



Dialogic[®] 4000 Media Gateway Series Integration Note

Mitel 3300 ICP

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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

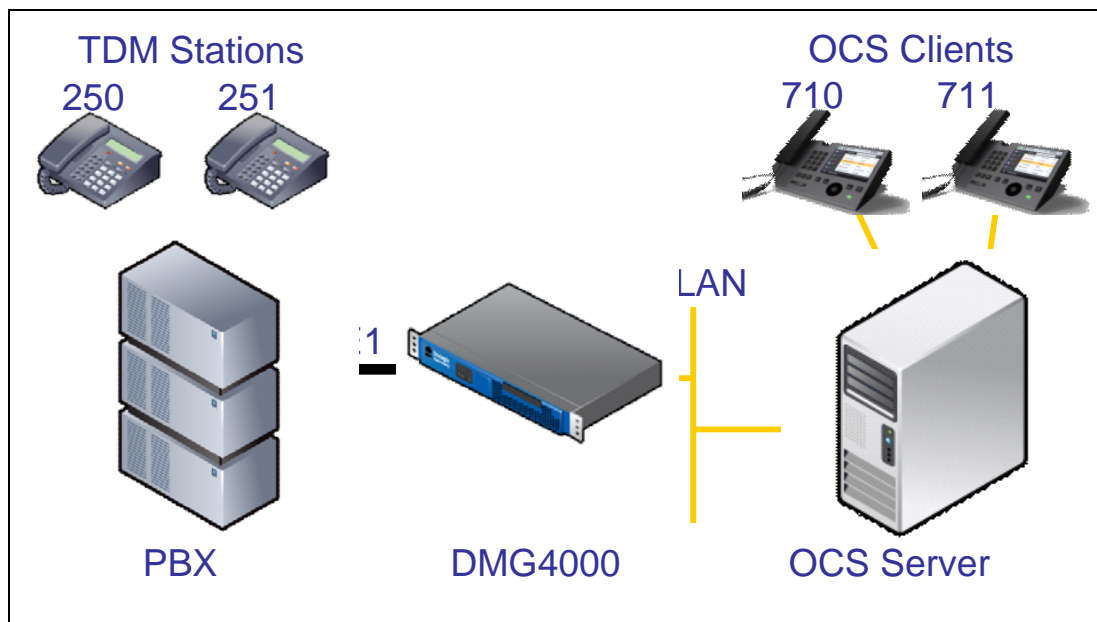
| | |
|---------------------|------------------------|
| PBX Vendor | Mitel |
| Model(s) | 3300 ICP |
| Software Version(s) | System Version 5.1.4.8 |
| Additional Notes | N/A |

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.

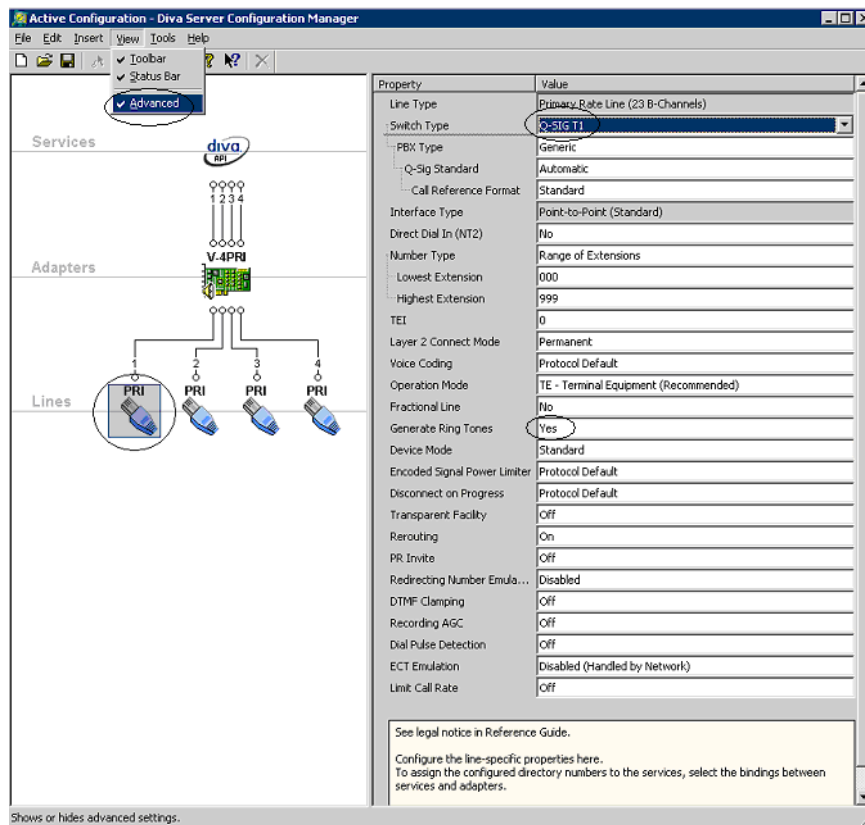
5.1 Dialogic® Diva® Media Board Configuration

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 screenshot displays the Dialogic SIPcontrol Configuration web interface. The main content area is divided into two sections: 'PSTN Interface Configuration' and 'Network Interface Configuration'.

PSTN Interface Configuration Table:

| Name | Nr | Hardware Description | Channels | Dialplan | Enabled | Details |
|-------------|----|---|----------|----------|-------------------------------------|---------|
| Controller1 | 1 | Eicon Diva Server V-4PRVE1/T1 - PORT 1 SN: 1223 | 23 | none | <input checked="" type="checkbox"/> | Details |
| Controller2 | 2 | Eicon Diva Server V-4PRVE1/T1 - PORT 2 SN: 1223 | 23 | none | <input checked="" type="checkbox"/> | Details |
| Controller3 | 3 | Eicon Diva Server V-4PRVE1/T1 - PORT 3 SN: 1223 | 23 | none | <input checked="" type="checkbox"/> | Details |
| Controller4 | 4 | Eicon Diva Server V-4PRVE1/T1 - PORT 4 SN: 1223 | 23 | none | <input checked="" type="checkbox"/> | Details |

Network Interface Configuration Table:

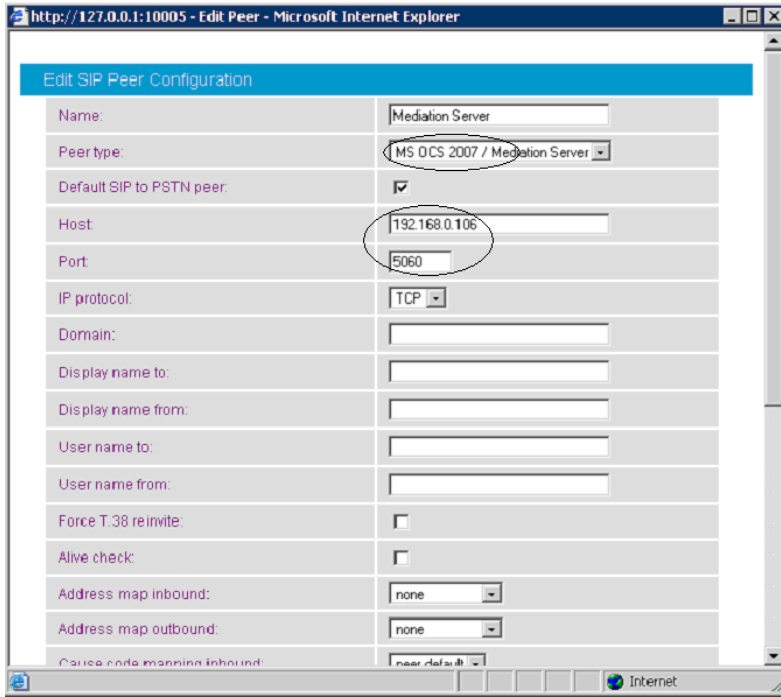
| Name | Device | IP Address | Protocol | SIP Listen Port | Enabled |
|---------------------------------|--|---------------|----------|-----------------|-------------------------------------|
| Inte[R] PRD1000 EB Network Conn | Inte[R] PRO/1000 EB Network Connection with I/O Acceleration | 192.168.0.106 | all | 9803 | <input checked="" type="checkbox"/> |
| Local Loopback Interface | Local Loopback Interface | 127.0.0.1 | all | 5060 | <input type="checkbox"/> |

Below the tables, there are input fields for 'RTP Start Port' (30000), 'RTP End Port' (39999), and 'Jitterbuffer Size Min [ms]' (0). There are also expandable sections for 'SIP Peer Configuration' and 'Routing 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.

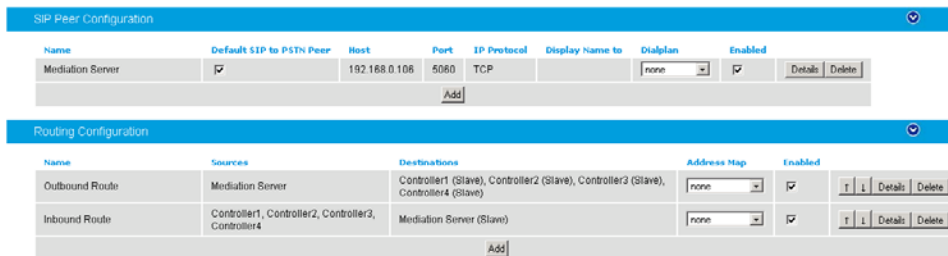
5.2.2 SIP Peer Configuration

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

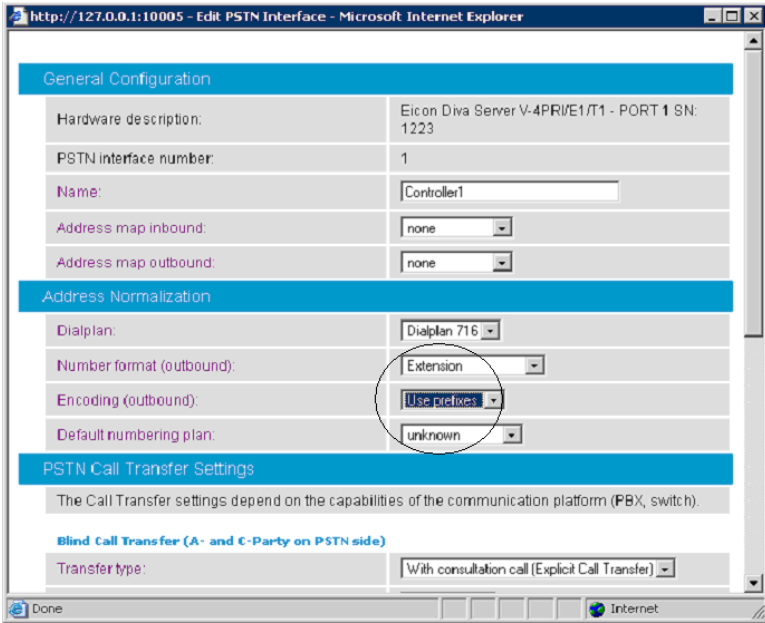
The screenshot shows a web browser window titled "http://127.0.0.1:10005 - Edit Dialplan - Microsoft Internet Explorer". The main content area is a form titled "Edit Dialplan Configuration". The form fields are as follows:

| | |
|--|-------------------------------------|
| Name: | Dialplan 716 |
| Country code: | 1 |
| North-American numbering plan: | <input checked="" type="checkbox"/> |
| Area code: | 716 With national prefix |
| Other local areas: | [Empty input boxes] |
| Base number: | 639 |
| Maximum extension digits: | 4 |
| International prefix: | 011 |
| National prefix: | 1 |
| Access code: | 9 |
| PSTN access code provided by SIP caller: | <input type="checkbox"/> |
| Incoming PSTN access code provided by PBX: | <input type="checkbox"/> |

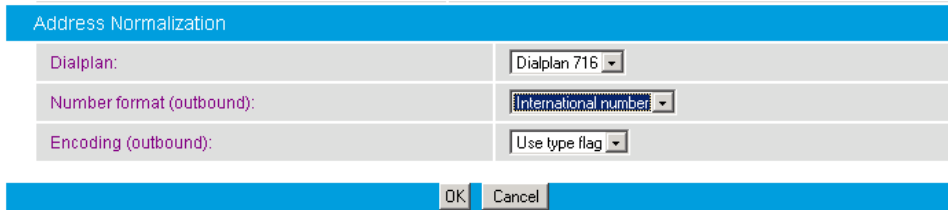
At the bottom of the form are "OK" and "Cancel" buttons. The browser status bar shows "Done" and "Internet".

