AT&T VoIP
Nortel BCM450 Release 1.0 SIP Configuration Guide
For Use with AT&T IP Flexible Reach

Issue 0.3
2/4/2010

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1 Introduction

This document provides a configuration guide to assist Nortel Networks BCM450 administrators in connecting to AT&T IP Flexible Reach via SIP trunks.

This document does not describe procedures to configure the BCM450 for advanced functionality. For more information and procedures, please refer to the Nortel technical documentation found on the Nortel website.

1.1 Pre-IP PBX Configuration Activity

This guide assumes that the administrator is knowledgeable in IP PBX programming and operations.

An important tool that the administrators should have at their disposal prior to testing their IP PBX with AT&T IP Flexible Reach is the Wireshark network protocol analyzer. This software can be used to run traces on problem calls so the information can be shared with equipment and network engineers. This free software can be obtained at http://www.wireshark.org/.

The customer may also use TCPDUMP which can be found on most UNIX and Linux systems. The customer should have Wireshark or TCPDUMP loaded on a server that is connected to a LAN switch or hub that can monitor both the signaling and media packets on any calls between the customer PBX and the IP Flexible Reach managed router.

1.2 Customer Questions

Section 4 of this guide provides screen shots and instructions for the configuration of your IP PBX. Should you have questions regarding these instructions, please contact Brian Stegemoller at +1 (972) 685-6629; ((972) 745-5139 after 2/22/2010). When calling this number please have the following information available:

- Company name
- Company location
- Administrator name and phone number
- IP PBX name and software version

- Customer Configuration Guide – Issue number and date

1.3 Trouble Reporting

Nortel and AT&T will make every effort to quickly resolve reported troubles. The time required for trouble shooting can be reduced if the customer has the necessary detailed information available when reporting a problem. Prior to reporting a problem please provide a Wireshark or TCPDUMP trace of the failed call.

1.4 Document Feedback

IP PBX administrators who would like to provide feedback on the contents of this document should send it to Brian Stegemoller (brianstegem@avaya.com) with a copy to Al Chee (alchee@ avaya.com) and Steven Chen (stevenchen@avaya.com).

1.5 Document Change History

Issue 0.0	October 1, 2008; Draft
Issue 0.1	October 8, 2008; Changes in wording on Special Notes, Added URL to download patches (pg 10)
Issue 0.2	February 4, 2009; Added patch information for disconnect issue on page 10.
Issue 0.3	February 4, 2010 Updated Contact Information to Reflect AVAYA Merger

2 Special Notes

Emergency 911/E911 Services Limitations

While AT&T IP Flexible Reach services support E911/911 calling capabilities in certain circumstances, there are significant limitations on how these capabilities are delivered. Please review the "AT&T IP Flexible Reach" Service Guide in detail to understand these limitations and restrictions.

BCM Soft Phones not currently supported

Nortel BCM soft phones are not currently supported with the AT&T IP Flexible Reach Services.

Ring back Issues with Unattended Transfers

An unattended transfer is one in which the party initiating the transfer hangs up prior to answer by the party to whom the call is being transferred. When 2 phones are in an active call on the BCM450 and one of those phones performs an unattended transfer to certain endpoints on the AT&T network, the BCM450 phone remaining on the call will not hear ring back prior to answer.

T.38 Fax must be used

T.38 fax should be configured on the BCM450 with AT&T IP Flexible Reach service. Fax via G.711 transport should not be used on the BCM450.

Fax Issues with the AT&T HIPCS Platform

Fax calls from the BCM450 to AT&T's HIPCS PSTN Hop-off platform where the destination is a super G3 fax machine is not supported.

3 Overview

This section provides a service overview of the Nortel Business Communication Manager 450 (BCM450) IP PBX integration with AT&T IP Flexible Reach services.

- Customer Premises
  - Phones and IP PBX Server in Private Address space
  - AT&T Managed Router
- Customer Sites connect to ATT IP Border Element (IPBE)

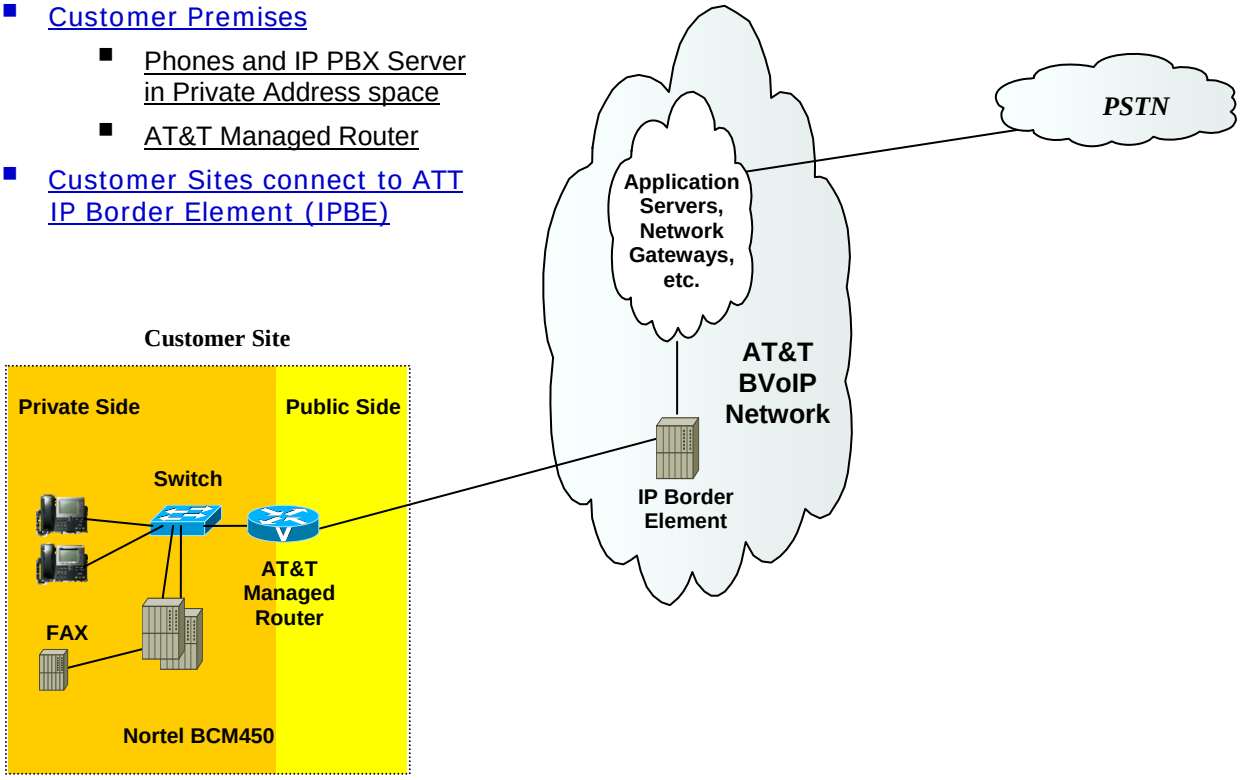

Figure 1: AT&T BVoIP Network

The Nortel BCM450 customer premises site shall consist of the following components.

- Nortel IP 200x, 11xx, 12xx phones* – These phones use the Nortel proprietary UNIStim signaling protocol to communicate to the Nortel BCM450 IP PBX for call feature and routing support. These phones can be connected to a Nortel Ethernet switch (ES 470, ERS 5520, etc.) that supplies in-line power (IEEE 802.3af) to the phones.

- The following interfaces and applications can also be used on the BCM450. They include (but are not limited to):

**Nortel, Nortel Networks, Nortel (Logo), the Globemark, CS1000 are trademarks of Nortel Networks.*

- Digital station ports supporting digital phones
- Analog station interfaces with message waiting and CLID support
- Integrated CallPilot voice mail system, auto attendant, and meet-me conferencing
- Analog trunk module for analog/POTS lines to PSTN
- Digital trunk module for PRI/T1 connection to PSTN

* RFC2833 DTMF is currently not supported on the IP Softphone 2050; thus the IP Softphone 2050 is not supported for AT&T IP Flexible Reach services.

The following routing scenarios are supported by the Nortel BCM450 IP PBX and **DO NOT** use the AT&T Call Control.

- Local Nortel BCM450 phone to other local Nortel BCM450 phone

The following routing scenarios are supported by the Nortel Networks BCM450 IP PBX and **DO** use the AT&T Call Control. For voice calls, the G.729 codec shall be used.

- Nortel BCM450 phones to PSTN (domestic US and international).
- Nortel BCM450 phones to legacy PBX site with Cisco gateway.
- Legacy PBX site with Cisco gateway to Nortel BCM450 phones.
- Nortel BCM450 phones at one Nortel BCM450 IP PBX site to Nortel BCM450 phones at another Nortel BCM450 IP PBX site

If the customer has subscribed to Calling Plans B and C (Local), then the following routing scenarios are supported by the BCM450 IP PBX and **DO** use the AT&T Call Control. For voice calls, the G.729 or G.711 codec may be used. BCM450 selects G.729 as the highest priority codec.

