

Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

Nortel Option 11c

with Dialogic[®] 2000 Media Gateway Series

(DMG2xxxDTI) using T1 QSIG

By : Dialogic

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READ THIS BEFORE YOU PROCEED

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Content

This document describes the configuration required to setup Nortel Option 11c and Dialogic® 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	Nortel
Model	Option 11c
Software Version	Release 25
Telephony Signaling	T1 QSIG
Additional Notes	N/A

1.2. VoIP Gateway

Gateway Vendor	Dialogic Corporation
Model	Dialogic® 2000 Media Gateway Series (DMG2xxxDT1)
Software Version	5.0.42
VoIP Protocol	SIP

1.3. Microsoft Exchange Server 2007 Unified Messaging

Version	RTM
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2. Prerequisites

2.1. Gateway Requirements

The gateway needs to support 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.

Listed below is a table of required software packages:

Package Name	Package Number
End to End Signaling package (EES)	10
Integrated Message System package (IMS)	35
Message Waiting Center package (MWC)	46
ISDN Signaling package (ISDN)	145
Advanced ISDN Network Services (NTWK)	148
1.5 Mb Primary Rate Access package (PRA)	146 or
2.0 Mb Primary Rate Interface package (PRI2)	154
International Primary Rate Interface package (IPRA)	202
Message Waiting Indication (MWI)	219
Multi Purpose Serial Data Link package (MSDL)	222
QM reference signaling point Interface package (QSIG)	263
QSIG Generic Functional protocol package (QSIGGF)	305
QSIG Supplementary Services package (QSIG-SS)	316
MCDN End to End Transparency package (MEET)	348

To connect to the PBX using T1 QSIG, you need the DTI/PRI - NTAK09BA 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.
- Configure your coder as either aLaw or uLaw as required.
- 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:

- Configuring the D-channel.
- Configuring the route data block.
- Adding the trunk members to the D-channel.
- Enabling the hardware and D-channel.
- Defining a route list and coordinated dialing plan.
- Setting up the subscribers stations.

All PBX programming is done via a serial terminal connected to the PBXs administration port.

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

Add Route Data Block	LD16
Add D-Channel	LD17
Add Trunk Members	LD14
Enable MSDL card	LD96
Enable D-Channel	LD96
Define Route List	LD86
Define Coordinated Dialing Plan	LD87

5.1. Configuring the D-Channel

Add the D-channel (ADAN) using overlay LD17. Several of the fields require site-specific entries, these are:

- Adan requires a D-channel number that is independent of other D-channel numbers on the switch.
- Cdno and Dchl require an independent trunk access code number.

The fields of this overlay that must be modified in this step are:

Type, Adan, Ctyp, Cdno, Des, Usr, Ifc, Pinx, Isdn_Mcnt, Clid, Dchl, Side.

The programming example below shows how to configure a D-Channel using LD17. For all other fields not noted in the example, press RETURN to use default values.

```
REQ chg
TYPE cfn
ADAN new dch 7
CTYP msdl
CDNO 7
PORT 1
```

```
DES
USR pri
IFC isgf
PINX_CUST 0
ISDN_MCNT 300
CLID opt0
DCHL 7
PRI
OTBF
DRAT
SIDE net
CNEG
RLS
RCAP COLP NDI CCBI CCWI PRI DV3I CTI QMWI
OVLN
OVLS
MBGA
TIMR
LAPD
```

- At the prompt REQ enter CHG to change an entry in the configuration record and press RETURN
- At the prompt ADAN enter NEW DCH XX
 - Where xx is an available d-channel number, press RETURN
- At the prompt CTYP enter MSDL press RETURN
- At the prompt CDNO enter XX
 - where xx is the card slot location of the T-1 card, press RETURN
- At the prompt DES enter XX
 - where xx is any name designation for the T1, press RETURN
- At the prompt USR enter PRI press RETURN
- At the prompt IFC enter ISGF press RETURN
- At the prompt PINX_CUST enter 0 press RETURN
- At the prompt ISDN_MCNT enter 300 press RETURN
- At the prompt CLID enter OPT0 press RETURN
- At the prompt DCHL enter XX
 - Where xx is the card slot location of the T-1 card, press RETURN
- At the prompt SIDE enter NET to set the PBX to the network side of the connection and press RETURN

5.2. Configuring the Route Data Block

Add the trunk route data block (RDB) using overlay LD16. In this overlay several of the fields require site-specific entries, these are:

- Rout requires a route number that is independent of other route numbers on the switch.
- Acod requires an independent trunk access code number.

