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Optimum Business Trunking and the Mitel 5000 IP PBX Configuration Guide

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1 Overview

The purpose of this configuration guide is to describe the steps needed to configure the Mitel 5000 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: s1ptrunk

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.

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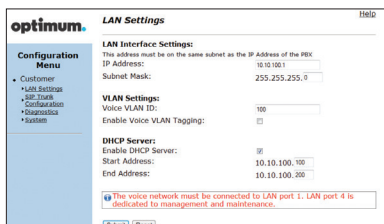
3. Optional: Enable the DHCP server. This will allow the SIP Trunk Adaptor to act as a DHCP server, which will provide IP addresses to the voice network, and create a dedicated voice LAN, as per diagram 2.
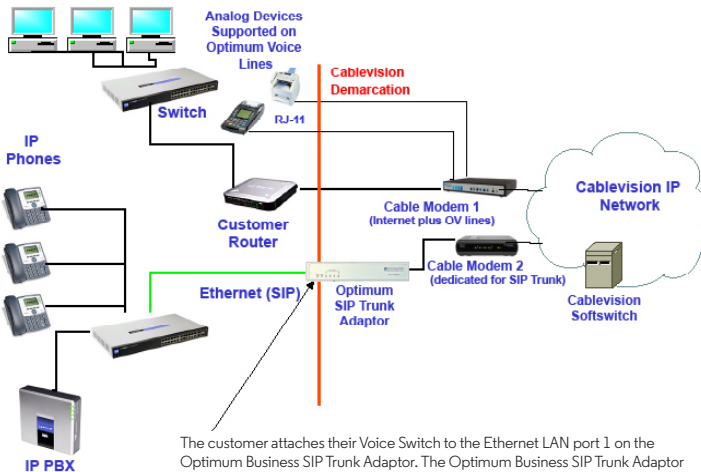
4. Click 'Submit'.

Diagram 1   SIP Trunk Adaptor for IP-PBXs
Example: Single LAN Configuration
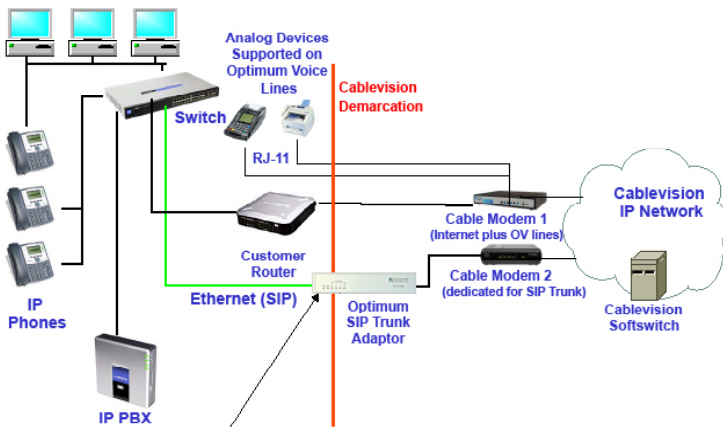
Using a connection from the customer's LAN, the SIP Trunk Adaptor's address can be a statically assigned private IP address. It may not be assigned a Public IP address.

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.

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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'.

Step 4:

Diagnostics Link

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

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

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

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.
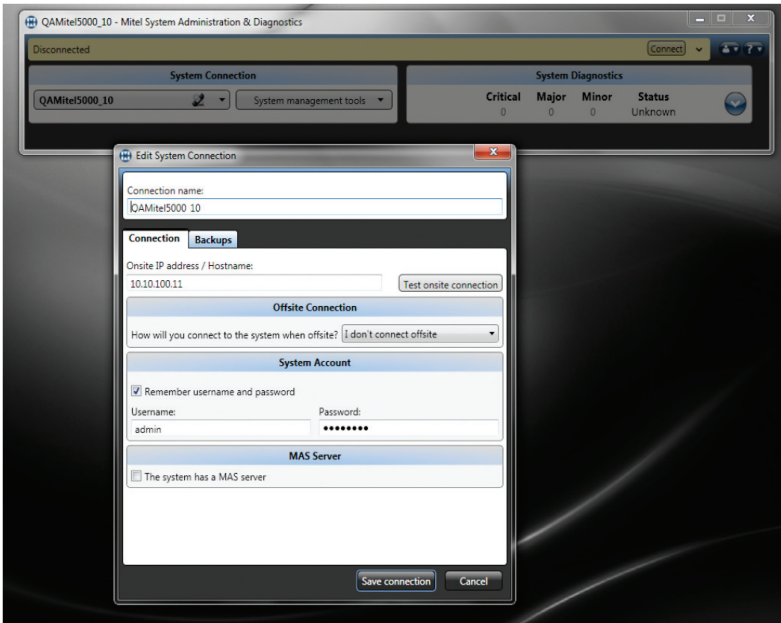
5 PBX Configuration

The steps on the next 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 Mitel 5000 product documentation for more information on advanced PBX features.

