Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

Siemens Hipath 4000

with Dialogic® 2000 Media Gateway Series

(DMG2xxxDTI) using T1 QSIG

By : Dialogic

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Content

This document describes the configuration required to setup Siemens Hipath 4000 and Dialogic® 2000 Media Gateway Series (DMG2xxxDTI) using T1 QSIG 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	Siemens
Model	Hipath 4000
Software Version	Version 2.0 SMR9 SMP0
Telephony Signaling	T1 QSIG
Additional Notes	N/A

1.2. VoIP Gateway

Gateway Vendor	Dialogic Corporation
Model	Dialogic® 2000 Media Gateway Series (DMG2xxxDTI)
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 T1 QSIG interface.

2.2. PBX Requirements

PBX must have all supplemental service packages installed for the QSIG protocol to operate properly and provide all advanced supplemental services.

To connect to the PBX using T1 QSIG, you need the ISDN T1 - DIU2U - Q2216 line card.

2.3. Cabling Requirements

Cabling for QSIG connections must be CAT5e or better. Standard voice quality cable will not provide optimum signal quality and the gateway will have problems establishing connection on the D-Channel.

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 ISDN - QSIG.

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 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.
- Configure the SIP Transport for TCP.

5. PBX Setup Notes

The basic steps of setting up the PBX for use with this gateway and a voice processing system are as follows:

- Activating the QSIG protocol.
- Administrating the Trunk configuration.
- Assigning an access code to the trunk.
- Setting up the subscriber station sets.

Programming on the Hipath 4000 can be done either by using a text based GUI application or a command line interface. The programming here will be shown using the GUI commands.

5.1. Activating QSIG Protocol

If required, you may need to turn on the QSIG protocol before use. The programming steps below show how it was done on the test system. If you have any questions, it is recommended to contact your Siemens representative.

Enter the DIS-PRODE:DB,PDSHORT; command and press RETURN. The following screen is displayed:

```
+------+---------------------------+-------+-------+---------------+
|PDNAME| PDSTRING                  | IDENT | ACTIV | VERSION       |
+------+---------------------------+-------+-------+---------------+
| PD14 | ISO QSIG Second ed. SS    | H/08  | Y     | B0-EL0.20.001 |
+------+---------------------------+-------+-------+---------------+
```

Make note of the number shown in the PDNAME column. You will need it in a future step.

Enter the DISPLAY-PRODE:SRC=HD,KIND=PDSHORT; command and press RETURN. The following screen will be displayed:

```
+------+---------------------------+-------+---------------+
| PDNO | PDSTRING                  | IDENT | VERSION       |
+------+---------------------------+-------+---------------+
|  32  | ISO QSIG Second ed. SS    | H/08  | B0-EL0.20.001 |
+------+---------------------------+-------+---------------+
```

Make note of the number shown in the PDNO column. You will need it in a future step.

Enter the REG-PRODE; command and press RETURN.

Enter the COPY-PRODE:PD,32,PD14; command and press RETURN.

Enter the CHANGE-PRODE:VARTAB,PSS1V2R,PD14,,; command and press RETURN. The following screen should be displayed:

```
AMO-PRODE-111       PROTOCOL DESCRIPTOR FOR NETWORKING
REGENERATE COMPLETED;
```

5.2. Administering Trunk Configuration

Use the add-buend command to configure a trunk group.

- Enter the add-buend command and press RETURN
- At the prompt TGRP = enter xxx
    - where xxx is any available trunk group number and press RETURN
- At the prompt NAME = enter xxx
    - where xxx is any assigned name for the trunk group and press RETURN
- At the prompt NO = enter 48 and press RETURN
- At the prompt TRACENO = enter 0 and press RETURN
- At the prompt ACDTHRH = enter * and press RETURN
- At the prompt PRIONO = enter 2 and press RETURN
- At the prompt TDDRFLAG = enter OFF and press RETURN
- At the prompt GDTRRULE = enter 0 and press RETURN
- At the prompt ACDPMGRP = enter 0 and press RETURN
- At the prompt CHARCON = enter NEUTRAL and press RETURN

