Avaya BCM Solutions Test Lab

Business Communication Manager Release 5.0 Configuration Guide for Skype for SIP R1.3

Issue 1.0

Abstract

This document provides guidelines for configuring a SIP Trunk between a BCM Release 5.0 and SIP For Skype.

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1.0 Introduction

This document provides a typical network deployment of Business Communication Manager (BCM) Release 5 utilizing the Skype SIP Trunking service offering. The document provides the software line up, supported features as well as procedures for configuring a SIP trunk between the BCM Release 5 and Skype. This document should serve as general guideline only, since it is not possible to document every possible variation of configuration. Further information may be obtained from your Avaya support representative.

1.1 Document Change History

Date Version	Summary of Changes
Issue 1.00	March 19, 2010, first publication

2.0 System Software / Loadware

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

System	Platform	Firmware
BCM 50	All supported platforms	• Base S/W: **9.0.1.00** • Patch: BCM050.R500.SU.System-001.200911 or higher
BCM450	All supported platforms	• Base S/W: **9.0.1.00** • Patch: **BCM450.R500.SU.System-001.200911** or higher
BCM Phones	All BCM Supported phones	As provided by most recent smart update
Skype for SIP Server		Release 1.3

Table 1 Validated Equipment and Software

3.0 Features

3.1 Features Supported

The following are capabilities provided by this solution:

- Basic calls (G711a-law 20 ms, G729 20ms)
- Calling line (number) identification presentation (CLIP)
- DTMF (RFC2833)
- Call hold
- Call transfer (Blind and consultative transfers)
- Ad hoc conference calls
- Meet-Me conference calls hosted on the BCM
- Call forward
- Call Redirection to Voice Mail on BCM
- Inter-office tandem calls
- Find Me Follow Me
- Silent Recording
- Incoming calls from Skype to BCM phones.

3.2 Technical Caveats

The following are limitations with this solution:

- Fax is not supported by any of Skype's SIP trunking service
- On outbound calls to PSTN, although BCM sends the correct Display name and CLID to Skype, Skype does not deliver neither Display name nor the CLID to PSTN
- On incoming calls from PSTN, Skype does not send the Display name of the actual caller. Instead Skype account name tied to the assigned DID is sent.

4.0 Network Diagram

Figure 1 shows a typical deployment of BCM Release 5 with S4S SIP trunking service. Two deployment scenarios are supported:

- BCM is assigned a publicly routable IP address and there are no NAT/Firewall devices between BCM and public Internet.
- BCM behind a non-SIP aware NAT/Firewall. BCM is located behind a NAT/Firewall with no SIP ALG or SIP border control functions. BCM is assigned an IP address from private address space managed by the NAT.

