



# **Dialogic<sup>®</sup> 4000 Media Gateway Series Integration Note**

**Avaya S8500**

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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

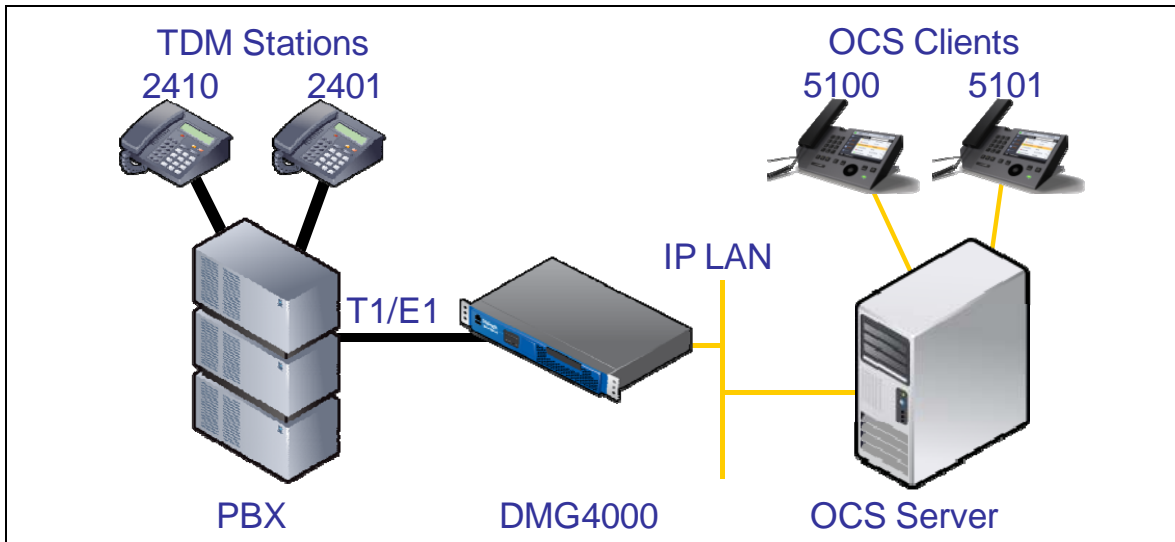
PBX Vendor	Avaya
Model(s)	S8500
Software Version(s)	Comm manager 3.0 – R013x.00.1.346.0
Additional Notes	N/A

### 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.

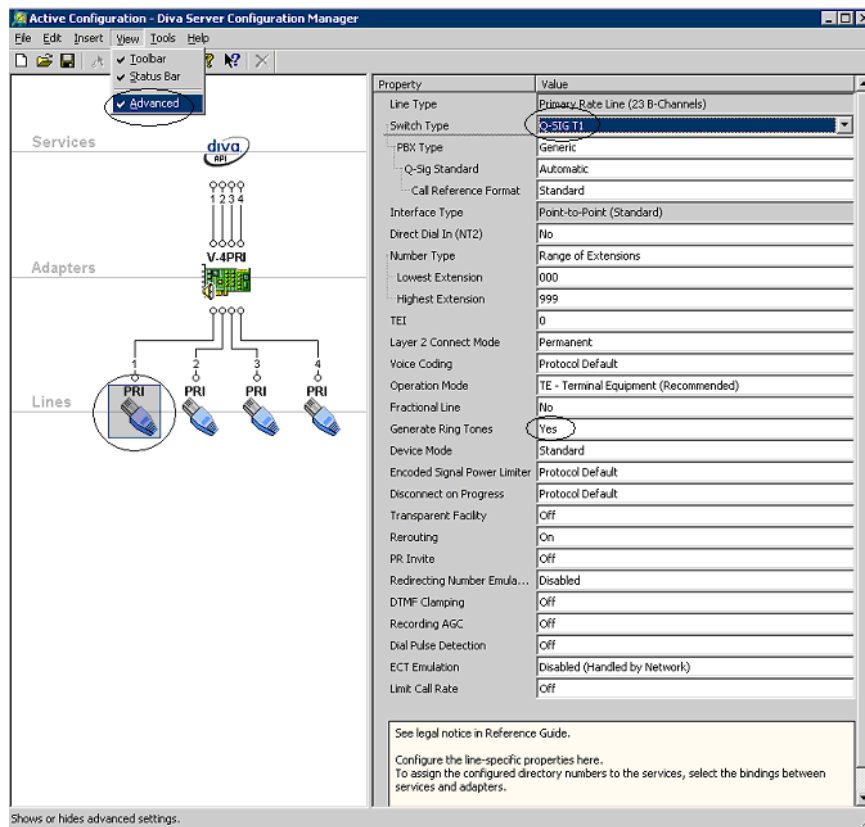
### 5.1 Dialogic® Diva® Media Board Configuration

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 screenshot shows the Dialogic SIPcontrol Configuration web interface. The main content area is titled 'Network Interface Configuration' and contains a table with the following data:

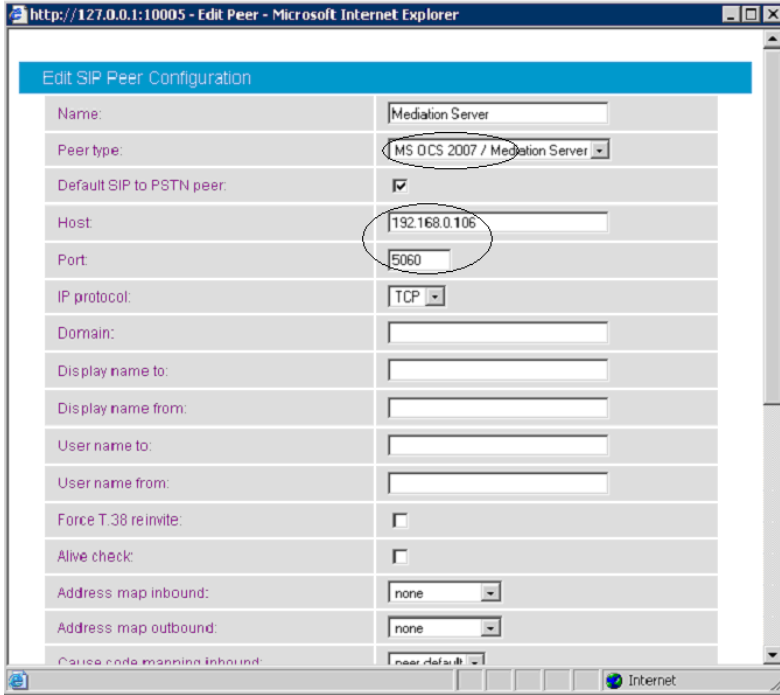
Name	Device	IP Address	Protocol	SIP Listen Port	Enabled
Inte(R) PRD1000 EB Network Conn	Inte(R) PRD1000 EB Network Connection with I/O Acceleration	192.168.0.106	all	9803	<input checked="" type="checkbox"/>
Local Loopback Interface	Local Loopback Interface	127.0.0.1	all	5060	<input type="checkbox"/>

Below the table, 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.

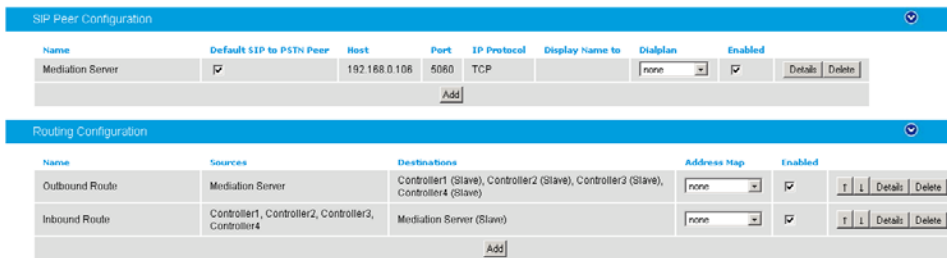
## 5.2.2 SIP Peer Configuration

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

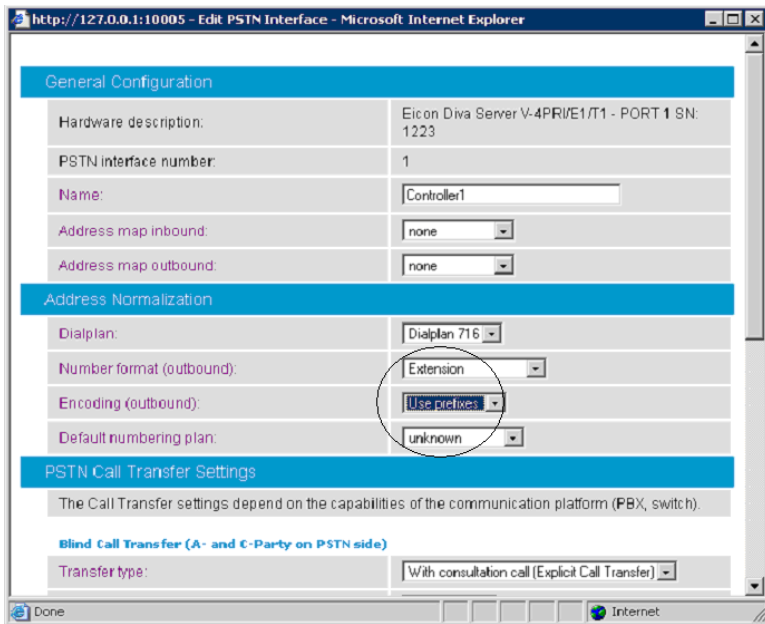
The screenshot shows a web browser window titled "http://127.0.0.1:10005 - Edit Dialplan - Microsoft Internet Explorer". The main content area is a form titled "Edit Dialplan Configuration". The form fields are as follows:

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"/>

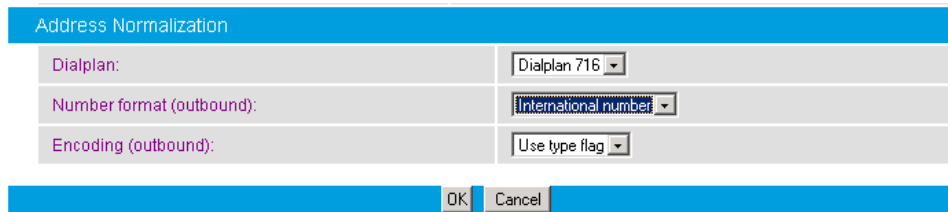
At the bottom of the form are "OK" and "Cancel" buttons. The browser's status bar shows "Done" and "Internet".

