Dialogic® 4000 Media Gateway Series Integration Note

Mitel 3300 ICP

August 2008 64-0355-01

www.dialogic.com

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1. Scope

This document is intended to detail a typical installation and configuration of the Dialogic® 4000 Media Gateway Series if used to interface between a PBX and the Microsoft® Office Communications Server (OCS) application.

2. Configuration Details

Listed below are the specific details of the PBX and gateways used in the testing to construct the following documentation.

2.1 PBX

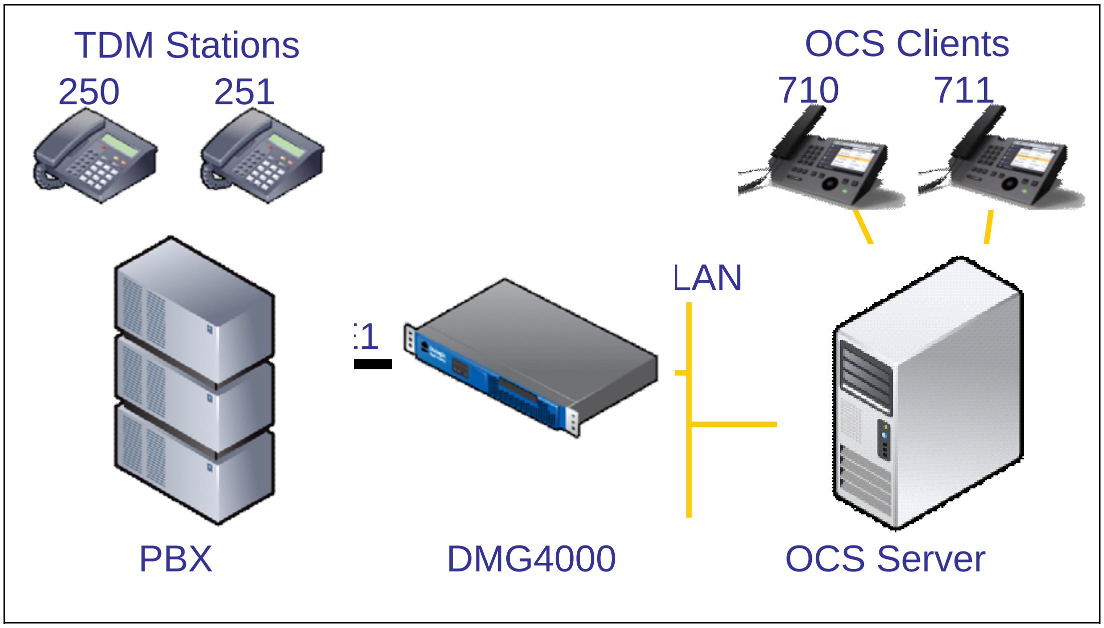
PBX Vendor	Mitel
Model(s)	3300 ICP
Software Version(s)	System Version 5.1.4.8
Additional Notes	N/A

2.2 Gateway

Gateway Model	Dialogic 4000 Media Gateway Series
Software Version(s)	Dialogic® Diva® System Release software version 8.3.2 build 459 (formerly called Diva® Server software) Dialogic® Diva® SIPcontrol™ Software version 1.6 build 46 (DSSIPControl.msi)
Protocol	E1 Q.SIG

2.3 System Diagram

The diagram below details the setup used in the testing and creation of this document. In the diagram, the abbreviation DMG4000 stands for the Dialogic® 4000 Media Gateway Series and OCS Server stands for Microsoft® Office Communications Server (OCS) 2007.



3. Prerequisites

3.1 PBX Prerequisites

The PBX must have all supplemental service packages installed for the Q.SIG protocol to operate properly and to provide all advanced supplemental services.

3.1.1 PBX Equipment Required

To support the E1 Q.SIG configuration as documented, you need a Mitel E1 line card.

3.1.2 PBX Cabling Requirements

The cabling for Q.SIG connections must be CAT5e or better. A standard voice quality cable will not provide desired signal quality and will cause the gateway to have issues establishing a connection on the D-channel.

3.2 Gateway Prerequisites

The gateway needs to support an E1 Q.SIG interface.

4. Summary of Limitations

No limitations noted as of the last update to this document.

5. Gateway Setup Notes

Steps for setting up the gateway are:

1. Configuration of the Dialogic® Diva® Media Board drivers.
2. Configuration of the Diva SIPcontrol software.

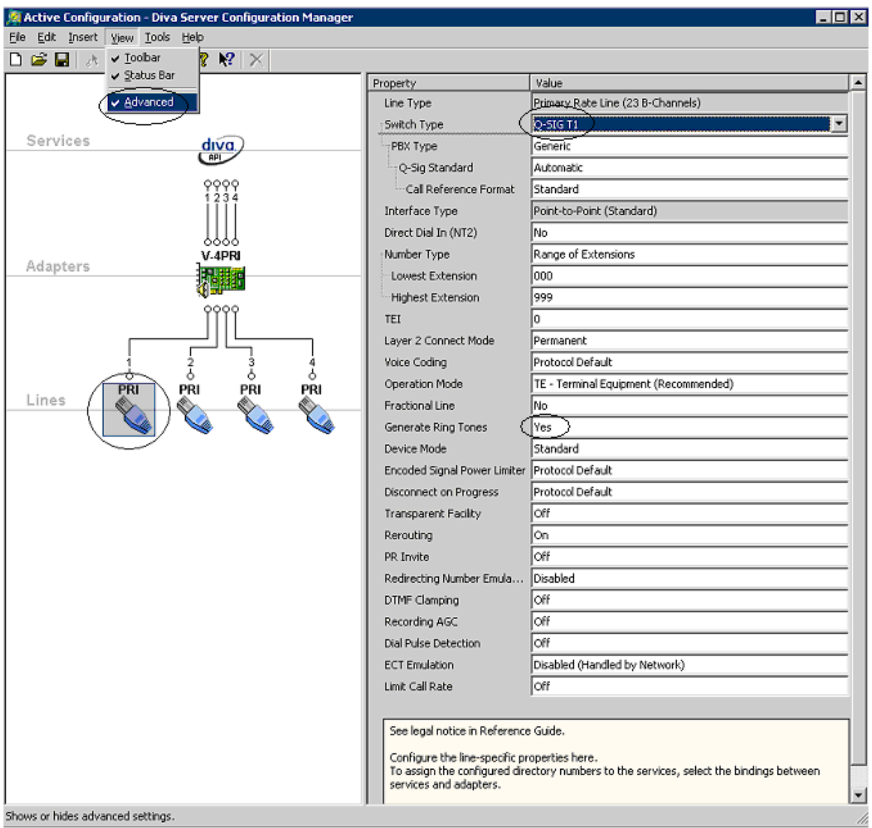
5.1 Dialogic® Diva® Media Board Configuration

The Diva Media Boards are configured in the Dialogic® Diva® Configuration Manager. To open the Configuration Manager, click:

`Start` > `Programs` > `Dialogic Diva > Configuration Manager`.

Note: In the Dialogic® Diva® software and documentation, Diva Media Boards are referred to as Diva Server adapters.

A screen similar to the one below will appear.



Note: The number of TDM circuits varies depending on the used Dialogic® Media Gateway model.

For this setup:

- Set the property `Switch Type` to `Q-SIG T1`.
- If your PBX does not provide ring tones to callers from TDM, set the property `Generate Ring Tones` to `Yes`.

To activate the change, click `File > Activate`.

Make these configuration changes for each TDM circuit you are going to use on the Dialogic® Media Gateway.

5.2 Dialogic® Diva® SIPcontrol™ Software Gateway Application

The Diva SIPcontrol software is configured via the web based interface. To open the web interface, click `Start > Dialogic Diva > SIPcontrol Configuration`.

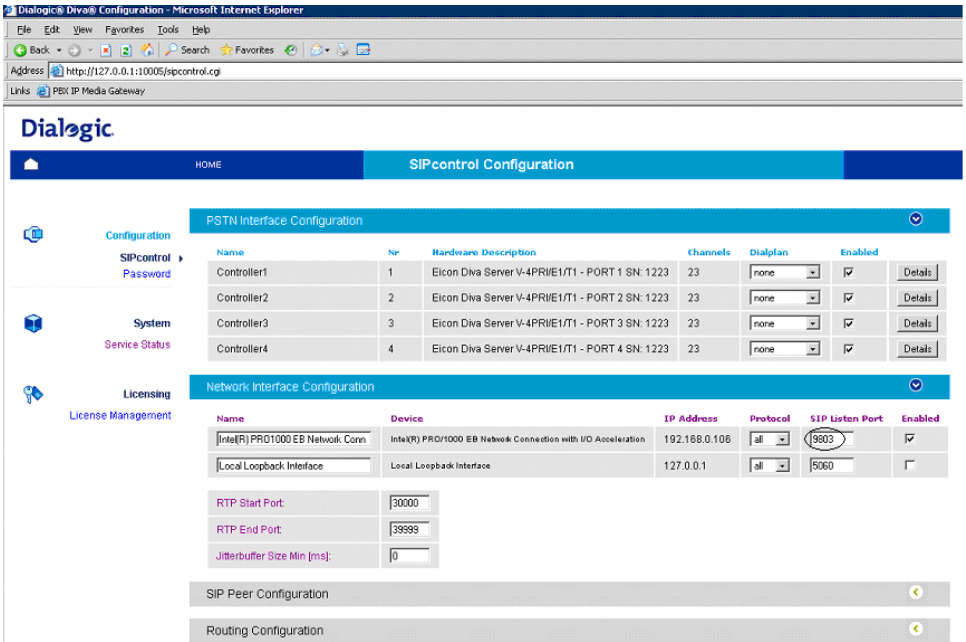
On the main page, click the `SIPControl` link to display the different configuration menus.
The `PSTN Interface Configuration` section should automatically include all ports detected in the system.

Note: If you do not see any detected ports, you may need to add http://127.0.0.1 as a trusted site. From Microsoft® Internet Explorer, click `Tools` > `Internet Options` > `Security` > `Trusted Sites`. Use http://127.0.0.1:10005 to get to the configuration.

In order for the Diva SIPcontrol software to route calls, the proper routes must be created and configured. Each route consists of a source interface and a destination interface. PSTN controllers and SIP peers are considered either a source interface or a destination interface depending on the call direction.

5.2.1 PSTN Interface and Network Interface Configuration

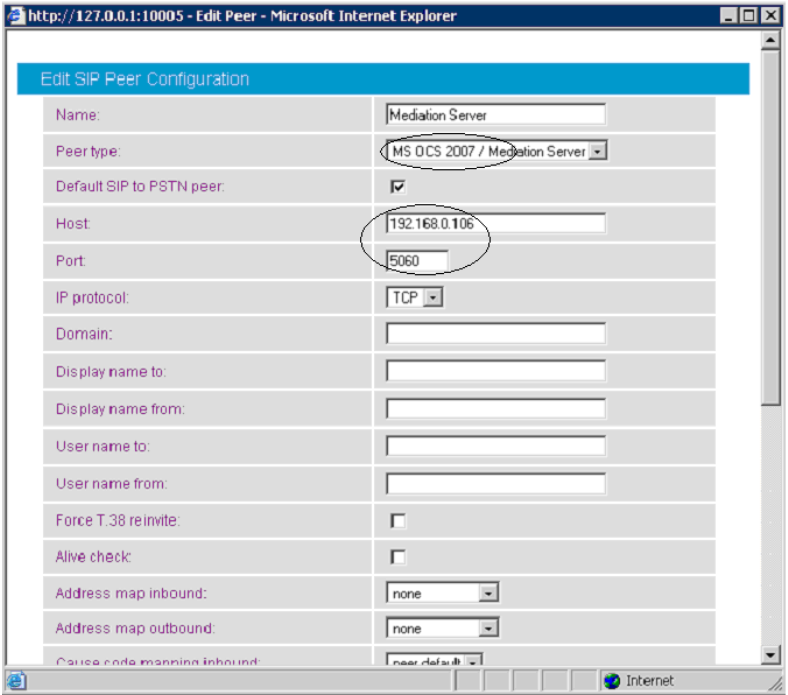
The following is a typical configuration.



The Network Interface Configuration will be used by the Diva SIPcontrol software for listening to the SIP traffic from Microsoft® Mediation Server. Given that on these gateways the Microsoft® Mediation Server component and the Diva SIPcontrol software are running in the same system, you will need to change SIP Listen port to 9803 or to an available un-used port. Later during the Microsoft® Mediation Server configuration, you will need to set the PSTN Gateway next hop setting to 9803 to match.

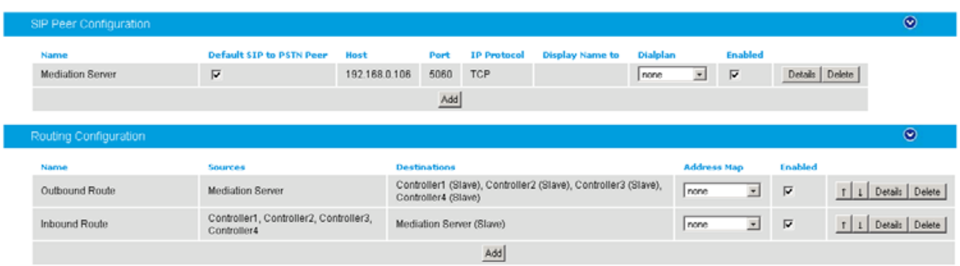
5.2.2 SIP Peer Configuration

Create one SIP peer to talk to Microsoft® Mediation Server as shown below.



5.2.3 Routing Configuration

In the `Routing Configuration` section, you must create two routes, one for the inbound direction (TDM to IP) and one for the outbound direction (IP to TDM). Once you have created the routes, click the `Save` button for the changes to take effect.



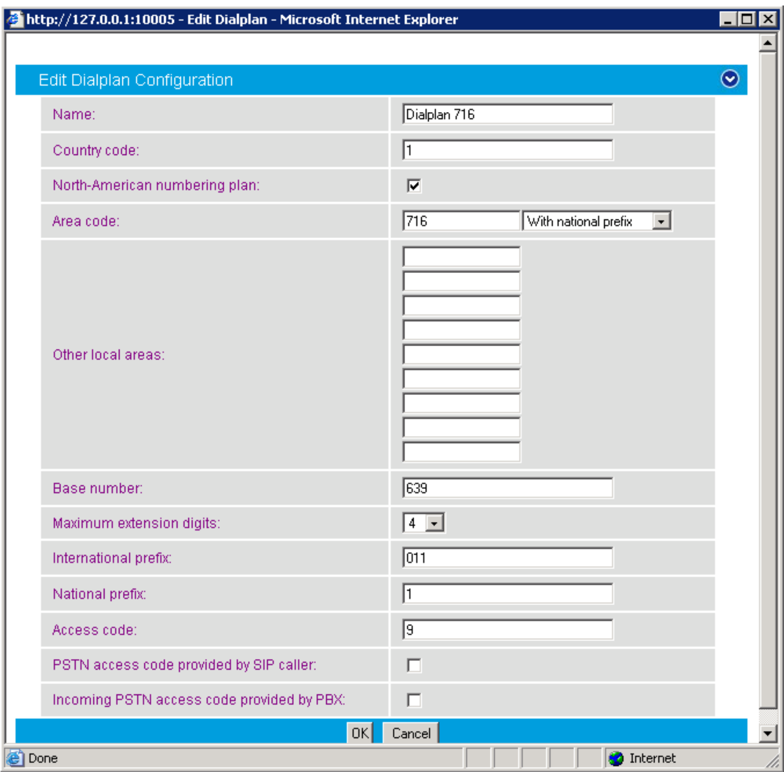
5.2.4 Number Normalization

The `Dialplan Configuration` and `Address Map Configuration` sections are used for manipulating dial numbers. For most PBX dialplans, an address map is required. See the following examples.

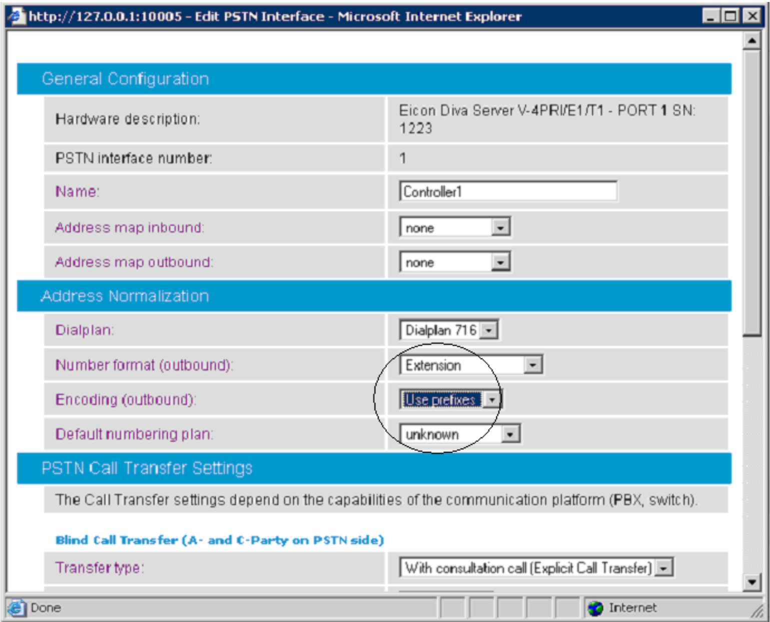
5.2.4.1 Dialplan Configuration Example

To create a dialplan, click `Add` from the `Dialplan Configuration`. The following screens show how to set up a dialplan for a Microsoft® Office Communications Server (OCS) 2007 application with the following dialplan from the PBX. (This may not match to the PBX programming in section 6 and the Setup in section 2.3).

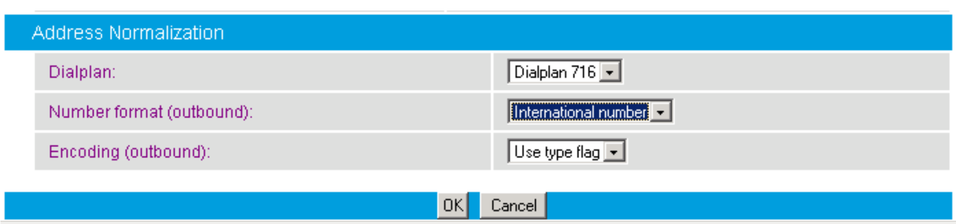
Area code: 716
Base number: 639
Extensions: 4 digits
Access code: 9



Complete the settings and click `OK`.
For the dialplan to be applied to outbound calls, click the `Details` button of the PSTN controller and configure the `Address Normalization` settings as shown in the screen below. This converts the dialed numbers into the format based on the dialplan for the PBX. If the dialed number is for an internal user, it is converted into a 4-digit extension. If the called number is for a national call, 91 is prepended. Click `OK` on this page, and `Save` on the next page for the changes to take effect.



For the dialplan to be applied to inbound calls, click the `Details` button of the configured SIP peer and configure the `Address Normalization` settings as in the screen below. This converts the phone number into the E.164 format as needed by Microsoft® Office Communications Server 2007. Click `OK` on this page, and `Save` on the next page for the changes to take effect.



5.2.4.2 Address Map Configuration Example

If the dialplan does not meet your setups special requirements, the `Address Map Configuration` can be used. An address map entry uses regular expressions (RegEx) (so does Microsoft® Office Communications Server 2007) for converting the call address format for inbound or/and outbound direction.

