Avaya BCM Solutions Test Lab

Business Communication Manager BCM 50 and BCM450 Release 6.0 Configuration Guide for Skype SIP Connect Service

Issue 1.0

Abstract

This document provides guidelines for configuring a SIP Trunk between a BCM50 or BCM450 Release 6.0 and Skype SIP Connect Service

Table of Contents

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 Skype. 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
July 23, 2010 0.1	Original publication
October 1, 2010 1.0	Release

2.0 System Software / Loadware

To achieve successful interoperability between the BCM and Skype 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

3.1 Features Supported

- The following are capabilities tested and validated:
- Basic calls (G711 a-law/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. FAX is not supported by Skype.
2. The Windows Skype client version 4.2.0.158 displays 'Failed Call' when calling a BCM set that is busy.
3. Skype does not support Early Media in SIP.
4. Calls from Skype users appear on BCM as 'Private Number' in the following cases:
    a. Skype user does not have an Online Number
    b. Skype User does not have a configured Caller ID

4.0 Overview

Figure 4-1 shows a typical deployment of a SIP trunk between BCM R6 and Skype. In the deployment, a SIP unaware layer 3 device with NAT and firewall is deployed between the BCM and Skype. The IP address assigned to the BCM and all IP phones is private and since Skype does not support remote NAT compensation, the BCM is configured to compensate for the presence of the local NAT.

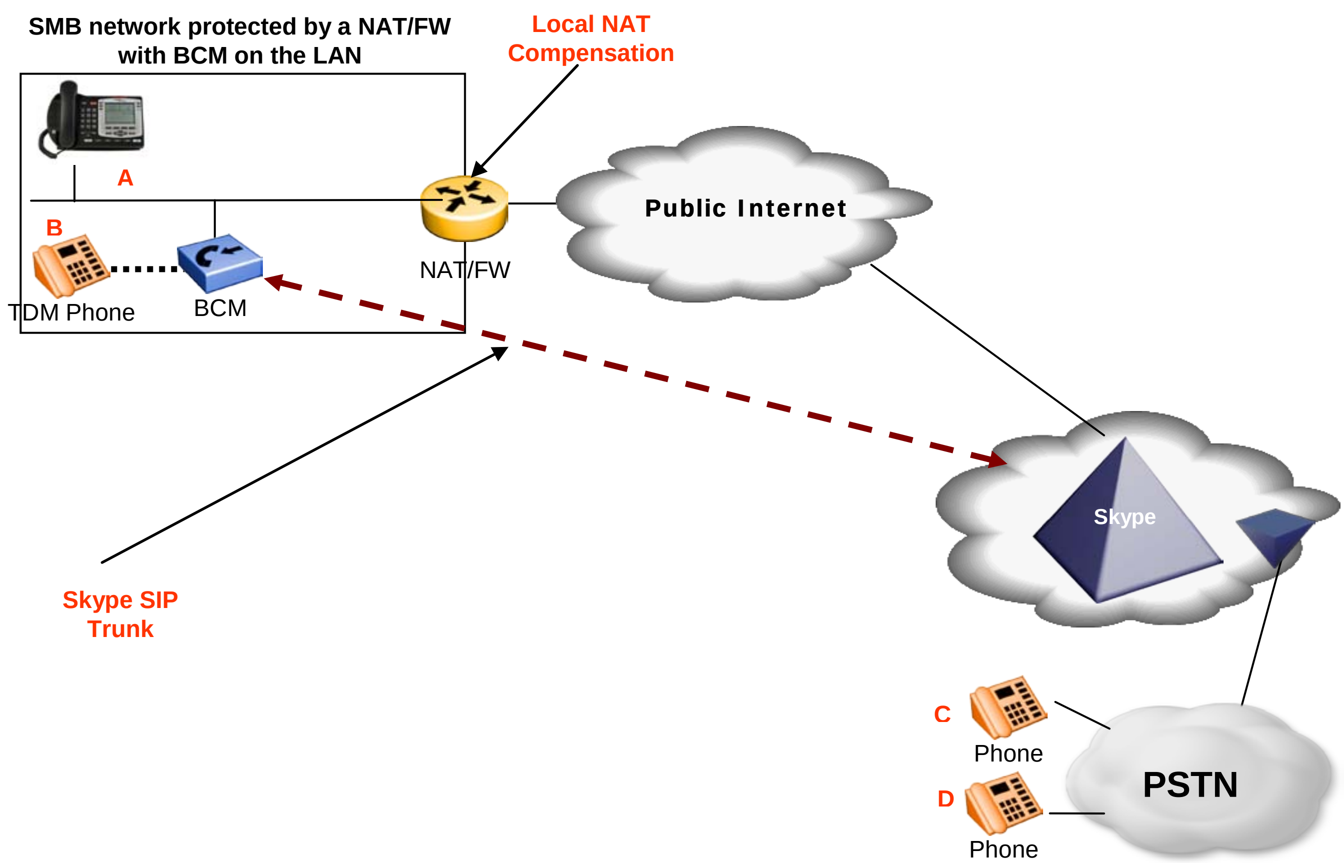

Figure 4-1 SIP Trunking between BCM and Skype

5.0 System Configuration

This section provides procedures for configuring a SIP trunk on BCM Rls. 6.0 to Skype.

5.1 BCM Configuration

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

5.1.1 IP Trunks Settings Configuration

1. Navigate to ***Configuration → Resources → IP Trunks→ General***
2. Click on the "**IP Trunks Settings**" tab
3. **Ignore in-band DTMF in RTP:** by clicking on the check box next to it.

5.1.2 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.
4. In the codec **Settings** section,
    - In the codec **Settings** section, select 20ms as the payload size for both G.729 and G.711
    - FAX is not supported by Skype.
    - **Provide in-band ringback** must be left unchecked.