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

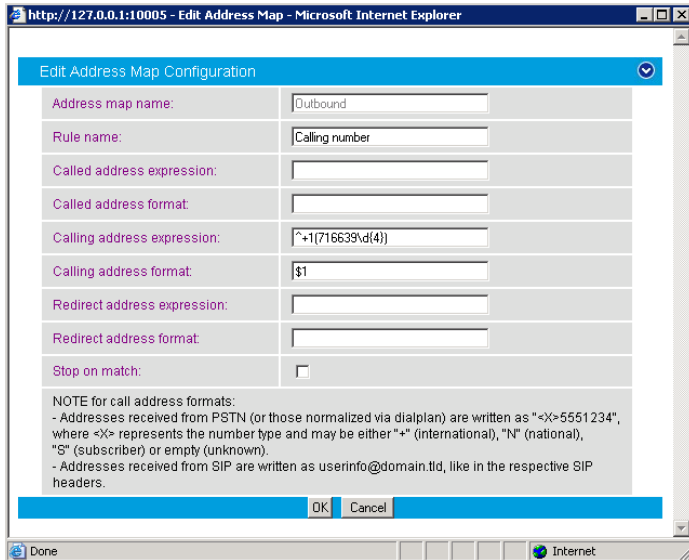
Below is RegEx for the conversion tables above.

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

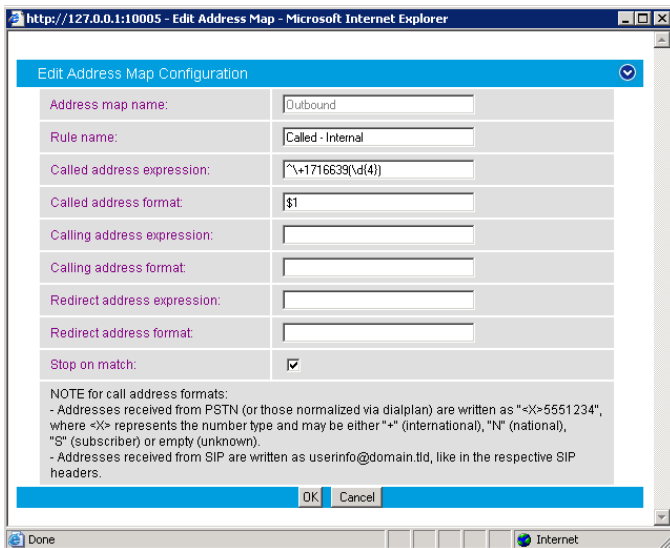
Below are the configured address maps for outbound calls. The order of the below four sub rules and the stop on match check mark are relevant:

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:

Routing Configuration					
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.

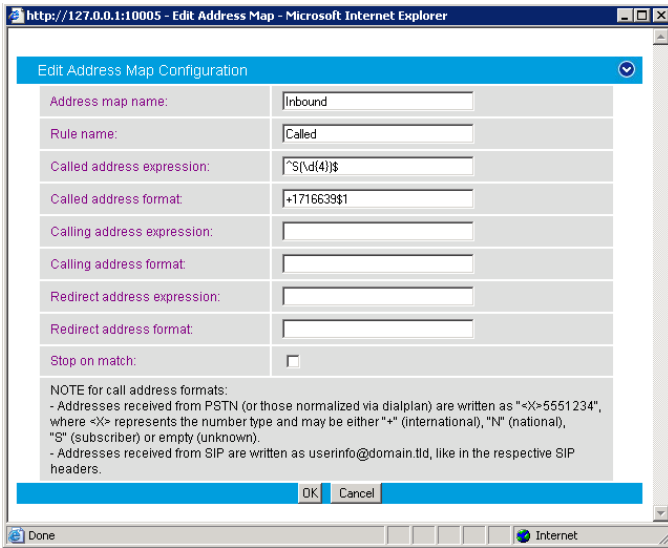
Called number	From PBX	To Microsoft® OCS
Internal	xxxx (with subscriber type of number)	+1716639xxxx

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:

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"/>	↑	↓
	Add Rule				
Inbound	Called	<input type="checkbox"/>	<input checked="" type="checkbox"/>	↑	↓
	Calling - Internal	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑	↓
	Calling - National	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑	↓
	Calling - International	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑	↓
	Add Rule				
Add					



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:	<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

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:	<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



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

Edit Address Map Configuration

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

Done Internet

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"/>	T L Details
Inbound Route	Controller1, Controller2, Controller3, Controller4	Mediation Server (Slave)	Inbound	<input checked="" type="checkbox"/>	T L 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

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.

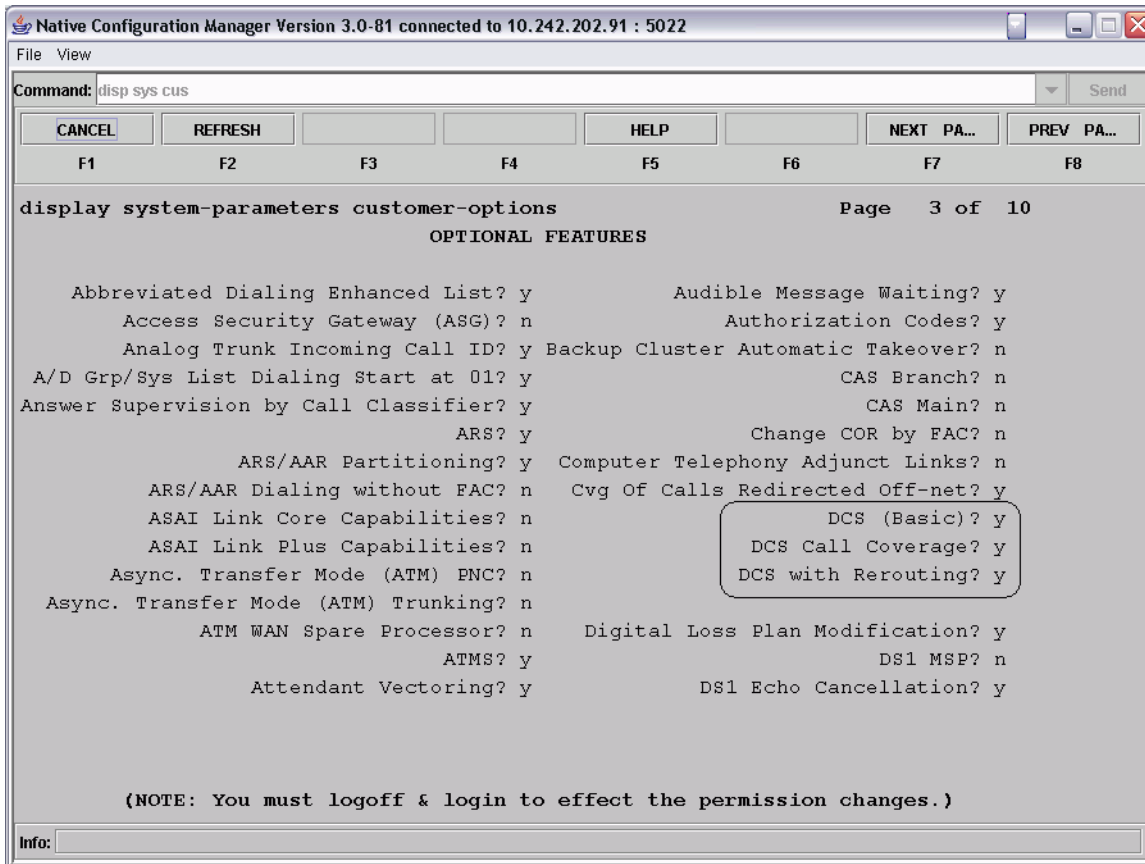
All PBX programming is done via a Windows<sup>®</sup>-based Java application. The basic commands that you will encounter on the PBX to perform these actions are:

change; delete; and add

### 6.1 Setting Up System Parameters and Customer Options

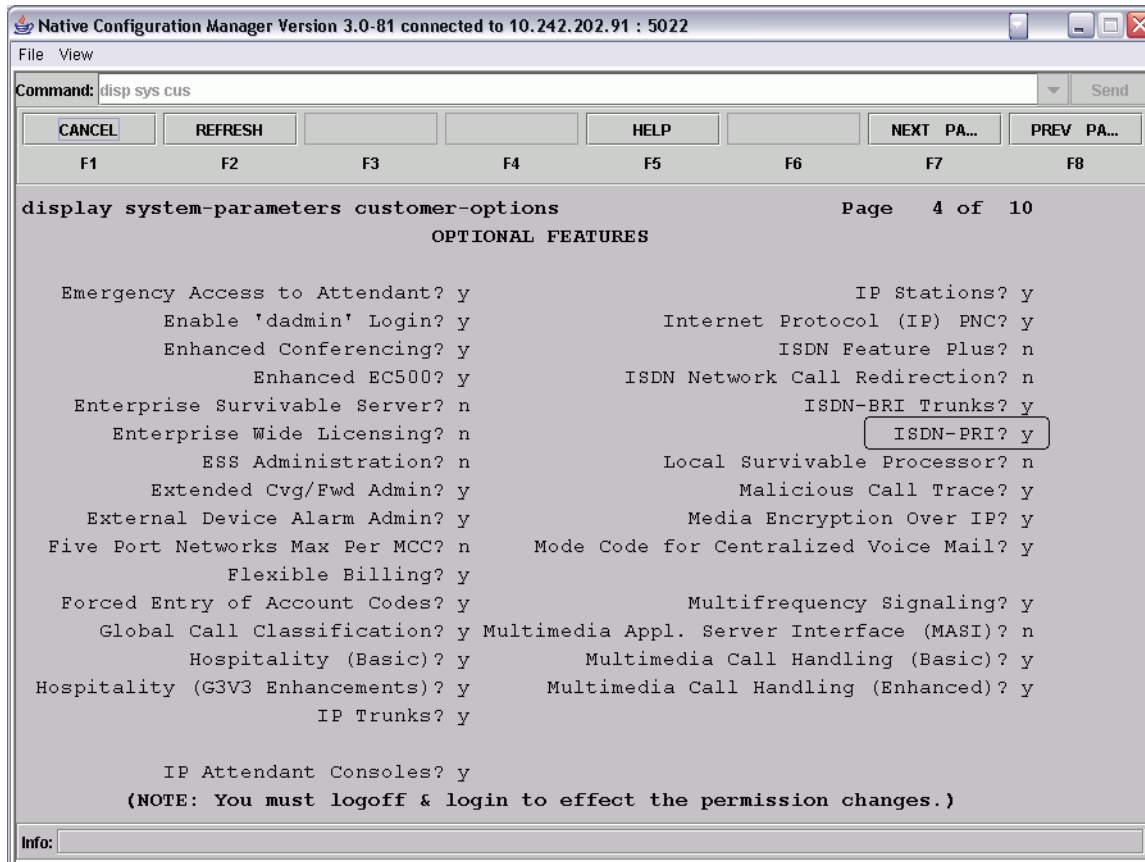
Use the `display system-parameters customer options` command such that several Q.SIG and trunk-specific parameters needed for the integration to operate properly are enabled.

