Avaya BCM Solutions Test Lab

Business Communication Manager BCM 50 Release 3.0 and BCM450 Release 1.0 Configuration Guide for Bell Canada SIP Trunking

Issue 1.0

Abstract

This document provides guidelines for configuring a SIP Trunk between a BCM50 Release 3.0 or BCM450 Release 1.0 and Bell Canada SIP Trunking Service

Table of Contents

1.0 Introduction

This document will cover the basic setup of the Nortel BCM50r3 and BCM450r1 for use with the Bell Canada SIP Trunking service.

- Testing was performed in accordance to the Bell Canada test plan and all the core features of the SIP Trunking service were verified.
- The BCM50 and 450 configuration detailed in this document was verified in a lab environment with a minimal configuration used to ensure proper interoperability between Bell's SIP network and the system under test.

DISCLAIMER: The configuration described in this document details only the minimum configuration required for interoperability to be successful; so care must be taken by the network administrator to ensure this configuration is valid for their deployment network, accounting for version differences, and possible feature conflicts with their CPE environment as well. Note all test cases where run BCM50 R3 hardware. Minimal testing was performed on the BCM450R1 system as the function of the BCM50R3 and BCM450R1 systems are identical this service.

2.0 Prerequisites

This document assumes the reader possesses administrator-level knowledge in regard to the deployment and configuration of the BCM line of products and there associated Management tools. Based on that assumption, this document will only cover what is necessary to connect the specified the BCM hardware to the Bell SIP Trunking service. The document does not cover any non-trunk-related configurations or any complex trunk-routing scenarios specific to a given customer deployment scenario.

2.1 Required Reading

The reader is urged to consult the Bell SIP Trunking Service Interface Document for more detailed coverage of the content in this guide. The Service Interface Document includes detailed coverage of the SIP –Trunking service parameters, including example SIP messages, full coverage of the production version of the dial-plan, and detailed coverage of codec support and policies for trans-coder invocation.

3.0 SIP Trunking Service Overview

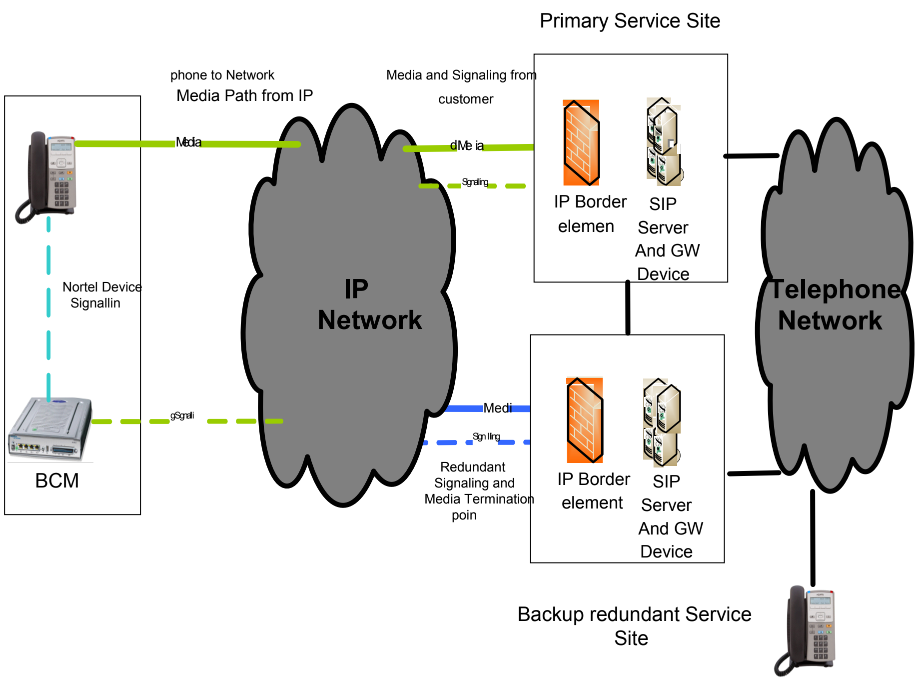
3.1 What is SIP Trunking?

The SIP Trunking Service is a critical element for customers migrating to VoIP. The SIP Trunking Service provides customers with a voice gateway (typically an IP-based PBX) over their data network for calls to and from the PSTN.

An IP Trunk consists of a single virtual voice channel with local calling rights in the rate centre in which it is associated. The terminology used to describe the virtual voice channel is a “concurrent call” (“DS-0 equivalent” is also used sometimes). Customers can buy one or more concurrent calls to enable more than one call to be established simultaneously. Purchasing multiple trunks is referred to as a trunk group, so purchasing a trunk-group with ten trunks would allow ten simultaneous calls to be carried on at once using that trunk group.

A rate centre represents the local calling area wherein customers can call each other without incurring toll charges. If a customer buys one trunk (one concurrent call) in the Toronto rate centre (416), they have purchased the capability for anyone on their corporate network to make one local call within that rate centre. If the customer places a trunk call to a location that is outside their local rate center, they would be charged for a long distance call. Callers in the Toronto 416 calling area must pay long distance to call the 514 rate centre in Montreal, for instance.

4.0 Reference Topology



5.0 Tested Equipment

Bell service is compatible with BCM50 and BCM450. The following list summarizes the version and patch levels of the hardware that has been validated as compatible in the Bell lab

- BCM50 R3 System Software Version 6.0.2.05.237 patch 203.200909-1

- BCM450 R1 System Software Version 9.0.1.20.533 patch 007.200908

6.0 Features

6.1 Basic Features Supported

The following list summarizes the base feature set for Bell SIP Trunking that has been verified on the tested equipment.

- Basic Call using G.729 or G.711ulaw
- Calling Party Number Presentation and Restriction
- Calling Name
- Intra- and Inter-site Call Transfer
- Intra- and Inter-site Conference.
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- Fax using G.711 passthru
- TTY using G.711 passthru
- Outbound calls to IP and TDM networks

6.2 Trunk Group Selection for Originating Calls

The charging model for Bell SIP Trunking service is based upon Bell's legacy PRI trunk model, and allows a trunk subscriber the ability to purchase trunks that provide local presence in one or more given toll rate centres. This implies the need for the subscriber to be able to originate outbound calls on the trunk of their choice so as to minimize the toll charges they would incur placing calls into any rate centre that they do not have a local presence (dedicated trunk) for. For instance, an Ottawa-based trunk subscriber can place local calls to Toronto customers if they have purchased a trunk for the 416 rate centre.

The ability to select an outbound trunk on a per-call basis allows a customer with multiple trunks to originate their call from within a rate centre that is local to the called party. This selection is accomplished through one of the following mechanisms (listed in order of precedence):

TGRP (RFC 3904)

RFC 4904 describes a standardized mechanism for conveying trunk group selection parameters within 'sip:' and 'tel:' URIs..

Bell Canada SIP Trunking service expects a trunk group selection to be conveyed in the contact header for calls originating on the PBX and destined for the PSTN. The trunk group parameters detailed below are specific to the Bell Canada SIP Trunking service. These parameters are defined during the service setup process.

Example SIP Header:

Contact: <sip:6131112222;**tgrp=rate_613;trunk-**

context=sipt.bell.ca@gw1.example.com;user=phone>

The example shows the trunk selection parameters are to be inserted between the user and domain parts of the contact address, and are semicolon-delimited. In this example, trunk group identified by the label 'rate_613' and trunk-context 'sipt.bell.ca' are used to route this particular outbound call.

OTG

Including an OTG parameter in the From:, P-Asserted-Identity:, or Diversion: SIP header fields will indicate to the SIP Trunking service that the customer wishes to use originate a call within the trunk group specified by the 'otg=' parameter. The value of this parameter is defined during the service setup process.

Example SIP Headers:

From: <sip:6135604063@company.ca;user=phone;**otg=rate_613**>

P-Asserted-Identity: <sip:6135604063@company.ca;**otg=rate_613**>

Diversion: <sip:6135604063@company.ca;user=phone;**otg=rate_613**>

Each of the header parameters above can be used to specify the trunk preference for this call to the SIP trunking service. In this example the customer wants the call to be placed using the settings for trunk group 613 (including billing and capacity management).

