Avaya Solution & Interoperability Test Lab

Application Notes for Configuring CenturyLink SIP Trunking with the Avaya Business Communication Manager 450 Release 6.0 – Issue 1.0

Abstract

These Application Notes describe a solution comprised of the Avaya Business Communication Manager 450 release 6.0 and the CenturyLink SIP Trunking. During the interoperability testing, Avaya Business Communication Manager 450 was able to interoperate with the CenturyLink Broadsoft via SIP trunk. This test was performed to verify SIP trunk features including basic call, call forward (all calls, busy, no answer), call transfer (blind and consult), conference, and voice mail. The calls are placed in both directions with various set types.

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

This document provides a typical network configuration deployment of the Avaya Business Communication Manager 450 (hereafter referred to as BCM450) and the CenturyLink SIP Trunking (hereafter referred to as CenturyLink system). During the interoperability testing, all SIP trunk applicable feature test cases were executed to ensure the interoperability between the CenturyLink system and the BCM450.

2. General Test Approach and Test Results

The Business Communication Manager 450 release 6.0 was connected to the CenturyLink system via a SIP trunk. Various call types were made from the BCM450 to the CenturyLink system and vice versa to ensure the interoperability between the BCM450 and the CenturyLink system.

2.1. Interoperability Compliance Testing

The focus of this testing is to verify that BCM450 release 6.0 can interoperate with the CenturyLink system. The following interoperability areas were covered.

- General call processing between BCM450 and CenturyLink systems including:
  - Codec/ptime (G.711 u-law)
  - Hold/Retrieve on both ends
  - CLID displays
  - Ring-back tone
  - Speech paths
  - Dialing plan support
  - Advanced features (Call on Mute, Call Park, Call Waiting)
  - Abandoned Call
- Call redirection verification: all supported methods (blind transfer, consultative transfer, call forward, and conference) including CLID. Call redirection is performed from both ends
- RFC2833/DTMF on both direction
- SIP Transport UDP
- Voice Mail Server CallPilot (hosted on BCM450)
- Meet-me conference (hosted on BCM450)
- Early Media Transmission

2.2. Test Results

The objectives outlined in **Section 2.1** were verified. All the applicable test cases were executed. However, the following observations were noted during the compliance testing:

01. Outbound call from BCM450 to PSTN, CLID Name is not delivered over CenturyLink SIP Trunk.
- Call scenario: Make an outbound SIP call from BCM450 phone to a PSTN phone.

- SIP observation: BCM450 sent "From" header with display name information.
- Expected result: Call is established with 2 way speech paths. CLID Number and Name are displayed correctly.
- Actual result: CLID Name is not delivered over CenturyLink SIP Trunk
- Recommendation: CenturyLink CLEC should populate CLID Name for the outgoing call to PSTN and CLID Name delivery needs to be supported by Local Exchange Carrier host PSTN test phones.

02. Inbound toll free call has not yet been tested due to lacking provision from CenturyLink.
- Call scenario: Make an inbound SIP call from PSTN phone to BCM450 toll free number.
- SIP observation: N/A.
- Expected result: Call is established with 2 way speech paths. CLID Number and Name are displayed correctly.
- Actual result: This case has not yet been tested on CenturyLink production network.
- Recommendation: Need CenturyLink to provide a toll free number associate to SIP Trunk to BCM450.

03. Inbound call waiting. CenturyLink should send INVITE for the 2nd inbound call to the same BCM450 phone.
- Call scenario: Make a SIP call from PSTN_1 to BCM450 phone, then make the 2nd inbound call from PSTN_2 to the same BCM450 phone.
- SIP observation: CenturyLink sent SIP/INFO – Call Waiting.
- Expected result: CenturyLink should send another INVITE for the 2nd call.
- Actual result: CenturyLink sent SIP/INFO – Call Waiting, BCM450 responded 501 Not Implemented.
- Resolution: Centurylink corrected the configuration and sent INVITE for the 2nd inbound call to BCM450. The call waiting feature on BCM450 was verified.

04. Off-net call forward. CenturyLink should increase Trunkgroup timer on SIP interface and disable Symmetric-Latching on CenturyLink SBC.
- Call scenario: Make a SIP call from PSTN_1 to BCM450 phone which will Call Forward All Call back to PSTN_2.
- SIP observation: CenturyLink sent CANCEL; this issue was corrected by increasing the Trunkgroup timer on the SIP interface at CenturyLink. With this setting, the call can be established, but there was an issue with RTP. Disabling Symmetric-Latching on the CenturyLink SBC fixed the RTP issue.
- Expected result: Call is established with 2 way speech paths. CLID Number and Name are displayed correctly.
- Actual result: The call was failed.
- Resolution: CenturyLink increased the Trunkgroup timer on the SIP interface and disabled Symmetric-Latching on the CenturyLink SBC and the test case passed.

05. BCM450 drops the 3 way conference if the 1st call leg was initialized by PSTN.
- Call scenario: Make a SIP call from PSTN_1 to BCM450_A which will conferences to PSTN_2. Then BCM450_A conferences to BCM450_B. BCM450_A and BCM450_B hang up.
- SIP observation: BCM450 sent BYE to drop the conference call.

- Expected result: The conference was successfully open for 4 call parties. The conference will be kept open even all BCM450 parties left.
- Actual result: The conference was dropped after all BCM450 parties left.
- Resolution: This is a known issue of BCM450.

06. Off-net call transfer.
- Call scenario: Make a SIP call from PSTN_1 to BCM450 phone which will blind transfer back to PSTN_2.
- SIP observation: BCM450 did not send INVITE/resume.
- Expected result: Call is established with 2 way speech paths. CLID Number and Name are displayed correctly.
- Actual result: The call was failed.
- Resolution: This issue has been fixed by patch BCM050.R600.IPTEL-37-3, applied to BCM450.

2.3. Support

For technical support on CenturyLink system, please contact CenturyLink technical support at:

- Toll Free: 1-877-290-5458
- http://www.centurylink.com/Pages/Support/

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliant testing event between the BCM450 and CenturyLink system.

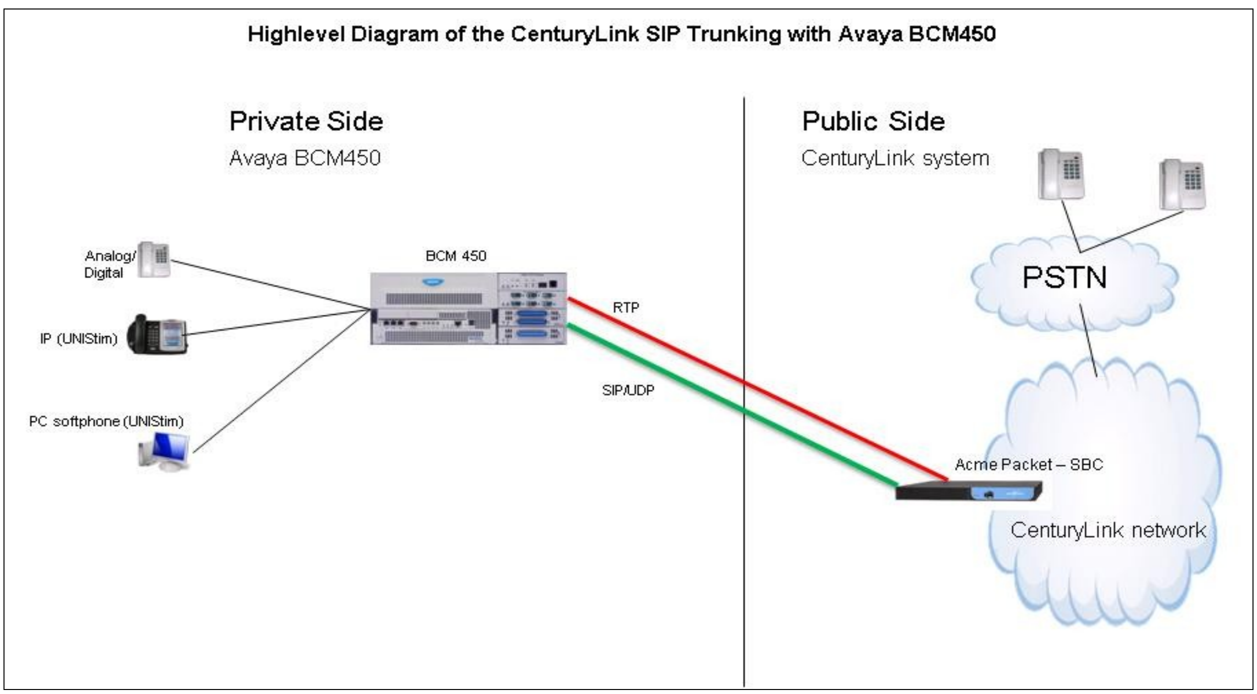

Figure 1- Network Diagram for BCM450 – CenturyLink System

The following assumptions were made for this lab test configuration.

1. BCM450 R6.0 software and implementation of latest patches
2. CenturyLink provides support to setup, configure, and troubleshoot on carrier switch for the duration of the testing.

During testing, the following activities were made to each test scenario:

1. Calls were checked for the correct call progress tones and cadences.
2. During the ringing state, the ring back tone and destination ringing were checked.
3. Calls were checked in both hands-free and handset mode due to internal Avaya requirements.
4. Calls were checked for speech path in both directions using spoken words to ensure clarity of speech.
5. The display(s) of the sets/clients involved were checked for consistent and expected CLID, name and redirection information both prior to answer and after call establishment.
6. The speech path and messaging system were observed for timely and quality End to End tone audio path generation and application responses.
7. The BCM450 Monitor was kept running to monitor SIP Trunking usage during the call and / the trunks were released after the call completed.
8. Speech path and display checked before and after calls were put on/off hold from each end.
9. Applicable files were screened on an hourly basis during the testing for any message that may indicate technical issues. This refers to Avaya PBX files.

4. Equipment and Software Validated

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

Avaya system:

System	Software/Loadware version
Avaya BCM450	Release 6.0 with following patches: • 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	• 2004 p2: 0604DCN (UNIStim) • 1140: 0625C7M (UNIStim) • 1120: 0624C7M (UNIStim) • 2007: 0621C7G (UNIStim) • 1220: 062AC7M (UNIStim) • Nortel Digital Phone M7310

	• I2050 PC softphone Release 3.2

CenturyLink system:

System	Software/Loadware version
Broadwork Broadsoft	R16 sp2
Sonus GSX9000	V07.02.05R000
Acme Packet Net-Net 4250	R6.1

5. Avaya Business Communication Manager 450 Configuration

These Application Notes assume that the basic configuration has already been administered. For further information on Avaya Business Communication Manager 450, please consult references in **Section 8.**

The below procedures describe the configuration details of BCM450 with a SIP trunk to CenturyLink system.

5.1. Login to BCM450

5.1.1. Install Business Element Manager and BCM450 Monitor

a) Open web browser and connect to the Web GUI http://<BCM450 IP address> as shown in **Figure 2**. Then log in using the appropriate Username and Password.

Note: The web browser has to enable Java Runtime Environment to support the BCM450 Web GUI.

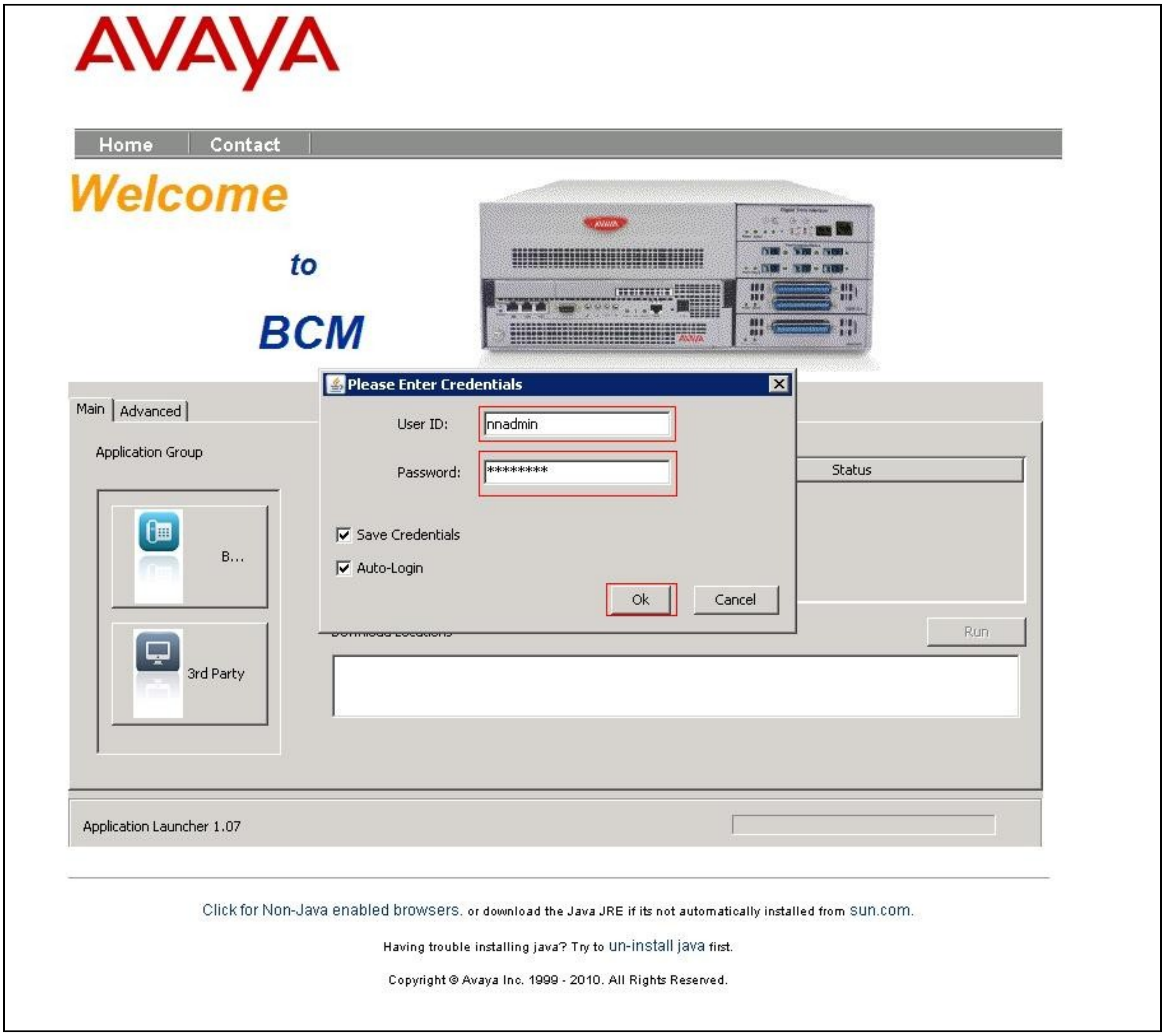

Figure 2 – Login to Business Communication Manager

b) The **Welcome to BCM450** page is displayed. Click on the **BCM450 applications/ web links**, select **Business Element Manager**, and then click **Run** as highlighted in red box as shown in **Figure 3**. This action will install **Business Element Manager** to the local PC. After the installation complete, **Business Element Manager** shortcut will be created on desktop.

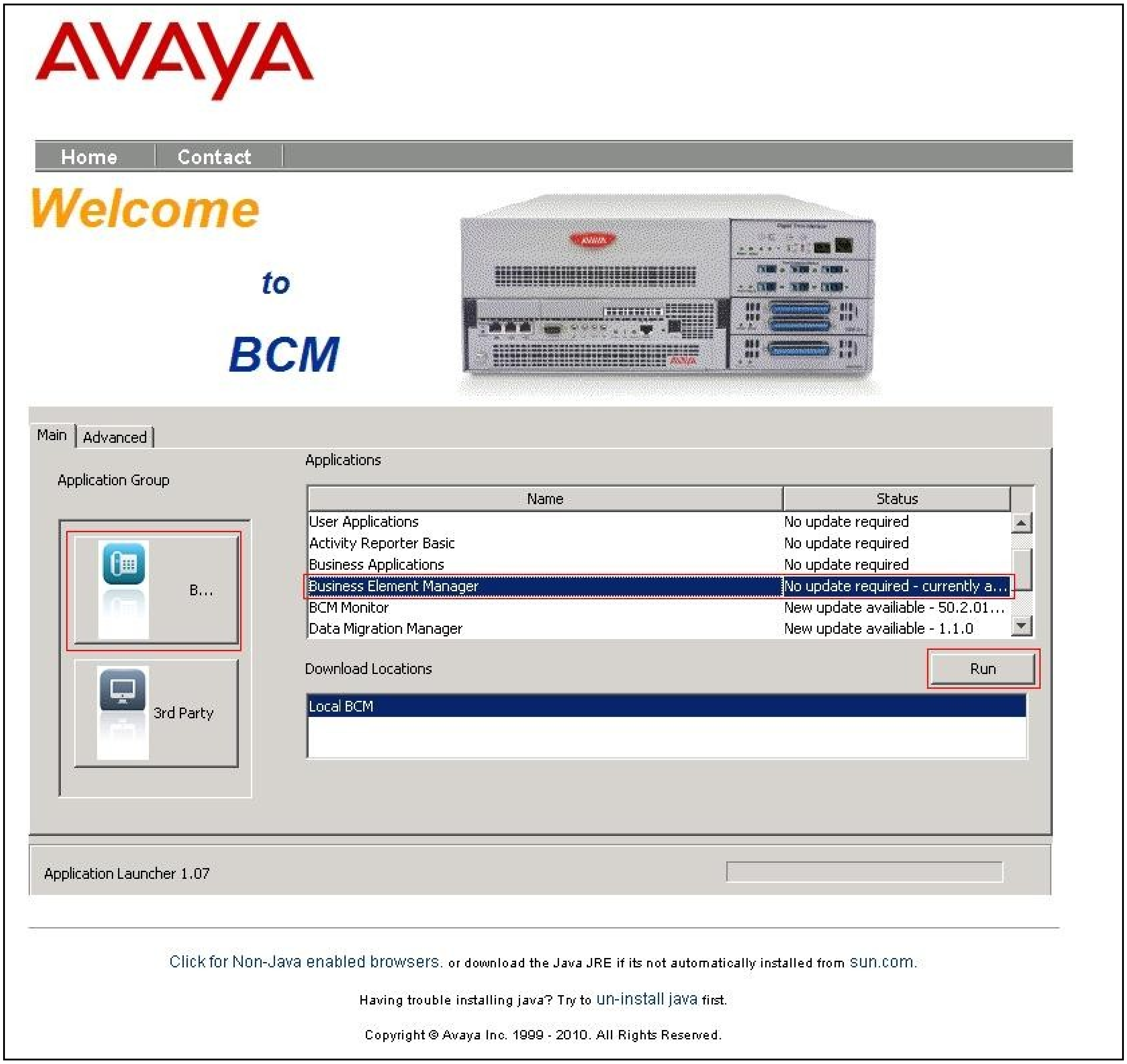

Figure 3 – Business Communication Manager

c) Continue with **Welcome to BCM450** page to install **BCM450 Monitor** as shown in **Figure 4**. After the installation complete, the **BCM450 Monitor** shortcut will be created on desktop.