*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.*

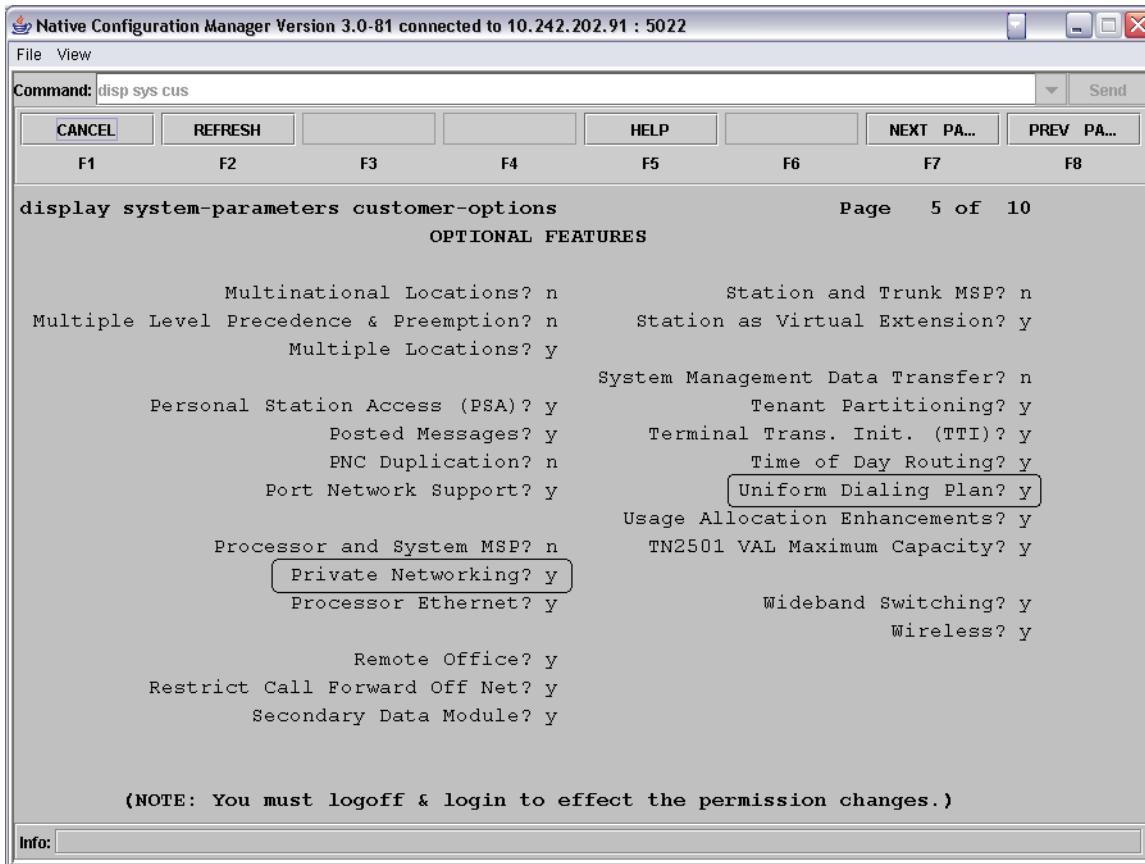


Review the DCS (Basic), DCS Call Coverage, and DCS with Rerouting options. They must be set to y for this integration to work properly. If they are not enabled, then contact your Avaya support representative to have these features enabled.

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

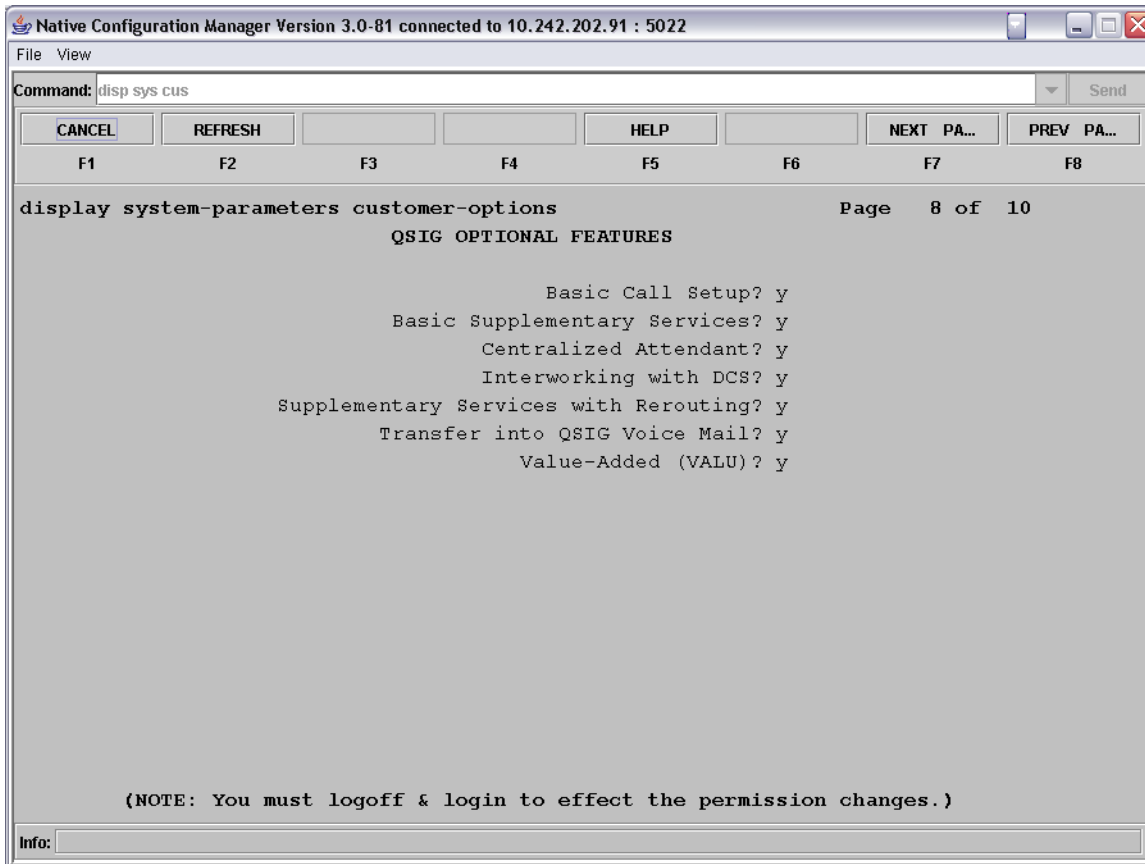


Review the ISDN-PRI option. It must be set to `y` for this integration to work properly. If this option is not enabled, then contact your Avaya support representative to have these features enabled. Press the NEXT PAGE soft key, and the next page is displayed.



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

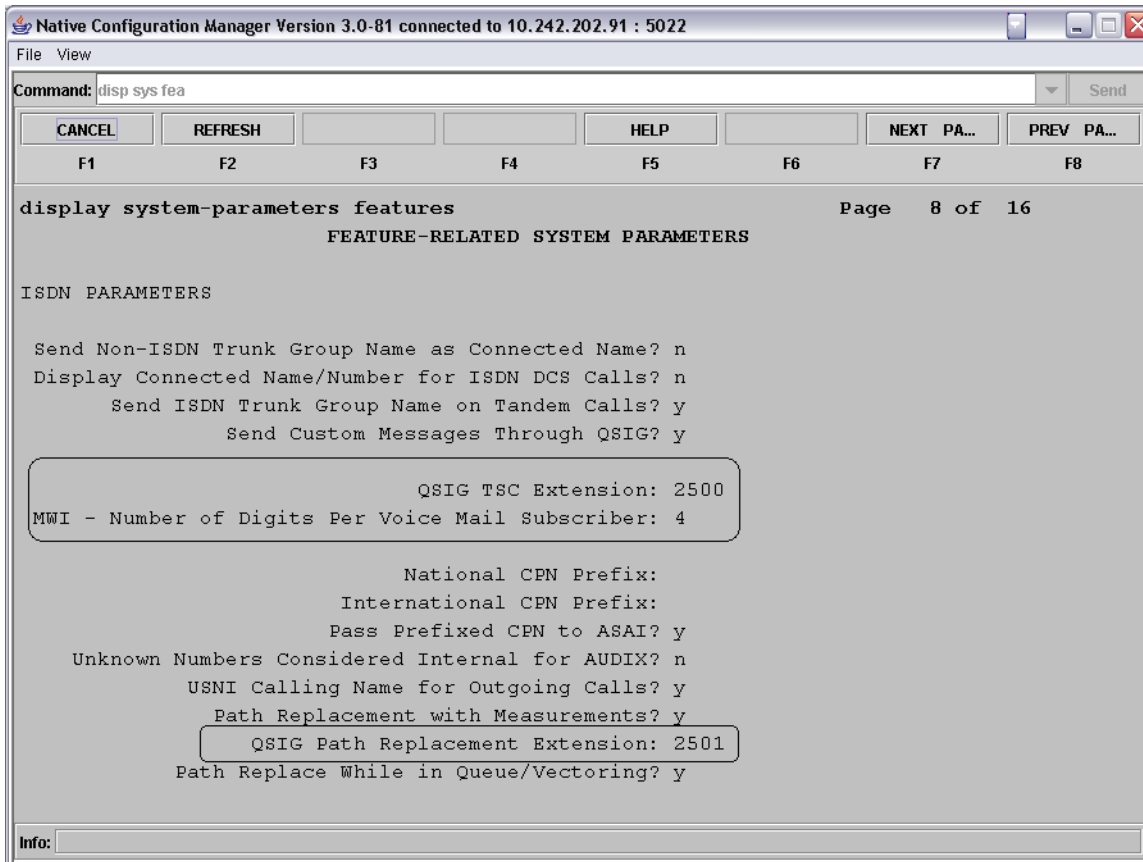
Press the NEXT PAGE soft key, until page #8 (QSIG Optional Features) is displayed.



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.

