

Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

Avaya Definity G3

with Dialogic® 1000 Media Gateway Series

(DMG1008DNI) using Digital Set Emulation

By : Dialogic

Updated Since : 12/19/2007

READ THIS BEFORE YOU PROCEED

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Content

This document describes the configuration required to setup Avaya Definity G3 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	Avaya
Model	Definity G3
Software Version	R009i.05.122.4
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 8434 digital station set emulation.

2.2. PBX Requirements

Configuration requires that the PBX support the 7434ND station set type. The 8434 station set can be used instead, but the 7434ND set type is comparatively preferred.

NOTE: The hardware on the gateway physically emulates a 2-wire 8434 station set but during the PBX configuration it is more preferred to set the station type for 7434ND as detailed later in this document.

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

- TN2181 (16 port)
- TN2224 (24 port)

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).

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 Avaya 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 hunt group in the PBX programming.
- Configure the SIP Transport for TCP.

5. PBX Setup Notes

The basic steps of setting up the PBX are as follows:

- Setting up each gateway station port.
- Defining 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 subscriber station sets.

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

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

5.1. Setting Up Each Gateway Station Port

Use the `change station` or `add station` 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.

```
change station 501                                     Page 1 of 5
                                                    STATION
Extension: 501                                         Lock Messages? n
  Type: 7434ND                                         Security Code:      TN: 1
  Port: 01A0501                                       Coverage Path 1:   COR: 1
  Name: VoiceMail                                       Coverage Path 2:   COS: 1
                                                    Hunt-to Station:
STATION OPTIONS
  Loss Group: 2                                       Personalized Ringing Pattern: 1
  Data Module? n                                       Message Lamp Ext: 575
  Display Module? y
  Display Language: english                           Coverage Module? n
```

```
change station 501                                     Page 2 of 5
                                                    STATION
FEATURE OPTIONS
  LWC Reception: msa-spe                               Auto Select Any Idle Appearance? n
  LWC Activation? y                                   Coverage Msg Retrieval? y
  LWC Log External Calls? n                           Auto Answer: none
  CDR Privacy? n                                       Data Restriction? n
  Redirect Notification? y                             Idle Appearance Preference? n
  Per Button Ring Control? n
  Bridged Call Alerting? n                             Restrict Last Appearance? y
  Active Station Ringing: single
```

H.320 Conversion? n

Per Station CPN - Send Calling Number? y

Audible Message Waiting? n

Display Client Redirection? y

Select Last Used Appearance? n

Coverage After Forwarding? S

change station 501

Page 3 of 5

STATION

SITE DATA

Room:	Headset? n
Jack:	Speaker? n
Cable:	Mounting: d
Floor:	Cord Length: 0
Building:	Set Color:

ABBREVIATED DIALING

List1:	List2:	List3:
--------	--------	--------

BUTTON ASSIGNMENTS

1: call-appr	6:
2: call-appr	7:
3:	8:
4:	9: lwc-store
5:	10: lwc-cancel

change station 501

Page 4 of 5

STATION

FEATURE BUTTON ASSIGNMENTS

1:	13:
2:	14:
3:	15:
4:	16:
5:	17:

6:	18:
7:	19:
8:	20:
9:	21:
10:	22:
11:	23:
12:	24:

```

change station 575                                     Page 5 of 5
                                                       STATION
DISPLAY BUTTON ASSIGNMENTS
1: normal
2:
3:
4:
5:
6:
7:

```

Important notes about the above programming:

Page 1

- The station Type is important. A beneficial configuration will have the gateway ports set up as '7434ND' (note the ND at the end of the type).
- Because you are selecting a station type that has an option display module, and the gateway requires that display to be present for integration data, you must set the Display Module option to 'y' and then set the Display Language to 'english'.

Page 2

- For a channel that is going to be used for processing MWIs, you must set the LWC Activation parameter to 'y'. This gives that specific port the ability to process MWI sets and clears.
- Because the Avaya PBX requires that you configure the station set with two call appearances (as seen on the next page), if you are going to do transfers you must configure each port with the Restrict Last Appearance parameter set to 'y'. This will ensure that you always have the second appearance in an idle state for doing transfers.

Page 3

- Because of a requirement of the Avaya PBX, you must configure each channel that is going to do transfers with two call appearances. These two appearances must be on the first 2 buttons (Buttons # 1 & 2) of the station.
- For a channel that will be used to process MWIs, you must administer both a LWC-Store and a LWC-Cancel key on the last two buttons (Buttons # 9 & 10) of the station set.

Page 4

- Ensure that all feature buttons listed on this page are blank.

Page 5

- Because you have administered a display module on this station set, you are required to administer a Normal key on the first button of the display module. Ensure that all other buttons are defined as blank. The PBX will not allow you to complete the programming until you set up the button as shown.

