

Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

Mitel 3300

with Dialogic® 2000 Media Gateway Series

(DMG2xxxDTI) using T1 QSIG

By : Dialogic

Updated Since : 12/19/2007

READ THIS BEFORE YOU PROCEED

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Content

This document describes the configuration required to setup Mitel 3300 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	Mitel
Model	3300
Software Version	3300 Universal NSU - 50001270
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
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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.

To connect to the PBX using T1 QSIG, you need the 3300 Universal NSU - 50001270 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 messaging system are as follows:

- Configuring hardware and class of service.
- Configuring ISDN-PRI interface.
- Configuring ARS options.
- Setting up subscriber station sets.

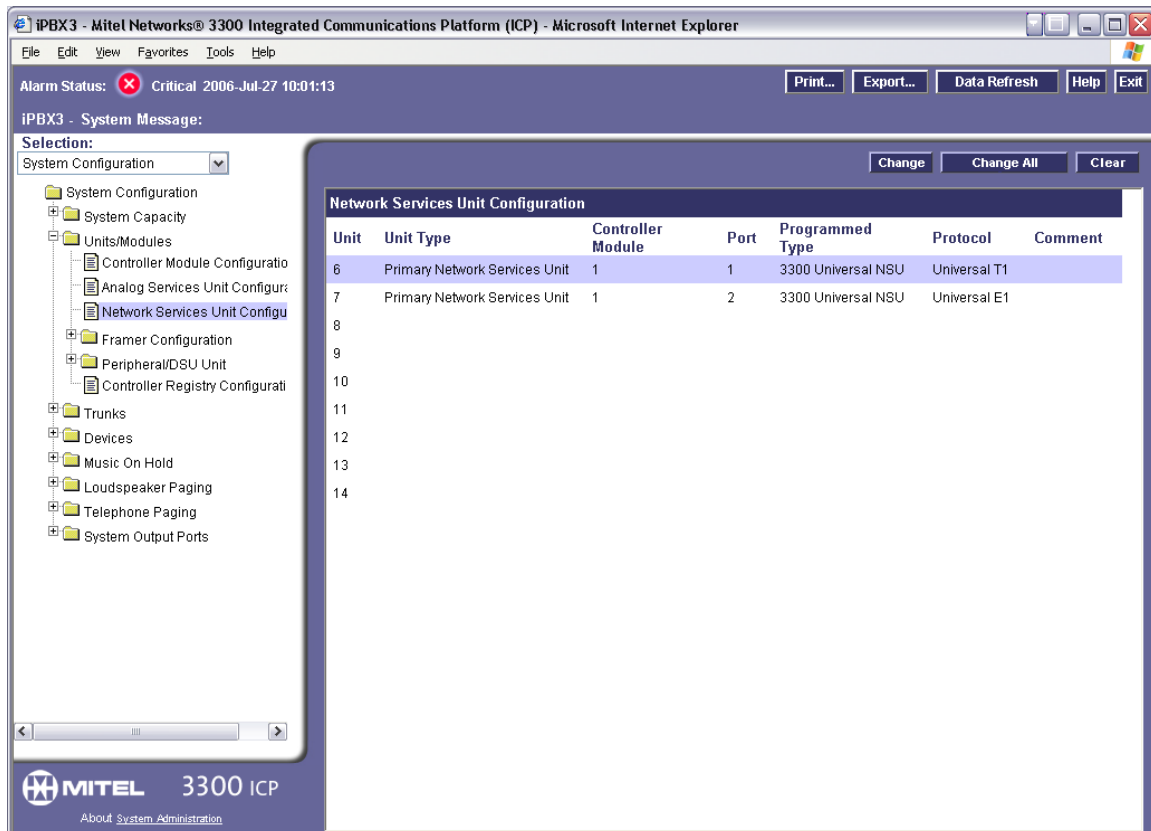
All PBX programming is done via a web browser by connecting to the network port of the PBX.

5.1. Configure Hardware and Class of Service

Use the Units and Modules Network Services Unit Configuration menu selection to configure a Network Service Unit.

This command sets the options on an installed NSU so it can be configured for a specific type and protocol to be used with the gateway.

Select an NSU from the list and click the Change button to configure the options.



The screenshot shows the IPBX3 web interface in Microsoft Internet Explorer. The browser title is "IPBX3 - Mitel Networks® 3300 Integrated Communications Platform (ICP) - Microsoft Internet Explorer". The interface includes a menu on the left, a main content area with a table, and a footer with the Mitel logo and "3300 ICP About System Administration".

Alarm Status: ✖ Critical 2006-Jul-27 10:01:13

IPBX3 - System Message:

Selection: System Configuration

Buttons: Change, Change All, Clear

Unit	Unit Type	Controller Module	Port	Programmed Type	Protocol	Comment
6	Primary Network Services Unit	1	1	3300 Universal NSU	Universal T1	
7	Primary Network Services Unit	1	2	3300 Universal NSU	Universal E1	
8						
9						
10						
11						
12						
13						
14						

See the screen below for the selected options.

