Dialogic® Media Gateway Installation and Configuration Integration Note

1. Scope

This document is intended to detail a typical installation and configuration of Dialogic® 2000 Media Gateway Series (DMG2000) when used to interface between PBX and Microsoft® Office Communications Server 2007 (OCS) application.

2. Configuration Details

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

2.1 PBX

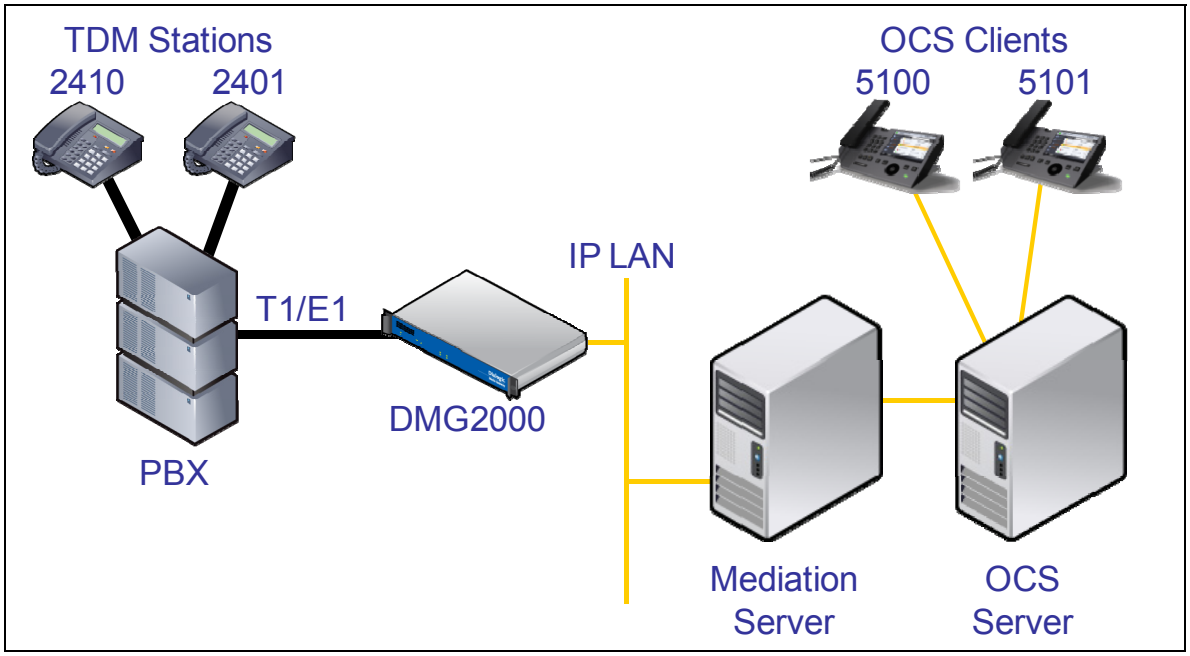
PBX Vendor	Avaya
Model	S8500
Software Version	Comm manager 3.0 – R013x.00.1.346.0
Additional Notes	N/A

2.2 Gateway

Gateway Model	Dialogic® 2000 Media Gateway Series (DMG2000)
Software Version	6.0 (6.0.103)
Protocol	E1 QSIG

2.3 System Diagram

The diagram below details the setup used in the testing and creation of the technical document.



3. Prerequisites

3.1 PBX Prerequisites

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

3.1.1 PBX Equipment Required

To support the E1 QSIG configuration as documented you need the TN464F DS1 interface card.

3.1.2 PBX 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.

3.2 Gateway Prerequisites

The gateway needs to support E1 QSIG interface.

4. Summary of Limitations

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

5. Gateway Setup Notes

Steps for setting up the gateway:

- Parameter Configuration
- Routing Engine Configuration

5.1 Parameter Configuration

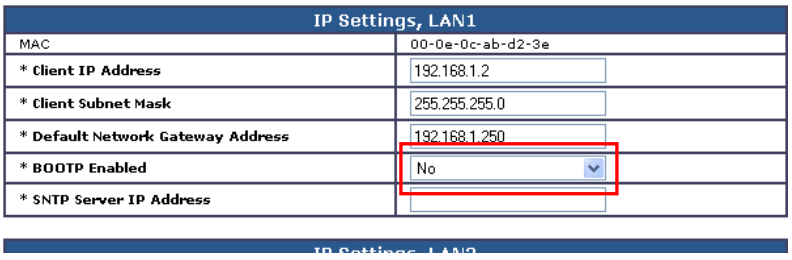
To get the gateway connected between the PBX and mediation server there are only a few configuration options that are required.

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

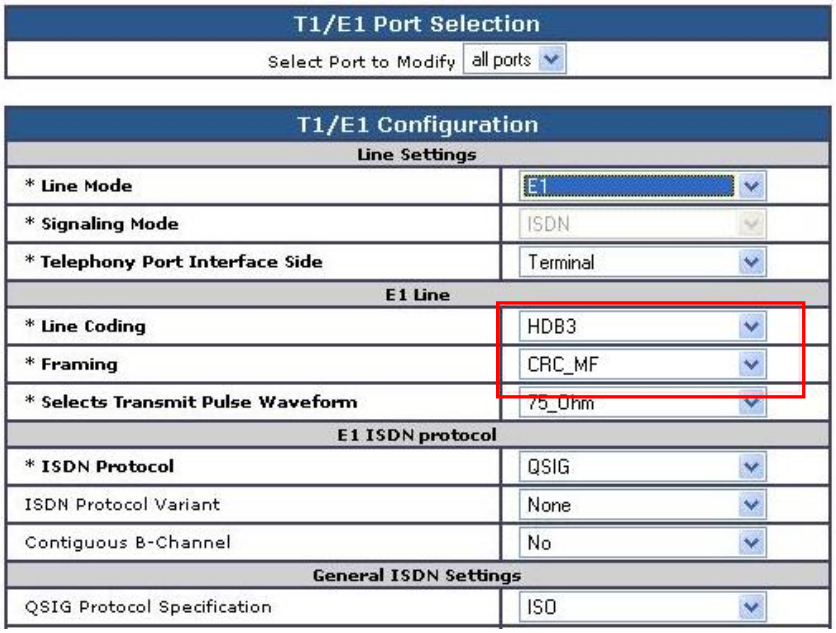
- Assign LAN 1 on the gateway a unique IP address, subnet mask and network gateway address (if the latter is required).
- Configure the gateway to use the SIP VoIP protocol.
- Set the Line Mode to E1.
- Set the Protocol to ISDN - QSIG.

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

- In the IP Settings page:
  - Set the `BOOTP Enabled` parameter to '`No`'. (the default is Yes)



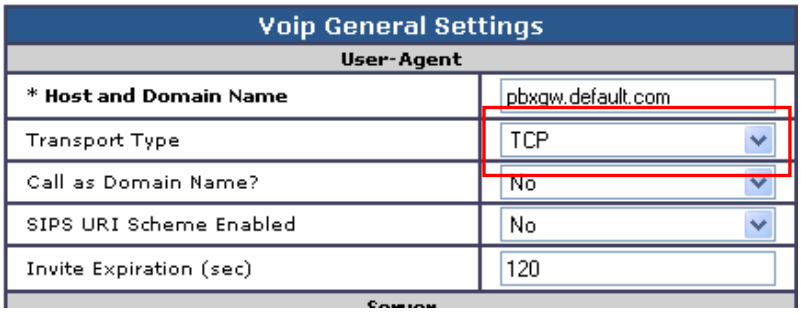
- In the T1/E1 General page:
  - Set the Line Encoding and Line Framing as required by your E1 Interface. Typical settings are Encoding = HDB3 and Framing = CRC_MF.



T1/E1 Port Selection	
Select Port to Modify	all ports

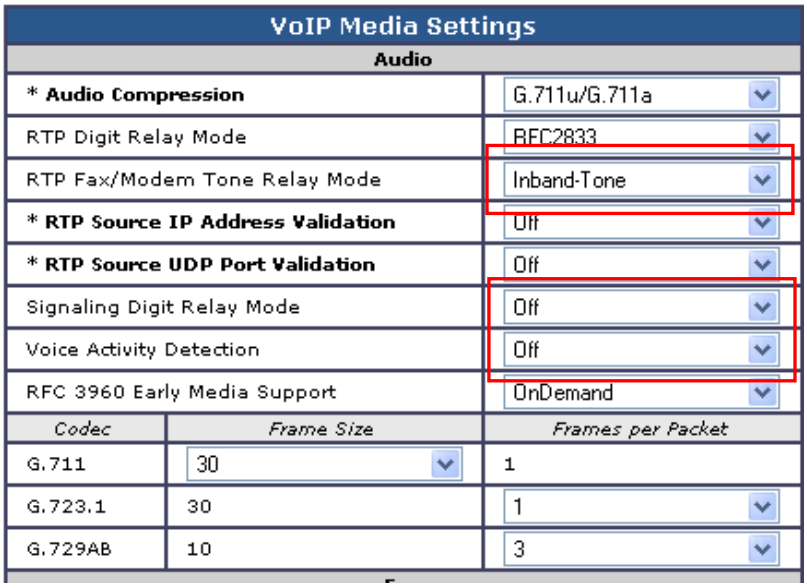
T1/E1 Configuration	
Line Settings	
* **Line Mode**	E1
* **Signaling Mode**	ISDN
* **Telephony Port Interface Side**	Terminal
E1 Line	
* **Line Coding**	HDB3
* **Framing**	CRC_MF
* **Selects Transmit Pulse Waveform**	75_Ohm
E1 ISDN protocol	
* **ISDN Protocol**	QSIG
ISDN Protocol Variant	None
Contiguous B-Channel	No
General ISDN Settings	
QSIG Protocol Specification	ISO

- In the VoIP General page:
  - Set the Transport Type parameter to TCP (the default is UDP)



Voip General Settings	
User-Agent	
* **Host and Domain Name**	pbxgw.default.com
Transport Type	TCP
Call as Domain Name?	No
SIPS URI Scheme Enabled	No
Invite Expiration (sec)	120

- In the VoIP Media page:
  - Set the `RTP Fax/Modem Tone Relay Mode` parameter to '`In band-Tone`' (the default is RFC2833)
  - Set the `Signaling Digit Relay Mode` parameter to '`Off`' (the default is On)
  - Set the `Voice Activity Detection` parameter to '`Off`' (the default is On)



5.2 Routing Engine Configuration

NOTE: For all the examples in this document going forward the term 'inbound call' refers to a call in the TDM to IP direction and the term 'outbound call' refers to a call in the IP to TDM direction.

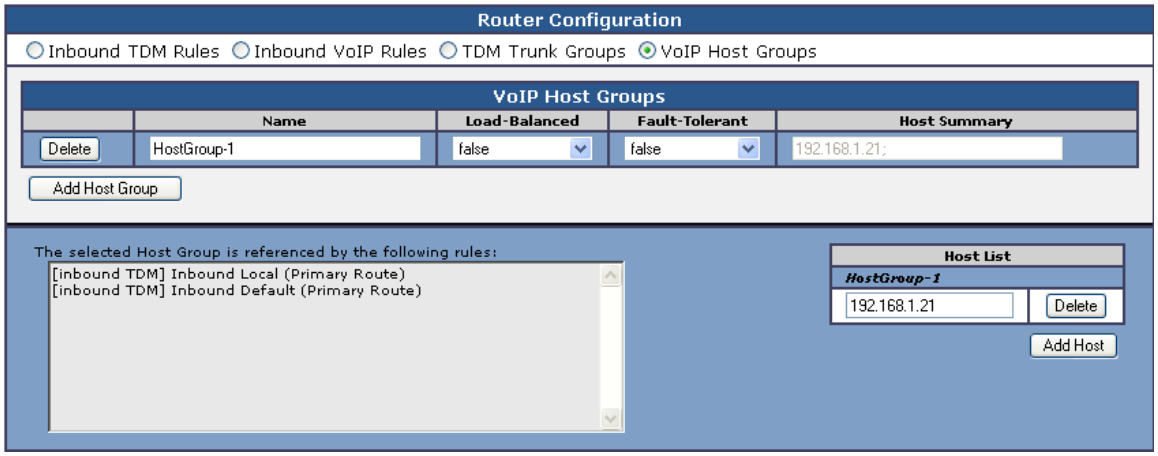
The example given in the system diagram at the start of this integration guide has the following dialing plans in the system:

- All TDM side stations have DID numbers assigned in the 2xxx extension range.
- All OCS side stations have DID numbers assigned in the 5xxx extension range.

All inbound calls need to be sent through to the Mediation Server at a specific IP address.

5.2.1 VoIP Host Group configuration

The first item to take care of is to set up the IP endpoint to use as the IP destination for all inbound calls. This is done in the routing table under the section VoIP Host Groups. Define a single host group (using the default group is fine) that includes the IP address of the gateway listening side of the Mediation Server; in this example case the IP address 192.168.1.21 is for this.



