



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.IPTTEL-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.

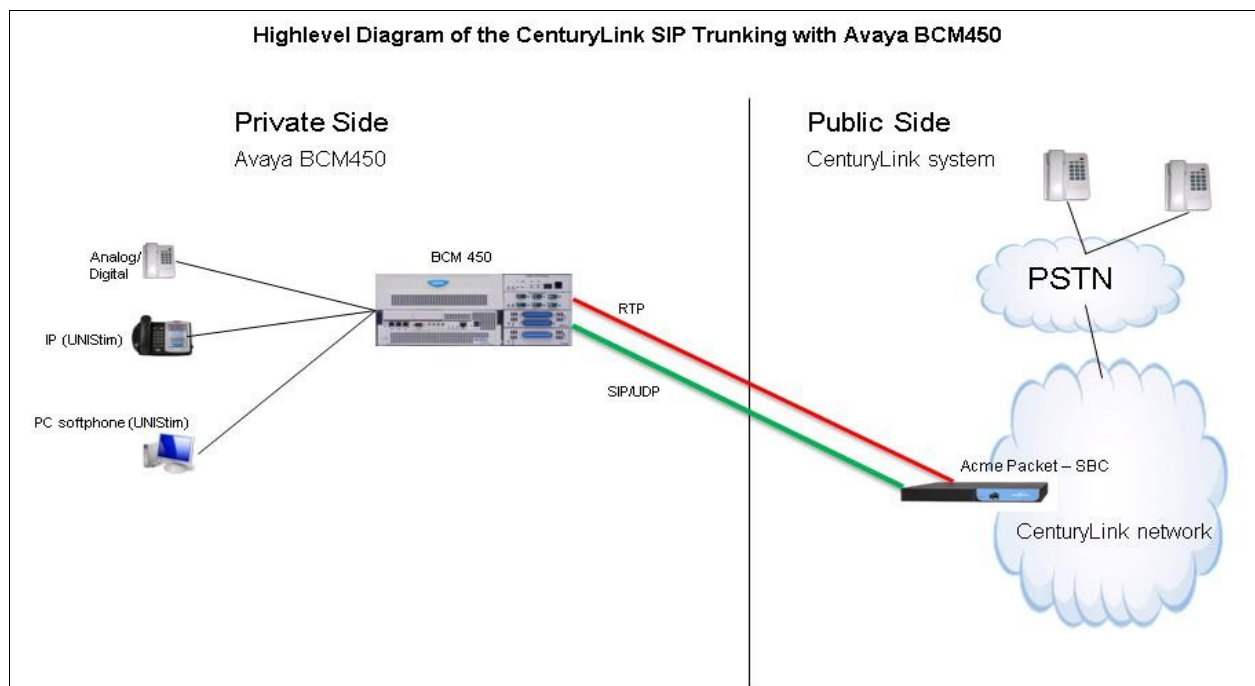


Figure 1- Network Diagram for BCM450 – CenturyLink System

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: <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)• Nortel Digital Phone M7310

	<ul style="list-style-type: none"> • I2050 PC softphone Release 3.2
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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.

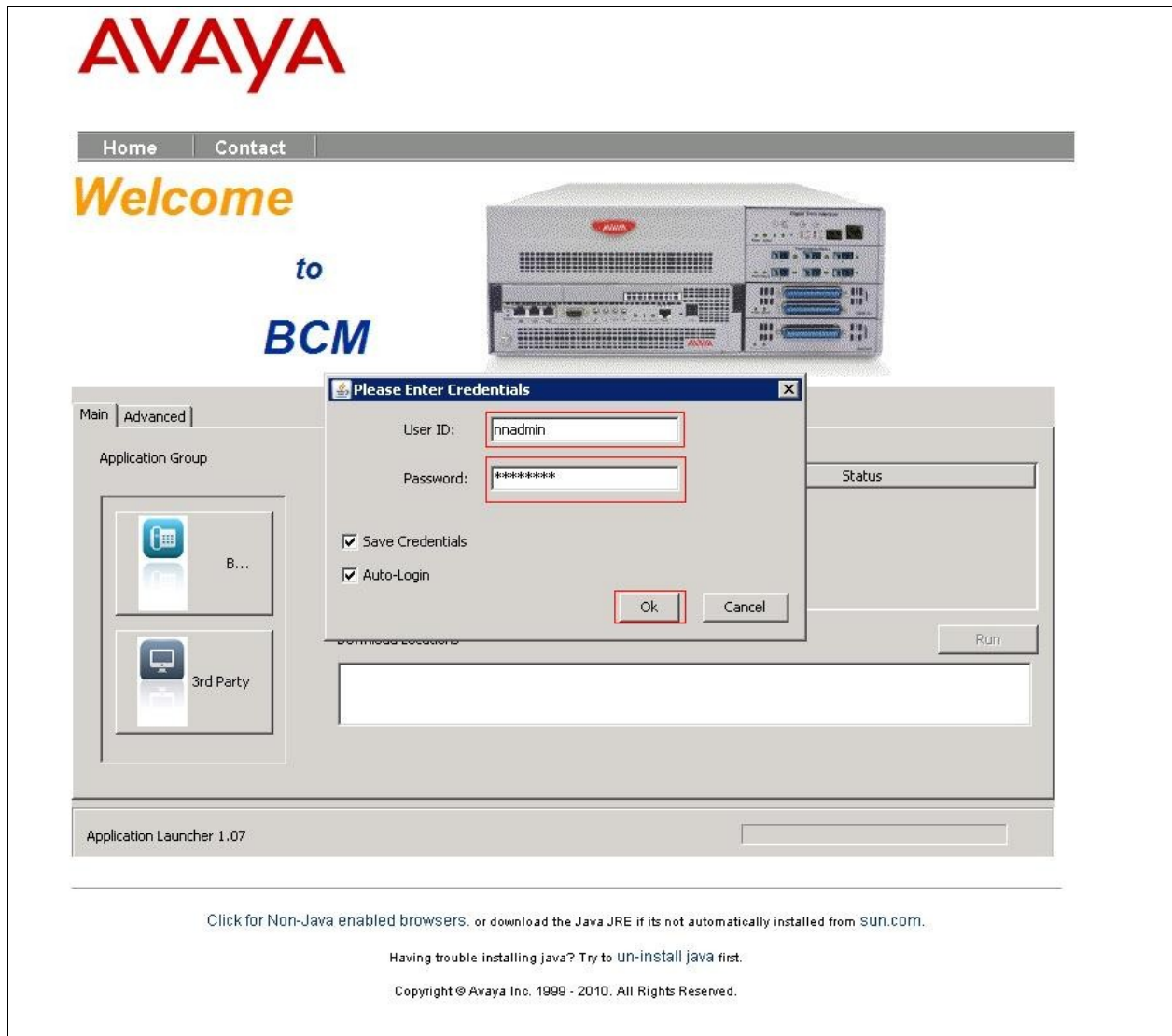


Figure 2 – Login to Business Communication Manager

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.

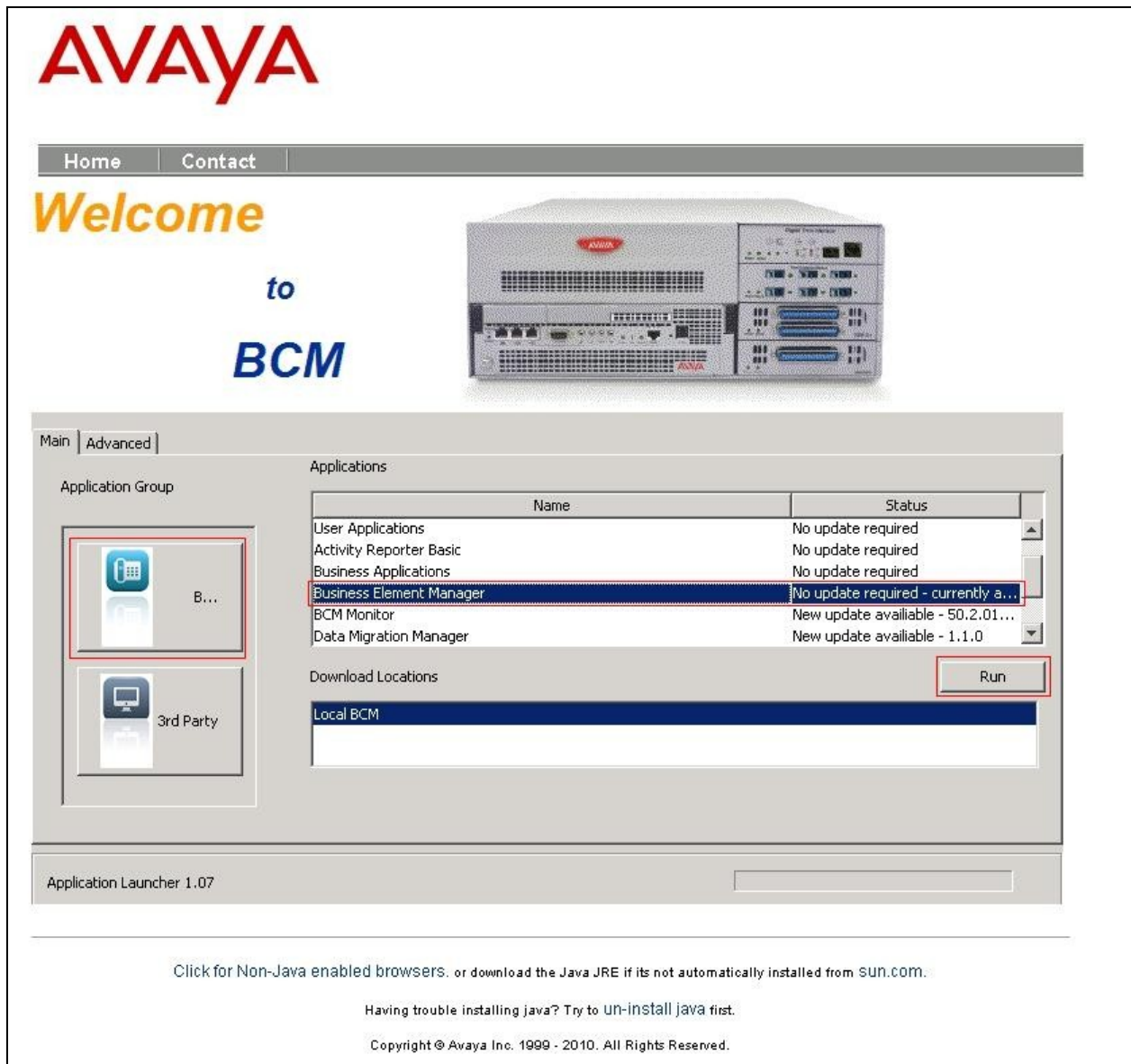


Figure 3 – Business Communication Manager

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.

