

Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

NEC NEAX2400 IPX

with Dialogic® 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 NEC NEAX2400 IPX and Dialogic® 2000 Media Gateway Series (DMG2xxxDT1) 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

PBX Vendor	NEC
Model	NEAX2400 IPX
Software Version	Ver.17 Rel.03.46.001
Telephony Signaling	T1 QSIG
Additional Notes	N/A

1.2. VoIP Gateway

Gateway Vendor	Dialogic Corporation
Model	Dialogic® 2000 Media Gateway Series (DMG2xxxDT1)
Software Version	5.0.42
VoIP Protocol	SIP

1.3. Microsoft Exchange Server 2007 Unified Messaging

Version	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 connect to the PBX using T1 QSIG, you need the PA24PRTB-A ISDN T1 line card.

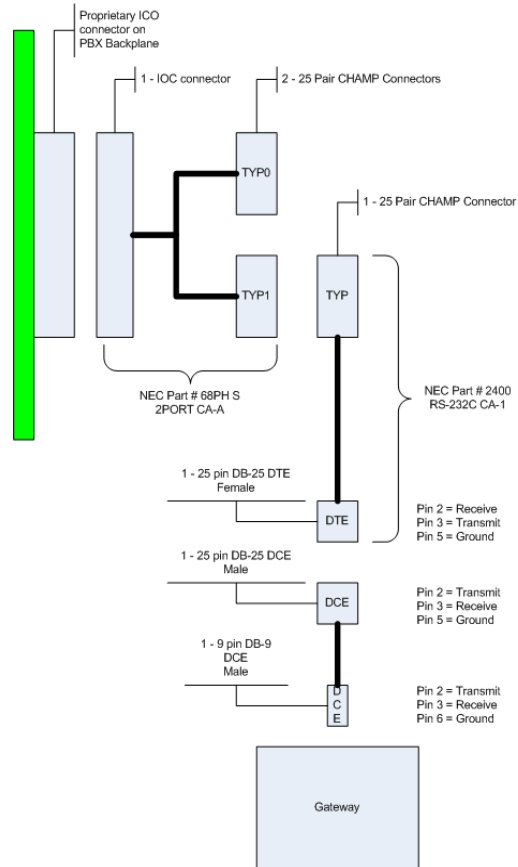
The table below shows the minimum firmware revision needed for specific line card models.

Card	Program	Minimum Revision	Function
PA-24DTIA	SP-3625	2A	Layer 1
PA-24PRTB	SP-3029	13A	Layer 1 & 2
PA-24DTR	SP-3010	15A	Layer 1
PA-24DTG-A	(SW-117) SP-716	(1L) 8A	Layer 1
PA-24DTG	SP-450	14A	Layer 1
PA-2DCHA	SP-3014	9A	Layer 2

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.

Even though the integration interface is using QSIG and the PBX supports QSIG supplemental services, MWIs are still done via the MCI serial interface.



The industry rule-of-thumb for RS232 serial communications is to keep the length of the cable no longer than 50 feet. Lengths longer than the recommended standard may need additional equipment to ensure correct transmission levels are maintained.

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.
- 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.

The NEC PBX, even though it has QSIG supplemental services available, still makes use of the MCI serial link for MWIs. You need to further configure the serial link on the gateway(s).

- Set the Serial Mode to Master for the single master gateway and to Slave for any of the additional slave gateways (if you are using any slaves).
- Ensure that the serial port settings (baud rate, parity and stop bits) match the port settings on the PBX.
- Set the Serial Interface Protocol to MCI.

