



# **Business Communication Manager BCM 50 and BCM450 Release 6.0 Configuration Guide for MTS Allstream SIP Trunking**

**Issue 1.0**

## **Abstract**

This document provides guidelines for configuring a SIP Trunk between a BCM50 or BCM450 Release 6.0 and MTS Allstream SIP Trunking

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# 1.0 Introduction

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This document is intended to provide information to installers configuring a BCM50 or a BCM450 Release 6.0 for SIP trunk inter-working with MTS Allstream. The information provided is specific for the SIP trunk inter-working, unrelated configuration is not considered within this guide. This document assumes that the installer has undergone Avaya approved BCM training and has a working knowledge of BCM installations.

## 1.1 Document Change History

Date	Version	Summary of Changes
September 3, 2010	0.1	Original publication
November 11, 2010	1.0	Issued version 1.0

## 2.0 System Software / Loadware

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To achieve successful interoperability between the BCM and MTS Allstream 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 GA build 1.02.
BCM450	All platforms	Release 6.0 GA build 1.02.
BCM Phones	All supported phones	As provided by release 6.0 GA build 1.02.

**Table 1 Validated Equipment and Software**

## 3.0 Features

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### 3.1 Features Supported

- The following are capabilities tested and validated:
- Basic calls (G711 u-law and G729 both with 20ms packetization)
- Calling line (number) identification presentation
- Calling line (number) presentation restriction
- DTMF (RFC 2833)
- Call hold
- Call transfer (attended and unattended)
- Conference calls
- Call forward
- Call waiting

### 3.2 Technical Caveats

1. BCM Find Me/Follow Me calls cannot be routed over MTS Allstream SIP trunks if the BCM SIP trunk is configured to use G.729 as the preferred codec. When G.711 is configured as the preferred codec, then FMFM calls can be setup successfully using G.711.
2. The CLID override set in the BCM Element Manager SIP trunk for the MTS Allstream account does not appear on Rogers Wireless phones.

# Overview

Figure 0-1 shows a typical deployment of a SIP trunk between BCM R6 and MTS Allstream. In the deployment, a SIP unaware router with NAT and firewall is deployed between the BCM and MTS Allstream.

