Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

ERICSSON MD-110 with AudioCodes MP-11x using Analog SMDI

By : AudioCodes

Updated Since : 2008-11-02

Content

This document describes the configuration required to setup ERICSSON MD-110 and AudioCodes MP-11x FXO using analog lines with 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, its 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**
02 November 2008	Version 1

1. Components Information

1.1. PBX or IP-PBX

PBX Vendor	ERICSSON
Model	MD-110
Software Version	MX-ONE TSW
Telephony Signaling	Analog with SMDI - ERICSSON
Additional Notes	None

1.2. VoIP Gateway

Gateway Vendor	AudioCodes
Model	MP-11x FXO (MP-114 / MP-118)
Software Version	5.20A.037.003
VoIP Protocol	SIP

1.3. Microsoft Exchange Server 2007 Unified Messaging

Version	RTM

2. Prerequisites

2.1. Gateway Prerequisites

- The gateway supports Ericsson's proprietary SMDI serial link. The parameter "SMDI" must be configured to 3 to enable this feature. For additional details, please refer to the gateway setup note in Section 6.

2.2. Cabling Requirements

- This integration uses standard RJ-11 telephone line cords to connect analog ports between the PBX and MP-11x FXO ports.
- This integration uses a serial cable:

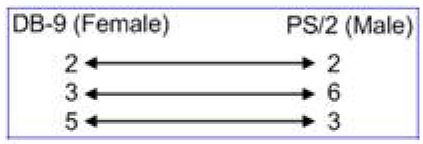
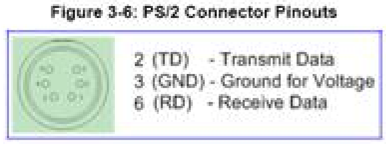


Figure 3-6: PS/2 Connector Pinouts



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

Step 1: SIP Environment Setup

Proxy & Registration	
Enable Proxy	Use Proxy
Proxy Name	
Proxy IP Address	10.10.0.186
First Redundant Proxy IP Address	0.0.0.0
Second Redundant Proxy IP Address	0.0.0.0
Third Redundant Proxy IP Address	0.0.0.0
Redundancy Mode	Parking
Proxy Load Balancing Method	Disable
Proxy IP List Refresh Time	60
Enable Proxy Keep Alive	Disable
Proxy Keep Alive Time	60
Enable Fallback to Routing Table	Disable
Prefer Routing Table	No
Use Routing Table for Host Names and Profiles	Disable
Always Use Proxy	Disable
Send All Invite to Proxy	No
Enable Proxy Hot -Swap	Disable
Enable Registration	Disable
Gateway Name	
Gateway Registration Name	
DNS Query Type	A-Record
Proxy DNS Query Type	A-Record
Subscription Mode	Per Gateway
Use Gateway Name for OPTIONS	No
Number of RTX Before Hot -Swap	3
User Name	
Password	••••••••••••••••
Cnonce	Default_Cnonce
Authentication Mode	Per Endpoint

Note: The Proxy IP Address must be one that corresponds to the network environment in which the Microsoft Unified Messaging server is installed (For example, 10.10.0.186 or the FQDN of the Microsoft Unified Messaging host).

Step 2: SIP General

General

PRACK Mode	Supported
Channel Select Mode	Ascending
Enable Early Media	Disable
183 Message Behavior	Progress
Session -Expires Time	0
Minimum Session -Expires	90
Session Expires Method	Re-Invite
Asserted Identity Mode	Disabled
Fax Signaling Method	T.38 Relay
! Detect Fax on Answer Tone	Initiate T.38 on Preamble
SIP Transport Type	TCP
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPS	Disable
Enable TCP Connection Reuse	Enable
SIP Destination Port	5060
Use "user=phone" in SIP URL	Yes
Use "user=phone" in From Header	No
! Use Tel URI for Asserted Identity	Disable
Tel to IP No Answer Timeout	180
Enable Remote Party ID	Disable
Add Number Plan and Type to Remote Party ID Header	Yes
Enable History -Info Header	Disable
Use Source Number as Display Name	No
Use Display Name as Source Number	No
Play Ringback Tone to IP	Don't Play
Play Ringback Tone to Tel	Play According to Early Med
Use Tgrp information	Disable
Enable GRUU	Disable
User-Agent Information	
Subject	
Multiple Packetization Time Format	None

