

SIP Trunking using Optimum Business® SIP Trunk Adaptor and the Mitel 5000 IP PBX

Goal

The purpose of this configuration guide is to describe the steps needed to configure the Mitel 5000 IP Version 4.0 for proper operation with Optimum Business SIP Trunking.

Prerequisites

Please follow the instructions in the Optimum Business SIP Trunk Set-Up Guide. The Set-Up Guide was left by the Optimum Business technician at installation. If you do not have the Set-Up Guide, go to www.optimumbusiness.com/SIP to download a copy.

Mitel PBX Configuration

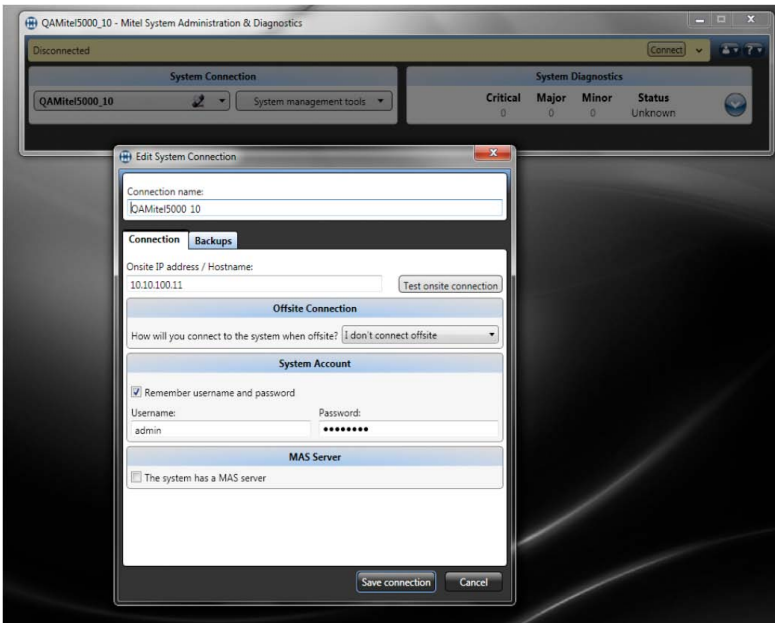
The steps on the next pages describe the minimum configuration required to enable the PBX to use Optimum Business SIP Trunking for inbound and outbound calling. Please refer to the Mitel 5000 product documentation for more information on advanced PBX features.

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

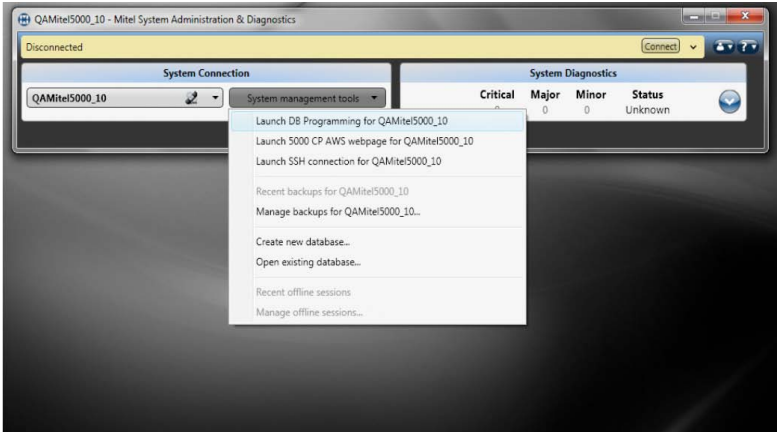
The configuration described here assumes that the Mitel PBX is already configured and operational with station side phones using assigned extensions or DIDs. This configuration is based on Mitel 5000 Version 4.0.

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 10.10.100.1/24.