-- Web Page Dialog

Network Services Unit Configuration

Unit: 6

Unit Type: Primary Network Services Unit

Controller Module: 1

Port: 1 2

Programmed Type: 3300 Universal NSU

Protocol: Universal T1

Comment:

Save Cancel

Before moving on to the next task be sure to click the Save button.

Configure a Class of Service Template for the Trunk. Shown below is the listing of all the enabled and disabled classes of service that will be configured on the trunk interface in this example. Yours may vary depending on site requirements but keep in mind that disabling certain classes of service will have an effect on certain available features.

Class of Service Options Assignment

Class Of Service Number: 1

Comment: Radbrook Config

Account Code Verified:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
ACD Silent Monitor Accept:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
ACD Silent Monitor Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
ACD Silent Monitor Notification:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
ANI/DNIS/ISDN Number Delivery Trunk:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Auto Answer Allowed:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Autovon Auto-preemption:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Autovon Trunk:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Brokers Call:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Busy Override Security:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Call Announce Line:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Forwarding Accept:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Forwarding (External Destination):	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Forwarding (Internal Destination):	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Forward Override:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Call Hold:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Hold Remote Retrieve:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Hold - Retrieve with Hold Key:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Pickup Dialed Accept:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Pickup Directed Accept:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Privacy:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Call Reroute after CFFM to Busy Destination:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Call Waiting Swap:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Calling Name Display - Internal - ONS:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Calling Number Display - Internal - ONS:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Campon Tone Security:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Check COR after PSTN Dial Tone:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Clear All Features Remote:	<input checked="" type="radio"/> No	<input type="radio"/> Yes

Conference Call:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
COV/ONS/E&M Voice Mail Port:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
DASS II OLI/TLI Provided:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Dialled Night Service:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Disable Send Message:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Display ANI/ISDN Calling Number Only:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Display ANI/DNIS/ISDN Calling/Called Number:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Display Caller ID on multicall/keylines:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Display DNIS/Called Number Before Digit Modification:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Display Dialed Digits during Outgoing Calls:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Display Held Call ID on Transfer:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Do Not Disturb:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Do Not Disturb - Access to Remote Phones:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Do Not Disturb Permanent:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Emergency Call Notification - Audio:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Emergency Call Notification - Visual:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Enable Call Duration Limit on External Calls:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Enable Call Duration Limit on Internal Calls:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Executive Busy Override:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
External Trunk Standard Ringback:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Flexible Answer Point:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Follow 2nd Alternate Reroute for Recall to Busy ACD Agent:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Forced Verified Account Code:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Forced Non-Verified Account Code:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Group Call Forward Follow Me Accept:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Group Call Forward Follow Me Allow:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Group Page Accept:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Group Page Allow:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Handset Volume Adjustment Saved:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Handsfree AnswerBack Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
HCI/CTI/TAPI Call Control Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes

HCI/CTI/TAPI Monitor Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Head Set Switch Mute:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Hot Desk Remote Logout Enabled:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Hot Desk Login Accept:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Hotel Room Extension:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Hotel Room Monitor Setup Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Hotel Room Monitoring Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Hotel/Motel Room Personal Wakeup Call Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Hotel/Motel Room Remote Wakeup Call Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Individual Trunk Access:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Keep TelDir Entry on Check Out:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Local Music On Hold source:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Loudspeaker Pager Override:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Loudspeaker Pager Equivalent Zone Override Security:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Message Waiting:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Message Waiting Audible Tone Notification:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Message Waiting Deactivate On Off-Hook:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Message Waiting Inquire:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Multiline Set Loop Test:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Multiline Set Message Center Remote Read Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Multiline Set Music:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Multiline Set On-hook Dialing:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Multiline Set Phonebook Allowed:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Multiline Set Voice Mail Callback Message Erasure Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Name Suppression on outgoing Trunk Call:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Non DID Extension:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Non-Prime Public Network Identity:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Non Verified Account Code:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Off-Hook Voice Announce Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
ONS CLASS/CLIP: Message Waiting Activate/Deactivate:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
ONS CLASS/CLIP: Set:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
ONS CLASS/CLIP: Visual Call Waiting:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
ONS/OPS Internal Ring Cadence for External Callers:	<input checked="" type="radio"/> No	<input type="radio"/> Yes

Override Interconnect Restriction on Transfer:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Pager Access All Zones:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Pager Access Individual Zones:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Privacy Released:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Public Network Access via DPNSS:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Public Network Identity Provided:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Public Network To Public Network Connection Allowed:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Public Trunk:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
R2 Call Progress Tone:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Record-A-Call Active:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Record-A-Call - Start Recording Automatically:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Record-A-Call - Save Recording on Hang-up:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Recorded Announcement Device:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Recorded Announcement Device - Advanced:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Redial Facilities:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Ringing Line Select:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
SC1000 Attendant Basic Function Key:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
SMDR External:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
SMDR Internal:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Speak@Ease Preferred:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Suite Services Enabled:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Suppress Simulated CCM after ISDN Progress:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Third Party Call Forward Follow Me Accept:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Third Party Call Forward Follow Me Allow:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Timed Reminder Allowed:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Trunk Calling Party Identification:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Trunk Flash Allowed:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Use Held Party Device for Call Re-routing:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Use Called Party Call Hold Timer:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Voice Mail Softkey:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Account Code Length:	<input type="text" value="12"/>	
After Answer Display Time:	<input type="text"/>	