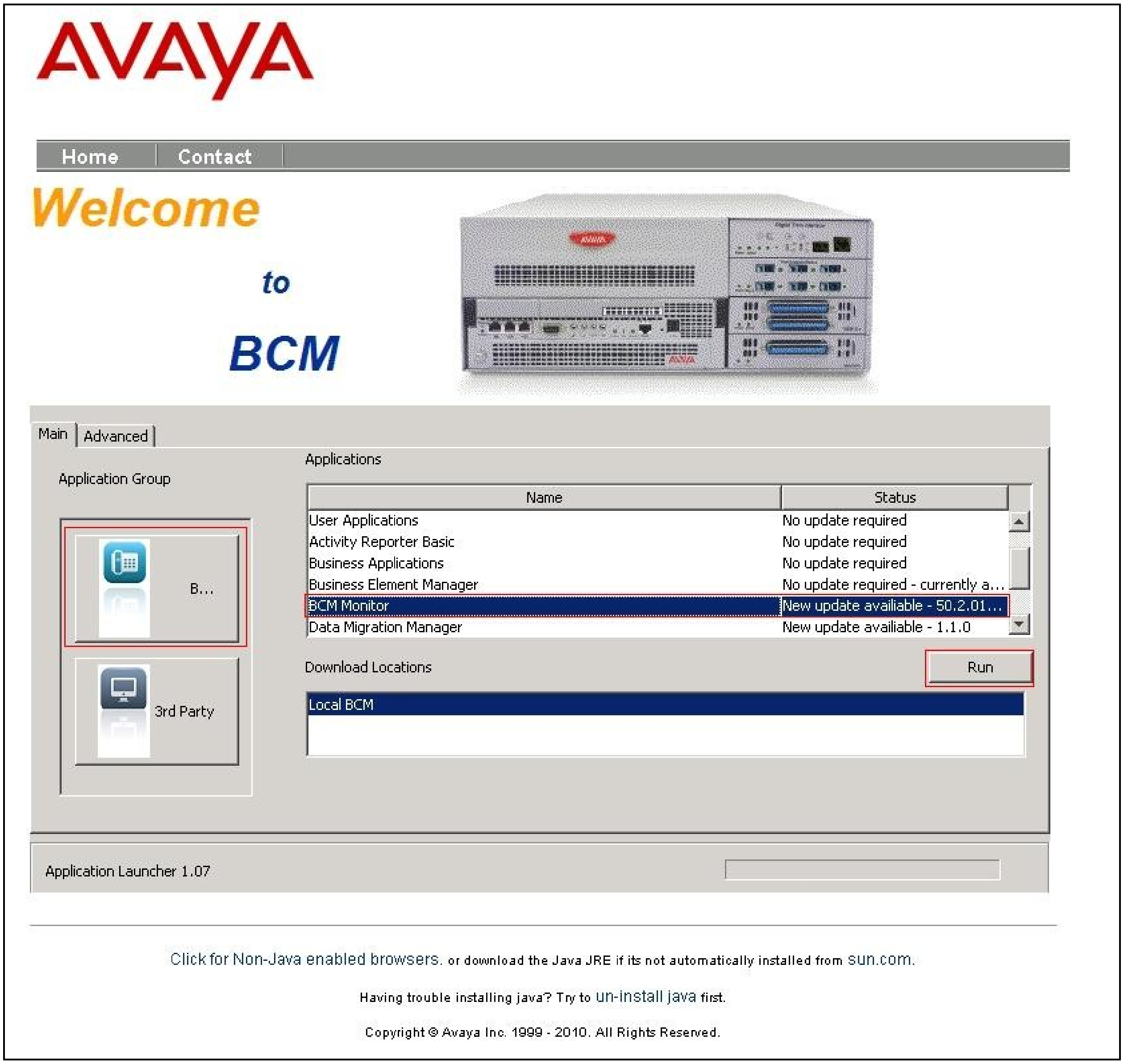

Figure 4 – Element Manager System Overview

5.1.2. Create a new Network Element Entry for Business Element Manager

a) Double click on the **Business Element Manager** desktop icon; the **Avaya Business Element Manager – Network Elements** will display. Create a new **Network Element** as shown in **Figure 5**.

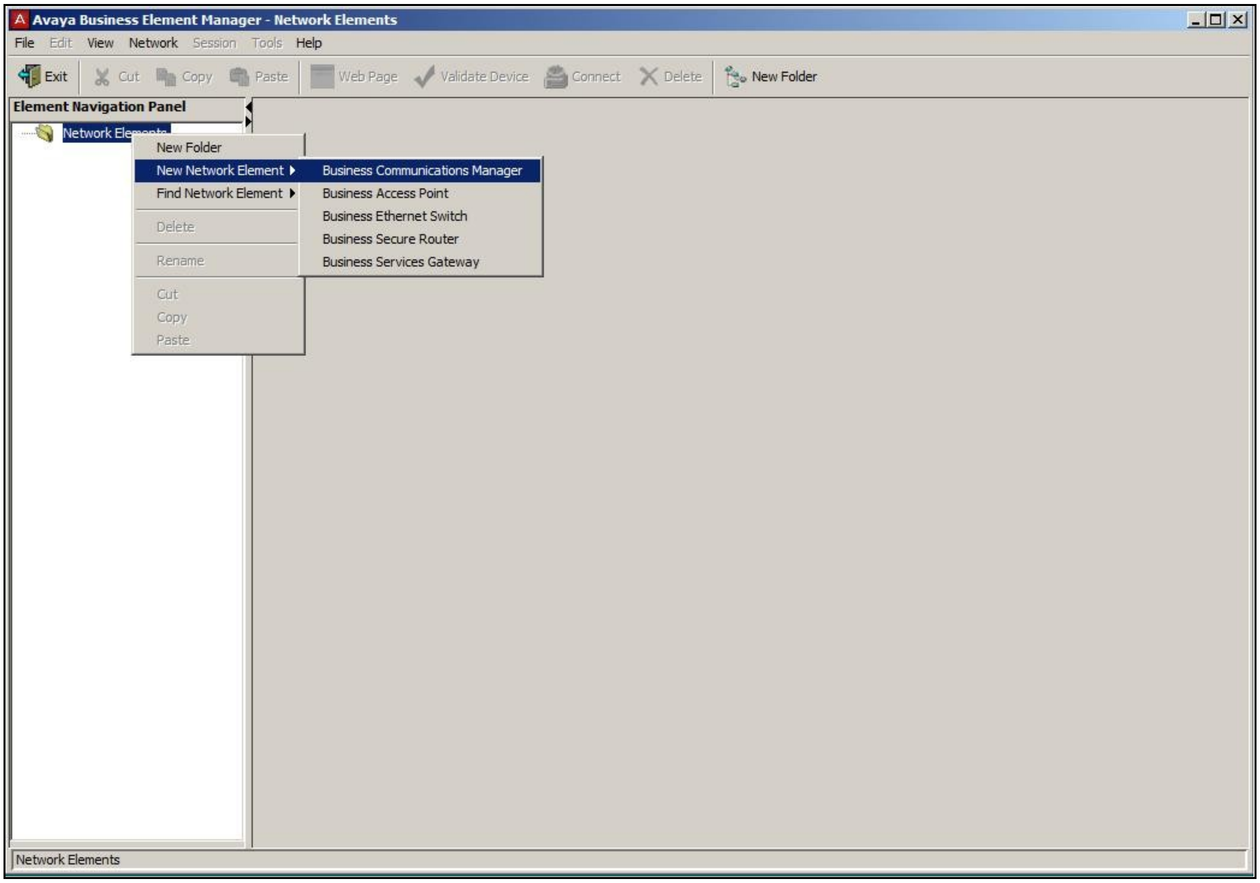

Figure 5: Create a New Network Element

b) Input IP address of BCM450, username: nnadmin and appropriate password to the red box as shown in **Figure 6**. Then click **OK**.

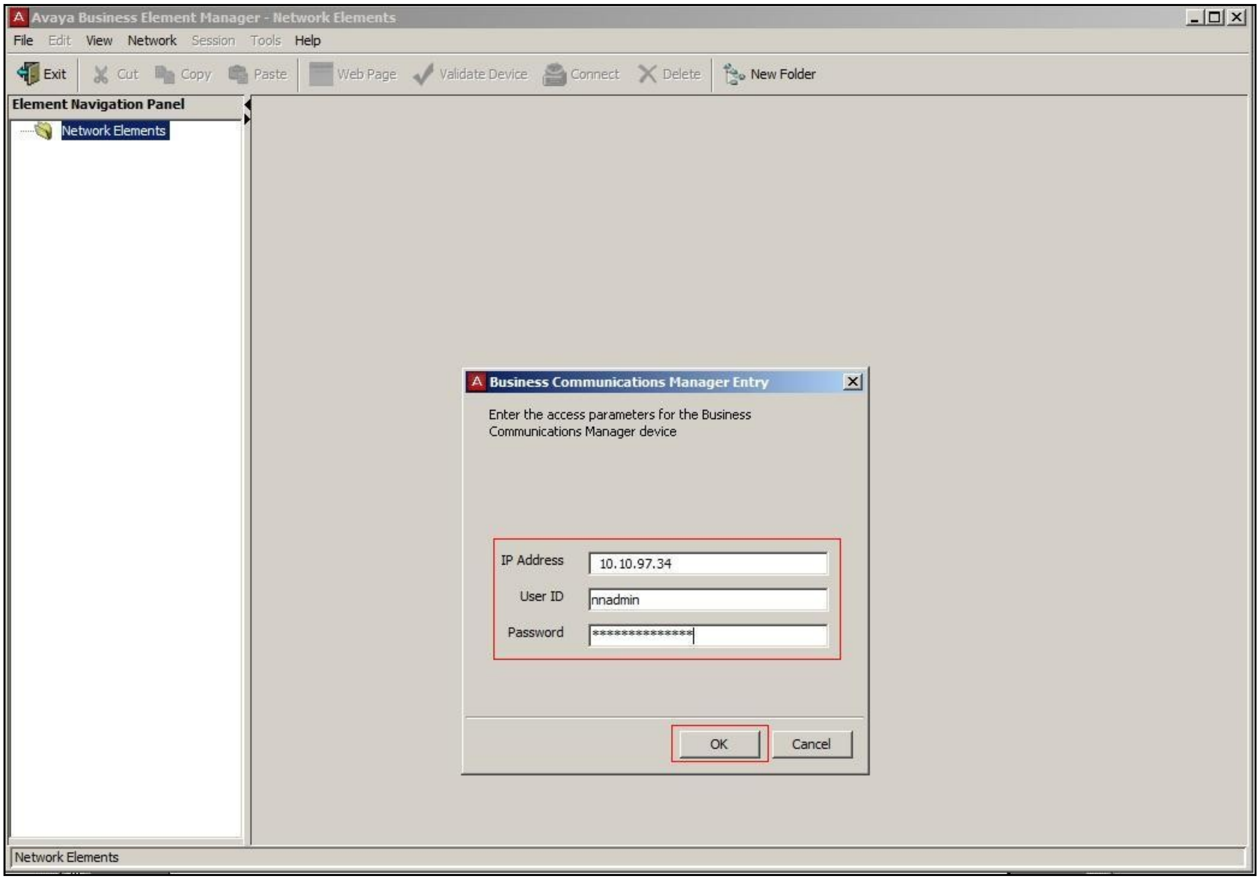

Figure 6: Business Communication Manager Entry

5.1.3. Login to Business Element Manager

a) Double click on the **Business Element Manager** desktop icon; select the **Network Element** then click **Connect** as shown in **Figure 7**.

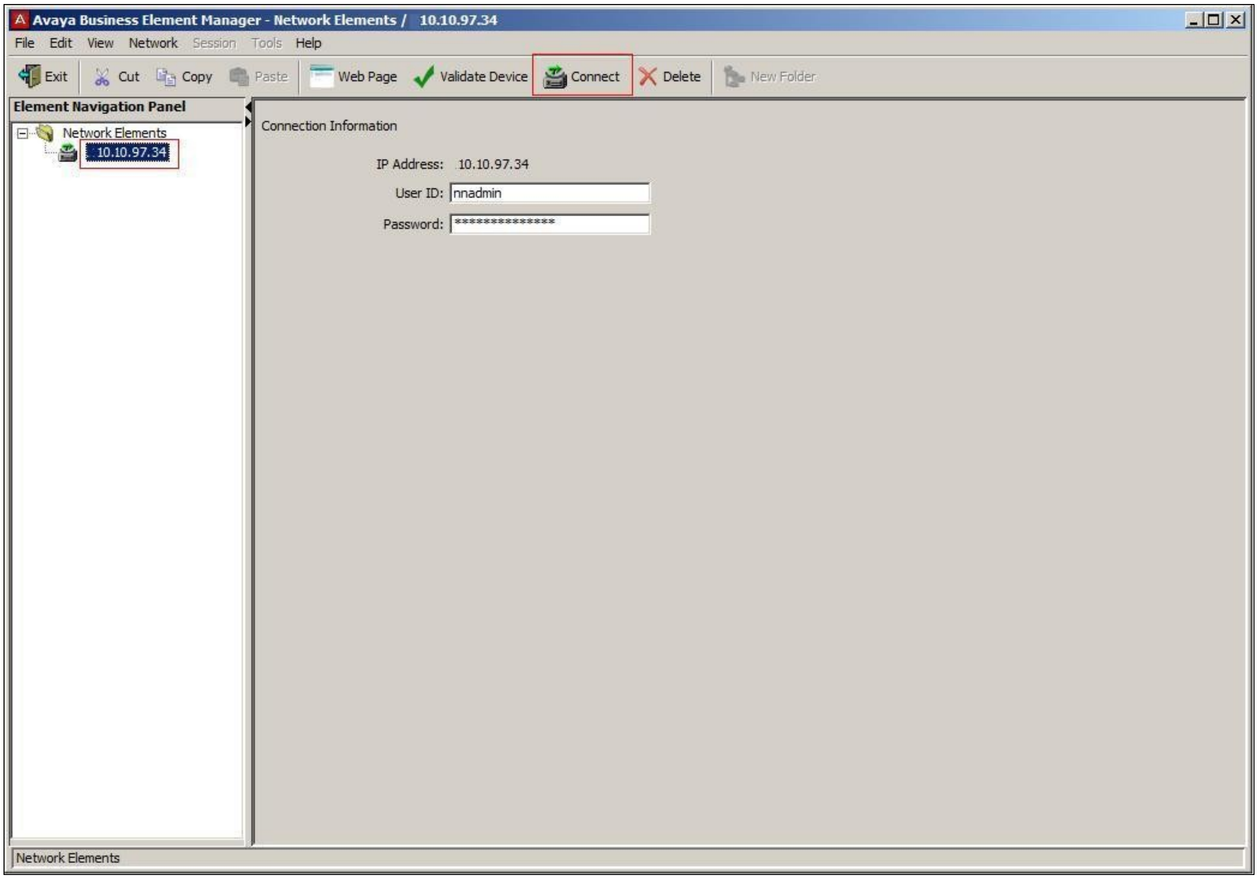

Figure 7: Connect to BCM450

b) After the connection has been established, click **OK** in the **Confirm** dialog (not shown). **Figure 8** shows **Business Element Manager** has been successfully logged on.

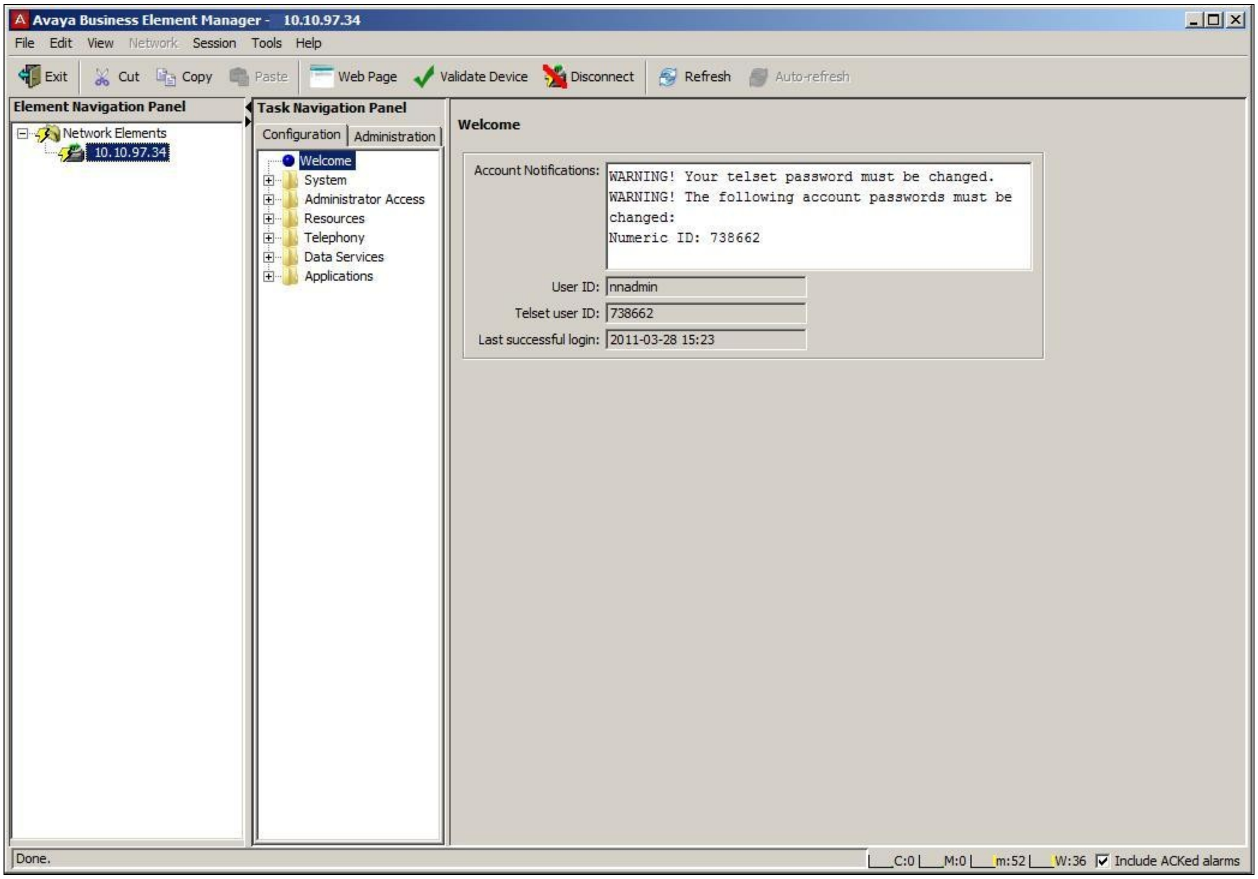

Figure 8: Avaya Business Element Manager

5.1.4. Login to BCM450 Monitor

a) Double click the **BCM450 Monitor** icon on the desktop. Then input IP address of BCM450, username: nnadmin and appropriate password as shown in **Figure 9**.

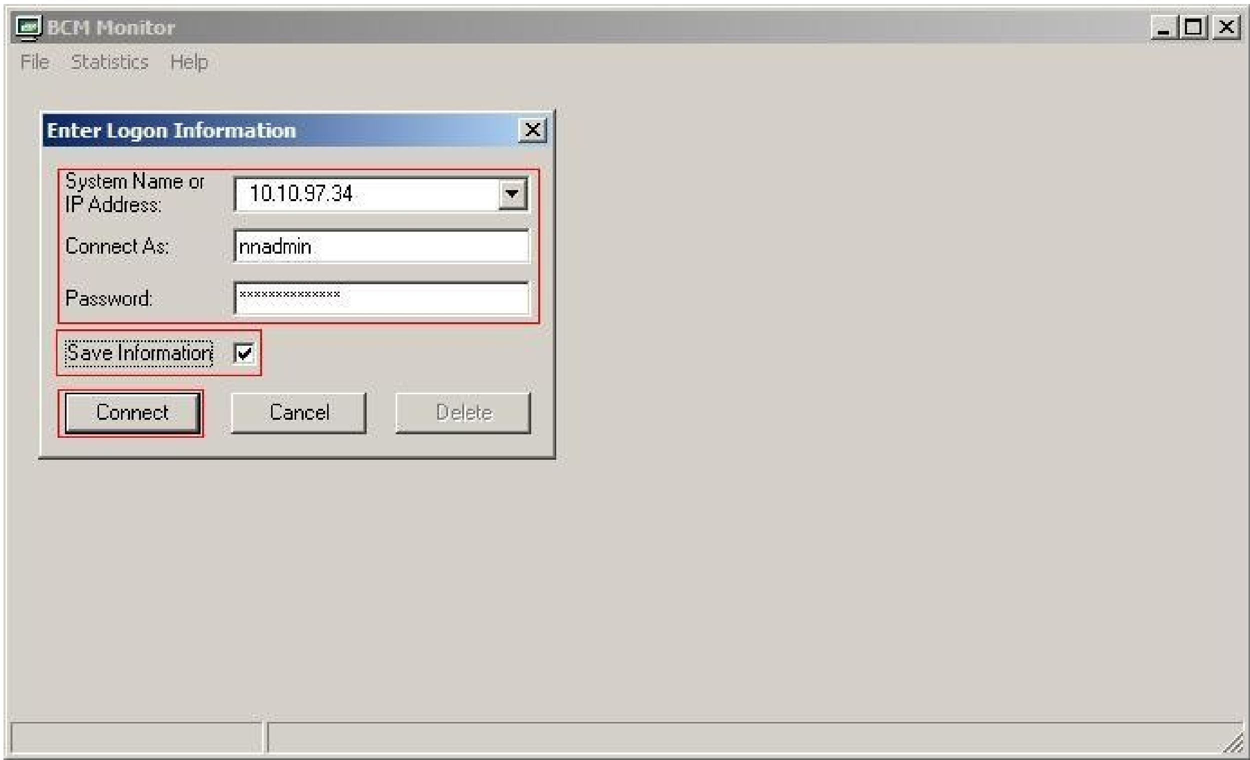

Figure 9: Enter Logon Information for BCM Monitor

b) Click **Connect**. The **BCM450 Monitor** GUI displays as shown in **Figure 10**.