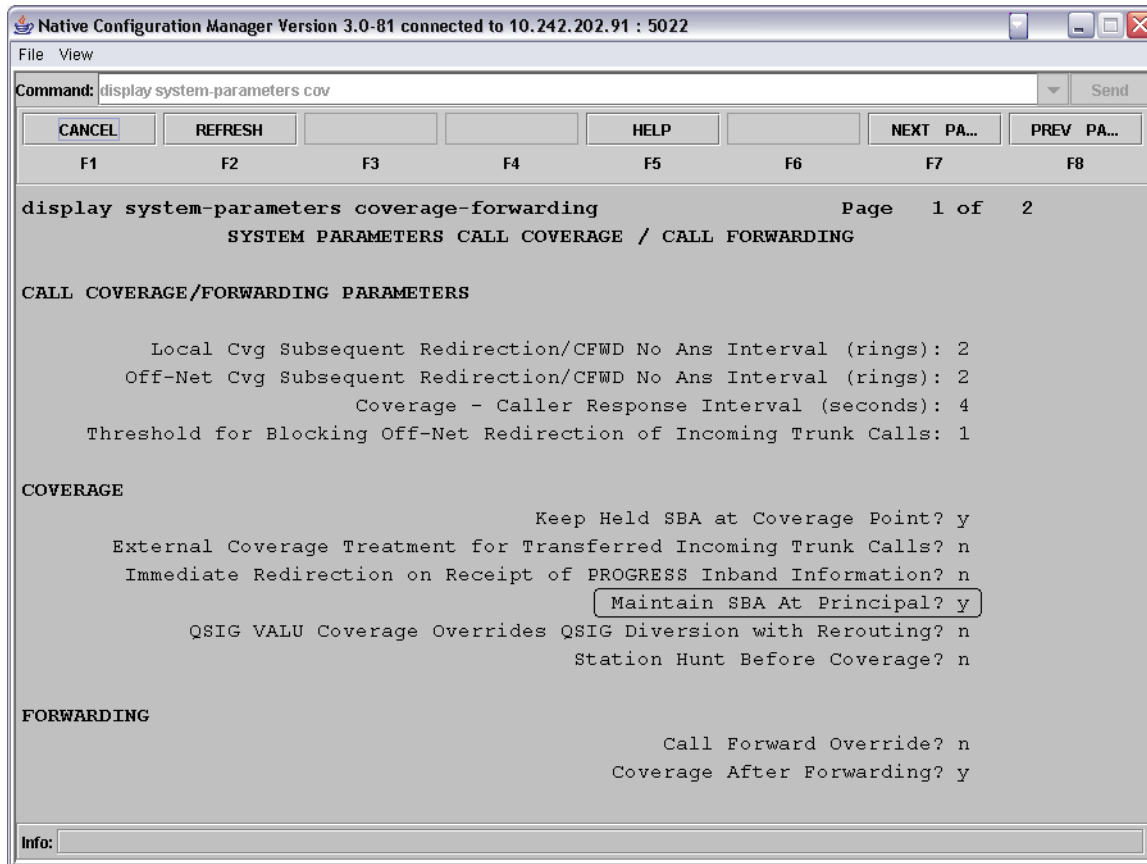
Press the `NEXT PAGE` soft key, until you get to the `ISDN PARAMETERS` page.



- 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.

