Dialogic® 4000 Media Gateway Series Integration Note

Avaya S8500

August 2008 64-0356-01

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) 2007 application.

2. Configuration Details

Listed below are details of the PBX and gateways used in the testing on which this document is based.

2.1 PBX

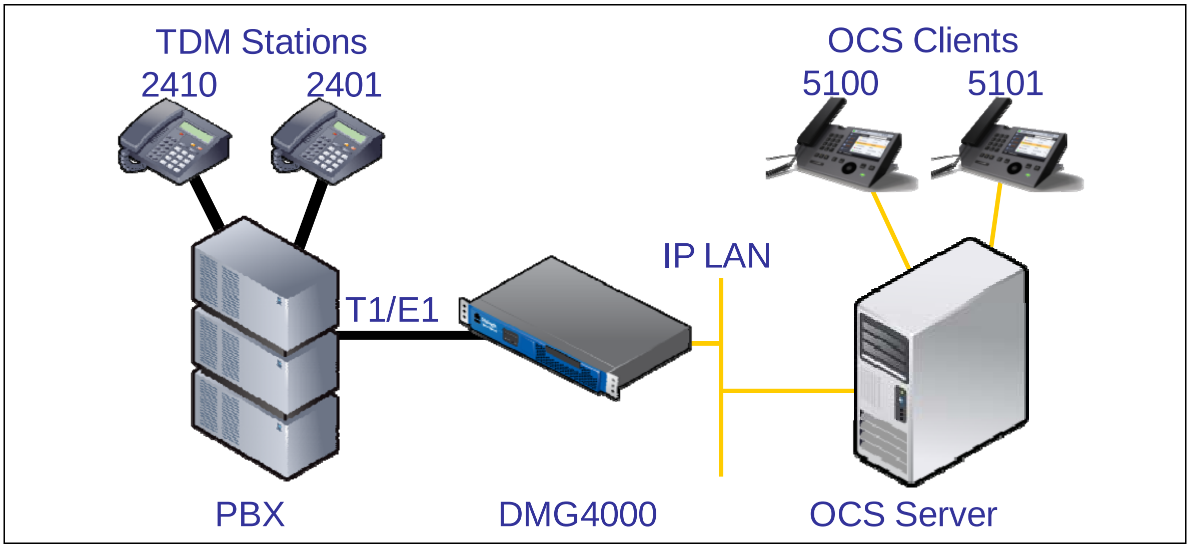
PBX Vendor	Avaya
Model(s)	S8500
Software Version(s)	Comm manager 3.0 – R013x.00.1.346.0
Additional Notes	N/A

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	T1 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 T1 Q.SIG configuration as documented, the TN464F DS1 interface card is needed.

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 the 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 a T1 Q.SIG interface.

4. Summary of Limitations

No limitations noted as of the most recent update to this document.

5. Gateway Setup Notes

Steps for setting up the gateway:

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

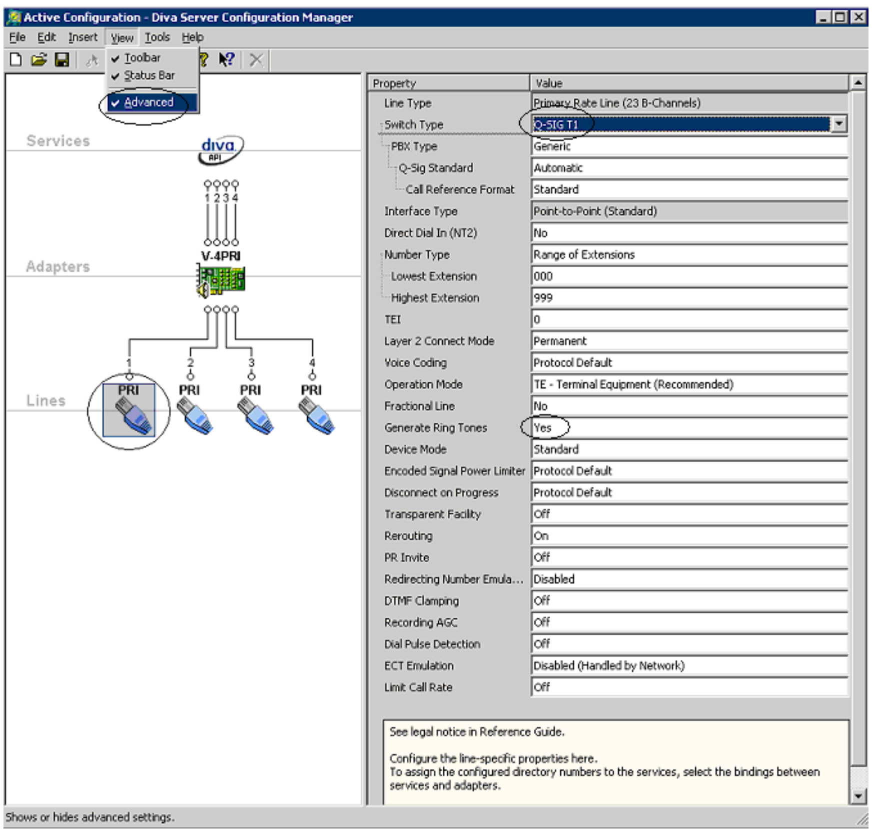
5.1 Dialogic® Diva® Media Board Configuration

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`.

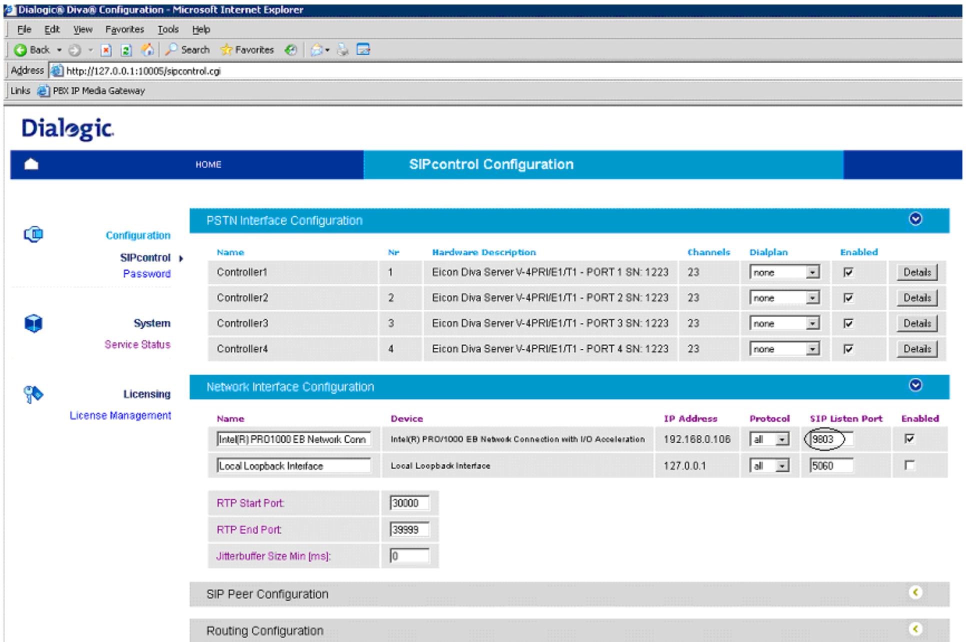
On the main page, click the `SIPControl` link to display the different configuration menus.
The `PSTN Interface Configuration` section should automatically include all ports detected in the system.

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

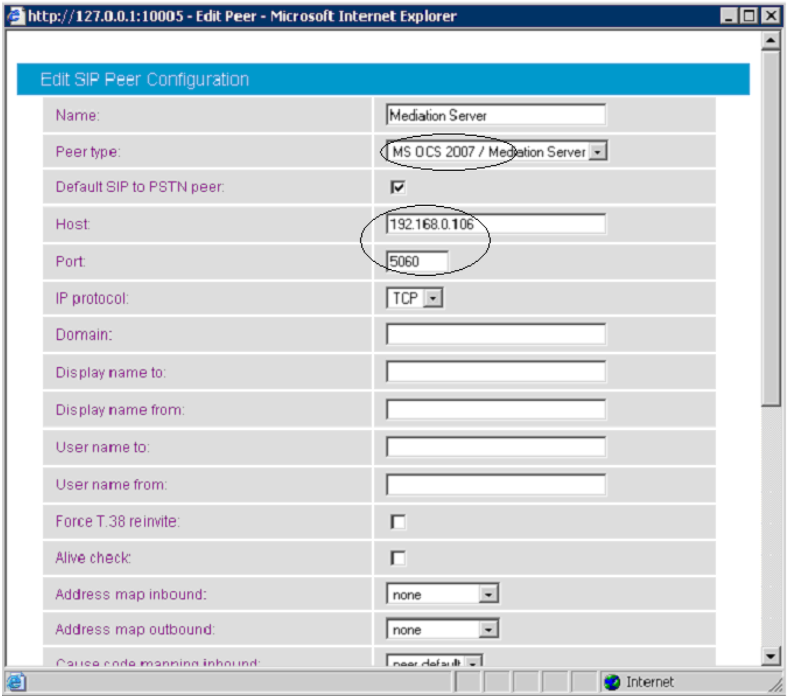
The following is a typical configuration.



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.

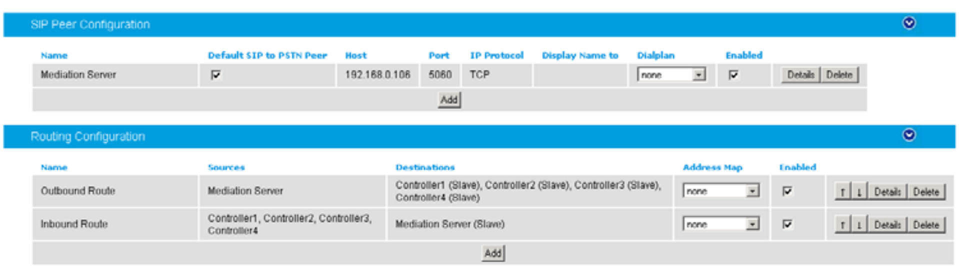
5.2.2 SIP Peer Configuration

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.



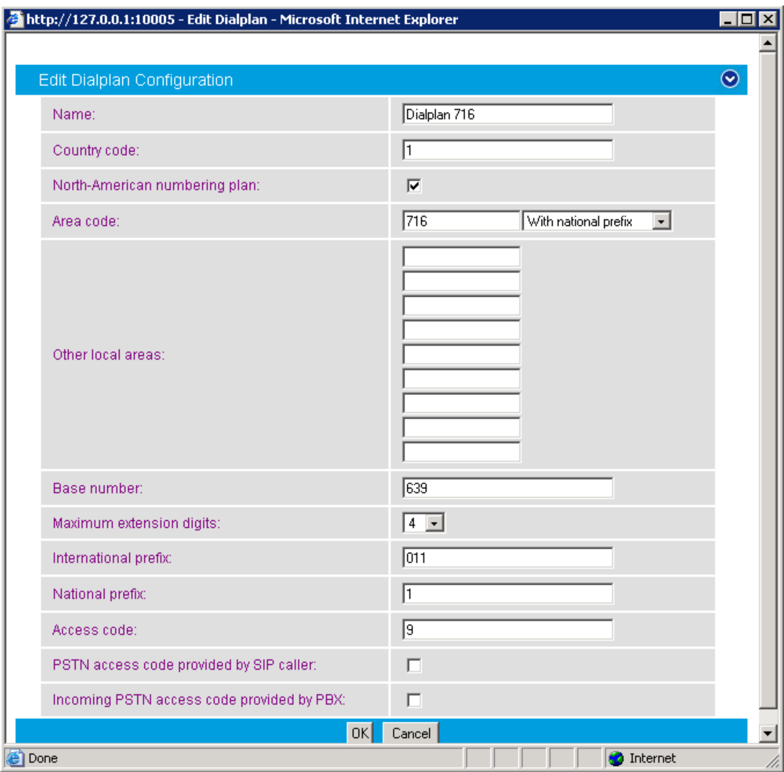
5.2.4 Number Normalization

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.

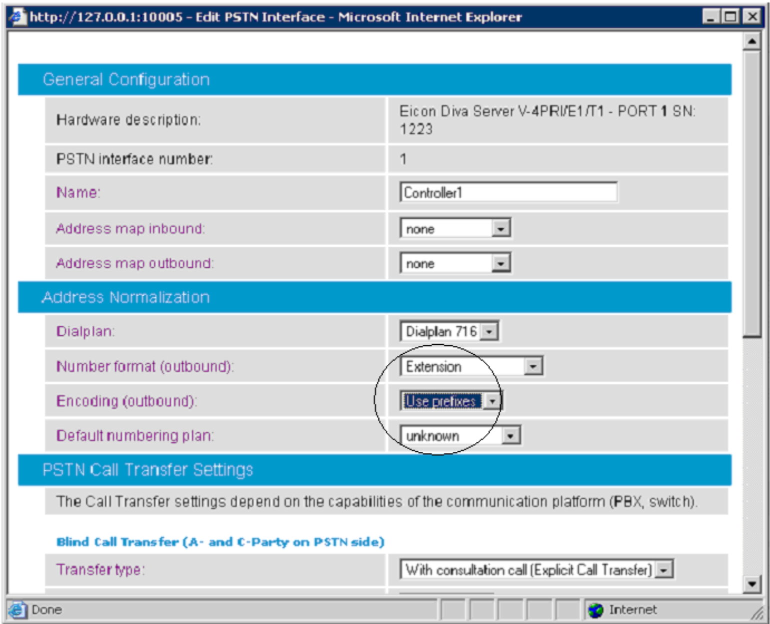
5.2.4.1 Dialplan Configuration Example

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).

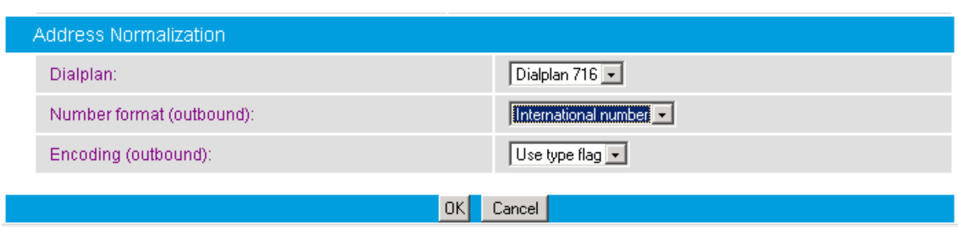
Area code: 716
Base number: 639
Extensions: 4 digits
Access code: 9



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.

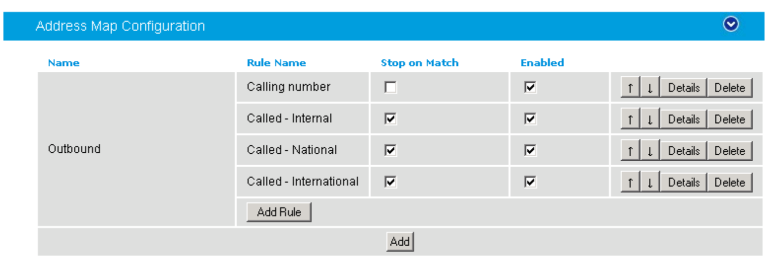
Calling number	**From Microsoft® OCS**	**To PBX**
Internal	+1716639xxxx	716639xxxx

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

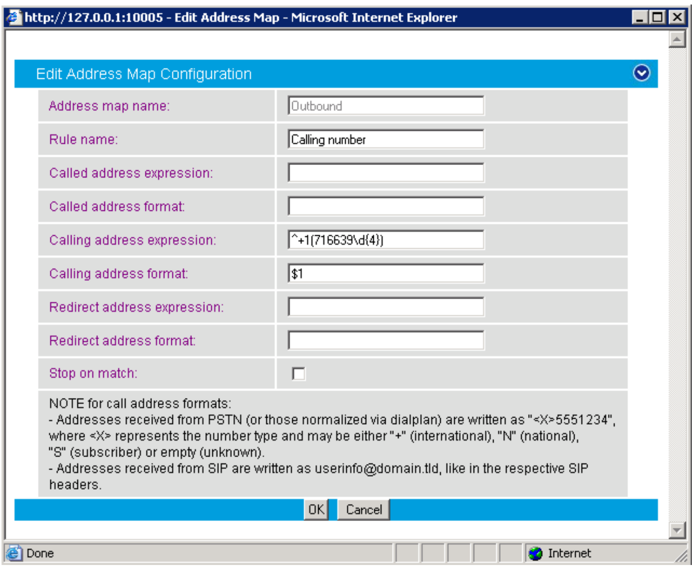
Below is RegEx for the conversion tables above.

