

# Microsoft Exchange Server 2007 Unified Messaging

## PBX Configuration Note:

**Avaya S8500**

**with Dialogic<sup>®</sup> 2000 Media Gateway Series**

**(DMG2xxxDTI) using T1 QSIG**

By : Dialogic

Updated Since : 12/19/2007

### **READ THIS BEFORE YOU PROCEED**

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## Content

This document describes the configuration required to setup Avaya S8500 and Dialogic® 2000 Media Gateway Series (DMG2xxxDTI) using T1 QSIG as the telephony signaling protocol. It also contains the results of the interoperability testing of Microsoft Exchange 2007 Unified Messaging based on this setup.

## Intended Audience

This document is intended for Systems Integrators with significant telephony knowledge.

## Technical Support

The information contained within this document has been provided by Microsoft partners or equipment manufacturers and is provided AS IS. This document contains information about how to modify the configuration of your PBX or VoIP gateway. Improper configuration may result in the loss of service of the PBX or gateway. Microsoft is unable to provide support or assistance with the configuration or troubleshooting of components described within. Microsoft recommends readers to engage the service of an Microsoft Exchange 2007 Unified Messaging Specialist or the manufacturers of the equipment(s) described within to assist with the planning and deployment of Exchange Unified Messaging.

## Microsoft Exchange 2007 Unified Messaging (UM) Specialists

These are Systems Integrators who have attended technical training on Exchange 2007 Unified Messaging conducted by Microsoft Exchange Engineering Team. For contact information, visit [here](#).

## Version Information

Date of Modification	Details of Modification
December 19, 2007	Initial version of this document.

## 1. Components Information

### 1.1. PBX or IP-PBX

<b>PBX Vendor</b>	Avaya
<b>Model</b>	S8500
<b>Software Version</b>	Comm manager 3.0 – R013x.00.1.346.0
<b>Telephony Signaling</b>	T1 QSIG
<b>Additional Notes</b>	N/A

### 1.2. VoIP Gateway

<b>Gateway Vendor</b>	Dialogic Corporation
<b>Model</b>	Dialogic® 2000 Media Gateway Series (DMG2xxxDTI)
<b>Software Version</b>	5.0.42
<b>VoIP Protocol</b>	SIP

### 1.3. Microsoft Exchange Server 2007 Unified Messaging

<b>Version</b>	RTM
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## 2. Prerequisites

### 2.1. Gateway Requirements

The gateway needs to support T1 QSIG interface.

### 2.2. PBX Requirements

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

To support the T1 QSIG configuration as documented, you need the TN464F DS1 interface card.

### 2.3. Cabling Requirements

Cabling for QSIG connections must be CAT5e or better. Standard voice quality cable will not provide optimum signal quality and the gateway will have problems establishing connection on the D-Channel.

### 3. Summary and Limitations

- A check in this box indicates the UM feature set is fully functional when using the PBX/gateway in question.

### 4. Gateway Setup Notes

During the initial setup of the gateway using the serial port, you must:

- Assign the gateway a Unique IP address, subnet mask and network gateway address (if the latter is required).
- Configure the gateway to use the SIP VoIP protocol.
- Set the Line Mode to T1.
- Set the Protocol to ISDN - QSIG.

During the solution-specific setup of the gateway using the web interface, you must:

- Configure the gateway with at least a single IP endpoint pointing to your voice server.
- Configure your coder as either aLaw or uLaw as required.
- Set the Voice coder to be either G.711 (default) or G.273 if required.
- Set the Line Encoding and Line Framing as required by your T1 Interface. Typical settings are Encoding = B8ZS and Framing = ESF.
- Configure the SIP Transport for TCP.

## 5. PBX Setup Notes

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

- Setting up system parameters and customer options.
- Administrating T1 interface.
- Programming AAR for trunk access.
- Setting up method of call direction and coverage.
- Setting up subscriber station sets.

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:

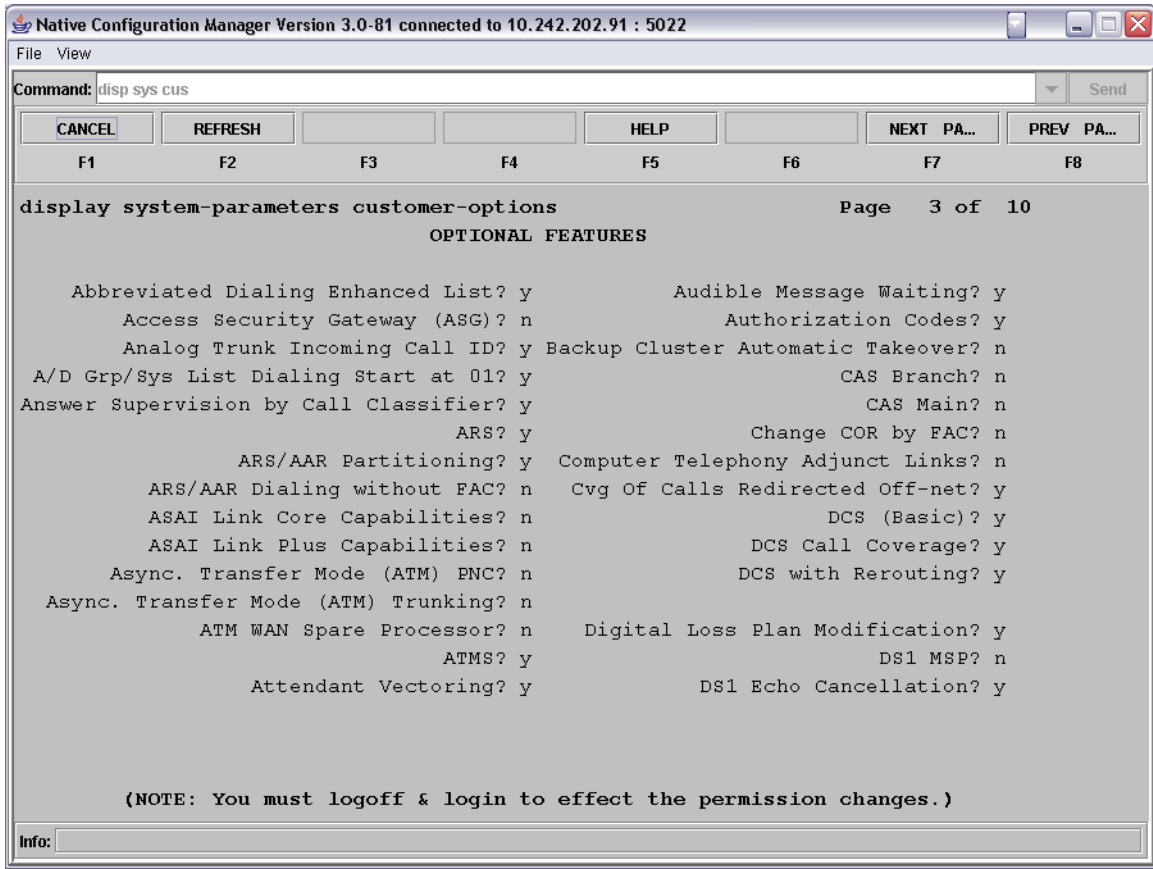
- `change`; `delete`; and `add`

### 5.1. Setting Up System Parameters and Customer Options

Use the `display system-parameters customer options` command to ensure that several QSIG and trunking specific parameters that are needed for the integration to operate properly are enabled.

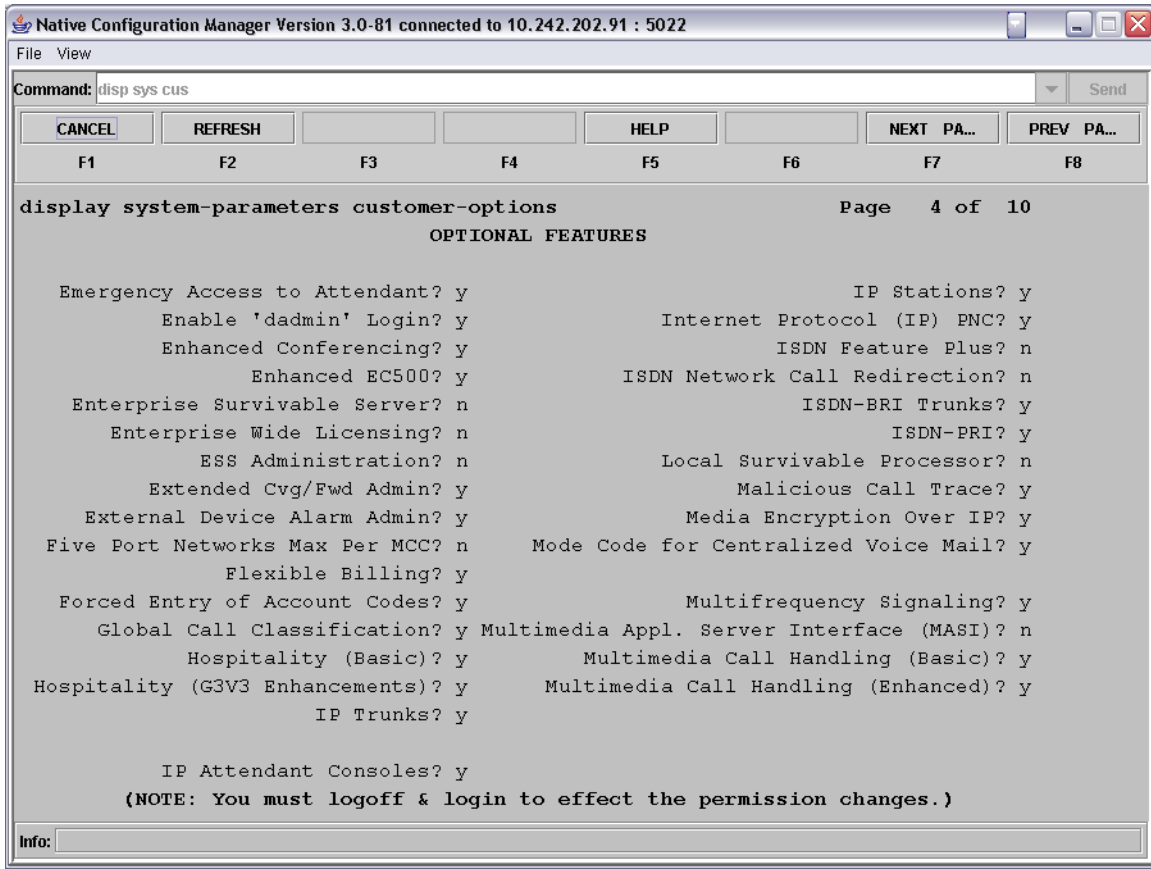
Note that any configuration parameters in this area which are not already enabled may require special maintenance or configuration by Avaya customer support and/or they may require additional software packages be installed to configure properly.