The fields of this overlay that must be modified in this step are:

Rout, Des, Tktp, Esn, Cnvt, Sat, Rcls, Dtrk, Brip, Dgtp, Isdn, Mode, Ifc, Pni, Chty, Ctyp, Inac, Cpxs, Dapc, Intc, Dsel, Ptyp, Auto, Dnis, Dcdr, Icog, Srch, Trmb, Acod, Clen Tcpp, Biln, Sigo, Drng, Cdr, Mus, Racd, Ohq, Ohqt, Cbq, Auth, Ttbl, Plev, Alm.

The programming example below shows how to configure the Route Data Block using LD16. For all other fields not noted in the example, press RETURN to use default values.

```
REQ new
TYPE rdb
CUST 0
DMOD
ROUT 7
DES 7
TKTP pri
TKTP pra
TKTP tie
ESN no
CNVT no
SAT no
RCLS ext
DTRK yes
BRIP no
DGTP pra
SCH0341
DGTP pri
ISDN YES
MODE pra
IFC isgf
PNI 00000
CHTY bch
```

CTYP ukwn
INAC no
CPFXS yes
DAPC no
INTC no
DSEL vod
PTYT pri
AUTO no
DNIS no
DCDR no
IANI
ICOG iao
SRCH rrb
TRMB yes
STEP
ACOD 7000
CLEN 1
SCH6710
TCPP no
TARG
BILN no
SGRP
OABS
INST
IDC
ANTK
SIGO std
CNTL
DRNG no
CDR no
MUS no
RACD no
FRL
OHQ no
OHQT n
SCH0030
OHQT o

```
SCH0030
OHQT
CBQ no
AUTH no
TTBL 0
ATAN
PLEV 2
ALRM no
```

- At the prompt REQ enter NEW press RETURN
- At the prompt TYPE enter RDB press RETURN
- At the prompt CUST enter XX
 - where xx is the defined customer number press RETURN
- At the prompt DMOD press RETURN
- At the prompt ROUT enter XX
 - where xx is an available route number then press RETURN
- At the prompt DES enter XX
 - where xx is any name designation for the trunk route press RETURN
- At the prompt TKTP enter TIE press RETURN
- At the prompt ESN enter NO press RETURN
- At the prompt CNVT enter NO press RETURN
- At the prompt SAT enter NO press RETURN
- At the prompt RCLS enter EXT press RETURN
- At the prompt DTRK enter YES press RETURN
- At the prompt BRIP enter NO press RETURN
- At the prompt DGTP enter PRI press RETURN
- At the prompt ISDN enter YES press RETURN
- At the prompt MODE enter PRA press RETURN
- At the prompt IFC type ISGF press RETURN
- At the prompt PNI enter 00000 press RETURN
- At the prompt CHTY enter BCH press RETURN
- At the prompt CTYP enter UKWN press RETURN
- At the prompt INAC enter NO press RETURN
- At the prompt CPFXS enter YES press RETURN
- At the prompt DAPC enter NO press RETURN
- At the prompt INTC enter NO press RETURN
- At the prompt DSEL enter VOD press RETURN
- At the prompt PTYP enter PRI press RETURN
- At the prompt AUTO enter NO press RETURN
- At the prompt DNIS enter NO press RETURN
- At the prompt DCDR enter NO press RETURN
- At the prompt ICOG enter IAO press RETURN
- At the prompt SRCH enter RRB press RETURN
- At the prompt TRMB enter YES press RETURN

- At the prompt ACOD enter XXXX
 - where xxxxx is an available trunk access code number the same length as the phone extension numbers and press RETURN
- At the prompt CLEN enter 1 press RETURN
- At the prompt TCPP enter NO press RETURN
- At the prompt BILN enter NO press RETURN
- At the prompt SIGO enter STD press RETURN
- At the prompt DRNG enter NO press RETURN
- At the prompt CDR enter NO press RETURN
- At the prompt MUS enter NO press RETURN
- At the prompt RACD enter NO press RETURN
- At the prompt OHQ enter NO press RETURN
- At the prompt OHQT enter 00 press RETURN
- At the prompt CBQ enter NO press RETURN
- At the prompt AUTH enter NO press RETURN
- At the prompt TTBL enter 0 press RETURN
- At the prompt PLEV enter 2 press RETURN
- At the prompt ALRM enter NO press RETURN

5.3. Adding Trunk Members to the D-Channel

Now that the trunk and D-Channel are created, you must assign each member of the trunk to this route group using overlay LD14.

The fields of this overlay that must be modified in this step are:

Type, TN, Cust, Cden, Trk, Pcml, Ncos, Rtmb, Tgar, Ast, Iapg, Cls.

