

Optimum Business Trunking and the Nortel BCM50 IP-PBX (Release 5.0)- V9.0.1.74.660 Configuration Guide

Table of Contents

1. Overview 3

2. SIP Trunk Adaptor Set-up Instructions..... 3

3. Additional Set-up Information..... 7

4. International Calling..... 8

5. PBX Configuration 9

1 Overview

The purpose of this configuration guide is to describe the steps needed to configure the Nortel PBX for proper operation Optimum Business Sip Trunking.

2 SIP Trunk Adaptor Set-up Instructions

These instructions describe the steps needed to configure the LAN side of the Optimum Business SIP Trunk Adaptor.

Step 1:

Log on to the Optimum Business SIP Trunk Adaptor

1. Connect a PC to port 4 of the Optimum Business SIP Trunk Adaptor, the silver device labeled Edgewater Networks, 4550 series.



2. Open a Web browser and go to IP Address <http://10.10.200.1>. A login box will appear.

3. Enter login and password and click 'OK'.

Login: pbxinstall

Password: slptrunk



Step 2:

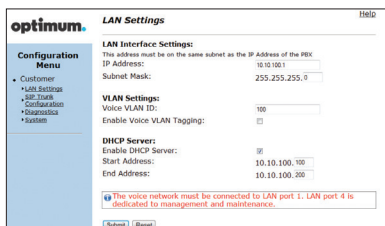
Click on the LAN Settings Link

1. Assign an IP Address to the LAN interface of the SIP Trunk Adaptor. The IP address must be on the same subnet as the IP PBX. This changes the address on port 1 of the Optimum Business SIP Trunk Adaptor.

Note: This will become your local SIP proxy IP address. No other IP addresses will be provided by Cablevision.

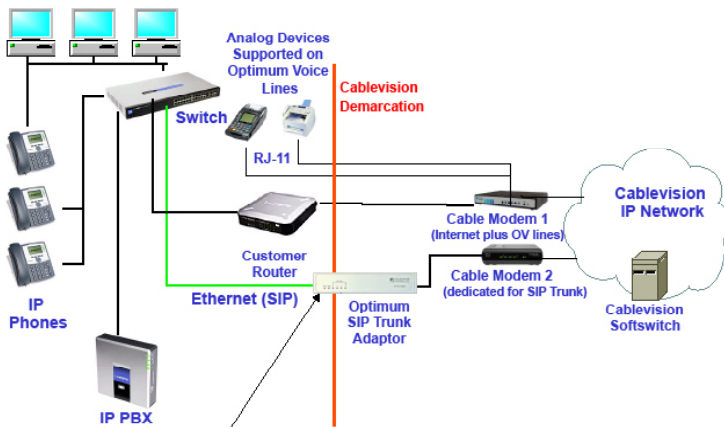
2. Optional: Specify a VLAN for your voice traffic. Click the 'Enable Voice VLAN Tagging' check box. The default VLAN ID is 100.

Note: VLAN 200 should not be used. It is dedicated to port 4 for management.



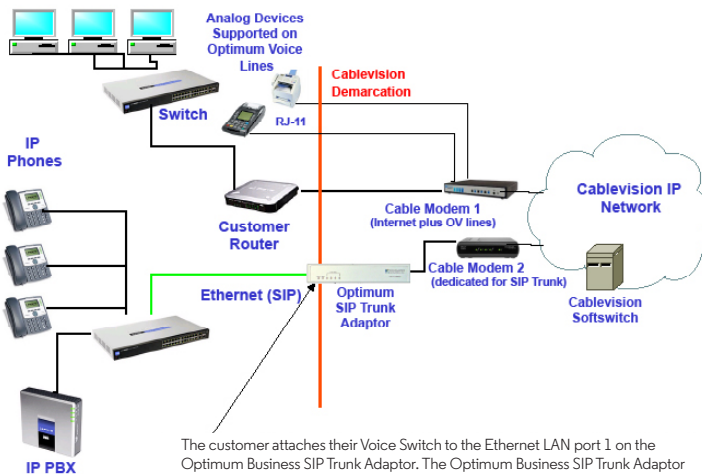
3. Optional: Enable the DHCP server. This will allow the SIP Trunk Adaptor to act as a DHCP server, which will provide IP addresses to the voice network, and create a dedicated voice LAN, as per diagram 2.
4. Click 'Submit'.

Diagram 1 SIP Trunk Adaptor for IP-PBXs
Example: Single LAN Configuration



Using a connection from the customer's LAN, the SIP Trunk Adaptor's address can be a statically assigned private IP address. It may not be assigned a Public IP address.

Diagram 2 SIP Trunk Adaptor for IP-PBXs
Example: Separate Voice and Data Networks Configuration



The customer attaches their Voice Switch to the Ethernet LAN port 1 on the Optimum Business SIP Trunk Adaptor. The Optimum Business SIP Trunk Adaptor can be enabled as a DHCP server to provide routing for the separate voice network.

Step 3:

Click on the SIP Trunk Configuration Link

1. Select your IP PBX make and model from the drop-down menu.
2. Specify how the IP PBX will register to the Optimum Business SIP Trunk Adaptor.
3. The Cablevision network only supports Inband DTMF. Click on the check box next to "Convert Inband DTMF" if you cannot configure your IP PBX to send out Inband DTMF. The DTMF tone duration generated by the phones and/or PBX may need to be increased from their default setting. Some phones and/or PBX have a default setting between 180ms to 200ms. This setting is too low. The recommended setting is 600ms.
4. Click 'Submit'.

The screenshot shows the 'SIP Trunk Configuration' page. On the left is a 'Configuration Menu' with links: Customer, LAN Settings, SIP Trunk Configuration (highlighted), Diagnostics, and System. The main content area has a 'Select your PBX:' dropdown menu with 'Asterisk' selected. Below this are two radio button options: 'Passive connection using the local, private IP address of the PBX interface' (selected) and 'Active connection using registration'. The 'Active connection' section includes fields for 'User Id:' (set to 'secret') and 'Password:' (masked with asterisks). There is a checkbox for 'Convert Inband DTMF:' which is currently unchecked. 'Submit' and 'Reset' buttons are present. A 'Status:' section shows 'Trunk Status:' as 'Not Registered' and a list of 'DID's' (0164030809, 0164030760, 0164030769, 0164030765, 0164030841).

Step 4:

Diagnostics Link

You can make a test call directly from your phone or use the test call application under the Diagnostics link.

The screenshot shows the 'Network Test Tools' page. The left 'Configuration Menu' is the same as in Step 3. The main content area has a description: 'A network administrator may use the test tools on this page to verify connectivity of the System and trace the path of data throughout the network.' It includes sections for 'Outbound Call Test:' (with a 'Pilot Number:' field set to 0164030809 and 'Call'/'Reset' buttons), 'Inbound Call Test:' (with a 'Submit' button), 'Ping Test:' (with an 'IP Address to Ping:' field and 'Ping'/'Reset' buttons), and 'Traceroute Test:' (with an 'IP Address to Trace:' field and 'Traceroute'/'Reset' buttons).

Step 4 continued

Field	Description
Outbound Call Test TelephoneNumber	Specifies an outside phone number to which an outbound call will be initiated. The pilot telephone number of the SIP Trunk will be prepopulated.
Pilot Number	Displays the provisioned pilot number, which is used for outbound and inbound call tests.
Call	Initiates a call outbound to a telephone number entered or inbound to the pilot number displayed.
Inbound Call Test (radio button)	Indicates whether inbound test call will be enabled or disabled. If inbound test calls are enabled, calls made to the pilot number will be redirected to the test UA for fifteen minutes. When the pilot number is dialed, you will hear a test message play.
Submit	Enables or disables the inbound call test.
IP Address to Ping	Verifies basic connectivity to a networking device. Successful ping test results indicate that both physical and virtual path connections exist between the system and the test IP address.
Ping Button	Sends a ping to the IP address specified in the field "IP Address to Ping".
IP Address to Trace	Tracks the progress of a packet through the network. The packet can be tracked through the WAN or LAN interfaces of the adaptor.
Interface (radio button)	Indicates whether a packet will be tracked through the LAN or the WAN.
Traceroute Button	Initiates a traceroute to the specified IP address on either the LAN or the WAN.
Reset	Clears all fields and selections and allows you to enter new information. Reset applies to outbound call test, ping and traceroute.

3 Additional Set-up Information
Systems

optimum.

Configuration Menu

- Customer
 - LAN Settings
 - SIP Trunk Configuration
 - Diagnostics
 - System

System

Help

Software Version:
Version 11.6.14.1 -- Fri Jan 4 17:49:28 PST 2013

Hostname:
5164939899

Model:
EdgeMarc 4552

Vendor:
Cablevision

LAN Interface MAC Address:
A8:70:A5:00:D8:18

Registration Status:
The ALG feature is registered. View [license key](#).

System Date:
02/29/2016 15:03:40 UTC

Change Password:

- [pbxinstall](#)

Field	Description
Pbxinstall Link	Select to change the default password for the pbxinstall login ID. Only the password can be changed. The login ID cannot be changed.

Password

optimum.

Configuration Menu

- Customer
 - LAN Settings
 - SIP Trunk Configuration
 - Diagnostics
 - System

Set Password

Help

Change the GUI password by filling in the fields below. The password must be between 6 and 8 characters in length.

Username:
Current Password:
New Password:
Confirm Password:

pbxinstall

Submit

Reset

7

Field	Description
Username	Specifies the username for which the password can be changed.
Current Password	Specifies the current password.
New Password	Specifies the new password.
Confirm Password	Confirms the new password.
Submit	Applies the settings configured on this page.
Reset	Clears all fields and selections and allows you to enter new information.

4 International Calling

Optimum Voice Business Trunking offers an optional International Calling Service for direct-dialed calls made from the Customer's business or from any phone via the Optimum Voice International Calling remote access number to destinations outside of the United States, Puerto Rico, Canada and the U.S. Virgin Islands at per minute rates. The Customer must login to the Optimum Business Account Center and activate the service on the Optimum Business Trunk Pilot telephone number to activate the service and manage the monthly International spending limit for the account.

Activating International calling on the Pilot TN will enable International calling for all Direct Inward Dial (DIDs) telephone numbers as well. Blocking International calling for one or more DIDs is managed by the customer directly from the PBX phone system configuration. To minimum the exposure to fraudulent calling, It is recommended to limit International calling capability to those DID's that require it and set up an account spending limit that reflects what is necessary to run the business.

It is the Customer (and/or the Customer Agent's) responsibility to properly secure the customer's PBX to prevent the PBX from being compromised and fraudulent calls from being made by unauthorized (internal or external) users. If fraudulent calls are detected, Cablevision reserves the right to disable International Calling until the PBX is properly secured by the customer.

5 PBX Configuration

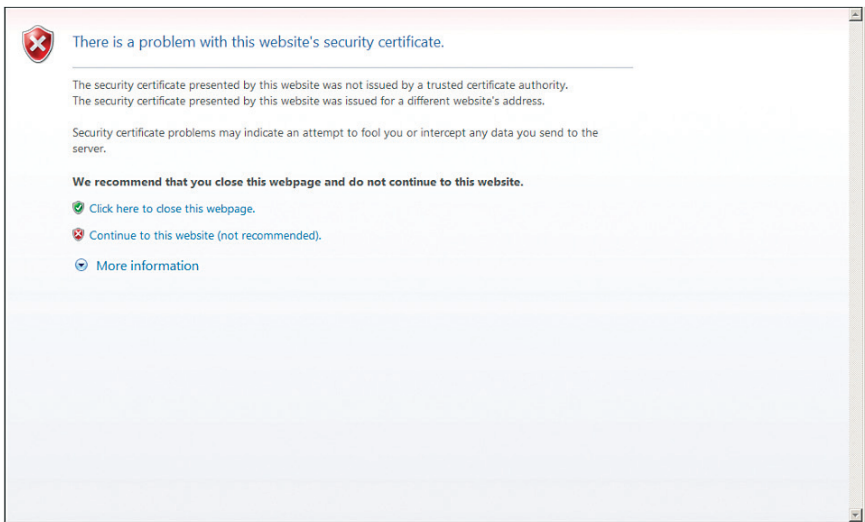
The steps on the following pages describe the minimum configuration required to enable the PBX to use Optimum Business SIP Trunking for inbound and outbound calling. Please refer to the Nortel BCM50 System product documentation for information on other advanced PBX features.