Step 3: Coder Setup

Coders

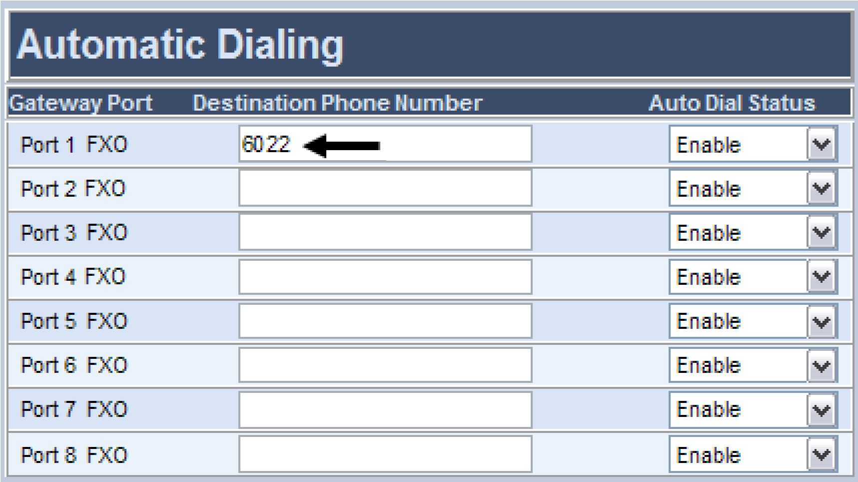
Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711U-law	20	64	0	Disabled
G.711A-law	20	64	8	Disabled
G.723.1	30	5.3	4	Disabled

Step 4: DTMF & Dialing

DTMF & Dialing

Max Digits In Phone Num	5
Inter Digit Timeout for Overlap Dialing [sec]	4
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option;	RFC 2833
2nd Tx DTMF Option;	Not Supported
3rd Tx DTMF Option;	Not Supported
4th Tx DTMF Option;	Not Supported
5th Tx DTMF Option;	Not Supported
RFC 2833 Payload Type	101
Hook-Flash Option	Not Supported
Digit Mapping Rules	
Dial Tone Duration [sec]	16
Hotline Dial Tone Duration [sec]	16
Enable Special Digits	Disable
Default Destination Number	1000
Dial Tone Duration [sec]	16
Hotline Dial Tone Duration [sec]	16
Hook-Flash Option	Not Supported

Step 5: Automatic Dialing Setup



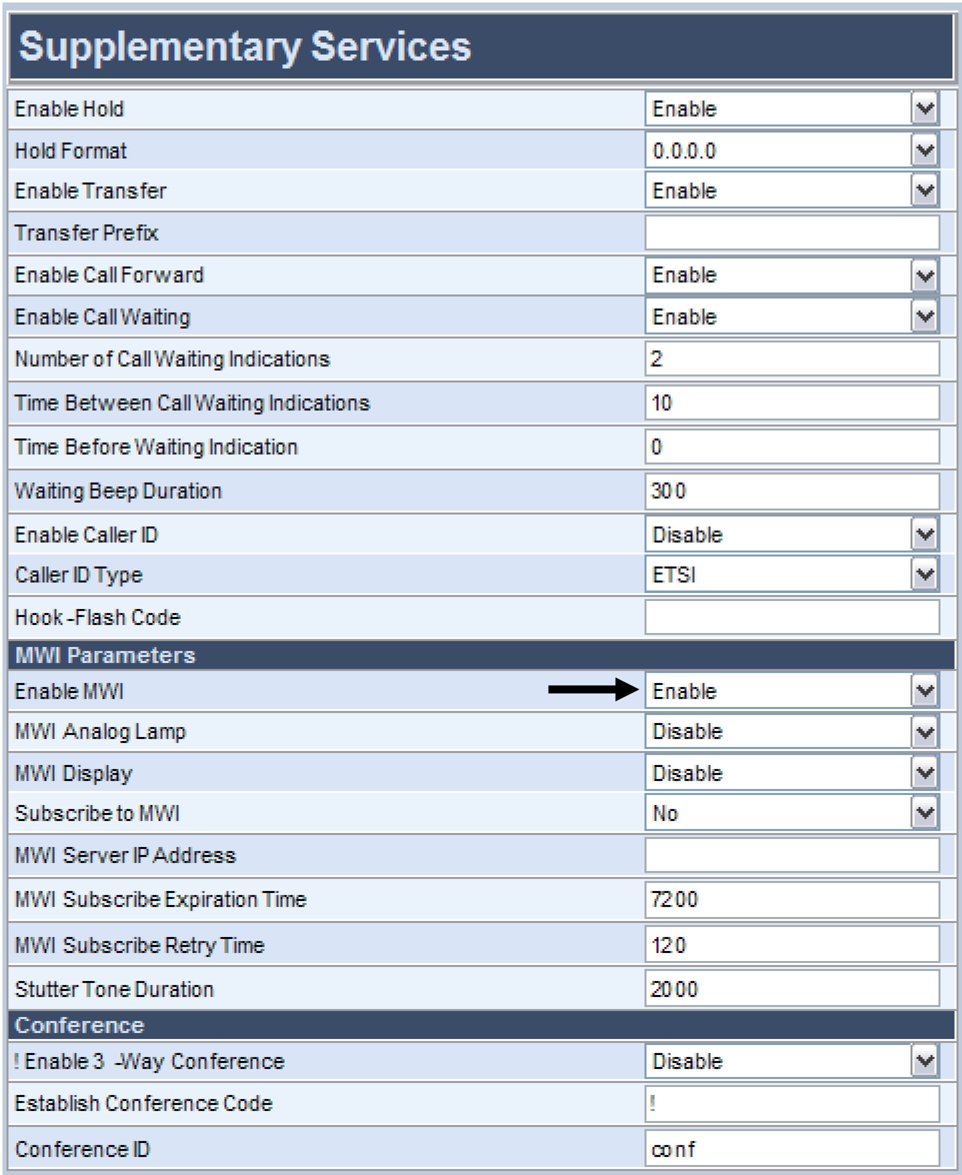
Note: The phone numbers must correspond to your network environment as the dial plan pilot number is configured for this PBX in the Microsoft Unified Messaging server (For example, 6022).