5.2.2 TDM and VoIP Routing Rule Configuration

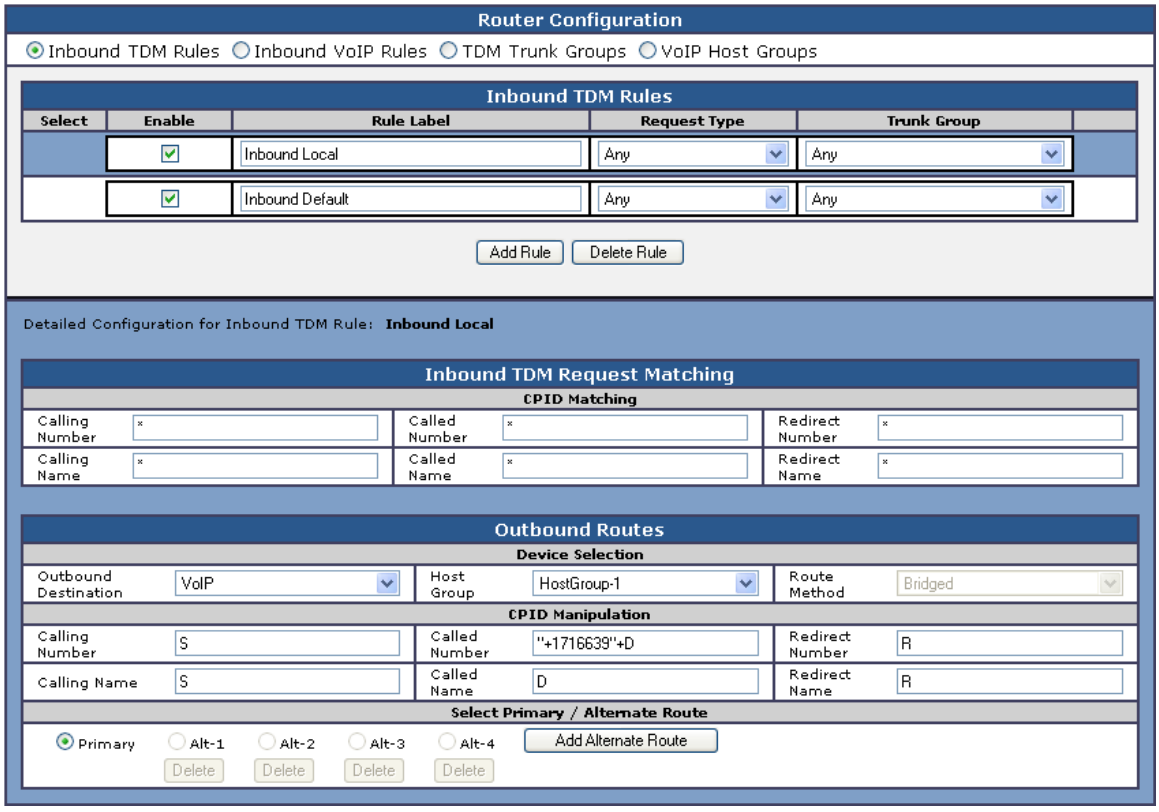
The second item is to configure the routing rules that will associate inbound or outbound calls with the proper digit manipulation rules for the type of call they need to service. This will require that the gateway perform some digit manipulation on calls that go from the TDM side to the IP side as well as in the reverse direction, IP to TDM.

The major idea here to remember is that OCS expects to get, and will send out, all addresses in E.164 format. This means that the gateway needs to recognize the need to convert up and down as needed to and from this format as calls pass through. To do this you make use of the Routing engine's CPID manipulation rules.

5.2.2.1 Inbound TDM Rules

When a local user on the PBX picks up their phone and calls one of the extensions on the OCS side within the 5xxx range the gateway will receive a call with a calling party of 4 digits. It then needs to convert that number up to full E.164 format and send the call on to OCS.

This example will take any number and then convert it into the full E.164 format by concatenating a prefix of '+1716639' onto the front of the number where 716 is the area code and 639 is the local exchange.
Other calls, such as DIDs that arrive over TDM trunks from the PSTN may provide a full 10 digits to the PBX or they may only provide the extension number after the prefix has been stripped off by the PBX. Depending on your site specific requirements you may need to add or build different rules to handle these cases. An example of the inbound rule for local PBX users is shown below:



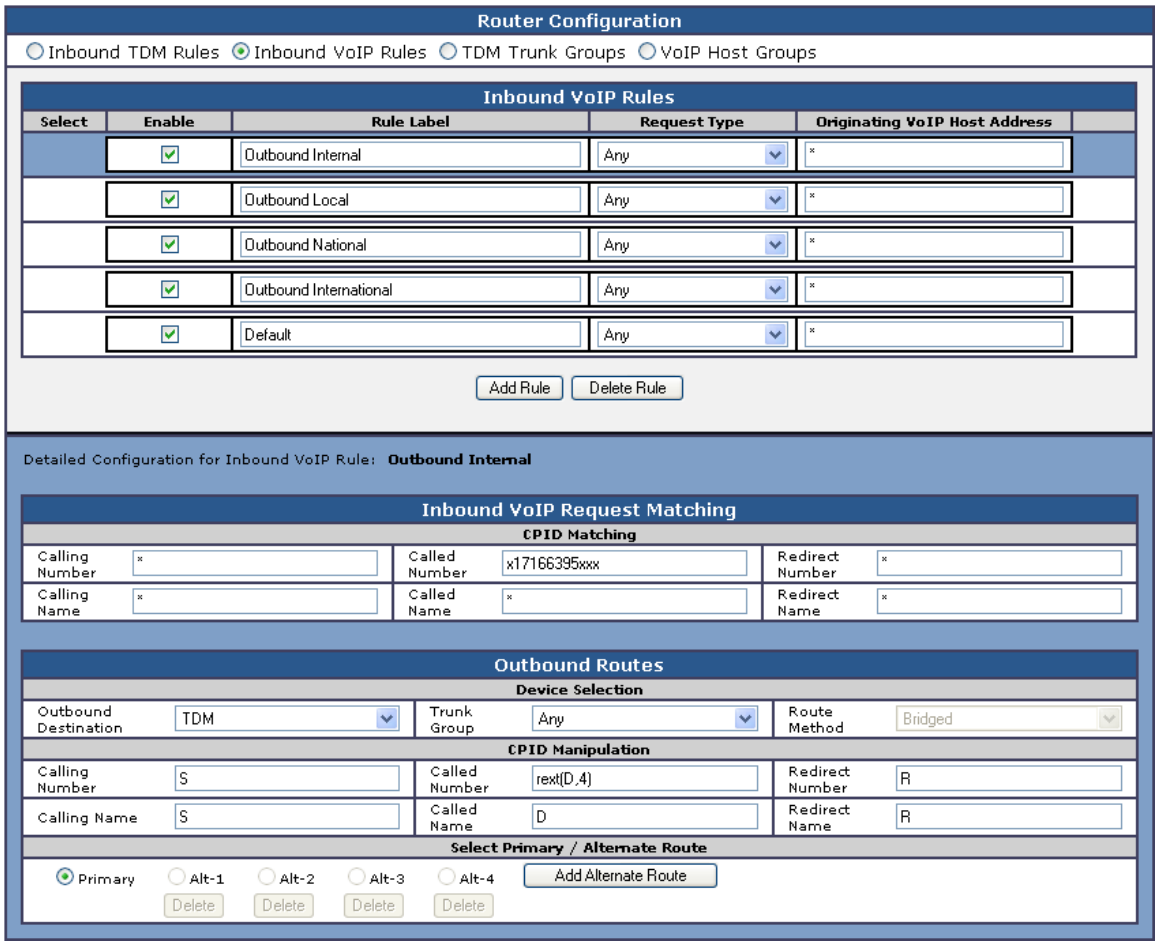
The CPID matching rule is simply a * meaning that any dialed number from a local user presented to this trunk will be seen by this rule. The CPID manipulation rule then uses the digits that are being seen (in this example it will be a 4 digit number because that is how the trunk is programmed) and then adds the prefix of "+1716639" onto it to build the full E.164 number that is needed for OCS. This rule also sets the destination to the VoIP Host group defined previously that points to the inbound IP address of the Mediation Server.

In addition to this rule a default rule has been left in place that acts as a catch all. This rule performs no CPID manipulation at all and just tries to send the call to the VoIP host group as dialed.

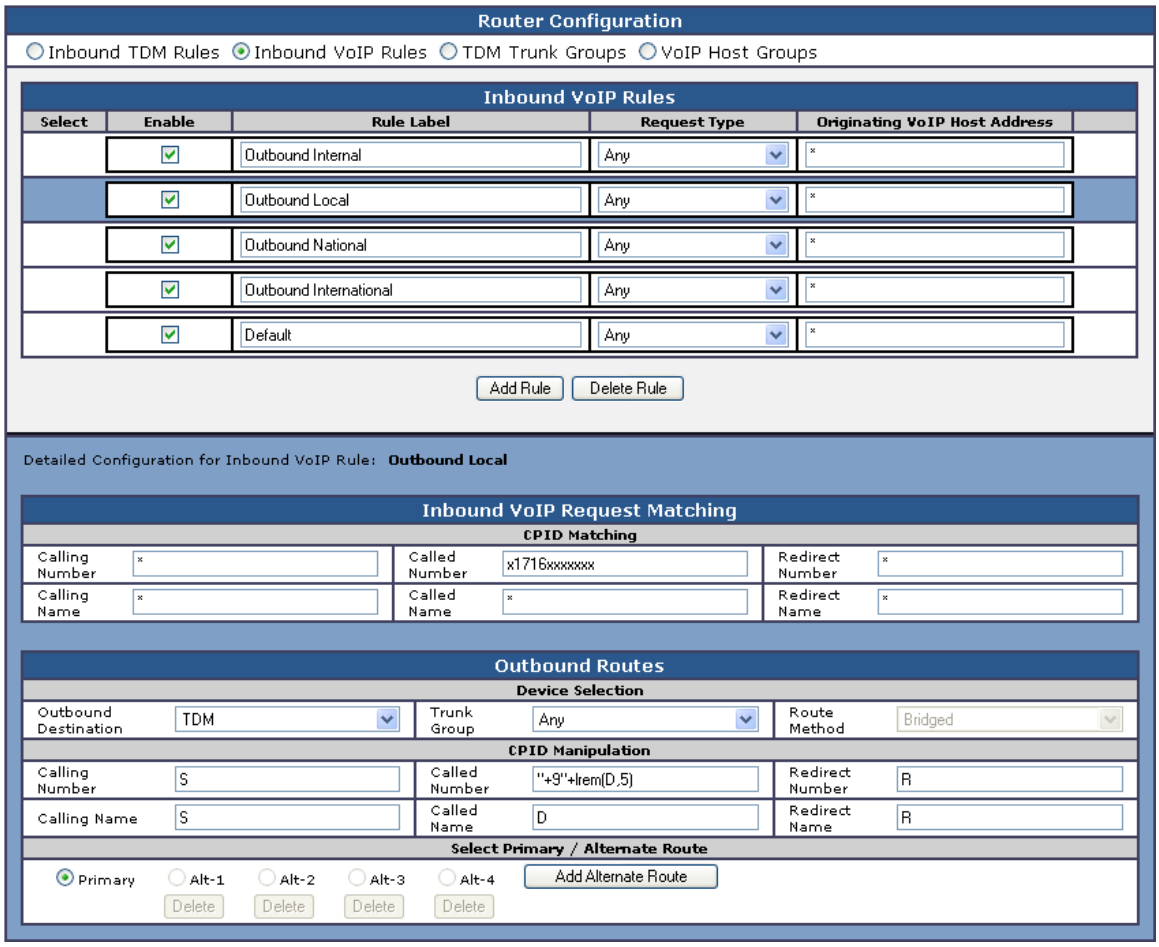
5.2.2.2 Inbound VoIP Rules

When an OCS user dials a number OCS will, through the use of normalization rules in the Location profile, provide the gateway with a number in full E.164 format. The gateway needs to be able to recognize various number patterns in inbound IP calls and properly manipulate them for the outbound TDM call that results.

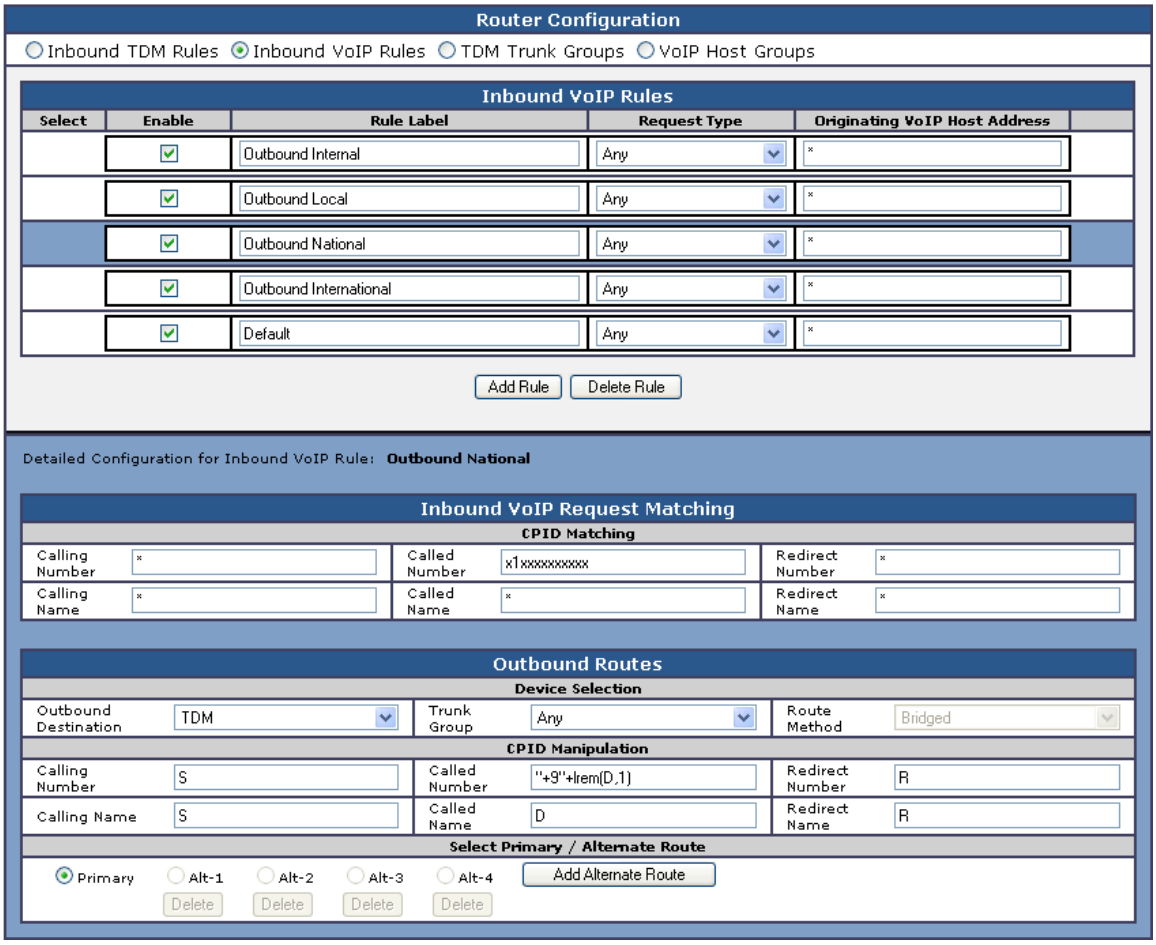
In the example here, OCS has been setup (as you will see later) with a route that directs all calls that meet the pattern 5xxx to the gateway in full E.164 format. The gateway then needs to know how to identify these numbers as extensions that are local on the PBX and manipulate them accordingly. To do this it needs to simply extract the right 4 digits from the called number provided to remove the prefix of "+1716639" and leave the last 4 digits remaining. Local, national and international numbers are going to need to be manipulated. At very least they will need a trunk access number, like a 9, pre-pended onto the front of them in order to dial an outside line. These can also be done using manipulation rules as follows:



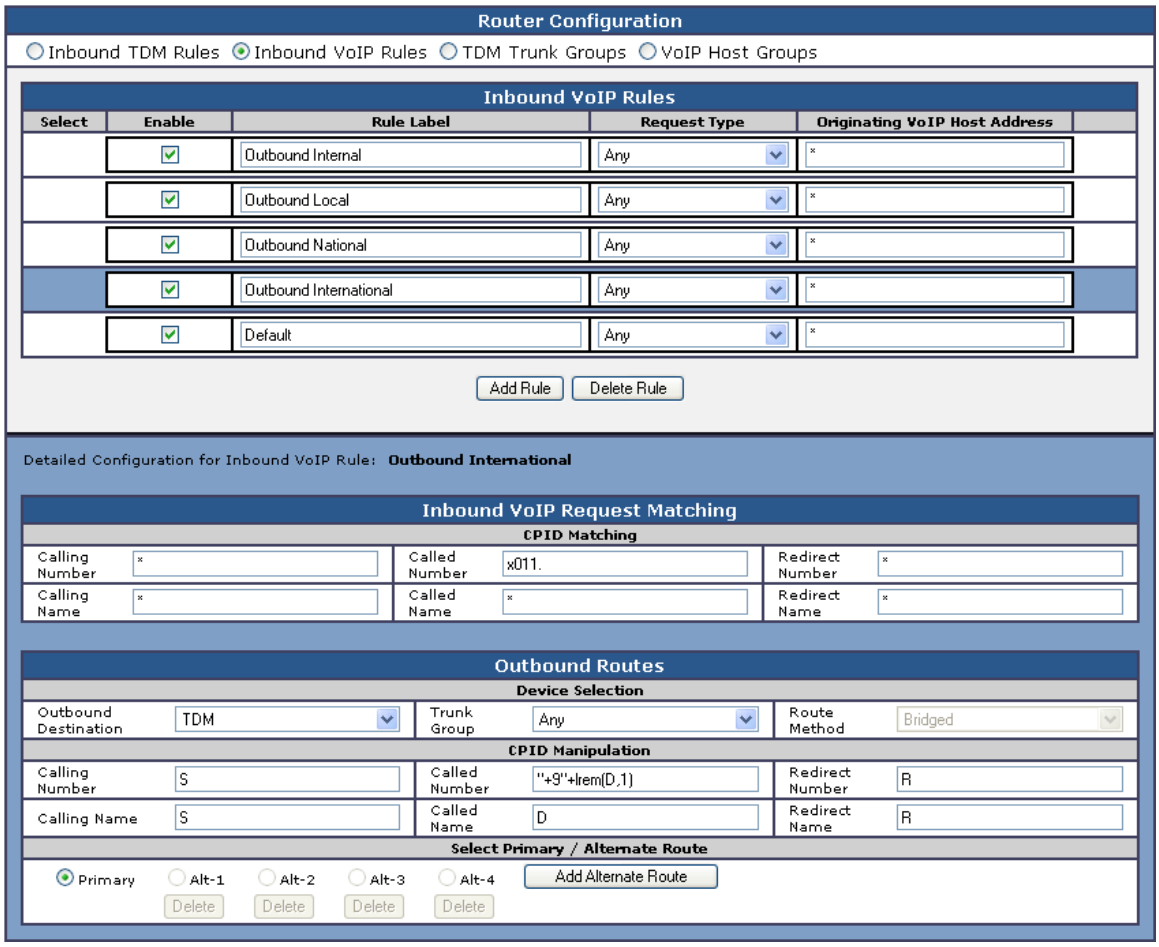
In the screen shot above, the first rule 'Outbound Internal' is selected. Notice that the blue bar near the top of the screen highlights this rule. The lower half of the screen displays the details of the currently selected rule. This rule matches outbound calls that have a called party number that starts with '+17166395' followed by any three digits. This rule is designed to match the locally defined TDM extensions as shown in the first figure in this document. Calls that match this rule are meant to go to a local user on the PBX. The CPID manipulation section of this rule extracts the last four digits from the called party number. The extracted four digits are then dialed as a local extension on the PBX.



In the screen shot above, the rule 'Outbound Local' is selected. This rule matches outbound calls that have a called party number that starts with '+1716' followed by seven digits. This rule is designed to match the calls within the same area code, but not from the same PBX. Calls that match this rule are meant to go to a local user that is not on the PBX. In the CPID manipulation area the trunk access code is added to the string and the leading 5 characters are stripped off (the '+1716'). The full string out as '+9xxxxxxx' is sent.



In this rule labeled as ‘Outbound National’ any number dialed that starts with ‘+1’ and includes 10 digits indicates a number that is not in the local area code. In this case the CPID manipulation simply adds a +9 to the start of the number and strips off the leading + creating a result of ‘+91xxxxxxxxxx’.



In this rule labeled as ‘Outbound International’ any number dialed that starts with ‘+011’ and includes any number of digits indicates a number that is not in the local area code. In this case the CPID manipulation simply adds a +9 to the start of the number and strips off the leading + creating a result of ‘+9011xxxxxxxxx’.

The last rule that is defined is another default rule that acts as a catch all and simply attempts to dial any number provided that has not matched the previous rules in the list.

Note 1: The last two rules labeled as ‘Outbound National’ and ‘Outbound International’ COULD have been combined into one rule since the CPID manipulation was the same in both. The rules have been split out here in this example simply for clarity of the example. Also, if the environment uses different trunks for local, national (long Distance) and international calls, breaking these rules out into separate segments allows you to also define trunk groups and direct calls of these specific types to those individual trunks.

Note 2: The rules are evaluated in the order they are listed, top down. The first rule that matches is used so the order is important. Always consider placing your more specific rules at the top of the order and the more general at the bottom.

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:

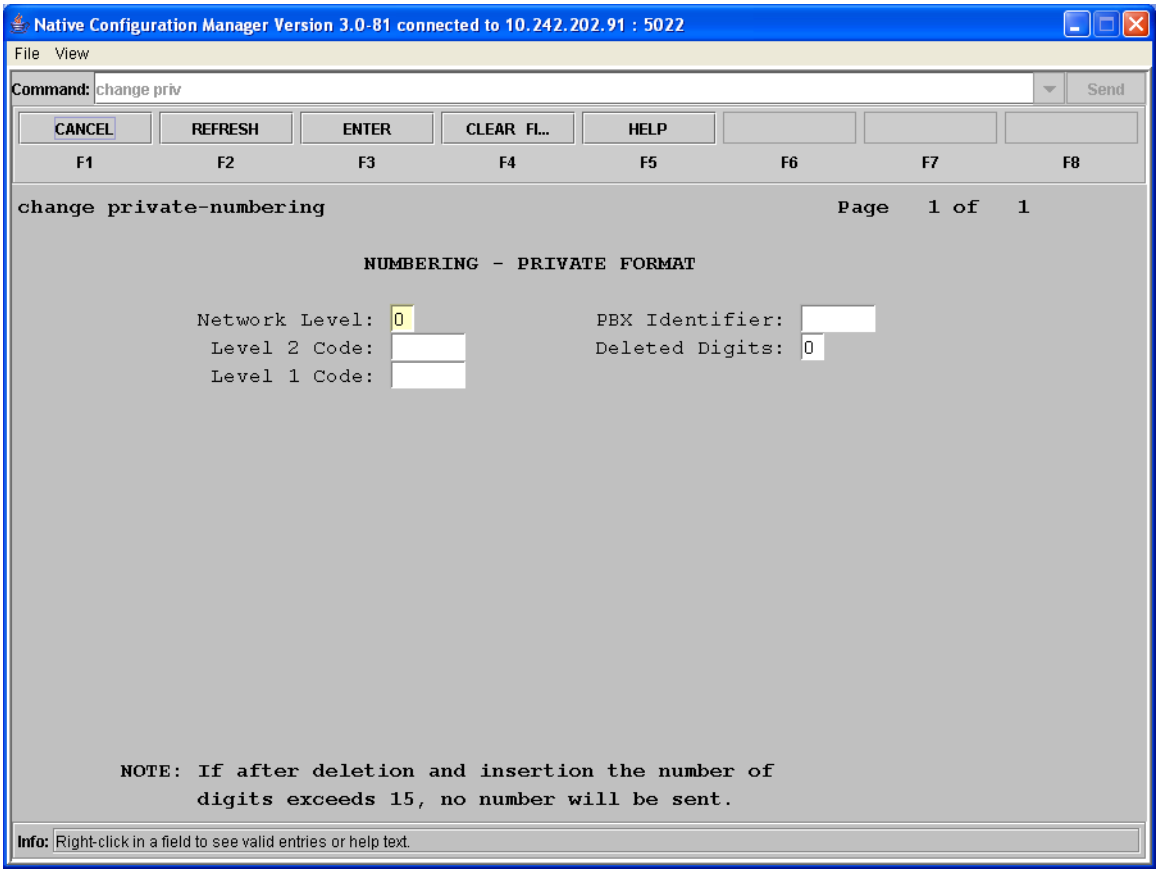
- Configuring private-numbering.
- Setting up system parameters and customer options.
- Administrating E1 interface.
- Programming AAR for trunk access.

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 Configuring Private-Numbering

Use the `change private-numbering` command to modify the Network Level to 0.



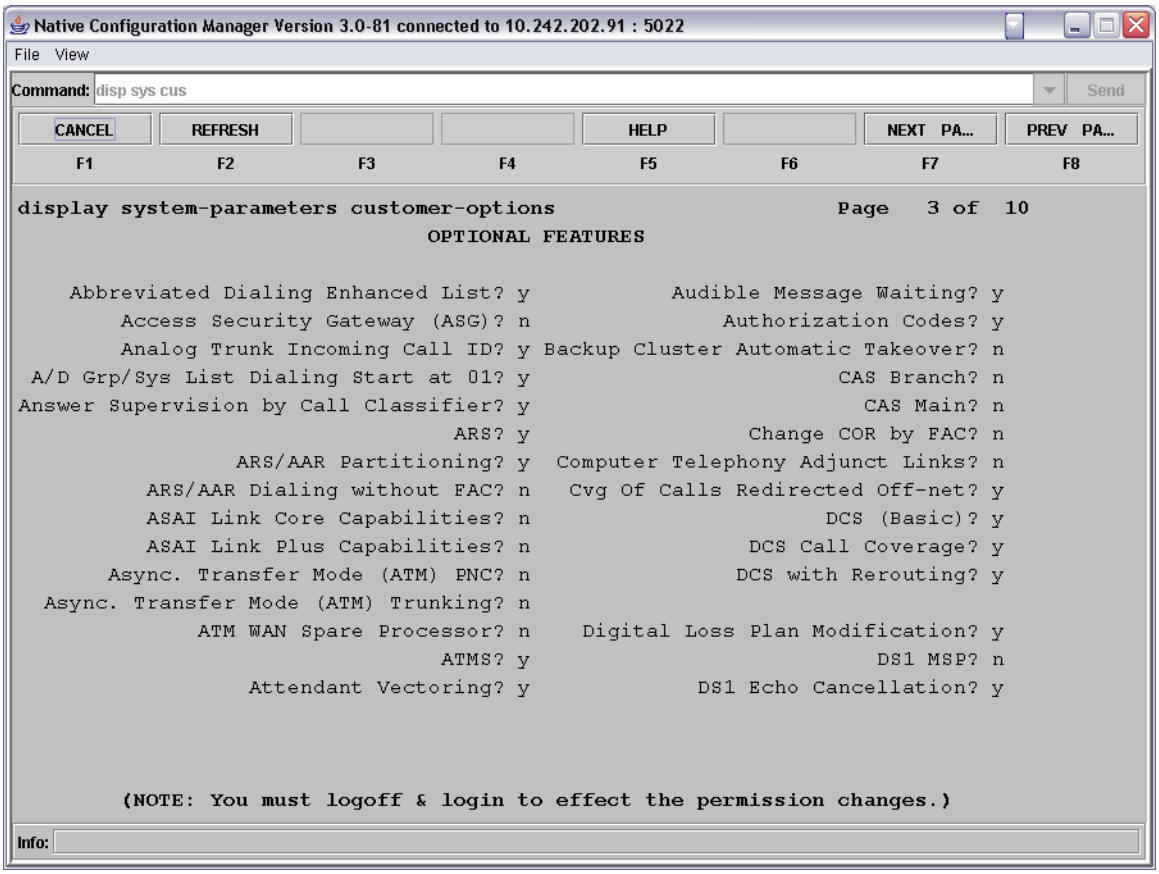
Before moving on to the next command, be sure to save your work.