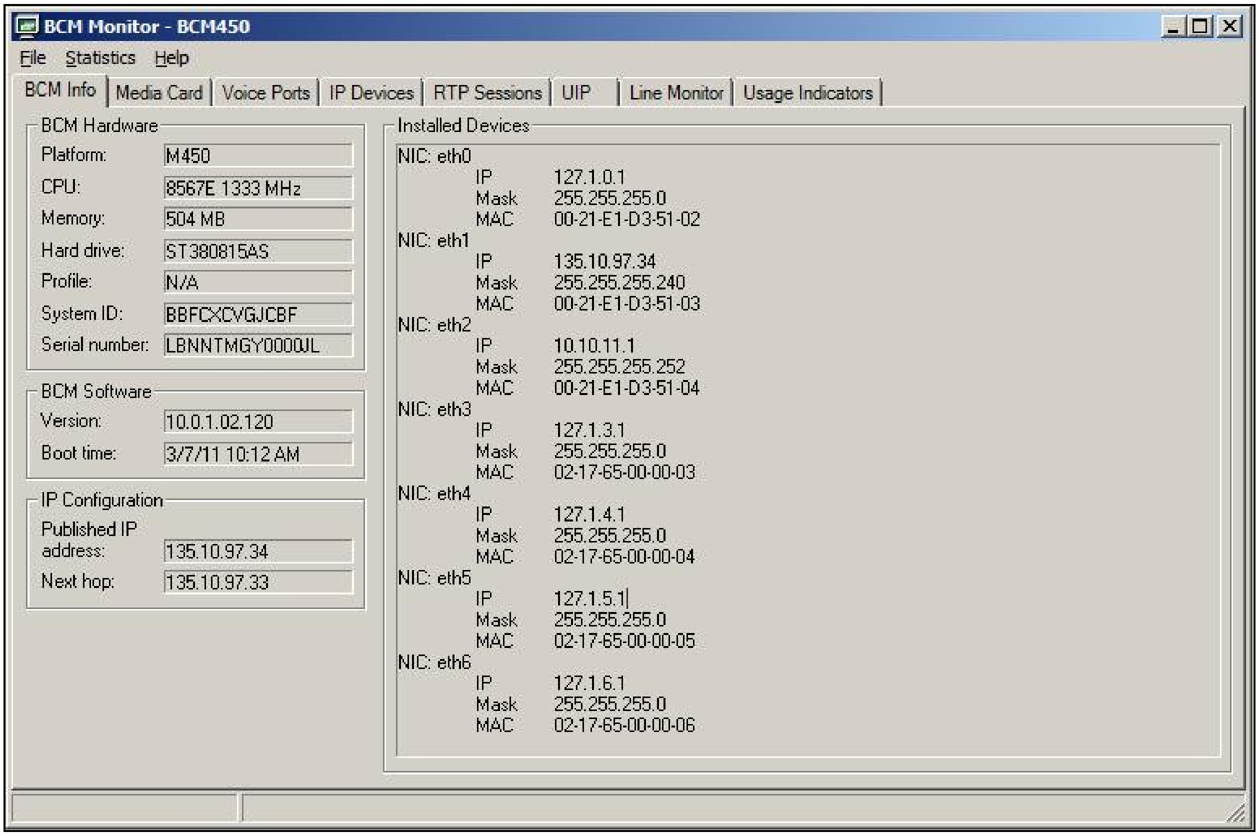

Figure 10: BCM450 Monitor GUI

5.2. Administer Resources

This section describes how to configure a SIP Trunk on BCM450 to CenturyLink system.

5.2.1. Administer Application Resource for SIP Trunks

These Application Notes assume that the basic configuration has already been administered. This section describes steps for configuring **Application Resource** for **SIP Trunks** on BCM450 to work with CenturyLink system. For further information on Avaya Business Communication Manager 450, please consult references in **Section 8**.

Select tab **Configuration** > **Resources** > **Application Resources.** Select **SIP Trunks** then click **Modify** button.

A new dialog displays with title "**Modify Resource Application Reservations**", input the configuration value as shown in **Figure 11**:

- Minimum: 0
- Maximum: MAX

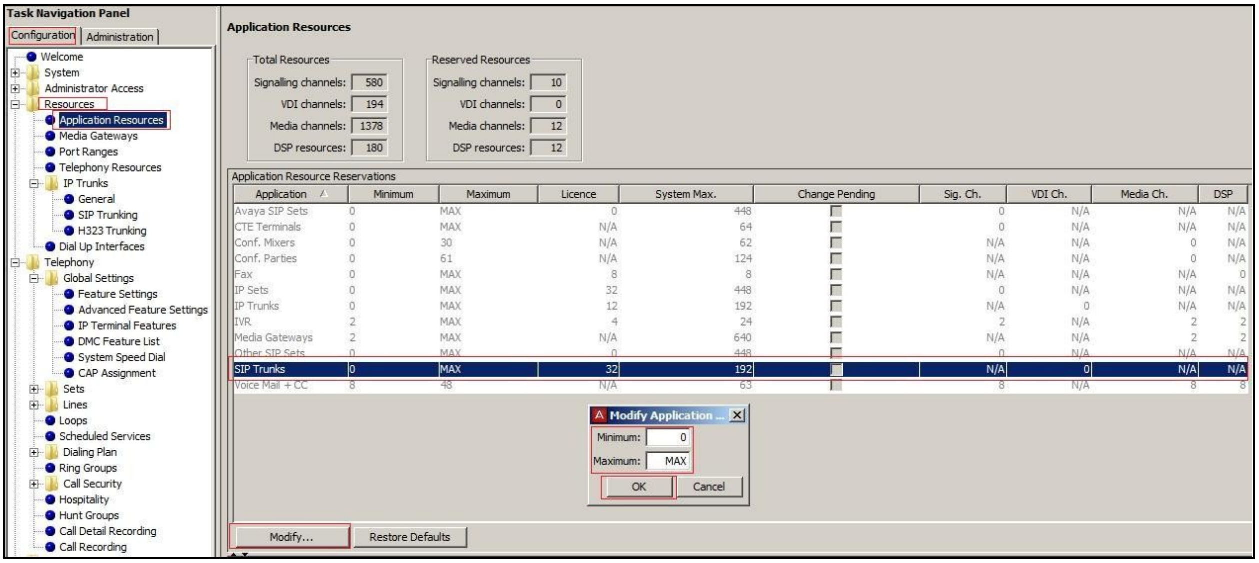

Figure 11 – Configuring Application Resources for SIP Trunks

5.2.2. Administer Media Gateway

Select tab **Configuration** > **Resources** > **Media Gateways.** Then select **Echo cancellation** enabled and as shown in red box in **Figure 12**.

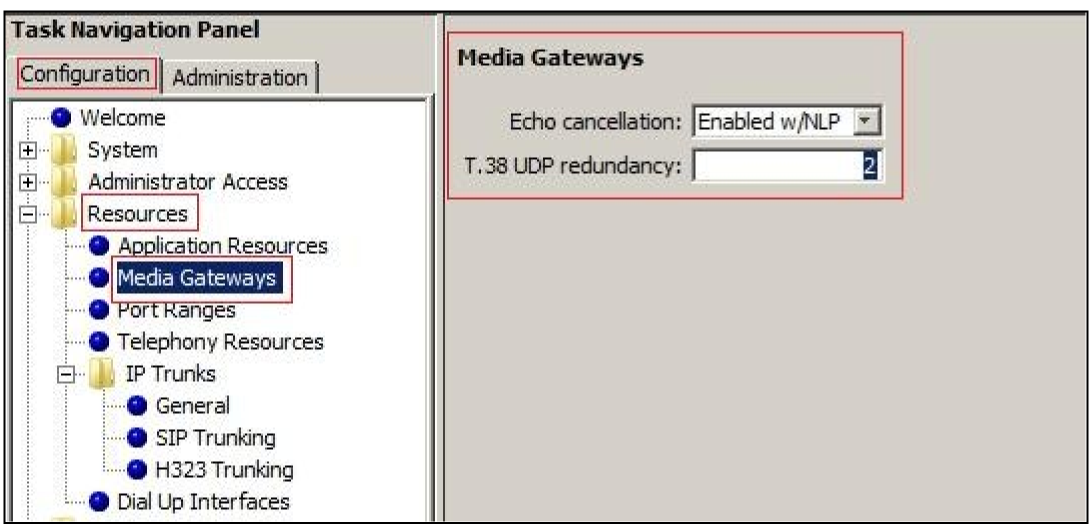

Figure 12 – Configuring Media Gateways

5.2.3. Administer Port Ranges

Select tab **Configuration** > **Resources** > **Port Ranges.** Then configure port ranges used for **RTP over UDP** traffic as show in red box in **Figure 13.**

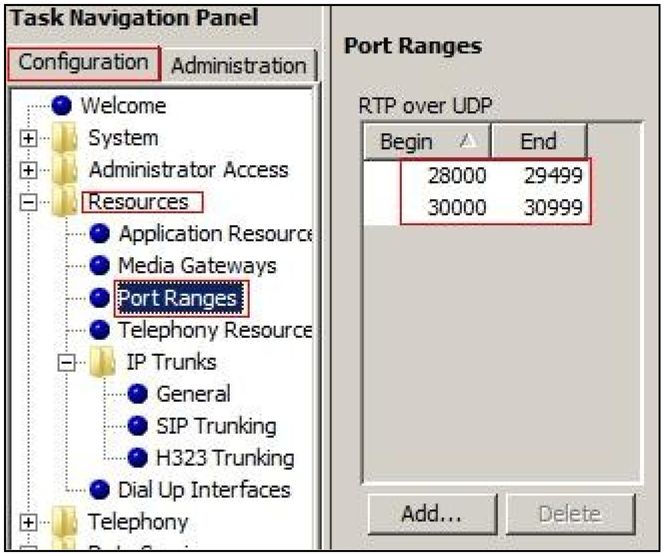

Figure 13 – Port Ranges for RTP over UDP

5.3. Administer SIP Trunk

This section describes the steps for configuring SIP Trunk between BCM450 and CenturyLink system.

5.3.1. General IP Trunk Settings

Select tab **Configuration** > **Resources** > **IP Trunks** > **General.** Select tab **IP Trunking Settings**; configure **Telephony Settings** as shown in **Figure 14**.

- **Forward redirected OLI**: First Redirect
- **Send name display**: checked (enabled)
- **Remote capacity MWI**: checked (enabled)
- **Ignore in-band DTMF in RTP**: unchecked (disabled)

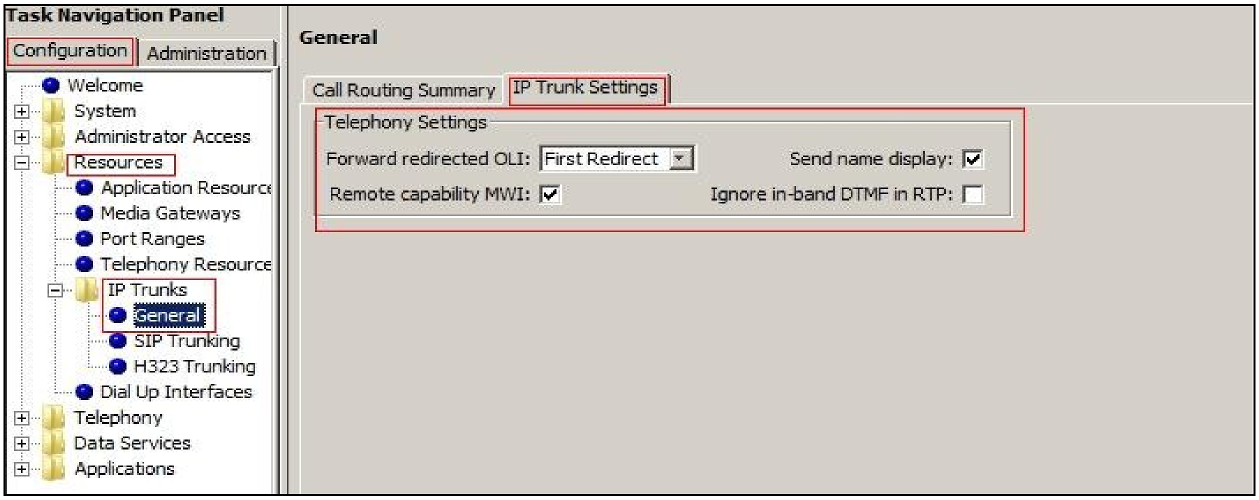

Figure 14 – IP Trunk Settings

5.3.2. Administer Global Settings

Select tab **Configuration** > **Resource s** > **IP Trunks** > **SIP Trunk.** Select tab **Global Settings**; **Figure 15** shows the detail configuration attributes.

- **SIP Settings** for **Local Domain**: bvwdev.com; **Call signaling port**: 5060
- **RTP Keepalives, Scope: None**
- **Telephony Settings, Fallback to circuit-switched:** Disabled
- **RFC2833 Dynamic payload: 101**

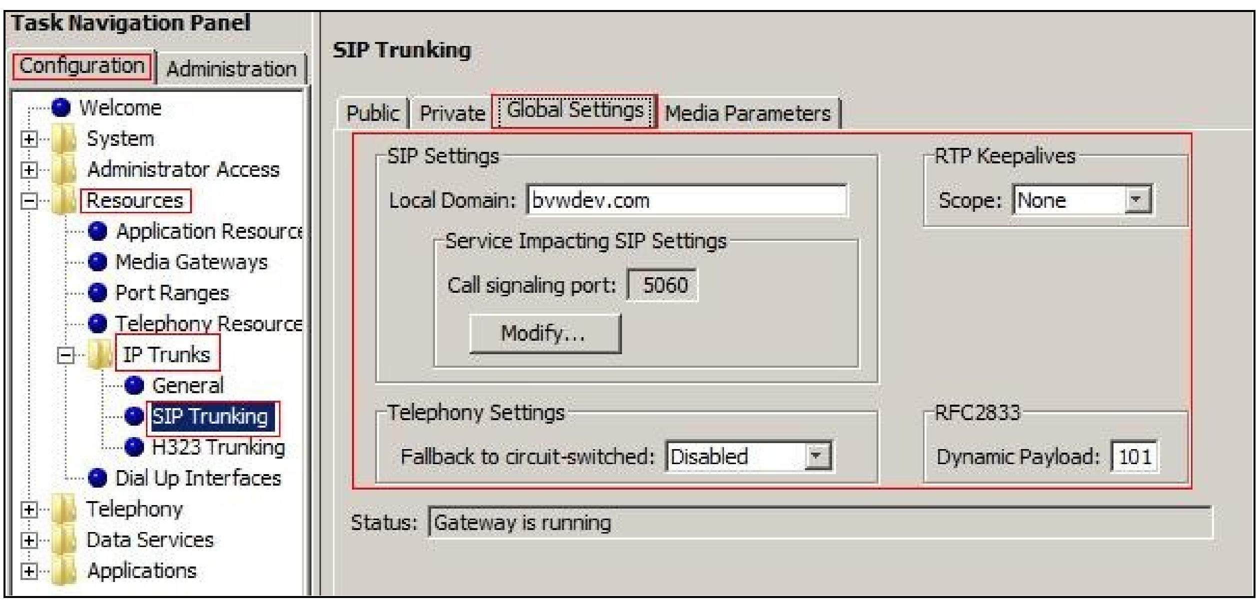

Figure 15 – Global Settings for SIP Trunk

5.3.3. Administer Public Port

Select tab **Configuration** > **Resources** > **IP Trunks** > **SIP Trunk.** Select tab **Public.** Click on tab **Settings** to set **Provisioned Public Port** to 5060 as shown in **Figure 16**.

Figure 16 – Provisioned Public Port Setting

5.3.4. Create a Public Account

Click on tab **Accounts**, and then click on **Add** button to create a public account for CenturyLink (**Figure 17**).

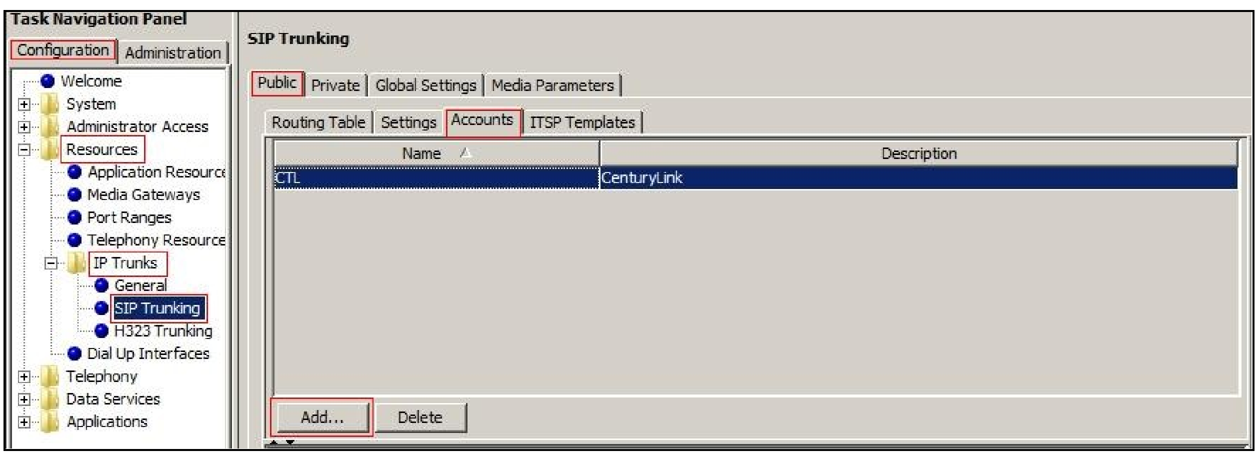

Figure 17 – Add a public account for SIP Trunk

The **Add Account** dialog displayed (not shown) to ask for the template; click on **No Template** and proceed to the next step.

The SIP Trunk to CenturyLink uses static IP endpoint and does not support registration. Thus, the detail configuration is shown in **Figure 18**.