Step 6: General Parameters Setup

General Parameters

IP Security	Disable
Filter Calls to IP	Don't Filter
! Enable Digit Delivery to Tel	Disable
! Enable Digit Delivery to IP	Disable
RTP Only Mode	Disable
Enable DID Wink	Disable
Delay Before DID Wink	0
Reanswer Time	0
Disconnect and Answer Supervision	
Enable Polarity Reversal	Disable
Enable Current Disconnect	Enable
Disconnect on Broken Connection	No
Broken Connection Timeout [100 msec]	3
Disconnect Call on Silence Detection	No
Silence Detection Period [sec]	120
Silence Detection Method	Voice/Energy Detectors
Send Digit Pattern on Connect	
CDR and Debug	
CDR Server IP Address	
CDR Report Level	End Call
Debug Level	5
Misc. Parameters	
Progress Indicator to IP	Not Configured
Enable Busy Out	Disable
Default Release Cause	3
Delay After Reset [sec]	7
Max Number of Active Calls	8
Max Call Duration [min]	0
Enable LAN Watchdog	Disable
Enable Calls Cut Through	Disable
Enable User -Information Usage	Disable
Out -Of-Service Behavior	Reorder Tone

Step 7: Message Waiting Indication Setup



Supplementary Services

Enable Hold	Enable
Hold Format	0.0.0.0
Enable Transfer	Enable
Transfer Prefix	
Enable Call Forward	Enable
Enable Call Waiting	Enable
Number of Call Waiting Indications	2
Time Between Call Waiting Indications	10
Time Before Waiting Indication	0
Waiting Beep Duration	300
Enable Caller ID	Disable
Caller ID Type	ETSI
Hook-Flash Code	
MWI Parameters	
Enable MWI	Enable
MWI Analog Lamp	Disable
MWI Display	Disable
Subscribe to MWI	No
MWI Server IP Address	
MWI Subscribe Expiration Time	7200
MWI Subscribe Retry Time	120
Stutter Tone Duration	2000
Conference	
! Enable 3-Way Conference	Disable
Establish Conference Code	!
Conference ID	conf

Step 8: Manipulation Routing Setup

Destination Phone Number Manipulation Table for Tel -> IP Calls

Table Index: 1-10

	Destination Prefix	Source Prefix	Number of stripped Digits	Prefix (Suffix) to Add	Number of Digits to Leave
1	*	*	0		0
2					
3					
4					
5					
6					
7					
8					
9					
10					