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

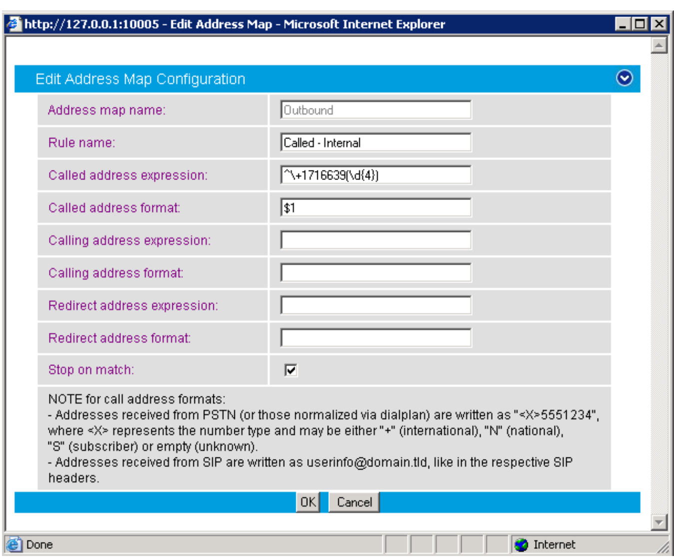
Below are the configured address maps for outbound calls. The order of the below four sub rules and the stop on match check mark are relevant:



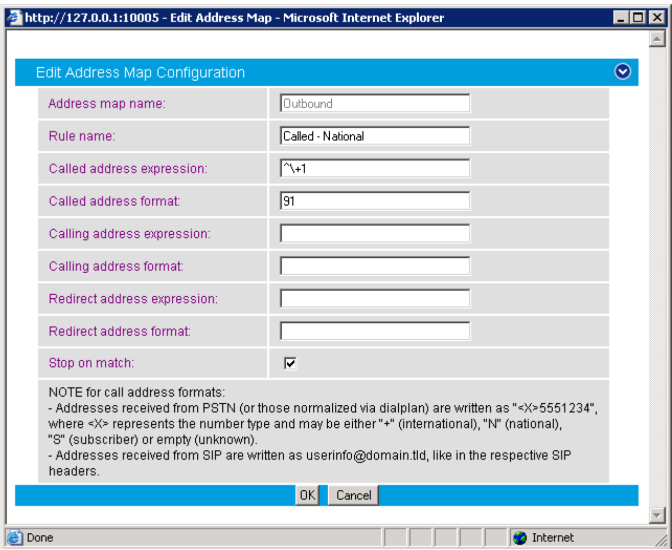
The following screen shows the first sub rule that converts the E.164 calling number into a 10-digit national number:



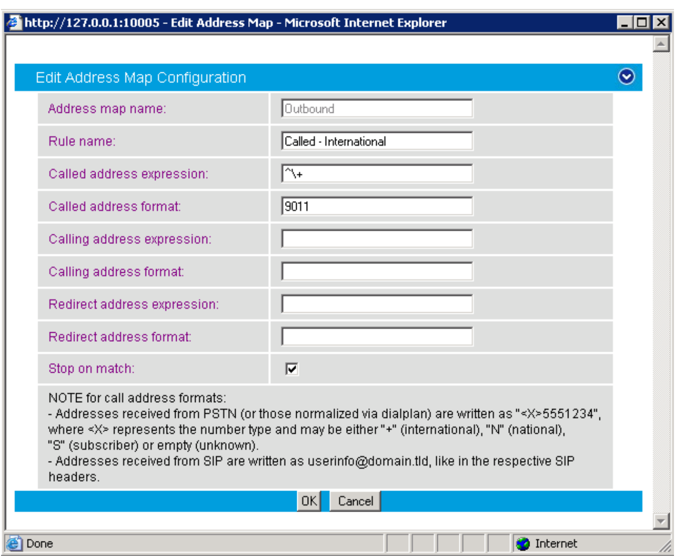
The following screen shows the second sub rule that converts E.164 for the internal called number into a 4-digit extension:



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:

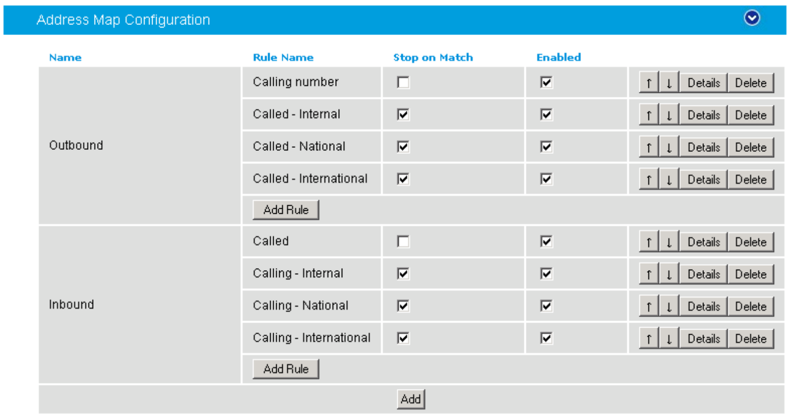
This example assumes that the PBX sends inbound calls using a 4-digit extension, with the ISDN type of number flag set to `Subscriber` for internal numbers, `National` for national calls, and `International` for international calls.

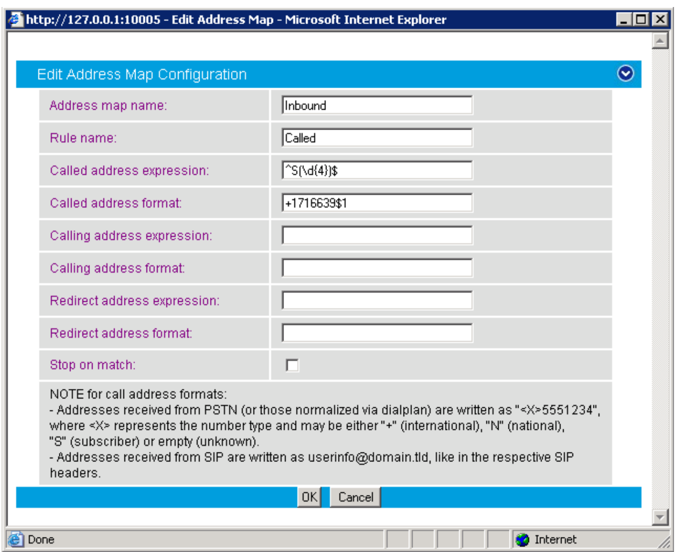
Called number	From PBX	To Microsoft® OCS
Internal	xxxx (with subscriber type of number)	+1716639xxxx

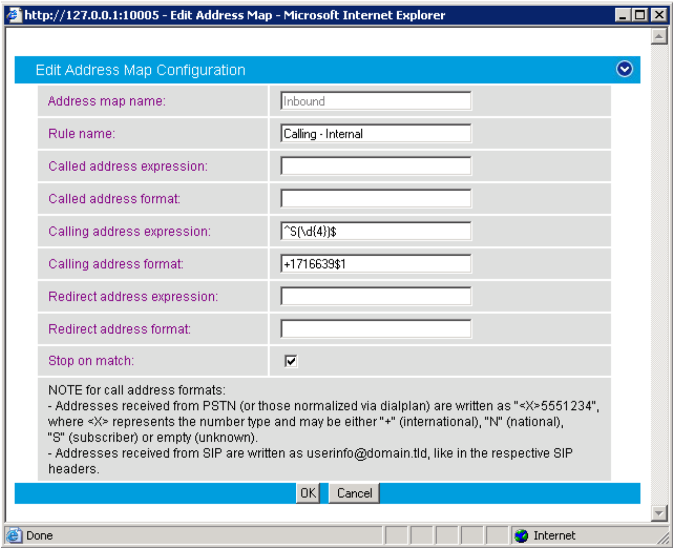
Calling number	From PBX	To Microsoft® OCS
Calling from internal	xxxx (with subscriber type of number)	+1716639xxxx
Calling from national	xxxxxxxxxx (with national type of number)	+1xxxxxxxxxx
Calling from international	xxx…xxx (with international type of number)	+xxx…xxx

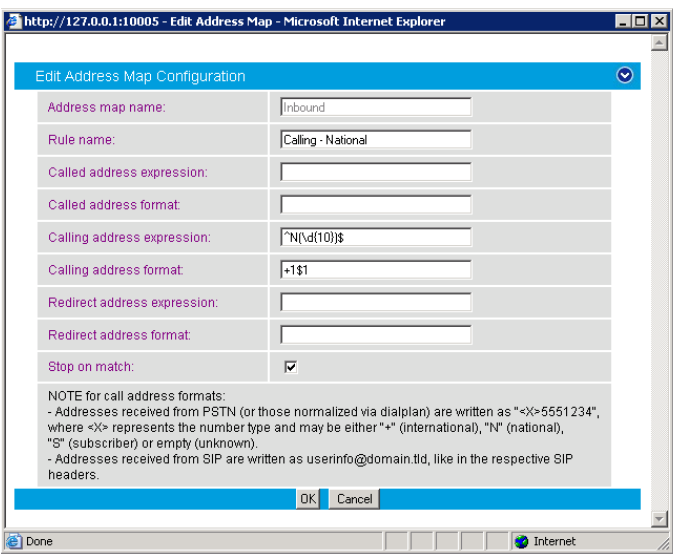
Sub rule name	Expression	Format	Stop on match
Called	^S(\d{4})$	+1716639$1	Not checked
Calling - internal	^S(\d{4})$	+1716639$1	Checked
Calling - national	^N(\d{10})$	+1$1	Checked
Calling - international	^\+	+	Checked

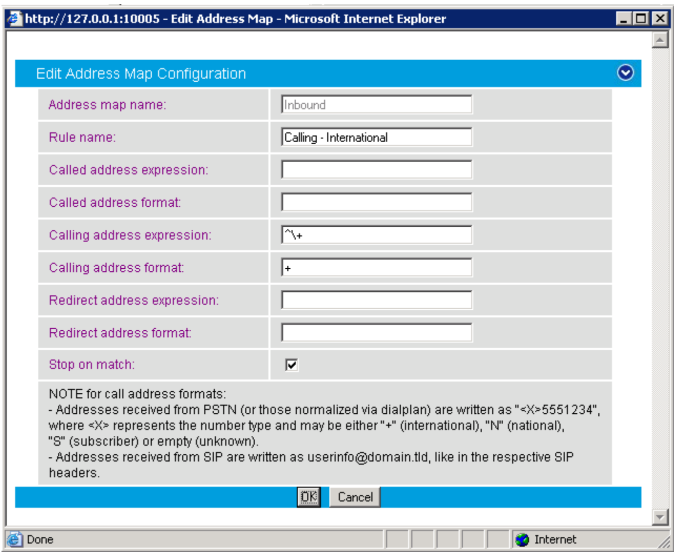
Create an address map named `Inbound` and its four sub rules as shown below:



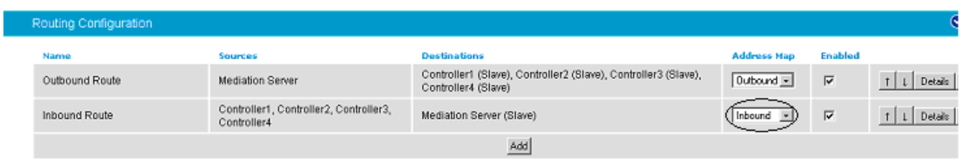



http://127.0.0.1:10005 - Edit Address Map - Microsoft Internet Explorer
Edit Address Map Configuration
Address map name:
Inbound
Rule name:
Calling - Internal
Called address expression:
Called address format:
Calling address expression:
^S(\d{4})$
Calling address format:
+1716639$1
Redirect address expression:
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).
- 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 - 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
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

The basic steps of setting up the PBX for use with this gateway and a voice messaging system are as follows:

1. Setting up system parameters and customer options.
2. Administrating T1 interface.
3. Programming AAR for trunk access.
4. Setting up method of call direction and coverage.

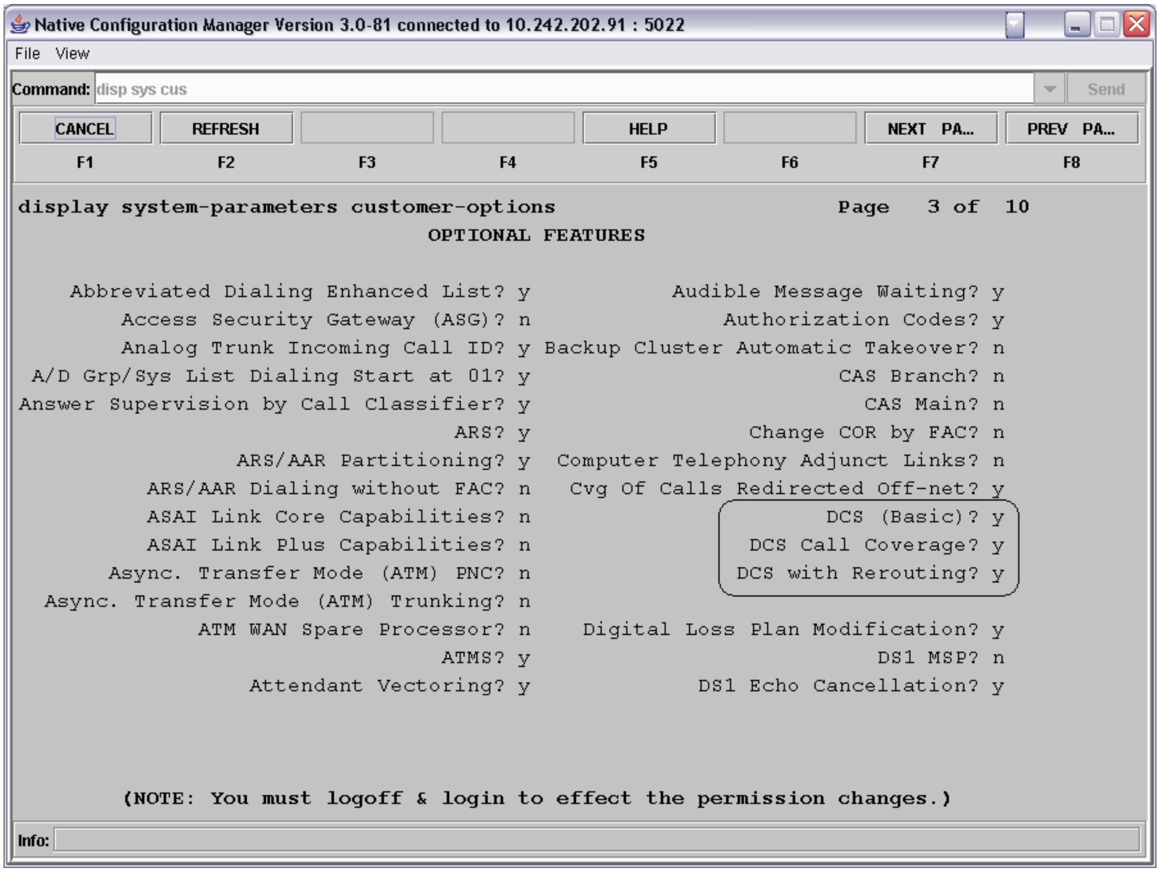
All PBX programming is done via a Windows®-based Java application. The basic commands that you will encounter on the PBX to perform these actions are:

`change`; `delete`; and `add`

6.1 Setting Up System Parameters and Customer Options

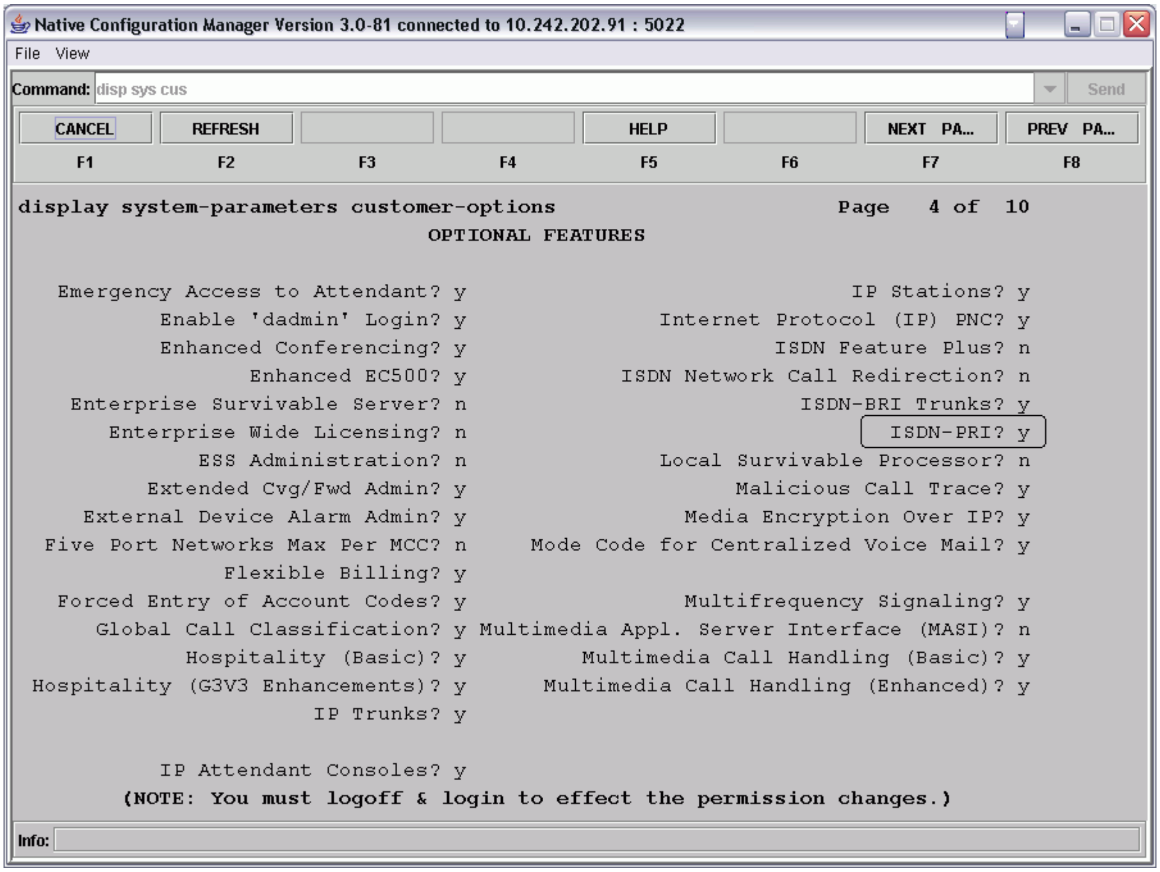
Use the `display system-parameters customer options` command such that several Q.SIG and trunk-specific parameters needed for the integration to operate properly are enabled.

Note that configuration parameters in this area that are not already enabled may require special maintenance or configuration by Avaya Customer support and/or may require additional software packages be installed to configure properly. Once the command has been entered, press the `NEXT PAGE` soft key until the `Optional Features` section is displayed as shown below.



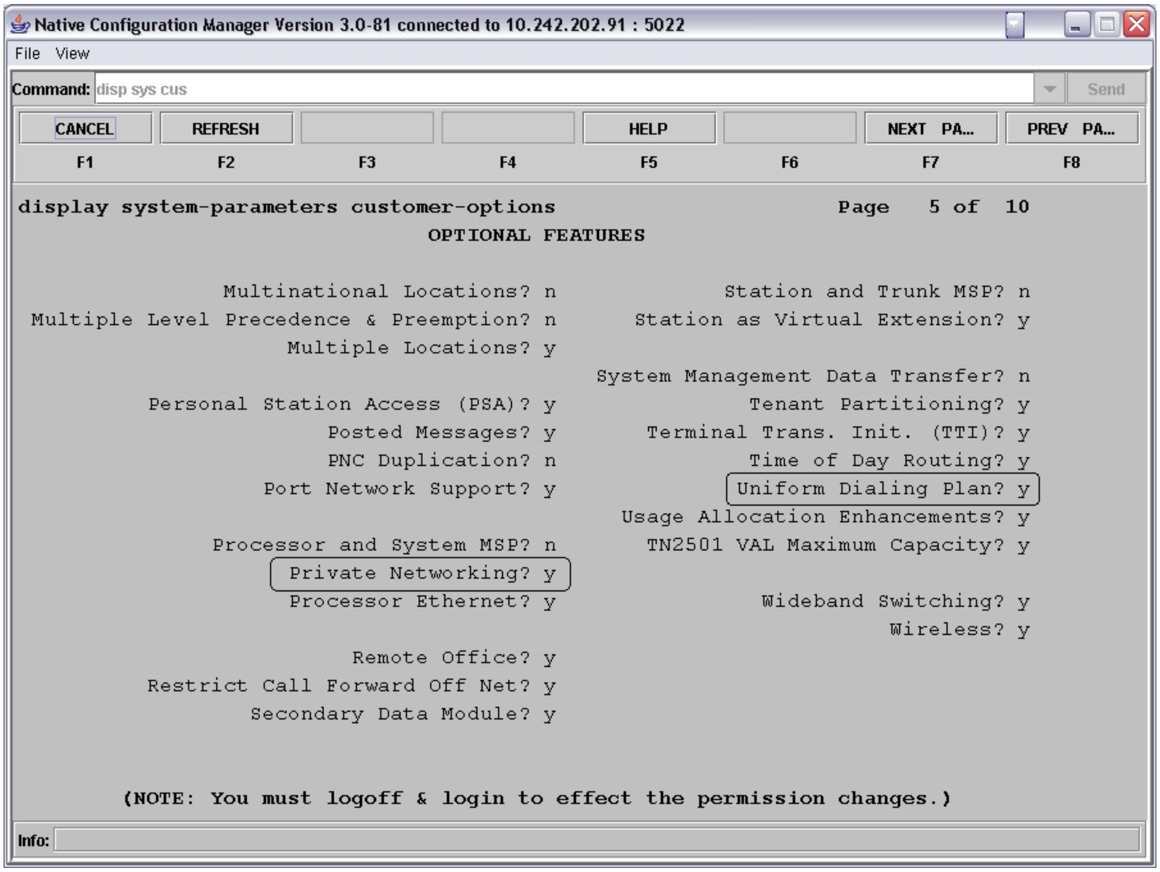
Review the `DCS (Basic)`, `DCS Call Coverage`, and `DCS with Rerouting` options. They must be set to `y` for this integration to work properly. If they are not enabled, then contact your Avaya support representative to have these features enabled.

Press the `NEXT PAGE` soft key, and the next page is displayed.



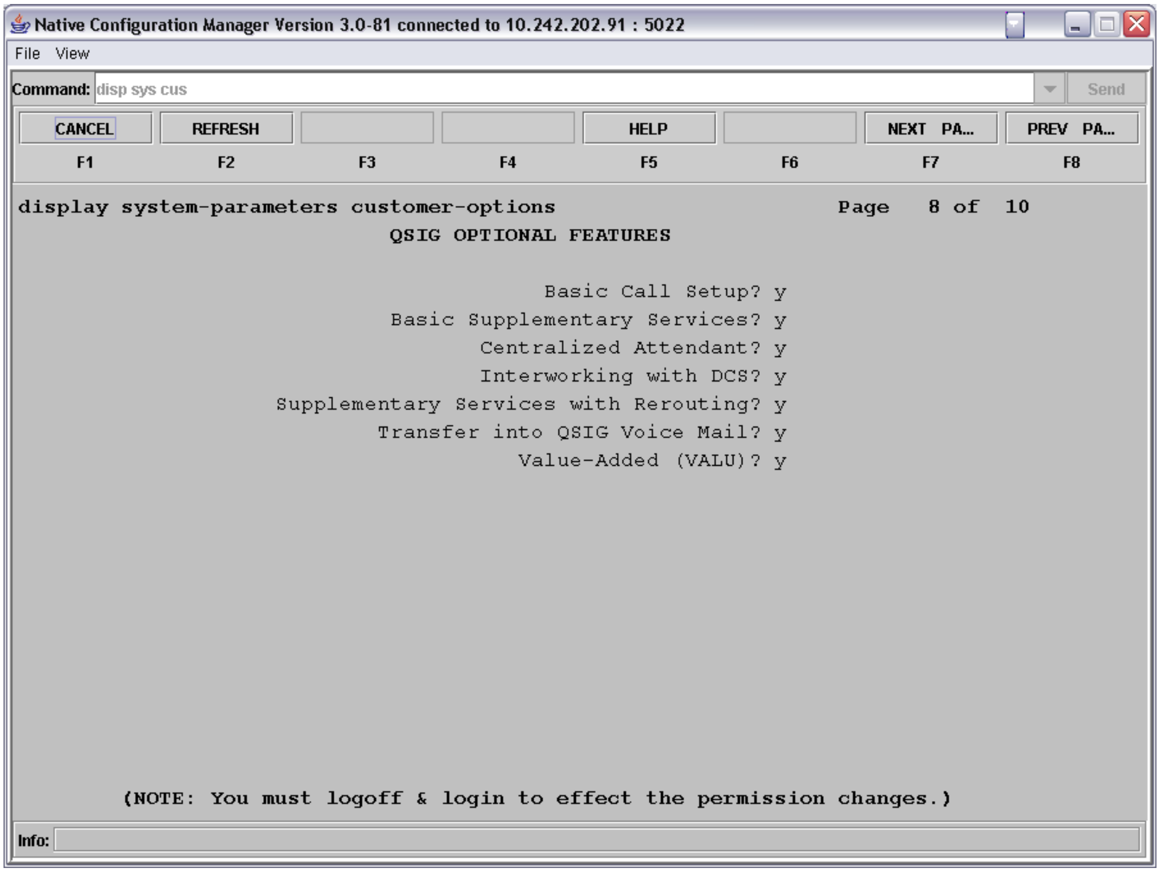
Review the `ISDN-PRI` option. It must be set to `y` for this integration to work properly. If this option is not enabled, then contact your Avaya support representative to have these features enabled.

Press the `NEXT PAGE` soft key, and the next page is displayed.



Review the `Private Networking` and `Uniform Dialing Plan` options. They must be set to `y` for this integration to work properly. It they are not enabled, then you must contact you Avaya support representative to have these features enabled.

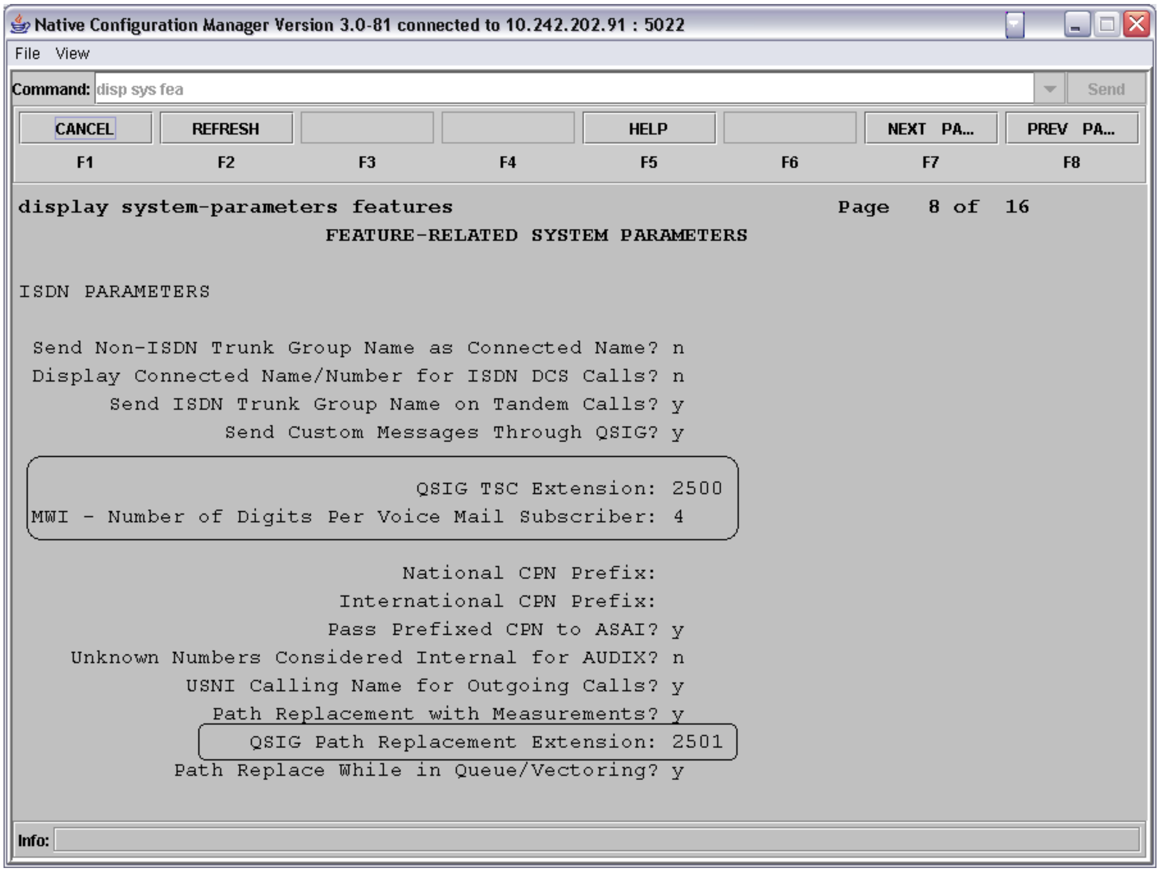
Press the `NEXT PAGE` soft key, until page #8 (`QSIG Optional Features`) is displayed.



Review the `Basic Call Setup`, `Basic Supplementary Services`, `Centralized Attendant`, `Interworking with DCS`, `Supplementary Services with Rerouting`, `Transfer into QSIG Voice Mail`, and `Value Added (VALU)` options. They must be set to `y` for this integration to work properly. If they are not enabled, then you need to contact your Avaya support representative to have these features enabled.

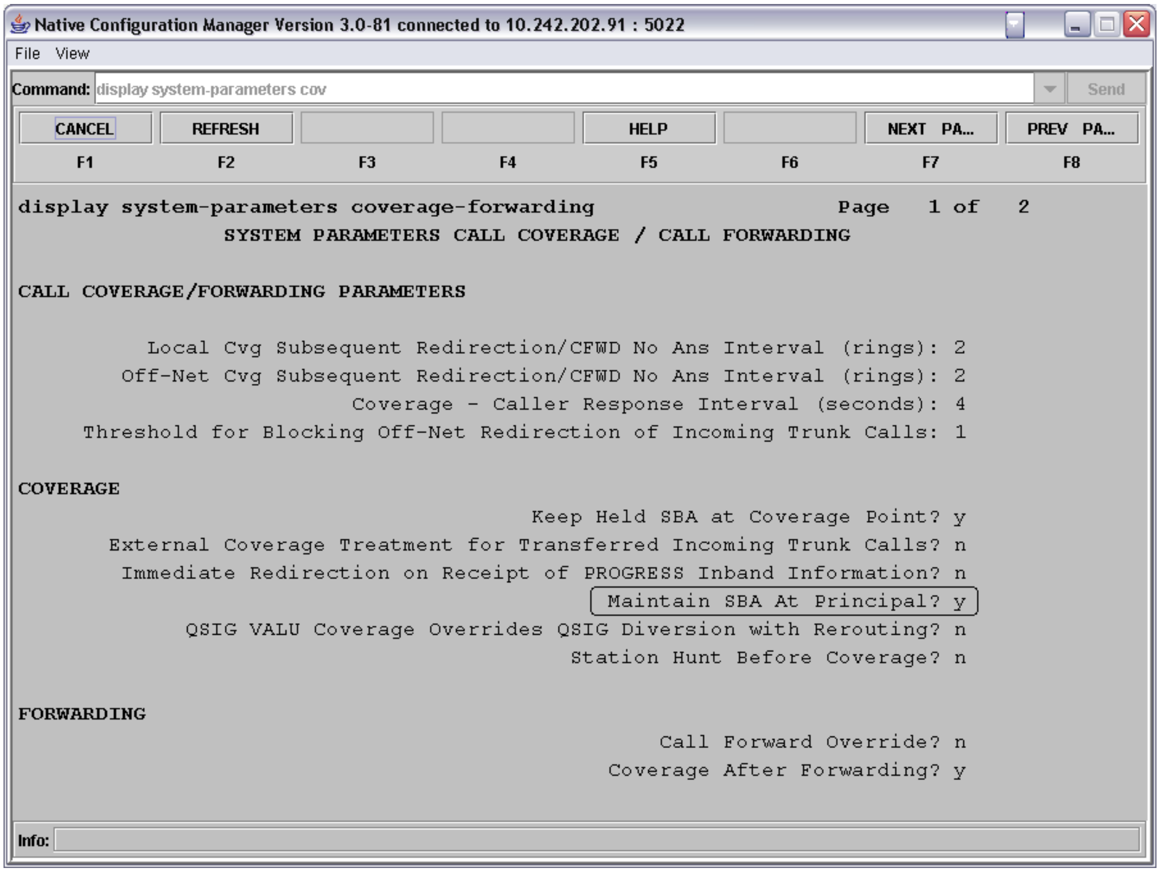
Use the `Change System-Parameter Features` command to make some required modifications to the systems advanced ISDN and Q.SIG features. In order to make changes to the following fields, you must have administrative rights for the Avaya switch. If you do not, then you need to contact your Avaya support representative.