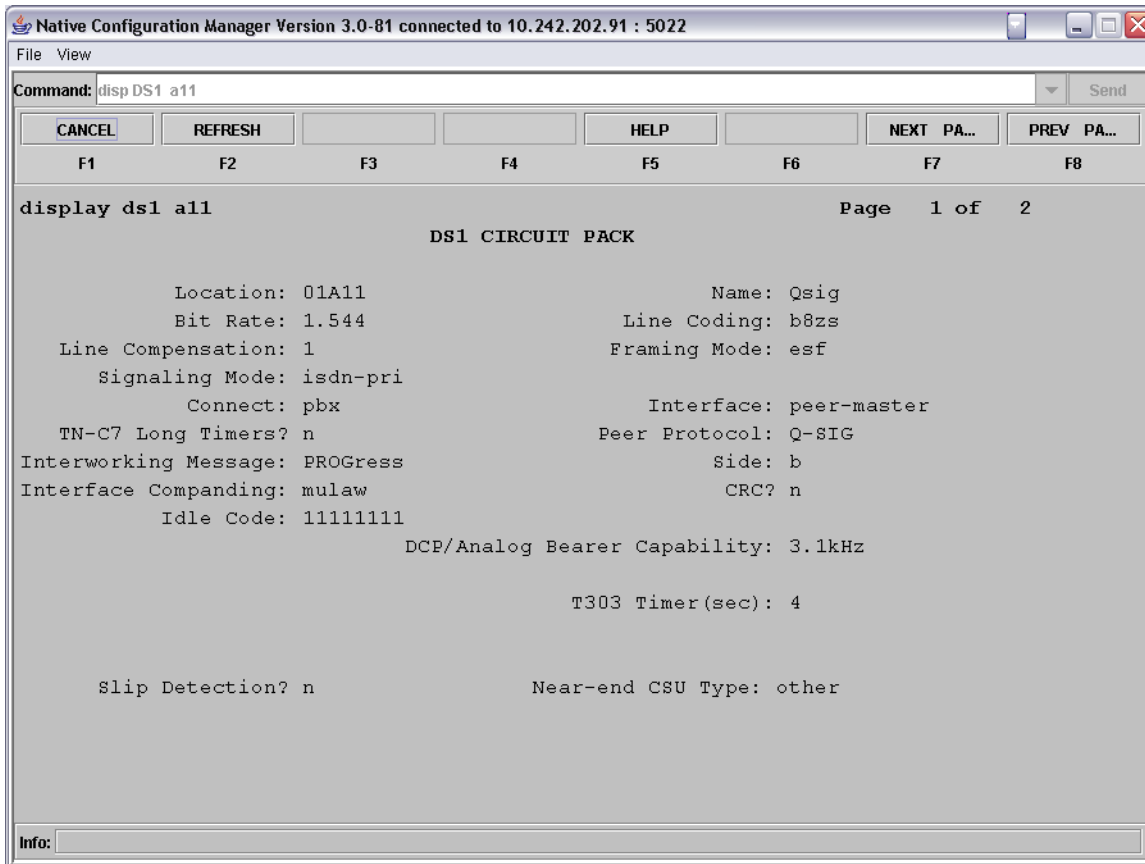
## 6.2 Administrating the T1 Interface

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
```

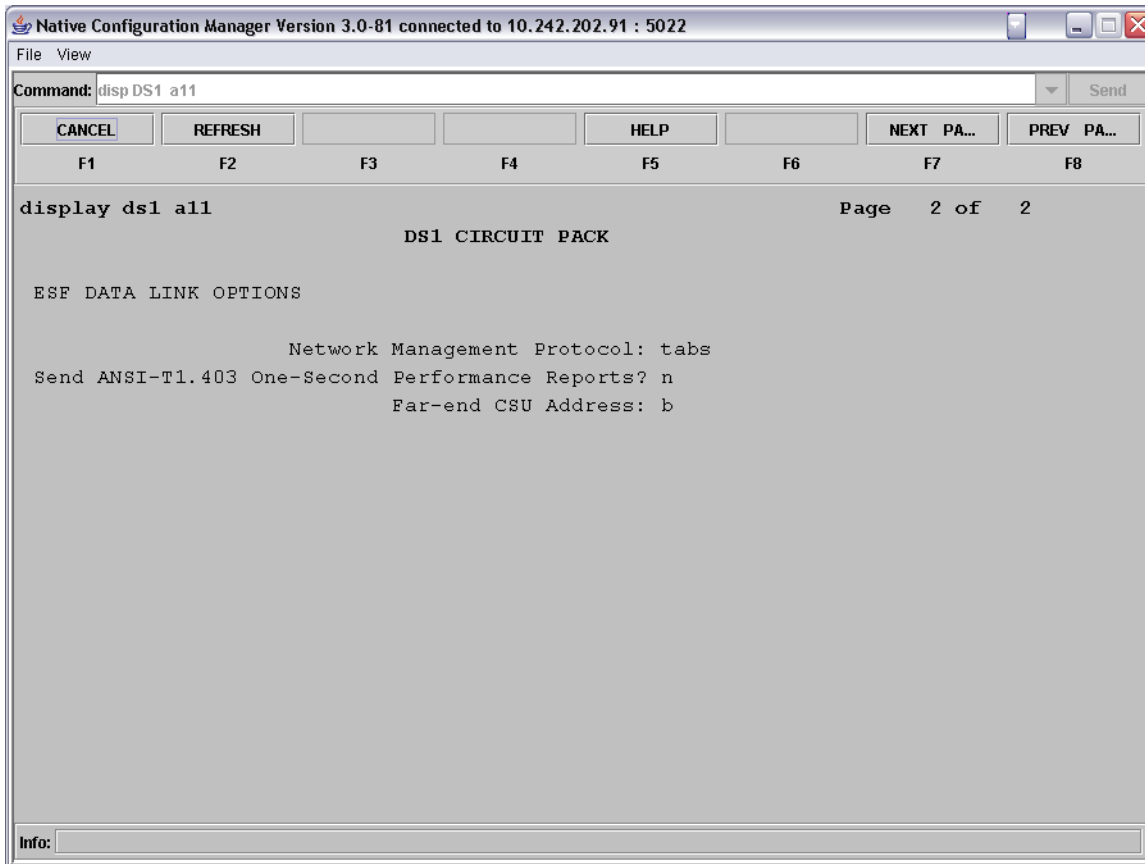
Add a ds1 circuit card into the configuration that is located in cabinet 'a' and is in slot #11 of that cabinet.





- 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*)

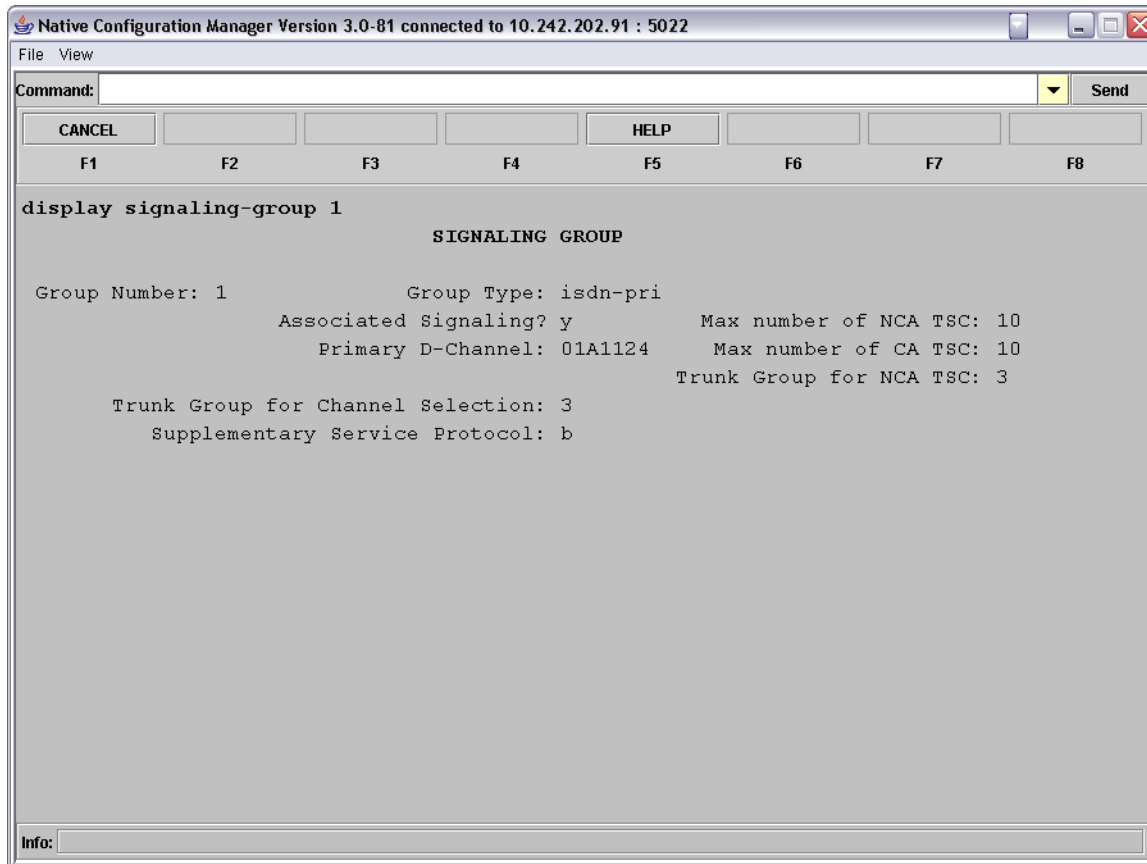
Press NEXT PAGE to move to the next page of settings.



- 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.

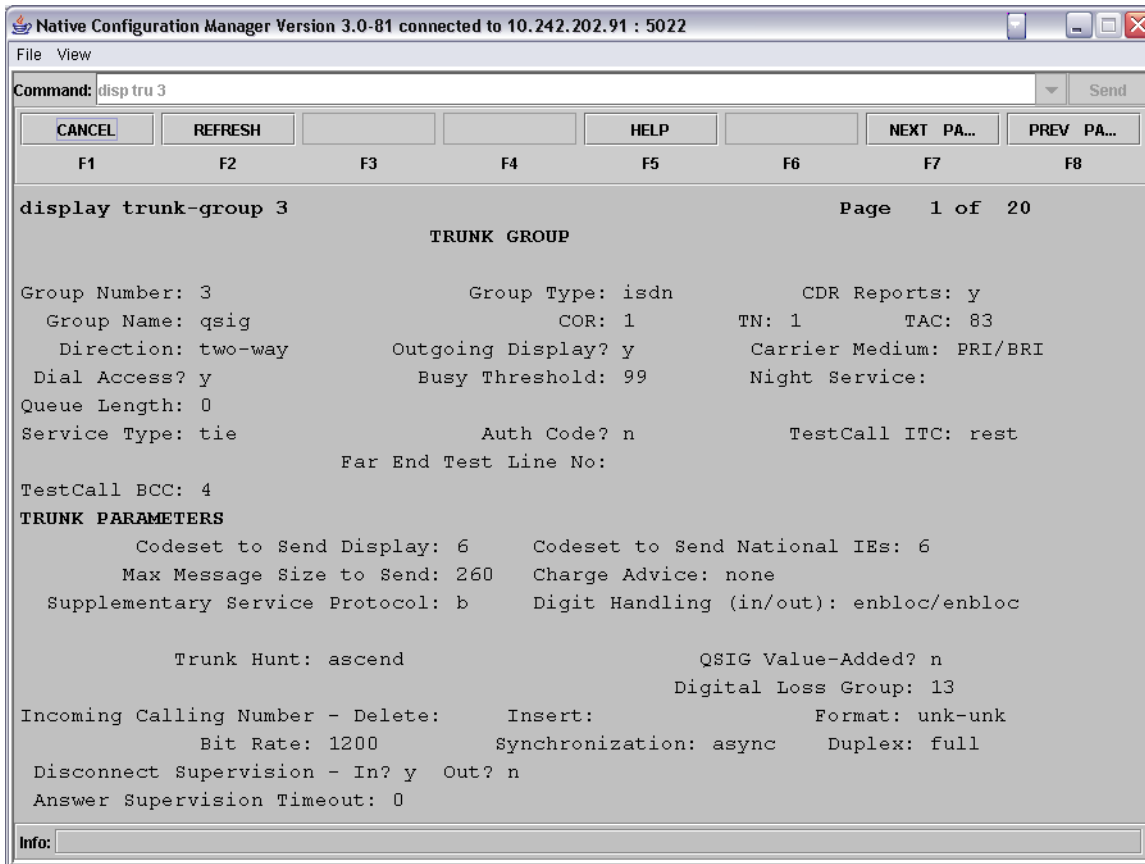


- In the Group Type field, enter isdn-pri.
- In the Associated Signaling field, enter y.
- In the Max number of NCA TSC field, enter 10.
- In the Primary D-Channel field, enter XXXXX24
  - where XXXXX24 is the 24<sup>th</sup> port number of the T1 board.
- In the Max number of CA TSC field, enter 10.
- 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.

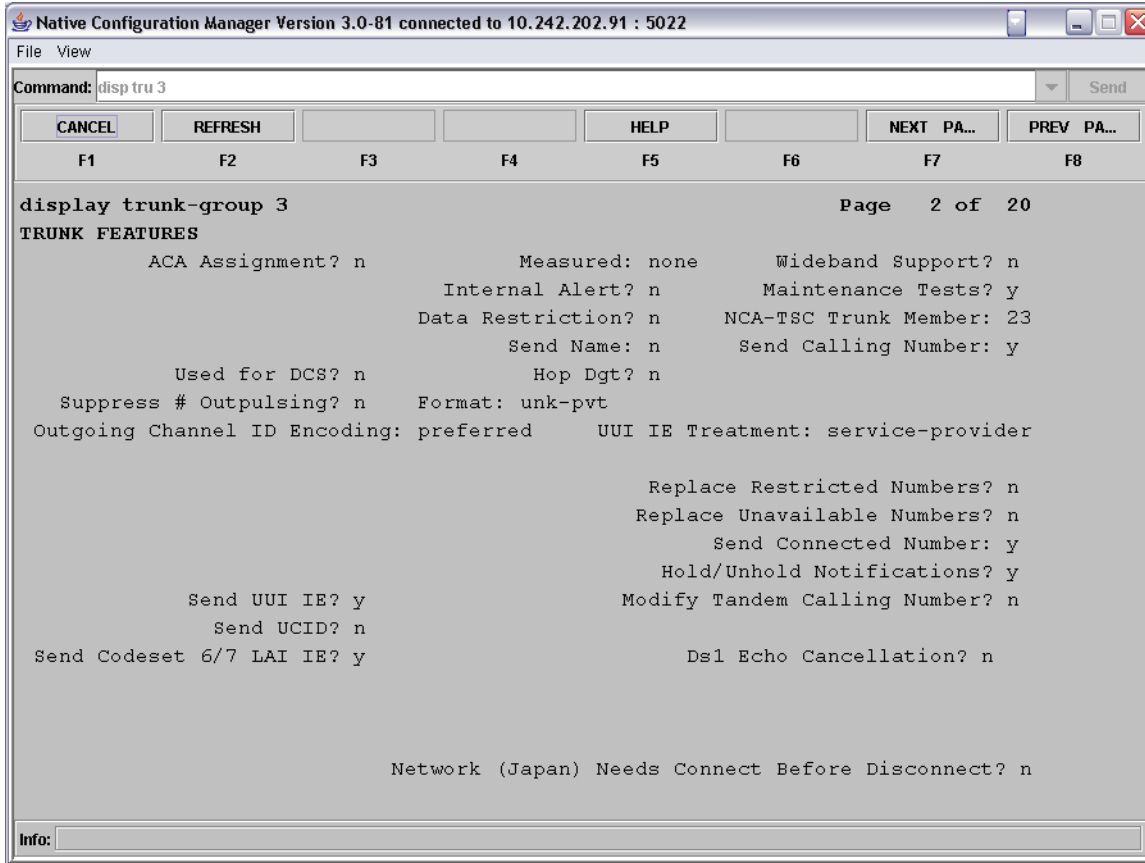
Before moving to the next command, be sure to save your work.

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.



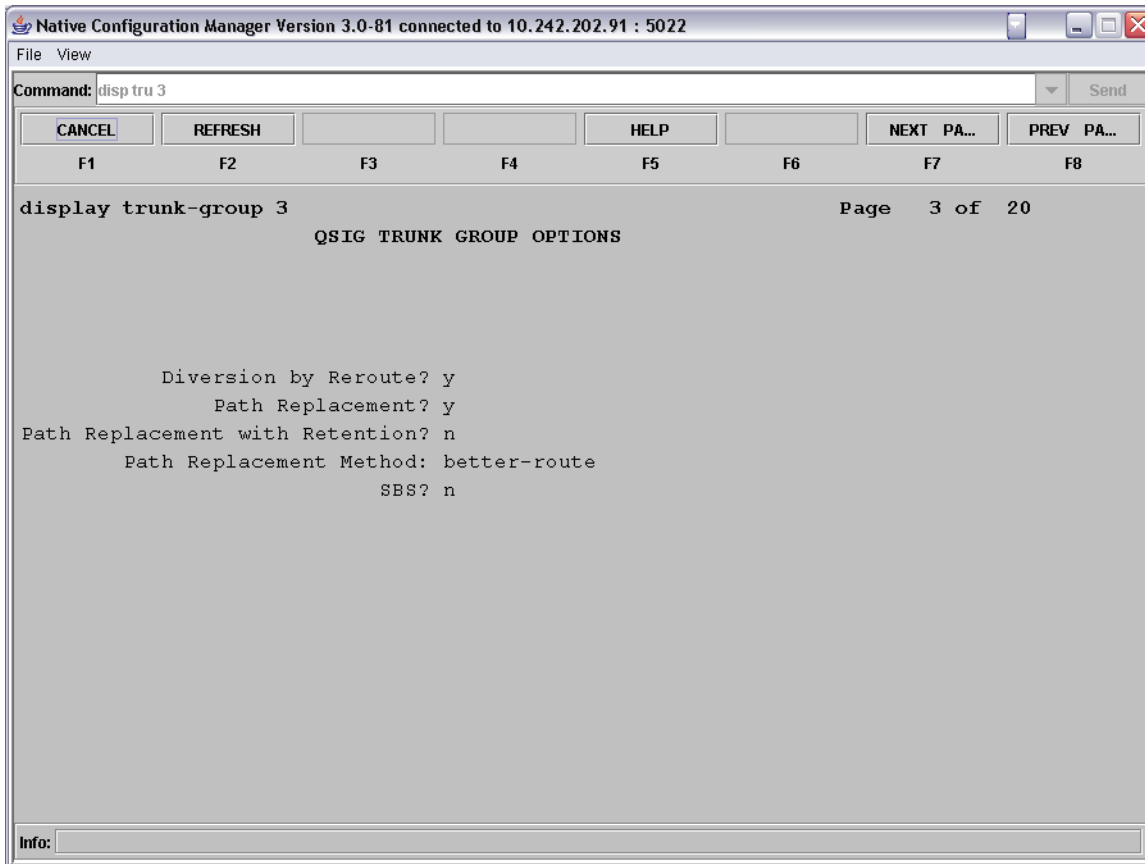
- In the Group Type field, enter isdn.
- In the Group Name field, enter a text string to give the group a name.
- In the COR field, enter the number of a class of restriction to be used, if there is going to be one.  