Answer Plus Delay To Message Timer:	20
Answer Plus Expected Off-hook Timer:	30
Answer Plus Message Length Timer:	10
Answer Plus System Reroute Timer:	0
Attendant Busy Out Timer:	10
Auto Campon Timer:	10
Autovon Precedence:	4
Busy Tone Timer:	30
Call Duration:	10
Call Duration Forced Cleardown Timer:	0
Call Forward - Delay:	0
Call Forward No Answer Timer:	15
Call Hold Timer:	30
Campon Recall Timer:	10
Delay Ring Timer:	10
Dialing Conflict Timer:	3
Display Caller ID On Multicall/Keylines Timer:	5
Emergency Call - Audio Level for Set:	Ringer ▼
First Digit Timer:	15
Inter Digit Timer:	10
Lockout Timer:	45
ACD 2000 Logout Agent No Answer Timer:	15
Message Waiting Ringing Start Time Hour:	▼
Message Waiting Ringing Start Time Minute:	▼
Message Waiting Ringing Stop Time Hour:	▼

Message Waiting Ringing Stop Time Minute:
No Answer Recall Timer:
ONS VMail-Delay Dial Tone Timer:
Ringing Timer:
Work Timer:
Key A:
Key B:
Key C:
Key D:

Before moving on to the next task be sure to click the Save button.

5.2. Configuring ISDN-PRI Interface

Use the ISDN PRI Link Descriptor Assignment menu selection to build a template that sets the various options for a trunk interface such as framing, coding and interface type.

Use the Add or Change buttons to either build a new Link Descriptor or modify an existing Link Descriptor.

The screenshot shows the IPBX3 configuration interface in Microsoft Internet Explorer. The left sidebar displays a tree view of system configuration options, with 'ISDN-PRI' selected. The main content area shows a table of Link Descriptor Assignments and a detailed configuration view for the selected descriptor.

Number	Address for Message Control	BER - Maintenance Limit, 10 ^{ms} .n	BER - Service Limit, 10 ^{ms} .n	Data Call Alternate Digit Inversion
1	A	4	3	Yes
2	A	4	3	Yes
3	A	4	3	Yes
4	A	4	3	Yes

Link Descriptor Assignment	
Number:	1
Address for Message Control:	A
BER - Maintenance Limit, 10 ^{ms} .n:	4
BER - Service Limit, 10 ^{ms} .n:	3
Data Call Alternate Digit Inversion:	Yes
Framing Losses in 24 hrs - Maintenance Limit:	255
Framing Losses in 24 hrs - Service Limit:	9000
Integrated Digital Access:	ISDN NODE
Vendor Inter-working Type (Philips SOPHO):	No
Satellite Link Delay:	No
Slip Data - Maintenance Limit (slip/24hr):	6000

See the screen below for the selected options.

The screenshot shows a web-based configuration dialog box titled "Link Descriptor Assignment". The dialog contains the following fields and options:

- Number:** 1
- Address for Message Control:** A
- BER - Maintenance Limit, 10⁻ⁿ-n:** 4
- BER - Service Limit, 10⁻ⁿ-n:** 3
- Data Call Alternate Digit Inversion:** No Yes
- Framing Losses in 24 hrs - Maintenance Limit:** 255
- Framing Losses in 24 hrs - Service Limit:** 9000
- Integrated Digital Access:** ISDN NODE
- Vendor Inter-working Type (Philips SOPHO):** No Yes
- Satellite Link Delay:** No Yes
- Slip Rate - Maintenance Limit (slips/24hr.):** 5000
- Slip Rate - Service Limit (slips/24hr.):** 7000
- Alarm Debounce Timer - Service Limit (millisec.):** 500
- Voice Encoding:** Invert
- Data Encoding:** Invert
- QSIG Private Network Access:** No Yes
- Digital Link Fault Delay Timer (sec.):** 240
- Termination Mode:** LT NT
- Send Malicious Call Indication to PSTN for Tagged Calls:** No Yes
- T1 Only:**
 - B8ZS Zero Code Suppression:** No Yes
 - Operation Mode:** CSU
 - CSU Tx Line Build-Out (dB.):** 0
 - DSX-1 Line Length (Ft.):** 0-133
 - Extended Super Frame:** No Yes
 - Inverted D channel (DPNSS only):** No Yes
- E1 Only:**
 - CRC-4 Enabled:** No Yes
 - E1 Line Length (Ft.):** 0-133
 - E1 Impedance (Ohms):** 75 120

At the bottom right of the dialog are "Save" and "Cancel" buttons.

Before moving on to the next task be sure to click the Save button.