The PBX used in the lab comprises of the following:

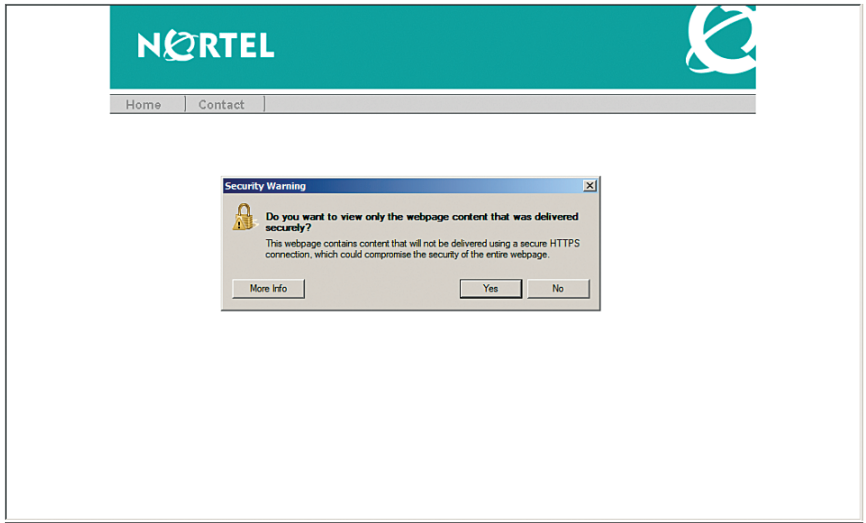
- Nortel BCM50 PBX
- 1 Nortel IP Phone 2004
- 1 Nortel IP Phone 1120E

The configuration described here assumes that the PBX is already configured and operational with station side phones using assigned extensions or DIDs. This configuration is based on BCM50 (Release 5.0) v9.0.1.74.660.

1. The Nortel BCM30 PBX comes with 4 Ethernet ports. Starting from the left side of the front panel, the first Ethernet port is for Administration only. This port has a default IP address of “10.10.11.1/255.255.255.252”. To access this OAM port for administration or configuration, the Windows PC must have an IP address of “10.10.11.2/29”, either by manually assigned or by sending DHCP request to the PBX. The second Ethernet port from the left side of the front panel is PBX’s LAN port. The PBX’s LAN port has a default IP address of “192.168.1.2/24” and a default gateway of “192.168.1.1/24”. The PBX’s LAN port, the local IP phones and the Optimum Business SIP Trunk Adaptor’s LAN port 1 should be on the same LAN segment. The Optimum Business SIP Trunk Adaptor’s LAN port 1 and the PBX’s LAN port have been assigned with IP addresses of “192.168.1.1/24” and “192.168.1.2/24” respectively. The third and the fourth ports are for connecting to expansion modules (i.e.: PRI) and are not needed for SIP trunk service.
2. To configure the PBX, first install the **Nortel Business Element Manager** from the CDROM to a Windows PC and then run “https://10.10.11.1/” from your browser.
 - a. Click the “**Continue to this website (not recommended)**” link.



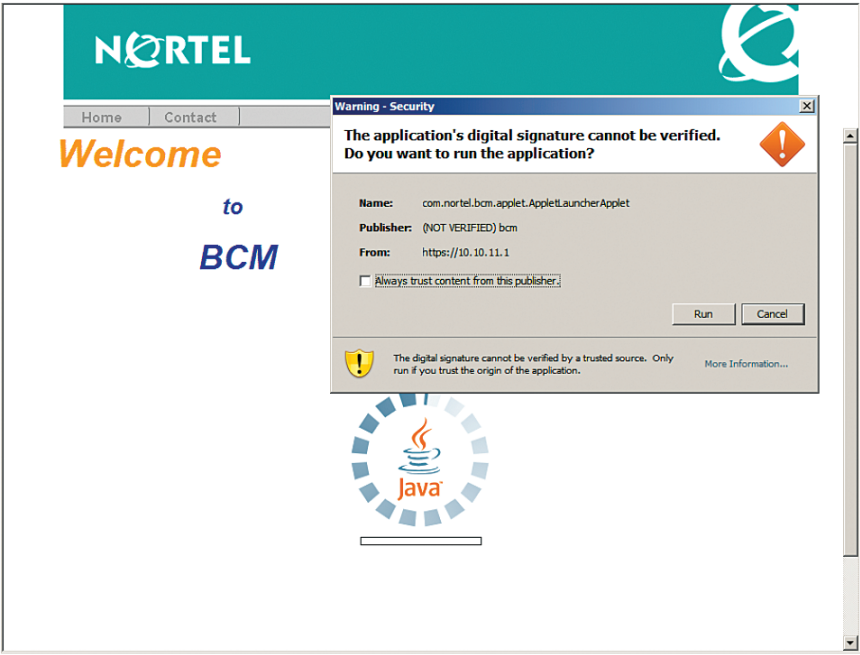
b. Click the “Yes” button.



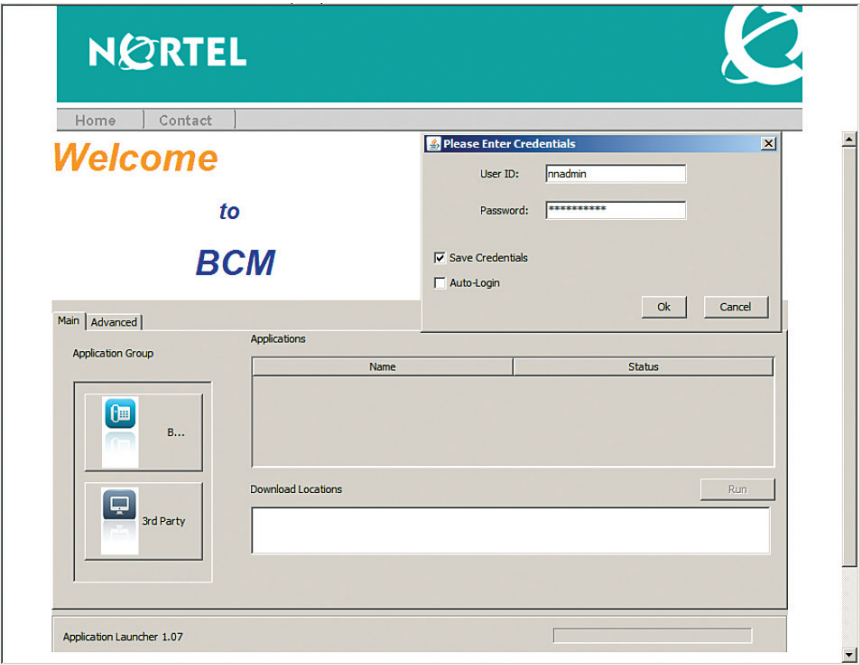
c. Click the “Yes” button.



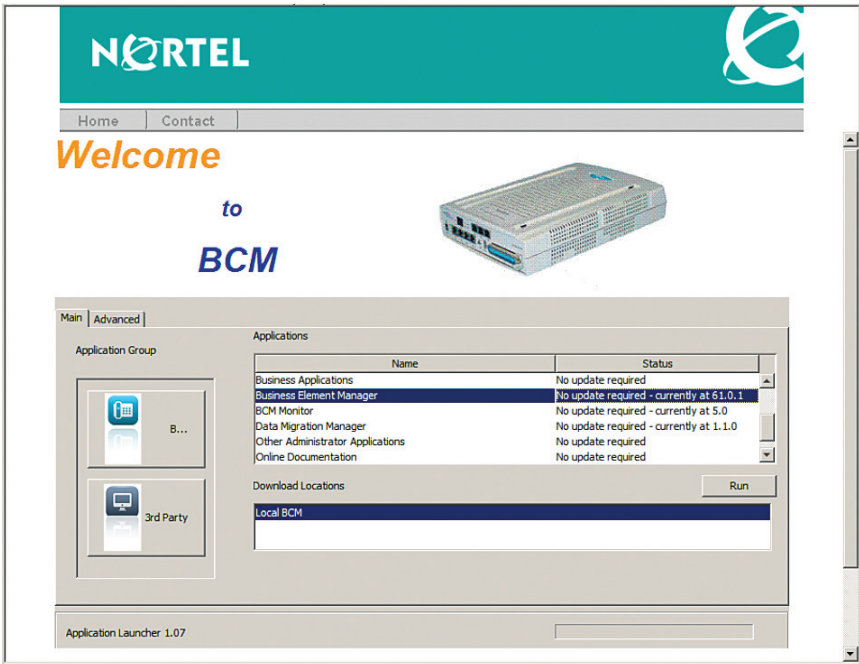
d. Click the **“Run”** button.



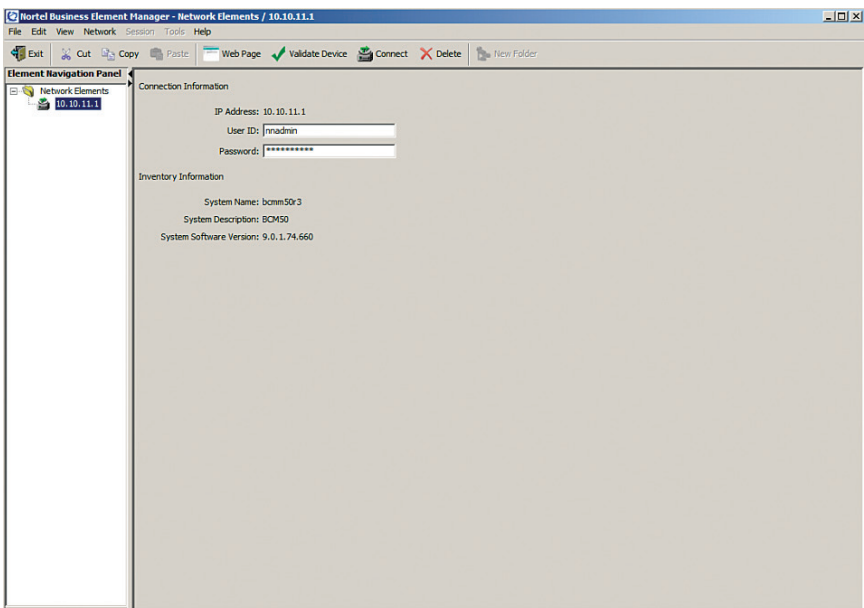
e. From the “Please Enter Credentials” pop-up box, enter the default user ID of “nnadmin” in the **“User ID”** field, enter the password (default password is “PlsChgMe!”) in the **“Password”** field and then click the **“Ok”** button.



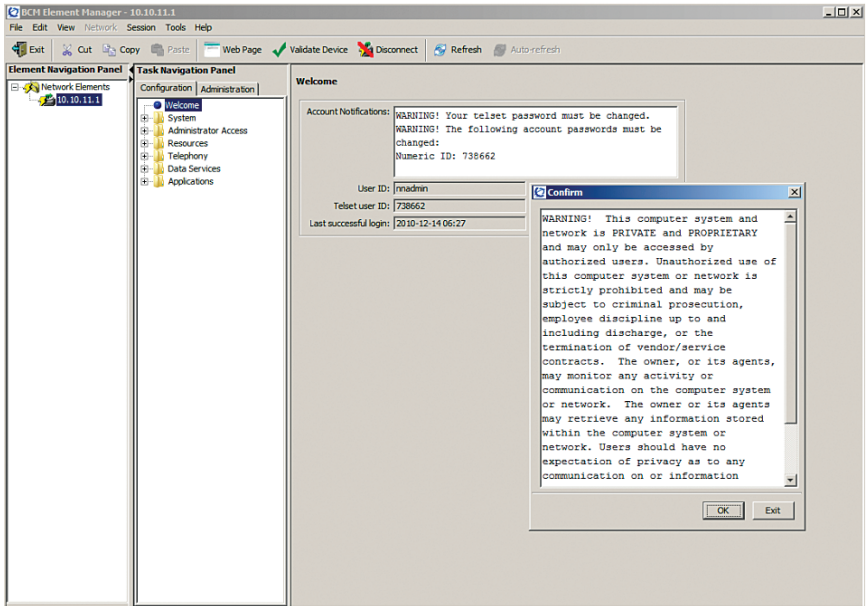
- f. Run Nortel Business Element Manager by selecting **“Business Element Manager”** in the Applications section and then clicking **“Local BCM”** in the Download Locations section.



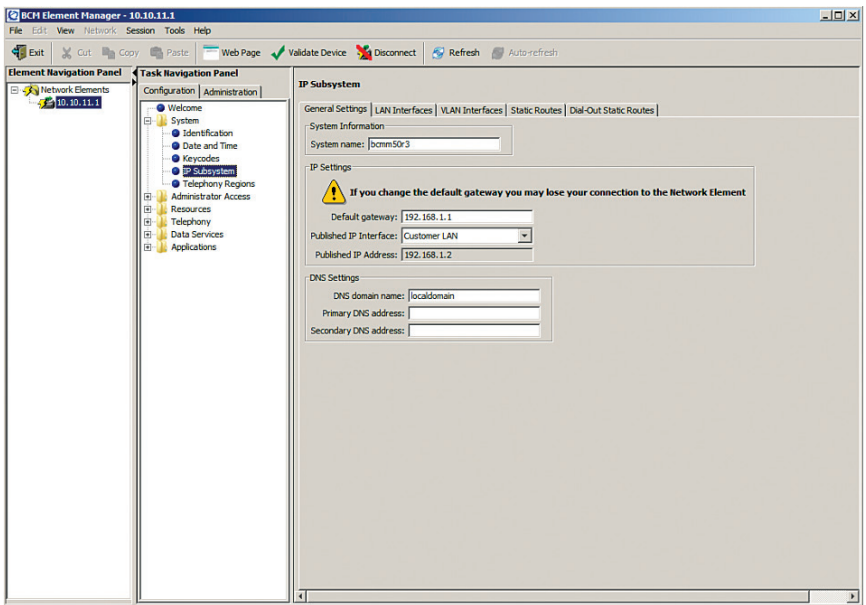
- g. A new window will pop up to run the Business Element Manager. Enter the default user ID of “nnadmin” in the **“User ID”** field, enter the password in the **“Password”** field and then click the **“Connect”** button.