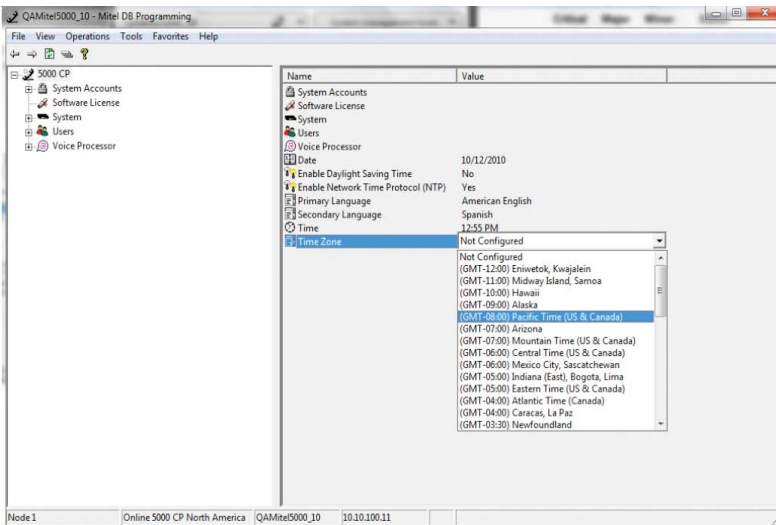
From the CD that is shipped with the PBX, double-click the SAaDDBP_4_0_5_1-core_DBP_4_0_3_80-plugin.exe program to install the configuration GUI (Mitel System Administration & Diagnostics) on a PC. The installer will prompt for the PBX's IP address during the installation process. The PBX's LAN port 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.



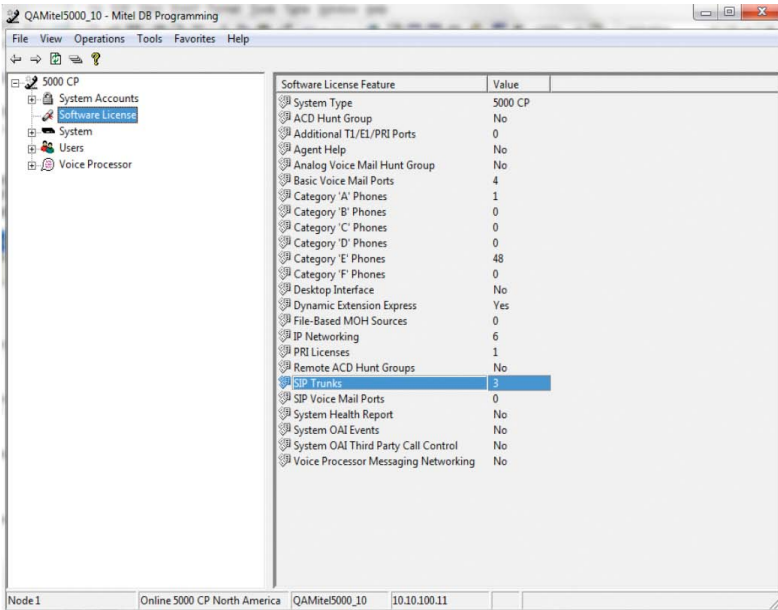
To connect to the PBX, click **“System management tools”** and, from the drop-down menu, click **“Launch DB Programming for QAMite!5000_10”**. You should see the initial configuration screen.



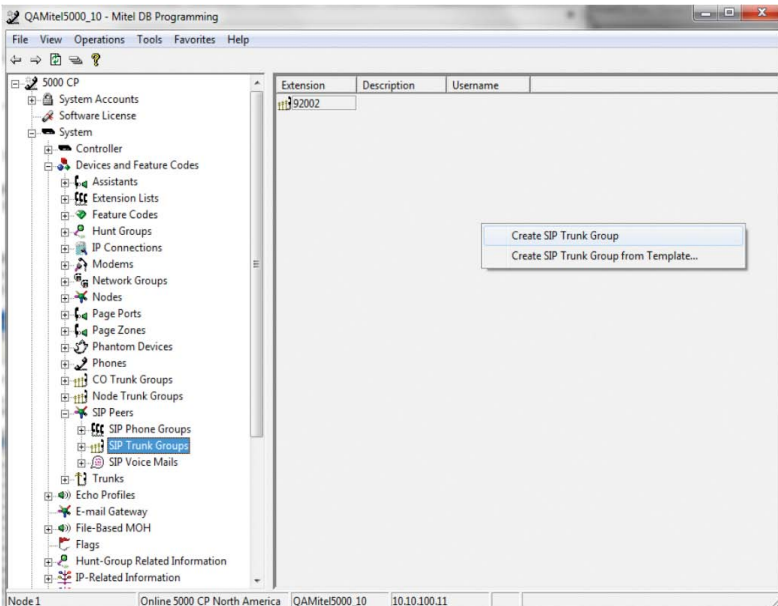
From the initial configuration screen, first configure a time zone for the PBX. Select a time zone from the drop-down menu of the Time Zone's **“Value”** field and hit the **“Enter”** key on the keyboard (or click anywhere on the right pane).