No Trunk Selected

If none of the preceding trunk selection mechanisms are used for a given outbound call, the SIP Trunking service will use the calling party's default trunk for that call. The default trunk is selected using the identity of the call originator, which is typically specified by the contents of the SIP From: header. If the From: header is encrypted or set to 'anonymous', then the service will use the contents of the P-Asserted-Identity: header instead.

* Note currently the BCM does not support trunk group selection within the IP trunk interface via the additions of specific SIP headers as described above, so by default the "No Trunk Selected" mechanism is used.

6.3 Features not supported

- **SIP REFER**
    - The REFER method is not supported under Bell's SIP Trunking service.
- **T.38 Fax Protocol**
    - T.38 Fax transmissions are not supported by Bell SIP Trunking service. In-band Fax transmissions using G.711 transport have been verified in Bell's SIP Trunking verification tests.
- **Modem transmissions not supported**
    - Modem connections are not supported over the Bell SIP Trunking at this time. Although these connections may work.
- **Call Connected Network Name Display**

  This feature is not supported by the SIP Trunking service. Name and Number display is presented as provisioned in the SIP Trunking core elements.

7.0 Deployment considerations

- A clear media path from IP sets to the sip trunk interface is required. The IP set determines the available codec used for media exchange.
- Trunk selection for the service is based on the From and PAI headers of the sip message. The headers are configured at a set level. Explicit trunk group selection is not supported on the BCM devices.
- If the call contains both G.711 and G.729 G711 will be used independent of the codec preference in the SDP to the PSTN.
- Call forwards do not include the original caller display information; instead these calls will appear as if they originated from the set doing the forwarding.
- When an INVITE request from the BCM50 R3 or BCM450 R1 is challenged, BCM responds to the challenge by sending an INVITE but the latter INVITE does not contain a PAI header.
- Additional licensing maybe required on the BCM to support deployment requirements. To connect the Bell SIP trunks you will need at a minimum the IP Trunks module

8.0 Media Codecs for Bell SIP Trunking

Bell SIP Trunking service provides support for G.711, and G.729 codecs. The G.711 codec is the default for SIP Trunking customers, but they may choose to sign up for SIP Trunking as G.729 subscribers in order to take advantage of the bandwidth saving a compressed codec offers them.

The SIP Trunking service infrastructure provides transcoding services to manage and alter the media path for calls where G.729 can't be supported end-to-end. The transcoder's intervention allows G.729 customers to view their service as a G.729-only service, and they need not consider the media capabilities of the called party's device or network, as the transcoder mediates the negotiation if required, and encapsulates any discrepancies between the customer's codec choice and that of the parties they place calls to, or receive calls from.

NOTE: The only valid ptime value for media codecs on the SIP Trunking service is 20 ms. All supported codecs must abide by this restriction. The 'ptime is specified in the SDP part of the SIP signaling, which is used to negotiate the media path for each session.

9.0 Required Service Configuration Information

Service domain

This is the domain name used by the service and will be inserted into the To: field of all sip messages generated by the PBX destined for the SIP Trunking interface, and in the From: header of all sip messages destined for the PBX from the SIP Trunking interface. The service domain for Bell SIP Trunking service is siptrunking.bell.ca

PBX domain

The PBX domain is the domain name that will be used by the SIP Trunking service to resolve the IP address of the customer PBX that is using the SIP Trunking service. This domain will be present in the To: header of all SIP messages generated by Bell SIP Trunking service to direct inbound calls to the customer's SIP trunks. This PBX domain is also expected to be the domain presented in the From: header of SIP messages entering the SIP trunk from the customer PBX. An example of a PBX domain specification would be: 'pbx1.customer.com'

Service IP address

The service IP address will be supplied to the customer by Bell. The service address will be the IP address (or addresses) of the customer-facing SIP trunk interface(s) on Bell's Session Border Controller(s).

PBX IP address

The IP address of the customer's PBX that inbound calls will be directed to. The customer's PBX domain must be resolved to this address when an inbound call is processed by the SIP Trunking service.

Authentication Credentials

This will be the authentication credentials used to authenticate SIP invite messages sent to Bell. This will consist of a user and password

10.0 SIP Trunking Dial Plan Overview

The SIP Trunking service will accept call requests with destinations according to the North American Number Plan (NANP: http://en.wikipedia.org/wiki/North_American_Numbering_Plan) If the service receives a request for a destination number that is not in service or has a malformed number, a proper announcement treatment will be applied.

** Please note that the service will accept 10 digit and 11 digit call requests for local and long distance calls. Depending on the purchased rate, long distance charges may apply. The following list shows a non-exhaustive dial plan example we used in our testing

- 10 Digits and 11 Digits for Local and Long Distance Call (long distance charge may apply)
- 0+10D Call (operator assistant NA LD Call)
- 01+International NDC (from 8 to 35 digits, Operator Assistant International Call)
- Toll Free Call 1-800,1-888,1-877
- 011+ International Call

- 101+xxxx+NDC call(from 13 to 40 digits, Casual Dial Call)
- 911
- 411, 611, 711, 211, 311, 511, 811
- 1-xxx-555-1212
- 310-xxxx Call
- 1-900 and 1-976 Call

11.0 Configure DNS (Optional)

There are two requirements for name resolution to support SIP Trunking calls. They are summarized briefly below. For more detailed information, please refer to the SIP Trunking Interface Document.

Bell resolves PBX domain

- A DNS resolution is needed for SIP trunking to reach the PBX. This DNS resolution can either be managed by customer DNS server or managed by Bell Canada.
- If Managed by Bell Canada the PBX destination IP address(es) needs to be provided to Bell Canada.
- If Managed by the customer, the customer needs to provide DNS access for the SBC to query the PBX domain. The domain is agreed upon during service setup and must resolve to the PBX IP address via an SRV and A records.

PBX resolves Bell service domain:

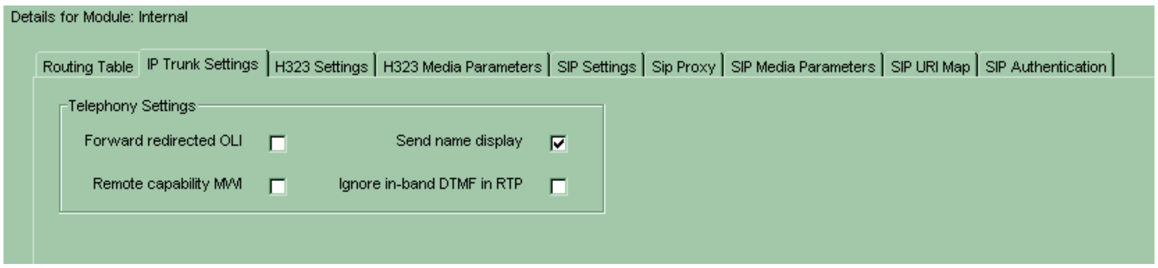
- Bell Canada provides redundant connections for the service. To provide redundancy the customer PBX can utilize DNS SRV records to resolve 1 or more Service interface IP addresses. These IP addresses are provided during service setup.

12.0 Configure the BCM

This section covers the base configuration of the BCM to interface with Bell Canada's SIP Trunking Service. The customer administrator will use the Business Element Manger to implement the following BCM configuration examples.

