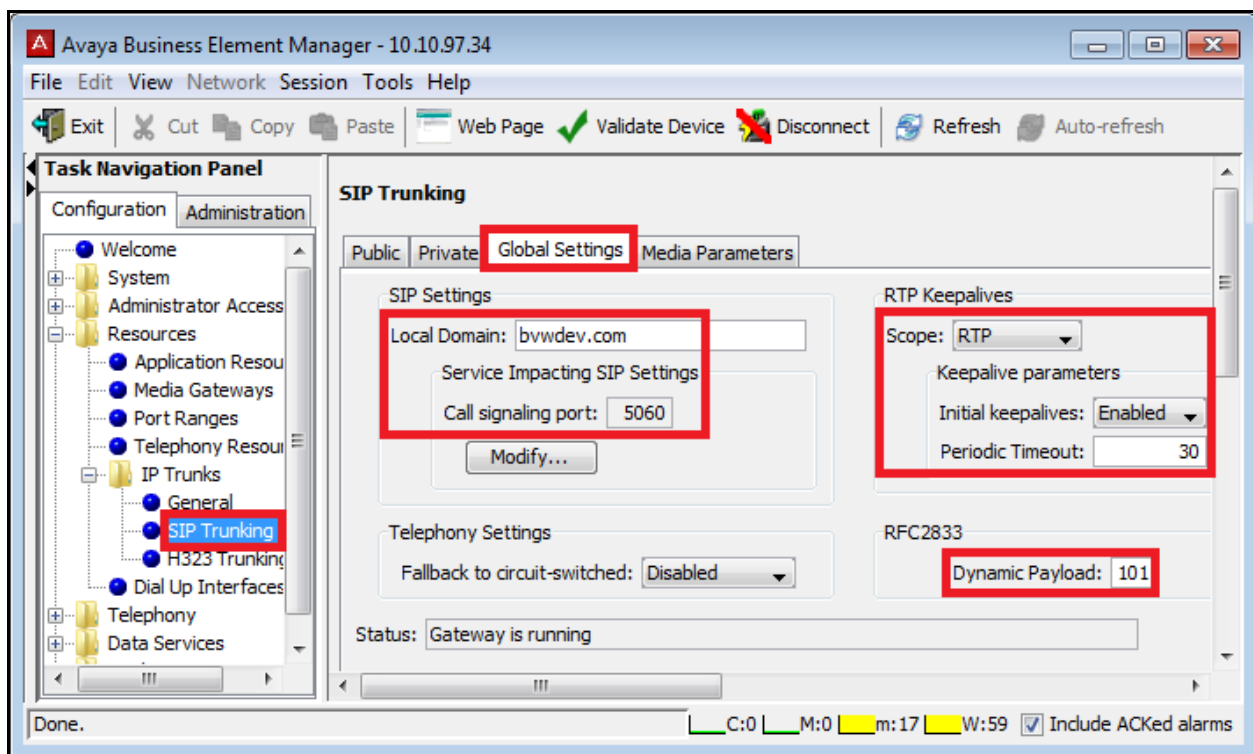
- **Local Domain:** Enter the private SIP domain for the enterprise .e.g. **bvwdev.com**.
- **Call signaling port:** Enter the port for SIP signaling .e.g. **5060**.

RTP Keepalives:

- **Scope:** RTP. This allows BCM450 to send RTP to keep the session active to prevent the termination by Bell Canada if the line is silent. In the compliance testing, Bell Canada detects the silence on the line, if it exceeds the allow duration then the active calls will be disconnected.
- **Initial keepalives:** Enabled. This allows BCM450 to send the initial RTP packet for keepalive purpose.
- **Periodic Timeout:** 30. The duration of sending out RTP keep alive was set to 30 seconds.

RFC2833:

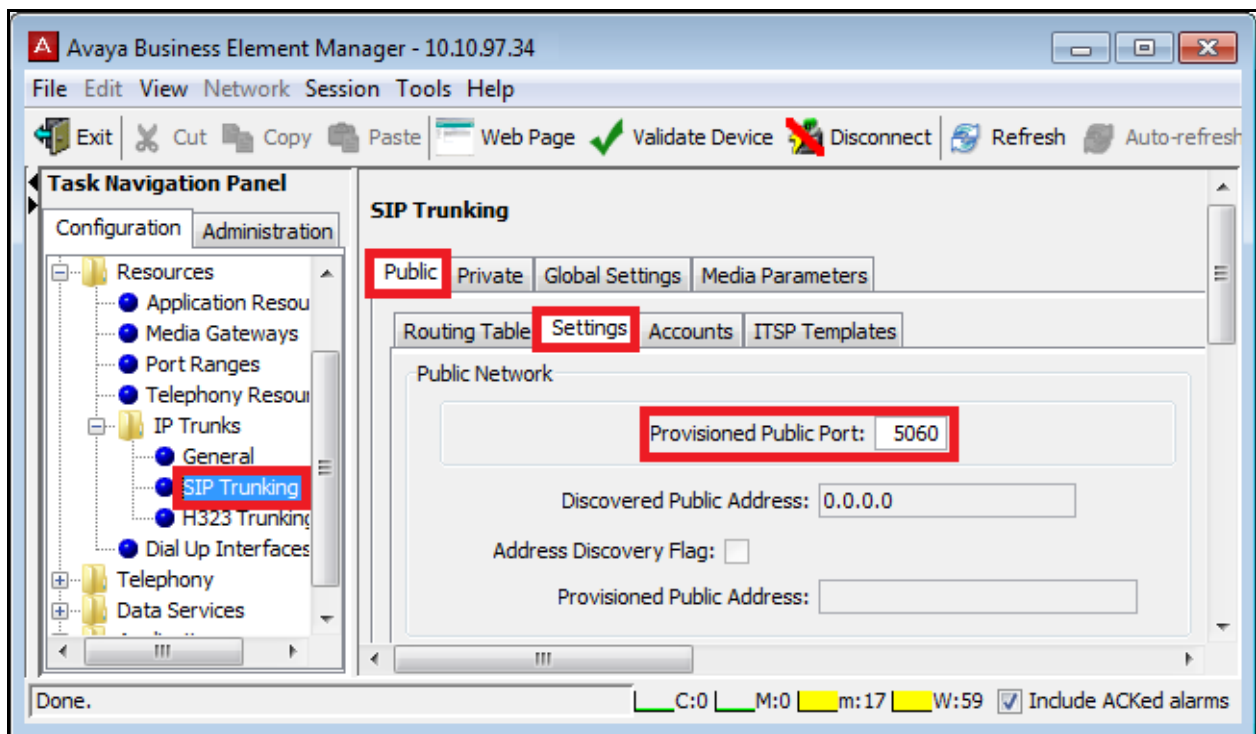
- **Dynamic payload:** 101. This payload value is preferred by Bell Canada.



5.3.3. Administer Public Port

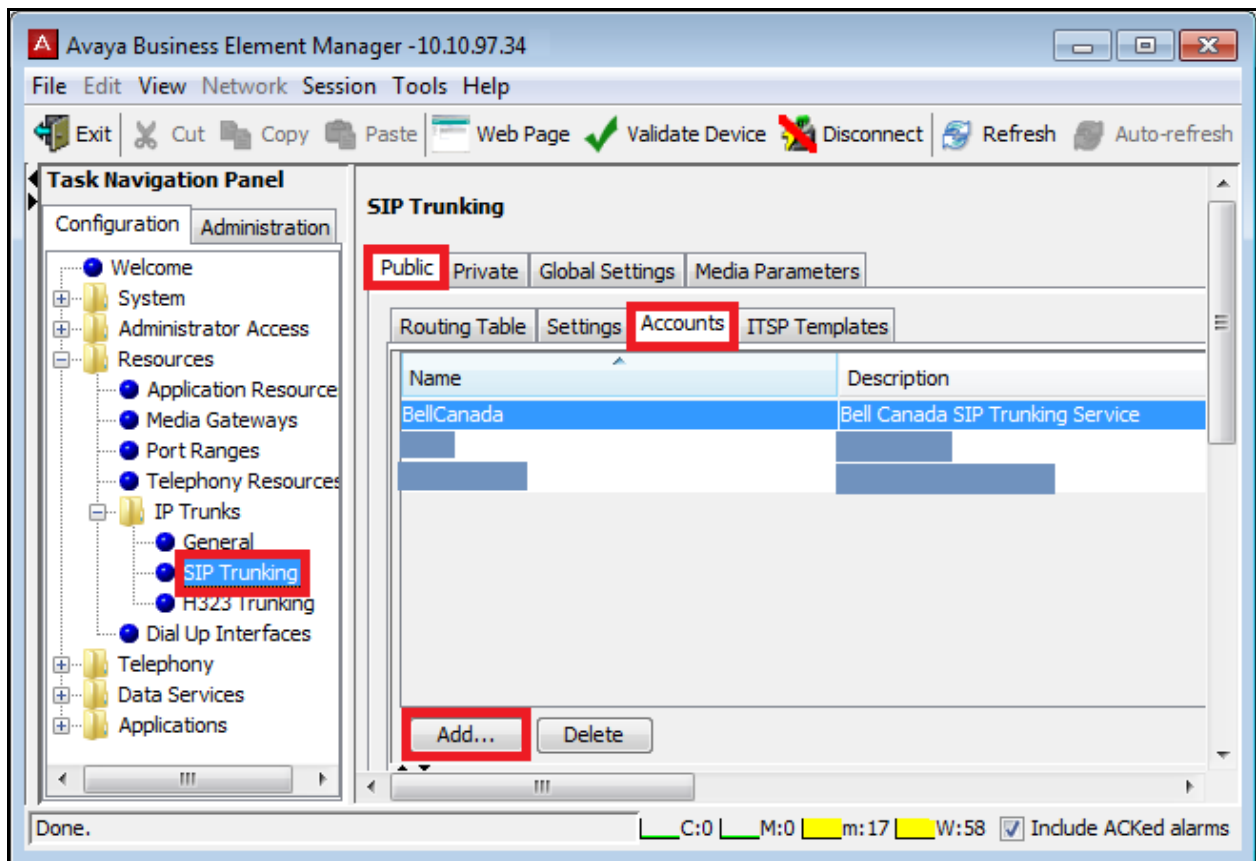
The Public Port is used to receive the SIP traffic from the service provider. In the compliance testing, it was set to **5060** as the default value that was preferred by Bell Canada.

To configure the Public Port, select **Configuration → Resources → IP Trunks → SIP Trunking**. Under **Public** tab, select **Settings** then configure the **Provisioned Public Port** to **5060** as shown in the screenshot below.