Important note before applying regular expression rules in address maps: The call address for outbound calls (IP to TDM) includes a "@hostname" part. For example, +17166391234@DMG4000.bufocs.local is the call address, not just +17166391234. For inbound calls (TDM to IP), the call address is the called or calling number, with a possible prefix "+", "N", or "S". For example, an inbound call has called number 1234 with ISDN type of numbering flag set to `Subscriber`*, and the calling number 49715233334444 with ISDN type of numbering flag set to* `International`*. The called address will be S1234 and the calling address will be +49715233334444.*

If the ISDN type of numbering flag is set to `National`, the prefix "N" will be used with the call number. If the type is `Unknown`, no prefix is used.

Outbound call example using address maps:

Microsoft® Office Communications Server 2007 sends the E.164 dial number format to the SIP gateway. Both called and calling numbers need to be converted into a format that the PBX can accept. If the same PBX dialplan as in the previous section is used, the following conversions are needed.

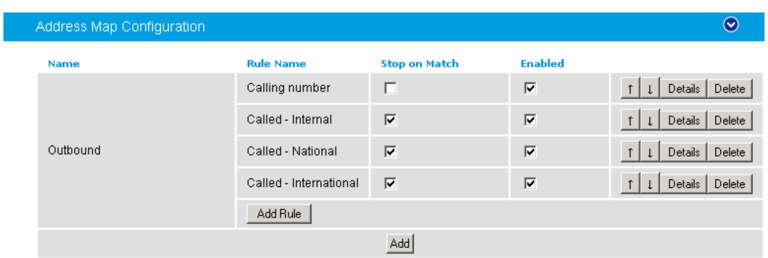
Calling number	From Microsoft® OCS	To PBX
Internal	+1716639xxxx	716639xxxx

Called number	From Microsoft® OCS	To PBX
To Internal	+1716639xxxx	xxxx
To National	+1xxxxxxxxxx	91xxxxxxxxxx
To International	+xxx…xxx	+xxx…xxx

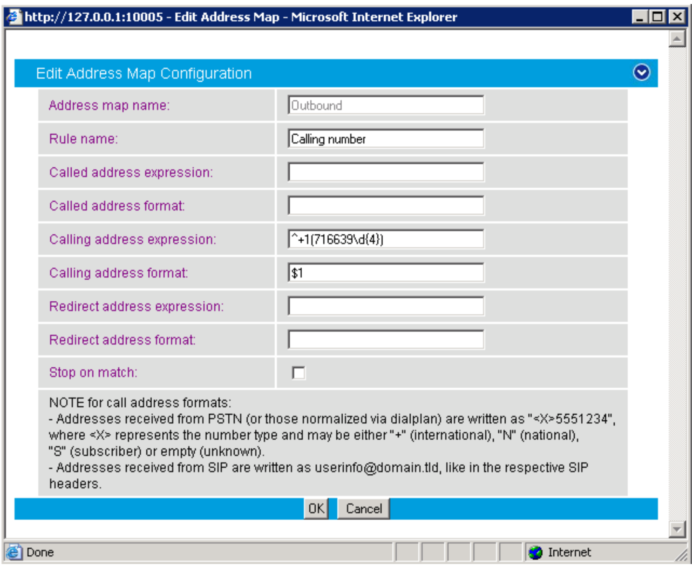
Below is RegEx for the conversion tables above.

Sub rule name	Expression	Format	Stop on match
Calling number	^\+1(716639\d{4})	$1	Not checked
Called - Internal	^\+1716639(\d{4})	$1	Checked
Called - National	^\+1	91	Checked
Called - International	^\+	9011	Checked

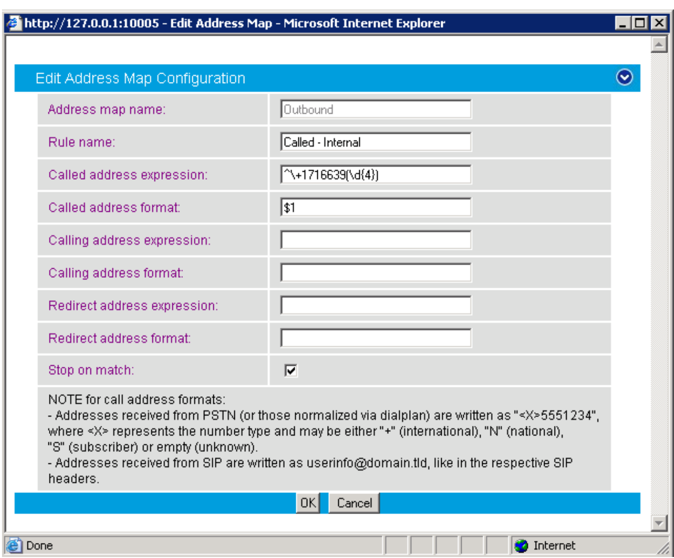
Below are the configured address maps for outbound calls. The order of the below four sub rules and the stop on match check mark are relevant:



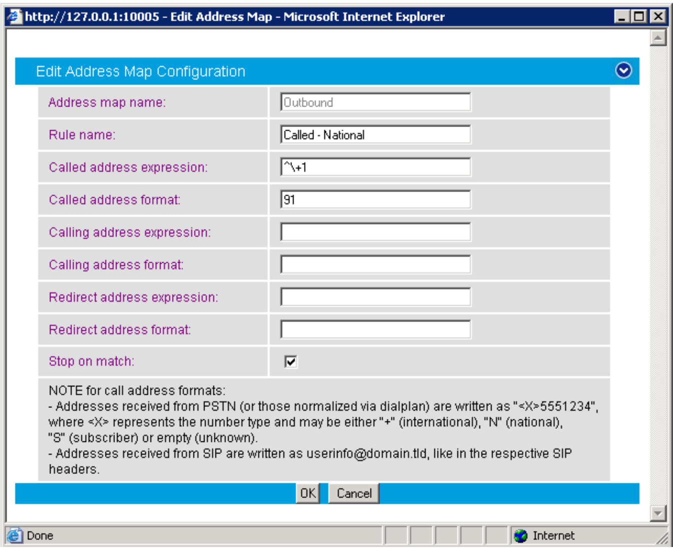
The following screen shows the first sub rule that converts the E.164 calling number into a 10-digit national number:



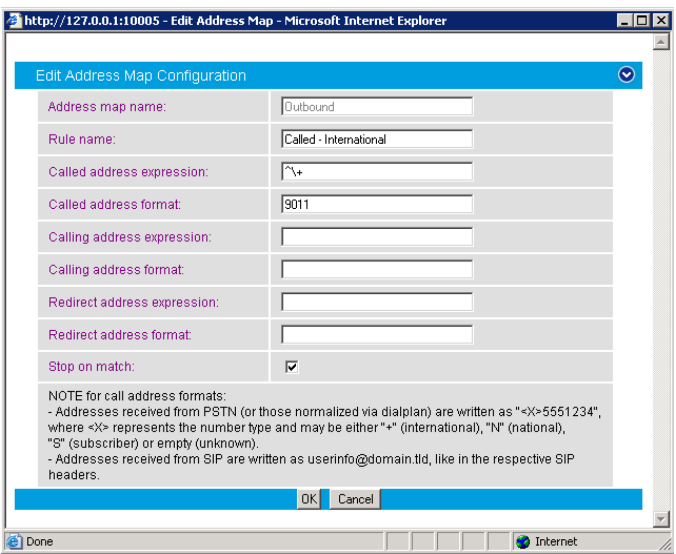
The following screen shows the second sub rule that converts E.164 for the internal called number into a 4-digit extension:



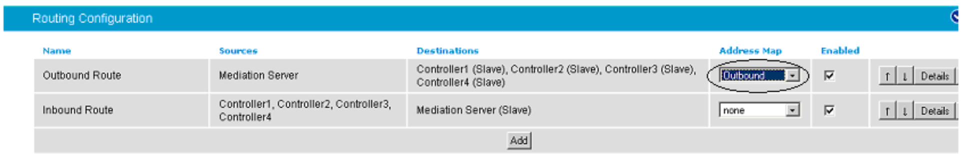
The following sub rule converts the E.164 national number into a 10-digit national number with prefix 91:



The following example converts international call numbers:



Once an address map rule is created, it can be applied in three different places. To ease the configuration and troubleshooting processes, apply the rule on the outbound route as shown below:



Inbound call example using address map:

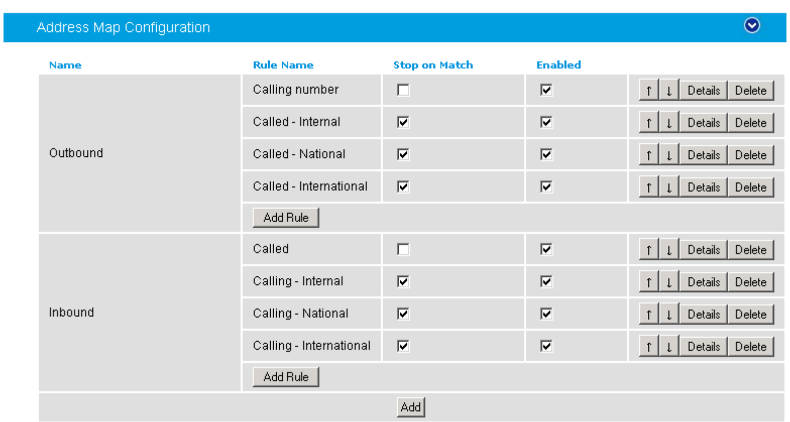
This example assumes that the PBX sends inbound calls using a 4-digit extension, with the ISDN type of number flag set to `Subscriber` for internal numbers, `National` for national calls, and `International` for international calls.

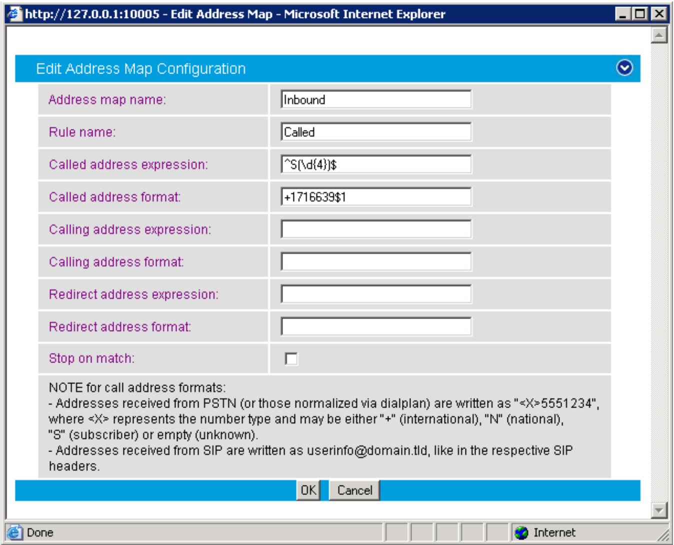
Called number	From PBX	To Microsoft® OCS
Internal	xxxx (with subscriber type of number)	+1716639xxxx

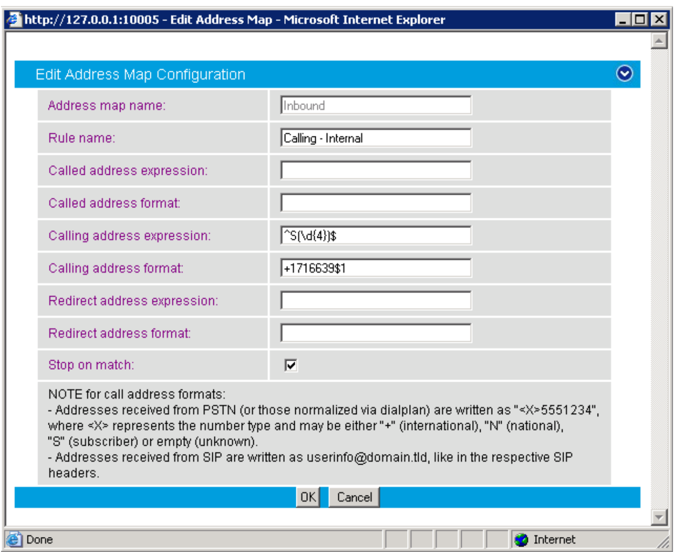
Calling number	From PBX	To Microsoft® OCS
Calling from internal	xxxx (with subscriber type of number)	+1716639xxxx
Calling from national	xxxxxxxxxx (with national type of number)	+1xxxxxxxxxx
Calling from international	xxx…xxx (with international type of number)	+xxx…xxx

Sub rule name	Expression	Format	Stop on match
Called	^S(\d{4})$	+1716639$1	Not checked
Calling - internal	^S(\d{4})$	+1716639$1	Checked
Calling - national	^N(\d{10})$	+1$1	Checked
Calling - international	^\+	+	Checked

Create an address map named `Inbound` and its four sub rules as shown below:




http://127.0.0.1:10005 - Edit Address Map - Microsoft Internet Explorer
Edit Address Map Configuration
Address map name:
Inbound
Rule name:
Called
Called address expression:
^S(\d{4})$
Called address format:
+1716639$1
Calling address expression:
Calling address format:
Redirect address expression:
Redirect address format:
Stop on match:
NOTE for call address formats:
- Addresses received from PSTN (or those normalized via dialplan) are written as "<X>5551234",
where <X> represents the number type and may be either "+" (international), "N" (national),
"S" (subscriber) or empty (unknown).
- Addresses received from SIP are written as userinfo@domain.tld, like in the respective SIP
headers.
OK
Cancel
Done
Internet


http://127.0.0.1:10005 - Edit Address Map - Microsoft Internet Explorer
Edit Address Map Configuration
Address map name:
Inbound
Rule name:
Calling - Internal
Called address expression:
Called address format:
Calling address expression:
^S(\d{4})$
Calling address format:
+1716639$1
Redirect address expression:
Redirect address format:
Stop on match:
NOTE for call address formats:
- Addresses received from PSTN (or those normalized via dialplan) are written as "<X>5551234",
where <X> represents the number type and may be either "+" (international), "N" (national),
"S" (subscriber) or empty (unknown).
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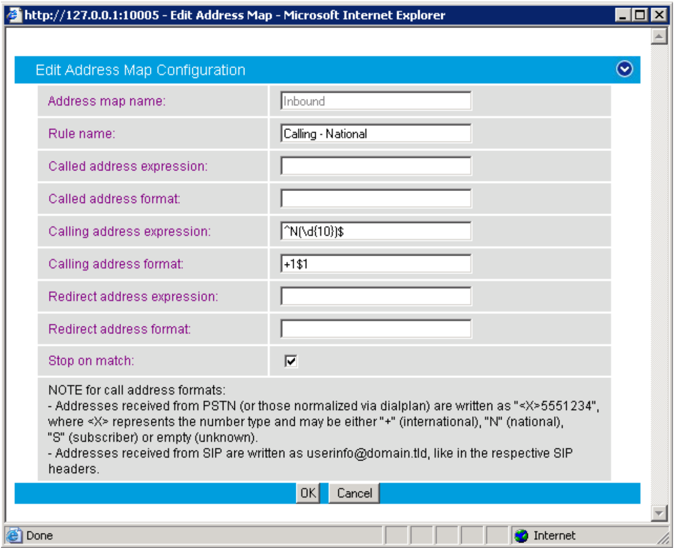
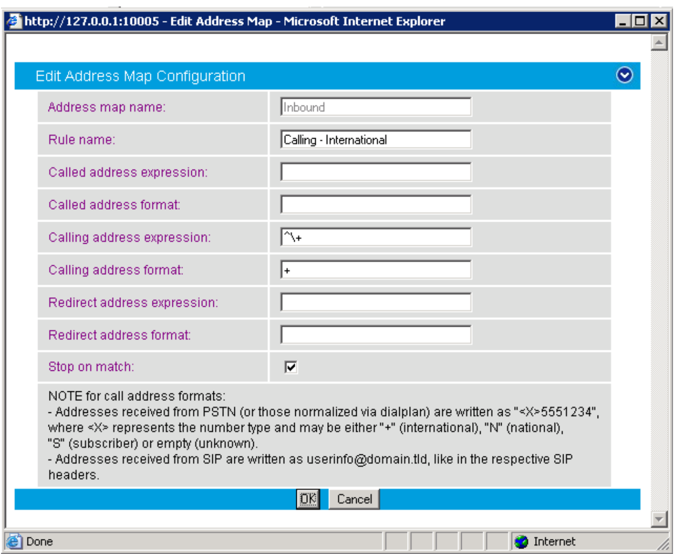
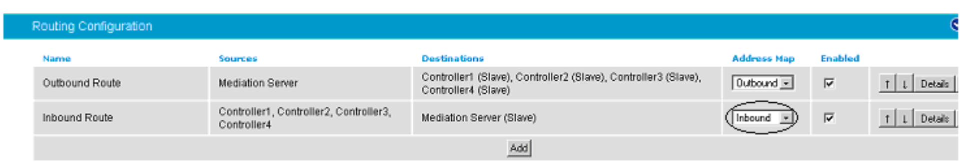
headers.
OK
Cancel
Done
Internet





Apply the address map inbound rule on the inbound route as follows:



5.2.5 Restarting the Dialogic® Diva® SIPcontrol™ Software

Note: A restart of the Diva SIPcontrol software service is needed only if the setting under `Network Interface` *is changed.*

Save the configuration and restart the Diva SIPcontrol software service for the changes to take effect. To do so, click `Service Status` on the left hand side of the main configuration page, and then click `Restart SIPcontrol`. The `Last operation log` will show that the service has been stopped and started again.

6. PBX Setup Notes

6.1 Mitel 3300 ICP

This section provides information about the PBX administration requirements for Mitel 3300 ICP systems. This information includes:

1. Configure the E1 Interface Administration.
2. Configure the Mitel 3300 ICP System Controller.
3. Program the Automatic Route Selection.
4. Configure the E1 protocol on NSU.

Notes:

- *This document was developed using the Mitel 3300 ICP system and the following software versions: ICP Release Software Version 5.1.4.8 and IMAT Version 7.5.1.4. Other software versions may not support certain features specified in this section.*

- *The installation of the E1 board must be validated prior to this configuration. Sometimes this can only be done by the switch vendor depending on the administration access capability.*

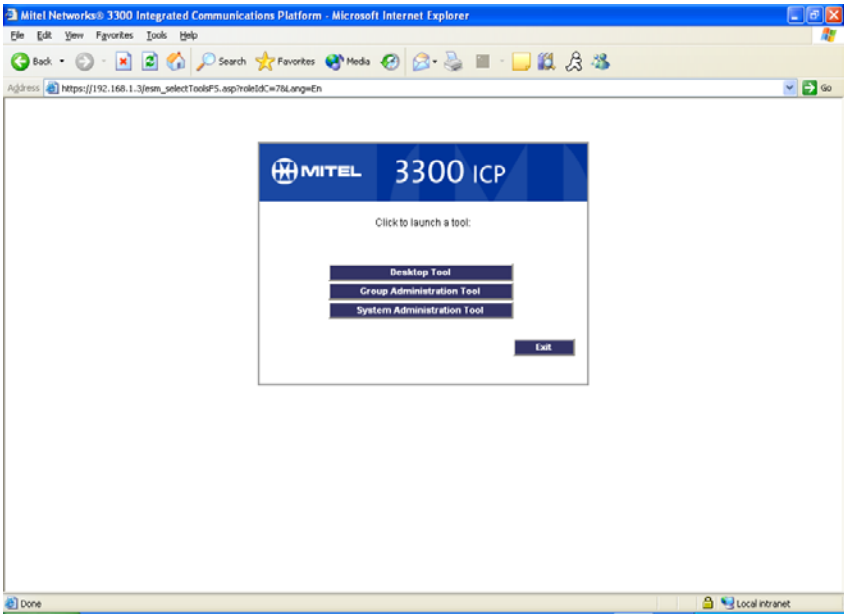
6.1.1 E1 Interface Administration

There are three types of E1 Interface Administration required to properly program the Q.SIG integration on the Mitel 3300 ICP system for the Dialogic® 4000 Media Gateway Series:

1. 3300 ICP Administration,
2. Automatic Route Selection (ARS), and
3. NSU Configurations / IMAT Programming.