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.

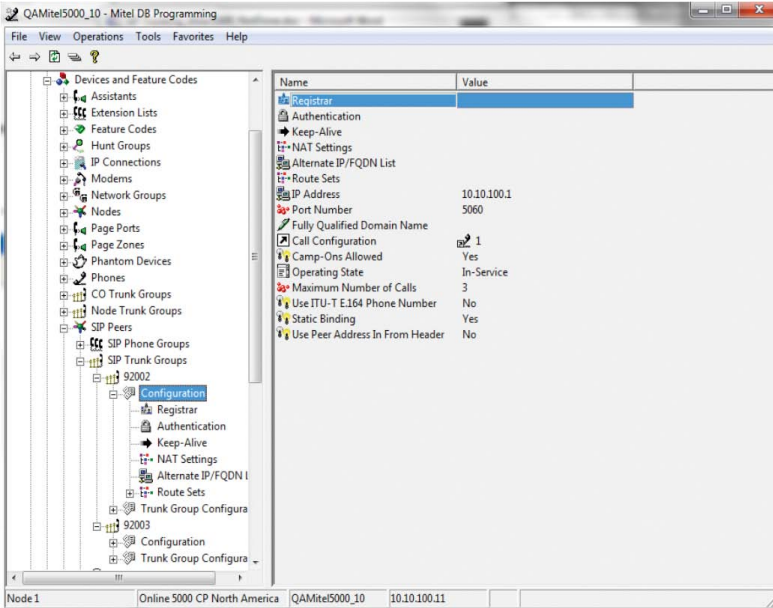
Enter the Optimum Business SIP Trunk Adaptor’s IP address in the “IP Address” field. This was Step 2 in the Optimum Business SIP Trunk Set-Up Guide.

Enter 5060 in the “Port Number” field.

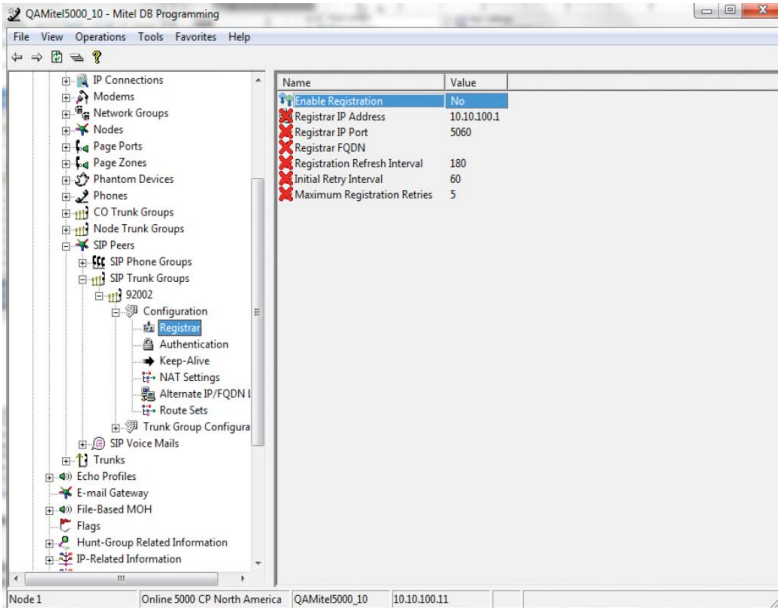
Make sure the “Operating State” field is set to “In-Service”.

