



Configuring Avaya BCM 6.0 for Spitfire SIP Trunks

This document is a guideline for configuring Spitfire SIP trunks onto Business Communication Manager release 6.0 and includes the settings required for Inbound DDI routing and Outbound CLI presentation. The settings contained within have been tested and are known to work at the time of testing.

SIP trunk details such as account number and password will be provided separately.

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Software level

Release 6.0 continues the common software release for both BCM50 and BCM450. Therefore BCM will refer to BCM Release 6.0 and will not refer to any particular hardware platform.

The Avaya BCM must be running at least software version 10.0.1.52.xxx. You can check the version of BCM by viewing the following screen under **Administration » Software Management » Software Update History**.

It is highly recommended that the latest software updates are applied.

This software release 10.0.1.52.xxx refers to BCM Release 6.

Software Update History

Current system software version: 10.0.1.52.176

Software Update History

Date ▼	Category	Name	Version	Description	Removeable
2011-12-30 23:48	Patch Applied	BCM050.R600.RCC	54-1	Update to RCC	<input type="checkbox"/>
2011-12-30 23:35	Patch Applied	BCM050.R600.SU.Desktop	002.201111-1	BCM50 R6 November 20...	<input type="checkbox"/>
2011-12-30 23:00	Patch Applied	BCM050.R600.CORE-TELEPHONY	51-1	Update to CORE-TELEP...	<input type="checkbox"/>
2011-12-30 22:25	Patch Applied	BCM050.R600.SU.System	008.201111-2	BCM50 R6 November 20...	<input type="checkbox"/>
2011-08-18 12:04	Patch Applied	BCM050.R600.SU.Desktop	001.201106-1	Update to SU.Desktop	<input type="checkbox"/>
2011-08-18 10:29	Patch Applied	BCM050.R600.SU.System	007.201108-1	BCM50 R6 August 2011 ...	<input type="checkbox"/>
2011-06-10 10:19	Patch Applied	BCM050.R600.SU.System	006.201106-1	BCM50 R5 June 2011 Sy...	<input type="checkbox"/>
2011-05-12 11:41	Patch Applied	BCM050.R600.SU.System	005.201105-1	BCM50 R6 May 2011 Sv...	<input type="checkbox"/>

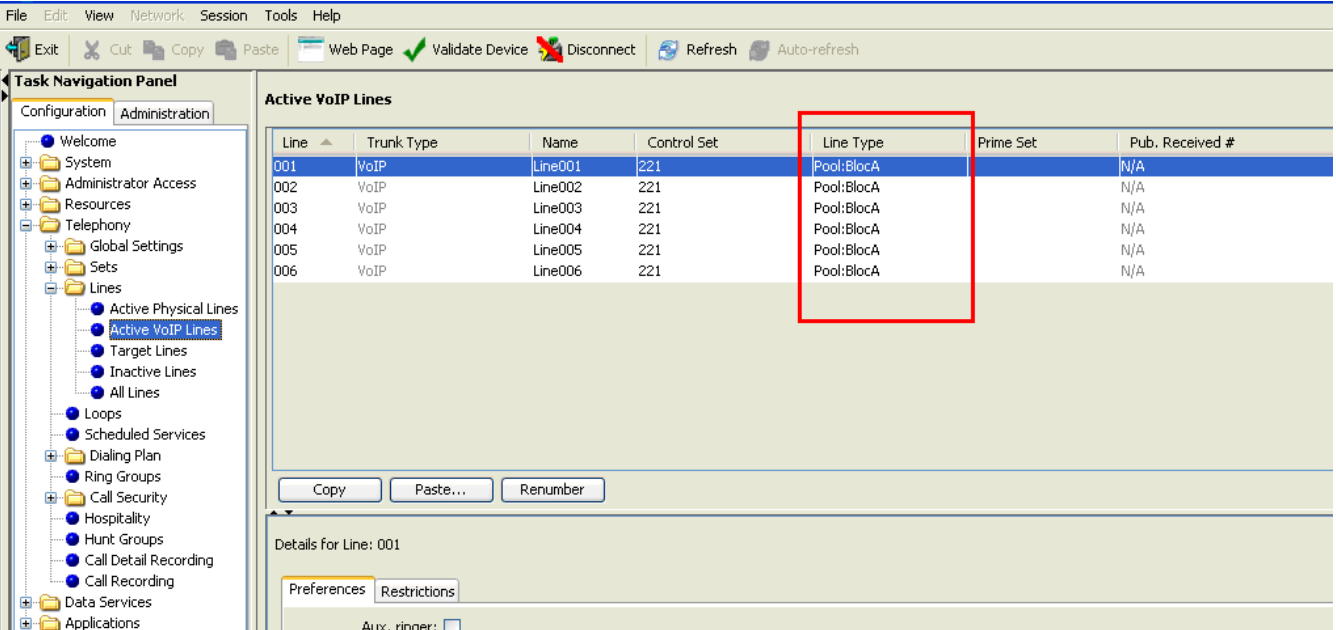
Remove Software Update

Licenses

There are two VoIP trunk license options: SIP Gateway Trunk License and VoIP Trunk Gateway License. The SIP Gateway Trunk License enables SIP-only trunks and the VoIP Trunk Gateway License enables SIP or H.323 trunks. Either type of trunk licenses can be used for SIP signaling.