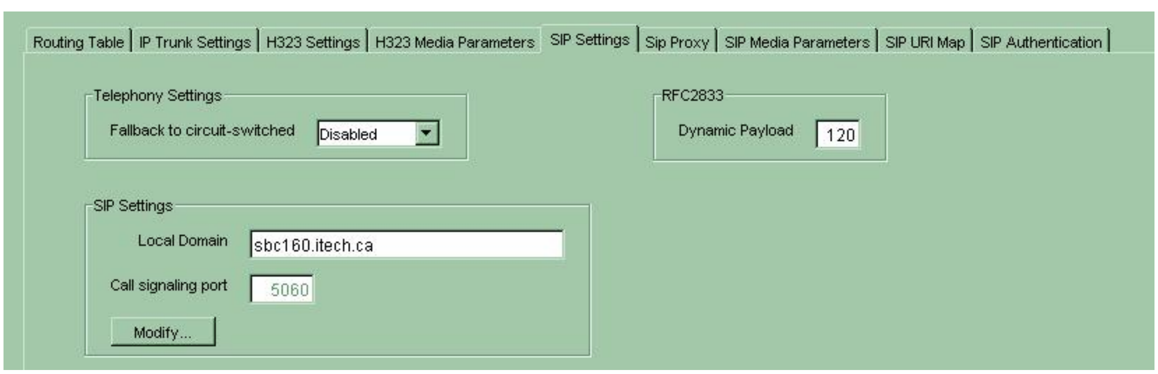
12.1 IP trunk Settings

Select Resources → telephony resources → Module IP trunks → IP Trunk Settings



Forward redirection OLI: off
Remote Capacity MWI: off
Send name display: on
Ignore in-band DTMF in RTP: off

Select Resources → telephony resources → Module IP trunks → SIP Settings

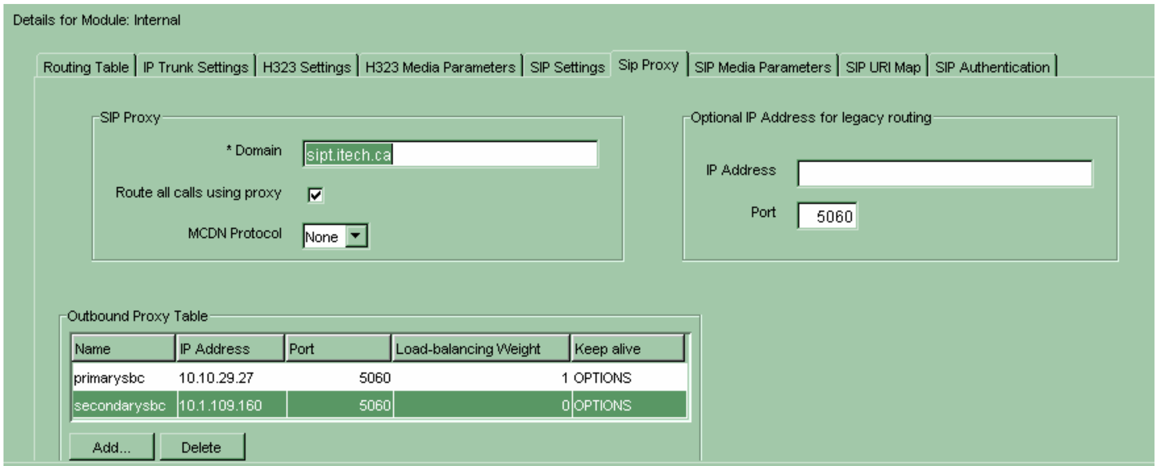


Fallback to circuit-switched: optional on or off depends on PBX deployment

Dynamic payload: 120
Local Domain: PBX domain
Call signaling port: 5060

* PBX domain will be determined and set as part of the activation process with Bell Canada.

Select Resources → telephony resources → Module IP trunks → SIP Proxy



SIP Proxy
    Domain: Service domain
    Route all calls using proxy: on
    MCDN: none

Optional IP address for legacy routing
    IP address: blank
    Port: 5060

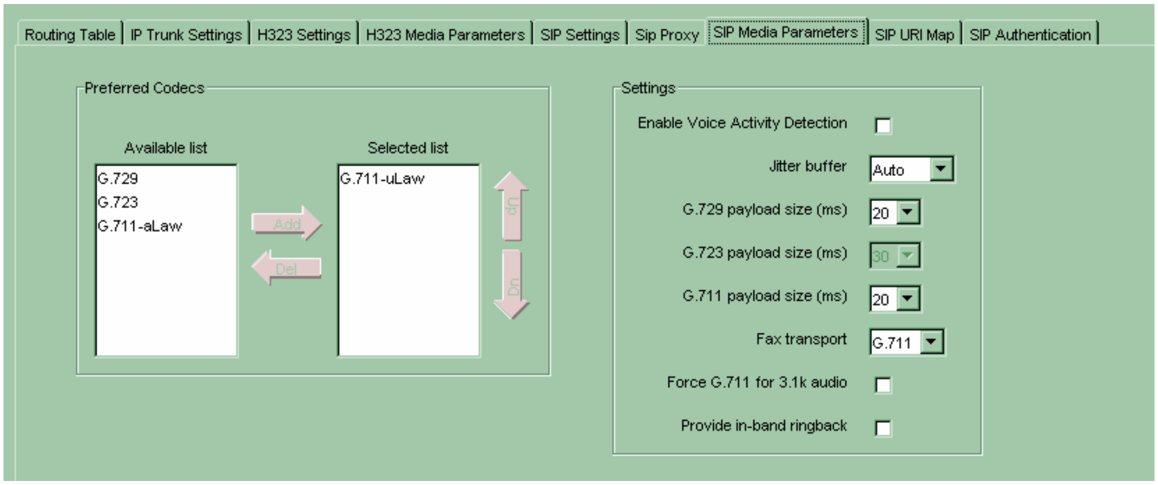
Outbound proxy table

Name: Service IP address or a name that represents the SBC
IP address: Service IP address
Port: 5060
Load balance weight: 1 or 0
When the weight is set to 1 all calls will use this device first
The device with a weight of 0 will be a failover device in the event the primary is unavailable.
Keep alive: options

If the redundancy option is not used us a weight of 1 and only 1 table entry.

* Service IP addresses and Domains will be provided by Bell Canada

Select Resources → telephony resources → Module IP trunks → SIP Media Parameters



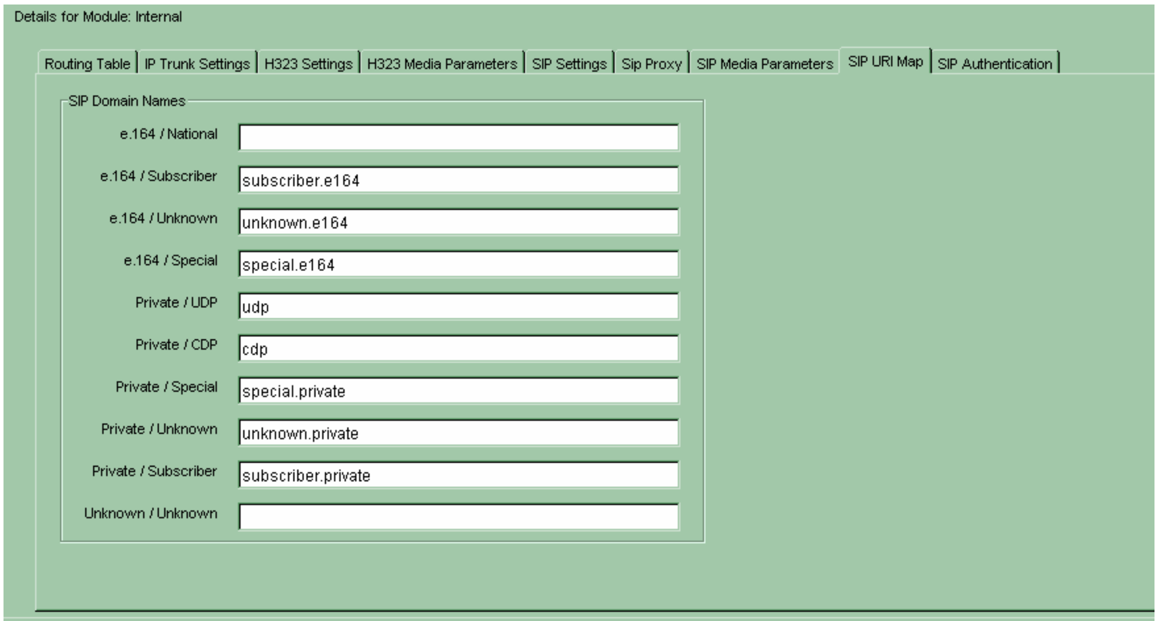
Preferred Codecs:

The selected list should only contain the codec that we want to use G711 u-law or G729 only.