5.3.4. Create a Public Account

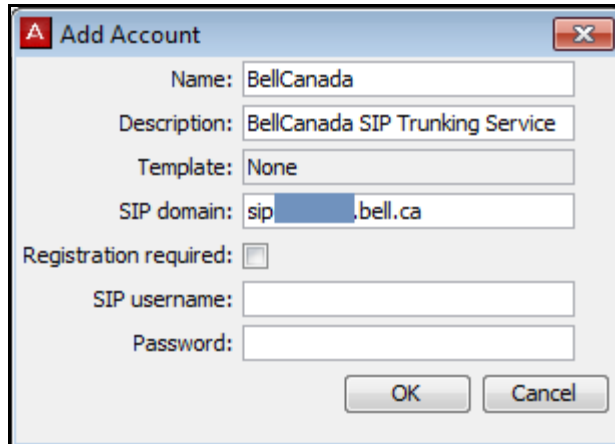
To create a Public Account, click **Accounts** tab then click **Add** button to add a public account for Bell Canada as shown in the screenshot below.



In the **Add Account** dialog, click **No Template** and proceed to the next steps (not shown).

The SIP Trunk to Bell Canada was configured as the static IP endpoints with the following parameters.

- **Name:** Enter the account name .e.g. **BellCanada**.
- **Description:** Enter the account description e.g. **BellCanada SIP Trunking Service**.
- **Template:** None.
- **SIP domain:** Enter the service provider public SIP domain name .e.g. **sipxxxxxxx.bell.ca**.
- **Registration required, SIP username and Password:** Leave these fields as blank. Bell Canada does not require the Registration on the SIP Trunk.



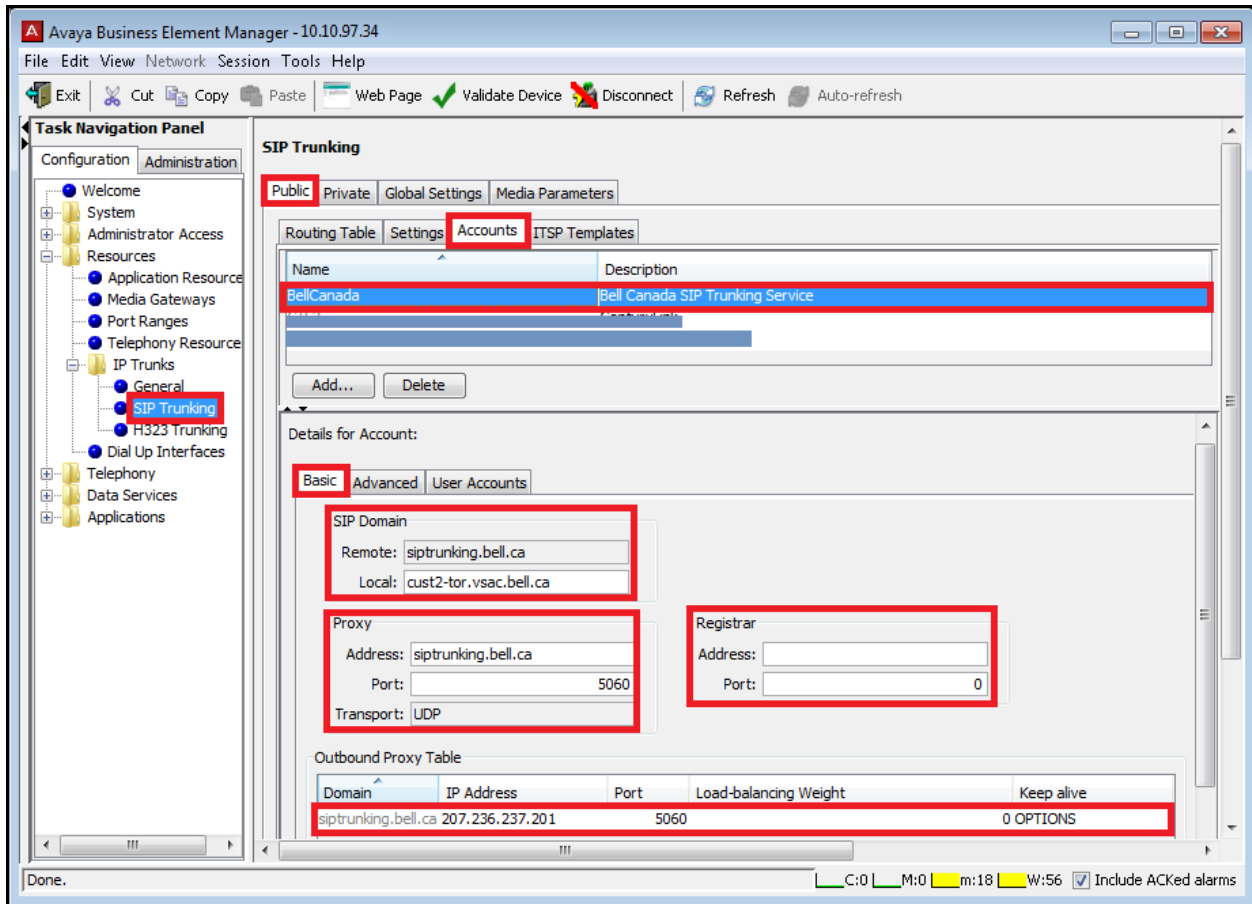
5.3.5. Basic settings

The Basic settings define parameters for the SIP Trunk including SIP domains, IP address of the far end proxies, Registration and OPTIONS keepalive.

To configure the Basic settings, select **Basic** tab of the account **BellCanada** created above. In the compliance testing, the Basic settings were configured with following values.

- **SIP Domain:**
 - **Remote:** The service provider public SIP domain e.g. **sipxxxxxxx.bell.ca** populated from the Public Account as shown in **Section 5.3.4**
 - **Local:** Enter the enterprise public SIP domain e.g. **cust2xxxx.xxxx.bell.ca**. This SIP domain is obtained through Bell Canada.
- **Proxy:**
 - **Address:** Enter the SIP domain that matches the entry defined next in **Outbound Proxy Table**. In the compliance testing, the SIP domain **sipxxxxxxx.bell.ca** was added with the Proxy Address is the IP address of Bell Canada Central Office e.g. **220.20.237.201**.
 - **Port:** Enter the port that Bell Canada is expecting to receive the SIP signaling from BCM450. In the compliance testing, Bell Canada preferred to port **5060**.
 - **Transport:** UDP. This is the default value on BCM450 and not configurable.
- **Registrar:** Leave these fields blank as default. Bell Canada does not require Registration on the SIP Trunk.
- **Outbound Proxy Table:**

- **Domain:** Enter the service provider public SIP domain e.g. **sipxxxxxxx.bell.ca**.
- **IP Address:** Enter the IP address of Bell Canada Central Office e.g. **220.20.237.201**.
- **Port:** Enter the port **5060** that Bell Canada is expecting to receive the SIP signaling from BCM450.
- **Load-balancing Weight:** Enter **0** to disable the Load-balancing on BCM450.
- **Keep Alive:** Select **OPTIONS** in the drop down list. This allows BCM450 to send OPTIONS heartbeat on the SIP Trunk for keep alive purpose.



The Advance settings define particular parameters for the SIP Trunk to meet the specification of the service provider.

- **Enable media relay:** Checked. This allows BCM450 to centralize the enterprise RTP traffic before sending to the service provider.
- **Support 100rel:** Checked. Bell Canada supports the “100rel” signaling as per the RFC 3262 on the SIP Trunk.
- **Allow UPDATE:** Checked. Bell Canada supports the UPDATE method on the SIP Trunk.
- **Use null IP to hold:** Checked. This allows BCM450 to send null IP to hold the call as per the RFC 2543.
- **Use user=phone:** Checked. This allows BCM450 to send URI parameter “user=phone” in the SIP signaling. Note: Without URI parameter “user=phone”, outgoing calls will be rejected by Bell Canada.
- **Enable Connected Identity:** Checked.
- **Standard SIP Caps Exchange:** Checked.
- **Session timer.** BCM450 was configured to refresh the Session Timer every 90 seconds.
 - **Session refresh method:** INVITE.
 - **Refresher:** Local.
 - **Session-Expires:** 1800
 - **Min-SE:** 90
- **Active call limit:** 0. This disables the active call limit on BCM450.
- **ITSP association method:** To header local domain match.

Basic

Advanced

User Accounts

Enable local NAT compensation:

☐

Enable media relay:

☒

Use maddr in R-URI:

☐

Use maddr in Contact:

☐

Support 100rel:

☒

Allow UPDATE:

☒

Use Null IP to hold:

☒

Use user=phone:

☒

Force E164 international dialing:

☐

Enable SDP OPTIONS query:

☐

Allow REFER:

☐

Support Replaces:

☐

Enable Connected Identity:

☒

Standard SIP Caps Exchange:

☒

NAT Pinhole Maintenance

Signaling method:

None

Signaling interval:

30

Session timer

Session refresh method:

INVITE

Refresher:

Local

Session-Expires:

1800

Min-SE:

90

Active call limit:

0

ITSP association method:

To header local domain match

Outbound Called characters to absorb:

0

Inbound Called prefix to prepend:

Authentication realm:

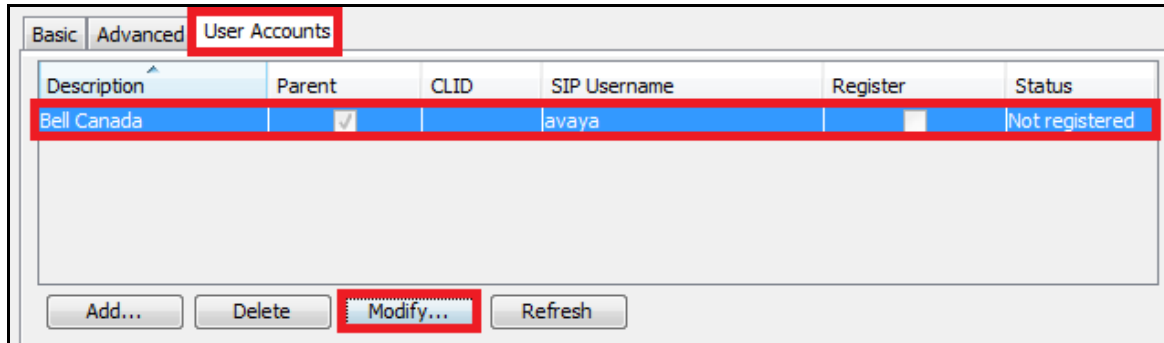
Note:

- The other options were left as blank as default.
- The ITSP association method is set to match the local SIP domain of the “To” header as defined as **cust2xxxx.xxxx.bell.ca** in **Section 5.3.5**.