This guide provides the configuration steps for both PBX registration and static or non-registration modes of PBX operation.

The configuration described here assumes that the Mitel PBX is already configured and operational with station side phones using assigned extensions or DIDs. This configuration is based on Mitel 5000 Version 4.0.

The PBX should be connected to the Optimum Business SIP Trunk Adaptor LAN port 1 (IP address=10.10.100.1/24 in the lab) in the same LAN segment. The PBX was shipped with a private IP address 192.168.200.201/24 for its LAN port. In the lab, its IP address has been changed to 10.10.100.11/24.

From the CD that is shipped with the PBX, double-click the SAaDDBP_4_0_5_1-core_DBP_4_0_3_80-plugin.exe program to install the configuration GUI (Mitel System Administration & Diagnostics) on a PC. The installer will prompt for the PBX's IP address during the installation process. The PBX's LAN port IP address can be found from the display in the front of the PBX. After the installation of the configuration GUI, invoke the GUI and select the **"System Connection"** entry that has been configured to point to the PBX.

To connect to the PBX, click **"System management tools"** and, from the drop-down menu, click **"Launch DB Programming for QAMitel5000_10"**. You should see the initial configuration screen.
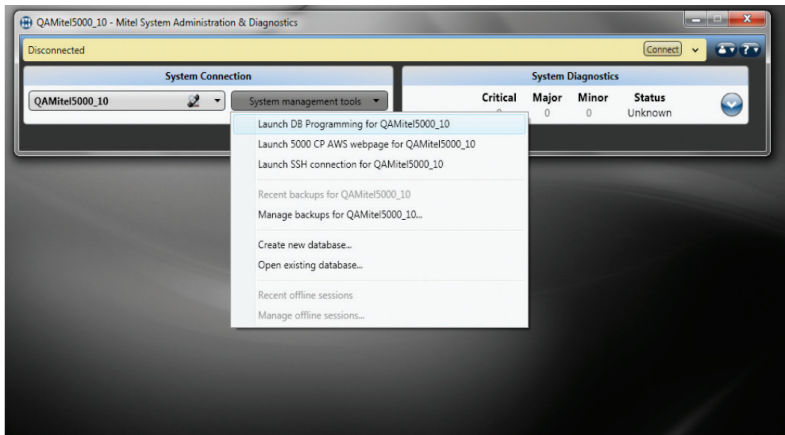