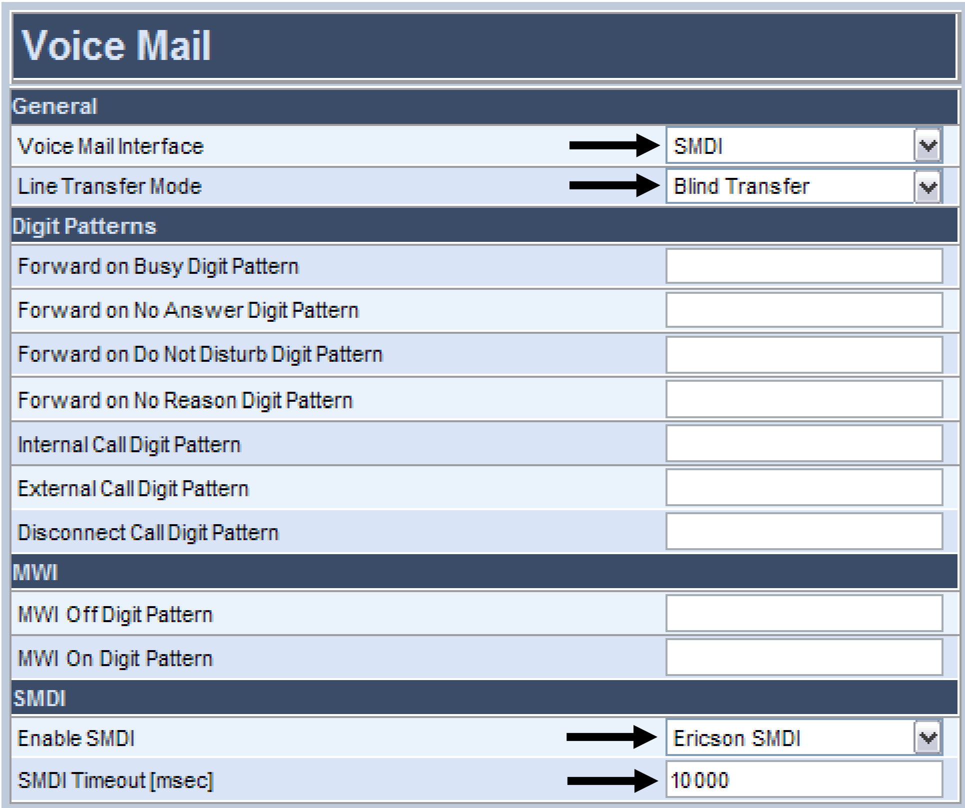
Step 9: Endpoints Setup

Endpoint Phone Number Table

	Channel(s)	Phone Number	Hunt Group ID	Profile ID
1	1	6014		0
2				
3				
4				
5				
6				
7				
8				

Note: The SMDI message's 'Line Identifier' field must match the gateway's local endpoint number defined in the screen above. This endpoint handles the SMDI message and is expected to receive the call. Therefore, the local number of the gateway's endpoints must correlate with SMDI messages identifiers. In the testing environment, the PBX is connected to MP-11x FXO using one voice mail line 6014, corresponding to port 1 of MP-11x FXO. Therefore, this is reflected in the endpoint settings. This endpoint number corresponds to the PBX analog voice mail extensions definition.

Step 10: Voice Mail SMDI Setup



Step 11: FAX Setup

Fax/Modem/CID Settings

Fax Transport Mode	T.38 Relay
Caller ID Transport Type	Mute
Caller ID Type	ETSI
V.21 Modem Transport Type	Disable
V.22 Modem Transport Type	Enable Bypass
V.23 Modem Transport Type	Enable Bypass
V.32 Modem Transport Type	Enable Bypass
V.34 Modem Transport Type	Enable Bypass
Fax Relay Redundancy Depth	0
Fax Relay Enhanced Redundancy Depth	4
Fax Relay ECM Enable	Enable
Fax Relay Max Rate (bps)	14400
Fax/Modem Bypass Coder Type	G711Alaw
Fax/Modem Bypass Packing Factor	1
CNG Detector Mode	Events Only