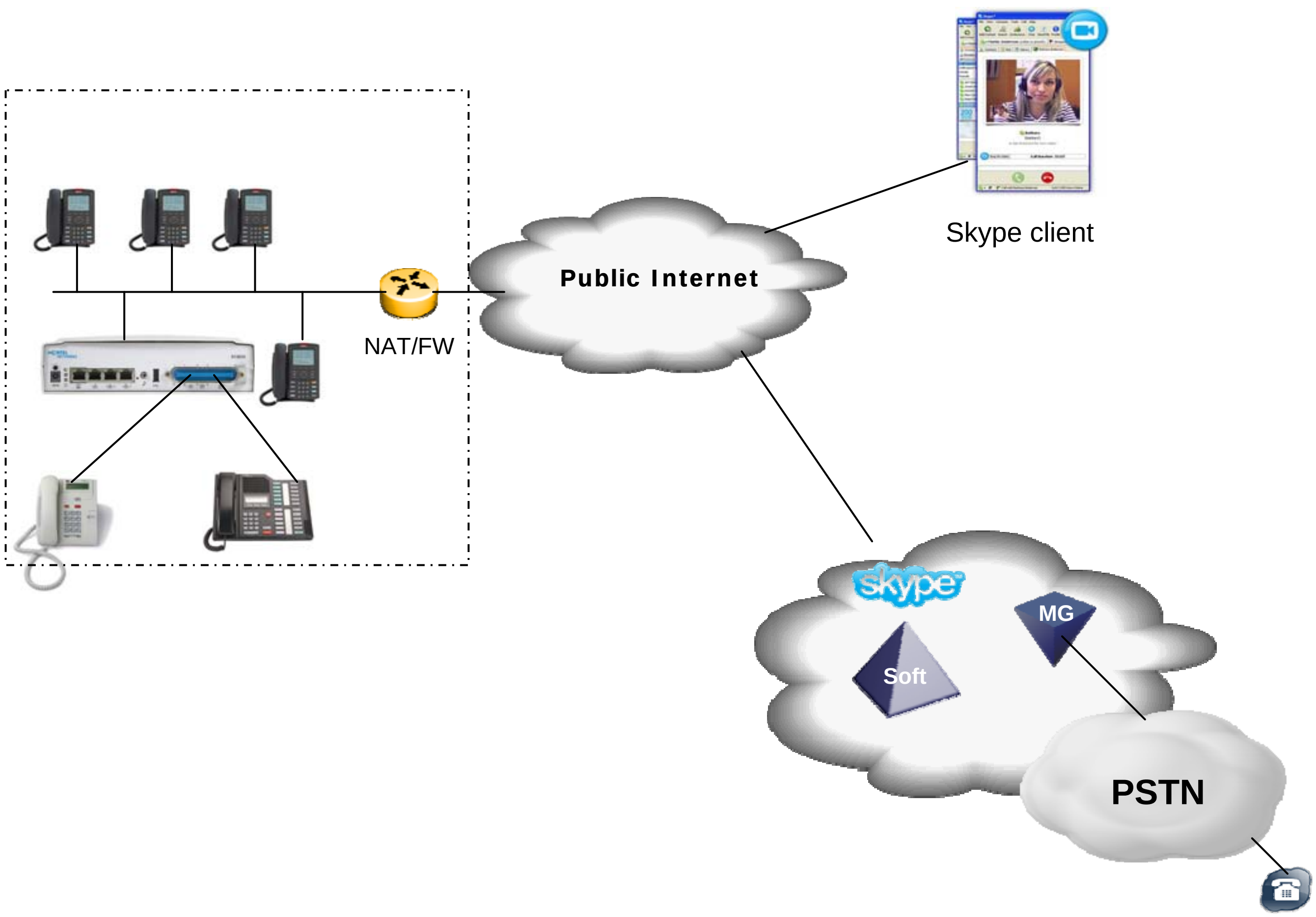

Figure 1 SIP connectivity between BCM Rls 5.0 and Skype

5.0 System Configuration

This section provides procedures for configuring a SIP trunk on BCM Rls 5.0 to connect to Skype

5.1 BCM Configuration

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

5.1.1 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.

5.1.2 Dial Plan Configuration

1. Under ***Configuration → Telephony → Dialing Plan → Public Network***, define the **Public Received** number length to “4” digits
2. Set the **Public Network Dialing Plan** to **Public (Unknown)**
3. Under ***Configuration → Telephony → Dialing Plan → Routing*****,** and select the **Routes** tab
4. Add a route by clicking on the **Add** button.
5. In the **Add Route** dialog box, provide an unused route and click on the **OK** button.
6. The **Dialing Plan – Routing** table will be displayed.
7. Click on the route just created
8. Under the **Use Pool** column, double click to select **BlocA** from the drop down list.
9. Under the **DN Type** column, double click to select Public **(Unknown)** from the drop down list
10. Click on the **Destination Codes** tab.
11. 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.
12. In the **Add Destination Code** dialog box, enter a numeric number for the destination code and click on the **OK** button.
13. Select the row representing the Destination Code entered in the previous step
14. Under **Normal Route** column, double click and enter the route entered in Step 4.
15. 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.

5.1.3 SIP Routing Table Configuration

There are two possible ways to configure the BCM to route outbound SIP calls to the Service Provider:

- Using the **Routing Table** tab found by clicking on IP Trunks module in ***Configuration → Resources → Telephony Resources.***
- Using the **SIP Proxy** tab found by clicking on IP Trunks module in ***Configuration → Resources → Telephony Resources.***

This guide illustrates how to do it using the Routing table. For details of how to use the SIP Proxy tab, see the BCM configuration guides.

