

Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

Toshiba CTX

with Dialogic® 1000 Media Gateway Series

(DMG1008LS) using Serial SMDI

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 Toshiba CTX and Dialogic® 1000 Media Gateway Series (DMG1008LS) using Serial SMDI 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	Toshiba
Model	CTX
Software Version	AR1ME021
Telephony Signaling	Serial SMDI
Additional Notes	N/A

1.2. VoIP Gateway

Gateway Vendor	Dialogic Corporation
Model	Dialogic® 1000 Media Gateway Series (DMG1008LS)
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

N/A

2.2. PBX Requirements

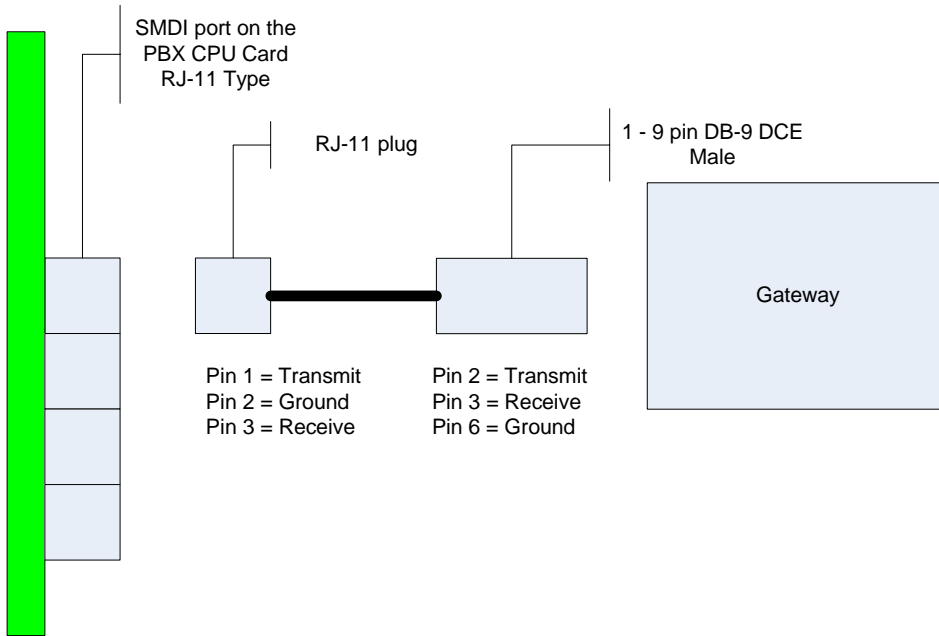
The CTX needs to be enabled / licensed for serial SMDI.

To support the 2-wire analog station interface, you need the RSTU3 analog PBX line card.

To support the SMDI interface on the PBX, you need a BSIS1A 4-port serial interface card connected to the CPU board of the PBX.

2.3. Cabling Requirements

Because the integration data is transmitted along a secondary serial data path between the PBX and the gateway, it will be required to either purchase or construct a cable that connects the serial port of the Toshiba PBX to the serial port of the gateway. The diagram below provides details on how this cable was constructed for the creation of this document.



The industry rule of thumb for RS232 serial communications is to keep the length of the cable no longer than 50 feet. Lengths longer than the recommended standard may need additional equipment to ensure correct transmission levels are maintained.

3. Summary and Limitations

- A check in this box indicates the UM feature set is fully functional when using the PBX/gateway in question.

4. Gateway Setup Notes

During the initial setup of the gateway using the serial port, you must:

- Assign the gateway a Unique IP address, subnet mask and network gateway address (if the latter is required).
- Configure the gateway to use the SIP VoIP protocol.

NOTE: The analog gateway is an interface specific SKU (only analog FXO) and therefore does not require being told what specific PBX it is being configured for.

During the solution-specific setup of the gateway using the web interface, you must:

- Configure the gateway with at least a single IP endpoint.
- Set the Voice coder to be either G.711 (default) or G.273 if required.
- Set the Serial Mode to Master for the single master gateway and to Slave for any of the additional slave gateways (if you are using any slaves).
- Ensure that the serial port settings (baud rate, parity and stop bits) match the port settings on the PBX.
- Set the Serial Interface Protocol to SMDI.
- Set the CPID length to 10.
- Set the CPID length to the same length as your PBXs extension numbers.
- Set the CPID Pad String to a string of characters that will match the characters that are left when you remove the extension numbers for your subscriber from the full string provided by the PBX. For example:
 - Your subscriber extensions are 3 digits long
 - The PBX sends you a full 10 digits of data.
 - The PBX data being sent to you looks like '0000000123' for extension number '123'.
 - Your pad string should be '0000000'
- Set each of the Logical extension Numbers to the extension number of each port in the proper order. These numbers for each analog line must match that ports extension number exactly.
- Configure the SIP Transport for TCP.