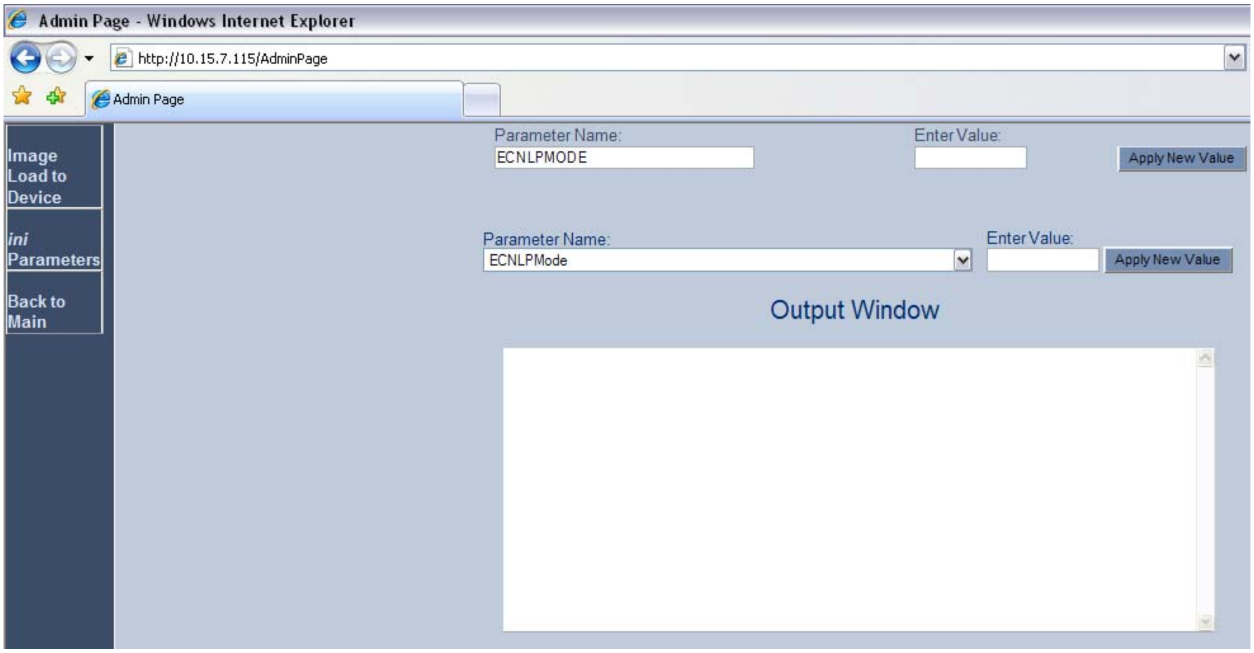
Step 12: FXO General Setup

FXO Settings

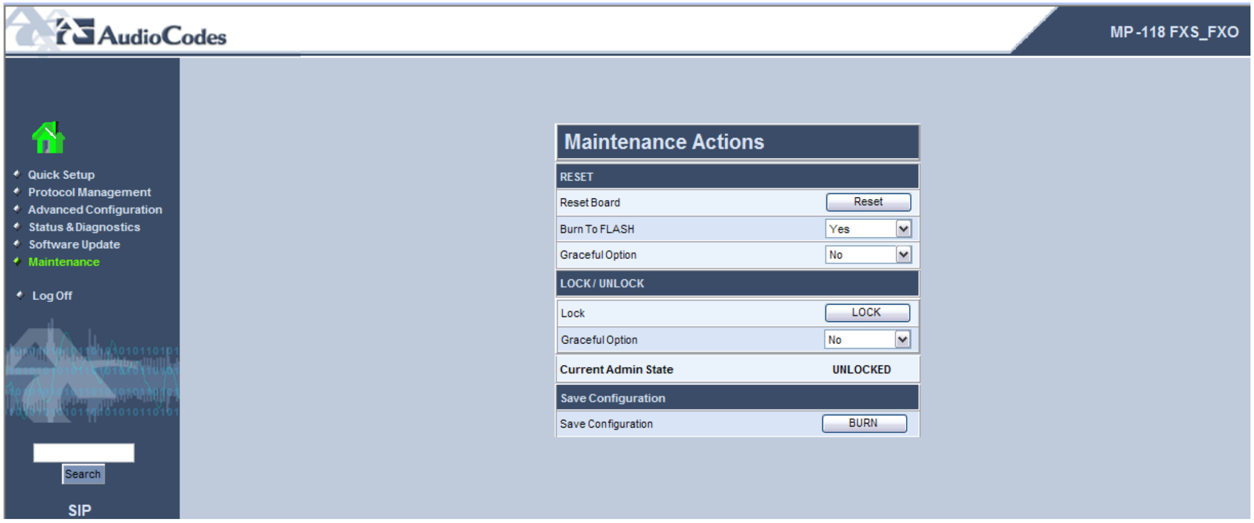
Dialing Mode	One Stage
Waiting for Dial Tone	No
Time to Wait before Dialing [msec]	1000
Ring Detection Timeout [sec]	8
Reorder Tone Duration [sec]	255
Answer Supervision	Yes
Rings before Detecting Caller ID	1
Send Metering Message to IP	No
Disconnect on Busy Tone	Yes
Disconnect On Dial Tone	Disable
Guard Time Between Calls	1

Step 13: AdminPage configurations:

- ECNLPMode = 1
- CurrentDisconnectDuration = 450
- FXSLoopCharacteristicsFilename = 'MP11x-02-1-FXS_16KHZ.dat'
- EnableDetectRemoteMACChange = 2
- CallerIDType = 1



Step 14: Reset Gateway



Click the **Reset** button to reset the gateway.

4.1. Configuration Files

- AudioCodes configuration ini file (.ini file extension).

MP118FXSFXO_with_
EriccsonMD110_SMDI

5. PBX Setup Notes

Information for this test case:

- Analog Voice Mail port: ext. 6014

System Parameters Setup

```
<ICFUP;
INFORMATION COMPUTER COMMON FUNCTIONS DATA
   UPDATING START TIME IS  01:30
   MESSAGE WAITING FUNCTIONALITY IS  ALL
INFORMATION COMPUTER EQUIPMENT DATA
   IFCIND  IODEV           EQU          RATE   DFMT  UPDFCN
PARITY  CCHECK
   1                       003-1-61-1   9600   5     NO      EVEN
NO
           TXC=YES    FILLER=32     ICEXG=NONE

<ICMWP:SID=ALL;;
INFORMATION COMPUTER MESSAGE WAITING DATA
SID  DTXT            KFCN DIG
1    6026            MWC  6026
END

<VMFUP;
VOICE MAIL FUNCTION DATA
IFCIND   VMF    POFMT
1        EXTN2  3
END

<VMPOP;
VOICE MAIL PORT DATA
DIR      PORT     IFCIND
VOICE MAIL PORT DATA
DIR      PORT     IFCIND
6014     100      1
VOICE MAIL GROUP DATA
GRP      IFCIND
6026     1
END
```