*Note: Setting any restrictions via a COR can effect how the trunk operates so these 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.

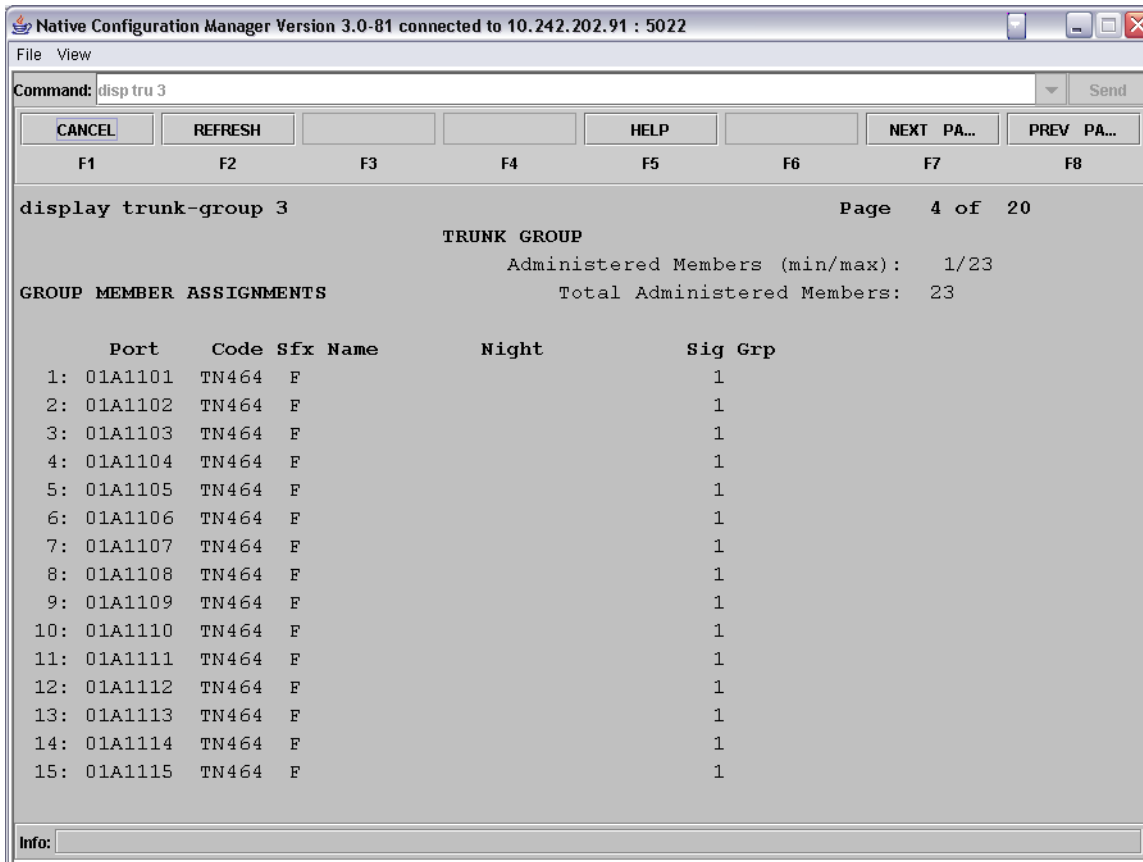
Press NEXT PAGE to advance to the next page of settings.



- 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.





Native Configuration Manager Version 3.0-81 connected to 10.242.202.91 : 5022

File View

Command: `cha rou 15` Send

CANCEL REFRESH ENTER CLEAR Fl... HELP NEXT PA... PREV PA...

F1 F2 F3 F4 F5 F6 F7 F8

change route-pattern 15 Page 1 of 3

Pattern Number: 15 Pattern Name: T1 qsig

SCCAN? n Secure SIP? n

Grp	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/	IXC
No	Mrk	Lmt	List	Del	Digits			QSIG	
					Dgts			Intw	
1:	3	0						y	
2:								n	
3:								n	
4:								n	
5:								n	
6:								n	

BCC	VALUE	TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	No.	Numbering	LAR	
0	1	2	3	4	W	Request		Dgts	Format	Subaddress	
1:	y	y	y	y	y	n	y	as-needed	rest	unk-unk	none
2:	y	y	y	y	y	n	n		rest		none
3:	y	y	y	y	y	n	n		rest		none
4:	y	y	y	y	y	n	n		rest		none
5:	y	y	y	y	y	n	n		rest		none
6:	y	y	y	y	y	n	n		rest		none

Info: Right-click in a field to see valid entries or help text.

- 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.