5.1.3 Importing Skype 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 Skype templates
8. Select the Skype template and click on Select Files
9. Click the OK button to import the template to the BCM

5.1.4 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 one of the Skype templates 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. Enter the Skype user id in the **SIP Username** field.
    d. Enter the Skype user id in the **Auth name** field.
    e. Enter the Skype user password in the **Auth password** field.
    f. Leave the rest of the parameters at their default values.
8. Click on the **OK** button

5.1.5 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 Skype 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
    - Set the **CLID** value equal to CLID of the BCM set, which is a concatenation of Public Network Code and Public OLI. The Public network code is configured in 5.2.2.
    - Within the Message Handling section;

- Set the **CLID Override** value to a Skype user id.
- Set the **PAI CLID Override** to the same value as the **CLID 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.
14. For each active set in the **Active sets** table, cconfigure the **Pub OLI**. CLID for outgoing calls from the set is constructed as: Public network code + Pub. OLI . The Public network code is configured in 5.2.2.

5.1.6 Caller ID Delivery for Outgoing Calls with Skype

Skype Connect uses the following mechanism to deliver Calling Line ID on outgoing PBX calls:

Skype Connect allows a PBX to select the outbound caller ID on a call by call basis using RFC3325 P-Asserted-ID and Privacy:id headers.

Skype Connect llows a business to register their landline DDI's with their SIP profile, the maximum number of landline numbers that can be registered is 99. In order for a business to register their landlines they must have passed our company verification process that can be located in the BCP under account details. The landline numbers must belong to the business and must be located in the same country as the business is registered.

When a business has been verified, any landline number that is registered is inserted into a virtual CLI database that also contains all Online numbers associated to the SIP profile. When the SIP PBX uses the P-Asserted-ID header, Skype check's the content of the P-Asserted-ID header against the users CLI database, if the value's match Skype will then use the number in the P-ID header as the outbound caller id. If the number does not match, Skype will set the outbound caller id as anonymous.

In order to control CallerID for outgoing calls via Skype Connect BCM administrator has to use SIP trunk user account **PAI CLID Override** field. The field can be set to the desired CallerID provided the CallerID is registered via Skype Manager as described above.

If a single SIP trunk user account is used for Skype ITSP Account only a single CLID can be specified for all calls from any extension on the BCM. If there is a need to set the CLID on a per-extension basis or for a group of BCM extensions multiple SIP trunk user accounts have to be created at the Skype ITSP account level.

Each SIP trunk user account is keyed by CLID field. In order for a BCM extension to be associated with a SIP trunk user account the CLID has to match the caller Public OLI (prefixed by Public Network Code ***Configuration → Telephony→Dialing Plan->Public Network->Public Network Settings***).

If multiple SIP trunk User Accounts are associated with the same Skype SIP profile, SIP registration can only be turned on for one of the SIP trunk user accounts associated with the same Skype SIP Profile. SIP Trunk User Accounts are associated with Skype SIP profiles by way of using SIP User credentials of the Skype SIP Profile.

BCM Outgoing Name and Number (ONN) Blocking feature (F819) is compatible with Skype Connect and can be used to block ONN on a call-by-call basis.

5.2 Dialing Plan

5.2.1 Line Pool Configuration

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

5.2.2 Dial Plan Configuration

1. Under ***Configuration → Telephony → Dialing Plan → Public Network***, define the **Public Received** number length. Check with Skype for the appropriate value.
2. Set the **Public Network Dialing Plan** to **Public (Unknown)**
3. Set the **Public network code** which is a Caller ID prefix for example 613763
4. **In the Public Network DN Lengths table,** configure the DN prefixes and their corresponding lengths.
5. Under ***Configuration → Telephony → Dialing Plan → Routing*****,** and select the **Routes** tab
6. Add a route by clicking on the **Add** button.
7. In the **Add Route** dialog box, provide an unused route and click on the **OK** button. This is the preferred route.
8. The **Dialing Plan – Routing** table will be displayed.
9. Click on the route just created
10. Under the **External Number** column of the newly created route, insert a number that will 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 Skype.
11. Under the **Use Pool** column of the newly created route, double click to select **BlocA** from the drop down list.
12. Under the **DN Type** column of the newly created route, double click to select **Public (Unknown)** from the drop down list.
13. Repeat Steps 6 to 11 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 10..
14. Click on the **Destination Codes** tab.

15. 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.
16. In the **Add Destination Code** dialog box, enter a numeric number for the destination code and click on the **OK** button.
17. Select the row representing the Destination Code entered in the previous step
18. Under **Normal Route** column, double click and enter the route entered in Step 7.
19. Under the **Absorbed Length** column, specify the number of digits that will be absorbed before pre-pending the configured External number configured in Step 10 and sending the rest of the digits to the SIP trunking component.
20. While the row representing the Destination Code enter in Step 16 is selected, in the **Alternate Routes** table that shows up below, click on an unused schedule
21. In the **Second Route** column of the selected row above, provide the route configure in Step 13.
22. Navigate to ***Configuration→ Telephony→Scheduled Services***
23. Click on the schedule used in Step 20 above.
24. Define the times when this schedule will be active by providing the start and stop times for each day of the week.
25. 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

5.2.3 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 Skype 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 Skype. The number selected here should be the number digits in the External Number configured in Step 10
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 10 of section 5.2.2
- In the **ITSP Account** drop down list, select the Skype account

9. Click **OK**

5.2.4 Configuring Incoming Calls from SKYPE 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

5.2.4.1 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-digis of the Skype 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 Skype 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.

5.2.4.2 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-digis of the Skype 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.

5.2.5 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.

If you have any issues with the solution described in this document, please contact 1-800-4-NORTEL