

# **Microsoft Exchange Server 2007 Unified Messaging**

## **PBX Configuration Note:**

### **Avaya S8300 with AudioCodes MP-11x FXO using Analog lines (In-band DTMF)**

By : AudioCodes

Updated Since : 2007-02-09

#### **READ THIS BEFORE YOU PROCEED**

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## Content

This document describes the configuration required to setup Avaya S8300 and AudioCodes MP-11x FXO using analog lines with inband DTMF 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
21 March 2007	Version 1

## 1. Components Information

### 1.1. PBX or IP-PBX

<b>PBX Vendor</b>	Avaya
<b>Model</b>	S8300
<b>Software Version</b>	G3xV11 Communication Manager 1.3
<b>Telephony Signaling</b>	Analog In-band DTMF Tones
<b>Additional Notes</b>	None

### 1.2. VoIP Gateway

<b>Gateway Vendor</b>	AudioCodes
<b>Model</b>	MP-11x FXO (MP-114 / MP-118)
<b>Software Version</b>	5.0
<b>VoIP Protocol</b>	SIP

### 1.3. Microsoft Exchange Server 2007 Unified Messaging

<b>Version</b>	RTM
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## 2. Prerequisites

### 2.1. Gateway Prerequisites

- Current Disconnect must be configured in the gateway and the duration must be adjusted for better performance for disconnecting calls when the PBX user has disconnected.
- The gateway also supports TLS (in addition to TCP). This provides security by enabling the encryption of SIP packets over the IP network.

### 2.2. PBX Prerequisites

- Enter the command "display system-parameters customer-options", and then ensure that the "Mode Code for Centralized Voice Mail?" parameter (on screen 3 of the PBX configuration tool) is set to "y".
- The PBX hardware includes an installed Analog Media Module MM711.

### 2.3. Cabling Requirements

This integration uses standard RJ-11 telephone line cords to connect analog ports between MM711 and MP-11x ports.

## 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

AudioCodes - Microsoft Internet Explorer

Address: http://172.20.22.200/

AudioCodes MP-118 FXO

Protocol Management | Advanced Parameters | Manipulation Tables | Routing Tables | Profile Definitions | Endpoint Phone Numbers | Hunt Group Settings | Endpoint Settings | FXO Settings

Quick Setup  
Protocol Management  
Advanced Configuration  
Status & Diagnostics  
Software Update  
Save Configuration  
Reset Device  
Log Off

**General**

PRACK Mode	Supported
Channel Select Mode	Ascending
Enable Early Media	Disable
Session-Expires Time	0
Minimum Session-Expires	90
Session Expires Method	Re-Invite
Asserted Identity Mode	Disabled
Fax Signaling Method	T.38 Relay
I Detect Fax on Answer Tone	Initiate T.38 on Preamble
SIP Transport Type	TCP
I SIP UDP Local Port	5060
I SIP TCP Local Port	5060
I SIP TLS Local Port	5061
Enable SIPS	Disable
SIP Destination Port	5060
Use "user=phone" in SIP URL	Yes

## Step 2: Routing Setup

AudioCodes - Microsoft Internet Explorer

Address: http://172.20.22.200/

AudioCodes MP-118 FXO

Protocol Definitions | Advanced Parameters | Manipulation Tables | **Routing Tables** | Profile Definitions | Endpoint Phone Numbers | Hunt Group Settings | Endpoint Settings | FXO Settings

Quick Setup  
Protocol Management  
Advanced Configuration  
Status & Diagnostics  
Software Update  
Save Configuration  
Reset Device  
Log Off

### Proxy & Registration

I Enable Proxy	Use Proxy
Proxy Name	
I Proxy IP Address	172.20.22.211
Gateway Name	
Gateway Registration Name	
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
Enable SRV Queries	Disable
Enable Proxy SRV Queries	Disable
Redundancy Mode	Parking
I Enable Registration	Disable
Registrar Name	
Registrar IP Address	
Registration Time	180

**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, 172.20.22.211 or the FQDN of the Microsoft Unified Messaging host).

### Step 3: SIP Environment Setup (Cont.)

AudioCodes - Microsoft Internet Explorer

Address: http://172.20.22.200/

AudioCodes MP-118 FXO

Navigation Menu:

- Quick Setup
- Protocol Management**
- Advanced Configuration
- Status & Diagnostics
- Software Update
- Save Configuration
- Reset Device
- Log Off

Configuration Parameters (Protocol Definition tab):

Parameter	Value
Re-registration Timing [%]	50
Registration Retry Time	30
Subscription Mode	Per Gateway
I Enable Proxy Keep Alive	Disable
Proxy Keep Alive Time	60
Use Gateway Name for OPTIONS	No
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
Number of RTX Before Hot-Swap	3
User Name	
Password	.....
Cnonce	Default_Cnonce
I Authentication Mode	Per Endpoint

SIP

Done Internet

#### Step 4: Coder Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Home Search Favorites Reload Print Mail New Tab

Address http://172.20.22.200/ Go Links

Google Go Bookmarks 6 blocked Check AutoLink AutoFill Send to Settings

### AudioCodes MP-118 FXO

- Protocol Definition
- Advanced Parameters
- Manipulation Tables
- Routing Tables
- Profile Definitions
- Endpoint Phone Numbers
- Hunt Group Settings
- Endpoint Settings
- FXO Settings

- Quick Setup
- Protocol Management**
- Advanced Configuration
- Status & Diagnostics
- Software Update
- Save Configuration
- Reset Device
- Log Off

#### 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

Submit

SIP

Done Internet



## Step 5: Digit Collection Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Reload Home Search Favorites RSS Print Mail AutoLink AutoFill Send to Settings

Links Interop Tasks Google G Go Bookmarks 156 blocked Check AutoLink AutoFill Send to Settings

Address http://10.15.6.2/ Go

**AudioCodes** **MP-114 FXO**

Protocol Definitions Advanced Parameters Manipulation Tables Routing Tables Profile Definitions Endpoint Phone Numbers Hunt Group Settings Endpoint Settings Advanced Applications RADIUS Parameters

**DTMF & Dialing**

Max Digits In Phone Num	15
Inter Digit Timeout for Overlap Dialing [sec]	2
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
Digit Mapping Rules	
Dial Tone Duration [sec]	16
Hotline Dial Tone Duration [sec]	16
Enable Special Digits	Enable
Hook-Flash Option	Not Supported
Default Destination Number	1000

Quick Setup  
Protocol Management  
Advanced Configuration  
Status & Diagnostics  
Software Update  
Maintenance  
Log Off

Search

Done Internet

## Step 6: Disconnect Supervision Setup

The screenshot shows the AudioCodes MP-114 FXO configuration page in Microsoft Internet Explorer. The browser window title is "AudioCodes - Microsoft Internet Explorer". The address bar shows "http://10.15.6.2/". The page has a navigation menu on the left with options: Quick Setup, Protocol Management (highlighted), Advanced Configuration, Status & Diagnostics, Software Update, Maintenance, and Log Off. The main content area is titled "General Parameters" and contains a table of settings. A red double-headed arrow points to the "Disconnect and Answer Supervision" section of the table.