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](tel:+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 | +1xxxxxxxxxxx | 91xxxxxxxxxxx |
| To International | +xxx...xxx | +xxx...xxx |

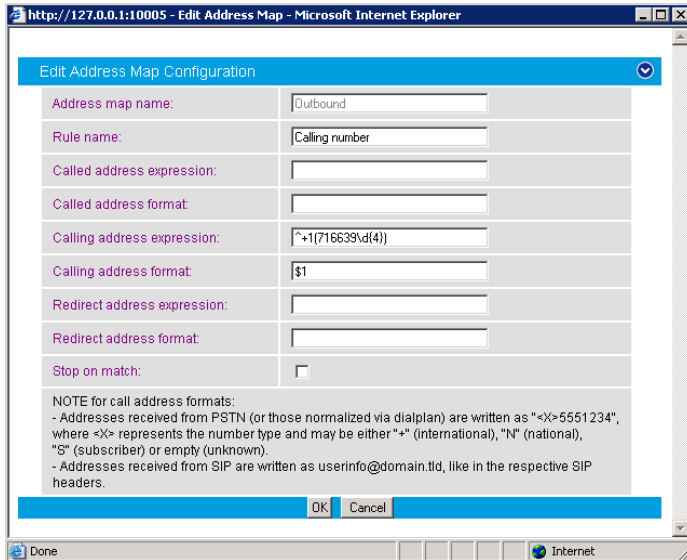
Below is RegEx for the conversion tables above.

| 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 |

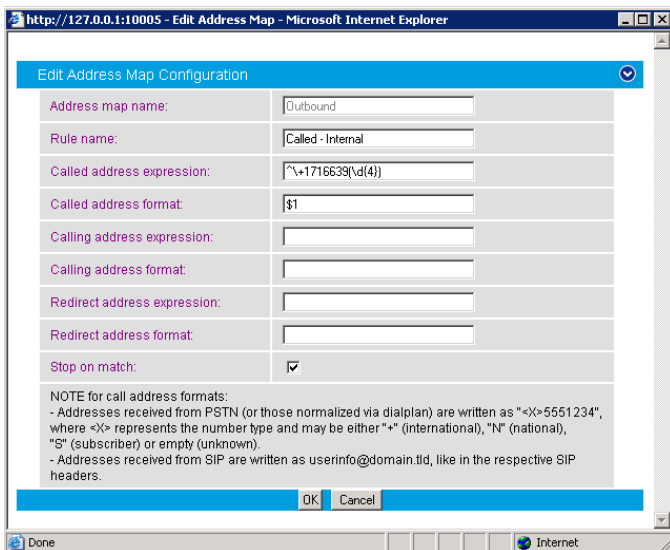
Below are the configured address maps for outbound calls. The order of the below four sub rules and the stop on match check mark are relevant:

| Address Map Configuration | | | |
|---|------------------------|-------------------------------------|-------------------------------------|
| Name | Rule Name | Stop on Match | Enabled |
| Outbound | Calling number | <input type="checkbox"/> | <input checked="" type="checkbox"/> |
| | Called - Internal | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |
| | Called - National | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |
| | Called - International | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |
| <input type="button" value="Add Rule"/> | | | |
| <input type="button" value="Add"/> | | | |

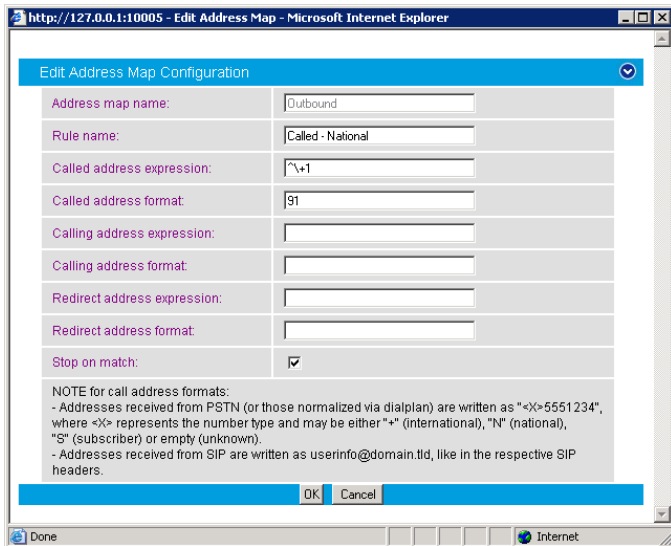
The following screen shows the first sub rule that converts the E.164 calling number into a 10-digit national number:



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:



http://127.0.0.1:10005 - Edit Address Map - Microsoft Internet Explorer

Edit Address Map Configuration

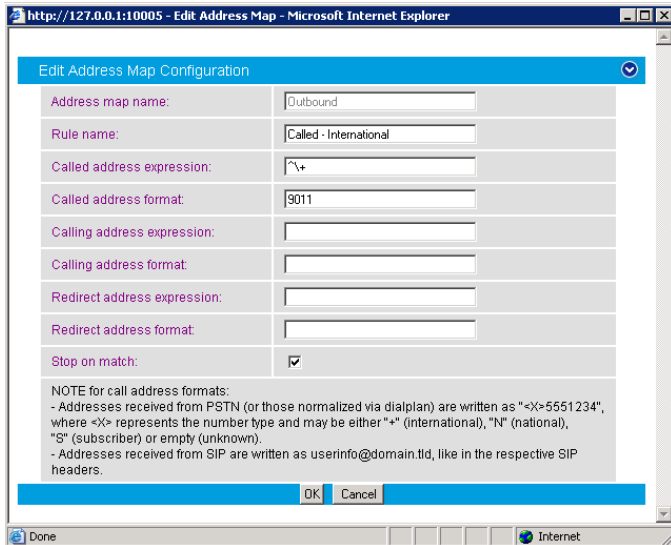
| | |
|------------------------------|-------------------------------------|
| Address map name: | Outbound |
| Rule name: | Called - National |
| Called address expression: | ^\\+1 |
| Called address format: | 91 |
| Calling address expression: | |
| Calling address format: | |
| Redirect address expression: | |
| Redirect address format: | |
| Stop on match: | <input checked="" type="checkbox"/> |

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

The following example converts international call numbers:



http://127.0.0.1:10005 - Edit Address Map - Microsoft Internet Explorer

Edit Address Map Configuration

| | |
|------------------------------|-------------------------------------|
| Address map name: | Outbound |
| Rule name: | Called - International |
| Called address expression: | ^\\+ |
| Called address format: | 9011 |
| Calling address expression: | |
| Calling address format: | |
| Redirect address expression: | |
| Redirect address format: | |
| Stop on match: | <input checked="" type="checkbox"/> |

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

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:

| Name | Sources | Destinations | Address Map | Enabled | |
|----------------|--|--|-------------|-------------------------------------|-------------|
| Outbound Route | Mediation Server | Controller1 (Slave), Controller2 (Slave), Controller3 (Slave), Controller4 (Slave) | Outbound | <input checked="" type="checkbox"/> | ↑ ↓ Details |
| Inbound Route | Controller1, Controller2, Controller3, Controller4 | Mediation Server (Slave) | none | <input checked="" type="checkbox"/> | ↑ ↓ Details |

[Add](#)

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 |

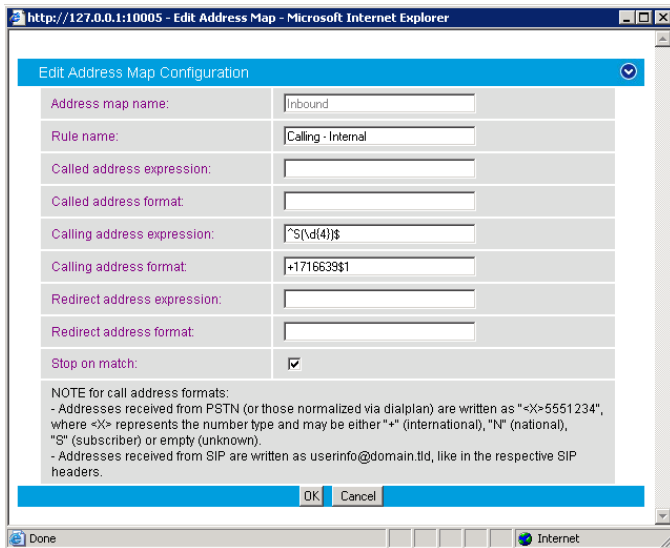
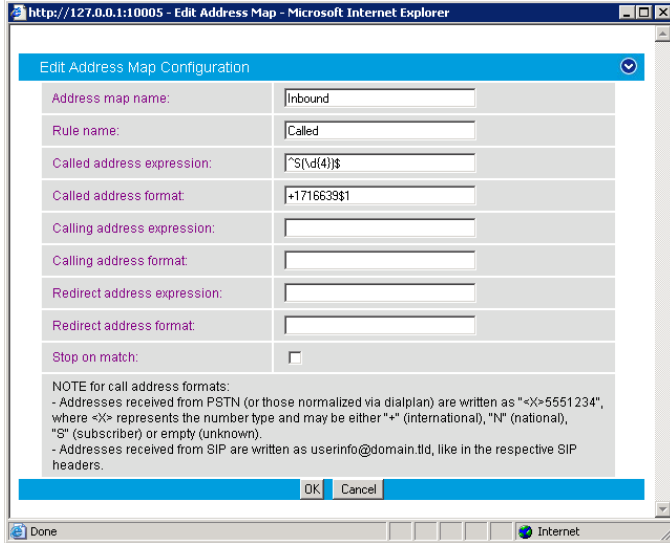
| Calling number | From PBX | To Microsoft® OCS |
|----------------------------|---|-------------------|
| Calling from internal | xxxx (with subscriber type of number) | +1716639xxxx |
| Calling from national | xxxxxxxxxx (with national type of number) | +1xxxxxxxxxx |
| Calling from international |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

| Name | Rule Name | Stop on Match | Enabled | |
|----------|--------------------------|-------------------------------------|-------------------------------------|--------------------|
| Outbound | Calling number | <input type="checkbox"/> | <input checked="" type="checkbox"/> | ↑ ↓ Details Delete |
| | Called - Internal | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | ↑ ↓ Details Delete |
| | Called - National | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | ↑ ↓ Details Delete |
| | Called - International | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | ↑ ↓ Details Delete |
| | Add Rule | | | |
| Inbound | Called | <input type="checkbox"/> | <input checked="" type="checkbox"/> | ↑ ↓ Details Delete |
| | Calling - Internal | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | ↑ ↓ Details Delete |
| | Calling - National | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | ↑ ↓ Details Delete |
| | Calling - International | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | ↑ ↓ Details Delete |
| | Add Rule | | | |

[Add](#)



Address map name: Inbound

Rule name: Calling - National

Called address expression:

Called address format:

Calling address expression: ^N(\d{10})\$

Calling address format: +1\$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).
 - Addresses received from SIP are written as userinfo@domain.tld, like in the respective SIP headers.