Once the licenses have been added you can check under **Configuration » Telephony » Lines » Active VoIP Lines** to see if trunks have been allocated. The total number of lines shown corresponds to the number of IP trunk licenses added.

When the VoIP trunks are set up you can assign them to line pools and program their behavior in the same way you would PRI lines.



The screenshot shows the CUCM configuration interface. On the left is the 'Task Navigation Panel' with a tree view containing 'Welcome', 'System', 'Administrator Access', 'Resources', 'Telephony', 'Global Settings', 'Sets', '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 VoIP Lines' section is selected. The main area displays a table titled 'Active VoIP Lines' with the following data:

Line	Trunk Type	Name	Control Set	Line Type	Prime Set	Pub. Received #
001	VoIP	Line001	221	Pool:BlocA		N/A
002	VoIP	Line002	221	Pool:BlocA		N/A
003	VoIP	Line003	221	Pool:BlocA		N/A
004	VoIP	Line004	221	Pool:BlocA		N/A
005	VoIP	Line005	221	Pool:BlocA		N/A
006	VoIP	Line006	221	Pool:BlocA		N/A

Below the table are buttons for 'Copy', 'Paste...', and 'Renumber'. At the bottom, there is a 'Details for Line: 001' section with tabs for 'Preferences' and 'Restrictions'. The 'Aux. ringer' checkbox is visible.

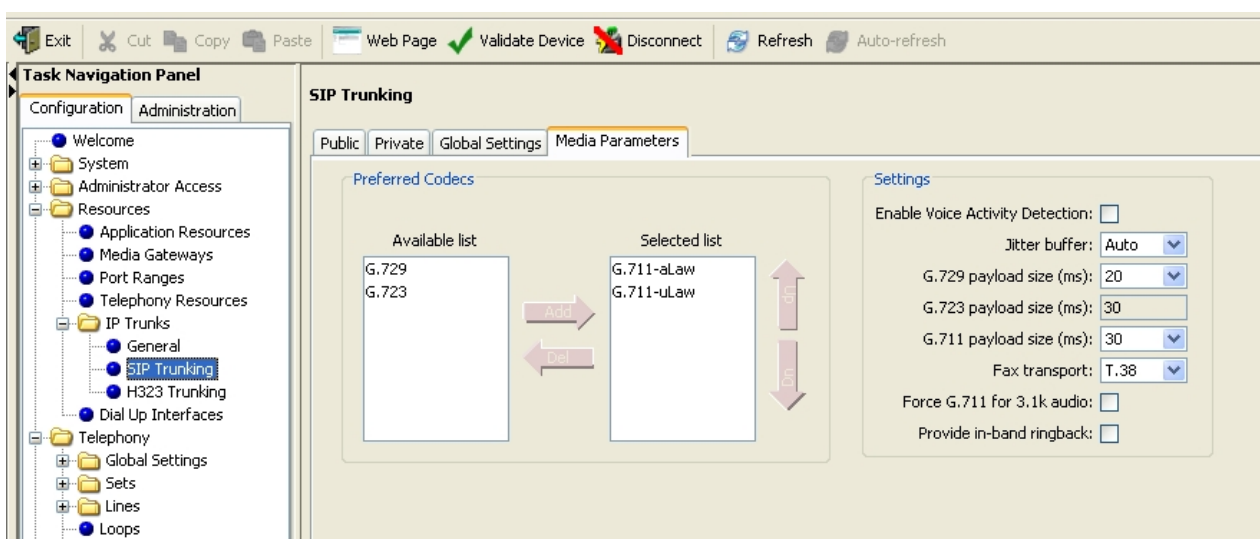
VoIP trunks should be configured to use a single line pool. Do not mix other trunk types on the same line pool (e.g. analog, PRI, etc).