SMDI configured

```
IFCIND=1
EQU=3-1-61-1
RATE=9600
DFMT=5
UPDFCN=NO
PARITY=EVEN
CCHECK=NO
TXC=YES
FILLER=32
ICEXG=NONE
```

6. Exchange 2007 UM Validation Test Matrix

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

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

Refer to:

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

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

5	Dial Auto Attendant (AA). Dial the extension for the AA and confirm the AA answers the call.	P	
6	Call Transfer by Directory Search.		
6a	Call Transfer by Directory Search and have the called party answer. Confirm the correct called party answers the phone.	NT	
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.	NT	
6c	Call Transfer by Directory Search when the called party does not answer. Confirm the call is routed to the called party's voicemail.	NT	
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.	NT	
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.	NT	
7b	Listen to voicemail using OWA's Play-On-Phone feature to an external number.	NT	
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."	NT	
9	Send a test FAX message to user	P	

	extension. Confirm the FAX is received in the user's inbox.		
10	Setup TLS between gateway/IP-PBX and Exchange UM. Replace this italicized text with your TLS configuration: self-signed certificates or Windows Certificate Authority (CA).		
10a	Dial the pilot number and logon to a user's mailbox. Confirm UM answers the call and confirm UM responds to DTMF input.	**NT**	
10b	Dial a user extension and leave a voicemail. Confirm the user receives the voicemail.	**NT**	
10c	Send a test FAX message to user extension. Confirm the FAX is received in the user's inbox.	**NT**	
11	Setup G.723.1 on the gateway. (If already using G.723.1, setup G.711 A Law or G.711 Mu Law for this step). Dial the pilot number and confirm the UM system answers the call.	**P**	
12	Setup Message Waiting Indicator (MWI). Geomant offers a third party solution: MWI 2007. Installation files and product documentation can be found on Geomant's [MWI 2007 website]().	**NT**	
13	Execute Test-UMConnectivity.	**NT**	
14	Setup and test fail-over configuration on the IP-PBX to work with two UM servers.	**NT**	

