

Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

Nortel Option 11c

with Dialogic® 1000 Media Gateway Series

(DMG1008DNI) using Digital Set Emulation

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READ THIS BEFORE YOU PROCEED

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Content

This document describes the configuration required to setup Nortel Option 11c and Dialogic® 1000 Media Gateway Series (DMG1008DNI) using digital set emulation 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	Nortel
Model	Option 11c
Software Version	Release 25
Telephony Signaling	Digital set emulation
Additional Notes	N/A

1.2. VoIP Gateway

Gateway Vendor	Dialogic Corporation
Model	Dialogic® 1000 Media Gateway Series (DMG1008DNI)
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 m2616 digital station set emulation.

2.2. PBX Requirements

PBX needs to have the following software packages installed:

- #19 (DDSP) Digit Display Software
- #46 (MWC) Message waiting software (only required to perform MWIs)

To support the 2-wire station interface as documented, you need one of the following PBX line card types:

- NT8D02xx (where xx = revision) – 16 port card
- NTDK16 - 48 port version built directly into the Option 11c mini

NOTES:

- Dialogic has been able to certify a wide range of NT8D02 revisions including EA and GA.
- Option 11c mini PBXs come with a 48 port digital card built-in. Additional digital ports can be added via standard NT8D02 type line cards.
- Nortel PBXs ranging from the Option 11 up to the Option 81 all use the NT8D02 type line cards.

2.3. Cabling Requirements

It is recommended that total loop length (cable distance between PBX connection and the gateway interface) be no longer than 3000 feet (915 meters) and no shorter than 6 feet (2 meters).

The Nortel digital set interface on the gateway is polarity sensitive.

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.
- Configure the gateway for the M1 integration.

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 messaging server. If multiple IP endpoints are to be used, then ensure they are configured as well.
- Activate fault tolerance and load balancing as required by the application and system requirements.
- Specify the required audio coders as required by the application.
- Set the hunt group extension number to the extension number you will be using for your primary DN in the PBX programming.
- 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 application are as follows:

- Validating required PBX software packages.
- Setting up each gateway station port.
- Setting up subscriber station sets.

The basic commands that you will encounter on the PBX to perform these actions are:

- LD10; LD11; LD20; LD22

5.1. Validating Required PBX Software Packages

Use the LD22 command to print out a listing of the all the installed software packages on the PBX. Once the listing scrolls by, review it to ensure that all the required packages are included. The example below shows the output of this command (truncated for brevity) and what to look for to locate if your PBX has the proper software loaded.

```
>LD                               22
REQ                               PRT
TYPE                              PKG
OPTF
CDR
CTY
DISA
NCOS
DDSP
MWC
```

Make note of the DDSP and MWC packages in the list. These are the require packages you must have. If you do not have these packages, contact your PBX vendor to have them installed.

5.2. Setting Up Each Gateway Station Port

Use the LD11 command to configure as many station ports as required for your configuration (8 ports per gateway). The example below shows performing this command on one station.

```
>LD 11
REQ  NEW
TYPE 2616
TN   6 0
DES
CUST 0
AOM
FDN
TGAR
NCOS
```

```
RNPG
SSU
CLS  ADD CNDA
HUNT 5001
LHK  0
KEY  00 SCR 5000
KEY  13 MIK
KEY  14 MCK
KEY  15 TRN
```

Important notes about the above programming:

1. The REQ field is where you specify the command for this overlay. You can choose to build a new station set using the NEW command or make changes to an existing station using the CHG command.
2. The TYPE field is critical as it specifies the station type configured for this sport. This MUST be set to 2616.
3. The CLS field is required to have both ADD and CNDA configured. ADD forces the display of the station to update as soon as the call rings to it, and CNDA forces the PBX to make sure that reason codes are present in the display when the call arrives.
4. The first key on the station set (KEY 00) must be configured as a SCR (Single Call Ringing) type.
5. The HUNT parameter is used to direct this station to forward to the next station in line under busy conditions. This is the setting that controls the grouping together of station sets in a hunting pattern for call distribution. This example shows station 5000 hunting to station 5001. Station 5001 would be configured to hunt to 5002 and so forth until your hunting pattern is completed at the last port to be built.
6. KEY 13 and KEY 14 must be set to MIK and MCK as shown for MWI functionality to operate properly.
7. KEY 15 must be configured to TRN (transfer) for transfers to operate properly.