Settings

Enable voice activity Detection: off
Jitter buffer: auto
G.729 payload size: 20
G.723 payload size: not applicable
G.711 payload size: 20
Fax transport: G.711
Force G.711 for 3.1k audio: off
Provide in-band ringback: off

Select Resources → telephony resources → Module IP trunks → SIP URI Map

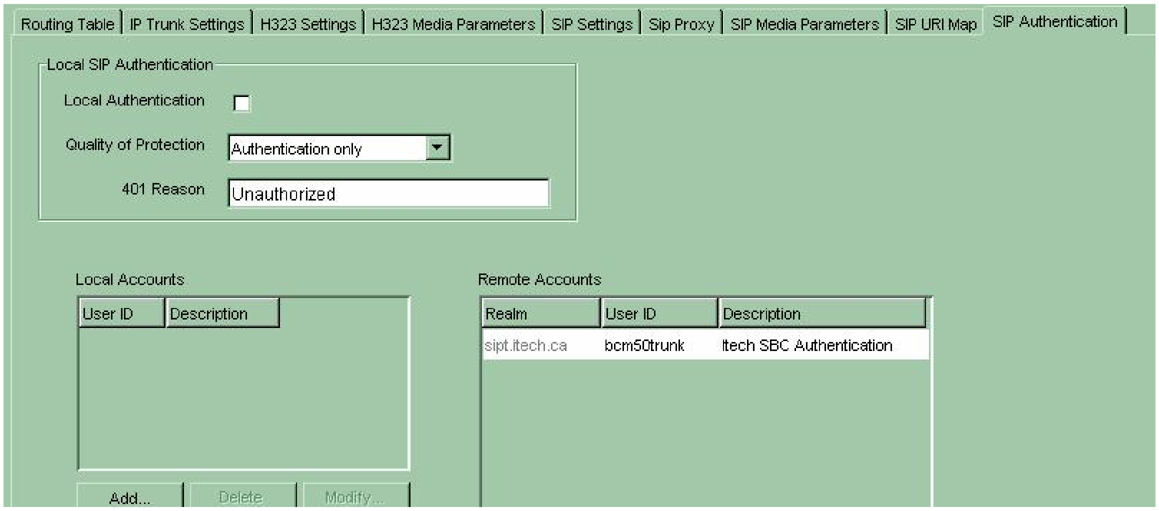


SIP Domain Names

e.164/National: blank this field
Unknow/Unknow: blank this field

All other fields should remain at the default settings.

Select Resources → telephony resources → Module IP Sets → SIP Authentication



Local SIP Authentication
Local Authentication: off

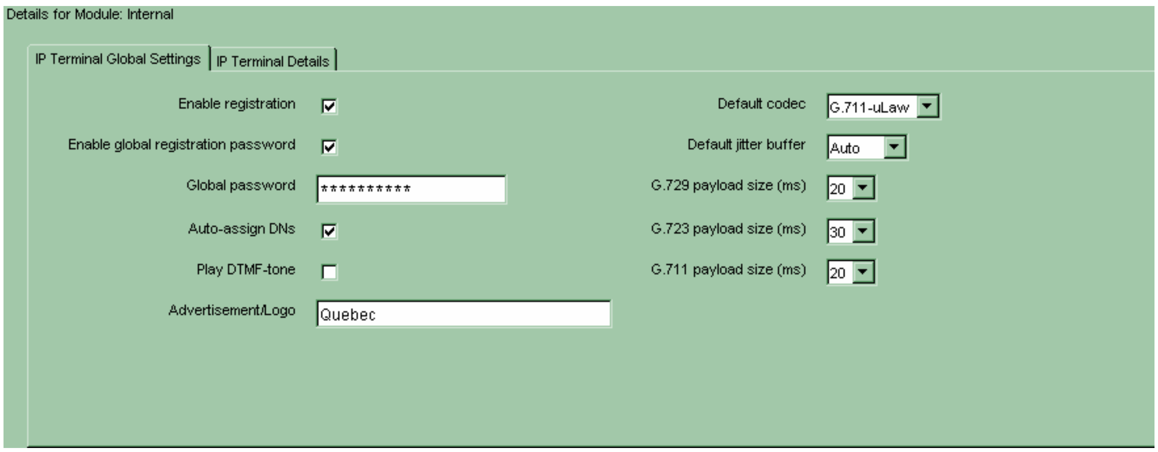
Remote Accounts

Realm: Service domain
User ID: Authentication Credentials user id
Description: a label describing the account

*Authentication Credentials will be determined and set as part of the activation process with Bell Canada

12.2 IP set Settings

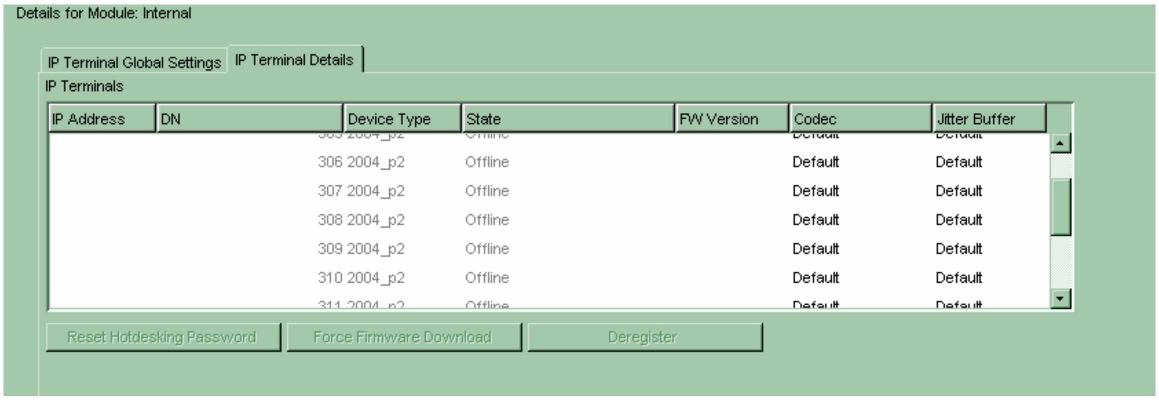
Select Resources → telephony resources → Module IP Sets → IP Terminal global Settings



In this section we are only concerned with the codec settings

Default Codec: G.711-uLaw or G.729
Default Jitter buffer: auto
G729 payload size:20
G723 payload size:30 ← G723 is not supported for the service.
G711 payload size:20