- Inbound PSTN to BCM450 phone
- Outbound local PSTN calls from the BCM450 phones
- Outbound local N11 (i.e. 411, 911) calls from the BCM450 phones

T.38 Fax was tested and is supported on the BCM450 with the AT&T IP Flexible Reach services to/from the following:

- PSTN
- Legacy PBX site with Cisco gateway
- Another BCM450 IP PBX site

4 Configuration Guide

This configuration guide specifies the Nortel BCM450 screens that must be configured and updated to support the AT&T IP Flexible Reach services.

4.1 Nortel BCM450 Version and Feature Requirements

The Nortel Networks BCM450 must be running at least software version 8.0.1.05.329. You can check the version of BCM450 by viewing the following screen under ***Administration → Software Management → Software Update History***.

This is the supported base release that is required for AT&T IP Flexible Reach services. Any 8.0.x.x.x software release greater than the aforementioned software version is acceptable.

This software release 8.0.x.x.x refers to BCM450 Release 1.0 SIP.

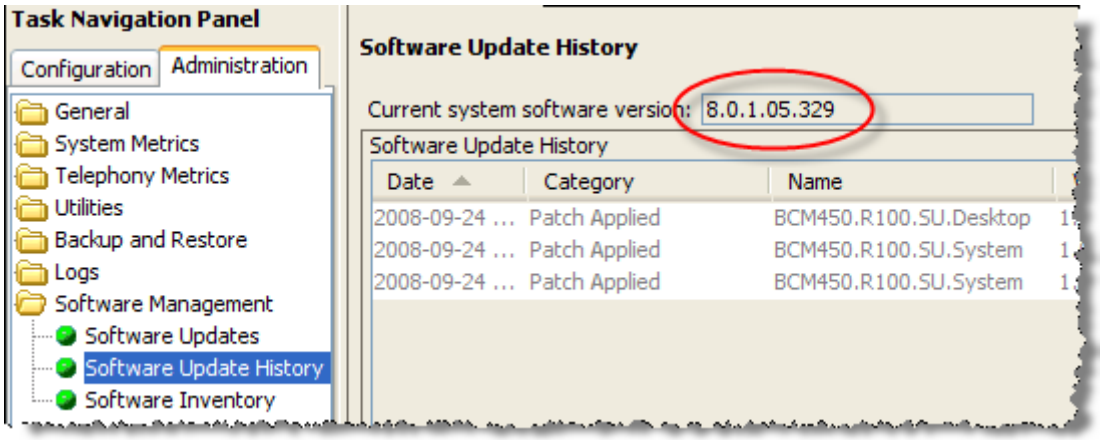

Figure 2: BCM450 Software Version Number and Patch List

The following BCM450 patches must be applied. To verify any installed patches on the system: under ***Administration → Software Management → Software Update History***.

Patch Name/ID	Version	Description
BCM450.R100.FEPS-22 (or any SUs that contain this)	--	Update to FEPS; contains the following fix: *SIP calls not disconnecting properly*

The patches can be downloaded on the Nortel Technical Support site: http://support.nortel.com/go/main.jsp

Then go to the “Voice, Multimedia & Unified Communications” section. Under “IP-Enabled & Pure IP Networks,” go to the “Business Communications Manager 450” link. Under “Software,” go to “Patches.”

4.2 IP Trunks

Voice over IP (VoIP) trunks, are signaling channels that simulate how CO lines work. However, VoIP trunks transmit data to the IP network over a LAN or IP network rather than over physical lines. Once 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.

VoIP trunks use line numbers, which appear under ***Configuration → Telephony → Lines → Active VoIP Lines***. To access VoIP lines, you need to enter software keycodes. Each keycode supports a specific number of trunks. No entries appear in the Enabled VoIP lines field until you complete the IP Trunks Settings field, which displays when you click IP Trunks under ***Configuration → Resources → Telephony Resources → IP trunks***.

Note: The BCM450 (Release 1.0) offers 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 with AT&T IP Flexible Reach service.

Customers that desire a lower cost or have no requirements for H.323 should choose the SIP Gateway Trunk License option.

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). The VoIP line pools are assigned to routes, which, in turn, are configured with destination codes that route calls to the BVoIP network.

Check under ***Configuration →Telephony → Lines → Active VoIP Lines*** to see if trunks have been allocated. You should have a number of IP trunks displayed. The total number of lines indicated corresponds to the number of IP trunks licensed by Nortel for your BCM450. See figure below.

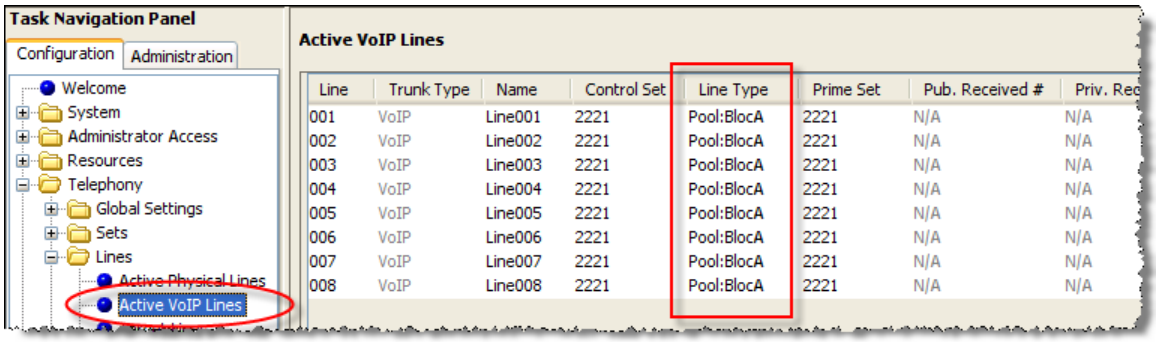

Figure 3: Available VoIP Trunks

Note: If no active VoIP lines are present, check to see if either SIP or IP Trunks licenses are installed. If so, try restarting the "feps" service on the BCM450. This can be done by going to ***Administration → General → Service Manager***, and restarting the "feps" service. Also, check under ***Configuration → Resources → Application Resources***, and ensure that the SIP (or IP) Trunks Minimum and Maximum values are set to 0 and MAX, respectively.

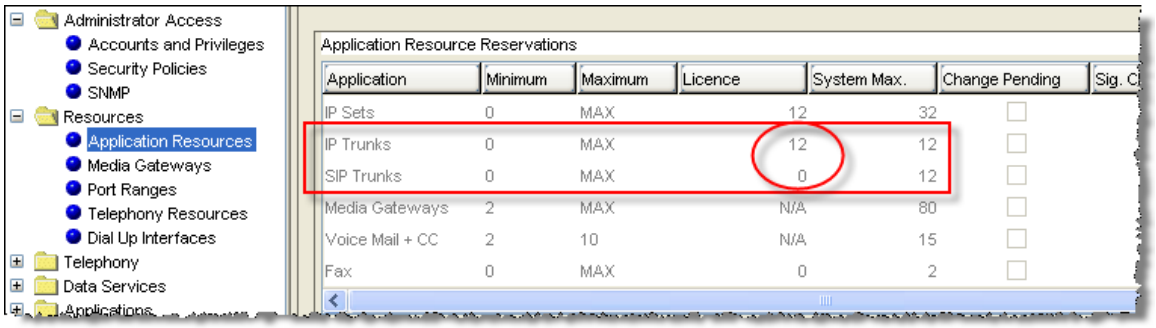

Figure 4: Application Resource availability of SIP (or IP) Trunks

Also ensure that there is a number under "**Licence**" for either trunk entries. Otherwise, new keycodes/licenses need to be retrieved for the BCM450.

Under ***Configuration → Telephony → Dialing Plan***; select "**Line Pools**"
In this case we selected "**BlocA**" under the "**Pool**" tab. We will use this line pool to access the VoIP trunks. Additionally, all DN numbers that need to access the VoIP trunks must be added to this pool. In other words, ensure that the DN numbers have the same line pool assigned as the VoIP trunks (in Figure 4). Please see the following screen shots for an example configuration.