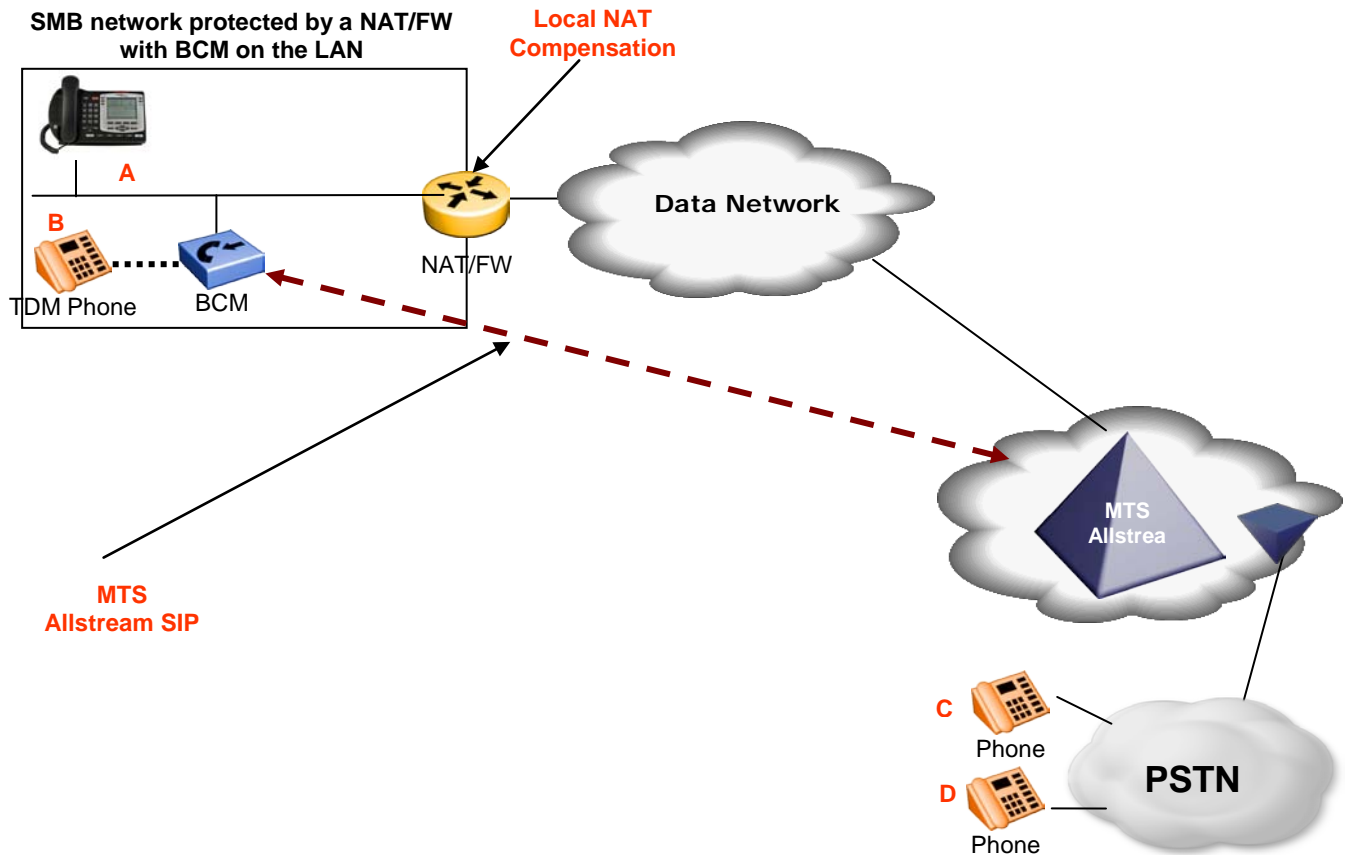


Figure 0-1 SIP Trunking between BCM and MTS Allstream

## 4.0 System Configuration

This section provides procedures for configuring a SIP trunk on BCM RIs. 6.0 to MTS Allstream.

### 4.1 BCM Configuration

In order to configure a SIP trunk between BCM and MTS Allstream, do the following:

#### 4.1.1 SIP Media Parameters Configuration

1. Navigate to **Configuration → Resources → IP Trunks → SIP Trunking**
2. Select the **Media Parameters** tab.
3. In the **Preferred Codecs** section, configure G.729, G.711-uLaw as the first and second preferred codec respectively. If the Find Me/Follow Me feature will be enabled then make G.711-uLaw first in the list as the preferred codec.
4. In the codec **Settings** section,
  - In the codec **Settings** section, select 20ms as the payload size for both G.729 and G.711.
  - Select G.711 for **Fax transport**.
  - **Provide in-band ringback** must be left unchecked.

#### 4.1.2 Importing MTS Allstream Template to BCM

1. Navigate to **Configuration → Resources → IP Trunks → SIP Trunking**
2. Select the **Public** tab.
3. Select the **ITSP Templates** tab
4. Click on the **Import** button
5. Select **Yes** to acknowledge the warning
6. An **Import Files** dialog box will open
7. Click on **Browse** and navigate to the directory containing the MTS Allstream templates
8. Select the MTS Allstream template and click on Select Files
9. Click the OK button to import the template to the BCM

#### 4.1.3 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 **Select Template**.

6. In the **Choose Template** dialog box, choose the MTS Allstream template (MTSAllstream.sipt) and click on the **Select** button.
7. 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. Leave the rest of the parameters at their default values.
8. Click on the **OK** button