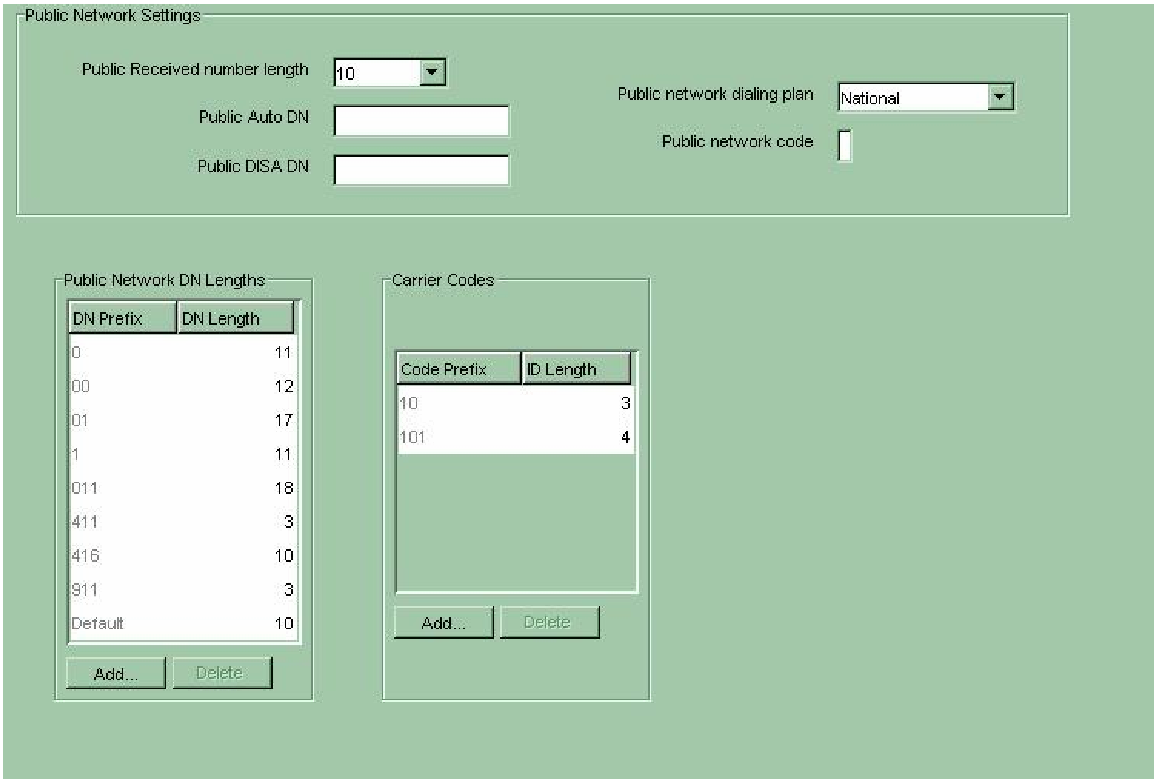
Select Resources → telephony resources → Module IP Sets → IP Terminal Details



In this section for all devices using the sip trunk these values should be set to Default or a codec and jitter buffer that is supported.

12.3 Dialing plan settings

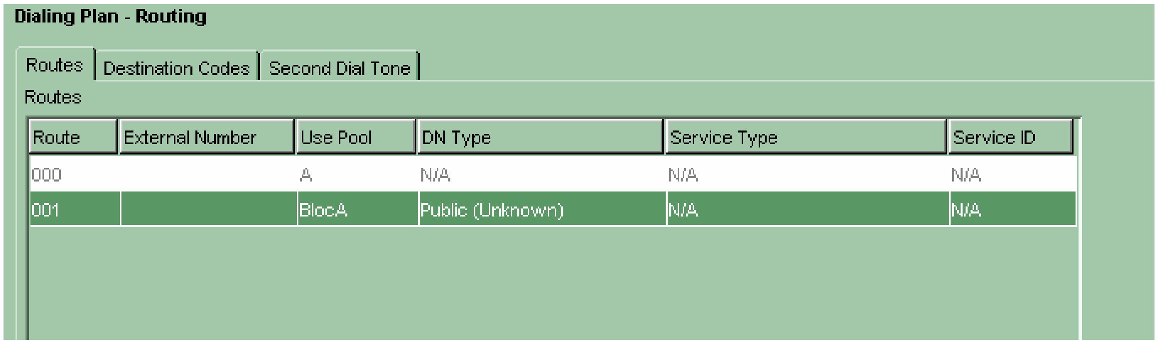
Select telephony → Dialing Plan → Public Network



Public Receive number: in our example we set this to 10 as it matches the 10 digit numbers sent from the SIP trunk service.

Public Network Dialing Plan: National ← this dictates the type of SIP messaging required for the service.

Select telephony → Dialing Plan → Routing → Routes

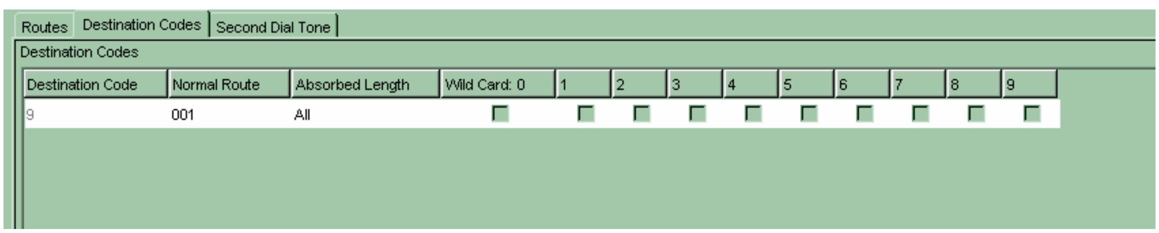


Route 001

User pool: BlocA

DN Type: Public (Unknown)

Select telephony → Dialing Plan → Routing → Destination Codes

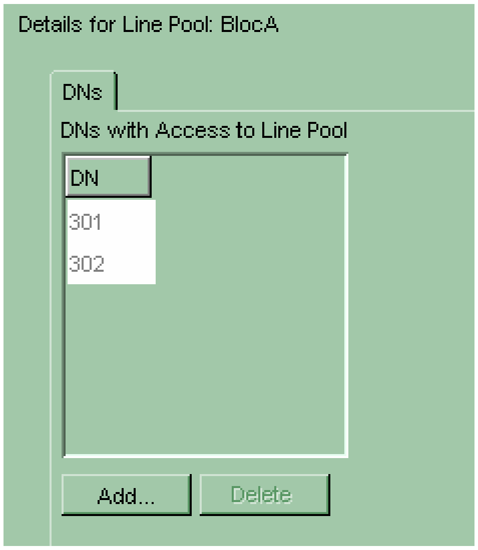


Destination Code: 9 ← this is the code to seize the SIP trunk

Normal Route: 001

Absorbed Length: all ← send all dialed digits except the 9

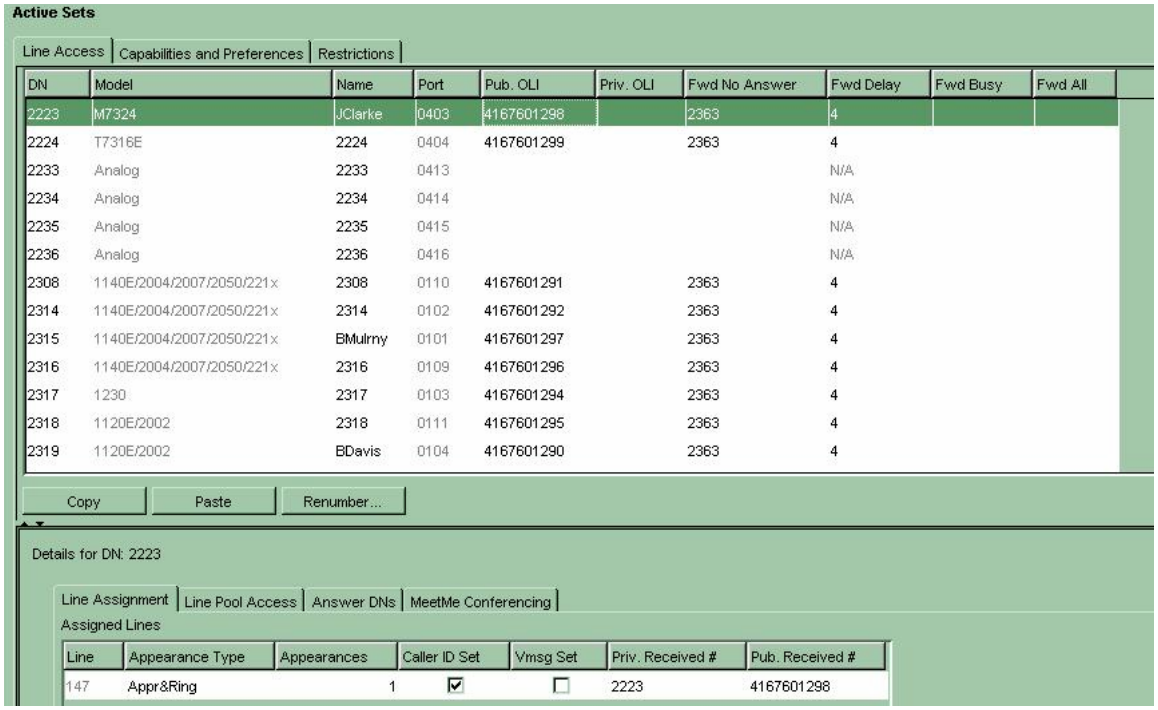
Select telephony → Dialing Plan →Line pools



