



Avaya BCM Solutions Test Lab

Business Communication Manager BCM 50 and BCM450 Release 6.0 Configuration Guide for Verizon Business SIP Trunking

Issue 1.1

Abstract

These Application Notes describe guidelines for using Session Initiation Protocol (SIP) trunking between the Verizon Business IP Trunk SIP Trunk Service Offer and an Avaya BCM50 or BCM450 Release 6.0.

The Verizon Business IP Trunk service offer referenced within these Application Notes is designed for business customers. The service enables local and long distance PSTN calling via standards-based SIP trunks directly, without the need for additional TDM enterprise gateways or TDM cards and the associated maintenance costs.

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Application Notes
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1.0 Introduction

This document is intended to provide information to installers configuring a BCM50 or a BCM450 Release 6.0 for SIP trunk inter-working with the Verizon Business IP network. The information in this chapter applies to both the BCM50 and the BCM450 platforms running Avaya Business Communications Manager 6.0 (Avaya BCM 6.0). This guide provides information on how to configure the BCM for:

- global regions
- SIP trunk inter-working with the Verizon Business IP network
- Contact Center capability on the BCM

1.1 Audience

This guide is directed to installers who configure Avaya BCM systems SIP trunks with the Verizon Business IP Trunk SIP Trunk Service Offer. To use this guide, you must

- be an authorized Avaya BCM installer or administrator within your organization
- know basic Avaya BCM terminology
- be knowledgeable about telephony and IP networking technology

For more information about how to configure and assign telephony features, and provide basic programming for the Avaya BCM, see Avaya Business Communications Manager 6.0 Configuration — Telephony (NN40170-502) for Release 6.0
<https://support.avaya.com/css/P8/documents/100122087>.

For more information about network planning information, planning and configuration prerequisites, and planning checklists, see Avaya Business Communications Manager 6.0 Planning and Engineering (NN40170-200) <https://support.avaya.com/css/P8/documents/100122271>.

For more information about upgrading to BCM50 6.0 or BCM450 6.0 see the Avaya Business Communications Manager 6.0 Upgrade Guide <http://support.avaya.com/css/P8/documents/100122469>

For all other information about Avaya BCM Installation, Migrations, Upgrades & Configurations, see documents at
<https://support.avaya.com/css/Products/P0863/Installation,%20Migrations,%20Upgrades%20&%20Configurations>

For all other information about Avaya BCM Customer and Partner Training, see
<http://www.avaya.com/learning>

For all other information about Avaya BCM Certification for Business Partners, see
<https://support.avaya.com/certification>

1.2 Prerequisites

Before you complete the following procedures, review the following prerequisites.

- Ensure the BCM initial installation and configuration is complete.
- Ensure all applicable keycodes are purchased and applied.
- Ensure all required MBMs are installed.

- Ensure market profile is selected.

1.3 Document Change History

The original version of this document was created from the “Business Communication Manager BCM 50 and BCM450 Release 5.0 Configuration Guide for Verizon Business SIP Trunking Issue 1.1” available at:

<http://support.avaya.com/css/P8/documents/100100648>

Date	Version	Summary of Changes
September 23, 2010	0.01	Original publication
February 15, 2011	0.02	Minor updates
March 30, 2011	0.03	Minor updates
May 06, 2011	0.04	Minor updates
November 17, 2011	1.0	Approved
March 15, 2012	1.1	Updated configuration steps without use of a template.

1.4 Support

1.4.1 Avaya

For technical support on the Avaya products described in these Application Notes visit

<http://support.avaya.com>.

1.4.2 Verizon

For technical support on Verizon Business IP Trunk service offer, visit the online support site at

<http://www.verizonbusiness.com/us/customer/>.