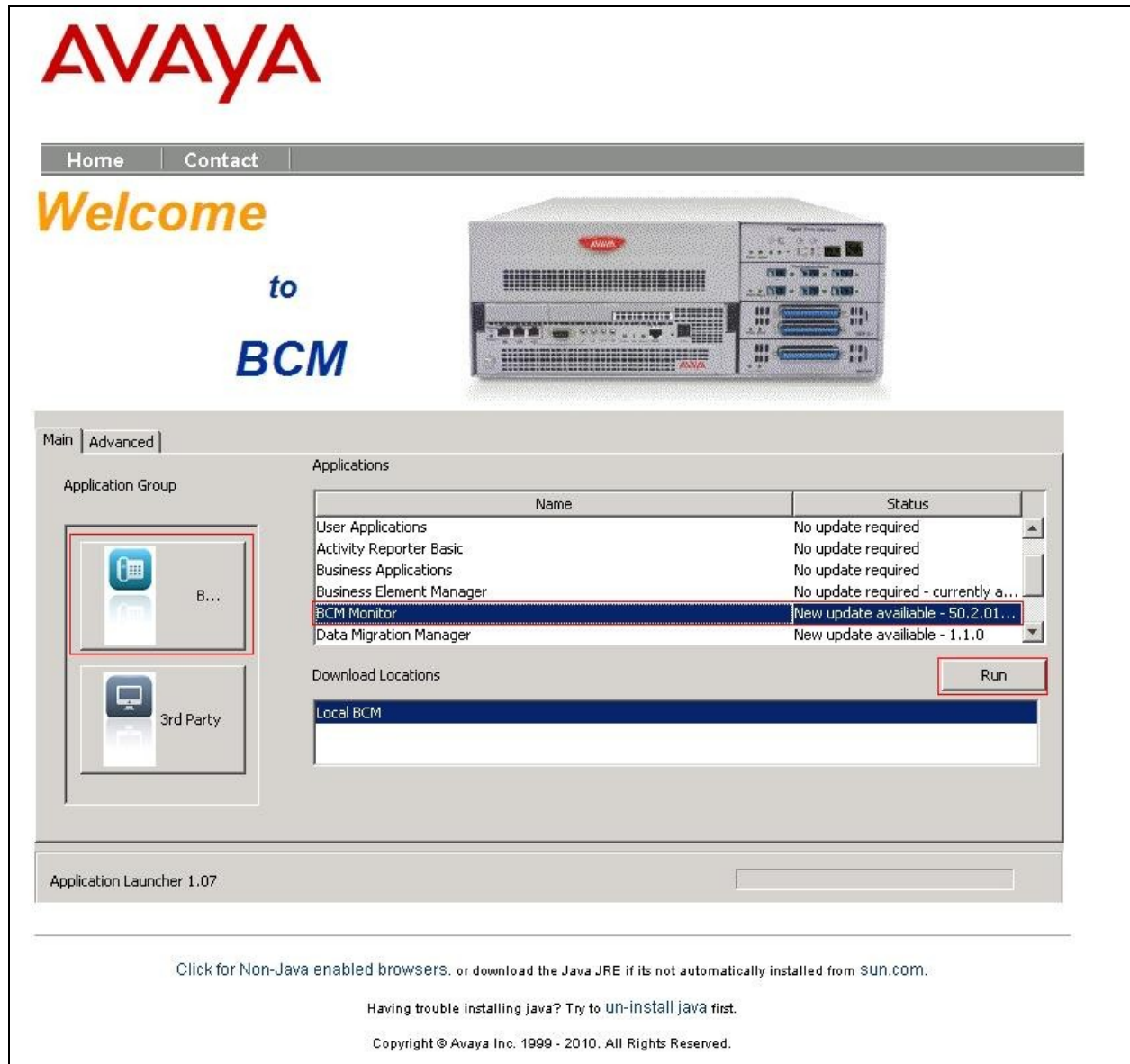


Figure 4 – Element Manager System Overview

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**.

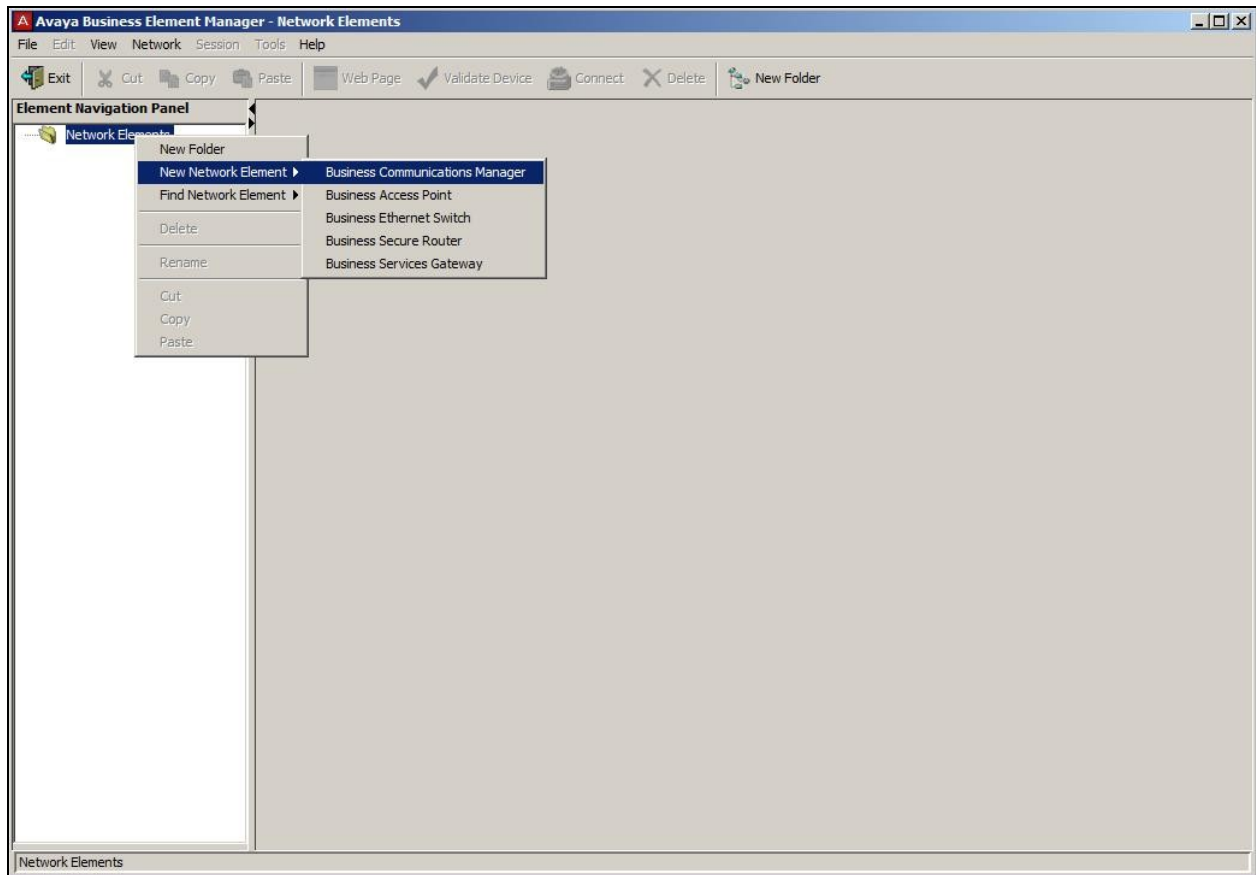


Figure 5: Create a New Network Element

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

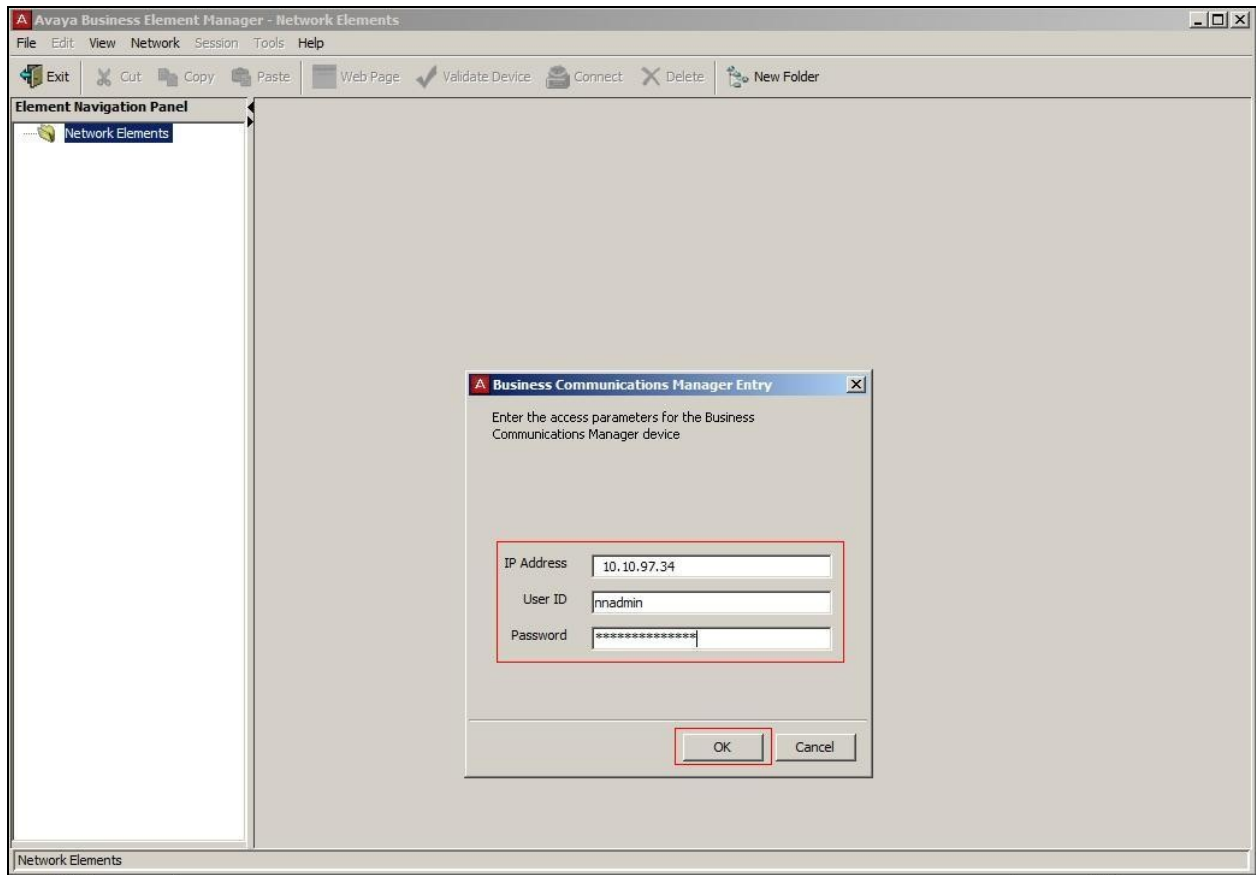


Figure 6: Business Communication Manager Entry

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**.

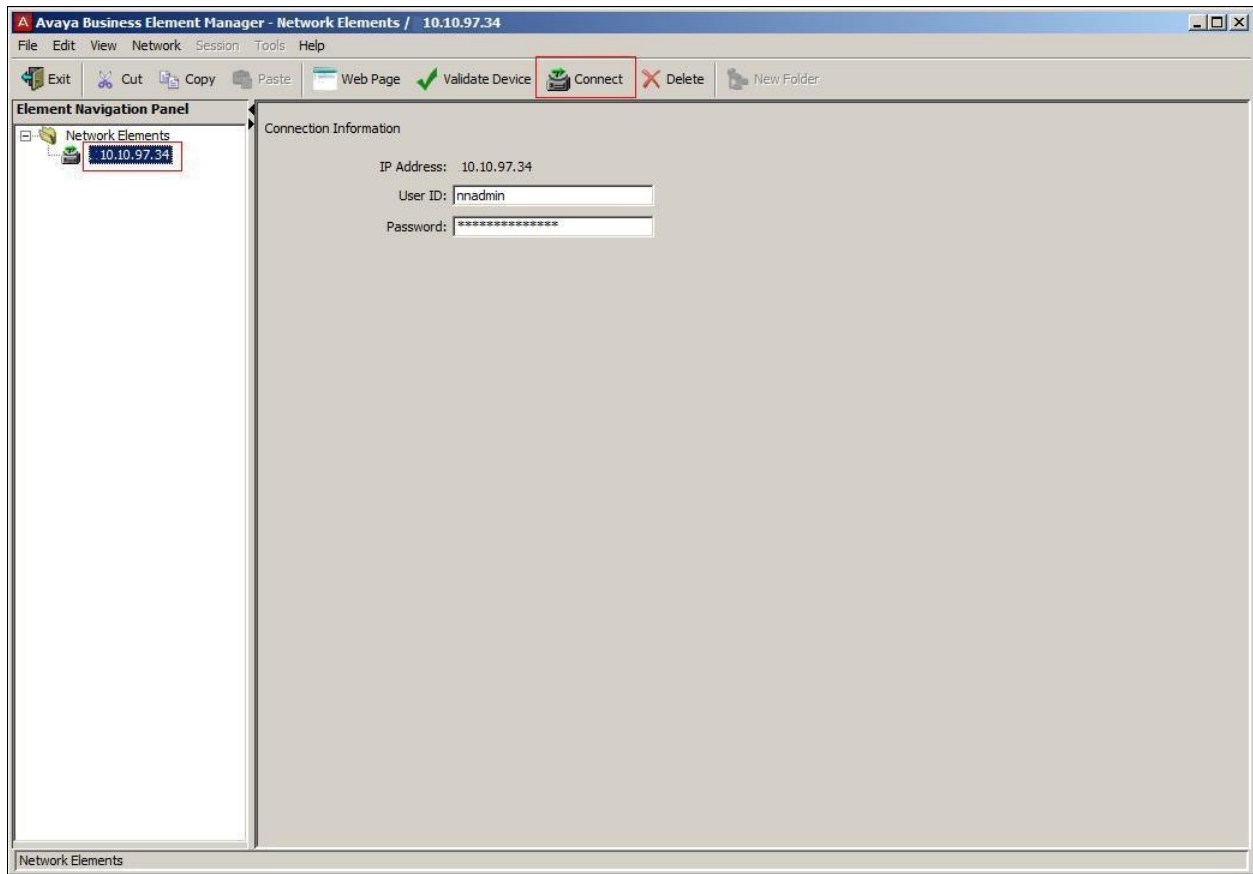


Figure 7: Connect to BCM450

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.