Adding a SIP trunk

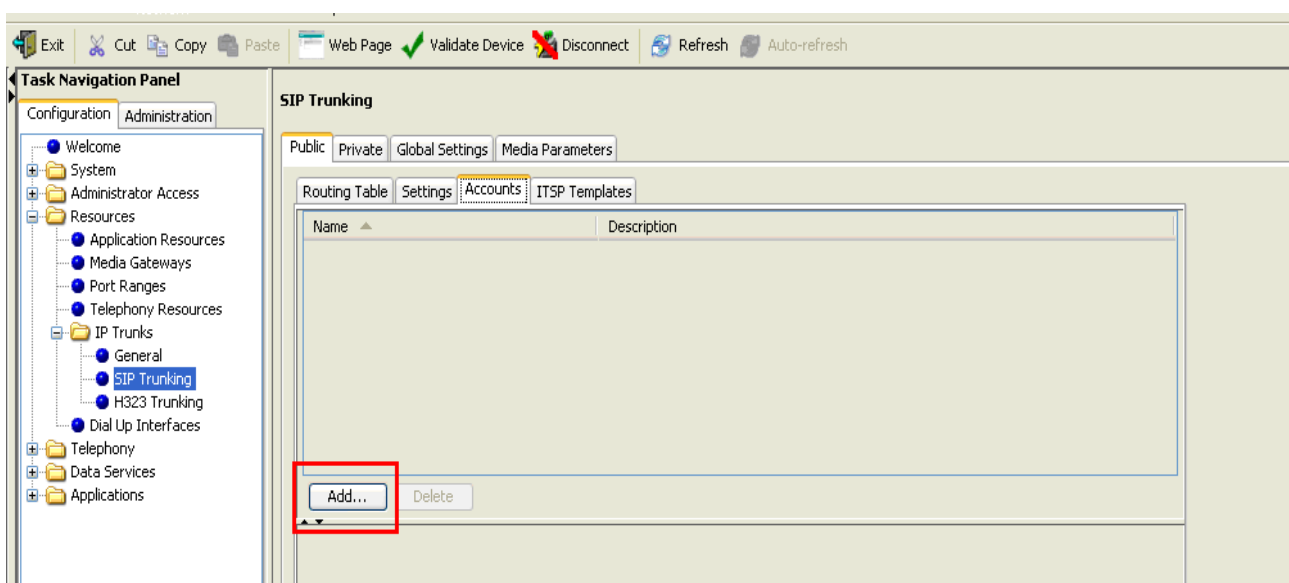
Go to **Resources** » **IP Trunks** » **SIP Trunking**

Select the **Media Parameters** tab and configure as below

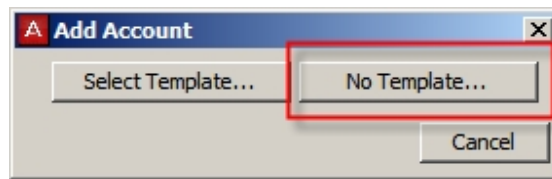
- 1st Preferred Codec: G.711-aLaw
- 2nd Preferred Codec: G.711-uLaw
- Voice Activity Detection: Disabled
- Jitter Buffer: Auto
- G.711 payload size: 30
- Fax transport: T.38



Select the **Public** tab, then the **Accounts** sub-tab and select **Add**

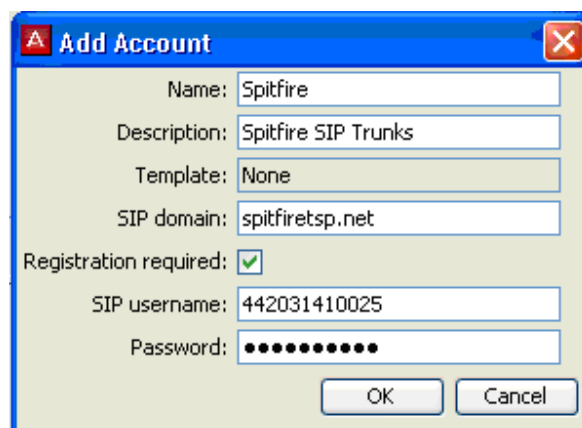


Select No Template

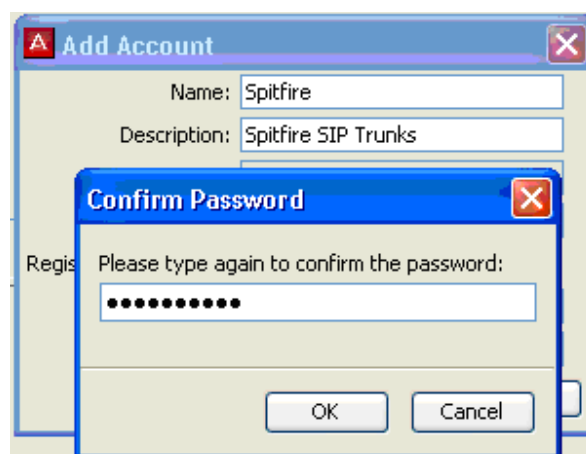


Enter the Spitfire SIP trunk account settings:

- Name: Enter a logical name for the Spitfire SIP trunk
- Description: Brief description of the account (optional)
- SIP domain: spitfiretsp.net
- Registration required: Checked
- SIP username: As supplied by Spitfire. **Note that it does not include the @spitfiretsp.net**
- SIP password: As supplied by Spitfire