Press the `NEXT PAGE` soft key, until you get to the `ISDN PARAMETERS` page.



- In the `QSIG TSC Extension` field enter `XXXX`
    - where `XXXX` is any unused extension number.
- In the `MWI-Number of Digits Per Voice Mail Subscriber` field enter `X`
    - where `X` is the length of digits of the subscribers extensions.
- In the `QSIG Path Replacement Extension` field enter `XXXX`
    - where `XXXX` is any unused extension number.

Before moving to the next command, be sure to save your work.
Use the `change system-parameters coverage-forwarding` command to modify the system features relevant for forwarding the extensions and call coverage such that the Q.SIG supplementary services work properly.



- In the `Maintain SBA At Principal` field enter `y` to enable this feature.

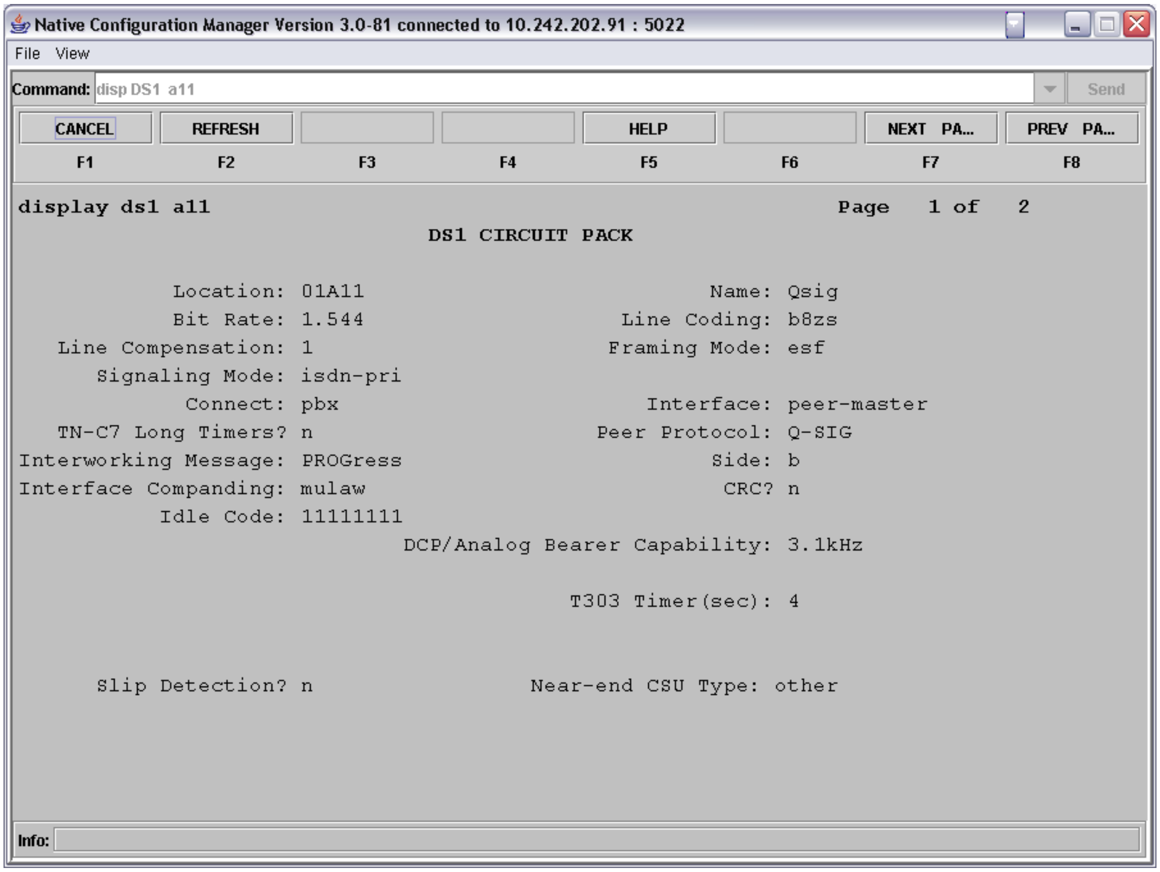
Before moving to the step, be sure to save your work.

6.2 Administrating the T1 Interface

Use the `add ds1` command to setup the DS1 card with the proper settings. The command takes an argument that includes the cabinet and slot number where the DS1 card is located. For example:

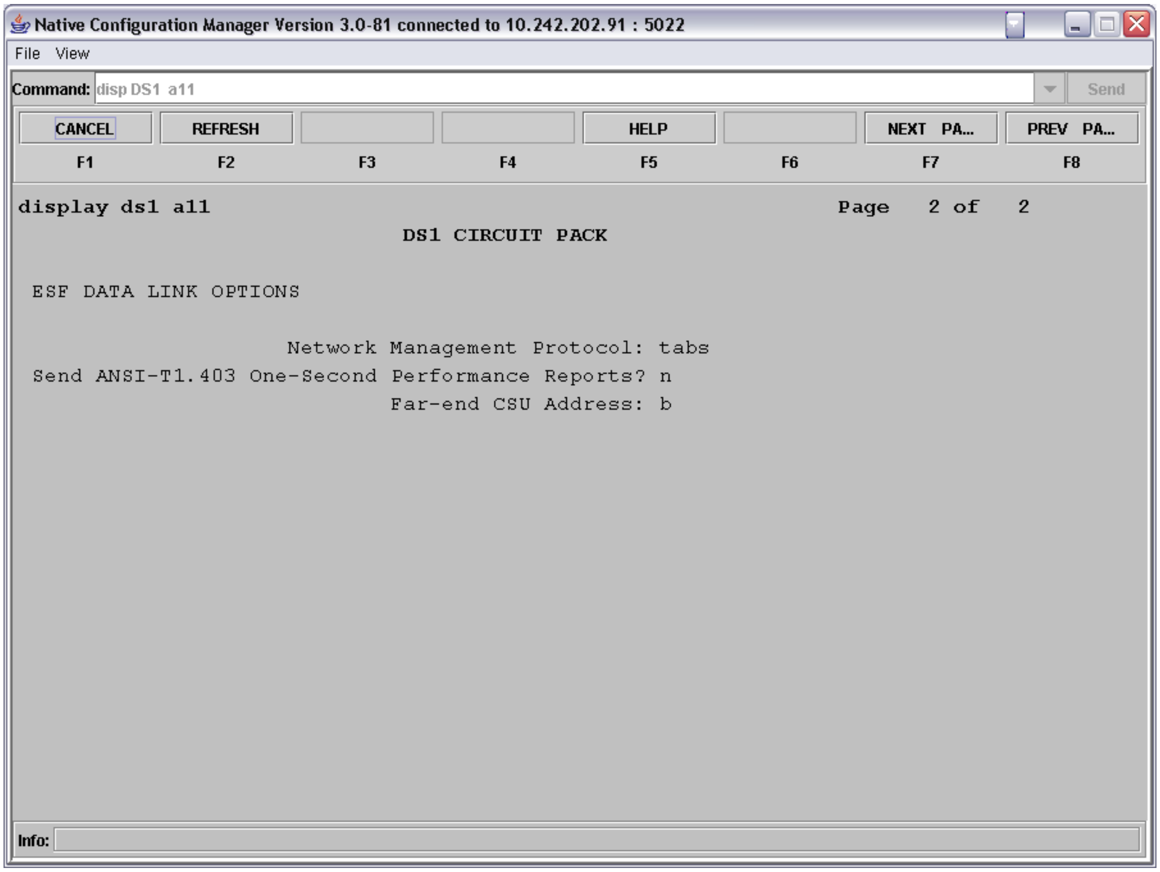
```
Command = add ds1 a11
```

Add a ds1 circuit card into the configuration that is located in cabinet ‘a’ and is in slot #11 of that cabinet.



- In the `Name` field, enter in any string to assign the DS1 a name
- In the `Bit Rate` field, enter `1.544`. *(The standard clock rate for T1)*
- In the `Line Coding` field, enter `b8zs`.
- In the `Line Compression` field, enter `1`.
- In the `Framing Mode` field, enter `esf`.
- In the `Signaling Mode` field, enter `isdn-pri`.
- In the `Connect` field, enter `pbx`.
- In the `Interface` field, enter `peer-master`.
- In the `Peer Protocol` field, enter `Q-SIG`.
- In the `Interworking Message` field, enter `PROGress`.
- In the `Side` field, enter `b`.
- In the `Interface Companding` field, enter `mulaw`.
- In the `CRC` field, enter `n`.
- In the `Idle Code` field, enter `11111111`. *(8 - 1's)*

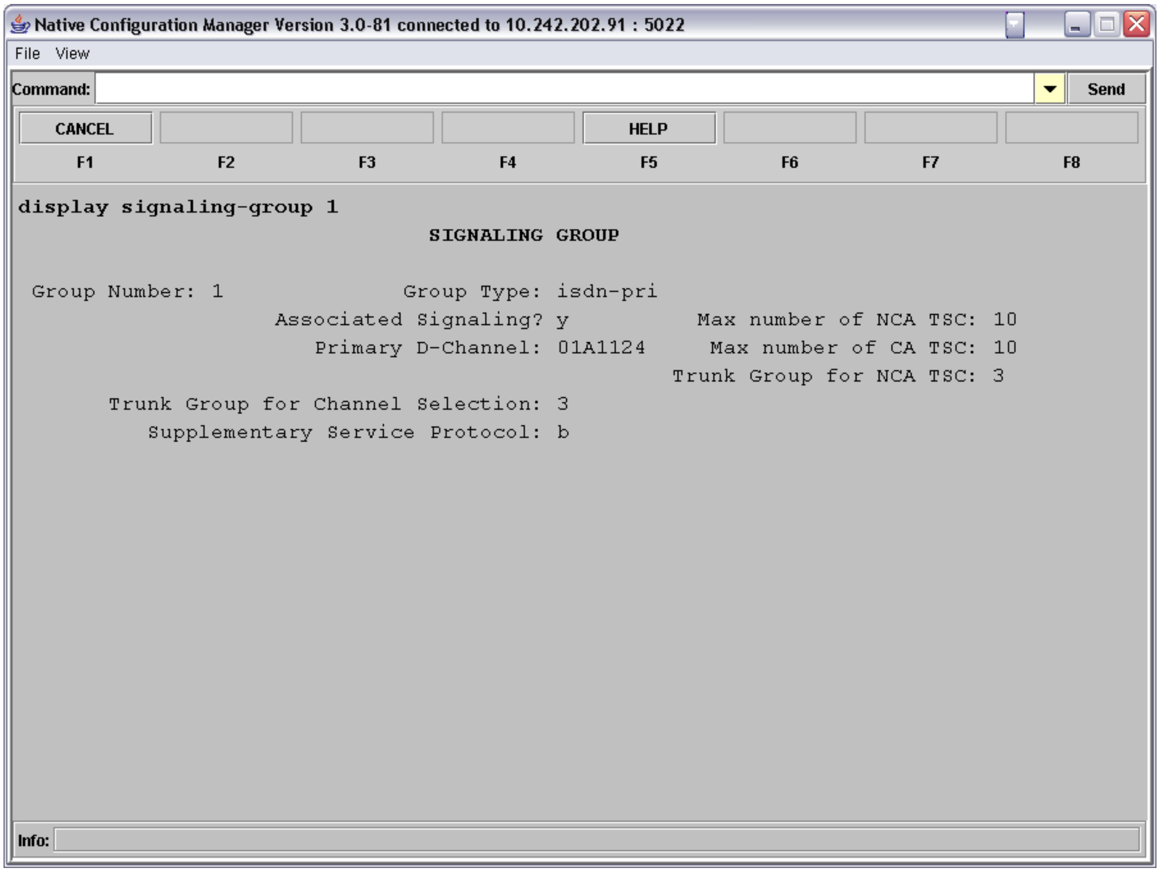
Press `NEXT PAGE` to move to the next page of settings.



- In the `Far-end CSU Address` field, enter `b`.

Before moving on to the next command, be sure to save your work.

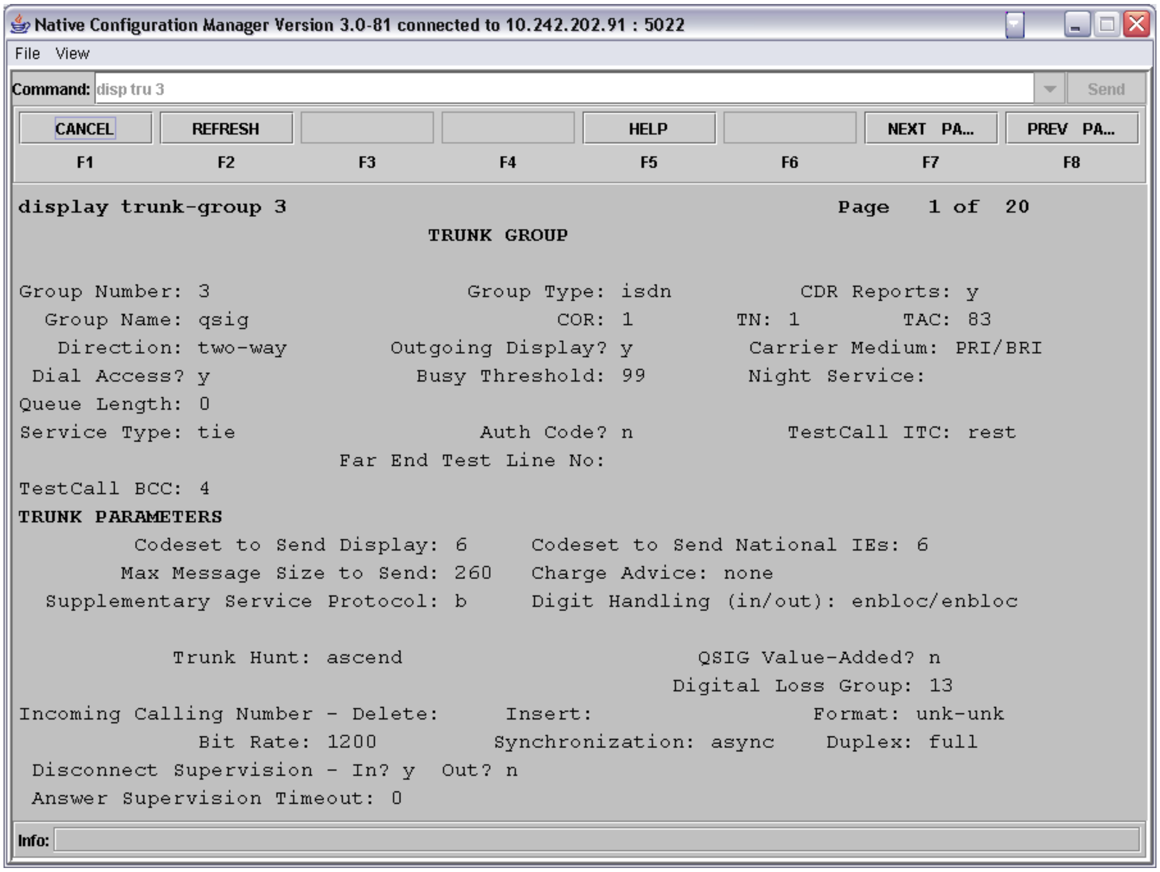
Use the `add signaling-group` command to build a signaling group for the Q.SIG link to the gateway. This signaling group specifies the type of interface being used, the signaling channel (D-channel) wiring address for the T1 span being used, and the protocol used on the signaling channel.



