

Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

NEC NEAX 2400 IMX

with Dialogic® 2000 Media Gateway Series

(DMG2xxxDTI) using T1 CAS MCI Serial Integration

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 NEAX 2400 IMX and Dialogic® 2000 Media Gateway Series (DMG2xxxDTI) using T1 CAS MCI Serial Integration 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	NEAX 2400 IMX
Software Version	Version 7400
Telephony Signaling	T1 CAS MCI Serial Integration
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

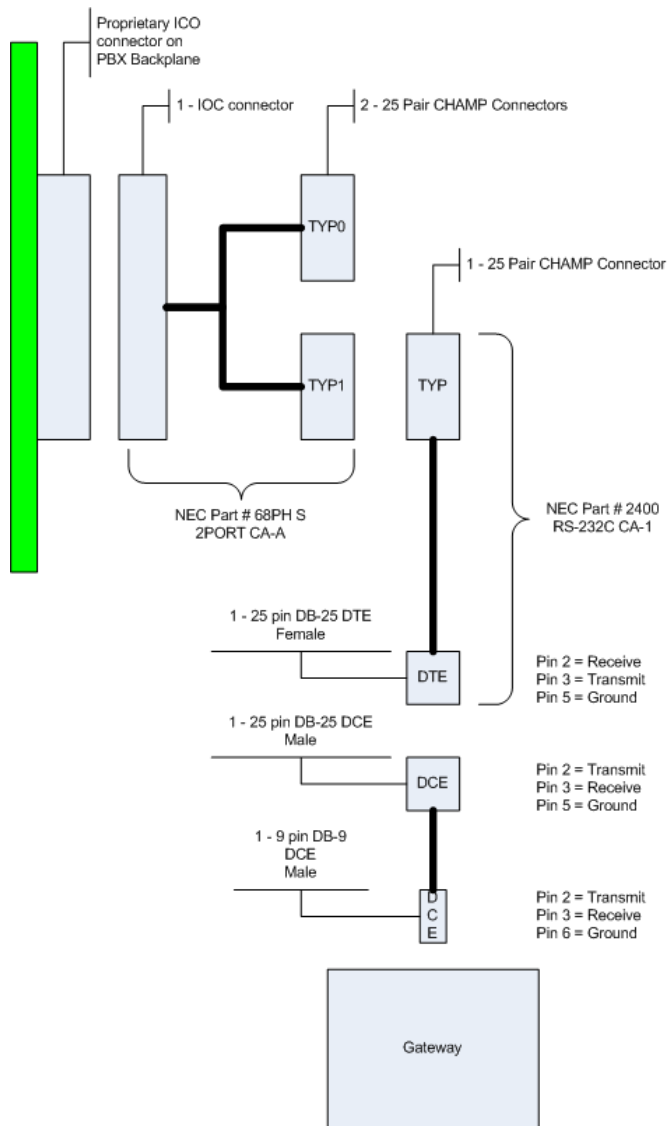
The gateway needs to support a T1 CAS robbed bit interface.

2.2. PBX Requirements

To support the T1 CAS configuration as documented, you need the SPA-24DLI Lineside T1 interface card.

2.3. Cabling Requirements

No special cabling is required. A standard T1 cable is used. It is recommended that a high quality cable be used for long loop lengths or loops that will be run through electronically noisy environments. Because the integration data is transmitted along a secondary serial data path between the PBX and the gateway, it will be required to either purchase or construct a cable that connects the serial port of the Toshiba PBX to the serial port of the gateway. The diagram below provides details on how this cable was constructed for the creation of this document.



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 CAS - Loop start.

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 messaging 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.
- Set the CAS Cpid type to TypeII_CPID.
- 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.
- 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 server are as follows:

- Programming system-wide settings.
- Setting up each gateway port.
- Defining a hunt group to act as a central point for incoming calls to the gateway and a coverage path for subscriber sets to use as a forwarding point.
- Setting up the subscriber stations.

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

5.1. Programming System-wide Settings

For the PBX to interact properly with the MCI link with respect to call data and MWI activity you must set certain specific system level parameters. The programming details below give you specifics on what needs to be enabled for this interaction to work properly.