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.



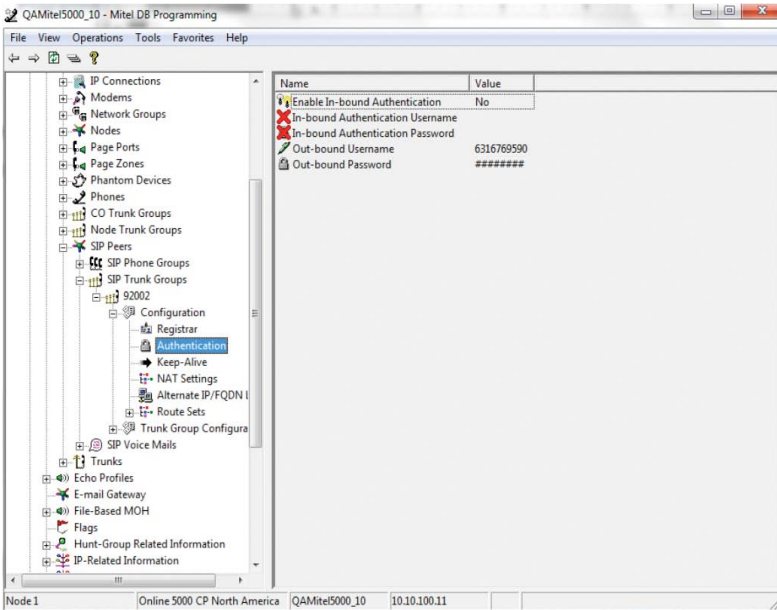
For **SIP registration mode**, select **“Yes”** in the **“Enable Registration”** field. Enter the Optimum Business SIP Trunk Adaptor’s IP address in the **“Registrar IP Address”** field. This is Step 2 in the Optimum Business SIP Trunk Adaptor Set-Up Guide.

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**.

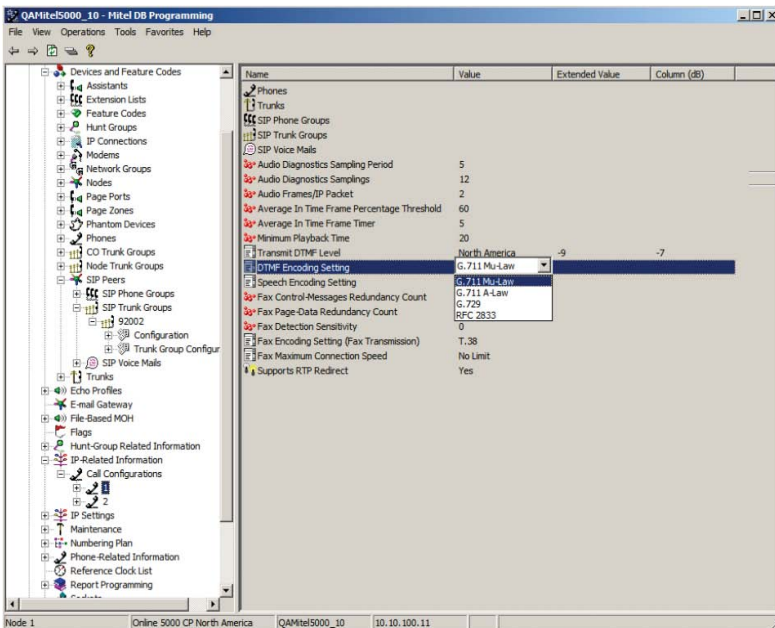


Select **"System"**, select **"IP-Related Information"**, select **"Call Configurations"** and select **"1"** (default call configuration index for the SIP trunk group).