Display and validate the changes with the following command:

Enter DIS-BUEND:100; and press RETURN. The following screen should be displayed:

```
+------------------------------ FORMAT = L ----------------------------------+
| TGRP NUMBER :   100   TGRP NAME   : QSIG              MAXIMUM NO.    :   48 |
|                       CHARCON     : NEUTRAL                                 |
| SUBGROUP NO.:     7   DEVICE TYPE : S1CONN            TRACENO        :    0 |
| SEARCH MODE : DESCENDING                              ACD THRESHOLD :    * |
| NUMBER OF ASSOCIATED ROUTES       : 2                 PRIORITY       :    2 |
| TDDRFLAG    :   OFF  TDDRTHRESHOLD:                  SOURCEGROUPIDX :    1 |
| GDTRRULE    :     0      ACDPMGRP : 0                                       |
| THE FOLLOWING TRUNKS (LTG-LTU-SLOT-CCT) HAVE BEEN ALLOCATED:                |
+-----------------------------------------------------------------------------+
|  1- 1- 97-0          1 |  1- 1- 97-0          2 |  1- 1- 97-0          3 |
|  1- 1- 97-0          4 |  1- 1- 97-0          5 |  1- 1- 97-0          6 |
|  1- 1- 97-0          7 |  1- 1- 97-0          8 |  1- 1- 97-0          9 |
|  1- 1- 97-0         10 |  1- 1- 97-0         11 |  1- 1- 97-0         12 |
|  1- 1- 97-0         13 |  1- 1- 97-0         14 |  1- 1- 97-0         15 |
|  1- 1- 97-0         16 |  1- 1- 97-0         17 |  1- 1- 97-0         18 |
|  1- 1- 97-0         19 |  1- 1- 97-0         20 |  1- 1- 97-0         21 |
|  1- 1- 97-0         22 |  1- 1- 97-0         23 |                        |
```

```
+--------------------------------------------------------------------------+


AMO-BUEND-111       TRUNK GROUP
       DISPLAY COMPLETED;
```

Use the add-tdcsu command to configure a trunk.

- Enter the add-tdcsu and press RETURN
- At the prompt OPT = enter new and press RETURN
- At the prompt PEN = enter x-xx-xxx-x
  - Where x-xx-xxx-x is the location of the installed T1 Port Equipment Number and press RETURN
- At the prompt COTNO = enter xxx
  - Where xxx is your selected class of trunk and press RETURN
- At the prompt COPNO = enter xxx
  - Where xxx is your selected class of parameter and press RETURN
- At the prompt DPLN = enter 0 and press RETURN
- At the prompt ITR = enter 0 and press RETURN
- At the prompt COS = enter 100 and press RETURN
- At the prompt LCOSV = enter 1 and press RETURN
- At the prompt LCOSD = enter 5 and press RETURN
- At the prompt CCT = enter CORNET-NQ and press RETURN
- At the prompt DESTNO = enter 100 and press RETURN
- At the prompt PROTVAR = enter PSS1V2R and press RETURN
  - This is the setting that specifies that you are using QSIG with supplemental services. The protocol you have specified is 'qsig iso iso/iec 11572 2nd. with ss'
- At the prompt SEGMENT = enter 1 and press RETURN
- At the prompt DEDSCC = and press RETURN
- At the prompt DEDSVC = enter NONE and press RETURN
- At the prompt FACILITY = and press RETURN
- At the prompt DITIDX = and press RETURN
- At the prompt SRTIDX = and press RETURN
- At the prompt TRTBL = enter GDTR and press RETURN
- At the prompt SIDANI = enter N and press RETURN
- At the prompt ATNTYP = enter TIE and press RETURN
- At the prompt CBMATTR = enter NONE and press RETURN
- At the prompt TCHARG = enter N and press RETURN
- At the prompt SUPPRESS = enter 0 and press RETURN
- At the prompt DGTPR = and press RETURN
- At the prompt ISDNIP = and press RETURN
- At the prompt ISDNNP = and press RETURN
- At the prompt PNPL2P = and press RETURN
- At the prompt PNPL1P = and press RETURN
- At the prompt PNPAC = and press RETURN
- At the prompt TRACOUNT = enter 31 and press RETURN
- At the prompt SATCOUNT = enter MANY and press RETURN
- At the prompt NNO = enter 1-1-100 and press RETURN
- At the prompt ALARMNO = enter 0 and press RETURN