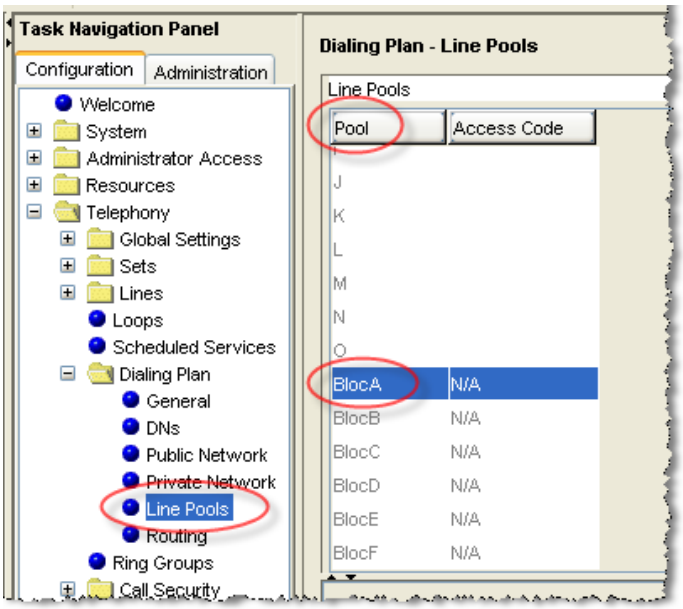

Figure 5: Assigning Line Pool to IP Trunks

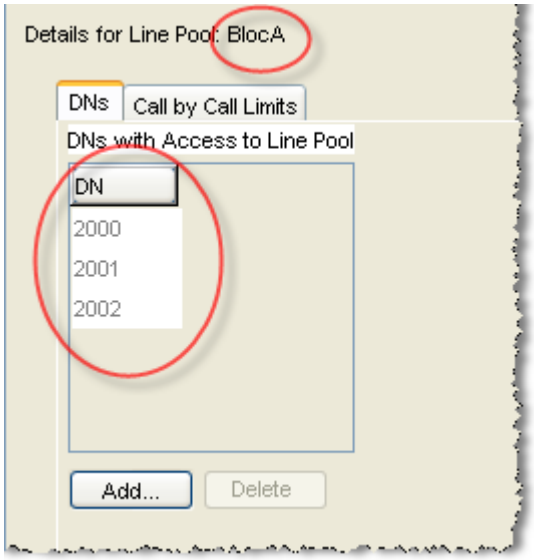

Figure 6: Assigning DN to Line Pool

Under ***Configuration → Telephony → Dialing Plan → Public Network***, we define the Public Received number length to "4" digits and Public network dialing plan to "National." This tells the BCM450 to look at the last four digits of a public received number, and alert the assigned DN of an incoming call.

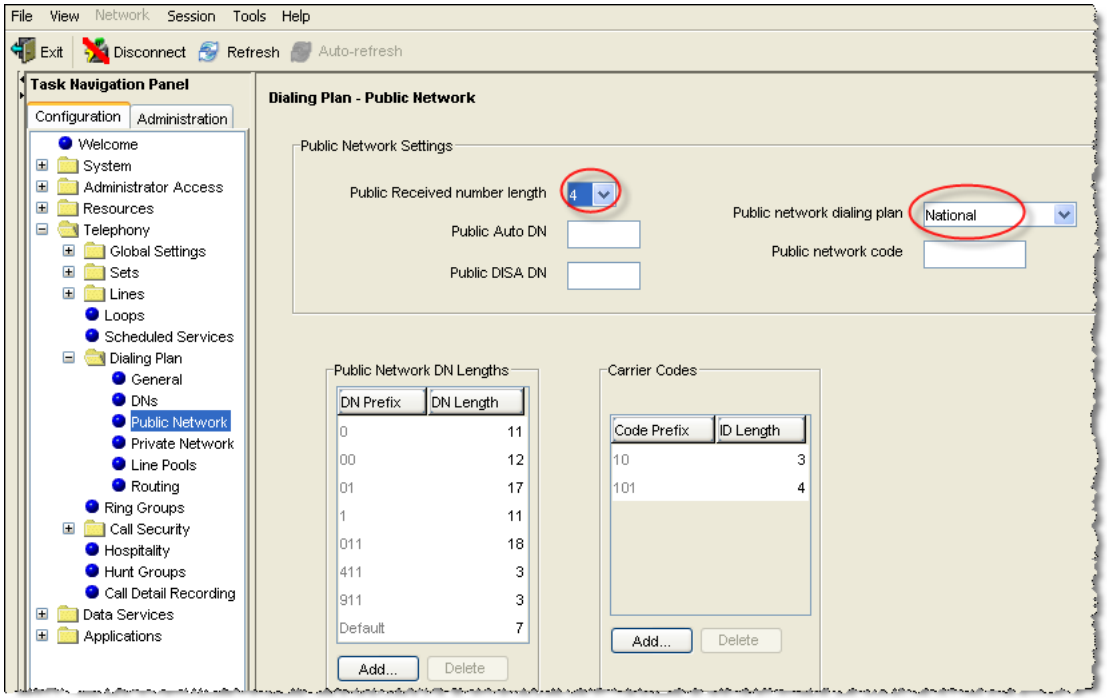

Figure 7: Configuring Public Received Number Length and Dialing Plan

Under ***Configuration → Telephony → Dialing Plan → Routing***: Select the "**Routes**" tab and ensure there is an entry for "**BlocA**" with DN Type "**National**." In this case "**001**" is the route number and the DN Type is specified as "National." See figure below.

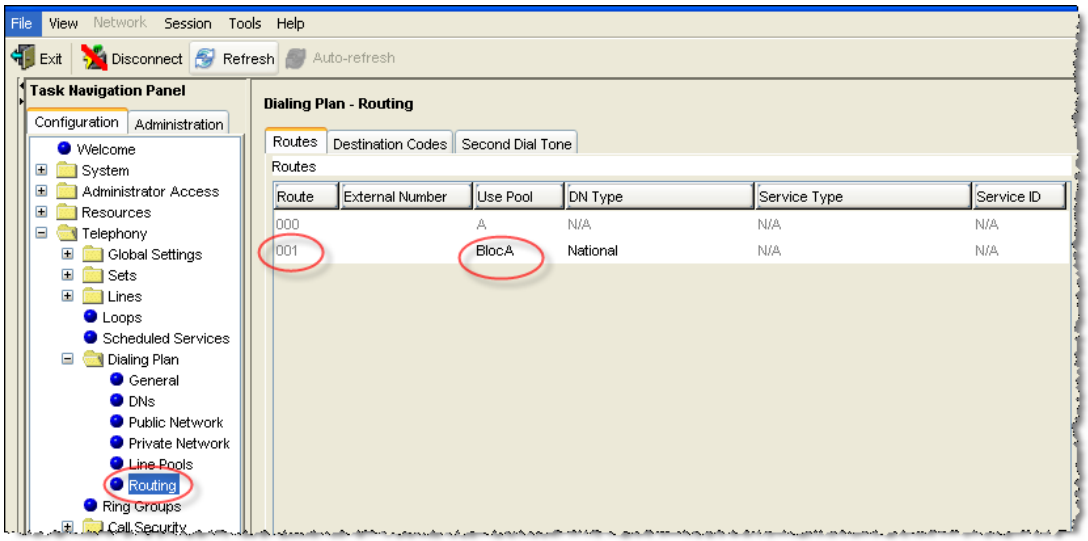

Figure 8: Assigning a Route for IP Trunks

Under ***Configuration → Telephony → Dialing Plan → Routing***, select the "**Destination Codes**" tab to assign a destination code for the IP trunks. Configure a destination code "**9**" or to whatever code you want to access for outside (IP off-net) calls that will be presented to the AT&T service for routing. In this case, when "**9**" is dialed we wish to push the dialed string to the IP trunk for routing. See figure below.

Note: When completing the Technical Questionnaire Section 6.0 Dial Plan Information and the **private dial plan** is selected (**YES**), then please refer to *Appendix 'A': Configuring Destination Code with Wildcard*.