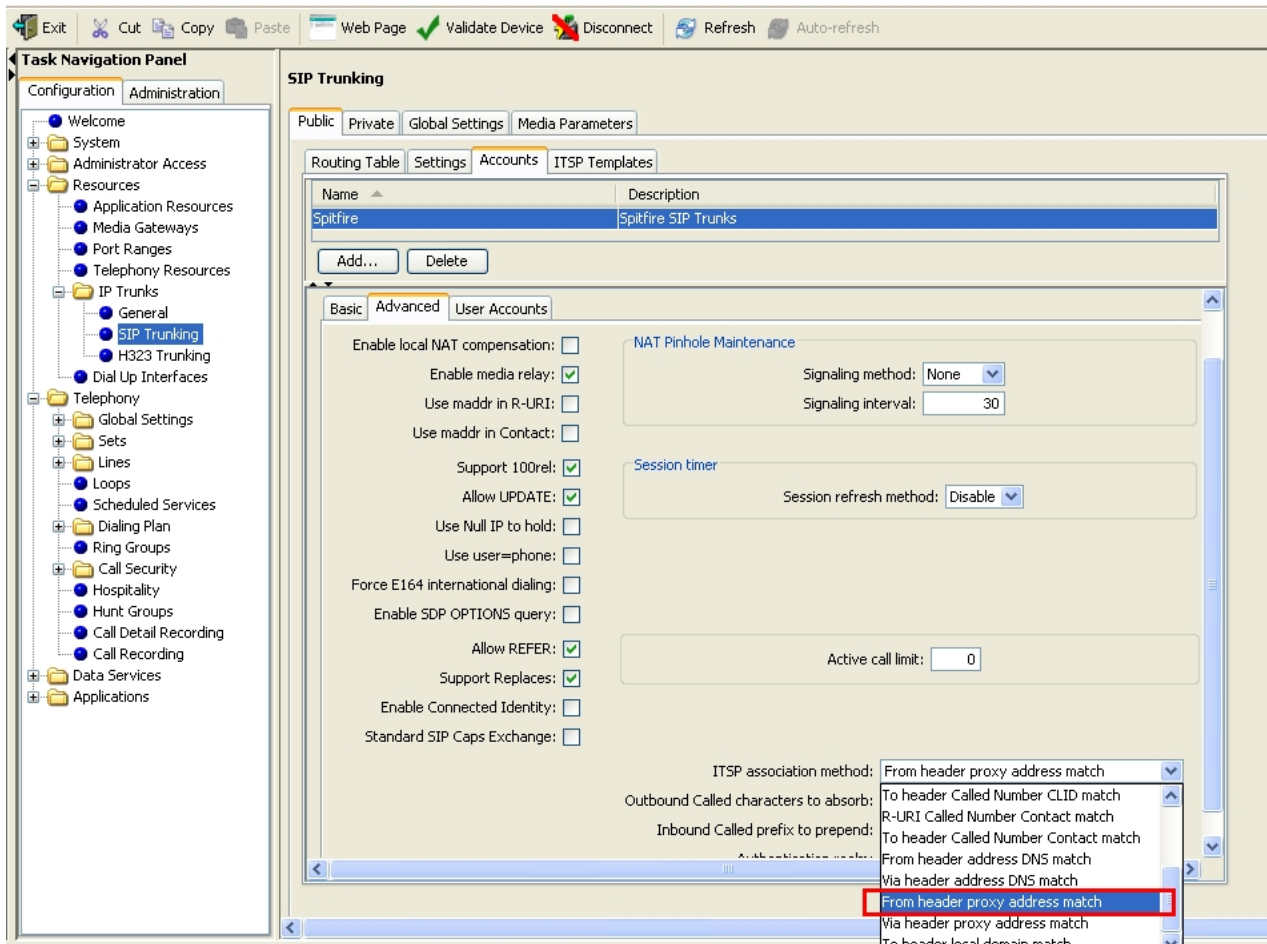
- Confirm Password: Enter the password again