General Parameters	
IP Security	Disable
Filter Calls to IP	Don't Filter
I Enable Digit Delivery to Tel	Disable
I Enable Digit Delivery to IP	Disable
RTP Only Mode	Disable
Enable DID Wink	Disable
<b>Disconnect and Answer Supervision</b>	
Enable Polarity Reversal	Disable
Enable Current Disconnect	Enable
Disconnect on Broken Connection	No
Broken Connection Timeout [100 msec]	100
Disconnect Call on Silence Detection	No
Silence Detection Period [sec]	120
Silence Detection Method	Voice/Energy Detectors
<b>CDR and Debug</b>	
CDR Server IP Address	
CDR Report Level	End Call

## Step 7: Message Waiting Indication Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Reload Home Search Favorites

Address http://172.20.22.200/ Go Links

Google Bookmarks 6 blocked Check AutoLink AutoFill Send to Settings

**AudioCodes** MP-118 FXO

Protocol Definition **Advanced Parameters** Manipulation Tables Routing Tables Profile Definitions Endpoint Phone Numbers Hunt Group Settings Endpoint Settings FXO Settings

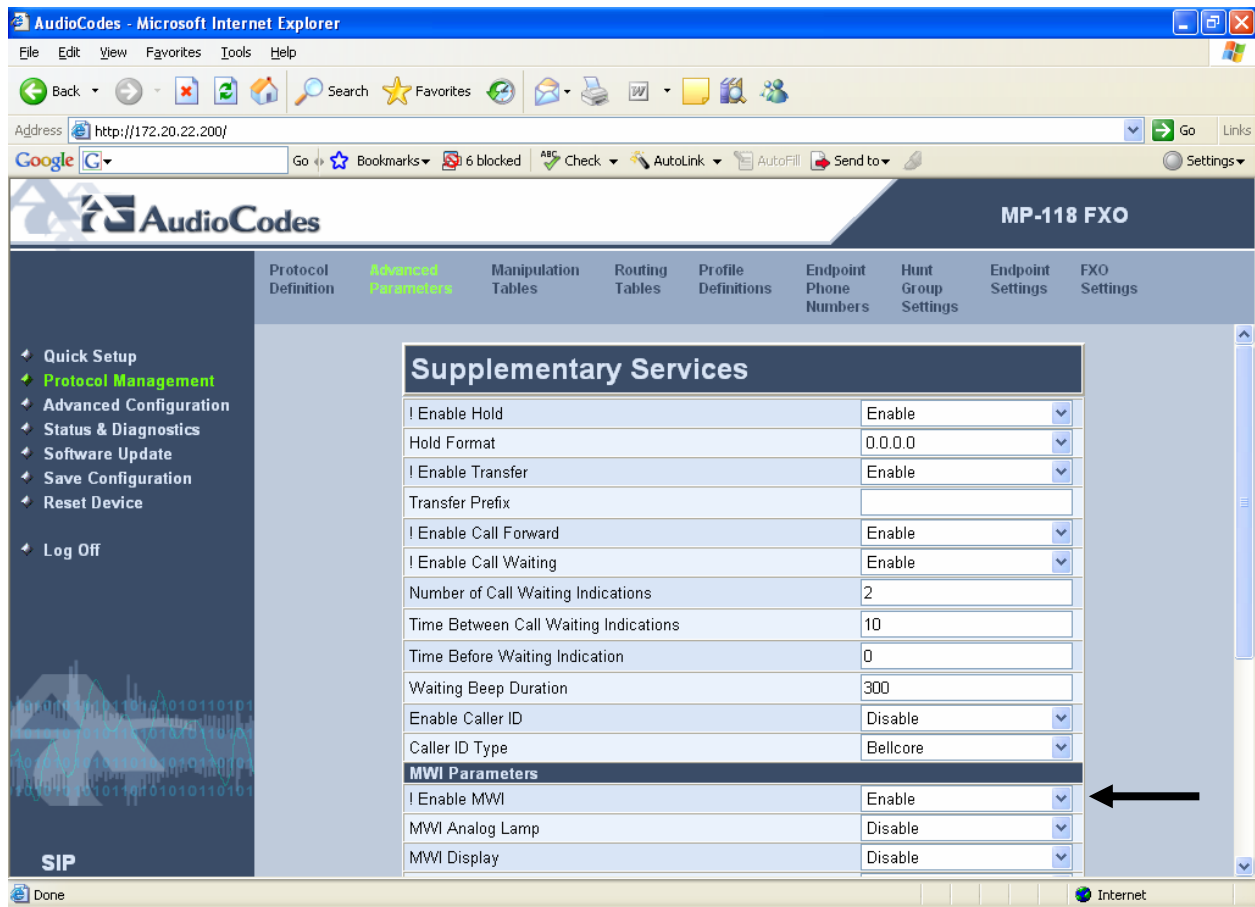
- Quick Setup
- Protocol Management**
- Advanced Configuration
- Status & Diagnostics
- Software Update
- Save Configuration
- Reset Device
- Log Off

### Supplementary Services

I Enable Hold	Enable
Hold Format	0.0.0.0
I Enable Transfer	Enable
Transfer Prefix	
I Enable Call Forward	Enable
I 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	Bellcore
<b>MWI Parameters</b>	
I Enable MWI	Enable
MWI Analog Lamp	Disable
MWI Display	Disable

SIP

Done Internet



## Step 8: Manipulation Routing Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Reload Home Search Favorites RSS Print Mail News Groups

Address http://172.20.22.200/ Go Links

Google G Go Bookmarks 9 blocked Check AutoLink AutoFill Send to Settings

**AudioCodes** **MP-118 FXO**

Protocol Definition Advanced Parameters **Manipulation Tables** Routing Tables Profile Definitions Endpoint Phone Numbers Hunt Group Settings Endpoint Settings FXO Settings