Once the command has been entered press the `NEXT PAGE` soft key until the `Optional Features` section is displayed as seen below.



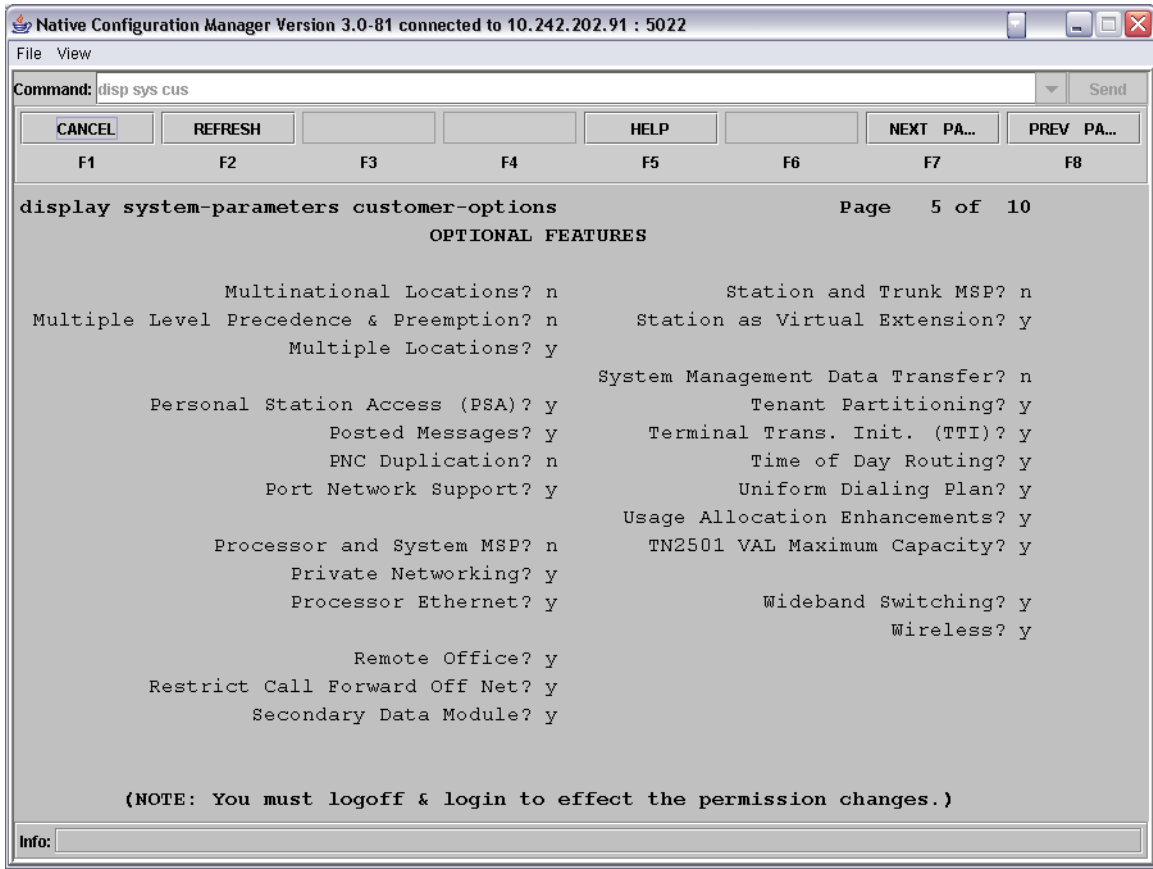
Review the DCS (Basic), DCS Call Coverage, and DCS with Rerouting options. They must be set to y for this integration to work properly. If they are not enabled, contact your Avaya support representative to have these features enabled.

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



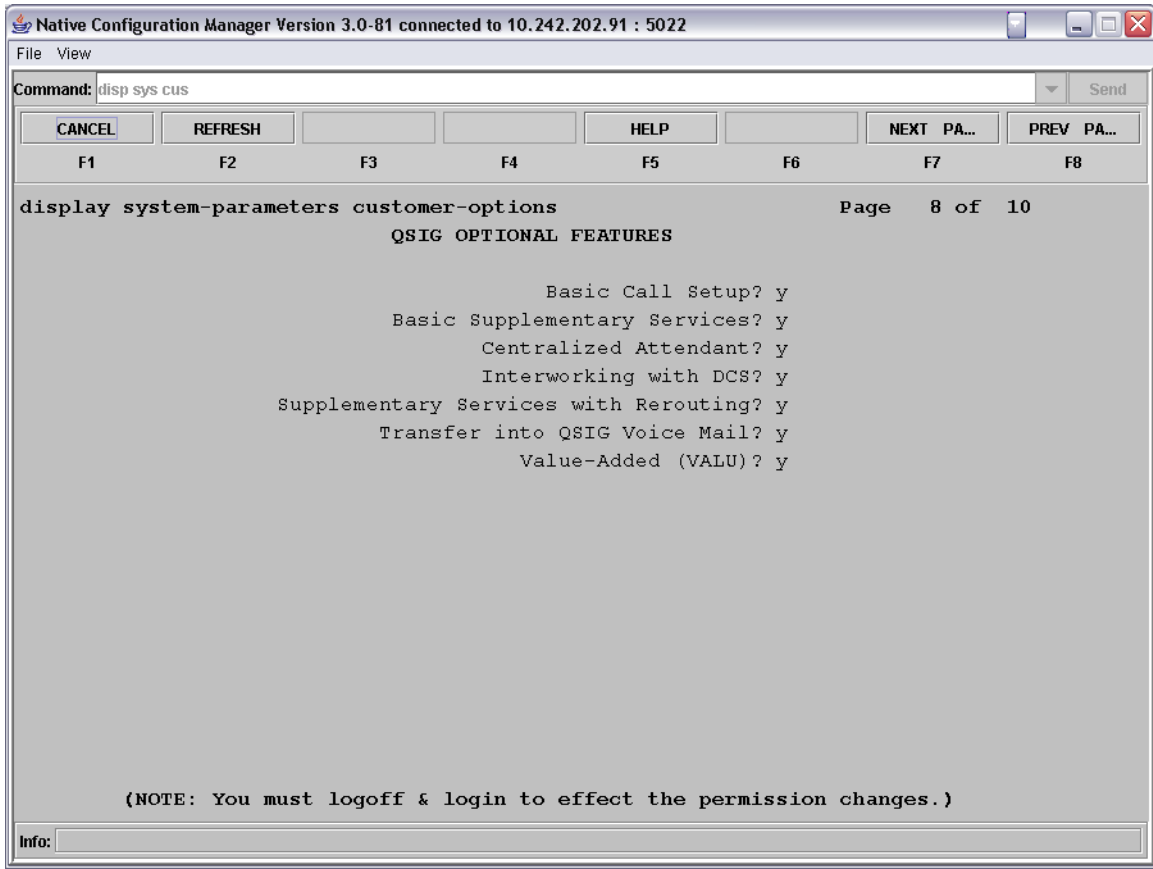
Review the ISDN-PRI option. It must be set to y for this integration to work properly. If this option is not enabled, contact your Avaya support representative to have these features enabled.

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, contact your Avaya support representative to have these features enabled.

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

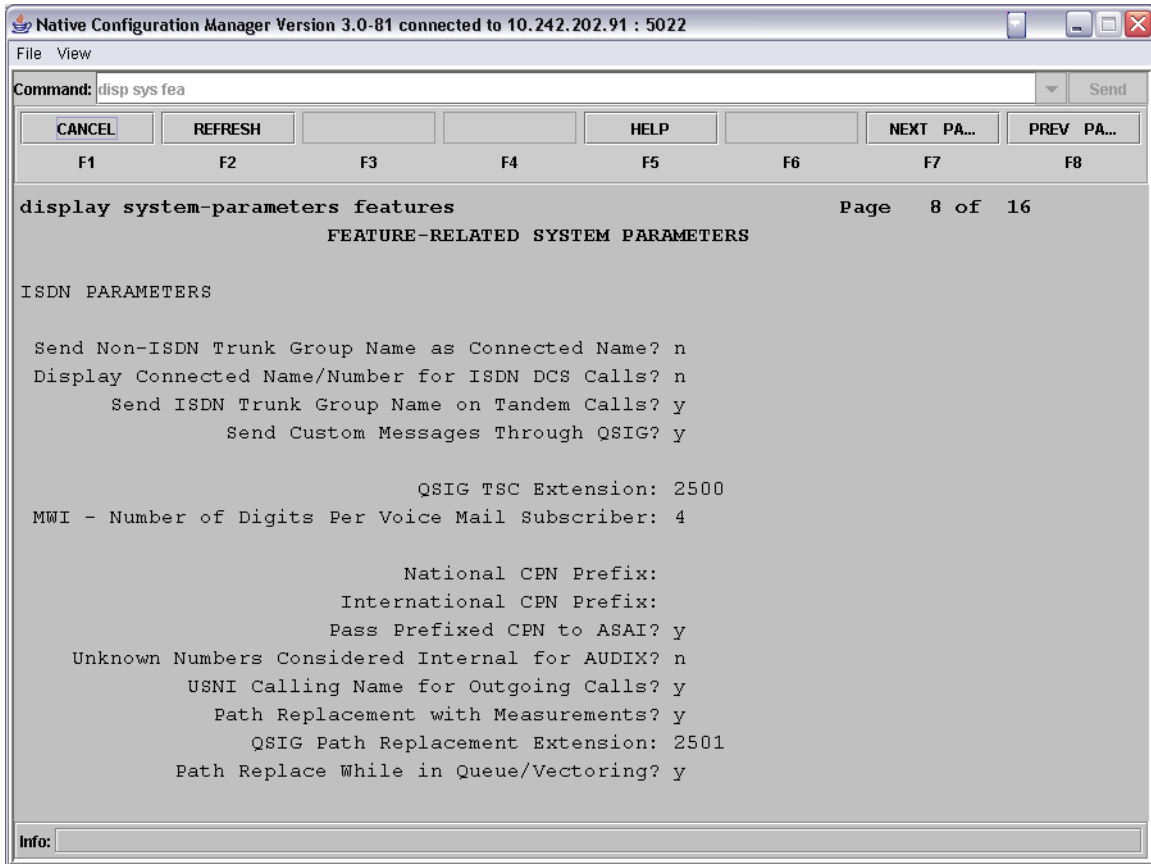


Review the Basic Call Setup, Basic Supplementary Services, Centralized Attendant, Interworking with DCS, Supplementary Services with Rerouting, Transfer into QSIG Voice Mail and Value Added (VALU) options. They must be set to y for this integration to work properly. If they are not enabled, contact your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

In order to make changes to the following fields, you must have administrative rights to the Avaya switch. If you do not, contact your Avaya support representative.