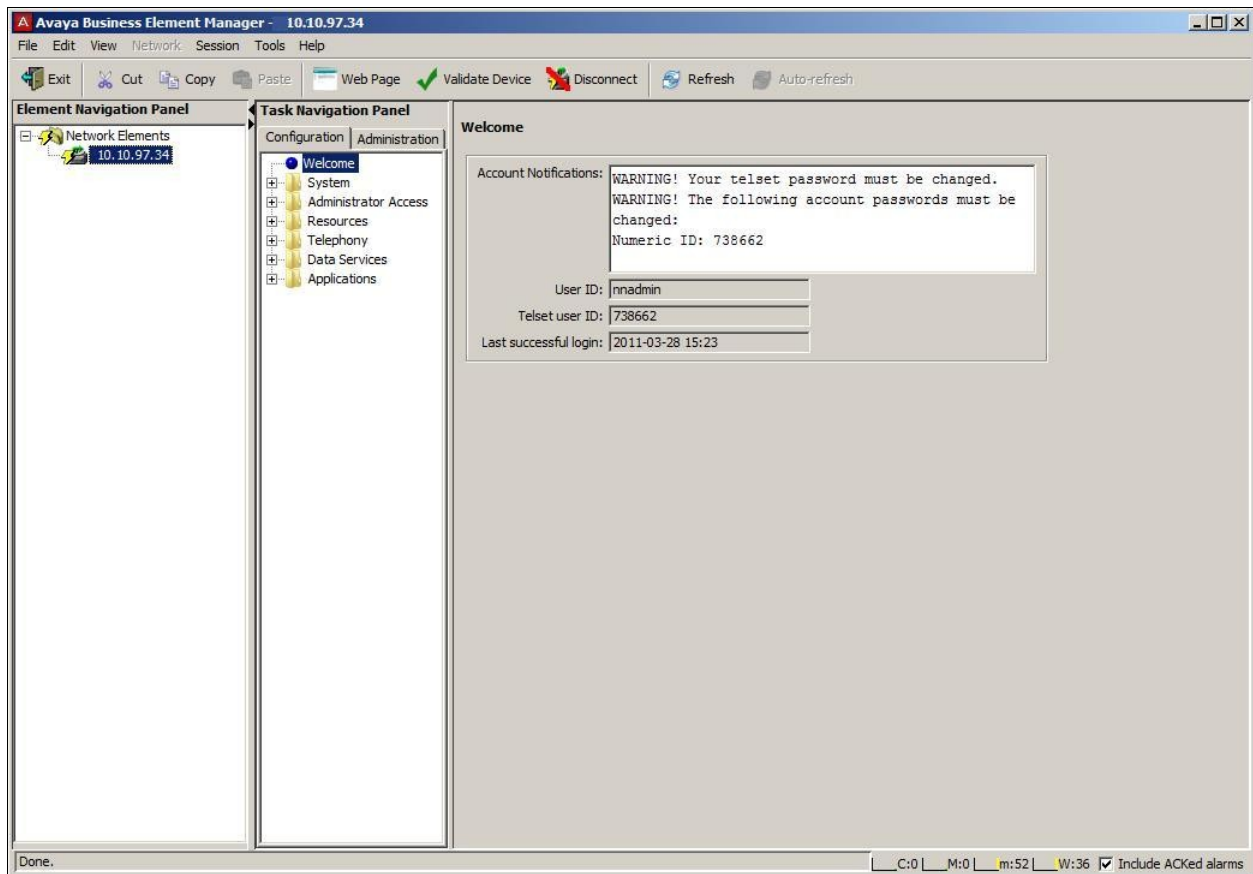


Figure 8: Avaya Business Element Manager

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**.

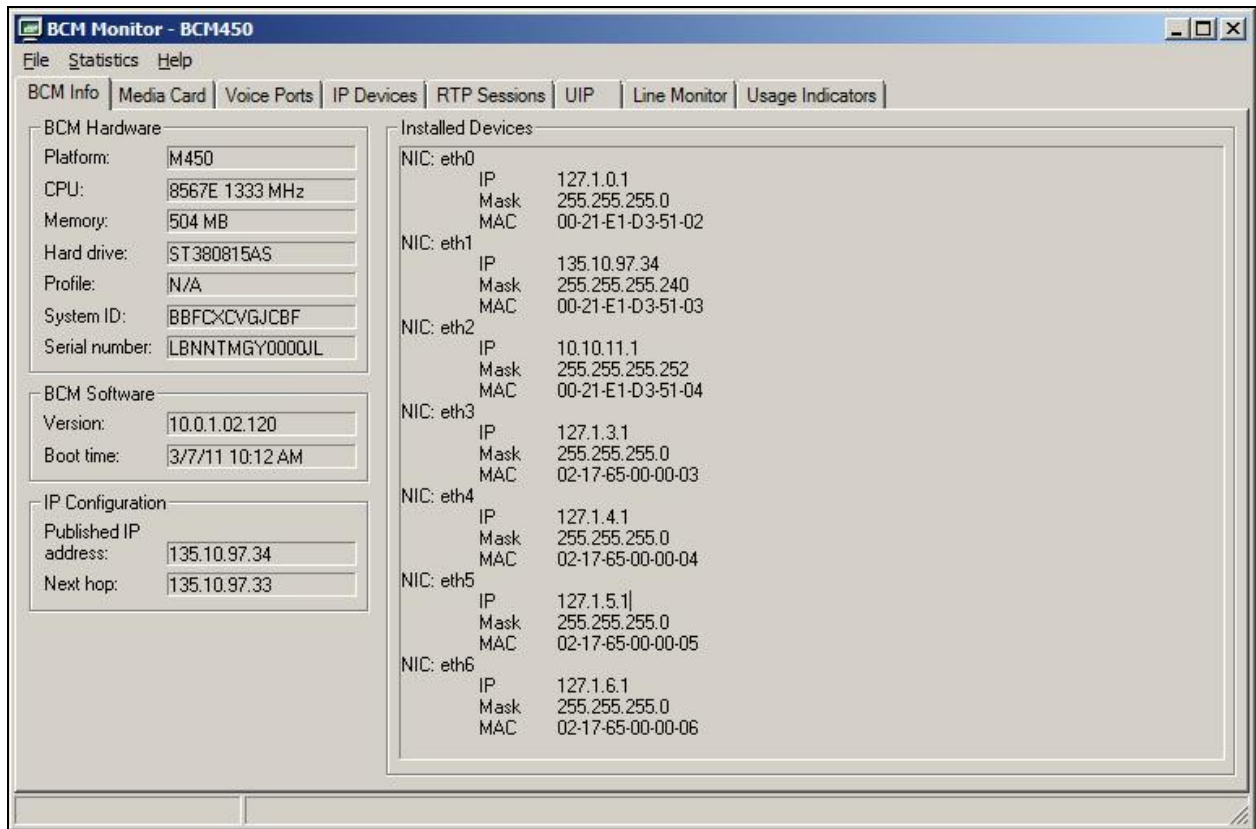


Figure 10: BCM450 Monitor GUI

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

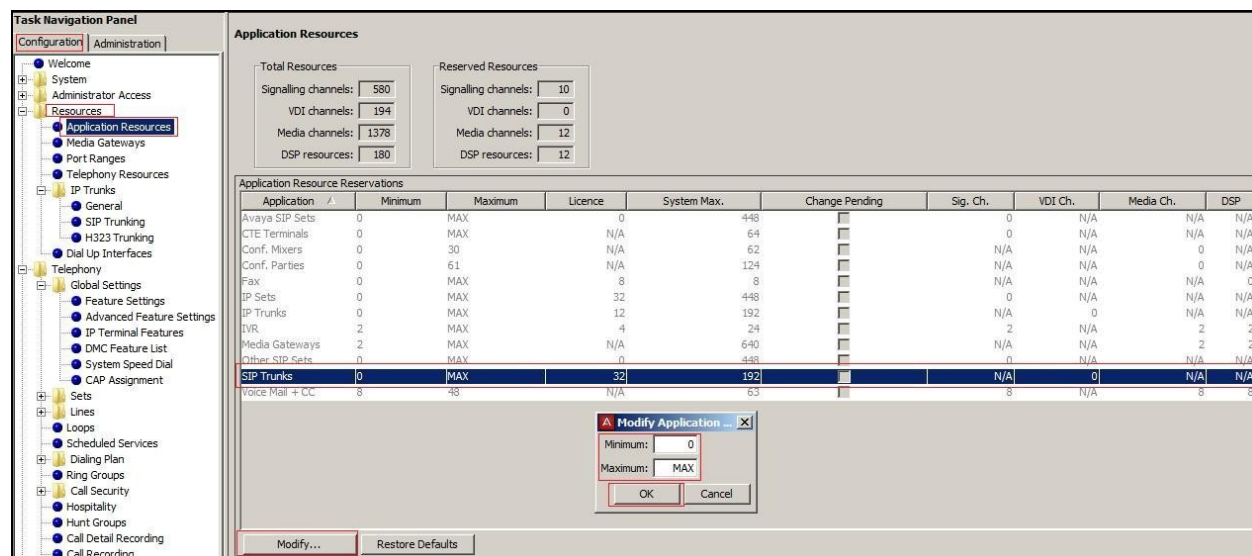


Figure 11 – Configuring Application Resources for SIP Trunks

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**.

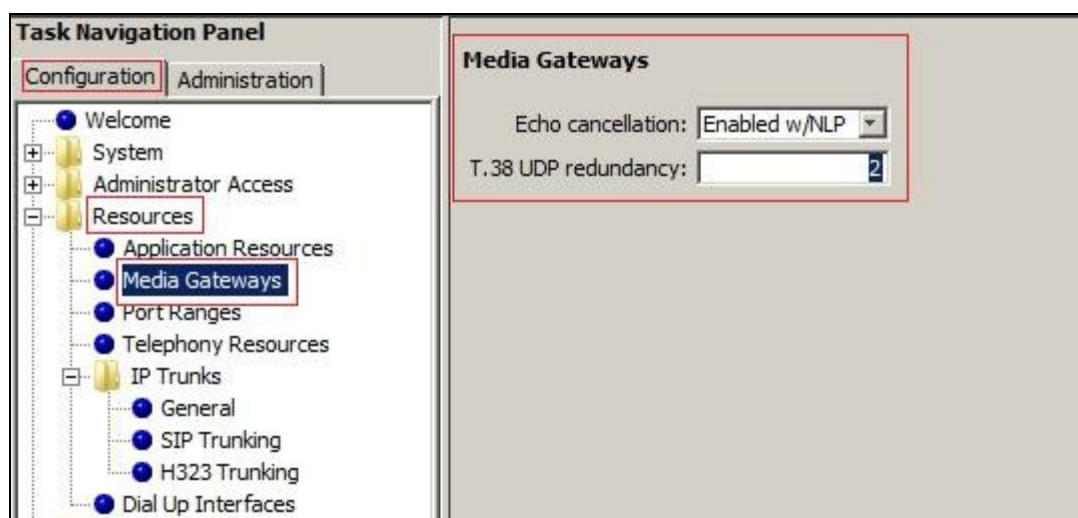


Figure 12 – Configuring Media Gateways

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**.