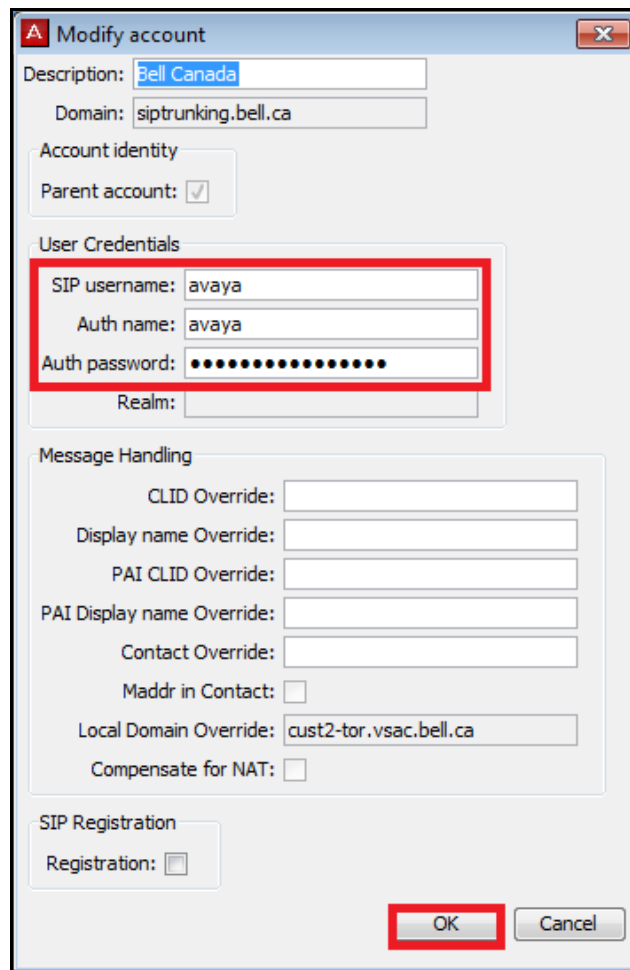
5.3.7. User Account settings

Under the User Account settings, the credential for Digest Authentication will be defined.

To change the User Account settings, select User Account **BellCanada** then click **Modify** button as shown in the screenshot below.



In the **Modify account** dialog, enter the **User Credentials** for Digest Authentication as shown in the screenshot below.



Notes:

- The credential was obtained through Bell Canada.
- The Message Handling fields and SIP Registration were kept blank as default. The certification testing does not require either the manipulation on the SIP message headers or the registration.

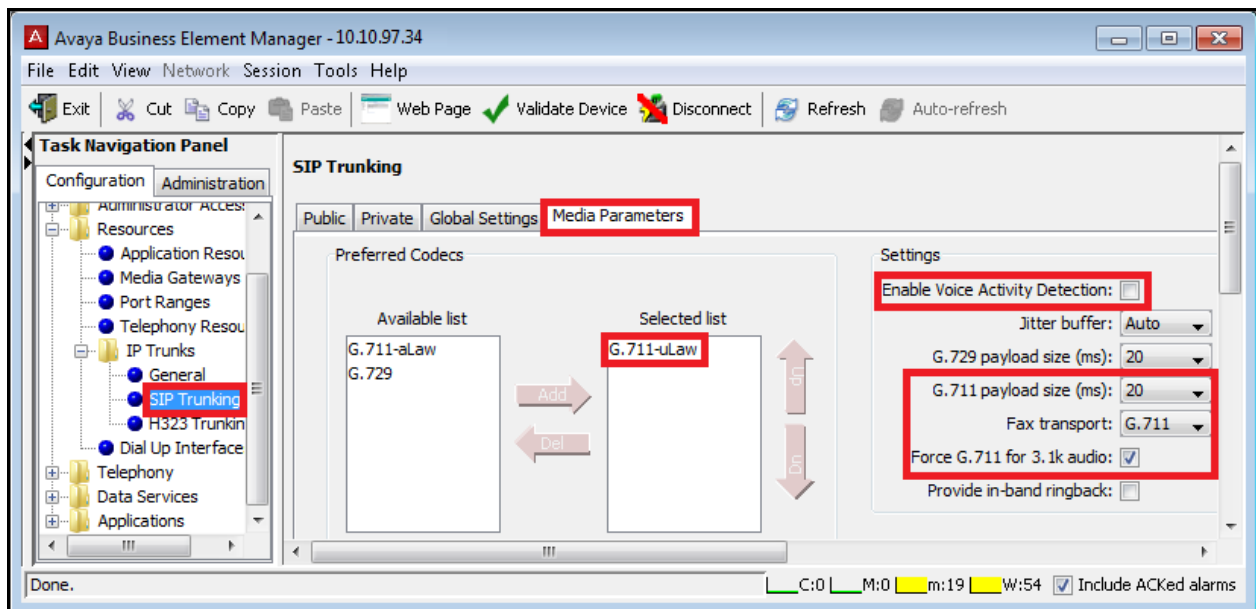
5.4. Administer Codec Profile

This section describes the steps for configuring the voice codec on BCM450 to interworking with Bell Canada.

5.4.1. Codec settings for SIP Trunk

To change the codec settings for SIP Trunk, select **Configuration → Resources → IP Trunks → SIP Trunk**. The **Media Parameters** tab is configured with the following parameters:

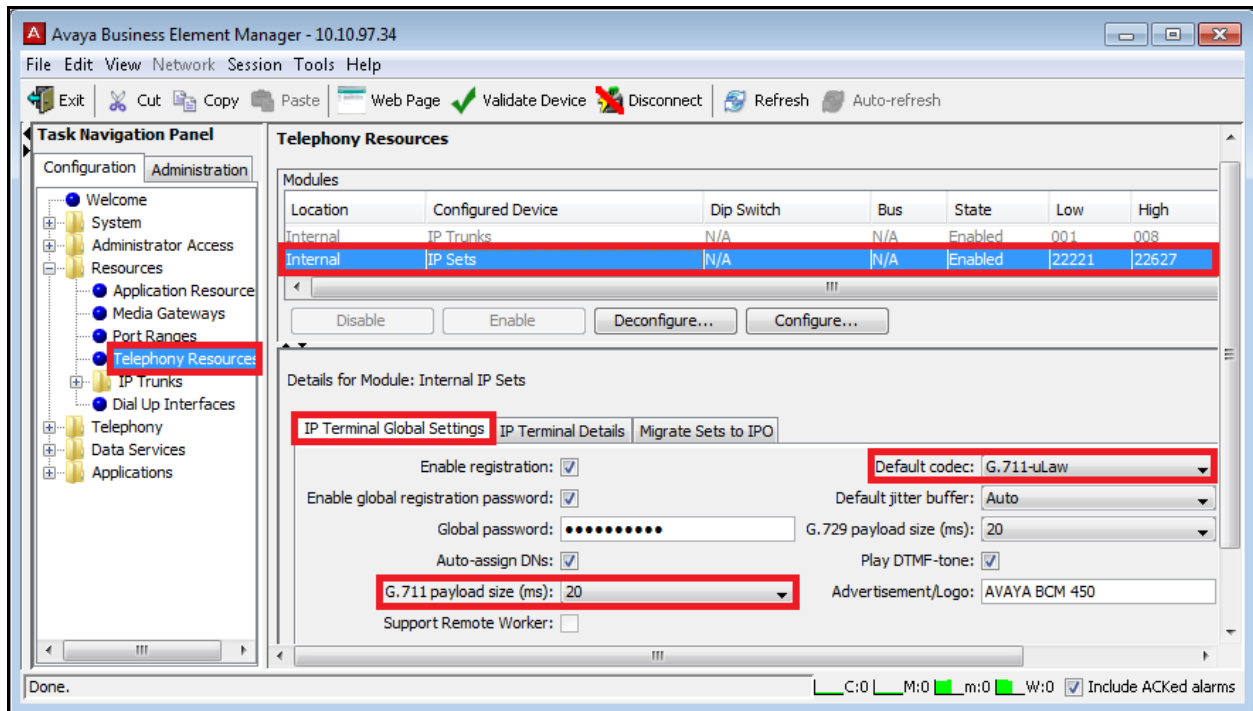
- **Selected list:** G.711-uLaw. The certification testing was conducted with only G.711 MU codec. Other voice codec e.g. G.729 or G.723 have not been tested.
- **G.711 payload size (ms):** 20.
- **Enable Voice Activity Detection:** Unchecked.
- **Fax transport:** G.711. The T.38 fax has not been tested.
- **Force G.711 for 3.1k audio:** Checked.



5.4.2. Codec settings for IP Sets

To define the default codec and payload size for IP Sets, select **Configuration → Resources → Telephony Resources**. Under **Internal/ IP Sets** module, the default codec and payload size were configured as G.711-uLaw and 20 ms as shown in the following screenshot.

- **Default codec:** G.711-uLaw.
- **G.711 payload size (ms):** 20.

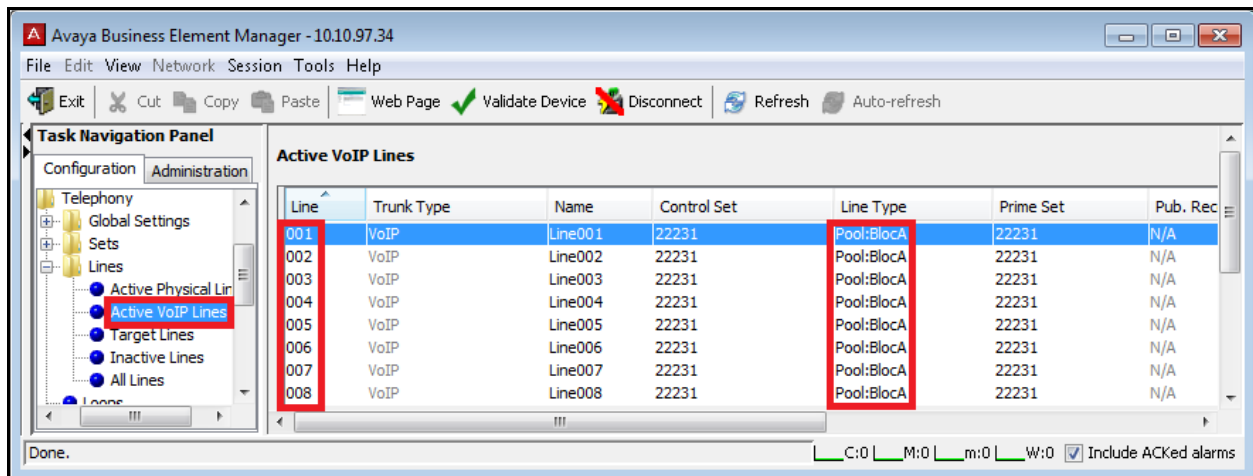


5.5. Administer Dialing Plan

This section shows how to configure BCM450 VoIP Lines for outgoing calls over the SIP Trunk to Bell Canada. For incoming calls, BCM450 uses Target Lines for call termination or redirection.