- Template: None
- SIP domain: IP address of CenturyLink system
- Registration required: leave as blank
- SIP username: leave as blank
- Password: leave as blank

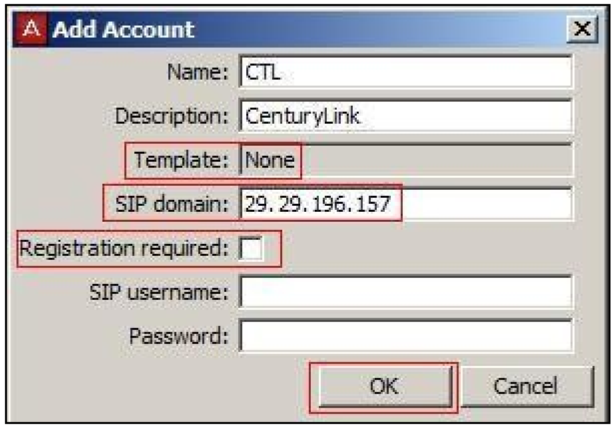

Figure 18 – Account Setting for CenturyLink SIP Trunk

5.3.5. Basic Settings

Select account CTL created in **Section 5.3.4.** Select **Basic** tab, the **Basic** settings are displayed as in **Figure 19**. Add an entry to **Outbound Proxy Table** associate to CenturyLink system, which is used by BCM450 to send OPTIONS to CenturyLink for keepalive purpose.

- **SIP Domain; Remote:** IP address of CenturyLink system
- **SIP Domain; Local:** IP address of BCM450
- **Proxy**: leave as blank (default)
- **Registrar**: leave as blank (default)
- **Outbound Proxy Table:**
  - **Domain**: IP address of CenturyLink system
  - **IP Address**: IP address of CenturyLink system
  - **Port**: 5060
  - **Load-balancing Weight**: 0
  - **Keep Alive: OPTIONS**

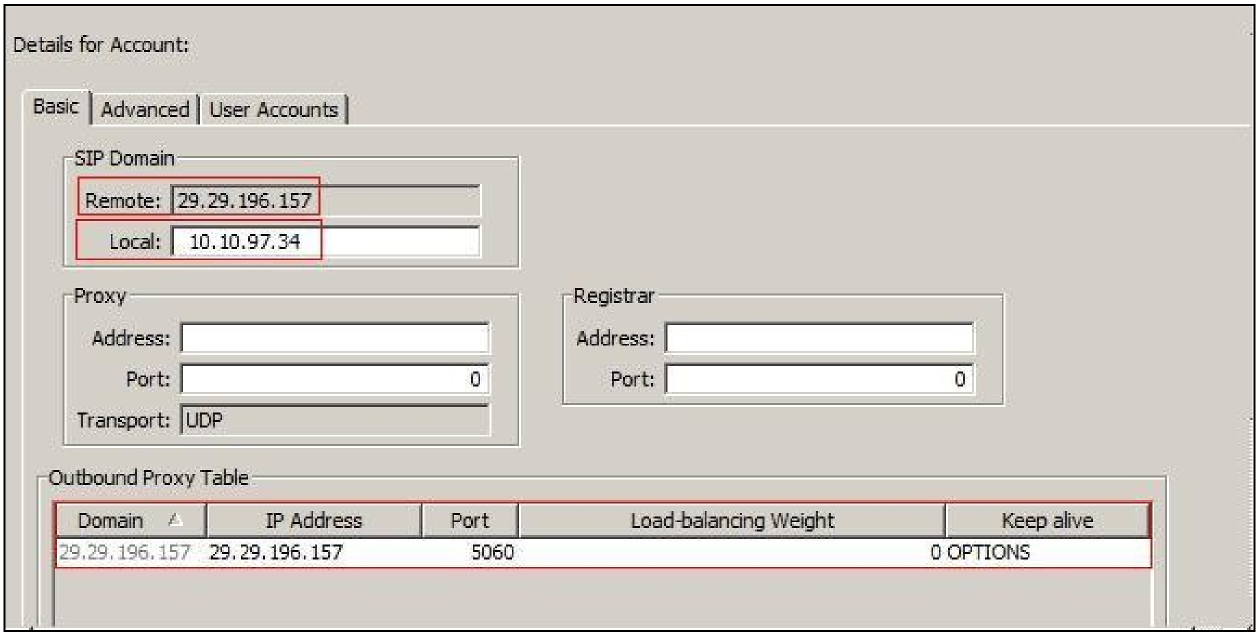

Figure 19 – Basic Configuration for Public SIP Trunk Account

5.3.6. Advance Settings

a) Select account CTL created in **Section 5.3.4**
b) Select **Advanced** tab, the **Advanced** settings are displayed as in **Figure 20**.

- **Enable media relay**: checked (enabled)
- **Support 100rel**: checked (enabled)
- **Allow UPDATE**: checked (enabled)
- **Use null IP to hold**: checked (enabled)
- **Allow REFER**: checked (enabled)
- **Support Replaces**: checked (enabled)
- **Enable Connected Identify**: checked (enabled)

Note: leave other fields as blank (default.)

NAT Pinhole Maintenance:

- **Signalling Method:** None

Session timer:

- **Session refresh method:** UPDATE
- **Refresher:** Local
- **Session-Expires:** 1800
- **Min-SE:** 90

Active call limit: 0

ITSP association method: From header domain match
Outbound Called characters to absorb: 0

Inbound Called prefix to prepend: leave as blank (default)
Authentication realm: leave as blank (default)

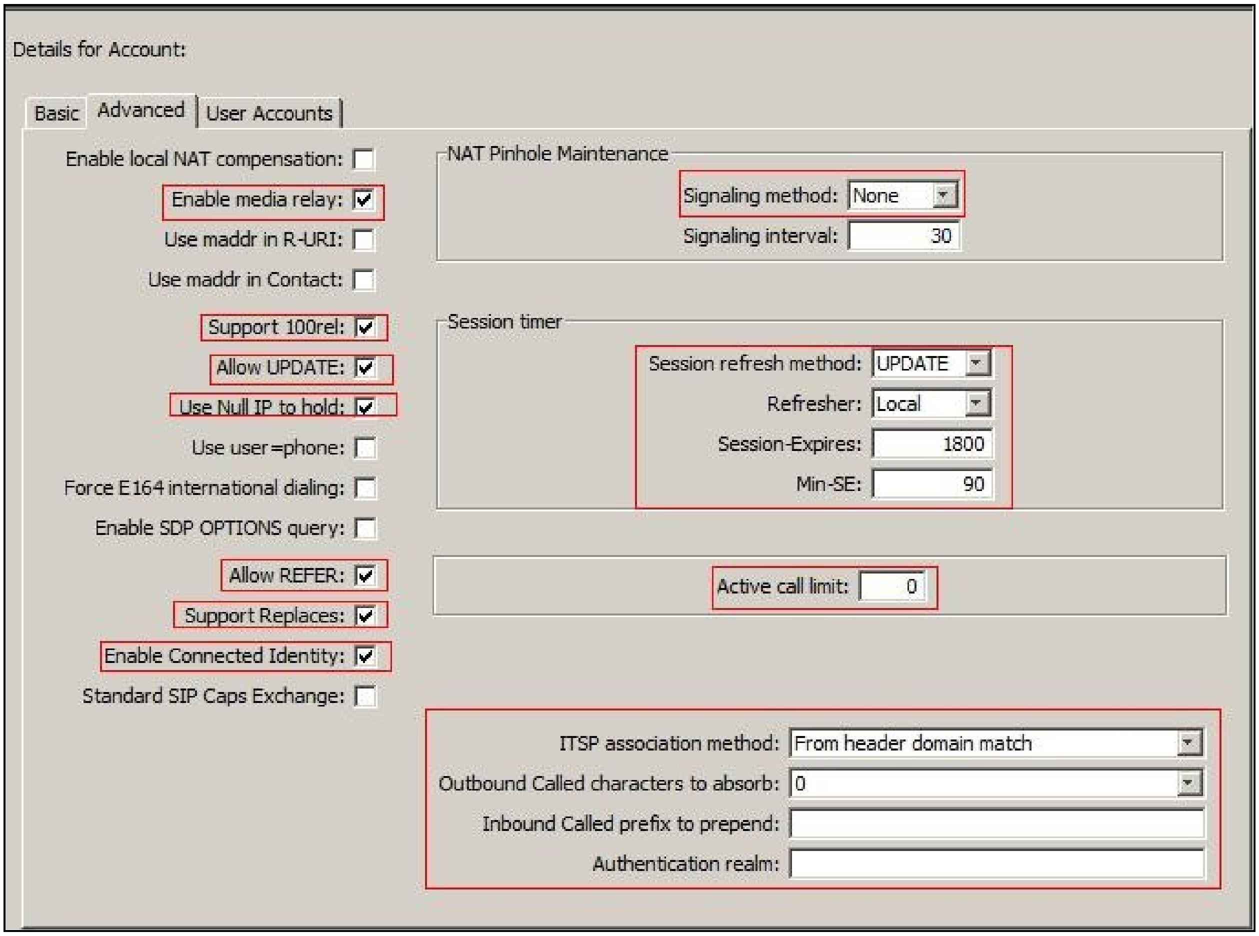

Figure 20 – Advanced Configuration for Public SIP Trunk Account

5.3.7. User Account Settings

a) Select account CTL created in **Section 5.3.4**

b) Select **User Account** tab, the **User Account** settings are displayed as in **Figure 21**. Click on **Modify** button.

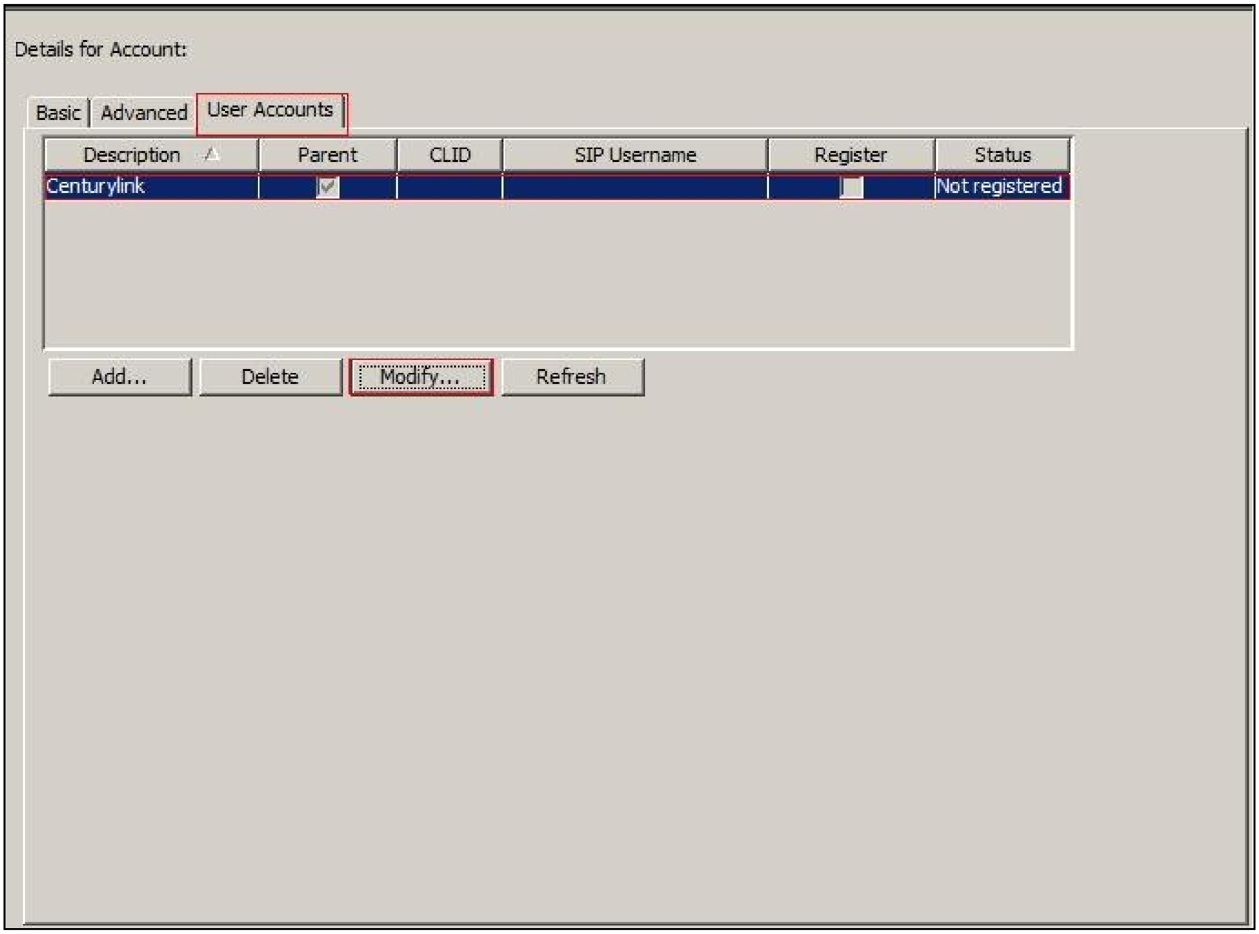

Figure 21 – Modify SIP Trunk User Account

c) **Modify account** dialog displays to show information of SIP Trunk account (as shown in **Figure 22**). This feature gives an option to manipulate SIP header before sending to Centurylink. In this testing, there is no SIP manipulation required, so leave all fields as blank.

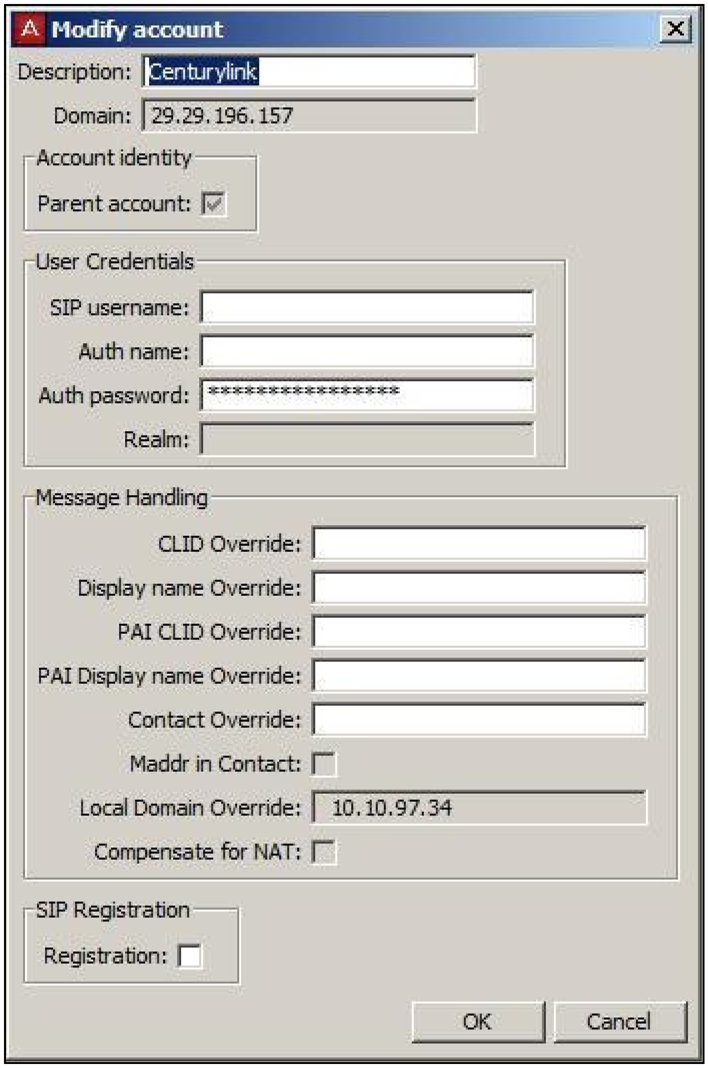

Figure 22 – Modify SIP Trunk User Account Details

5.4. Administer Codec Profile

5.4.1. Codec Settings for SIP Trunk

Select tab **Configuration > Resources > IP Trunks > SIP Trunk.** Select tab **Media Parameters**; the detail configuration attributes is in **Figure 23**.
CenturyLink does not support G.729, therefore only G.711u has been selected in **Seleted list**, with **Voice Activity Detection** disabled; **G.711 payload size (ms)** is set to **20ms**.

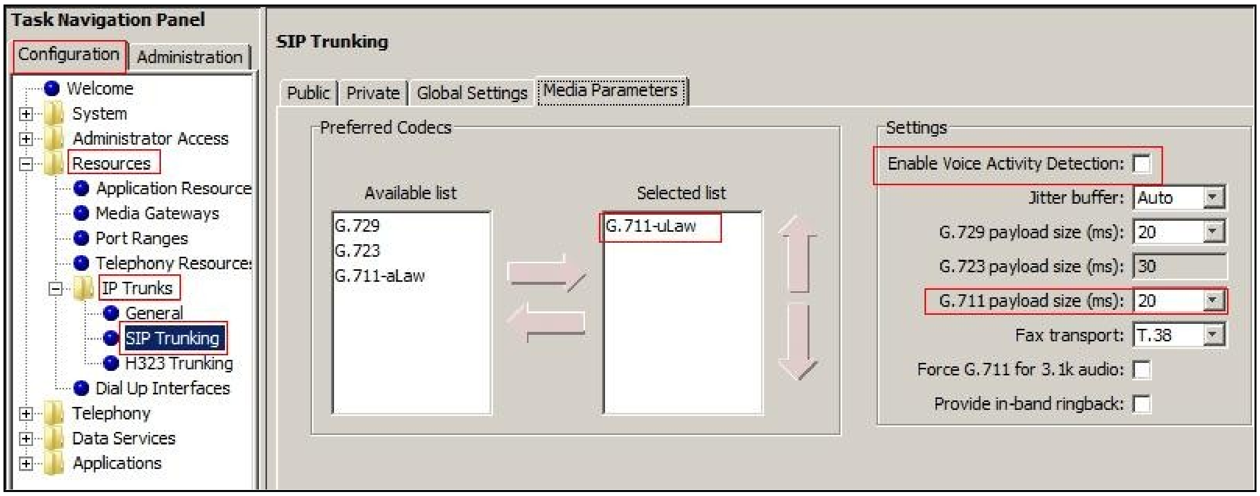

Figure 23 – Codec Settings for SIP Trunk

5.4.2. Codec Settings for IP Sets

Select tab **Configuration** > **Resources** > **Telephony Resources.** Select line **Internal / IP Sets**; and then configure supported codec for IP Sets as shown in **Figure 24**.

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

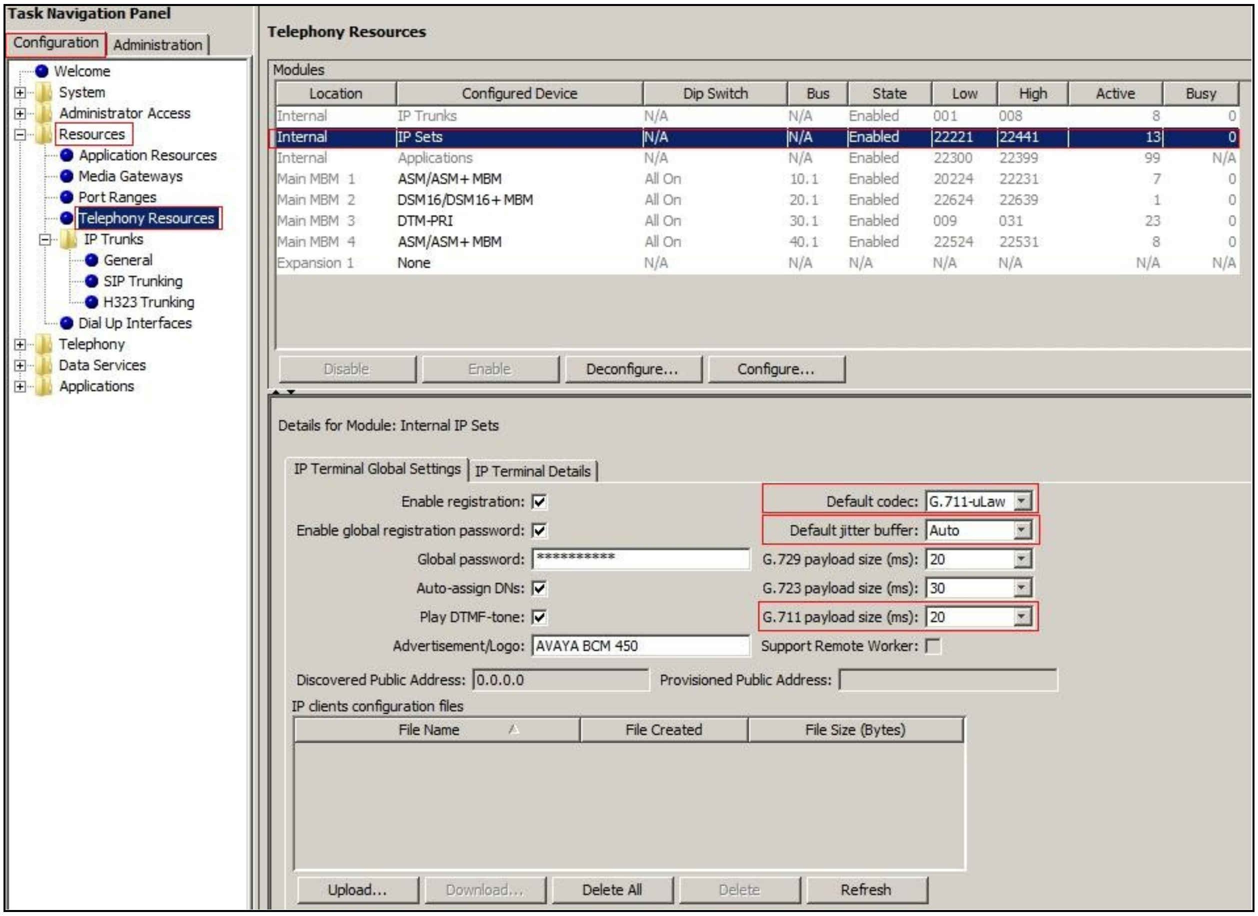

Figure 24 – Codec setting for IP Sets

5.5. Administer Dialing Plan

This section shows how to configure BCM450 VoIP lines to be used for outbound calls over the SIP Trunk via CenturyLink system. The public dialing plan will be provisioned with appropriate parameters as detail as below. BCM450 uses target lines to either terminate or redirect the inbound SIP calls.