5. PBX Setup Notes

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

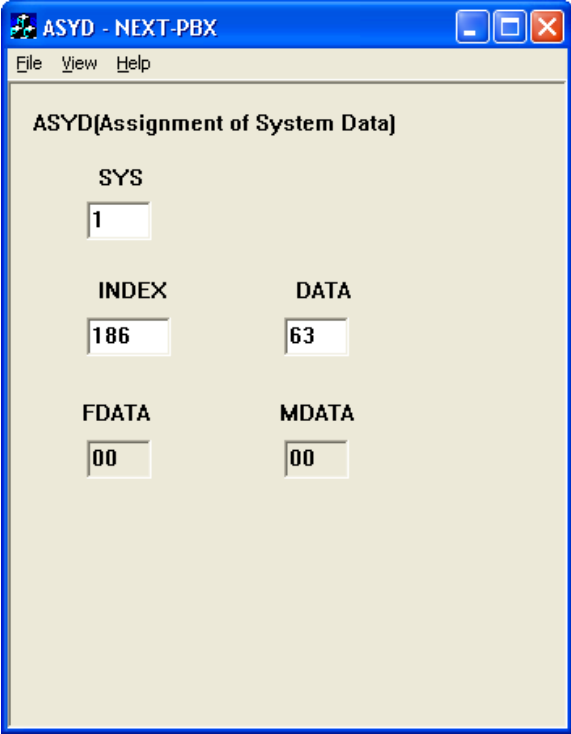
- Initializing the ISDN services.
- Enabling ANI pass-through.
- Building and configuring trunk routing.
- Setting up the subscriber station sets.

All PBX programming is done via a Windows-based application that connects to the PBXs serial administration port.

5.1. Initializing the ISDN Services

Using the ASYD (Assignment of System Data), set the system data bytes that are required to bring the ISDN system into service.

Note that not all fields will initially be visible. As you start to enter data new fields will appear.



The screenshot shows a Windows application window titled "ASYD - NEXT-PBX". The window has a menu bar with "File", "View", and "Help". The main content area is titled "ASYD(Assignment of System Data)". It contains several input fields for system data:

SYS	
<input type="text" value="1"/>	
INDEX	DATA
<input type="text" value="186"/>	<input type="text" value="63"/>
FDATA	MDATA
<input type="text" value="00"/>	<input type="text" value="00"/>

- In the ASYD Window:
 - In the SYS field, enter 1
 - in the INDEX field, enter 186
 - in the DATA field, enter 63
- Hit Enter to save and move on to Index 187
- In the ASYD Window in the DATA field, enter 00
- Hit Enter to save Index 187
- Close the ASYD Window.

5.2. Enabling ANI Pass-Through

Use the SFC (Assignment of Service Feature Restriction Class) command to allow ANI to pass from the PBX to TELCO.

Note that for this to work, SFI 94 must be set to 0. For extensions that want outward ANI to be blocked, assign them to an SFC that has SFI 94 set to a 1. Always assign SFI's 97 & 98 as 0.

D/N	TN	SFI
D	1	94

SFC	0	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
RES	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0

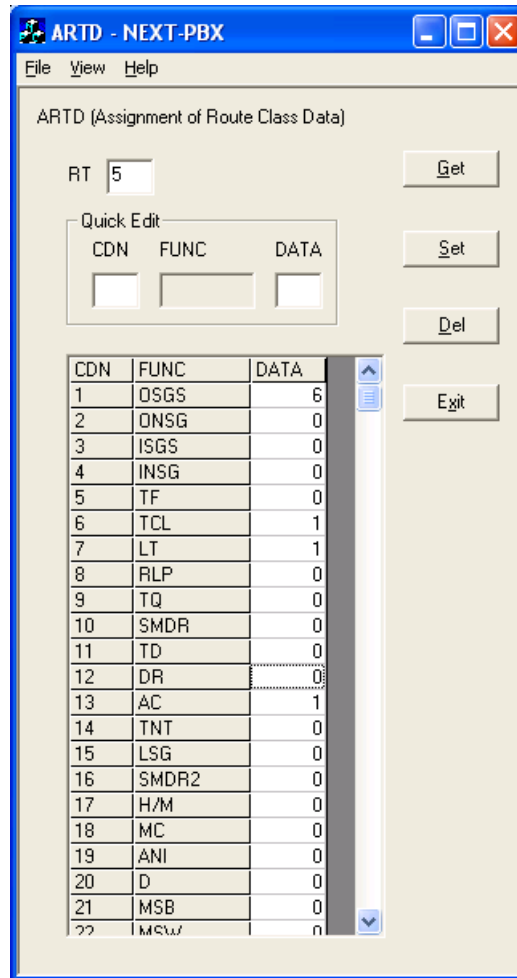
- In the ASFC Window:
 - In the D/N field, enter D (Day time configuration)
 - In the TN field, enter 1
 - In the SFI field, enter 94
 - In the RES fields, enter 0 for every field
- Close the ASFC Window.