5.5.1. Associate Line Pool to VoIP Lines

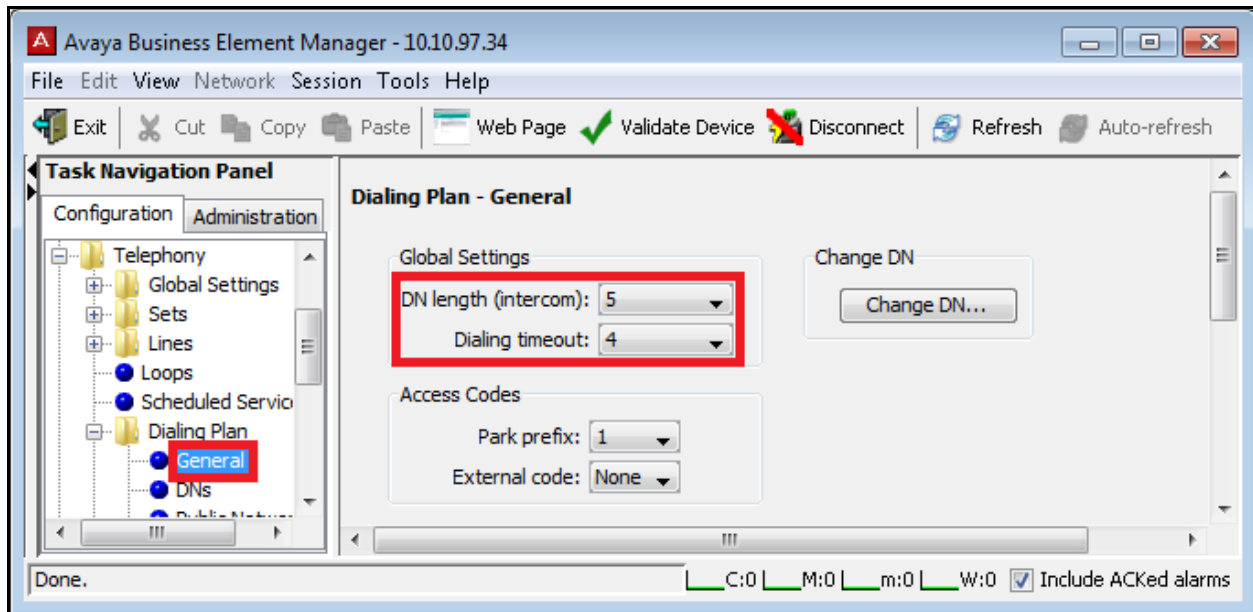
On BCM450, the VoIP Lines which are the resources of the SIP Trunk, are assigned to a Line Pool which can be used to for outgoing call routing. To associate a Line Pool to VoIP Lines, select **Configuration → Telephony → Lines → Active VoIP Lines**. Under the **Line Type** of each VoIP Line, select the pool name **Pool: BlocA** as shown in the screenshot below.



5.5.2. Administer DN Length

To configure DN length for BCM450 station, select **Configuration → Telephony → Dialing Plan → General**. In the compliance testing, **DN length** for BCM450 stations was set to 5 with **Dialing timeout** was set to 4 rings as shown in the screenshot below.

- **DN length (intercom): 5.**
- **Dialing timeout: 4.**

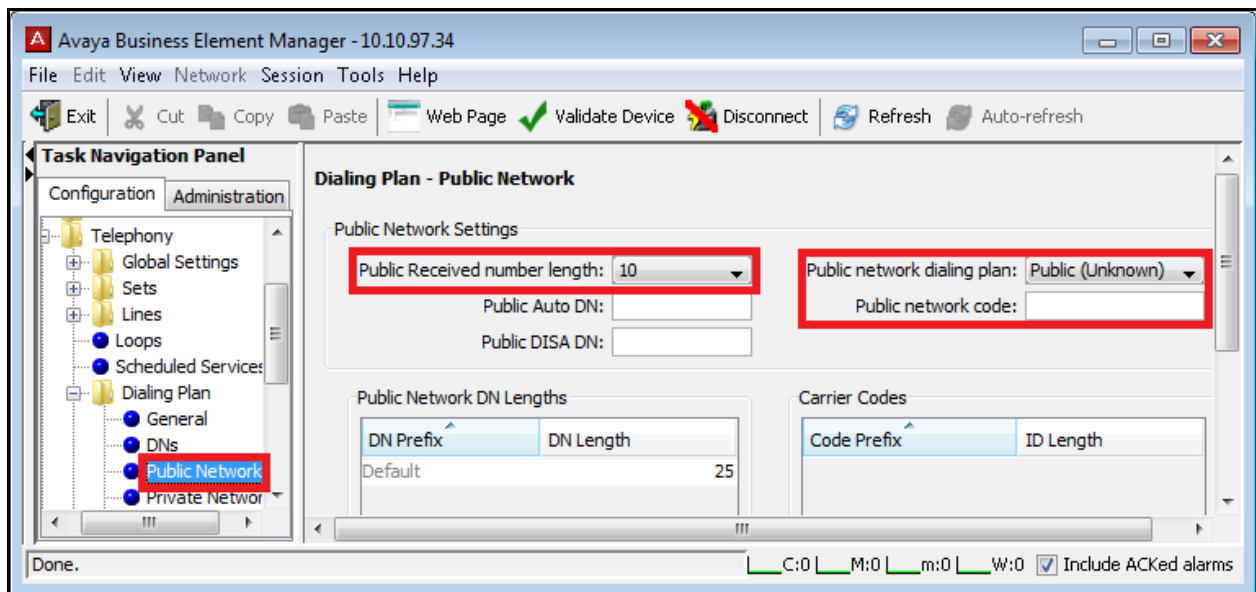


5.5.3. Administer Public Network

In the compliance testing, BCM450 was configured to receive 10-digit incoming calls and use **Public (Unknown)** as the call type for outgoing calls.

To configure the Public Network, select **Configuration → Telephony → Dialing Plan → Public Network**. Then configure the **Public Receive number length**, **Public network dialing plan** and **Public network code** as following.

- **Public Receive number length:** 10.
- **Public network dialing plan:** Public (Unknown).
- **Public network code:** Leave blank as default.



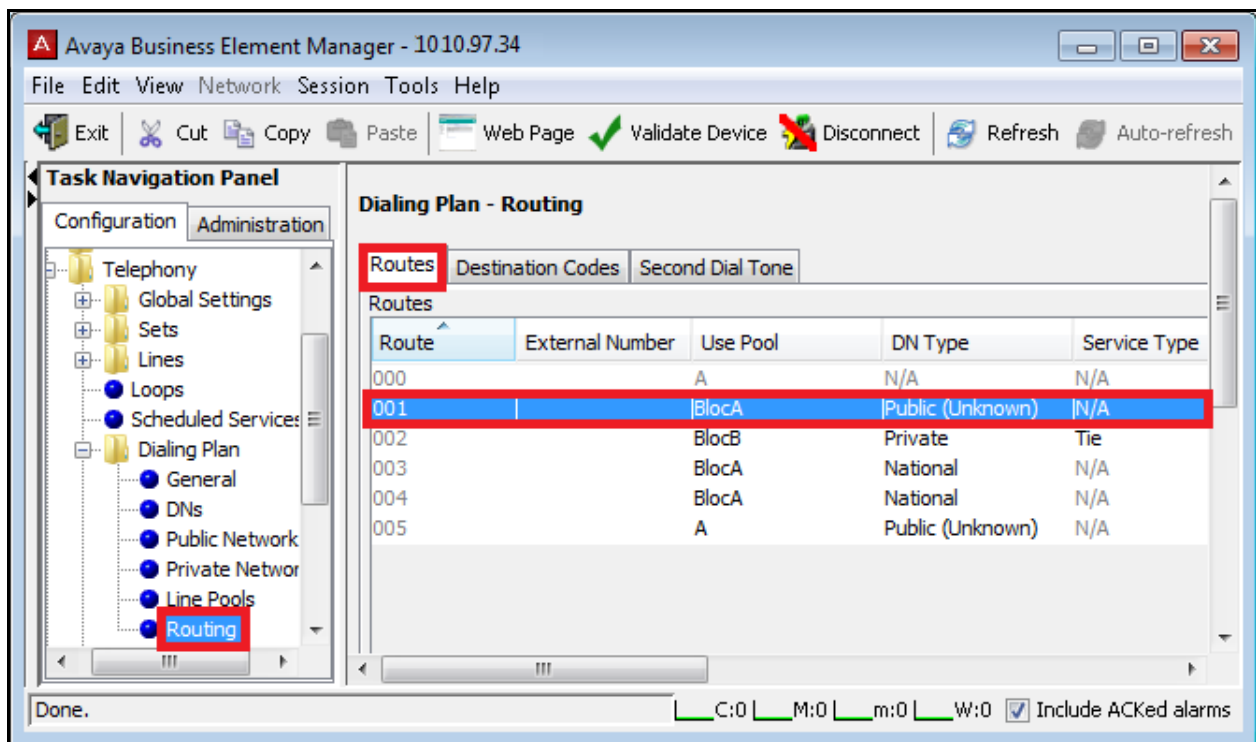
Note: The **Public network code** will affect the Calling Party Number of outgoing calls. If specified, it will be used in combination with the **Pub. OLI** setting for the phone to form the Calling Party Number. In this testing, the Calling Party Number is set on each phone individually, therefore it is unnecessary to define the **Public network code**.

5.5.4. Administer Outgoing Call - Routing

A route was created for outgoing calls to Bell Canada. To create a route, select **Configuration → Telephony → Dialing Plan → Routing**. Under the **Routes** tab, click **Add** to create a new route (not shown).

In the compliance testing, route **001** was added to use pool **BlocA** as show in the screenshot below:

- **Route 001:**
 - **Use Pool:** BlocA.
 - **DN Type:** Public (Unknown).



5.5.5. Administer Outgoing Call - Destination Codes

Destination Codes define the dialed prefixes for outgoing calls. In the compliance testing, a digit **9** was used as an internal access code for outgoing calls over the SIP trunk. This access code was configured to be deleted before sending out to Bell Canada.