OK Cancel

Address map name: Inbound

Rule name: Calling - International

Called address expression:

Called address format:

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

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

| Name | Sources | Destinations | Address Map | Enabled | |
|----------------|--|--|-------------|-------------------------------------|-----------------|
| Outbound Route | Mediation Server | Controller1 (Slave), Controller2 (Slave), Controller3 (Slave), Controller4 (Slave) | Outbound | <input checked="" type="checkbox"/> | [↑] [↓] Details |
| Inbound Route | Controller1, Controller2, Controller3, Controller4 | Mediation Server (Slave) | Inbound | <input checked="" type="checkbox"/> | [↑] [↓] Details |

Add

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.*

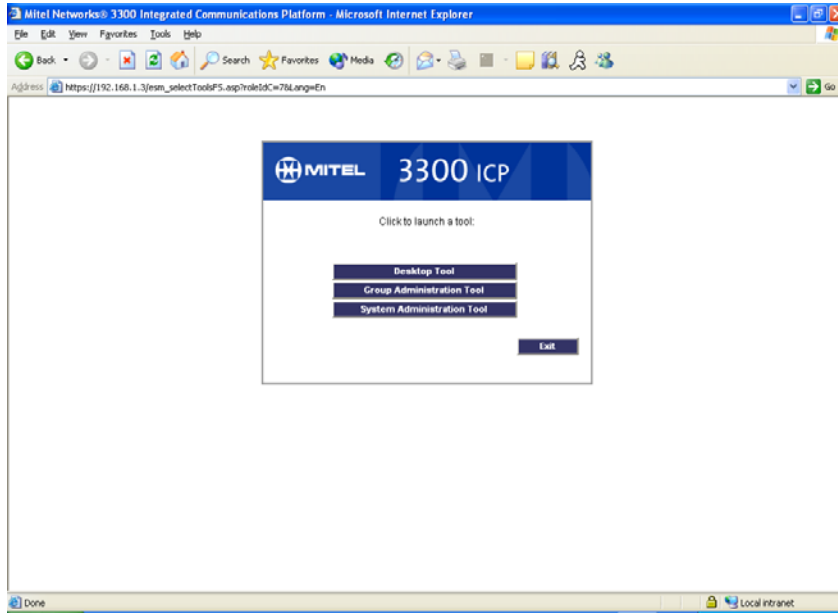
6.1.1 E1 Interface Administration

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.

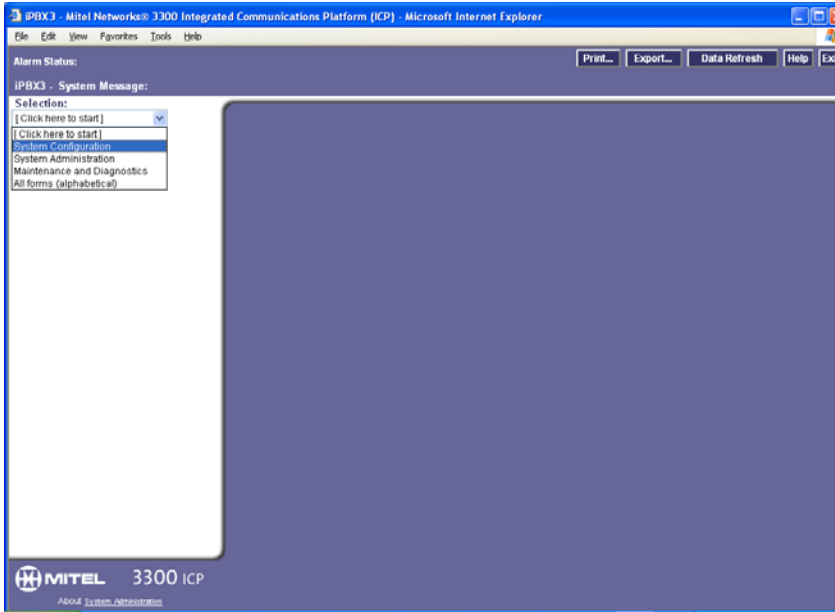
6.1.1.1 3300 ICP Administration

Step 1: Log into the Mitel 3300 ICP system.

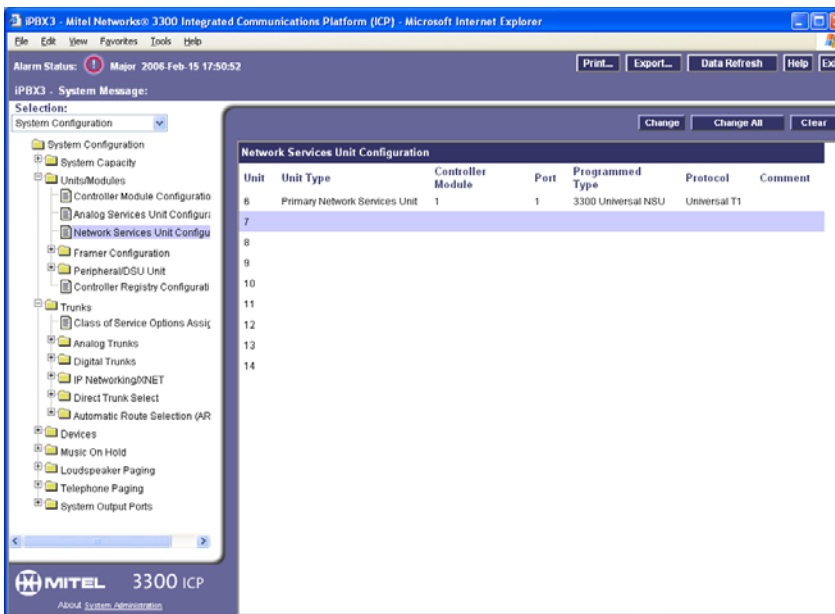


Select the System Administration Tool option.

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

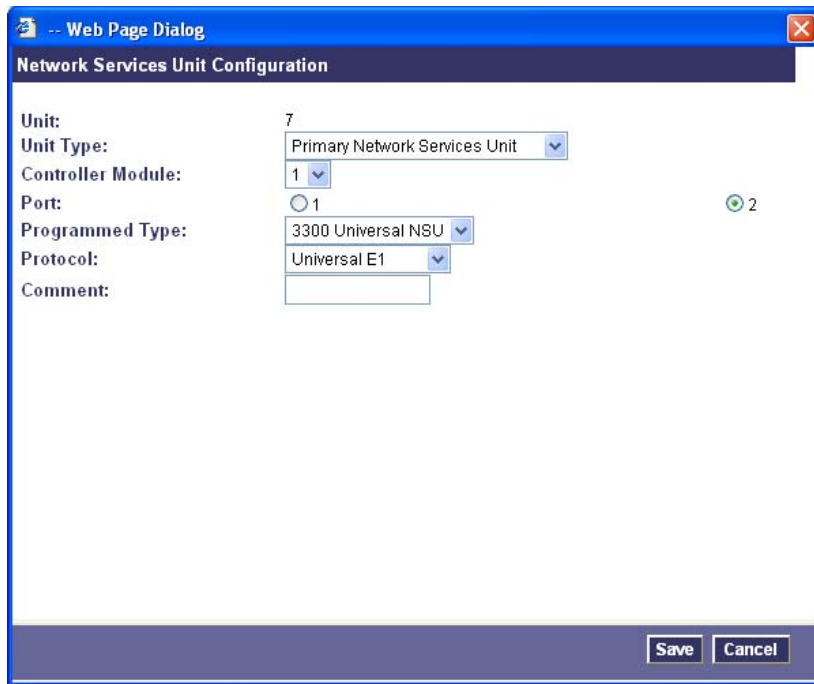


Step 3: In the System Configuration browser, expand Units/Modules and select Network Services Unit Configuration.



Select an unused Unit Number and click Change.

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



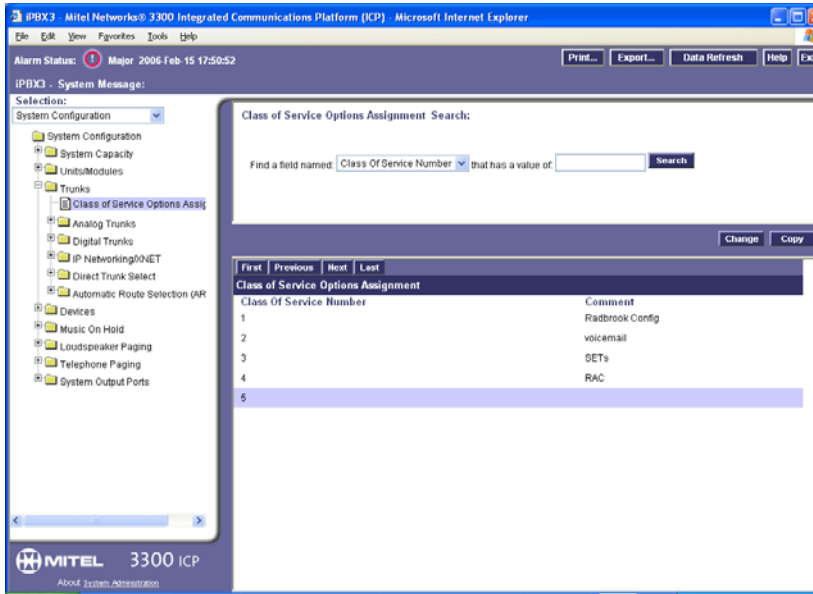
The screenshot shows a web page dialog titled "Network Services Unit Configuration". The dialog contains the following fields and values:

| | |
|--------------------|--|
| Unit: | 7 |
| Unit Type: | Primary Network Services Unit |
| Controller Module: | 1 |
| Port: | <input type="radio"/> 1 <input checked="" type="radio"/> 2 |
| Programmed Type: | 3300 Universal NSU |
| Protocol: | Universal E1 |
| Comment: | |

At the bottom of the dialog are "Save" and "Cancel" buttons.

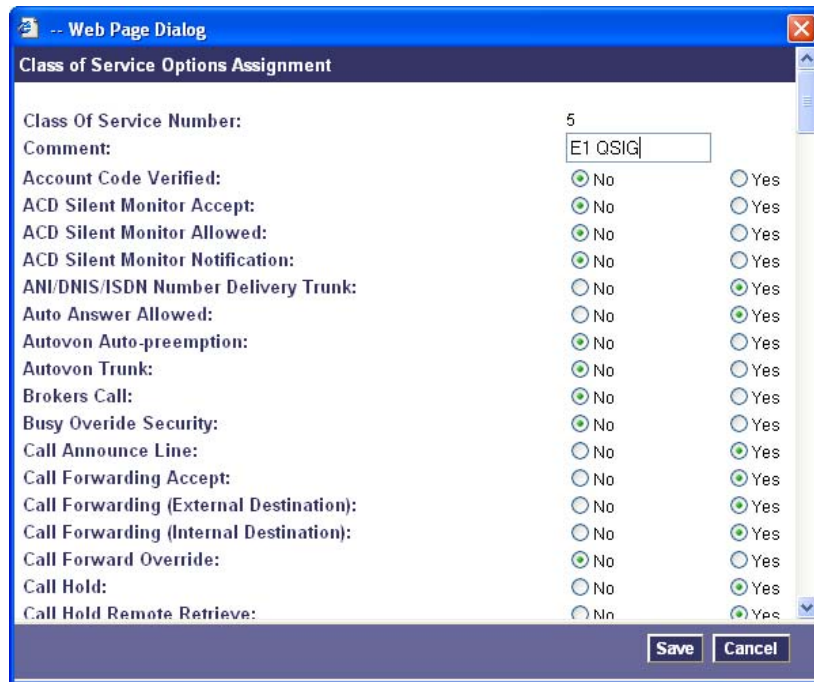
- Set Unit Type to Primary Network Services Unit.
- Set Controller Module to the Controller Module Number.
- Set Port to the port number where the NSU is installed.
- Set Programmed Type to 3300 Universal NSU.
- Set Protocol to Universal E1.
- Click Save.

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.

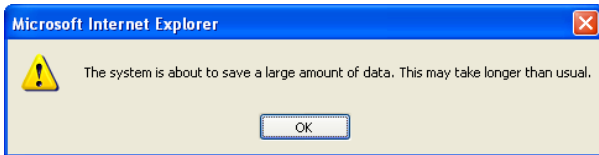
Set the following parameters to Yes:

- 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.