System Data

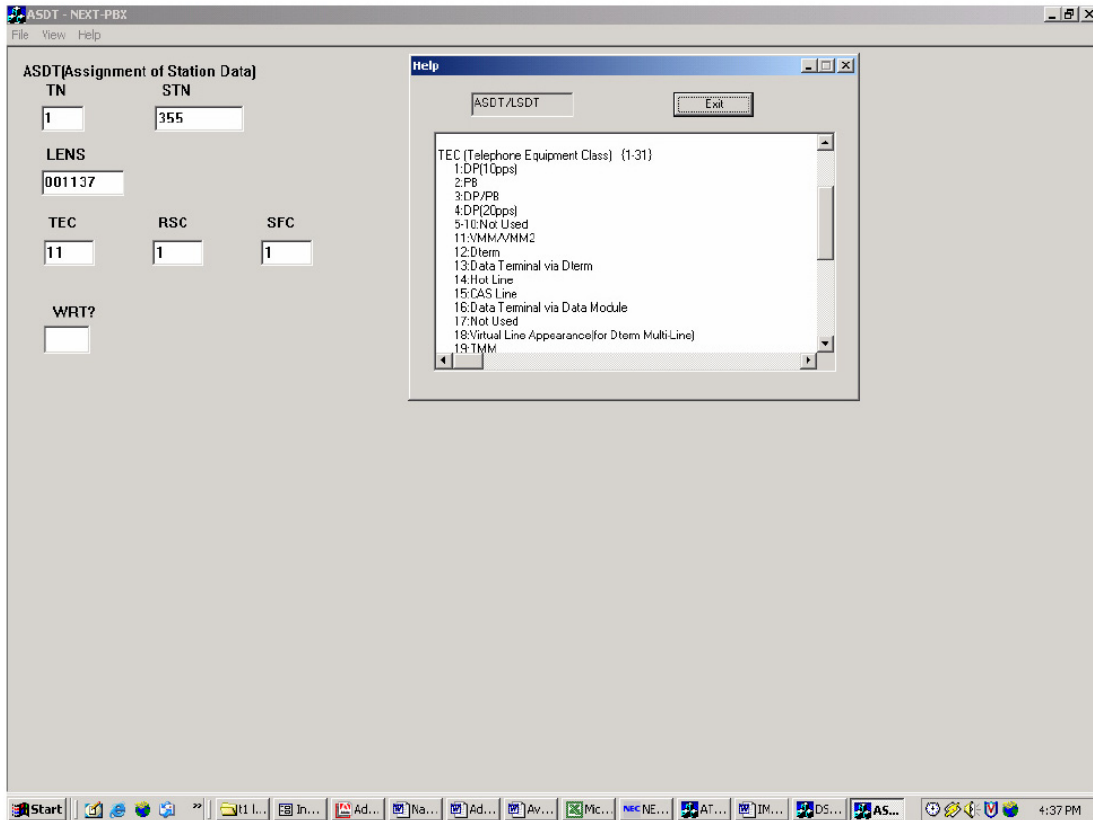
System	Index	Bit	Value	Description
1	17	b4	1	Release (blind) transfer to attendant console
	28	b0-4	0	Guard timer no required
		b5	1	MWI controlled by MCI
	29	b1-7	0/1	No/Yes: Assign I/O port fro MCI ouptut Port 1 = b1, port 2 = b2, etc.
	34	b1-4	0	Set output to no parity and 1 stop bit
	60	b3	0	UCD queuing required
	63	b0	1	Release transfer for stations in service
	69	b0	1	No recall, execute call forwarding on no answer
	70	b0	1	Called number display, when forwarding to attendant console
	77	b2	0	MWI refresh required
	78	b0	1	Calling number display enabled
		b1	1	Calling station status display enabled
	238	b0-7	0	Lamp flash rate
	246	b3	0	MCI expansion set to normal
400	b2	1	Calling number information sent to MCI	
2	6	b0	1	MCI In service when terminating to a UCD group

Programming System Data Local Data

Table 7 Programming System Data Local Data				
System	Index	Bit	Value	Description
1	641	b1	0/1	0/1: MCI/IMX station number/phone number
	832	b0-7	00 FD	Assign the FPC of the node connected MC
	833	b0	0	MWI controlled by MCI

5.2. Setting Up Each Gateway Station Port

Use the Assignment of Station Data command to create as many station ports as required for your configuration (24 ports per span). The example below shows performing this command on one station.