Use the ISDN-PRI Digital Link Assignment menu selection to connect a physical location (port, unit, shelf, slot and link) within the PBX with the NSU and Link Descriptor configured in the previous steps.

Select a Controller Module and click the Change button to configure the options.

The screenshot shows the IPBX3 - System Message configuration interface. The left sidebar contains a tree view of system configuration options, with 'Digital Link Assignment' selected under 'ISDN-PRI'. The main content area displays a table titled 'Digital Link Assignment' with the following data:

Controller Module	Port	Unit	Shelf	Slot	Link	Interface Type	Digital Link Descriptor	Comment
1	1	6	1	1	1	UNIVERSAL T1	1	T1 QSIG
1	1	6	1	1	2	UNIVERSAL T1	3	T1 NI2
1	2	7	1	1	1	UNIVERSAL E1	2	E1 QSIG
1	2	7	1	1	2	UNIVERSAL E1	4	DK &RS

See the screen below for the selected options.

The screenshot shows a web browser dialog box titled "-- Web Page Dialog" with a close button in the top right corner. The main content area is titled "Digital Link Assignment" and contains the following configuration fields:

Controller Module:	1
Port:	1
Unit:	6
Shelf:	1
Slot:	1
Link:	1
Interface Type:	UNIVERSAL T1
Digital Link Descriptor:	<input type="text" value="T1"/>
Comment:	<input type="text" value="T1 QSIG"/>

At the bottom right of the dialog box, there are two buttons: "Save" and "Cancel".

Before moving on to the next task be sure to click the Save button.

Use the ISDN-PRI MSDN-DPNSS-DASSII Trunk Circuit Descriptor menu selection to assign direction and protocols to the individual trunk cards in the PBX.

Select a trunk number and click the Change button to configure the options.

The screenshot shows the Mitel Networks 3300 ICP web interface in Microsoft Internet Explorer. The browser title is "IPBX3 - Mitel Networks® 3300 Integrated Communications Platform (ICP) - Microsoft Internet Explorer". The page displays an "Alarm Status: Critical 2006-Jul-27 10:01:13" and a "System Message" section. A navigation menu on the left includes "System Configuration" and "MSDN-DPNSS-DASSII Trunk Circuit Des". The main content area shows a table titled "MSDN-DPNSS-DASSII Trunk Circuit Descriptor" with the following data:

Number	Card Type	Dual Seizure Priority	Far End Connection	Signalling Protocol	ISDN BRI Mode
1	UNIVERSAL T1	Incoming	Local Office	MSDN-DPNSS	
2	UNIVERSAL E1	Incoming	Local Office	MSDN-DPNSS	
3	UNIVERSAL E1	Incoming	Local Office	MSDN-DPNSS	

The interface also includes buttons for "Add", "Change", "Copy", and "Delete", and a "Go to" search field. The Mitel logo and "3300 ICP" are visible at the bottom left.

See the screen below for the selected options.

-- Web Page Dialog

MSDN-DPNSS-DASSII Trunk Circuit Descriptor

Number: 1

Card Type: UNIVERSAL T1

Dual Seizure Priority: Incoming Outgoing

Far End Connection: Local Office

Signalling Protocol: MSDN-DPNSS DASS II

ISDN BRI Mode:

Save Cancel

Before moving on to the next task be sure to click the Save button.

Use the ISDN-PRI Trunk Service Assignment menu selection to set up a template that contains various service levels, for example the trunks class of service that was previously defined, into a template that will latter on be assigned to a trunk.