The screenshot shows the Mitel Networks 3300 ICP System Configuration browser. The left-hand navigation tree is expanded to 'ISDN-PRI' > 'Link Descriptor Assig'. The main content area displays a table with the following data:

| Number | Address for Message Control | BER - Maintenance Limit, 10 ⁻ⁿ | BER - Service Limit, 10 ⁻ⁿ | Data Call Alternate Digit Inversion |
|--------|-----------------------------|---|---------------------------------------|-------------------------------------|
| 1 | A | 4 | 3 | Yes |
| 2 | A | 4 | 3 | Yes |

Below the table, a detailed view of the selected entry (Number: 2) is shown:

- Number: 2
- Address for Message Control: A
- BER - Maintenance Limit, 10⁻ⁿ: 4
- BER - Service Limit, 10⁻ⁿ: 3
- Data Call Alternate Digit Inversion: Yes
- Framing Losses in 24 hrs - Maintenance Limit: 255
- Framing Losses in 24 hrs - Service Limit: 9000
- Integrated Digital Access: ISDN NODE
- Vendor Inter working Type (Philips SOPHO): No
- Satellite Link Delay: No
- Site Date - Maintenance Limit (Site Date): 4/2/01

Click Add to create a new entry.

Step 10: Configure the Link Descriptor Assignment as follows:

The screenshot shows a 'Web Page Dialog' window with the following configuration options:

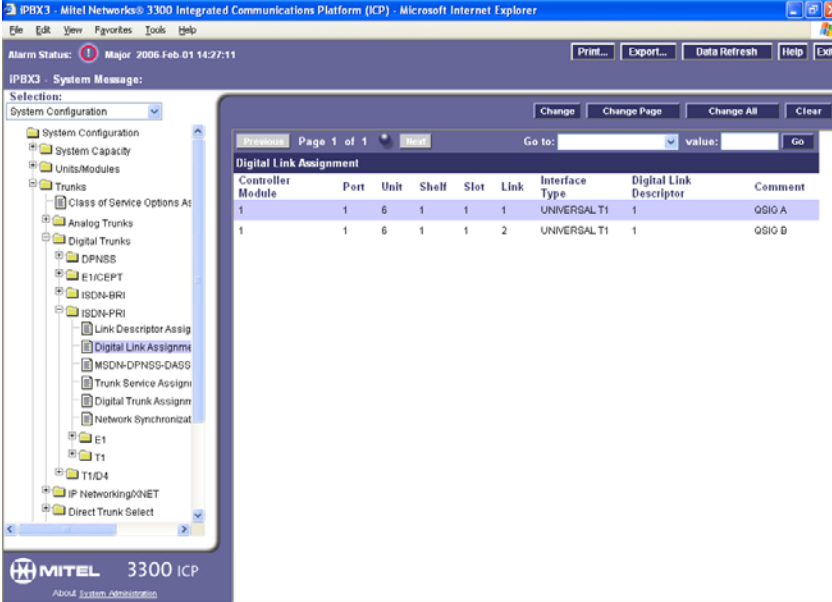
- Slip Rate - Maintenance Limit (slips/24hr.): 5000
- Slip Rate - Service Limit (slips/24hr.): 7000
- Alarm Debounce Timer - Service Limit (millisec.): 500
- Voice Encoding: ADI
- Data Encoding: ADI
- QSIG Private Network Access: No Yes
- Digital Link Fault Delay Timer (sec.): 240
- Termination Mode: LT NT
- Send Malicious Call Indication to PSTN for Tagged Calls: No Yes
- T1 Only:
 - B8ZS Zero Code Suppression: No Yes
 - Operation Mode: CSU
 - CSU Tx Line Build-Out (dB.): 0
 - DSX-1 Line Length (Ft.): 0-133
 - Extended Super Frame: No Yes
 - Inverted D channel (DPNSS only): No Yes
- E1 Only:
 - CRC-4 Enabled: No Yes
 - E1 Line Length (Ft.): 0-133
 - E1 Impedance (Ohms): 75 120

Buttons: Save, Cancel

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 Impedance (Ohms) to 120.
- Click Save.

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



The screenshot shows the MITEL 3300 ICP System Configuration browser. The left sidebar displays a tree view of system configuration options, with 'Digital Link Assignment' selected under 'ISDN-PRI'. The main content area displays a table titled 'Digital Link Assignment' with the following data:

| Controller Module | Port | Unit | Shelf | Slot | Link | Interface Type | Digital Link Descriptor | Comment |
|-------------------|------|------|-------|------|------|----------------|-------------------------|---------|
| 1 | 1 | 6 | 1 | 1 | 1 | UNIVERSAL T1 | 1 | QSIO A |
| 1 | 1 | 6 | 1 | 1 | 2 | UNIVERSAL T1 | 1 | QSIO B |

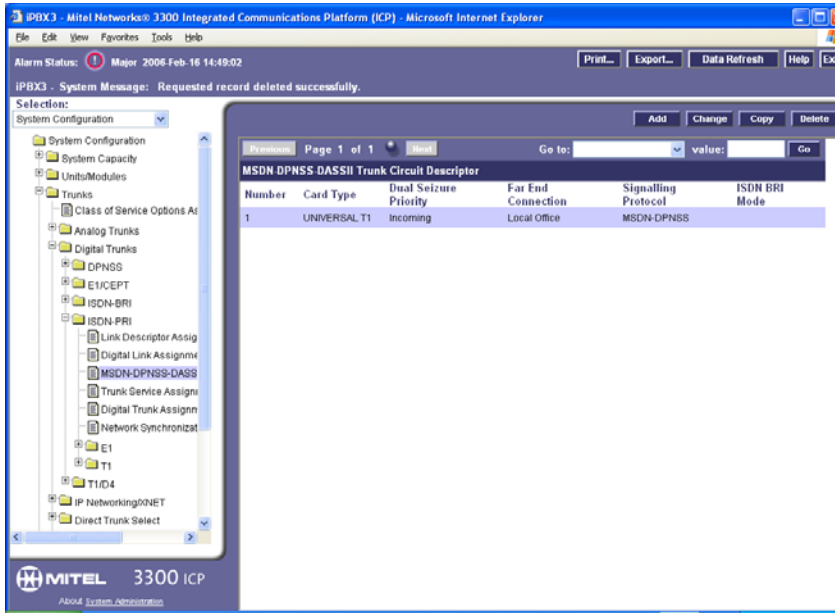
Select the link on the NSU on which the E1 Q.SIG line will be installed and click Change.

Step 12: Configure the Digital Link Assignment as follows:

| | |
|--------------------------|--------------|
| Controller Module: | 1 |
| Port: | 2 |
| Unit: | 7 |
| Shelf: | 1 |
| Slot: | 1 |
| Link: | 1 |
| Interface Type: | UNIVERSAL E1 |
| Digital Link Descriptor: | 2 |
| Comment: | E1 QSIG |

- 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.



The screenshot shows the Mitel Networks 3300 ICP System Configuration browser. The left sidebar contains a tree view with the following structure:

- System Configuration
 - System Capacity
 - Units/Modules
 - Trunks
 - Class of Service Options As
 - Analog Trunks
 - Digital Trunks
 - DPNSS
 - E1/CEPT
 - ISDN-BRI
 - ISDN-PRI
 - Link Descriptor Assign
 - Digital Link Assignme
 - MSDN-DPNSS-DASSII**
 - Trunk Service Assign
 - Digital Trunk Assignm
 - Network Synchronizat
 - E1
 - T1
 - T1/D4
 - IP Networking/NET
 - Direct Trunk Select

The main area displays a table titled "MSDN DPNSS DASSII Trunk Circuit Descriptor" with the following data:

| Number | Card Type | Dual Seizure Priority | Far End Connection | Signalling Protocol | ISDN BRI Mode |
|--------|--------------|-----------------------|--------------------|---------------------|---------------|
| 1 | UNIVERSAL T1 | Incoming | Local Office | MSDN-DPNSS | |

Click **Add** to create a new entry.

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

The screenshot shows a web page dialog box titled "MSDN-DPNSS-DASSII Trunk Circuit Descriptor". The dialog contains the following fields and options:

- Number:** A text input field containing the value "2".
- Card Type:** A dropdown menu set to "UNIVERSAL E1".
- Dual Seizure Priority:** Two radio buttons: "Incoming" (selected) and "Outgoing".
- Far End Connection:** A dropdown menu set to "Local Office".
- Signalling Protocol:** Two radio buttons: "MSDN-DPNSS" (selected) and "DASS II".
- ISDN BRI Mode:** A dropdown menu with a blank entry selected.

At the bottom right of the dialog, there are "Save" and "Cancel" buttons.

- 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**.

The screenshot shows the Mitel Networks 3300 ICP System Configuration browser. The left sidebar displays a tree view of system configuration options, with 'Trunk Service Assignment' selected. The main content area displays a table of Trunk Service Assignments. The table has the following columns: Trunk Service Number, Release Link Trunk, Class of Service, Class of Restriction, Baud Rate, Intercept Number, and Trunk Label. The table contains 10 rows of data. Row 5 is highlighted in blue. Below the table, there is a summary section for the selected row (Trunk Service Number: 5) with the following values: Release Link Trunk: No, Class of Service: 5, Class of Restriction: 1, Baud Rate: 300, Intercept Number: 1, and various other parameters.

| Trunk Service Number | Release Link Trunk | Class of Service | Class of Restriction | Baud Rate | Intercept Number | Trunk Label |
|----------------------|--------------------|------------------|----------------------|-----------|------------------|-------------|
| 1 | No | 1 | 1 | 300 | 1 | Radbrook |
| 2 | No | 1 | 1 | 300 | 1 | 3 to 4 |
| 3 | No | 1 | 1 | 300 | 1 | 3 to 5 |
| 4 | No | 1 | 1 | 300 | 1 | ASU Lp Bk |
| 5 | No | 5 | 1 | 300 | 1 | E1 QSIG |
| 6 | No | 1 | 1 | 300 | 1 | |
| 7 | No | 1 | 1 | 300 | 1 | |
| 8 | No | 1 | 1 | 300 | 1 | |
| 9 | No | 1 | 1 | 300 | 1 | |
| 10 | No | 1 | 1 | 300 | 1 | |

Trunk Service Assignment

Trunk Service Number: 5
Release Link Trunk: No
Class of Service: 5
Class of Restriction: 1
Baud Rate: 300
Intercept Number: 1
Non-dial In Trunks Answer Point - Day:
Non-dial In Trunks Answer Point - Night 1:
Non-dial In Trunks Answer Point - Night 2:
Dial In Trunks Incoming Digit Modification - Absorb: 0
Dial In Trunks Incoming Digit Modification - Insert:

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

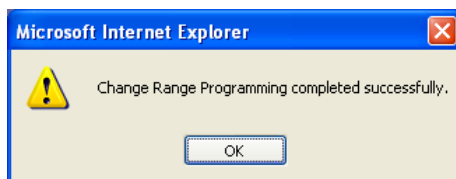
Step 16: Configure the Trunk Service Assignment as follows:

| | |
|--|---------|
| Trunk Service Number: | 5 |
| Release Link Trunk: | No |
| Class of Service: | 5 |
| Class of Restriction: | 1 |
| Baud Rate: | 300 |
| Intercept Number: | 1 |
| Non-dial In Trunks Answer Point - Day: | |
| Non-dial In Trunks Answer Point - Night 1: | |
| Non-dial In Trunks Answer Point - Night 2: | |
| Dial In Trunks Incoming Digit Modification - Absorb: | 0 |
| Dial In Trunks Incoming Digit Modification - Insert: | |
| Trunk Label: | E1 QSIG |

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.