2.0 System Software / Loadware

To achieve successful interoperability between the BCM and Verizon SIP Trunking, the various network elements must be running the version of software as shown below:

System	Platform	Firmware
BCM 50	All platforms	Release 6.0 plus smart update BCM050.R600.SU.System-006
BCM450	All platforms	Release 6.0 plus smart update BCM450.R600.SU.System-006
BCM Phones	All supported phones	As provided by smart update

Table 1 Validated Equipment and Software

2.1 Hardware Bill of Materials tested

Part Number	Description
NTC06601BUE6	BCM450 6.0 STD NA PCord Bund
AL4500E14-E6	4548GT-PWR (NA PC)
NT5B16AAAJE5	BCM GASM8 MBM
NTC02008KC	BCM450 VM 1Se eAuth
NTC02009KC	BCM450 UM 1Se eAuth
NTC02024KC	BCM450 SIP Gateway 1Tr eAuth
NTC02025KC	BCM450 IP Tele 1CI EndPt eAuth
NTC02031KC	BCM450 FindFollow Me - 1 eAuth
NTC03110SYE6	BCM450 Capacity Exp Card
NTYS02BBE6	1110 - Eng keys no ps
NTYS03BFE6	1120E - Eng keys no ps
NTYS05BFE6	1140E - Eng keys no ps
NTYS18BC70E6	1210 - Eng keys no ps
NTYS19BC70E6	1220 - Eng keys no ps
NTYS20BC70E6	1230 - Eng keys no ps

Hardware specifications can be found at the following link on the Support Portal.

<http://portal.avaya.com/ptlWeb/products/P0633/AllCollateral>

3.0 Features

3.1 Capacity

BCM450 with the CEC installed supports 130 SIP trunks or 48 without the CEC installed. The BCM50 supports 12 SIP trunks.

For more information about capacity, see the Avaya Business Communications Manager Release 6.0 System Overview at <http://support.avaya.com/css/P8/documents/100121908>

3.2 Features Supported

3.2.1 SIP Trunking

The following are capabilities tested and validated:

- Security
 - Layer 2 IPSec Authentication
- DNS SRV
 - Service Protocols/Port Adherence
- Inbound Calls
 - Inbound Call Loop Avoidance
 - Inbound call with originator (PSTN) release
 - Inbound call with terminator (CPE) release
 - Inbound call - Hang-up during Ring phase
 - Inbound Call - customer phone not registered/online
 - Inbound Calling Line Identification (Caller-ID)
 - Inbound Call Waiting
 - Inbound FAX
 - Inbound Call from PSTN with Privacy Restricted
 - Inbound call - User Busy
 - Inbound Call - Ring No Answer Timer Expire
 - Inbound Call - Long Call Duration
- Outbound Calls
 - Outbound call with Originator (CPE) release
 - Outbound call with Terminator (PSTN) release
 - Outbound call - Hangup during ring phase
 - Outbound 1+10digit call
 - Outbound International Call

- Outbound 311 Non Emergency call
- Outbound 555-1212 Directory Assistance
- Outbound 411 Directory Assistance
- Outbound 1411 Directory Assistance
- Outbound 711 Telephone Relay Services (Hearing Impaired)
- 911 Emergency Service
- Outbound 511 Information Line
- Outbound Toll- Free Call
- Operator assistance (0+ Local)
- Operator assistance (0+ Toll)
- Operator assistance (0 Minus)
- Operator assistance (00 Minus)
- Operator assistance (01+ international)
- Outbound FAX
- Outbound Calling Line Identifier (Caller ID)
- Outbound Fast Answer
- Outbound Call to PSTN with Privacy Requested
- Calling Party Number not provisioned
- Premium Call (900)
- Premium Call (976)
- Outbound Call - Long Call Duration
- Outbound Call – User Busy
- Outbound Call – Ring No Answer Timer Expire
- Private Dialing Plan
- Protocols
 - UDP for SIP
 - SDP support (RFC2327)
 - RTP and RTCP support (RFC 3550)
 - SIP Headers
 - 18x Behavior
 - 302 Behavior
 - Diversion Header
 - DTMF RFC 2833— Outbound
 - DTMF RFC 2833— Inbound
 - Offer/Answer with SDP (RFC3264)