5.2. Defining Hunt Group and Coverage Path

Use the `change hunt-group` or `add hunt-group` command to a single hunt group to act as a central inbound route for all calls.

```
change hunt-group 10                                     Page 1 of 10
                                                         HUNT GROUP
Group Number: 10                                         ACD? n
Group Name: UM Hunt                                       Queue? n
Group Extension: 500                                     Vector? n
Group Type: ucd-mia                                       Coverage Path:
TN: 1                                                    Night Service Destination:
COR: 1                                                  MM Early Answer? n
Security Code:
ISDN Caller Display:
```

```
change hunt-group 10                                     Page 3 of 10
                                                         HUNT GROUP
Group Number: 10   Group Extension: 500   Group Type: ucd-mia
Voice Mail Number:   Administered Members (min/max): 1 /8
Total Administered Members: 8
GROUP MEMBER ASSIGNMENTS
Ext  Name                               Ext  Name
1 : 501  VoiceMail                          14 :
2 : 502  VoiceMail                          15 :
3 : 503  VoiceMail                          16 :
4 : 504  VoiceMail                          17 :
5 : 505  VoiceMail                          18 :
6 : 506  VoiceMail                          19 :
7 : 507  VoiceMail                          20 :
8 : 508  VoiceMail                          21 :
9 :                               22 :
```



```

10 :                               23 :
11 :                               24 :
12 :                               25 :
13 :                               26 :

At End of Member List

```

Important notes about the above programming:

Page 1

- To change the type of hunting performed by this group, you can change the Group Type parameter. These settings can be changed between a uniform style of call distribution (round robin) and a linear style.
- If desired, you can also activate queuing on the hunt group to allow a call to ring into the group even if all the ports in the group are busy.

Page 2

- This is the page where you add each of the gateway ports that are going to be part of your inbound hunt group. If required, you may add additional channels past the limits of this page.

Use the change coverage path or add coverage path command to configure a single coverage path to act as a forwarding rule to be assigned to each subscriber.

```

change coverage path 20                               Page 1 of 1
              COVERAGE PATH

Coverage Path Number: 20

Next Path Number:                               Hunt after Coverage? n
                                                Linkage

COVERAGE CRITERIA

Station/Group Status  Inside Call  Outside Call
Active?                n             n
Busy?                  y             y
Don't Answer?         y             y             Number of Rings: 4
All?                   n             n
DND/SAC/Goto Cover?  y             y

COVERAGE POINTS

Terminate to Coverage Pts. with Bridged Appearances? n

```

Point1: h10	Point2:	Point3:
Point4:	Point5:	Point6:

Important notes about the above programming:

Page 1

- To set forwarding for both internal and external busy and ring no answer calls, set the appropriate configuration sections to 'y'.
- To set the number of rings that constitute a ring no answer condition, set the required number of rings in the Number of Rings field.
- To route any calls that will follow this coverage path, place the hunt group number into the Point1 field.

5.3. Setting Up Subscriber Station Sets

This is an example of how to set up a subscriber's station set to forward correctly to the server via the coverage path that was administered. Use the `change station` command to alter an existing station or the `add station` command to add a new users station. The only modification needed is to ensure that the Coverage Path 1 configuration section is set to use the coverage path that has been defined.

```

change station 585                                     Page 1 of 5

                                     STATION

Extension: 585                                       Lock Messages? n
  Type: 8434D                                       Security Code:      TN: 1
  Port: 01A0511                                     Coverage Path 1: 20 COR: 1
  Name: Smith, John                                 Coverage Path 2:    COS: 1
                                                    Hunt-to Station:

STATION OPTIONS
  Loss Group: 2                                     Personalized Ringing Pattern: 1
  Data Module? n                                   Message Lamp Ext: 585
  Speakerphone: 2-way                               Mute Button Enabled? y
  Display Language: english                         Expansion Module? n

```

5.4. Additional Comments

Phone Set Type Selection

This configuration requires the use of the 7434ND station set type for each of the gateway ports. If you do not have this station type as an option when doing your configuration, then you may need to enable this option in the PBX's system configuration area. You should be able to locate an option there that specifically states 'Enable 7434ND support?' and should set that parameter to 'y'. This station type is important because it allows you to not have to make any changes to the subscriber's

name fields in the PBX to get integration to work. If you do not use this (7434ND) station type and instead opt to use the 8434 station type, then you will be required to configure the name field of each subscriber's phone set to make sure that it includes their extension number in the name field, somewhere within the first 15 characters. Below is an example of what to change on all your subscribers (change shown in red type) if you do not use the 7434ND station set type on the gateway ports.

```
change station 585                                     Page 1 of 5

                                STATION

Extension: 585                                         Lock Messages? n
  Type: 8434D                                         Security Code:      TN: 1
  Port: 01A0511                                       Coverage Path 1: 20  COR: 1
  Name: 585 Smith, John                                   Coverage Path 2:      COS: 1
                                                         Hunt-to Station:

STATION OPTIONS
  Loss Group: 2                                         Personalized Ringing Pattern: 1
  Data Module? n                                       Message Lamp Ext: 585
  Speakerphone: 2-way                                   Mute Button Enabled? y
  Display Language: english                             Expansion Module? n
```

MWI Considerations

The Avaya PBX requires that the same port be used to turn off an MWI that is turned on. Keep this in mind when troubleshooting any MWI issues. Consider this requirement when performing a system cut-over from another solution. Message indicators on subscriber phones that are already activated prior to a cut-over will most likely not be able to be turned off by the gateway. These lights will need to be manually cleared by the PBX administrator. For these same reasons, any ports of the gateway that are not planned to be used should be marked as disabled in the port capabilities configuration area of the gateway.

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	

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...)

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.