From the initial configuration screen, first configure a time zone for the PBX. Select a time zone from the drop-down menu of the Time Zone's **"Value"** field and hit the **"Enter"** key on the keyboard (or click anywhere on the right pane).
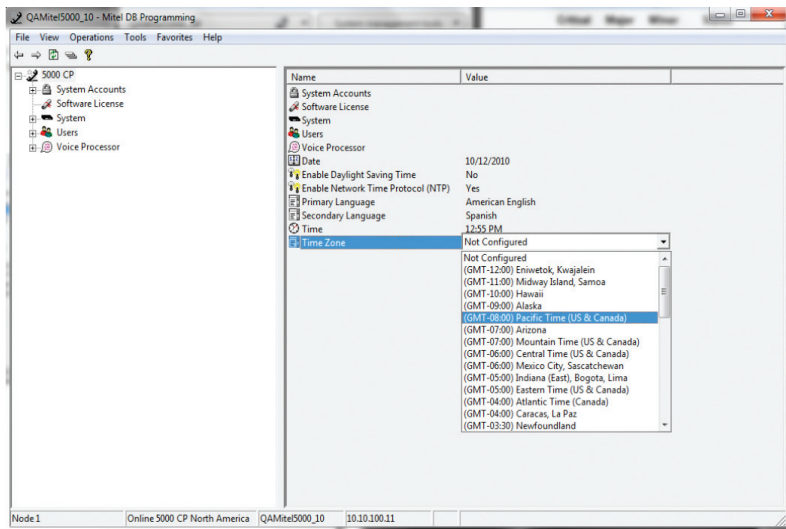
Before configuring the PBX for SIP trunk service, ensure that the PBX is equipped with enough SIP trunking licenses. Select **"Software License"** and check the value of the **"SIP Trunks"** field. This value represents the maximum number of concurrent calls allowed for the SIP trunk service.
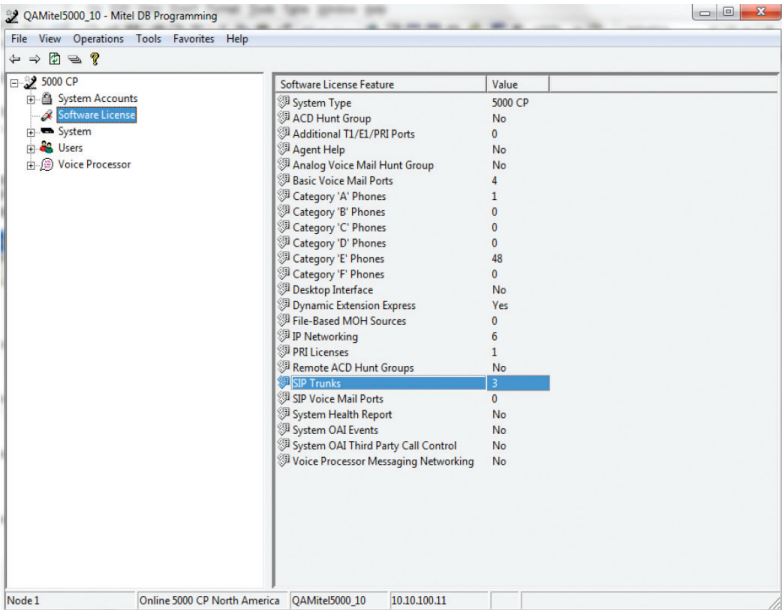
To start configuring the PBX for SIP trunk service, select **"System"**, select **"Device and Feature Codes"**, select **"SIP Peers"** and select **"SIP Trunk Groups"** to create a SIP Trunk Group. **Right-click** in the right-hand window panel and then select **"Create SIP Trunk Group"**.
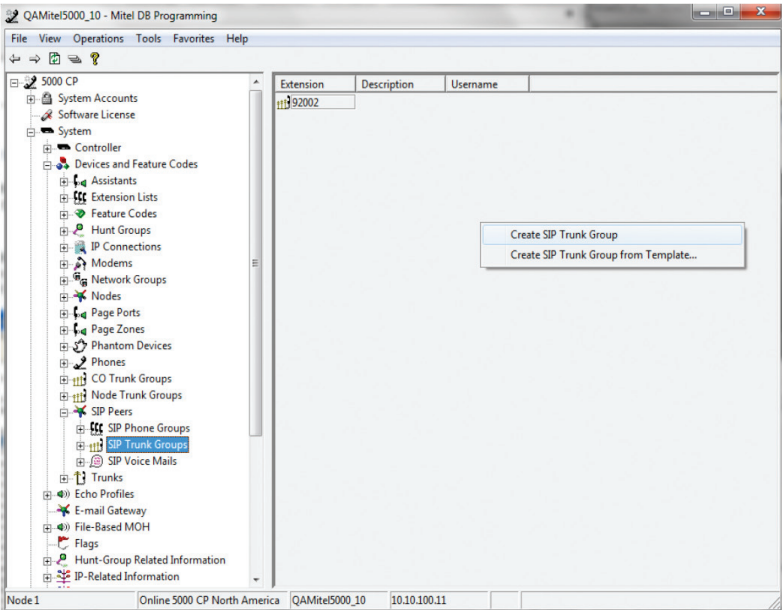
Select **"92002"** (the newly created SIP Trunk Group in this example) and select **"Configuration"** to configure the SIP trunk parameters.
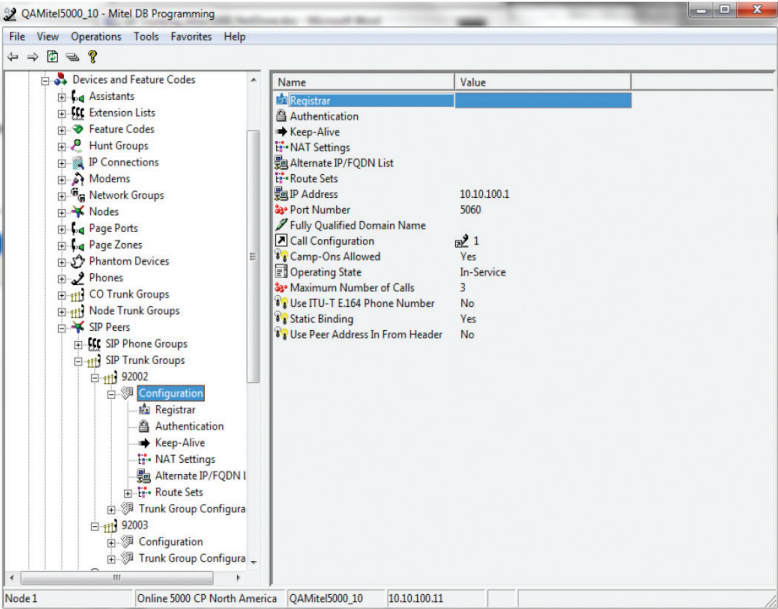
Enter the Optimum Business SIP Trunk Adaptor's IP address in the **"IP Address"** field. This was Step 2 in the Optimum Business SIP Trunk Set-Up Guide.

Enter 5060 in the **"Port Number"** field.

Make sure the **"Operating State"** field is set to **"In-Service"**.

Configure the **"Maximum Number of calls"** field to the maximum number of concurrent calls allowed according to the SIP trunks license.