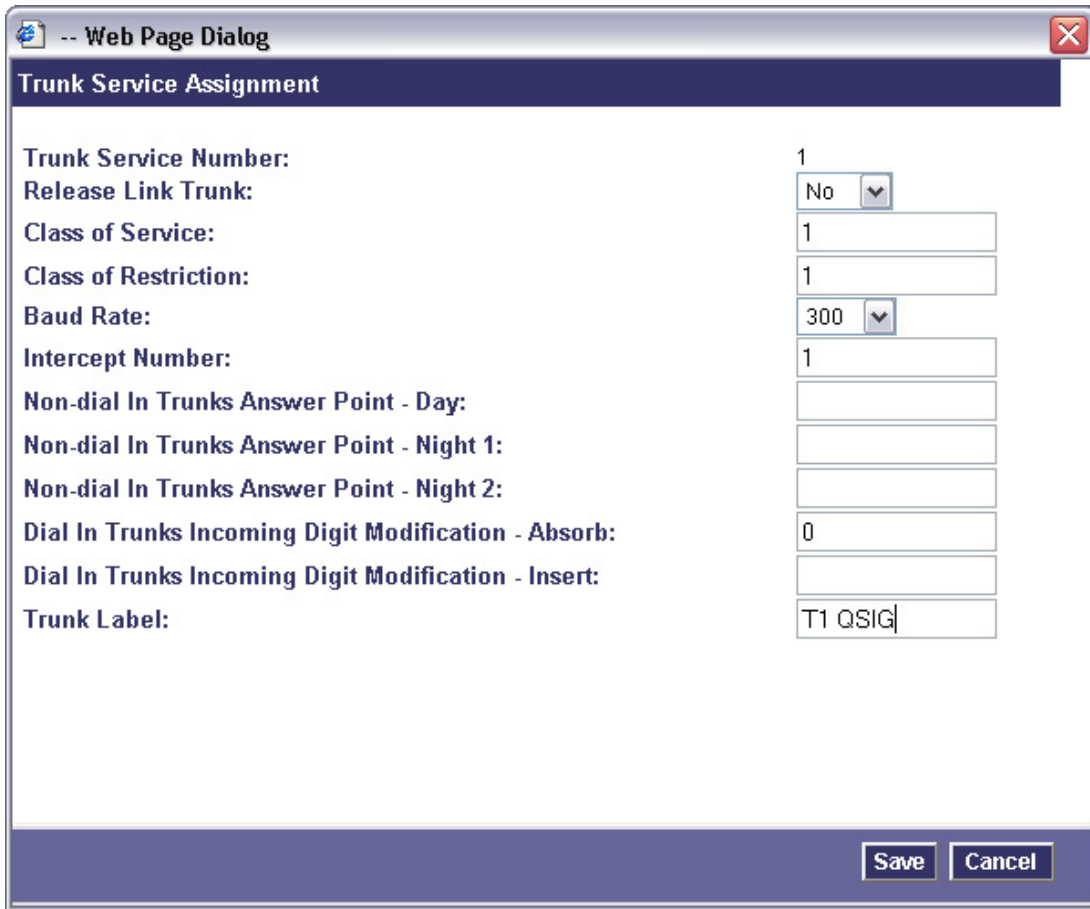
Select the trunk service number and click the Change button to configure the options.

The screenshot shows the IPBX3 configuration interface in Microsoft Internet Explorer. The left sidebar displays a tree view of system configuration options, with 'ISDN-PRI' > 'Trunk Service Assignment' selected. The main content area displays a table of trunk service assignments and a configuration summary below it.

Trunk Service Number	Release Link Trunk	Class of Service	Class of Restriction	Baud Rate	Intercept Number	Trunk Label
1	No	1		300	1	Radbrook
2	No	1	1	300	1	3 to 4
3	No	1	1	300	1	3 to 5
4	No	1	1	300	1	ASU Lp Bk
5	No	5	1	300	1	E1 QSIG
6	No	6	1	300	1	T1 NI2
7	No	1	1	300	1	
8	No	1	1	300	1	
9	No	1	1	300	1	
10	No	5	1	300	1	DK &RS

Trunk Service Number:	1
Release Link Trunk:	No
Class of Service:	1
Class of Restriction:	1
Baud Rate:	300
Intercept Number:	1
Non-dial In Trunks Answer Point - Day:	
Non-dial In Trunks Answer Point - Night 1:	
Non-dial In Trunks Answer Point - Night 2:	
Dial In Trunks Incoming Digit Modification - Absorb:	0
Dial In Trunks Incoming Digit Modification - Insert:	

See the screen below for the selected options.



The image shows a web browser dialog window titled "-- Web Page Dialog" with a close button in the top right corner. The main content area is titled "Trunk Service Assignment" and contains the following fields and values:

Trunk Service Number:	1
Release Link Trunk:	No
Class of Service:	1
Class of Restriction:	1
Baud Rate:	300
Intercept Number:	1
Non-dial In Trunks Answer Point - Day:	
Non-dial In Trunks Answer Point - Night 1:	
Non-dial In Trunks Answer Point - Night 2:	
Dial In Trunks Incoming Digit Modification - Absorb:	0
Dial In Trunks Incoming Digit Modification - Insert:	
Trunk Label:	T1 QSIG

At the bottom right of the dialog, there are two buttons: "Save" and "Cancel".

Before moving on to the next task be sure to click the Save button.

Use the ISDN-PRI Digital Trunk Assignment menu selection to configure the individual trunk members and assign them the defined Trunk Service Assignment template and Trunk Circuit Descriptor template that were configured in the previous steps. This gets done for each member of a trunk interface.

Select each individual trunk member and click the Change button to configure the options.

The screenshot shows the IPBX3 - Mitel Networks 3300 ICP web interface. The left-hand navigation pane is expanded to show the 'Digital Trunk Assignment' option under the 'ISDN-PRI' category. The main content area displays a table of trunk assignments with the following data:

Cabinet	Shelf	Slot	Circuit	Card Type	Trunk Number
6	1	1	1	UNIVERSAL T1	100
6	1	1	2	UNIVERSAL T1	101
6	1	1	3	UNIVERSAL T1	102
6	1	1	4	UNIVERSAL T1	103
6	1	1	5	UNIVERSAL T1	104
6	1	1	6	UNIVERSAL T1	105
6	1	1	7	UNIVERSAL T1	106
6	1	1	8	UNIVERSAL T1	107
6	1	1	9	UNIVERSAL T1	108
6	1	1	10	UNIVERSAL T1	109

Below the table, a summary of the selected entry is shown:

Digital Trunk Assignment

Cabinet: 6
 Shelf: 1
 Slot: 1
 Circuit: 1
 Card Type: UNIVERSAL T1
 Trunk Number: 100
 Trunk Service Number: 1
 DTS Service Number:
 Circuit Descriptor Number: 1
 Interconnect Number: 1

See the screen below for the selected options.