- Call Hold (RFC 3264)
 - Media Inactivity
 - FQDN
- Media
 - G.711 ulaw
 - G.729 and G.729a
 - Codec Negotiation
 - Early Media Support
- Diffserv
 - RTP
 - SIP
- Call Transfer
 - Attended Call Transfer Re-INVITE Method
 - Semi-Attended Call Transfer Re-INVITE Method
 - Blind Call Transfer Re-INVITE Method
- Locally chaired Conference Calls
- Ambient Noise fax tone detection

3.2.2 Verizon's Retail VoIP Trunking service

- Security Test Cases
 - Layer-2 IPSec Authentication
- DNS-SRV Test Cases
 - DNS-SRV – Service Protocols/Port Adherence
- Inbound Calls
 - Inbound Call Loop Avoidance verification
 - Inbound Call with Originator (PSTN) release
 - Inbound Call with Terminator (CPE) release
 - Inbound Call—Hang-up during ring phase (Cancel Call)
 - Inbound Call —Customer Phone Not Registered/Online
 - Inbound Calling Line Identification (Caller ID)
 - Inbound Call Waiting Support
 - Inbound FAX
 - Inbound Call from PSTN with Privacy Requested
 - Inbound Call - User Busy

- Inbound Call - Ring No Answer Timer Expire
 - Inbound Call - Long Call Duration
- Outbound Calls
 - Unscreened ANI using Diversion header
 - Unscreened ANI using P-Asserted-Identity
 - Outbound Call with Originator (CPE) Release
 - Outbound Call with Terminator (PSTN) release
 - Outbound Call - Hang up During Ring Phase
 - Outbound 1+10 Digit Call Terminates
 - Outbound International Call
 - Outbound '311' Non Emergency call
 - Outbound 555-1212 Directory Assistance
 - Outbound '411' Directory Assistance
 - Outbound '1411' Directory Assistance
 - Outbound 711 Telephone Relay Services (Hearing Impaired Services)
 - 911 Emergency Service
 - Outbound 511 Information Line
 - Outbound Toll-Free call
 - Operator Services Call (0+Local)
 - Operator Assistance (0+ Toll)
 - Operator Assistance (0 Minus)
 - Operator Assistance (00 Minus)
 - Operator Assistance (01+ international)
 - Outbound FAX
 - Outbound Calling Line Identifier (Caller ID)
 - Outbound Fast Answer
 - Outbound Call to PSTN with Privacy Requested
 - Calling Party Number not provisioned
 - Premium Call (900)
 - Premium Call (976)
 - Outbound Call - Long Call Duration
 - Outbound Call - User Busy
 - Outbound Call - Ring No Answer Timer Expire
 - Private Dialing Plan
- Protocols

- UDP for SIP
- SDP Support (RFC 2327)
- RTP and RTCP (RFC 3550)
- SIP Headers
- '18x' Behavior
- '302' Behavior
- 'Diversion' Header
- DTMF (RFC 2833) Outbound
- DTMF (RFC 2833) Inbound
- Support for Offer/Answer with SDP (RFC 3264)
- Call Hold (RFC 3264)
- Media Inactivity
- Use of FQDN IP Addressing in SIP Messaging
- Media Test Cases
 - G.711 μ -law codec (AVT payload type 0)
 - G.729 and G.729a support
 - Codec Negotiation
 - Early Media Support
- Differentiated Services (DiffServ)
 - RTP media marked with DSCP EF or CS5
 - SIP signaling marked with DSCP AF32 or CS3
- Re-Invite Call Transfer
- Attended Call Transfer
- Semi-Attended Call Transfer
- Blind Call Transfer
- Attended Call Transfer
- Semi-Attended Call Transfer
- Blind Call Transfer
- Locally chaired Conference Calls
- Ambient Noise fax tone detection

3.2.3 Verizon Business's IP Enabled Contact Center

- Security – Layer IPsec Authentication
- OPTIONS Method Request and Response
- IP Toll Free