Quick Setup  
Protocol Management  
Advanced Configuration  
Status & Diagnostics  
Software Update  
Save Configuration  
Reset Device  
Log Off

**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					

Submit

Done Internet

## Step 9: Endpoints Setup

AudioCodes - Microsoft Internet Explorer

Address: <http://172.20.22.200/>

AudioCodes MP-118 FXO

Protocol Definition | Advanced Parameters | Manipulation Tables | Routing Tables | Profile Definitions | **Endpoint Phone Numbers** | Hunt Group Settings | Endpoint Settings | FXO Settings

Quick Setup  
Protocol Management  
Advanced Configuration  
Status & Diagnostics  
Software Update  
Save Configuration  
Reset Device  
Log Off

**Endpoint Phone Number Table**

Channel(s)	Phone Number	Hunt Group ID	Profile ID
1	11111		0
2	11111		0
3			
4			
5			
6			
7			
8			

Register Un-Register  
Submit

SIP

Web Server Internet

**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, 11111).

## Step 10: Voicemail In-Band DTMF Setup

The screenshot shows the AudioCodes MP-118 FXO configuration page in Microsoft Internet Explorer. The browser's address bar shows the URL `http://172.20.22.200/`. The page has a navigation menu on the left with options like Quick Setup, Protocol Management, Advanced Configuration, Status & Diagnostics, Software Update, Save Configuration, Reset Device, and Log Off. The main content area is titled 'Voice Mail' and contains several sections: General, Digit Patterns, MWI, and SMDI. The 'General' section has two dropdown menus: 'Voice Mail Interface' set to 'DTMF' and 'Line Transfer Mode' set to 'Blind Transfer'. The 'Digit Patterns' section has seven text input fields for various digit patterns. The 'MWI' section has two text input fields for 'MWI Off Digit Pattern' and 'MWI On Digit Pattern'. The 'SMDI' section has one dropdown menu for 'Enable SMDI' set to 'Disable'. Two arrows point to the 'Voice Mail Interface' and 'Line Transfer Mode' dropdown menus. The status bar at the bottom shows 'Done' and 'Internet'.

Voice Mail	
<b>General</b>	
Voice Mail Interface	DTMF
Line Transfer Mode	Blind Transfer
<b>Digit Patterns</b>	
Forward on Busy Digit Pattern	#02#S.#R.#
Forward on No Answer Digit Pattern	#02#S.#R.#
Forward on Do Not Disturb Digit Pattern	#03##R.#
Forward on No Reason Digit Pattern	#02#S.#R.#
Internal Call Digit Pattern	#00#S.##
External Call Digit Pattern	#01#S.##
Disconnect Call Digit Pattern	
<b>MWI</b>	
MWI Off Digit Pattern	#2
MWI On Digit Pattern	*2
<b>SMDI</b>	
Enable SMDI	Disable

## Step 11: FAX Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Reload Home Search Favorites RSS Print Mail Wordpad PDF Settings

Address http://172.20.22.200/ Go Links

Google Go Bookmarks 6 blocked Check AutoLink AutoFill Send to Settings

**AudioCodes** MP-118 FXO

Network Settings **Media Settings** Configuration File Regional Settings Security Settings Management Settings

- Quick Setup
- Protocol Management
- Advanced Configuration**
- Status & Diagnostics
- Software Update
- Save Configuration
- Reset Device
- Log Off

**Fax/Modem/CID Settings**

Fax Transport Mode	T.38 Relay
Caller ID Transport Type	Mute
Caller ID Type	Bellcore
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

SIP

Done Internet

## Step 12: FXO General Setup

The screenshot shows the AudioCodes MP-118 FXO configuration page. The browser is Microsoft Internet Explorer, and the address bar shows <http://172.20.22.200/>. The page has a navigation menu on the left with options like Quick Setup, Protocol Management, Advanced Configuration, Status & Diagnostics, Software Update, Save Configuration, Reset Device, and Log Off. The main content area is titled 'FXO Settings' and contains a table of configuration parameters. Two arrows point to the 'One Stage' value for 'Dialing Mode' and the 'Yes' value for 'Answer Supervision'.

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]	0
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