Native Configuration Manager Version 3.0-81 connected to 10.242.202.91 : 5022

File View

Command: busy bo a11 Send

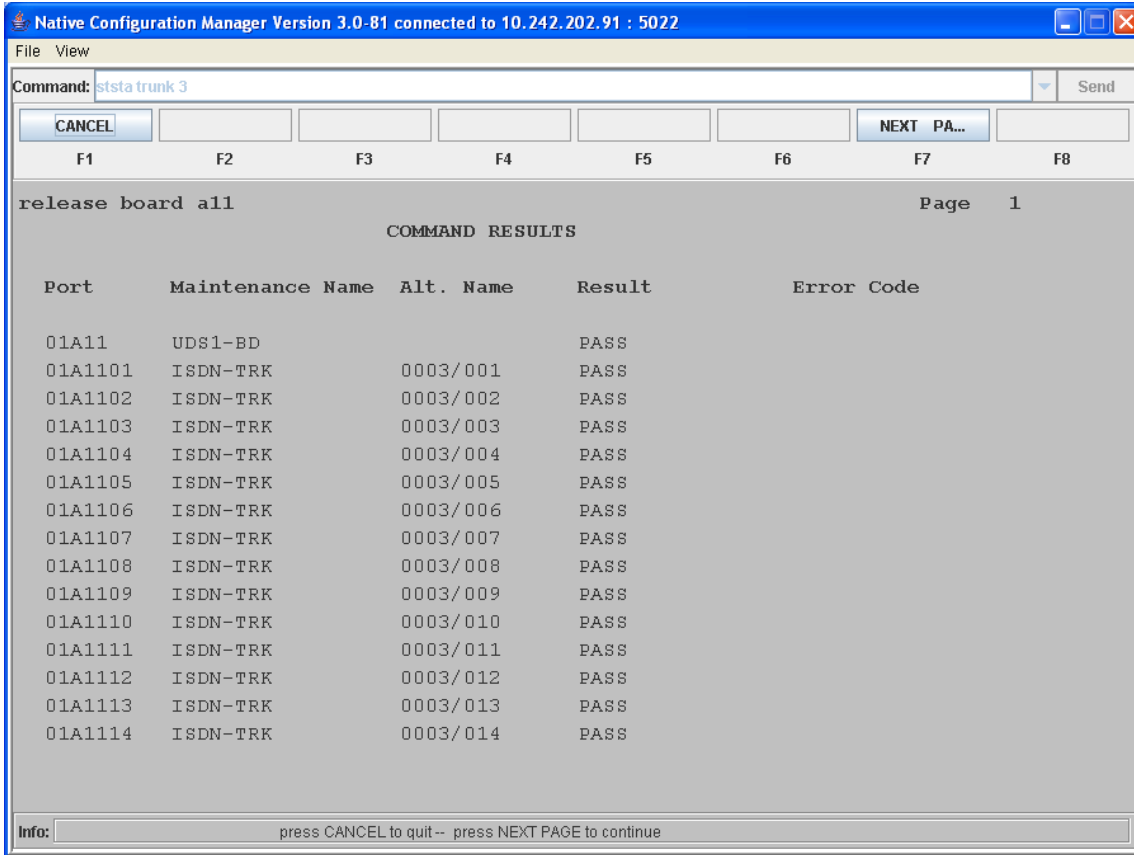
CANCEL F1 F2 F3 F4 F5 F6 F7 F8 NEXT PA...

busyout board all 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

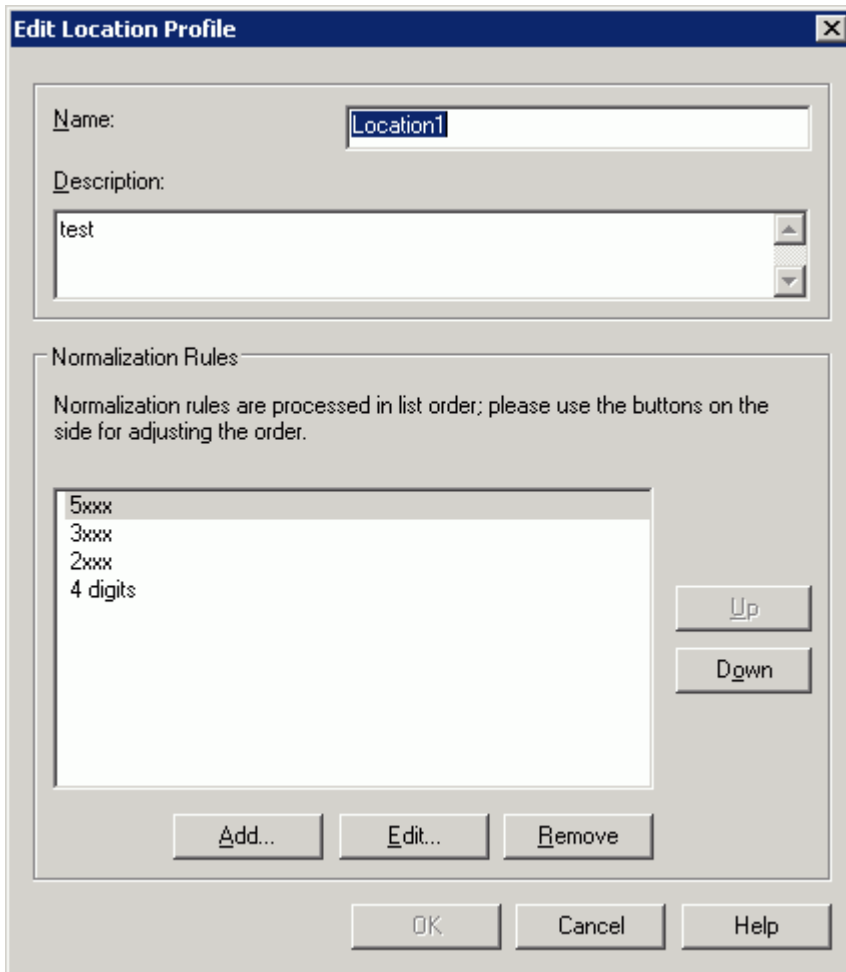
### 7.1 Steps for configuring Microsoft® OCS

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` 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:

<b>Name</b>	<b>Phone Pattern</b>	<b>Translation Pattern</b>	<b>Comments</b>
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.

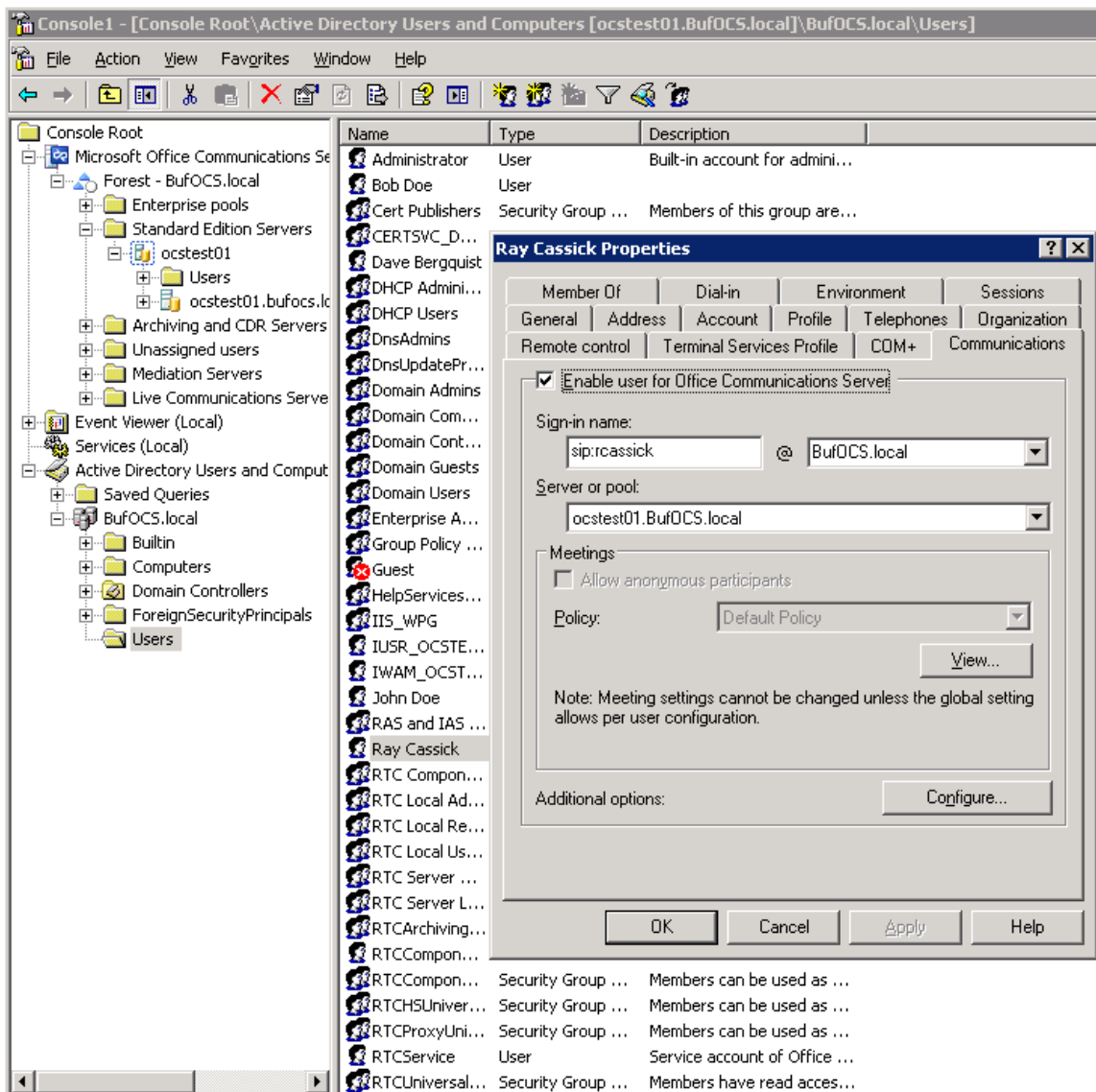
Restart the Front End Services for the above changes to take effect, including all normalization rules. This can be done from the window Services.

*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*

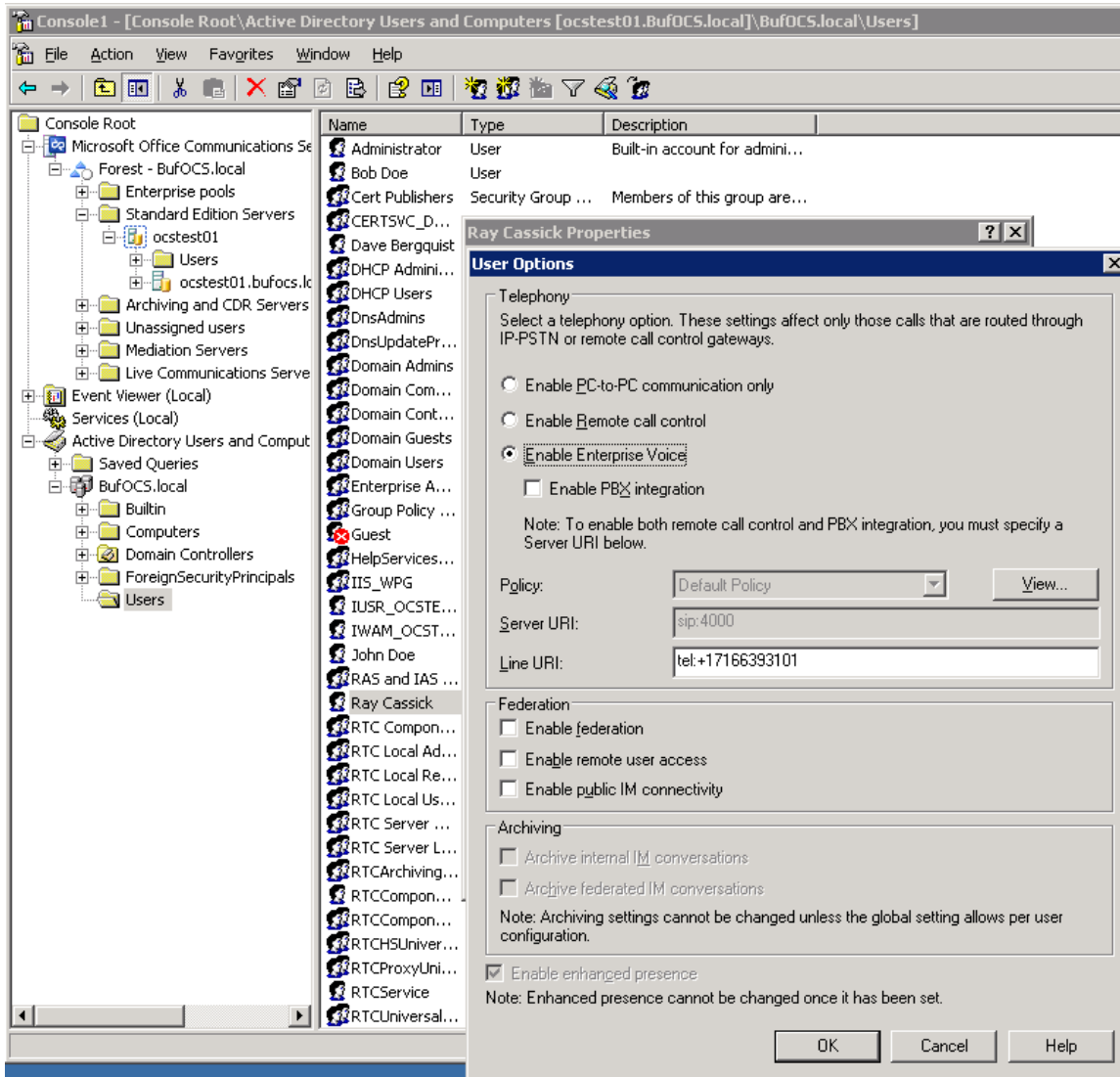
or matched, outbound calls will fail. In this case, Dialogic® Diva® Diagnostics trace will not receive an outbound SIP message, since the call will not yet have reached the SIP gateway.

## 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:

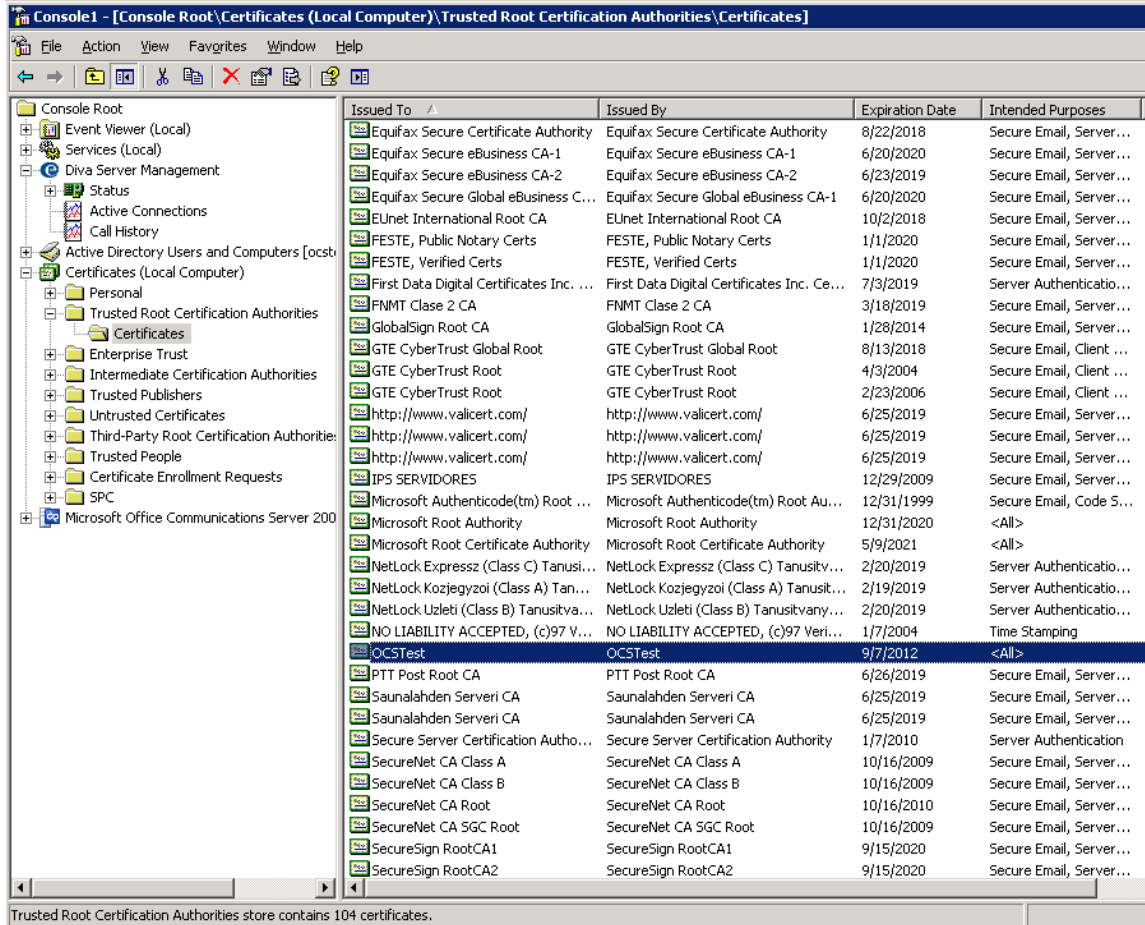
**Step 1:** Install the Microsoft® Mediation Server software.

**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.

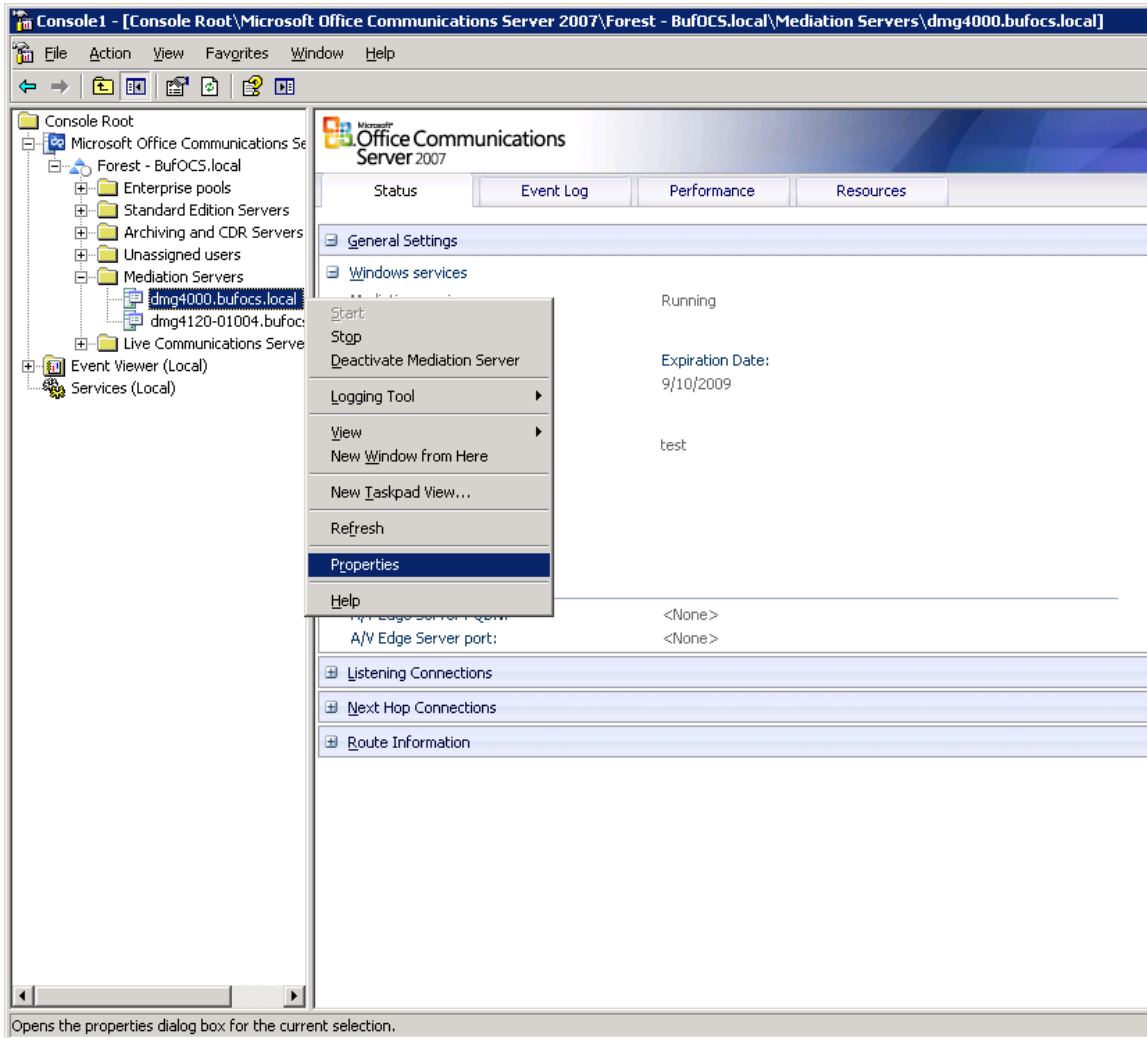