- At the prompt FIDX = enter 1 and press RETURN
- At the prompt CARRIER = enter 1 and press RETURN
- At the prompt ZONE = enter EMPTY and press RETURN
- At the prompt COTX = enter 100 and press RETURN
- At the prompt FWDX = enter 5 and press RETURN
- At the prompt CHIMAP = enter N and press RETURN
- At the prompt INIGHT = and press RETURN
- At the prompt DOMTYPE = enter PRIVATE and press RETURN
- At the prompt DOMAINNO = enter 0 and press RETURN
- At the prompt TPROFNO = and press RETURN
- At the prompt CCHDL = and press RETURN
- At the prompt UUSCCX = enter 16 and press RETURN
- At the prompt UUSCCY = enter 8 and press RETURN
- At the prompt FNIDX = enter 0 and press RETURN
- At the prompt NWMUXTIM = enter 10 and press RETURN
- At the prompt SRCGRP = enter 1 and press RETURN
- At the prompt CLASSMRK = enter EC&G711 and press RETURN
- At the prompt TCCID = and press RETURN
- At the prompt TGRP = enter 100 and press RETURN
- At the prompt SRCHMODE = enter DSC and press RETURN
- At the prompt INS = enter Y and press RETURN
- At the prompt DEV = enter S1CONN and press RETURN
- At the prompt BCHAN = enter 1&&23 and press RETURN
- At the prompt BCNEG = enter N and press RETURN
- At the prompt BCGR = enter 1 and press RETURN
- At the prompt LWPAR = enter 100 and press RETURN
- At the prompt LWPP = and press RETURN
- At the prompt LWLT = and press RETURN
- At the prompt LWPS = and press RETURN
- At the prompt LWR1 = and press RETURN
- At the prompt LWR2 = and press RETURN
- At the prompt SVCDOM = and press RETURN

Once completed, you can validate the settings by using the DIS-TDCSU:<pen> command where the PEN is your Peripheral Equipment Number of your trunk card. Press enter and the following screen should be displayed.

```
H500:  AMO TDCSU STARTED
  +------------------------ DIGITAL TRUNK (FORMAT=L) ------------------------+
  | DEV      = S1CONN         PEN      =  1-01-097-0   TGRP     = 100          |
  |--------------------------------------------------------------------------|
  | PROTVAR  = PSS1V2R        INS      = N             SRCHMODE = DSC          |
  | COTNO    = 100            COPNO    = 100           DPLN     = 0            |
  | ITR      = 0              COS      = 100           LCOSV    = 1            |
  | LCOSD    = 5              CCT      = CORNET-NQ     DESTNO   = 100          |
  | SEGMENT  = 1              DEDSCC   =               DEDSVC   = NONE         |
  | FACILITY =                DITIDX   =               SRTIDX   =              |
  | TRTBL    = GDTR           SIDANI   = N             ATNTYP   = TIE          |
  | CBMATTR  = NONE           NWMUXTIM = 10            TCHARG   = N            |
  | SUPPRESS = 0              DGTPR    =               CHIMAP   = N            |
  | ISDNIP   =                ISDNNP   =                                       |
  | PNPL2P   =                PNPL1P   =               PNPAC    =              |
  | TRACOUNT = 31             SATCOUNT = MANY          NNO      = 1  -1  -100  |
  | ALARMNO  = 0              FIDX     = 1             CARRIER  = 1            |
  | ZONE     = EMPTY          COTX     = 100           FWDX     = 5            |
  | DOMTYPE  = PRIVATE        DOMAINNO = 0             TPROFNO  =              |
  | INIGHT   =                                         CCHDL    =              |
  | UUSCCX   = 16             UUSCCY   = 8             FNIDX    = 0            |
  | CLASSMRK = EC     & G711                           SRCGRP   = 1            |
  | TCCID    =                                                                 |
  |--------------------------------------------------------------------------|
  | BCNEG    = N              BCGR     = 1             LWPAR    = 100          |
  | LWPP     =                LWLT     =               LWPS     =              |
  | LWR1     =                LWR2     =                                       |
  | SVCDOM   =                                                                 |
  | BCHAN    =  1 && 23                                                        |
  |                                                                          |
  +--------------------------------------------------------------------------+
  AMOUNT OF B-CHANNELS IN THIS DISPLAY-OUTPUT: 23

AMO-TDCSU-111       DIGITAL TRUNKS
DISPLAY COMPLETED;
```