If night time configuration is also required, enter N in the D/N field and repeat the steps above.

5.3. Building and Configuring Trunk Routing

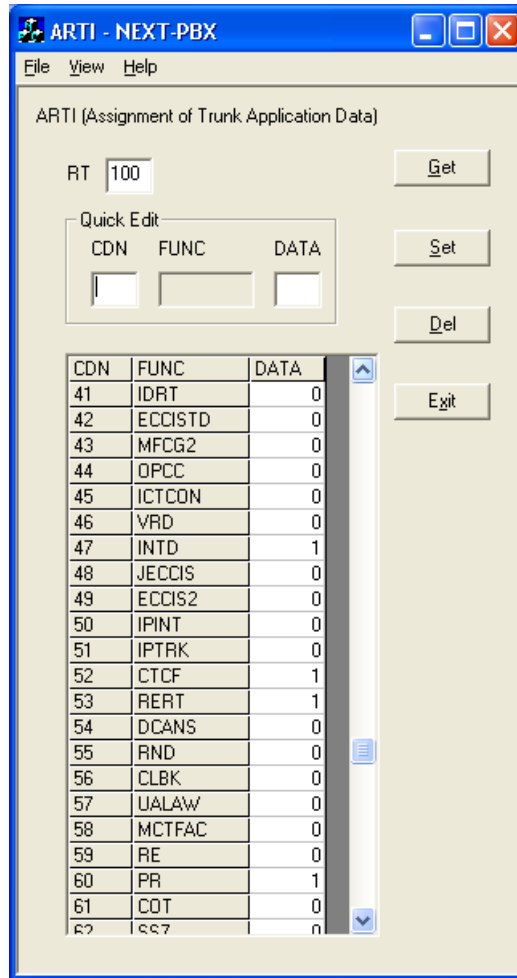
Use the ARTD (Assign Rout Data) command to assign route data for each B-Channel and D-Channel route.

For B-Channels, setting CDN to 1 allows INCOMING ANI to D-term displays (does not affect analog stations). This does NOT affect outgoing ANI. Other CDNs may be left default (data 0) for the route.



- In the ARTD Window under RT field, enter a number from 1–255 (Note: 31 is the unofficial convention. Also, remember this route number for latter steps in the configuration).
- In the ARTD Window hit the GET button to initiate the command.
- In the ARTD Window:
 - In the TCL field, enter 1
 - In the LT field, enter 1
 - In the AC field, enter 1
- In the ARTD Window hit the SET button to initiate the command.
- Close the ARTD Window.

Use the ARTI (Assignment of Trunk Application Data) command to configure your OSIG line for ISO standard protocol.