6.2 Setting Up System Parameters and Customer Options

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

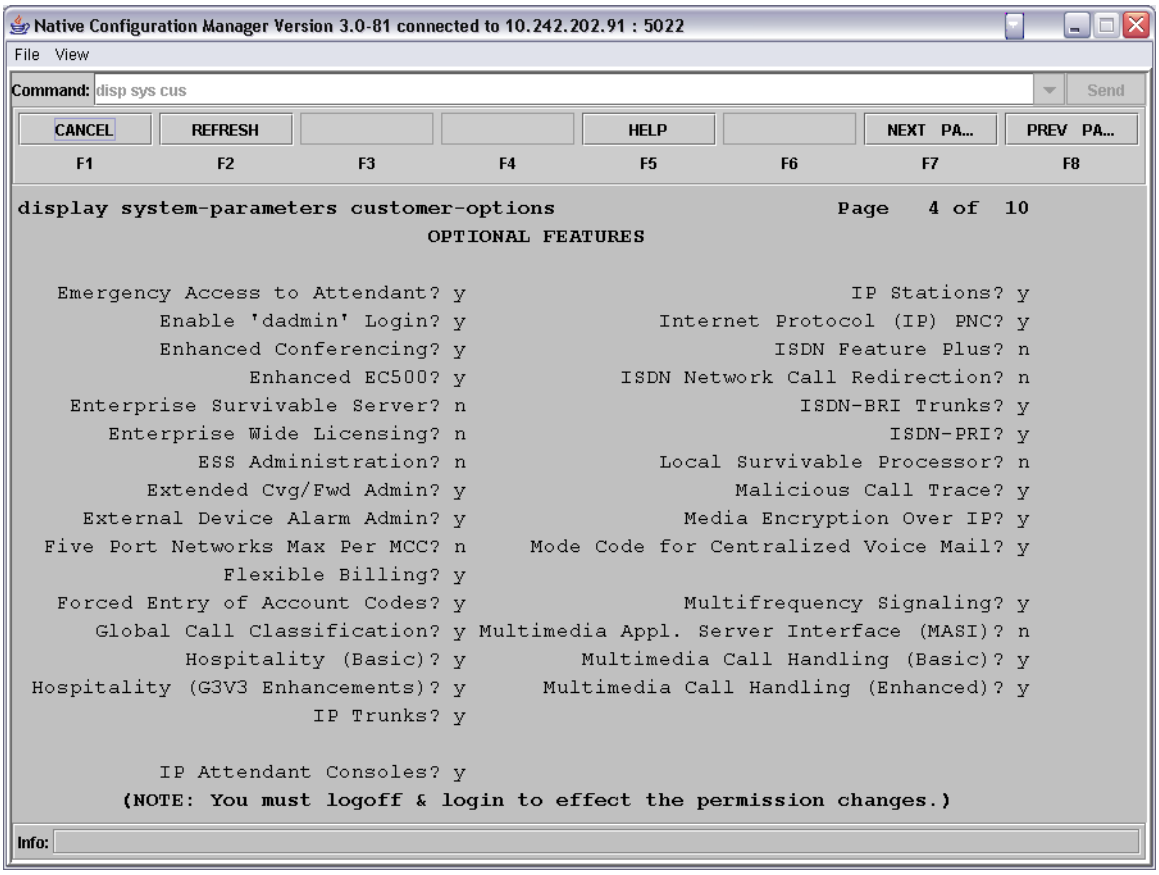
Note that any configuration parameters in this area that are not already enabled may require special maintenance or configuration by Avaya Customer support and/or they may require additional software packages to be installed to configure properly. Once the command has been

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



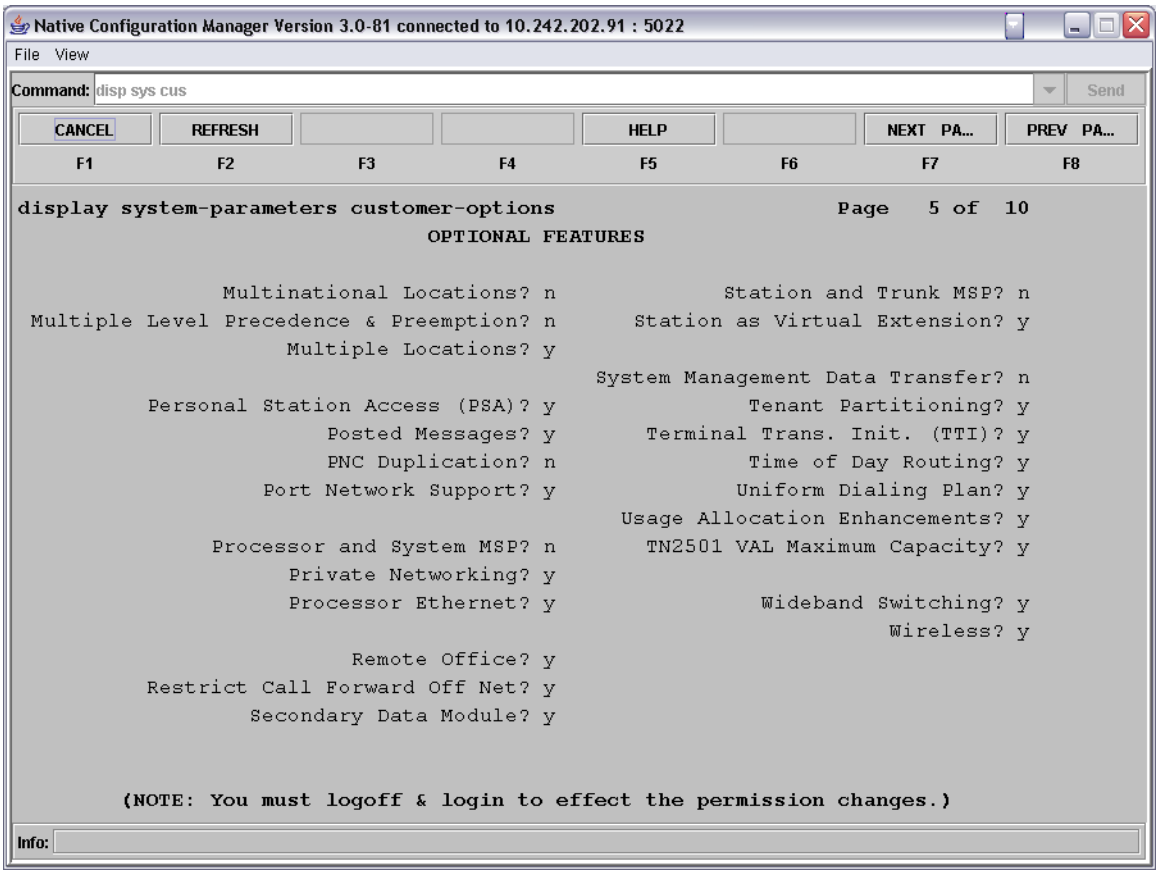
Review the `DCS (Basic)`, `DCS Call Coverage`, and `DCS with Rerouting` options. They must be set to `y` for this integration to work properly. It they are not enabled then you must contact your Avaya support representative to have these features enabled.

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



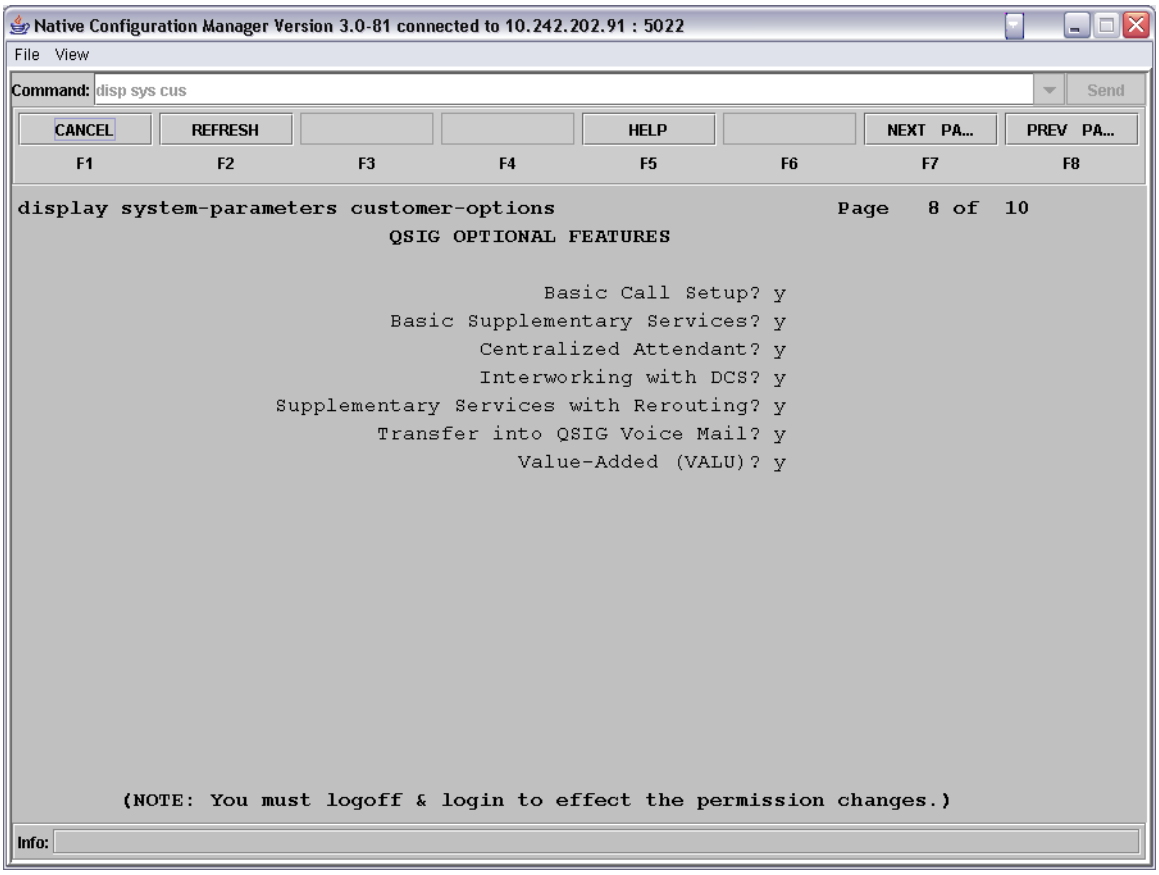
Review the `ISDN-PRI` option. It must be set to y for this integration to work properly. If this option is not enabled then you must contact your Avaya support representative to have these features enabled.

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 your Avaya support representative to have these features enabled.

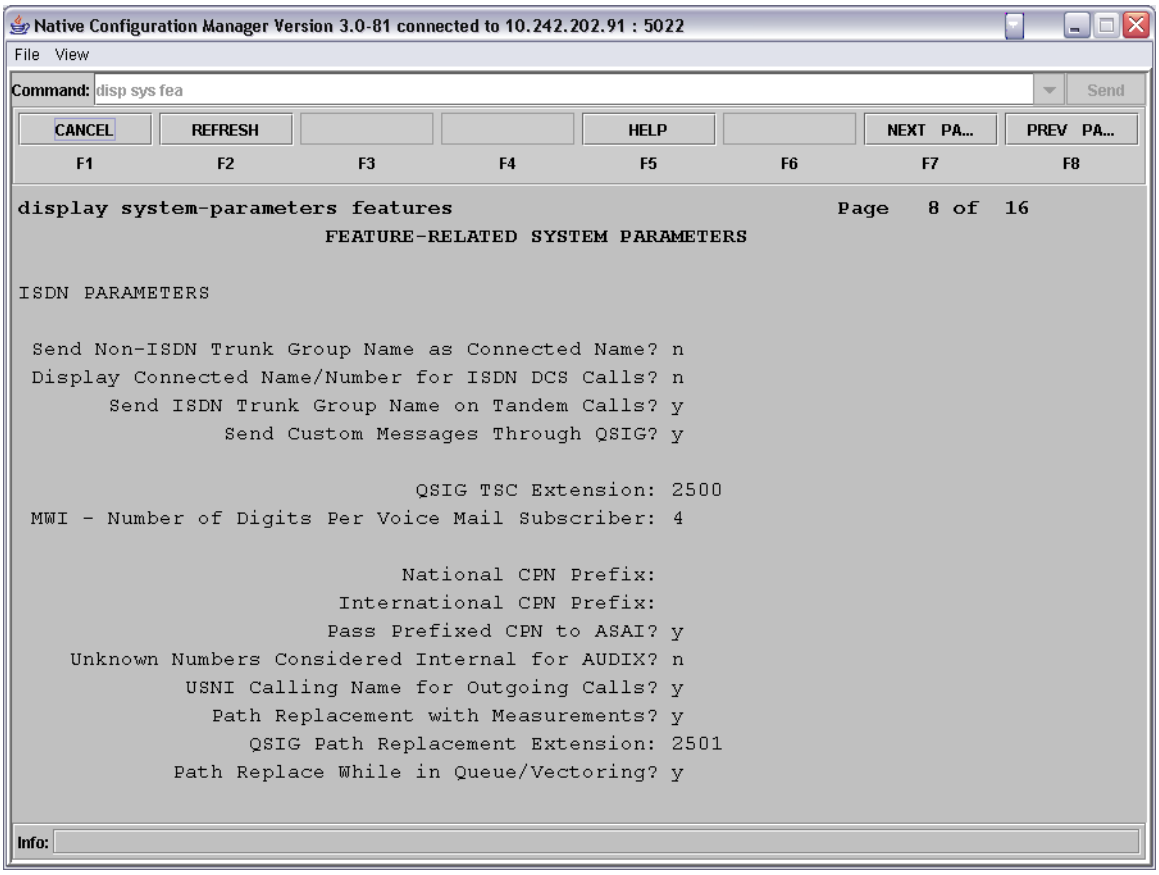
Press the `NEXT PAGE` soft key until page #8 (`QSIG Optional Features`) is displayed.



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. It they are not enabled then you must 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 then you need to contact your Avaya support representative.

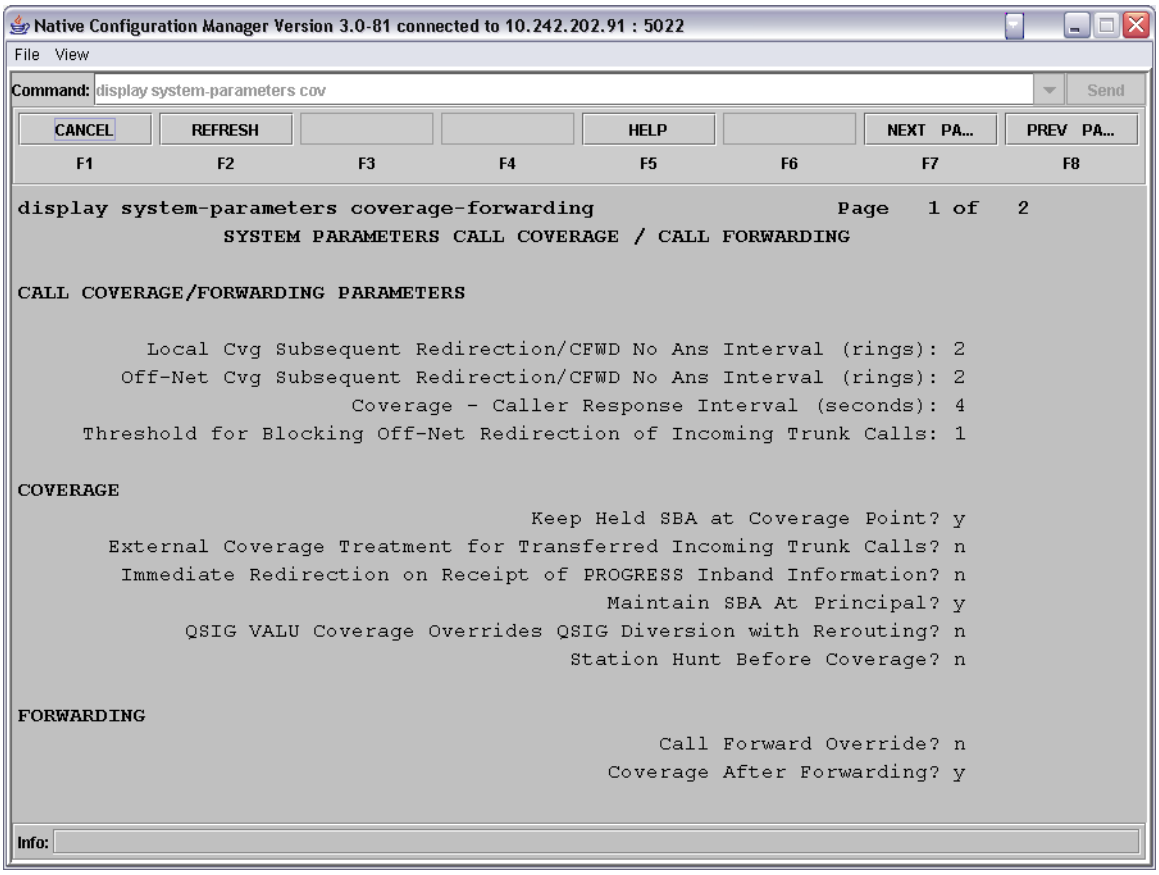
Press the `NEXT PAGE` soft key until you get to the ISDN Parameters page.