Use the add-cot to adjust the class of trunk setting (COTNO) that you are using in the trunk configuration.

- Enter the add-cot command and press RETURN
- At the prompt COTNO = enter xxx
    - Where xxx is the class of trunk (COTNO) that you have used in the trunk configuration using the add-tdcsu command above and press RETURN

- At the prompt PAR = press RETURN
- At the prompt DEV = enter s1conn and press RETURN
- At the prompt INFO = press RETURN

Repeat these steps for each of these parameters

PRI,RCL,XFER,KNOR,CEBC&CBBN&CBFN&IEVT&BSHT&BLOC&PROV&ATRS&ROPT&TSCS&TRSC&CFOS&PINR&AOCC&CTLS&AMFC&NTON;

Once completed, you can display class of trunk configuration using the DISP-COT:100,L,,,,,; command. You should see a screen that looks like the following:

```
   COT: 100  INFO:
   DEVICE: INDEP        SOURCE: DB
   PARAMETER:
        PRIORITY FOR AC WILL BE DETERMINED FROM MESSAGE             PRI
        RECALL IF USER HANGS UP IN CONSULTATION CALL                RCL
        TRUNK CALL TRANSFER                                         XFER
        KNOCKING OVERRIDE POSSIBLE                                  KNOR
        CALL EXTEND FOR BUSY, RING OR CALL STATE                    CEBC
        NETWORKWIDE AUTOMATIC CALLBACK ON BUSY                      CBBN
        NETWORKWIDE AUTOMATIC CALLBACK ON FREE                      CBFN
        REGISTRATION OF IMPLAUSIBLE EVENTS                          IEVT
        DON'T RELEASE CALL TO BUSY HUNT GROUP                       BSHT
        END-OF-DIAL FOR BLOCK IS SET                                BLOC
        EMERGENCY OVERRIDE/DISCONNECT VIA S0/S2 LINE                PROV
        ACTIVATE TRANSIT COUNTER ADMINISTRATION FOR S0/S2 LINE      ATRS
        CONNECTION TO ROUTE OPTIMIZATION NODE                       ROPT
        TSC-SIGNALING FOR NETWORKWIDE FEATURES (MANDATORY)          TSCS
        TRUNK SENDS CALL CHARGES TO ORIGINATING NODE NUMBER         TRSC
        CALL FORWARDING PROGRAMING FOR OTHER SUBSCRIBERS            CFOS
        PIN NETWORKWIDE POSSIBLE                                    PINR
        AOC PER CALL (AUTOMATICAL OR ON REQUEST), MAND. CORNET-NQ   AOCC
        CONTROLLED TRUNK AND LINE SELECTION                         CTLS
        AUTOM.DTMF CONVERSION ON INCOM.CALL WHILE IN TALK STATE     AMFC
        NO TONE                                                     NTON

AMO-COT  -111       CLASS OF TRUNK FOR CALL PROCESSING
DISPLAY COMPLETED;
```

Use the add-cop command to configure the class of parameter setting (COPNO) that you are using in the trunk configuration.