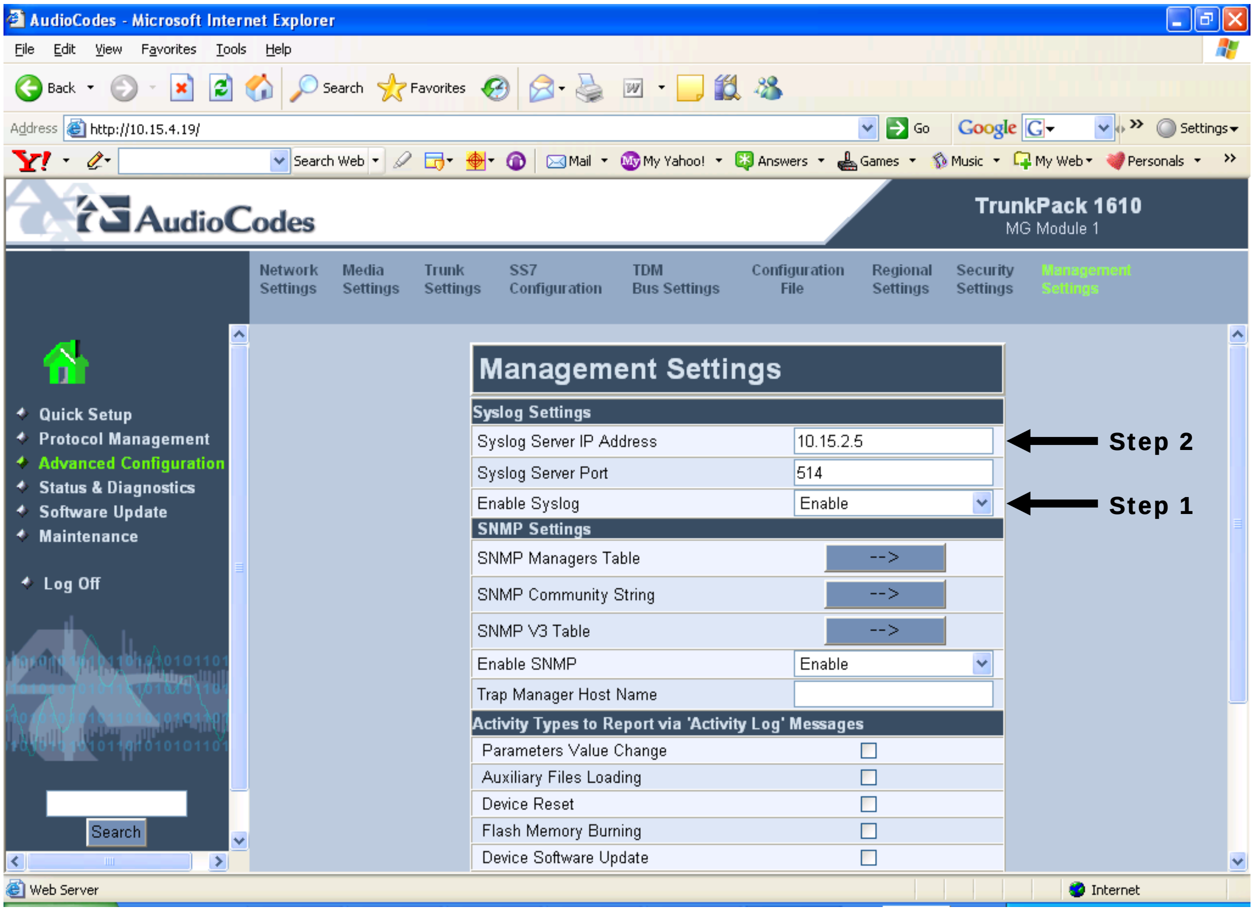
7. Troubleshooting

The tools used for debugging include network sniffer applications (such as Ethereal) and AudioCodes' Syslog protocol.

The Syslog client, embedded in the AudioCodes gateways (MP-11x, Mediant 1000, and Mediant 2000), sends error reports/events generated by the gateway application to a Syslog server, using IP/UDP protocol.

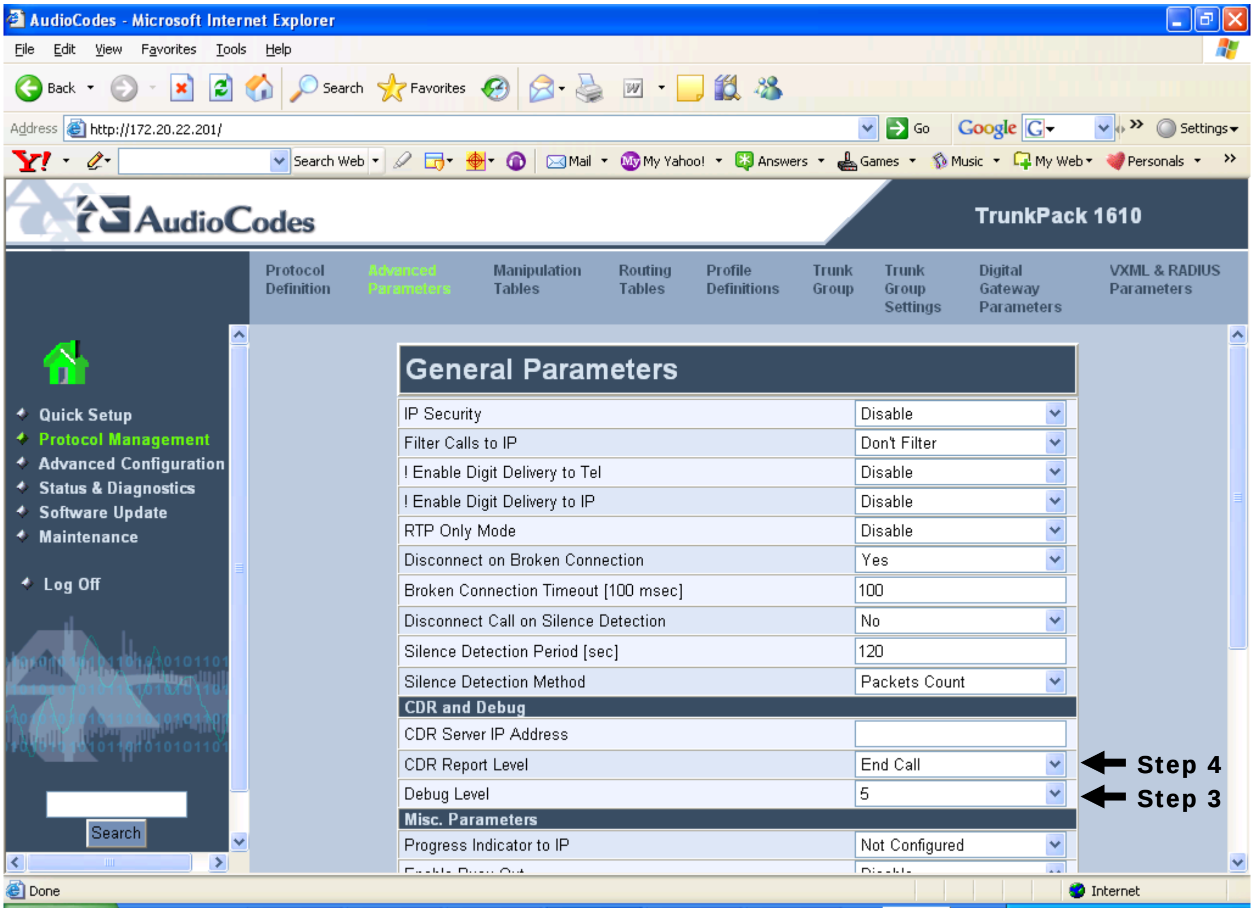
To activate the Syslog client on the AudioCodes gateways:

1. Set the parameter **Enable Syslog** to 'Enable'.
2. Use the parameter **Syslog Server IP Address** to define the IP address of the Syslog server you use.



Note: The Syslog Server IP address must be one that corresponds to your network environment in which the Syslog server is installed (for example, 10.15.2.5).

3. To determine the Syslog logging level, use the parameter **Debug Level** and set this parameter to '5'.
4. Change the **CDR Report Level** to 'End Call' to enable additional call information.



AudioCodes has also developed advanced diagnostic tools that may be used for high-level troubleshooting. These tools include the following:

- Call Progress Tone wizard (CPTWizard): helps detect the Call Progress Tones generated by the PBX. The software automatically creates a basic Call Progress Tones file.
- DSP Recording: DSP recording is a procedure used to monitor the DSP operation (e.g., RTP packets and events).

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

a. From an internal extension, dial the extension for a UM-enabled user and leave a voicemail message.

b. Confirm the voicemail message arrives in the called user's inbox.

c. Confirm this message displays a valid Active Directory name as the sender of this voicemail.

b. From an External Phone

a. From an external phone, dial the extension for a UM-enabled user and leave a voicemail message.

b. Confirm the voicemail message arrives in the called user's inbox.

c. 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:

  a. Under the Exchange Management Console, expand "Organizational Configuration" and then click on "Unified Messaging".

  b. Go to the Auto Attendant tab under the results pane.

  c. Click on the "New Auto Attendant…" under the action pane to invoke the AA wizard.

  d. Associate the AA with the appropriate dial plan and assign an extension for the AA.

  e. Create PBX dialing rules to always forward calls for the AA extension to the UM server.

  f. 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.
    a. UM Enable a user by invoking the “Enable-UMMailbox” wizard.
    b. Assign an unused extension to the user.
    c. Do not map the extension on the PBX to any user or device.
    d. Call Transfer by Directory Search to this user.
    e. Confirm the call fails and the caller is prompted with appropriate messages.

7. Play-On-Phone

- To access play-on-phone:
    a. Logon to Outlook Web Access (OWA) by going to URL https://<server name>/owa.
    b. After receiving a voicemail in the OWA inbox, open this voicemail message.
    c. At the top of this message, look for the Play-On-Phone field (Play on Phone...).
    d. 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:
    a. 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:
    b. $dp = get-umdialplan -id <dial plan ID>
    c. $dp.ConfiguredInCountryOrRegionGroups.Clear()
    d. $dp.ConfiguredInCountryOrRegionGroups.Add("anywhere,*,*,")
    e. $dp.AllowedInCountryOrRegionGroups.Clear()
    f. $dp.AllowedInCountryOrRegionGroups.Add(“anywhere")
    g. $dp|set-umdialplan
    h. $mp = get-ummailboxpolicy -id <mailbox policy ID>
    i. $mp.AllowedInCountryGroups.Clear()
    j. $mp.AllowedInCountryGroups.Add("anywhere")
    k. $mp|set-ummailboxpolicy
    l. The user must be enabled for external dialing on the PBX.
    m. 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:
    a. Double click on a user's mailbox and go to Mailbox Features tab.
    b. Click Unified Messaging and then click the properties button.
    c. Check the box "Allow faxes to be received".
- Management Shell - execute the following command:
    a. Set-UMMailbox –identity UMUser –FaxEnabled:$true
- To test fax functionality:
    a. Dial the extension for this fax-enabled UM user from a fax machine.
    b. Confirm the fax message is received in the user's inbox.
    c. 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.
    d. 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):
    a. Provide the configuration steps in Section 5.
    b. Configure the IP-PBX to work with two UM servers.
    c. Simulate a failure in one UM server.
    d. Confirm the IP-PBX transfers new calls to the other UM server successfully.