Leave the other fields as default. Note that the other call parameters (i.e.: DTMF Encoding Setting) are stored in a call configuration file identified by the **"Call Configuration"** number. This number is the default number of **"1"** in this example.

Select **"92002"**, select **"Configuration"** and select **"Registrar"** to configure the PBX for SIP registration mode or static IP address mode (i.e.: no SIP registration).

For **static IP address mode**, select **"No"** in the **"Enable Registration"** field.
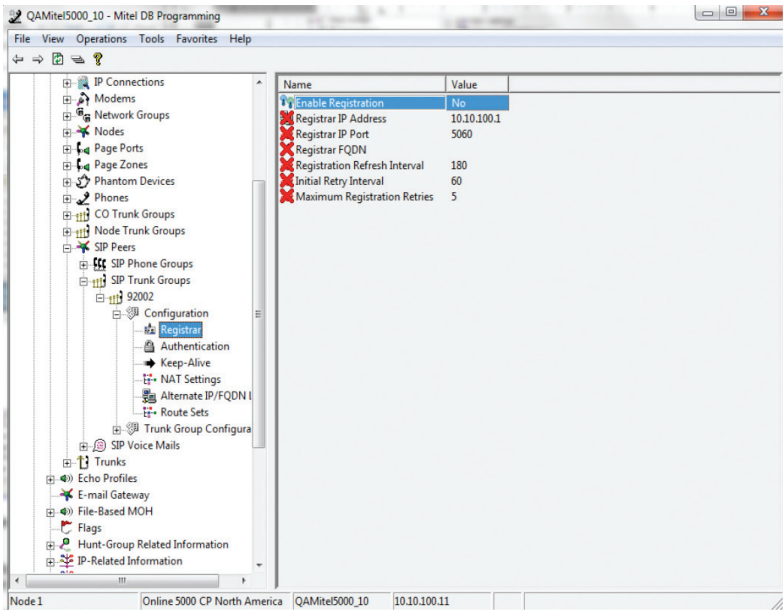
For **SIP registration mode**, select **"Yes"** in the **"Enable Registration"** field. Enter the Optimum Business SIP Trunk Adaptor's IP address in the **"Registrar IP Address"** field. This is Step 2 in the Optimum Business SIP Trunk Adaptor Set-Up Guide.

Enter 5060 in the **"Registrar IP Port"** field.

Leave the other fields as default.

Select **"Configuration"**, select **"Authentication"**, enter the username in the **"Out-bound Username"** field and the password in the **"Out-bound Password"** field. The username and password must match the **"User ID"** field and **"Password"** field configured on the Optimum Business SIP Trunk Adaptor. This was Step 3 in the Optimum Business SIP Trunk Set-Up Guide.
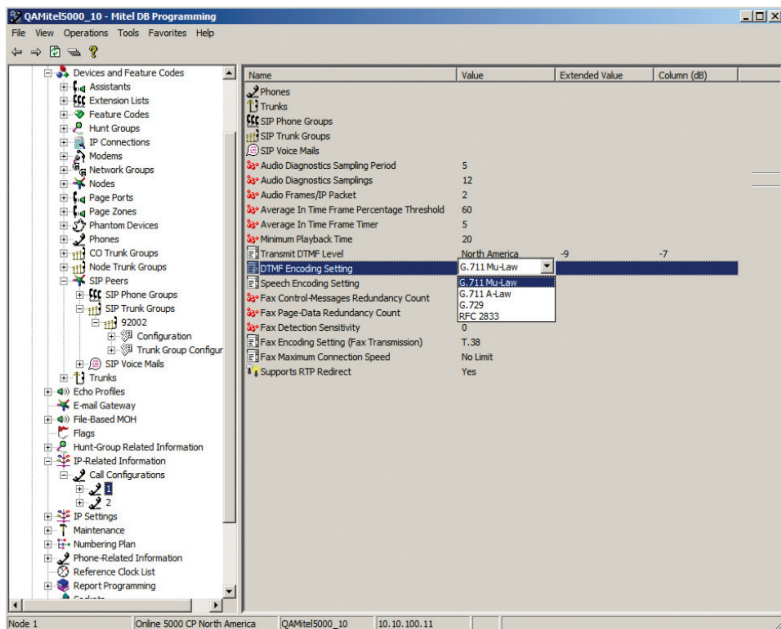
Leave **the other fields as default**.
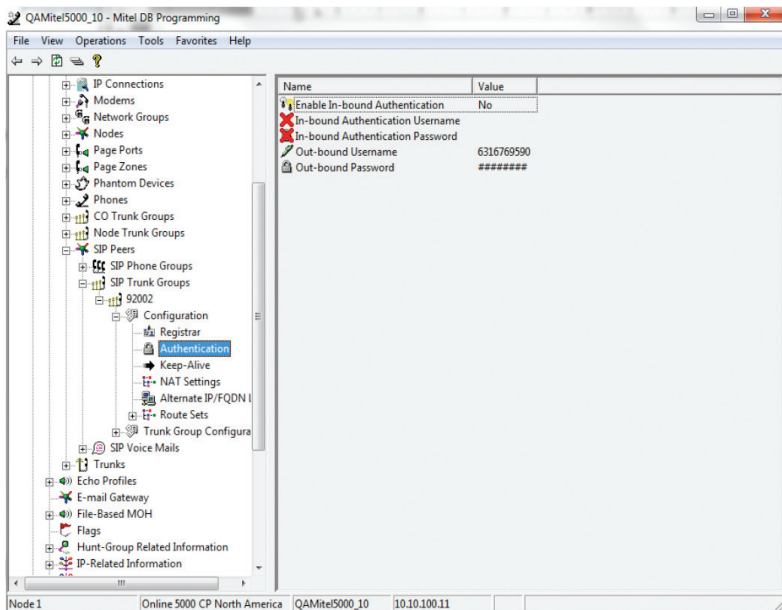
Select **"System"**, select **"IP-Related Information"**, select **"Call Configurations"** and select **"1"** (default call configuration index for the SIP trunk group).