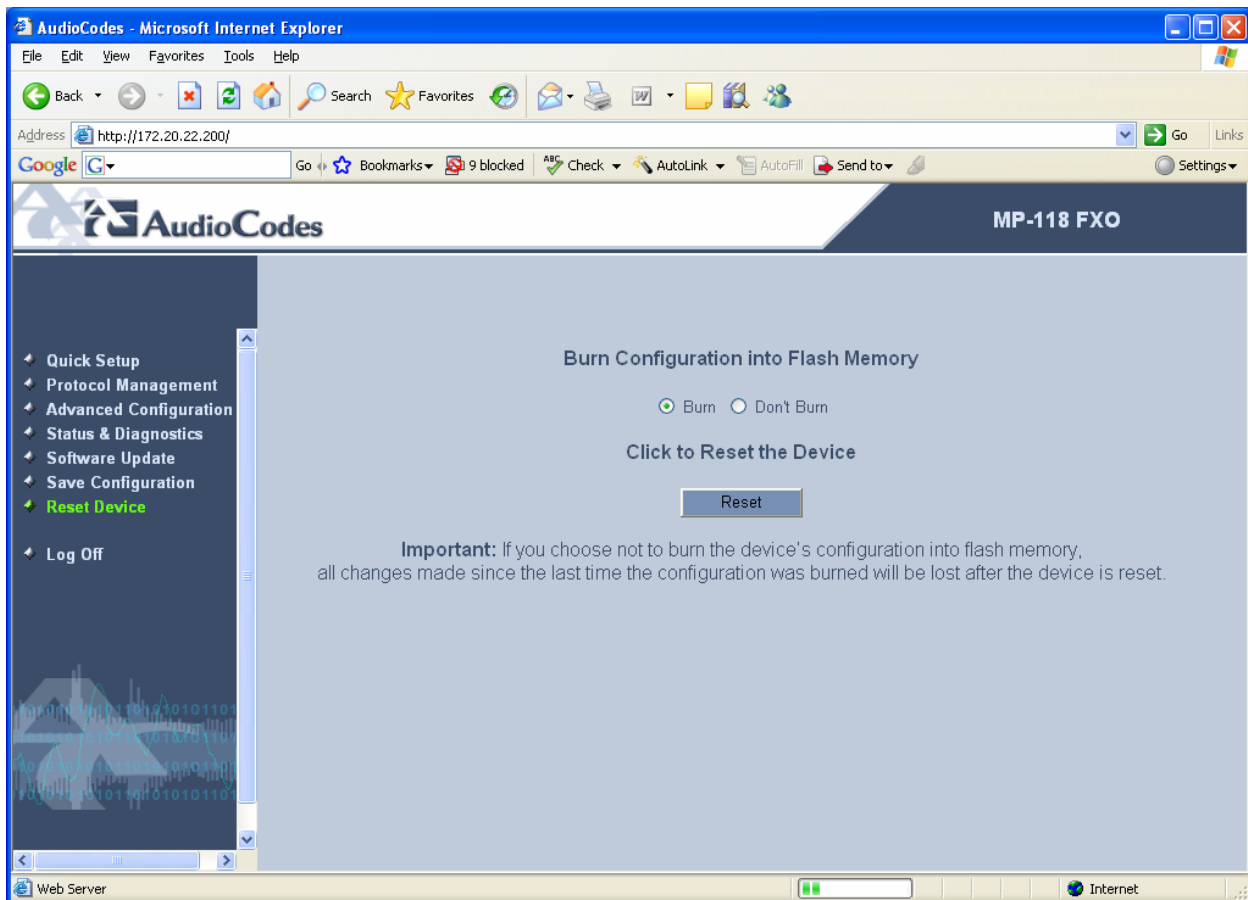
Submit

## Step 13: FXO General Setup (Cont.)

- CurrentDisconnectDuration = 450
- EchoCancellerAggressiveNLP = 1
- EnableDetectRemoteMACChange = 2
- ECNLPMODE = 1



## Step 14: Reset FXO

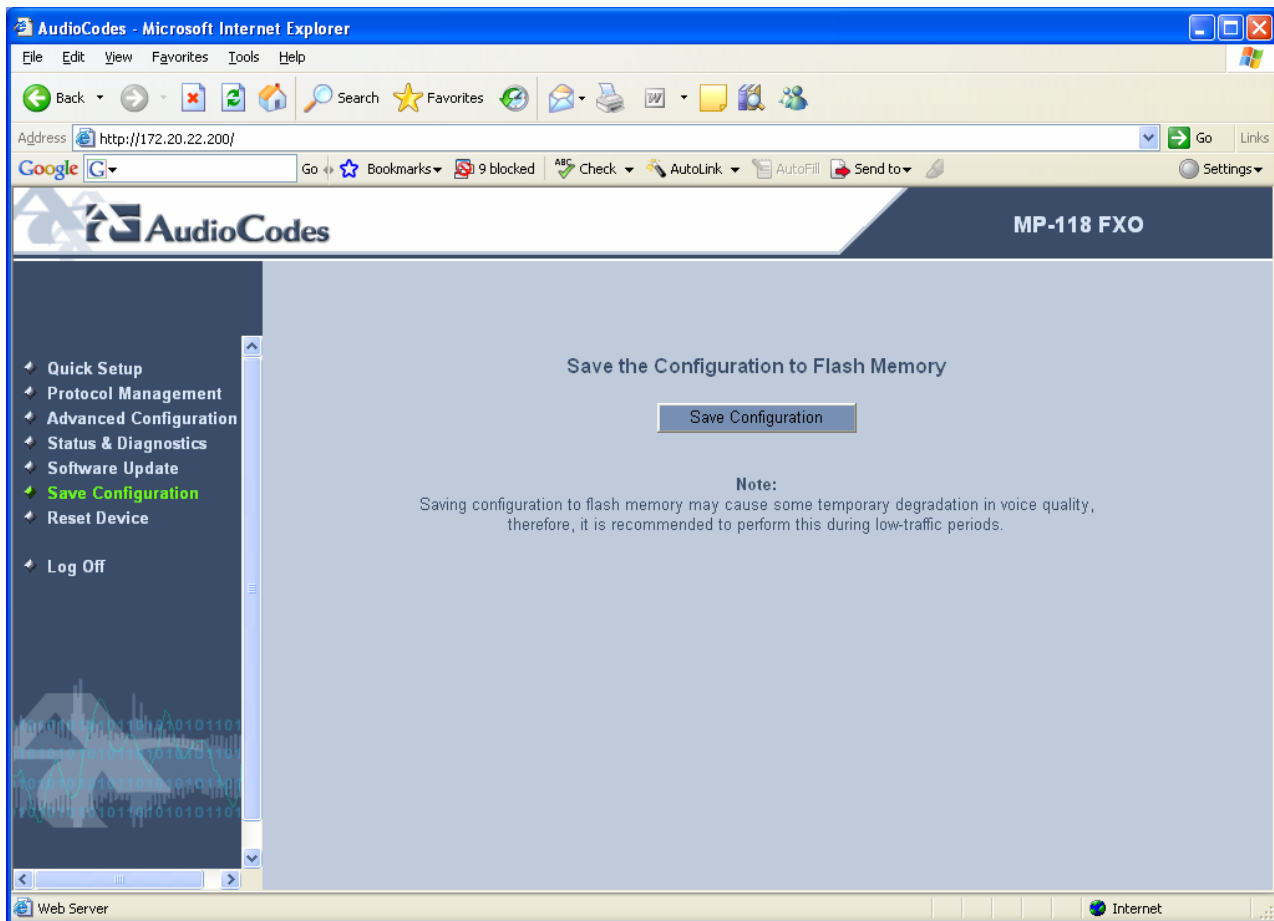


**Note:** Steps 1 and 7 involve core configuration changes (versus default settings):

- Proxy IP Address (Microsoft Unified Messaging IP address)
- Enabling Message Waiting process

These changes require a gateway reset (by default, when performing a gateway reset, the configuration is burnt to flash memory). If no change is made to these two core configuration parameters, skip to Step 15.