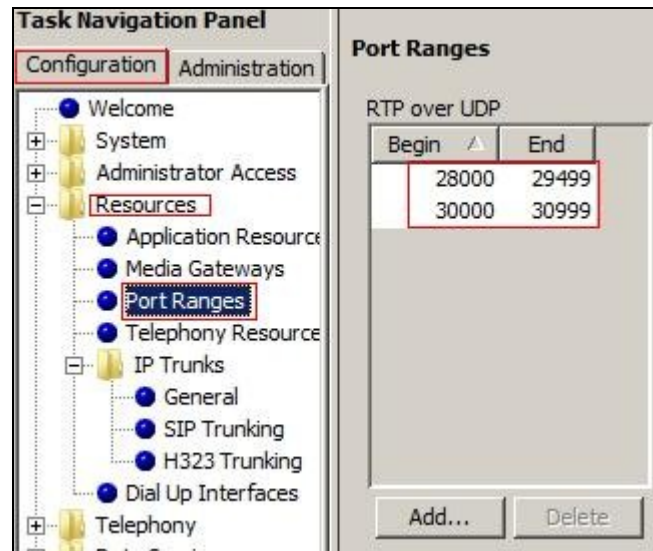


Figure 13 – Port Ranges for RTP over UDP

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)

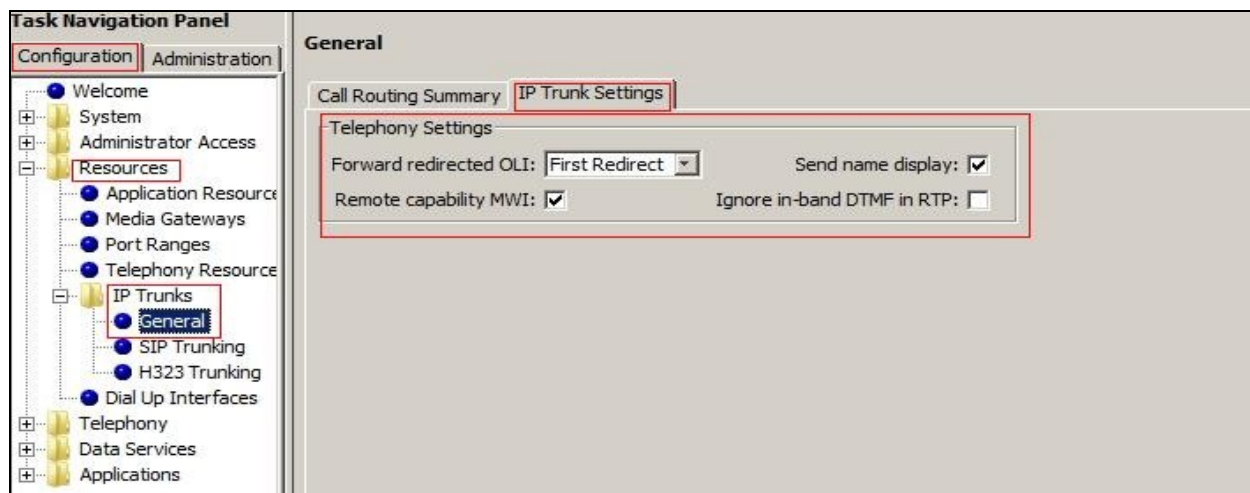


Figure 14 – IP Trunk Settings

5.3.2. Administer Global Settings

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

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

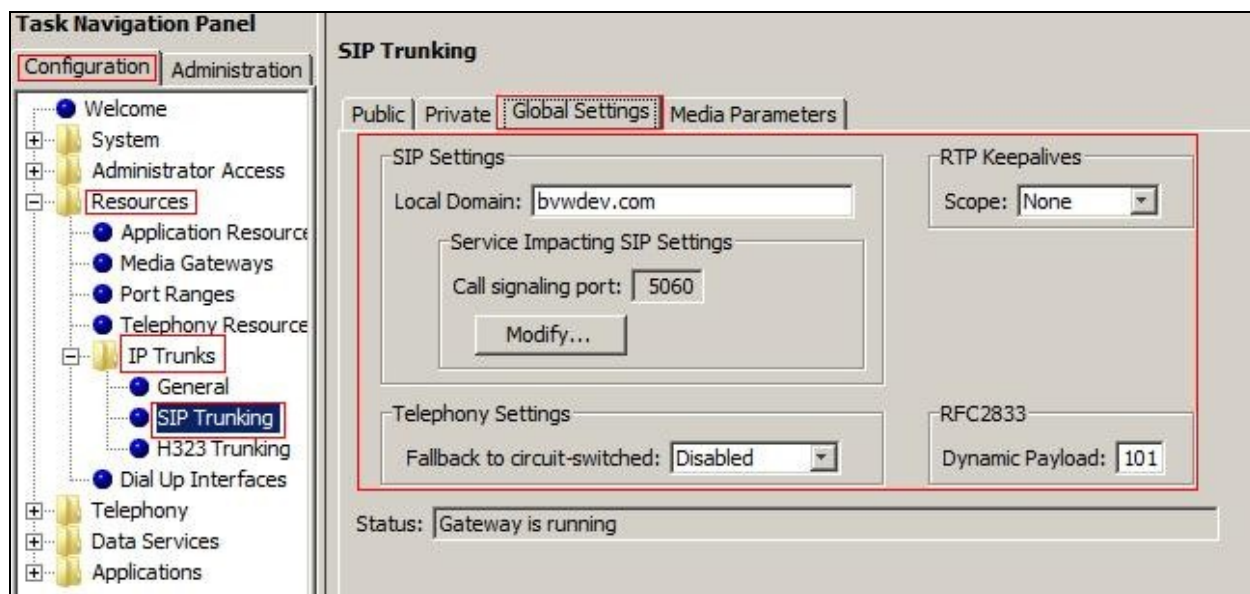


Figure 15 – Global Settings for SIP Trunk

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**.

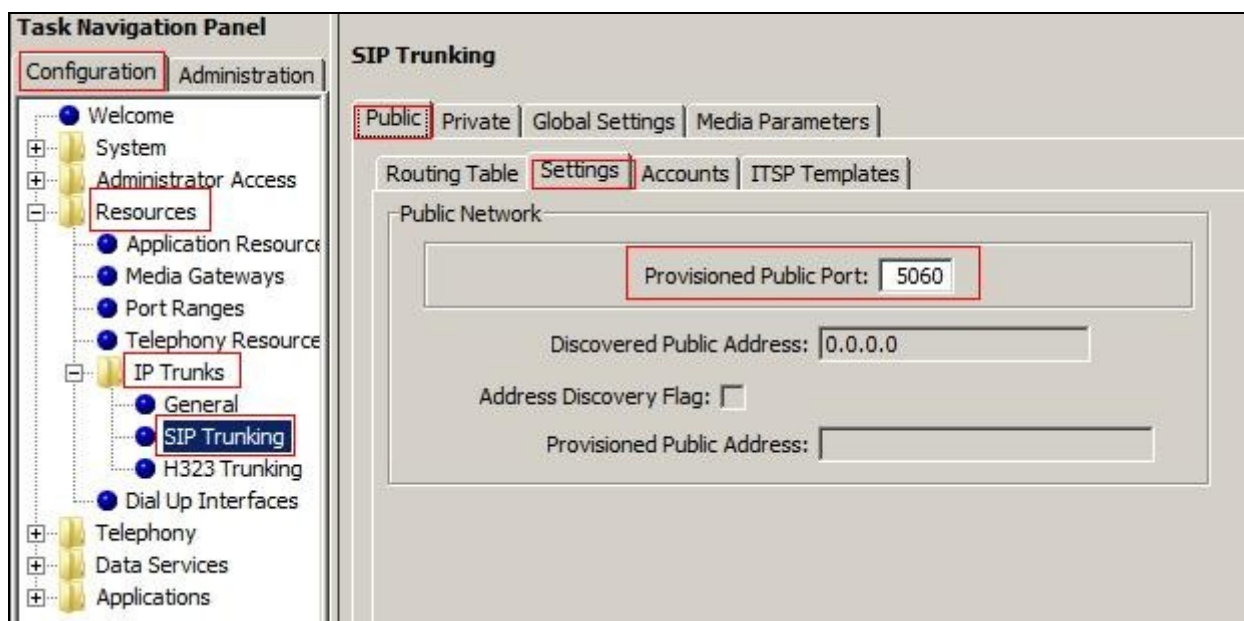


Figure 16 – Provisioned Public Port Setting

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).

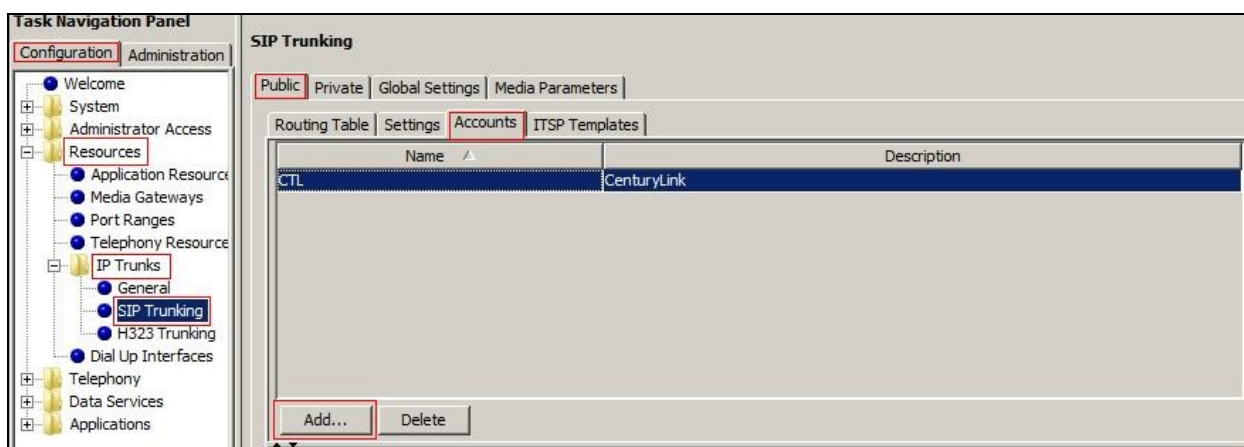


Figure 17 – Add a public account for SIP Trunk

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

Details for Account:

Basic | Advanced | User Accounts

SIP Domain

Remote: 29.29.196.157

Local: 10.10.97.34

Proxy

Address:

Port: 0

Transport: UDP

Registrar

Address:

Port: 0

Outbound Proxy Table

Domain	IP Address	Port	Load-balancing Weight	Keep alive
29.29.196.157	29.29.196.157	5060		0 OPTIONS

Figure 19 – Basic Configuration for Public SIP Trunk Account

5.3.6. Advance Settings

- Select account CTL created in Section 5.3.4
- 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)

Details for Account:

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:

Signaling interval:

Session timer

Session refresh method:

Refresher:

Session-Expires:

Min-SE:

Active call limit:

ITSP association method:

Outbound Called characters to absorb:

Inbound Called prefix to prepend:

Authentication realm:

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.

Details for Account:

Basic | Advanced | **User Accounts**

Description	Parent	CLID	SIP Username	Register	Status
Centurylink	<input checked="" type="checkbox"/>			<input type="checkbox"/>	Not registered

Add... Delete **Modify...** Refresh

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.

The screenshot shows a 'Modify account' dialog box with the following sections and fields:

- Description:** Centurylink
- Domain:** 29.29.196.157
- Account identity:**
 - Parent account: ☒
- User Credentials:**
 - SIP username:
 - Auth name:
 - Auth password:
 - Realm:
- Message Handling:**
 - CLID Override:
 - Display name Override:
 - PAI CLID Override:
 - PAI Display name Override:
 - Contact Override:
 - Maddr in Contact: ☐
 - Local Domain Override: 10.10.97.34
 - Compensate for NAT: ☐
- SIP Registration:**
 - Registration: ☐

At the bottom right are 'OK' and 'Cancel' buttons.

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 **Selected list**, with **Voice Activity Detection** disabled; **G.711 payload size (ms)** is set to **20ms**.