The screenshot shows the Mitel Networks 3300 ICP System Configuration browser. The left sidebar displays a tree view of system configuration options, with 'Digital Trunk Assignment' selected. The main content area displays a table of Digital Trunk Assignments. The table has columns for Cabinet, Shelf, Slot, Circuit, Card Type, and Trunk Number. The row with Cabinet 7, Shelf 1, Slot 1, and Circuit 1 is highlighted. Below the table, a summary of the selected assignment is shown.

| Cabinet | Shelf | Slot | Circuit | Card Type | Trunk Number |
|---------|-------|------|---------|--------------|--------------|
| 6 | 1 | 1 | 21 | UNIVERSAL T1 | 120 |
| 6 | 1 | 1 | 22 | UNIVERSAL T1 | 121 |
| 6 | 1 | 1 | 23 | UNIVERSAL T1 | 122 |
| 7 | 1 | 1 | 1 | UNIVERSAL E1 | |
| 7 | 1 | 1 | 2 | UNIVERSAL E1 | |
| 7 | 1 | 1 | 3 | UNIVERSAL E1 | |
| 7 | 1 | 1 | 4 | UNIVERSAL E1 | |
| 7 | 1 | 1 | 5 | UNIVERSAL E1 | |
| 7 | 1 | 1 | 6 | UNIVERSAL E1 | |
| 7 | 1 | 1 | 7 | UNIVERSAL E1 | |

Digital Trunk Assignment

Cabinet: 7
Shelf: 1
Slot: 1
Circuit: 1
Card Type: UNIVERSAL E1
Trunk Number:
Trunk Service Number:
DTS Service Number:
Circuit Descriptor Number:
Interconnect Number:

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:

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

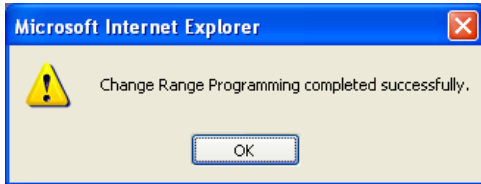
| Field Name | Change action | Value to change | Increment by |
|----------------------------|---------------------|-----------------|--------------|
| Cabinet: | - | 7 | - |
| Shelf: | - | 1 | - |
| Slot: | - | 1 | - |
| Circuit: | - | 1 | - |
| Card Type: | - | UNIVERSALE1 | - |
| Trunk Number: | Increment | 150 | by 1 |
| Trunk Service Number: | Change all to | 5 | |
| DTS Service Number: | Leave all unchanged | | |
| Circuit Descriptor Number: | Change all to | 2 | |
| Interconnect Number: | Change all to | 1 | |

Buttons: Preview, Save, Cancel

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

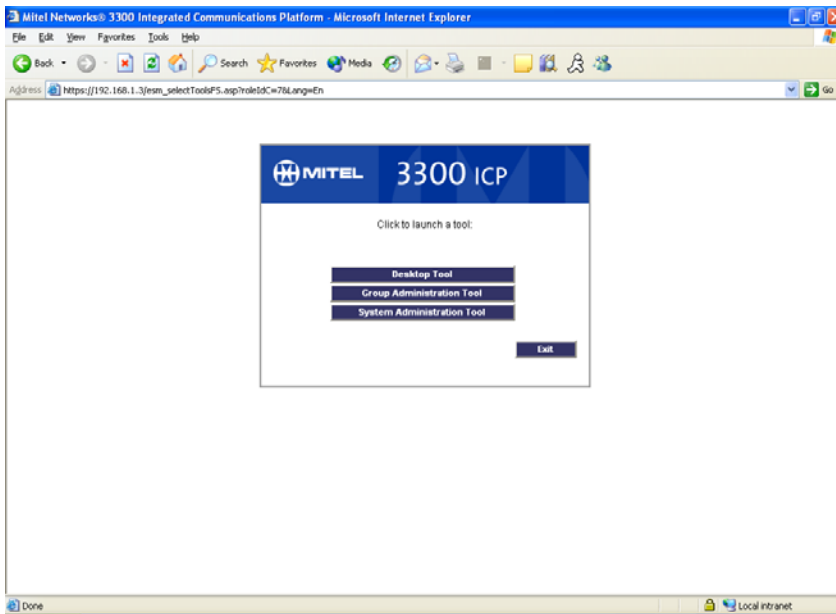
- Set Enter the number of records to change to 30.
- Set Trunk Number:
 - under Change action to Increment.
 - under Value to change to an unused trunk number.
 - under Increment by to 1.
- Set Trunk Service Number:
 - under Change action to Change all to.
 - under Value to change to the trunk service 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.



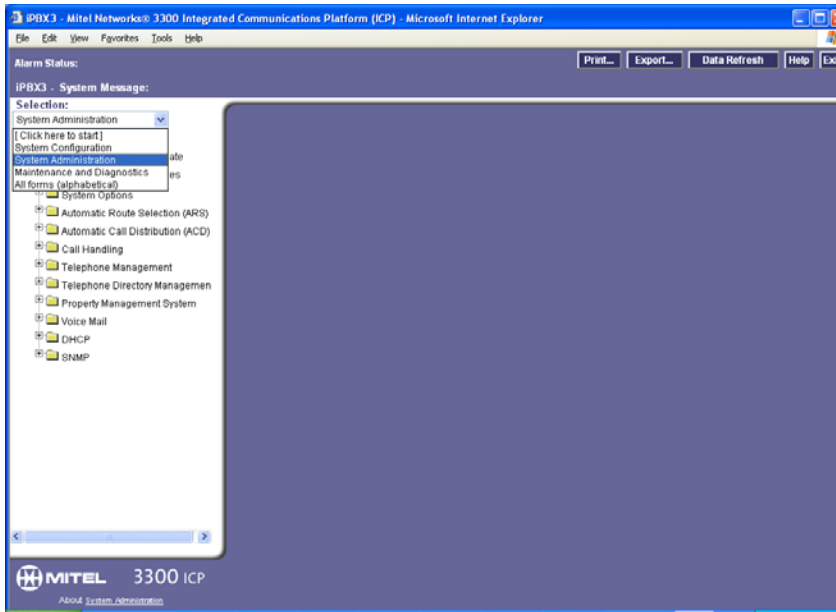
6.1.1.2 Automatic Route Selection (ARS)

Step 1: Log into the Mitel 3300 ICP system.

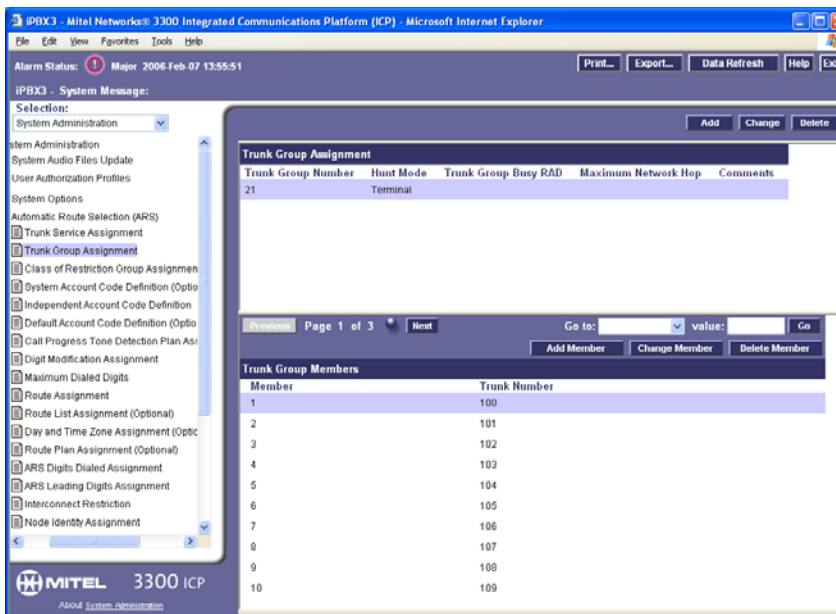


Click the **System Administration Tool** option.

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:

| Field Name | Value to Add | Increment by |
|-----------------------|--|--------------|
| Trunk Group Number: | 22 | |
| Hunt Mode: | <input checked="" type="radio"/> Terminal <input type="radio"/> Circular | - |
| Trunk Group Busy RAD: | | |
| Maximum Network Hop: | | |
| Comments: | | - |

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.

The screenshot shows the Mitel 3300 ICP System Administration web interface. The top navigation bar includes 'System Administration' and various system options. The main content area is divided into two sections:

- Trunk Group Assignment:** A table with columns: Trunk Group Number, Hunt Mode, Trunk Group Busy RAD, Maximum Network Hop, and Comments. One entry is visible: Trunk Group Number 21, Hunt Mode Terminal.
- Trunk Group Members:** A table with columns: Member and Trunk Number. It lists members 1 through 10, each assigned to a specific trunk number (100-109).

Navigation controls include 'Add Member', 'Change Member', and 'Delete Member' buttons. The interface also shows a 'Page 1 of 3' indicator and a search field.

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

Range Programming -- Web Page Dialog

Add Range Programming - *Trunk Group Members* Help

This form allows you to add one or more records.

1. Enter the number of records to add:

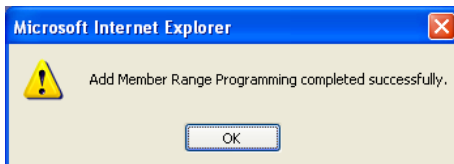
2. Define the Add Range Programming Pattern:

| Field Name | Value to Add | Increment by |
|---------------|----------------------------------|--------------------------------|
| Trunk Number: | <input type="text" value="150"/> | <input type="text" value="1"/> |

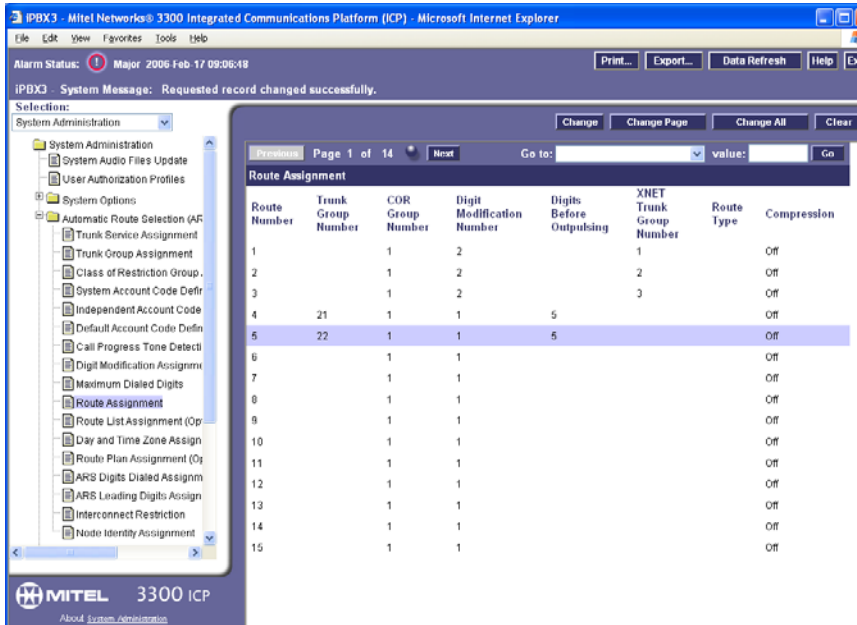
Preview Save Cancel

- 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.

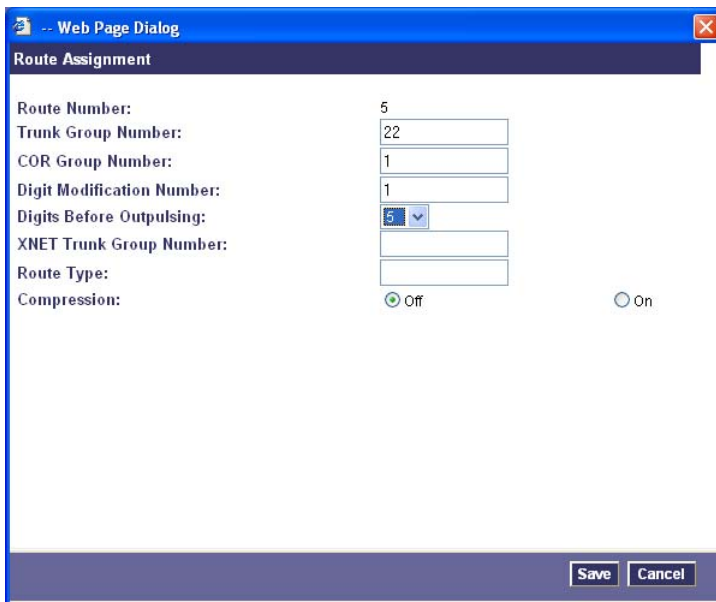


Step 9: In the System Administration browser, select Route Assignment.