## Step 15: Save Gateway Configuration



**Note:** This step is optional and is not required if you performed Step 14.

## 4.1. Configuration Files

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



INI Avaya S8300.zip

## 4.2. TLS Setup

### Step 1: PBX to IP Routing Setup

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.6.1/

AudioCodes MP-114 FXO

Protocol Definitions | Advanced Parameters | Manipulation Tables | Routing Tables | Profile Definitions | Endpoint Phone Numbers | Hunt Group Settings | Endpoint Settings | Advanced Applications | RADIUS Parameters

**Proxy & Registration**

Enable Proxy	Use Proxy
Proxy Name	exchange2007.com
Proxy IP Address	10.15.3.207
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 Invites to Proxy	No

**Note:** The Proxy IP Address and Name must be one that corresponds to the network environment in which the Microsoft Unified Messaging server is installed (For example, 10.15.3.207 for IP Address and exchaneg2007.com for the FQDN of the Microsoft Unified Messaging host).

## Step 2: SIP Environment and Gateway Name Setup

The screenshot shows the AudioCodes MP-114 FXO configuration interface. The browser window is titled 'AudioCodes - Microsoft Internet Explorer' and the address bar shows 'http://10.15.6.1/'. The page has a navigation menu on the left with options like 'Quick Setup', 'Protocol Management', 'Advanced Configuration', 'Status & Diagnostics', 'Software Update', 'Maintenance', and 'Log Off'. The main content area has tabs for 'Protocol Definition', 'Advanced Parameters', 'Manipulation Tables', 'Routing Tables', 'Profile Definitions', 'Endpoint Phone Numbers', 'Hunt Group Settings', 'Endpoint Settings', 'Advanced Applications', and 'RADIUS Parameters'. The 'Protocol Definition' tab is active, showing a list of settings for the gateway. Two black arrows point to the 'Gateway Name' and 'Subscription Mode' fields.

Setting	Value
Send All Invite to Proxy	No
Enable Proxy Hot-Swap	Disable
Enable Registration	Disable
Gateway Name	gw2.fxo.audiocodes.com
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

Buttons: Register, Un-Register, Submit

**Note:** Assign an FQDN name to the gateway (for example, gw2.fxoaudiocodes.com). Any gateway name that corresponds to your network environment is applicable; the only limitation is not to include underscores in the name (Windows Certification server limitation).

### Step 3: SIP Environment Setup (Cont.)

The screenshot shows the AudioCodes MP-114 FXO configuration interface. The browser window is titled 'AudioCodes - Microsoft Internet Explorer' and the address bar shows 'http://10.15.6.2/'. The page has a navigation menu on the left and a main configuration area. The 'General' tab is selected under the 'Protocol Definitions' section. The configuration table lists various SIP parameters and their values. Black arrows on the right side of the table point to the following settings:

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
I Detect Fax on Answer Tone	Initiate T.38 on Preamble
SIP Transport Type	TLS
SIP UDP Local Port	5000
SIP TCP Local Port	5040
SIP TLS Local Port	5060
Enable SIPs	Disable
Enable TCP Connection Reuse	Enable
SIP Destination Port	5061

#### Step 4: DNS Servers Setup

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.6.2/

AudioCodes MP-114 FXO

Network Settings | Media Settings | Configuration File | Regional Settings | Security Settings | Management Settings

Quick Setup  
Protocol Management  
Advanced Configuration  
Status & Diagnostics  
Software Update  
Maintenance  
Log Off

Search

IP Settings

IP Networking Mode	Single IP Network
IP Address	10.15.6.2
Subnet Mask	255.255.0.0
Default Gateway Address	10.15.0.1

DNS Settings

DNS Primary Server IP	10.1.1.11
DNS Secondary Server IP	10.1.1.10

DHCP Settings

Enable DHCP	Disable
-------------	---------

NAT Settings

I NAT IP Address	0.0.0.0
------------------	---------

Differential Services

Network QoS	48
Media Premium QoS	46
Control Premium QoS	40
Gold QoS	25

Done Internet

**Note:** Define the primary and secondary DNS servers' IP addresses so that they correspond to your network environment (for example, 10.1.1.11 and 10.1.1.10). If no DNS server is available in the network, then skip this step.

## Step 5: Internal DNS Setup

The screenshot shows the AudioCodes MP-114 FXO web interface in a Microsoft Internet Explorer browser. The address bar shows <http://10.15.6.2/>. The interface has a top navigation bar with the AudioCodes logo and the model name MP-114 FXO. Below this is a menu bar with the following items: Protocol Definition, Advanced Parameters, Manipulation Tables, **Routing Tables** (highlighted), Profile Definitions, Endpoint Phone Numbers, Hunt Group Settings, Endpoint Settings, Advanced Applications, and RADIUS Parameters. On the left side, there is a sidebar menu with a home icon and the following options: Quick Setup, **Protocol Management** (highlighted), Advanced Configuration, Status & Diagnostics, Software Update, Maintenance, and Log Off. Below the sidebar menu is a search bar with a search button. The main content area displays the 'Internal DNS Table' configuration. It consists of a table with 10 rows and 3 columns: Domain Name, First IP Address, and Second IP Address. The first row is pre-filled with 'exchange2007.com' and '10.15.3.207'. The other rows are empty. Below the table is a 'Submit' button.

	Domain Name	First IP Address	Second IP Address
1	exchange2007.com	10.15.3.207	
2			
3			
4			
5			
6			
7			
8			
9			
10			

Submit

**Note:** If no DNS server is available in the network, define the internal DNS table where the domain name is the FQDN of the Microsoft Unified Messaging server and the First IP Address corresponds to its IP address (for example, exchange2007.com and 10.15.3.207).



## Step 6: NTP Server Setup

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.6.2/

AudioCodes MP-114 FXO

Network Settings Media Settings Configuration File Regional Settings Security Settings Management Settings

Quick Setup  
Protocol Management  
Advanced Configuration  
Status & Diagnostics  
Software Update  
Maintenance  
Log Off

Search

Application Settings

NTP Settings

NTP Server IP Address 10.15.6.50

NTP UTC Offset Hours 0 Minutes 0

NTP Update Interval Hours 24 Minutes 0

Telnet Settings

I Embedded Telnet Server Disable

I Telnet Server TCP Port 23

I Telnet Server Idle Timeout 0

STUN Settings

Enable STUN Disable

STUN Server Primary IP 0.0.0.0

STUN Server Secondary IP 0.0.0.0

NFS Settings

NFS Table -->

Submit

Done Internet

**Note:** Define the NTP server's IP address so that it corresponds to your network environment (for example, 10.15.3.50). If no NTP server is available in the network, then skip this step (as the gateway uses its internal clock).

## Step 7: Generate Certificate Setup

Use the screen below to generate CSR. Copy the certificate signing request and send it to your Certification Authority for signing.