- In the `Group Type` field, enter `isdn-pri`.
- In the `Associated Signaling` field, enter `y`.
- In the `Max number of NCA TSC` field, enter `10`.
- In the `Primary D-Channel` field, enter `XXXXX24`
    - where `XXXXX24` is the 24th port number of the T1 board.
- In the `Max number of CA TSC` field, enter `10`.
- In the `Trunk Group for Channel Selection` field, enter `X`.
    - where `X` is the trunk group number created in the next step.
      Note: You will need to return to here and edit this number after you have created the trunk group.
- In the `Supplementary Service Protocol` field, enter `b`.

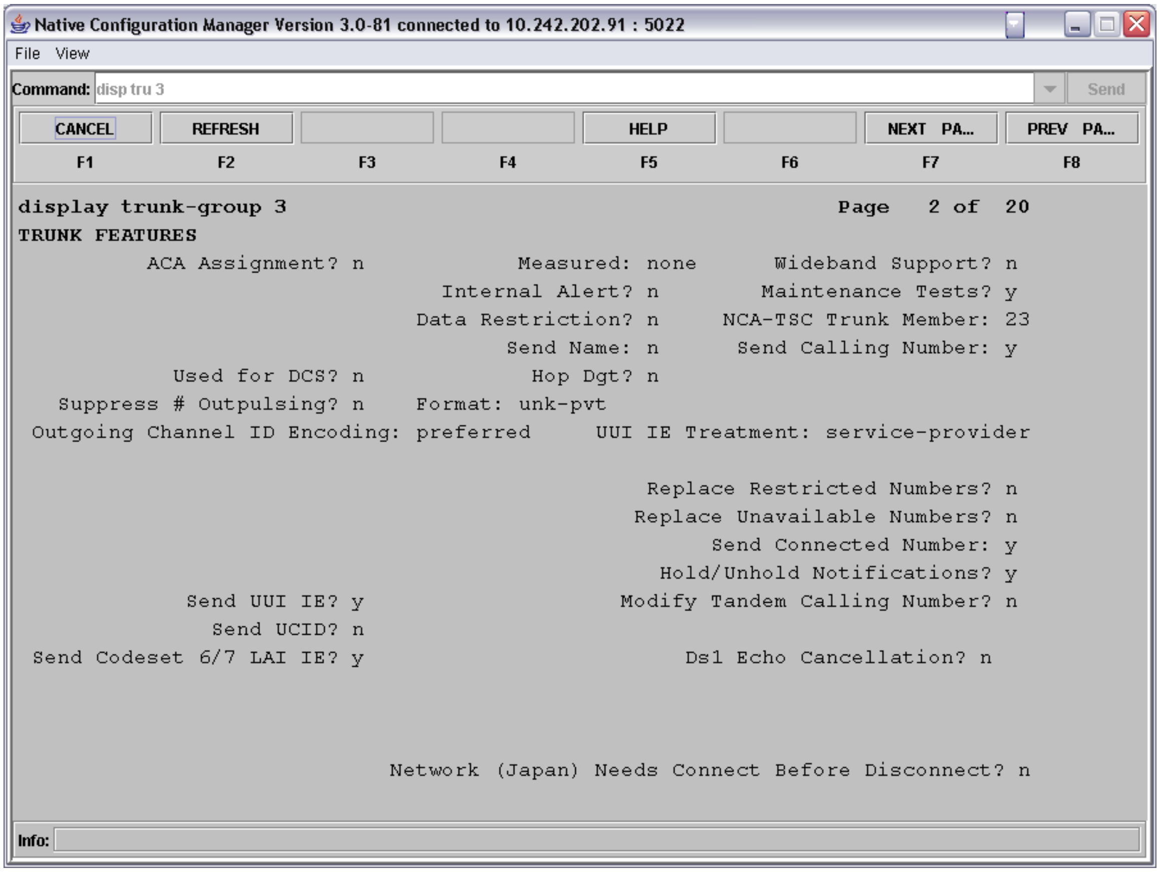
Before moving to the next command, be sure to save your work.

Use the `add trunk-group` command to add a new trunk group that will contain all the members of the ISDN line and tie them together with the signaling group created above.



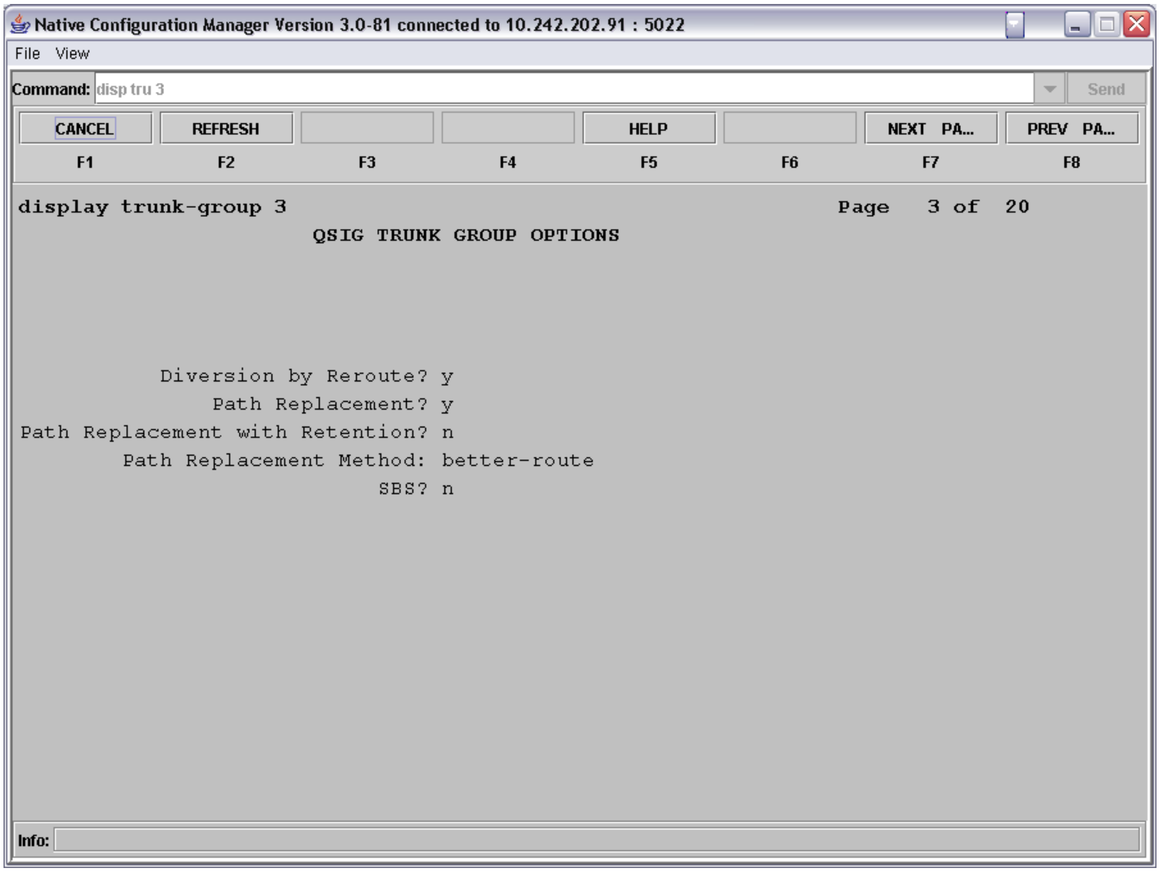
- In the `Group Type` field, enter `isdn`.
- In the `Group Name` field, enter a text string to give the group a name.
- In the `COR` field, enter the number of a class of restriction to be used, if there is going to be one.
  Note: Setting any restrictions via a COR can effect how the trunk operates so these should be as liberal as possible.
- In the `TN` field, enter the proper tenant number.
- In the `TAC` field, enter `XX`
  - where `XX` is an available DAC (used as a trunk access code) number from the dial-plan.
- In the `Direction` field, enter `two-way`.
- In the `Outgoing Display` field, enter `y`.
- In the `Carrier Medium` field, enter `PRI/BRI`.
- In the `Service Type` field, enter `tie`.
- In the `Supplementary Service Protocol` field, enter `b`.
- In the `Digit Handling (in/out)` field, enter `enbloc/enbloc`.
- In the `Format` field, enter `unk-unk`.

Press `NEXT PAGE` to advance to the next page of settings.



- In the `NCA-TSC Trunk Member` field, enter `23`.
- In the `Send Calling Number` field, enter `y`.
- In the `Format` field, enter `unk-pvt`.
- In the `Outgoing Channel ID Encoding` field, enter `preferred`.
- In the `Send Connected Number` field, enter `y`.
- In the `Send UUI IE` field, enter `y`.

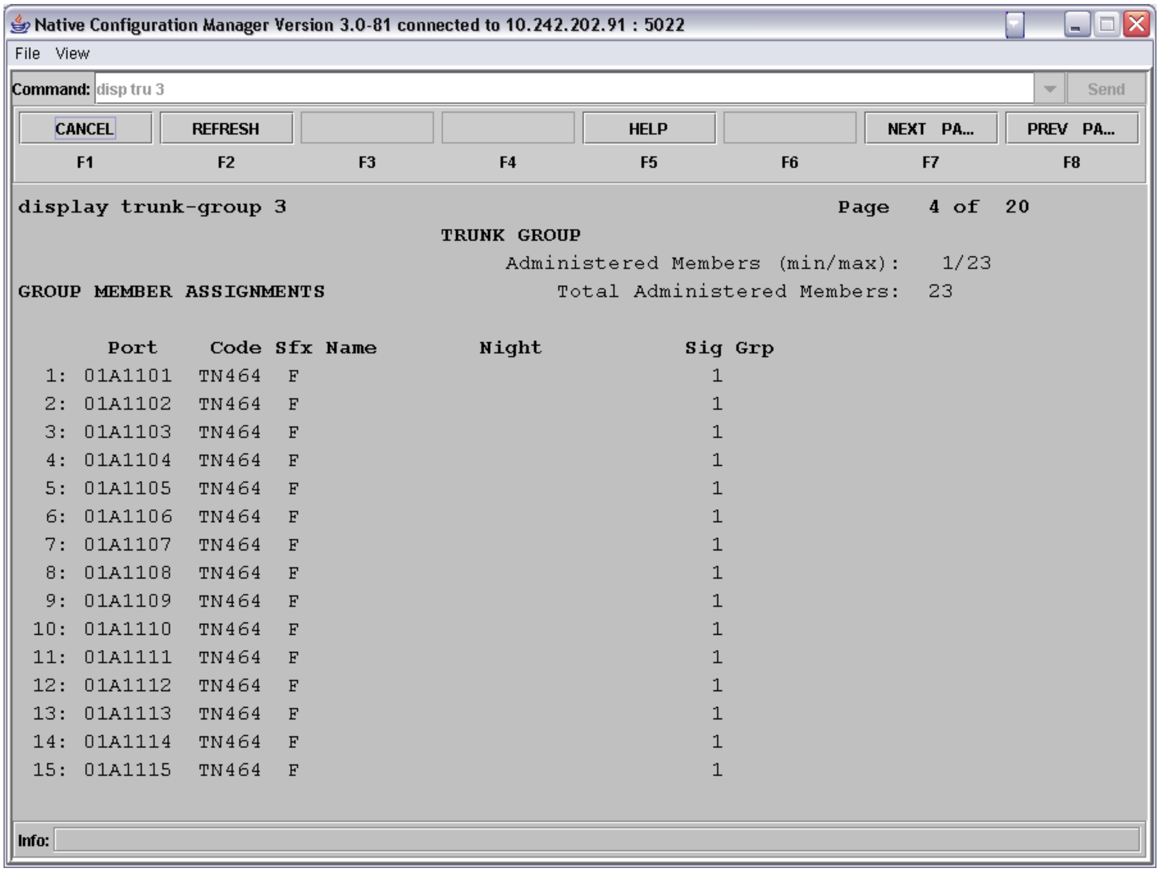
Press `NEXT PAGE` to advance to the next page of settings.



- In the `Diversion by Reroute` field, enter `y`.
- In the `Path Replacement` field, enter `y`.
  Note: This parameter enables the path replacement functionality of the Q.SIG supplemental services package.
- In the `Path Replacement Method` field, you can enter a selection to tune how your path replacement optimization is done.

Press `NEXT PAGE` to advance to the next page of settings.

On this page, enter each bearer channel of the T1 span, leaving off the D-channel since that is connected to the group via the Signaling Group and is not used to carry voice.



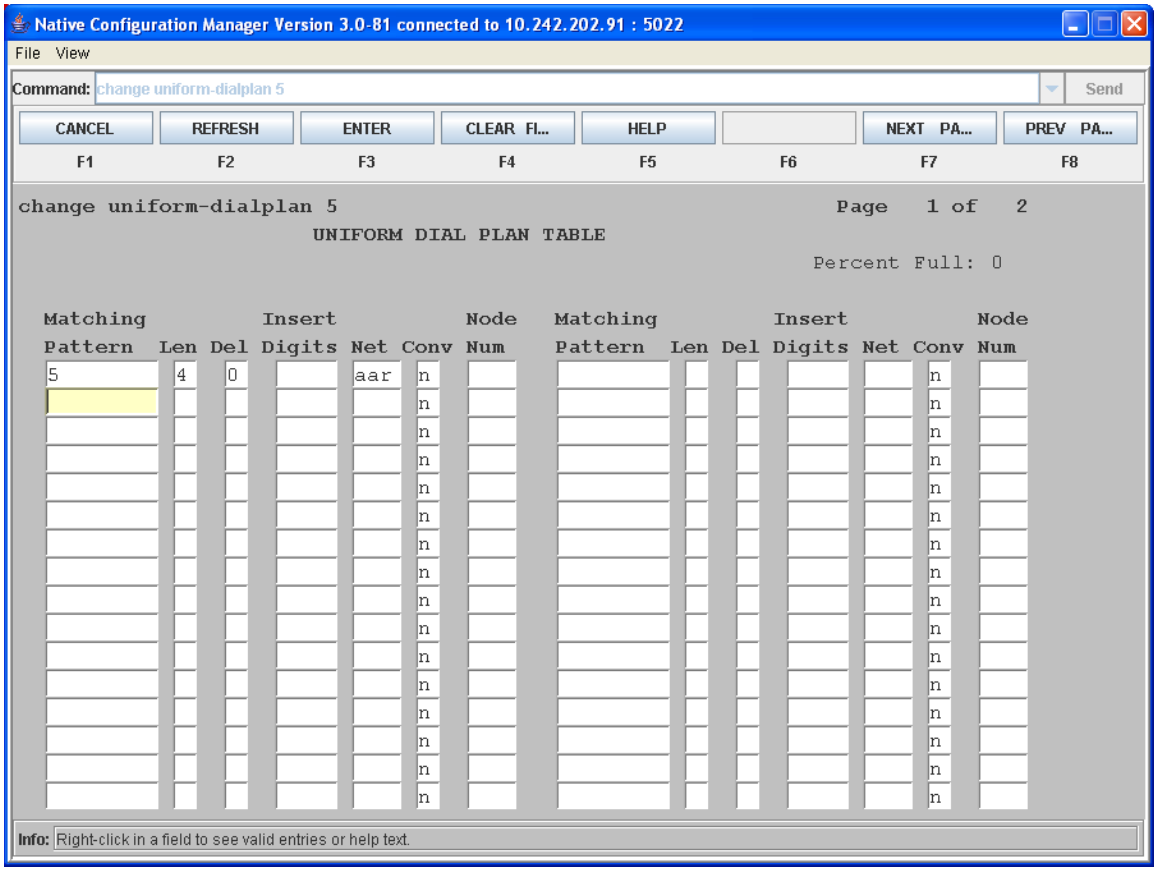
- In each of the `Port` fields, enter the port number that designates the physical wiring address of each of the bearer channels in the span.
- In each of the corresponding `Sig Grp` fields, for each port enter in the number of the Signaling Group created above.

Before moving on to the next command, be sure to save your work.

6.3 Programming AAR for Trunk Access

Important Note: In order for Path Replacement to work on Join transfers, the inbound calls MUST use an AAR number to access the trunk, and NOT the assigned Trunk Access Code (TAC). If an outgoing call is made using a TAC, or a call was extended by an attendant using DTGS (Direct Trunk Group Selection), the user has intentionally chosen a particular Trunk Group for the outgoing call, and the PBX will not use Path Replacement optimization in these cases.