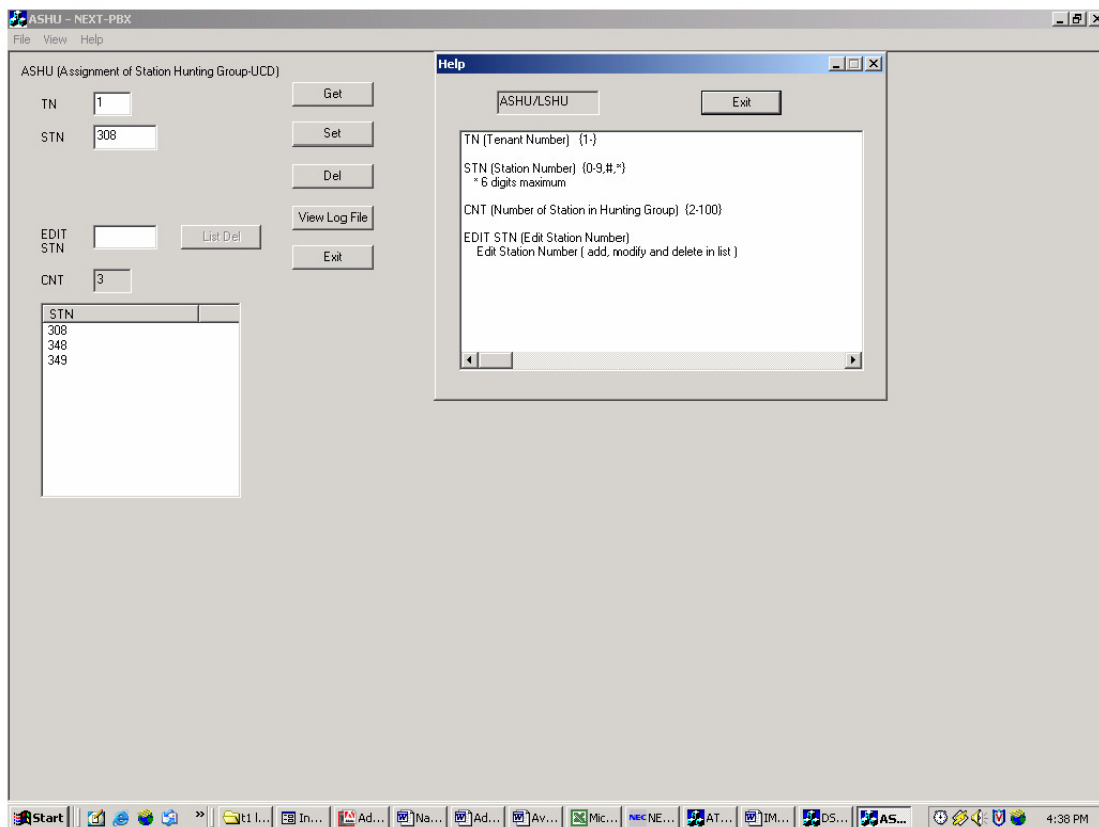


Important notes about the above programming:

1. The STN field assigns an extension to the T1 bearer channel connected to this station set definition.
2. The TN and LENS fields together specify the physical wiring address of each bearer channel on a T1 interface.
3. The TEC should be set to 11 and this specifies the port is connected to a voice mail device. This enables the integration via the MCI serial link.

5.3. Setting Up Hunt Group and Coverage Path

Use the Assignment of Station Hunting Group - UCD or Assignment of Station Hunting Group - Circular commands to define a hunt group to contain all the defined gateway ports. Depending on what command you select will have an impact on the hunting pattern you will get, either linear or round robin style. The programming example below is for a UCD style hunt group.



Important notes about the above programming:

1. The STN field specifies the extension assigned to the hunt group and should be used as the forwarding destination for all subscribers as well as the inbound call destination for accessing the server.

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 subscriber phones is defined on the subscriber station set directly. The subscriber should be directed to set their internal and external ring no answer and busy forwarding conditions to the Pilot Number setting defined to the hunt group.

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.	P	
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	P	

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

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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.
- Microsoft HyperTerminal – Used to test the output of the PBXs serial port to determine if data is being sent by the PBX. This helps in the validation of the cabling and all connects on the serial interface between the PBX and the gateway.
- RS232 breakout box – This tool can be very valuable in doing serial cabling work. It helps you determine what signals are available and if data is being either transmitted or received.

7.2. Important Gateway Trace Masks

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

- `voip prot` and `voip code` – this allows the capture 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 server.
- `tel event` and `tel code` – This allows the capture 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...)

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.

- `dspcpi` (all keys) – This allows the capture of tone-related audio 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. This data is very important in the following scenarios:
 - Failing transfers
 - Failing outbound calls (play to phone)
 - Dropped calls (callers cut off while leaving messages, etc...)
- `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.

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.