- 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 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.

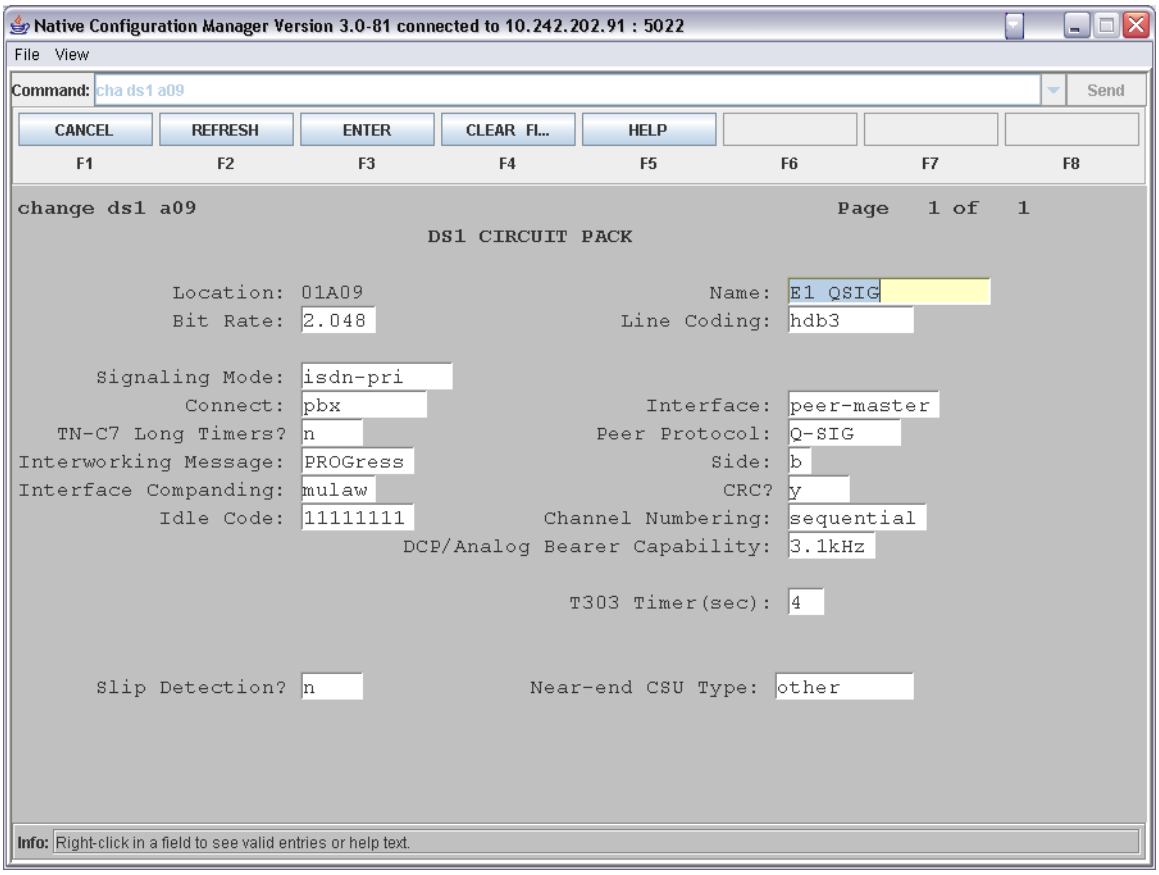
Before moving on to the step, be sure to save your work.