- Inbound Calls (Verizon Business PSTN to Customer)
 - Inbound Calls with Request-URI Set to Customer's Provisioned URL Address
 - Inbound Call with Originator (PSTN) Release
 - Inbound Call with Terminator (SIP) Release
 - Inbound Call with Disconnect During Ring Phase (Cancel Call)
 - Inbound Call with Customer Phone Not Registered with SIP PBX
 - Inbound Call with Ring No Answer
 - Inbound Call with User Busy
 - Inbound Call with CPN Allowed – Privacy Null
 - Inbound Call with CPN Restricted – Privacy “id”
 - Inbound Call with Long Duration
 - Inbound Call with Proprietary Headers
 - Agent to Agent Proprietary Headers
 - Network Call Redirect (NCR) Testing with no Enhanced Transfer
 - Inbound Call with NCR with Answer
 - Inbound Call with NCR with Ring No Answer
 - Inbound Call with NCR with User Busy
- Outbound Calls (Customer to Verizon Business)
 - Locating IPCC SIP Server (Customer DNS Resolution of Verizon Business VoIP Proxy) – IDA Only
 - Using DNS Resolution
 - Load Sharing
 - Failover
 - Outbound Call with CPN Allowed – Privacy Null
 - Outbound Call with CPN Restricted – Privacy “id”
- Protocols
 - Supported SIP Methods (RFC)
 - UDP for SIP and Long Message Support
 - SIP Ports
 - CPE Must Accept Receipt of Full and Compact Headers
 - x Behavior
 - Behavior
 - Draft-levy-sip-diversion- Diversion Header
 - Call Hold
 - Media Inactivity (Call Hold Long Duration)
- Media

- Verifying Session Description Protocol (SDP)
- SDP Support (RFC)
- Verizon Business SDP Offer – Customer SDP Answer
- Customer SDP Offer – Verizon Business SDP Answer
- Verifying Real Time Protocol (RTP)
- Verifying RTP – Basic Inbound Call
- Verifying RTP – Inbound Call with Enhanced Transfer
- DTMF (RFC)
- Verifying RTP – Phone on Mute
- RTP and RP Support (RFC)
- Early Media
- Conference – DTMF
- IVR
 - Inbound Calls (Verizon Business PSTN to Customer)
 - Inbound Call with Originator (PSTN) Release
 - Inbound Call with Terminator (SIP) Release
 - Inbound Call with Disconnect During Ring Phase (Cancel Call)
 - Inbound Call with Customer Phone Not Registered with SIP PBX
 - Inbound Call with Ring No Answer Timer Expire
 - Inbound Call with User Busy
 - Inbound Call with CPN Allowed – Privacy Null
 - Inbound Call with CPN Restricted – Privacy “id”
 - Inbound Call with Long Duration
 - Agent to Agent Proprietary Headers
 - Network Call Redirect (NCR) Testing
 - Inbound Call with NCR with Answer
 - Inbound Call with NCR with Ring No Answer
 - Inbound Call with NCR with User Busy
 - Inbound Call with Release Link Trunking (RLT)
 - Outbound Calls (Customer to Verizon Business)
 - Protocols
 - Supported SIP Methods (RFC)
 - UDP for SIP and Long Message Support
 - SIP Ports
 - CPE Must Accept Receipt of Full and Compact Headers

- Behavior
- Behavior
- Draft-levy-sip-diversion- Diversion Header
- Call Hold
- Media Inactivity (Call Hold Long Duration)
- Media
 - Verifying Session Description Protocol (SDP)
 - SDP Support (RFC)
 - Verizon Business SDP Offer – Customer SDP Answer
 - Customer SDP Offer – Verizon Business SDP Answer
 - Verifying Real Time Protocol (RTP)
 - Verifying RTP – Basic Inbound Call
 - Verifying RTP – Inbound Call with NCR
 - DTMF (RFC)
 - Verifying RTP – Phone on Mute
 - RTP and RP Support (RFC)
 - Early Media
- IP IVR Custom Transfers
 - Blind – DTMF
 - Attended – DTMF

3.3 Technical Caveats

- The BCM SIP trunking, Contact Center and IVR do not support attended or semi-attended basic or enhanced call transfer using the REFER Method. The BCM supports transfer using INVITE instead of using REFER.
- The BCM does not support failover between two BCMs.
- FAX using T.38 is supported with the following software update applied:
BCM50.R600.SU.System-006

4.0 Overview

Figure 4-1 shows a typical deployment of a SIP trunk between BCM R6 and Verizon Business Network. In the test configuration, there were no NAT between the BCM and Verizon's SIP network. All IP addresses assigned to the BCM as well as BCM IP phones were routable from Verizon's SIP network.