The screenshot shows the AudioCodes MP-114 FXO web interface in a Microsoft Internet Explorer browser. The address bar shows the URL `http://10.15.6.2/`. The page title is "AudioCodes" and the model "MP-114 FXO" is displayed in the top right corner. The navigation menu includes "Network Settings", "Media Settings", "Configuration File", "Regional Settings", "Security Settings" (highlighted), and "Management Settings". The left sidebar contains a "Home" icon and a list of options: "Quick Setup", "Protocol Management", "Advanced Configuration" (highlighted), "Status & Diagnostics", "Software Update", "Maintenance", and "Log Off".

The main content area is titled "Certificate Signing Request". It features a text input field for "Subject Name" containing the value `gw2.fxo.audiocodes.com` and a "Generate CSR" button. Below this, a message states: "Copy the certificate signing request and send it to your Certification Authority for signing."

The generated CSR text is as follows:

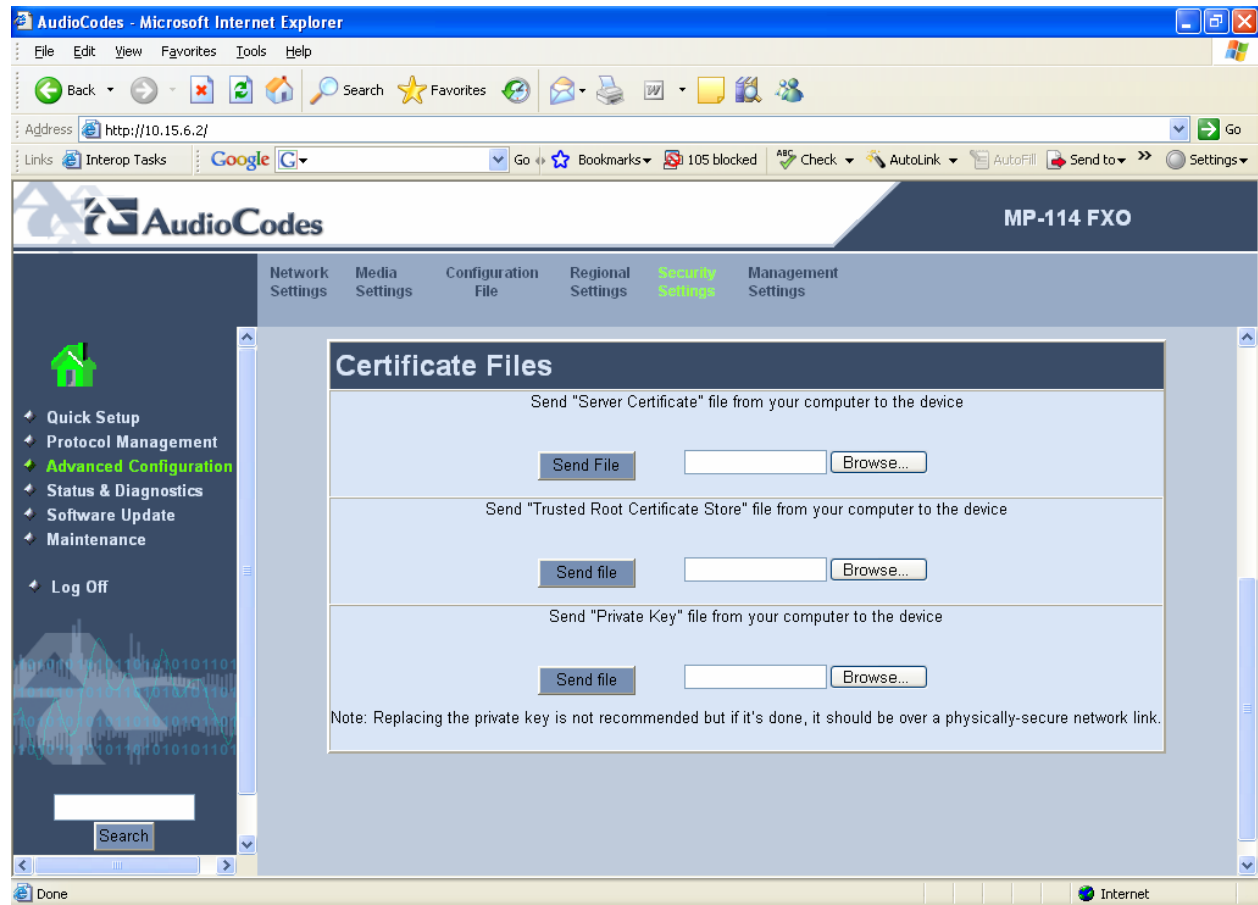
```
-----BEGIN CERTIFICATE REQUEST-----
MIIBYDCBygIBADAhMR8wHQYDVQQDEx2ndzIubTJrLmF1ZG1vY29kZXMuY29tMIGf
MA0GCSqGSIb3DQEBAQUAA4GNADCB1QKBgQCw7x/nSvJJzUOKsSRvYoGS9LguLjGJ
oK4td5mMRE9isdMVy4iogiMEJLG51BSQ5Jta9J8Sd/d7munYt3utpUnIjbbfG5OG
K1br4pR3jG9jot2oWjqhemH34hTouhjc1dJQLY21e12RrOXcL6Qa1AQ1pZPrMnLM
eBFdwSPJ8bGAQwIDAQABoAAwDQYJKoZIhvcNAQEEBQADgYEAHxJZM48IXV+MjiPe
Qnn580m9Xs1ht2S1E075G6O/5su+tZuV10X1ss1hvlorqiOJxWJRgH4bBP+G8g8F
sSLG2Bkq08QghFTT2PXp1CiqqTyI3Ru6Ge4UCOTWtb/DRmUU1qZtMfuHy7DqcmHd
gcdLo3ncyAgVoOMLOy8LPreKqPc=
-----END CERTIFICATE REQUEST-----
```

## Step 8: Uploading Certificates Setup

The screen below is used to upload the sign certificates.

In the "Server Certificate" area, upload the gateway certificate signed by the CA.

In the "Trusted Root Certificate Store" area, upload the CA certificate.



