

# Microsoft Exchange Server 2007 Unified Messaging

## PBX Configuration Note:

**Avaya S8500**

**with Dialogic<sup>®</sup> 2000 Media Gateway Series**

**(DMG2xxxDTI) using T1 CAS In-band 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 Avaya S8500 and Dialogic® 2000 Media Gateway Series (DMG2xxxDTI) using T1 CAS In-band 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

|                            |                                     |
|----------------------------|-------------------------------------|
| <b>PBX Vendor</b>          | Avaya                               |
| <b>Model</b>               | S8500                               |
| <b>Software Version</b>    | Comm manager 3.0 – R013x.00.1.346.0 |
| <b>Telephony Signaling</b> | T1 CAS In-band Integration          |
| <b>Additional Notes</b>    | N/A                                 |

### 1.2. VoIP Gateway

|                         |  |
|-------------------------|--|
| <b>Gateway Vendor</b>   | Dialogic Corporation                             |
| <b>Model</b>            | Dialogic® 2000 Media Gateway Series (DMG2xxxDTI) |
| <b>Software Version</b> | 5.1.10   |
| <b>VoIP Protocol</b>    | SIP  |

### 1.3. Microsoft Exchange Server 2007 Unified Messaging

|                |     |
|----------------|-----|
| <b>Version</b> | RTM |
|----------------|-----|

## 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 TN464F DS1 interface card.

### 2.3. Cabling Requirements

No special cabling was required. A standard T1 cable was 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.

### 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 MWI on and MWI off feature access codes according to what your PBX requires. The defaults are MWI On = \*4 and MWI Off = #4.
- Set the Maximum Call Party delay to 4000 msec.
- Set the Incoming Rings Before Answer to 1.
- Set the feature access code to Drop a transfer while the destination is busy to '!'.  
• Configure the SIP Transport for TCP.

#### 4.1. Parsing rules

In order to gather proper integration data from this PBX, the following CPID parsing rules are required for all calls. These rules allow the gateway to collect and extract the In-band integration data for the call as it arrives at the gateway.

```
rule #00#\d(3-10)##
src_number 1
reason direct

rule #00#\d(3-10)#\d(3-10)#
src_number 1
dst_number 2
reason no-answer

rule #02#\d(3-10)#\d(3-10)#
```

```
src_number 1
dst_number 2
reason no-answer

rule #01#\d(1-2)##
reason trunk

rule #01#\d(1-2)
reason opening

rule #01#
reason direct

rule #03##\d(3-10)#
dst_number 1
reason no-answer

rule #04##\d(1-2)#
reason trunk
```

The above rules are contained in the ini file named 'T1\_LS\_Cfg\_Avaya8500.ini' that is part of the gateway software installation package.

## 5. PBX Setup Notes

The basic steps of setting up the PBX for use with this gateway and a voice messaging system 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 the subscriber station sets.

All PBX programming is done via a Windows-based Java application. The basic commands that you will encounter on the PBX to perform these actions are:

- change; delete; and add

### 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 (24 ports per span). The example below shows performing this command on one station.

The screenshot shows a window titled "Native Configuration Manager Version 3.0-81 connected to 10.242.202.91 : 5022". The interface includes a menu bar with "File" and "View", a command input field containing "cha stat 2101", and a toolbar with buttons for "CANCEL", "REFRESH", "ENTER", "CLEAR FL...", "HELP", "NEXT PA...", and "PREV PA...". Below the toolbar are function key labels F1 through F8. The main display area shows the command "change station 2101" and "Page 1 of 3". The configuration details for station 2101 are as follows:

|            |         |                |   |        |   |
|------------|---------|----------------|---|--------|---|
| Extension: | 2101    | Lock Messages? | n | BCC:   | 0 |
| Type:      | VMI     | Security Code: |   | TN:    | 1 |
| Port:      | 01A0601 |                |   | COR:   | 1 |
| Name:      | 2101    |                |   | COS:   | 1 |
|            |         |                |   | Tests? | y |

STATION OPTIONS

|                        |          |
|------------------------|----------|
| Loss Group:            | 4        |
| Off Premises Station?  | y        |
| R Balance Network?     | n        |
| Survivable COR:        | internal |
| Survivable Trunk Dest? | y        |

Info: Right-click in a field to see valid entries or help text.

Important notes about the above programming:

1. The Type field must be set to VMI. If this is not set correctly, the PBX will not send In-band integration data.
2. The Port field points this station definition to a specific bearer channel of the DS1 interface circuit.