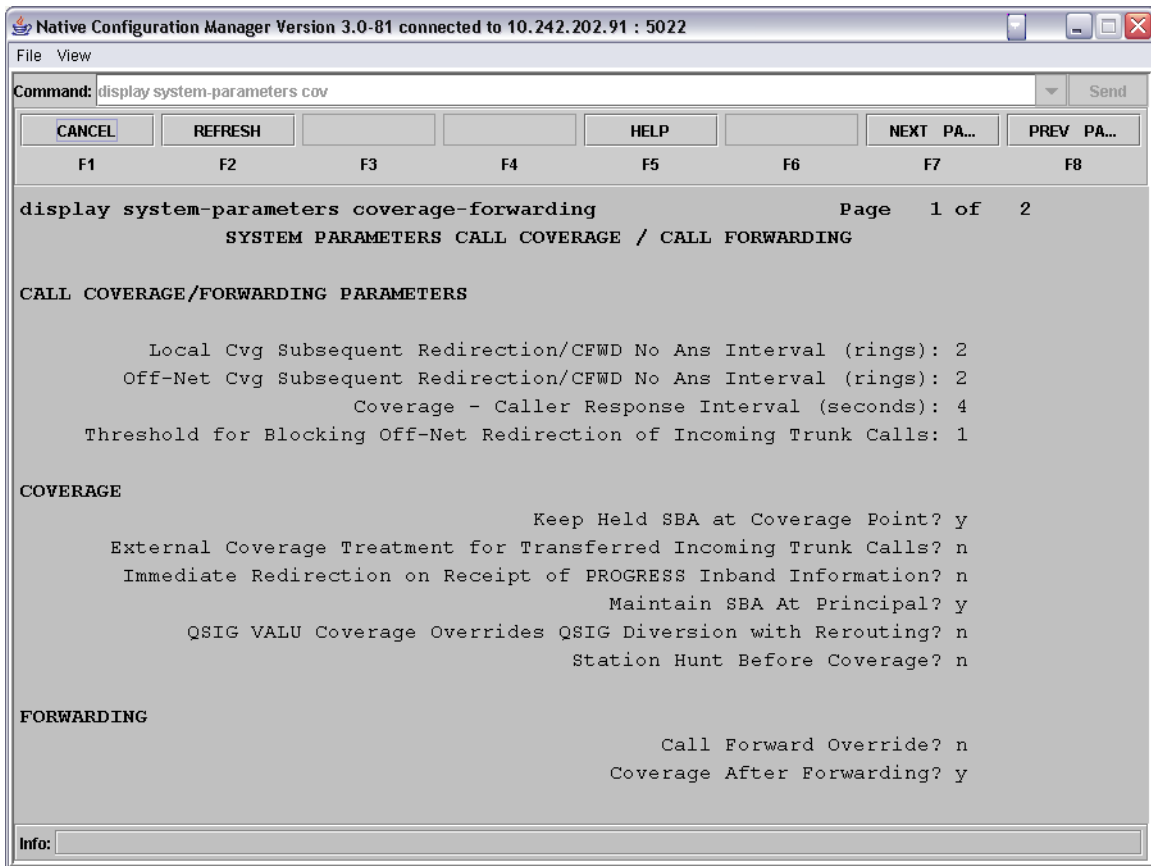
Press the NEXT PAGE soft key until you get to the ISDN Parameters page.



- 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 subscriber's extensions.
- In the QSIG Path Replacement Extension field enter XXXX
  - where XXXX is any unused extension number.

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

Use the change system-parameters coverage-forwarding command to modify the systems features relevant to forwarding of extensions and call coverage to ensure the QSIG supplementary services work properly.



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

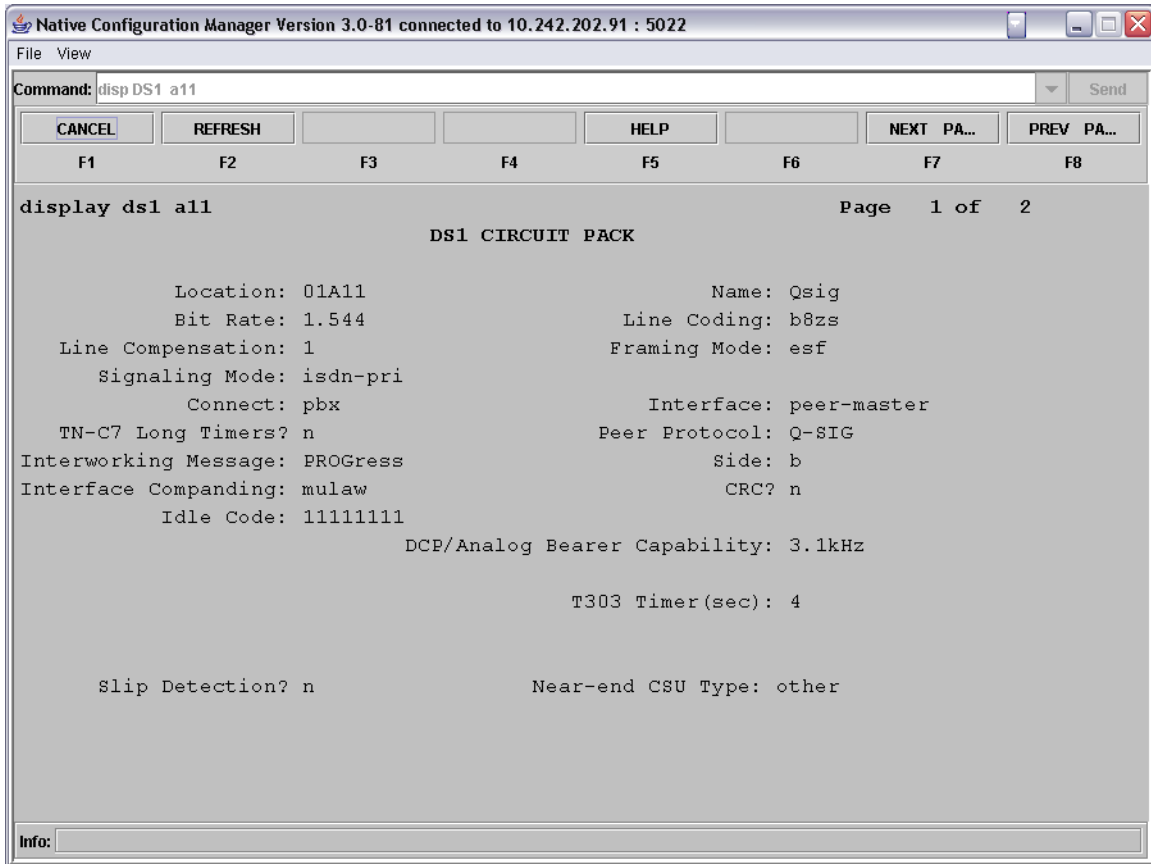
Before moving on to the step, be sure to save your work.

## 5.2. Administrating T1 Interface

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

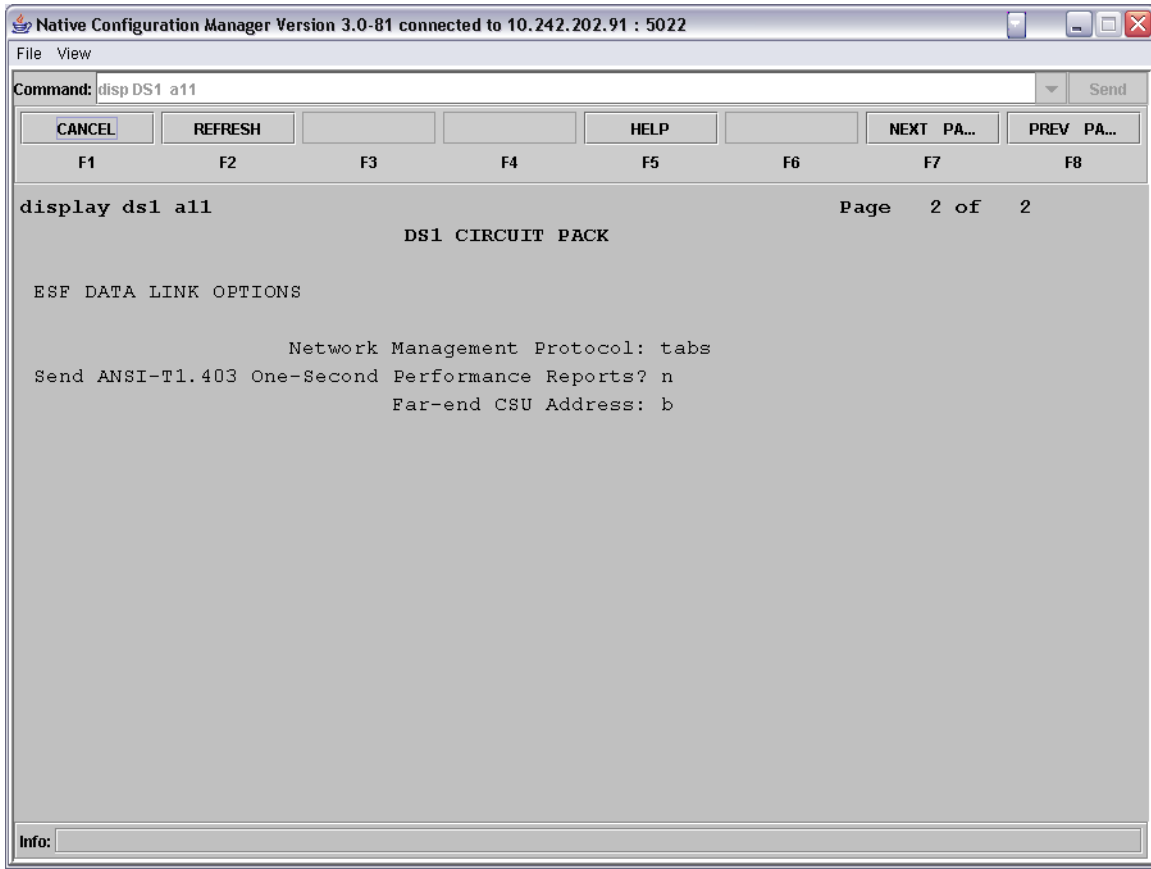
Command = add ds1 a11

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*)