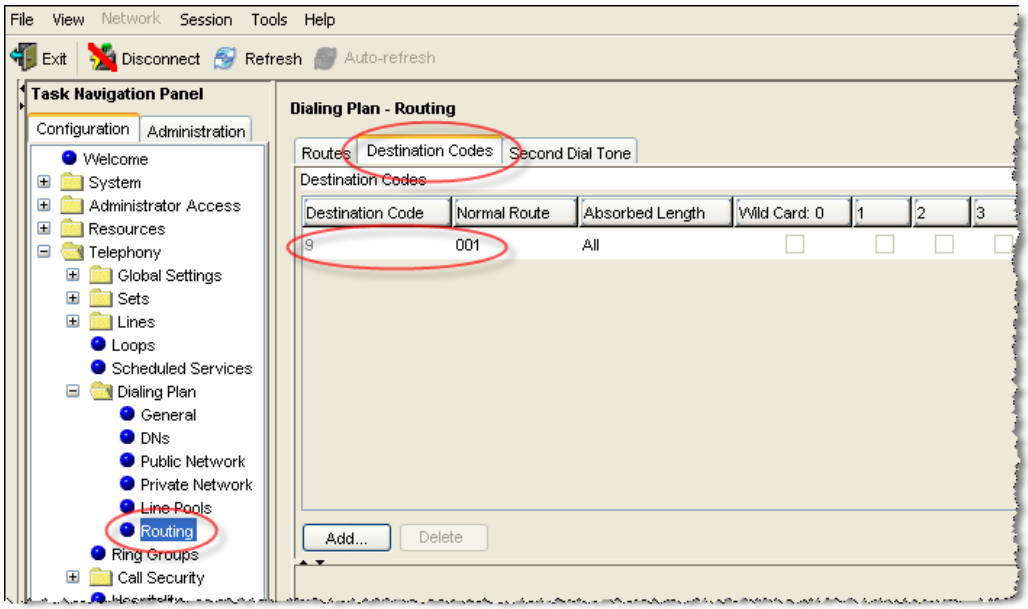

Figure 9: Assign Destination Code to Access IP Trunk Routes

4.3 SIP Proxy Parameters

Under ***Configuration → Resources → Telephony Resources:*** Select module type "**IP Trunks**" and click on the "**Sip Proxy**" tab (see figures below).

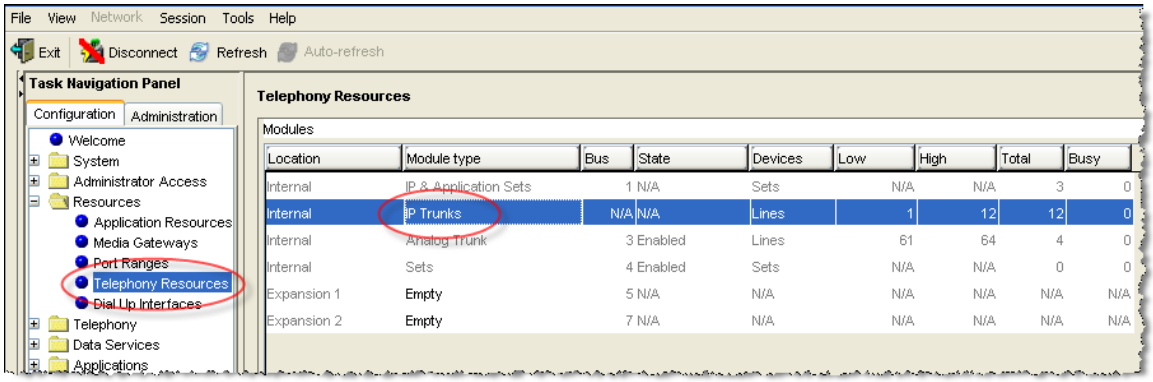

Figure 10: Selecting IP Trunk Module

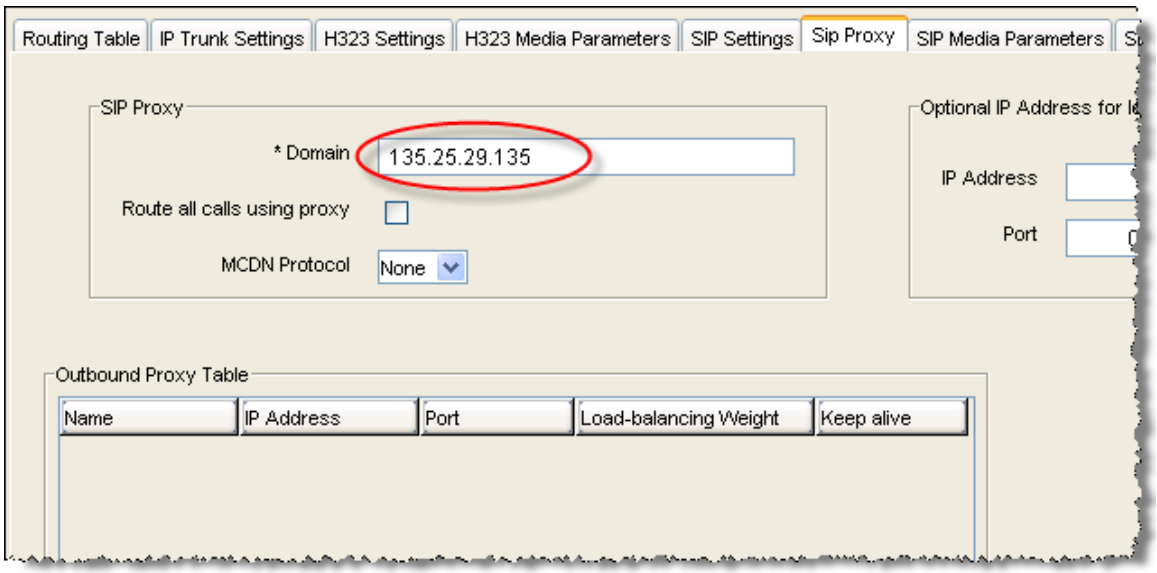

Figure 11: Sip Proxy Parameters

For IP Flexible Reach: Populate the "**Domain**" as the IP address of the AT&T IP Border Element.

Contact your local AT&T Customer Care representative for the IP addresses and more information.

4.3.1 Failover to Secondary AT&T IP Border Element (IPBE)

Backup to a secondary AT&T IP Border Elements is supported on the BCM450 using the SIP "OPTIONS" keep-alive approach. The BCM450 will send SIP "OPTIONS" messages to the AT&T IPBEs listed in the Outbound Proxy Table and send VoIP calls based on whether or not the BCM450 receives a SIP response from the IPBEs and the Load-balancing Weight values set for each entry.

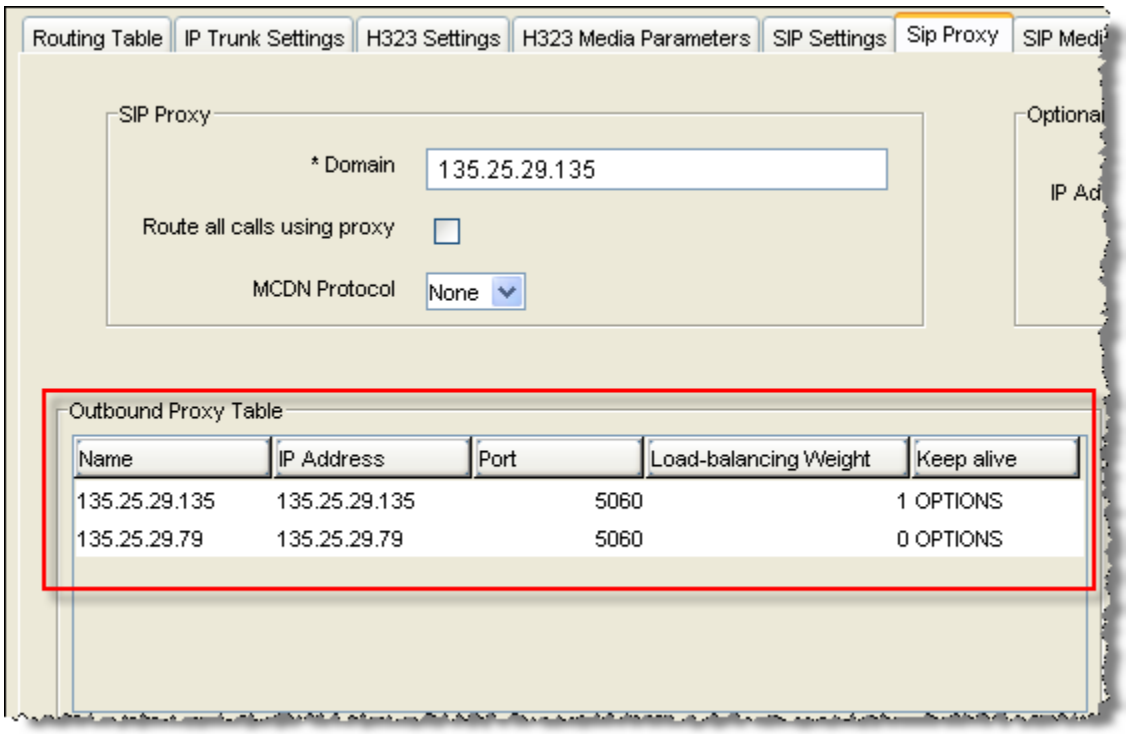

Figure 12: Outbound Proxy Table for failover to secondary AT&T IPBEs

A Load-balancing weight value of 0 means the IPBE is used as last resort. Any non-zero number value indicates the ratio of calls (the specific IPBE's load-balancing weight to sum of all IPBE's load-balancing weights) the BCM450 will send out to each IPBE. In this example, the BCM450 will send 1 call for every 1 call made (in essence, all calls) to the IPBE with IP address 135.25.29.135 and use 135.25.29.79 IPBE as backup.