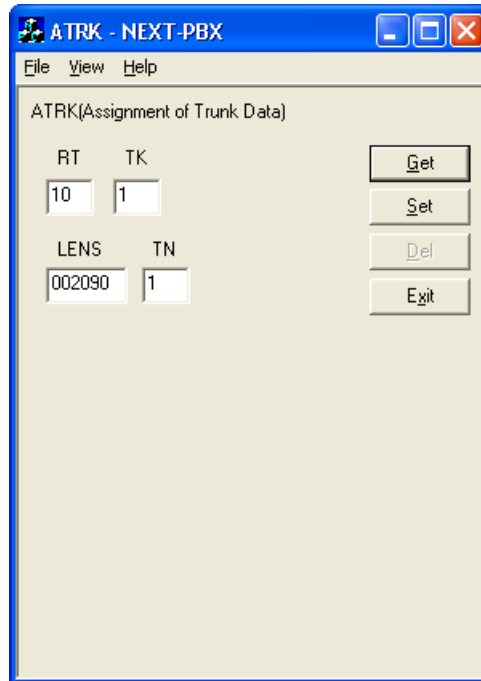


- In the ARTI Window under RT field, enter a B-Channel route number within the range of 1 through 255.
- In the ARTI Window hit the GET button to initiate the command.
- In the ARTI Window:
 - In the INTD field, enter 1
 - In the CTCF field, enter 1
 - In the RERT field, enter 1
 - In the PR field, enter 1
- In the ARTI Window hit the SET button to initiate the command.
- Close the ARTI Window.

All other CDNs may be left at there default (data 0) for the route.

Use the ATRK (Assignment of Trunk Data) command to assign trunks to the defined B-Channel and D-Channel routes.

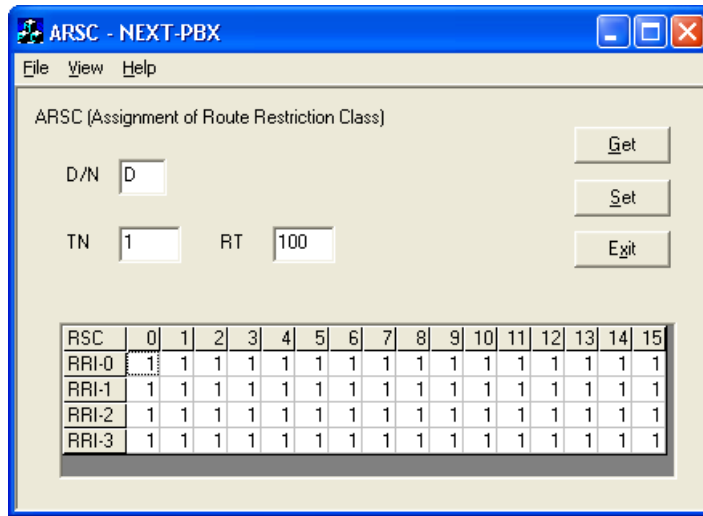
There should be two trunks programmed for the D-Channel route. The trunks assigned will depend on which circuit cards being used.



- In the ATRK Window:
 - in the RT field, enter the Route Number previously assigned in the ARTD step.
 - in the TK field, enter xxx where xxx is a trunk number used for calls going to and from the T1.
 - in the LENS field, enter xxxxxxx where xxxxxxx is the LENS (card address) number of the T1 card. a trunk number used for calls going to and from T1)
 - in the TN field, enter xxx where xxx is a tenant number used for calls going to and from the T1.
- In the ATRK Window hit the SET button to initiate the command.
- Close the ATRK Window.

Repeat these steps to assign a second trunk (TK) on the T1 card.

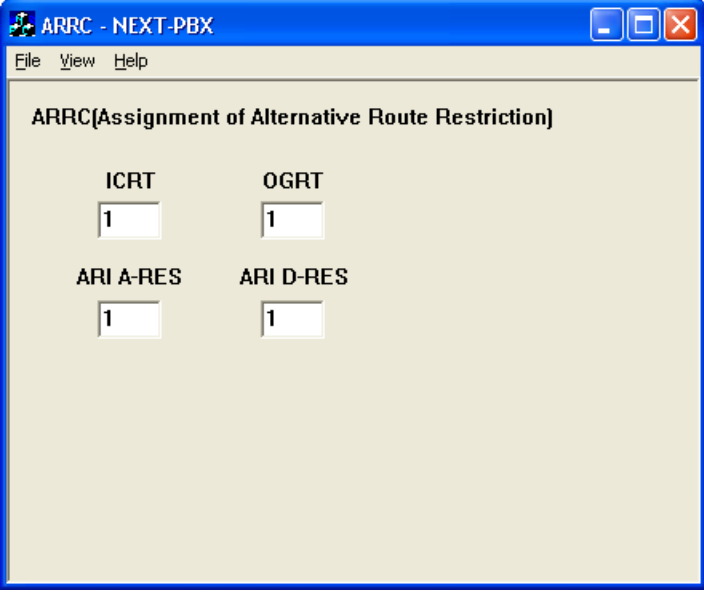
Use the ARSC (Assignment of Route Restriction Class) to assign route restriction data to allow, restrict, or toll restrict calls to the B-Channel route.