5.5.1. Associate a Line Pool to VoIP Lines

Select tab **Configuration** > **Telephony** > **Lines** > **Active VoIP Lines.** Assigned a pool name where the **VoIP Lines** are associated with. In this case it is **Pool: BlocA** (**Figure 25**).

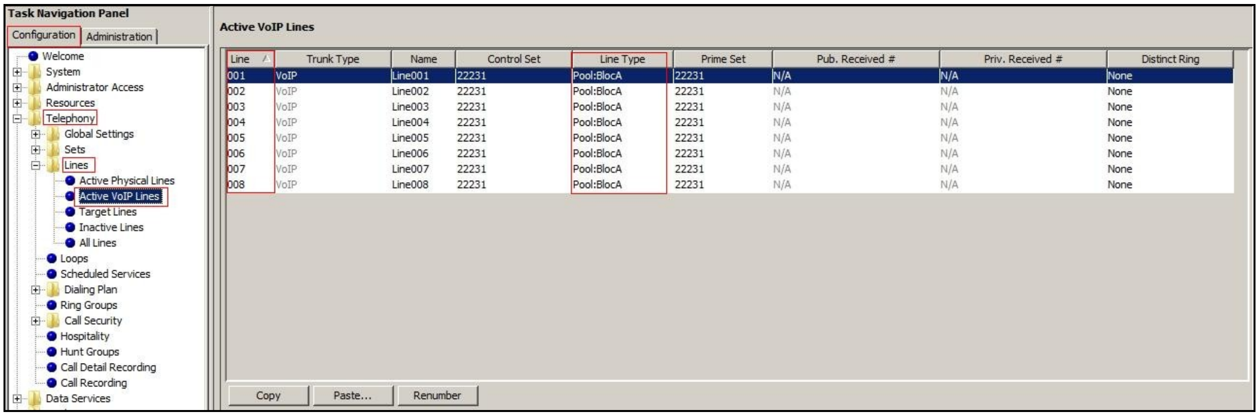

Figure 25 – Line Pool Assignment for VoIP Lines

5.5.2. Administer DN Length

This section shows how to configure intercom DN length for BCM450 phone. In this testing, DN length is set to 5.

Select tab **Configuration** > **Telephony** > **Dialing Plan** > **General.** Configure **DN length (intercom)** and **Dialing timeout** as shown in **Figure 26**.

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

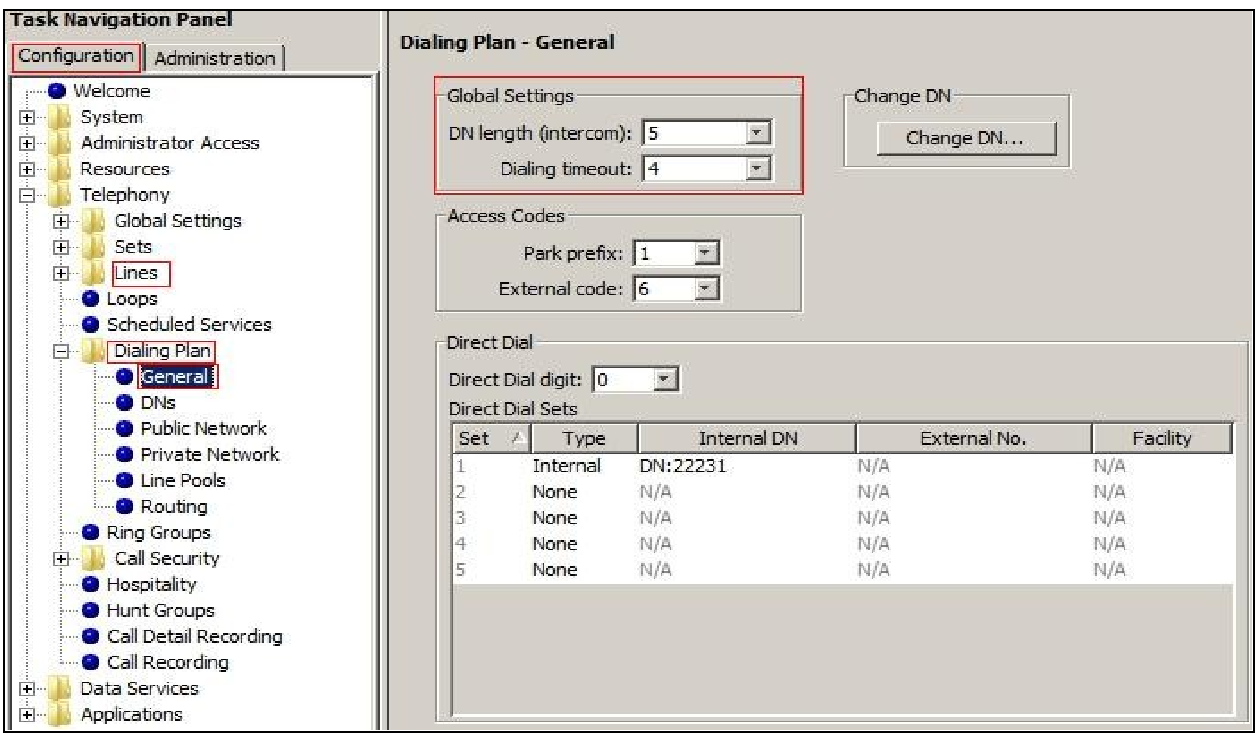

Figure 26 – Define DN length (intercom)

5.5.3. Administer Public Network

Select tab **Configuration** > **Telephony** > **Dialing Plan** > **Public Network.** Configure **Public Receive number length**, **Public network dialing plan** and **Public network code** as shown in **Figure 27**.

- **Public Receive number length**: 10
- **Public network dialing plan**: Public (unknown)
- **Public network code**: leave as blank

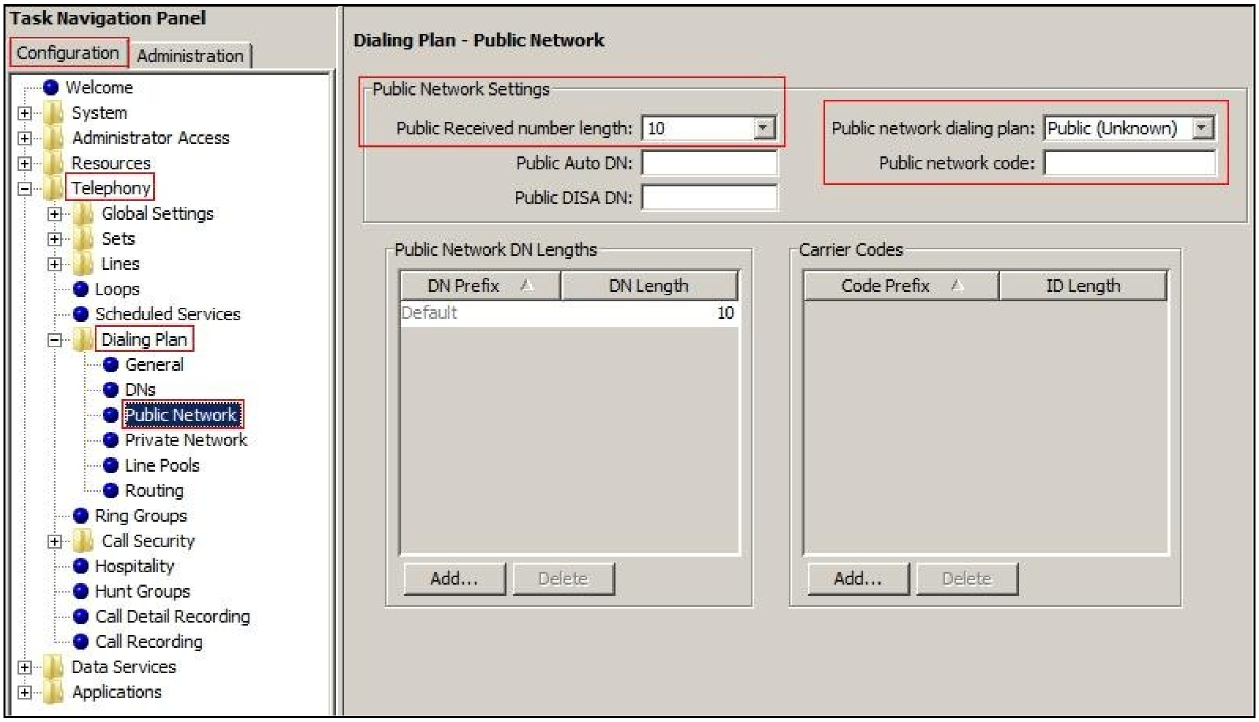

Figure 27 – Public Network Settings

Note: **Public network code** will affect the CLID of an outbound call. If specified, it will combine with Public OLI setting for the phone to form the CLID. In this testing, CLID is set on each phone individually, therefore it is unnecessary to define **Public network code**.

5.5.4. Administer Routing

Select tab **Configuration** > **Telephony** > **Dialing Plan** > **Routing.** On tab **Routes**, click **Add** to create a new route. **Figure 28** shows route 3 was created.

Route 3:

- **Use Pool: BlocA**
- **DN Type: National**

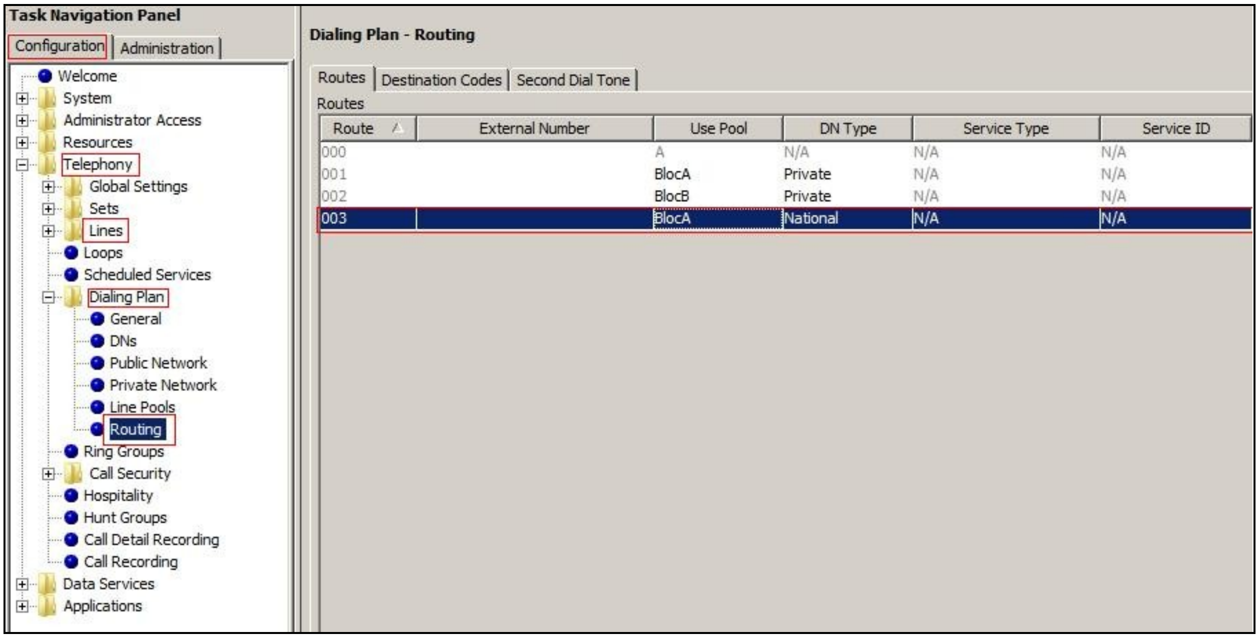

Figure 28 – Create a route

5.5.5. Administer Outbound Call - Destination Codes

Destination Codes define the prefix for outbound call. This testing uses internal access code 9 to access to the SIP trunk. The code will be trimmed out before sending to CenturyLink. In order to exercise different dialing plans over the SIP Trunk, multiple **Destination Codes** were added.

Select tab **Configuration** > **Telephony** > **Dialing Plan** > **Routing.** On tab **Destination Codes**, click **Add** to create a new route. **Figure 29** shows following **Destination Codes** were created.

Destination Codes: 90.

- Purpose: BCM450 will use these codes:
    - To reach CenturyLink's operator by dialing 0
    - To reach operator by dialing 0+10 digits
    - To make international call by dialing 011+CountryCode+AreaCode+DN.
- Configuration attributes:
    - Normal Route: 003 (created in **Section 5.5.4**)
    - Absorbed Length: 1 (digit 9 will be deleted)

Destination Codes: 9411.

- Purpose: BCM450 will use this code to reach 411 services.
- Configuration attributes:
    - Normal Route: 003 (created in **Section 5.5.4**)
    - Absorbed Length: 1 (digit 9 will be deleted)

Destination Codes: 9911.

- Purpose: BCM450 will use this code to reach 911 services.
- Configuration attributes:

- Normal Route: 003 (created in **Section 5.5.4**)
- Absorbed Length: 1 (digit 9 will be deleted)

Destination Codes: 9613.

- Purpose: BCM450 will use this code to make NPA call by dialing 10 digits (Area Code 613 + DN).
- Configuration attributes:
    - Normal Route: 003 (created in **Section 5.5.4**)
    - Absorbed Length: 1 (digit 9 will be deleted)

Destination Codes: 91613.

- Purpose: BCM450 will use this code to make North America long distance call by dialing 1+10 digits (1+Area Code 613 + DN).
- Configuration attributes:
    - Normal Route: 003 (created in **Section 5.5.4**)
    - Absorbed Length: 1 (digit 9 will be deleted)

Destination Codes: 91800; 91866

- Purpose: BCM450 will use these codes to make toll free call with prefix 1800, 1866.
- Configuration:
    - Normal Route: 003 (created in **Section 5.5.4**)
    - Absorbed Length: 1 (digit 9 will be deleted)

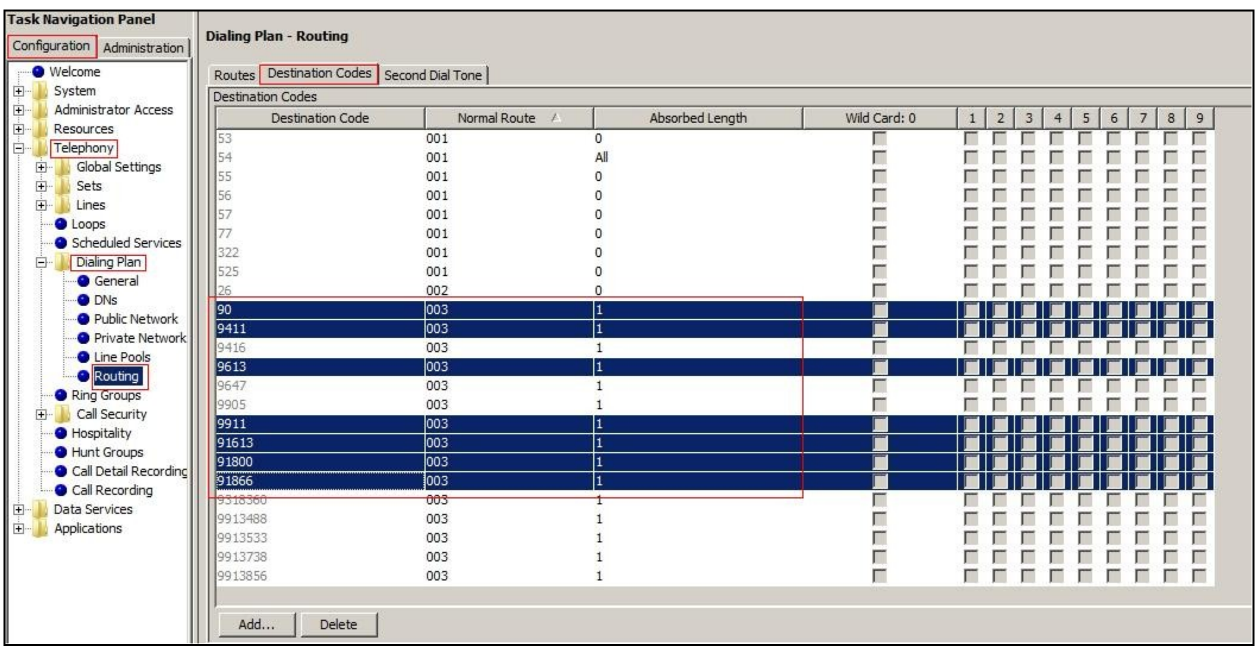

Figure 29 – Administer Destination Codes

5.5.6. Administer Outbound Call - SIP Trunk Routing Table

The Destination Codes are associated with a VoIP pool. BCM450 needs to specify which signaling protocols it going to use, SIP or H323. This section shows how to configure a SIP route on BCM450.

Select tab **Configuration** > **Resources** > **IP Trunk** > **SIP Trunking.** On tab **Public**, select tab **Routing Table** and then click **Add** to create a new route. The new route is configured and appropriated with the **Destination Codes** defined in **Section 5.5.5,** and attached to SIP Trunk public account CTL defined in **Section 5.3.4. Figure 30** shows routes **0**, **1**, **411**, **911**, and **613** were created.