Additionally, the AT&T IP Flexible Reach service will send incoming calls to the BCM450 from multiple IP Border Elements. The BCM450 will accept calls from any Border Element without additional configuration.

4.4 Media Parameters

Configuration → Resources → Telephony Resources:
Select module type "**IP Trunks**" and click on the "**SIP Media Parameters**" tab. Within this screen; ensure that all values are exactly as the sample screen shot shown below:

- **1st Preferred Codec:** **G.729***
- **2nd Preferred Codec:** **G.711-uLaw**
- **Voice Activity Detection:** **Disabled***
- **Jitter Buffer:** **Auto**
- **Fax transport:** **T.38**
- **G.729 payload size:** **20**
- **G.711 payload size:** **20**

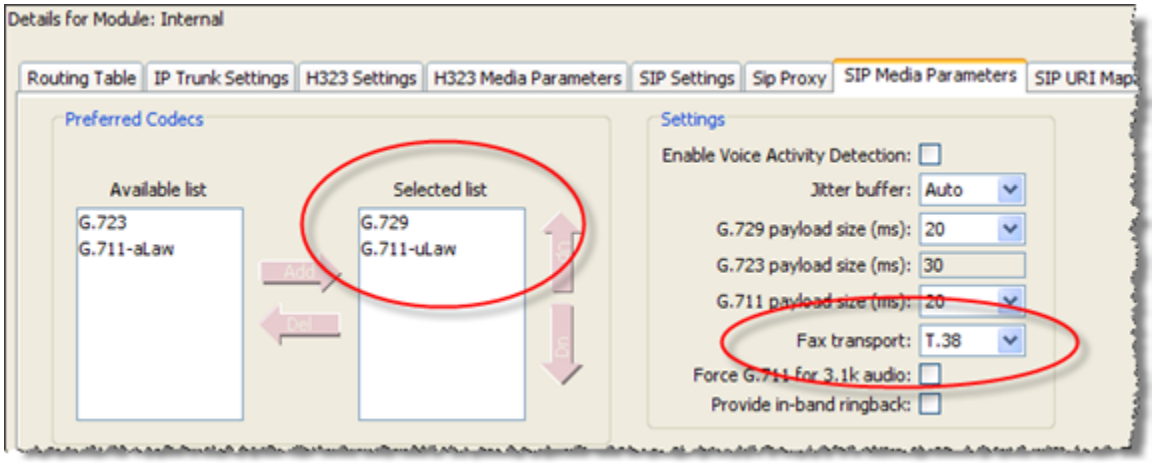

Figure 13: SIP Media Parameters

***** For default configurations, G.729A codec will be used for voice calls. However, in the case that G.729B needs to be configured instead of G.729A, ensure that "Enable Voice Activity Detection" is checked.

4.5 SIP URI Map

Configuration → Resources → Telephony Resources:
Select module type "**IP Trunks**" and click on the "**SIP URI Map**" tab.

Ensure that the e.164 / National SIP domain name is blank.

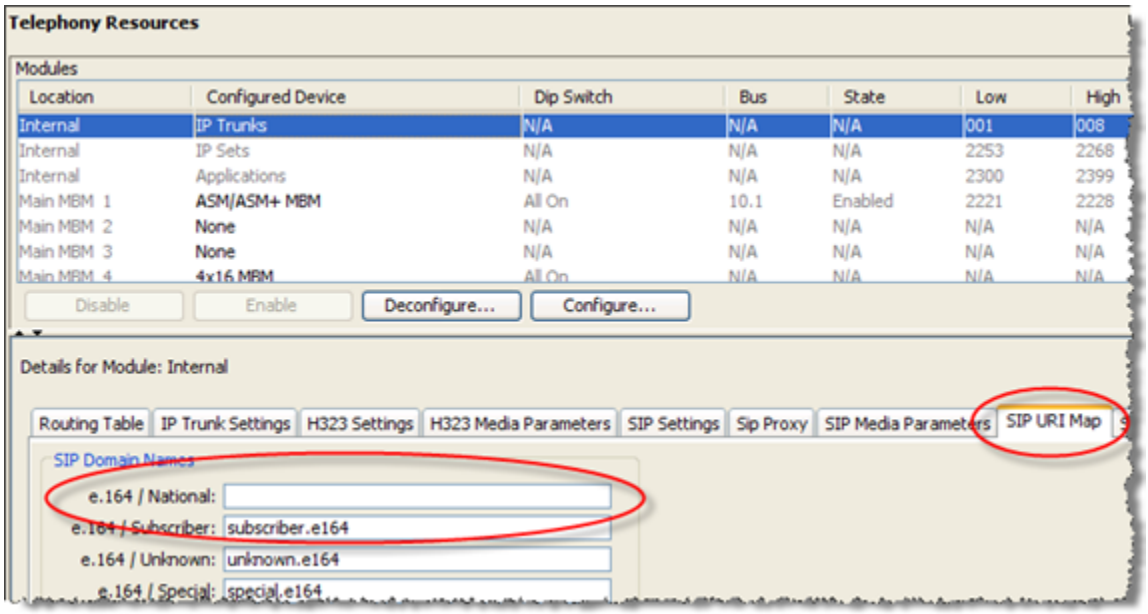

Figure 14: SIP URI Map

4.6 Port Ranges

Configuration → Resources → Telephony Resources:

Select "**Port Ranges**" and use the values shown below. The default RTP ranges are from 28000 to 28255. This range is used for fax (T.38), digital phones and analog phones. The media gateway port ranges are configurable.

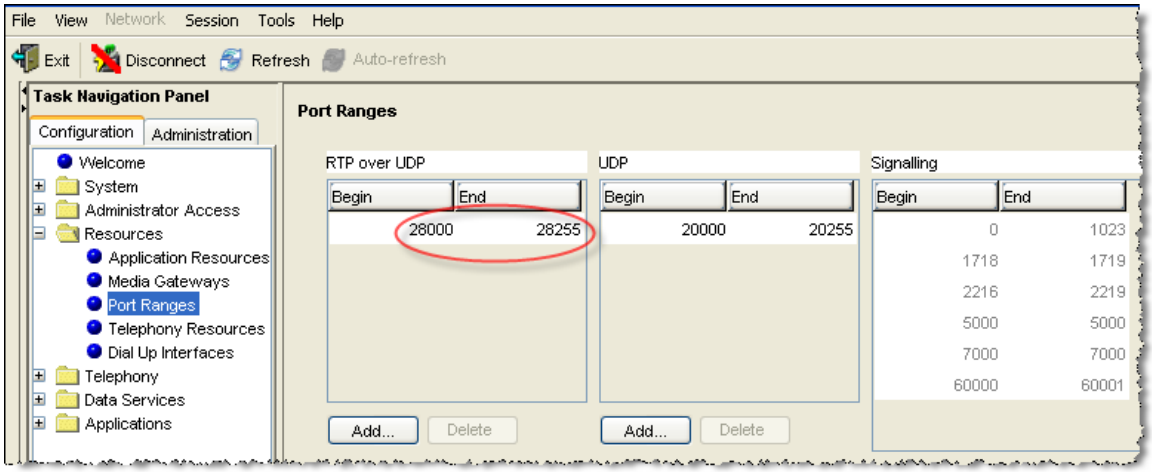

Figure 15: Media Gateway Port Ranges

The BCM450 IP phone's RTP and RTCP port range are 51000-51399. Each IP phone call uses two ports. The default port range for RTP and RTCP are not configurable.

4.7 Configuring Outgoing Calls from a BCM450 DN to AT&T IP Flex Reach

Configuration→ Telephony→ Sets → All DNs:

We will now associate the private DN number with the DID number. In the example below; 2000 is entered in the "**Private OLI**" field and 7323683478 is entered in the "**Public OLI**" field. This example enables "calling number translation" (outgoing) for this particular DN number. See figure below.