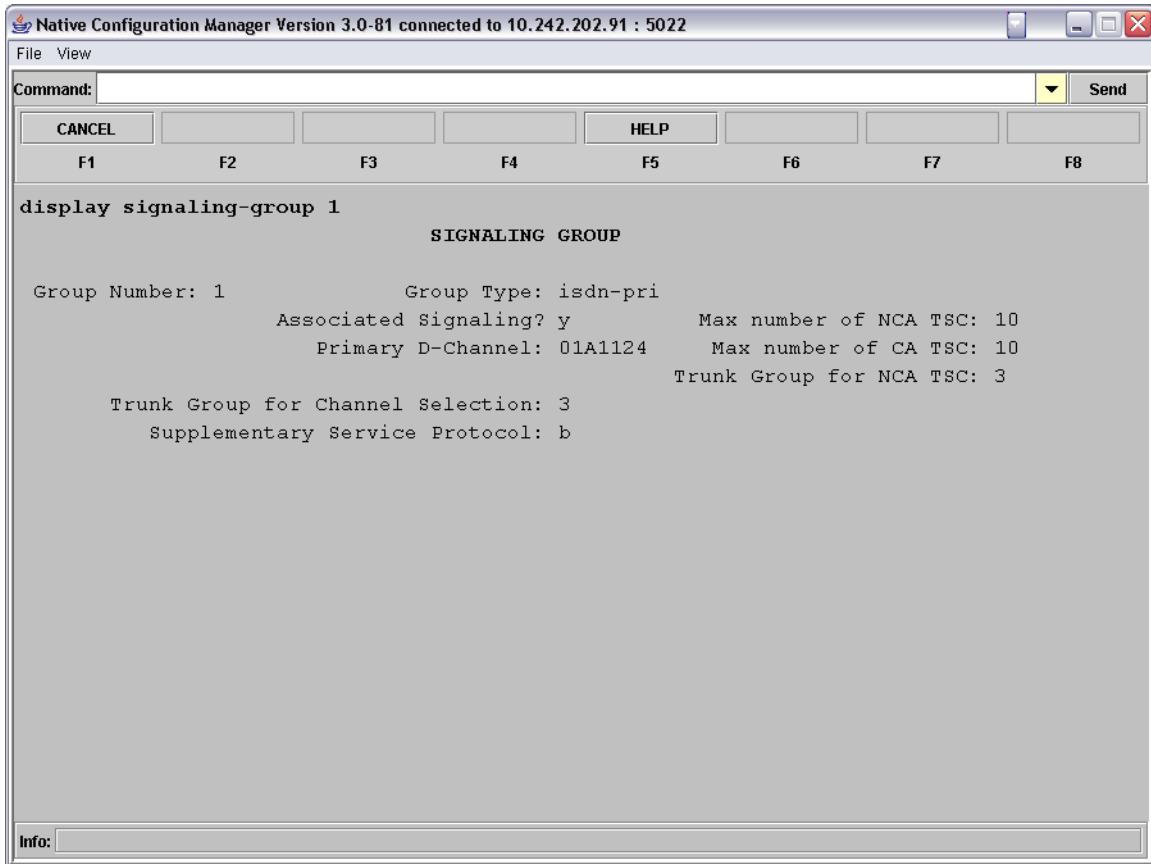
Press the NEXT PAGE soft key to move to the next page of settings.



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

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

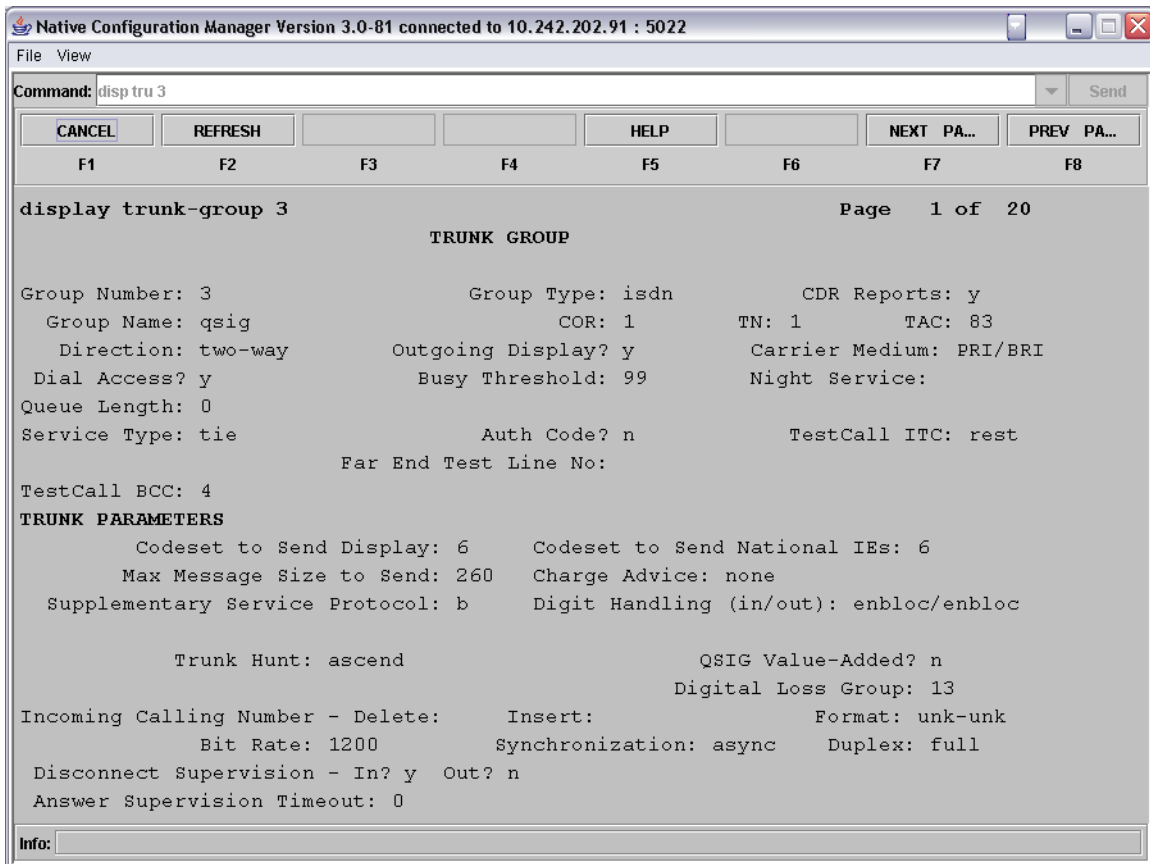
Use the `add signaling-group` command to build a signaling group for the QSIG link to the gateway. This signaling group is what specifies the type of interface being used, the signaling channel (D-Channel) wiring address for the T1 span being used, and the protocol used on the signaling channel.



- In the Group Type field enter isdn-pri.
- In the Associated Signaling field enter y.
- In the Max number of NCA TSC 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. *You will need to return to here and edit this number after you create the trunk group.*
- In the Supplementary Service Protocol field enter b.

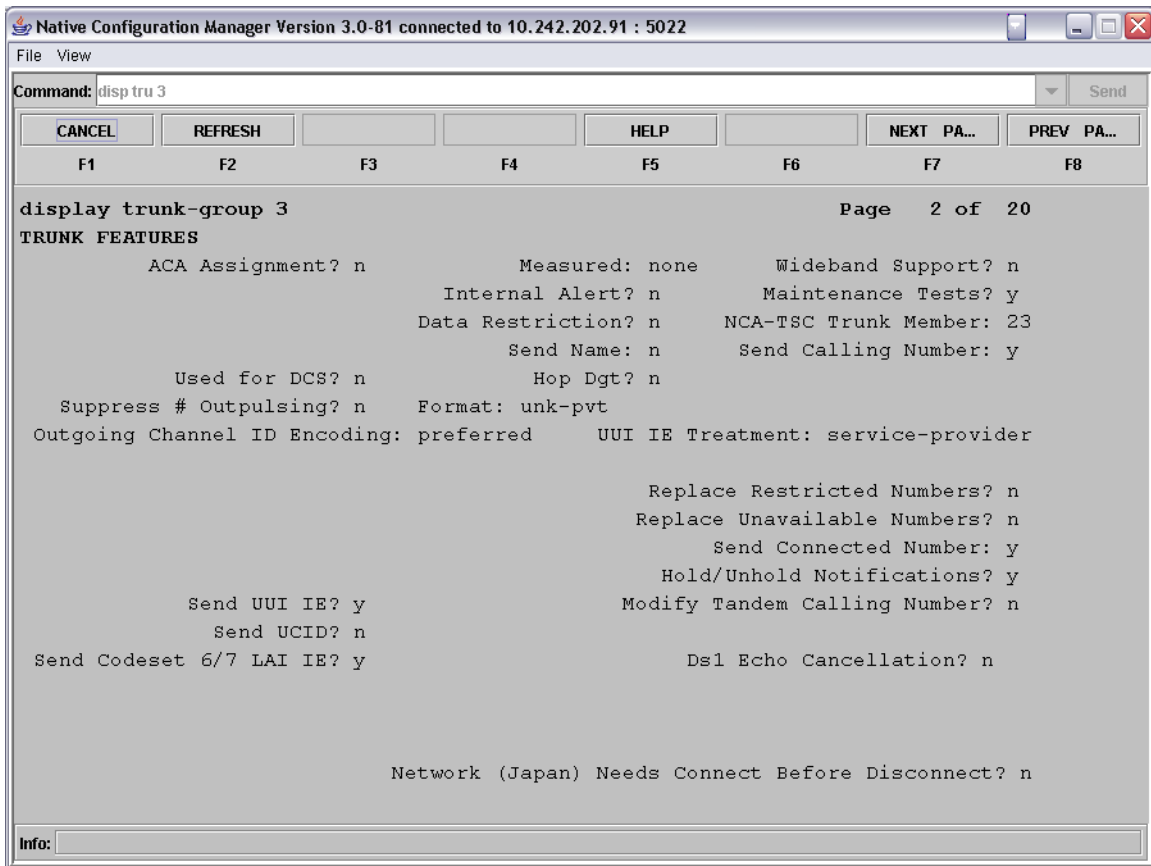
Before moving on to the next command, be sure to save your work.

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.