4.1. Parsing Rules

In order to correctly judge remote disconnection of TDM side devices, the gateway CPID Parser needs to be configured to recognize the digit sent to it from the PBX to signal disconnection. The setup details below show the PBX is configured to send a DTNF 'D' tone to the gateway. For the gateway to correctly judge this tone as a disconnection tone, you would need to set up a parsing rule to handle it.

```
#Remote disconnection
rule D
reason disconnect
```

These rules can be edited directly into the Analog web page.

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:

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

All PBX programming is done via a web-based interface accessible via a web browser.

5.1. Setting Up Serial Integration Interface

Use the `System Voice Mail Data` menu selection to configure the system data settings required for the system to provide the SMDI data. An example of this is shown below:

The screenshot shows a web browser window titled "CTX WinAdmin 1.16 - Microsoft Internet Explorer" displaying the "System Voice Mail Data" configuration page. The page has a left-hand navigation menu and a main content area with a table of settings.

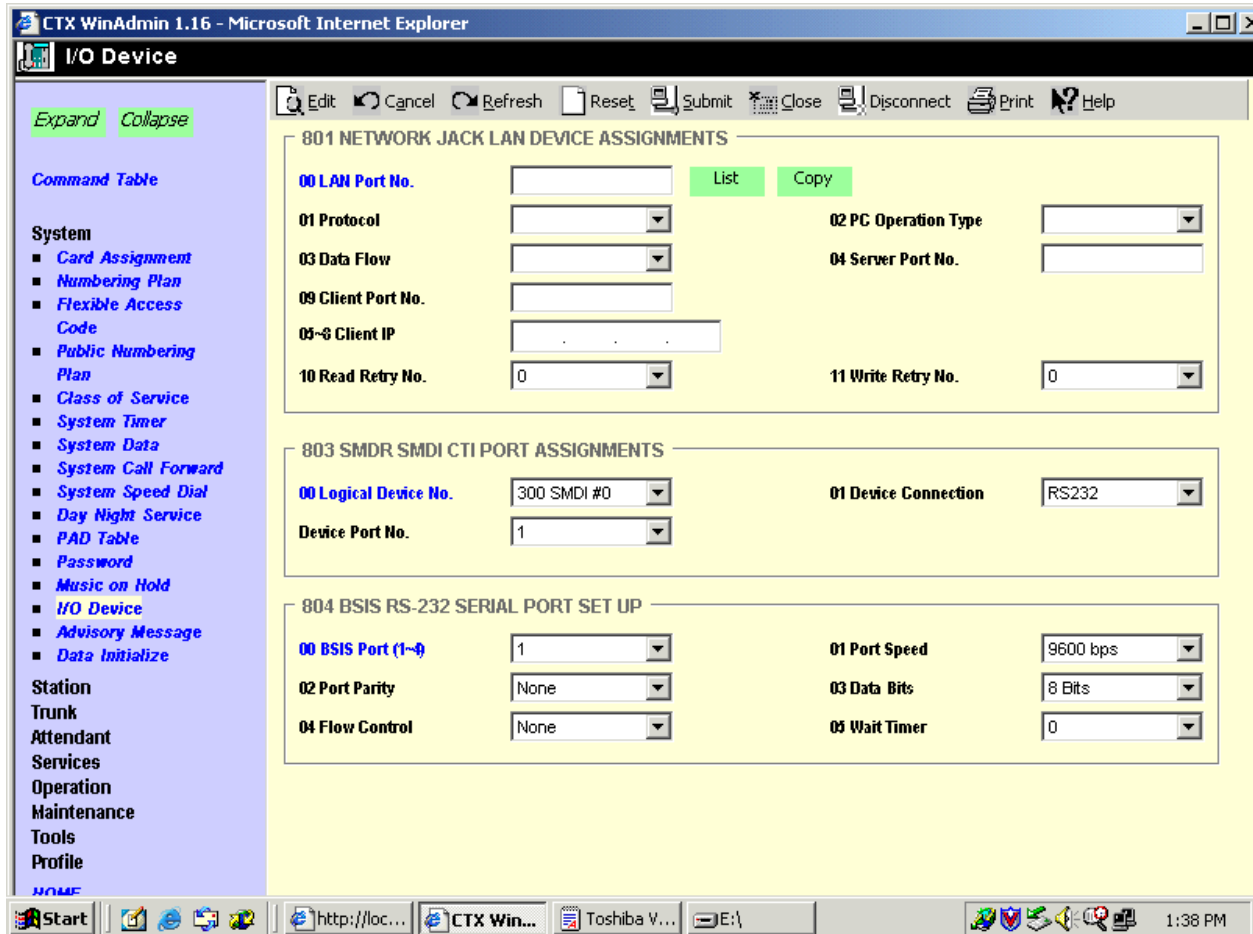
System Voice Mail Data Configuration Table:

Setting ID	Setting Name	Value / Option
01	DID/DNIS VMID Option	DN VMID
02	Cancellation Method for VM MW	Access Code Cancel
03	Message Desk Number	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
04	Output of CLASS / ANI and DNIS	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
05	Calling Number Digits Sent to VM	10
06	Blank Digits Sent to VM	2
07	Auto Cancel of VM and MW	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
08	DTMF Duration	80msec
09	LCD Control of Voice Mail	<input checked="" type="radio"/> Enable <input type="radio"/> Disable
10	Central VM Callback	
11	CF - All Call Record	91
12	CF - Busy Record	91
13	CF - No Answer Record	91
14	Direct Call to VM	91
15	Retrieve Messages	92
16	Transfer Direct to VM DN	
17	Length of VM-ID	10

Important notes about the above programming:

1. The Calling Number Digits Sent to VM should be sent to 10. This controls how long the calling party field is in the SMDI message.
2. The Length of VM ID should be set to 10. This controls how long the called party field is in the SMDI message.

In addition to setting up this configuration, you also need to use the I/O device menu selection to set up the physical serial port on the PBX. An example of this is shown below:



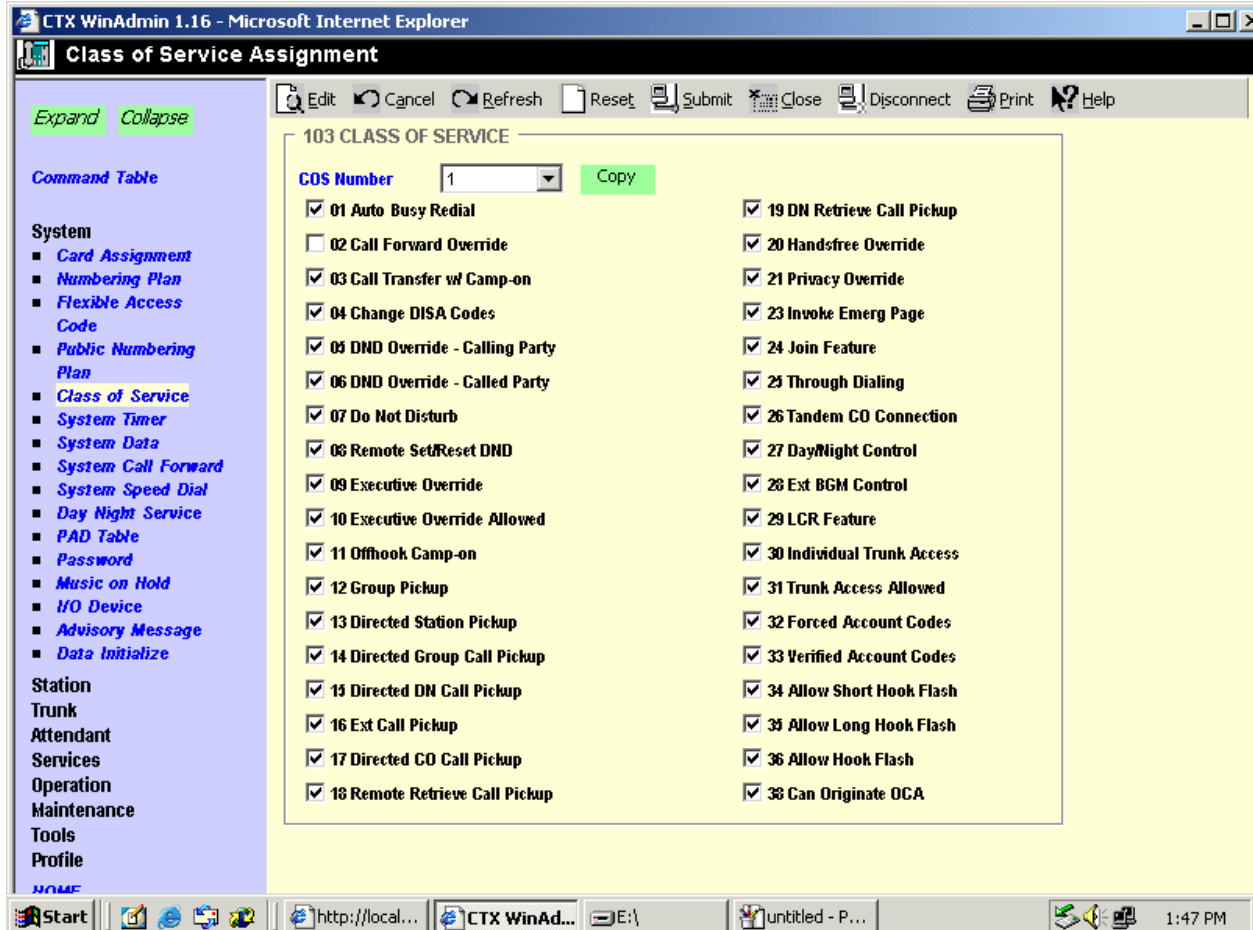
Important notes about the above programming:

1. The Device Connection setting should be set to RS232.
2. Ensure that your BSIS port is set to the number of the serial interface you are planning to use on the PBX.
3. Ensure that the Port Parity, Port Speed, and Data Bits settings are the same as you have set up in the gateway programming. The attached config.ini file sets these up for you but you can change them if needed.
4. Ensure that the Flow Control setting is set to None.

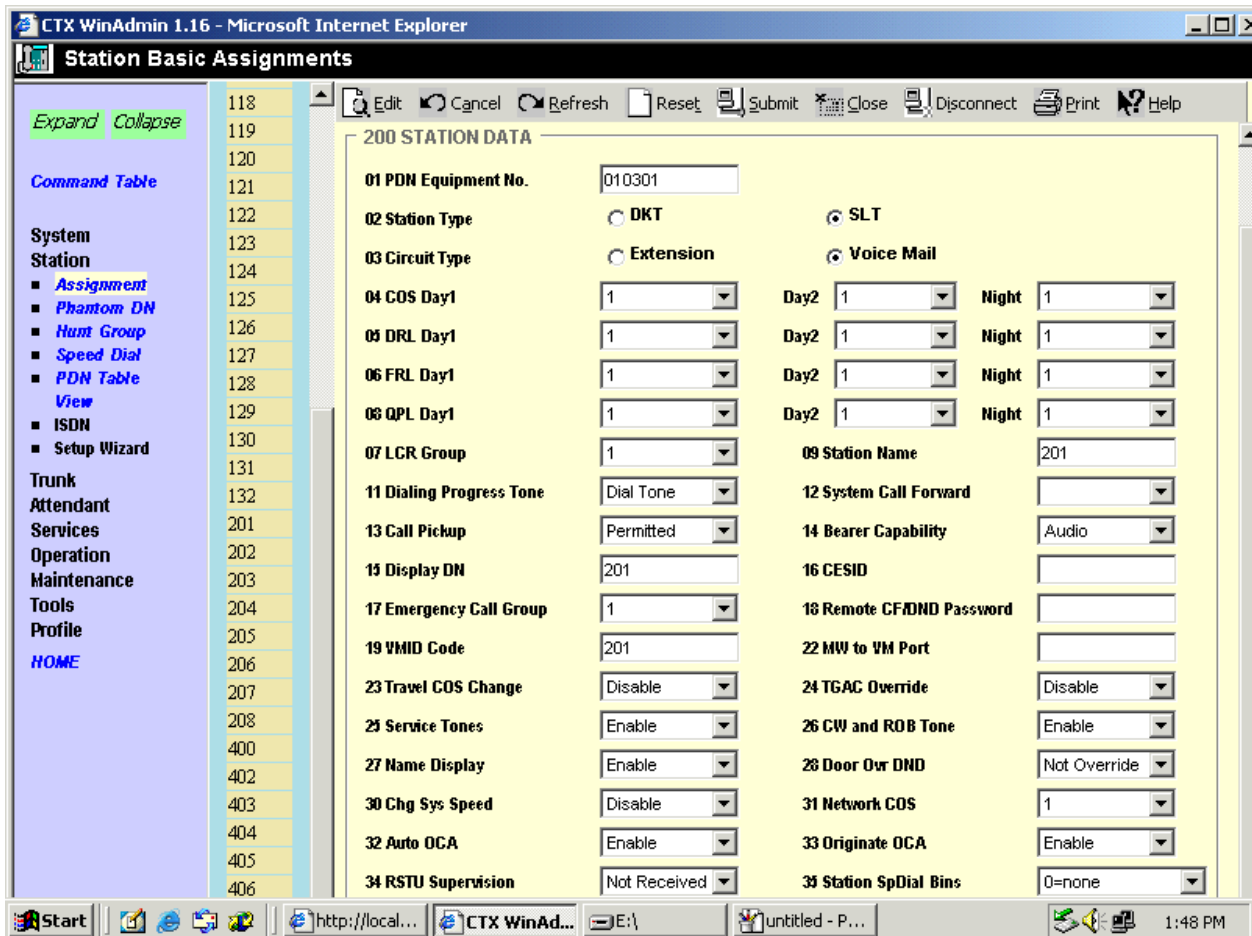
NOTE: If you do not set the flow control setting to None as recommended then you will need to ensure that the cable used provides the proper loop back connections on the PBX side of the cable. This note does not detail those required settings.