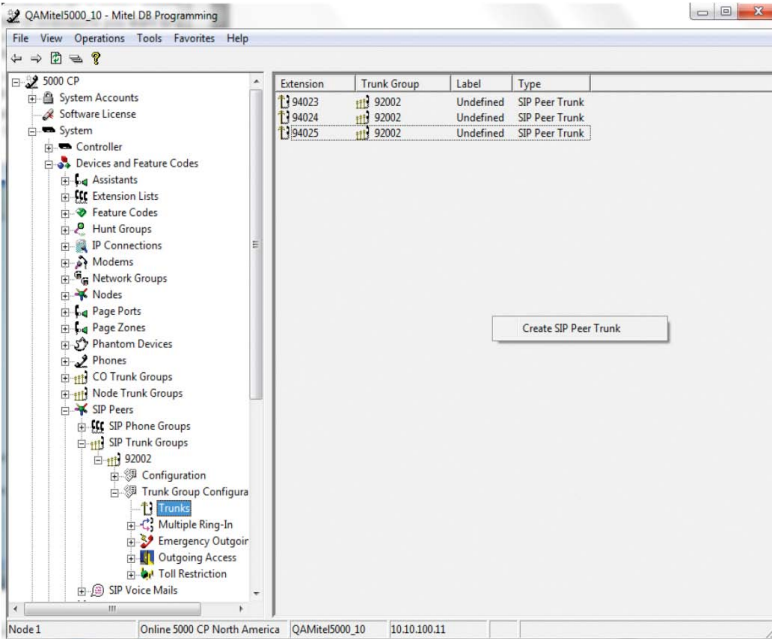
Note: The Cablevision network only supports Inband DTMF tones.

Select **"G.711 Mu-Law"** in the **"DTMF Encoding Setting"** field.

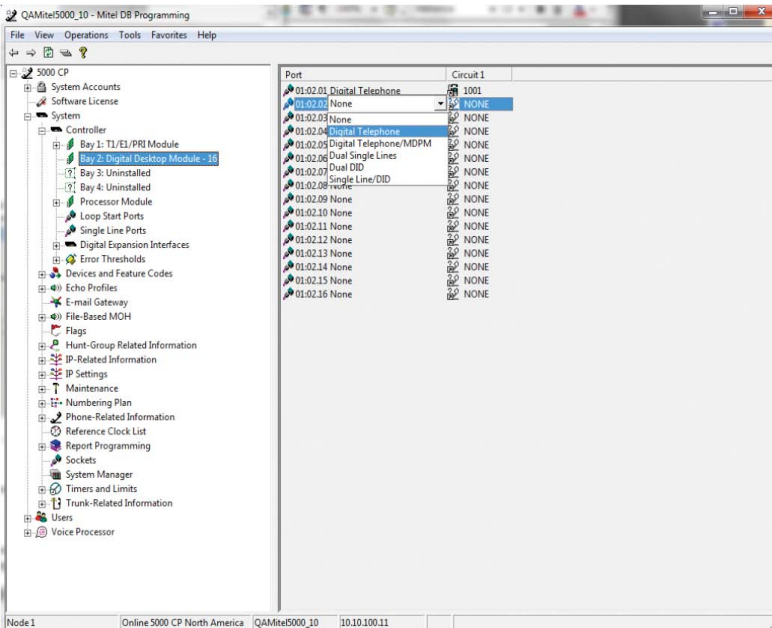
Leave other fields as default.