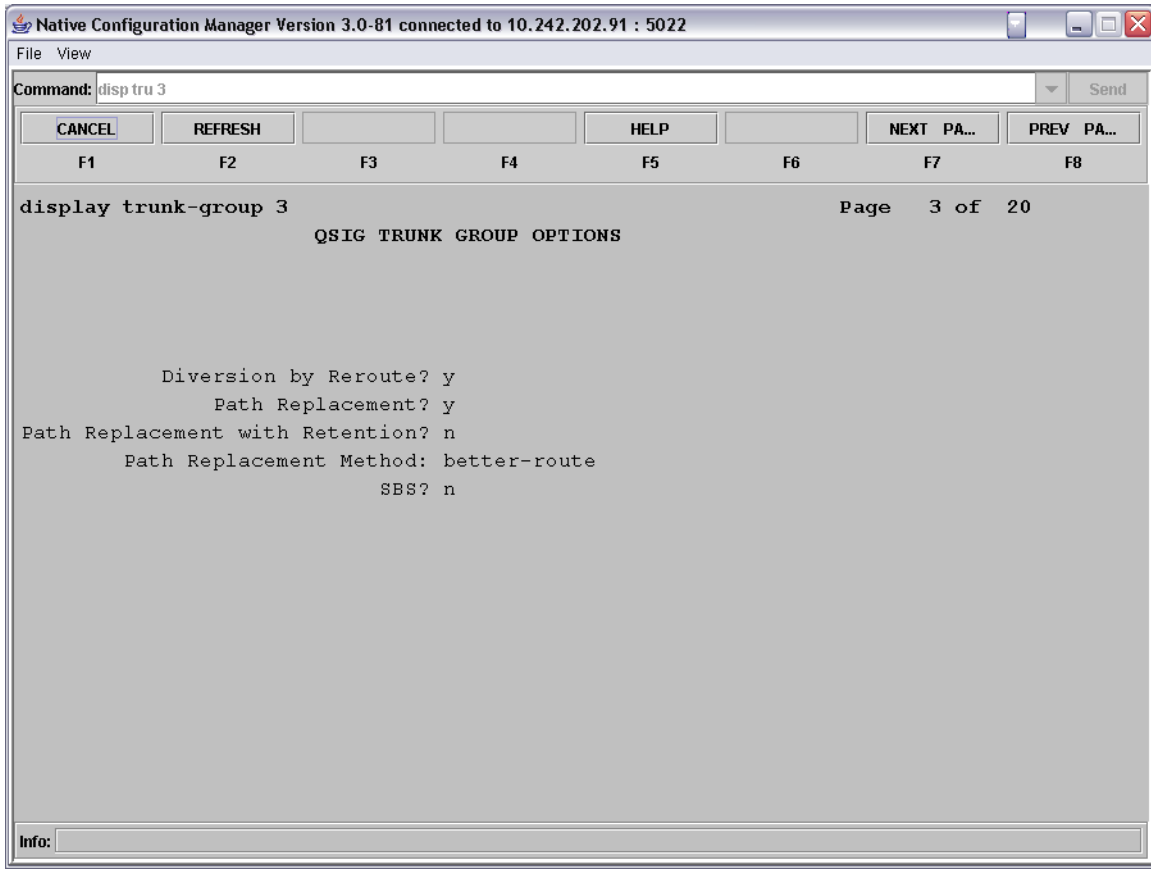
- In the Group Type field enter isdn.
- In the Group Name field enter in a text string to give the group a name.
- In the COR field enter the number of a class of restriction to be used, if there is going to be one. *Understand that setting any restrictions via a COR can effect how the trunk operates so these should be as liberal as possible.*
- In the TN field enter in 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 exclusive.
- In the Send Connected Number field enter y.
- In the Send UUI IE field enter y.

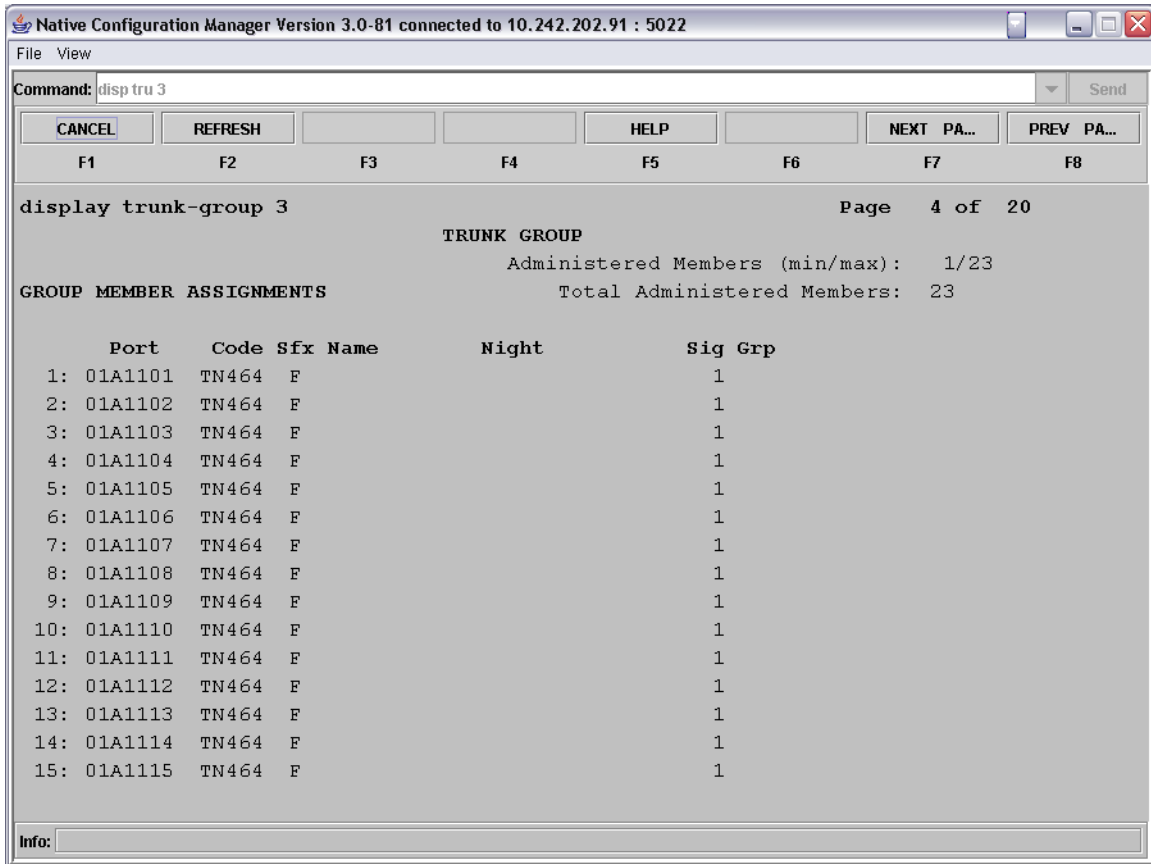
Press NEXT PAGE to advance to the next page of settings.



- In the Diversion by Reroute field enter `y`.
- In the Path Replacement field enter `y`. *This is what enables the path replacement functionality of the SQIG supplemental services package.*
- In the Path Replacement Method field you can enter in 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 we enter in 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 in 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.

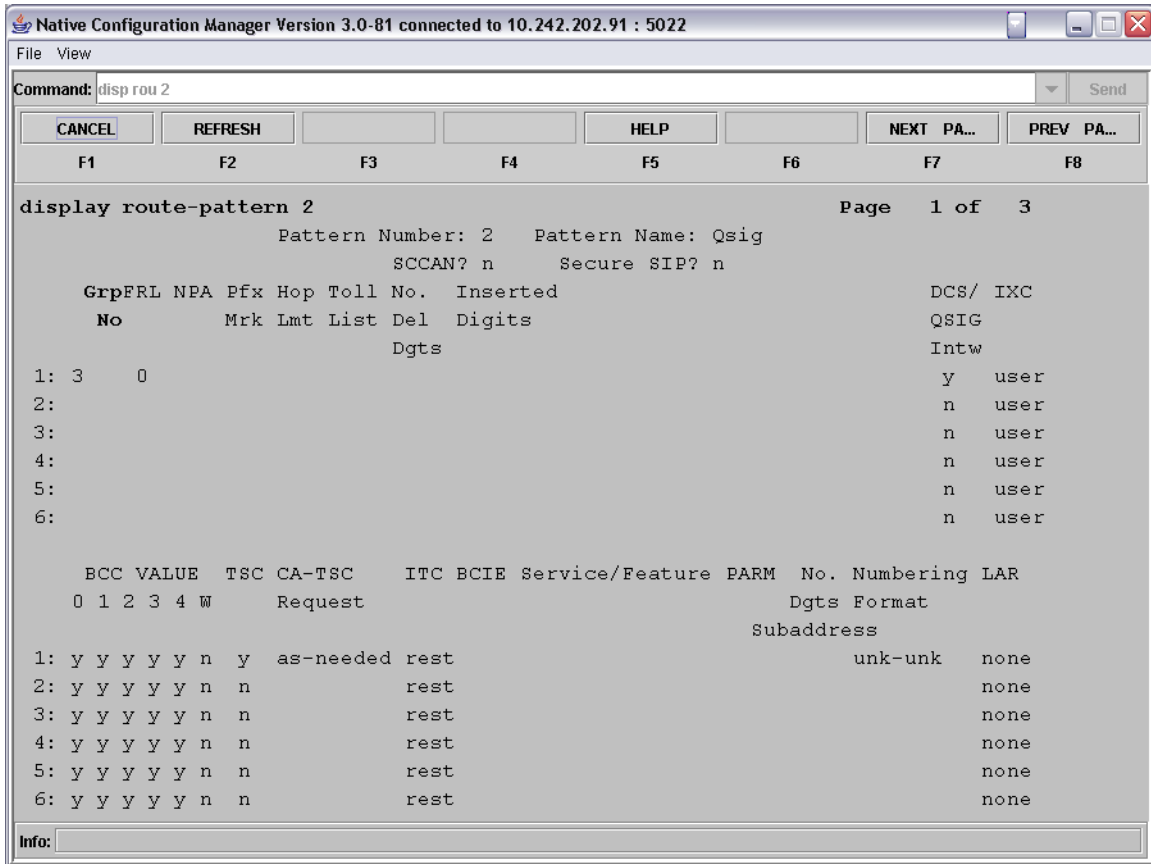
Before moving on to the next command, be sure to save your work.

### 5.3. 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). When 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 set up an extension that can be used to access the trunks via AAR.





- 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 TSC Request field enter as-needed.
- In the Dgts 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 we just defined.



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

File View

Command: bus tru 5 Send

CANCEL NEXT PA...