Range Programming -- Web Page Dialog

Change Range Programming - Digital Trunk Assignment

This form allows you to change one or more records, starting at the following record:

Cabinet	Shelf	Slot	Circuit	Card Type	Trunk Number	Trunk Service Number	DTS Service Number
6	1	1	1	UNIVERSAL T1	100	1	

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Cabinet:	-	6	-
Shelf:	-	1	-
Slot:	-	1	-
Circuit:	-	1	-
Card Type:	-	UNIVERSAL T1	-
Trunk Number:	Change to	<input type="text" value="100"/>	<input type="text"/>
Trunk Service Number:	Change to	<input type="text" value="1"/>	<input type="text"/>
DTS Service Number:	Change to	<input type="text"/>	<input type="text"/>
Circuit Descriptor Number:	Change to	<input type="text" value="1"/>	<input type="text"/>
Interconnect Number:	Change to	<input type="text" value="1"/>	<input type="text"/>

Before moving on to the next task be sure to click the Save button.

5.3. Configuring ARS Options

Use the ARS Trunk Group Assignment menu selection to edit your trunk group configuration.

Use the Add or Change buttons to either build a new trunk group or modify an existing group.

The screenshot displays the Mitel Networks 3300 ICP web interface in Microsoft Internet Explorer. The page title is "iPBX3 - Mitel Networks® 3300 Integrated Communications Platform (ICP) - Microsoft Internet Explorer". The interface includes a navigation tree on the left, a main content area with a table, and a footer with the Mitel logo and "3300 ICP About System Administration".

Alarm Status: Critical 2006-Jul-27 10:01:13

Selection: System Administration

Trunk Group Assignment

Trunk Group Number	Hunt Mode	Trunk Group Busy RAD	Maximum Network Hop	Comments
15	Terminal			
21	Terminal			
22	Terminal			
23	Terminal			
24	Terminal			

Page 1 of 3

Go to: [] value: [] Go

Trunk Group Members

Member	Trunk Number
1	100
2	101
3	102
4	103
5	104
6	105
7	106
8	107
9	108
10	109

See the screen below for the selected options.

The screenshot shows a web dialog box titled "Range Programming -- Web Page Dialog" with a sub-header "Change Range Programming - Trunk Group Assignment". It includes a "Help" button and a message: "This form allows you to change one or more records, starting at the following record:". Below this is a table with columns: Trunk Group Number, Hunt Mode, Trunk Group Busy RAD, Maximum Network Hop, and Comments. The first row shows values: 21, Terminal, (empty), (empty), and (empty). The dialog has two main sections: "1. Enter the number of records to change:" with a text box containing "1", and "2. Define the Change Range Programming Pattern:". This second section contains a table with headers: Field Name, Change action, Value to change, and Increment by. The rows are: Trunk Group Number (Change to, 21, increment box), Hunt Mode (Change to, Terminal selected, Circular unselected, -), Trunk Group Busy RAD (Change to, empty, increment box), Maximum Network Hop (Change to, empty, increment box), and Comments (Change to, empty, -). At the bottom are "Preview", "Save", and "Cancel" buttons.

Click the Save button when you are finished editing the options and wish to save your configuration. When you have configured a trunk group you use the Add Member button to add individual trunk members to the group.

The screenshot shows a web dialog box titled "Range Programming -- Web Page Dialog" with a sub-header "Change Range Programming - Trunk Group Members". It includes a "Help" button and a message: "This form allows you to change one or more records, starting at the following record:". Below this is a table with columns: Trunk Number and (empty). The first row shows values: 100, and (empty). The dialog has two main sections: "1. Enter the number of records to change:" with a text box containing "1", and "2. Define the Change Range Programming Pattern:". This second section contains a table with headers: Field Name, Change action, Value to change, and Increment by. The rows are: Trunk Number (Change to, 100, increment box). At the bottom are "Preview", "Save", and "Cancel" buttons.

Before moving on to the next task be sure to click the Save button.

