Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

Mitel SX200

with Dialogic® 1000 Media Gateway Series

(DMG1008MTLDNI) using Digital Set Emulation

By : Dialogic

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Content

This document describes the configuration required to setup Mitel SX200 and Dialogic® 1000 Media Gateway Series (DMG1008MTLDNI) 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
March 17, 2008	Initial version of this document.

1. Components Information

1.1. PBX or IP-PBX

PBX Vendor	Mitel
Model	SX200
Software Version	Lightware 19
Telephony Signaling	Digital set emulation
Additional Notes	N/A

1.2. VoIP Gateway

Gateway Vendor	Dialogic Corporation
Model	Dialogic® 1000 Media Gateway Series (DMG1008MTLDNI)
Software Version	5.0.42
VoIP Protocol	SIP

1.3. Microsoft Exchange Server 2007 Unified Messaging

Version	RTM

2. Prerequisites

2.1. Gateway Requirements

The gateway needs to support Super Set 430 digital station set emulation.

2.2. PBX Requirements

To support the 2-wire station interface as documented, you need the DNI Line MC330 line card.

2.3. Cabling Requirements

No specific cabling requirements have been noted.

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.

NOTE: The Mitel gateway is a PBX-specific SKU and therefore does not require being told what specific PBX it is being configured for.

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 Exchange UM are as follows:

- Setting up each gateway station port.
- Defining hunt group to act as a central point for incoming calls to the gateway.
- Setting up subscriber station sets.

You will do all your programming using a menu system provided by the Mitel PBX. This interface is typically accessed via a serial or network terminal.