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.

The screenshot shows the IPBX3 System Administration web interface. The left sidebar contains a tree view of system administration options, with 'ARS Digits Dialed Assignment' selected. The main content area displays a table titled 'ARS Digits Dialed Assignment' with the following data:

| Digits Dialed | Number of Digits to Follow | Termination Type | Termination Number |
|---------------|----------------------------|------------------|--------------------|
| 8 | 2 | Route | 21 |
| 92 | 4 | Route | 1 |
| 94 | 4 | Route | 2 |
| 95 | 4 | Route | 3 |

The interface also includes a navigation menu at the top with options like 'Print...', 'Export...', 'Data Refresh', 'Help', and 'Exit'. The bottom left corner features the MITEL 3300 ICP logo and 'About System Administration' text.

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

| Field Name | Value to Add | Increment by |
|-----------------------------|--------------|--------------|
| Digits Dialed: | 7 | |
| Number of Digits to Follow: | 2 | - |
| Termination Type: | Route | - |
| Termination Number: | 5 | |

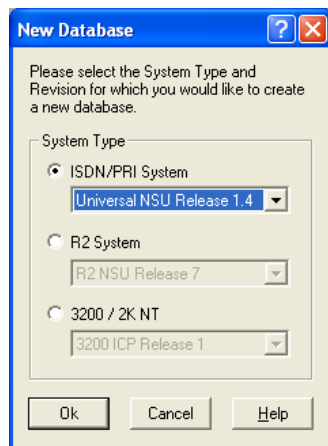
- Set Enter the number of records to add to 1.
- Under Value to Add set:
 - Digits Dialed to the leading digit(s) to be assigned.
 - Number of Digits to Follow to 2.
 - Termination Type to Route.
 - Termination Number to the route number that was defined in Step 10 of the Automatic Route Selection.
-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

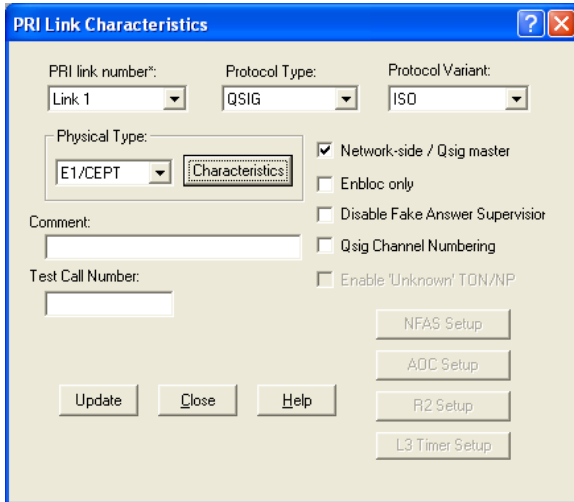
The screenshot shows the 'Site Options' configuration window. It is divided into several sections:

- System Type:** Radio buttons for PRI Card, 3200 ICP, Gateway, R2 Card, and Universal NSU (selected).
- Connected Platform:** Radio buttons for SX-2000, SX-2000 Light, SX-200 Light, 3300 ICP (selected), and SX-200 EL/ML/ICP.
- Site ID:** A text input field.
- Passwords:** Six empty text input fields.
- Options:** Checkboxes for Min/Max, Automated Min/Max, NFAS, D-Channel Backup, Network-side Interface (checked), and Qsig (checked).
- Configurable PRA Links:** A text input field containing the number '2'.

At the bottom of the window are three buttons: 'Update', 'Close', and 'Help'.

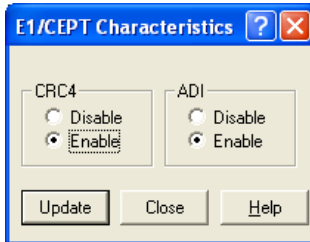
- Under System Type, select Universal NSU.
- Under Connected Platform, select 3300 ICP.
- Under Options, select Network-side Interface and Qsig.
- Click Update and then Close.

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



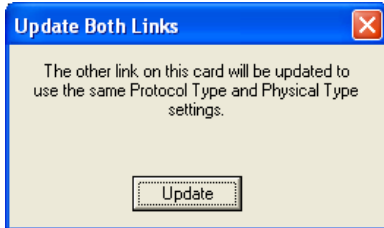
Note: Parameters not listed below can be left at their default settings.

- 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.

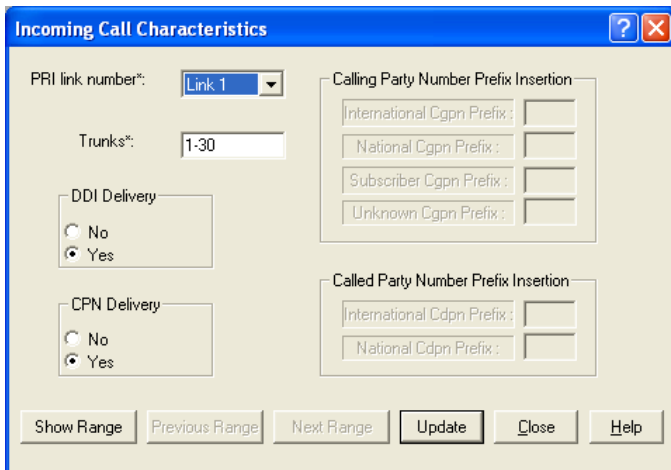
- The Update Both Links dialog box appears.



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.

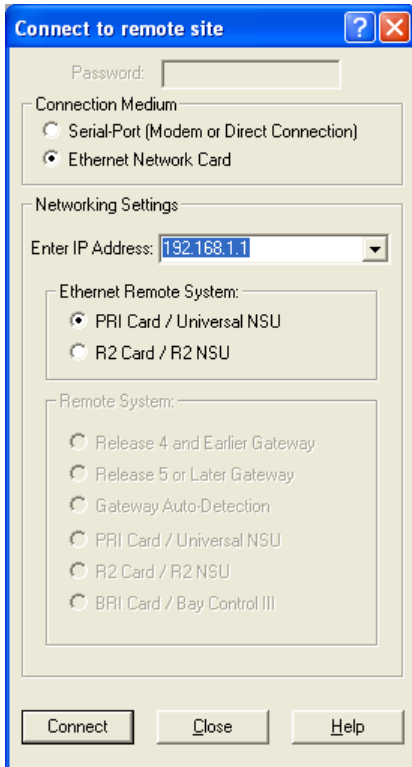
Step 5: Click Config > Outgoing Call Characteristics to configure the outgoing call characteristics as follows:

The screenshot shows the 'Outgoing Call Characteristics' configuration window. At the top, 'PRI link number:' is set to 'Link 1' and 'Numbering Plan/Type' is empty. The window is divided into several sections: 'Fixed Bearer Capability' with 'Voice' set to 'Speech' and 'Data' to 'Null'; 'Fixed High Layer Compatibility' with a single 'Unassigned' dropdown; 'Fixed CLIR' with 'Voice' and 'Data' both set to 'Null'; 'Per Call Bearer Capability' with a list of options (0: Speech, 1: 3.1 kHz, 2: UDI, 3: RDI, 5: Video); 'Per Call High Layer Compatibility' with a 2x5 grid of 'Unassigned' dropdowns; 'Per Call CLIR' with '0: Allow' and '1: Restrict' options; and 'Trunk CPN Tables' with 'Voice' and 'Data' buttons. At the bottom are 'Update', 'Close', and 'Help' buttons.

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

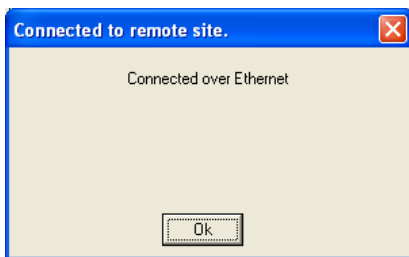
- Set PRI link number* to the link number that has Q.SIG installed.
- Set Voice to Speech.
- Click Update and then Close.

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

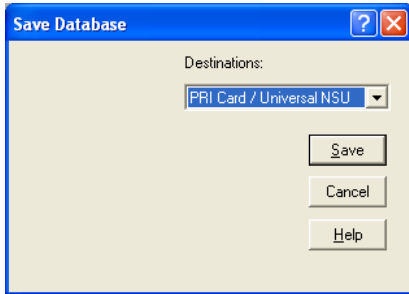


Note: Parameters not listed below can be left at their default settings.

- 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.

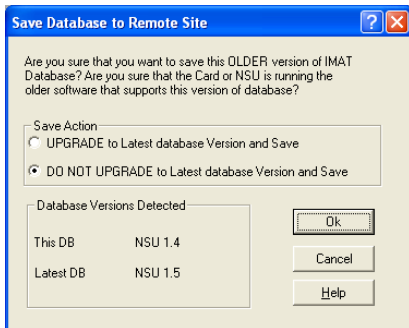


Step 7: Click **File > Save > Database** to save the database as follows:

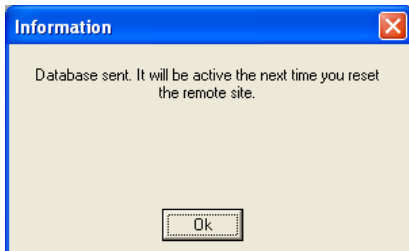


- Set Destinations to PRI Card / Universal NSU.
- Click Save.

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.

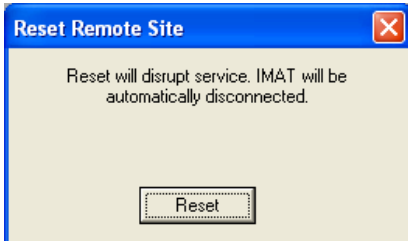


The Information dialog box appears.

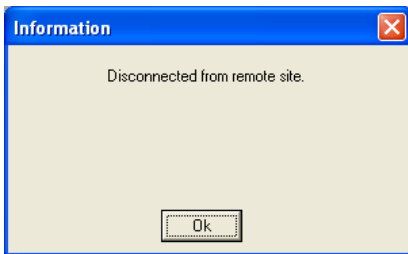


Click **OK**.

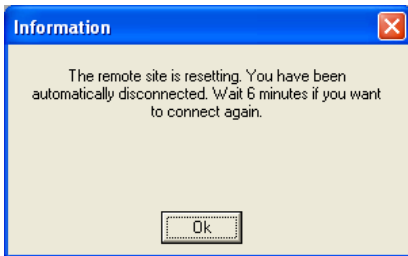
Click Maintenance > Remote Site Reset.