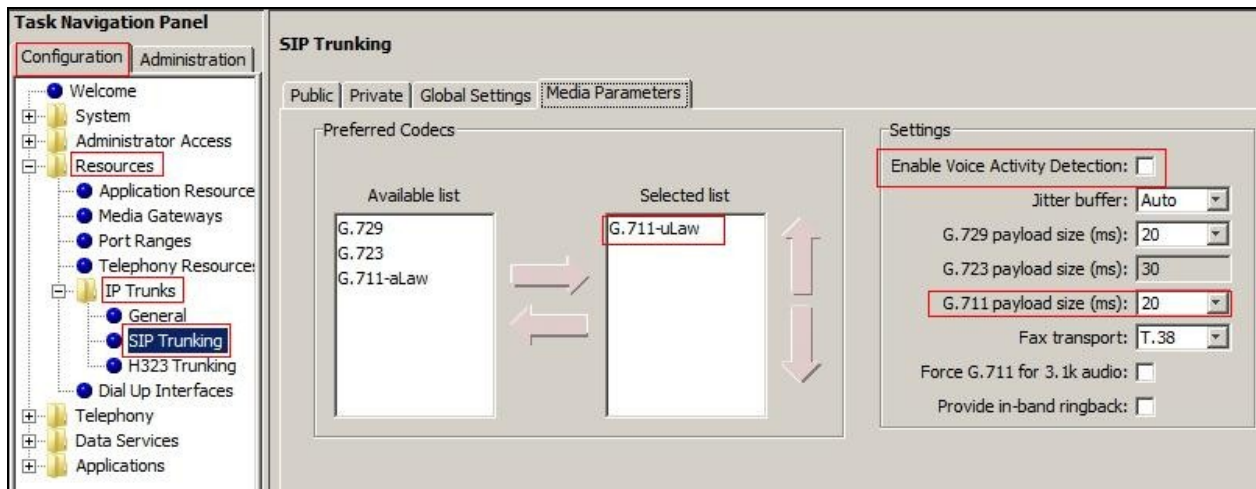


Figure 23 – Codec Settings for SIP Trunk

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**

Task Navigation Panel

Configuration Administration

- Welcome
- System
 - Administrator Access
- Resources
 - Application Resources
 - Media Gateways
 - Port Ranges
 - Telephony Resources**
 - IP Trunks
 - General
 - SIP Trunking
 - H323 Trunking
 - Dial Up Interfaces
- Telephony
- Data Services
- Applications

Telephony Resources

Modules

Location	Configured Device	Dip Switch	Bus	State	Low	High	Active	Busy
Internal	IP Trunks	N/A	N/A	Enabled	001	008	8	0
Internal	IP Sets	N/A	N/A	Enabled	22221	22441	13	0
Internal	Applications	N/A	N/A	Enabled	22300	22399	99	N/A
Main MBM 1	ASM/ASM+ MBM	All On	10.1	Enabled	20224	22231	7	0
Main MBM 2	DSM16/DSM16+ MBM	All On	20.1	Enabled	22624	22639	1	0
Main MBM 3	DTM-PRI	All On	30.1	Enabled	009	031	23	0
Main MBM 4	ASM/ASM+ MBM	All On	40.1	Enabled	22524	22531	8	0
Expansion 1	None	N/A	N/A	N/A	N/A	N/A	N/A	N/A

Disable Enable Deconfigure... Configure...

Details for Module: Internal IP Sets

IP Terminal Global Settings IP Terminal Details

Enable registration: ☒

Enable global registration password: ☒

Global password: *****

Auto-assign DNS: ☒

Play DTMF-tone: ☒

Advertisement/Logo: AVAYA BCM 450

Discovered Public Address: 0.0.0.0

Provisioned Public Address:

Support Remote Worker: ☐

G.729 payload size (ms): 20

G.723 payload size (ms): 30

G.711 payload size (ms): 20

Default codec: G.711-uLaw

Default jitter buffer: Auto

IP clients configuration files

File Name	File Created	File Size (Bytes)
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Upload... Download... Delete All Delete Refresh

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**).

Task Navigation Panel									
<ul style="list-style-type: none"> Welcome System Administrator Access Resources Telephony <ul style="list-style-type: none"> Global Settings Sets Lines <ul style="list-style-type: none"> Active Physical Lines Active VoIP Lines Target Lines Inactive Lines All Lines Loops Scheduled Services Dialing Plan <ul style="list-style-type: none"> Ring Groups Call Security Hospitality Hunt Groups Call Detail Recording Call Recording Data Services 									

Active VoIP Lines									
Line	Trunk Type	Name	Control Set	Line Type	Prime Set	Pub. Received #	Priv. Received #	Distinct Ring	
001	VoIP	Line001	22231	Pool:BlocA	22231	N/A	N/A	None	
002	VoIP	Line002	22231	Pool:BlocA	22231	N/A	N/A	None	
003	VoIP	Line003	22231	Pool:BlocA	22231	N/A	N/A	None	
004	VoIP	Line004	22231	Pool:BlocA	22231	N/A	N/A	None	
005	VoIP	Line005	22231	Pool:BlocA	22231	N/A	N/A	None	
006	VoIP	Line006	22231	Pool:BlocA	22231	N/A	N/A	None	
007	VoIP	Line007	22231	Pool:BlocA	22231	N/A	N/A	None	
008	VoIP	Line008	22231	Pool:BlocA	22231	N/A	N/A	None	

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

Task Navigation Panel		Dialing Plan - General																															
<ul style="list-style-type: none"> Welcome System Administrator Access Resources Telephony <ul style="list-style-type: none"> Global Settings Sets Lines Loops Scheduled Services Dialing Plan <ul style="list-style-type: none"> General DNs Public Network Private Network Line Pools Routing Ring Groups Call Security Hospitality Hunt Groups Call Detail Recording Call Recording Data Services Applications 		<div>Global Settings</div> <div> DN length (intercom): 5 Dialing timeout: 4 </div> <div>Access Codes</div> <div> Park prefix: 1 External code: 6 </div> <div>Direct Dial</div> <div>Direct Dial digit: 0</div> <div>Direct Dial Sets</div> <table> <tr> <th>Set</th><th>Type</th><th>Internal DN</th><th>External No.</th><th>Facility</th></tr> <tr> <td>1</td><td>Internal</td><td>DN:22231</td><td>N/A</td><td>N/A</td></tr> <tr> <td>2</td><td>None</td><td>N/A</td><td>N/A</td><td>N/A</td></tr> <tr> <td>3</td><td>None</td><td>N/A</td><td>N/A</td><td>N/A</td></tr> <tr> <td>4</td><td>None</td><td>N/A</td><td>N/A</td><td>N/A</td></tr> <tr> <td>5</td><td>None</td><td>N/A</td><td>N/A</td><td>N/A</td></tr> </table>		Set	Type	Internal DN	External No.	Facility	1	Internal	DN:22231	N/A	N/A	2	None	N/A	N/A	N/A	3	None	N/A	N/A	N/A	4	None	N/A	N/A	N/A	5	None	N/A	N/A	N/A
Set	Type	Internal DN	External No.	Facility																													
1	Internal	DN:22231	N/A	N/A																													
2	None	N/A	N/A	N/A																													
3	None	N/A	N/A	N/A																													
4	None	N/A	N/A	N/A																													
5	None	N/A	N/A	N/A																													

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

Task Navigation Panel

Configuration Administration

- Welcome
- System
- Administrator Access
- Resources
- Telephony
 - Global Settings
 - Sets
 - Lines
 - Loops
 - Scheduled Services
 - Dialing Plan
 - General
 - DNs
 - Public Network**
 - Private Network
 - Line Pools
 - Routing
 - Ring Groups
- Call Security
 - Hospitality
 - Hunt Groups
 - Call Detail Recording
 - Call Recording
- Data Services
- Applications

Dialing Plan - Public Network

Public Network Settings

Public Received number length: 10

Public Auto DN:

Public DISA DN:

Public network dialing plan: Public (Unknown)

Public network code:

Public Network DN Lengths

DN Prefix	DN Length
Default	10

Add... Delete

Carrier Codes

Code Prefix	ID Length
-------------	-----------

Add... Delete

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

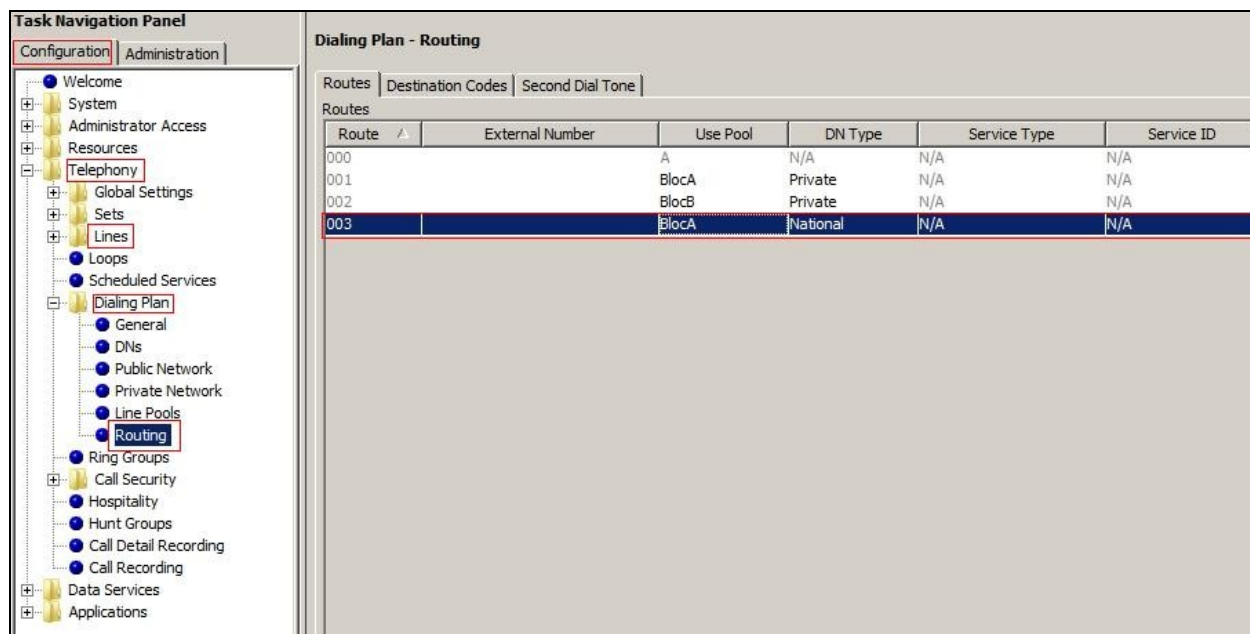


Figure 28 – Create a route

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)

Destination Code	Normal Route	Absorbed Length	Wild Card: 0	1	2	3	4	5	6	7	8	9
53	001	0										
54	001	All										
55	001	0										
56	001	0										
57	001	0										
77	001	0										
322	001	0										
525	001	0										
26	002	0										
90	003	1										
9411	003	1										
9416	003	1										
9613	003	1										
9647	003	1										
9905	003	1										
9911	003	1										
91613	003	1										
91800	003	1										
91866	003	1										
9913360	003	1										
9913488	003	1										
9913533	003	1										
9913738	003	1										
9913856	003	1										

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.

Name	Destination Digits	Account
0	0	CTL
1	1	CTL
411	411	CTL
911	911	CTL
ctl_318360	318360	CTL
ctl_913488	913488	CTL
ctl_913533	913533	CTL
ctl_913738	913738	CTL
ctl_913856	913856	CTL
ont	647	CTL
ont_416	416	CTL
ont_613	613	CTL
ont_905	905	CTL

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.

- Select **Configuration > Telephony > Sets > Active Sets**.
- On tab **Line Access**, chose a DN e.g. 22264.
- 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.