Note: The Cablevision network only supports Inband DTMF tones.
Select **"G.711 Mu-Law"** in the **"DTMF Encoding Setting"** field.
Leave other fields as default.

Select **"System"**, select **"Devices and Feature Codes"**, select **"SIP Peers"**, select **"SIP Trunk Groups"**, select **"92002"**, select **"Trunk Group Configuration"** and select **"Trunks"** to create SIP Peer trunks. **Right-click** on the right pane and then click the **"Create SIP Peer Trunk"** link.
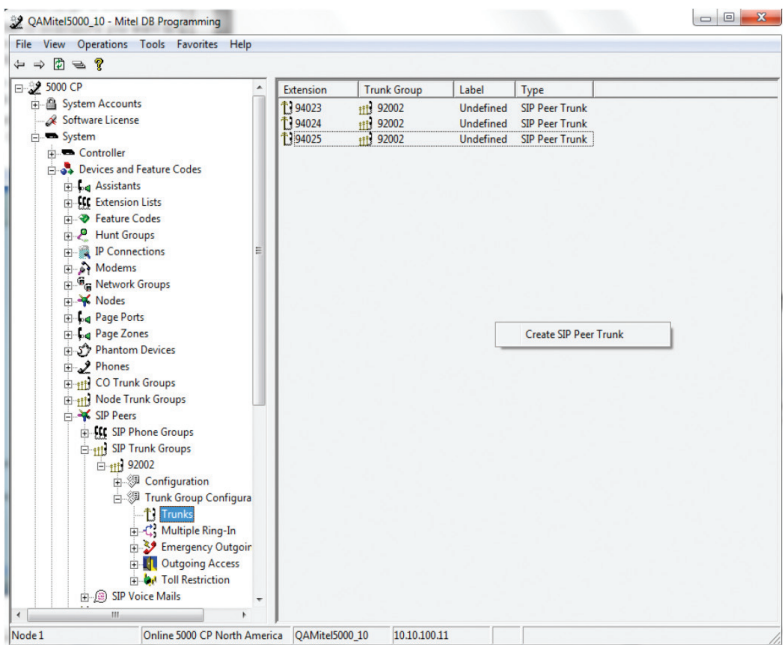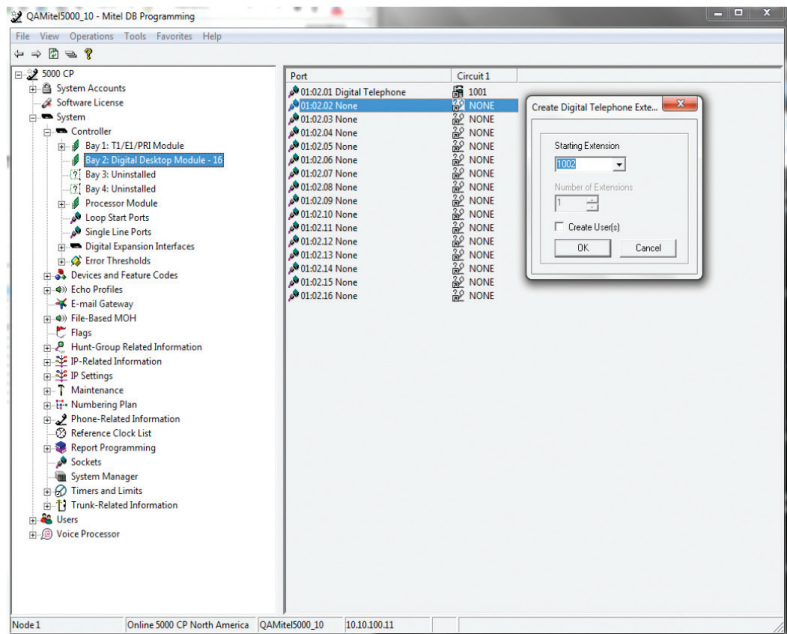