5.2. Setting Up Each Gateway Station Port

Use the **System Class of Service** menu selection to set up a class of service temple that will be shared across all the gateway station ports. The example below shows performing the command.



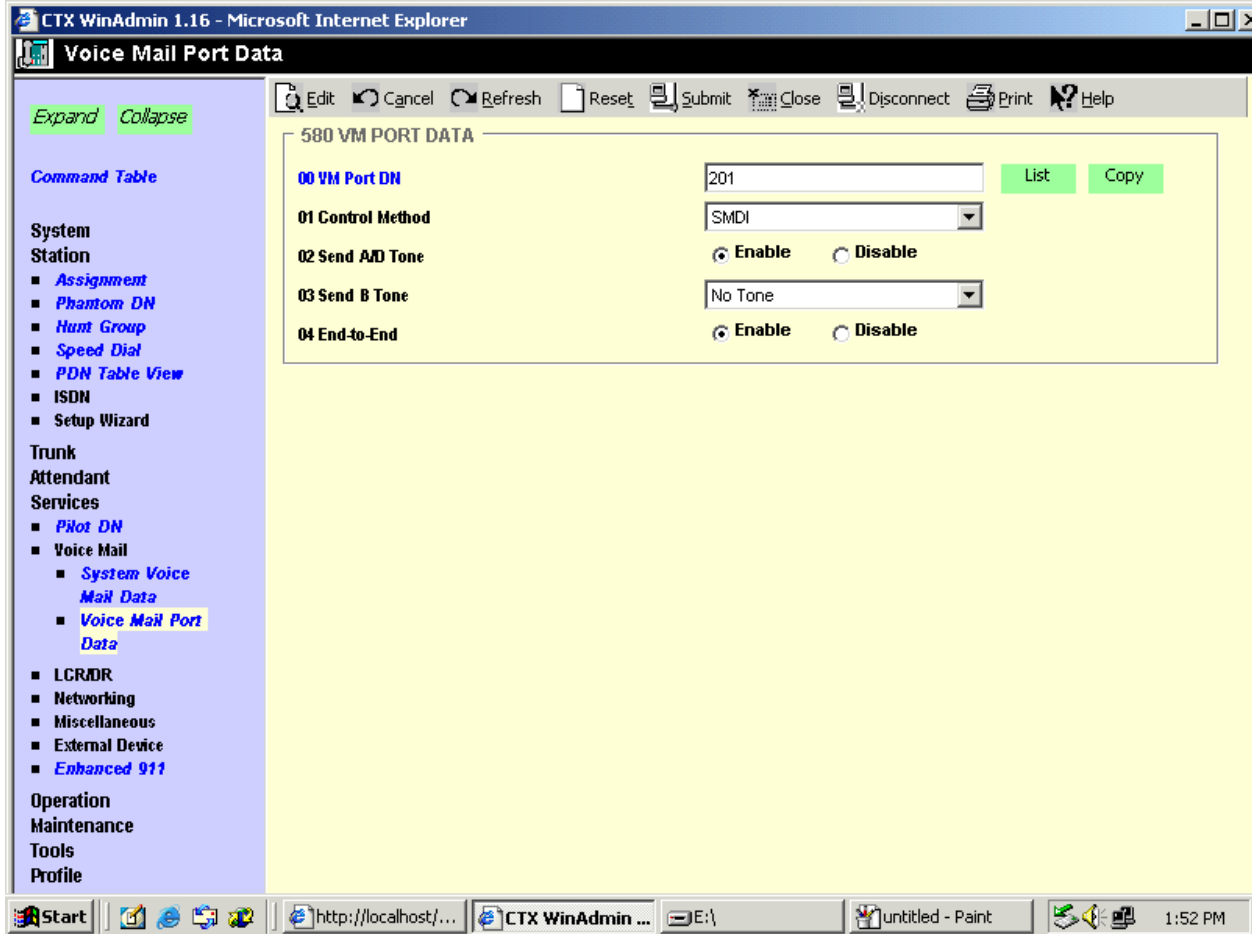
Use the Station Assignment menu selection to build as many analog station ports as required for your configuration (8 ports per gateway). This command is what builds the physical interface on the PBX. The example below shows performing this command on one station.