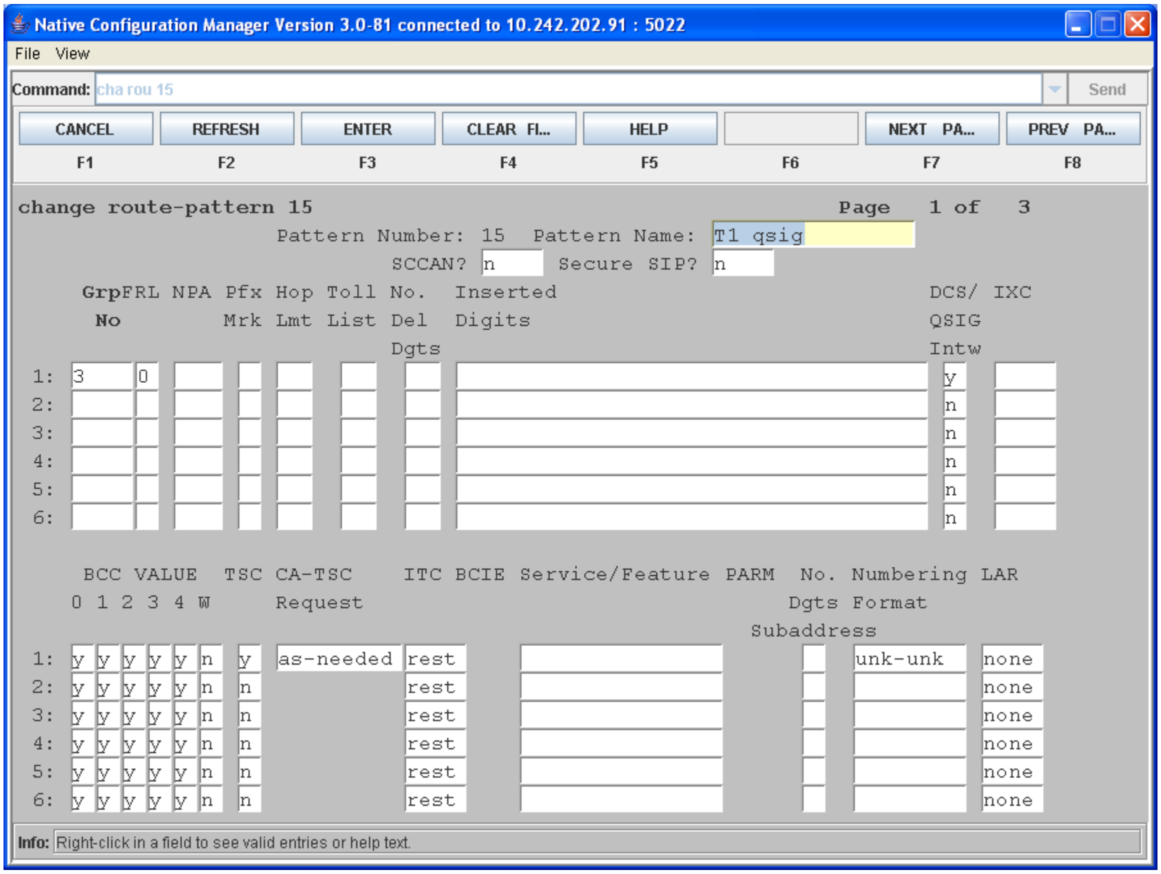
Use the `change uniform-dialplan` command to setup an extension that can be used to access the trunks via AAR.



- In the `Matching Pattern` field, enter a number to define as a dialable AAR number. *In this example it is 5.*
- In the `Len` field, enter `X`
  - o where `X` is the length of your dialable number. *In this example 4.*
- In the `Del` field, enter `0`.
- In the `Net` field, enter `aar`.
- In the `Conv` field, enter `n`.

Before moving on to the next command, be sure to save your work.

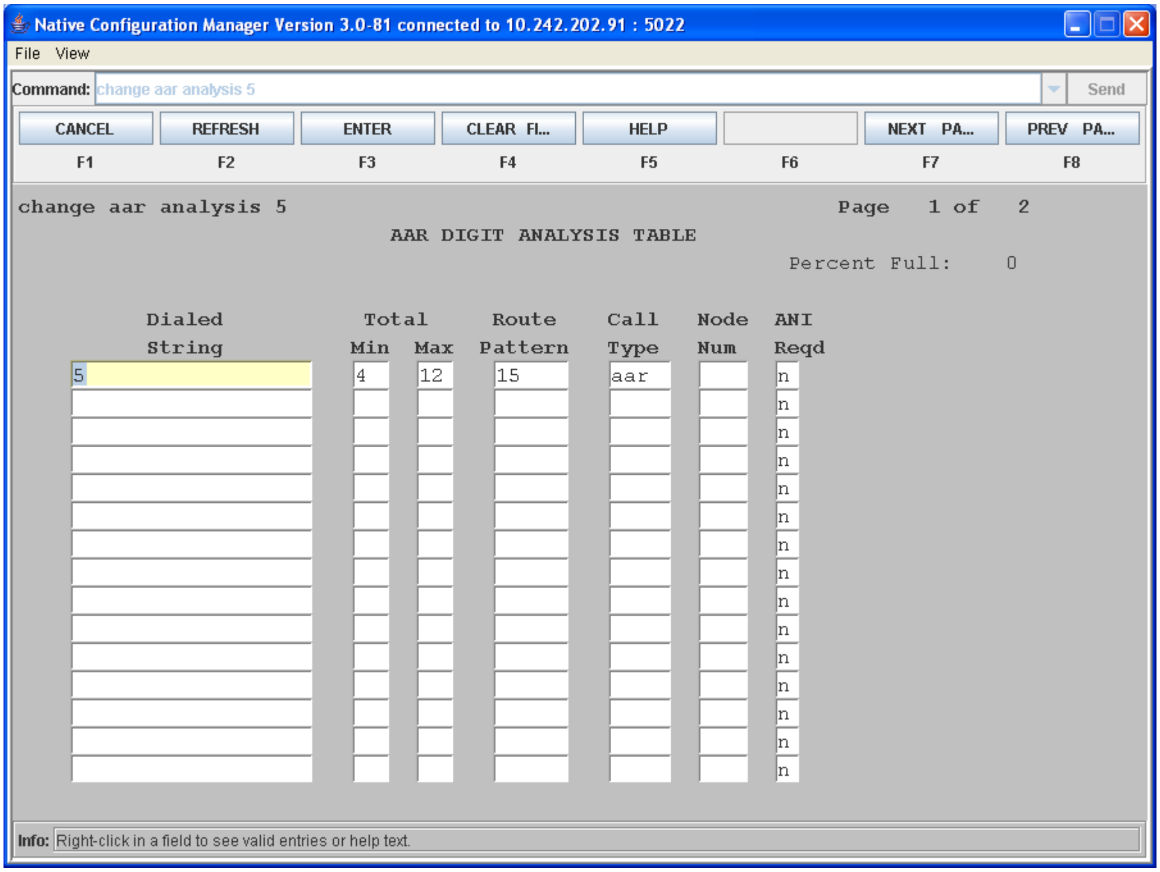
Use the `change route-pattern` command to define a route rule pattern that when referenced will direct calls to the trunk group you created above.



- In the `Grp` field, enter `X`.
  - where `X` is the number of the trunk group assigned above.
- In the `FRL` field, enter `0`.
- In the `Intw` field, enter `y`.
- In the `CA-TSC Request` field, enter `as-needed`.
- In the `Numbering Format` field, enter `unk-unk`.

Before moving on to the next command, be sure to save your work.

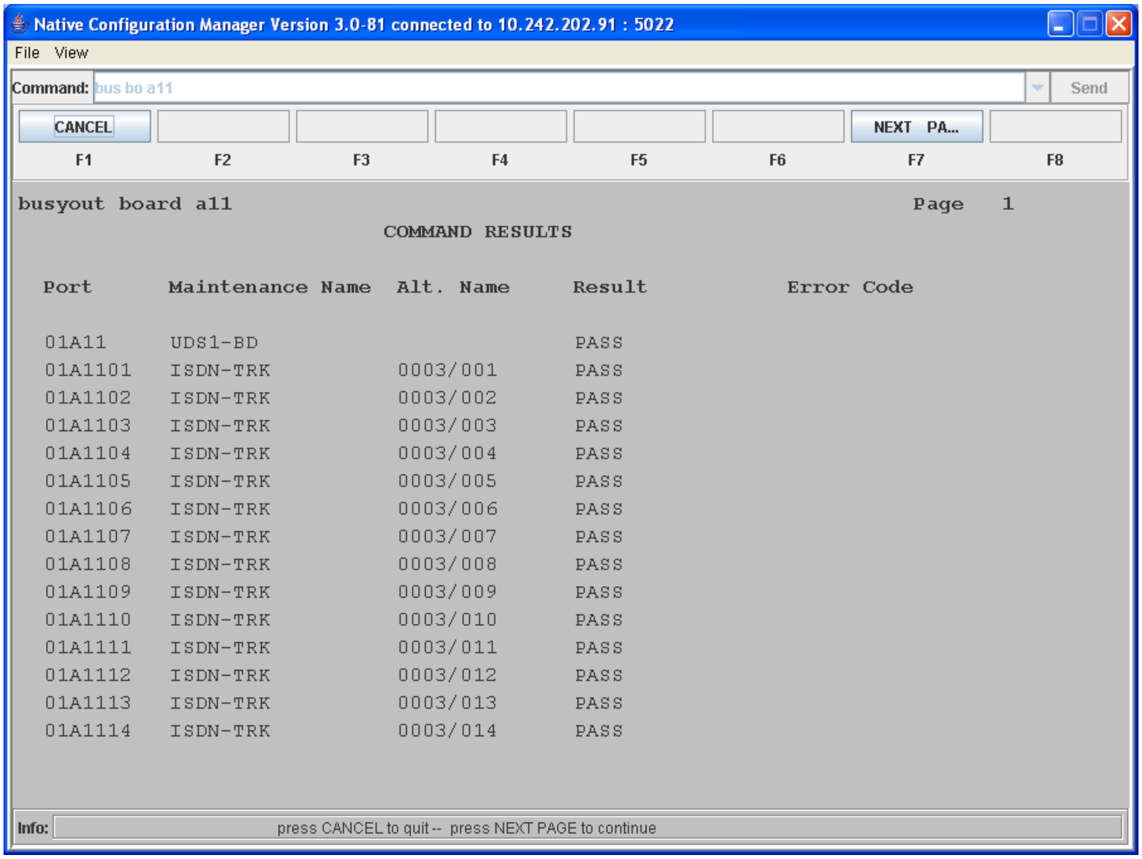
Use the `change AAR Digit Analysis Table` command to set up AAR to bind your dialable number setup in UDP to the route pattern you have just defined.

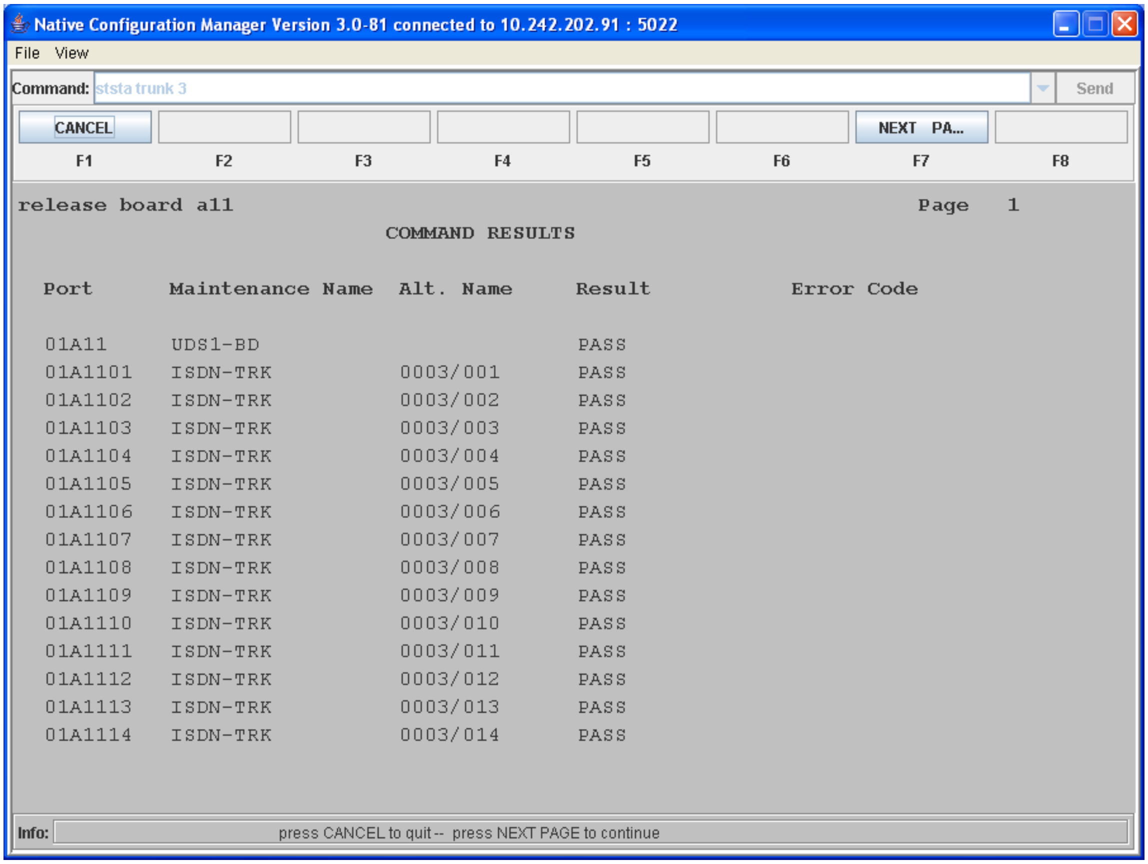


- In the Dialed String field, enter the number you defined in UDP as the AAR number. *In this example we use 5.*
- In the Min and Max fields, enter the length of the number you are going to dial. *In this example 4 for the Min and 12 for the Max.*
- In the Route Pattern field, enter the number of the route pattern that you defined above. *In this example 2.*
- In the Call Type field, enter aar.
- In the ANI Reqd field, enter n.

Before moving on to the next command, be sure to save your work.

Use the busyout board and the release board commands to reset the trunk device that has been defined. When a T1 board is initially programmed, by default it is in a busy state. In order to enable the T1 board, it must be set to busy and then released.


Native Configuration Manager Version 3.0-81 connected to 10.242.202.91 : 5022
File View
Command: bus bo a11
Send
CANCEL
NEXT PA...
F1 F2 F3 F4 F5 F6 F7 F8
busyout board a11 Page 1
COMMAND RESULTS
Port Maintenance Name Alt. Name Result Error Code
01A11 UDS1-BD PASS
01A1101 ISDN-TRK 0003/001 PASS
01A1102 ISDN-TRK 0003/002 PASS
01A1103 ISDN-TRK 0003/003 PASS
01A1104 ISDN-TRK 0003/004 PASS
01A1105 ISDN-TRK 0003/005 PASS
01A1106 ISDN-TRK 0003/006 PASS
01A1107 ISDN-TRK 0003/007 PASS
01A1108 ISDN-TRK 0003/008 PASS
01A1109 ISDN-TRK 0003/009 PASS
01A1110 ISDN-TRK 0003/010 PASS
01A1111 ISDN-TRK 0003/011 PASS
01A1112 ISDN-TRK 0003/012 PASS
01A1113 ISDN-TRK 0003/013 PASS
01A1114 ISDN-TRK 0003/014 PASS
Info: press CANCEL to quit -- press NEXT PAGE to continue



In both commands, you can use `NEXT PAGE` to view the trunk channels that are not visible on the page.