- In the ARSC Window
 - in the D/N field, enter D for Day time configuration
 - in the TN field, enter xxx where xxx is the tenant number used for calls going to and from the T1.
 - in the RT field, enter the Route Number previously assigned in the ARTD step.
 - in the RSC table, enter 1 for all calls
- In the ARSC Window hit the SET button to initiate the command.
- Close the ARSC Window.

If night time configuration is also required, enter N in the D/N field and repeat the steps above.

Use the ARRC (Assignment of Alternative Route Restriction) command to assign Route to Route Connection data to allow tandem connections between existing routes and the B-Channel route.



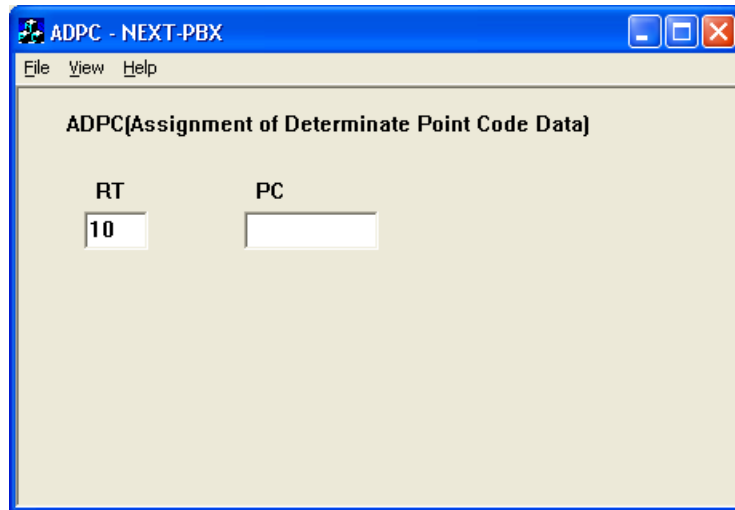
The screenshot shows a window titled "ARRC - NEXT-PBX" with a menu bar containing "File", "View", and "Help". The main area is titled "ARRC(Assignment of Alternative Route Restriction)". It contains four input fields arranged in a 2x2 grid:

ICRT	OGRT
1	1
ARI A-RES	ARI D-RES
1	1

- In the ARRC Window:
 - in the ICRT field, enter 1 (Incoming route number)
 - in the OGRT field, enter 1 (Outgoing route number)
- Close the ARRC Window.