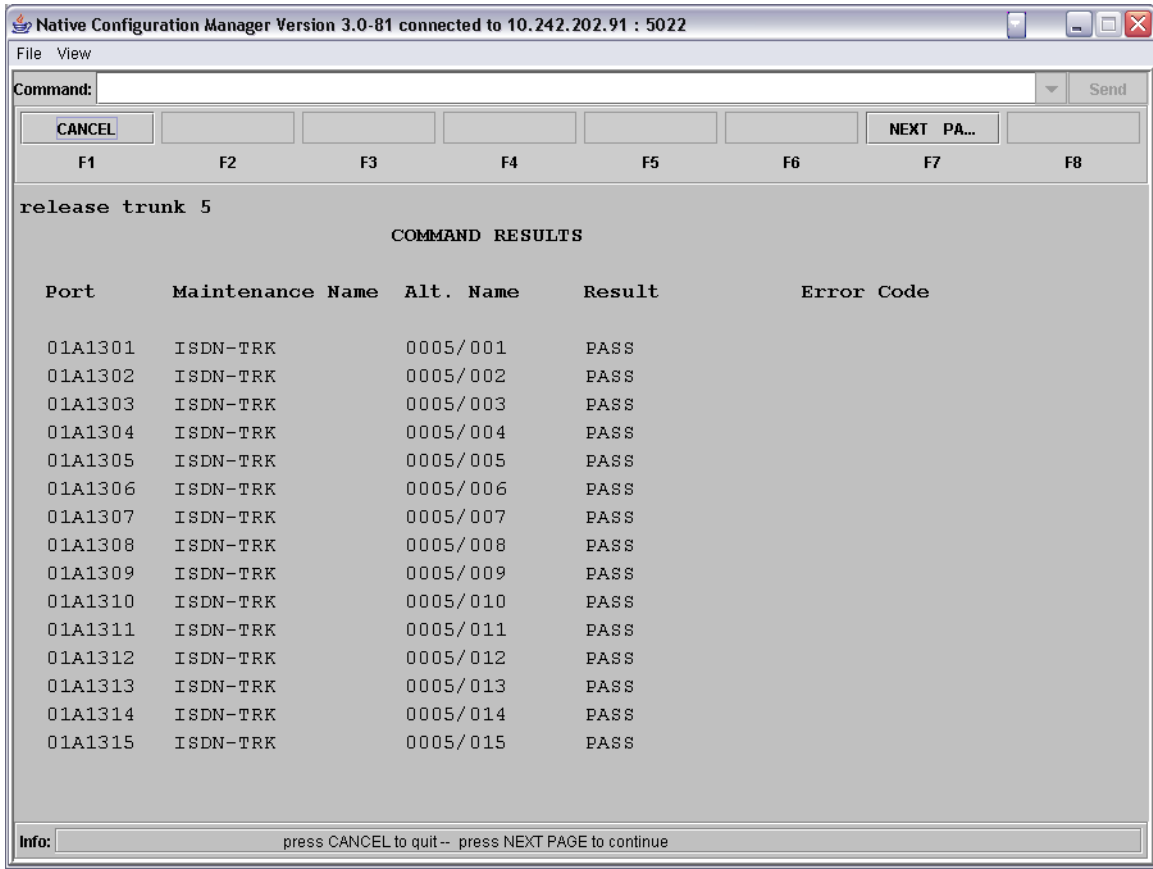
F1 F2 F3 F4 F5 F6 F7 F8

busyout trunk 5 Page 1

COMMAND RESULTS

Port	Maintenance Name	Alt. Name	Result	Error Code
01A1301	ISDN-TRK	0005/001	PASS	
01A1302	ISDN-TRK	0005/002	PASS	
01A1303	ISDN-TRK	0005/003	PASS	
01A1304	ISDN-TRK	0005/004	PASS	
01A1305	ISDN-TRK	0005/005	PASS	
01A1306	ISDN-TRK	0005/006	PASS	
01A1307	ISDN-TRK	0005/007	PASS	
01A1308	ISDN-TRK	0005/008	PASS	
01A1309	ISDN-TRK	0005/009	PASS	
01A1310	ISDN-TRK	0005/010	PASS	
01A1311	ISDN-TRK	0005/011	PASS	
01A1312	ISDN-TRK	0005/012	PASS	
01A1313	ISDN-TRK	0005/013	PASS	
01A1314	ISDN-TRK	0005/014	PASS	
01A1315	ISDN-TRK	0005/015	PASS	

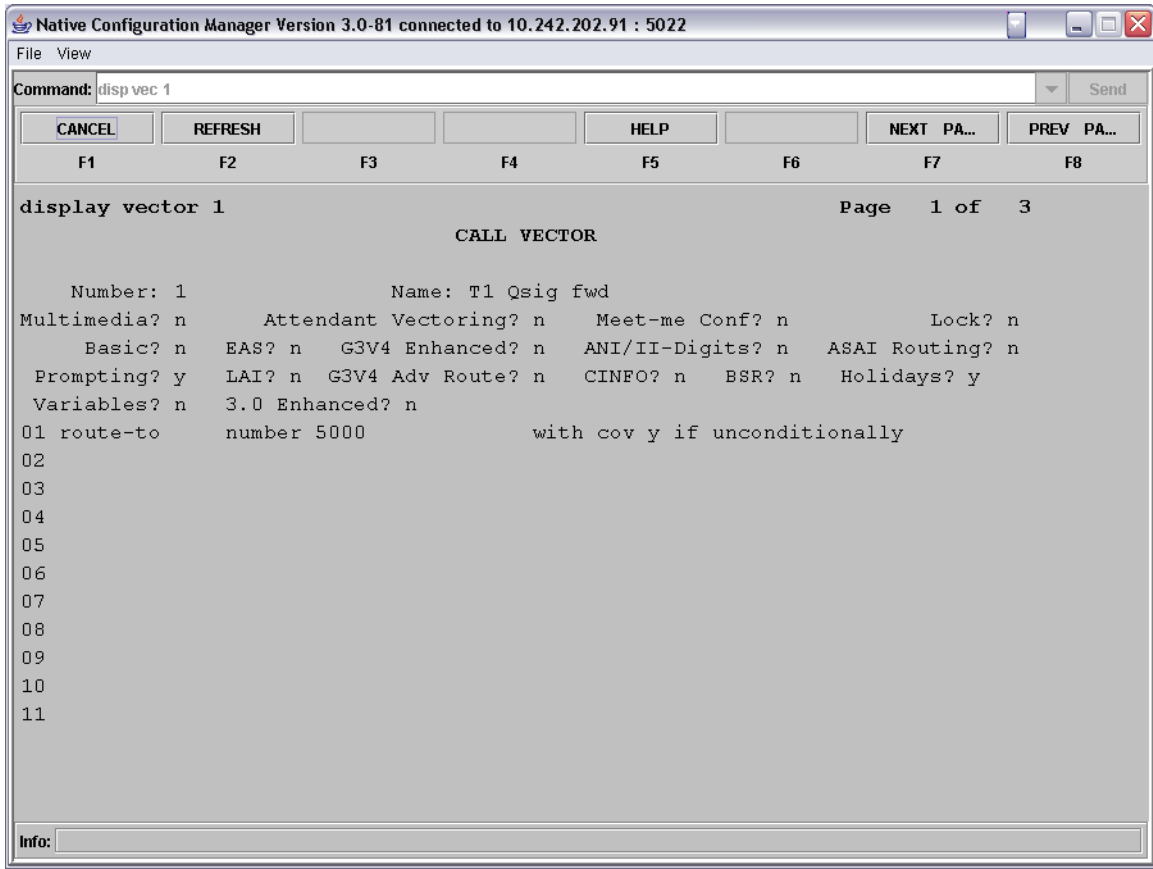
Info: press CANCEL to quit-- press NEXT PAGE to continue



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

#### 5.4. Setting Up Method of Call Direction and Coverage

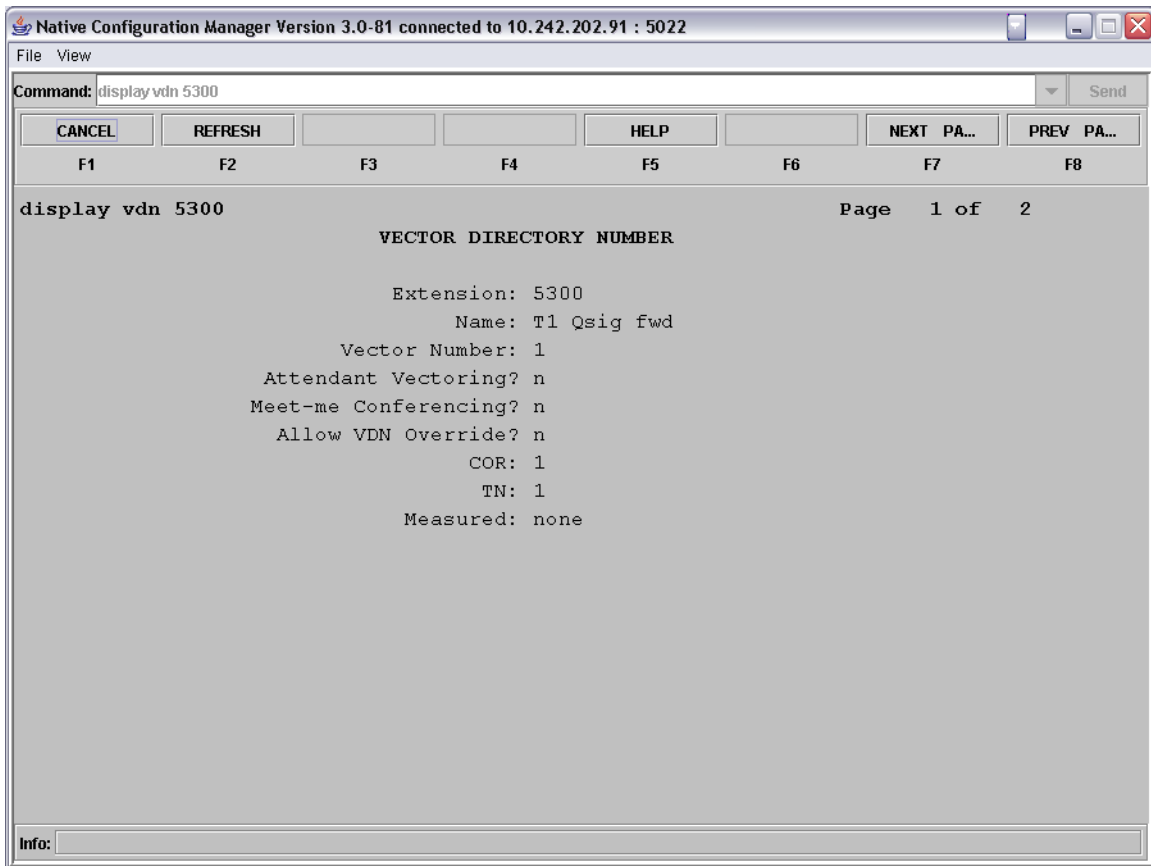
Use the add vector or change vector commands to either add a new or change an existing vector script. This script will be run by a Vector Directory Number (VDN) to be defined next.



- In the Name field enter in a string to give the Vector a descriptive name.
- On the first script line enter the following code:
  - route-to number XXXX with cov y if unconditionally
    - where XXXX is the dialable number that we setup in UDP and assigned as an AAR code.

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

Use the add vdn or change vdn to either add a new, or change an existing Vector Directory Number that will use the Call Vector we just defined above.