5.1. Setting Up Each Gateway Station Port

Use the `Class of Service Options Assignment` menu selection to build a class of service template that will be shared between all the gateways station ports. The example below shows performing this command.

```
|          CLASS          OF          SERVICE          OPTIONS          ASSIGNMENT
|
|Class of service number : 1
|Comments :
|Option                                                          Select
+-----------------------------------------------------------------
|Account Code Verified. . . . . . . . . . . . . . . . . . .     No
|ACD Silent Monitoring - Accept . . . . . . . . . . . . . .     No
|ACD Silent Monitoring - Allowed. . . . . . . . . . . . . .     No
|ACD Silent Monitoring - Notification . . . . . . . . . . .     No
|ANI/DNIS/ISDN Number Delivery Trunk. . . . . . . . . . . .    Yes
|Auto Answer Allowed. . . . . . . . . . . . . . . . . . . .    Yes
|Brokers Call . . . . . . . . . . . . . . . . . . . . . . .     No
|Busy Override Security . . . . . . . . . . . . . . . . . .     No
|Call Announce Line . . . . . . . . . . . . . . . . . . . .    Yes
|Call Forwarding - Accept . . . . . . . . . . . . . . . . .    Yes
|Call Forwarding (External Destination) . . . . . . . . . .    Yes
|Call Forwarding (Internal Destination) . . . . . . . . . .    Yes
|Call Forwarding - Override . . . . . . . . . . . . . . . .     No
|Call Hold. . . . . . . . . . . . . . . . . . . . . . . . .    Yes
|Call Hold - Remote Retrieve. . . . . . . . . . . . . . . .    Yes
|Call Hold - Retrieve with Hold Key . . . . . . . . . . . .     No
|Call Pickup - Dialed : Accept. . . . . . . . . . . . . . .    Yes
|Call Pickup - Directed : Accept. . . . . . . . . . . . . .    Yes
|Call Privacy . . . . . . . . . . . . . . . . . . . . . . .     No
|Call Reroute after CFFM to busy destination. . . . . . . .     No
|Call Waiting - Swap. . . . . . . . . . . . . . . . . . . .     No
|Calling Name Display - Internal - ONS  . . . . . . . . . .    Yes
|Calling Number Display - Internal - ONS  . . . . . . . . .    Yes
|Camp-on Tone Security. . . . . . . . . . . . . . . . . . .     No
```

```
|Check COR after PSTN Dial Tone . . . . . . . . . . . . . .    No
|Clear All Features Remote. . . . . . . . . . . . . . . . .    No
|Conference Call. . . . . . . . . . . . . . . . . . . . . .   Yes
|COV/ONS/E&M Voice Mail Port. . . . . . . . . . . . . . . .    No
|DASS II OLI/TLI Provided . . . . . . . . . . . . . . . . .    No
|Dialed Night Service . . . . . . . . . . . . . . . . . . .   Yes
|Disable Call Reroute Chaining On Diversion . . . . . . . .    No
|Disable Send Message.  . . . . . . . . . . . . . . . . . .    No
|Display ANI/ISDN Calling Number Only . . . . . . . . . . .    No
|Display ANI/DNIS/ISDN Calling/Called Number. . . . . . . .   Yes
|Display Caller ID on multicall/keylines. . . . . . . . . .   Yes
|Display DNIS/Called Number Before Digit Modification . . .   Yes
|Display Dialed Digits during Outgoing Calls. . . . . . . .   Yes
|Display Held Call ID on Transfer . . . . . . . . . . . . .    No
|Do Not Disturb . . . . . . . . . . . . . . . . . . . . . .   Yes
|Do Not Disturb - Access to Remote Phones . . . . . . . . .   Yes
|Do Not Disturb - Permanent . . . . . . . . . . . . . . . .    No
|Emergency Call Notification - Audio. . . . . . . . . . . .    No
|Emergency Call Notification - Visual . . . . . . . . . . .    No
|Enable Call Duration Limit on External Calls . . . . . . .    No
|Enable Call Duration Limit on Internal Calls . . . . . . .    No
|Executive Busy Override. . . . . . . . . . . . . . . . . .    No
|External Trunk Standard Ringback . . . . . . . . . . . . .    No
|Flexible Answer Point. . . . . . . . . . . . . . . . . . .    No
|Follow 2nd Alternate Reroute for Recall to Busy ACD Agent     No
|Forced Verified Account Code . . . . . . . . . . . . . . .    No
|Forced Non-Verified Account Code . . . . . . . . . . . . .    No
|Group Call Forward Follow Me - Accept. . . . . . . . . . .    No
|Group Call Forward Follow Me - Allow . . . . . . . . . . .    No
|Group Page - Accept. . . . . . . . . . . . . . . . . . . .    No
|Group Page - Allow . . . . . . . . . . . . . . . . . . . .    No
|Handset Volume Adjustment - Saved. . . . . . . . . . . . .    No
|Handsfree AnswerBack Allowed . . . . . . . . . . . . . . .   Yes
|HCI/CTI/TAPI Call Control Allowed. . . . . . . . . . . . .    No
|HCI/CTI/TAPI Monitor Allowed . . . . . . . . . . . . . . .    No
|Head Set Switch Mute . . . . . . . . . . . . . . . . . . .    No
|Hot Desk Remote Logout Enabled . . . . . . . . . . . . . .    No
|Hot Desk Login Accept  . . . . . . . . . . . . . . . . . .    No
|Hotel Room Extension . . . . . . . . . . . . . . . . . . .    No
|Hotel Room Monitor Setup Allowed . . . . . . . . . . . . .    No
|Hotel Room Monitoring Allowed. . . . . . . . . . . . . . .    No
|Hotel/Motel Room Personal Wakeup Call Allowed  . . . . . .    No
|Hotel/Motel Room Remote Wakeup Call Allowed. . . . . . . .    No
|Individual Trunk Access. . . . . . . . . . . . . . . . . .   Yes
|Keep TelDir Entry on Check Out . . . . . . . . . . . . . .    No
```

```
|Local Music On Hold source . . . . . . . . . . . . . . . .   No
|Loudspeaker Pager Override . . . . . . . . . . . . . . . .  Yes
|Loudspeaker Pager Equivalent Zone Override Security. . . .   No
|Message Waiting. . . . . . . . . . . . . . . . . . . . . .  Yes