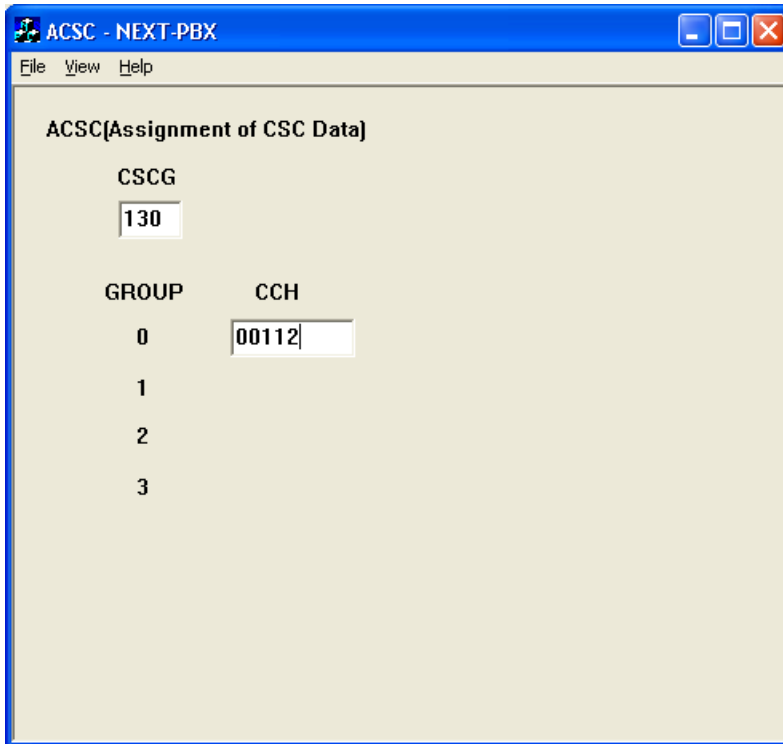
Use the ADPC (Assignment of Determinate Point Code Data) command to assign a separate and unique point code for each ISDN span.

The allowed range for point codes is 1 through 16383. Any number in that range can be used, however it is recommended to use a point code under 200. Also you can not use any point code already assigned in the PBX which is used for CCIS or ISDN. To find point codes which are in use, look in AYSD Index 180 & 181 as well as LDPC.



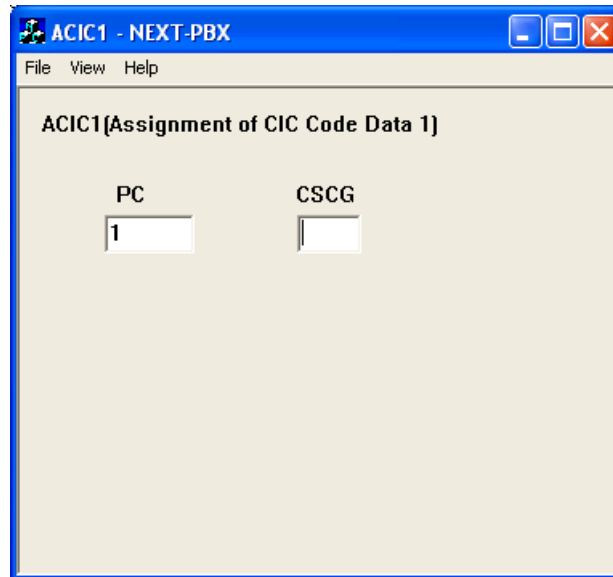
- In the ADPC Window
 - in the RT field, enter Route Number previously assigned in the ARTD step.
 - in the PC field, enter xxx where xxx is the point code you wish to use.
- Close the ARSC Window.

Use the ACSC (Assignment of CSC Data) to assign only an even CSCG number. If for any reason ACSC data must be deleted; first delete ACIC/ACIC1 data, then delete ACSC.



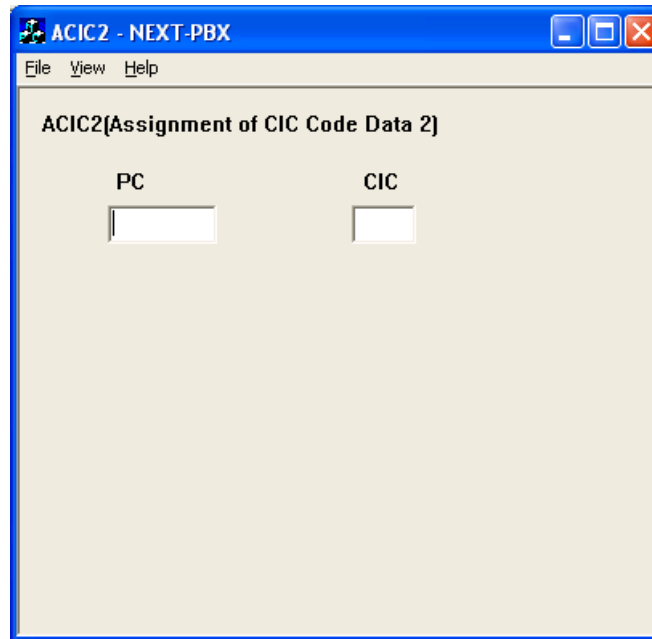
- In the ACSC Window:
 - in the CSCG field, enter xxx where xxx is an available Common Channel signaling controller group number from 2 through 255 (do not use 128 or 129)
 - in the CCH field, enter xxx where xxx is the slot address (MG, UNIT, and GROUP)
- Close the ARSC Window.