The programming example below shows how to add trunk members to the D-Channel using LD14. This needs to be repeated for each B-Channel you are adding to the D-Channel (23 times per span). For all other fields not noted in the example, press RETURN to use default values.

```

REQ  new
TYPE tie
TN   7 1
DES
PDCA
PCML
CUST 0
NCOS 0
RTMB 7 1
B-CHANNEL SIGNALING
MNDN
TGAR 1

```

```
AST
CLS unr dtn
TKID
```

- At the prompt TYPE enter TIE press RETURN
- At the prompt TN enter XX XX
 - where xx xx is the slot and port number of each channel of the T1 hardware press RETURN
- At the prompt CUST enter XX
 - where xx is the defined customer number press RETURN
- At the prompt CDEN press RETURN
- At the prompt TRK enter PRI press RETURN
- At the prompt PCML press RETURN
- At the prompt NCOS enter 0 press RETURN
- At the prompt RTMB enter XX XX
 - where xx xx is the rout number and member defined previously in LD16 press RETURN
- At the prompt TGAR enter 1 press RETURN
- At the prompt AST enter NO press RETURN
- At the prompt IAPG enter 0 press RETURN
- At the prompt CLS enter UNR DTN press RETURN

5.4. Enabling the MSDL Board and D-Channel

To use the newly added card and D-Channel, you need to enable both of them using overlay LD96.

- Enter the command `enl msdl xx`
 - where xx is the D-Channel number defined in LD17 and press RETURN
- Enter the command `enl dch xx`
 - Where xx is the D-Channel; number assigned in LD17 and press RETURN

5.5. Defining a Route List

Use overlay LD86 to define a route list.

The fields of this overlay that must be modified in this step are:

Req, Cust, Feat, Rli, Entr, Lter, Rout, Tod, Cnv, Exp, Frl, Dmi, Fci, Fsni, Ohq, Cbq, Iset, Mfrl, Ovll.

The programming example below shows how to define a rout list using LD86. For all other fields not noted in the example, press RETURN to use default values.

```
>ld 86
ESN000
REQ new
```

```
CUST 0
FEAT
FEAT rlb
RLI 1
ENTR 0
LTER no
ROUT 7
TOD
CNV no
EXP no
FRL 0
DMI 0
FCI 0
FSNI 0
SBOC
OHQ no
CBQ no
ENTR 0
ESN001
ENTR
ISET 0
NALT
MFRL 0
OVLL 0
```

- At the prompt REQ enter NEW press RETURN
- At the prompt CUST enter XX
 - where xx is the defined customer number press RETURN
- At the prompt FEAT enter RLB press RETURN
- At the prompt RLI enter X
 - where x is the next available route list index number press RETURN
- At the prompt ENTR enter X
 - where x is the entry number for the NARS/BARS route list and press RETURN
- At the prompt LTER enter NO press RETURN
- At the prompt ROUT enter X
 - where x is the route number defined in the previous steps and press RETURN
- At the prompt CNV enter NO press RETURN
- At the prompt EXP enter NO press RETURN
- At the prompt FRL enter 0
 - Where Facility restriction level it should be set as low as possible. press RETURN

- At the prompt DMI enter 0 press RETURN
- At the prompt FCI enter 0 press RETURN
- At the prompt FSNI enter 0 press RETURN
- At the prompt OHQ enter NO press RETURN
- At the prompt CBQ enter NO press RETURN
- At the prompt ISET enter 0 press RETURN
- At the prompt MFRL enter 0 press RETURN
- At the prompt OVLL enter 0 press RETURN

5.6. Defining the Coordinated Dialing Plan

Use overlay LD87 to define your CDP (Coordinated Dialing Plan). This is the method used to be able to access the trunk as a forwarding point for station sets using an extension number.

The fields of this overlay that must be modified in this step are:

Req, Cust, Feat, Type, Dsc, Flen, Dsp, Rli.

The programming example below shows how to define a CDP using LD87. For all other fields not noted in the example, press RETURN to use default values.

```
>ld 87
ESN000
REQ new
CUST 0
FEAT cdp
TYPE dsc
DSC 5000
FLEN 4
DSP 1sc
RLI 1
NPA
NXX
DSC
```

- At the prompt REQ enter NEW press RETURN
- At the prompt CUST enter XX
 - where xx is the defined customer number press RETURN
- At the prompt FEAT enter CDP press RETURN
- At the prompt TYPE enter DSC press RETURN
- At the prompt DSC enter XXXX

- Where `xxxx` is the extension you want to use to access the trunk route list and press RETURN
- At the prompt `FLEN` enter `X`
 - Where `x` is the length of the extensions in this CDP and press RETURN
- At the prompt `DSP` enter `LSC` press RETURN
- At the prompt `RLI` enter `X`
 - Where `x` is the rout list index created in LD86 and press RETURN

5.7. Setting Up Subscriber Station Sets

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

```
>LD 11
REQ   CHG
TYPE  2008
TN    0 1 8 3
ECHG
DES
FDN   5000
TGAR
HUNT  5000
NCOS
RNPG
SSU
CLS   HTA FNA MWA CFTA SFA
EFD   5000
EHT   5000
.
.
.
```

Important notes about the above programming:

1. The `FDN` field is where you specify the destination for this station set to forward to under ring no answer conditions. It should be configured to send the calls to the CDP defined to access the trunk route list.
2. The `CLS` field is required to have `HTA`, `FNA`, `MWA`, `CFTA` and `SFA` configured. If these are not configured properly the remainder of the programming is not going to provide you with the proper prompts to continue.