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

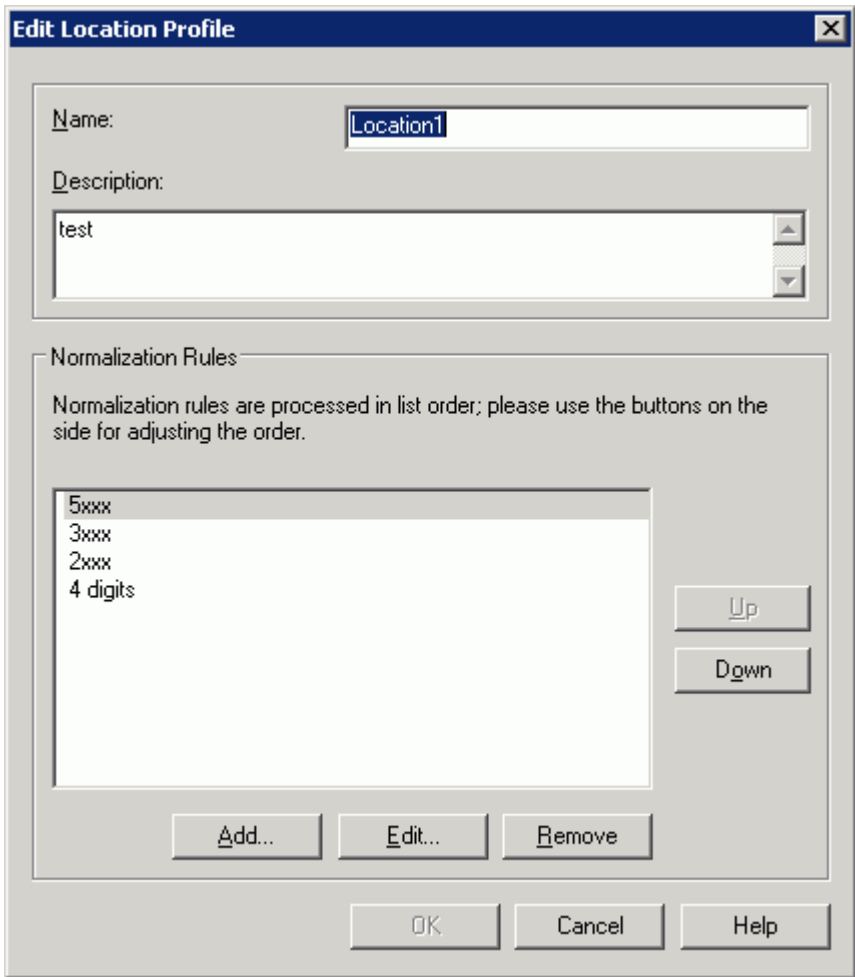
7.1 Steps for configuring Microsoft® OCS

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` from the menu provided. Edit a location profile as shown in the following example:



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

Edit Phone Number Normalization Rule

Name:

Click to copy an existing rule.

Description:

Translation

Phone pattern regular expression:

Translation pattern regular expression:

Valid translation characters are +, numbers, and \$. Example: +1425\$1.

Click Helper for assistance in creating common phone number regular expressions and translations.

Test translation

To test the translation, enter a sample dialed number. If it matches the phone pattern, the translation will be shown.

Sample dialed number:

Translated number:

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` from the menu provided. Edit a route as shown in the example below.

Edit Route

Name:

Description:

A route requires a target phone number regular expression, one or more gateways, and one or more phone usages.

Target phone numbers:

Target regular expression:

Helper...

Gateways

| Address |
|---------------------------|
| dmg4000.BufOCS.local:5061 |

Add... Remove

Phone usages

| |
|---------------|
| Default Usage |
|---------------|

Configure...

OK Cancel Help

This entry routes numbers with or without “+” prefix followed by any digits to Microsoft® Mediation Server dm4000.bufocs.local.

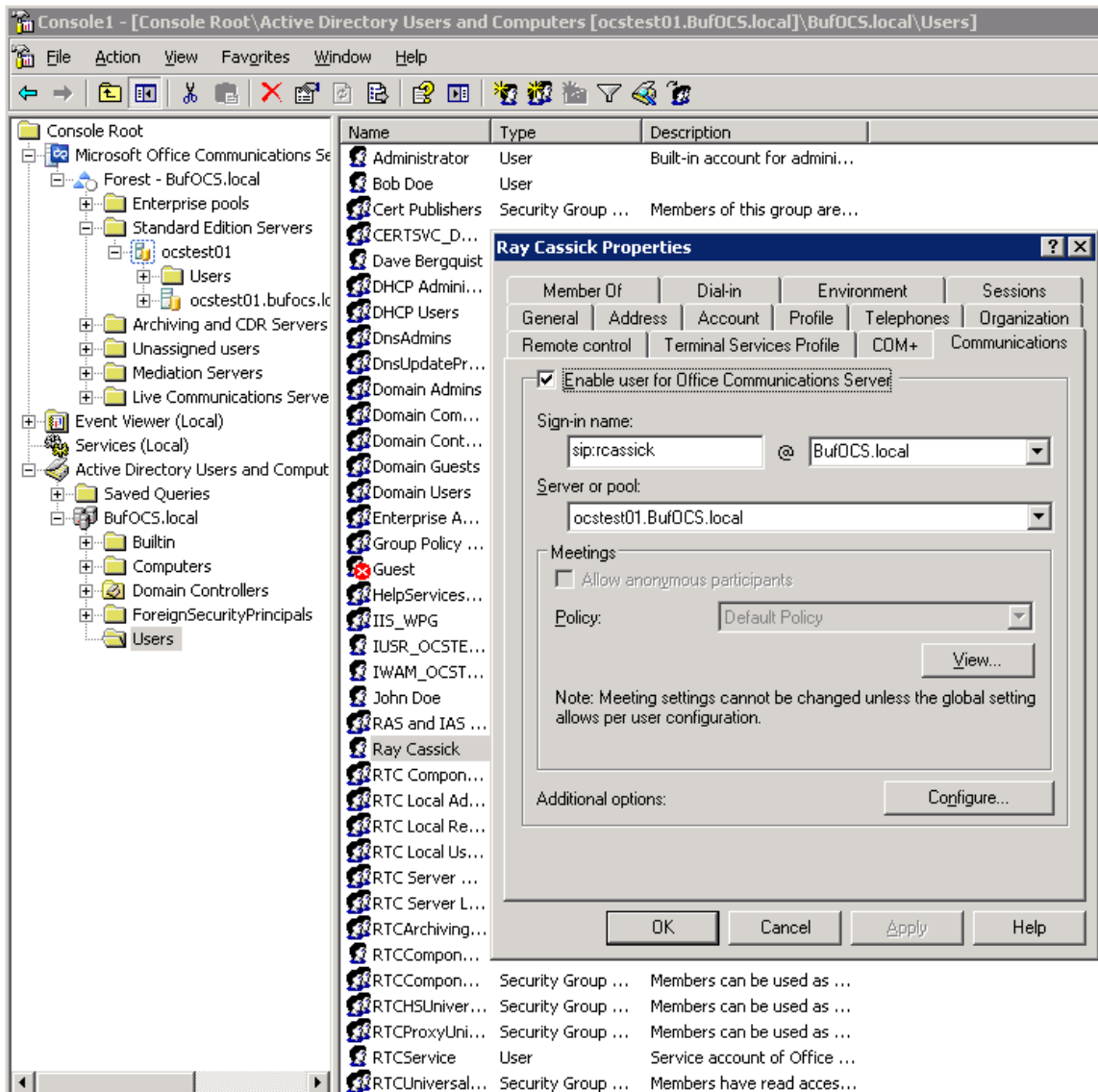
Restart the Front End Services for the above changes to take effect, including all normalization rules. This can be done from the window Services.

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

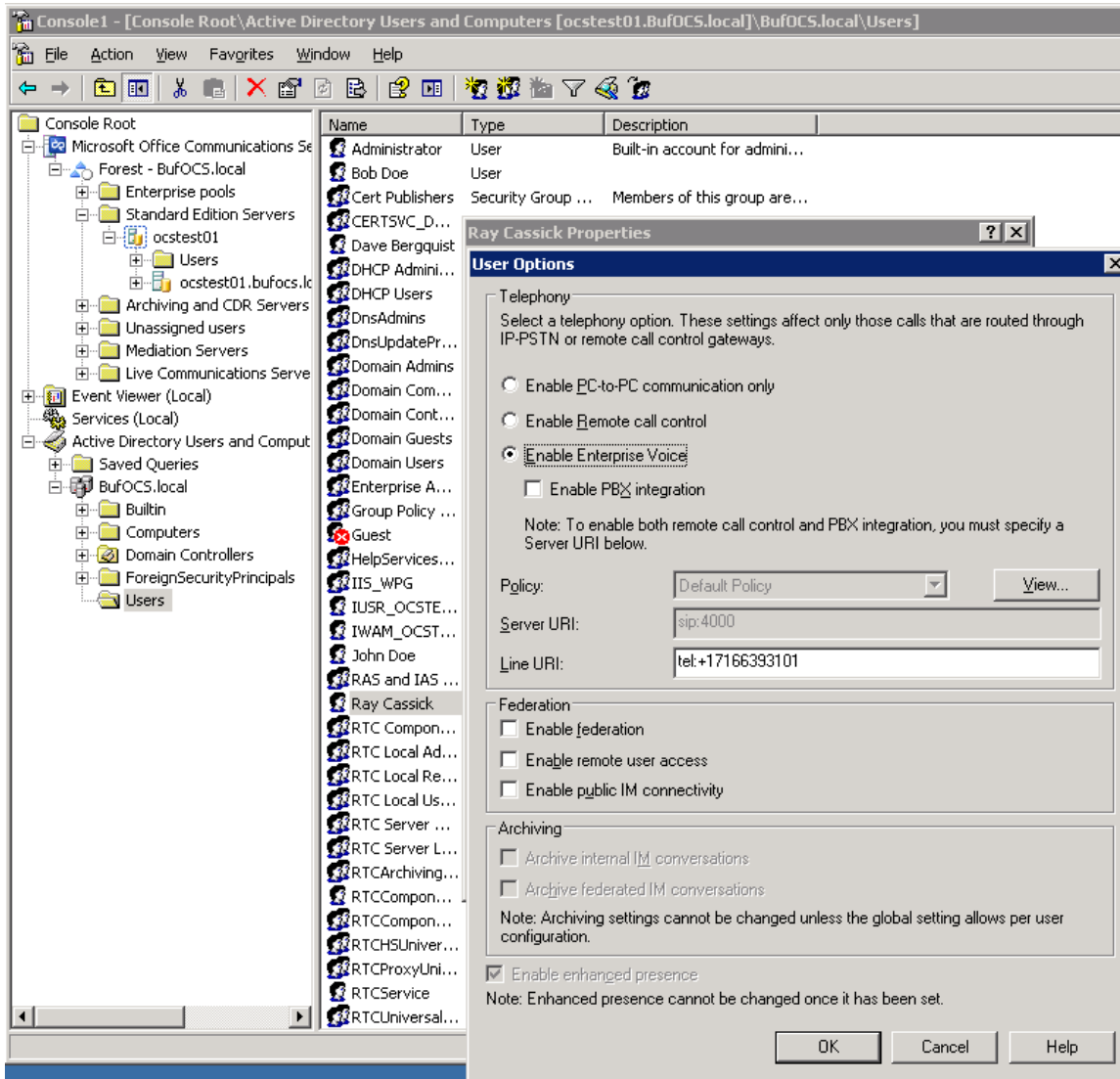
or matched, outbound calls will fail. In this case, Dialogic® Diva® Diagnostics trace will not receive an outbound SIP message, since the call will not yet have reached the SIP gateway.