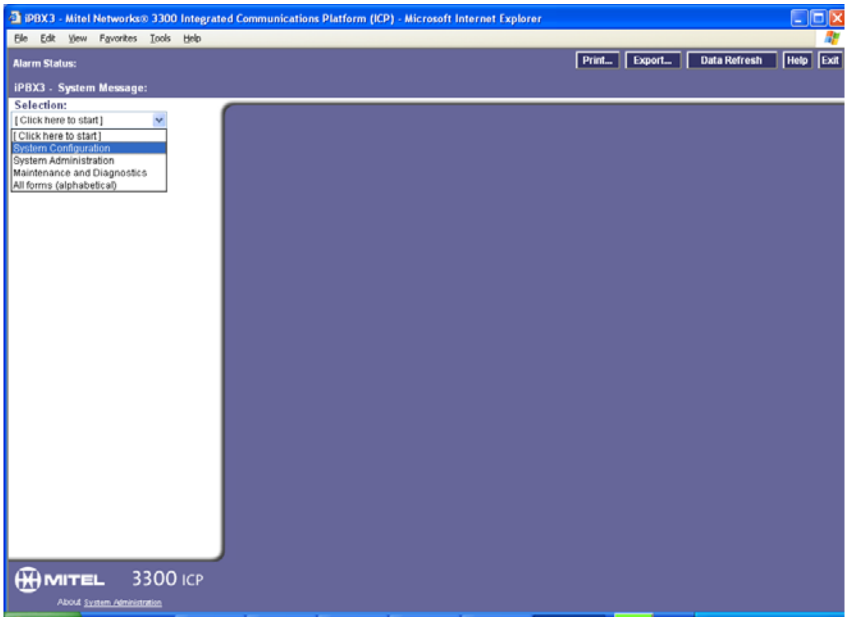
6.1.1.1 3300 ICP Administration

Step 1: Log into the Mitel 3300 ICP system.

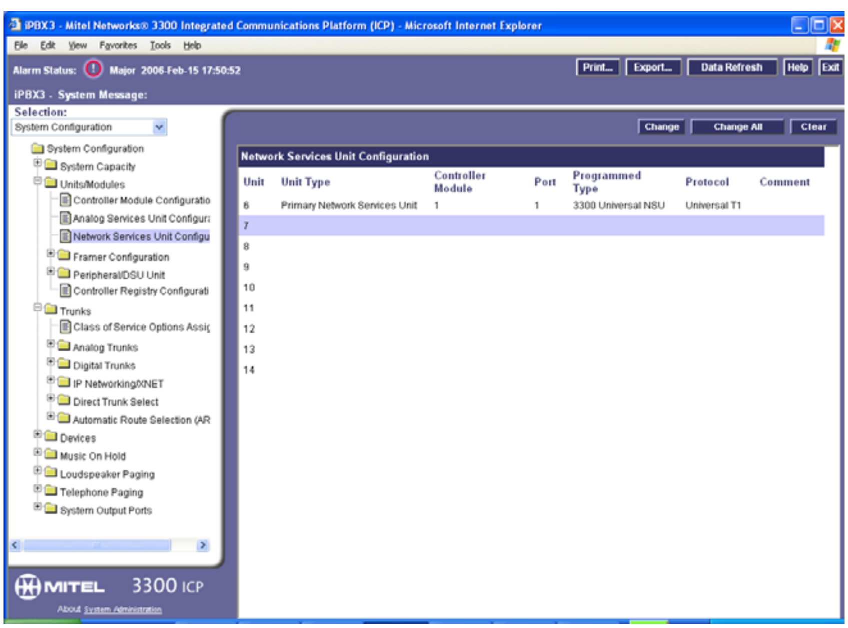


Select the `System Administration Tool` option.

Step 2: Select `System Configuration` from the `Selection` drop-down menu to open the Systems Configuration browser.

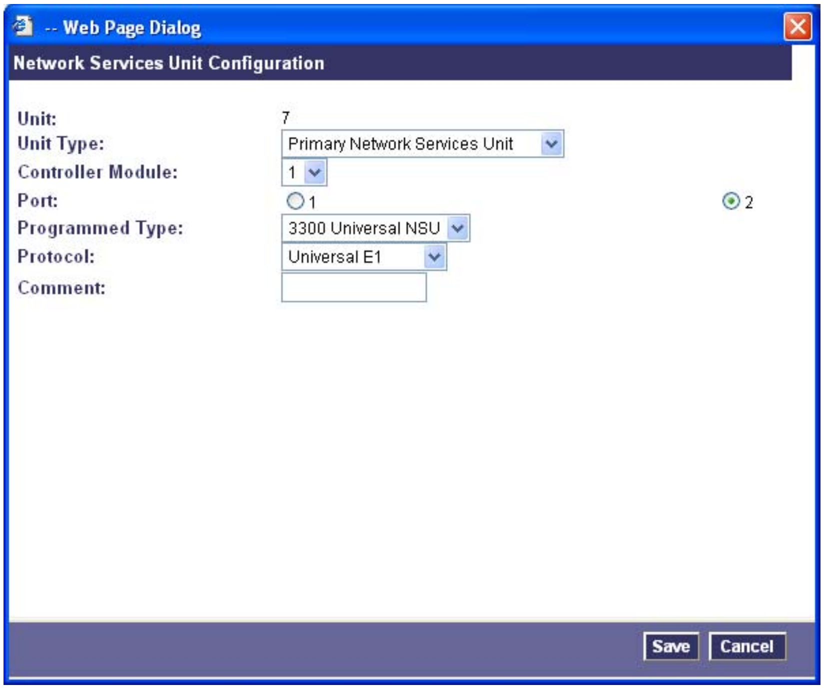


Step 3: In the System Configuration browser, expand `Units/Modules` and select `Network Services Unit Configuration`.



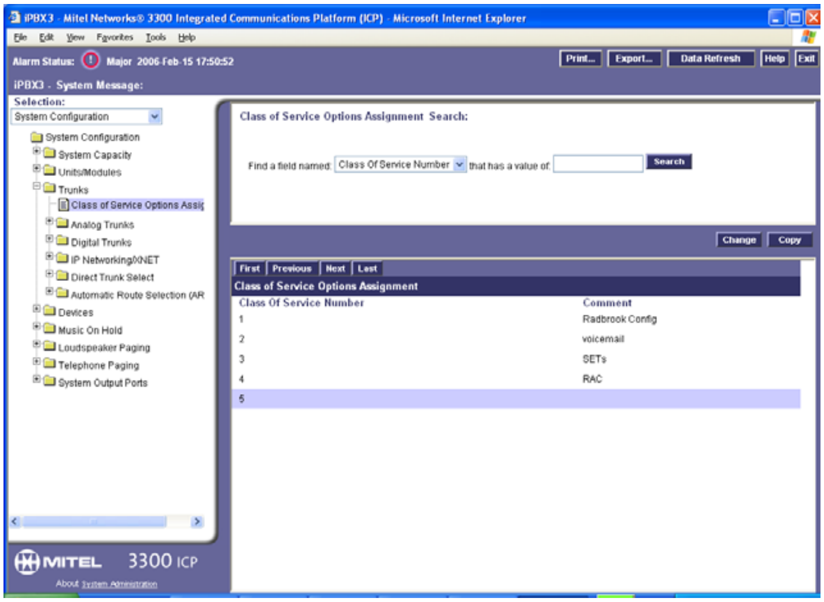
Select an unused `Unit Number` and click `Change`.

Step 4: Configure the `Network Services Unit Configuration` as follows:



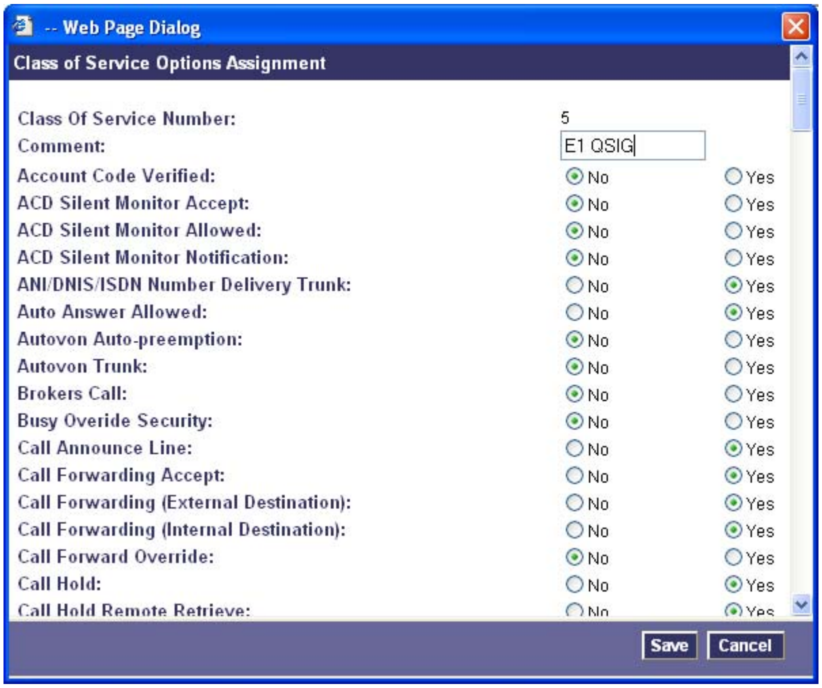
- Set `Unit Type` to `Primary Network Services Unit`.
- Set `Controller Module` to the Controller Module Number.
- Set `Port` to the port number where the NSU is installed.
- Set `Programmed Type` to `3300 Universal NSU`.
- Set `Protocol` to `Universal E1`.
- Click `Save`.

Step 5: In the System Configuration browser, expand `Trunks` and select `Class of Service Options Assignment`.



Select an unused `Class of Service Number` and click `Change`.

Step 6: Configure the `Class of Service Options Assignment` as follows:



Note: Parameters not listed below can be left at their default settings.

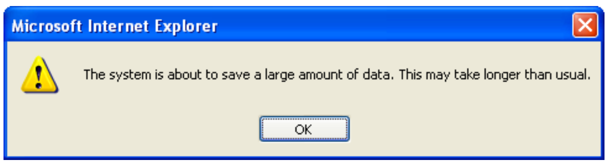
Set the following parameters to `Yes`:

- `ANI/DNIS/ISDN Number Delivery Trunk`
- `Call Announce Line`
- `Call Forwarding (External Destination)`
- `Call Hold – Retrieve with Hold Key`
- `Call Reroute after CFFM to busy Destination`
- `Display ANI/DNIS/ISDN Calling/Called Number`
- `Display DNIS/Called Number Before Digit Modification`
- `Display Dialed Digits During Outgoing Calls`
- `Display Held Call ID on Transfer`
- `Message Waiting – Audible Tone Notification`
- `ONS CLASS/CLIP: Message Waiting Activate/Deactivate`
- `ONS CLASS/CLIP: Set`
- `Public Network Access via DPNSS`
- `Public Network Identity Provided`
- `Public Network to Public Network Connection Allowed`
- `Public Trunk`
- `R2 Call Progress Tone`

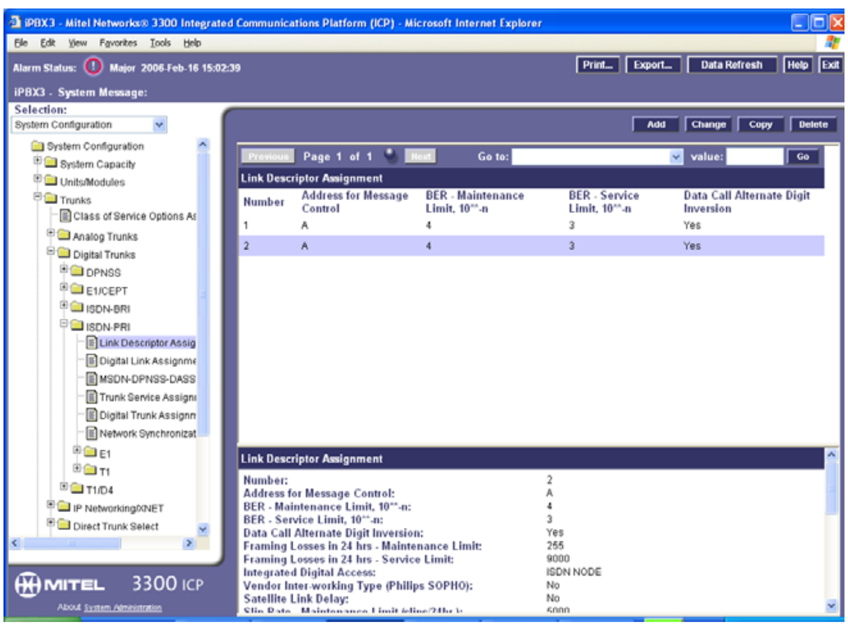
- `Third Party Call Forward Follow Me – Accept`
- `Third Party Call Forward Follow Me – Allow`
- `Trunk Flash Allowed`

Step 7: Click `Save`.

Step 8: Click `OK` in the following prompt:

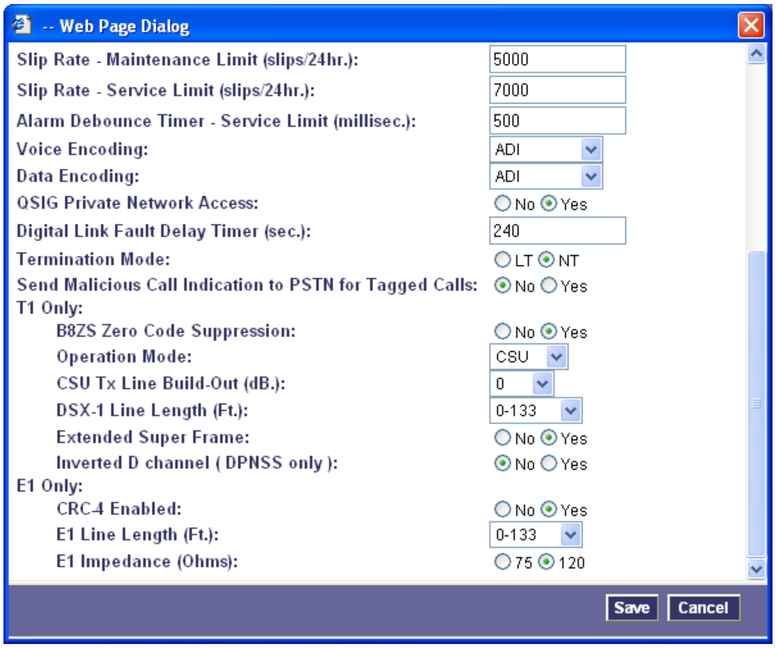


Step 9: In the System Configuration browser, expand `Digital Trunks`, then `ISDN-PRI`, and select `Link Descriptor Assignment`.



Click `Add` to create a new entry.

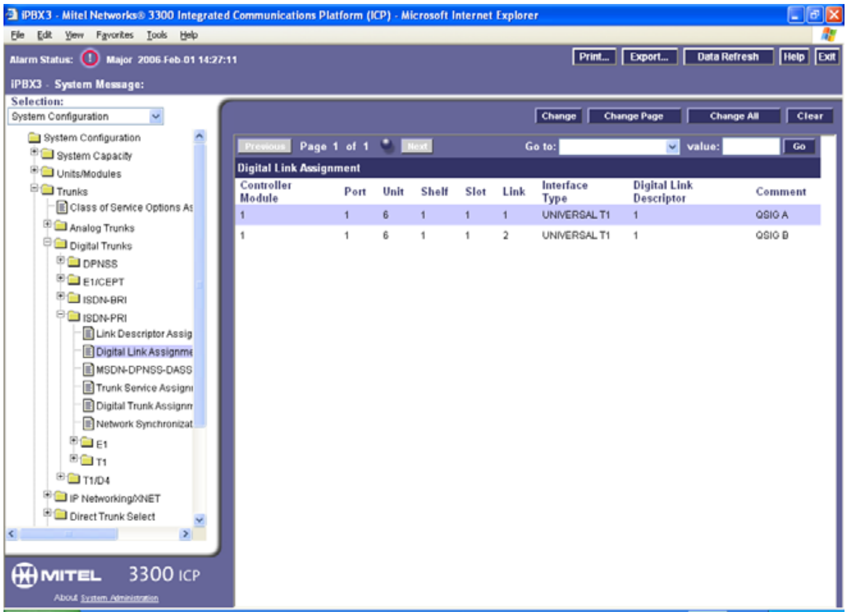
Step 10: Configure the `Link Descriptor Assignment` as follows:



Note: Parameters not listed below can be left at their default settings.

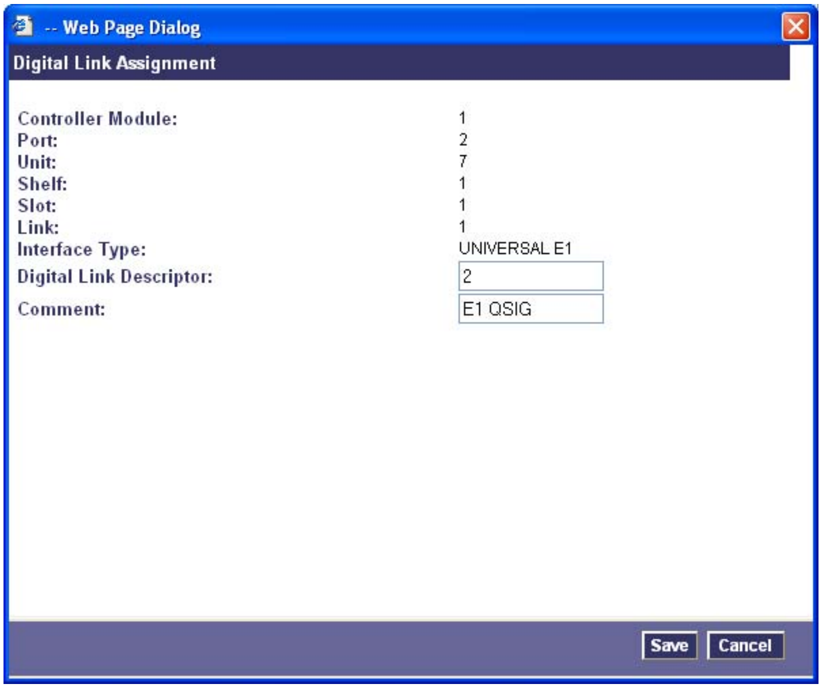
- Set `Number Field` to an unused link descriptor assignment number.
- Set `Address for Message Control` to `A`.
- Set `Integrated Digital Access` to `ISDN Node`.
- Set `Voice Encoding` and `Data Encoding` to `ADI`.
- Set `QSIG Private Network Access` to `Yes`.
- Set `E1 Only: CRC-4 Enabled` to `Yes`.
- Set `E1 Only: E1 Line Length (Ft.)` to `0-133`.
- Set `E1 Only: E1 Impednace (Ohms)` to `120`.
- Click `Save`.