## 5. PBX Setup Notes

Information used for this test case:

- Analog VoiceMail ports: ext. 5098 and 5099
- VoiceMail Hunt Group Pilot: ext. 5095
- VoiceMail User Phone: ext. 5006
- Coverage Path for ext. 5006 is 1
- User Test Phone Avaya 2420

**Step 1:** Define analog voice mail ports:

“add station” or “cha station” command:

Page 1: Set port type as VMI

The screenshot shows the Avaya Site Administration - [S8300 GEDI] window. The title bar includes the application name and standard window controls. The menu bar contains File, Edit, View, System, Action, Tools, Window, and Help. Below the menu bar is a toolbar with various icons. A status bar at the top of the main area shows 'change station 5098' and several function keys: send (return), help (F5), cancel (esc), enter (F3), schedule (F9), next (F7), and previous (F8). The main area is divided into two sections: 'STATION' and 'STATION OPTIONS'. The 'STATION' section contains fields for Extension (5098), Type (VMI), Port (0010207), Name (Analog 5098), Lock Messages? (n), Security Code (empty), BCC (0), TN (1), COR (1), COS (1), and Tests? (y). The 'STATION OPTIONS' section contains fields for Loss Group (1) and Off Premises Station? (n). At the bottom of the window, there is a status bar with the text 'Right-click in a field to see a list of valid entries or help text' and 'Ready'. The Windows taskbar is visible at the very bottom, showing the Start button, several open applications, and the system clock displaying 2:54 PM.

Avaya Site Administration - [S8300 GEDI]

File Edit View System Action Tools Window Help

change station 5098 send (return) help (F5) cancel (esc) enter (F3) schedule (F9) next (F7) previous (F8)

1 2 3

STATION

Extension: 5098 Lock Messages?  BCC: 0  
Type:  Security Code:  TN:   
Port:  COR:   
Name:  COS:   
Tests?

STATION OPTIONS

Loss Group:   
Off Premises Station?

Right-click in a field to see a list of valid entries or help text  
Ready

Start Win... A.. C.. M.. C.. S.. u... 2:54 PM

Page 2: Set adjunct supervision to Y

**Avaya Site Administration - [S8300 GEDII]**

File Edit View System Action Tools Window Help

change station 5098 send (return) help (f5) cancel (esc) enter (f3) schedule (f9) next (f7) previous (f8)

1 2 3

**STATION**

**FEATURE OPTIONS**

LWC Activation?	<input checked="" type="checkbox"/>	Coverage Msg Retrieval?	<input checked="" type="checkbox"/>
LWC Log External Calls?	<input type="checkbox"/>	Auto Answer:	none
CDR Privacy?	<input type="checkbox"/>	Data Restriction?	<input type="checkbox"/>
Redirect Notification?	<input checked="" type="checkbox"/>		
Per Button Ring Control?	<input type="checkbox"/>		
Bridged Call Alerting?	<input type="checkbox"/>	Distinctive Audible Alert?	<input checked="" type="checkbox"/>
Switchhook Flash?	<input checked="" type="checkbox"/>	Adjunct Supervision?	<input checked="" type="checkbox"/>
Ignore Rotary Digits?	<input type="checkbox"/>		
H.320 Conversion?	<input type="checkbox"/>	Per Station CPN - Send Calling Number?	<input type="checkbox"/>
Service Link Mode:	as-needed		
Multimedia Mode:	basic	Audible Message Waiting?	<input type="checkbox"/>
MWI Served User Type:			
AUDIX Name:		Coverage After Forwarding?	<input checked="" type="checkbox"/>
		Multimedia Early Answer?	<input type="checkbox"/>

Emergency Location Ext: 5098

Right-click in a field to see a list of valid entries or help text

Ready

Start Win... A.. C.. M.. C.. S.. u... 2:55 PM

Repeat this process for all analog VoiceMail ports.

**Step 2:** Define a hunt group for analog VoiceMail ports:

“add hunt-group” or “cha hunt-group” command

Page 1: Set group type as UCD-MIA

The screenshot displays the Avaya Site Administration - [S8300 GEDI] window. The interface includes a menu bar (File, Edit, View, System, Action, Tools, Window, Help) and a toolbar with various icons. Below the toolbar is a navigation bar with buttons for "change hunt-group 1", "send (return)", "help (f5)", "cancel (esc)", "enter (f3)", "schedule (f9)", "next (f7)", and "previous (f8)". A tab bar at the top shows tabs numbered 1 through 32, with tab 1 selected. The main area is titled "HUNT GROUP" and contains the following configuration fields:

Group Number:	1	ACD?	<input type="checkbox"/>
Group Name:	UH Analog	Queue?	<input type="checkbox"/>
Group Extension:	5095	Vector?	<input type="checkbox"/>
Group Type:	ucd-mia	Coverage Path:	
TN:	1	Night Service Destination:	
COR:	1	MM Early Answer?	<input type="checkbox"/>
Security Code:			
ISDN Caller Display:	grp-name		
Queue Length:	2		
Calls Warning Threshold:		Port:	
Time Warning Threshold:		Port:	

At the bottom of the window, there is a status bar with the text "Right-click in a field to see a list of valid entries or help text" and "Ready". The Windows taskbar at the very bottom shows the Start button, several application icons, and the system clock displaying "3:01 PM".

Add analog voice mail ports to this hunt group.

Page 3.

**Avaya Site Administration - [S8300 GEDI]**

File Edit View System Action Tools Window Help

S8300

change hunt-group 1 send (return) help (F5) cancel (esc) enter (F3) schedule (F9) next (F7) previous (F8)

1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 31 32 3: ▶

**HUNT GROUP**

Group Number: 1 Group Extension: 5095 Group Type: ucd-mia  
Member Range Allowed: 1 - 1500 Administered Members (min/max): 1 /2  
Total Administered Members: 2

**GROUP MEMBER ASSIGNMENTS**

Ext	Name (24 characters)	Ext	Name (24 characters)
1: 5098	Analog 5098	14:	
2: 5099	Analog 5099	15:	
3:		16:	
4:		17:	
5:		18:	
6:		19:	
7:		20:	
8:		21:	
9:		22:	
10:		23:	
11:		24:	
12:		25:	
13:		26:	

At End of Member List

Right-click in a field to see a list of valid entries or help text

Ready

Start W... A C M C S u 3:05 PM



**Step 3:** Set the coverage path

"add coverage-path" or "cha coverage-path" commands

Point to the analog voice mail hunt group 5095

H1 in point1

The screenshot shows the Avaya Site Administration - [S8300 GED] window. The main area is titled "COVERAGE PATH" and displays the following configuration:

Coverage Path Number: 1  
Next Path Number:   
Hunt after Coverage? ☐ n  
Linkage

**COVERAGE CRITERIA**

Station/Group Status	Inside Call	Outside Call
Active?	<input type="checkbox"/> n	<input type="checkbox"/> n
Busy?	<input type="checkbox"/> u	<input type="checkbox"/> u
Don't Answer?	<input type="checkbox"/> u	<input type="checkbox"/> u
All?	<input type="checkbox"/> n	<input type="checkbox"/> n
DND/SAC/Goto Cover?	<input type="checkbox"/> u	<input type="checkbox"/> u

Number of Rings:  4

**COVERAGE POINTS**

Terminate to Coverage Pts. with Bridged Appearances? ☐ n

Point1:  h1 Rng: ☐ Point2:   
Point4:  Point5:  Point6:

Right-click in a field to see a list of valid entries or help text

Ready

The taskbar at the bottom shows the Start button, several application icons, and the system clock displaying 2:48 PM.