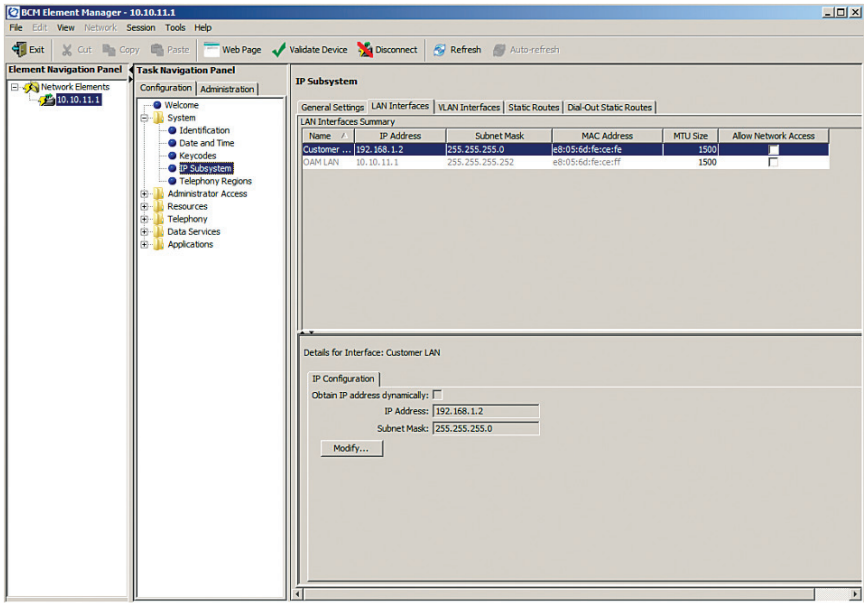
h. Click the **“OK”** button.



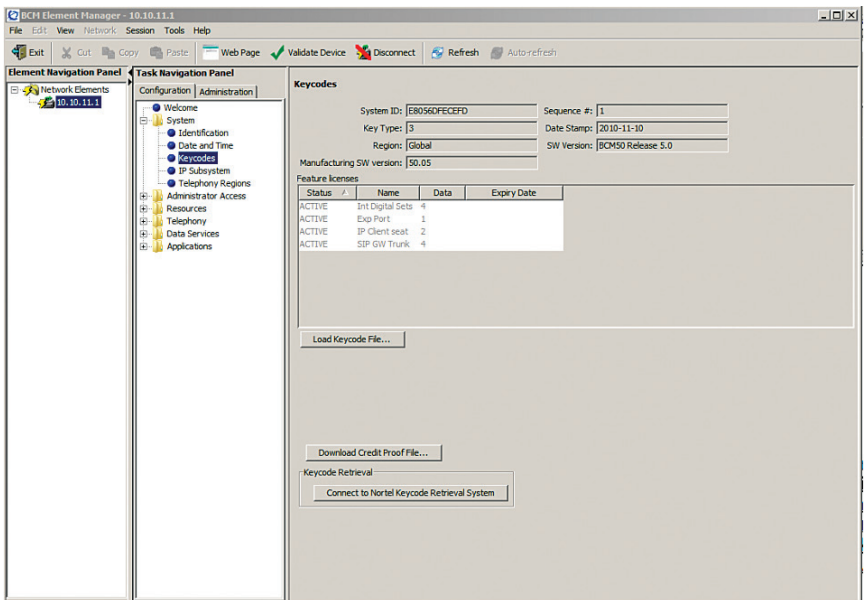
3. Navigate to **“Configuration”>“System”>“IP Subsystem”** to check and/or change IP address of the PBX and its default gateway.
 - a. Select the **“General Settings”** tab to make sure the Optimum Business SIP Trunk Adaptor’s IP address is set in the **“Default gateway”** field.



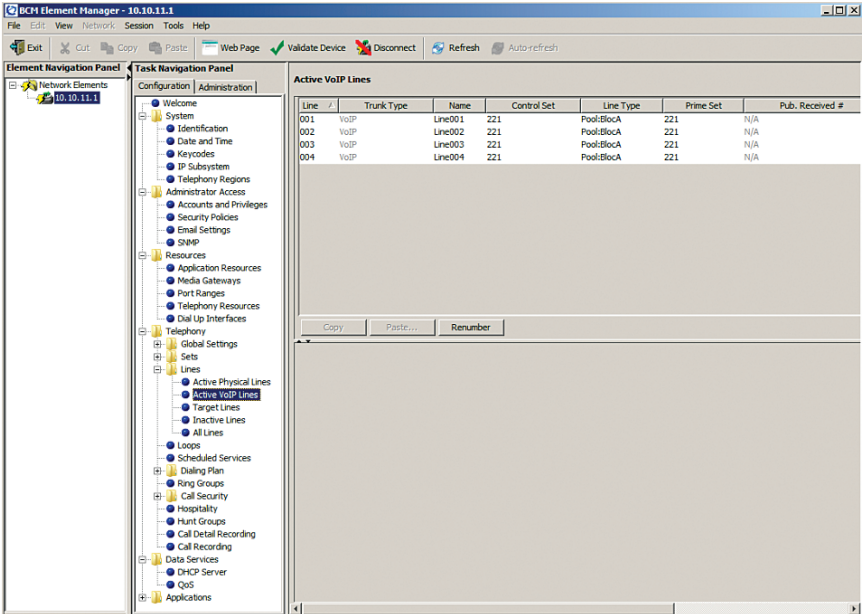
- b. Select the “**LAN Interfaces**” tab to make sure the PBX’s LAN IP address is correct.



4. Before configuring the PBX, first make sure the PBX has license for SIP trunk. To enable the licensed features, certain keycodes will need to be installed in the PBX. Users should require these keycodes be installed before the PBX is delivered.
- a. Navigate to “**Configuration**”>“**System**”>“**Keycodes**” to check the licensed features. PBX should have license for SIP trunks.

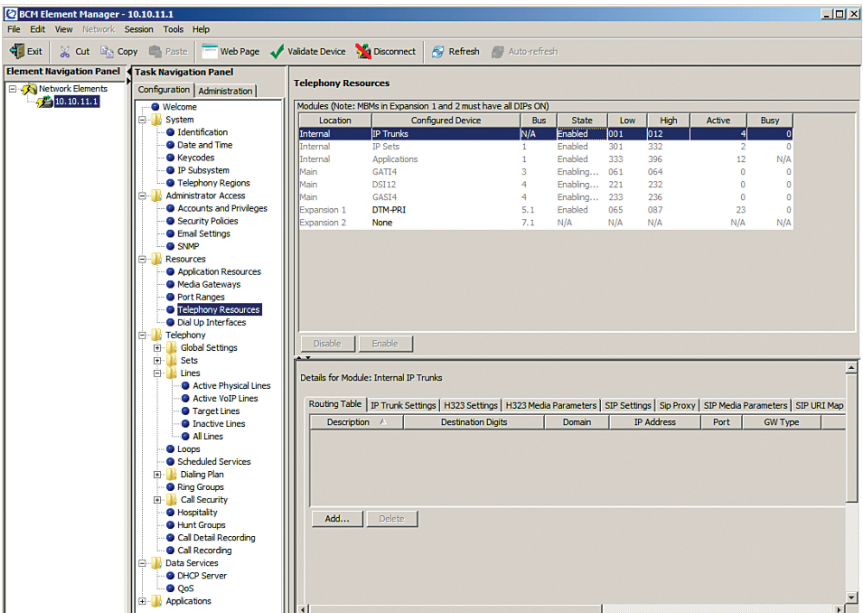


- b. Navigate to **“Configuration”>“Telephony”>“Lines”** and then select **“Active VoIP Lines”** to find out which line pool (‘‘BlocA’’ in this example) has been assigned for the VoIP/SIP trunks.

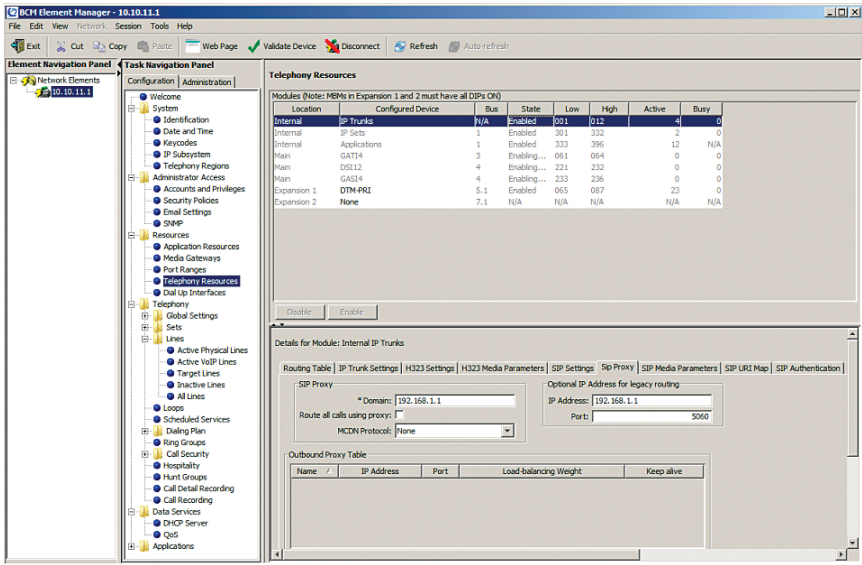


5. To configure the Optimum Business SIP Trunk Adaptor as the PBX’s SIP trunk service provider, navigate to **“Configuration”>“Resources”>“Telephony Resources”** and select **“IP Trunks”** entry under the **“Configured Device”** column.

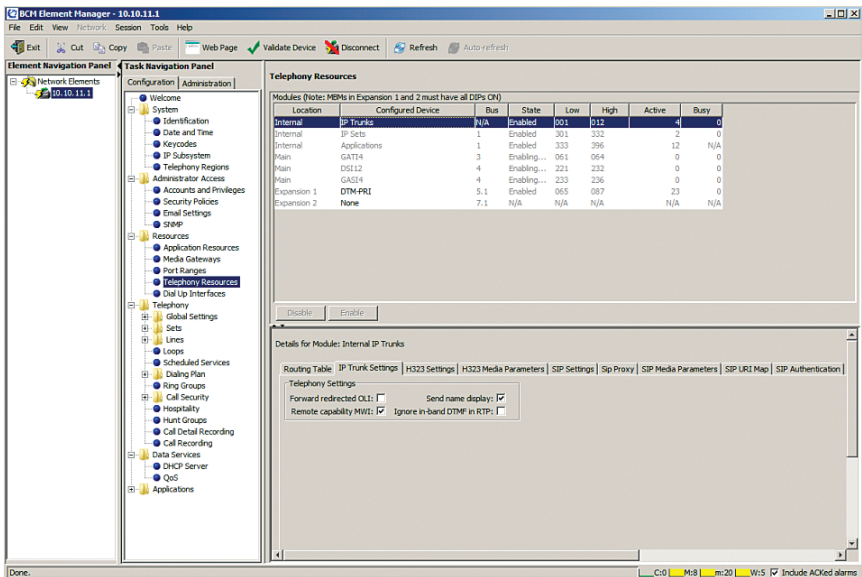
- a. SIP trunk service may be set up from either the **“Routing Table”** tab or the **“Sip Proxy”** tab. If using the **“Sip Proxy”** tab, make sure nothing is set up for the Optimum Business SIP Trunk Adaptor under the **“Routing Table”** tab.