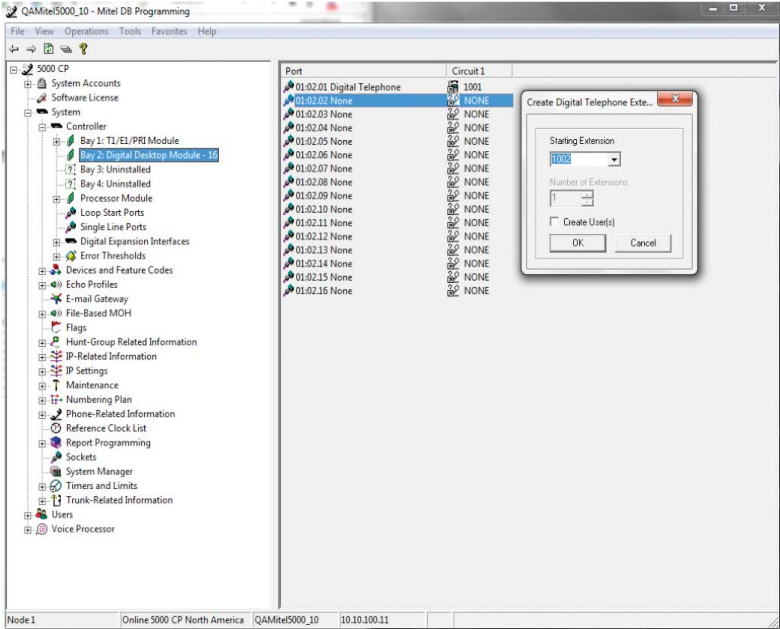
Select **"System"**, select **"Devices and Feature Codes"**, select **"SIP Peers"**, select **"SIP Trunk Groups"**, select **"92002"**, select **"Trunk Group Configuration"** and select **"Trunks"** to create SIP Peer trunks. **Right-click** on the right pane and then click the **"Create SIP Peer Trunk"**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

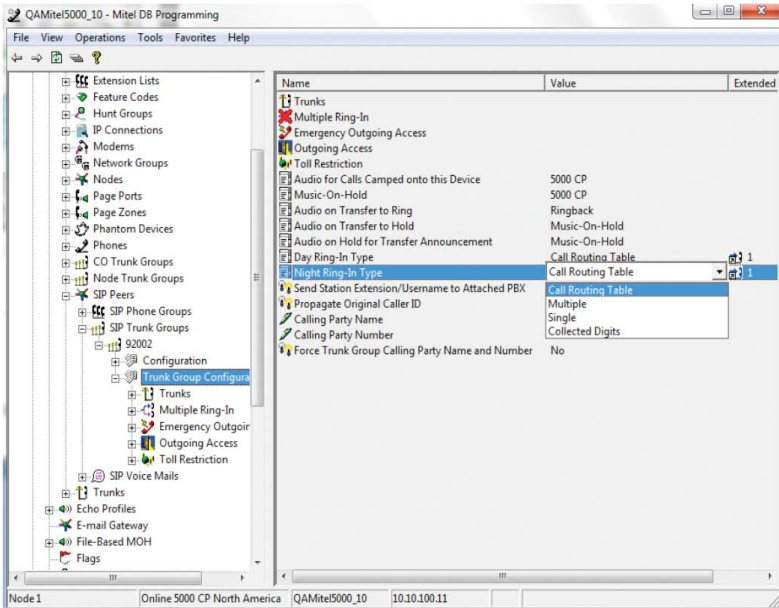


Select **"System"**, select **"Devices and Feature Codes"**, select **"SIP Peers"**, select **"SIP Trunk Groups"**, select **"92002"** and select **"Trunk Group Configuration"** to use routing table for mapping incoming calls to the extensions.

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.

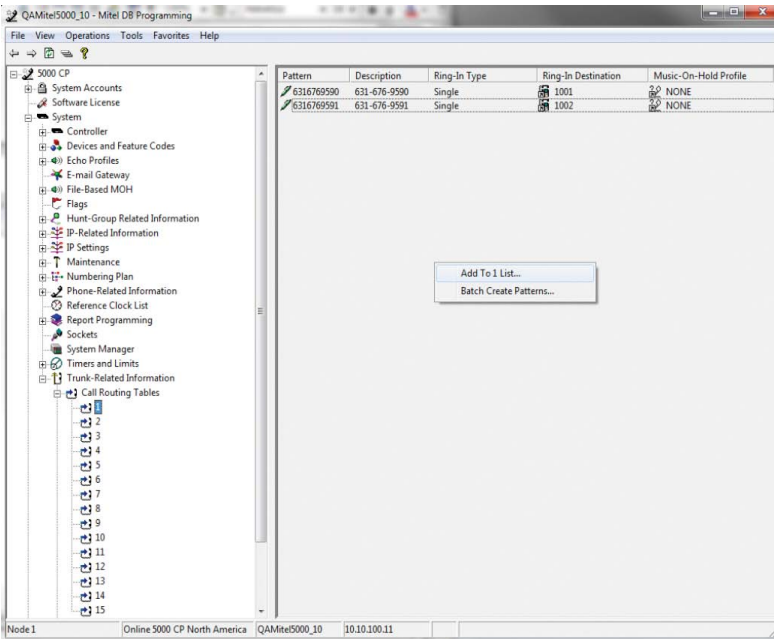
Enter the DID assigned to the extension in the **“Pattern”** field.