Task Navigation Panel

- Configuration
 - System
 - Administrator Access
 - Resources
 - Telephony
 - Global Settings
 - Sets
 - Active Sets
 - Active Applicat...
 - Inactive DN...
 - All DN...
 - Lines
 - Loops
 - Scheduled Services
 - Dialing Plan
 - Ring Groups
 - Call Security
 - Hospitality
 - Hunt Groups
 - Call Detail Recording
 - Call Recording
 - Data Services
 - Applications

Active Sets

Line Access | Capabilities and Preferences | Restrictions

DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Answer	Fwd Delay	Fwd Busy	Fwd All
22255	1140E/2004/2007/2050/221x	chau	0234						
22259	1140E/2004/2007/2050/221x	dat	0236						
22260	1140E/2004/2007/2050/221x	22260	0244						
22261	1140E/2004/2007/2050/221x	22261	0237						
22262	1140E/2004/2007/2050/221x	22262	0245						
22263	1120E/2002	22263	0238	9134400059	22263	77777		77777	
22264	1140E/2004/2007/2050/221x	22264	0246	9134400061	22264	22263	4	22265	
22265	1140E/2004/2007/2050/221x	22265	0239	9134400150	22265	96139675279	4		
22441	1120E/2002	22441	0235	22441					
22524	Analog	22524	4001	9134404664	22524	22301	2	22301	
22525	Analog	22525	4002	22525		77777	2	77777	
22526	Analog	22526	4003						
22527	Analog	22527	4004						
22528	Analog	22528	4005						
22529	Analog	22529	4006						
22530	Analog	22530	4007						
22531	Analog	22531	4008						
22624	T7316/M7310	22624	2001	9134404975	22624	96139675279	2	22301	22263

Copy Paste... Renumber...

Details for DN: 22264

Line Assignment | Line Pool Access | Answer DN | MeetMe Conferencing

Assigned Lines

Line	Appearance Type	Appearances	Caller ID Set	Vmsg Set	Priv. Received #	Pub. Received #
998	Appr&Ring	1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	22264	9134400061

Add... Delete

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**.

Task Navigation Panel

- Configuration
- Administration
- Welcome
- System
- Administrator Access
- Resources
- Telephony
 - Global Settings
 - Feature Settings**
 - Advanced Feature Settings
 - IP Terminal Features
 - DMC Feature List
 - System Speed Dial
 - CAP Assignment
 - Sets
 - Lines
 - Loops
 - Scheduled Services
 - Dialing Plan
 - Ring Groups
 - Call Security
 - Hospitality
 - Hunt Groups
 - Call Detail Recording
 - Call Recording
- Data Services
- Applications

Feature Settings

Business Names

1: BCM450 2: 3: 4: 5:

Feature Settings

Background music: ☒ On hold: Music Held line reminder: Off Answer keys: Basic
Page tone: ☒ Delayed ring transfer: After 4 rings Receiver volume: Use sys volume
Message reply enhancement: ☐ Park mode: Lowest Directed pickup: ☒
Force auto/spd dial over ic/conf: ☐ Maximum CLI per line: 50 Set relocation: ☐
Alarm set: 22231

Timers

Camp timeout (sec.): 45 Transfer callback timeout: After 4 rings Host delay (ms.): 1000
Park timeout (sec.): 45 Link time (ms.): 600
Page timeout (sec.): 180

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**.

Task Navigation Panel

- Configuration
- Administration
- Welcome
- System
- Administrator Access
- Resources
- Telephony
 - Global Settings
 - Sets**
 - Active Sets
 - Active Application DNs
 - Inactive DNs
 - All DNs
 - Lines
 - Loops
 - Scheduled Services
 - Dialing Plan
 - Ring Groups
 - Call Security
 - Hospitality
 - Hunt Groups
 - Call Detail Recording
- Data Services
- Applications

Active Sets

Line Access Capabilities and Preferences Restrictions

DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Answer	Fwd Delay	Fwd Busy	Fwd All
22255	1140E/2004/2007/2050/221x	chau	0234				N/A		
22259	1140E/2004/2007/2050/221x	dat	0236				N/A		
22260	1140E/2004/2007/2050/221x	22260	0244				N/A		
22261	1140E/2004/2007/2050/221x	22261	0237				N/A		
22262	1140E/2004/2007/2050/221x	22262	0245				N/A		
22263	1120E/2002	22263	0238				N/A	77777	
22264	1140E/2004/2007/2050/221x	22264	0246	9134400059	22263	77777	4	22265	
22265	1140E/2004/2007/2050/221x	22265	0239	9134400150	22265	96139675279			
22441	1120E/2002	22441	0235				N/A		
22524	Analog	22524	4001	9134404664	22524	22301	2	22301	
22525	Analog	22525	4002	22525	22525	77777	2	77777	
22526	Analog	22526	4003				N/A		
22527	Analog	22527	4004				N/A		
22528	Analog	22528	4005				N/A		
22529	Analog	22529	4006				N/A		
22530	Analog	22530	4007				N/A		
22531	Analog	22531	4008				N/A		
22624	T7316/M7310	22624	2001	9134404975	22624	96139675279	2	22301	22263

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

The screenshot displays the Avaya configuration interface. On the left is the 'Task Navigation Panel' with a tree view showing 'Configuration' > 'Telephony' > 'Sets' > 'Active Sets' selected. The main area is titled 'Active Sets' and contains a table with columns: DN, Model, Name, Prime Line, Intercom Keys, Control Set, First Display, and Auto Called ID. The table lists various DNs, with DN 22264 highlighted. Below the table are 'Copy' and 'Paste...' buttons. Below the table is a section titled 'Details for DN: 22264' with tabs for 'Capabilities', 'SWCA Call Group', 'Preferences', 'IP Terminal Details', 'Button Programming Table', 'Button Programming', and 'User Speed Dial'. The 'Preferences' tab is active, showing settings for 'Language' (English), 'Dialing options' (Standard dial), 'Contrast' (4), 'Ring type' (1), 'Distinct rings in use' (None), 'Aux. ringer' (unchecked), 'Business name' (Business name 1), 'Long name' (longname), and 'Send long name' (unchecked). There are also fields for 'Call log options' (No one answered), 'Log space' (0), 'Available log space' (3000), and 'Hotline' (None).

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**.

DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Answer	Fwd Delay	Fwd Busy	Fwd All
22255	1140E/2004/2007/2050/221x	chau	0234						
22259	1140E/2004/2007/2050/221x	dat	0236						
22260	1140E/2004/2007/2050/221x	22260	0244						
22261	1140E/2004/2007/2050/221x	22261	0237						
22262	1140E/2004/2007/2050/221x	22262	0245						
22263	1120E/2002	22263	0238	9134400059	22263	77777		77777	
22264	1140E/2004/2007/2050/221x	22264	0246	9134400061	22264				
22265	1140E/2004/2007/2050/221x	22265	0239	9134400150	22265	96139675279			
22441	1120E/2002	22441	0235	22441					
22524	Analog	22524	4001	9134404664	22524	22301			
22525	Analog	22525	4002	22525					
22526	Analog	22526	4003						
22527	Analog	22527	4004						
22528	Analog	22528	4005						
22529	Analog	22529	4006						
22530	Analog	22530	4007						
22531	Analog	22531	4008						
22624	T7316/M7310	22624	2001	9134404975	22624	96139675279			

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**.

Line	Trunk Type	Name	Control Set	Line Type	Prime Set	Pub. Received #	Priv. Received #	Distinct Ring
981	Target line	Line981	20224	Public	22231			None
982	Target line	Line982	20224	Public	22231			None
983	Target line	Line983	20224	Public	22231			None
984	Target line	Line984	20224	Public	22231			None
985	Target line	Line985	20224	Public	22231			None
986	Target line	Line986	20224	Public	22231			None
987	Target line	Line987	20224	Public	22231			None
988	Target line	Line988	20224	Public	22231			None
989	Target line	Line989	20224	Public	22231			None
990	Target line	Line990	20224	Public	22231			None
991	Target line	Line991	20224	Public	22231			None
992	Target line	Line992	20224	Public	22231			None
993	Target line	Line993	20224	Public	22231			None
994	Target line	Line994	20224	Public	22231	9134400059	22349	None
995	Target line	Line995	22263	Public	22263	9134404975	22624	None
996	Target line	Line996	22263	Public	22263	9134404664	22524	None
997	Target line	Line997	22263	Public	22263	9134400150	22265	None
998	Target line	Line998	22263	Public	22263	9134400061	22264	None
999	Target line	Line999	22263	Public	22263		22263	None

Details for Line: 998

Preferences Assigned DNs

Aux. ringer: ☐ If Busy: **Busy tone**

Distinct rings in use: None Voice message center: 1

Redirect to:

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**.

The screenshot displays the Avaya configuration interface. On the left is the 'Task Navigation Panel' with a tree view containing categories like System, Administrator Access, Resources, Telephony, Global Settings, Sets, Templates, Active Sets, Active Applicatic, Inactive DNs, All DNs, Lines, Active Physical L, Active VoIP Line, Target Lines, Inactive Lines, All Lines, Loops, Scheduled Services, Dialing Plan, Ring Groups, Call Security, Hospitality, Hunt Groups, Call Detail Recording, Call Recording, Data Services, and Applications. The 'Target Lines' table is the main focus, listing lines 980 through 999. Line 998 is selected. Below the table are buttons for 'Copy', 'Paste...', and 'Renumber'. At the bottom, the 'Details for Line: 998' panel is open, showing the 'Preferences' tab. In this tab, the 'If Busy:' dropdown is set to 'To prime'. Other fields include 'Aux. ringer:' (unchecked), 'Distinct rings in use:' (set to 'None'), 'Voice message center:' (set to '1'), and 'Redirect to:' (a dropdown menu).

Line	Trunk Type	Name	Control Set	Line Type	Prime Set	Pub. Received #	Priv. Received #	Distinct Ring
980	Target line	Line980	20224	Public	22231			None
981	Target line	Line981	20224	Public	22231			None
982	Target line	Line982	20224	Public	22231			None
983	Target line	Line983	20224	Public	22231			None
984	Target line	Line984	20224	Public	22231			None
985	Target line	Line985	20224	Public	22231			None
986	Target line	Line986	20224	Public	22231			None
987	Target line	Line987	20224	Public	22231			None
988	Target line	Line988	20224	Public	22231			None
989	Target line	Line989	20224	Public	22231			None
990	Target line	Line990	20224	Public	22231			None
991	Target line	Line991	20224	Public	22231			None
992	Target line	Line992	20224	Public	22231			None
993	Target line	Line993	20224	Public	22231			None
994	Target line	Line994	20224	Public	22231	9134400059	22349	None
995	Target line	Line995	22263	Public	22263	9134404975	22624	None
996	Target line	Line996	22263	Public	22263	9134404664	22524	None
997	Target line	Line997	22263	Public	22263	9134400150	22265	None
998	Target line	Line998	22263	Public	22263	9134400061	22264	None
999	Target line	Line999	22263	Public	22263		22263	None

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...** button (not shown). **Figure 38** shows a **MeetMe Conference Bridge** has been enabled for DN 22264.

The screenshot displays the Avaya Configuration Manager interface. On the left is the 'Task Navigation Panel' with a tree view containing categories like System, Resources, Telephony, and Lines. The 'Active Sets' section is expanded under 'Telephony'. The main area shows a table of 'Active Sets' with columns for DN, Model, Name, Port, Pub. OLI, Priv. OLI, Fwd No Answer, Fwd Delay, Fwd Busy, and Fwd All. Row 22264 is highlighted. Below the table, the 'Details for DN: 22264' section is visible, showing the 'MeetMe Conferencing' tab. Under this tab, the 'Class of Service' is set to '10'. At the bottom of this section, the 'Remove MeetMe Conference Bridge' button is highlighted with a red box.

DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Answer	Fwd Delay	Fwd Busy	Fwd All
22221	1140E/2004/2007/2050/221x	22221	0242			22301	4	22301	
22223	1140E/2004/2007/2050/221x	22223	0241				N/A	22301	
22225	Analog	22225	1002				N/A		
22226	Analog	22226	1003				N/A		
22227	Analog	22227	1004				N/A		
22228	Analog	22228	1005			22301	4	22301	
22229	Analog	22229	1006				N/A		
22230	Analog	22230	1007				N/A		
22231	Analog	22231	1001			22301	3		
22233	1140E/2004/2007/2050/221x	22233	0233			22301	4	22301	
22254	1140E/2004/2007/2050/221x	22254	0243				N/A		
22255	1140E/2004/2007/2050/221x	chau	0234				N/A		
22259	1140E/2004/2007/2050/221x	dat	0236				N/A		
22260	1140E/2004/2007/2050/221x	22260	0244				N/A		
22261	1140E/2004/2007/2050/221x	22261	0237				N/A		
22262	1140E/2004/2007/2050/221x	22262	0245			77777	2	77777	
22263	1120E/2002	22263	0238	9134400059	22263		N/A		
22264	1140E/2004/2007/2050/221x	22264	0246	9134400061	22264	22263	4	22265	

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.