- Enter the add-cop command and press RETURN
- At the prompt COPNO = enter xxx

    o Where xxx is the class of parameter (COPNO) that you have used in the trunk configuration using the add-tdcsu command above and press RETURN

- At the prompt PAR = enter L3AR and press RETURN
- At the prompt TRK = press RETURN
- At the prompt TOLL = press RETURN
- At the prompt DEV = press RETURN
- At the prompt INFO = press RETURN

Repeat these steps for each of these parameters

LKNQ, RRST

Once completed, you can display class of parameter configuration using the DISP-COP:100,L,,,,,,,; command. You should see a screen that looks like the following:

```
H500:  AMO COP   STARTED


   COP: 100  INFO:
   DEVICE: INDEP        SOURCE: DB
   PARAMETER:
        REGISTRATION OF LAYER 3 ADVISORIES                            L3AR
        LINK OF 2 CORNET-NQ PABX VIA INTER-LINK                       LKNQ
        REFLECT RESTART INDICATOR AND B-CHANNEL BY RESTART            RRST

AMO-COP  -111       CLASS OF PARAMETER FOR DEVICE HANDLER
DISPLAY COMPLETED;
```

Shown below is an example of the trunk routing set up on a switch using a T1 Qsig trunk. This is to be used as an example only, as many parameters will be site-specific and should be configured by a vendor technician.

```
DIS-RICHT:ALL;
H500:  AMO RICHT STARTED
+--------------------------------------------------------------------------------+
| ROUTES FOR ALL DPLN                                                  SVC = VCE |
|                                                                                |
+-------+------------------+----+-+------------------------------+-----+----+--+
|CODE   |NAME, CQMAX,      |TGRP|P|          DTMF                | LRTE|CPAR| F|
|       |DESTNO AND CPS    |CCNO|L+---+---+----------------+-----+     |    | W|
|       |          1 111112|    |B|CNV|DSP|      TEXT      |PULS |     |    | D|
|       |12345 67890 123452|    | |   |   |                |PAUSE|     |    | B|
+-------+------------------+----+-+---+---+----------------+-----+-----+----+--+
|20     |..... ..... ......| 100| | W | W |                |     |   20|    |  |
|NEUTRAL|QSIG FWD          |    | |   |   |                |     |     |    |  |
```

```
|       |PDNNO:        0   |    | |   |   |                |     |     |    |  |
|       |DESTNO   :20      |    | |   |   |                |     |     |    |  |
|       |REROUT   :YES     |    | |   |   |                |     |     |    |  |
+-------+------------------+----+-+---+---+----------------+-----+-----+----+--+


+-----------------------------------------------------------------+
| LRTE = 100    NAME = QSIG             (NEUTRAL)  LSVC = ALL     |
| DNNO =       100 PDNNO =       100 DESTNO =100                  |
| ROUTOPT = YES    REROUT = YES   PLB = NO     FWDBL = NO         |
| DTMFCNV = FIX     DTMFDSP = WITHOUT DTMFTEXT =                  |
| DTMFPULS = PP80  BUGS = LIN  ROUTATT = NO       MAINGRP =    4  |
| EMCYRTT = NO       CONFTONE = NO   RERINGRP = NO   RTENO =    4 |
| INFO =                                                          |
| NOPRCFWD = NO                                                   |
| NITO = NO                                                       |
| CLNAMEDL = NO                                                   |
+-----------------------------------------------------------------+
| TGRP = 100  LDAT  QSIG               (NEUTRAL)  SUBGROUP =    7 |
+-----------------------------------------------------------------+
| LRTE = 101    NAME = NI2              (NEUTRAL)  LSVC = ALL     |
| DNNO =       101 PDNNO =       0   DESTNO =101                  |
| ROUTOPT = YES    REROUT = YES   PLB = NO     FWDBL = NO         |
| DTMFCNV = FIX     DTMFDSP = WITHOUT DTMFTEXT =                  |
| DTMFPULS = PP80  BUGS = LIN  ROUTATT = NO       MAINGRP =    5  |
| EMCYRTT = NO       CONFTONE = NO   RERINGRP = NO   RTENO =    5 |
| INFO =                                                          |
| NOPRCFWD = NO                                                   |
| NITO = NO                                                       |
| CLNAMEDL = NO                                                   |
+-----------------------------------------------------------------+
| TGRP = 101  LDAT  NI2                (NEUTRAL)  SUBGROUP =    8 |
+-----------------------------------------------------------------+


AMO-RICHT-111       TRUNK ROUTING
DISPLAY COMPLETED;
```