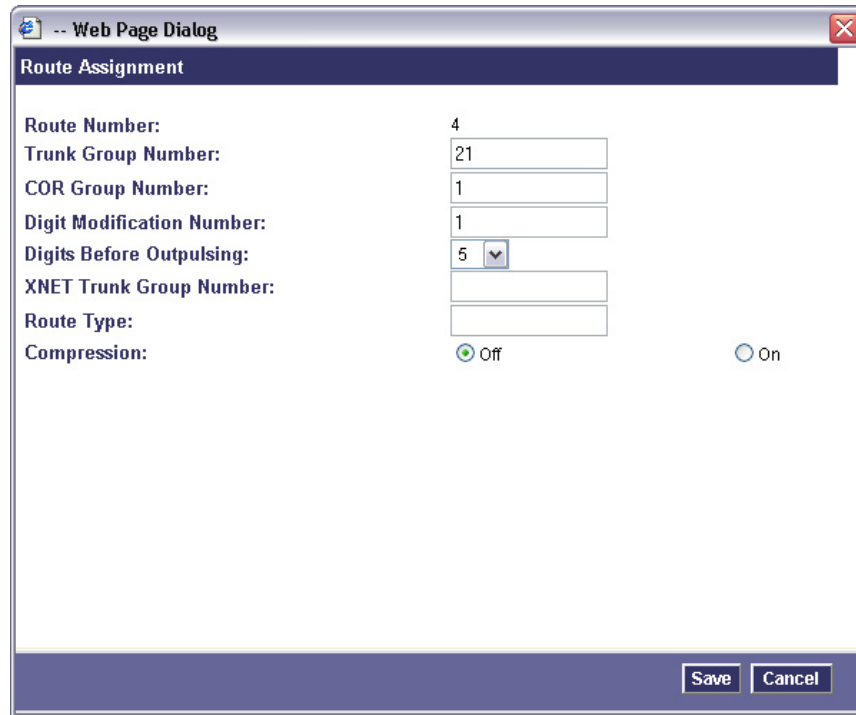
Use the ARS Route Assignment menu selection to define a route to direct calls to a specific trunk.

Select a Route Number and click the Change button to configure the options.

The screenshot shows the IPBX3 System Administration web interface. The left sidebar contains a tree view with 'Automatic Route Selection (ARS)' expanded to 'Route Assignment'. The main content area displays a table of route assignments. The table has columns for Route Number, Trunk Group Number, COR Group Number, Digit Modification Number, Digits Before Outpulsing, XNET Trunk Group Number, Route Type, and Compression. Row 4 is highlighted in blue. Above the table are navigation buttons: Change, Change Page, Change All, and Clear. Below the table are pagination controls: Previous, Page 1 of 8, Next, Go to: [dropdown], value: [input], and Go. The bottom of the interface features the MITEL 3300 ICP logo and 'About System Administration' text.

Route Number	Trunk Group Number	COR Group Number	Digit Modification Number	Digits Before Outpulsing	XNET Trunk Group Number	Route Type	Compression
1		1	2		1		Off
2		1	2		2		Off
3		1	2		3		Off
4	21	1	1	5			Off
5	22	1	1	5			Off
6	23	1	1	5			Off
7		1	1				Off
8		1	1				Off
9		1	1				Off
10	24	1	1	5			Off
11		1	1				Off
12		1	1				Off
13		1	1				Off
14		1	1				Off
15	15	1	1				Off

See the screen below for the selected options.



The screenshot shows a web-based dialog box titled "-- Web Page Dialog" with a close button in the top right corner. The main heading is "Route Assignment". The form contains the following fields and controls:

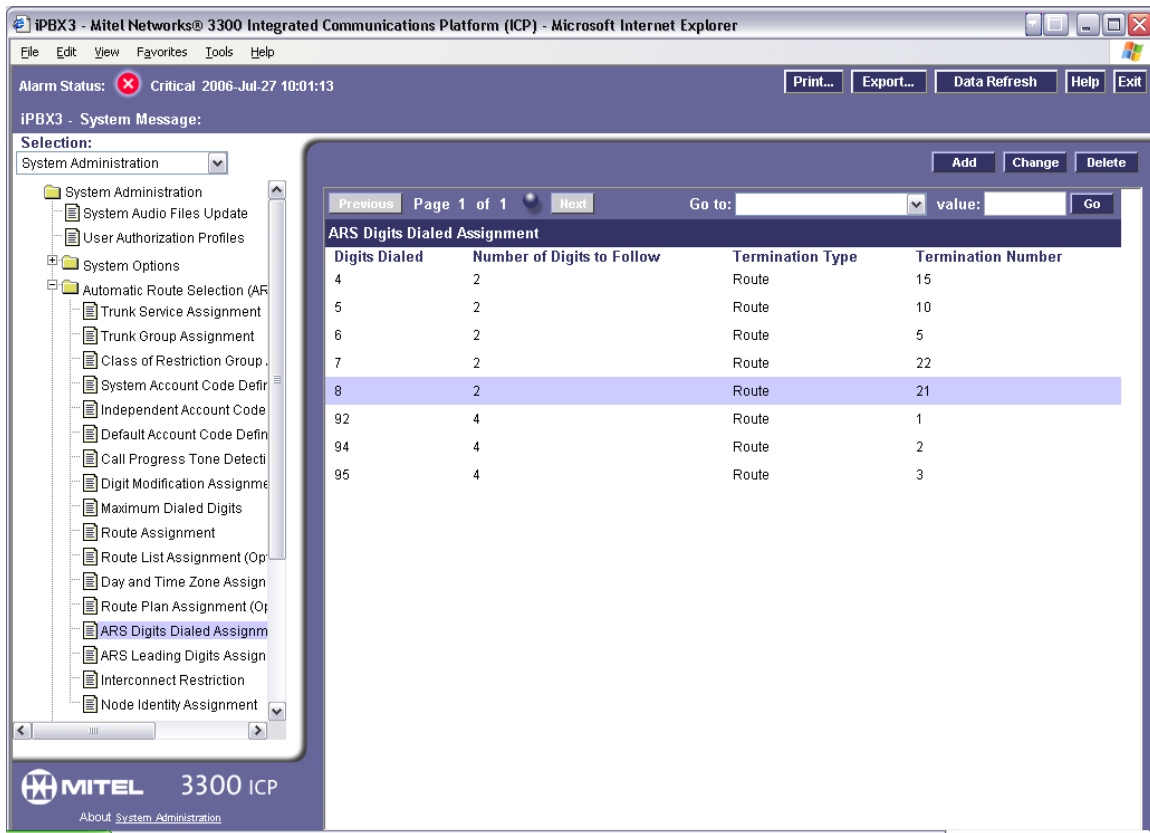
- Route Number:** A text input field containing the value "4".
- Trunk Group Number:** A text input field containing the value "21".
- COR Group Number:** A text input field containing the value "1".
- Digit Modification Number:** A text input field containing the value "1".
- Digits Before Outpulsing:** A dropdown menu with "5" selected.
- XNET Trunk Group Number:** An empty text input field.
- Route Type:** An empty text input field.
- Compression:** Two radio buttons, "Off" (which is selected) and "On".