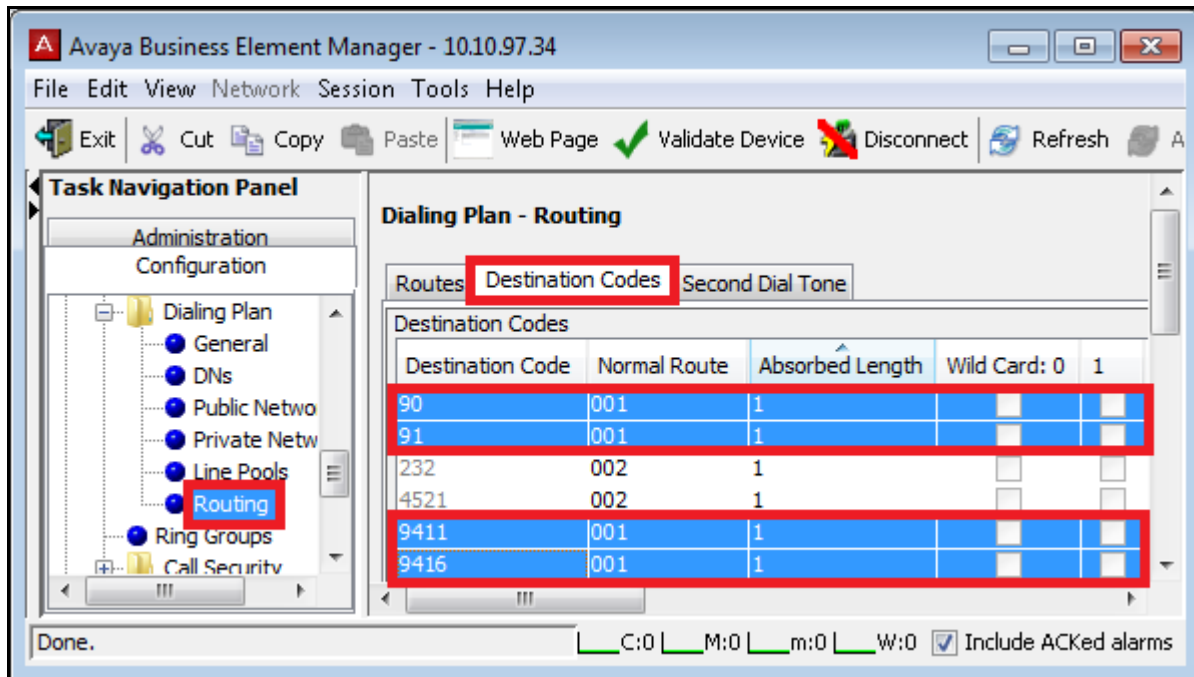
To create a Destination Code, select **Configuration → Telephony → Dialing Plan → Routing**. Under the **Destination Codes** tab, click **Add** to create a new route (not shown).

To include dialing plans as described in **Section 2.1**, multiple Destination Codes e.g. **90**, **91**, **9411**, and **9416** were added as shown in the screenshot below. Each Destination Code was configured with following paramters.

- **Normal Route:** 001 (created in **Section 5.5.4**).
- **Absorbed Length:** 1. The leading digit 9 will be deleted.

The dialing plan on BCM450 was configured with above Destination Codes to do following:

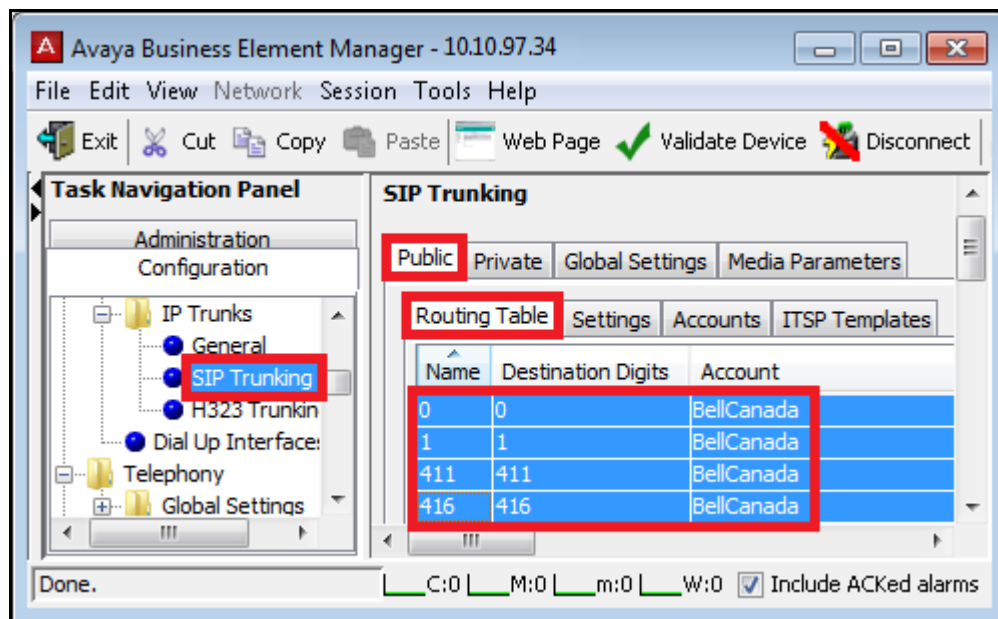
- Destination Code **90**: To make operator assisted calls by dialing 0 or 0 + 10 digits. It is also be used to make an international call by dialing 011 + Country Code + Area Code + DN.
- Destination Code **91**: To make long distance call by dialing 1 + 10 digits.
- Destination Code **9411**: To make directory assisted calls by dialing 411.
- Destination Code **9416**: To make 10 digit local calls with are code 416 by dialing 416 + 6 digit extension.



5.5.6. Administer Outgoing Call - SIP Trunk Routing Table

As the Destination Codes are defined to use Line Pool **BlocA** that contains VoIP Lines which may be shared between H.323 and SIP Trunks. In order to route outgoing calls with the defined Destination Codes over the SIP Trunk, the appropriate routes need to be defined under the Routing Table for SIP Trunk.

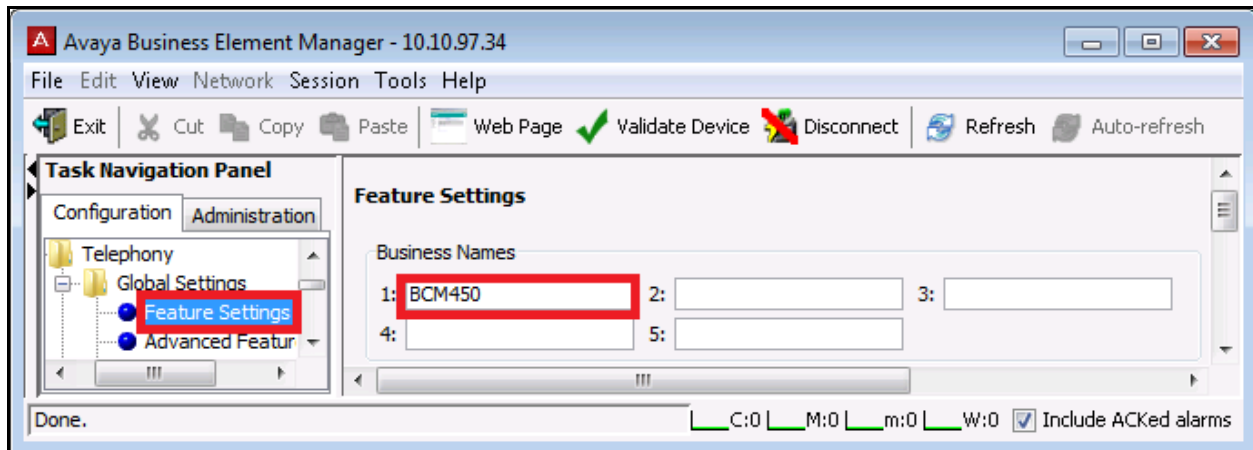
To create an entry in the Routing Table for the SIP Trunk, select **Configuration → Resources → IP Trunk → SIP Trunking**. Under the **Public** tab, select **Routing Table** and then click **Add** (not shown). In the compliance testing, multiple routes e.g. **0**, **1**, **411**, and **416** were created for Public Account **BellCanada** (see **Section 5.3.4**) as shown in the screenshot below.



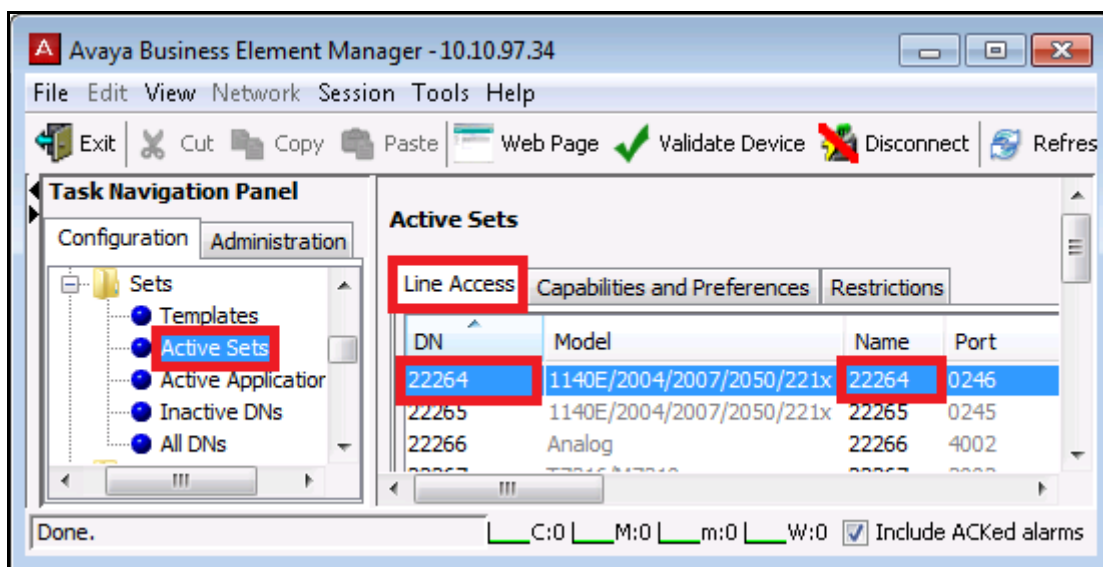
5.5.7. Administer Outgoing Call - Calling Party Name Delivery

This section shows how to configure Calling Party Name delivery over the SIP Trunk. For example, when an active DN e.g. **22264** makes an outgoing call, the display name in the “From” header will be constructed using the **Business Name** and the phone **Name** configured as following.

To define the **Business Name**, select **Configuration → Telephony → Global Settings → Feature Settings**. Then define the **Business Names Entry 1** as **BCM450** as shown in the screenshot below.