1. Under ***Configuration → Resources → Telephony Resources;*** select module type **"IP Trunks"** and click on the "**Routing Table**" tab.
2. Add a "**Remote Gateway**" by clicking on the "**Add**" button.
3. In the "**Add Remote Gateway**" dialog box that is displayed, configure the fields as follows:
    - **Description**: Enter the logical name for the trunk destination
    - **Destination Digits**: Enter digits that identify the remote system as the call destination.
    - **VoIP Protocol**: SIP
    - **Domain**: SIP Domain name of Skype. This is **sip.skype.com**
    - **IP Address**: Leave this field blank
    - **Port**: Leave this field blank
    - **GW Type**: Set this field to **Other**.
    - **MCDN Protocol**: Set to **None**.
    - **QoS Monitor**: Leave unchecked
    - **Tx Threshold**: Leave this field at its default value of 0.0
4. Click the **OK** button

5.1.4 SIP Settings Configuration

1. Under ***Configuration → Resources → Telephony Resources:*** Select module type "**IP Trunks**" and click on the "**SIP Settings**" tab
2. In the **SIP Settings** sections configure the different fields as follows:
    - **Local Domain**: sip.skype.com
    - **Disable maddr in Contact**: **Checked** to disabled maddr in Contact header
    - **Disable OPTIONS Caps**: **Checked** to disable OPTIONS capability request by the BCM
3. If the BCM is located behind NAT and remote NAT compensation is provided by Skype, then in the **RTP Keepalives** section enable the sending of RTP keepalive by setting the following parameters as shown below:
    - **Scope**: RTP-RTCP
    - **Initial Keepalives**: Enabled

- **Periodic Timeout**: 0

4. Leave the rest of parameters under the **SIP Settings** tab at the default values

5.1.5 SIP Media Parameters Configuration

1. Under ***Configuration → Resources → Telephony Resources:*** Select module type "**IP Trunks**" and click on the "**SIP Media Parameters**" tab.
2. In the **Preferred Codecs** section, configure G.729, G.711-uLaw, G.711-aLaw as the first, second and third preferred codecs respectively.
3. In the codec **Settings** section, **Disable Enable Voice Activity Detection**.
4. In the codec **Settings** section, select 20ms as the payload size for both G.729 and G.711
5. Again in the **codec settings** section set G.711 as the **Fax transport**[1].

5.1.6 SIP Authentication Configuration

1. Under ***Configuration → Resources → Telephony Resources**:* Select module type "**IP Trunks**" and click on the "**SIP Authentication**" tab.
2. Create a new SIP Account by clicking on the "**Add**" button.
3. In the **Add Auth Account** dialog box that opens up, configure the parameters as follows:

- **Description**: Provide a descriptive name for this SIP account.
- **Domain**: Set this field to **sip.skype.com**
- In the Account Identity section, select configure this SIP Account as either a **Parent** or **Child** account. A Parent account can be used by BCM Phone to place out going calls via the SIP Trunk between the BCM and Skype whereas as a Child account can only be used by a given BCM Phone. A Parent account will work for most deployments.
- In the **User Credentials** section, configure the following parameters as shown below:
    - **SIP Username**: set this field to your SIP for Skype Account username (provided by Skype)
    - **Auth Username**: set this field to your SIP for Skype Account username (provided by Skype)
    - **Auth Password:** set this to the password for the SIP For Skype user account (provided by Skype)
- Configure the parameters in the **Message Handling** section as follows:
    - **CLID Override**: set this field to your Skype for SIP Account username (provided by Skype in your Business Control Panel under Skype for SIP settings)
    - **Display name Override**: Provide the Display Name that will be presented to the called party.
    - **Contact Override**: Leave blank
    - **Maddr in Contact**: Leave unchecked
    - **Local Domain Override**: set this field to **sip.skype.com**
- Enable SIP Registration the account by checking the checkbox next **Registration**.
    - Leave all parameters in the **Registration Details** section at their default values.
- Click the **OK** button

[1] Fax is not supported however

5.1.7 IP Sets Media Parameters

1. Under ***Configuration → Resources → Telephony Resources***: Select module type "**IP Sets**" and click on "**IP Terminals Global Settings**".
2. Set the "**Default Codec**" for IP sets to **G.729** and set the payload size (ms) for G.729 and G.711 to 20

5.1.8 Configuring Incoming Calls from Skype to BCM

This can be done in one of two ways;

1. The DID or Skype Online number associated with the Skype for SIP profile 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 routed those set(s).
2. All calls to the DID associated with the Skype SIP account assigned to the BCM can be answered by the Auto Attendant (AA) and from there, a DN can be entered to reach phone on the BCM

5.1.8.1 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 SIP for Skype Account User name.
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.

5.1.8.2 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 SIP for Skype Account User name.
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.

5.1.9 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.

6.0 References

The following are useful references to assist in this solution:

1. BCM Release 5.0 Technical Documents, http://support.nortel.com/go/main.jsp?cscat=DOCUMENTATION&poid=15181&catOID=-9602&viewOptSelect=&viewOpt1=5.0&viewOpt2=DEFAULT&searchText=&searchType=fulltext&x=57&y=17