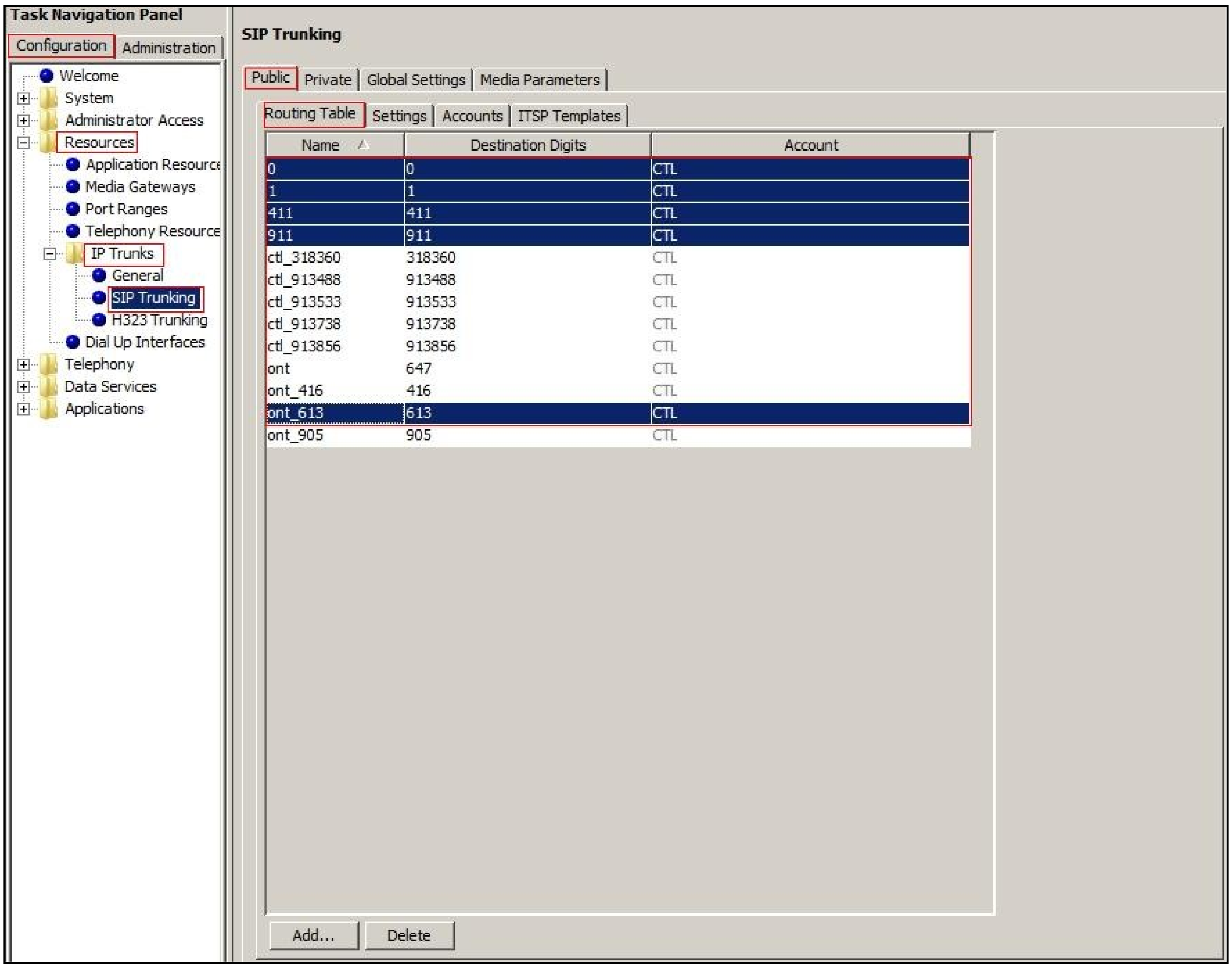

Figure 30 – Administer SIP Trunk Routing Table

5.5.7. Administer Inbound Call - Target Line

BCM450 uses a virtual target line to receive VoIP inbound calls. The maximum target line available is defined by the license. In this section, the target line is configured to terminate the call to a specific DID number. The target line is assigned to a key on the phone set.

a) Select **Configuration > Telephony > Sets > Active Sets.**
b) On tab **Line Access**, chose a DN e.g. 22264.
c) On tab **Line Assignment**, click **Add**. Then in **Add Line Assignment** dialog (not shown), input the target line number. **Figure 31** shows target line 998 was being added for DN 22264.

Target Line: 998
Appearance Type: Appr&Ring
Appearance: 1
Caller IP Set: checked (enable CLID delivery)
Vmsg Set: checked if want to register voice mail service for the set, uncheck if voicemail is not being registered.
Priv. Received #: input the private number assigned to the DN. BCM450 will ring the phone if receiving the private call to this number.
Pub. Received #: input the public DID number assigned to the DN. BCM450 will ring the phone if receiving the public call with this number. Target line will use this input to terminate a SIP call from CenturyLink. In this case, a target line is configured to terminate a SIP call from CenturyLink to DID number 9134400061.

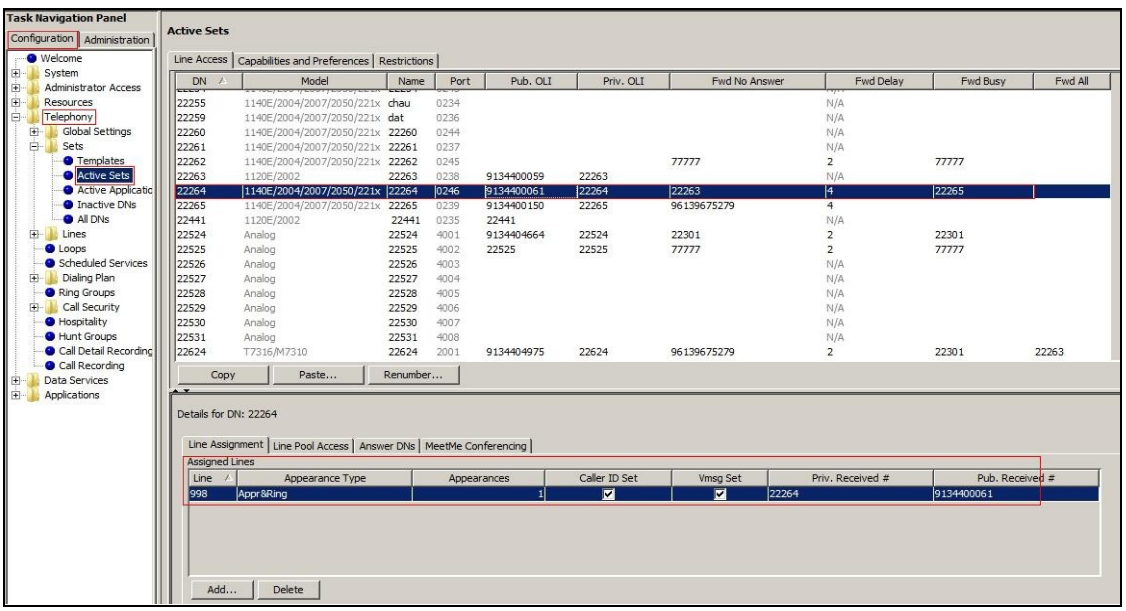

Figure 31 – Administer Target Lines

5.6. Administer Outbound CLID Delivery

5.6.1. Administer Outbound CLID-Name Delivery

This section shows how to configure CLID-Name delivery for BCM450. When DN 22264 makes an outbound call, the display name in **From** header will be constructed using the **Business Name** (created in step a as below) and set **Name** (created in step b as below).

a) Select **Configuration > Telephony > Global Settings > Feature Settings.** Then define **Business Names** Entry 1 as "BCM450" as shown in **Figure 32**.

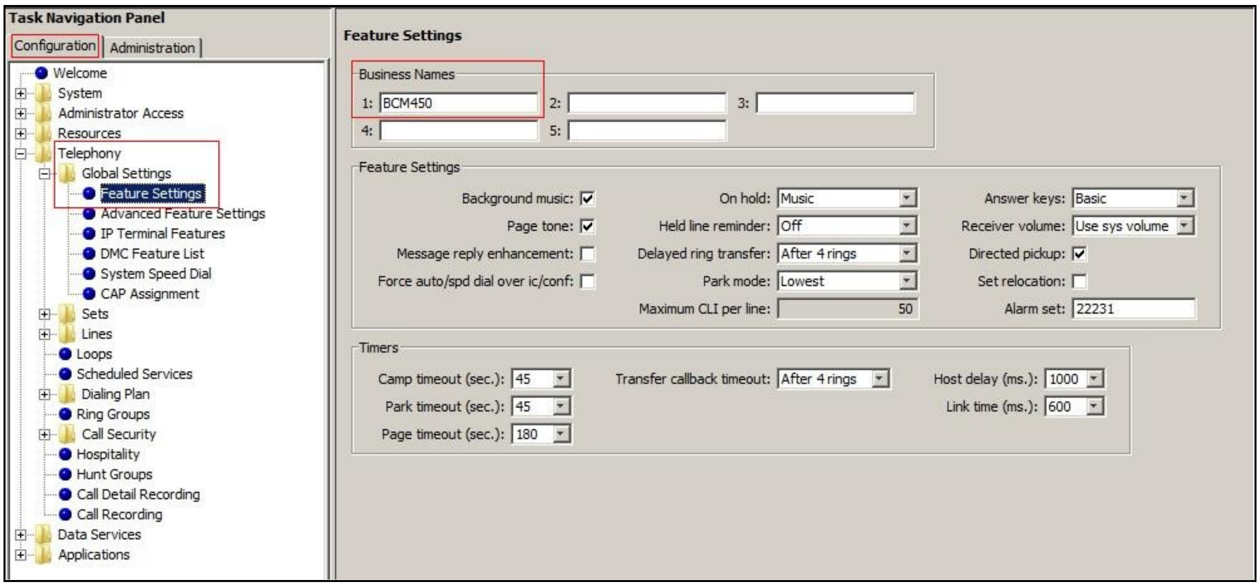

Figure 32 – Administer Business Name Entry 1 for BCM450

b) Then select **Configuration > Telephony > Sets > Active Sets.** On **Line Access** tab, the defined **Name** for DN 22264 is "22264" as shown in **Figure 33**.

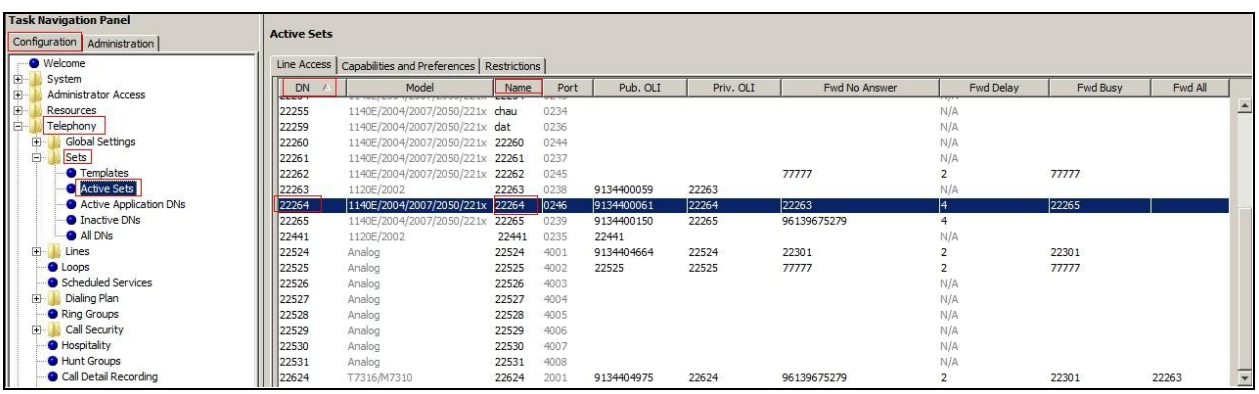

Figure 33 – Define a Name for DN 22264

c) Continue to select **Configuration > Telephony > Sets > Active Sets.** On **Capabilities and Preferences** tab, check **Auto Called ID** to enable CLID display before the call is being answered. Then click the **Preferences** tab; select **Business name 1** and uncheck **Send long name. Figure 34** illustrates the configuration in detail.

Business name: Business name 1
Send long name: unchecked

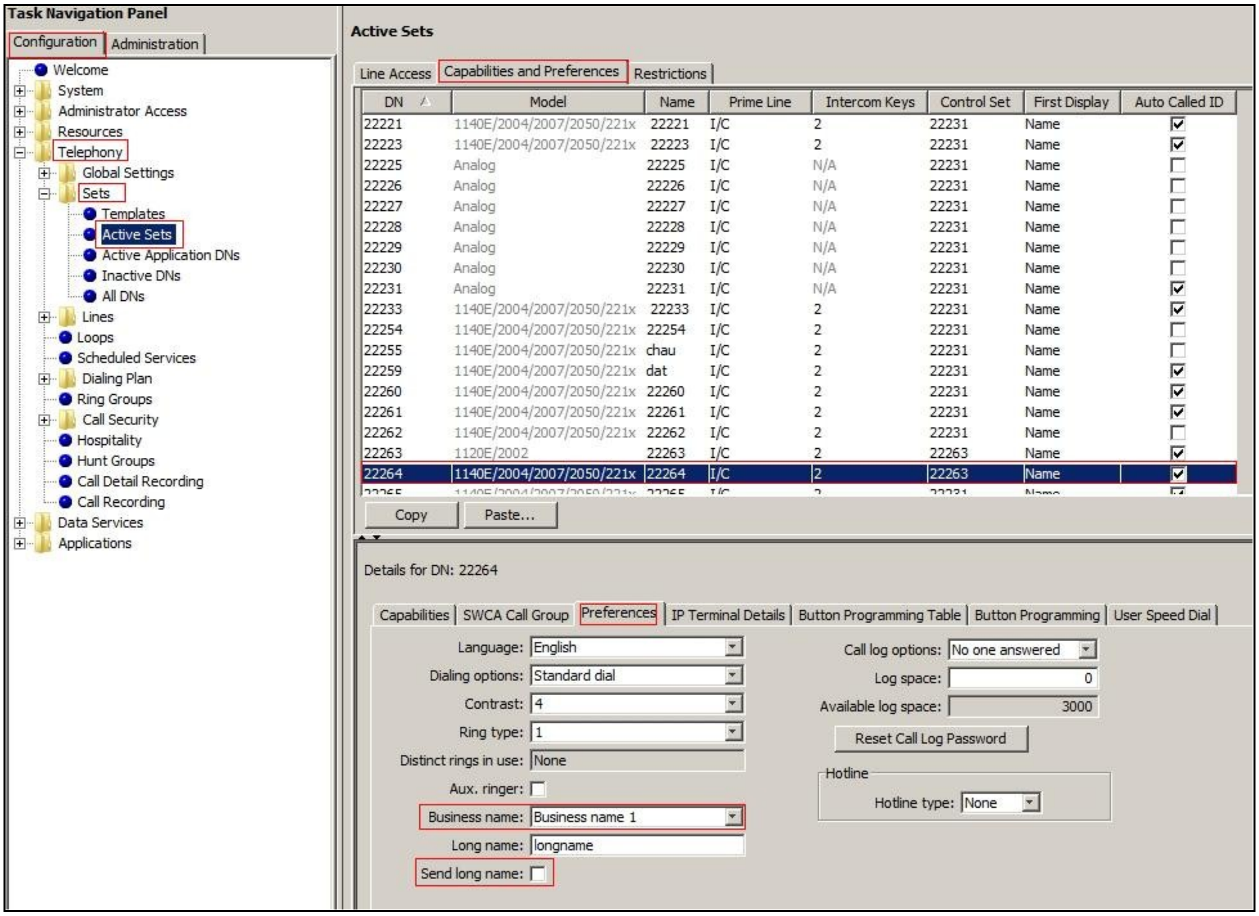

Figure 34 – Define CLID Name for DN 22264

5.6.2. Administer Outbound CLID-Number Delivery

This section shows how to configure CLID-Number delivery for BCM450. When DN 22264 makes an outbound call, the display number in the **From** header will be constructed using Pub. OLI.

Select **Configuration > Telephony > Sets > Active Sets.** On **Line Access** tab, define **Pub. OLI** for DN 22264 as “9134400061” as shown in **Figure 35**.

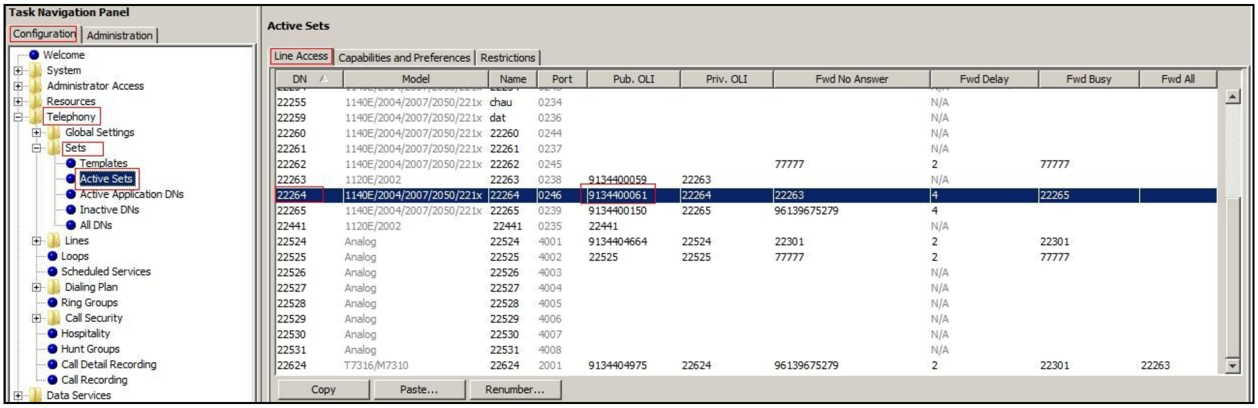

Figure 35 – Define a Pub. OLI for DN 22264

5.7. Administer Phone Sets

5.7.1. Configure Target Line to send busy tone

This section shows how to configure DN 22264 to return **Busy tone** if target line 998 was seized.

Select **Configuration > Telephony > Lines > Target Lines.** In **Section 5.5.7**, target line 998 was assigned to DN 22264, so scroll down to select Line 998. On **Preferences** tab, configure **If Busy** option to **Busy tone** as shown in **Figure 36**.

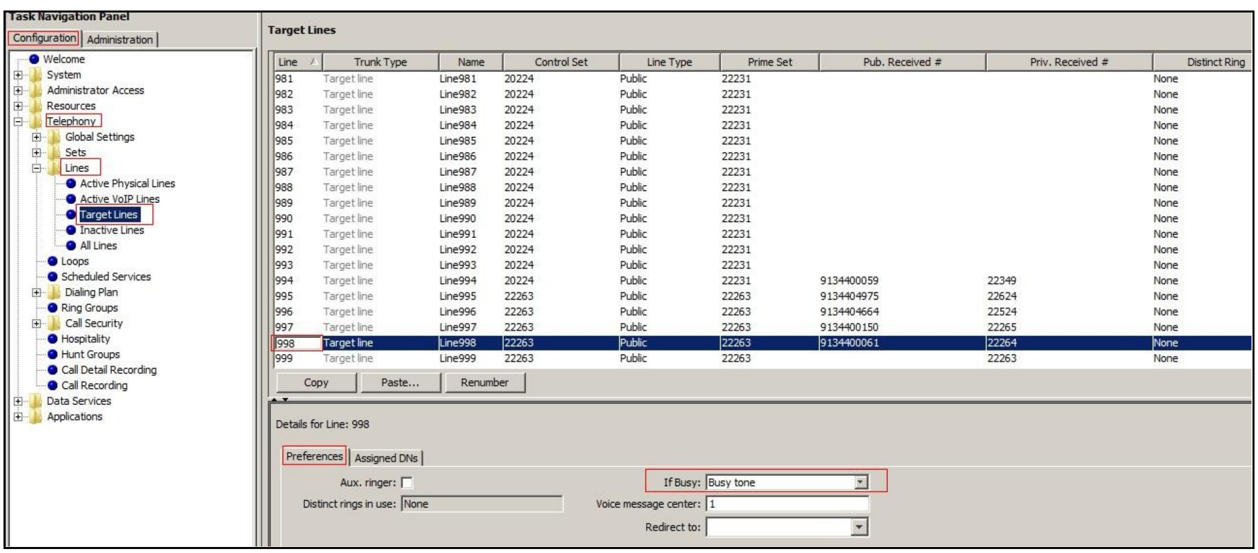

Figure 36 – Configure Target Line to send busy tone

5.7.2. Configure Target Line to ring Prime Set if busy

In this example, DN 22264 was set with a **Prime Set** of DN 22263.

To configure, select **Configuration > Telephony > Lines > Target Lines.** Then scroll down to select Line 998. Define **Prime Set** with DN 22263 and select option **If Busy: To Prime** as shown in the in **Figure 37**.