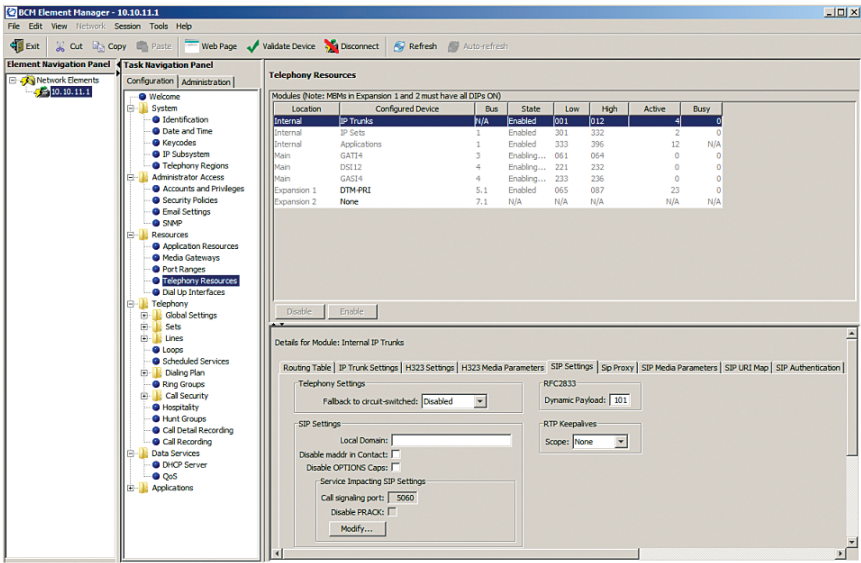
- b. Select the “**Sip Proxy**” tab to set up the Optimum Business SIP Trunk Adaptor as the PBX’s SIP trunk service provider. Enter Optimum Business SIP Trunk Adaptors IP address (“192.168.1.1” in this example) in both the “**Domain**” field and the “**IP address**” field, enter “5060” in the “**Port**” field and leave other fields as default.



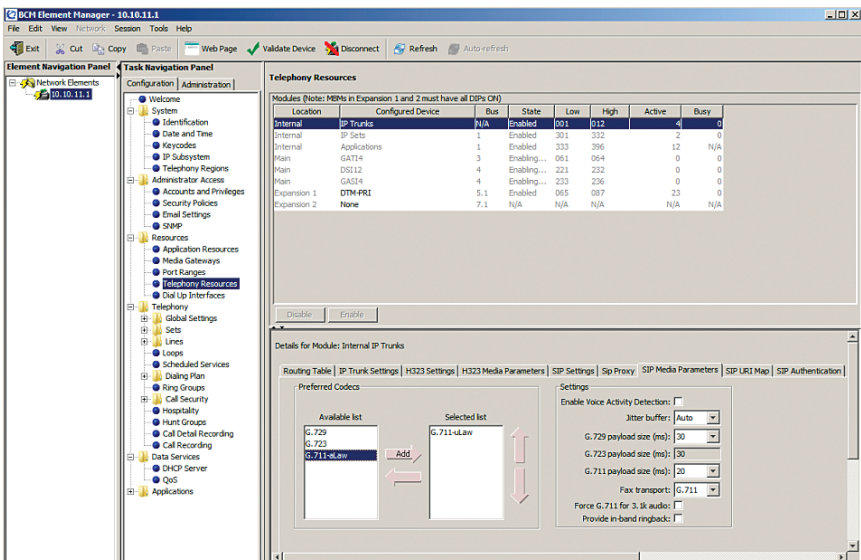
- c. Select the “**IP Trunk Settings**” tab, make sure the “**Ignore in-band DTMF in RTP**” checkbox is **unchecked** and leave other fields as default. This is a mandatory setting for the service to work properly. If you set up an Auto-Attendant, it will recognize either in-band DTMF or RFC-2833 DTMF.



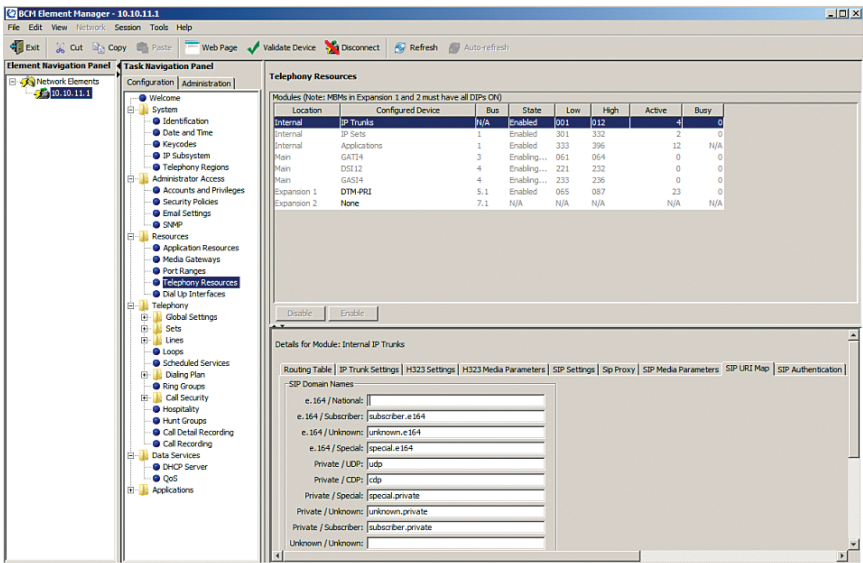
- d. Select the **“SIP Settings”** tab, enter **“101”** in the **“Dynamic Payload”** field for RFC-2833 DTMF and leave other fields as default.



- e. Select the **“SIP Media Parameters”** tab, move the G.711-uLaw and the G.711-aLaw codec to the **“Selected list”** by selecting the G.711 codec from the **“Available list”** and then click the **“Add”** arrow. Select **“20”** in the **“G.711 payload size (ms)”** field and leave other fields as default.

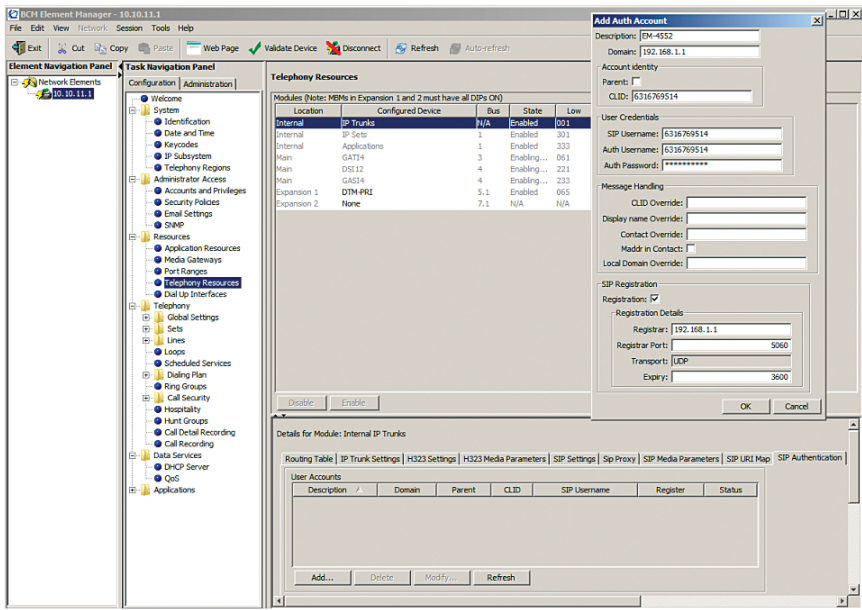


- f. Select the “**SIP URI Map**” tab, blank out both the “**e.164 / National**” field and the “**Unknown / Unknown**” field and leave other fields as default.

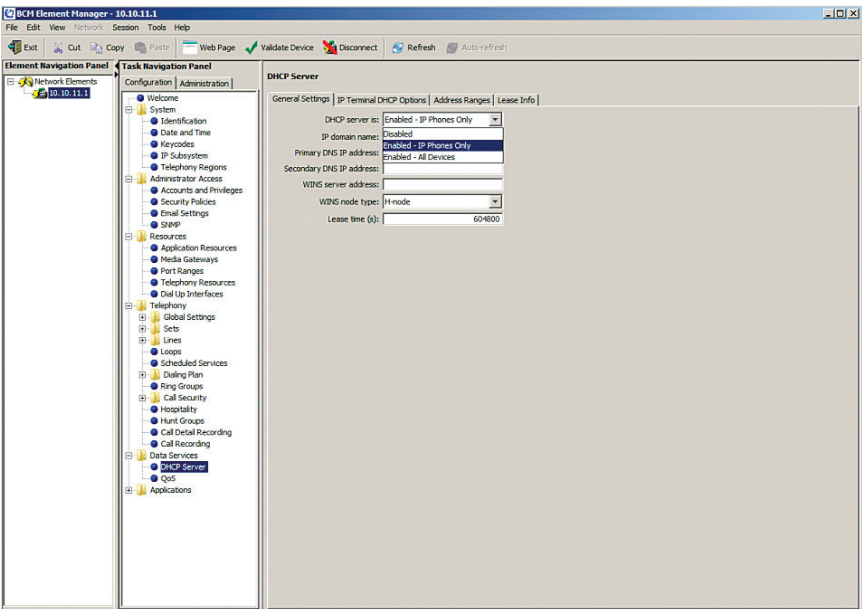


- g. Note that this step is for PBX registration mode only; skip this step if the PBX is communicating with the Optimum Business SIP Trunk Adaptor using static IP mode. Select the “**SIP Authentication**” tab and then click the “**Add**” button. From the “**Add Auth Account**” pop-up box:

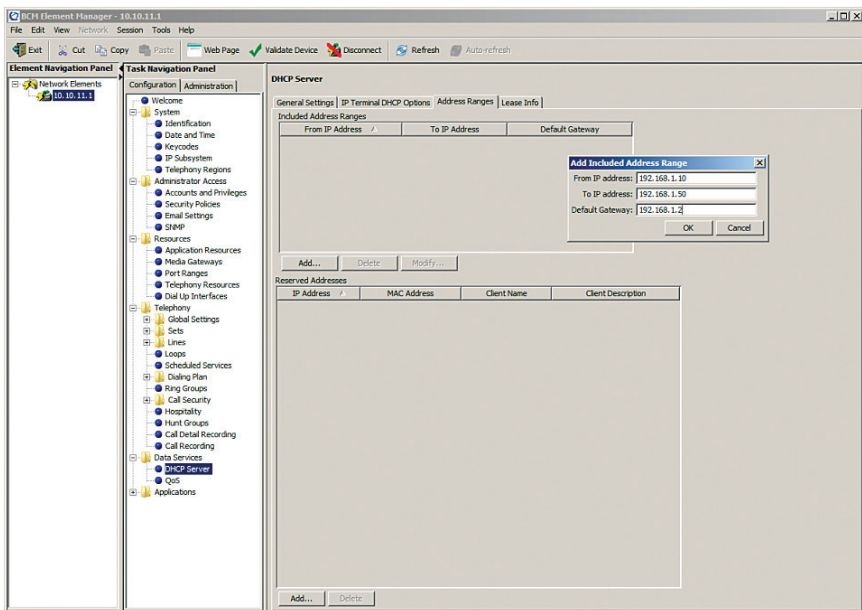
- 1) Enter a descriptive name in the “**Description**” field (EM-4552 was used in this example).
- 2) Enter the Optimum Business SIP Trunk Adaptor’s IP address in the “**Domain**” field.
- 3) Enter the pilot DID in the “**CLID**” field.
- 4) Enter the SIP username in the “**SIP Username**” field, enter the Auth. Username in the “**Auth. Username**” field (if needed), and enter the Auth. Password in the “**Auth. Password**” field. **Note:** The user credential info **must match** what has been configured on the Optimum Business SIP Trunk Adaptor.
- 5) Check the “**Registration**” checkbox.
- 6) Leave other fields as default and click the “**OK**” button.



6. Navigate to **“Configuration” > “Data Services” > “DHCP Server”** to configure DHCP server for the IP phones. Note that the PBX’s DHCP server is enabled by default. Make sure the Optimum Business SIP Trunk Adaptor’s DHCP server is disabled. This is part of step 2 of the Optimum Business SIP Trunk Set-Up Guide.
 - a. Select the **“General Settings”** tab, select **“Enabled – IP Phones Only”** from the drop-down list of the **“DHCP server is”** field.

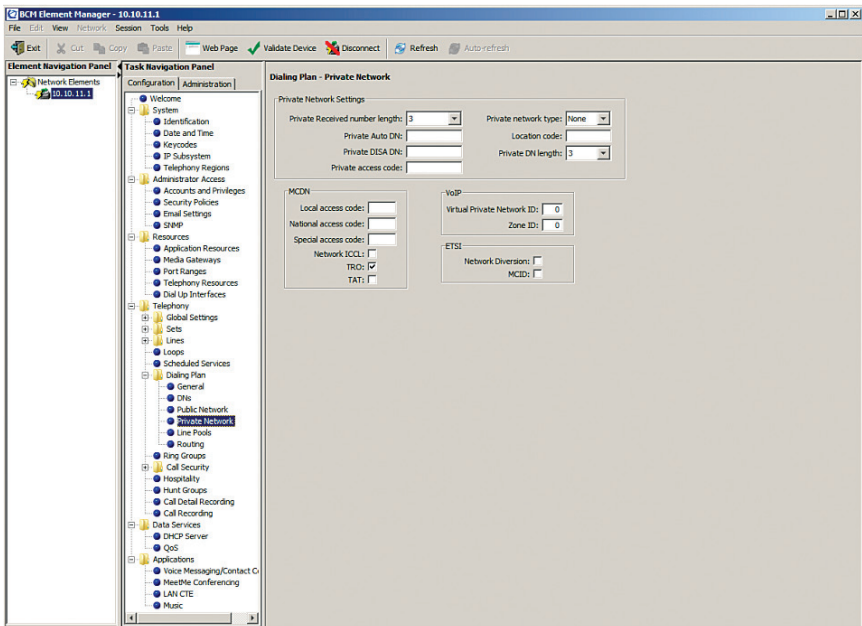


- b. Select the **“Address Ranges”** tab and then click the **“Add”** button. From the **“Add Included Address Range”** pop-up box, enter the starting IP address in the **“From IP address”** field, enter the ending IP address in the **“To IP address”** field and enter the PBX’s LAN IP address in the **“Default Gateway”** field.