4. Create the certificate request for the Microsoft® Mediation Server:
  - Run Deployment Wizard, click step 4.
  - Select the option Create a new certificate.
  - Select the option Send the request immediately to an online CA.
  - Complete the settings in the blank.
  - Click Assign to complete the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



Opens the properties dialog box for the current selection.

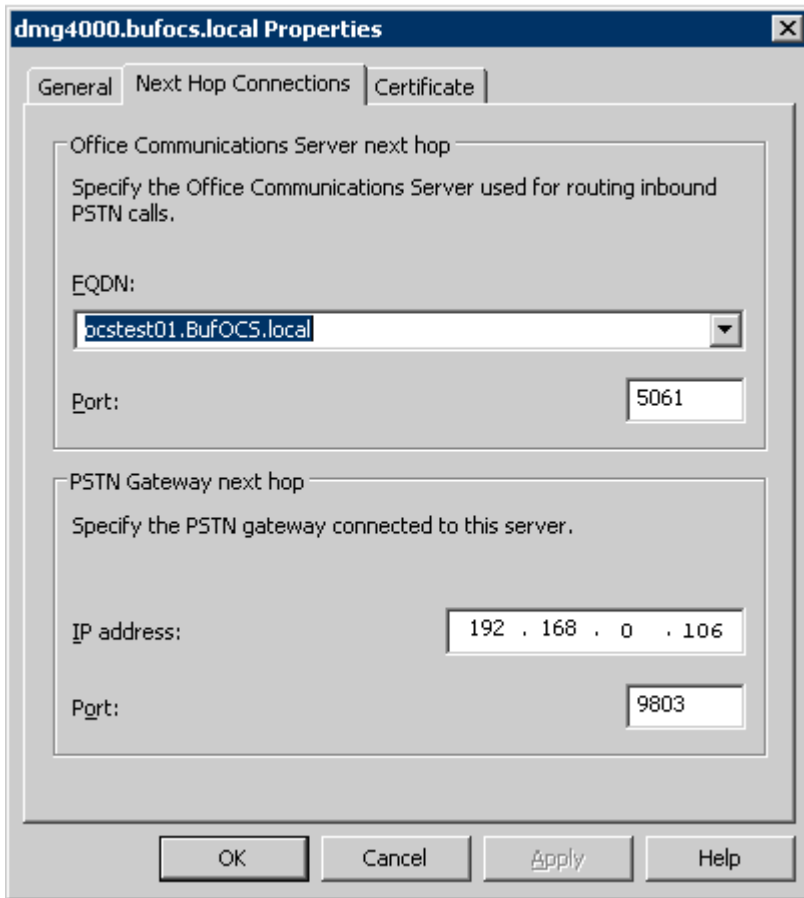
Configure the following settings on the General tab:

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 folder icon.
- EQDN:** A text box 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".

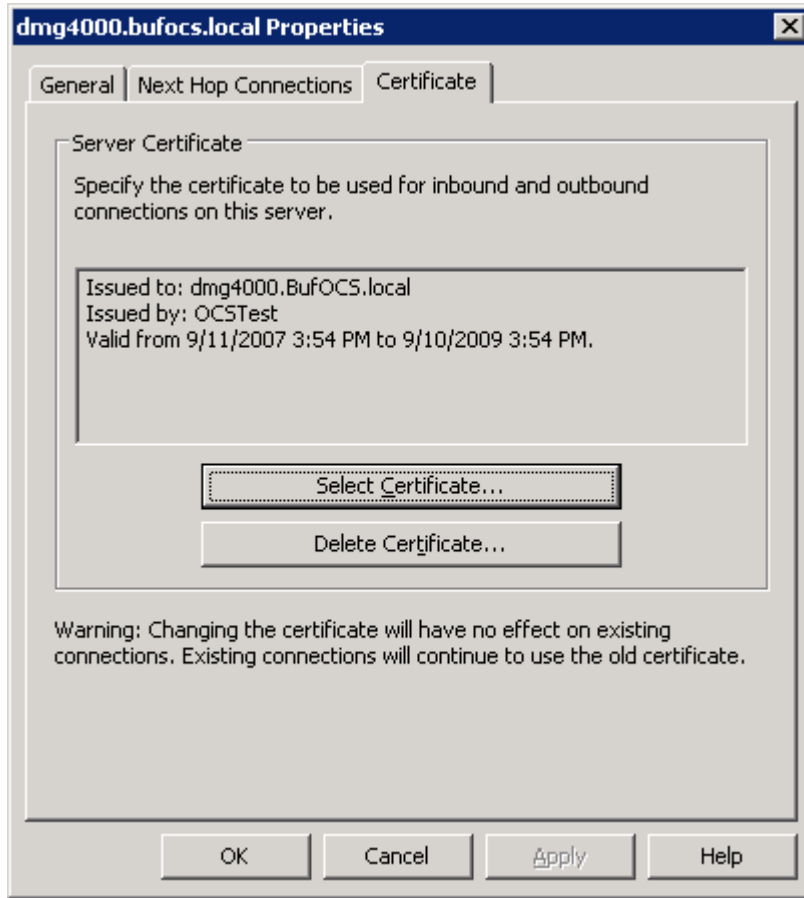
Click the Next Hop Connections tab and configure the following:



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
<b>Inbound call scenarios</b>		
1	Direct call from TDM station set to Microsoft® OCS client.	
2	Direct call from Microsoft® OCS client to TDM station set.	

## 10. Troubleshooting

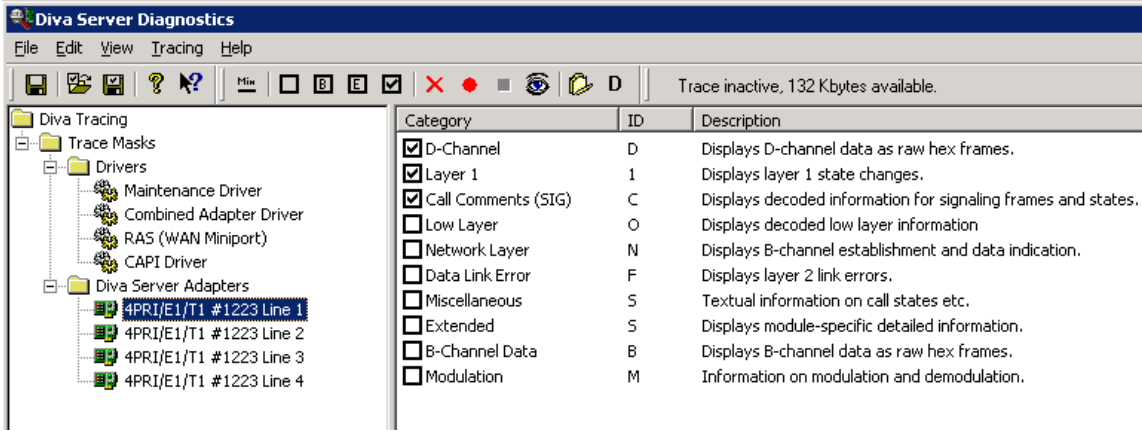
### 10.1 Important Debugging Tools




- **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
```

...

## &lt;Below is a RX SIP message&gt;

```

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- >aptime: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
```