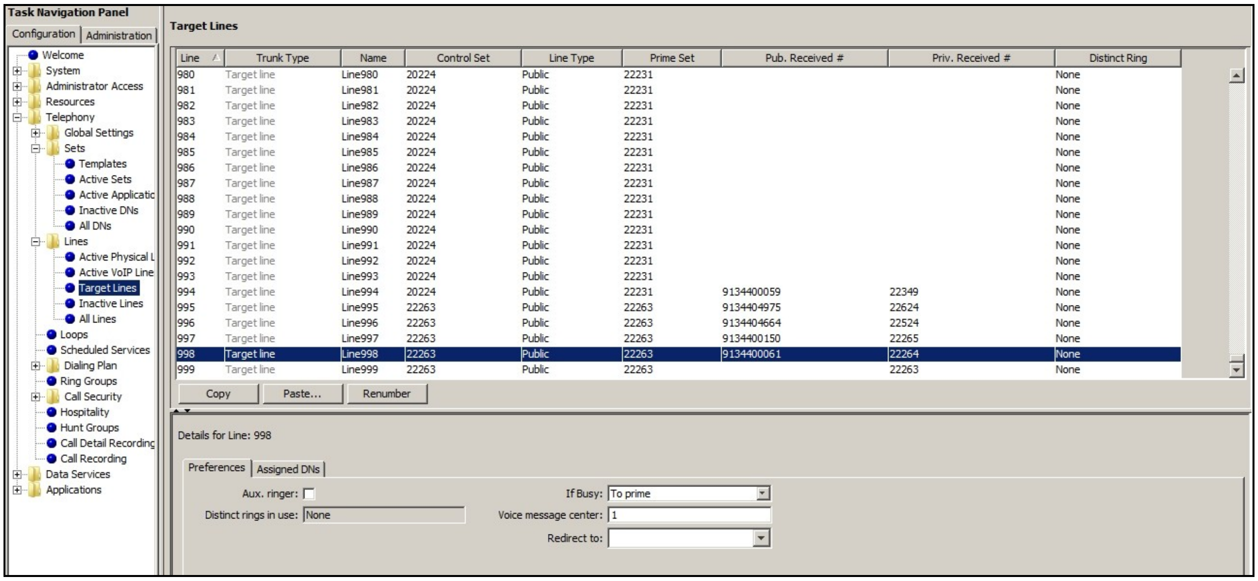

Figure 37 – Configure Target Line to ring Prime Set if busy

5.7.3. Configure Target Line for MeetMe Conferencing hosted on BCM450.

This section shows the configuration of BCM450 Target Line #10 to receive an incoming call to DID 9134400059 which is assigned to the MeetMe Conferencing service.

a) To enable the MeetMe conference, select **Configuration** > **Telephony** > **Sets** > **Active Sets**. On **Line Access** tab, select a DN e.g. 22264, then click on tab **MeetMe Conferencing**. Then click **Create MeetMe Conference Bridge...** buton (not shown). **Figure 38** shows a **MeetMe Conference Bridge** has been enabled for DN 22264.

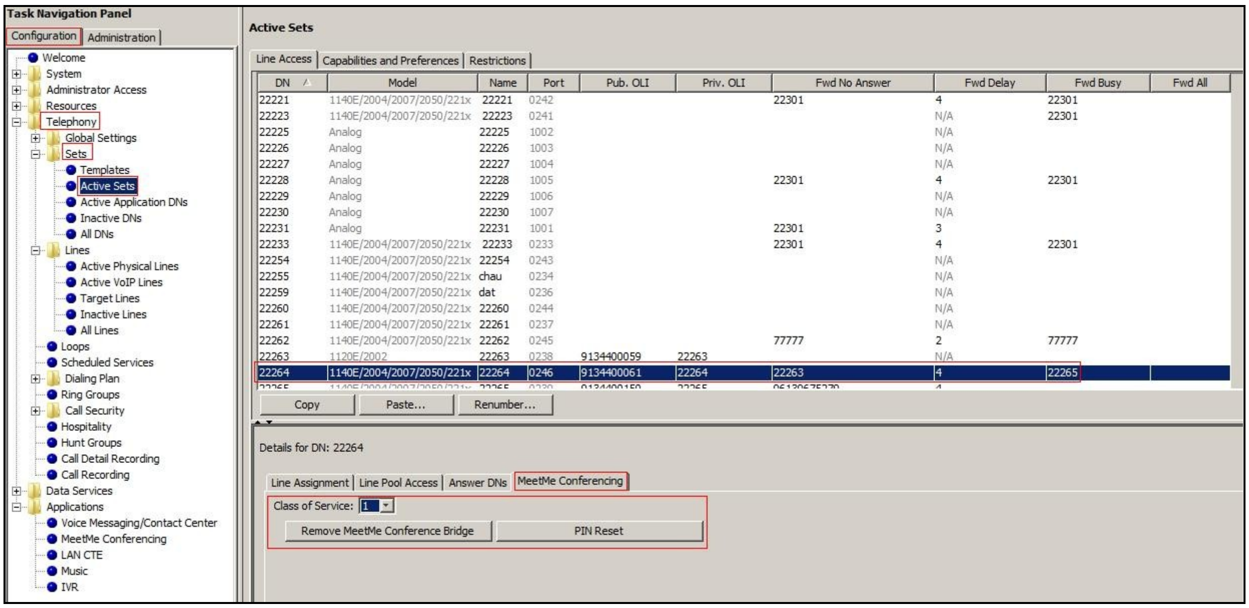

Figure 38 – Enable MeetMe Conference Bridge for a set

b) To custom change the **Access Code** for **MeetMe Conference**, select **Configuration > Applications > MeetMe Conferencing**. **Conference Bridges** tab is shown in **Figure 39**. By default, the **Access Code** is as same as DN of the set. However, it can be different.

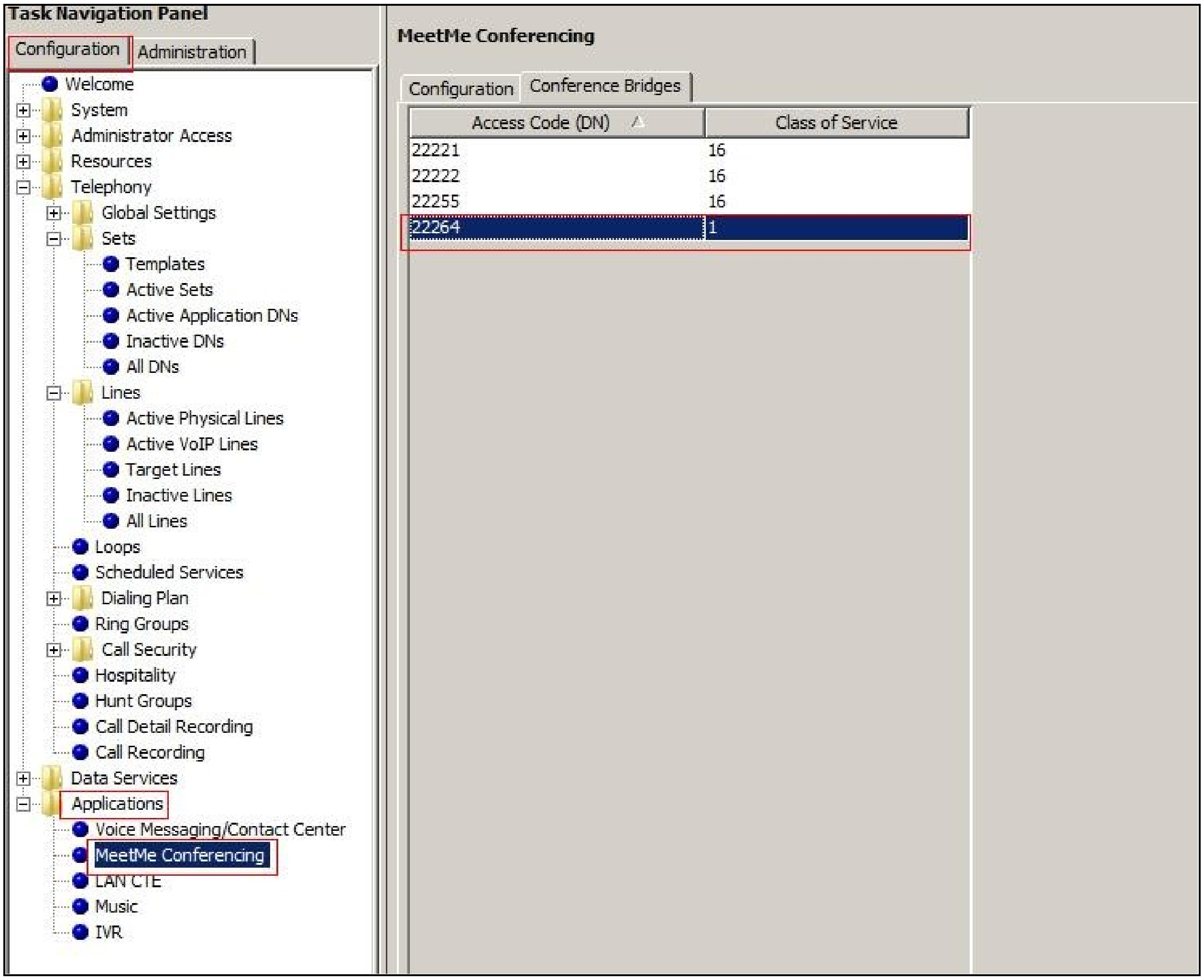

Figure 39 – Change the Access Code for a MeetingMe Conferencing Bridge.

c) Please refer to **Section 5.5.7** to configure **Target Line #10** to receive incoming call to **DID 9134400059**.
d) Launch **CallPilot Manager** by select **Configuration > Applications > Voice Messaging/ Contact Center**. Click **Launch CallPilot Manager** button as shown in **Figure 40**.

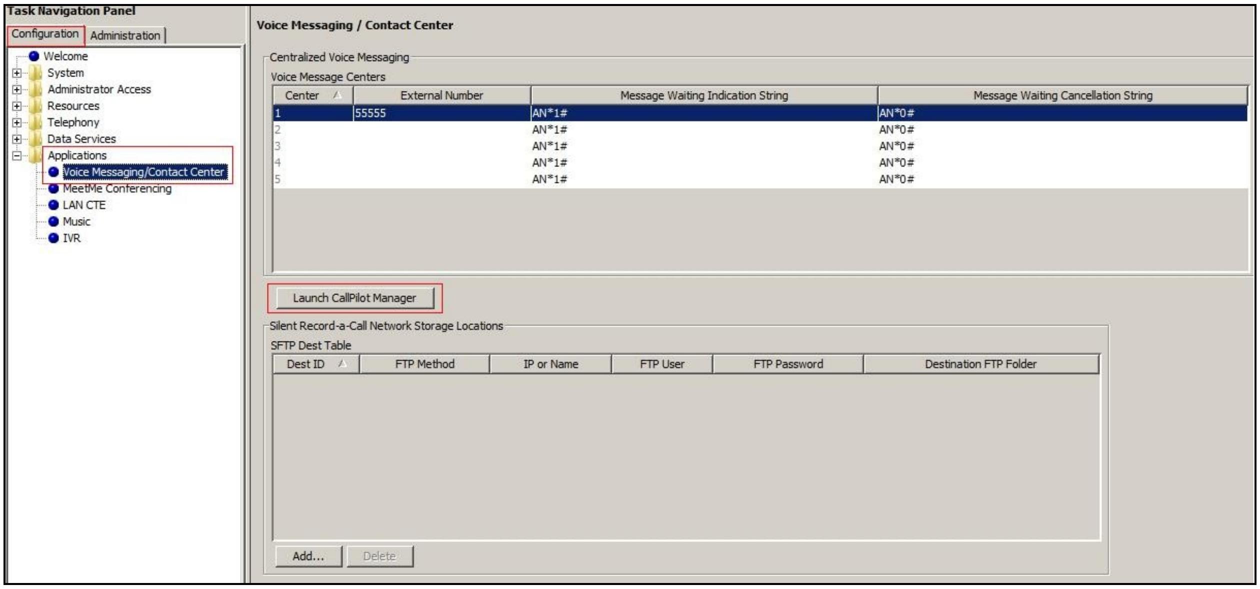

Figure 40 – Launch CallPilot Manager

e) The web browser displays **CallPilot Manager**. Then select **Auto-Attendent > Lines Administration** as shown in **Figure 41**. Scroll down to line 10 than click on link **Change**.

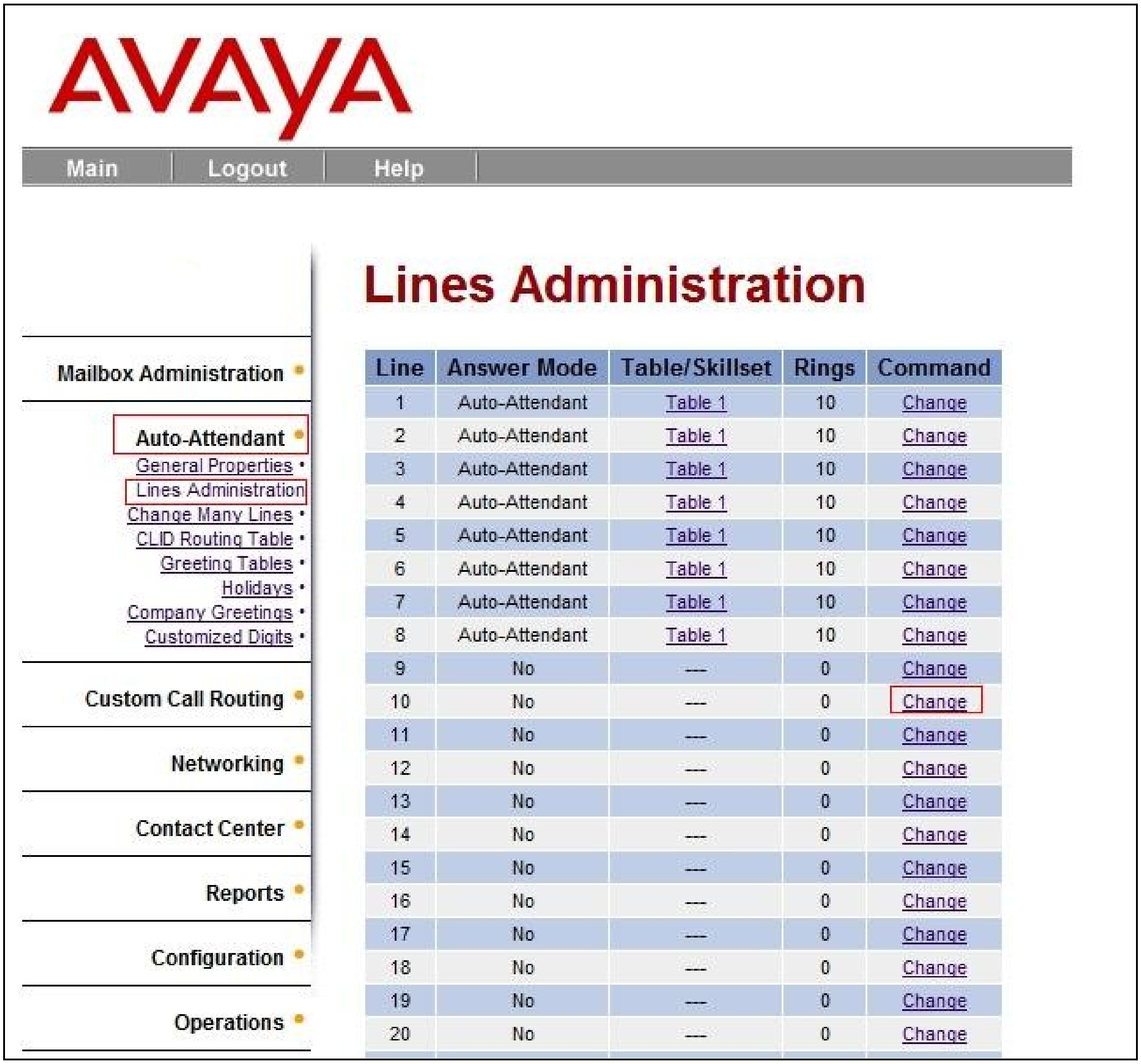

Figure 41: Line Administration

f) On the **Line Properties** page, select **Answer Mode** as **MeetMe Conferencing** and **Number of Rings** as 4, and then click **Submit** (**Figure 42**).

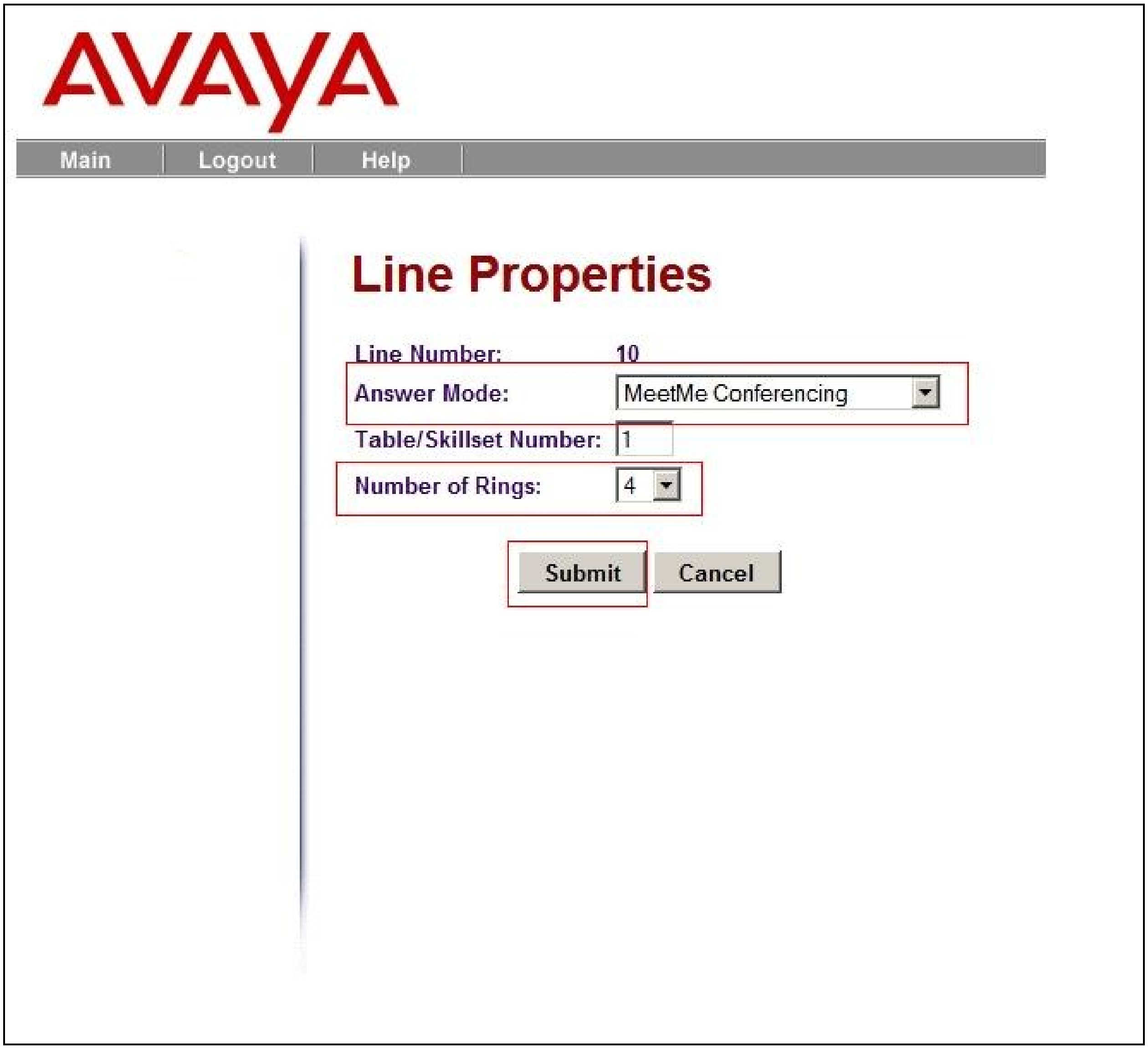

Figure 41: Line Properties

5.7.4. Call Redirection setting

Select **Configuration > Telephony > Sets > Active Set.** Then select **Capabilities and Preferences** tab. Click on a DN e.g 22264 to modify. On sub-tab **Capabilities**, check **Allow redirect** and **Redirect ring** as shown on **Figure 42**.