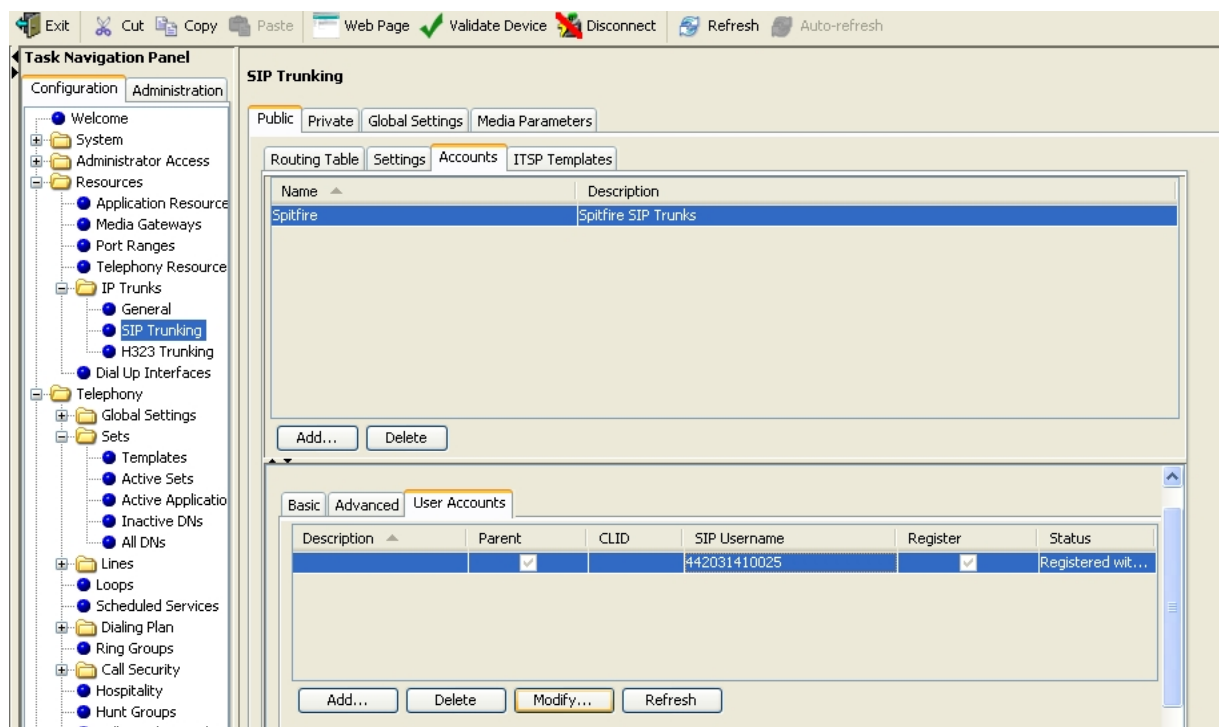
Select OK to Add the account

Once the account has been added highlight the account and select the **Advanced** sub-tab.

- Enabled media relay: If you have audio transmission problems when using IP sets switch on media relay
- ITSP association method: Ensure this is set to “**From header proxy address match**”

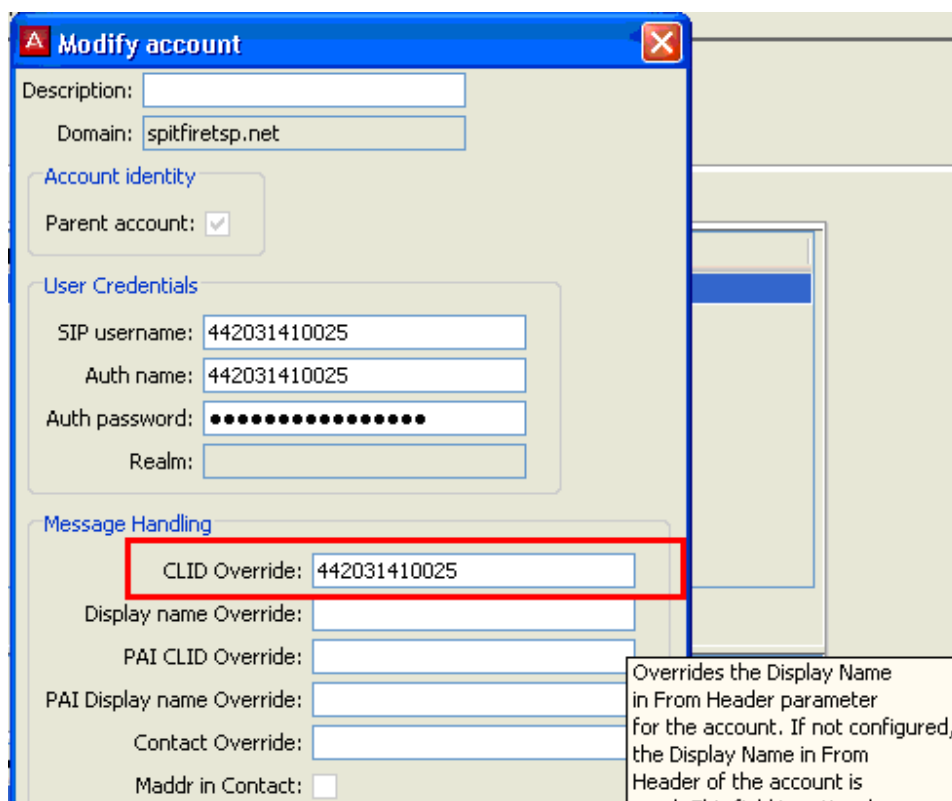


Select the **User Accounts** sub-tab.