## 5.2. Defining Hunt Group and Coverage Path

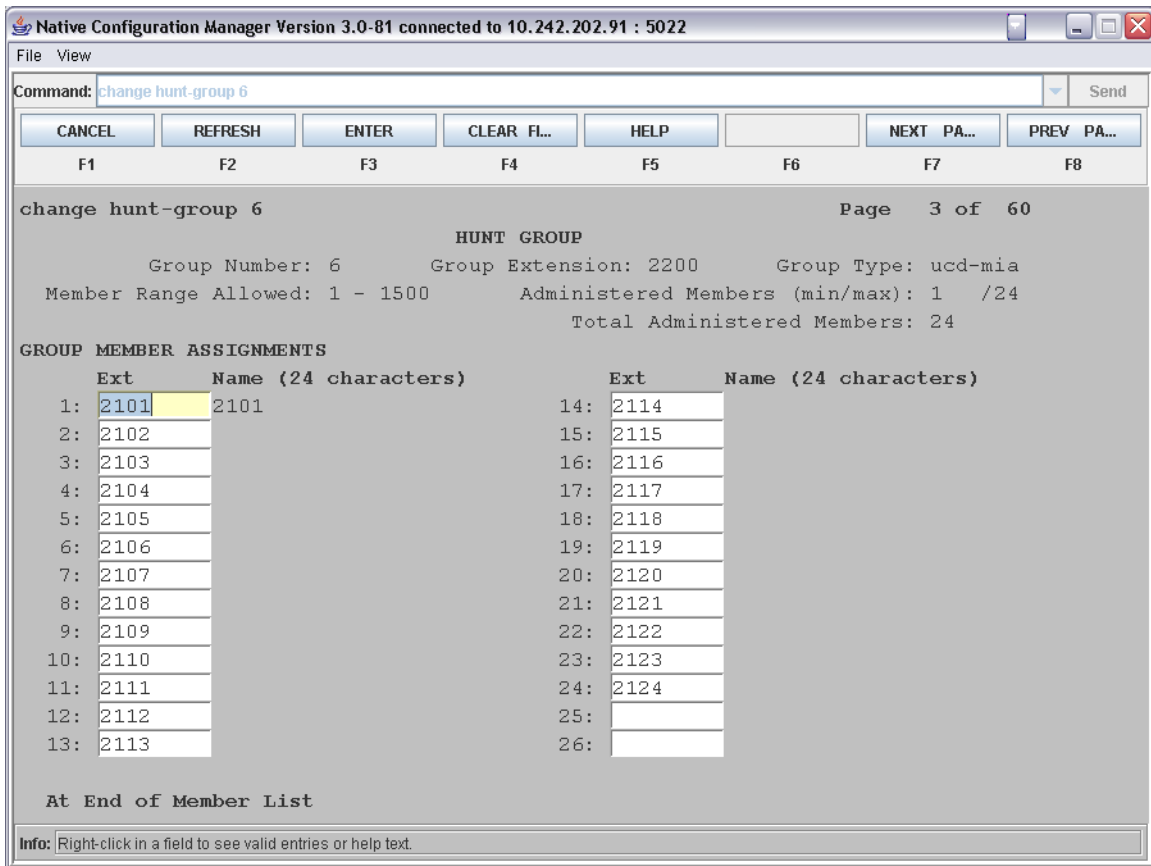
Use the `change hunt-group` or `add hunt-group` command to build a single hunt group to act as a central inbound route for all calls.

The screenshot shows the Native Configuration Manager interface for configuring Hunt Group 6. The window title is "Native Configuration Manager Version 3.0-81 connected to 10.242.202.91 : 5022". The command entered is "change hunt-group 6". The interface includes a command line, a toolbar with function keys (F1-F8), and a configuration area with various fields and their values.

| HUNT GROUP                 |         |
|----------------------------|---------|
| Group Number:              | 6       |
| Group Name:                | VM      |
| Group Extension:           | 2200    |
| Group Type:                | ucd-mia |
| TN:                        | 1       |
| COR:                       | 1       |
| Security Code:             |         |
| ISDN/SIP Caller Display:   |         |
| ACD?                       | n       |
| Queue?                     | n       |
| Vector?                    | n       |
| Coverage Path:             |         |
| Night Service Destination: |         |
| MM Early Answer?           | n       |
| Local Agent Preference?    | n       |

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Info: Right-click in a field to see valid entries or help text.



Important notes about the above programming:

**Page 1**

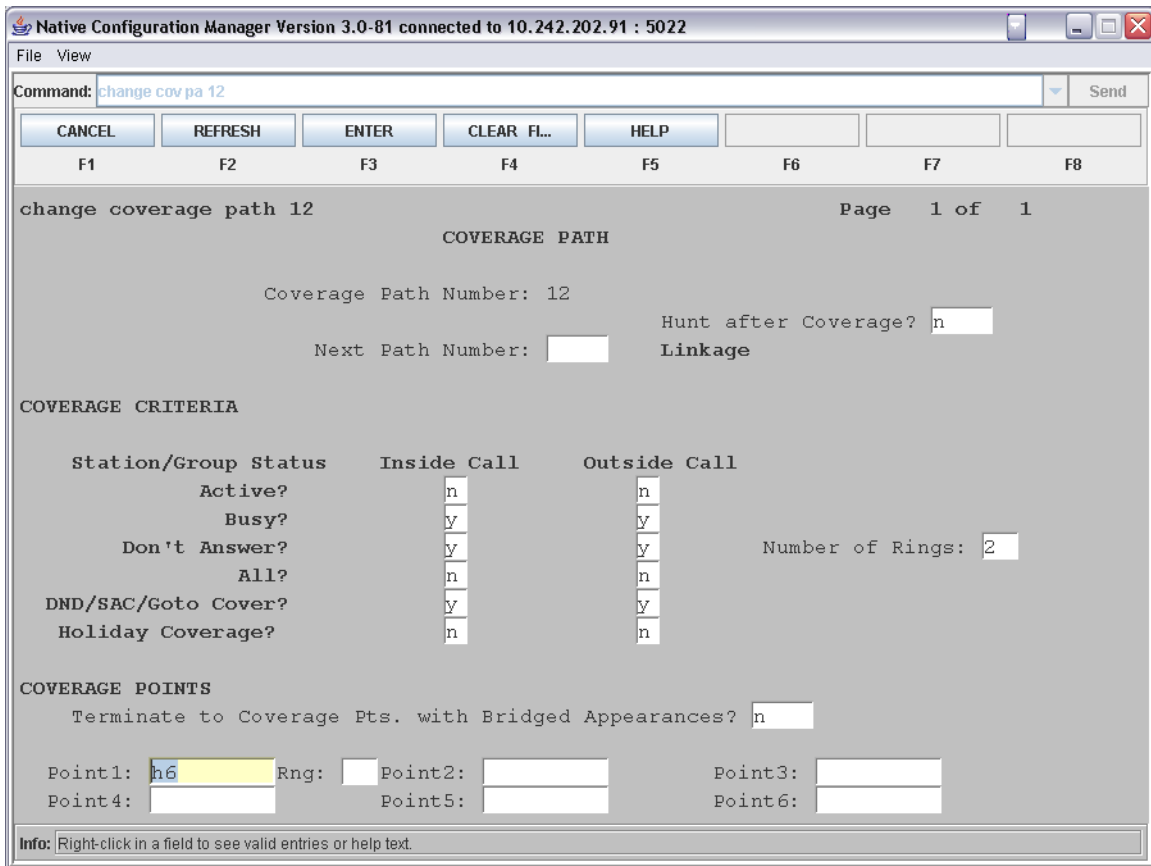
1. To change the type of hunting performed by this group, you can change the Group Type parameter. These settings can be changed between a uniform style of call distribution (round robin) and a linear style.
2. If desired, you can also activate queuing on the hunt group that will allow a call to ring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

**Page 3**

3. This is the page where you add each of the gateway ports that are going to be part of your inbound hunt group.

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





Important notes about the above programming:

**Page 1**

1. To set forwarding for both internal and external busy and ring no answer calls, set the appropriate configuration sections to 'y'.
2. To set the number of rings that constitutes a ring no answer condition, set the required number of rings in the Number of Rings field.
3. 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 station set to forward correctly to the server via the coverage path that was administered. The only modification that is needed is to ensure that the Coverage Path 1 configuration section is set to use the coverage path defined above.

The screenshot shows the 'Native Configuration Manager' interface. The title bar indicates it is connected to 10.242.202.91 on port 5022. The command entered is 'cha sta 2011'. The interface displays configuration for 'change station 2011' on 'Page 1 of 5'. The configuration is divided into 'STATION' and 'STATION OPTIONS' sections.

| STATION    |         |                  |    |      |   |
|------------|---------|------------------|----|------|---|
| Extension: | 2011    | Lock Messages?   | n  | BCC: | 0 |
| Type:      | 8434D   | Security Code:   |    | TN:  | 1 |
| Port:      | 01A0512 | Coverage Path 1: | 12 | COR: | 1 |
| Name:      | 2011    | Coverage Path 2: |    | COS: | 1 |
|            |         | Hunt-to Station: |    |      |   |

| STATION OPTIONS   |         |                               |      |
|-------------------|---------|-------------------------------|------|
| Loss Group:       | 2       | Personalized Ringing Pattern: | 1    |
| Data Module?      | n       | Message Lamp Ext:             | 2011 |
| Speakerphone:     | 2-way   | Mute Button Enabled?          | y    |
| Display Language: | english | Expansion Module?             | n    |
|                   |         | Media Complex Ext:            |      |
|                   |         | IP SoftPhone?                 | n    |
|                   |         | Remote Office Phone?          | n    |

Info: Right-click in a field to see valid entries or help text.

### 5.4. Additional Comments

#### 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.

## 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.<br><br>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.<br><br>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.<br><br>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).<br><br>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.<br><br>Confirm the correct called party answers the phone.  | P |   |
| 6b | Call Transfer by Directory Search when the called party's phone is busy.<br><br>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.<br><br>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.<br><br>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.<br><br>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>  |    | 5.1.10 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>  |    | 5.1.10 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>  |    | 5.1.10 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 <a href="#">MWI 2007 website</a>.</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

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

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

## 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.