The screenshot displays the Avaya Management System interface. On the left, the 'Task Navigation Panel' shows a tree structure with 'Configuration' selected. Under 'Configuration', 'Applications' is expanded, and 'MeetMe Conferencing' is highlighted. The main panel, titled 'MeetMe Conferencing', has two tabs: 'Configuration' and 'Conference Bridges'. The 'Conference Bridges' tab is active, showing a table with two columns: 'Access Code (DN)' and 'Class of Service'. The table contains four rows of data, with the first row highlighted in blue.

Access Code (DN)	Class of Service
22221	16
22222	16
22255	16
22264	1

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**.

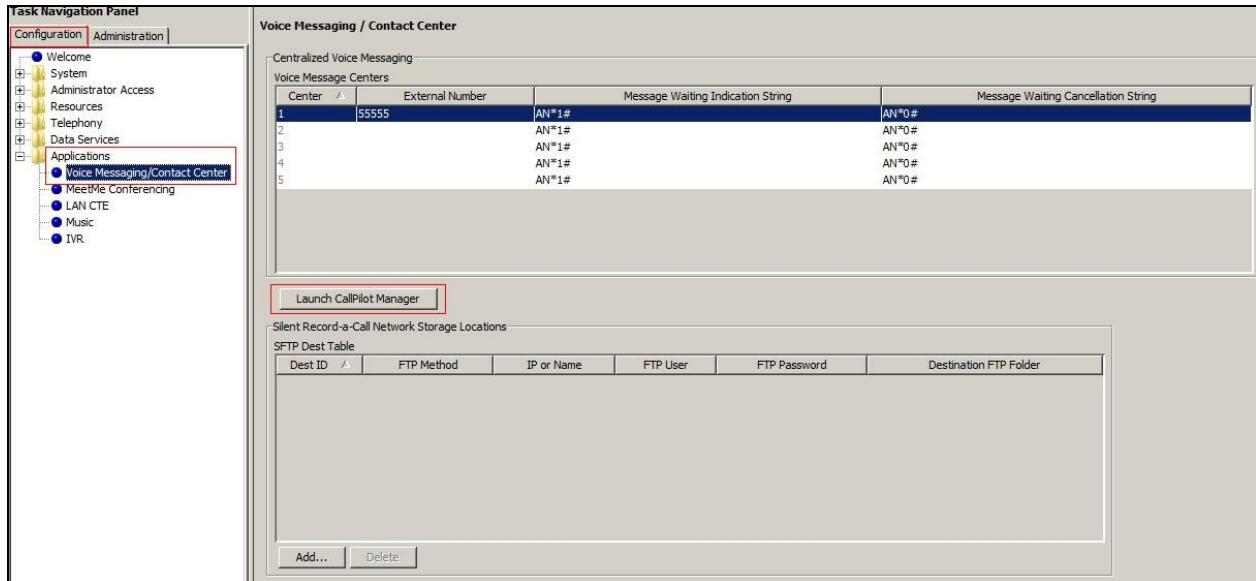


Figure 40 – Launch CallPilot Manager

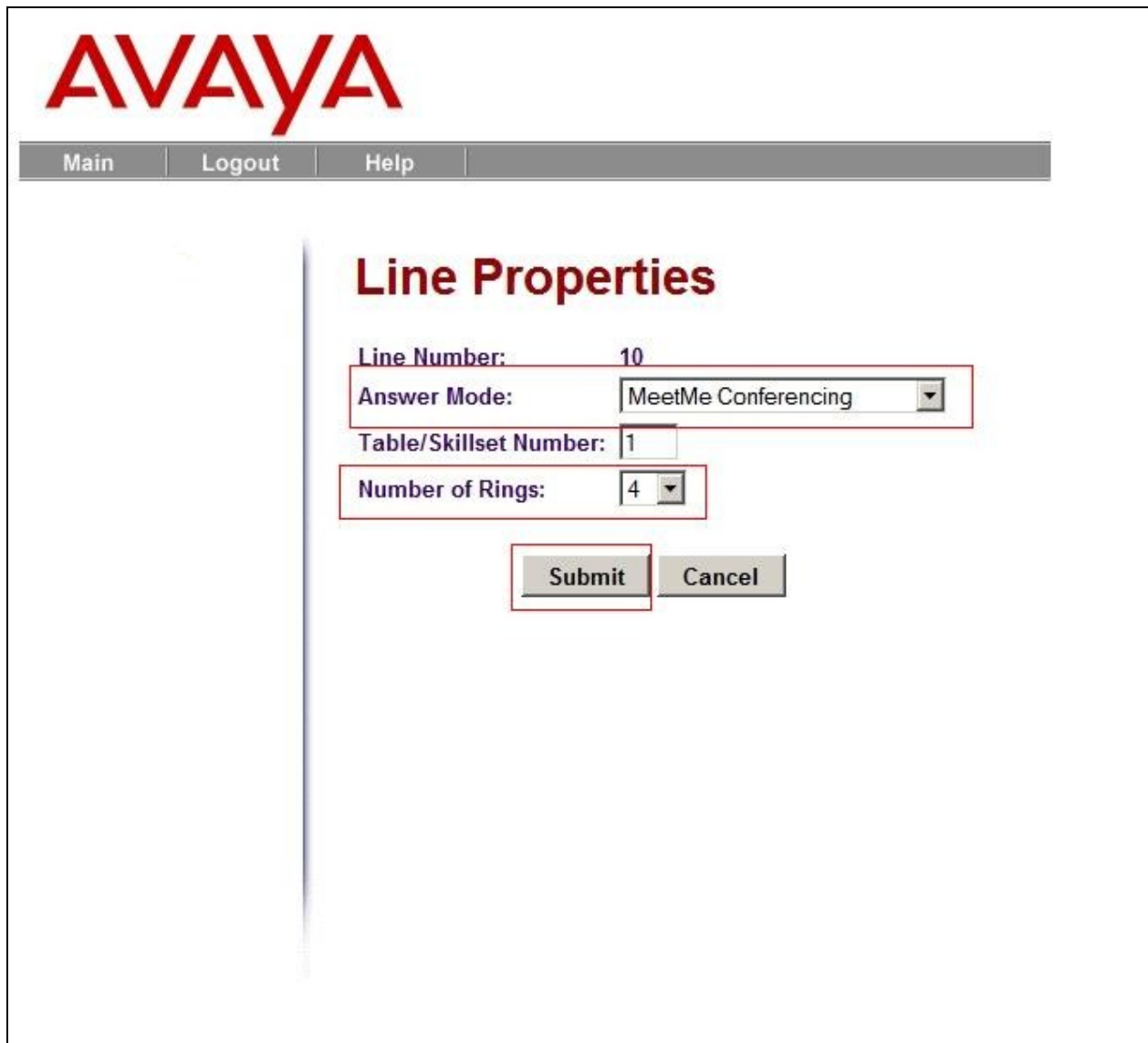
e) The web browser displays **CallPilot Manager**. Then select **Auto-Attendant > Lines Administration** as shown in **Figure 41**. Scroll down to line 10 than click on link **Change**.

The screenshot shows the Avaya CallPilot Manager interface. The top navigation bar includes 'Main', 'Logout', and 'Help'. The sidebar on the left contains several categories: 'Mailbox Administration', 'Custom Call Routing', 'Networking', 'Contact Center', 'Reports', 'Configuration', and 'Operations'. Under 'Mailbox Administration', 'Auto-Attendant' is selected, and 'Lines Administration' is highlighted. The main content area is titled 'Lines Administration' and displays a table with 20 lines. The table has columns for 'Line', 'Answer Mode', 'Table/Skillset', 'Rings', and 'Command'. Line 10 is highlighted, and its 'Change' link is circled.

Line	Answer Mode	Table/Skillset	Rings	Command
1	Auto-Attendant	Table 1	10	Change
2	Auto-Attendant	Table 1	10	Change
3	Auto-Attendant	Table 1	10	Change
4	Auto-Attendant	Table 1	10	Change
5	Auto-Attendant	Table 1	10	Change
6	Auto-Attendant	Table 1	10	Change
7	Auto-Attendant	Table 1	10	Change
8	Auto-Attendant	Table 1	10	Change
9	No	---	0	Change
10	No	---	0	Change
11	No	---	0	Change
12	No	---	0	Change
13	No	---	0	Change
14	No	---	0	Change
15	No	---	0	Change
16	No	---	0	Change
17	No	---	0	Change
18	No	---	0	Change
19	No	---	0	Change
20	No	---	0	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).



The screenshot shows the Avaya web interface. At the top is the red Avaya logo. Below it is a navigation bar with links for 'Main', 'Logout', and 'Help'. The main content area is titled 'Line Properties'. It contains several form fields: 'Line Number' with the value '10', 'Answer Mode' with a dropdown menu set to 'MeetMe Conferencing', 'Table/Skillset Number' with the value '1', and 'Number of Rings' with a dropdown menu set to '4'. At the bottom of the form are two buttons: 'Submit' and 'Cancel'. Red rectangular boxes are drawn around the 'Answer Mode' dropdown, the 'Number of Rings' dropdown, and the 'Submit' button to highlight the required configuration steps.

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**.

The screenshot displays the Avaya configuration interface. On the left is the 'Task Navigation Panel' with a tree view containing 'System', 'Administrator Access', 'Resources', 'Telephony', 'Global Settings', 'Sets', 'Templates', 'Active Sets', 'Active Application DNs', 'Inactive DNs', 'All DNs', 'Lines', 'Active Physical Lines', 'Active VoIP Lines', 'Target Lines', 'Inactive Lines', 'All Lines', 'Loops', 'Scheduled Services', 'Dialing Plan', 'Ring Groups', 'Call Security', 'Hospitality', 'Hunt Groups', 'Call Detail Recording', 'Call Recording', 'Data Services', and 'Applications'. The 'Active Sets' sub-tab is selected under 'Sets'. The main area shows a table of active sets with columns: DN, Model, Name, Prime Line, Intercom Keys, Control Set, First Display, and Auto Called ID. The row for DN 22264 is highlighted. Below the table, the 'Details for DN: 22264' section is visible, with the 'Capabilities' sub-tab selected. In the 'Capabilities' sub-tab, the 'Allow redirect' and 'Redirect ring' checkboxes are checked.