Highlight the Spitfire account and select **Modify**

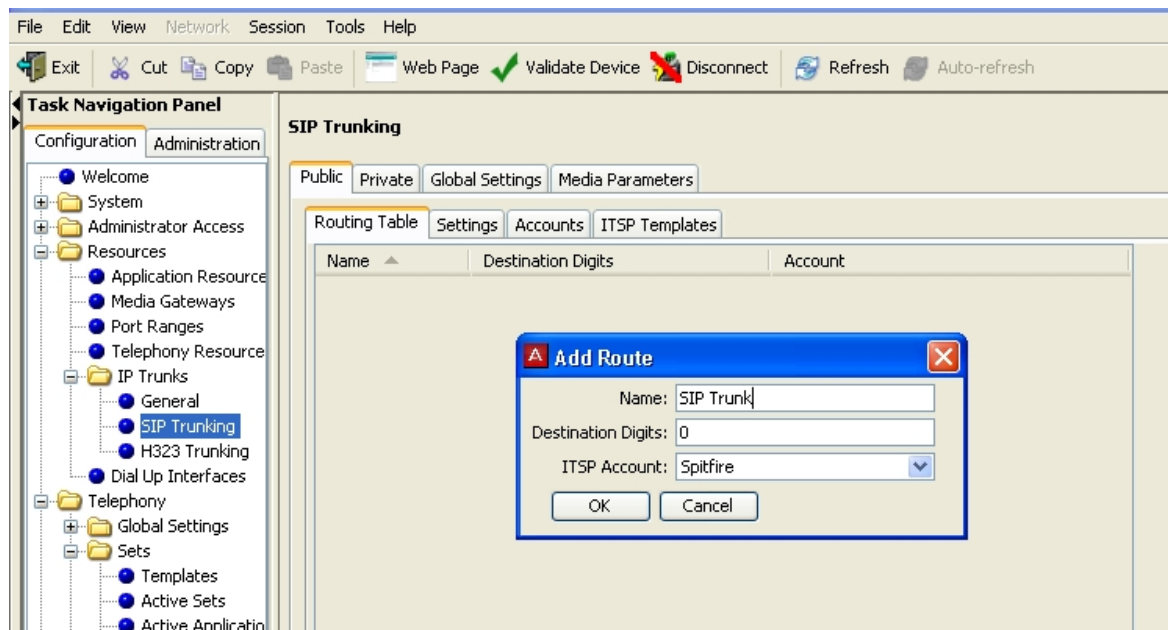
In the Message Handling field set "**CLID Override**" to your SIP trunk account number as supplied by Spitfire **without** the @spitfiretsp.net



Select the **Routing Table** sub-tab

Add a SIP trunk route by clicking on the “Add” button. A new window dialog appears. Enter the following:

- Name: Identifier for the route
- Destination Digits: Enter the leading destination digits i.e. 0 for 01, 02 etc. You may enter multiple leading digits separated by a space
- ITSP Account: Select the Spitfire ITSP account created previously



Inbound DDI Routing

There are obviously several ways to program inbound DDI's onto the BCM so this example is for reference only.

In this example I am programming the full received number (12 digits) for clarity in Element manager and for ease of programming the Outbound CLI. The received public number is programmed into a Target Line which is in turn assigned to active extension 302.

Go to **Telephony** ➤ **Dialing Plan** ➤ **Public Network**

Change **Public Received number length** to “12”

The screenshot shows the Cisco Element Manager web interface. On the left is the 'Task Navigation Panel' with a tree view containing folders like System, Administrator Access, Resources, Telephony, Global Settings, Sets, Lines, Loops, Scheduled Services, and Dialing Plan. Under 'Dialing Plan', 'Public Network' is selected. The main area is titled 'Dialing Plan - Public Network' and contains three sections: 'Public Network Settings', 'Public Network DN Lengths', and 'Carrier Codes'. In the 'Public Network Settings' section, the 'Public Received number length' dropdown is highlighted with a red box and set to '12'. Other fields include 'Public network dialing plan' (set to 'Public (Unknown)'), 'Public Auto DN', and 'Public DISA DN'. The 'Public Network DN Lengths' section contains a table with two columns: 'DN Prefix' and 'DN Length'. The 'Carrier Codes' section contains a table with two columns: 'Code Prefix' and 'ID Length'. Both tables have 'Add...' and 'Delete' buttons below them.

DN Prefix	DN Length
0	11
00	17
9	11
80	12
118	6
0844	12
907	12
9020	12
80442	7
Default	8

Code Prefix	ID Length
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Go to **Telephony » Lines » Target Lines**

Enter the DDI's allocated by Spitfire in the **Pub. Received #** field, 1 per Target Line

Line	Trunk Type	Name	Control Set	Line Type	Prime Set	Pub. Received #
125	Target line	Line125	221	Public	221	442031410025
126	Target line	Line126	221	Public	221	442031410026
127	Target line	Line127	221	Public	221	442031410027
128	Target line	Line128	221	Public	221	442031410028
129	Target line	Line129	221	Public	221	442031410029
130	Target line	Line130	221	Public	221	
131	Target line	Line131	221	Public	221	
132	Target line	Line132	221	Public	221	
133	Target line	Line133	221	Public	221	
134	Target line	Line134	221	Public	221	
135	Target line	Line135	221	Public	221	
136	Target line	Line136	221	Public	221	
137	Target line	Line137	221	Public	221	
138	Target line	Line138	221	Public	221	
139	Target line	Line139	221	Public	221	

Go to **Telephony » Sets » Active Sets**

Assign the applicable Target line to the DN you wish the number to ring on.

DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Answer	Fwd Delay	Fwd Busy	Fwd All
233	Analog	233	0413				N/A		
234	Analog	234	0414				N/A		
302	1230	302	0101	2031410026			N/A		
303	1140E/2004/2007/2050/221x	303	0109	2031410027			N/A		

Details for DN: 302

Line	Appearance Type	Appearances	Caller ID Set	Vmsg Set	Priv. Received #	Pub. Received #
126	Appr&Ring	1	<input checked="" type="checkbox"/>	<input type="checkbox"/>		442031410026

Outbound Routing

The BCM can be configured with a very complex routing table. This guide is not intended to cover all aspects of ARS/TRS on the BCM, it simply gives a very basic example on how to send outbound calls over the Spitfire SIP trunks.

Go to **Configuration** \Rightarrow **Telephony** \Rightarrow **Lines** \Rightarrow **Active VoIP Lines** and confirm which Bloc the VoIP lines are in

The screenshot shows the 'Active VoIP Lines' configuration page. On the left is a 'Task Navigation Panel' with a tree view containing 'Welcome', 'System', 'Administrator Access', 'Resources', 'Telephony', 'Global Settings', 'Sets', 'Lines', 'Active Physical Lines', 'Active VoIP Lines' (selected), 'Target Lines', 'Inactive Lines', 'All Lines', and 'Loops'. The main area displays a table of active VoIP lines. A red rectangle highlights the 'Line Type' column, which contains the value 'Pool:BlocA' for all six lines.

Line	Trunk Type	Name	Control Set	Line Type	Prime Set
001	VoIP	Line001	221	Pool:BlocA	
002	VoIP	Line002	221	Pool:BlocA	
003	VoIP	Line003	221	Pool:BlocA	
004	VoIP	Line004	221	Pool:BlocA	
005	VoIP	Line005	221	Pool:BlocA	
006	VoIP	Line006	221	Pool:BlocA	

Got to **Configuration** \Rightarrow **Telephony** \Rightarrow **Dialing Plan** \Rightarrow **Line Pools** and ensure that the extensions have access to the pool that contains the VoIP lines

The screenshot shows the 'Line Pools' configuration page. The 'Task Navigation Panel' on the left has 'Line Pools' selected under the 'Dialing Plan' section. The main area shows a table of line pools. A red rectangle highlights the 'BlocA' pool with an 'N/A' access code. Below the table, the 'Details for Line Pool: BlocA' section is visible, showing tabs for 'DNs' and 'Call by Call Limits'. The 'DNs' tab is active, displaying a list of DNs with access to the line pool: 221, 222, 223, 233, 302, and 303. There are 'Add...' and 'Delete' buttons at the bottom of the DN list.