In this example, the route has been set up to take calls delivered to route # 20 (as shown in the `CODE` field) and send them to trunk group (`TRGP`) 100 that we have defined in the prior steps.

5.3. Accessing Code Assigned to the Trunk

Shown below is how an access code is assigned to the trunk routing.

```
DIS-WABE:GEN,3002;
H500:  AMO WABE  STARTED
  ------------------------------------------------------------------------------
  | DIGIT INTERPRETATION                   VALID FOR ALL DIAL PLANS            |
  ------------------------------------------------------------------------------
  |                 |    CALL PROGRESS STATE     |  DIGIT   | RESERVED/CONVERT |
  |      CODE       |            1 11111 11112 22| ANALYSIS | DNI/ADD-INFO     |
  |                 |0 12345 67890 12345 67890 12|  RESULT  | *=OWN NODE       |
  ------------------------------------------------------------------------------
  | 3002            |. .**** ***** **... ..... .*| STN      |                  |
  |                 |                            |          |DESTNO  20        |
  |                 |                            |          |DNNO    0-  0-  0 |
  |                 |                            |          |PDNNO   0-  0-  0 |
  ------------------------------------------------------------------------------

AMO-WABE -111       DIALLING PLANS, FEATURE ACCESS CODES
DISPLAY COMPLETED;

DIS-RICHT:PM;
+------+----------------------+------------------------------+--------+
| IDX  |         SAN          |              NAME            |  TYPE  |
+------+----------------------+------------------------------+--------+
|     1|3002                  |QSIG                          |OTHER   |
+------+----------------------+------------------------------+--------+
```

In this example, the access code 3002 has been assigned to route calls made to it to trunk route 20 as defined in the DESTNO field. This setup allows both subscribers to call the trunk access code 3002 and get through to the gateway as well as allow subscriber station sets to be forwarded to the access code for converge under busy and ring no answer conditions.

5.4. Setting Up Subscriber Station Sets

There is no PBX-side programming for setting up the subscriber station sets. All the forwarding of the subscriber station sets is defined directly on subscriber station set using either feature access codes or the phones soft menu keys. Subscribers should be directed to set their internal and external ring no answer and busy forwarding conditions to the extension number assigned to the access code assigned to the trunk route.

5.5. Additional Comments

- Phonemail access must be configured in `RICHT` (Parameter PM)
- Index of `RICHT:PM` must be setup for every extension that needs access `(SBCSU => Parameter PMIDX)`
- `COS` of the extension must contain `TTT` (Trunk to Trunk Transfer) and `FWDEXT` (ForWarDdingEXTernal / AMO COSSU) to forward to the server
- Make sure you don’t have CFVA set for the Trunk the server is connected to (`AMO COT` - This parameter will check the availability of the forwarding target)

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	

	extension. Confirm the FAX is received in the user's inbox.		
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…)
- `teldrv prot` – This allows the collection of all ISDN messages both transmitted and received on the gateways front-end interface. 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.

- `dspif` (all keys) – This allows the collection of tone-related data. This data is helpful in cases where you think you have problems detection specific tones that should be, should not be, or are expected to be present at specific times during the call. If you do not suspect a tone-related issue, this key may be left disabled.

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.