|Message Waiting - Audible Tone Notification. . . . . . . .  Yes
|Message Waiting - Deactivate On Off-Hook . . . . . . . . .   No
|Message Waiting - Inquire. . . . . . . . . . . . . . . . .  Yes
|Multiline Set Loop Test. . . . . . . . . . . . . . . . . .   No
|Multiline Set Message Center Remote Read Allowed . . . . .   No
|Multiline Set Music. . . . . . . . . . . . . . . . . . . .   No
|Multiline Set On Hook Dialing. . . . . . . . . . . . . . .  Yes
|Multiline Set Phonebook Allowed. . . . . . . . . . . . . .  Yes
|Multiline Set Voice Mail Callback Message Erasure Allowed    No
|Name Suppression on outgoing Trunk Call. . . . . . . . . .   No
|Non-DID Extension. . . . . . . . . . . . . . . . . . . . .   No
|Non-Prime Public Network Identity. . . . . . . . . . . . .   No
|Non-Verified Account Code. . . . . . . . . . . . . . . . .  Yes
|Off-Hook Voice Announce Allowed. . . . . . . . . . . . . .   No
|ONS CLASS/CLIP: Message Waiting Activate/Deactivate. . . .   No
|ONS CLASS/CLIP: Set. . . . . . . . . . . . . . . . . . . .   No
|ONS CLASS/CLIP: Visual Call Waiting  . . . . . . . . . . .  Yes
|ONS/OPS Internal Ring Cadence for External Callers . . . .   No
|Override Interconnect Restriction on Transfer. . . . . . .   No
|Pager Access - All Zones . . . . . . . . . . . . . . . . .  Yes
|Pager Access - Individual Zones. . . . . . . . . . . . . .   No
|Privacy Released . . . . . . . . . . . . . . . . . . . . .   No
|Public Network Access via DPNSS  . . . . . . . . . . . . .  Yes
|Public Network Identity Provided . . . . . . . . . . . . .  Yes
|Public Network To Public Network Connection Allowed. . . .  Yes
|Public Trunk . . . . . . . . . . . . . . . . . . . . . . .  Yes
|R2 Call Progress Tones . . . . . . . . . . . . . . . . . .   No
|Record-A-Call Active . . . . . . . . . . . . . . . . . . .   No
|Record-A-Call - Start Recording Automatically  . . . . . .   No
|Record-A-Call - Save Recording on Hang-up  . . . . . . . .   No
|Recorded Announcement Device . . . . . . . . . . . . . . .   No
|Recorded Announcement Device - Advanced  . . . . . . . . .   No
|Redial Facilities. . . . . . . . . . . . . . . . . . . . .  Yes
|Ringing Line Select. . . . . . . . . . . . . . . . . . . .   No
|SC1000 Attendant Basic Function Key (Yes/NO) . . . . . . .   No
|SMDR - External. . . . . . . . . . . . . . . . . . . . . .   No
|SMDR - Internal. . . . . . . . . . . . . . . . . . . . . .   No
|Speak@Ease Preferred . . . . . . . . . . . . . . . . . . .   No
|Suite Services Enabled . . . . . . . . . . . . . . . . . .   No
|Suppress Simulated CCM after ISDN Progress . . . . . . . .   No
|Third Party Call Forward Follow Me - Accept. . . . . . . .  Yes
```

```
|Third Party Call Forward Follow Me - Allow . . . . . . . .   Yes
|Timed Reminder Allowed . . . . . . . . . . . . . . . . . .   Yes
|Trunk Calling Party Identification . . . . . . . . . . . .   Yes
|Trunk Flash Allowed. . . . . . . . . . . . . . . . . . . .    No
|Use Held Party Device for Call Re-routing. . . . . . . . .   Yes
|Use Called Party Call Hold Timer . . . . . . . . . . . . .    No
|Voice Mail Softkey . . . . . . . . . . . . . . . . . . . .    No
|Timer Options  / Account Code Length                       Value
|Account Code Length (2-12) . . . . . . . . . . . . . . . .    12
|After Answer Display Time(0-60 secs, Blank=off). . . . . .
|ANSWER PLUS - Delay To Message Timer (0-300 secs). . . . .    20
|ANSWER PLUS - Expected Offhook Timer (0-255 secs). . . . .    30
|ANSWER PLUS - Message Length Timer (0-120 secs). . . . . .    10
|ANSWER PLUS - System Reroute Timer (0-720 secs). . . . . .     0
|Attendant Busy-out Timer (1-1440 mins) . . . . . . . . . .    10
|Auto Camp-on Timer (0-30 secs, Blank for Off). . . . . . .    10
|Busy Tone Timer ( 1-120 secs). . . . . . . . . . . . . . .    30
|Call Duration  (2-120 mins). . . . . . . . . . . . .          10
|Call Duration Forced Cleardown Timer (0-10 mins) . .           0
|Call Forward - Delay (0-125 secs). . . . . . . . . . . . .     0
|Call Forward No Answer Timer (0-125 secs). . . . . . . . .    15
|Call Hold Timer (10-600 secs). . . . . . . . . . . . . . .    30
|Camp-on Recall Timer (0-180 secs). . . . . . . . . . . . .   110
|Delay Ring Timer (5-60 secs) . . . . . . . . . . . . . . .    10
|Dialing Conflict Timer (2-5 secs). . . . . . . . . . . . .     3
|Display Caller ID On Multicall/Keylines timer (3-125 secs)     5
|Emergency Call - Audio Level for Set(Ringer/Medium/High) . Ringer
|First Digit Timer (5-60 secs). . . . . . . . . . . . . . .    15
|Inter-Digit Timer (3-60 secs). . . . . . . . . . . . . . .    10
|Lockout Timer ( 1-60 secs) . . . . . . . . . . . . . . . .    45
|ACD 2000 Logout Agent No Answer Timer (0-125 secs) . . . .    15
|Message Waiting Ringing Start Time (00:00 to 23:59). . . .     :
|Message Waiting Ringing Stop Time  (00:00 to 23:59). . . .
|No Answer Recall Timer (0-125 secs). . . . . . . . . . . .     17
|ONS VMail-Delay Dial Tone Timer (5-20 secs). . . . . . . .      5
|Ringing Timer (60-300 secs). . . . . . . . . . . . . . . .   180
|Work Timer (0-600 secs). . . . . . . . . . . . . . . . . .      0
  DTMF Key Assignments                                       Value
|Key A. . . . . . . . . . . . . . . . . . . . . . . . . . .
|Key B. . . . . . . . . . . . . . . . . . . . . . . . . . .
|Key C. . . . . . . . . . . . . . . . . . . . . . . . . . .
|Key D. . . . . . . . . . . . . . . . . . . . . . . . . . .
```