Enter a description in the **“Description”** field.

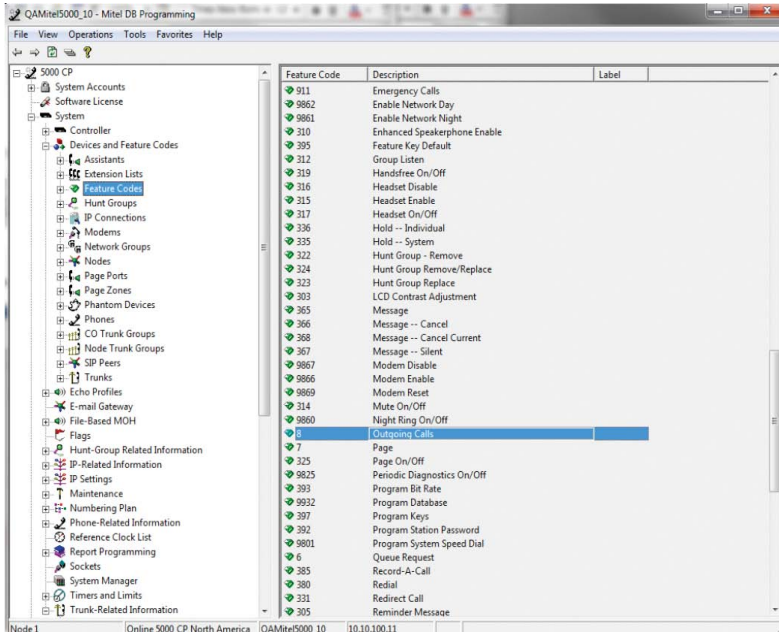
Select **“Single”** in the **“Ring-In Type”** field.

Enter the extension in the **“Ring-In Destination”** field.

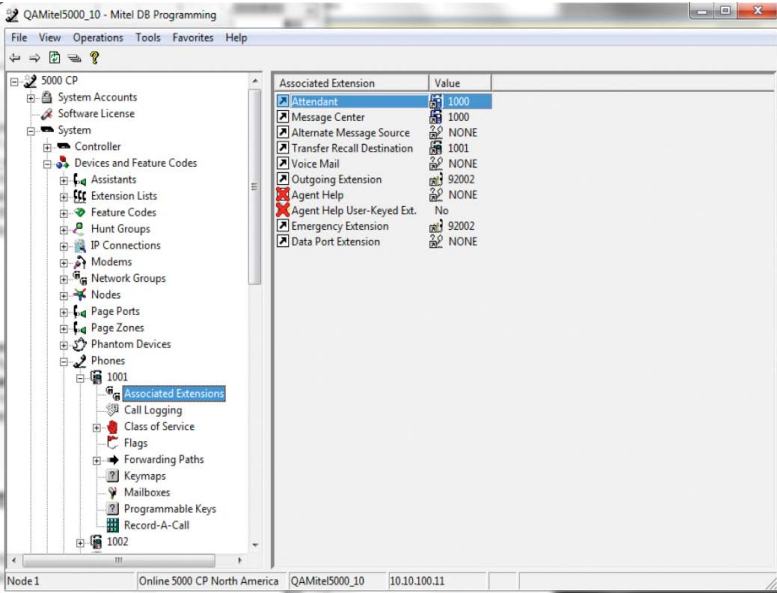
Note: If the extension is set up for Auto-Attendant, set the **“DTMF Encoding Setting”** field to **“G.711 Mu-Law”** to allow the Auto-Attendant to recognize in-band DTMF.