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 dialplan 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 Server`. Follow the steps in the Wizard to complete the setup:

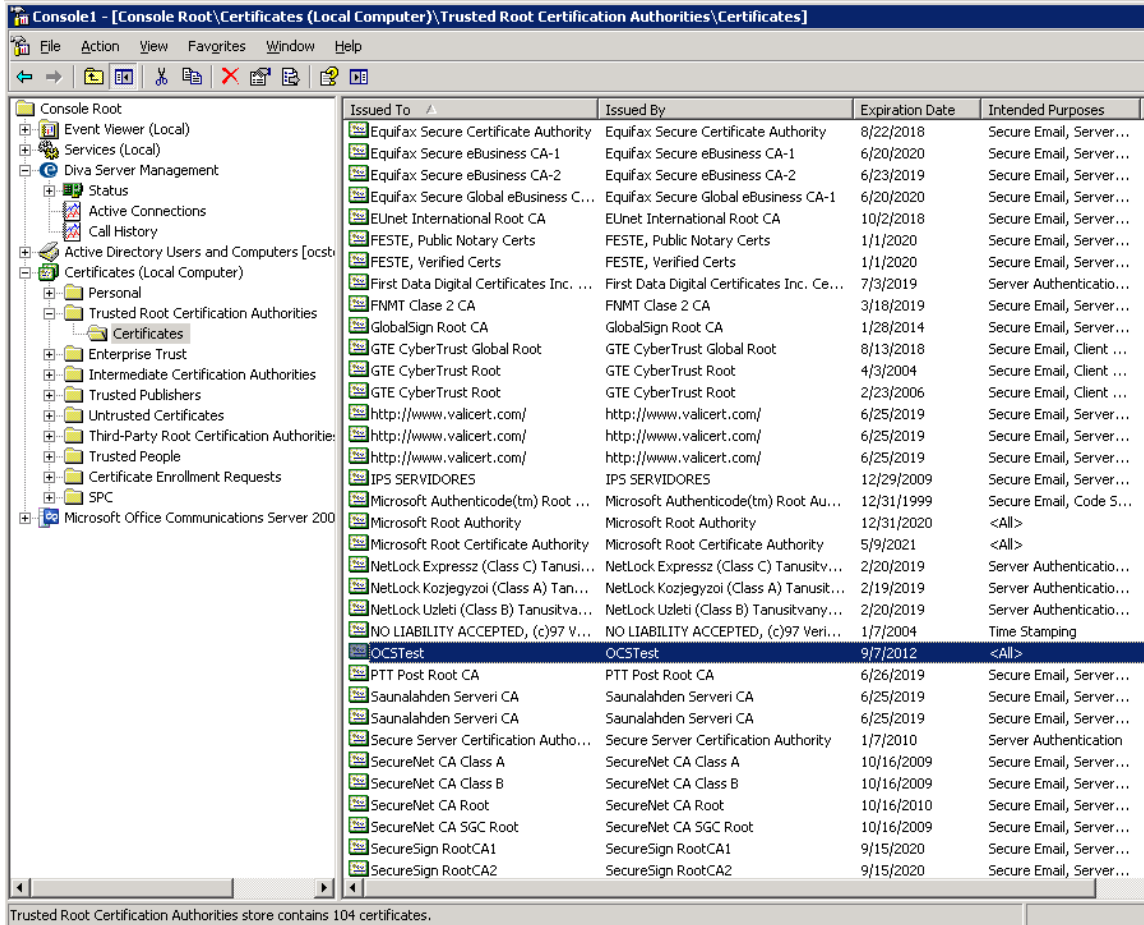
Step 1: Install the Microsoft® Mediation Server software.

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.



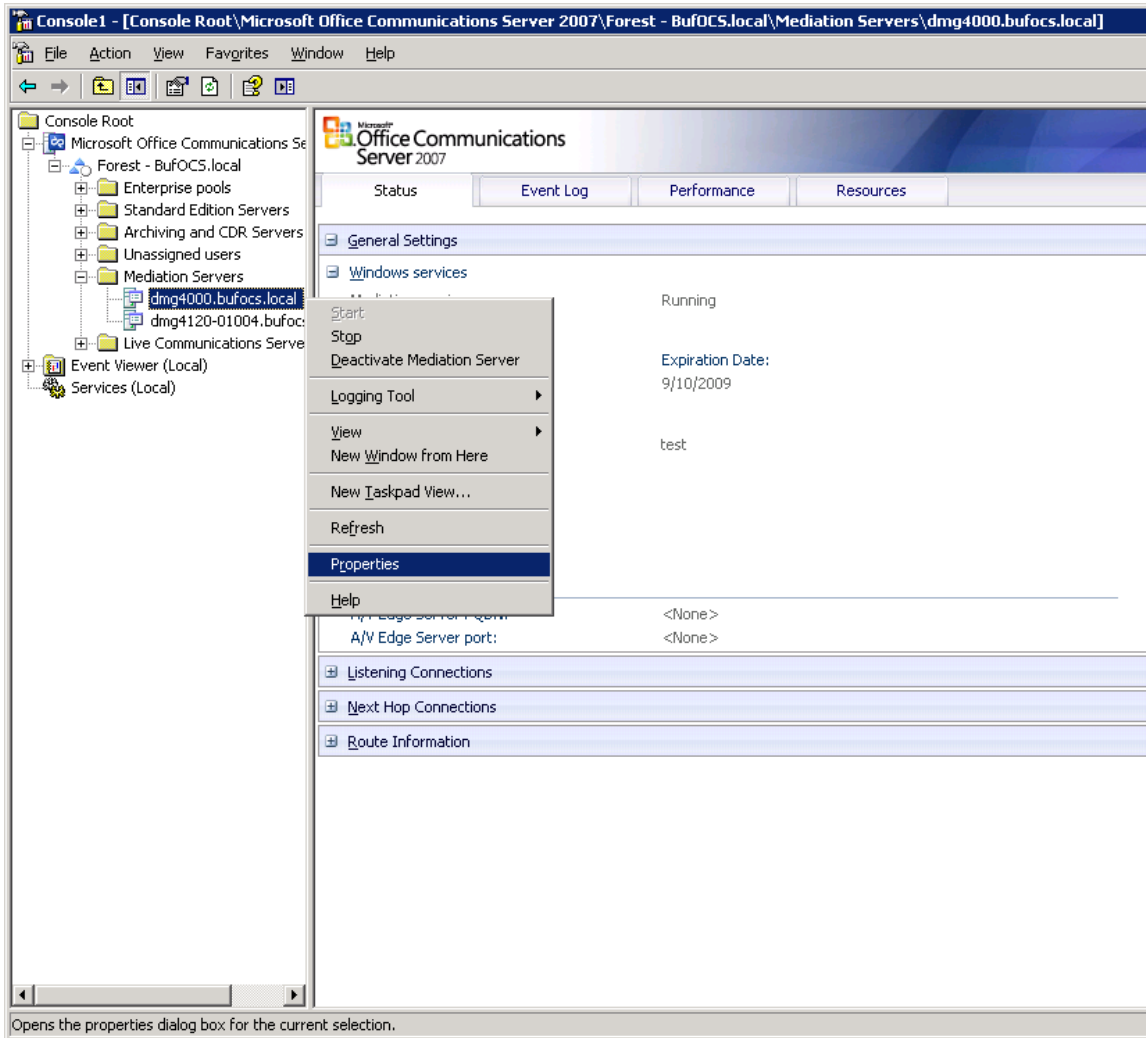
4. Create the certificate request for the Microsoft® Mediation Server:

- 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.

The screenshot shows a Windows-style dialog box titled "dmg4000.bufocs.local Properties". It has three tabs: "General", "Next Hop Connections", and "Certificate". The "General" tab is selected. The dialog contains the following fields and controls:

- Mediation Server:** Represented by a server icon.
- EQDN:** A text field containing "dmg4000.BufOCS.local".
- Communications Server listening IP address:** A dropdown menu showing "192.168.0.106".
- Gateway listening IP address:** A dropdown menu showing "192.168.0.106".
- A/V Edge Server:** A dropdown menu showing "(None)".
- Default location profile:** A dropdown menu showing "Location1" and a "View" button.
- Media port range:** Two text boxes containing "60000" and "64000" with a "to" label between them.

At the bottom of the dialog are four buttons: "OK", "Cancel", "Apply", and "Help".

Click the Next Hop Connections tab and configure the following:

dmg4000.bufocs.local Properties

General | **Next Hop Connections** | Certificate

Office Communications Server next hop
Specify the Office Communications Server used for routing inbound PSTN calls.

EQDN:
ocstest01.BuFOCS.local

Port: 5061

PSTN Gateway next hop
Specify the PSTN gateway connected to this server.

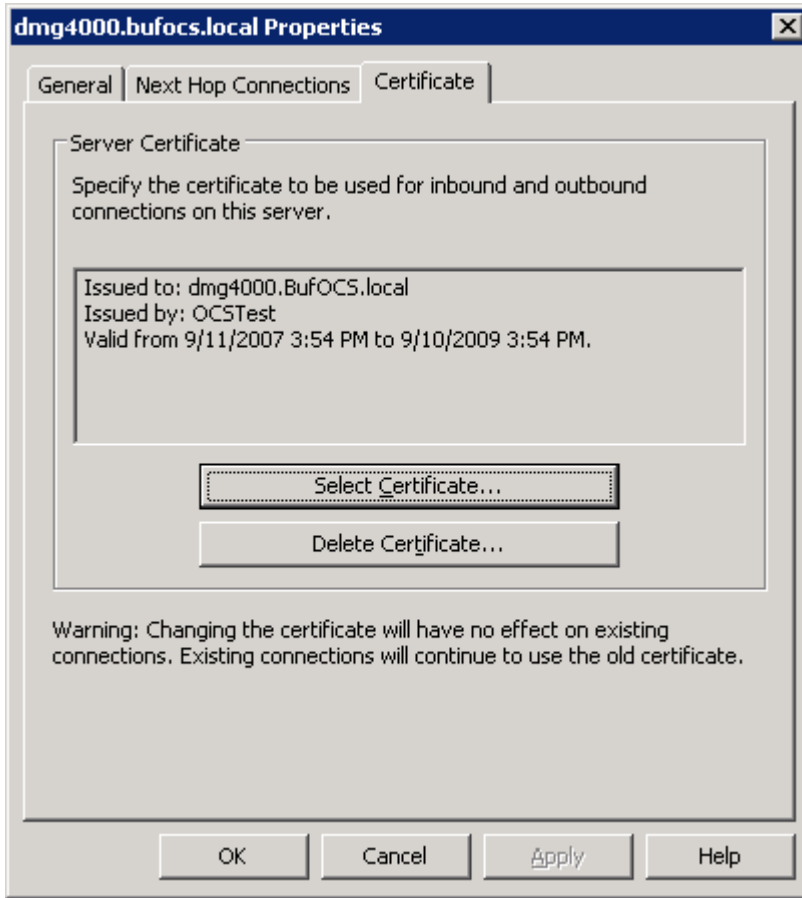
IP address: 192 . 168 . 0 . 106

Port: 9803

OK Cancel Apply Help

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

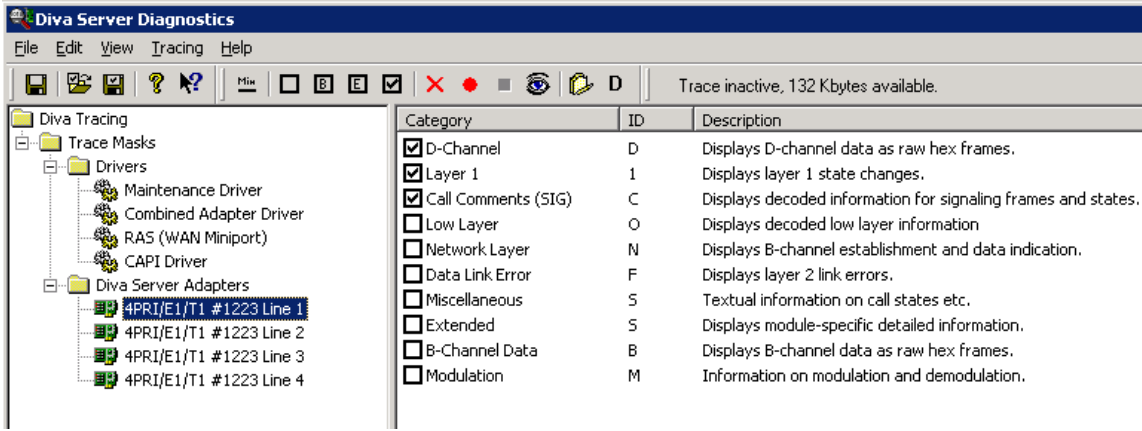
10.1 Important Debugging Tools




- **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 from 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 from 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
```