Select **"System"**, select **"Controller"** and select **"Digital Desktop Module - 16"** to set up digital phones. Select **"Digital Telephone"** in the drop-down list from the next available **"Port"** field.
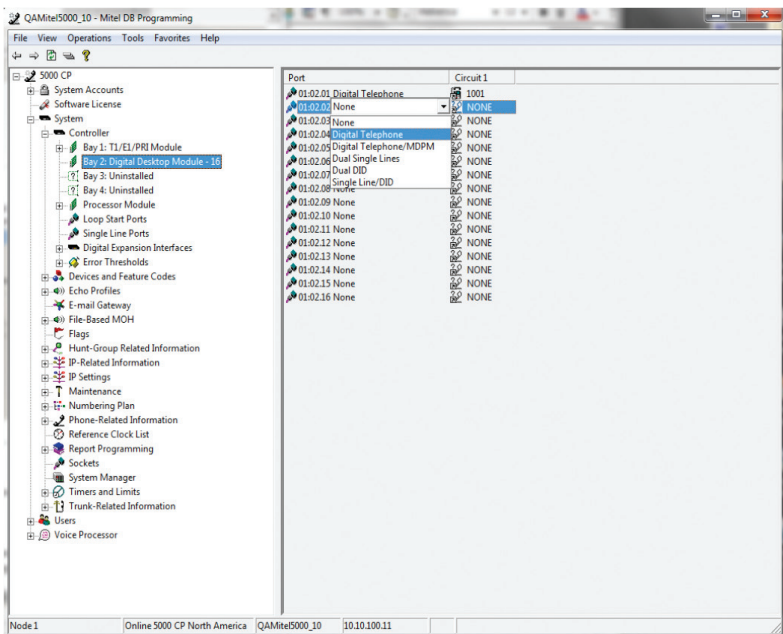
Click on the **"Circuit 1"** field, select the phone extension from the **"Starting Extension"** field of the pop-up box and click the **"OK"** button.


Select **"System"**, select **"Devices and Feature Codes"**, select **"SIP Peers"**, select **"SIP Trunk Groups"**, select **"92002"** and select **"Trunk Group Configuration"** to use routing table for mapping incoming calls to the extensions.

From both the **"Day Ring-In Type"** entry and the **"Night Ring-In Type"** entry, select **"Call Routing Table"** from the **"Value"** field and enter **"1"** in the **"Extended Value"** field. Note that the extended value of **"1"** means using the Call Routing Table 1.

Enter the calling party number in the **"Calling Party Number"** field. The calling party number is the pilot DID in this example.

Leave other fields as default.
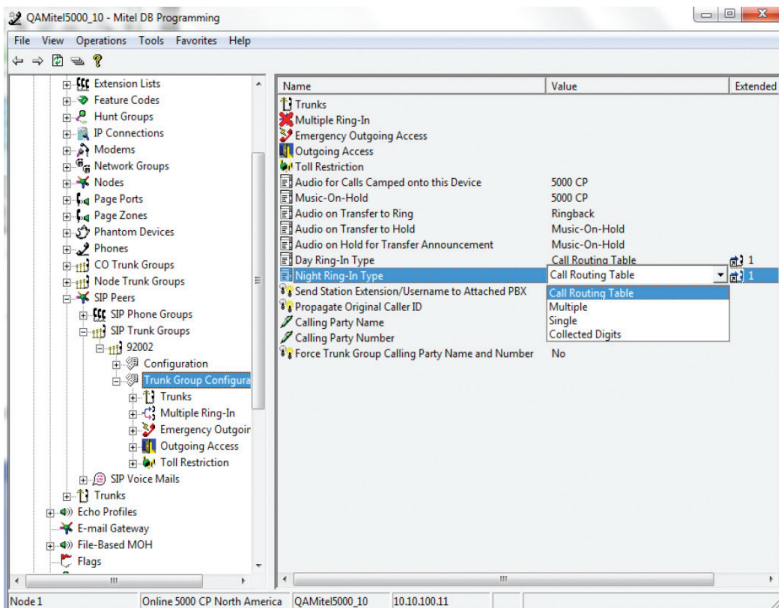
Select **"System"**, select **"Trunk-Related Information"**, select **"Call Routing Tables"** and select table entry **"1"** to configure mapping of incoming calls to extensions. For each mapping:

**Right-click** on the right pane and click on **"Add To 1 list…"** from the pop-up box.

Enter the DID assigned to the extension in the **"Pattern"** field.

Enter a description in the **"Description"** field.

Select **"Single"** in the **"Ring-In Type"** field.

Enter the extension in the **"Ring-In Destination"** field.