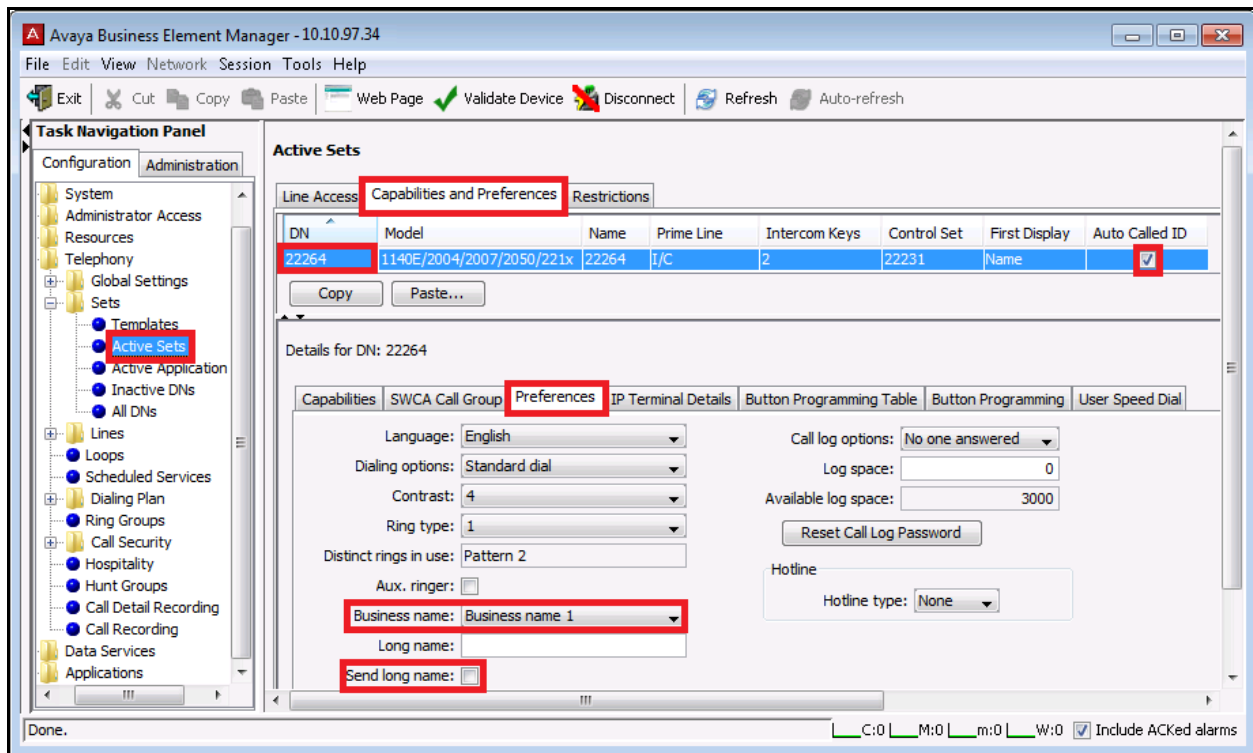
To define the phone **Name**, select **Configuration → Telephony → Sets → Active Sets**. Under the **Line Access** tab, define **Name** for an active DN e.g. **22264** as **22264** as shown in screenshot below.



Continue with the **Active Sets** settings, under the **Capabilities and Preferences** tab, check **Auto Called ID** to enable the Calling Party Name to display before the call is being answered. In the

the **Preferences** sub tab, select the **Business name** as **Business name 1** and uncheck **Send long name** as shown in the screenshot below.

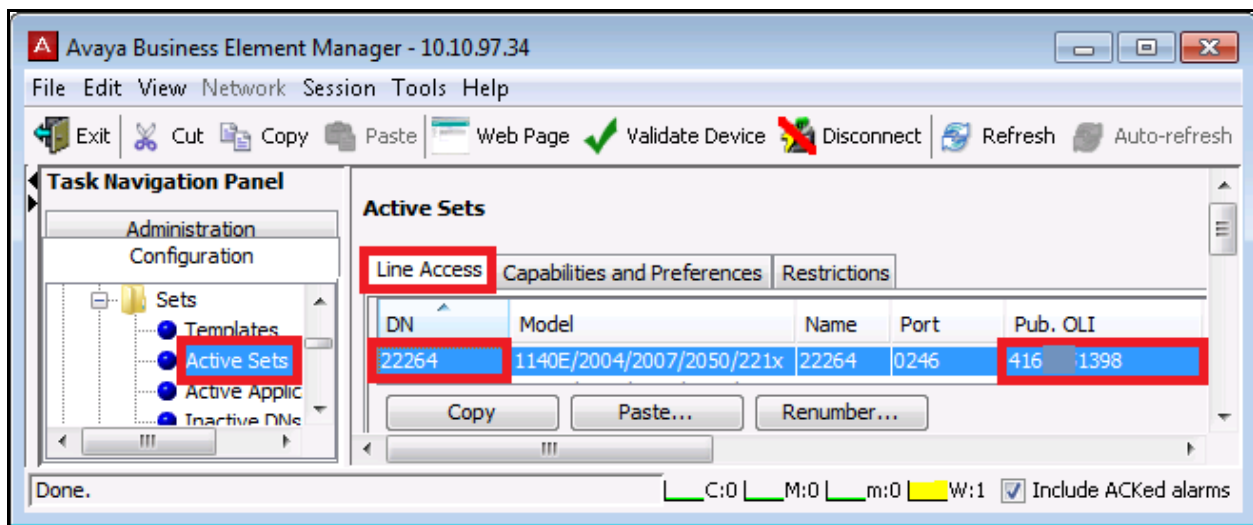
- **Business name:** Business name 1.
- **Send long name:** Unchecked. This disables the predefined **Long name** to be sent.



5.5.8. Administer Outgoing Call - Calling Party Number Delivery

For outgoing calls, the Calling Party Number will be derived from the **Pub. OLI** setting of an active DN. It will be presented as the URI-User of the “From” header.

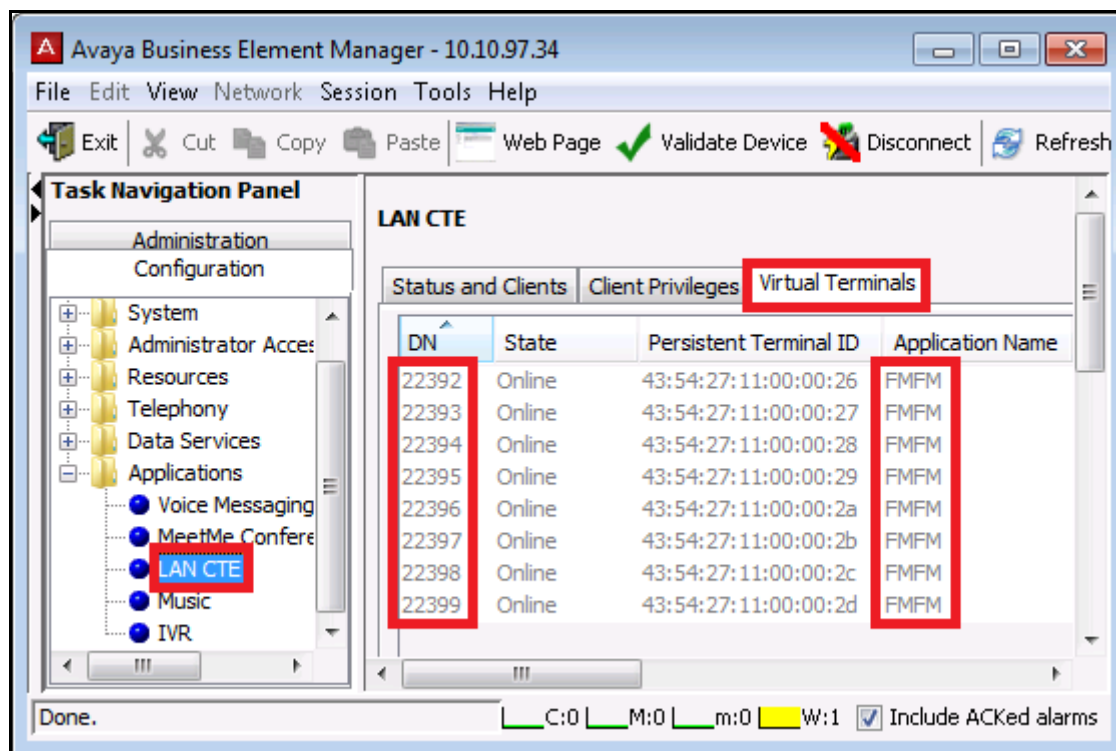
To configure the **Pub. OLI**, select **Configuration → Telephony → Sets → Active Sets**. Under **Line Access** tab, define **Pub. OLI** for an active DN e.g. **22264** as **416XXX1398** as shown in the screenshot below.



5.5.9. Administer Outgoing Call - Find Me Follow Me

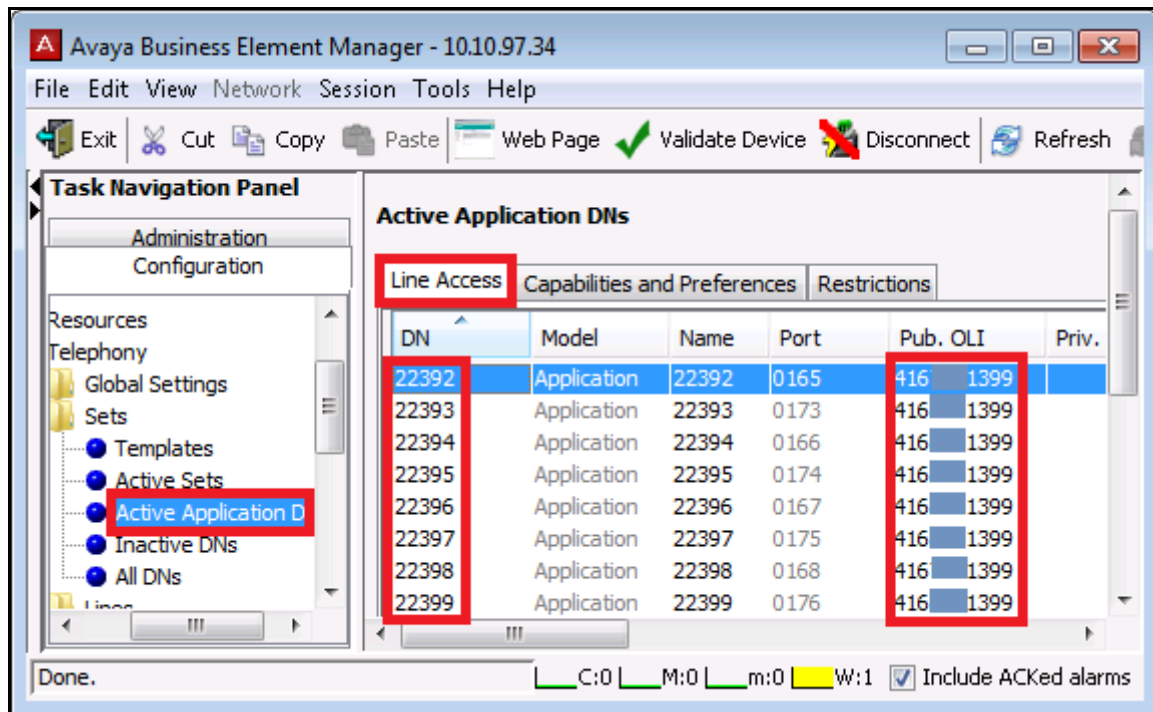
Find Me Follow Me (FMFM) is a feature on BCM450 to allow an incoming call to a local phone can be twinned to a PSTN party which may be an cellular extension. This section shows additional configuration for Application DNs to support the FMFM calls. It is assumed that the basic configuration for FMFM has already been provisioned and will not discuss here.