Use the ACIC1 (Assignment of CIC Code Data 1) command on IMX/IPX PBX's using Windows Commands (ACIC1 / ACIC2)



- In the ACIC1 Window:
 - in the PC field, enter xxxx where xxxx is point code for the ISDN span that was assigned in ADPC.
 - in the CSCG field, enter xxxx where xxxx is an even number that was assigned in ACSC.
- Close the ACIC1 Window.

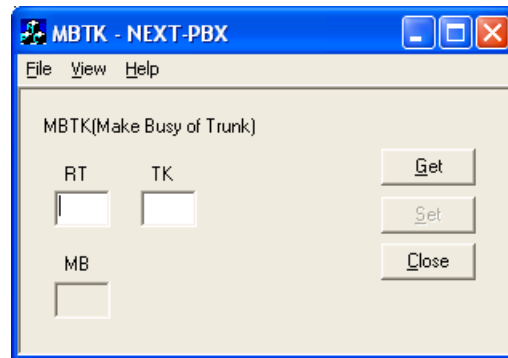
Use the ACIC2 (Assignment of CIC Code Data 2) command on IMX/IPX PBX's using Windows Commands (ACIC1 / ACIC2)



- In the ACIC2 Window
 - in the PC field, enter xxxx where xxxx is a point code for the ISDN span that was assigned in ADPC.
 - in the CIC field, enter 1.
 - in the LENS field, enter xxxx where xxxx is the first B-Channel Trunk assigned in ATRK.
- Close the ACIC2 Window.

Note: Repeat the ACIC2 steps for every circuit identification code (CIC) and its corresponding line equipment number (LENS). Make sure you only assign a CIC to a B-Channel trunk.

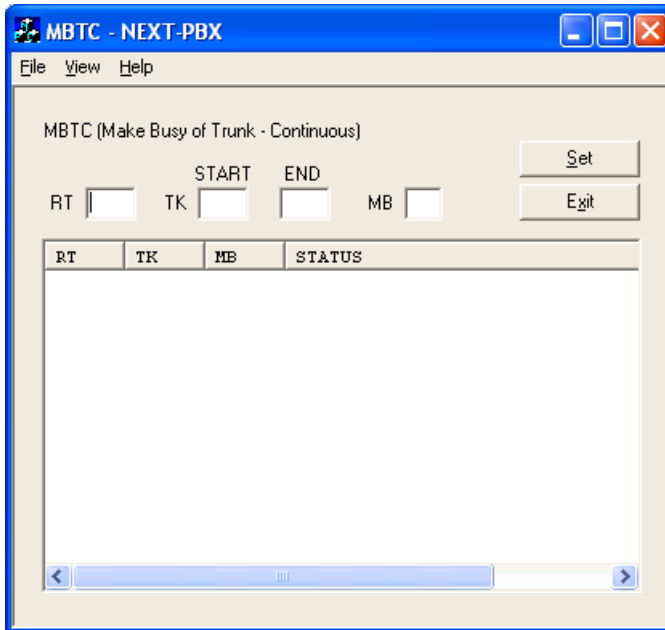
Use the MBTK (Make Busy of Trunk) command to set the B-Channels into an idle state.



- In the MBTK Window
 - in RT field, enter xxxx where xxxx is a route number entered in the ARTD step.
 - in TK field, enter xxxx where xxxx is a trunk number entered in the ATRK step.
 - in MB field, enter 0, where 0 sets the trunk to idle on the T1.
- In the MBTK Window hit SET, to initiate the command.
- Close the MBTK Window.