Use the `DNI Circuit Assignment` menu selection to set a specific wiring address to use the Superset 430 station type. Configure as many wiring addresses as required for your configuration (8 ports per gateway). The example below shows performing this command on one wire address.

```
                          DNI CIRCUIT ASSIGNMENT                               |
|                                                                              |
|       |     |    |       |               |           Device Type            |
|Cabinet|Shelf|Slot|Circuit|   Card Type   |   Channel #1    |   Channel #2    |
|-------|-----|----|-------|---------------|-----------------|-----------------|
|   2   |  1  |  1 |    1  |DNI Line       | Superset 430    |                 |
```

Important notes about the above programming:

1. The station type setting under the `Channel #1` column is important and must be set for `Superset 430`.

Use the `Multiline Set Key Assignment` menu selection to assign each wiring address to an individual digital station set and extension number. Configure as many digital stations as required for your configuration (8 ports per gateway). The example below shows performing this command on one station.

```
                     MULTILINE SET KEY ASSIGNMENT                              |
|                                                                              |
| Prime Directory Number: 501       Prime Line Type:    single line            |
| Prime Ring Type: ring             Cab,Shlf,Slot,Circ:   2  1  1   1          |
| Prime Name:                                                                  |
|                                                                              |
|  Key Number   |Directory Number |    Line Type       |       Ring Type       |
|---------------|-----------------|--------------------|-----------------------|
|       2       |                 |not assigned        |                       |
|       3       |                 |not assigned        |                       |
|       4       |                 |not assigned        |                       |
|       5       |                 |not assigned        |                       |
|       6       |                 |not assigned        |                       |
|       7       |                 |not assigned        |                       |
|       8       |                 |not assigned        |                       |
|       9       |                 |not assigned        |                       |
|      10       |                 |not assigned        |                       |
|      11       |                 |not assigned        |                       |
```

Important notes about the above programming:

1. The `Prime Directory Number` is what assigns this station its dial-able number.
2. Set the `Prime Ring Type` to `ring`.