DN	Model	Name	Prime Line	Intercom Keys	Control Set	First Display	Auto Called ID
22229	Analog	22229	I/C	N/A	22231	Name	<input type="checkbox"/>
22230	Analog	22230	I/C	N/A	22231	Name	<input type="checkbox"/>
22231	Analog	22231	I/C	N/A	22231	Name	<input type="checkbox"/>
22233	1140E/2004/2007/2050/221x	22233	I/C	2	22231	Name	<input checked="" type="checkbox"/>
22254	1140E/2004/2007/2050/221x	22254	I/C	2	22231	Name	<input type="checkbox"/>
22255	1140E/2004/2007/2050/221x	chau	I/C	2	22231	Name	<input type="checkbox"/>
22259	1140E/2004/2007/2050/221x	dat	I/C	2	22231	Name	<input checked="" type="checkbox"/>
22260	1140E/2004/2007/2050/221x	22260	I/C	2	22231	Name	<input checked="" type="checkbox"/>
22261	1140E/2004/2007/2050/221x	22261	I/C	2	22231	Name	<input checked="" type="checkbox"/>
22262	1140E/2004/2007/2050/221x	22262	I/C	2	22231	Name	<input type="checkbox"/>
22263	1120E/2002	22263	I/C	2	22263	Name	<input checked="" type="checkbox"/>
22264	1140E/2004/2007/2050/221x	22264	I/C	2	22263	Name	<input checked="" type="checkbox"/>
22265	1140E/2004/2007/2050/221x	22265	I/C	2	22231	Name	<input type="checkbox"/>
22441	1120E/2002	22441	I/C	2	22231	Name	<input checked="" type="checkbox"/>
22524	Analog	22524	I/C	N/A	22231	Name	<input type="checkbox"/>
22525	Analog	22525	I/C	N/A	22231	Name	<input type="checkbox"/>
22526	Analog	22526	I/C	N/A	22231	Name	<input type="checkbox"/>
22527	Analog	22527	I/C	N/A	22231	Name	<input type="checkbox"/>

Details for DN: 22264

Capabilities | SWCA Call Group | Preferences | IP Terminal Details | Button Programming Table | Button Programming | User Speed Dial

Handsfree: Auto | HF answerback: ☒ | Allow redirect: ☒
Pickup group: | DND on Busy: ☐ | Redirect ring: ☒
Page zone: 1 | Paging: ☒ | Silent monitor supervisor: ☐
Direct dial: 1 | Auto hold for incoming page: ☐ |
Intrusion protection level: None | Priority call: ☐ | Auto hold: ☒

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

Task Navigation Panel

Configuration Administration

Active Sets

Line Access Capabilities and Preferences Restrictions

DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Answer	Fwd Delay	Fwd Busy	Fwd All
22255	1140E/2004/2007/2050/221x	chau	0234			N/A			
22259	1140E/2004/2007/2050/221x	dat	0236			N/A			
22260	1140E/2004/2007/2050/221x	22260	0244			N/A			
22261	1140E/2004/2007/2050/221x	22261	0237			N/A			
22262	1140E/2004/2007/2050/221x	22262	0245			N/A			
22263	1120E/2002	22263	0238	9134400059	22263	77777	2	77777	
22264	1140E/2004/2007/2050/221x	22264	0246	9134400061	22264	961396750001	4	961396750002	961396750003
22265	1140E/2004/2007/2050/221x	22265	0239	9134400150	22265	96139675279	4		
22441	1120E/2002	22441	0235	22441		N/A			
22524	Analog	22524	-4001	9134404664	22524	22301	2	22301	
22525	Analog	22525	-4002	22525	22525	77777	2	77777	
22526	Analog	22526	-4003			N/A			
22527	Analog	22527	-4004			N/A			
22528	Analog	22528	-4005			N/A			
22529	Analog	22529	-4006			N/A			
22530	Analog	22530	-4007			N/A			
22531	Analog	22531	-4008			N/A			
22624	T7316/M7310	22624	2001	9134404975	22624	96139675279	2	22301	22263

Copy Paste... Renumber...

Details for DN: 22264

Line Assignment Line Pool Access Answer DN's MeetMe Conferencing

Assigned Lines

Line	Appearance Type	Appearances	Caller ID Set	Vmsg Set	Priv. Received #	Pub. Received #
998	Appr8Ring	1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	22264	9134400061

Add... Delete

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.

Line	Direction	Start Time	User	State	Duration	Number and Name
1 - Line001	Incoming	Fri May 6 10:58:04 2011	22264 - 22264	Connected	00:00:31	6139675258 - Unavailable DNIS: 9134...
2 - Line002		Mon Apr 25 11:32:42 2011	22263 - 22263	Idle		6139675279 - Unavailable DNIS: 9134...
7 - Line007		Fri Apr 29 16:53:23 2011	L1 - ?	Idle		6139675279
8 - Line008	Outgoing	Fri May 6 10:58:15 2011	22263 - 22263	Connected	00:00:07	6139675279

Figure 44: VoIP Line monitoring

- Use BCM450 Monitor to verify RTP

Figure 45 shows an active SIP call with RTP G.711

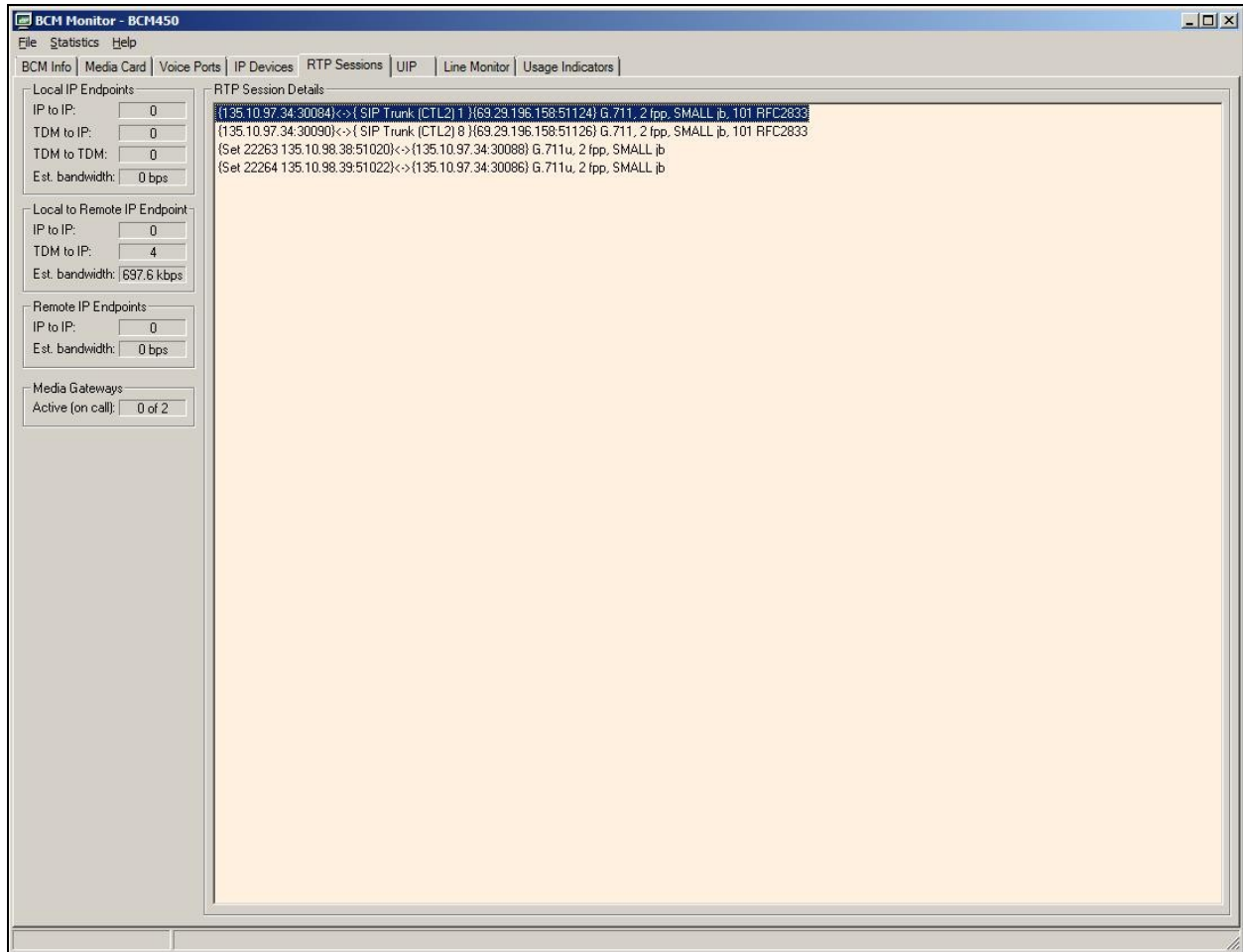


Figure 45: RTP monitoring

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.

No.	Time	Source	Destination	Protocol	Info
11	2.667086	135.10.97.34	69.29.196.158	SIP	Request: OPTIONS sip:anonymous@69.29.196.158;transport=udp
12	2.732799	69.29.196.158	135.10.97.34	SIP	Status: 200 OK
46	11.820414	69.29.196.158	135.10.97.34	SIP/SDP	Request: INVITE sip:9134400061@135.10.97.34:5060, with session description
47	11.848845	135.10.97.34	69.29.196.158	SIP	Status: 100 Trying
57	12.138108	135.10.97.34	69.29.196.158	SIP	Status: 180 Ringing
80	13.966590	135.10.97.34	69.29.196.158	SIP/SDP	Status: 200 OK, with session description
97	14.248041	69.29.196.158	135.10.97.34	SIP	Request: ACK sip:9134400061@135.10.97.34:5060;transport=udp
4639	28.359735	135.10.97.34	69.29.196.158	SIP/SDP	Request: INVITE sip:6139675279@69.29.196.158;transport=udp, with session description
4708	28.377500	135.10.97.34	69.29.196.158	SIP	Request: OPTIONS sip:anonymous@69.29.196.158;transport=udp
4729	28.642472	69.29.196.158	135.10.97.34	SIP	Status: 200 OK
4797	28.866187	135.10.97.34	69.29.196.158	SIP/SDP	Request: INVITE sip:6139675279@69.29.196.158;transport=udp, with session description
4884	29.147425	69.29.196.158	135.10.97.34	SIP	Status: 100 Trying
6582	34.655075	69.29.196.158	135.10.97.34	SIP/SDP	Status: 183 Session Progress, with session description
8147	37.810763	69.29.196.158	135.10.97.34	SIP/SDP	Status: 200 OK, with session description
8158	37.823670	135.10.97.34	69.29.196.158	SIP	Request: ACK sip:6139675279@69.29.196.158;transport=udp
18076	57.485776	135.10.97.34	69.29.196.158	SIP	Request: OPTIONS sip:anonymous@69.29.196.158;transport=udp
18107	57.548220	69.29.196.158	135.10.97.34	SIP	Status: 200 OK
23342	71.892504	69.29.196.158	135.10.97.34	SIP	Request: OPTIONS sip:135.10.97.34:5060
25351	71.909047	135.10.97.34	69.29.196.158	SIP	Status: 200 OK
32170	85.395867	135.10.97.34	69.29.196.158	SIP	Request: OPTIONS sip:anonymous@69.29.196.158;transport=udp
32204	85.457875	69.29.196.158	135.10.97.34	SIP	Status: 200 OK
34112	89.283058	69.29.196.158	135.10.97.34	SIP	Request: BYE sip:9134400059@135.10.97.34:5060;transport=udp
34115	89.294292	135.10.97.34	69.29.196.158	SIP	Status: 200 OK
34436	90.308474	69.29.196.158	135.10.97.34	SIP	Request: BYE sip:9134400061@135.10.97.34:5060;transport=udp
34439	90.319904	135.10.97.34	69.29.196.158	SIP	Status: 200 OK

Frame 46: 977 bytes on wire (7816 bits), 977 bytes captured (7816 bits)

Ethernet II, Src: Nortel_01:b4:42 (00:17:65:01:b4:42), Dst: NortelNe_d3:51:03 (00:21:ef:d3:51:03)

Internet Protocol, Src: 69.29.196.158 (69.29.196.158), Dst: 135.10.97.34 (135.10.97.34)

User Datagram Protocol, Src Port: sip (5060), Dst Port: sip (5060)

Session Initiation Protocol

```

0000  00 21 e1 d3 51 03 00 17 65 01 b4 42 08 00 45 00  .!..Q...e..B...E.
0010  01 c3 00 00 00 00 6e 11 57 42 45 1d c4 9e 87 0a  .....f..wBE....
0020  61 22 13 c4 13 c4 03 af c8 c5 49 4e 56 49 54 45  a"......INVITE
0030  20 73 69 70 3a 39 31 33 34 34 30 30 30 36 31 40  sip:9134400061@
0040  31 33 35 2e 31 30 2e 29 37 2e 33 34 3a 35 30 36  135.10.97.34:506
0050  30 20 53 49 50 2f 32 2e 30 0d 0a 56 69 61 3a 20  0 SIP/2.0, O.Via:
0060  53 49 50 2f 32 2e 30 2f 55 44 50 20 36 39 2e 32  SIP/2.0/UDP 69.2
0070  39 2e 31 39 36 2e 31 35 38 3a 33 30 36 30 3b 62  9.196.158;S:5060;b
0080  77 61 e1 c2 e8 3d 7a 30 68 47 34 d7 4b 66 6a 63  ranch=70;hcfhxfn

```

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.*

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