Create DNS/extensions for the IP phones and map the inbound called number to the DNS/extensions. Note that a 3-digit DN/extension is used in this example.

- Navigate to **“Configuration” > “Telephony” > “Dialing Plan”** and then select the **“Private Network”** tab. Select **“3”** from the drop-down list of the **“Private Received number length”** field, select **“3”** from the drop-down list of the **“Private DN length”** field and leave other fields as default.



- b. Navigate to **“Configuration” > “Telephony” > “Sets”** and then select the **“All DNs”** tab. For each **selected DN**, select the model of the IP phone in the **“Model”** field and enter the assigned DID in the **“Pub. OLI”** field. Note that for PBX registration mode, the **“Pub. OLI”** field must always be set to be the same as the pilot DID. Also note that while static IP mode allows all the different DIDs to be used as the caller ID, PBX registration mode allows only the pilot DID as the caller ID.

UICH Element Manager - 10.10.11.1

FileEditViewNetworkSessionToolsHelp

ExitCutCopyPasteValidate DeviceDisconnectRefreshAuto-refresh

Web Page

Element Navigation Panel

Network Elements

10.10.11.1

- System
 - Identification
 - Date and Time
 - Keywords
 - IP Subsystem
 - Telephony Regions
 - Administrator Access
 - Accounts and Privileges
 - Security Policies
 - Email Settings
 - DHCP
 - Resources
 - Application Resources
 - Media Gateways
 - Port Ranges
 - Telephony Resources
 - Dial Up Interfaces
 - Telephony
 - Global Settings
 - Sets
 - Templates
 - Active Sets
 - Active Application DNs
 - Inactive DNs
 - DNs
 - Lines
 - Loops
 - Scheduled Services
 - Dialing Plan
 - Ring Groups
 - Call Security
 - Hospitality
 - Hunt Groups
 - Call Detail Recording
 - Call Recording
 - Data Services
 - DHCP Server
 - QoS
 - Applications
 - Voice Messaging/Contact O
 - MeetMe Conferencing
 - LAN CTE
 - Music

Task Navigation Panel

ConfigurationAdministration

All DNs

Line AccessCapabilities and PreferencesRestrictions

DN	Model	Name	Port	Pub. CLI	Priv. CLI	Fwd No Answer	Fwd Delay
292	17208/M7208	292	0809	292			N/A
293	17208/M7208	293	0810	294			N/A
294	17208/M7208	295	0811	295			N/A
295	17208/M7208	296	0812	296			N/A
296	17208/M7208	297	0813	297			N/A
297	17208/M7208	298	0814	298			N/A
298	17208/M7208	299	0815	299			N/A
300	17208/M7208	300	0816	300			N/A
301	11402/2004/2007/2050/221x	301	0101	6316/769514	301		N/A
302	1120E/2002	302	0109	6316/769515	302		N/A
303	11402/2004/2007/2050/221x	303	0103	6316/769516	303		N/A
304	11402/2004/2007/2050/221x	304	0104	304			N/A
305	1120E/2002	305	0105	305			N/A
306	1110/2001/2003	306	0106	306			N/A
307	1230	307	0107	307			N/A
308	1230	308	0108	308			N/A
	1210						

CopyName...Renum...

Details for DN: 303

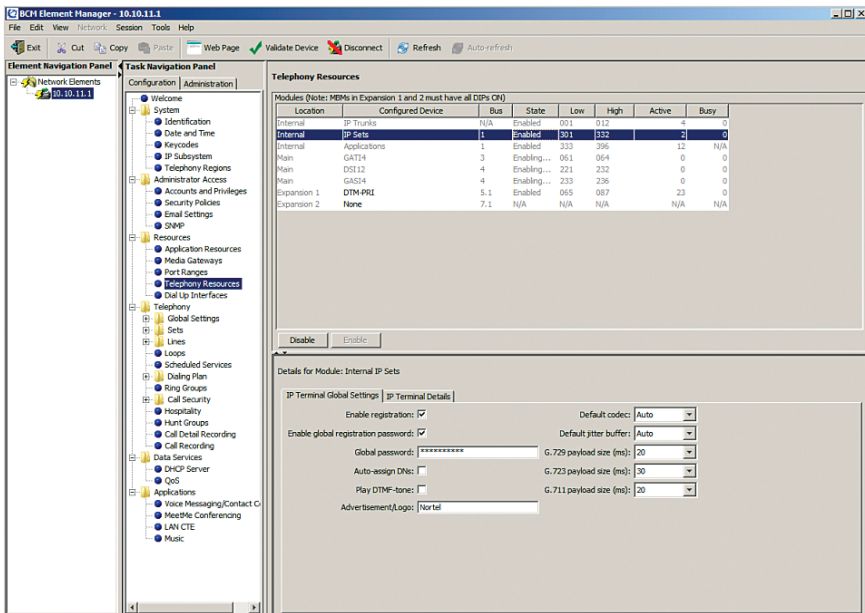
Line AssignmentLine Pool AccessAnswer DNsMeetMe Conferencing

Assigned Lines

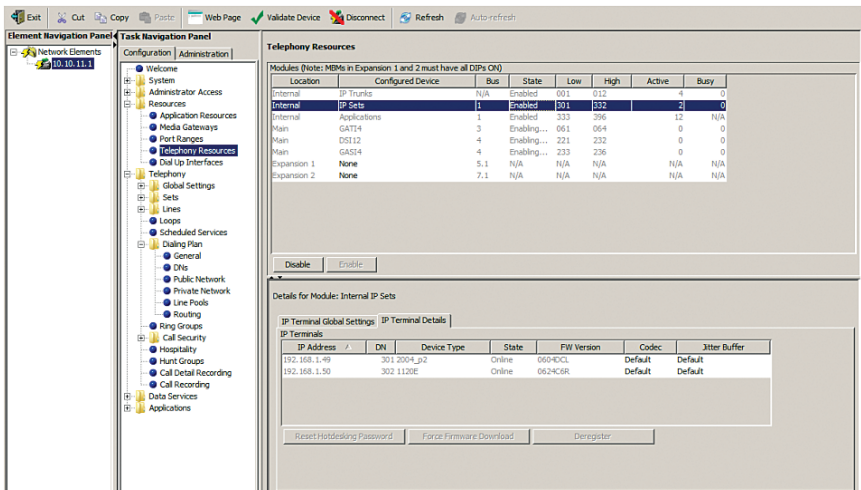
Line	Appearance Type	Appearances	Caller ID Set	Vmsg Set	Priv. Received #

Add...Delete

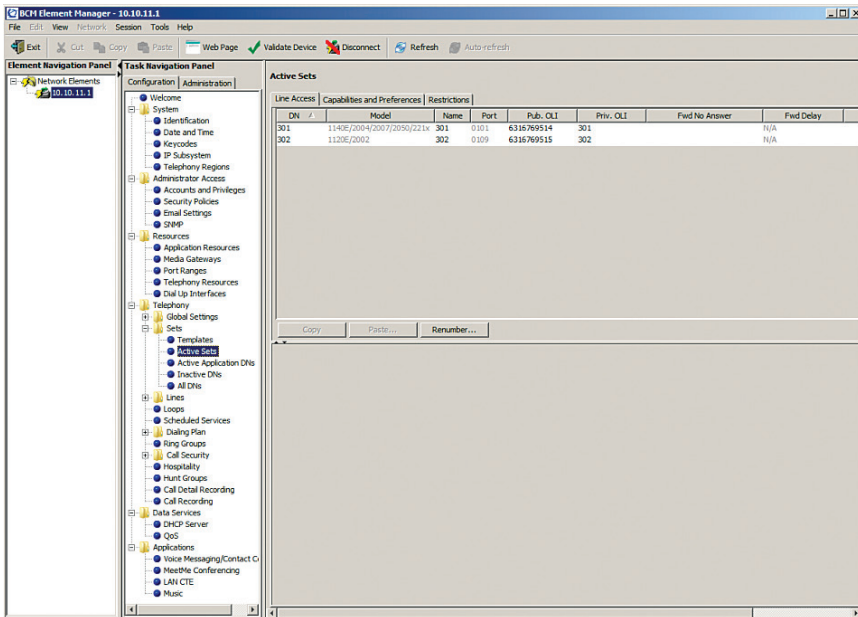
- c. Navigate to **“Configuration”>“Resources”>“Telephony Resources”** and then select **“IP Sets”** under the Configured Device column to enable registration from the phones. From the **“IP Terminal Global Settings”** tab:
 - 1) Check the **“Enable registration”** checkbox.
 - 2) Check the **“Enable global registration password”** checkbox.
 - 3) Enter a global password in the **“Global password”** field. When you add a new IP phone, the IP phone will be prompted for the password.
 - 4) Select **“20”** from the drop-down list of the **“G.711 payload size (ms)”** field.
 - 5) Leave other fields as default.



- d. Navigate to “**Configuration**”>“**Resources**”>“**Telephony Resources**” and then select “**IP Sets**” under the Configured Device column after rebooting the IP phones. The IP phones should boot up and successfully obtain an assigned IP address from the PBX. Select the “**IP Terminal Details**” tab to see the IP addresses assigned to the IP phones.

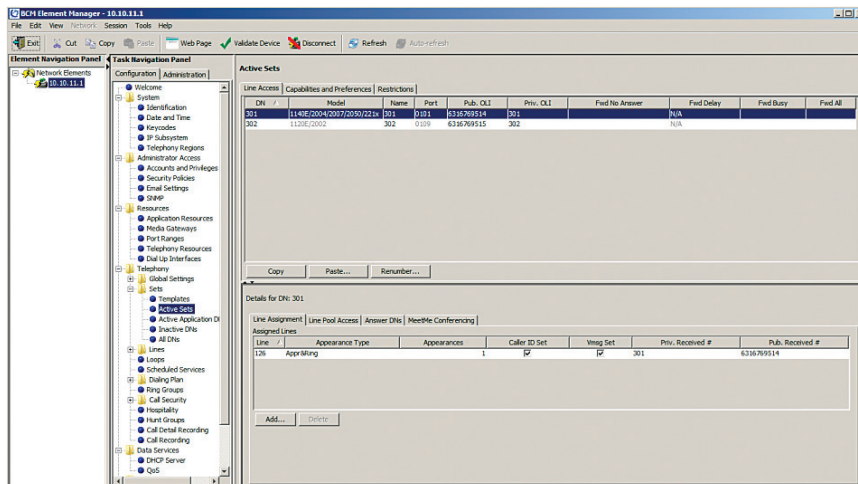


- e. Navigate to “**Configuration**”>“**Telephony**”>“**Sets**” and then select “**Active Sets**” to see if the IP phones are up and running. **Note:** when the IP phone boots up and before it is operational, it will prompt for the DN/extension. You must enter the DN assigned for the IP phone from the IP phone itself. Nortel IP 200x, 11xx and 12xx phones use the Nortel proprietary UNISim signaling protocol to communicate to the BCM50 PBX.

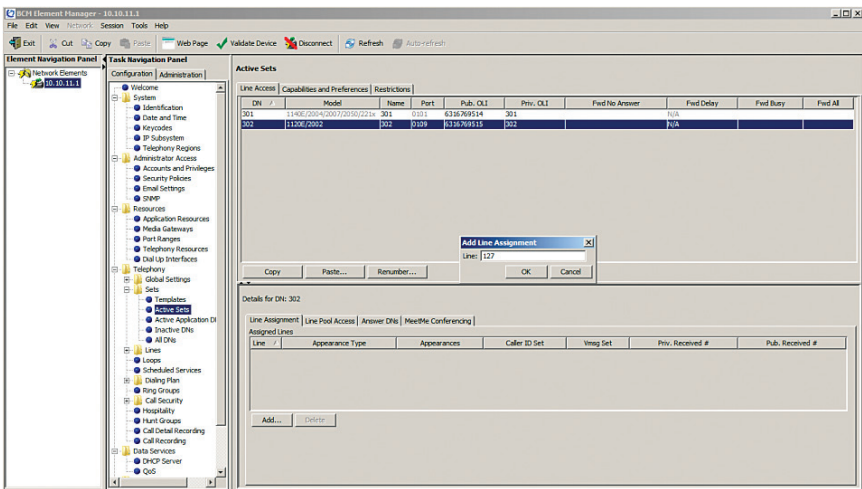


To map the inbound calls to the DNs/extensions, navigate to **“Configuration” > “Telephony” > “Sets”** and then select **“Active Sets”** to assign the DIDs to each DN/extension.