6.3 Administrating E1 Interface

`Use the add ds1` command to setup 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 a09
```

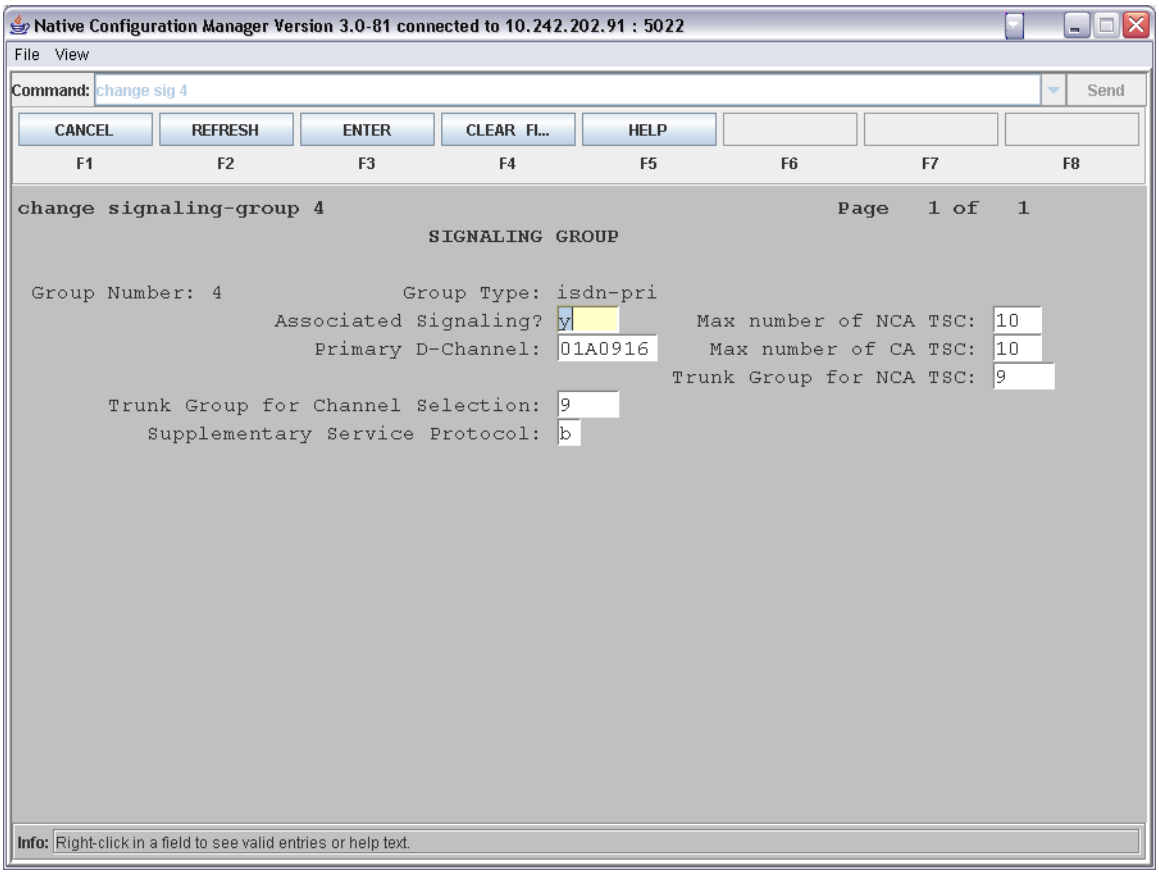
Add a ds1 circuit card into the configuration that is located in cabinet ‘a’ and is in slot #09 of that cabinet.



- In the `Name` field enter in any string to assign the DS1 a name
- In the `Bit Rate` field enter `2.048`. *(The standard clock rate for E1)*
- In the `Line Coding` field enter `hdb3`.
- 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)*

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

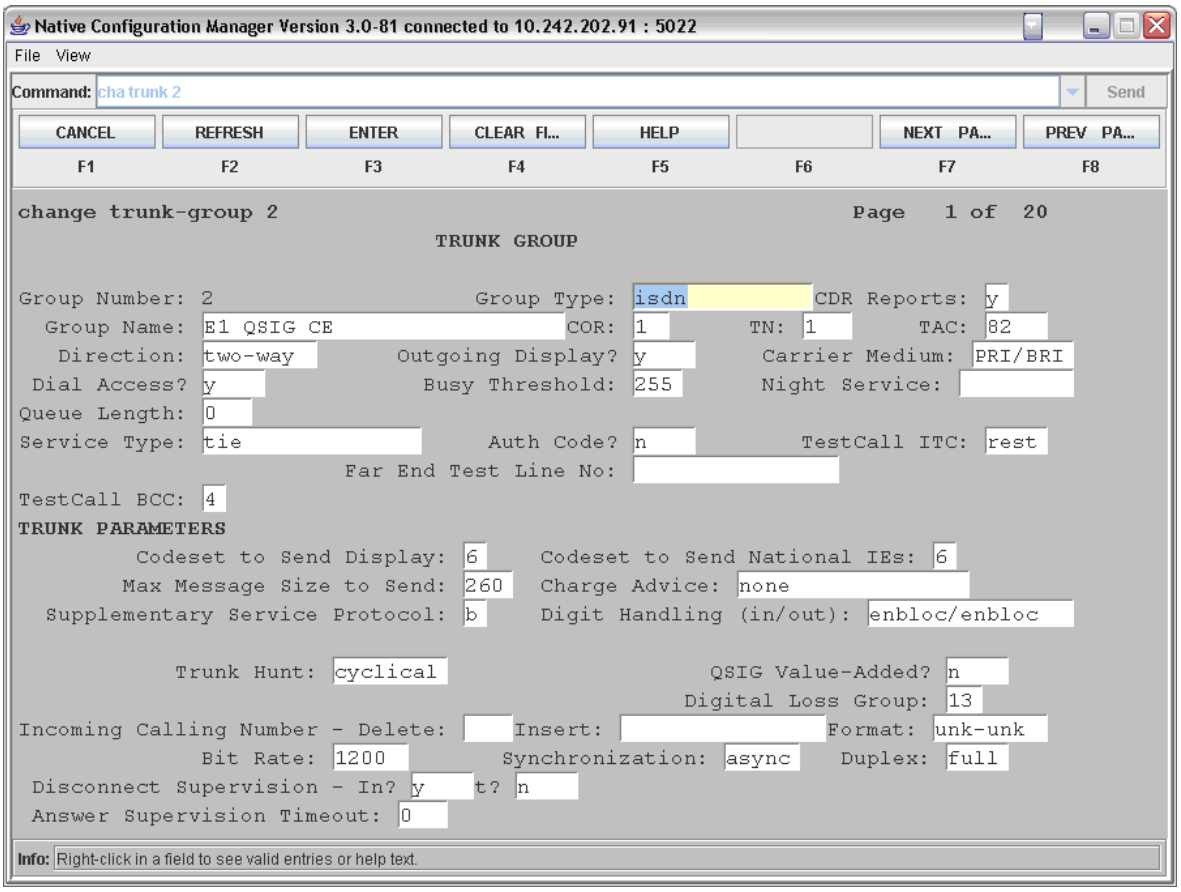
Use the `add signaling-group` command to build a signaling group for the QSIG link to the gateway. This signaling group is what specifies the type of interface being used, the signaling channel (D-Channel) wiring address for the E1 span being used, and the protocol used on the signaling channel.



- 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 `XXXXX16`
  - Where XXXXX16 is the 16th port number of the E1 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`

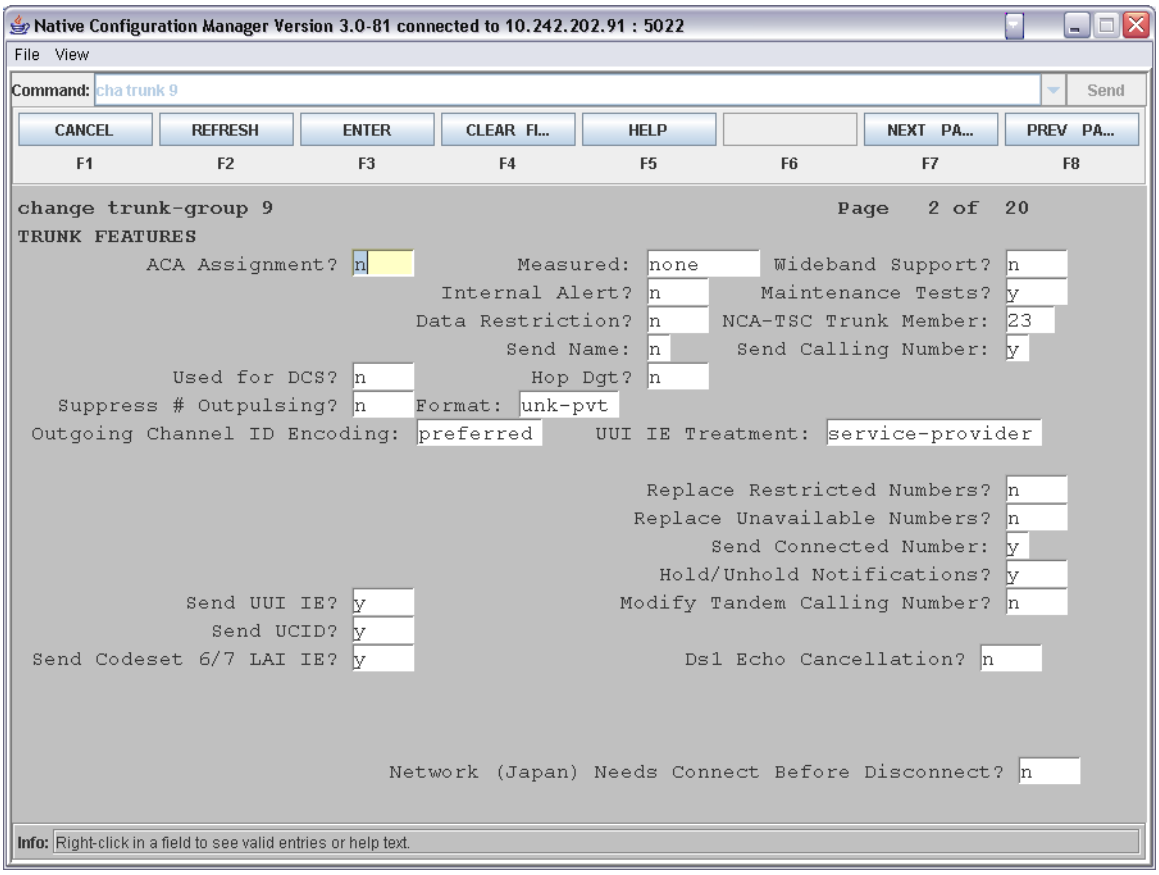
Before moving on to the next command, be sure to save your work.

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.



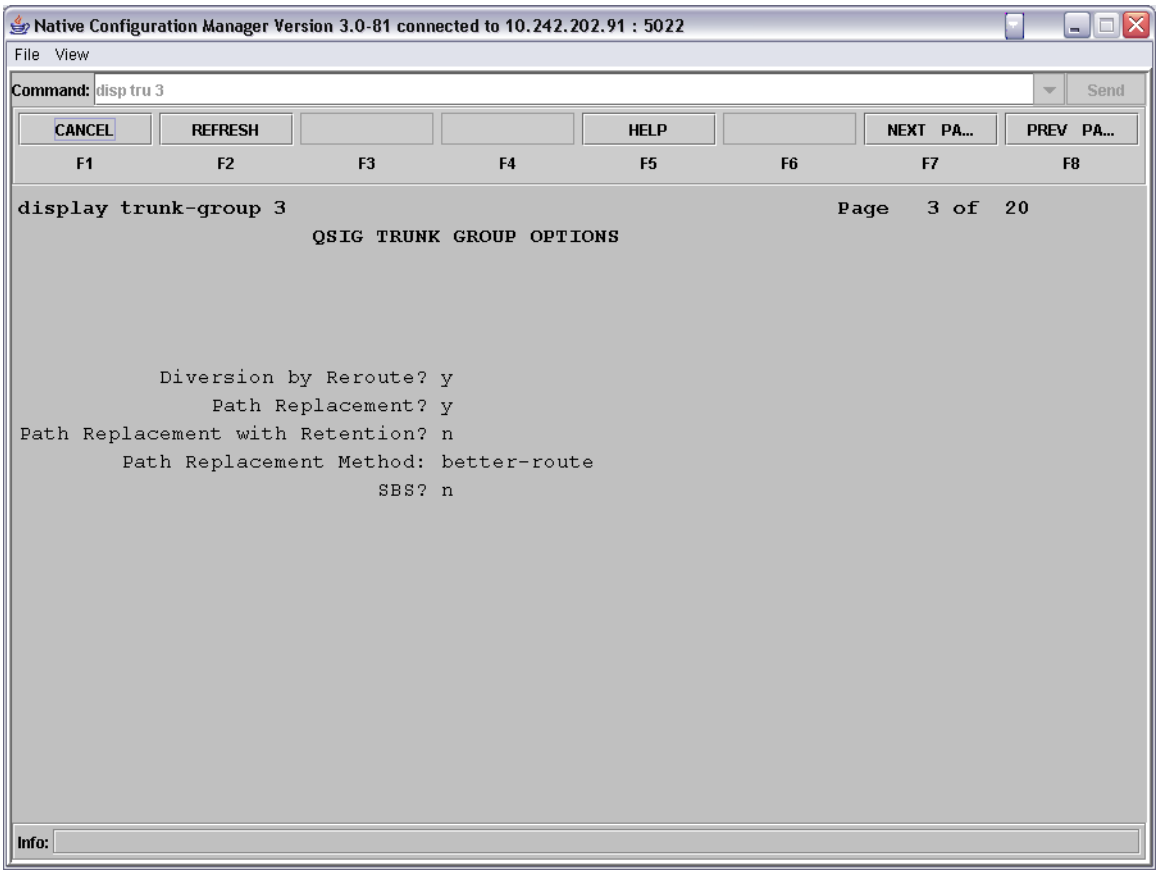
- In the `Group Type` field enter `isdn`.
- In the `Group Name` field enter in a text string to give the group a name.
- In the `COR` field enter the number of a class of restriction to be used, if there is going to be one. *Understand that setting any restrictions via a COR can effect how the trunk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

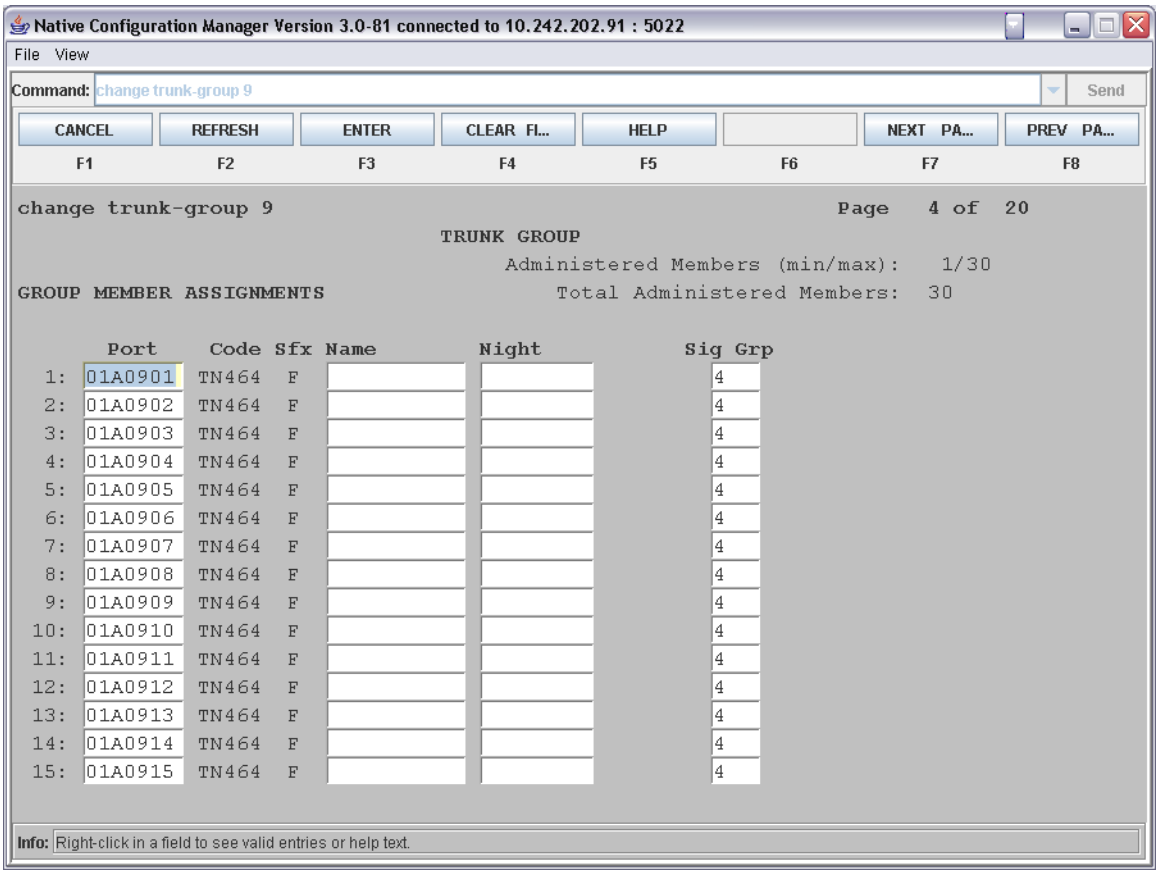
Press `NEXT PAGE` to advance to the next page of settings.



- In the `Diversion by Reroute` field enter `y`.
- In the `Path Replacement` field enter `y`. *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 enter in each bearer channel of the E1 span, leaving off the D-Channel since that is connected to the group via the Signaling Group and is not used to carry voice.



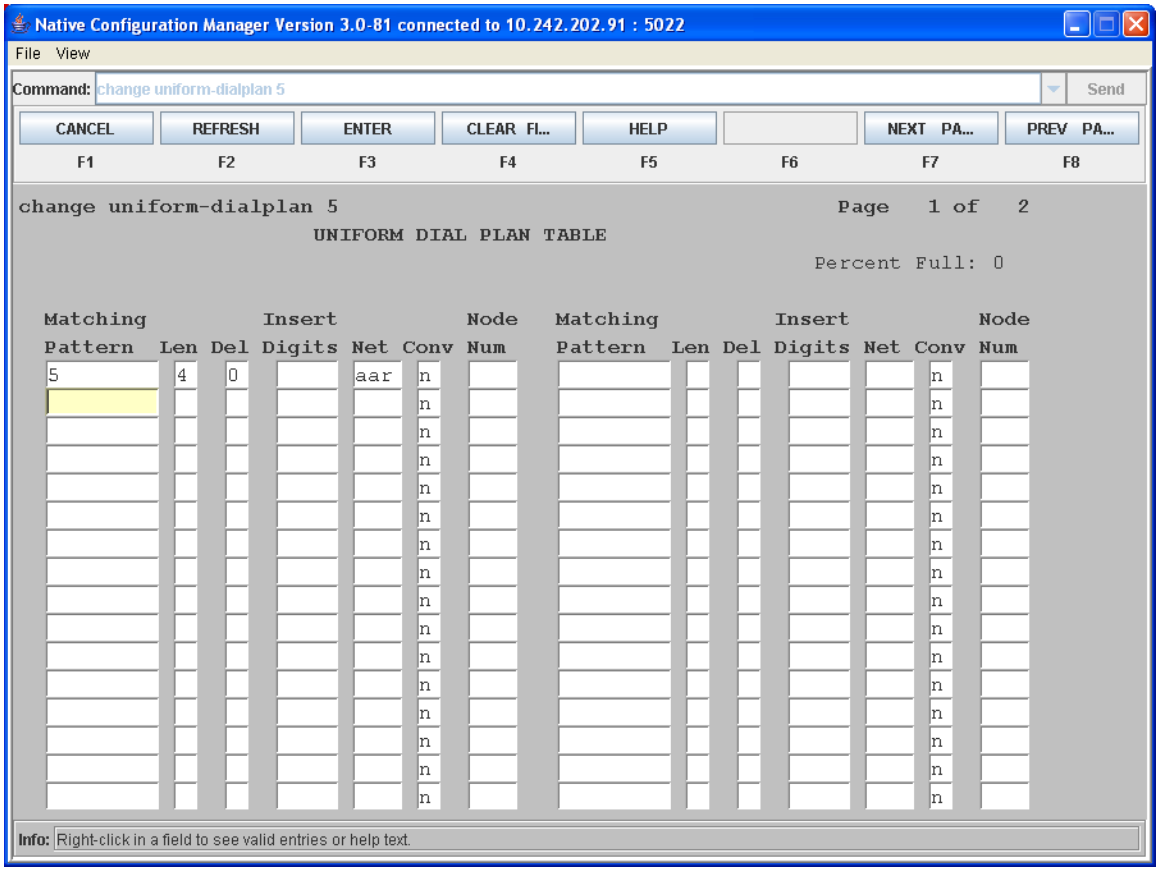
- In each of the `port` fields enter in the port number that designates the physical wiring address of each of the bearer channels in the span.
- In each of the corresponding `Sig Grp` fields for each port enter in the number of the Signaling Group created above.

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

6.4 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.

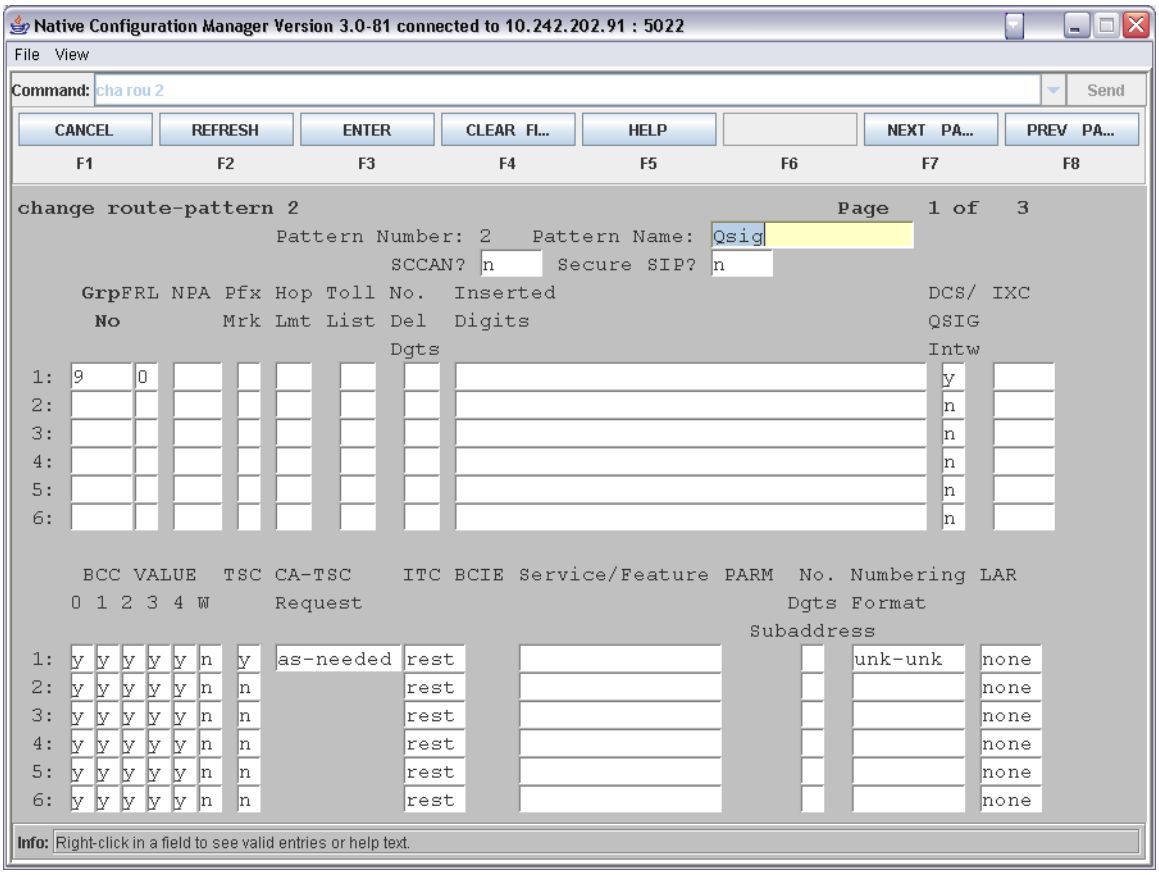