#### 4.1.4 Customize the Account

1. Navigate to **Configuration → Resources → IP Trunks → SIP Trunking**
2. Select the **Public** tab
3. Next select the **Accounts** tab
4. Select the MTS Allstream account
5. The **Details of the Accounts** appears at the bottom
6. Select the **User Accounts** tab in the **Details of the Accounts** table.
7. Select the first row in the **User Accounts** table.
8. Click on the **Modify** button and this should open the **Modify Account** dialog box.
9. 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 MTS Allstream 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
10. Navigate to **Configuration → Telephony → Global Settings → Feature Settings**
11. Provide a name in the **Business Names** section
12. Navigate to **Configuration → Telephony → Sets → Active Sets**
13. For each active set in the **Active sets** table, change the **Name** column to name of user of that BCM phone



## 4.2 Dialing Plan

### 4.2.1 Line Pool Configuration

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

### 4.2.2 Dial Plan Configuration

1. Under **Configuration → Telephony → Dialing Plan → Public Network**, define the **Public Received** number length. Check with MTS Allstream for the appropriate value.
2. Set the **Public Network Dialing Plan** to **Public (Unknown)**
3. In the **Public Network DN Lengths table**, configure the DN prefixes and their corresponding lengths.
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. This is the preferred route.
7. The **Dialing Plan – Routing** table will be displayed.
8. Click on the route just created
9. Under the **External Number** column of the newly created route, insert a number that will be appended to the dialed number and presented to the SIP trunking module on the BCM for routing. This number will be stripped off before routing the call to MTS Allstream.
10. Under the **Use Pool** column of the newly created route, double click to select **BlocA** from the drop down list.
11. Under the **DN Type** column of the newly created route, double click to select **Public (Unknown)** from the drop down list.
12. Repeat Steps 5 to 10 to add a backup route. This route will be used if the attempt to route the call via the preferred route fails. In the External Number column for this secondary route, provide a number which is different from that used in Step 9..
13. Click on the **Destination Codes** tab.
14. 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.
15. In the **Add Destination Code** dialog box, enter a numeric number for the destination code and click on the **OK** button.
16. Select the row representing the Destination Code entered in the previous step

17. Under **Normal Route** column, double click and enter the route entered in Step 6.
18. Under the **Absorbed Length** column, specify the number of digits that will be absorbed before pre-pending the configured External number configured in Step 9 and sending the rest of the digits to the SIP trunking component.
19. While the row representing the Destination Code enter in Step 15 is selected, in the **Alternate Routes** table that shows up below, click on an unused schedule
20. In the **Second Route** column of the selected row above, provide the route configure in Step 12.
21. Navigate to **Configuration → Telephony → Scheduled Services**
22. Click on the schedule used in Step 19 above.
23. Define the times when this schedule will be active by providing the start and stop times for each day of the week.
24. In the **Services** section that appears below, select the schedule used in previous steps and do the following:
  - double click the **Routing Svc** column and select **Auto** to active the schedule.
  - Enable overflow by putting a check in the **Overflow** checkbox

### 4.2.3 SIP Trunk Routing table Configuration

1. Navigate to **Configuration → Resources → IP Trunks → SIP Trunking**
2. Select the **Public** tab
3. Click on the **Accounts** tab
4. Select the first MTS Allstream account
5. In the Details for Accounts table below, in the drop down list next to Called Number Characters to absorb select the number of digits that will be absorbed by the SIP trunking component before sending the dialed digits to MTS Allstream. The number selected here should be the number digits in the External Number configured in Step 9
6. Click on the **Routing Table** tab
7. Click on the **Add** button
8. In the **Add Route** dialog box;
  - Provide a descriptive name for the new route in the **Name**.
  - Set the **Destination Digits** to the value provide as the External Number in Step 9 of section 4.2.2
  - In the **ITSP Account** drop down list, select the MTS Allstream account
9. Click **OK**

## 4.2.4 Configuring Incoming Calls from MTS ALLSTREAM to BCM

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

### 4.2.4.1 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 MTS Allstream 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 MTS Allstream 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.

### 4.2.4.2 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
2. On the selected Target Line, set the “**Pub. Received #**” to the last 4-digits of the MTS Allstream 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.

#### 4.2.5 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.

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