Step 11: In the System Configuration browser, select `Digital Link Assignment`.



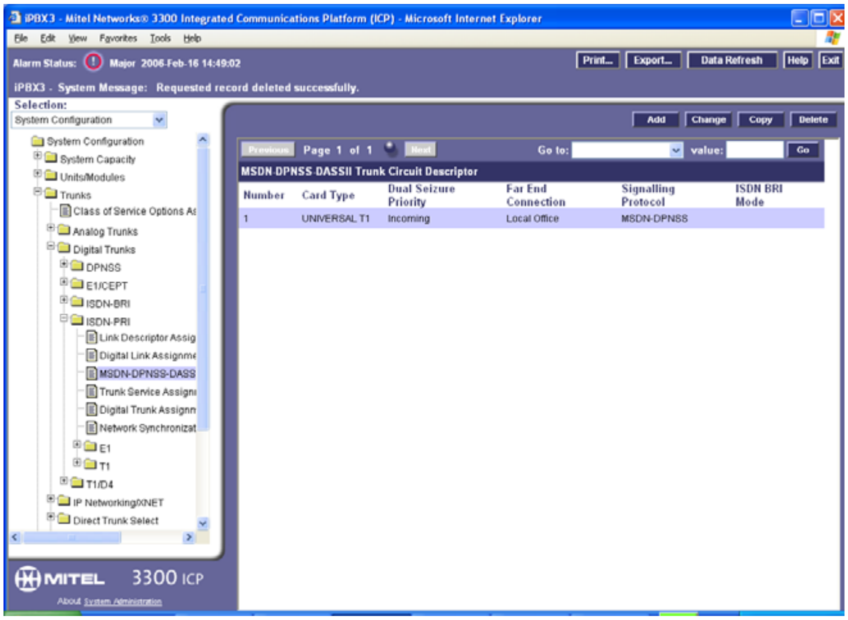
Select the link on the NSU on which the E1 Q.SIG line will be installed and click `Change`.

Step 12: Configure the `Digital Link Assignment` as follows:



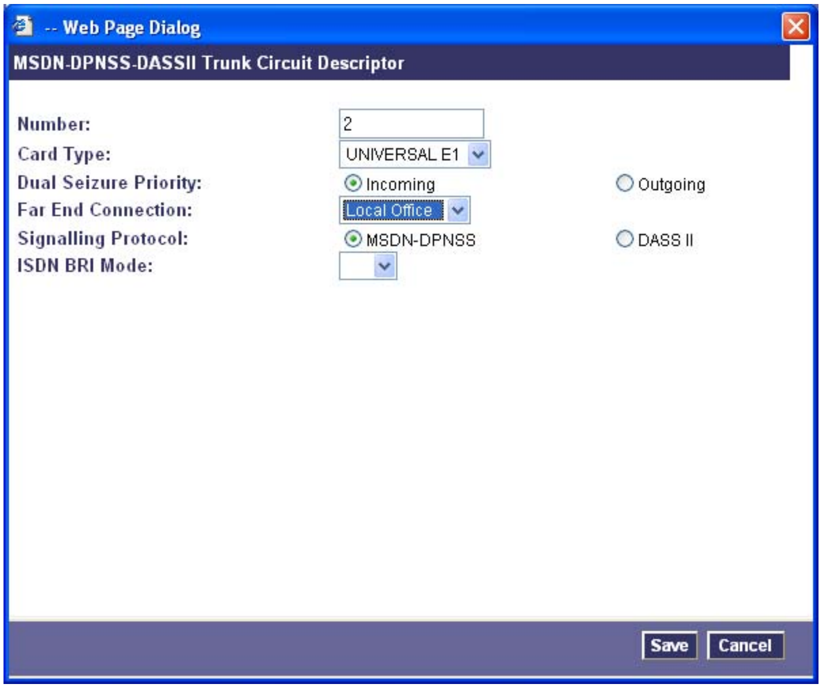
- Set `Digital Link Descriptor` to the link descriptor assignment number defined in Step 10 of the 3300 ICP Administration.
- The `Comment` field can be left blank, or a description of the link can be entered.
- Click `Save`.

Step 13: In the System Configuration browser, select `MSDN-DPNSS-DASSII Trunk Circuit Descriptor`.



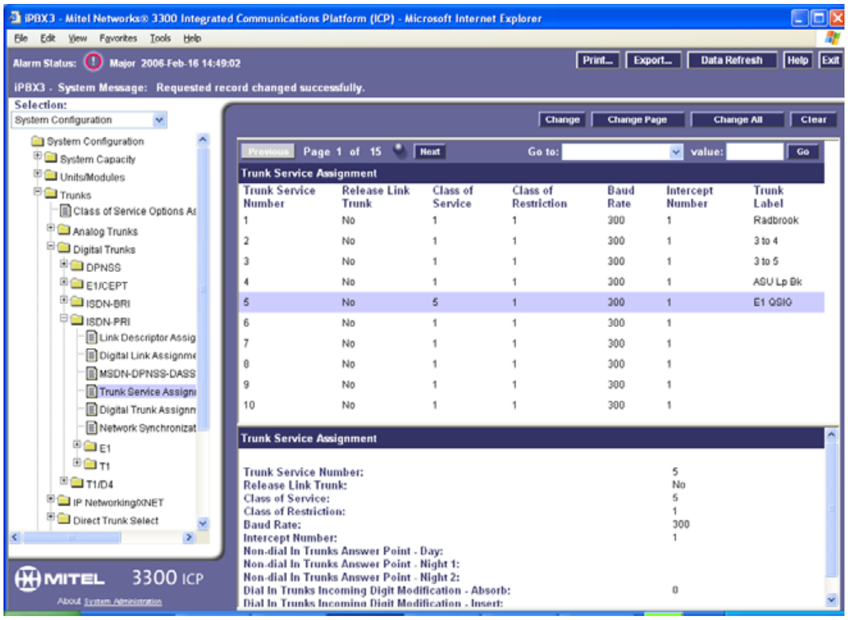
Click `Add` to create a new entry.

Step 14: Configure the `MSDN-DPNSS-DASSII Trunk Circuit Descriptor` as follows:



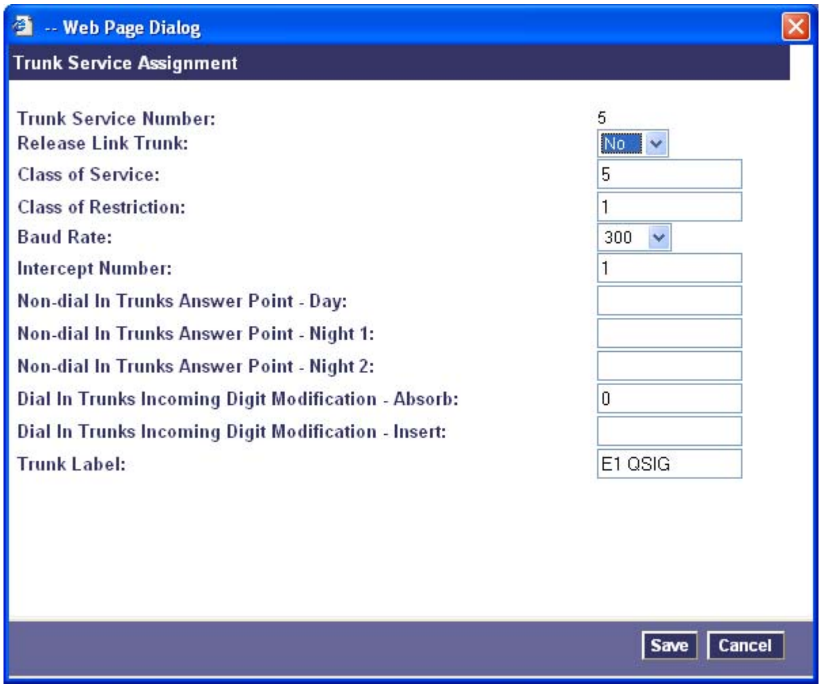
- Set `Number` to an unused trunk circuit descriptor number.
- Set `Card Type` to `Universal E1`.
- Set `Dual Seizure Priority` to `Incoming`.
- Set `Far End Connection` to `Local Office`.
- Set `Signaling Protocol` to `MSDN-DPNSS`.
- Set `ISDN BRI Mode` to `<blank entry>`.
- Click `Save`.

Step 15: In the System Configuration browser, select `Trunk Service Assignment`.



Select an unused `Trunk Service Number` and then click `Change`.

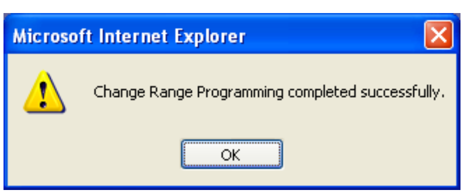
Step 16: Configure the `Trunk Service Assignment` as follows:



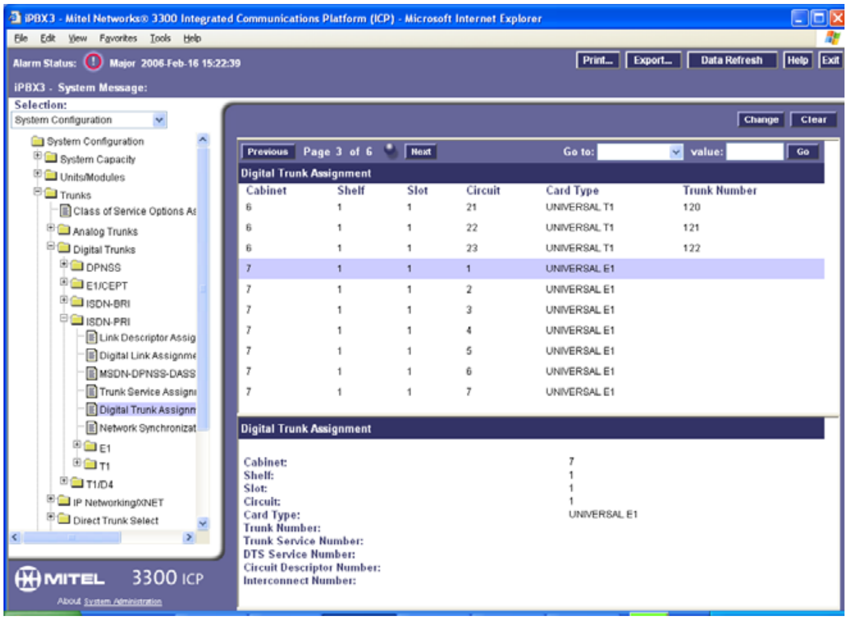
Note: Parameters not listed below can be left at their default settings.

- Set `Release Link Trunk` to `No`.
- Set `Class of Service` to the class of service number defined in Step 9 of the 3300 ICP Administration.
- Set `Class of Restriction` to `1`.
- Set `Baud Rate` to `300`.
- Set `Interceptor Number` to `1`.
- Set `Dial In Trunks Incoming Digit Modification – Absorb` to `0`.
- Click `Save`.

Step 17: Click `OK` in the prompt stating the programming is completed.

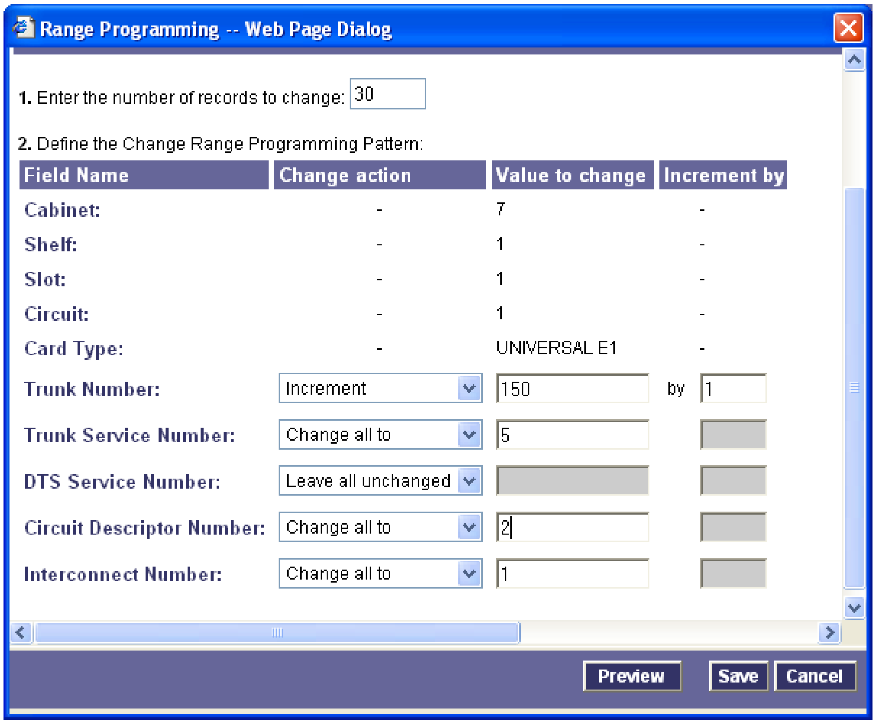


Step 18: In the System Configuration browser, select `Digital Trunk Assignment`.



Select the line, where the `Circuit` number is 1, and where the `Cabinet` number, the `Shelf` number, and the `Slot` number are the numbers in which the Q.SIG line is being installed, and click `Change`.

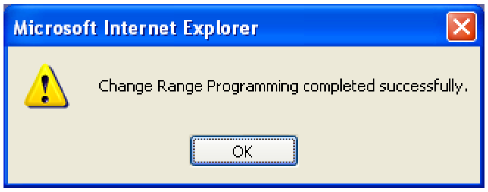
Step 19: Configure the `Digital Trunk Assignment` as follows:



Note: Parameters not listed below can be left at their default settings.

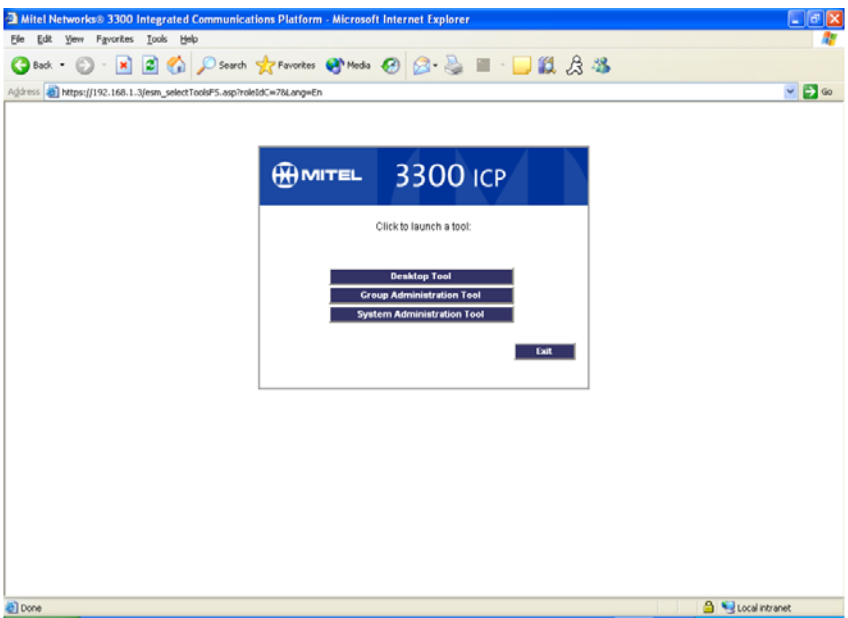
- Set `Enter the number of records to change` to `30`.
- Set `Trunk Number:`
  - under `Change action` to `Increment`.
  - under `Value to change` to an unused trunk number.
  - under `Increment by` to `1`.
- Set `Trunk Service Number:`
  - under `Change action` to `Change all to`.
  - under `Value to change` to the trunk service number defined in Step 16 of the 3300 ICP Administration.
- Set `DTS Service Number:` under `Change action` to `Leave all unchanged`.
- Set `Circuit Descriptor Number:`
  - under `Change action` to `Change all to`.
  - under `Value to change` to the circuit descriptor number defined in Step 14 of the 3300 ICP Administration.
- Set `Interconnect Number:`
  - under `Change action` to `Change all to`.
  - under `Value to change` to a valid inter-connect number.
- Click `Save`.

Step 20: Click `OK` on the prompt stating the programming is completed successfully.



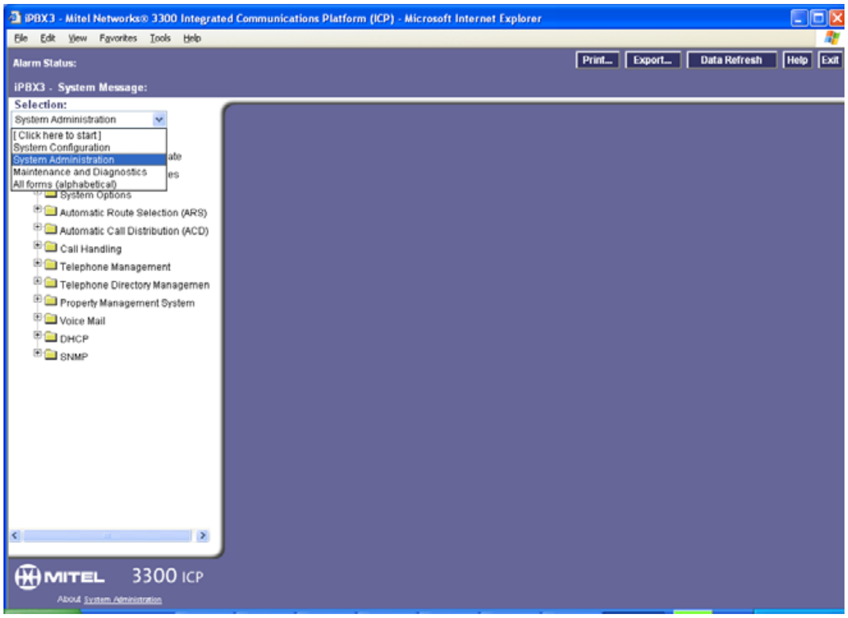
6.1.1.2 Automatic Route Selection (ARS)

Step 1: Log into the Mitel 3300 ICP system.