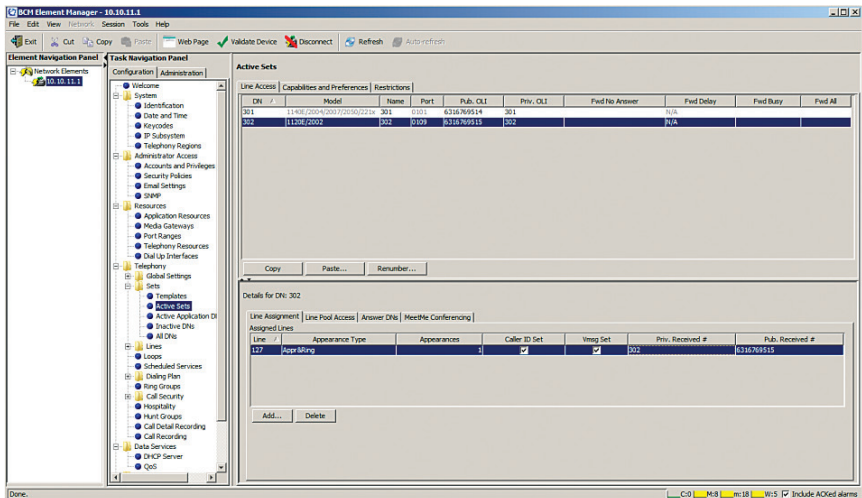
- Select the **“Line Access”** tab and select the first DN from the list. In this example, the pilot DID/“631-676-9514” has been assigned to DN “301”, using a target line of “126”.



- Select **DN “302”** and then click the **“Add”** button to select an unused target line for assigning another DID (“6316769515” in this example) for DN “302”. The unused target lines can be selected from **“Configuration” > “Telephony” > “Lines” > “Target Lines”**. In this example, the target line we want to use is “127”. Enter “127” in the **“Line”** field of the pop-up box and then click the **“OK”** button.

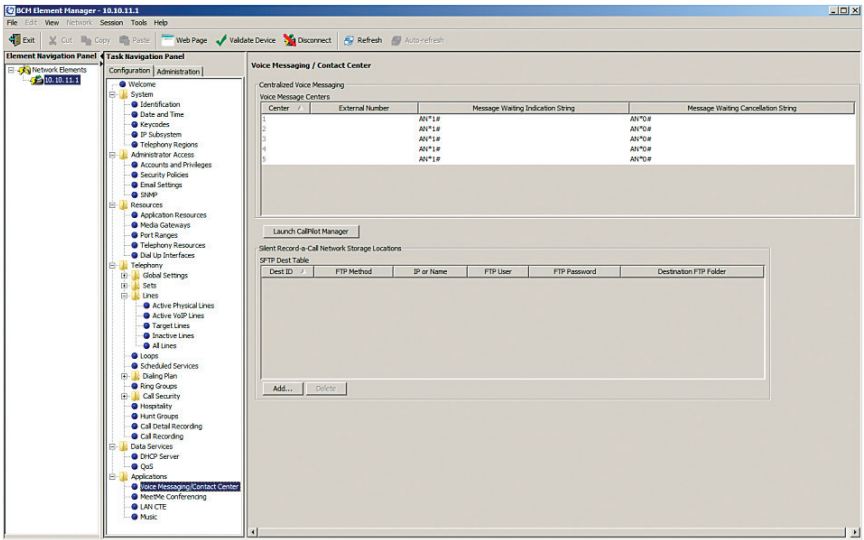


- c. From the Assigned Lines section, check the “**Caller ID Set**” checkbox, check the “**Vmsg Set**” checkbox, enter “302” in the “**Priv. Received #**” field, enter “6316769515” in the “**Pub. Received #**” field and leave other fields as default.

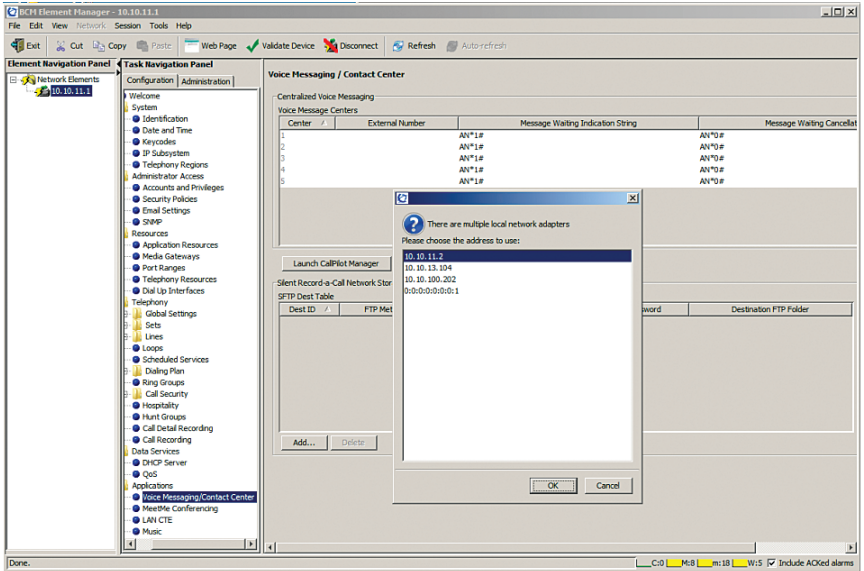


To set up an Auto-Attendant, navigate to “**Configuration**”>“**Applications**” and then select the “**Voice Messaging/Contact Center**” to launch the CallPilot Manager.

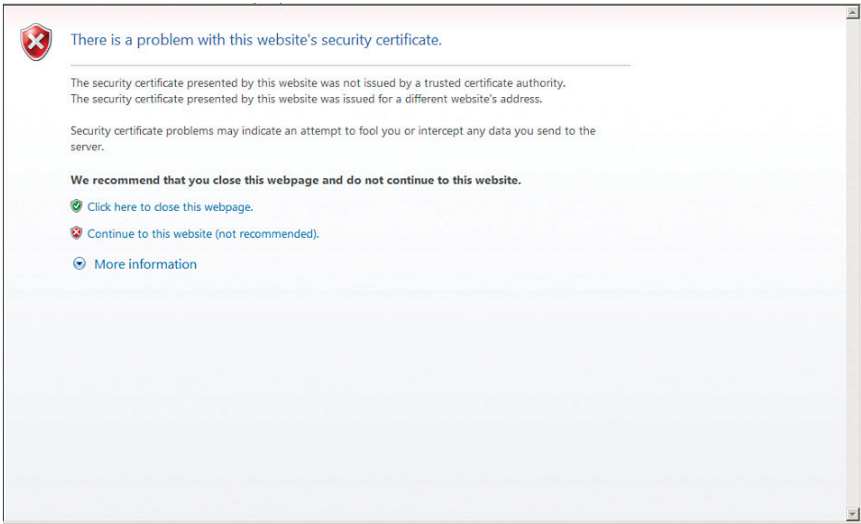
- a. Click the “**Launch CallPilot Manager**” button.



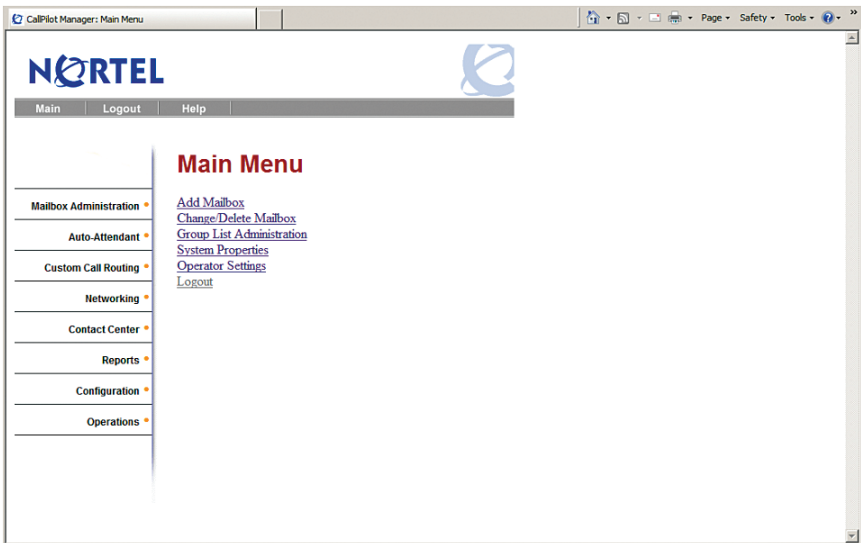
b. Click the “OK” button.



c. Click the “Continue to this website (not recommended)” link.



- d. A new window will come up when the CallPilot Manager starts running.
Click the **“Auto-Attendant”** tab.



- e. Click the **“Company Greetings”** link.

CallPilot Manager: Lines Administration

NORTEL

Main Logout Help

Lines Administration

Mailbox Administration

- Auto-Attendant
 - General Properties
 - Lines Administration
 - Change Many Lines
 - CLD Routing Table
 - Greeting Tables
 - Holidays
 - Company Greetings
 - Customized Digits
- Custom Call Routing
- Networking
- Contact Center
- Reports
- Configuration
- Operations

Line	Answer Mode	Table/Skillset	Rings	Command
1	No	---	0	Change
2	No	---	0	Change
3	No	---	0	Change
4	No	---	0	Change
5	No	---	0	Change
6	No	---	0	Change
7	No	---	0	Change
8	No	---	0	Change
9	No	---	0	Change
10	No	---	0	Change
11	No	---	0	Change
12	No	---	0	Change
13	No	---	0	Change
14	No	---	0	Change
15	No	---	0	Change
16	No	---	0	Change
17	No	---	0	Change
18	No	---	0	Change
19	No	---	0	Change
20	No	---	0	Change
21	No	---	0	Change

f. Click the **“Voice”** link from **Company Greeting 2**, an unused Company Greeting entry/table in this example.

CallPilot Manager: Company Greetings

NORTEL

Main Logout Help

Company Greetings

Mailbox Administration

- Auto-Attendant
 - General Properties
 - Lines Administration
 - Change Many Lines
 - CLD Routing Table
 - Greeting Tables
 - Holidays
 - Company Greetings
 - Customized Digits
- Custom Call Routing
- Networking
- Contact Center
- Reports
- Configuration
- Operations

Company Greeting	Status	Caption	Commands
1	Recorded		Add Caption Voice
2	Not Recorded		Voice
3	Not Recorded		Voice
4	Not Recorded		Voice
5	Not Recorded		Voice
6	Not Recorded		Voice
7	Not Recorded		Voice
8	Not Recorded		Voice
9	Not Recorded		Voice
10	Not Recorded		Voice
11	Not Recorded		Voice
12	Not Recorded		Voice
13	Not Recorded		Voice
14	Not Recorded		Voice
15	Not Recorded		Voice
16	Not Recorded		Voice
17	Not Recorded		Voice
18	Not Recorded		Voice
19	Not Recorded		Voice
20	Not Recorded		Voice

- g. A new window will pop up for Company Greeting 2. You will need a working phone and its DN to record the message for the Auto-Attendatant.
 - a. Enter “301” (or other DN of a working phone) in the “**Connect To**” field and then click the “**Dial**” button. The called phone should ring and it should show “Stopped” on the display when answered.
 - b. Click the “**Record**” button to prompt the user to enter an extension and the called phone should show “Recording” on its display.
 - c. Record your message from the called phone.
 - d. Click the “**Stop**” button as soon as you finish recording the message.
 - e. Click the “**Save**” button to save the recording.
 - f. Click the “**Hang Up**” button to hang up the call.
 - g. Click the “**Close**” button to close the window.

https://10.10.11.1/Voicemail-cgi-bin/MyPhone.exe?SecCon=m_rmisMSPiXmxvwFLbmkzvsPL Certificate Error

Company Greeting 2

Phone Set:

Connect To:

Changes to the recording are applied ONLY when the SAVE button is pressed BEFORE hanging up.