Use the `change uniform-dialplan` command to setup an extension that can be used to access the trunks via AAR.



- In the `Matching Pattern` field enter a number to define as a dialable AAR number. *In this example use 5.*
- In the `Len` field enter `X`.
    - 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.

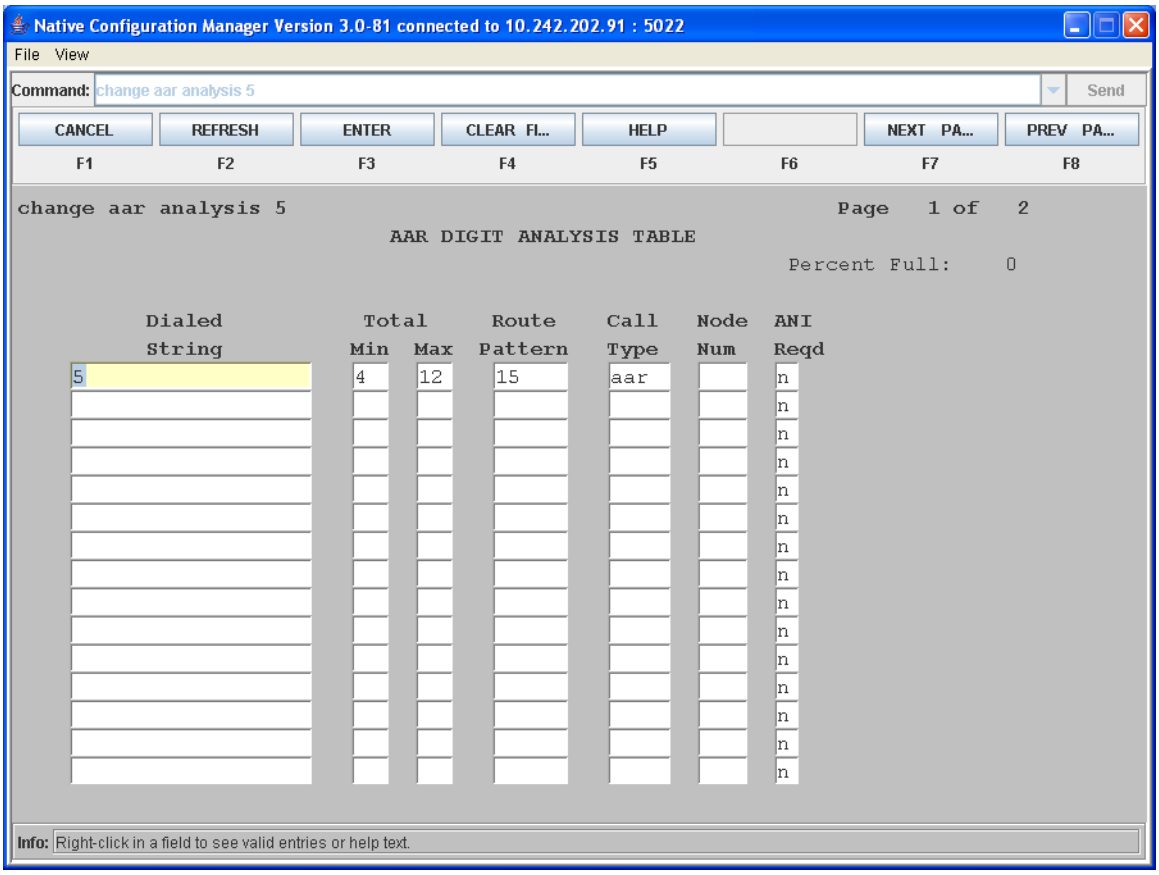
Use the `change route-pattern` command to define a route rule pattern that when referenced will direct calls to the trunk group you created above.



- 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 `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.

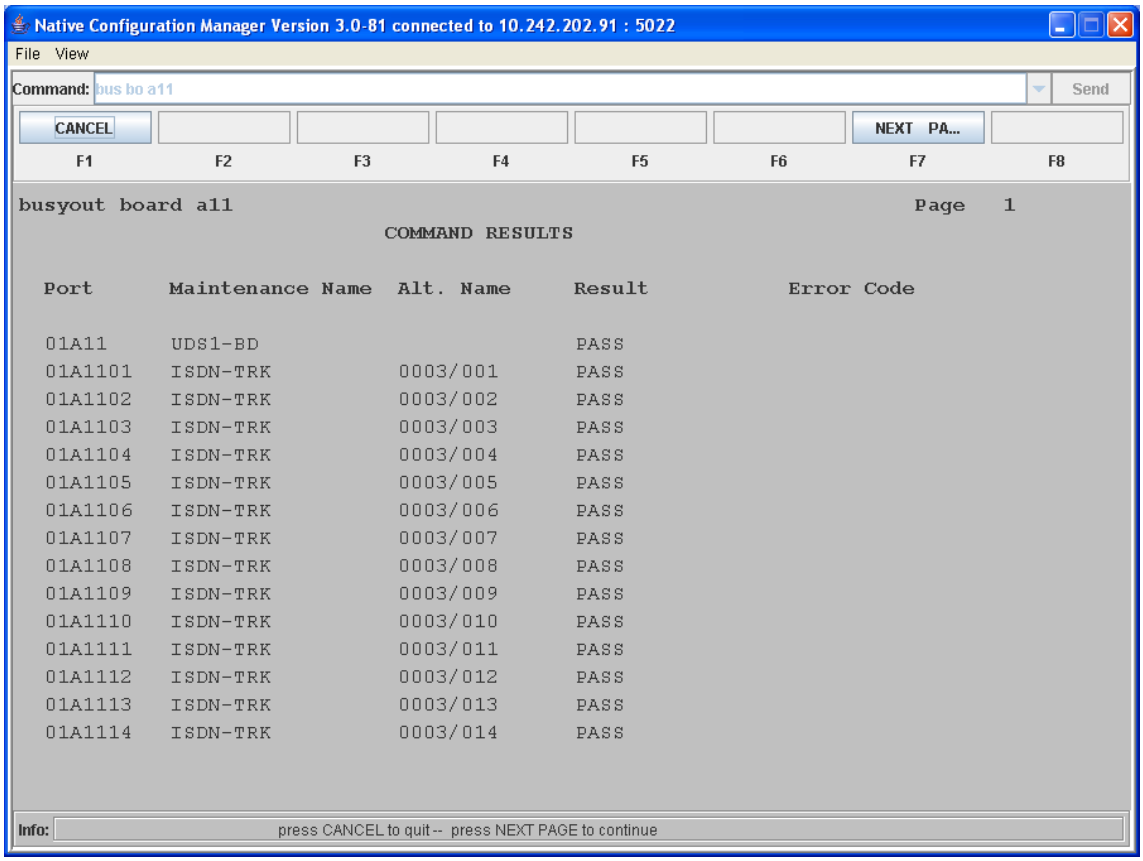
Use the `change AAR Digit Analysis Table` command to set up AAR to bind your dialable number setup in UDP to the route pattern just defined.

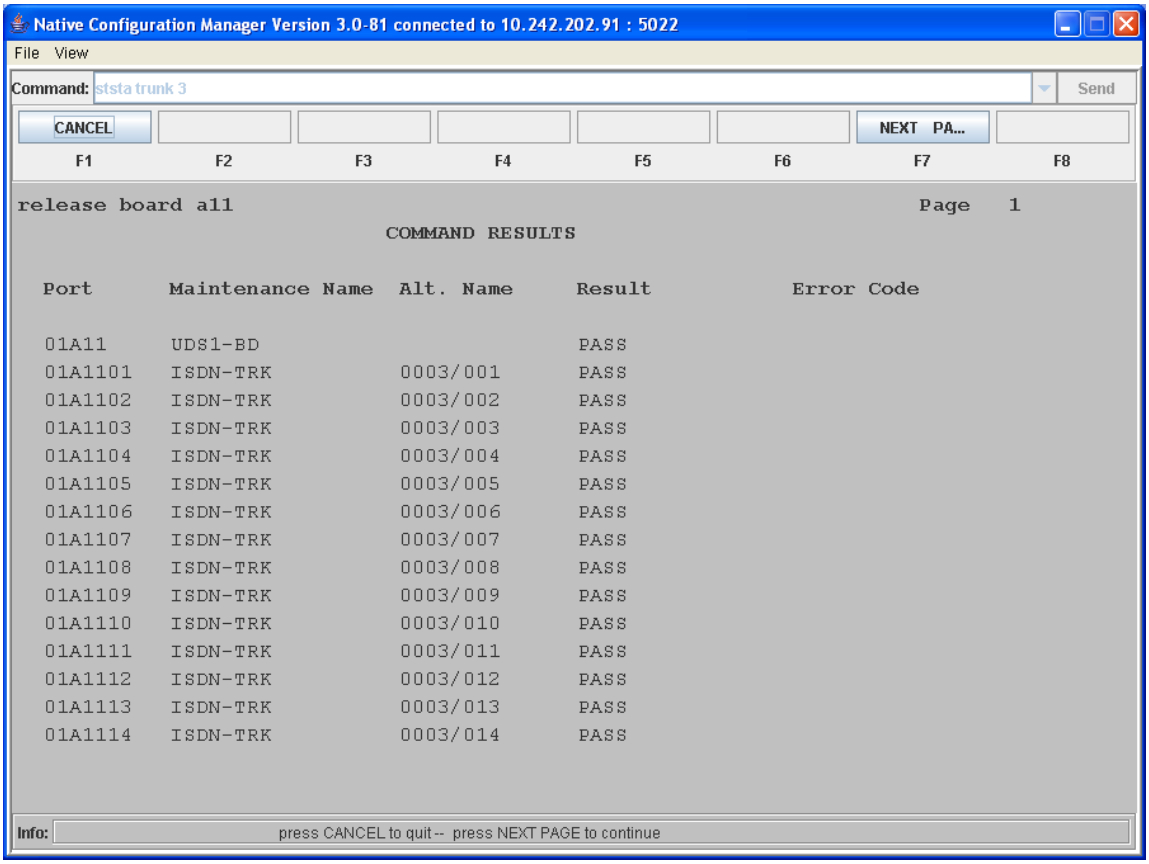


- In the `Dialed String` field enter in the number you defined in UDP as the AAR number. *In this example use 5.*
- In the `Min` and `Max` fields enter in the length of the number you are going to dial. *In this example 4 for the min and 12 for the max.*
- In the `Route Pattern` field enter in the number of the route pattern that was defined above. *In this example 15.*
- 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 an E1 card is initially programmed, by default it is in a busy state. In order to enable the E1 card, it must be set to busy and then released.





In both commands you can use the `NEXT PAGE` soft key 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 OCS

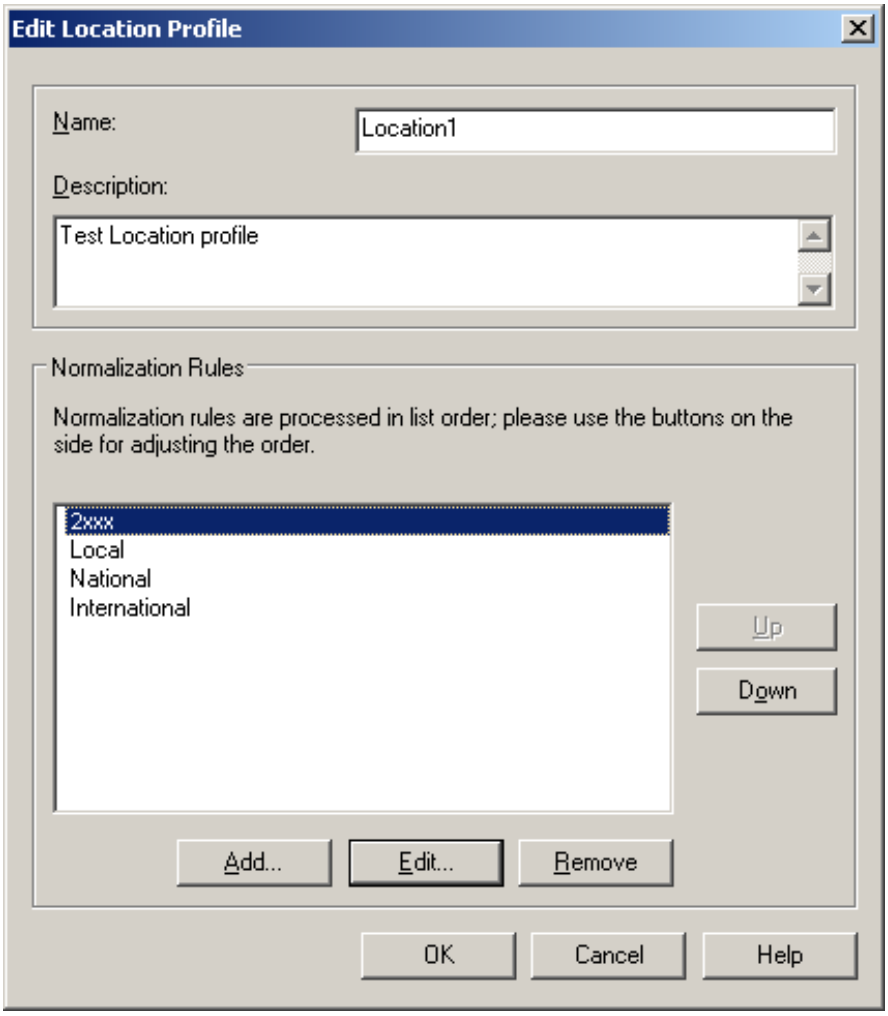
Normalization rules are used to convert all possible dial numbers into full E.164 formatted numbers. OCS uses the standard E.164 format to search for all users listed in Active Directory (AD).

When an 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 that are for internal extensions, local numbers, long distance numbers, and international numbers.

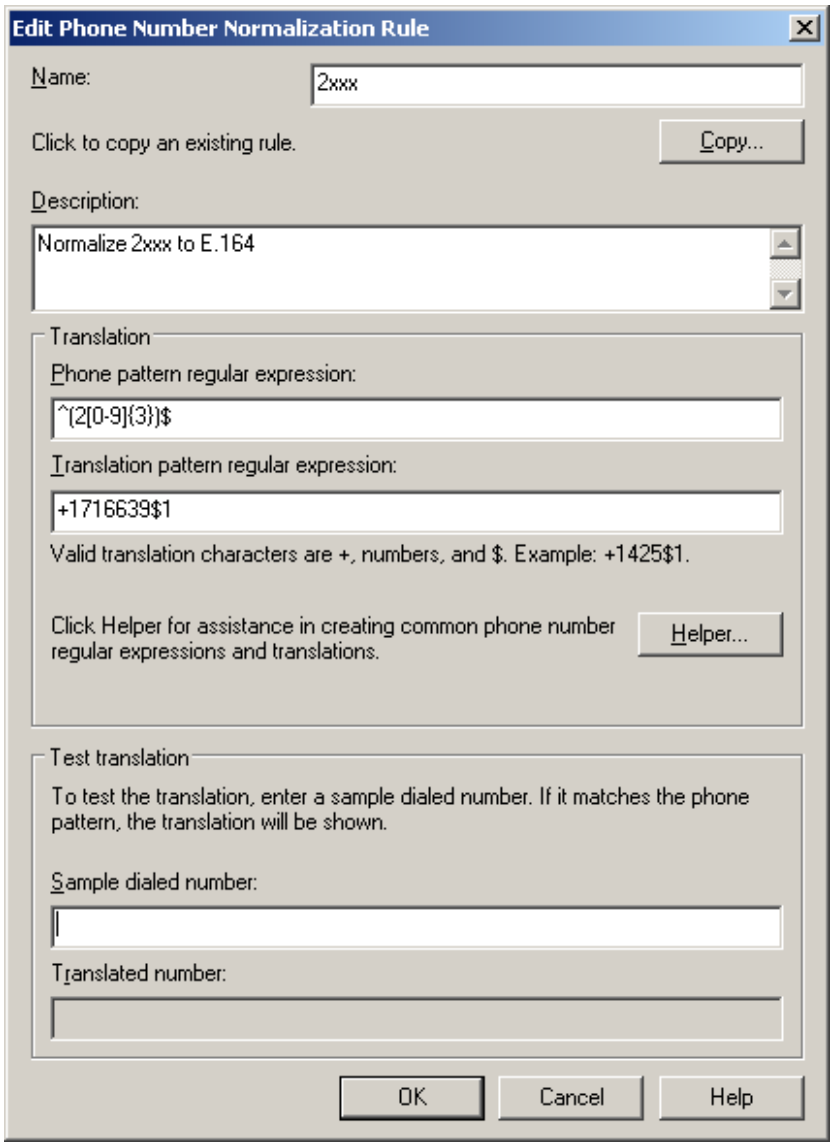
From the Start menu select the following to configure the OCS server:

- Programs → Administrative Tools → OCS 2007

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



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



In this example, when a user dials any 4-digit number starting with 2, 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:

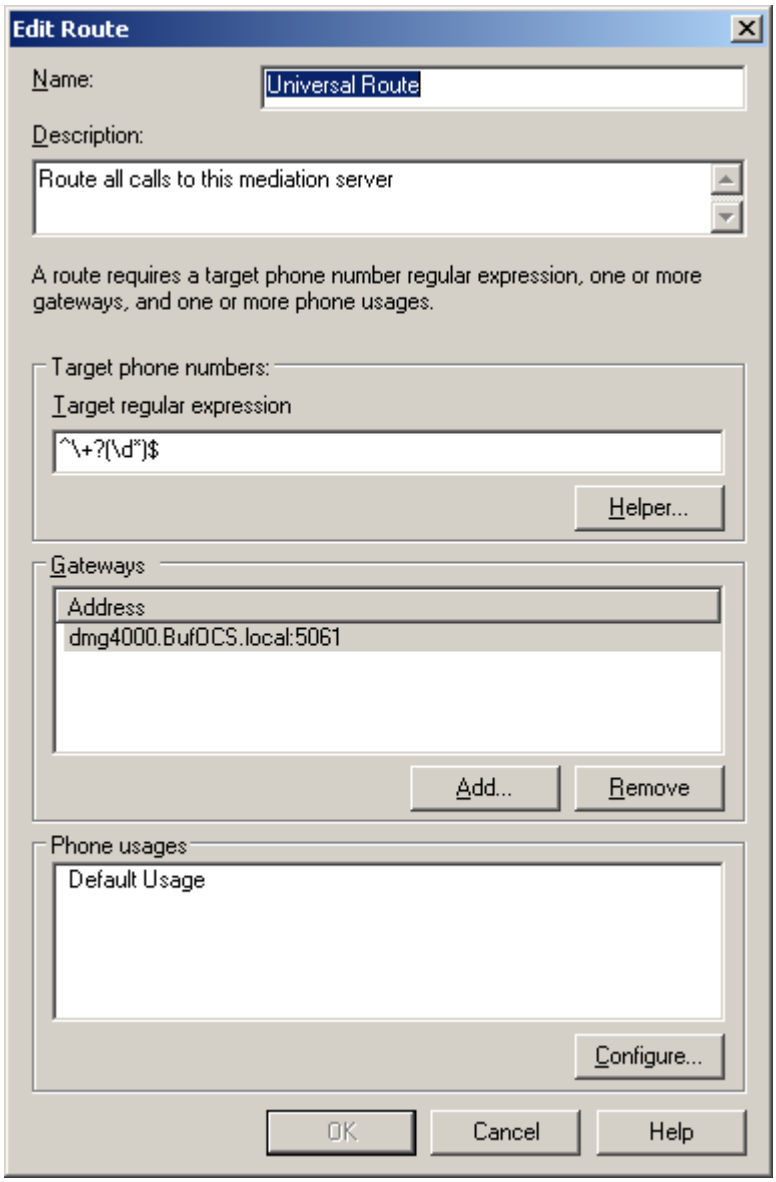
Name	Phone Pattern	Translation Pattern	Descriptions
2xxx	^(2[0-9]{3})$	+1716639$1	Normalize 2xxx to E.164
Local	^(\d{7})$	+1716$1	Local number
National	^1(\d*)$	+1$1	Long distance number
International	^011(\d*)	+011$1	International number

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