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

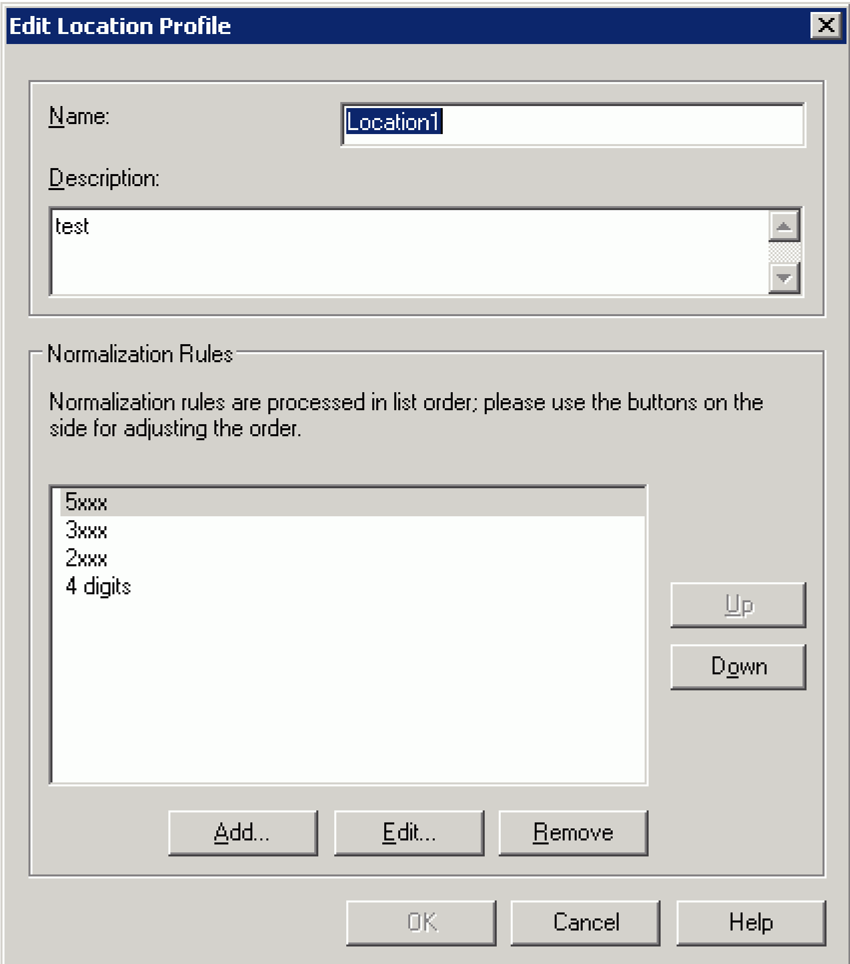
7.1 Steps for configuring Microsoft® OCS

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).

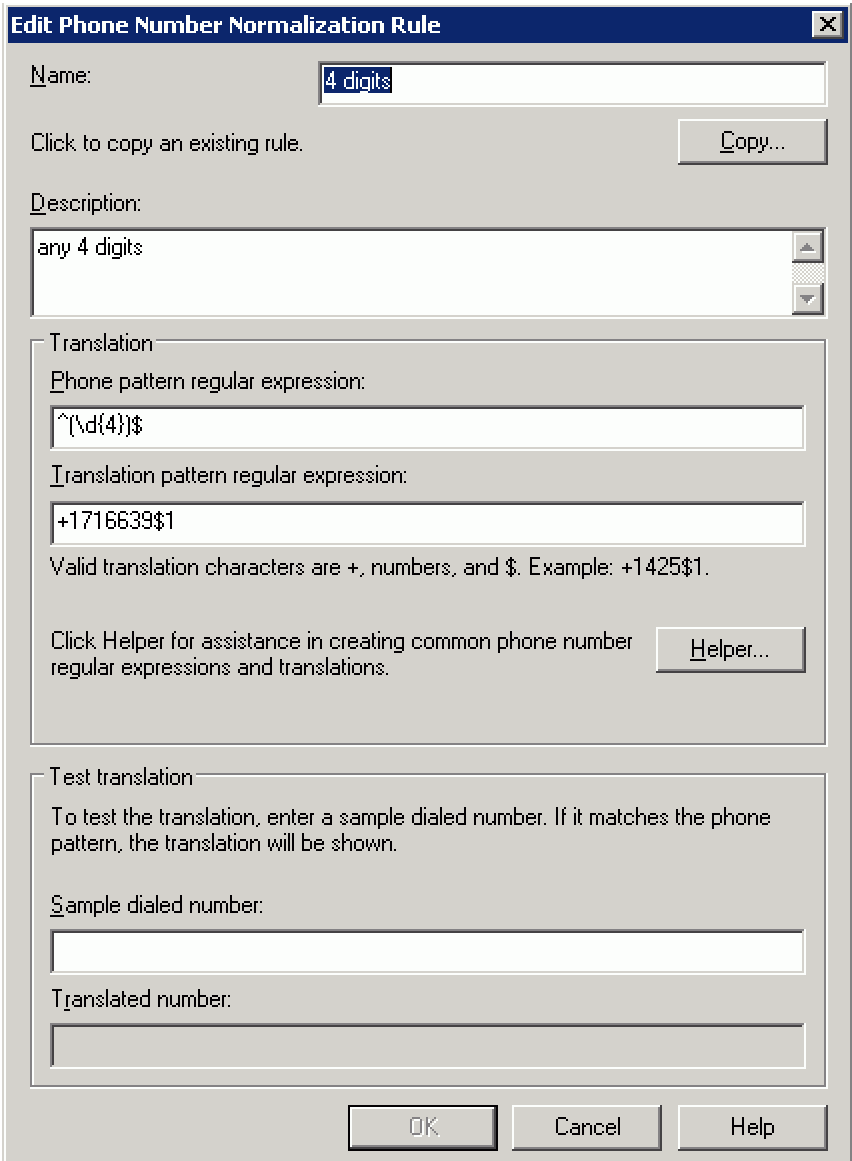
If a Microsoft® OCS user dials an internal extension number (normally 3-5 digits), the normalization rules convert it into full E.164 format. These normalization rules should cover dialed digits for internal extensions, local numbers, long distance numbers, and international numbers.

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.

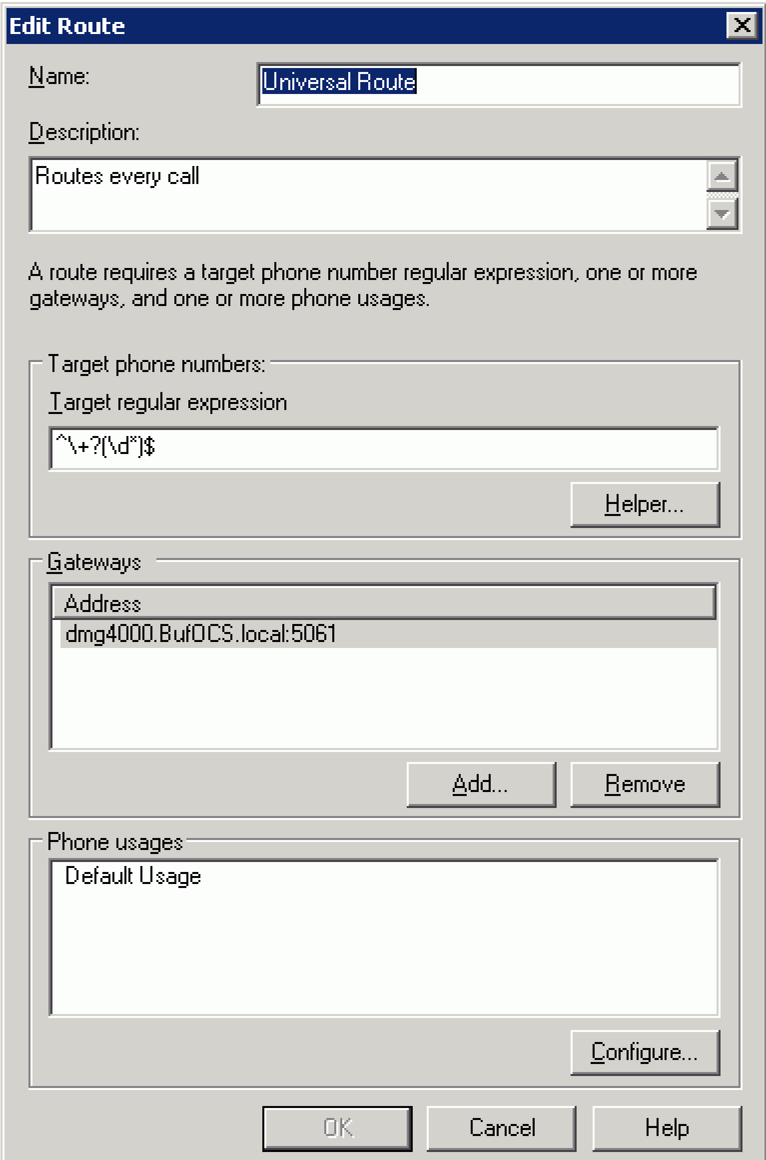
More examples are shown in the following table:

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.



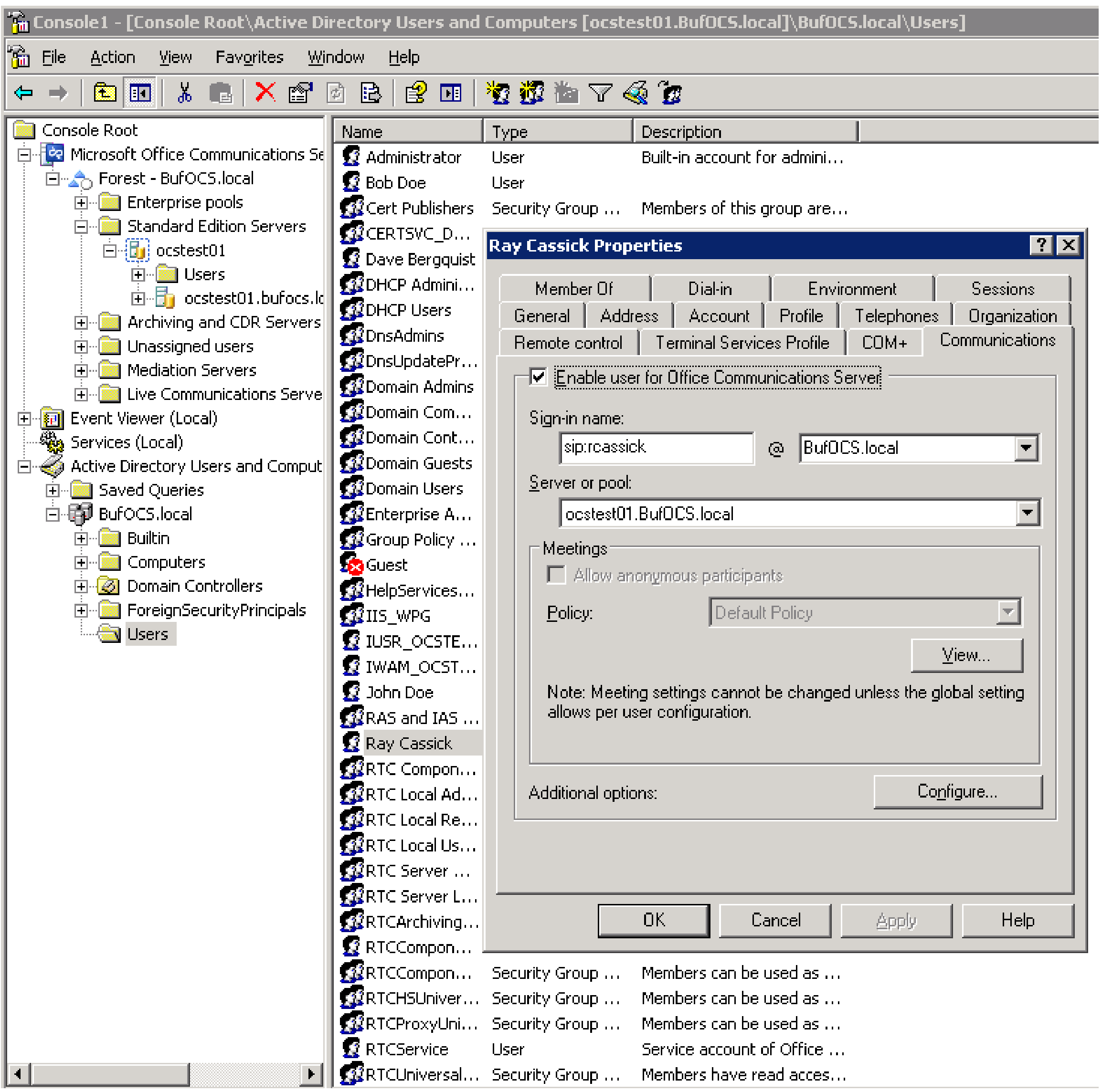
This entry routes numbers with or without "+" prefix followed by any digits to Microsoft® Mediation Server dmg4000.bufocs.local.

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

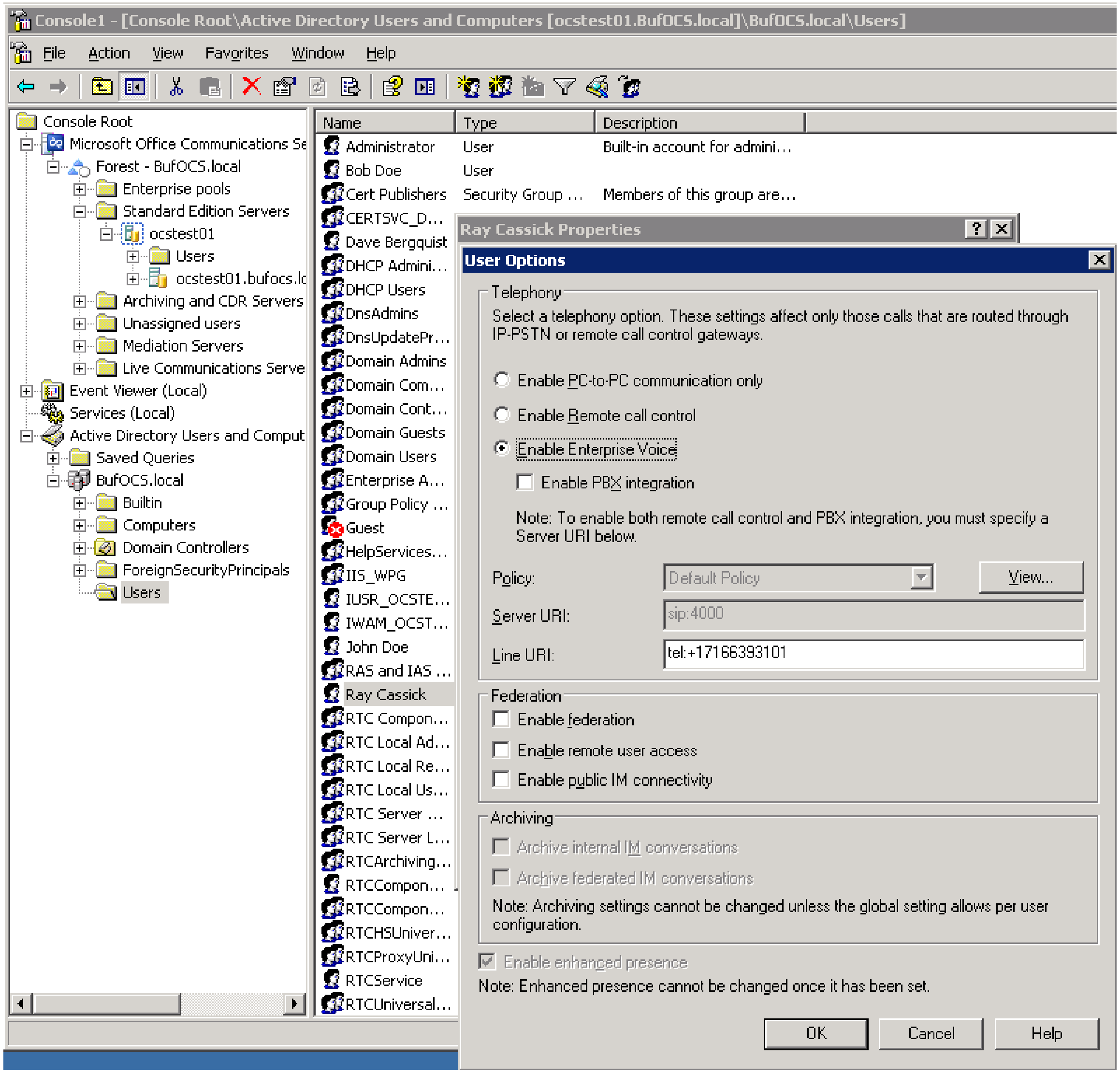
Note: Unless the dialed number from Microsoft® OCS client (such as Microsoft® Office Communicator) is in E.164 format, Microsoft® OCS must find a normalization rule to convert the dialed number to E.164. If no rule is found *or matched, outbound calls will fail. In this case, Dialogic® Diva® Diagnostics 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:

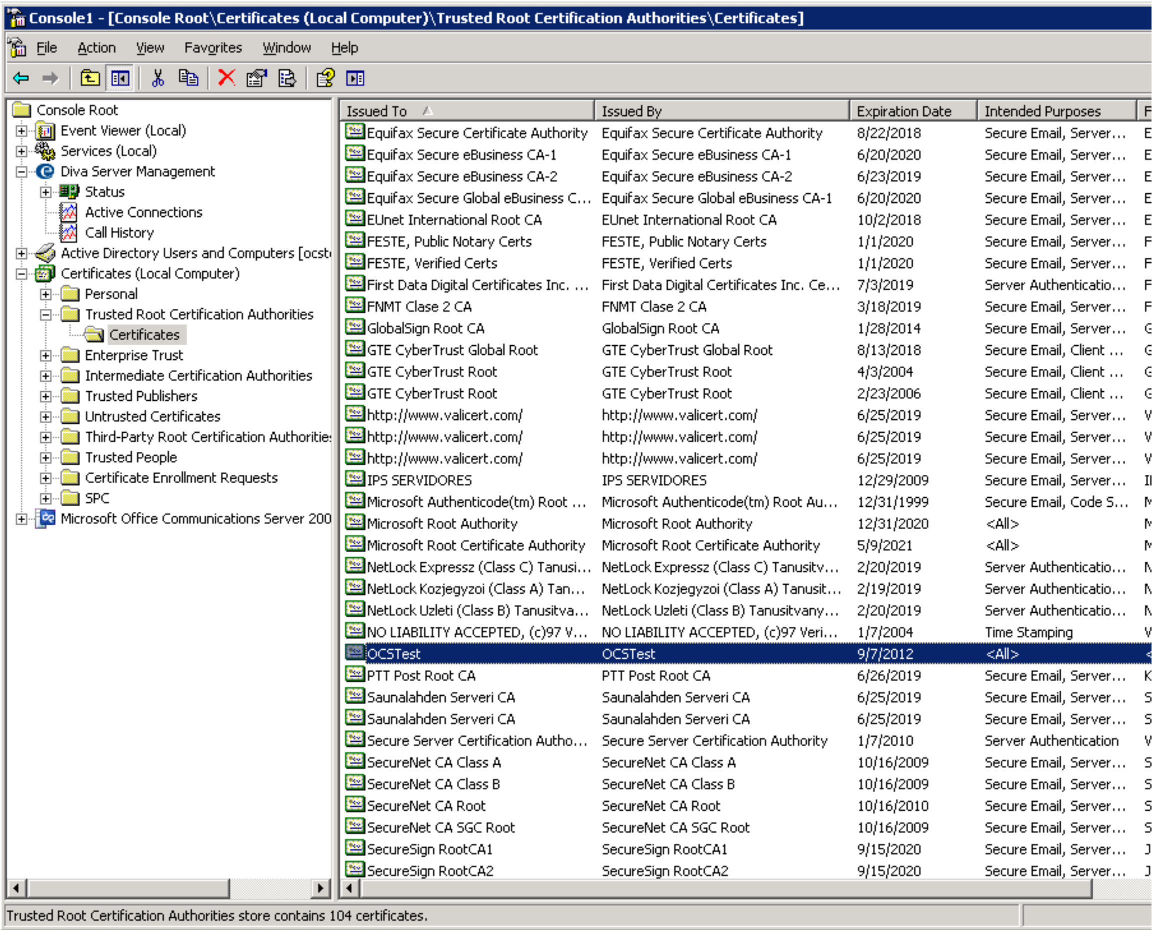
Step 1: Install the Microsoft® Mediation Server software.

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.

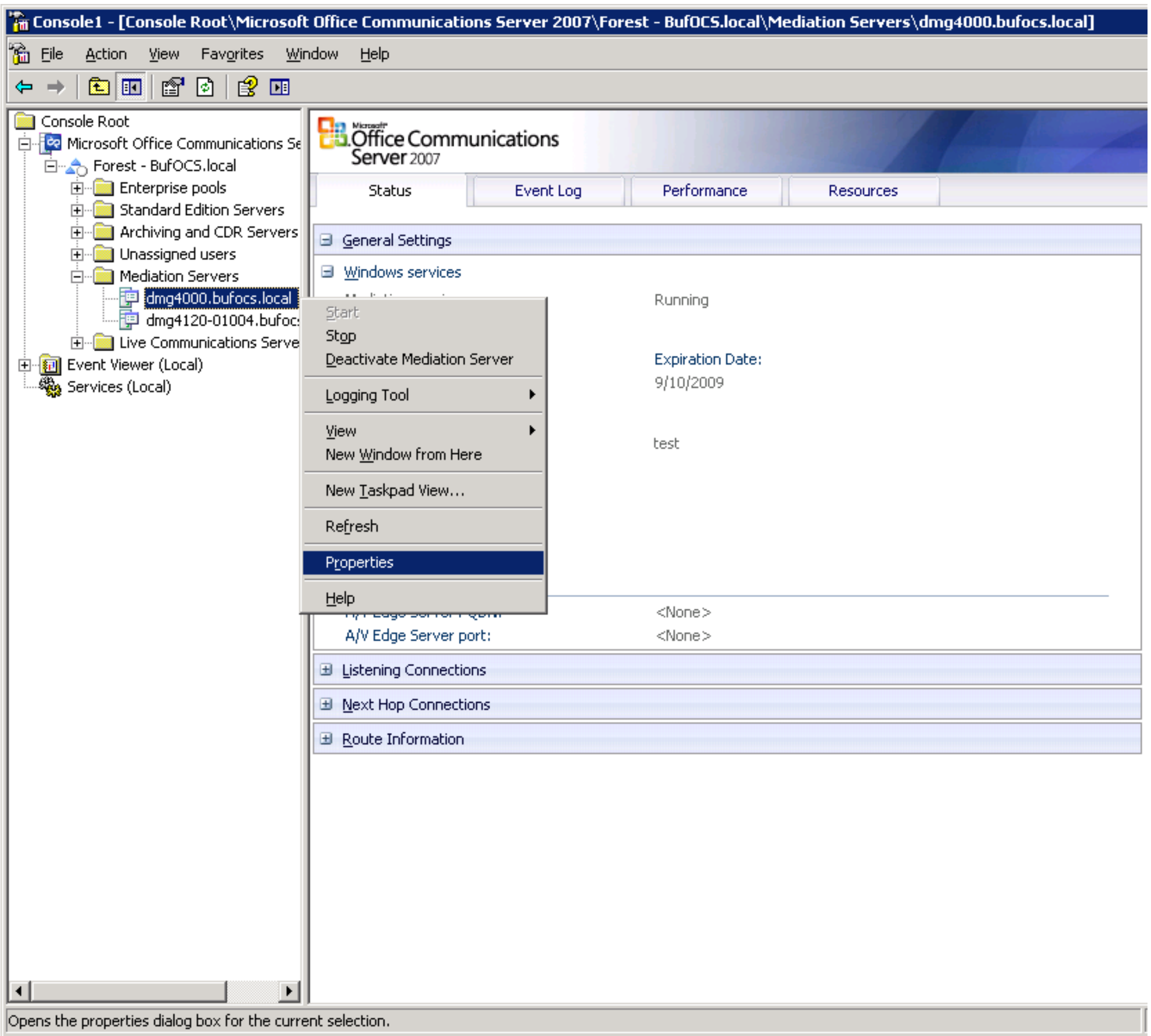


4. Create the certificate request for the Microsoft® Mediation Server:
   - Run Deployment Wizard, click step 4.
   - Select the option `Create a new certificate`.
   - Select the option `Send the request immediately to an online CA`.
   - Complete the settings in the blank.
   - Click `Assign` to complete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

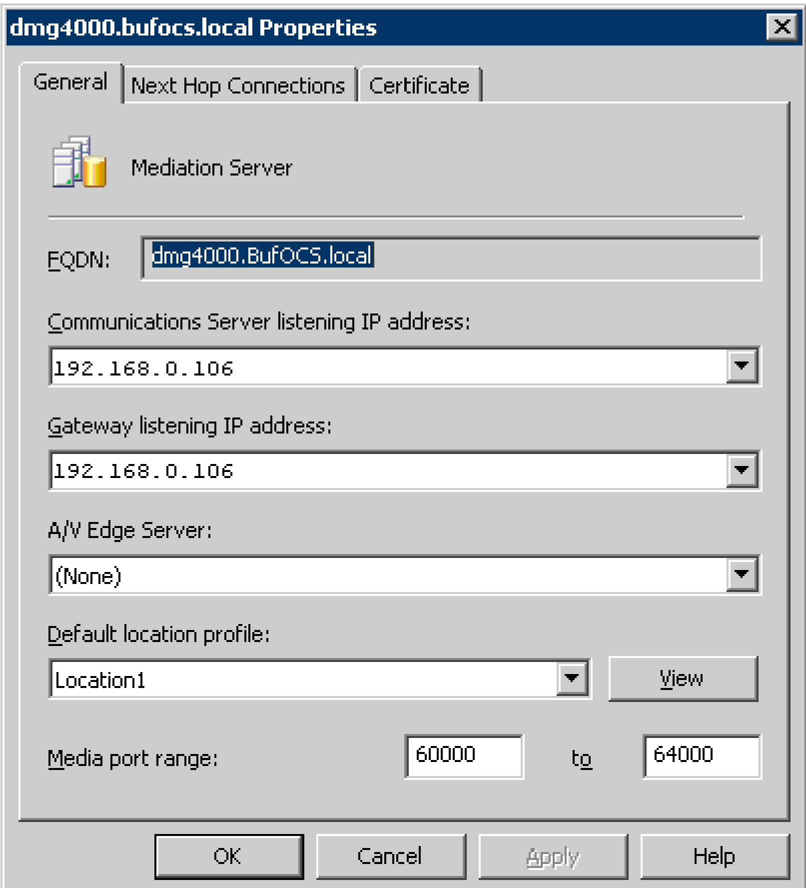
Note: If you receive the error message "certificate expired or is not yet valid" when you click the assign button at the end of step 4, check the time/time zone configured for your Microsoft® Mediation Server is correct, then run the Deployment Wizard again or click `Certificates` *in* `Available tasks`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

8.2 Configuration

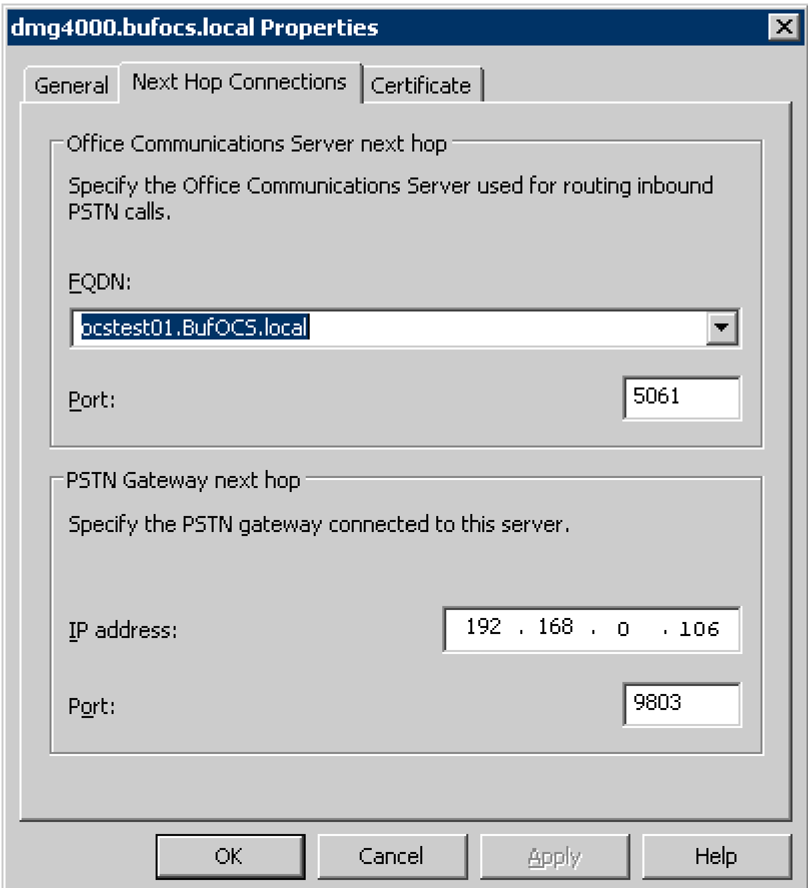
From the MMC snap-in, right-click the detected Microsoft® Mediation Server and select `Properties`.



Configure the following settings on the `General` tab:

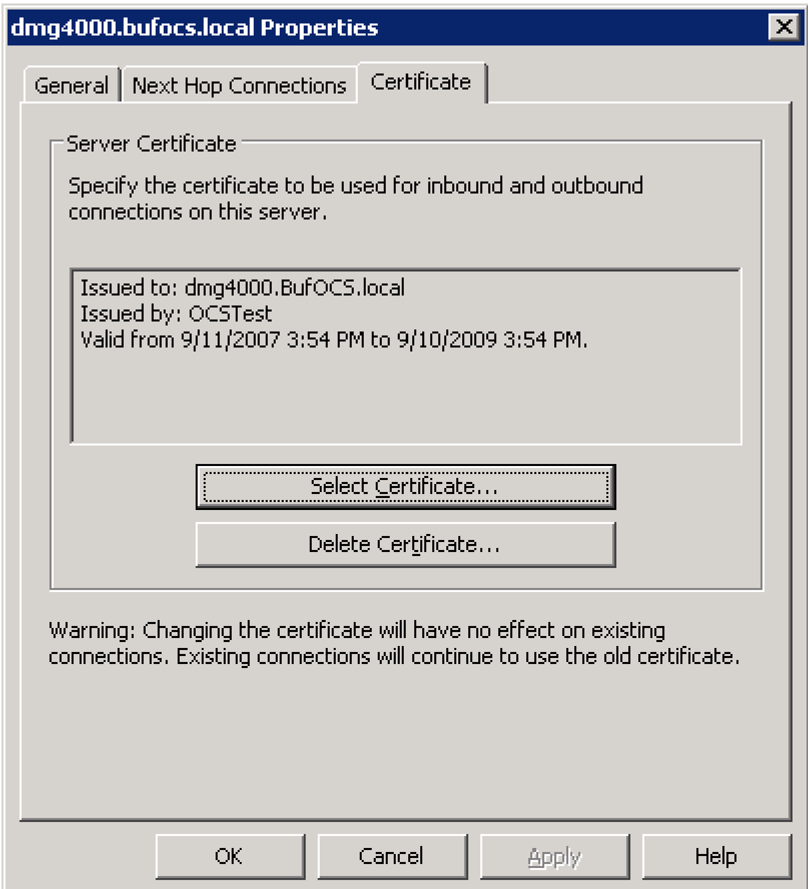


Click the `Next Hop Connections` tab and configure the following:



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

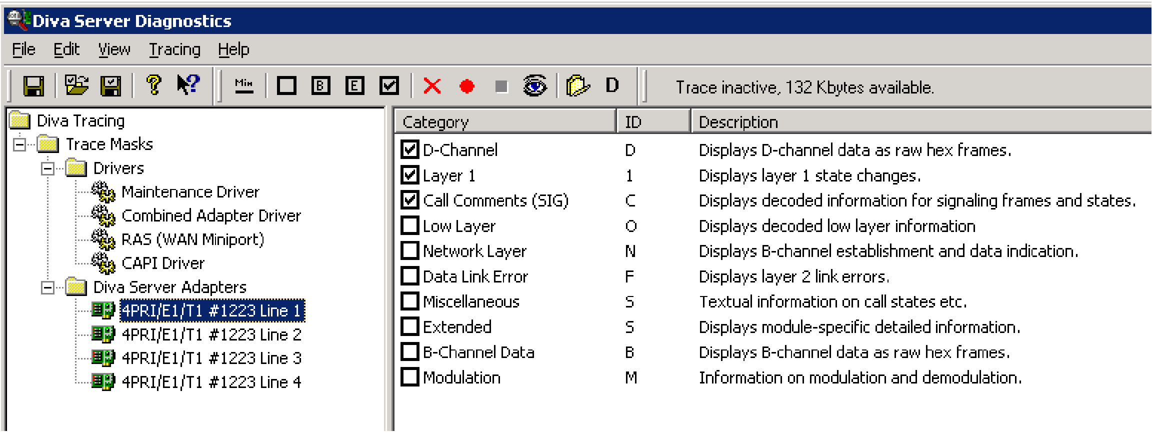
10.1 Important Debugging Tools

- **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
```