Note: If the extension is set up for Auto-Attendant, set the **"DTMF Encoding Setting"** field to **"G.711 Mu-Law"** to allow the Auto-Attendant to recognize in-band DTMF.
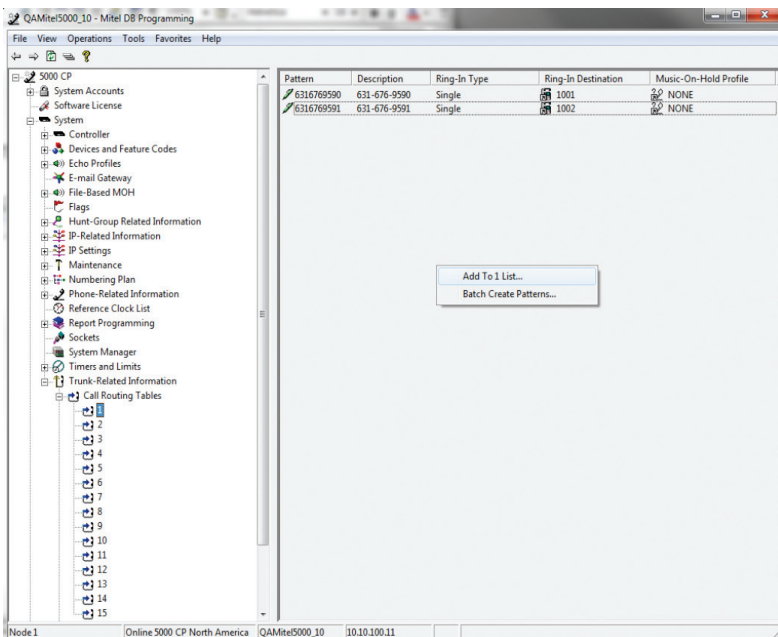
Select "**System**", select "**Devices and Feature Codes**" and select "**Feature Codes**" to make sure that Feature Code "**8**" is used for making outgoing calls. Note that by default, the user can dial the Feature Code "**8**", followed by "**1**" and the phone number to place outside calls (i.e.: 8 1 4083517255). The user may also dial by the SIP trunk group's extension (i.e.: 92002 1 4083517255) or dial by each SIP trunk extension (i.e.: 94023 1 4083517255).
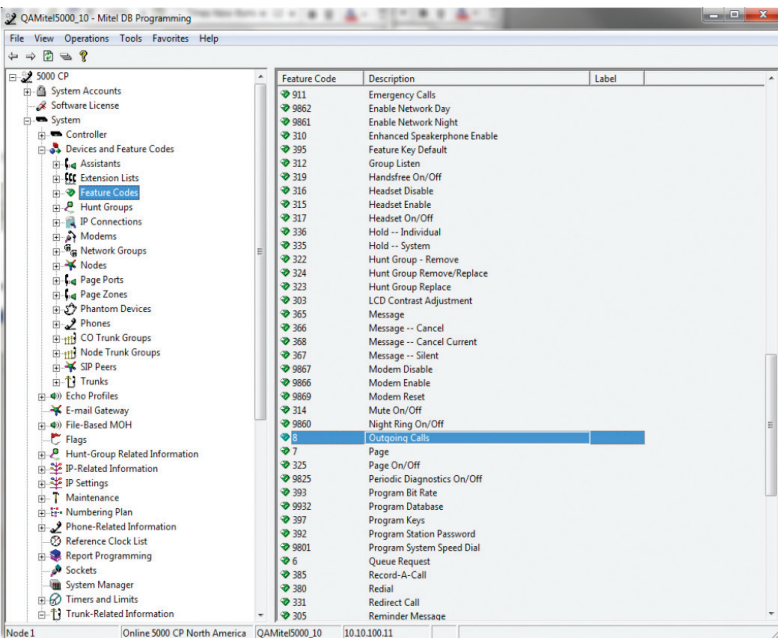
Select **"System"**, select **"Devices and Feature Codes"**, select **"Phones"**, select the **phone's extension** and select **"Associated Extensions"** to verify the **"Outgoing Extension"** field is set to the SIP trunk group extension ("92002" in this example).
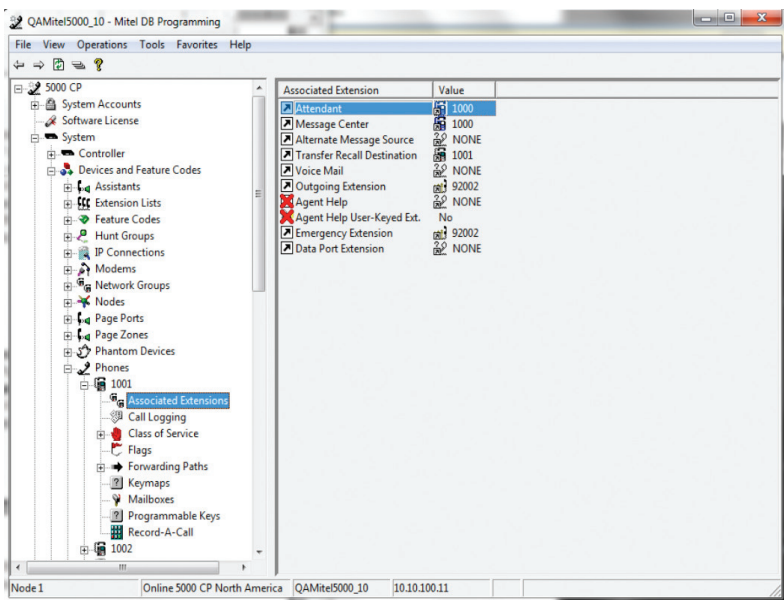