From the Start menu select the following to configure the OCS server:

- Programs → Administrative Tools → OCS 2007

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



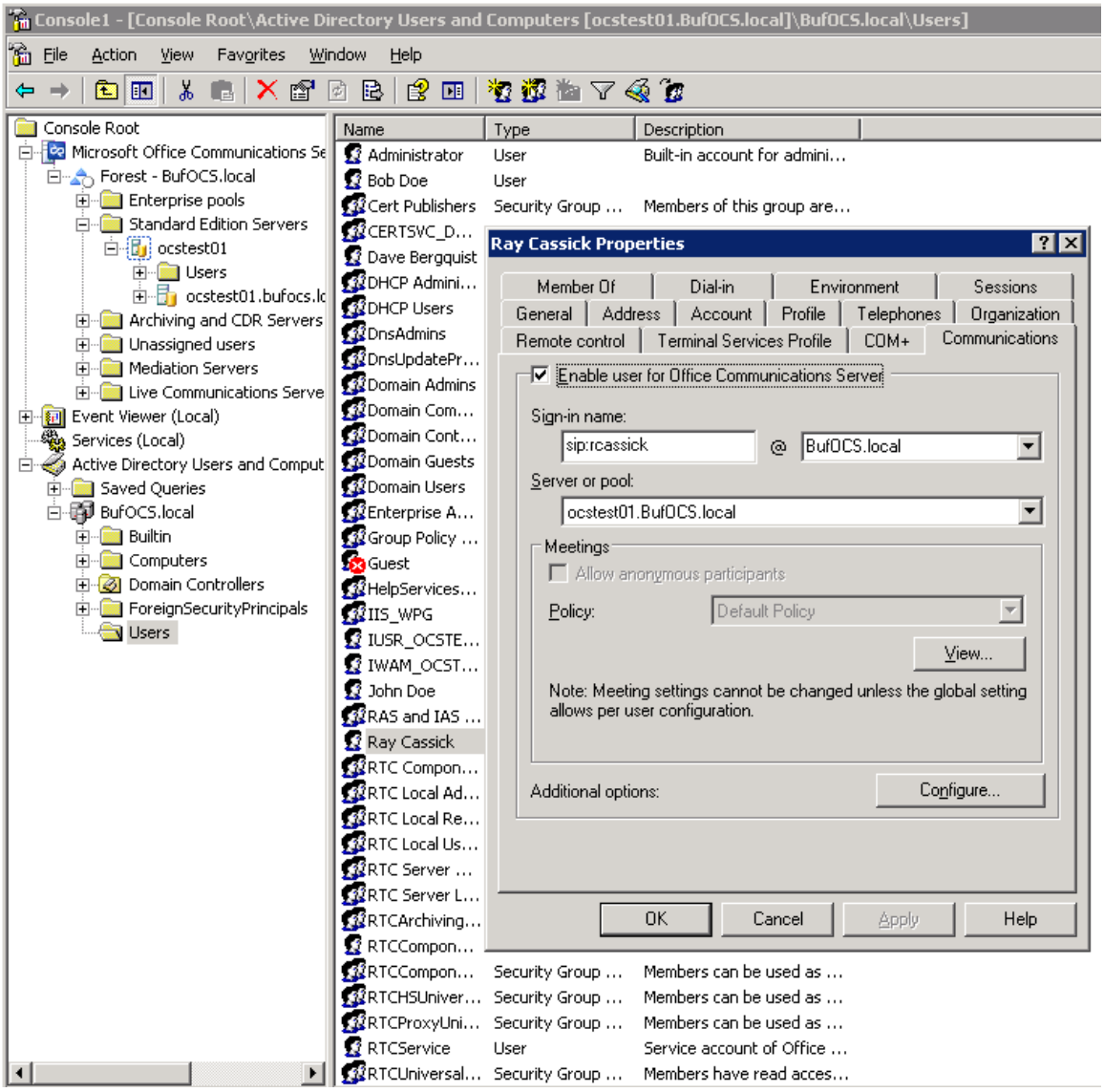
This entry routes any number with or without '+' prefix followed by any digits to Mediation server dmg4000.bufocs.local

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

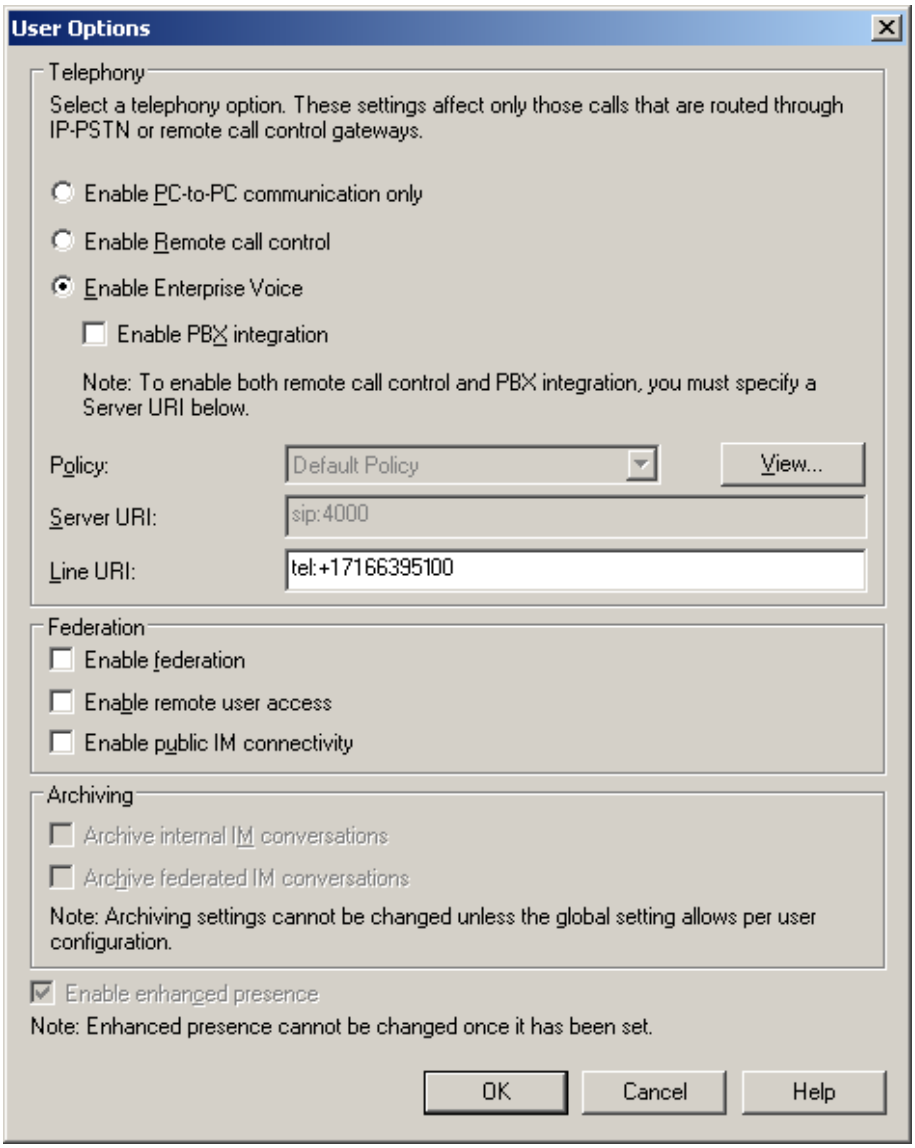
Note: Unless the dialed number from OCS client (such as Office Communicator) is in E.164 format, OCS must find a normalization rule to convert the dialed number to E.164.

7.2 Steps for Configuring OCS Clients

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



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



In the above configuration for user Ray Cassick, when an inbound PSTN call for 5100, it will be converted by the gateway CPID manipulation and routing rules into +17166395100. OCS will match that number provided by the gateway to the `Line URI` parameter for this user and ring Ray Cassick if he is logged on to OCS from Office Communicator or any OCS supported device.

8. Testing Validation Matrix

The table below shows various test scenarios that are run as typical validation scenarios when the 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 complete 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 OCS client.	
2	Direct call from OCS client to TDM station set.	

9. Troubleshooting

9.1 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.

9.2 Important Gateway Trace Masks

These keys are helpful during all troubleshooting scenarios and should be considered keys to activate by default fro 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 all circuit-side activity of the emulated station set such as display updates, key presses, light transitions and hook state changes. This data is very 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 very important in the following scenarios:
    - Call control problems (dropped calls, failing transfers, etc…)
    - Integration problems (incorrect mailbox placement, missed auto-attendant greetings etc…)

- `Routingtable (all keys)` – This allows you to look inside the routing table engine and see how matching rules and CPID manipulation rules work with respect to your call. This data is very important in the following scenarios:
    - Call routing problem (reaching the incorrect OCS client or no client at all, etc…)

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.

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