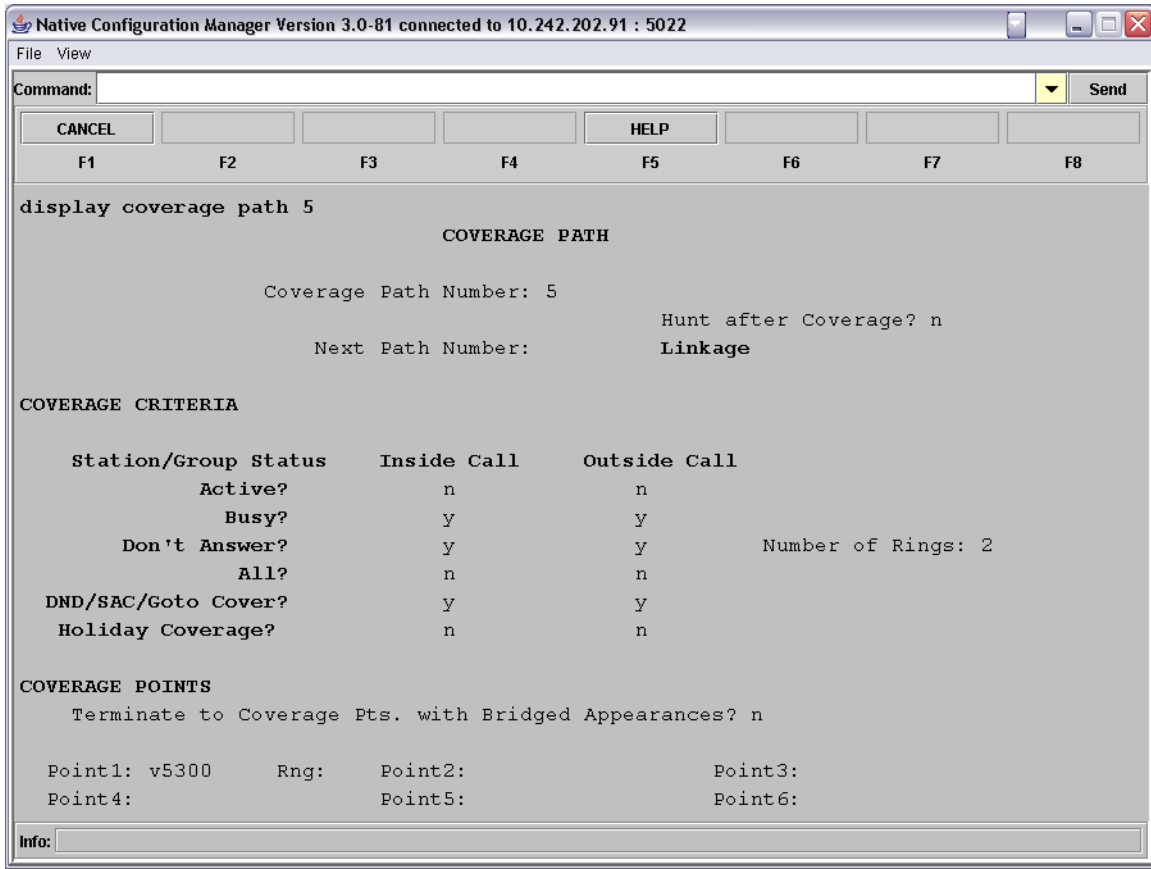


- In the Name field enter in a string to give the VDN a descriptive name.
- In the Vector Number field enter the number of the call Vector that we just defined above.  
*In this example 1.*

We know have a dialable number that when called will route the call over our defined trunks using AAR.

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

Use the `add coverage path` or `change coverage path` commands to either add a new, or change an existing coverage path template. This coverage path will be assigned to any subscriber station that are required to forward to the server for ring no answer and busy conditions.



- In the Forwarding Criteria section, enter y for any/all of the desired forwarding scenarios.
- In the Point1 field enter vXXXX.
  - Where vXXXX is the VDN defined above. In this example 5300.

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

## 5.5. Setting Up Subscriber Station Sets

This is an example of how to set up a subscriber's station set to forward correctly to the server via the coverage path that was administered. The only modification that is needed is to ensure that the Coverage Path 1 configuration section is set to use the coverage path defined above.

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

File View

Command: cha sta 2011 Send

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

F1 F2 F3 F4 F5 F6 F7 F8

change station 2011 Page 1 of 5

STATION

Extension: 2011 Lock Messages? n BCC: 0  
 Type: 8434D Security Code: TN: 1  
 Port: 01A0512 Coverage Path 1: 5 COR: 1  
 Name: 2011 Coverage Path 2: COS: 1  
 Hunt-to Station:

STATION OPTIONS

Loss Group: 2 Personalized Ringing Pattern: 1  
 Data Module? n Message Lamp Ext: 2011  
 Speakerphone: 2-way Mute Button Enabled? y  
 Display Language: english Expansion Module? n  
 Media Complex Ext:  
 IP SoftPhone? n  
 Remote Office Phone? n

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

## 5.6. Additional Comments

N/A

## 6. Exchange 2007 UM Validation Test Matrix

The following table contains a set of tests for assessing the functionality of the UM core feature set. The results are recorded as either:

- Pass (**P**)
- Conditional Pass (**CP**)
- Fail (**F**)
- Not Tested (**NT**)
- Not Applicable (**NA**)

Refer to:

- Appendix for a more detailed description of how to perform each call scenario.
- Section 6.1 for detailed descriptions of call scenario failures, if any.

No.	Call Scenarios (see appendix for more detailed instructions)	(P/CP/F/NT)	Reason for Failure (see 6.1 for more detailed descriptions)
1	Dial the pilot number from a phone extension that is NOT enabled for Unified Messaging and logon to a user's mailbox.  Confirm hearing the prompt: "Welcome, you are connected to Microsoft Exchange. To access your mailbox, enter your extension..."	P	
2	Navigate mailbox using the Voice User Interface (VUI).	P	
3	Navigate mailbox using the Telephony User Interface (TUI).	P	
4	Dial user extension and leave a voicemail.		
4a	Dial user extension and leave a voicemail from an internal extension.  Confirm the Active Directory name of the calling party is displayed in the sender field of the voicemail message.	P	
4b	Dial user extension and leave a voicemail from an external phone.  Confirm the correct phone number of the calling party is displayed in the sender field of the voicemail message.		No External trunk was available for testing.

5	Dial Auto Attendant (AA).  Dial the extension for the AA and confirm the AA answers the call.	P	
6	Call Transfer by Directory Search.		
6a	Call Transfer by Directory Search and have the called party answer.  Confirm the correct called party answers the phone.	P	Transfer was done as a path replacement transfer.
6b	Call Transfer by Directory Search when the called party's phone is busy.  Confirm the call is routed to the called party's voicemail.	P	
6c	Call Transfer by Directory Search when the called party does not answer.  Confirm the call is routed to the called party's voicemail.	P	Transfer was done as a path replacement transfer.
6d	Setup an invalid extension number for a particular user. Call Transfer by Directory Search to this user.  Confirm the number is reported as invalid.	P	
7	Outlook Web Access (OWA) Play-On-Phone Feature.		
7a	Listen to voicemail using OWA's Play-On-Phone feature to a user's extension.	P	
7b	Listen to voicemail using OWA's Play-On-Phone feature to an external number.		No External trunk was available for testing.
8	Configure a button on the phone of a UM-enabled user to forward the user to the pilot number. Press the voicemail button.  Confirm you are sent to the prompt: "Welcome, you are connected to Microsoft Exchange. <User>. Please enter your pin and press the pound key."	P	No speed dial button was available. Testing was done by making a direct call to the hunt group.
9	Send a test FAX message to user		No External trunks or analog lines were available for testing so we could not test fax.

	<p>extension.</p> <p>Confirm the FAX is received in the user's inbox.</p>		
10	<p>Setup TLS between gateway/IP-PBX and Exchange UM.</p> <p>Replace this italicized text with your TLS configuration: self-signed certificates or Windows Certificate Authority (CA).</p>		
10a	<p>Dial the pilot number and logon to a user's mailbox.</p> <p>Confirm UM answers the call and confirm UM responds to DTMF input.</p>		5.0.42 gateway firmware does not yet implement TLS so this feature was not tested.
10b	<p>Dial a user extension and leave a voicemail.</p> <p>Confirm the user receives the voicemail.</p>		5.0.42 gateway firmware does not yet implement TLS so this feature was not tested.
10c	<p>Send a test FAX message to user extension.</p> <p>Confirm the FAX is received in the user's inbox.</p>		5.0.42 gateway firmware does not yet implement TLS so this feature was not tested.
11	<p>Setup G.723.1 on the gateway. (If already using G.723.1, setup G.711 A Law or G.711 Mu Law for this step).</p> <p>Dial the pilot number and confirm the UM system answers the call.</p>	P	
12	<p>Setup Message Waiting Indicator (MWI).</p> <p>Geomant offers a third party solution: MWI 2007. Installation files and product documentation can be found on Geomant's <a href="#">MWI 2007 website</a>.</p>		The Geomant software was not available at the time of validation so this feature was not tested.
13	Execute Test-UMConnectivity.	NT	
14	Setup and test fail-over configuration on the IP-PBX to work with two UM servers.	NA	

## 6.1. Detailed Description of Limitations

<b>Failure Point</b>	
<b>Phone type (if phone-specific)</b>	
<b>Call scenarios(s) associated with failure point</b>	
<b>List of UM features affected by failure point</b>	
<b>Additional Comments</b>	

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<b>List of UM features affected by failure point</b>	
<b>Additional Comments</b>	

## 7. Troubleshooting

### 7.1. Important Debugging Tools

- Ethereal/Wireshark – Used to view and analyze the network captures provided by the Dialogic® gateway diagnostic firmware.
- Adobe Audition -- Used to review and analyze the audio extracted from the network captures to troubleshoot any audio-related issues.

### 7.2. Important Gateway Trace Masks

These keys are helpful during troubleshooting scenarios and should be considered keys to activate by default for all troubleshooting cases.

- `voip prot` and `voip code` – this allows the collection of all SIP-related messages as they are sent from and received by the gateway. This data is important in cases where you feel that the gateway is not able to communicate properly with the messaging server.
- `tel event` and `tel code` – This allows the collection of circuit-side activity of the emulated station set, such as display updates, key presses, light transitions and hook state changes. This data is important in the following scenarios:
  - Call control problems (dropped calls, failing transfers, etc...)
  - Integration problems (incorrect mailbox placement, missed auto-attendant greetings etc...)
- `teldrv prot` – This allows the collection of all ISDN messages both transmitted and received on the gateways front-end interface. This data is important in the following scenarios:
  - Call control problems (dropped calls, failing transfers, etc...)
  - Integration problems (incorrect mailbox placement, missed auto-attendant greetings etc...)