5.3. Setting Up Subscriber Station Sets

This is an example of how to set up a subscriber that uses a digital station set to forward correctly to the voice messaging server. Use the LD11 command to change the stations parameters as shown below.

```
>LD 11
REQ  CHG
TYPE 2008
TN   0 1 8 3
ECHG
DES
FDN  5000
TGAR
HUNT 5000
NCOS
RNPG
```

```
SSU
CLS   HTA FNA MWA CFTA SFA
EFD   5000
EHT   5000
.
.
.
```

Important notes about the above programming:

1. The FDN field is where you specify the destination for this station set to forward to under ring no answer conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
2. The CLS field is required to have HTA, FNA, MWA, CFTA and SFA configured. If these are not configured properly, the remainder of the programming is not going to provide you with the proper prompts to continue.
3. The EFD field (only seen if the CLS has been set up properly) is where you specify the destination for external calls to the station to forward under ring no answer conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
4. The HUNT field is where you specify the destination for internal calls to the station to forward under busy conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
5. The EHT field (only seen if the CLS has been set up properly) is where you specify the destination for external calls to the station to forward under busy conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
6. The MWA setting in the CLS field allows this station to make use of its MWI light. If this is not configured, the stations MWI lamp will not work.

This is an example of how to set up a subscriber that uses an analog station set to forward correctly to the voice messaging server. Use the LD10 command to change the stations parameters as shown below.

```
>LD 10
REQ   CHG
TYPE  500
TN    0 0 7 1
CDEN
DES
FDN   5000
CUST
DIG
DN
HUNT  5000
TGAR
NCOS
RNPG
CLS   HTA FNA MWA LPA CFTA SFA
```

FTR

EFD 5000

EHT 5000

.

.

.

Important notes about the above programming:

1. The FDN field is where you specify the destination for this station set to forward to under ring no answer conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
2. The CLS field is required to have HTA, FNA, MWA, LPA, CFTA and SFA configured. If these are not configured properly, the remainder of the programming is not going to provide you with the correct prompts to continue.
3. The EFD field (only seen if the CLS has been set up properly) is where you specify the destination for external calls to the station to forward under ring no answer conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
4. The HUNT field is where you specify the destination for internal calls to the station to forward under busy conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
5. The EHT field (only seen if the CLS has been set up properly) is where you specify the destination for external calls to the station to forward under busy conditions. It should be configured to send the calls to the primary dn, the first configured gateway port.
6. The MWA setting in the CLS field allows this station to make use of the MWI feature.
7. The LPA setting in the CLS field controls the phones MWI notification method. On analog stations with a neon MWI lamp, this setting must be included to use it. Without this setting, the analog station will only have stutter dial tone as its notification method.

5.4. 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.	F	The caller is routed to the incorrect prompt "Please select a name..." During the test the PBX was not following the defined busy forwarding and this was causing the transfer to fail with a busy tone. Test failed due to a PBX problem.
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.	F	The caller is routed to the incorrect prompt "Please select a name..." The Nortel provides standard busy tone for invalid extensions and Exchange UM did not handle the 'Busy here' failure due to the busy tone properly.
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	<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	PBX was not following the busy forwarding that was configured.
Phone type (if phone-specific)	
Call scenarios(s) associated with failure point	6b
List of UM features affected by failure point	Transfers to busy stations via AA.
<p>Additional Comments The caller is routed to the incorrect prompt "Please select a name..."</p> <p>Durring the test the PBX was not following the defined busy forwarding and this was causing the transfer to fail with a busy tone. Test failed due to a PBX problem.</p>	

Failure Point	Exchange UM not properly handling the 'busy here' return from the call attempt.
Phone type (if phone-specific)	
Call scenarios(s) associated with failure point	6d
List of UM features affected by failure point	Outbound calls (Play to Phone) to invalid numbers.
<p>Additional Comments The caller is routed to the incorrect prompt "Please select a name..."</p> <p>The Nortel provides standard busy tone for invalid extensions and Exchange UM did not properly handle the 'Busy here' failure due to the busy tone.</p>	

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...)

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

- `dsppci` (all keys) – This allows the collection of tone-related data. This data is helpful in cases where you think you have problems detecting specific tones that should be, should not be, or are expected to be present at specific times during the call. If you do not suspect a tone-related issue, this key may be left disabled. This data is important in the following scenarios:
 - Failing transfers
 - Failing outbound calls (play to phone)
 - Dropped calls (callers cut off while leaving messages, etc...)
- `adept` (all keys) – This allows the collection of rule-parsing data. This data is required in the troubleshooting of integration-related issues.

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.