Pool	Access Code
BlocA	N/A

Details for Line Pool: BlocA

DNs | Call by Call Limits

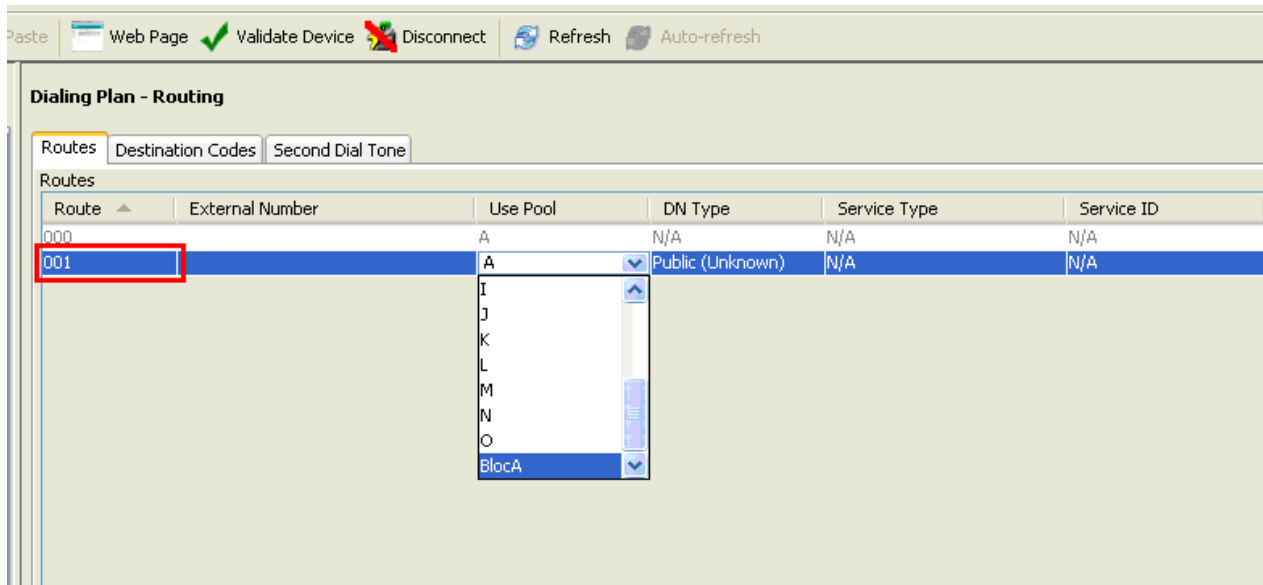
DNs with Access to Line Pool

DN
221
222
223
233
302
303

Add... Delete

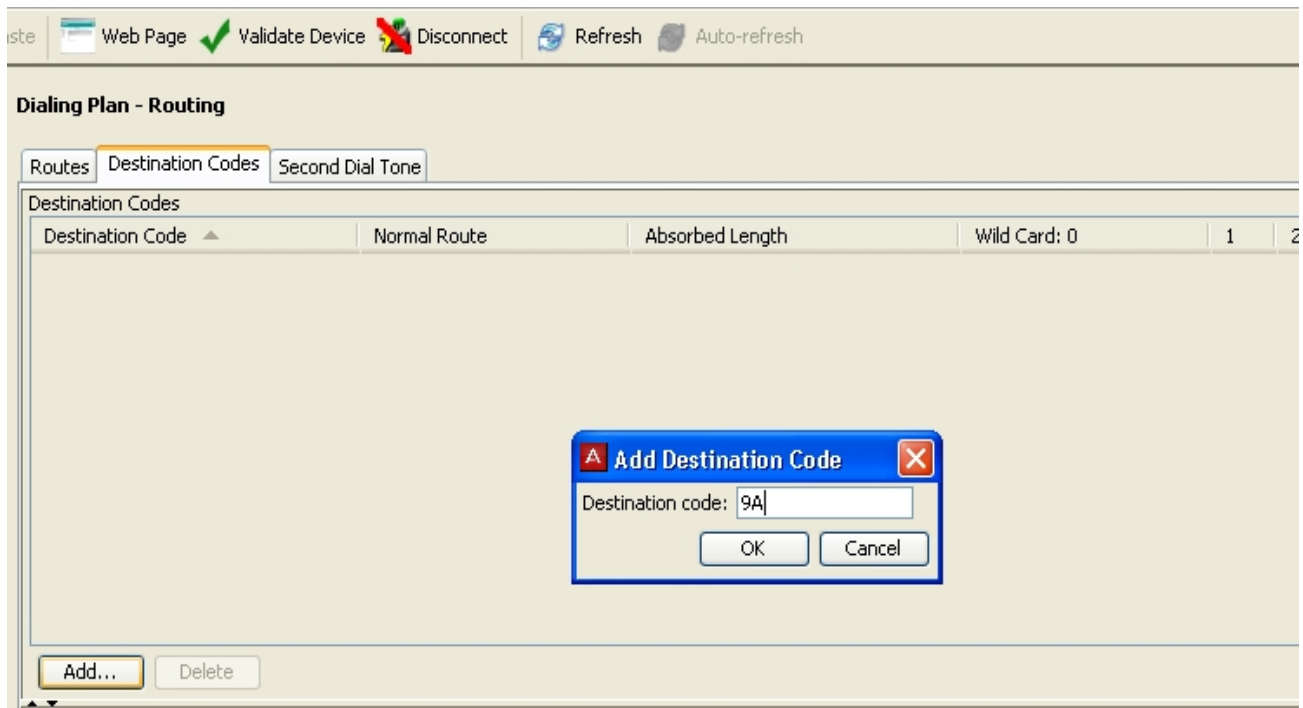
Select **Configuration** → **Telephony** → **Dialing Plan** → **Routing** and add a new route or amend an existing route on the “**Routes**” tab. In this example I am modifying “**Route 1**”

- Name: Select the Pool associated with the SIP trunk (BlocA in this case).
- DN Type: Public (Unknown)



Now select the “**Destination Codes**” tab

Add the required destination code – here I am simply using 9 followed by the wildcard for anything



Change the Route (Normal, Night etc) to the Route created for your SIP trunks. In this case it is Route 1 as created on the previous page.

Absorb the applicable leading digits – i.e. 1 for the leading 9 as I do not want this sent to line.

Paste Web Page Validate Device Disconnect Refresh Auto-refresh

Dialing Plan - Routing

Routes Destination Codes Second Dial Tone

Destination Codes

Destination Code	Normal Route	Absorbed Length	Wild Card: 0
9A	001	1	<input checked="" type="checkbox"/>

Add... Delete

Outbound CLI Presentation

Select **Configuration** → **Telephony** → **Sets** → **Active Sets** and select the extension you wish to modify. Enter the number you wish to present in the Pub. OLI field. This should be a DDI associated with the Spitfire SIP trunk **without** the leading 44, as below.

The screenshot shows a web interface for configuring Active Sets. The left sidebar contains a 'Task Navigation Panel' with a tree view showing the path: Configuration > Telephony > Sets > Active Sets. The main content area is titled 'Active Sets' and has three tabs: 'Line Access', 'Capabilities and Preferences', and 'Restrictions'. The 'Line Access' tab is active, displaying a table with the following data:

DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Ans
233	Analog	233	0413			
234	Analog	234	0414			
302	1230	302	0101	2031410026		
303	1140E/2004/2007/2050/221x	303	0109	2031410027		

The 'Pub. OLI' field for extension 302 is highlighted with a red box.