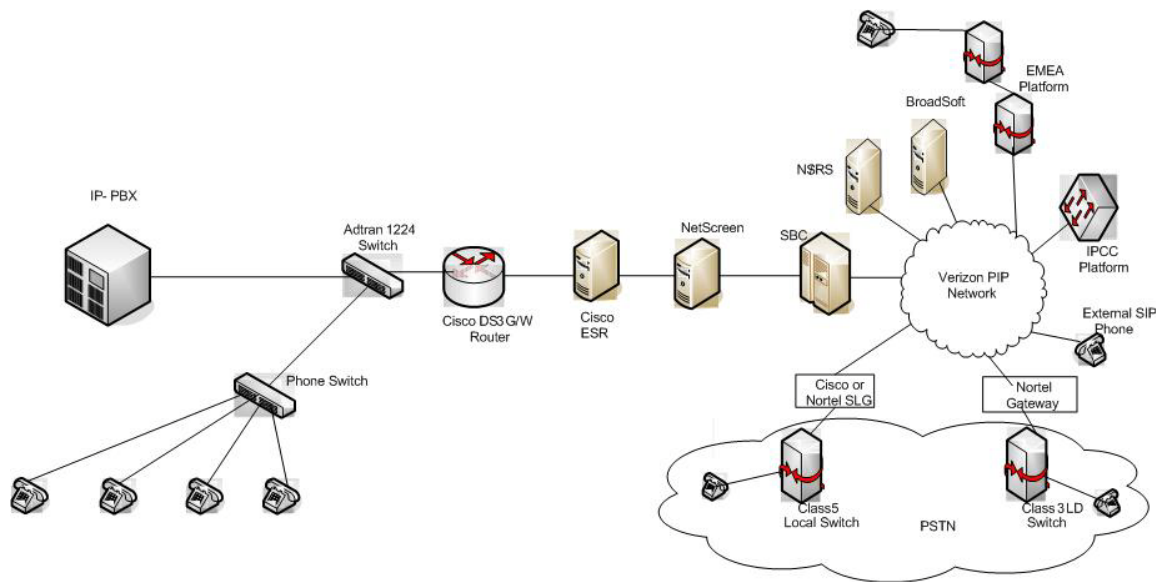


Figure 4-1 SIP Trunking between BCM and Verizon Business Network

5.0 System Configuration

5.1 BCM Configuration for global regions

The System Configuration in this guide is the same for all global regions. The administrator can select the Region of the system during initial system start up using Business Element Manager > Administration > Utilities > Reset > Cold Reset Telephony Services. For more information on selecting a global Region, see the “BCM450 RIs 6.0 System Start Up Task Based Guide” <http://support.avaya.com/css/P8/documents/100125306>

5.2 BCM Configuration for SIP trunking

This section provides procedures for configuring a SIP trunk on BCM RIs. 6.0 to Verizon Business IP network. In order to configure a SIP trunk between BCM and Verizon Business network, do the following using the Business Element Manager:

5.2.1 IP Trunks Settings Configuration

1. Under **Configuration → Resources → IP Trunks → General → IP Trunk Settings**
2. Click on the “**IP Trunks Settings**” tab
3. **Enable Forward redirected OLI:** by selecting “**Last Redirect**”.

5.2.2 SIP Settings Configuration

1. Under **Configuration → Resources → IP Trunks → SIP Trunking →** Click on the “**Global Settings**” tab
2. In the **SIP Settings** section:
 - **Local Domain:** The IP address of the Session Border Controller. Obtain this information from Verizon.
 - **Call Signaling Port:** This port is defined by Verizon. Obtain this information from Verizon
3. In RFC2833 section
 - **Dynamic Payload:** Set this value to 101
4. Leave the rest of parameters under the **Global Settings** tab at the default values.