3. Set the Prime Line Type to single line.
4. Ensure that all the other available key assignments are set to not assigned.

Use the Station Service Assignment menu selection to attach the station that has been created to the defined class-of-service template.

STATION SERVICE ASSIGNMENT

Directory Number	Intercept Number	Class of Service DAY	Class of Service NIGHT1	Class of Service NIGHT2	Class of Restriction DAY	Class of Restriction NIGHT1	Class of Restriction NIGHT2	Default Account Code Index	COMP Zone ID
501	1	1	1	1	1	1	1		1
502	1	1	1	1	1	1	1		1
503	1	1	1	1	1	1	1		1
504	1	1	1	1	1	1	1		1
505	1	1	1	1	1	1	1		1
506	1	1	1	1	1	1	1		1
507	1	1	1	1	1	1	1		1
508	1	1	1	1	1	1	1		1

5.2. Defining Hunt Group

Use the Hunt Group Assignment menu selection to add a new or change an existing hunt group to act as a central inbound route for all calls.

```
                        HUNT GROUP ASSIGNMENT
   Pilot Number : 500                        Name :
   Hunt Mode :    Circular                   Priority :
   Group Type :   Voice                      1st Threshold (%) :
   RAD1 :                                    2nd Threshold (%) :
   RAD2 :                                    Alert Device :
   NIGHT RAD :                               Phase Timer :
```

Member	Directory Number	Name
1	501	
2	502	
3	503	
4	504	
5	505	

```
|        6        |        506         |                                          |
|        7        |        507         |                                          |
|        8        |        508         |                                          |
|        9        |                    |                                          |
```

Important notes about the above programming:

1. The `Pilot Number` is what assigns this hunt group a dial-able number and is used to access this hunt group. This is the number to publish to users to access the system for forwarding and direct dialing into the UM server.
2. The `Hunt Mode` setting is what allows you to adjust the hunting pattern within the group. You can set this to be either a circular or linear style of hunting.
3. Set the `Group Type` to `Voice`.
4. Add each of your configured gateway ports as members to this group.

5.3. Setting Up Subscriber Station Sets

There is no PBX-side programming for setting up the subscriber station sets. All the forwarding of the subscriber station sets is defined directly on subscriber station set using the phone's soft menu keys. The subscriber should be directed to set their internal and external ring no answer and busy forwarding conditions to the Pilot Number setting defined in the hunt group configuration.

5.4. Additional Comments

Because of the way forwarding needs to be configured at each individual subscriber station, it is important to remind the subscriber that it is critical that they keep their phones forwarded, and that they are forwarded to the proper destination for Exchange UM to function properly for them.

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.	**CP**	Trunk used did not provide call id data so UM indicated "Anonymous" as the calling party.

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.	CP	Trunk issues prevented this test from completing. The error message provided by the CO was judged as connection and the message was played out.
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 extension. Confirm the FAX is received in the user's inbox.	**P**	
10	Setup TLS between gateway/IP-PBX and Exchange UM. Replace this italicized text with your TLS configuration: self-signed certificates or Windows Certificate Authority (CA).		
10a	Dial the pilot number and logon to a user's mailbox. Confirm UM answers the call and confirm UM responds to DTMF input.		5.0.42 gateway firmware does not yet implement TLS so this feature was not tested.
10b	Dial a user extension and leave a voicemail. Confirm the user receives the voicemail.		5.0.42 gateway firmware does not yet implement TLS so this feature was not tested.
10c	Send a test FAX message to user extension. Confirm the FAX is received in the user's inbox.		5.0.42 gateway firmware does not yet implement TLS so this feature was not tested.
11	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). Dial the pilot number and confirm the UM system answers the call.	**P**	
12	Setup Message Waiting Indicator (MWI). Geomant offers a third party solution: MWI 2007. Installation files and product documentation can be found on Geomant's MWI 2007 website.		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.

- `dspcpi` (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.