Play	Stop
Record	Save

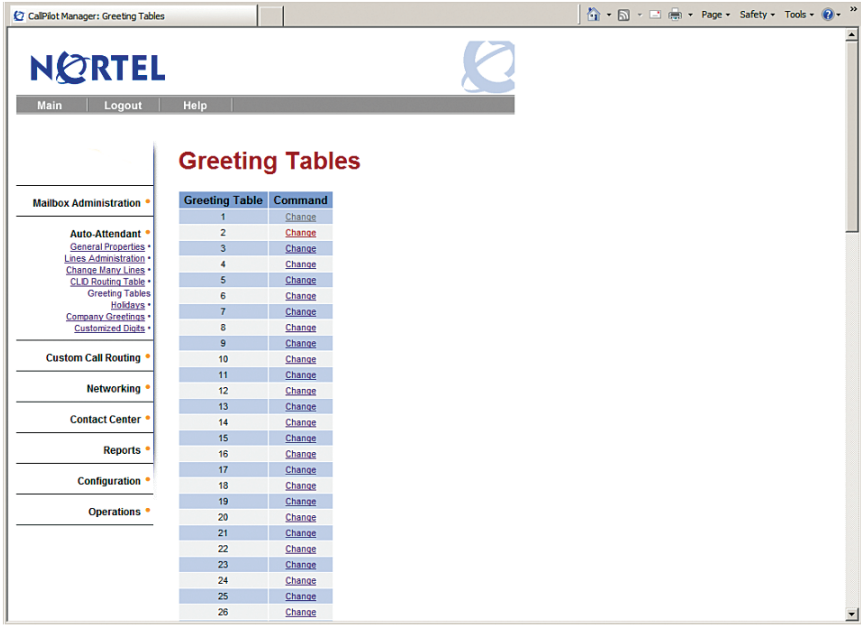
Import:

From:

Export:

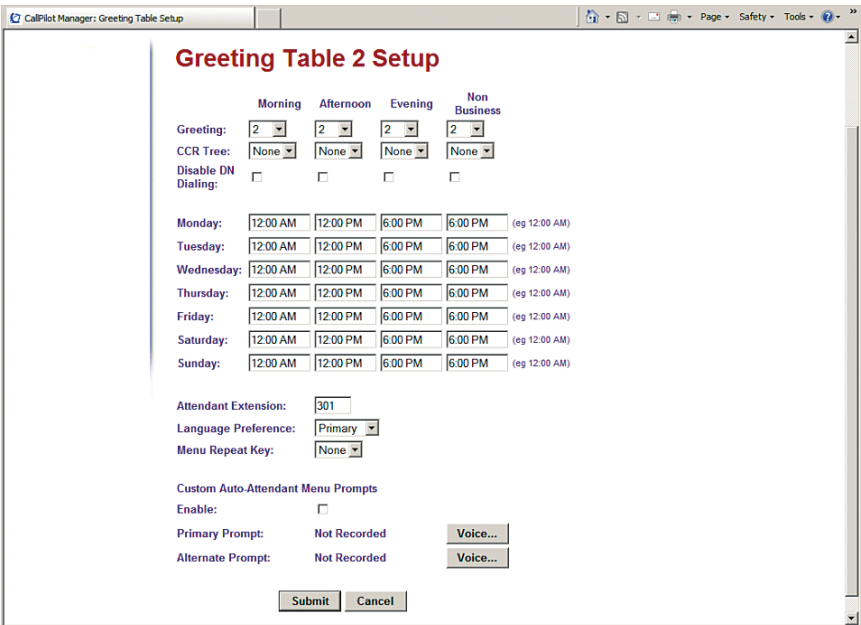
[Native Encoding](#) [Wav Encoding](#)

h. Click **“Greeting Tables”** and then click **“Change”** from **Greeting Table 2**.

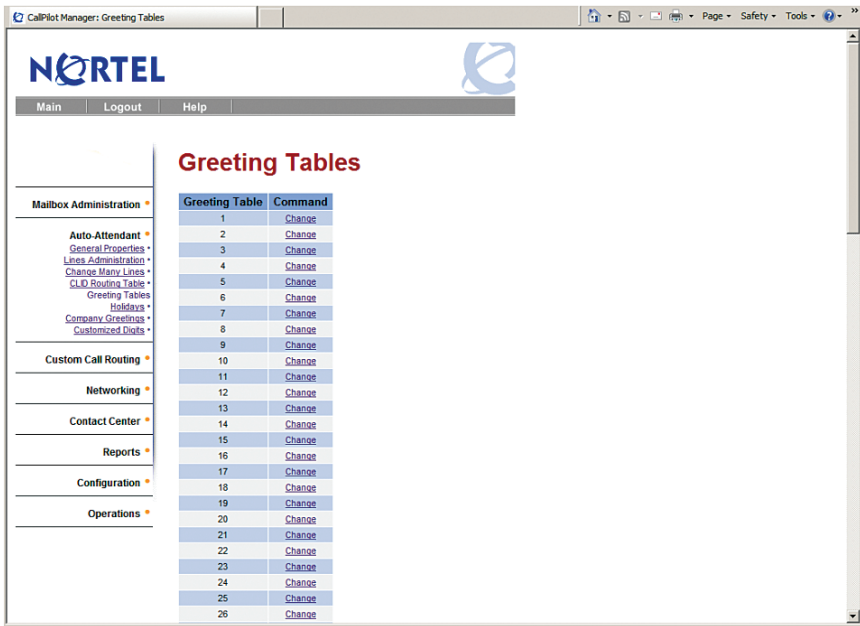


i. From the Greeting Table 2 Setup screen:

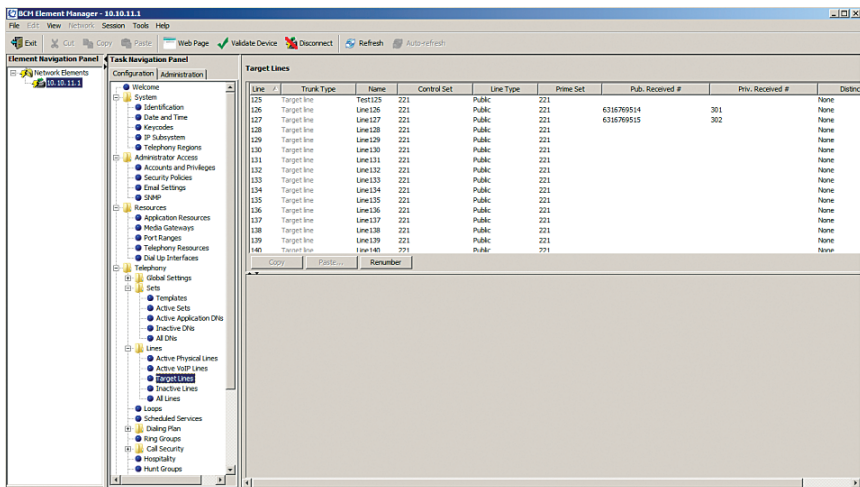
- 1) Select “2” in the **“Greeting”** fields under the Morning, Afternoon, Evening and Non Business column.
- 2) Enter “301” in the **“Attendant Extension”** field.
- 3) Leave other fields as default and click the **“Submit”** at the bottom of the screen.



j. Click the **“Lines Administration”** link.



k. Navigate to **“Configuration”>“Telephony”>“Lines”** and then select an **available target line** for assigning it to the Auto-Attendant. We will use **“125”** in this example.



l. Select **Line 125** and then click the **“Change”** link.

Line	Answer Mode	Table/Skillset	Rings	Command
1	No	---	0	Change
2	No	---	0	Change
3	No	---	0	Change
4	No	---	0	Change
5	No	---	0	Change
6	No	---	0	Change
7	No	---	0	Change
8	No	---	0	Change
9	No	---	0	Change
10	No	---	0	Change
11	No	---	0	Change
12	No	---	0	Change
13	No	---	0	Change
14	No	---	0	Change
15	No	---	0	Change
16	No	---	0	Change
17	No	---	0	Change
18	No	---	0	Change
19	No	---	0	Change
20	No	---	0	Change
21	No	---	0	Change
22	No	---	0	Change
23	No	---	0	Change
24	No	---	0	Change
25	No	---	0	Change

m. Select “Auto-Attendant” from the drop-down list of the **“Answer Mode”** field, enter “2” in the **“Table/Skillset Number”** field. Leave other fields as default and click the **“Submit”** button.

NORTEL

Main Logout Help

Line Properties

Line Number: 125

Answer Mode: Auto-Attendant

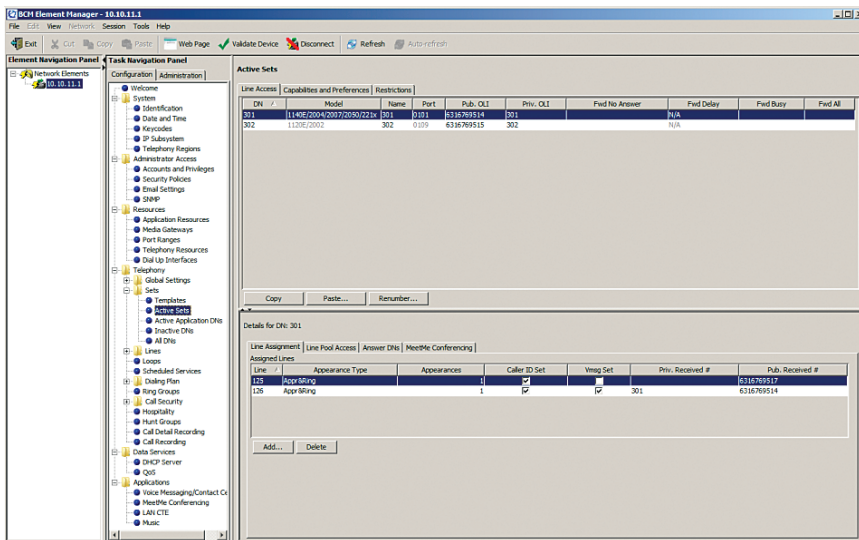
Table/Skillset Number: 2

Number of Rings: 0

Submit Cancel

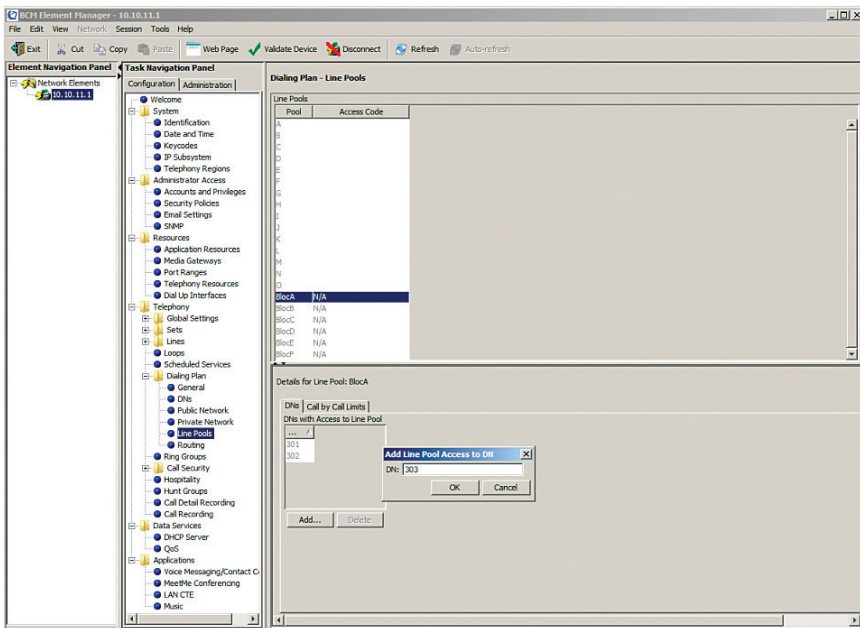
n. Navigate to **“Configuration”>“Telephony”>“Sets”** and then select the **“Active Sets”** tab to assign a DID (“631-676-9517” in this example) for the Auto-Attendant.

- 1) Select the **“Line Access”** tab and then select **DN 301**.
- 2) From the Assigned Lines section, click the **“Add”** button and follow the previous example to add target line “125” which has been set up for an Auto-Attendant.
- 3) Follow the previous example to assign the DID to target line “125”.

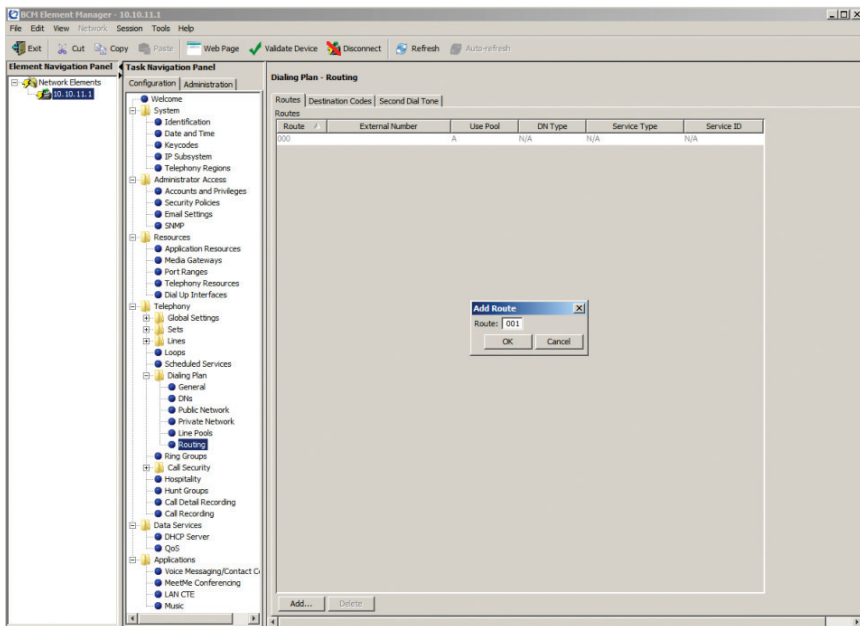


Setting up PBX to dial 9 first to place outbound calls via SIP trunks.