5.2.3 Create Public SIP Trunk Accounts

1. Navigate to **Configuration → Resources → IP Trunks → SIP Trunking**
2. Select the **Public** tab.
3. Next select the **Accounts** tab.
4. Click on the **Add** button.
5. In the **Add Account** dialog box that pops up, click on **No Template**.
6. In the **Add Account** dialog box that opens up

- a. Provide a unique name for this account in the **Name field**
 - b. Provide a descriptive name for this account in the **Description field**
 - c. SIP domain: The IP address of the Session Border Controller. Obtain this information from Verizon.
 - d. Uncheck the **Registration field**.
7. Click on the **OK** button
 8. Select the Verizon account in the Accounts list.
 9. The **Details of the Accounts** appears at the bottom
 10. Select the **Advanced** tab in the **Details of the Accounts** table.
 11. Uncheck the “**Support 100rel**” checkbox to disable PRACK.
 12. Select the **User Accounts** tab in the **Details of the Accounts** table.
 13. Select the first row in the **User Accounts table**.
 14. Click on the **Modify** button and this should open the **Modify Account** dialog box.
 15. In the **Modify Account** dialog box:
 - Provide a description for this user account in the **Description field**
 - Within the Message Handling section;
 - Set the **CLID Override** value to the Verizon DID phone number.
 - Set the **Display name Override** to the desired display name.
 - The **PAI CLID Override** must be any 10 digit national phone number.
 - Set the **PAI Display name Override** to the same value as the **Display name Override**.
 - Click the **OK** button
 16. Navigate to **Configuration → Telephony→Global Settings→Feature Settings**
 17. Provide a name in the **Business Names** section
 18. Navigate to **Configuration → Telephony→ Sets→Active Sets**
 19. For each active set in the **Active sets** table, change the **Name** column to name of user of that BCM phone

5.2.4 SIP Media Parameters Configuration

1. Navigate to **Configuration → Resources → IP Trunks→ SIP Trunking**
2. Click on the **Media Parameters** tab.
3. In the **Preferred Codecs** section, configure G.729, G.711-uLaw as the first and second preferred codec respectively.
4. In the codec **Settings** section, uncheck **Enable Voice Activity Detection**.
5. In the codec **Settings** section, select 20ms as the payload size for both G.729 and G.711
6. Again in the codec **Settings** section set G.711 as the **Fax transport**.

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7. **Force G.711 for 3.1KHz audio** must be **checked**.
8. **Provide in-band ringback** must be left **unchecked**.

5.2.5 Configuring Analog Sets

1. Under **Configuration → Telephony → Sets → Active Sets**, click on the **Capabilities and Preferences** tab
2. Click on the DN of the first analog set
3. In the Capabilities tab that shows up at the bottom, configure this analog device to receive short tones by checking the checkbox next to Receive Short Tones.
4. In the ATA Settings tab that shows up at the bottom, configure this analog device to 'Modem' and non-fax analog devices as 'Telephone' to allow the use of G.729.
5. Repeat Steps 2 to 4 for each of the analog DNs

5.2.6 Dialing Plan

5.2.6.1 Line Pool Configuration

1. Under **Configuration → Telephony → Dialing Plan**, select **Line Pools**.
2. Select **BlocA**.
3. Click on the **"Add"** button to add DNs of sets that need to access the above line pool.

5.2.6.2 Dial Plan Configuration

1. Under **Configuration → Telephony → Dialing Plan → Public Network**, define the **Public Received** number length. Check with Verizon for the appropriate value.
2. Set the **Public Network Dialing Plan** to **Public (Unknown)**
3. **In the Public Network DN Lengths table,**
 - Enter a DN prefix of 011 and set the length to 14
 - Enter a DN prefix of 0019 and set the length to 14
 - Change the default length to 10
4. Under **Configuration → Telephony → Dialing Plan → Routing**, and select the **Routes** tab
5. Add a route by clicking on the **Add** button.
6. In the **Add Route** dialog box, provide an unused route and click on the **OK** button.
7. The **Dialing Plan – Routing** table will be displayed.
8. Click on the route just created