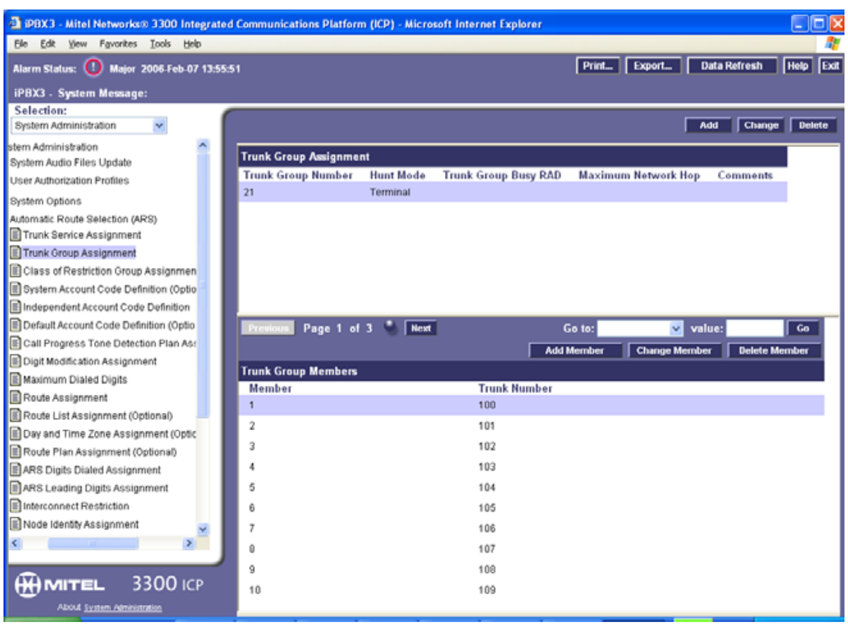


Click the `System Administration Tool` option.

Step 2: To open the System Administration browser, select `System Administration` from the `Selection` drop-down menu

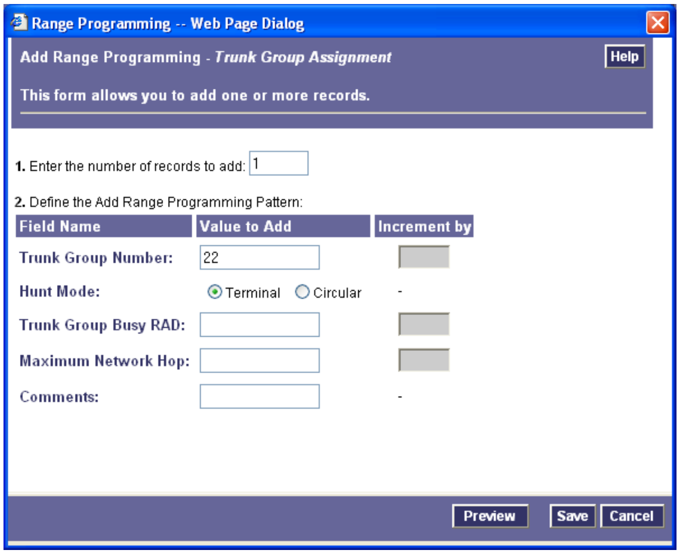


Step 3: In the System Administration browser, expand `Automatic Route Selection (ARS)` and select `Trunk Group Assignment`.



Click `Add` to create a new entry.

Step 4: Configure the `Trunk Group Assignment` as follows:

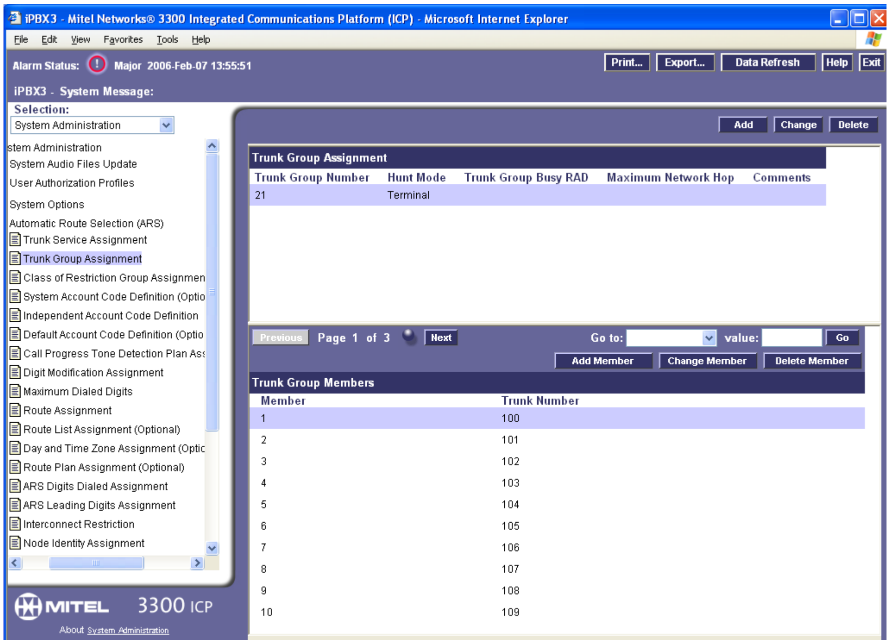


Note: Parameters not listed below can be left at their default settings.

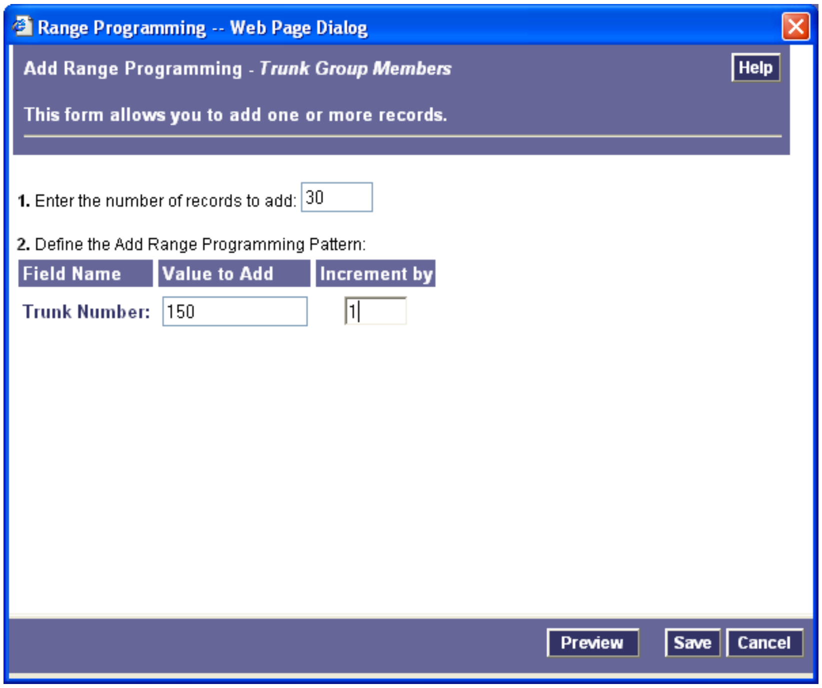
- Set `Enter the number of records to change` to `1`.
- Set `Trunk Group Number:` under `Value to Add` to an unused trunk group number.
- Set `Hunt Mode` to `Terminal`.
- Click `Save`.

Step 5: Click `OK` in the prompt stating that the new entry was created successfully.

Step 6: Under `Trunk Group Number`, select the trunk group number just created in step 4. Click `Add Member` to create a new member.

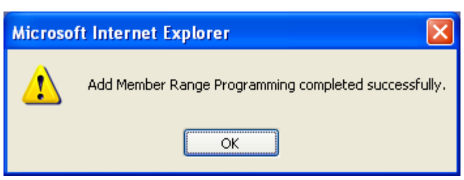


Step 7: Configure the new member `Trunk Group Assignment` as follows:

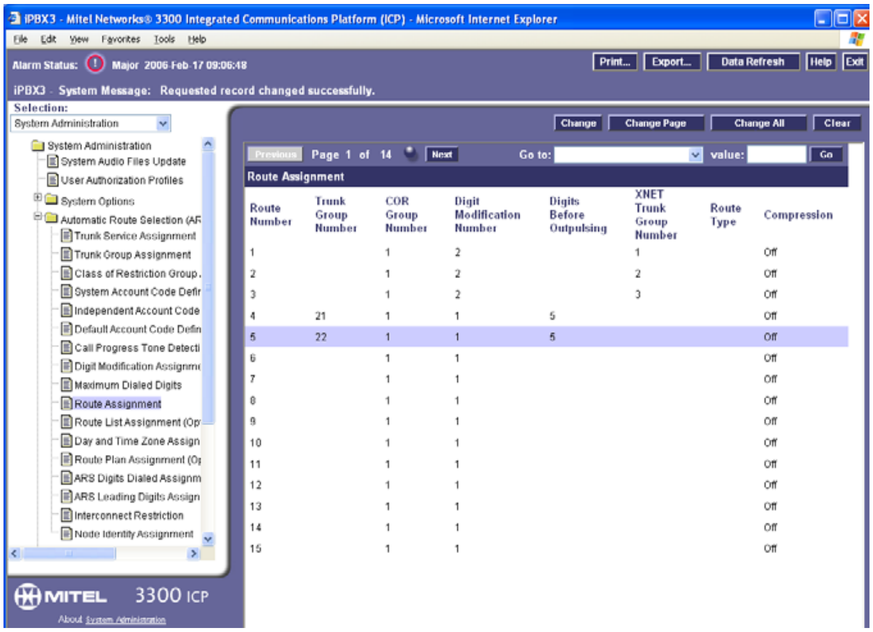


- Set `Enter the number of records to add` to `30`.
- Set `Trunk Number` under `Value to Add` to the number of the first trunk defined in Step 17 of the 3300 ICP Administration.
- Set `Trunk Number` under `Increment by` to `1`.
- Click `Save`.

Step 8: Click `OK` In the prompt stating that the programming was completed successfully.

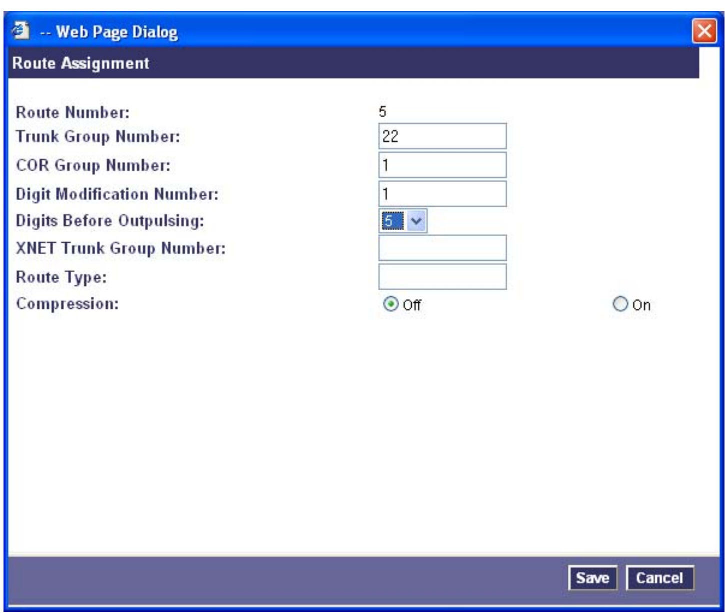


Step 9: In the System Administration browser, select `Route Assignment`.



Under `Route Number`, select an unused route number and click `Change`.

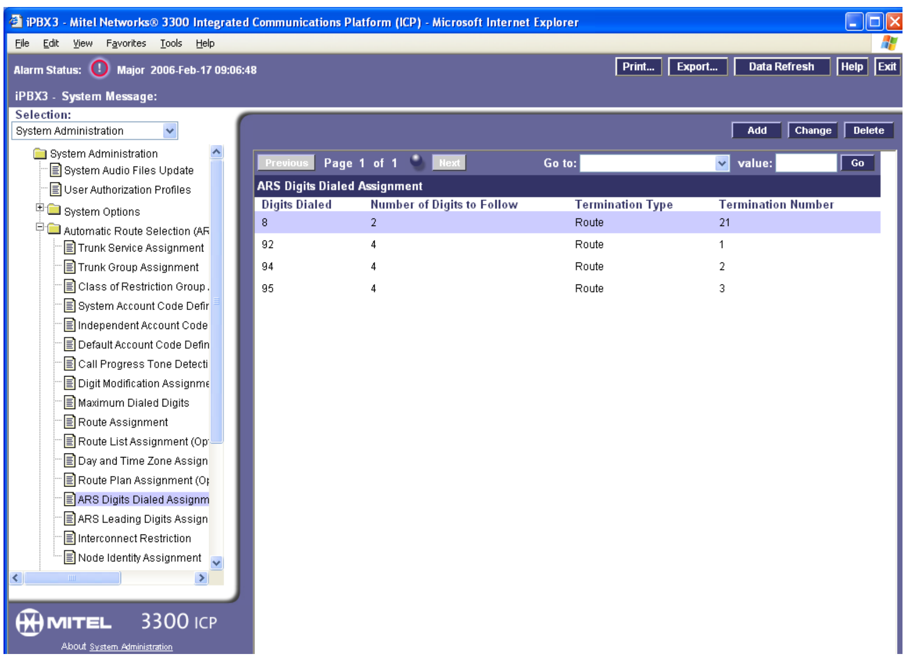
Step 10: Configure the `Route Assignment` as follows:



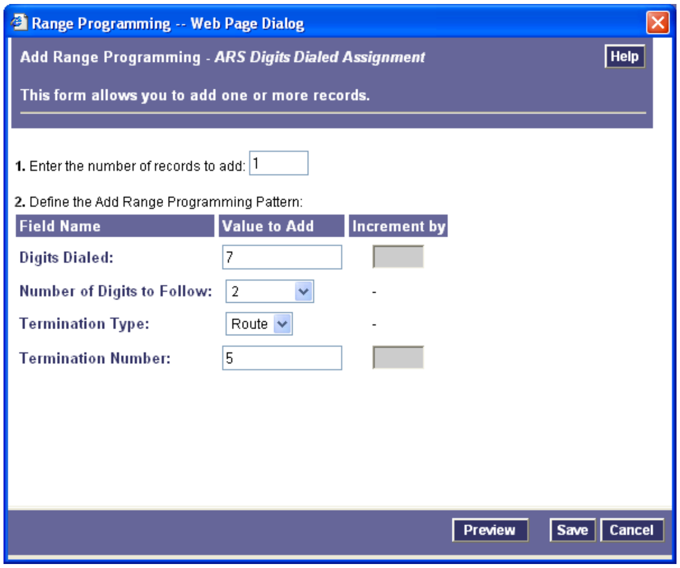
Note: Parameters not listed below can be left at their default settings.

- Set `Trunk Group Number` to an unused trunk group number defined in Step 4 of the Automatic Route Selection.
- Set `COR Group Number` to a valid class of restriction number.
- Set `Digit Modification Number` to `1`.
- Set `Digits Before Outpulsing` to `5`.
- Set `Compression` to `Off`.
- Click `Save`.

Step 11: In the System Administration browser, select `ARS Digits Dialed Assignment`. Click `Add` to create a new entry.



Step 12: Configure the `ARS Digits Dialed Assignment` as follows:

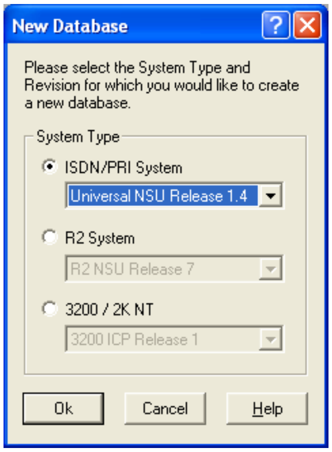


- Set `Enter the number of records to add` to `1`.
- Under `Value to Add` set:
  - `Digits Dialed` to the leading digit(s) to be assigned.
  - `Number of Digits to Follow` to `2`.
  - `Termination Type` to `Route`.
  - `Termination Number` to the route number that was defined in Step 10 of the Automatic Route Selection.
- Click `Save`.

Step 13: Click `OK` in the prompt stating that new entry was added successfully.

6.1.1.3 NSU Configuration/IMAT Programming

Step 1: Open up IMAT and click `File` > `New Database` to create a new database:

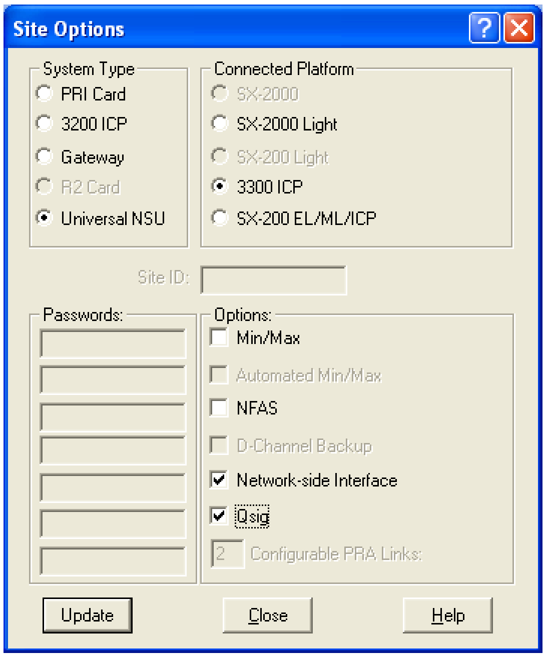


Select the appropriate version of the database from the `ISDN/PRI System` dropdown menu based on the criteria in the table below. For example, for ICP software version 5.1.4.8, the `Universal NSU Release 1.4` needs to be selected

Note: IMAT Version 7.5.1.4 is database version backward compatible.

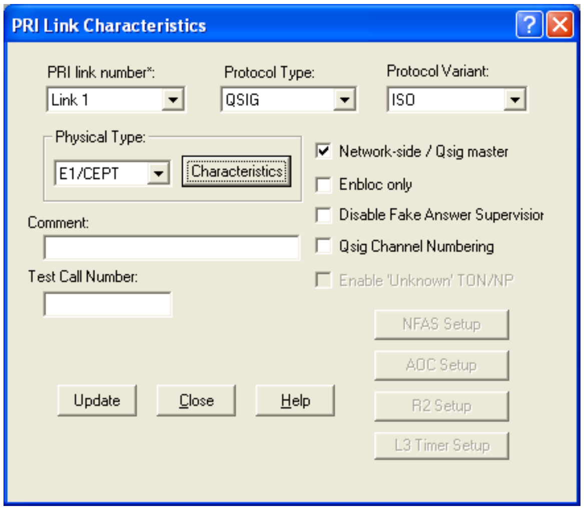
ICP Software Version	IMAT Database Version
Release 3.0 – Release 3.3	PRI 8.0 or NSU 1.1
Release 4.0	PRI 8.1 or NSU 1.2
Release 4.1 – Release 5.0	PRI 8.2 or NSU 1.3
Release 5.1 NSU	NSU 1.4
Release 5.1 PRI	PRI 8.2 or NSU 1.3
Release 5.2 NSU	NSU 1.5
Release 6.0 NSU	NSU 1.5
Release 6.1 NSU	NSU 1.5
Release 6.1 PRI	PRI 8.2 or NSU 1.3

Step 2: Click `Config` > `System Configuration` > `Site Options` to configure the `Site Options`:



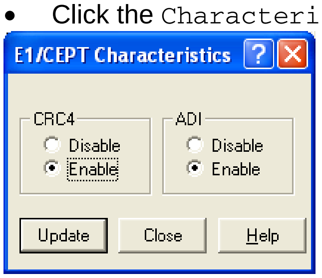
- Under `System Type`, select `Universal NSU`.
- Under `Connected Platform`, select `3300 ICP`.
- Under `Options`, select `Network-side Interface` and `Qsig`.
- Click `Update` and then `Close`.

Step 3: Click `Config` > `System Configuration` > `PRI Link Characteristics` to configure the `PRI Link Characteristics`:



Note: Parameters not listed below can be left at their default settings.