Important notes about the above programming:

1. The PDN Equipment No. field is the physical wiring address of the port on the PBX.
2. The Station Type field should be set to SLT.
3. The Circuit Type field should be set to Voice Mail.
4. All the class of service (COS) fields should be set to use the class of service defined in the previous step.
5. The Dialing Progress Tone field should be set to Dial Tone.
6. The Station Name field, VMID Code field, and Display DN field should be configured for the same number as is where you specify the extension number for each analog station port.
7. The Service Tones field should be set to Enabled.

In addition to the physical definition, you need to tell the PBX that each port is part of a voice mail type and how to present the integration data to the automated system. This is done by using the Voice Mail Port Data menu selection. This should be done for each of the analog station ports you defined to use with the gateway. The example below shows performing this command on one station.

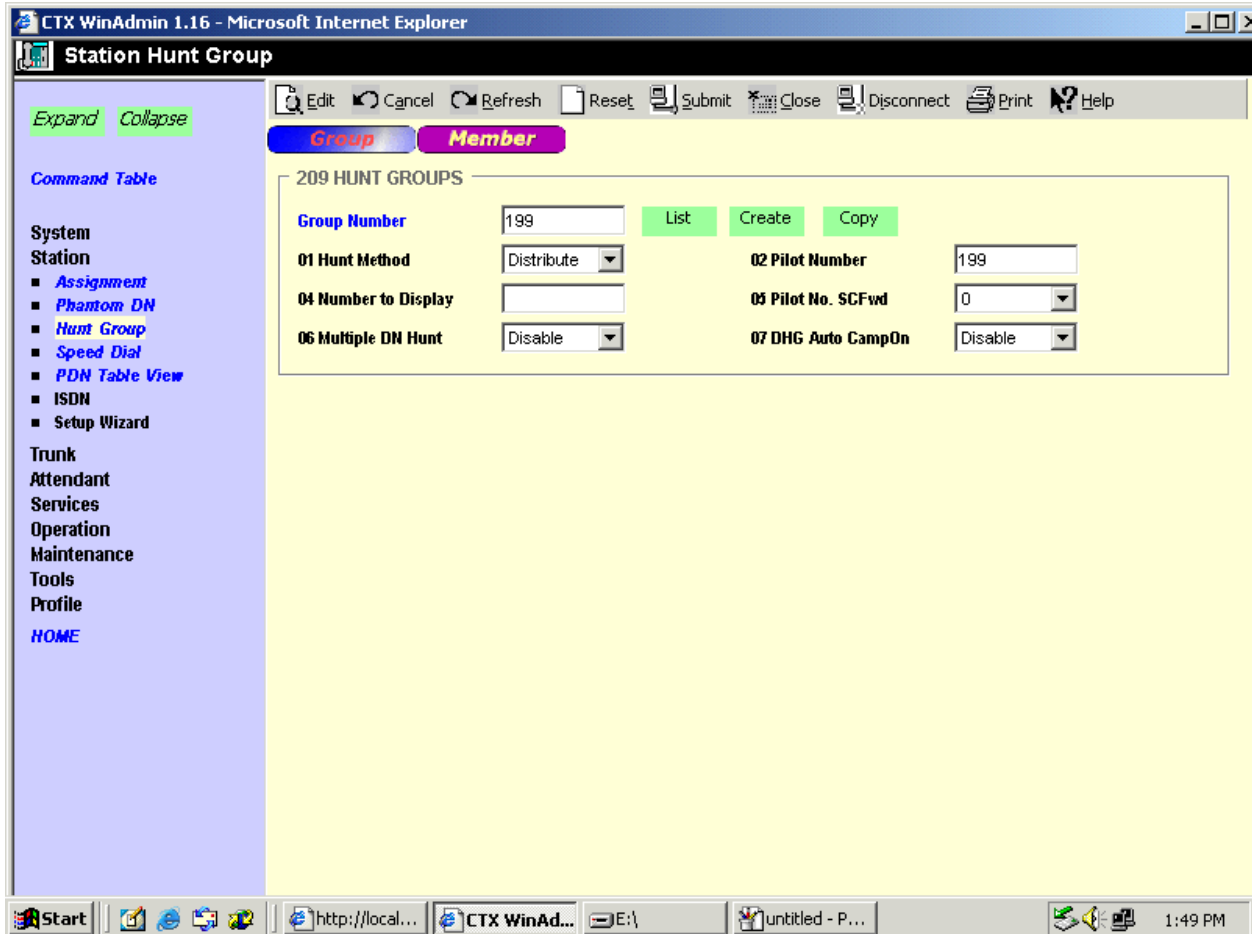


Important notes about the above programming:

1. The Control Method field should be set to SMDI.
2. The Send A/D Tone field should be set to Enabled.
3. The End-to-End field should be set to Enabled.

5.3. Defining Hunt Group

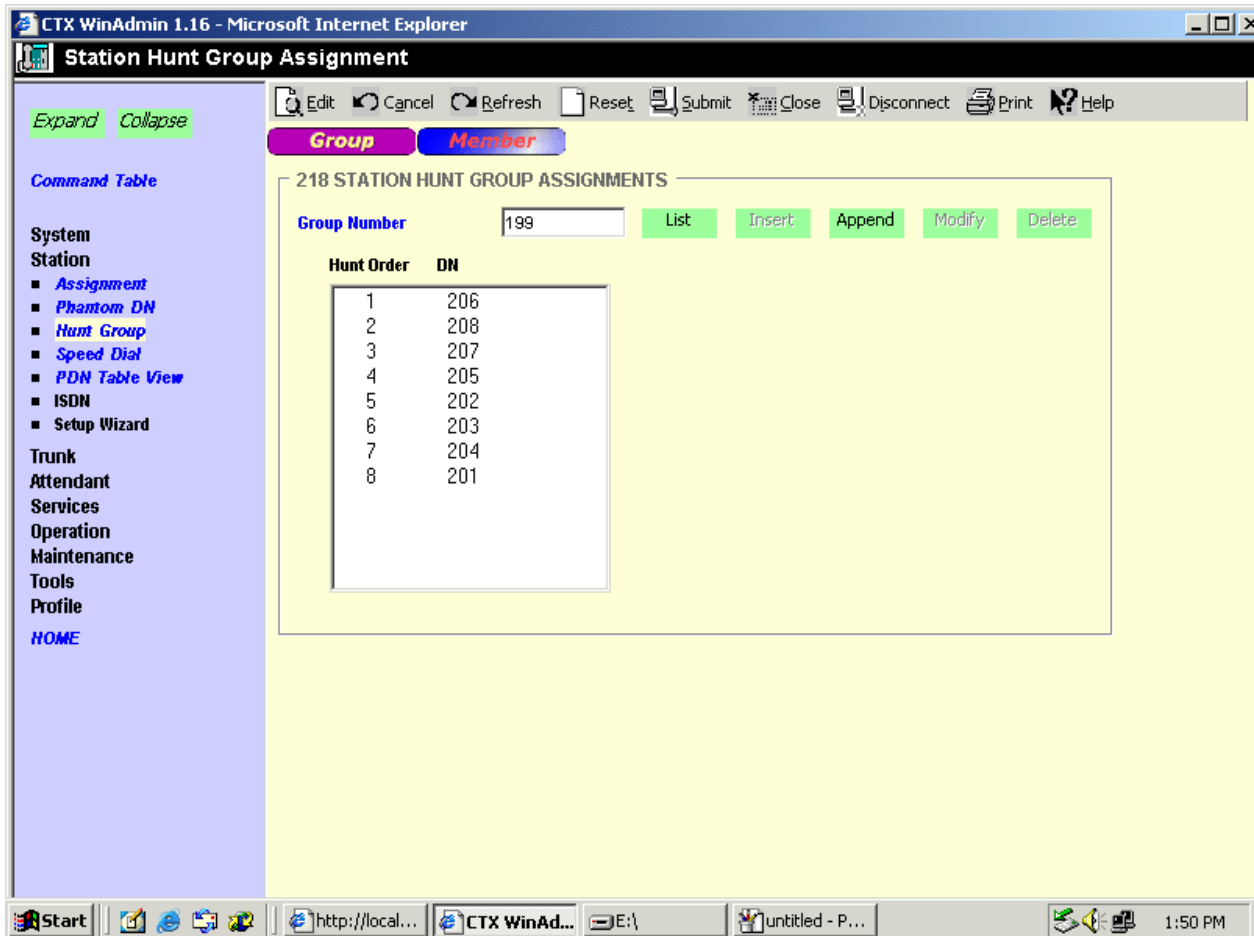
Use the Station Hunt Group menu selection to define a hunt group that includes all the configured analog station ports you wish to use for the gateway. The example below shows performing this command.