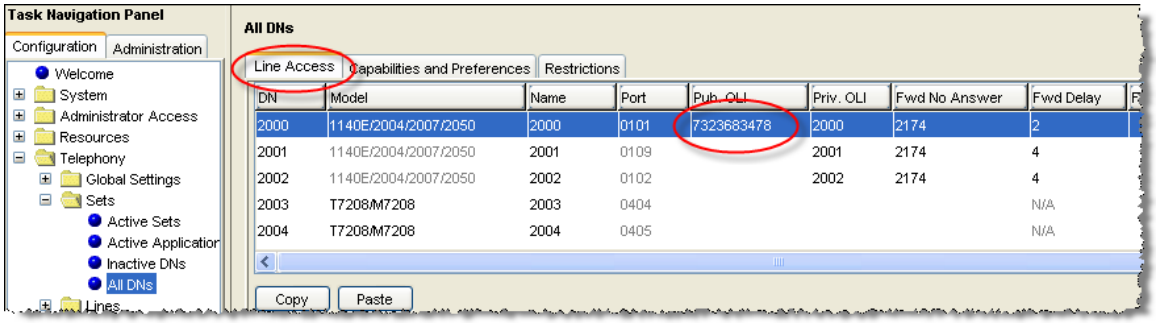

Figure 16: Configuring DID for Outgoing Calls

4.8 Configuring Incoming Calls from AT&T IP Flex Reach to a BCM450 DN

Configuration→ Telephony→ Sets → All DNs:

We will now configure the "called number translation" (incoming) for the DN number. In our example, go to the "**Line Assignment**" tab located at the bottom of the "**Line Access**" page. Enter 2000 in the "**Priv. Received #**" field; then enter the last four digits (or however many digits specified in Figure 9) of the DID (Public number) in the "**Pub. Received #**" column. Incoming DID calls will be routed to telephones, based on the trailing portion of the digits received by the network. For example, incoming calls from the AT&T IP Flexible Reach network will deliver a ten digit DID number, e.g. 7323683478. The BCM450 will route the call using the last four digits, e.g. 3478. Additionally, this configuration will allow incoming 4-digit dialing plan calls from the IP Flexible Reach network, e.g. 3478.

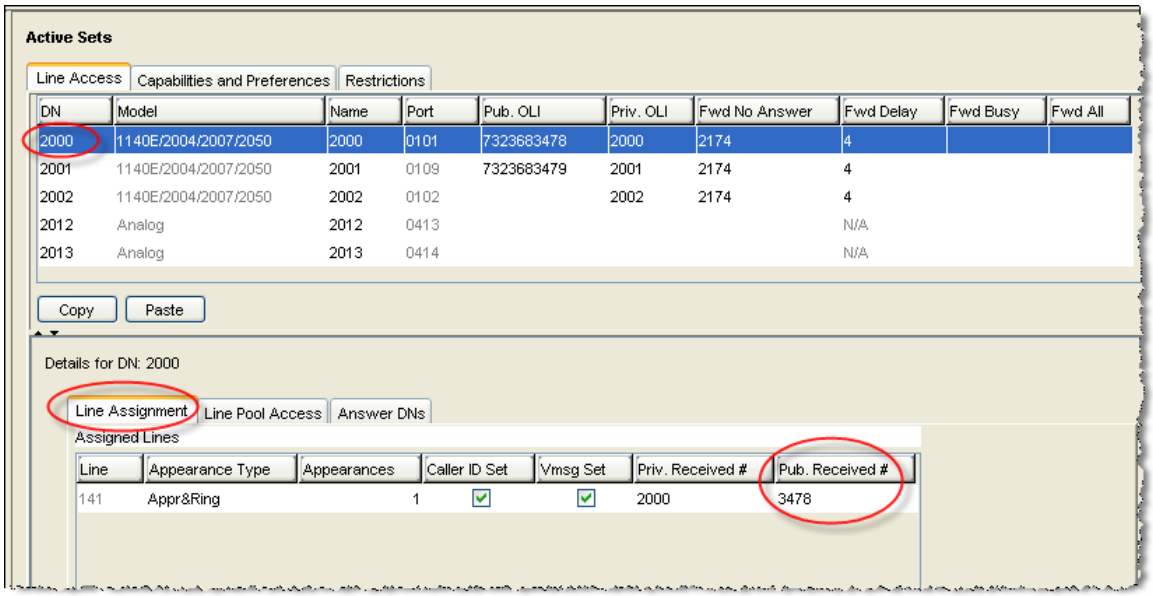

Figure 17: Configuring DID for Incoming Calls

Configuration→ Telephony→ Lines → Target Lines

To display the DID number on the IP phone LCD screen; under the "**Target Lines**" tab click on the assigned "Line" number of the DN you want to program. In our example below we click on "**Line 141**"; enter "3683478" in the "**Name**" field. See figure below.

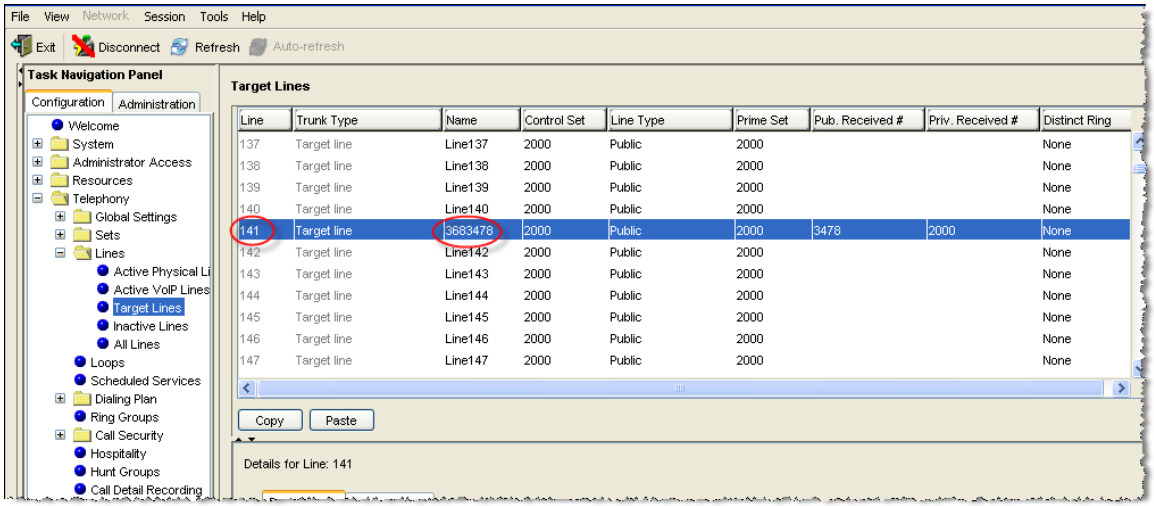

Figure 18: Display DID on IP Set LCD

Additionally, all telephone sets that need to access the VoIP trunks needs to be configured with the designated "**Line Pool" code**. In our example, we defined "**BlocA**" as the code to access the VOIP trunks. See figure 19 below.

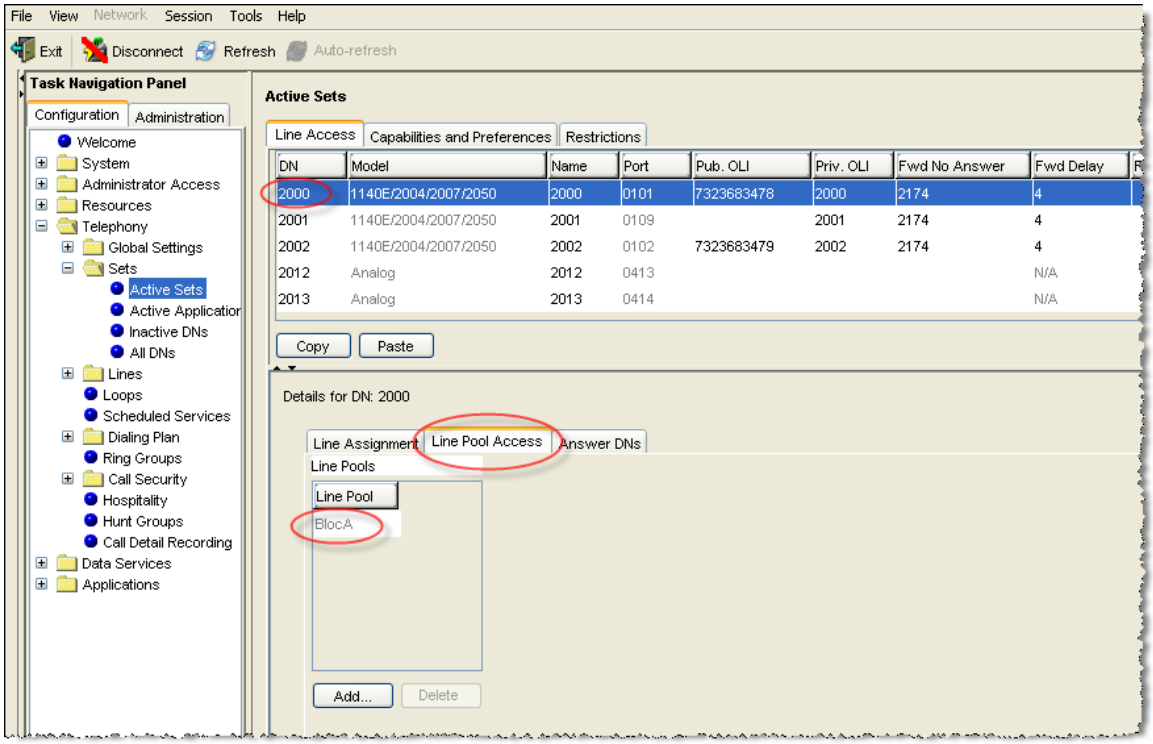

Figure 19: Assign Line Pool to IP Sets

5 Troubleshooting

This section provides some tips about troubleshooting problems

5.1 System Monitoring with BCM Monitor

A valuable application for performance monitoring is the BCM Monitor. It allows the BCM administrator to see the current status of various parts of the BCM system. Statistical information is provided on system throughput and other performance-related information, including system CPU usage (graph or table format) and memory usage (graph or table format).