These keys are helpful during specific issues and can be enabled for targeted troubleshooting of very specific cases. Activation of these keys may generate large amounts of data on busy systems and increase the size of the collected log files, but should not harm system performance.

- `dspif` (all keys) – This allows the collection of tone-related data. This data is helpful in cases where you think you have problems detection specific tones that should be, should not be, or are expected to be present at specific times during the call. If you do not suspect a tone-related issue, this key may be left disabled.

*NOTE: Turning on all traces is not recommended. Doing this floods the debug stream with significant amounts of information that can cause delays in determining the root cause of a problem.*

## Appendix

### 1. Dial Pilot Number and Mailbox Login

- Dial the pilot number of the UM server from an extension that is NOT enabled for UM.
- Confirm hearing the greeting prompt: "Welcome, you are connected to Microsoft Exchange. To access your mailbox, enter your extension..."
- Enter the extension, followed by the mailbox PIN of an UM-enabled user.
- Confirm successful logon to the user's mailbox.

### 2. Navigate Mailbox using Voice User Interface (VUI)

- Logon to a user's UM mailbox.
- If the user preference has been set to DTMF tones, activate the Voice User Interface (VUI) under personal options.
- Navigate through the mailbox and try out various voice commands to confirm that the VUI is working properly.
- This test confirms that the RTP is flowing in both directions and speech recognition is working properly.

### 3. Navigate Mailbox using Telephony User Interface (TUI)

- Logon to a user's UM mailbox.
- If the user preference has been set to voice, press "#0" to activate the Telephony User Interface (TUI).
- Navigate through the mailbox and try out the various key commands to confirm that the TUI is working properly.
- This test confirms that both the voice RTP and DTMF RTP (RFC 2833) are flowing in both directions.

### 4. Dial User Extension and Leave Voicemail

- Note: If you are having difficulty reaching the user's UM voicemail, verify that the coverage path for the UM-enabled user's phone is set to the pilot number of the UM server.

#### a. From an Internal Extension

- From an internal extension, dial the extension for a UM-enabled user and leave a voicemail message.
- Confirm the voicemail message arrives in the called user's inbox.
- Confirm this message displays a valid Active Directory name as the sender of this voicemail.

## **b. From an External Phone**

- From an external phone, dial the extension for a UM-enabled user and leave a voicemail message.
- Confirm the voicemail message arrives in the called user's inbox.
- Confirm this message displays the phone number as the sender of this voicemail.

## **5. Dial Auto Attendant(AA)**

- Create an Auto Attendant using the Exchange Management Console:
  - Under the Exchange Management Console, expand "Organizational Configuration" and then click on "Unified Messaging".
  - Go to the Auto Attendant tab under the results pane.
  - Click on the "New Auto Attendant..." under the action pane to invoke the AA wizard.
  - Associate the AA with the appropriate dial plan and assign an extension for the AA.
  - Create PBX dialing rules to always forward calls for the AA extension to the UM server.
  - Confirm the AA extension is displayed in the diversion information of the SIP Invite.
- Dial the extension of Auto Attendant.
- Confirm the AA answers the call.

## **6. Call Transfer by Directory Search**

- Method one: Pilot Number Access
  - Dial the pilot number for the UM server from a phone that is NOT enabled for UM.
  - To search for a user by name:
    - Press # to be transferred to name Directory Search.
      - Call Transfer by Directory Search by entering the name of a user in the same Dial Plan using the telephone keypad, last name first.
  - To search for a user by email alias:
    - Press "# " to be transferred to name Directory Search
    - Press "# #" to be transferred to email alias Directory Search
    - Call Transfer by Directory Search by entering the email alias of a user in the same Dial Plan using the telephone keypad, last name first.
- Method two: Auto Attendant
  - Follow the instructions in appendix section 5 to setup the AA.
  - Call Transfer by Directory Search by speaking the name of a user in the same Dial Plan. If the AA is not speech enabled, type in the name using the telephone keypad.

- Note: Even though some keys are associated with three or four numbers, for each letter, each key only needs to be pressed once regardless of the letter you want. Ignore spaces and symbols when spelling the name or email alias.

#### **a. Called Party Answers**

- Call Transfer by Directory Search to a user in the same dial plan and have the called party answer.
- Confirm the call is transferred successfully.

#### **b. Called Party is Busy**

- Call Transfer by Directory Search to a user in the same dial plan when the called party is busy.
- Confirm the calling user is routed to the correct voicemail.

#### **c. Called Party does not Answer**

- Call Transfer by Directory Search to a user in the same dial plan and have the called party not answer the call.
- Confirm the calling user is routed to the correct voicemail.

#### **d. The Extension is Invalid**

- Assign an invalid extension to a user in the same dial plan. An invalid extension has the same number of digits as the user's dial plan and has not been mapped on the PBX to any user or device.
  - UM Enable a user by invoking the "Enable-UMMailbox" wizard.
  - Assign an unused extension to the user.
  - Do not map the extension on the PBX to any user or device.
  - Call Transfer by Directory Search to this user.
  - Confirm the call fails and the caller is prompted with appropriate messages.

### **7. Play-On-Phone**

- To access play-on-phone:
  - Logon to Outlook Web Access (OWA) by going to URL <https://<server name>/owa>.
  - After receiving a voicemail in the OWA inbox, open this voicemail message.
  - At the top of this message, look for the Play-On-Phone field ( Play on Phone...).
  - Click this field to access the Play-On-Phone feature.

#### **a. To an Internal Extension**

- Dial the extension for a UM-enabled user and leave a voicemail message.
- Logon to this called user's mailbox in OWA.

- Once it is received in the user's inbox, use OWA's Play-On-Phone to dial an internal extension.
- Confirm the voicemail is delivered to the correct internal extension.

#### **b. To an External Phone number**

- Dial the extension for a UM-enabled user and leave a voicemail message.
- Logon to the UM-enabled user's mailbox in OWA.
- Confirm the voicemail is received in the user's mailbox.
- Use OWA's Play-On-Phone to dial an external phone number.
- Confirm the voicemail is delivered to the correct external phone number.
- Troubleshooting:
  - Make sure the appropriate UMMailboxPolicy dialing rule is configured to make this call. As an example, open an Exchange Management Shell and type in the following commands:
    - `$dp = get-umdialplan -id <dial plan ID>`
    - `$dp.ConfiguredInCountryOrRegionGroups.Clear()`
    - `$dp.ConfiguredInCountryOrRegionGroups.Add("anywhere,*,*,")`
    - `$dp.AllowedInCountryOrRegionGroups.Clear()`
    - `$dp.AllowedInCountryOrRegionGroups.Add("anywhere")`
    - `$dp|set-umdialplan`
    - `$mp = get-ummailboxpolicy -id <mailbox policy ID>`
    - `$mp.AllowedInCountryGroups.Clear()`
    - `$mp.AllowedInCountryGroups.Add("anywhere")`
    - `$mp|set-ummailboxpolicy`
  - The user must be enabled for external dialing on the PBX.
  - Depending on how the PBX is configured, you may need to prepend the trunk access code (e.g. 9) to the external phone number.

### **8. Voicemail Button**

- Configure a button on the phone of a UM-enabled user to route the user to the pilot number of the UM server.
- Press this voicemail button on the phone of an UM-enabled user.
- Confirm you are sent to the prompt: "Welcome, you are connected to Microsoft Exchange. <User Name>. Please enter your pin and press the pound key."
- Note: If you are not hearing this prompt, verify that the button configured on the phone passes the user's extension as the redirect number. This means that the user extension should appear in the diversion information of the SIP invite.

## 9. FAX

- Use the Management Console or the Management Shell to FAX-enable a user.
- Management Console:
  - Double click on a user's mailbox and go to Mailbox Features tab.
  - Click Unified Messaging and then click the properties button.
  - Check the box "Allow faxes to be received".
- Management Shell - execute the following command:
  - Set-UMMailbox -identity UMUser -FaxEnabled:\$true
- To test fax functionality:
  - Dial the extension for this fax-enabled UM user from a fax machine.
  - Confirm the fax message is received in the user's inbox.
  - Note: You may notice that the UM server answers the call as though it is a voice call (i.e. you will hear: "Please leave a message for..."). When the UM server detects the fax CNG tones, it switches into fax receiving mode, and the voice prompts terminate.
  - Note: UM only support T.38 for sending fax.

## 10. TRANSPORT SECURITY LAYER (TLS)

- Setup TLS on the gateway/IP-PBX and Exchange 2007 UM.
- Import/Export all the appropriate certificates.

### a. Dial Pilot Number and Mailbox Login

- Execute the steps in scenario 1 (above) with TLS turned on.

### b. Dial User Extension and Leave a Voicemail

- Execute the steps in scenario 4 (above) with TLS turned on.

### c. FAX

- Execute the steps in scenario 9 (above) with TLS turned on.

## 11.G.723.1

- Configure the gateway to use the G.723.1 codec for sending audio to the UM server.
- If already using G.723.1 for the previous set of tests, use this step to test G.711 A Law or G.711 Mu Law instead.
- Call the pilot number and verify the UM server answers the call.
- Note: If the gateway is configured to use multiple codecs, the UM server, by default, will use the G.723.1 codec if it is available.

## 12. Message Waiting Indicator (MWI)

- Although Exchange 2007 UM does not natively support MWI, Geomant has created a 3rd party solution - MWI2007. This product also supports SMS message notification.
- Installation files and product documentation can be found on Geomant's [MWI 2007 website](#).

## 13. Test-UMConnectivity

- Run the Test-UMConnectivity diagnostic cmdlet by executing the following command in Exchange Management Shell:
- Test-UMConnectivity –UMIPGateway: <Gateway> -Phone: <Phone> |fl
- <Gateway> is the name (or IP address) of the gateway which is connected to UM, and through which you want to check the connectivity to the UM server. Make sure the gateway is configured to route calls to UM.
- <Phone> is a valid UM extension. First, try using the UM pilot number for the hunt-group linked to the gateway. Next, try using a CFNA number configured for the gateway. Please ensure that a user or an AA is present on the UM server with that number.
- The output shows the latency and reports if it was successful or there were any errors.

## 14. Test Fail-Over Configuration on IP-PBX with Two UM Servers

- This is only required for direct SIP integration with IP-PBX. If the IP-PBX supports fail-over configuration (e.g., round-robin calls between two or more UM servers):
  - Provide the configuration steps in Section 5.
  - Configure the IP-PBX to work with two UM servers.
  - Simulate a failure in one UM server.
  - Confirm the IP-PBX transfers new calls to the other UM server successfully.