Important notes about the above programming:

1. The `Group Number` defines the hunt group index number in the PBX system.
2. The `Pilot Number` field defines the extension provided to the hunt group and should be used as the forwarding point for all subscriber stations as well as the inbound call point o access the server.
3. The `Hunt Method` field can be adjusted as required to change between various hunting patterns.

Once the hunt group is defined, you use the **Member** tab to add all your analog voice mail ports to the group. The example below shows performing this command.



NOTE: The order that the ports appear on this page (the order that the ports are included in the hunt group) needs to match the order that the physical ports are plugged into the gateway as well as the order that the extension numbers are configured into the LTN fields on the gateway. If there is a misalignment between the PBX configuration, the gateway configuration and or the physical connections then the data will not be properly aligned with the call when it arrives to the gateway. This can result in either a lack of integration or cross-integrated calls.

5.4. Setting Up Subscriber Station Sets

All the forwarding of the subscriber station sets can be defined directly on subscriber station set using feature access codes. The subscriber should be directed to set their internal and external ring no answer and busy forwarding conditions to the extension number assigned to the hunt group configuration.

5.5. 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.		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.		No trunks were available at the time of testing.
7b	Listen to voicemail using OWA's Play-On-Phone feature to an external number.	P	
8	Configure a button on the phone of a UM-enabled user to forward the user to the pilot number. Press the voicemail button. Confirm you are sent to the prompt: "Welcome, you are connected to Microsoft Exchange. <User>. Please enter your pin and press the pound key."	P	No speed dial button was available. Testing was done by making a direct call to the hunt group.
9	Send a test FAX message to user	P	Faxing was done via an internal analog extension.

	<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.
- Microsoft HyperTerminal – Used to test the output of the PBXs serial port to determine if data is being sent by the PBX. This helps in the validation of the cabling and all connects on the serial interface between the PBX and the gateway.
- RS232 breakout box – This tool can be valuable in doing serial cabling work. It helps you determine what signals are available and if data is being either transmitted or received.

7.2. Important Gateway Trace Masks

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

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

These keys are helpful during specific issues and can be enabled for targeted troubleshooting of very specific cases. Activation of these keys may generate large amounts of data on busy systems and increase the size of the collected log files, but should not harm system performance.

- `dspcpi` (all keys) – This allows the collection of tone-related data. This data is helpful in cases where you think you have problems detecting specific tones that should be, should not be, or are expected to be present at specific times during the call. If you do not suspect a tone-related issue, this key may be left disabled. This data is important in the following scenarios:
 - Failing transfers
 - Failing outbound calls (play to phone)
 - Dropped calls (callers cut off while leaving messages, etc...)
- `si` – This allows the collection of all inbound and outbound serial data on the serial master gateway. This data is required in the troubleshooting of all integration-related issues seen on the gateway designated as the serial master.
- `siip` – This allows the collection of all inbound and outbound serial data on any of the serial slave gateways. This data is required in the troubleshooting of all integration-related issues seen on the gateway designated as a serial slave.
- `simwi` – This allows the collection of all activity reacted to the processing of MWIs using the serial port. This data is required for any MWI-related issues while using the serial interface.

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.