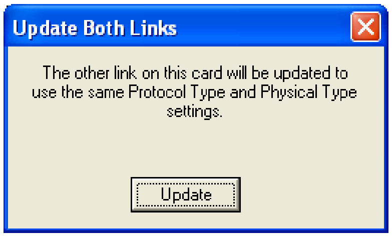
- Set `PRI Link Number*` to the link number on which Q.SIG is installed.
- Set `Protocol Type` to `QSIG`.
- Set `Protocol Variant` to `ISO`.
- Set `Physical Type` to `E1/CEPT`.
- Select `Network-side / Qsig master`.
- Click the `Characteristics` button and configure the `E1/CEPT Characteristics` as follows:



- Under `CRC4`, select `Enable`.
- Under `ADI`, select `Enable`.
- Click `Update` and then `Close`.

- In the `PRI Link Characteristics` dialog box, click `Update`.

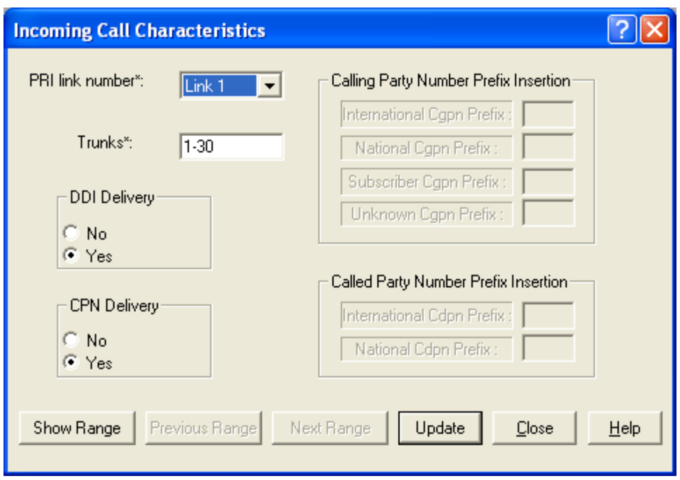
- The Update Both Links dialog box appears.



Click Update.

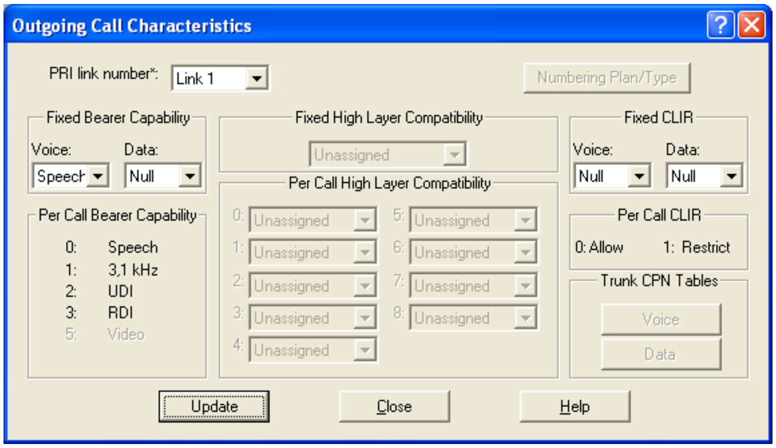
- In the PRI Link Characteristics dialog box, click Close.

Step 4: Click Config > Incoming Call Characteristics to configure the Incoming Call Characteristics as follows:



- Set PRI link number* to the link number that has Q.SIG installed.
- Set Trunks* to 1-30.
- Set DDI Delivery to Yes.
- Set CPN Delivery to Yes.
- Click Update and then Close.

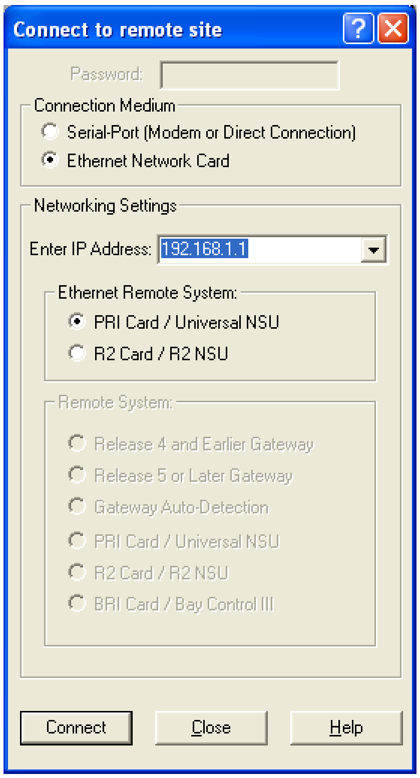
Step 5: Click `Config` > `Outgoing Call Characteristics` to configure the outgoing call characteristics as follows:



Note: Parameters not listed below can be left at their default settings.

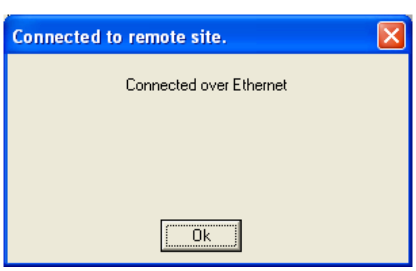
- Set `PRI link number*` to the link number that has Q.SIG installed.
- Set `Voice` to `Speech`.
- Click `Update` and then `Close`.

Step 6: Click `File` > `Connect to Remote Site` to configure the `Connection to Remote Sites` as follows:

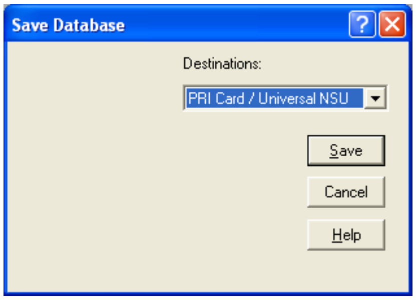


Note: Parameters not listed below can be left at their default settings.

- Set `Connection Medium` to `Ethernet Network Card`.
- Under `Network Settings`, enter the IP address of the 3300 Universal NSU. The default address is 192.168.1.1).
- Set `Ethernet Remote System` to `PRI Card / Universal NSU`.
- Click `Connect`.
- Click `OK` in the `Connected over Ethernet` dialog box.

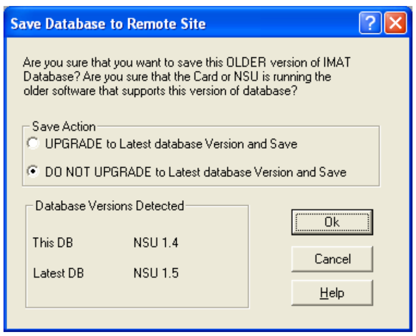


Step 7: Click `File` > `Save` > `Database` to save the database as follows:

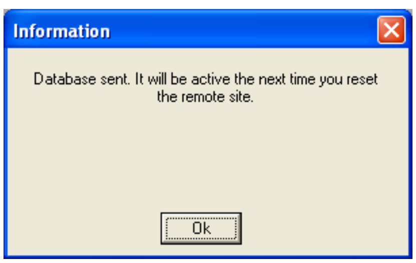


- Set `Destinations` to `PRI Card / Universal NSU`.
- Click `Save`.

Note: If a dialog box appears stating "Are you sure you want to save this OLDER version of IMAT Database?" select `DO NOT UPGRADE to Latest database Version and Save` and click `OK`.

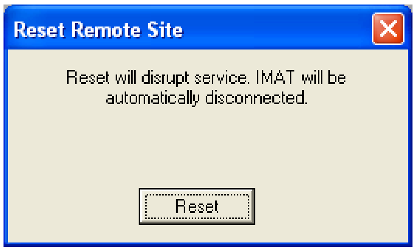


The Information dialog box appears.



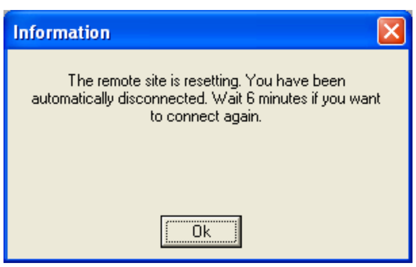
Click `OK`.

Click Maintenance > Remote Site Reset.



Click Reset.

Click OK in the Information dialog box.



Click OK in the dialog box that states the remote site is resetting.

7. Microsoft® Office Communications Server 2007 (OCS) Setup

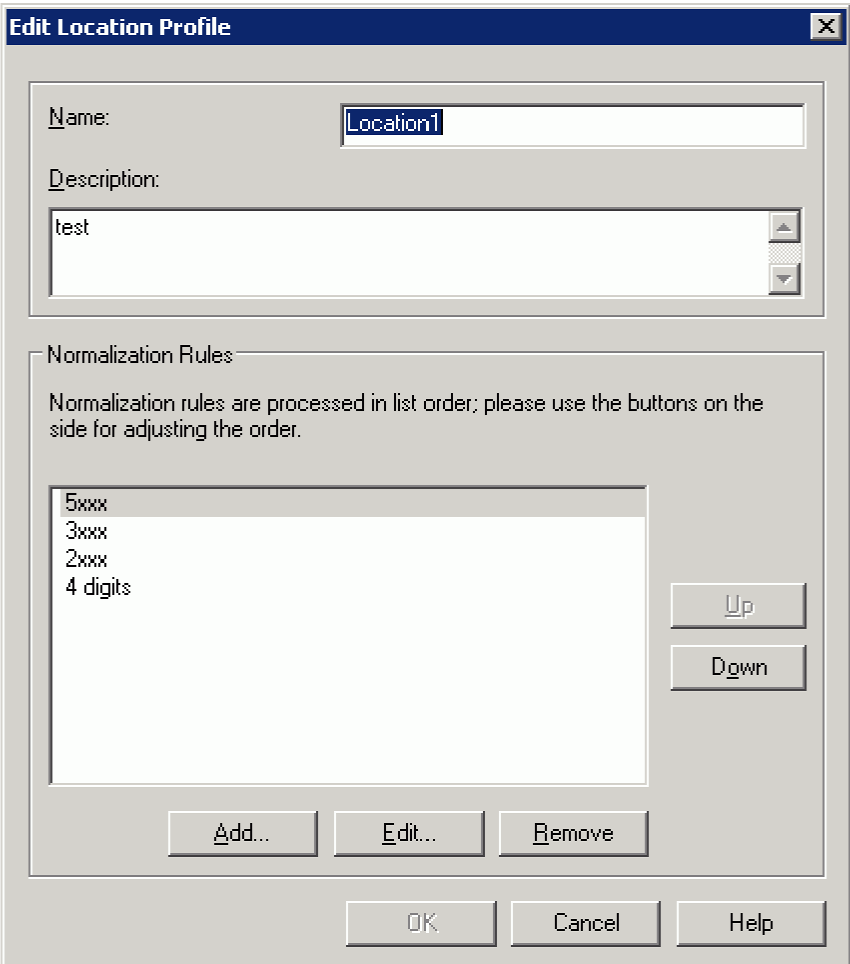
7.1 Steps for configuring Microsoft® OCS

Normalization rules are used to convert dial numbers into full E.164 formatted numbers. Microsoft® OCS uses the standard E.164 format to search for users listed in the Active Directory (AD).

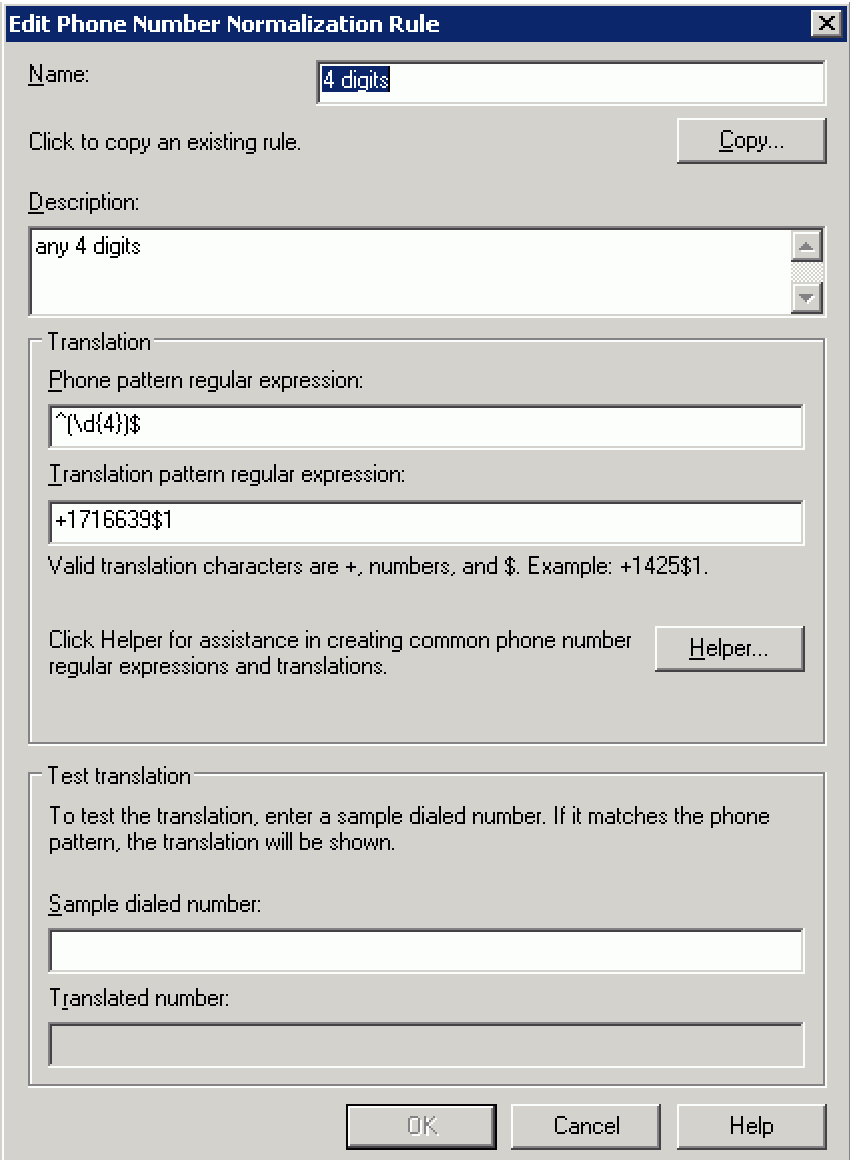
If a Microsoft® OCS user dials an internal extension number (normally 3-5 digits), the normalization rules convert it into full E.164 format. These normalization rules should cover dialed digits for internal extensions, local numbers, long distance numbers, and international numbers.

To configure Microsoft® OCS, click
`Start` > `Programs` > `Administrative Tools` > `OCS 2007`.

On the tree presented in the configuration window, right-click `Forest` then select `Properties` and then `Voice Properties` form the menu provided. Edit a location profile as shown in the following example:



Click `Add` or `Edit` to create or change a particular rule.



In this example, when a user dials any 4-digit number, it will be converted to its E.164 equivalent of +1716639xxxx and then that number will be searched for in AD.

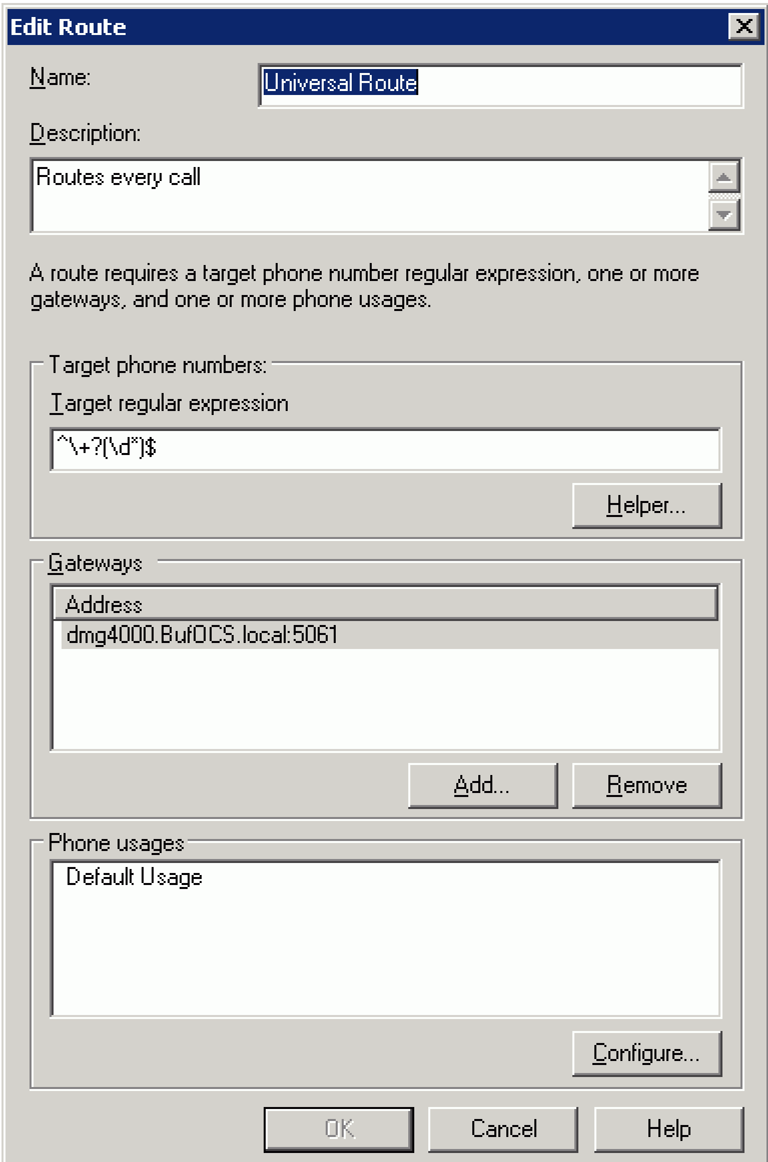
More examples are shown in the following table:

Name	Phone Pattern	Translation Pattern	Comments
Extensions	^(\d{4})$	+1716639$1	Internal extensions
Local	^(\d{7})$	+1716$1	Local number
National	^1(\d*)$	+1$1	Long distance number
International	^011(\d*)	+$1	International number

A default route is used to route all calls to Microsoft® Mediation Server. If you need to route some calls to a different Microsoft® Mediation Server, configure the `Target phone numbers` field accordingly.

To configure Microsoft® OCS, click
`Start` > `Programs` > `Administrative Tools` > `OCS 2007`.

On the tree presented in the configuration window, right-click `Forest` then select `Properties` and then `Voice Properties` form the menu provided. Edit a route as shown in the example below.