To locate the Application DNs which were used for FMFM, select **Application** → **LAN CTE** → **Virtual Terminals**. The Virtual Terminals tab displays a list of Application DNs where used for FMFM call. The screenshot below shows DN range **22392-22399** with Application Name **FMFM**.



When making an FMFM call to cellular extension, BCM450 will use the **Pub. OLI** assigned to the Application DN as the Calling Party Number. It will be included as the URI-User of the “From” header. In the compliance testing, Bell Canada examined the URI-User of the “From” header for call authentication purpose, it only grants access for the call if the URI-User is one of the assigned DID number, otherwise outgoing calls will be rejected. Thus, the **Pub. OLI** of the appropriate Application DNs has to be set to a typical value e.g. **416XXX1399** which is one of the assigned DID numbers. However, this implementation is limited as per the observation 6 in **Section 2.2**, the FMFM users will display the **Pub. OLI 416XXX1399** instead of the original PSTN caller.

To set the **Pub. OLI** for a DN, refer to **Section 5.5.8**. The following screenshot shows the Application DN range **22392-22399** with the **Pub. OLI** was set to **416XXX1399**.



5.5.10. Administer Incoming Call - Target Lines

For incoming calls, BCM450 uses Target Lines to terminate the calls that match the configured **Pub. Received** numbers. The Target Line is assigned as a key on the phone. The maximum available Target Line is ruled by the system license.

To add a Target Line to a a phone, select **Configuration → Telephony → Sets → Active Sets**. Under the **Line Access** tab, select an active DN e.g. **22264** then go to the **Line Assignment**, and click **Add** (not shown). In the **Add Line Assignment** dialog, enter the available Target Line e.g. **964**.

The sample configuration shows active DN **22264** was configured with Target Line **964** to receive incoming call to DID number **416XXX1398**. It is shown in the screenshot below.

- **Target Line:** 964.
- **Appearance Type:** Appr&Ring. This allows the Target Line to display as a key on the phone and it will ring per incoming call that matches the configured Priv. Received # or Pub. Received #.
- **Appearance:** 1. The Target Line will be displayed as 1 particular key on the phone. Increase the Appearance if more than 1 key are needed.
- **Caller IP Set:** Checked. This enables the Calling Party Name and Number to display on the phone.
- **Vmsg Set:** Checked if the phone has voice mail enabled on the integrated CallPilot, otherwise leave this option as unchecked.

- **Priv. Received #:** Enter the private number assigned to the active DN e.g. **22264** to receive local system calls or calls over private trunk.
- **Pub. Received #:** Enter the public DID number assigned to the active DN e.g. **416XXX1398** to receive calls over public trunk.

The screenshot shows the Avaya Business Element Manager interface. On the left is the Task Navigation Panel with a tree view containing 'Administration', 'Configuration', 'Sets', 'Lines', 'Dialing Plan', 'Call Security', and 'Hospitality'. The 'Active Sets' option under 'Sets' is highlighted. The main window is titled 'Active Sets' and has three tabs: 'Line Access' (selected), 'Capabilities and Preferences', and 'Restrictions'. Below the tabs is a table with columns: DN, Model, Name, Port, Pub. OLI, and Priv. OLI. The first row is highlighted with a red box and contains the values: 22264, 1140E/2004/2007/2050/221x, 22264, 0246, 416 1398, and 22264. Below the table are buttons for 'Copy', 'Paste...', and 'Renumber...'. Underneath is a section titled 'Details for DN: 22264' with four tabs: 'Line Assignment' (selected), 'Line Pool Access', 'Answer DNs', and 'MeetMe Conferencing'. Below these tabs is a table titled 'Assigned Lines' with columns: Line, Appearance Type, Appearances, Caller ID Set, Vmsg Set, Priv. Receive..., and Pub. Received #. The first row is highlighted with a red box and contains the values: 964, Appr&Ring, 1, [checked], [checked], 22264, and 416 1398. At the bottom of the window, there is a status bar with 'Done.' and a row of indicators: C:0, M:0, m:0, W:1, and a checked box for 'Include ACKed alarms'.

6. Verification and Troubleshooting

This section provides verification steps that may be performed in the field to verify that the solution is configured properly. This section also provides a list of useful commands that can be used to troubleshoot the solution.

6.1. Verification Steps

The following activities are made to each test scenario:

- Verify that endpoints at the enterprise site can place calls to PSTN and that the call remains active for more than 35 seconds. This time period is included to verify that proper routing of the SIP messaging has satisfied SIP protocol timers.
- Verify that endpoints at the enterprise site can receive calls from PSTN and that the call can remain active for more than 35 seconds.
- Verify that the user on PSTN can end an active call by hanging up.
- Verify that an endpoint at the enterprise site can end an active call by hanging up.

6.2. Protocol Traces

The following SIP message headers are inspected using sniffer traces:

- Request-URI: Verify the request number and the SIP domain.
- From: Verify the display name and display number.
- To: Verify the display name and display number.
- P-Asserted-Identity: Verify the display name and display number.
- Privacy: Verify the privacy masking with “user, id”.
- Diversion: Verify the DID number.
- Authorization: Verify the Digest Authentication implementation.

The following attributes in SIP message body are inspected using sniffer traces:

- Connection Information (c line): Verify the IP addresses of near and far endpoints.
- Time Description (t line): Verify the session timeout value of near and far endpoints.
- Media Description (m line): Verify the audio port, codec, DTMF event description.
- Media Attribute (a line): Verify the specific audio port, codec,ptime, send/ receive abilities, DTMF event and fax attributes.

Use a network sniffing tool e.g. Wireshark to monitor the SIP signaling between the enterprise and Bell Canada. The sniffer traces are captured at the public interface of BCM450.

Following screenshots show an example incoming call from Bell Canada to the enterprise.
Incoming INVITE request from Bell Canada.

```
INVITE sip:416XXX1398@cust2xxxx.xxxx.bell.ca;transport=udp SIP/2.0
Via: SIP/2.0/UDP 220.20.237.201:5060;branch=z9hG4bKi0m9ki3038a1o186n0c1.1
From: <sip:+1613XXX5258@sipxxxxxxxx.bell.ca;user=phone>;tag=SDocfbb01-453254403-1350677715771-
To: "Bell Demo12345"<sip:416XXX1398@cust2xxxx.xxxx.bell.ca>
Call-ID: SDocfbb01-09eee563259466f2de0a5057d0f97d6f-a0n8330
CSeq: 1028992414 INVITE
Contact: <sip:+1613XXX5258@220.20.237.201:5060;transport=udp>
Supported: 100rel
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Accept: multipart/mixed,application/media_control+xml,application/sdp
Max-Forwards: 18
Content-Type: application/sdp
Content-Length: 207

v=0
o=BroadWorks 6273513 1 IN IP4 220.20.237.201
s=-
c=IN IP4 220.20.237.201
t=0 0
m=audio 20396 RTP/AVP 0 18 101
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
aptime:20
a=fmtp:18 annexb=no
```

- 200OK response from the enterprise.

```
SIP/2.0 200 OK
From: <sip:+1613XXX5258@sipxxxxxxxx.bell.ca;user=phone>;tag=SDocfbb01-453254403-1350677715771-
To: "Bell Demo12345"<sip:416XXX1398@cust2xxxx.xxxx.bell.ca>;tag=344575a8-870a6122-13c4-55013-204e-13ec910e-204e
Call-ID: SDocfbb01-09eee563259466f2de0a5057d0f97d6f-a0n8330
CSeq: 1028992414 INVITE
Via: SIP/2.0/UDP 220.20.237.201:5060;branch=z9hG4bKi0m9ki3038a1ol86n0c1.1
Supported: sipvc,x-nortel-sipvc,100rel,replaces,timer
User-Agent: Nortel Networks BCM VoIP Gateway release_46 version_46.51.0.33
P-Asserted-Identity: "BCM450 22264"<sip:416XXX1398@cust2xxxx.xxxx.bell.ca>
x-nt-corr-id: SDocfbb01-09eee563259466f2de0a5057d0f97d6f-a0n8330
Contact: <sip:416XXX1398@10.10.97.34:5060;transport=udp>
Session-Expires: 90;refresher=uas
Allow: INVITE,INFO,ACK,OPTIONS,CANCEL,BYE,NOTIFY,PRACK,UPDATE,REFER
Content-Type: application/sdp
Content-Length: 241

v=0
o=- 1350679313 1350679313 IN IP4 10.10.97.34
s=-
c=IN IP4 10.10.97.34
t=0 0
m=audio 30160 RTP/AVP 0 101
c=IN IP4 10.10.97.34
a=fmtp:101 0-15
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

Following screenshots show an example outgoing call from the enterprise to Bell Canada.

- Outgoing INVITE request from the enterprise.

```
INVITE sip:1613XXX5279@sipxxxxxxxx.bell.ca;transport=udp SIP/2.0
From: "BCM450 22264"<sip:416XXX1398@cust2xxxx.xxxx.bell.ca>;tag=34414cd8-870a6122-13c4-55013-19b8-44869384-19b8
To: <sip:1613XXX5279@sipxxxxxxxx.bell.ca>
Call-ID: 345a1170-870a6122-13c4-55013-19b8-6f233c7e-19b8
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.10.97.34:5060;branch=z9hG4bK-19b8-6479a3-c1d6dbd
Max-Forwards: 70
Supported: sipvc,x-nortel-sipvc,100rel,replaces,timer
User-Agent: Nortel Networks BCM VoIP Gateway release_46 version_46.51.0.33
P-Asserted-Identity: "BCM450 22264"<sip:416XXX1398@cust2xxxx.xxxx.bell.ca>
Diversion: <sip:4167751881@cust2xxxx.xxxx.bell.ca>;reason=unconditional
Route: <sip:220.20.237.201:5060;lr>
x-nt-corr-id: 345a1170-870a6122-13c4-55013-19b8-6f233c7e-19b8
Contact: <sip:416XXX1398@10.10.97.34:5060;transport=udp>
Session-Expires: 90
Allow: INVITE,INFO,ACK,OPTIONS,CANCEL,BYE,NOTIFY,PRACK,UPDATE,REFER
Content-Type: application/sdp
Content-Length: 263

v=0
o=- 1350677620 1350677620 IN IP4 10.10.97.34
s=-
c=IN IP4 10.10.97.34
t=0 0
m=audio 30142 RTP/AVP 0 18 101 111
c=IN IP4 10.10.97.34
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:111 X-nt-inforeq/8000
a=sendrecv
```

- Incoming 401 challenge to request the Digest Authentication.