The Line Pool BlocA must contain the DN's of all sets to access the SIP trunk

12.4 Set configuration

Select telephony → Sets → Active Sets → Line access → Line assignment



Active Sets

Line Access | Capabilities and Preferences | Restrictions

DN	Model	Name	Port	Pub. OLI	Priv. OLI	Fwd No Answer	Fwd Delay	Fwd Busy	Fwd All
2223	M7324	JClarke	0403	4167601298		2363	4		
2224	T7316E	2224	0404	4167601299		2363	4		
2233	Analog	2233	0413				N/A		
2234	Analog	2234	0414				N/A		
2235	Analog	2235	0415				N/A		
2236	Analog	2236	0416				N/A		
2308	1140E/2004/2007/2050/221x	2308	0110	4167601291		2363	4		
2314	1140E/2004/2007/2050/221x	2314	0102	4167601292		2363	4		
2315	1140E/2004/2007/2050/221x	BMulrny	0101	4167601297		2363	4		
2316	1140E/2004/2007/2050/221x	2316	0109	4167601296		2363	4		
2317	1230	2317	0103	4167601294		2363	4		
2318	1120E/2002	2318	0111	4167601295		2363	4		
2319	1120E/2002	BDavis	0104	4167601290		2363	4		

Copy | Paste | Renumber...

Details for DN: 2223

Line Assignment | Line Pool Access | Answer DNs | MeetMe Conferencing

Assigned Lines

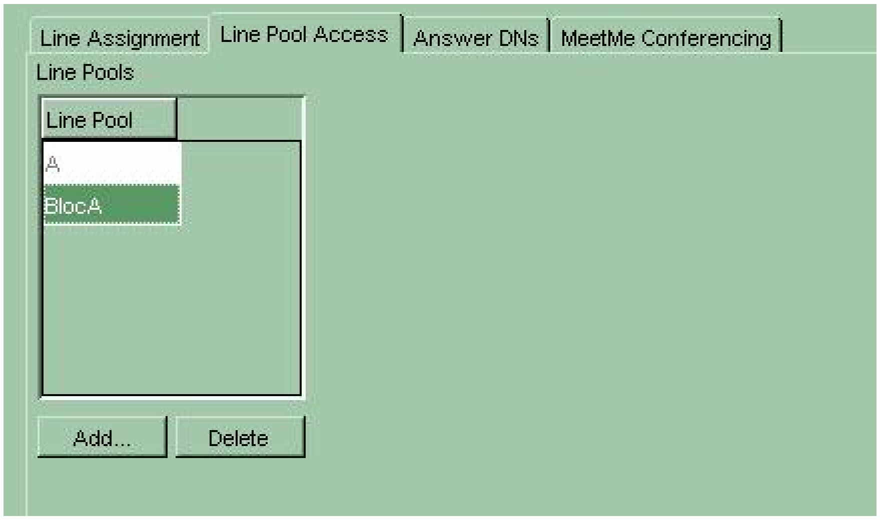
Line	Appearance Type	Appearances	Caller ID Set	Vmsg Set	Priv. Received #	Pub. Received #
147	Appr&Ring	1	☑	☐	2223	4167601298

What is important on this tab is that Pub OLI is set and the Pub Received number is set.

Pub OLI. : This is set to the number that you want to use when a call leaves the PBX toward the SIP trunk.

Pub. Received #: This is the number assigned to this set on inbound calls and will cause the set to ring.

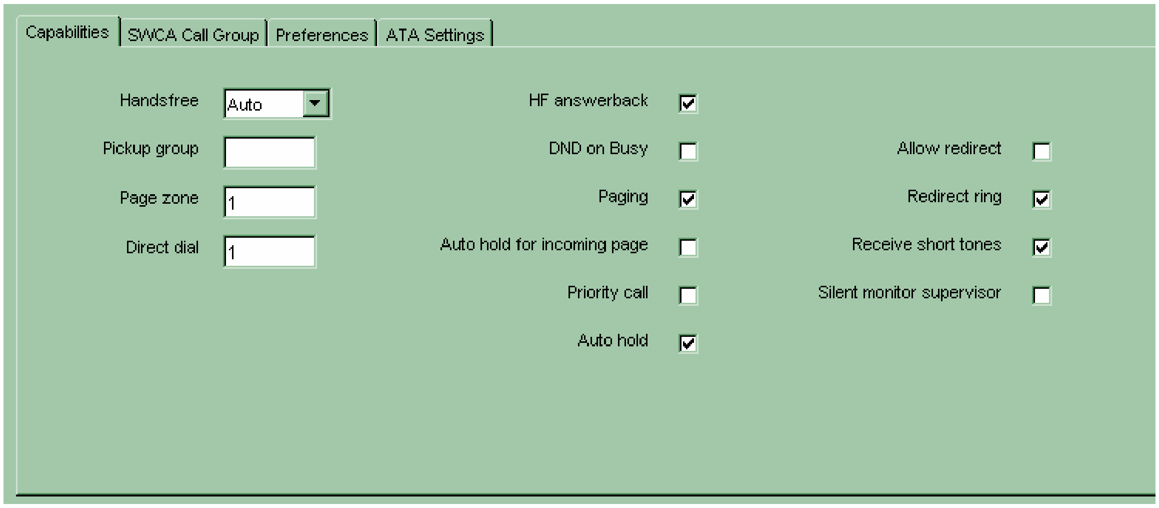
Select telephony → Sets → Active Sets → Line access → Line pool Access



For sets that need access to the SIP trunk they must have access to BlocA

12.5 Analog Sets

Select telephony → Sets → Active Sets →Active Sets→ Capabilities and preferences→Capabilities