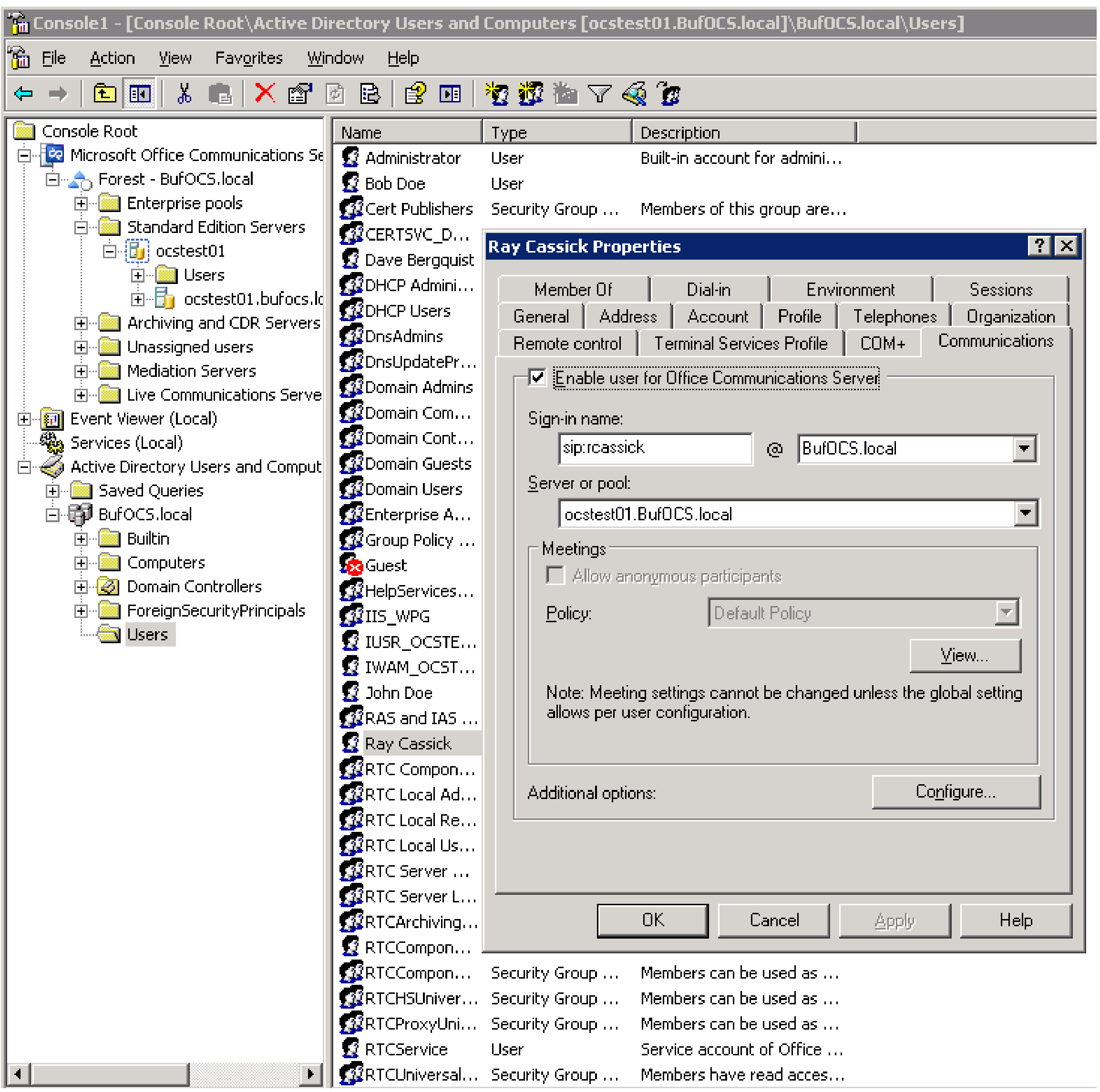
This entry routes numbers with or without "+" prefix followed by any digits to Microsoft® Mediation Server dmg4000.bufocs.local.

Restart the Front End Services for the above changes to take effect, including all normalization rules. This can be done from the window Services.

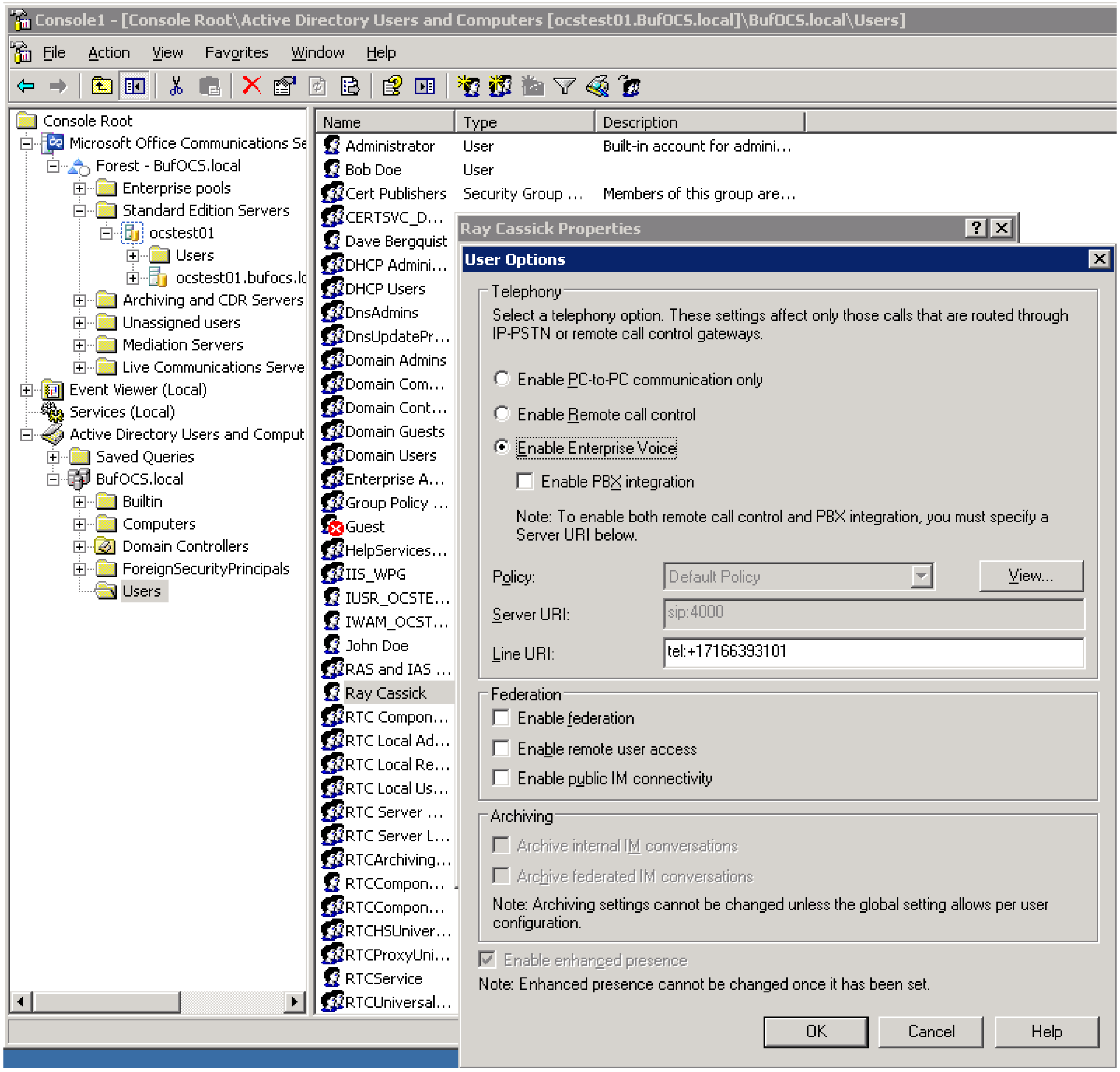
Note: Unless the dialed number from Microsoft® OCS client (such as Microsoft® Office Communicator) is in E.164 format, Microsoft® OCS must find a normalization rule to convert the dialed number to E.164. If no rule is found *or matched, outbound calls will fail. In this case, Dialogic® Diva® Diagnostics trace will not receive an outbound SIP message, since the call will not yet have reached the SIP gateway.*

7.2 Steps for configuring Microsoft® Office Communications Server 2007 (OCS) clients

The domain users need to be enabled for making calls through Microsoft® OCS.



Under the `Communications` tab, check the `Enable user for Office Communications Server` option and then click the `Configure` button.



In the above configuration for the hypothetical user Ray Cassick, an inbound PSTN call for 3101 will be converted by the Dialogic® Diva® SIPcontrol™ Software to +17166393101 because in the Diva SIPcontrol software dialpan in the `SIP Peer Configuration` section under `Address Normalization` the:

- `Number format (outbound)` is set to `International number`, and
- `Encoding (outbound)` is set to `Use type flag`.

Microsoft® OCS will ring the user Ray Cassick if he is logged on to Microsoft® OCS from Microsoft® Office Communicator or any Microsoft® OCS supported device.

8. Microsoft® Mediation Server Installation and Configuration

8.1 Installation

The gateways of the Dialogic® 4000 Media Gateway Series (DMG4000 Gateways) are shipped with pre-installed Microsoft® Mediation Server software. You can complete the Microsoft® Mediation Server configuration by running Microsoft® Office Communications Server 2007 (OCS) “Setup.exe” in the DMG4000 Gateways. In the Microsoft® OCS Deployment Wizard, select `Deploy Other Server Roles`, then select `Deploy Mediation Sever`. Follow the steps in the Wizard to complete the setup:

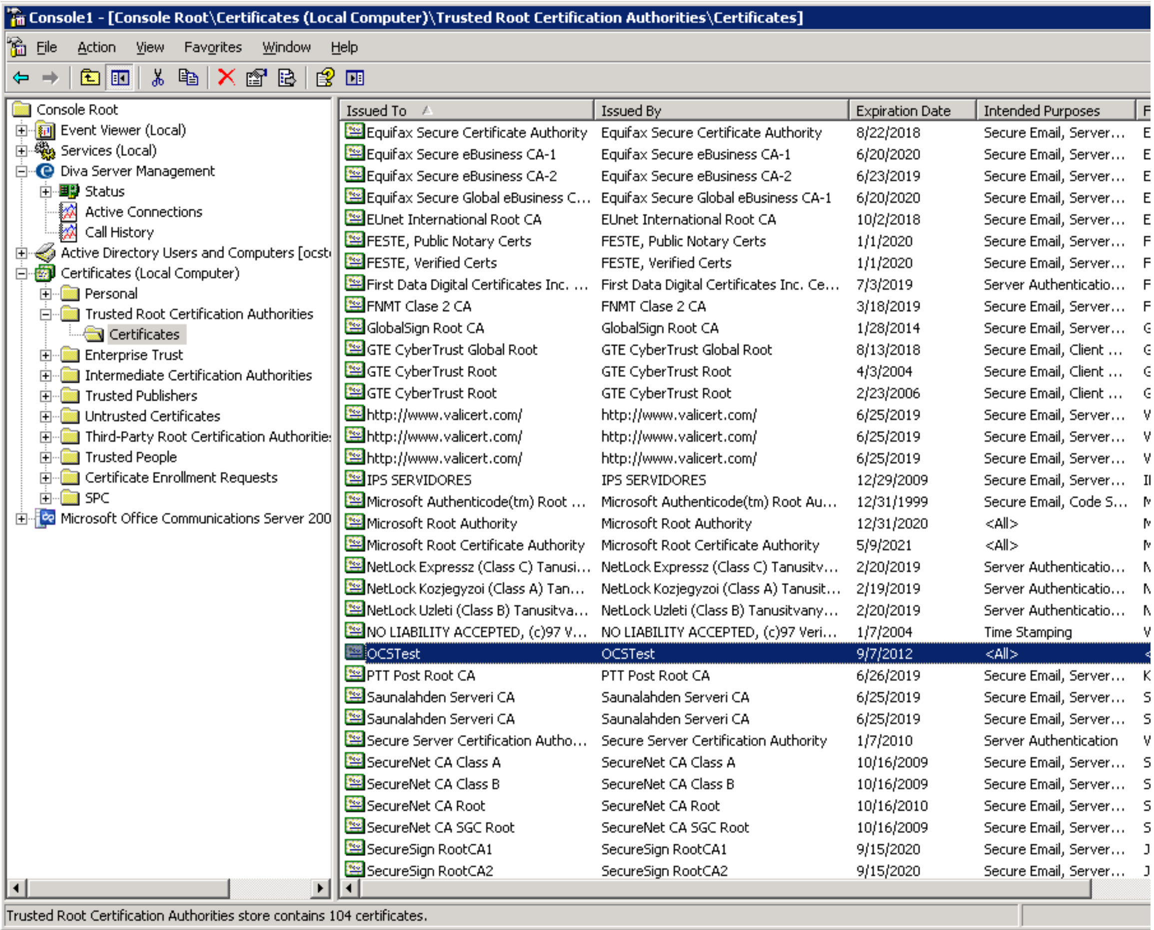
Step 1: Install the Microsoft® Mediation Server software.

Step 2: Activate Microsoft® Mediation Server. Use the existing account and enter the password for the service account.

Step 3: No action needed. Do this step when the installation is complete.

Step 4: Configure Certificate.

1. Download the CA certification path for Microsoft® Mediation Server.
    - From `Start` > `Run`, enter http://<CA server>/certsrv
    - Select to download a CA certificate, chain or CRL.
    - Click `Download CA certificate chain`.
    - In `File Download`, click `Save`.
2. Install the certificate chain for the Microsoft® Mediation Server:
    - In the Deployment Wizard, run step 4 again.
    - Select `Import a certificate chain from a .p7b file` in step 1.
3. Verify that your CA is in the list of Trusted root CAs:
    - In the Microsoft® Management Console (MMC) snap-in, click `Certificates` (If not already done, add it.)
    - Verify that CA is on the list of trusted CAs as shown in the example below.

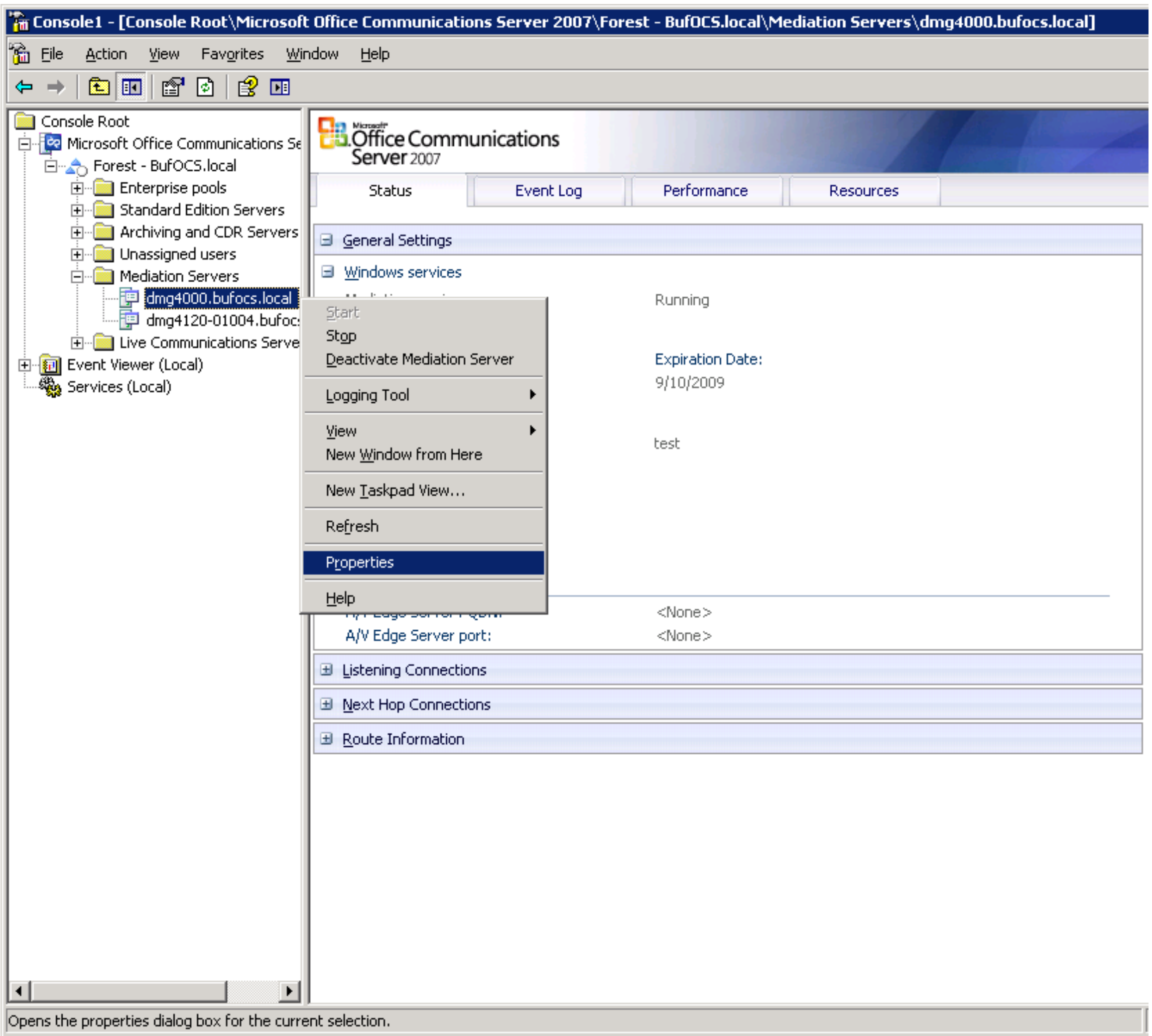


4. Create the certificate request for the Microsoft® Mediation Server:
   - Run Deployment Wizard, click step 4.
   - Select the option `Create a new certificate`.
   - Select the option `Send the request immediately to an online CA`.
   - Complete the settings in the blank.
   - Click `Assign` to complete the task.

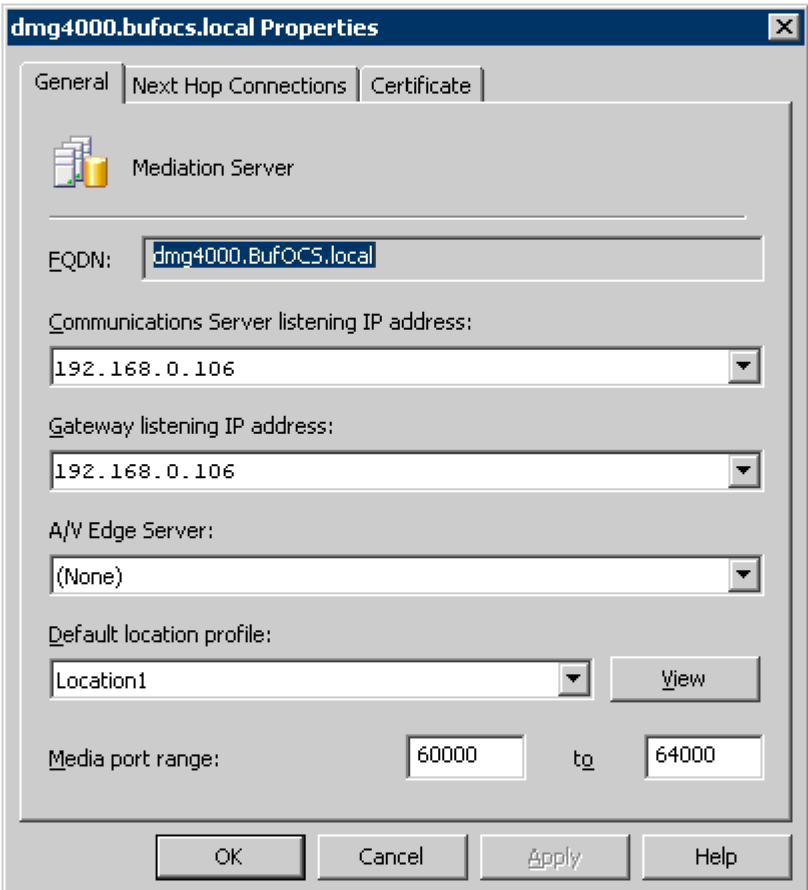
Note: If you receive the error message "certificate expired or is not yet valid" when you click the assign button at the end of step 4, check the time/time zone configured for your Microsoft® Mediation Server is correct, then run the Deployment Wizard again or click `Certificates` in `Available tasks` in Microsoft® Mediation Server MMC snap-in.