At the bottom right of the dialog, there are two buttons: "Save" and "Cancel".

Before moving on to the next task be sure to click the Save button.

Use the ARS Digits Dialed Assignment menu selection to configure an ARS number to use to place and forward calls to a specific trunk group. The ARS number is used as the forwarding target for subscriber station sets and the inbound entry point for direct calls to the server.

Use the Add or Change buttons to either add a new ARS number or modify an existing one.



This example shows setting up ARS to except any 3 digit number that starts with an 8 as a dialable number. The ASR table will then take the call and route it to trunk group 21. Useable numbers in this example would be any number between 800 and 899 all inclusive.

An alternate method of configuration would be to define a very specific number, for example 800, not an entire range, and not define any following digits.

The method you choose is up to what your sites configuration will support.

See the screen below for the selected options.

Range Programming -- Web Page Dialog

Change Range Programming - ARS Digits Dialed Assignment Help

This form allows you to change one or more records, starting at the following record:

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
8	2	Route	21

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Digits Dialed:	Change to	<input type="text" value="8"/>	<input type="text"/>
Number of Digits to Follow:	Change to	<input type="text" value="2"/>	-
Termination Type:	Change to	<input type="text" value="Route"/>	-
Termination Number:	Change to	<input type="text" value="21"/>	<input type="text"/>

Preview Save Cancel

Below is an example of the configuration using the described alternative method.

Range Programming -- Web Page Dialog

Change Range Programming - ARS Digits Dialed Assignment Help

This form allows you to change one or more records, starting at the following record:

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
800		Route	21

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Digits Dialed:	Change to	<input type="text" value="800"/>	<input type="text"/>
Number of Digits to Follow:	Change to	<input type="text"/>	-
Termination Type:	Change to	<input type="text" value="Route"/>	-
Termination Number:	Change to	<input type="text" value="21"/>	<input type="text"/>

Preview Save Cancel

Before moving on to the next task be sure to click the Save button.

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 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.

5.5. Additional Comments

Ensure that the Node Identity assignment has not been entered. If this is entered, it will append extra digits onto extensions as they pass across the trunk to the gateway.

6. Exchange 2007 UM Validation Test Matrix

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

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

Refer to:

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

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

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.		No trunks were available at the time of testing.
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."		No speed dial button was available. Testing was done by making a direct call to the hunt group.
9	Send a test FAX message to user	P	

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

6.1. Detailed Description of Limitations

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

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

7. Troubleshooting

7.1. Important Debugging Tools

- Ethereal/Wireshark – Used to view and analyze the network captures provided by the Dialogic® gateway diagnostic firmware.
- Adobe Audition -- Used to review and analyze the audio extracted from the network captures to troubleshoot any audio-related issues.

7.2. Important Gateway Trace Masks

These keys are helpful during troubleshooting scenarios and should be considered keys to activate by default for all troubleshooting cases.

- `voip prot` and `voip code` – this allows the collection of all SIP-related messages as they are sent from and received by the gateway. This data is important in cases where you feel that the gateway is not able to communicate properly with the messaging server.
- `tel event` and `tel code` – This allows the collection of circuit-side activity of the emulated station set, such as display updates, key presses, light transitions and hook state changes. This data is important in the following scenarios:
 - Call control problems (dropped calls, failing transfers, etc...)
 - Integration problems (incorrect mailbox placement, missed auto-attendant greetings etc...)
- `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.