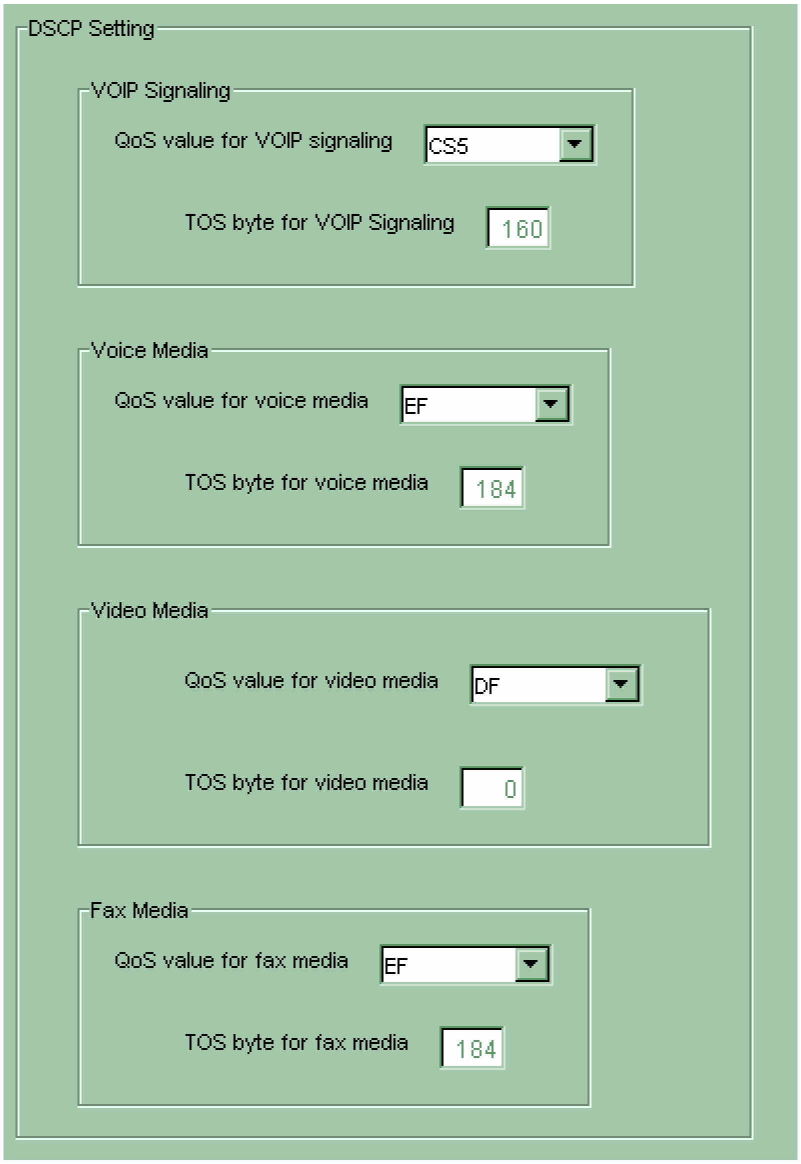


To enable DTMF you must turn on Receive short tones. If turned off DTMF do not work correctly in all cases. This setting only applies to analog sets.

Receive short tones: on

12.6 QOS

DATA Services → QOS



The QOS values for signaling and media are supported by the service. Signaling will use a value of CS5 and Media will use a value of EF. On the BCM these can be configured in the Data Services → QOS section.

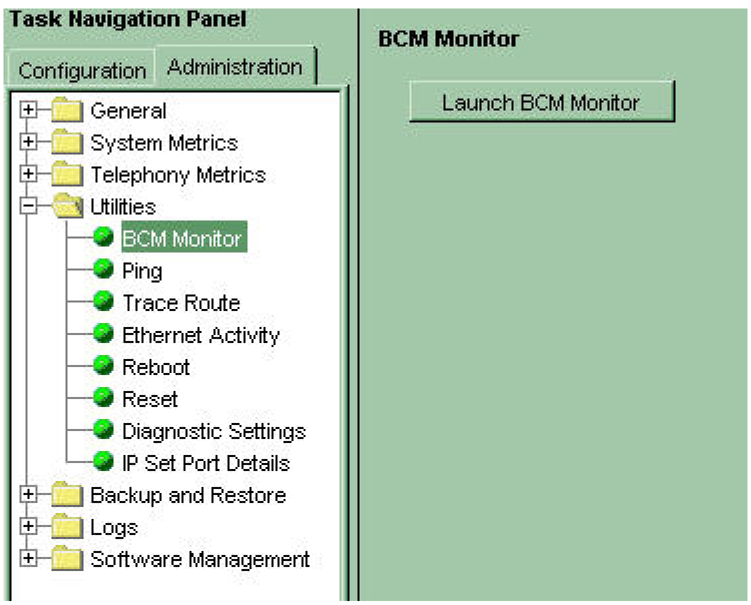
To provide QOS end to end it may be required to set similar parameters to the above on the IP sets. The configuration of these values is not covered in this document. Device specific documents should be reference for the procedures on setting these values.

13.0 FAQ & Troubleshooting

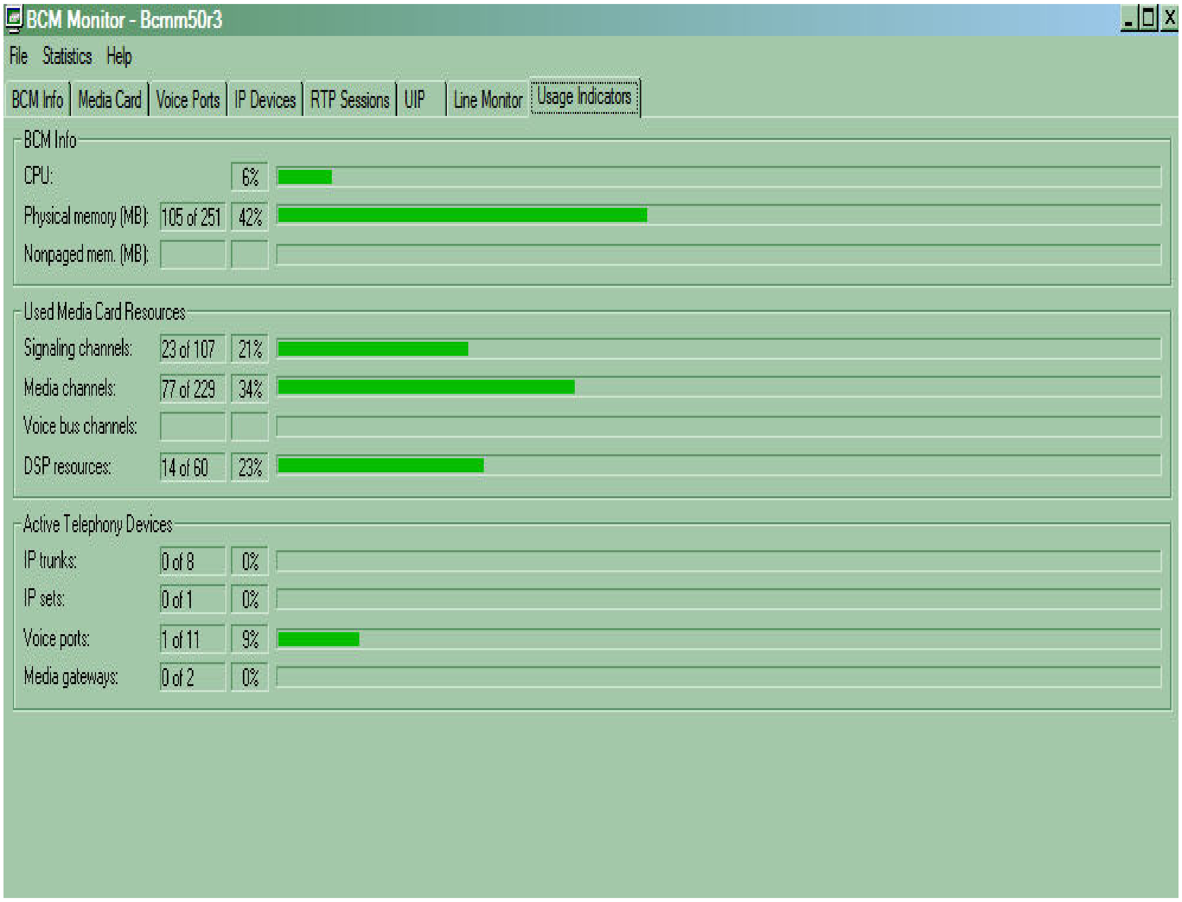
Several commands and tools can be used to troubleshoot the SIP functionality from the CPE point of view. This section identifies a few of them but many other commands and tools are available. Please refer to vendor documentation for more detailed explanation of features and functions.