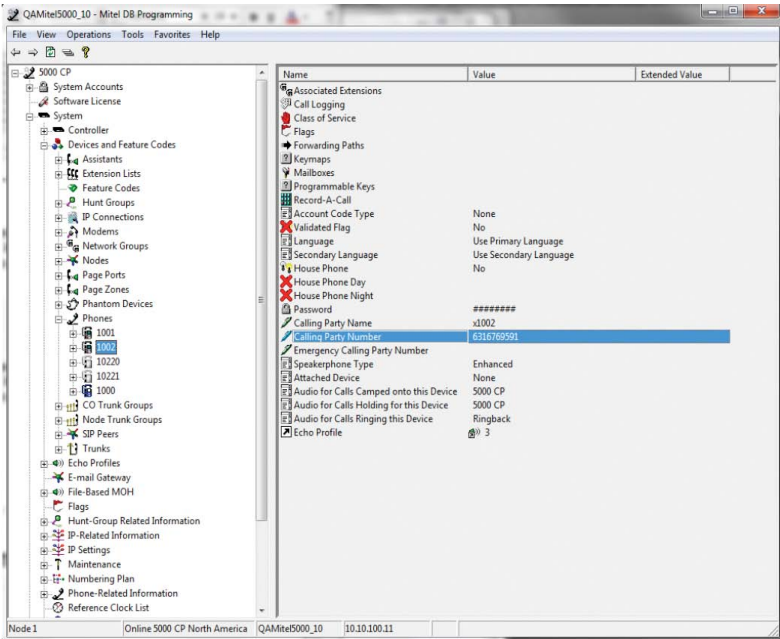
Select **"System"**, select **"Devices and Feature Codes"** and select **"Feature Codes"** to make sure that Feature Code **"8"** is used for making outgoing calls. Note that by default, the user can dial the Feature Code **"8"**, followed by **"1"** and the phone number to place outside calls (i.e.: 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).

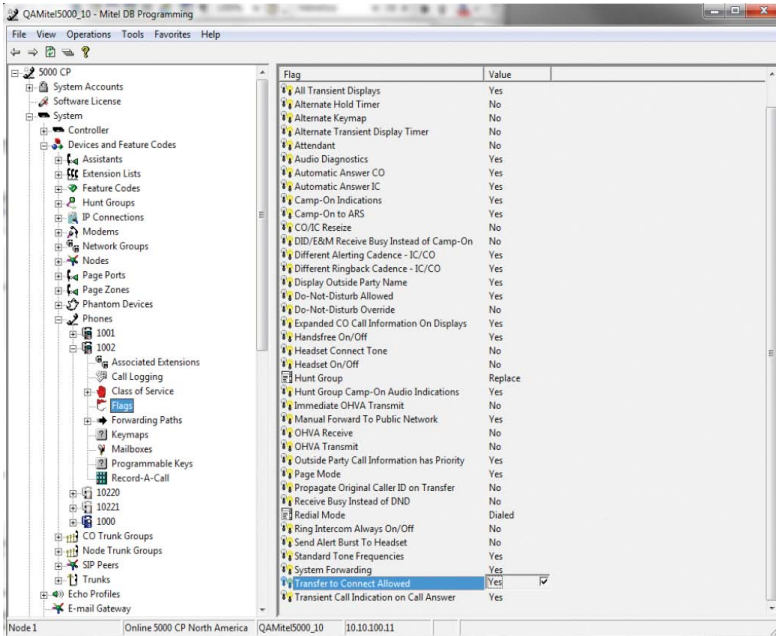


Select **“System”**, select **“Devices and Feature Codes”**, select **“Phones”**, select the **phone’s extension** to configure Calling Party Name and Calling Party Number for each extension. Enter a name in the **“Value”** field for the Calling Party Name and enter the DID assigned for the extension 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.