If a performance display is active, it is automatically updated with real-time performance information in user-selectable time increments.

The focus of the real-time monitoring capabilities is:

- Overall system status
- Utilization of resources on the Media Services Card (e.g. signaling channel usage)
- Operation of telephony applications (e.g., Messaging, Call Center, etc.).
- IP telephony activity
- D-channel monitoring for PRI, BRI and VoIP trunks

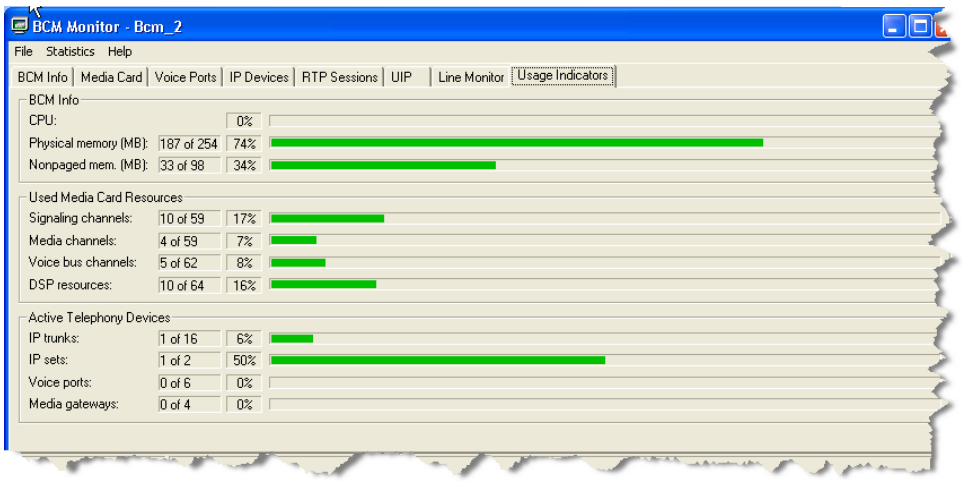

Figure 20: System Monitoring Example

The BCM Monitor application can be downloaded to an administrator's PC from the BCM and pointed at a specific BCM's IP address for monitoring. Multiple

instances of the BCM Monitor application can be used on a single PC to monitor several remote BCM systems at the same time.
Backward version compatibility is supported.

All of the registered IP devices can be viewed with the BCM Monitor. The screen shot below depicts IP Phone type, DN number and IP address of each registered IP phone. Additionally, if the device is active on a call the RTP session information is also displayed.

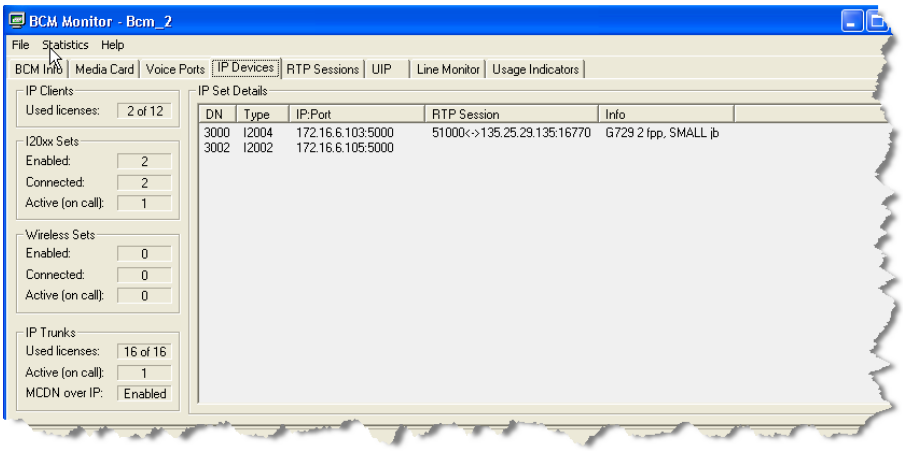

Figure 21: IP Device Listing

The end-to-end RTP sessions per IP call can also be displayed with the BCM Monitor. The example below depicts an end-to-end call.

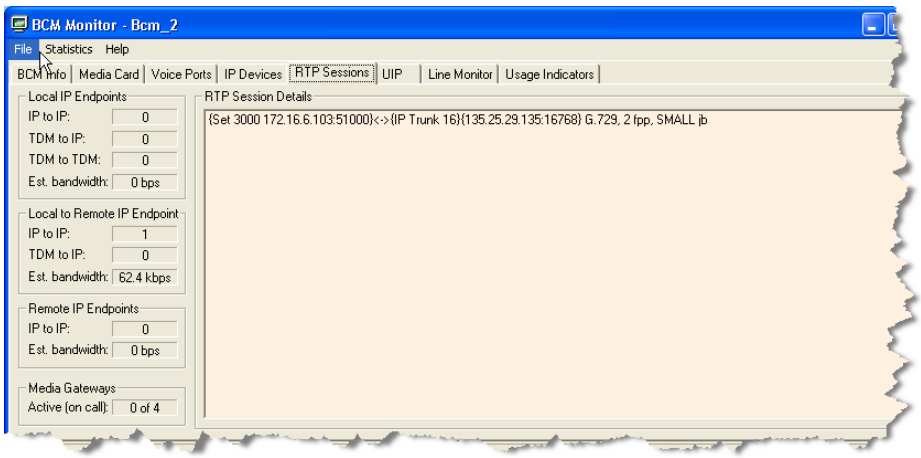

Figure 22: RTP Session Information

The BCM Monitor can be used to monitor incoming and outgoing trunks to determine if trunks are being busy or if they are idle. The example below depicts utilized lines used by local and remote telephone/DN numbers.

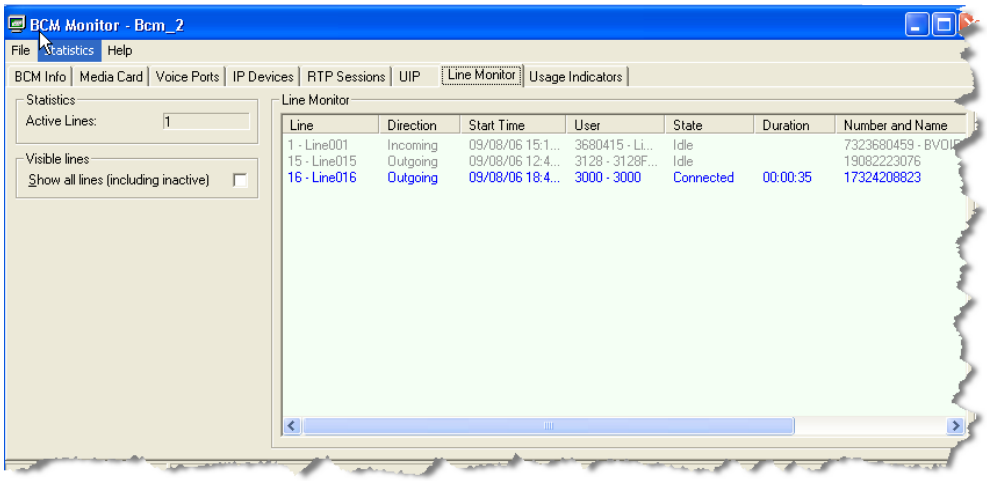

Figure 23: Line Monitor Information

The BCM Monitor can also be used to monitor all types of system usages. The following are some parameters that can be monitored:

- CPU utilization
- Physical memory
- Media card DSP utilization
- IP sets and IP Trunks
- Voice ports and media gateway usage