8.2 Configuration

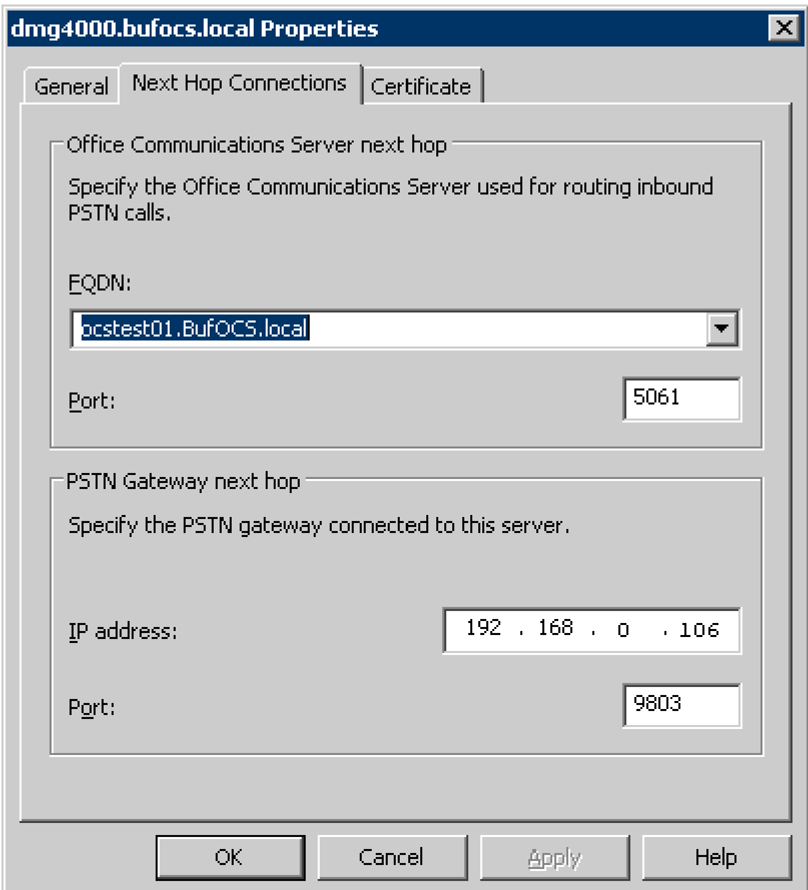
From the MMC snap-in, right-click the detected Microsoft® Mediation Server and select `Properties`.



Configure the following settings on the `General` tab.

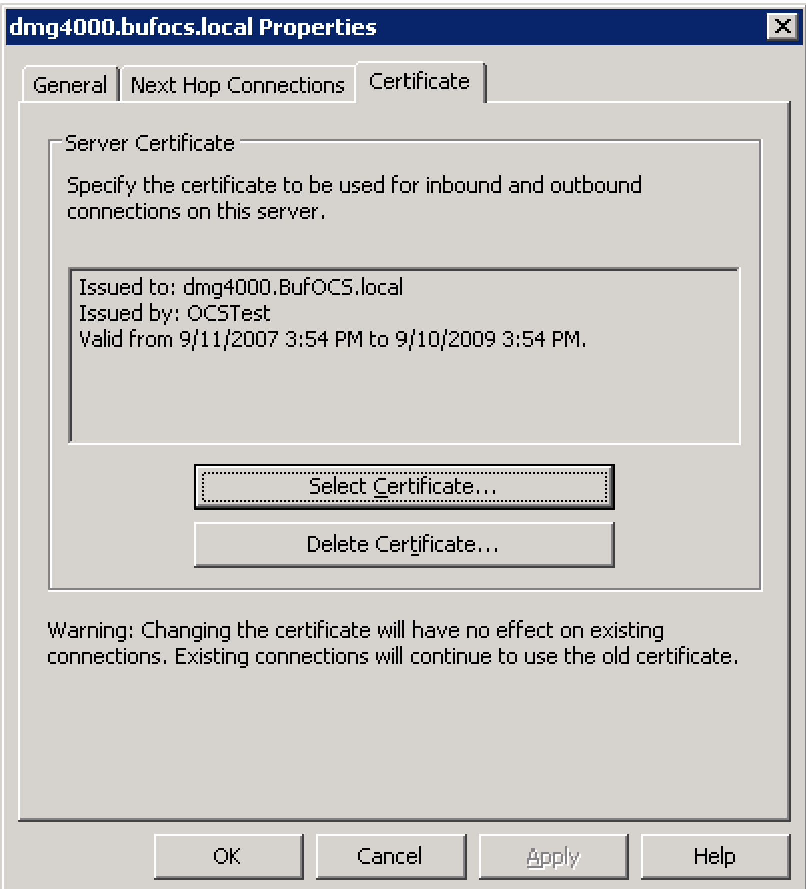


Click the `Next Hop Connections` tab and configure the following:



The `Port` entry under `PSTN Gateway Next hop` has to match the configuration in the Dialogic® Diva® SIPcontrol™ Software under `Network Interface Configuration` > `SIP Listen Port`.

Click the `Certificate` tab.



Select the certificate that will be used to communicate with Microsoft® OCS. Microsoft® Mediation Server will need to restart for these changes to properly take effect.

9. Testing the Validation Matrix

The table below shows various test scenarios that are run as typical validation scenarios if the Dialogic® Media Gateway is used in a voice messaging situation. The notes column specifies any notable parts of the test.

The test scenarios below assume that all gateway configuration parameters are at their default values. For a sample showing call flows and states please consult the Gateway SIP Compatibility Guide.

Test Number	Call Scenario Description	Notes
Inbound call scenarios		
1	Direct call from TDM station set to Microsoft® OCS client.	
2	Direct call from Microsoft® OCS client to TDM station set.	

10. Troubleshooting

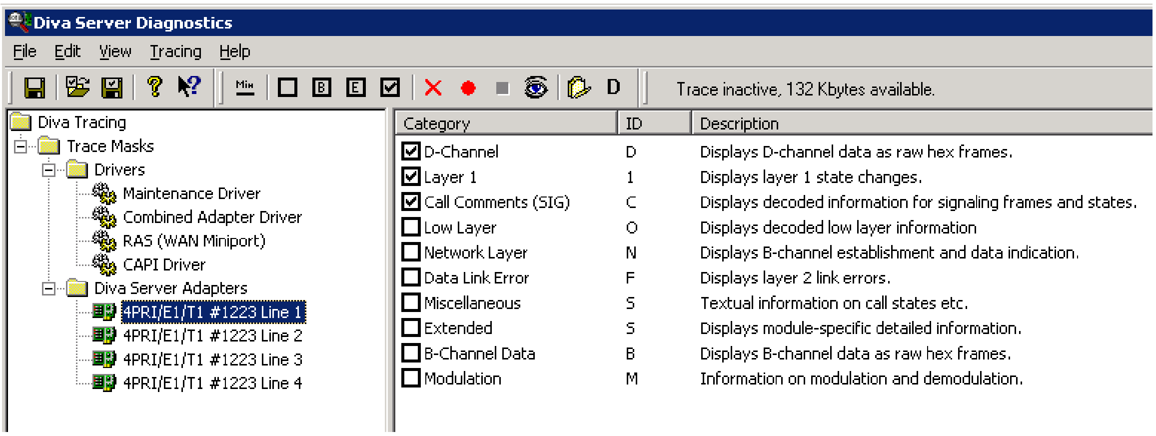
10.1 Important Debugging Tools

- **Ethereal/Wireshark**: Can be used to view and analyze the network captures provided by the Dialogic gateway diagnostic firmware.
- **Adobe Audition**: Can be used to review and analyze the audio extracted from the network captures to troubleshoot any audio related issues.
- **Dialogic® Diva® Diagnostics tool**: Used to review and analyze all SIP and ISDN traffic that relates to calls going into and leaving the Dialogic® 4000 Media Gateway.

10.2 Using the Dialogic® Diva® Diagnostics Tool

Before using the Dialogic® Diva® Diagnostics tool, you would need to enable it by setting the Dialogic® Diva® SIPcontrol™ Software debug. To do so, open the Diva SIPcontrol software web interface, click the link `System Settings`, and set `Debug Level` to `Extended`. Click the `Save` button for the changes to take effect.

Now, you can start the Diva Diagnostics tool. To do so, click:
`Start` > `Programs` > `Dialogic Diva > Diagnostics`.



1. Click one line of your Dialogic® Diva® Media Board in the left pane and click B on the toolbar to activate the Basic tracing level. This level captures Q.931 ISDN messages.

2. Click CAPI driver in the left pane and activate the Basic tracing level as explained in step 1.

3. Start tracing. To do so, click the start icon or select the `Start Tracing` option form the `Tracing` menu.

4. Reproduce the issue.

5. To stop tracing, click the stop icon on the tool bar or select the `Stop Tracing` option form the `Tracing` menu.

6. To view your collected trace, click the view icon on the toolbar or select the `View Recorded Trace` option from the `View` menu. A notepad window will open with the recorded log.

Examples of Dialogic® Diva® Diagnostics traces for an inbound (TDM to IP) call to Microsoft® Office Communications Server 2007 (OCS)

Basic notations for reading the trace:

- SIG-R: RX Q.931 ISDN message
- SIG-X: TX Q.931 ISDN message
- SIPR: RX SIP message
- SIPX: TX SIP message

< Below is a RX Q.931 ISDN message for an inbound call >

```
…
9:16:28.431 C  3 21:2389:383 - SIG-R(030) 08 02 00 17 05 04 03 80 90 A2 18 03 A9 83 81 6C 06 01 A0 33
30 30 32 70 05 C1 35 31 30 31
                     Q.931  CR0017 SETUP
                            Bearer Capability 80 90 a2
                            Channel Id a9 83 81
                            Calling Party Number 01 a0 '2401'
                            Called Party Number c1 '5101'

…
```

<Below is a TX SIP message with SDP>

```
9:16:28.431 1 L 12 00010000-SIPX begin to IP:192.168.0.106 port:5060 socket:3 Proto:TCP
 9:16:28.431 1 L 12 00010000-    >INVITE sip:+17166395101@dmg4000.bufocs.local:5060 SIP/2.0
 9:16:28.431 1 L 12 00010000-    >Via: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
 9:16:28.431 1 L 12 00010000-    >Max-Forwards: 70
 9:16:28.431 1 L 12 00010000-    >Allow: INVITE,ACK,CANCEL,BYE,OPTIONS,NOTIFY,REFER
 9:16:28.431 1 L 12 00010000-    >Accept: application/sdp,application/simple-message-summary
 9:16:28.431 1 L 12 00010000-    >Supported: timer,replaces
 9:16:28.431 1 L 12 00010000-    >From: "Dialogic Diva SIPcontrol"
<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
 9:16:28.431 1 L 12 00010000-    >To: "Default" <sip:+17166395101@bufocs.local;user=phone>
 9:16:28.431 1 L 12 00010000-    >Call-ID: 9c046698-730448-17@dmg4000
 9:16:28.431 1 L 12 00010000-    >CSeq: 1 INVITE
 9:16:28.431 1 L 12 00010000-    >Min-SE: 90
 9:16:28.431 1 L 12 00010000-    >Session-Expires: 600;refresher=uac
 9:16:28.431 1 L 12 00010000-    >Contact: <sip:+17166392401@192.168.0.106:9803>
 9:16:28.431 1 L 12 00010000-    >Content-Type: application/sdp
 9:16:28.431 1 L 12 00010000-    >Content-Length: 253
 9:16:28.431 1 L 12 00010000-    >
 9:16:28.431 1 L 12 00010000-    >v=0
 9:16:28.431 1 L 12 00010000-    >o=SIPcontrol 7472200 7472200 IN IP4 192.168.0.106
 9:16:28.431 1 L 12 00010000-    >s=-
 9:16:28.431 1 L 12 00010000-    >c=IN IP4 192.168.0.106
 9:16:28.431 1 L 12 00010000-    >t=0 0
 9:16:28.431 1 L 12 00010000-    >m=audio 30060 RTP/AVP 8 0 101 13
 9:16:28.431 1 L 12 00010000-    >a=rtpmap:8 PCMA/8000
 9:16:28.431 1 L 12 00010000-    >a=rtpmap:0 PCMU/8000
 9:16:28.431 1 L 12 00010000-    >a=rtpmap:101 telephone-event/8000
 9:16:28.431 1 L 12 00010000-    >a=fmtp:101 0-15
 9:16:28.431 1 L 12 00010000-    >a=rtpmap:13 CN/8000
 9:16:28.431 1 L 12 00010000-    >a=sendrecv
 9:16:28.431 1 L 12 00010000-SIPX end
…
```

<Below is a RX SIP message>

```
9:16:28.431 1 L 12 00010000-SIPR begin (331 byte) from IP:192.168.0.106 PORT:5060 on socket 3 port 5060
TCP
 9:16:28.431 1 L 12 00010000-    >SIP/2.0 100 Trying
 9:16:28.431 1 L 12 00010000-    >FROM: "Dialogic Diva
SIPcontrol"<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
 9:16:28.431 1 L 12 00010000-    >TO: "Default"<sip:+17166395101@bufocs.local;user=phone>
 9:16:28.431 1 L 12 00010000-    >CSEQ: 1 INVITE
 9:16:28.431 1 L 12 00010000-    >CALL-ID: 9c046698-730448-17@dmg4000
 9:16:28.431 1 L 12 00010000-    >VIA: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
 9:16:28.431 1 L 12 00010000-    >CONTENT-LENGTH: 0
 9:16:28.431 1 L 12 00010000-    >
 9:16:28.431 1 L 12 00010000-SIPR end
…
9:16:28.665 0 L 12 00010000-SIPR begin (408 byte) from IP:192.168.0.106 PORT:5060 on socket 3 port 5060
TCP
 9:16:28.665 0 L 12 00010000-    >SIP/2.0 183 Session Progress
 9:16:28.665 0 L 12 00010000-    >FROM: "Dialogic Diva
SIPcontrol"<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
 9:16:28.665 0 L 12 00010000-    >TO:
Default<sip:+17166395101@bufocs.local;user=phone>;epid=CE4C602FA5;tag=3f5ea65423
 9:16:28.665 0 L 12 00010000-    >CSEQ: 1 INVITE
 9:16:28.665 0 L 12 00010000-    >CALL-ID: 9c046698-730448-17@dmg4000
 9:16:28.665 0 L 12 00010000-    >VIA: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
 9:16:28.665 0 L 12 00010000-    >CONTENT-LENGTH: 0
 9:16:28.665 0 L 12 00010000-    >SERVER: RTCC/3.0.0.0 MediationServer
 9:16:28.665 0 L 12 00010000-    >
 9:16:28.665 0 L 12 00010000-SIPR end
…
9:16:28.869 1 L 12 00010000-SIPR begin (399 byte) from IP:192.168.0.106 PORT:5060 on socket 3 port 5060
TCP
 9:16:28.869 1 L 12 00010000-    >SIP/2.0 180 Ringing
 9:16:28.869 1 L 12 00010000-    >FROM: "Dialogic Diva
SIPcontrol"<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
 9:16:28.869 1 L 12 00010000-    >TO:
Default<sip:+17166395101@bufocs.local;user=phone>;epid=CE4C602FA5;tag=3f5ea65423
 9:16:28.869 1 L 12 00010000-    >CSEQ: 1 INVITE
 9:16:28.869 1 L 12 00010000-    >CALL-ID: 9c046698-730448-17@dmg4000
 9:16:28.869 1 L 12 00010000-    >VIA: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
 9:16:28.869 1 L 12 00010000-    >CONTENT-LENGTH: 0
 9:16:28.869 1 L 12 00010000-    >SERVER: RTCC/3.0.0.0 MediationServer
 9:16:28.869 1 L 12 00010000-    >
 9:16:28.869 1 L 12 00010000-SIPR end
…
9:16:30.197 1 L 12 00010000-SIPR begin (836 byte) from IP:192.168.0.106 PORT:5060 on socket 3 port 5060
TCP
 9:16:30.197 1 L 12 00010000-    >SIP/2.0 200 OK
 9:16:30.197 1 L 12 00010000-    >FROM: "Dialogic Diva
SIPcontrol"<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
 9:16:30.197 1 L 12 00010000-    >TO:
Default<sip:+17166395101@bufocs.local;user=phone>;epid=CE4C602FA5;tag=3f5ea65423
 9:16:30.197 1 L 12 00010000-    >CSEQ: 1 INVITE
 9:16:30.197 1 L 12 00010000-    >CALL-ID: 9c046698-730448-17@dmg4000
 9:16:30.197 1 L 12 00010000-    >VIA: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
 9:16:30.197 1 L 12 00010000-    >CONTACT:
<sip:dmg4000.BufOCS.local:5060;transport=Tcp;maddr=192.168.0.106>
 9:16:30.197 1 L 12 00010000-    >CONTENT-LENGTH: 253
 9:16:30.197 1 L 12 00010000-    >SUPPORTED: 100rel
 9:16:30.197 1 L 12 00010000-    >CONTENT-TYPE: application/sdp; charset=utf-8
```

```
 9:16:30.197 1 L 12 00010000-    >ALLOW: UPDATE
 9:16:30.197 1 L 12 00010000-    >SERVER: RTCC/3.0.0.0 MediationServer
 9:16:30.197 1 L 12 00010000-    >ALLOW: Ack, Cancel, Bye,Invite
 9:16:30.197 1 L 12 00010000-    >
 9:16:30.197 1 L 12 00010000-    >v=0
 9:16:30.197 1 L 12 00010000-    >o=- 0 0 IN IP4 192.168.0.106
 9:16:30.197 1 L 12 00010000-    >s=session
 9:16:30.197 1 L 12 00010000-    >c=IN IP4 192.168.0.106
 9:16:30.197 1 L 12 00010000-    >b=CT:1000
 9:16:30.197 1 L 12 00010000-    >t=0 0
 9:16:30.197 1 L 12 00010000-    >m=audio 62438 RTP/AVP 8 101
 9:16:30.197 1 L 12 00010000-    >c=IN IP4 192.168.0.106
 9:16:30.197 1 L 12 00010000-    >a=rtcp:62439
 9:16:30.197 1 L 12 00010000-    >a=label:Audio
 9:16:30.197 1 L 12 00010000-    >a=rtpmap:8 PCMA/8000
 9:16:30.197 1 L 12 00010000-    >a=rtpmap:101 telephone-event/8000
 9:16:30.197 1 L 12 00010000-    >a=fmtp:101 0-16
 9:16:30.197 1 L 12 00010000-    >a=ptime:20
 9:16:30.197 1 L 12 00010000-SIPR end
```

…

<Bellow is a TX Q.931 ISDN message, after SIP session is established>

```
9:16:30.212 C  3 21:2391:136 - SIG-X(005) 08 02 80 17 07
            Q.931  CR8017 CONN
```