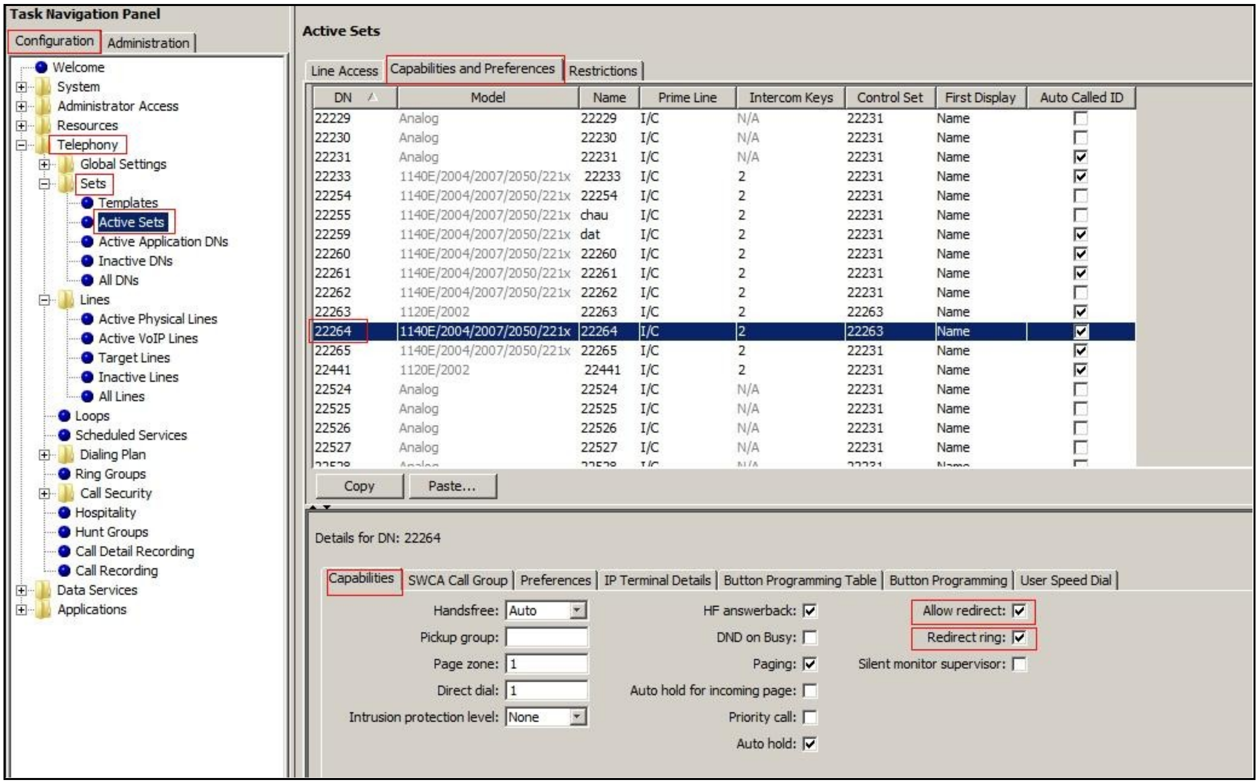

Figure 42: Call Redirection settings

5.7.5. Call Forward settings

Select **Configuration > Telephony > Sets > Active Set.** Then select **Line Access** tab. **Figure 43** shows the configuration for DN 22264 to forward the call after 4 rings (Fwd Delay), forward no answer to 6139675001, forward busy to 6139675002 and forward all call to 6139675003.

- Fwd No Answer: 6139675001
- Fwd Delay: 4
- Fwd Busy: 6139675002
- Fwd All: 6139675003

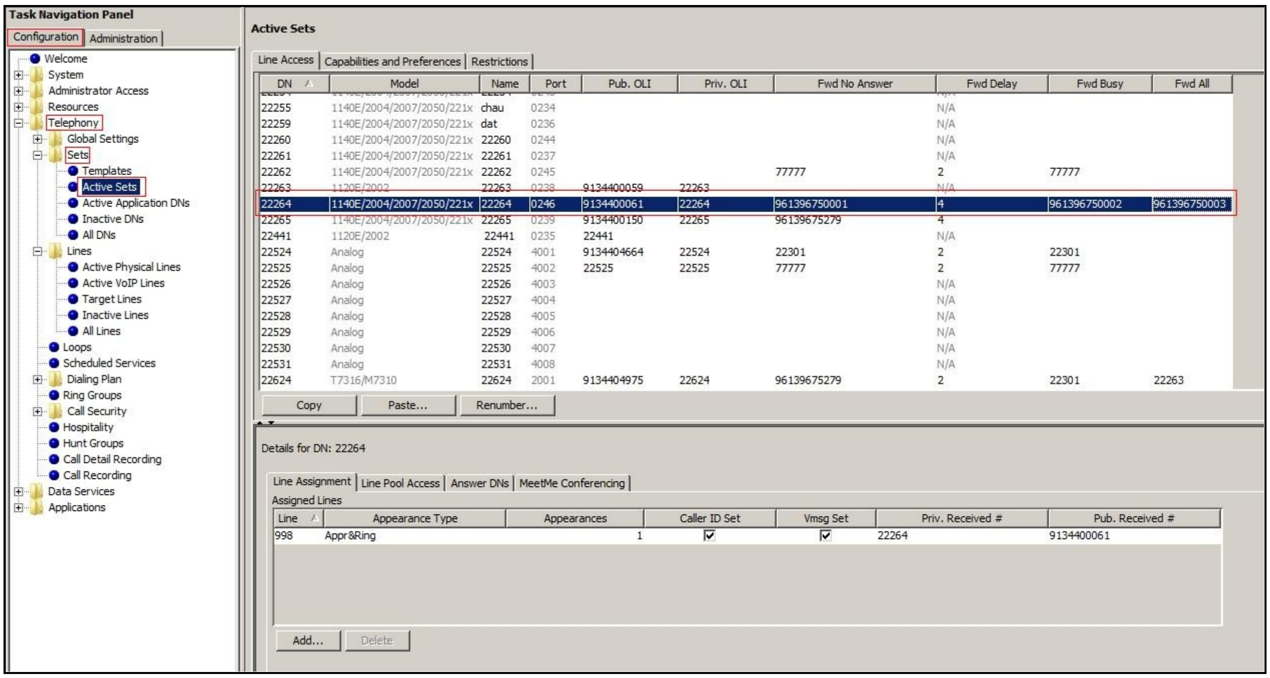

Figure 43: Call forward settings

6. Verification Steps

The following steps may be used to verify the configuration:

6.1. General

Place an **inbound/ outbound** call from/to a PSTN phone to/from an internal BCM450 phone, answer the call, and verify that two-way speech path exists. Check call display name and number to ensure the correct info was sent/received. Perform hold/ retrieve. Verify the call remains stable for several minutes and disconnect properly.

6.2. Verify Call Establishment on BCM450

- Use BCM450 Monitor to verify VoIP Line
Figure 44 shows Line001 was used for an incoming call and Line008 was used for an outgoing call.

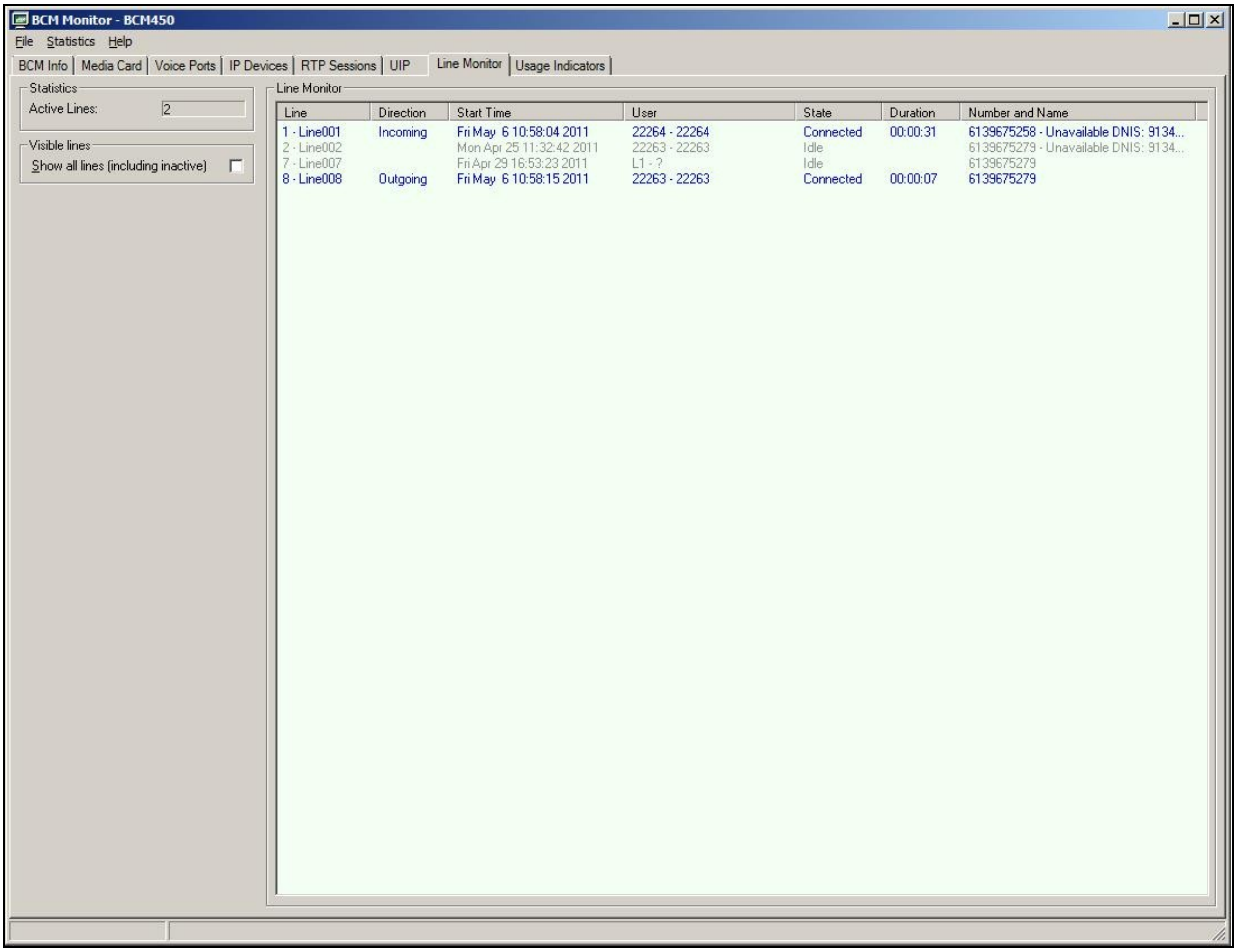

Figure 44: VoIP Line monitoring

- Use BCM450 Monitor to verify RTP
Figure 45 shows an active SIP call with RTP G.711

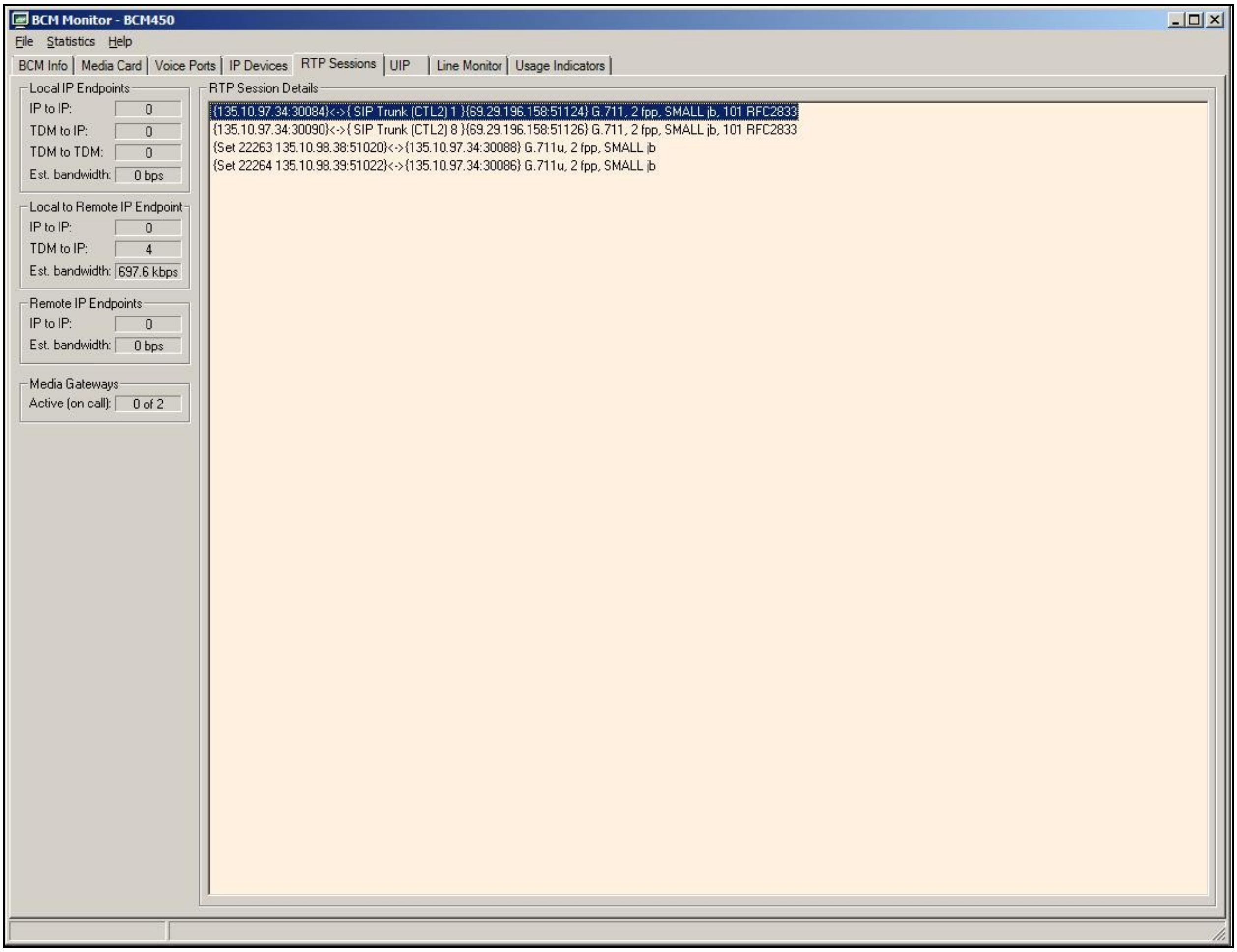

Figure 45: RTP monitoring

6.3. Protocol Traces

Ethereal traces are captured and analyzed. The following SIP headers are inspected:

- RequestURI: verify the request number and either SIP domain
- From: verify the display name and display number.
- To: verify the display name and display number.
- Diversion: verify the call forward information and reason code.
- P-Asserted-Identity: verify the display name and display number.
- Privacy: verify the “user, id” masking.

The following attributes in SIP message body are inspected:

- Connection Information (c): verify IP address of far end endpoint
- Time Description (t): verify session timeout of far end endpoint

- Media Description (m): verify audio port, codec, DTMF event description
- Media Attribute (a): verify specific audio port, codec, ptime, send/ receive ability, DTMF event and fax attributes.

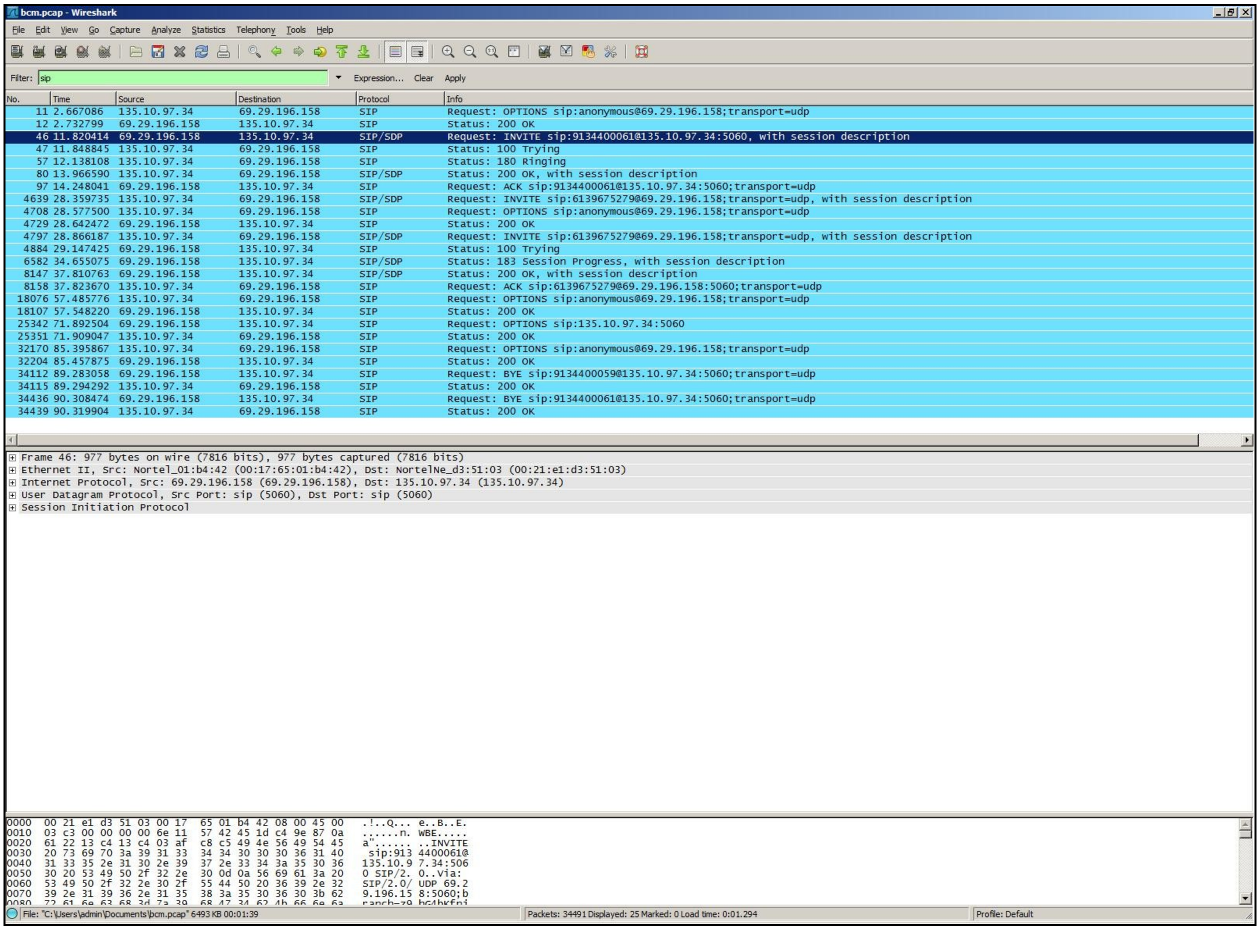

Figure 46: Ethereal trace analysis

7. Conclusion

All of the test cases have been executed. The test result met the objectives outlined in **Section 2.1** with observations seen during testing as noted in **Section 2.2**. The CenturyLink system is considered **compliant** with the Avaya Business Communication Manager 450 Release 6.0.

8. Additional References

Product documentation for Avaya products may be found at:
http://support.avaya.com/css/appmanager/public/support

[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.*

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.