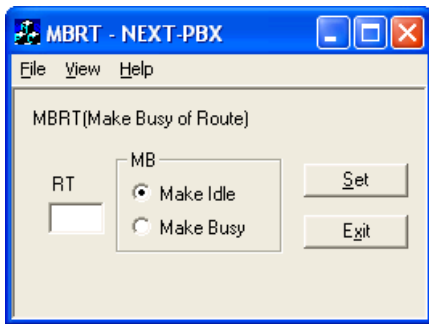
Repeat this step for each for each trunk used.

Use the MBTC (Make Busy of Trunk - Continuous) command to set all the B-Channels idle.



- In the MBTC Window:
 - In the RT field, enter xxxx where xxxx is a route number entered in the ARTD step.
 - in the TK START field, enter xxxx where xxxx is the first trunk number entered in the ATRK step.
 - in the TK END field, enter xxxx where xxxx is the last trunk number entered in the ATRK step.
 - in the MB field, enter 0, where 0 sets the trunk to idle on T1.
- In the MBTC Window hit SET, to initiate the command.
- Close the MBTC Window.

Use the MBRT (Make Busy or Route) command to set the B-Channels idle.



- In the MBTC Window:
 - in the RT field, enter xxxx where xxxx is a route number entered in the ARTD step.
 - in the MB field, select Make Idle radio button to set the trunk to idle on T1.
- Close the MBTC Window.

The next step to perform is to verify that the D-Channel is stable. The connecting PBX should see the D-Channel linked and at this time the B-Channels should come online (this is sometimes referred to as B-Channel Restarts) and test calls can be made on any Incoming routes.

The next task is to set up to Least Cost Routing rules. It is suggested that you see the NEAX 2400 ISDN Installation manual, or contact local NEC vendor representative for proper configuration, as many of these parameters are site specific.

Once you have completed the LCR setup it is time to program ANI "Caller ID" to assign caller id data for outgoing calls. It is suggested that you see the NEC2400 ISDN Installation manual, or contact local NEC vendor representative for proper configuration, as many parameters are site specific.

5.4. Setting Up Subscriber Station Sets

There is no PBX-side programming for setting up the subscriber station sets. All the forwarding of the user's phones is defined on the users station set directly. Users should be direction to set their internal and external ring no answer and busy forwarding conditions to an LCR number assigned to direct calls to the defined trunks.

For phones that need to have their MWI lights set and cleared, you need to enable SFI 74 and SFI 82.

5.5. 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.	P	

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	
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	
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.	F	Testing showed that the gateway is getting a connection message from the local PBX and this is causing the application to start playing the outgoing message too quickly.
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	<p>Send a test FAX message to user extension.</p> <p>Confirm the FAX is received in the user's inbox.</p>	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 MWI 2007 website.</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

Failure Point	7b
Phone type (if phone-specific)	
Call scenarios(s) associated with failure point	Play to phone feature to an external number.
List of UM features affected by failure point	Only play to phone to an external number.
Additional Comments The gateway is getting an ISDN connect message form the local PBX before the remote party even picks up the call. This causes the gateway to signal the application that the call has gone connected causing it to start playing the message before a party has actually answered the call.	

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

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 will not harm system performance.

- `dspif` (all keys) – This allows the collection of tone-related data. This data is very 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 issues this key may be left disabled.
- `si` – This allows the collection of all inbound and outbound serial data on the serial master gateway. This data is required in the troubleshooting of all integration-related issues seen on the gateway designated as the serial master.
- `siip` – This allows the collection of all inbound and outbound serial data on any of the serial slave gateways. This data is required in the troubleshooting of all integration-related issues seen on the gateway designated as a serial slave.
- `simwi` – This allows the collection of all activity reacted to the processing of MWIs using the serial port. This data is required for any MWI-related issues while using the serial interface.

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.