```
SIP/2.0 401 Unauthorized
From: "BCM450 22264"<sip:416XXX1398@cust2xxxx.xxxx.bell.ca>;tag=34414cd8-870a6122-13c4-55013-19b8-44869384-19b8
To: <sip:1613XXX5279@sipxxxxxxxx.bell.ca>;tag=SDdsnqe99-2020444452-1350676030100
Call-ID: 345a1170-870a6122-13c4-55013-19b8-6f233c7e-19b8
CSeq: 1 INVITE
Via: SIP/2.0/UDP 10.10.97.34:5060;branch=z9hG4bK-19b8-6479a3-c1d6dbd
WWW-Authenticate: DIGEST
qop="auth",nonce="BroadWorksXh8hpmutwT1lz2y4BW",realm="sipxxxxxxxx.bell.ca",algorithm=MD5
Content-Length: 0
```

- Outgoing subsequent INVITE with Authorization header to respond to the Digest Authentication.

```

INVITE sip:1613XXX5279@sipxxxxxxx.bell.ca;transport=udp SIP/2.0
From: "BCM450 22264"<sip:416XXX1398@cust2xxxx.xxxx.bell.ca>;tag=34414cd8-870a6122-13c4-55013-19b8-44869384-19b8
To: <sip:1613XXX5279@sipxxxxxxx.bell.ca>
Call-ID: 345a1170-870a6122-13c4-55013-19b8-6f233c7e-19b8
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.10.97.34:5060;branch=z9hG4bK-19b8-6479ea-1208da2b
Max-Forwards: 70
Supported: sipvc,x-nortel-sipvc,100rel,replaces,timer
User-Agent: Nortel Networks BCM VoIP Gateway release_46 version_46.51.0.33
P-Asserted-Identity: "BCM450 22264"<sip:416XXX1398@cust2xxxx.xxxx.bell.ca>
Diversion: <sip:4167751881@cust2xxxx.xxxx.bell.ca>;reason=unconditional
Route: <sip:220.20.237.201:5060;lr>
x-nt-corr-id: 345a1170-870a6122-13c4-55013-19b8-6f233c7e-19b8
Contact: <sip:416XXX1398@10.10.97.34:5060;transport=udp>
Authorization: Digest
username="avaya",realm="sipxxxxxxx.bell.ca",nonce="BroadWorksXh8hpmutwT11z2y4BW",u
ri="sip:1613XXX5279@sipxxxxxxx.bell.ca;transport=udp",response="14a9e80efc158f4dce
a494848675b059",algorithm=MD5,cnonce="6479ec",qop=auth,nc=00000001
Session-Expires: 90
Allow: INVITE,INFO,ACK,OPTIONS,CANCEL,BYE,NOTIFY,PRACK,UPDATE,REFER
Content-Type: application/sdp
Content-Length: 263

v=0
o=- 1350677620 1350677620 IN IP4 10.10.97.34
s=-
c=IN IP4 10.10.97.34
t=0 0
m=audio 30142 RTP/AVP 0 18 101 111
c=IN IP4 10.10.97.34
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:111 X-nt-inforeq/8000
a=sendrecv

```


- Incoming 200OK response from Bell Canada.

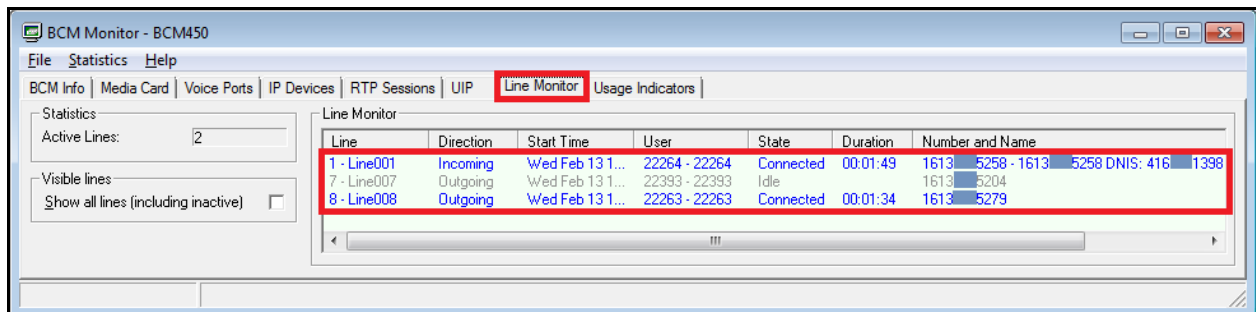
```
SIP/2.0 200 OK
From: "BCM450 22264"<sip:416XXX1398@cust2xxxx.xxxx.bell.ca>;tag=34414cd8-870a6122-13c4-55013-19b8-44869384-19b8
To: <sip:1613XXX5279@sipxxxxxxxxx.bell.ca>;tag=SDdsnqe99-1312326424-1350676031266
Call-ID: 345a1170-870a6122-13c4-55013-19b8-6f233c7e-19b8
CSeq: 2 INVITE
Via: SIP/2.0/UDP 10.10.97.34:5060;branch=z9hG4bK-19b8-6479ea-1208da2b
Supported:
Contact: <sip:1613XXX5279@220.20.237.201:5060;transport=udp>
P-Asserted-Identity: <sip:1613XXX5279@sipxxxxxxxxx.bell.ca;user=phone>
Allow: ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Accept: multipart/mixed,application/media_control+xml,application/sdp
Content-Type: application/sdp
Content-Length: 183

v=0
o=BroadWorks 6267245 2 IN IP4 220.20.237.201
s=-
c=IN IP4 220.20.237.201
t=0 0
m=audio 20386 RTP/AVP 0 101
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:20
```

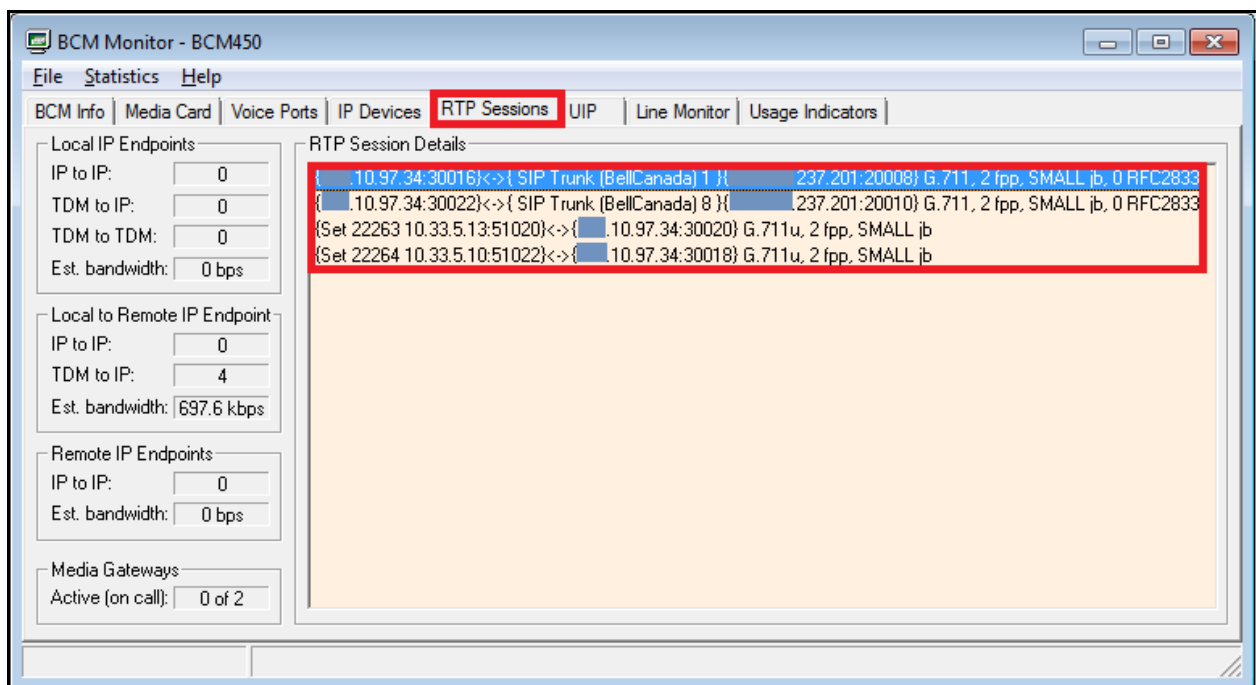
6.3. Troubleshooting

The BCM Monitor is a builtin troubleshooting tool for BCM450. This section shows how to use this tool to monitor incoming and outgoing calls. For further information, see the **References** in **Section 8**.

Use BCM Monitor tool to monitor VoIP Lines for incoming and outgoing calls. The following screenshot shows Line001 was used for an active incoming call and Line008 was used for an active outgoing call.



Use BCM450 Monitor to monitor RTPs. The following screenshot shows 2 external RTP streams connecting to Bell Canada and 2 internal RTP streams connecting to local phones. In the compliance testing, BCM450 was configured to relay the media using G.711 codec.



7. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Business Communication Manager 450 Release 6.0 to Bell Canada SIP Trunking Service. Bell Canada SIP Trunking Service is a SIP-based Voice over IP solution for customers ranging from small businesses to large enterprises. Bell Canada SIP Trunking Service provides a flexible, cost-saving alternative to traditional hardwired telephony trunks.

All of the test cases have been executed. Despite the number of observations seen during testing as noted in **Section 2.2** the test results met the objectives outlined in **Section 2.1**. Bell Canada SIP Trunking Service is considered **compliant** with Avaya Business Communication Manager 450 Release 6.0.

8. References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Configuration —Telephony*, Avaya Business Communication Manager, Release 6.0, Document Number NN40170-502, Revision: 03.01, May 2010.
- [2] *Planning and Engineering*, Avaya Business Communication Manager, Release 6.0, Document Number NN40170-200, Revision: 02.01, May 2010.
- [3] *CallPilot Reference Guide*, Avaya Business Communication Manager, Release 6.0, Document Number NN40170-100, Revision 02.01, May 2010.
- [4] *RFC 3261 SIP: Session Initiation Protocol*, <http://www.ietf.org/>.
- [5] *RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, <http://www.ietf.org/>.

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