3. The EFD field (only seen if the CLS has been set up properly) is where you specify the destination for external calls to the station to forward under ring no answer conditions. It should be configured to send the calls to the CDP defined to access the trunk route list.
4. The HUNT field is where you specify the destination for internal calls to the station to forward under busy conditions. It should be configured to send the calls to the CDP defined to access the trunk route list.
5. The EHT field (only seen if the CLS has been set up properly) is where you specify the destination for external calls to the station to forward under busy conditions. It should be configured to send the calls to the CDP defined to access the trunk route list.
6. The MWA setting in the CLS field allows this station to make use of its MWI light. If this is not configured the stations MWI lamp will not work.

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

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

Important notes about the above programming:

1. The FDN field is where you specify the destination for this station set to forward to under ring no answer conditions. It should be configured to send the calls to the CDP defined to access the trunk route list.

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

5.8. Additional Comments

N/A

6. Exchange 2007 UM Validation Test Matrix

The following table contains a set of tests for assessing the functionality of the UM core feature set. The results are recorded as either:

- Pass (**P**)
- Conditional Pass (**CP**)
- Fail (**F**)
- Not Tested (**NT**)
- Not Applicable (**NA**)

Refer to:

- Appendix for a more detailed description of how to perform each call scenario.
- Section 6.1 for detailed descriptions of call scenario failures, if any.

No.	Call Scenarios (see appendix for more detailed instructions)	(P/CP/F/NT)	Reason for Failure (see 6.1 for more detailed descriptions)
1	Dial the pilot number from a phone extension that is NOT enabled for Unified Messaging and logon to a user's mailbox. Confirm hearing the prompt: "Welcome, you are connected to Microsoft Exchange. To access your mailbox, enter your extension..."	P	
2	Navigate mailbox using the Voice User Interface (VUI).	P	
3	Navigate mailbox using the Telephony User Interface (TUI).	P	
4	Dial user extension and leave a voicemail.		
4a	Dial user extension and leave a voicemail from an internal extension. Confirm the Active Directory name of the calling party is displayed in the sender field of the voicemail message.	P	
4b	Dial user extension and leave a voicemail from an external phone. Confirm the correct phone number of the calling party is displayed in the sender field of the voicemail message.		Not tested. The external trunk available would not allow the phone to follow its forwarding path so this could not be tested.

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.	CP	No calling party id. The message was form anonymous. The transfer is a hair-pin, NOT a path replacement.
6c	Call Transfer by Directory Search when the called party does not answer. Confirm the call is routed to the called party's voicemail.	CP	No calling party id. The message was form anonymous. The transfer is a hair-pin, NOT a path replacement.
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		Not tested. The external trunk available would not allow the phone to follow its forwarding

	<p>extension.</p> <p>Confirm the FAX is received in the user's inbox.</p>		<p>path so this could not be tested.</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>		<p>5.0.42 gateway firmware does not yet implement TLS so this feature was not tested.</p>
10b	<p>Dial a user extension and leave a voicemail.</p> <p>Confirm the user receives the voicemail.</p>		<p>5.0.42 gateway firmware does not yet implement TLS so this feature was not tested.</p>
10c	<p>Send a test FAX message to user extension.</p> <p>Confirm the FAX is received in the user's inbox.</p>		<p>5.0.42 gateway firmware does not yet implement TLS so this feature was not tested.</p>
11	<p>Setup G.723.1 on the gateway. (If already using G.723.1, setup G.711 A Law or G.711 Mu Law for this step).</p> <p>Dial the pilot number and confirm the UM system answers the call.</p>	P	
12	<p>Setup Message Waiting Indicator (MWI).</p> <p>Geomant offers a third party solution: MWI 2007. Installation files and product documentation can be found on Geomant's MWI 2007 website.</p>		<p>The Geomant software was not available at the time of validation so this feature was not tested.</p>
13	<p>Execute Test-UMConnectivity.</p>	NT	
14	<p>Setup and test fail-over configuration on the IP-PBX to work with two UM servers.</p>	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.