9. Under the **Use Pool** column, double click to select **BlocA** from the drop down list.
10. Under the **DN Type** column, double click to select Public (**Unknown**) from the drop down list
11. Click on the **Destination Codes** tab.
12. Configure a destination code to route dialed digits by clicking on the **Add** button. Digits that begin with this destination code will be presented to the SIP trunking component on the BCM for routing towards the Service Provider.
13. In the **Add Destination Code** dialog box, enter a numeric number for the destination code and click on the **OK** button.
14. Select the row representing the Destination Code entered in the previous step
15. Under **Normal Route** column, double click and enter the route entered in Step 5.
16. Under the Absorbed Length column, specify the number of digits that will be absorbed before sending the rest of the digits to the service provider.

5.2.7 Configuring Incoming Calls from Verizon to BCM

The BCM must be configured for incoming calls for each DID for any combination of a system DID, multiple individual DID's or one or more group DID's (e.g. for a department). This can be done in one of two ways;

1. The DID assigned to the BCM can be associated with a target line assigned to a group of set(s) and all calls to the DID will be routed those set(s).
2. All calls to the DID assigned to the BCM can be answered by the Auto Attendant (AA) and from there, a DN can be entered to reach a phone on the BCM

5.2.7.1 Assigning DID to BCM Phones for Incoming Call

1. Navigate to **Configuration → Telephony → Lines → Target Line** and click on an unused target line
2. On the selected Target Line, set the "**Pub. Received #**" to the last 4-digits of the Verizon assigned DID.
3. Assign the DN of phones on the BCM that require an appearance on this target line. This will be the phones that will be alerted when call to the Verizon assign DID is received.
 - a. Navigate to **Configuration → Telephony → Lines → Target Line** and click on the Target Line configured in Step 2 above
 - b. Click on the **Assigned DNs** tab
 - c. Click the **Add** button to add the DN of set(s) to this Target Line.

5.2.7.2 Configuring AA to Answer Incoming Calls

Alternatively, the AA on the BCM can be configured to answer incoming calls and then call routed to a target phone on the BCM by entering the extension of the set at the AA prompt. To do this,

1. Navigate to **Configuration → Telephony → Lines → Target Line** and click on an unused target line

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2. On the selected Target Line, set the “**Pub. Received #**” to the last 4-digits of the Verizon assigned DID.
3. Navigate to **Configuration→Application→Voice Messaging/Contact Center**
4. Click on the **Launch CallPilot Manager**.
5. This launches a web browser to the BCM. Log in with the administrator credentials
6. On the left hand navigation menu, click on **Auto Attendant**
7. In the **Line Administration** web page, scroll down to the Target Line configured in Step 1.
8. Under the **Command Column**, click **Change**
9. In the Line Properties web page, select **Auto-Attendant** as the **Answer Mode**
10. Click the **Submit** button.

5.2.8 Giving Access to SIP Trunks

To give access to BCM phones to make outgoing calls across the SIP trunk;

1. Navigate to **Configuration→Telephony→Sets→Active Sets**
2. Click on the Line Access tab
3. Click on the DN of each the registered BCM phones in turn and click on the **Line Pool Access** tab
4. Click on the **Add** button
5. In the **Add Line Pool** dialog box, type **bloca**
6. Click **OK**
7. Repeat steps 3 to 6 for each of active sets on the BCM.

5.3 BCM Configuration for Contact Center

Business Communications Manager utilizes Intelligent Contact Center, which is the evolution of the existing Contact Center capability on the BCM. Contact Center answers calls, then routes the calls to agents in a skillset that most closely matches the needs of the caller.

Calls can be routed based on:

- the origin of the call
- the destination of the call
- or the information entered by the caller

Callers can be given:

- high or low priorities.

Callers can overflow to:

- different groups
- or skillsets of agents
- transfer out of the system
- leave a message

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- hear announcements or informative messages.

For more information about how to configure BCM Contact Center, see BCM RIs 6.0 Contact Center Task Based Guide (NN40011-023) <http://support.avaya.com/css/P8/documents/100125752>

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