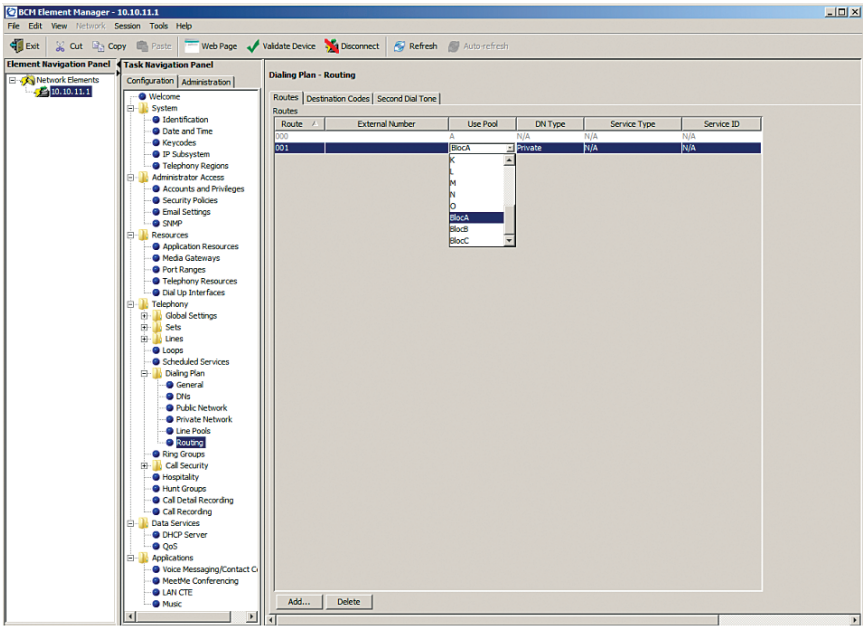
- a. Navigate to **“Configuration”>“Telephony”>“Dialing Plan”** and then select **“Line Pools”** to ensure the DN numbers have access to the same line pool assigned as the SIP trunks (line pool BlocA in this example). To add the DN for the line pool access, select **“BlocA”** from **“Pool”** column, click the **“Add”** button, enter the DN in the **“DN”** field and then click the **“OK”** button.



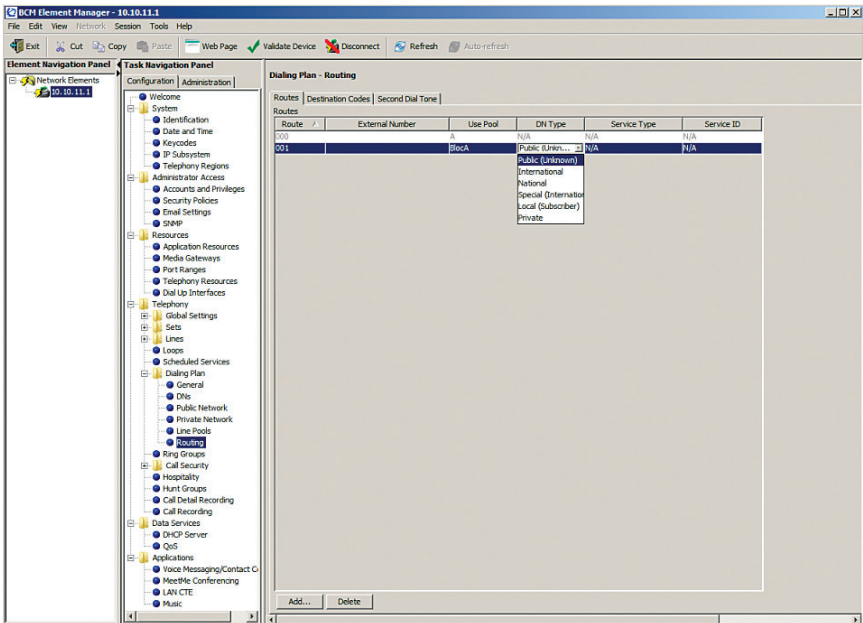
- b. Navigate to **“Configuration”>“Telephony”>“Dialing Plan”** and then select **“Routing”** to create a route for accessing the SIP trunks via line pool BlocA. Click the **“Add”** button, enter the next available route (**“001”** in this example) in the **“Route”** field of the **“Add Route”** pop-up box and then click the **“OK”** button.



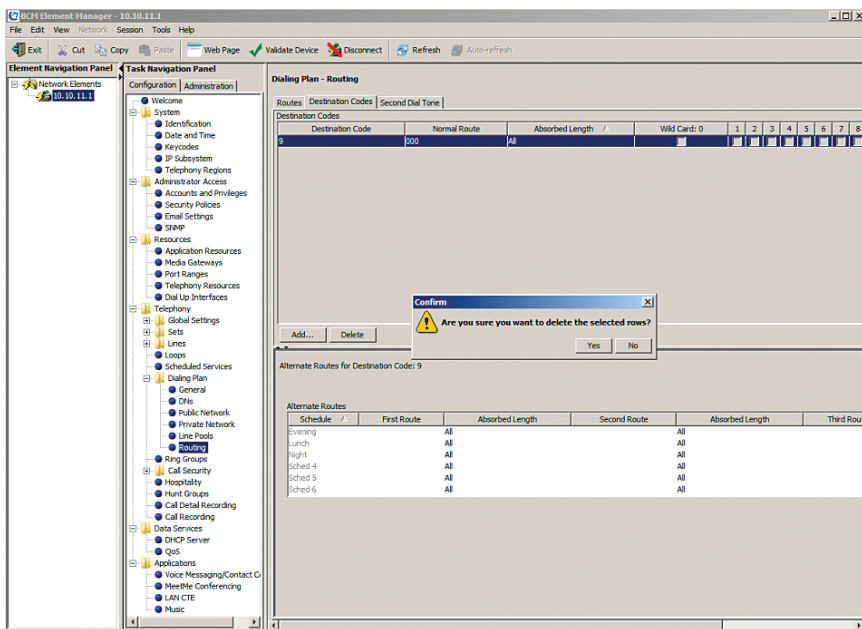
- c. Select **Route “001”** and then select “BlocA” from the drop-down list of its **“Use Pool”** field.



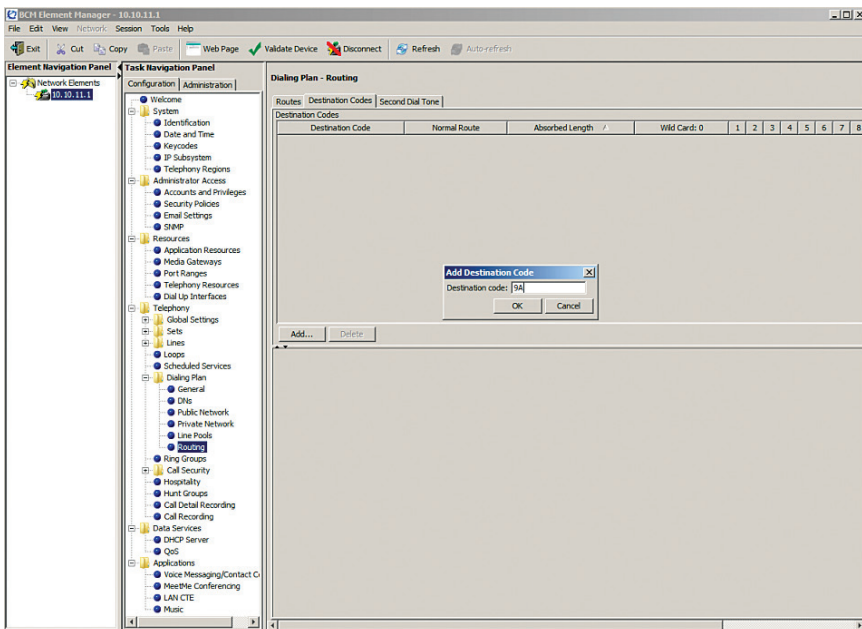
- d. Select **Route “001”** and then select “Public (Unknown)” from the drop-down list of its **“DN Type”** field.



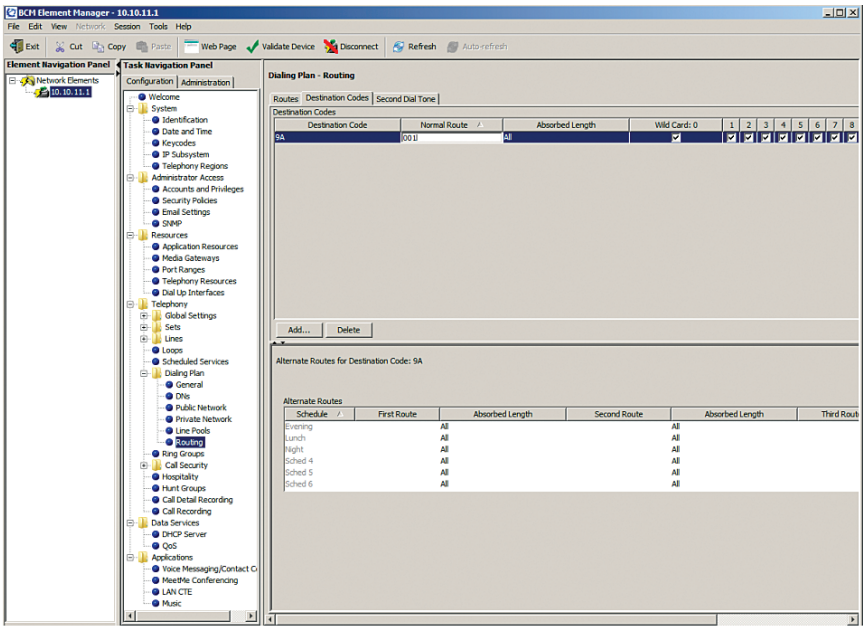
- e. Navigate to **“Configuration”>“Telephony”>“Dialing Plan”>“Routing”** and then select the **“Destination Codes”** tab to create a destination code of “9A” (“9” followed by any digits). First, we need to delete the default destination code of “9” (“9” followed by nothing). Select the **“9”** destination code and then click the **“Delete”** button.



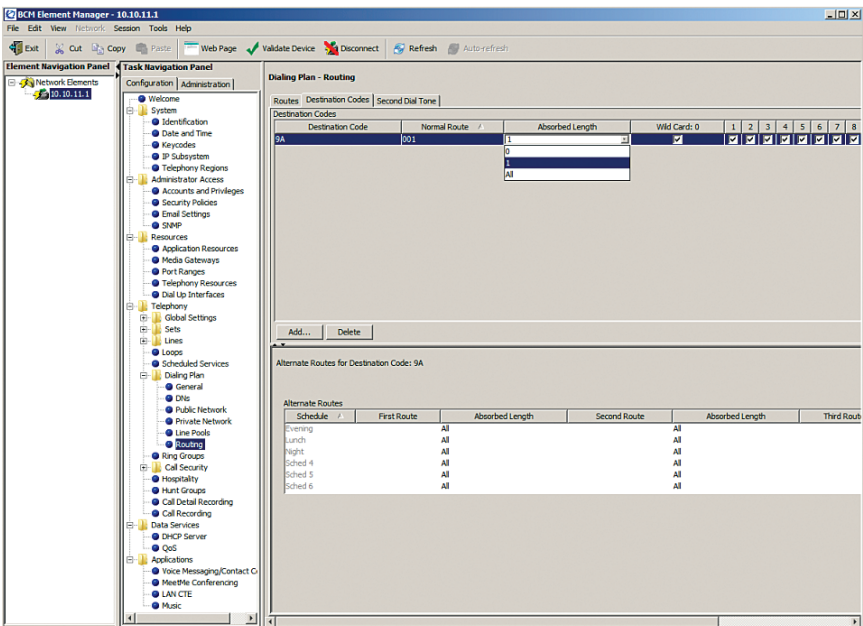
- f. Now click the **“Add”** button. From the **“Add Destination Code”** pop-up box, enter “9A” in the **“Destination Code”** field and then click the **“OK”** button.



g. Select **destination code “9A”** and then enter “001” in the **“Normal Route”** field.



h. Select **destination code “9A”** and then enter “1” in its **“Absorbed Length”** field (the PBX will strip “9” from the called number before sending the outbound call to the SIP trunk) and leave the other checkboxes checked.



- i. Navigate to **“Configuration”>“Telephony”>“Dialing Plan”** and then select the **“Public Network”** tab. From the Public Network Setting, elect **“10”** from the drop-down list of the **“Public Received number length”** field. From the Public Network DN Length section, select the **Default DN Prefix** and change the **“DN Length”** field from **“7”** (default) to **“10”**.