Select **"System"**, select **"Devices and Feature Codes"**, select **"Phones"**, select the phone's extension to configure Calling Party Name and Calling Party Number for each extension. Enter a name in the **"Value"** field for the Calling Party Name and enter the DID assigned for the extension in the **"Value"** field for the Calling Party Number only when the PBX is configured for static IP address mode.
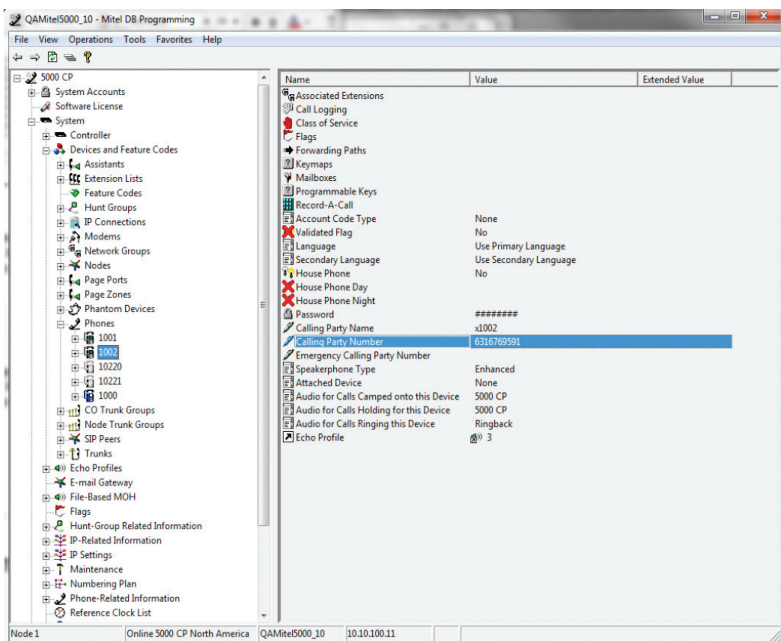
If the PBX is configured for SIP registration mode, the calling party number must be the same as the username (i.e.: the pilot DID of 6316769590 used in the PBX registration example) used for SIP registration. In this example, the PBX has been configured for static IP address mode.

Select **"System"**, select **"Devices and Feature Codes"**, select **"Phones"**, select the phone's extension and select **"Flags"** to enable unattended transfer for the phone extension (disabled by default). In the **"Value"** field for the **"Transfer to Connect Allowed"** entry, change **"No"** to **"Yes"**.
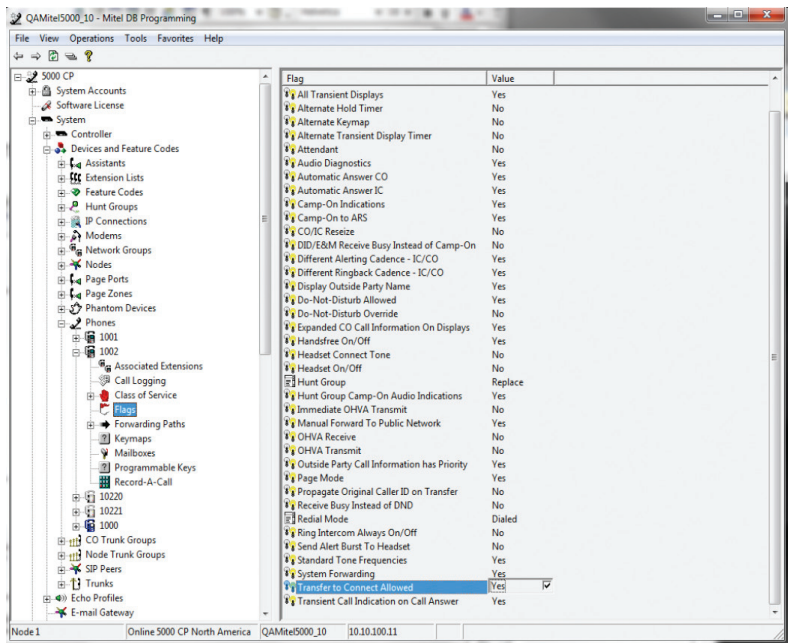
Note: When making configuration changes to the PBX, some may take effect right away, and some may require a reset. When in doubt, save all the changes (**"Operations** > **Database Operations** > **Database Save..."**) and restart (**"Operations** > **Reset System"**) the PBX.