13.1 System Monitoring with BCM Monitor

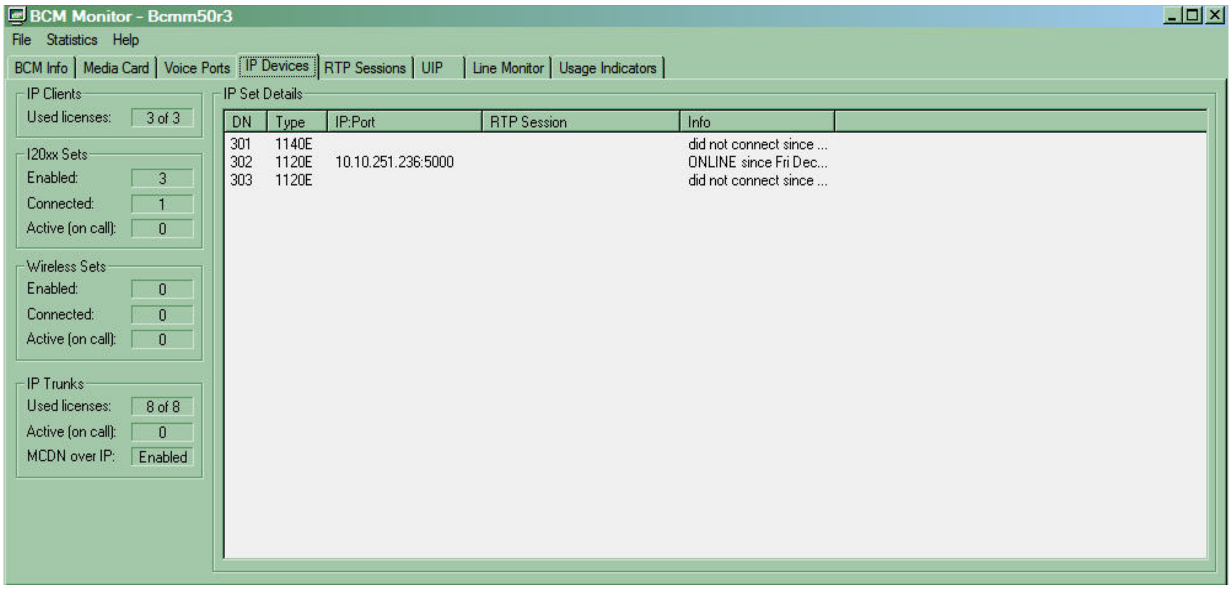
The BCM monitor tool provides information on the general health of the BCM and utilization of system resources. To launch the BCM Monitor go to administration options in the BCM element manager then choose the launch button. You can also start The BCM monitor tool for the windows start menu.



The below usage indictors tab Gives general information on CPU Memory, Media resources and Telephony Devices.

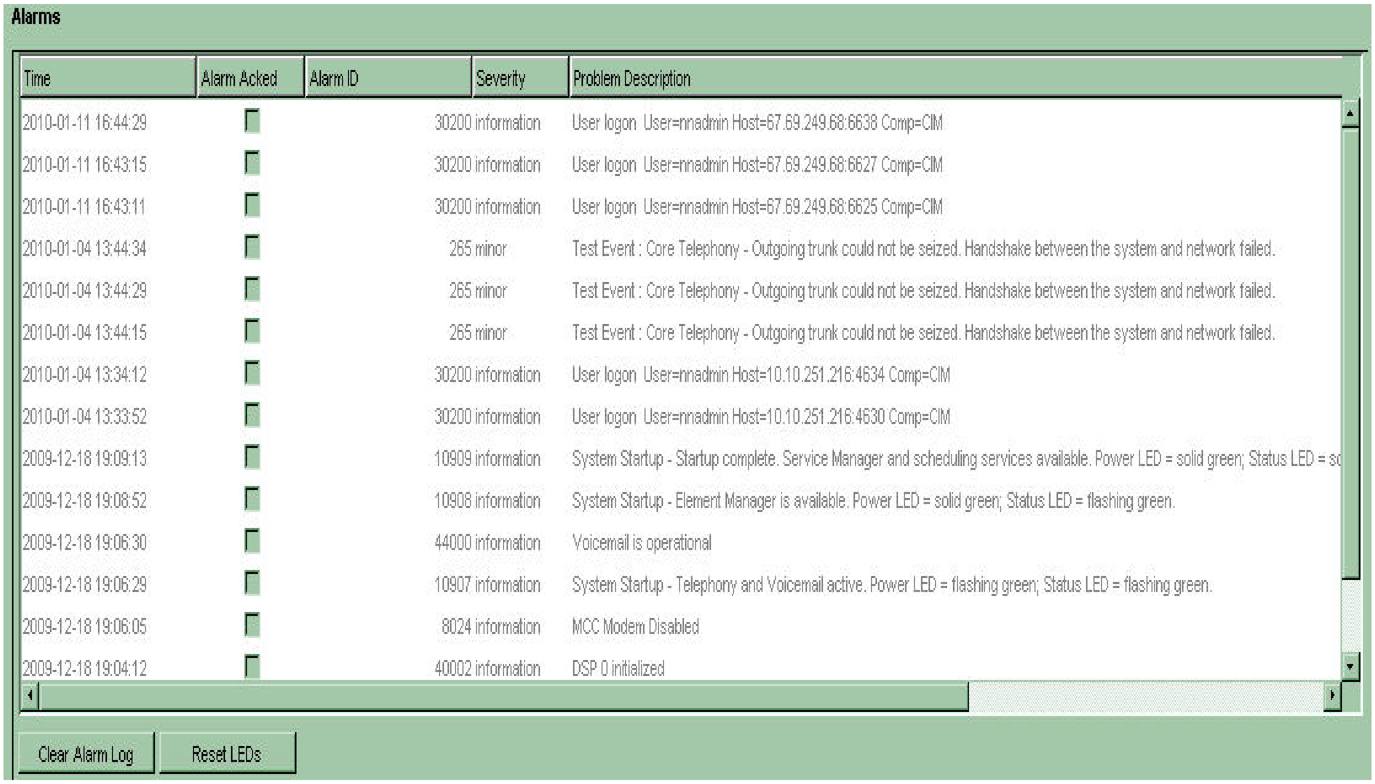


The IP devices tab show the status of IP devices connected to the system as well as devices that are not currently connected to the system. If a device is in an active call then additional information is show in the RTP Sessions and Info columns. Additional RTP session information is available on the RTP Sessions TAB.



13.2 Real-time display of BCM Alarms

The BCM generates many types of alarms looking at alarms generated can give clues to the state of the BCM and errors that may have occurred. These alarms are visible in the Administration tab under General. Various alarms can be additionally configured in the Alarm Settings section.



Alarms

Time	Alarm Acked	Alarm ID	Severity	Problem Description
2010-01-11 16:44:29		30200	information	User logon User=nnadmin Host=67.69.249.68:6638 Comp=CIM
2010-01-11 16:43:15		30200	information	User logon User=nnadmin Host=67.69.249.68:6627 Comp=CIM
2010-01-11 16:43:11		30200	information	User logon User=nnadmin Host=67.69.249.68:6625 Comp=CIM
2010-01-04 13:44:34		265	minor	Test Event : Core Telephony - Outgoing trunk could not be seized. Handshake between the system and network failed.
2010-01-04 13:44:29		265	minor	Test Event : Core Telephony - Outgoing trunk could not be seized. Handshake between the system and network failed.
2010-01-04 13:44:15		265	minor	Test Event : Core Telephony - Outgoing trunk could not be seized. Handshake between the system and network failed.
2010-01-04 13:34:12		30200	information	User logon User=nnadmin Host=10.10.251.216:4634 Comp=CIM
2010-01-04 13:33:52		30200	information	User logon User=nnadmin Host=10.10.251.216:4630 Comp=CIM
2009-12-18 19:09:13		10909	information	System Startup - Startup complete. Service Manager and scheduling services available. Power LED = solid green; Status LED = sc
2009-12-18 19:08:52		10908	information	System Startup - Element Manager is available. Power LED = solid green; Status LED = flashing green.
2009-12-18 19:06:30		44000	information	Voicemail is operational
2009-12-18 19:06:29		10907	information	System Startup - Telephony and Voicemail active. Power LED = flashing green; Status LED = flashing green.
2009-12-18 19:06:05		8024	information	MCC Modem Disabled
2009-12-18 19:04:12		40002	information	DSP 0 initialized

Clear Alarm Log | Reset LEDs

14.0 Acronyms

Acronym	Definition
DNS	Domain Name Resolution
G.711	Voice Codec (Uncompressed)
G.729	Voice Codec (Compressed)
OTG	Originating Trunk Group
PAI	P-Asserted Identity
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RFC	Request For Comment
RTP	Real Time Protocol
T.38	Fax over IP protocol
TGRP	SIP Trunk Group selection convention
URI	Uniform Resource Indicator
BCM	Business Communications Manager

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