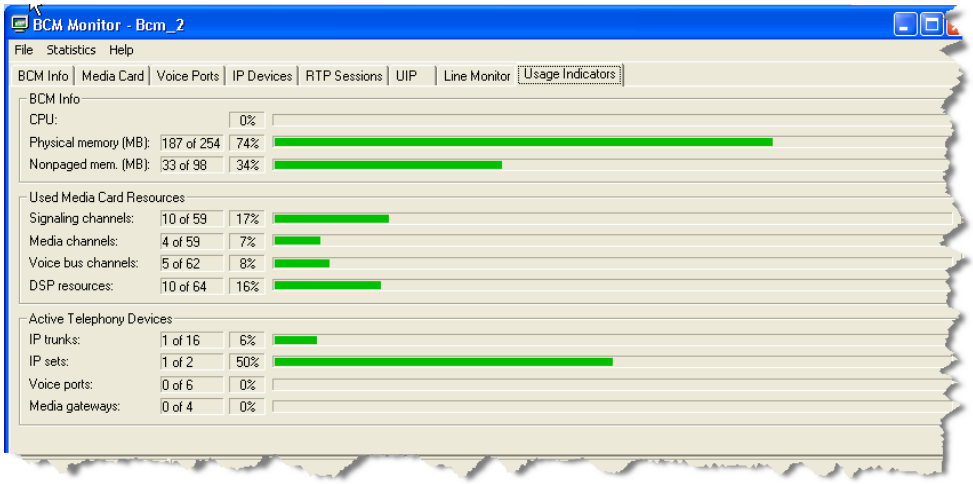

Figure 24: System Resources

5.2 Real-time display of BCM450 Alarms

Administration→ General →Alarms

The BCM450 provides extensive alarm logs along with severity and problem descriptions. The following is an example screen shot of the "Alarms" display:

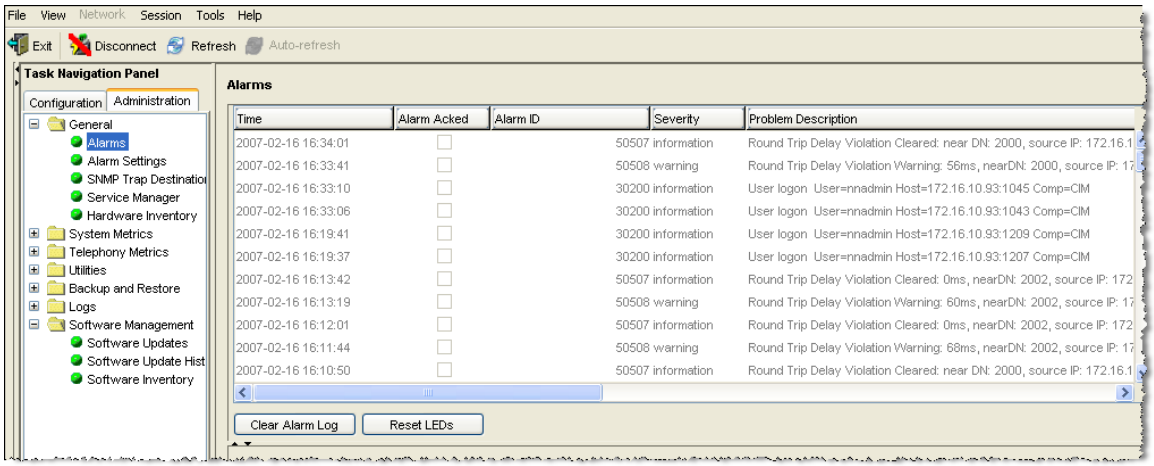

Figure 25: BCM450 Alarms Page

5.3 Log Management

Another extremely useful tool is the “Log Management.” This allows you to quickly and easily collect all relevant logs files and other information to help the various support teams debug any problems you may have with your BCM450. The entire log files required to diagnose a problem is consolidated into a single file.

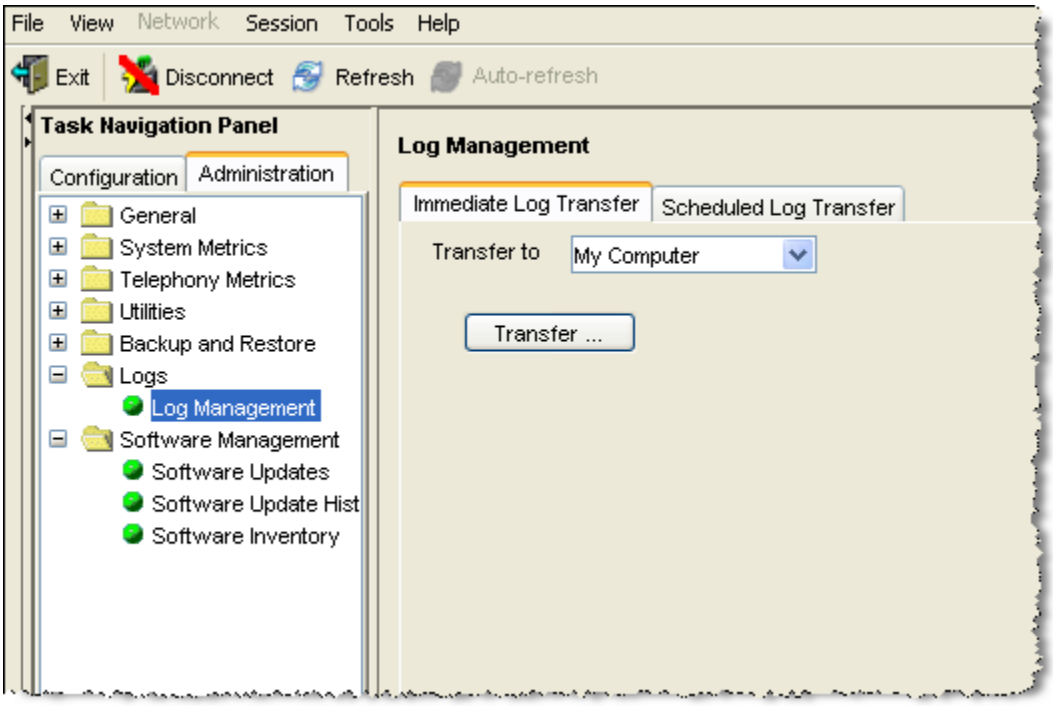

Figure 26: Log Management

When you first suspect a problem with your BCM450, it is important that you go into the “Log Management” screen and download the log file to your PC. Even if you end up resolving the issue, it is good to know that this information has been captured if it does end up being required.

6 APPENDIX A: Configuring Destination Code with Wildcard

In an inbound call scenario, the leading digit sent to the BCM450 may be the same as the digit used in the destination code. Without configuring for a wildcard; the BCM450 will interpret the call as a tandem call, and will fail to terminate the call on the BCM450. To remedy this, it is recommended to configure destination code with wildcard.

If the IP Flexible Reach sends a site prefix that is the same as the digit being used for the destination code; please use the following configuration example.

In this example, the number sent to the BCM450 is the following: "961170". To configure the BCM450 for this call, use the following wildcard configuration.

Under ***Configuration → Telephony → Dialing Plan → Routing***, under the ***Destination Codes*** tab, add the destination code "9A" to use Normal Route "001." Configure the absorbed length to 1 so that the BCM450 will absorb the '9' only in an outbound call scenario. Uncheck the digit following the '9' in the incoming digits to the BCM450 (in this example, the '6' in "961170").

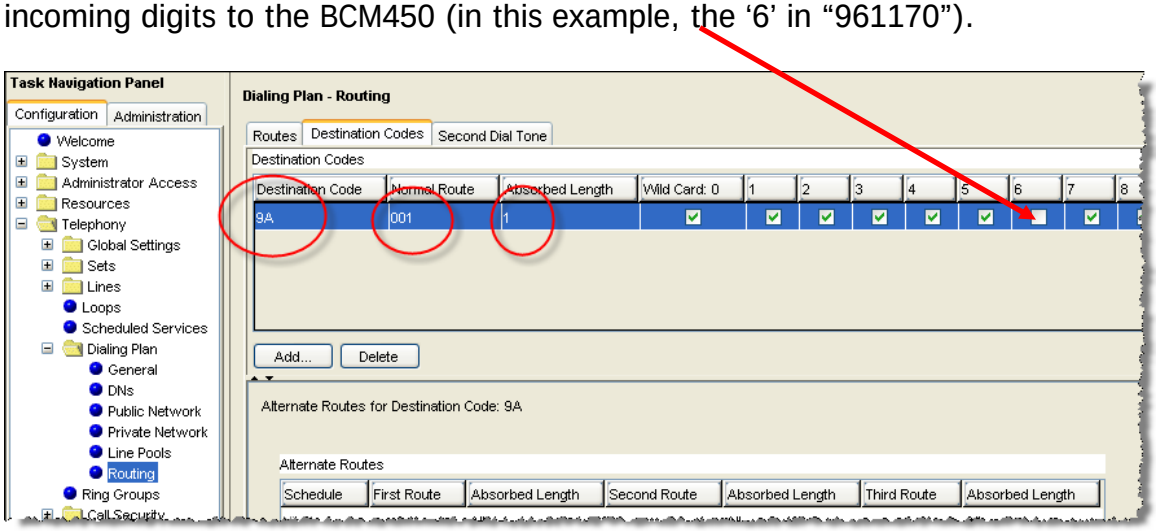

Figure 27: Configuring Destination Code with Wildcard

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