**Step 4:** Add or change station coverage path:

Define user station (analog or digital) with coverage path:

Use "add station" or "change station" command

The screenshot shows the Avaya Site Administration interface for station configuration. The window title is "Avaya Site Administration - [S8300 GEDI]". The menu bar includes File, Edit, View, System, Action, Tools, Window, and Help. The toolbar contains various icons for file operations and navigation. Below the toolbar, there is a status bar with the text "change station 5006" and several function keys: send (return), help (F5), cancel (esc), enter (F3), schedule (F9), next (F7), and previous (F8).

The main configuration area is titled "STATION" and contains the following fields:

- Extension: 5006
- Type: 2420
- Port: 0010401
- Name: 2420
- Lock Messages?: n
- Security Code: [empty]
- Coverage Path 1: 1
- Coverage Path 2: [empty]
- Hunt-to Station: [empty]
- BCC: 0
- TN: 1
- COR: 1
- COS: 1

Below the "STATION" section is the "STATION OPTIONS" section with the following fields:

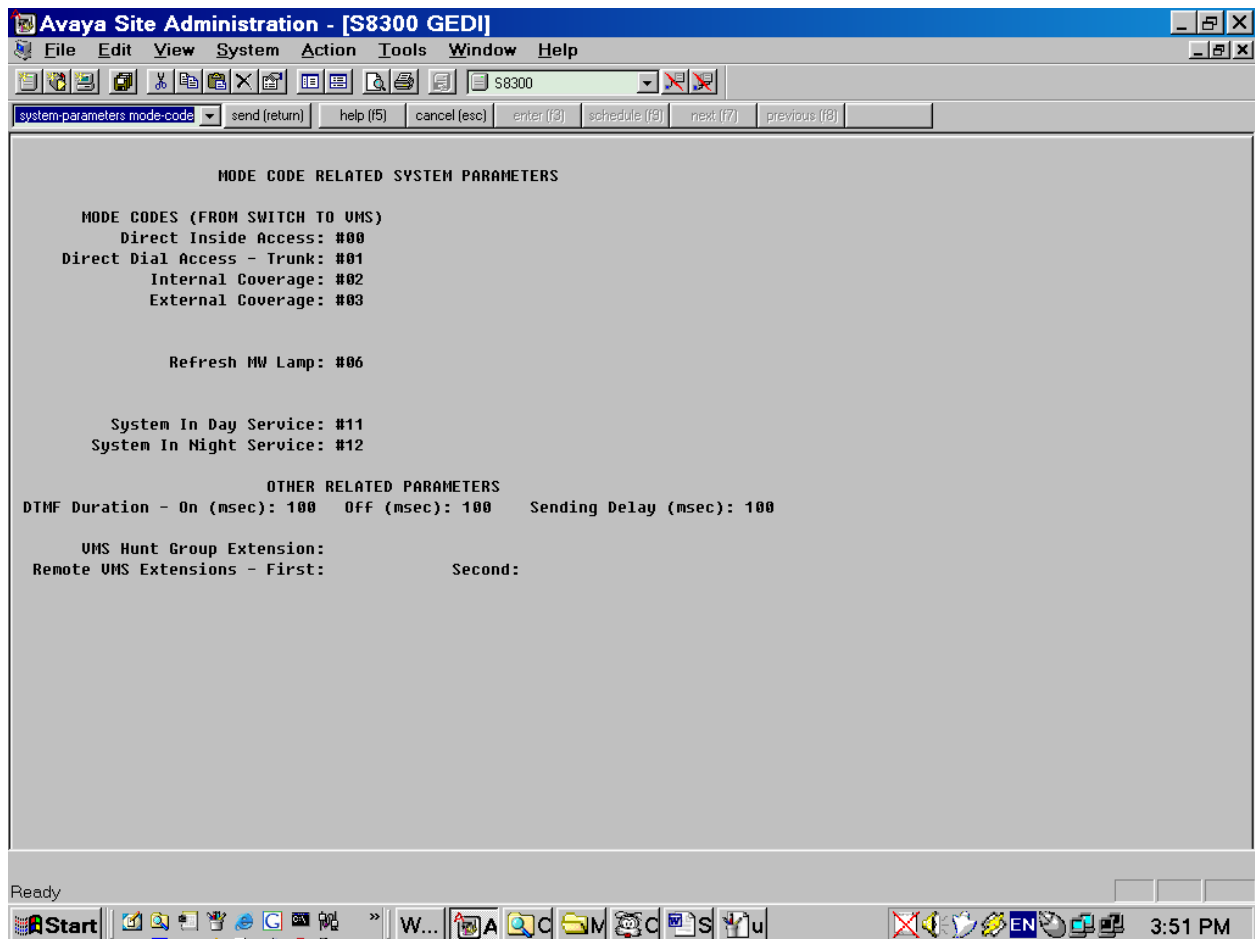
- Loss Group: 2
- Data Option: none
- Speakerphone: 2-way
- Display Language: english
- Personalized Ringing Pattern: 2
- Message Lamp Ext: 5006
- Mute Button Enabled?: y
- Expansion Module?: n
- Media Complex Ext: [empty]
- IP SoftPhone?: n
- Remote Office Phone?: n

At the bottom of the window, there is a status bar with the text "Right-click in a field to see a list of valid entries or help text" and "Ready". The taskbar at the very bottom shows the Start button, several open applications, and the system clock displaying "2:46 PM".

**Step 5:** Display mode-code information.

It shows DTMF codes sent by the PBX associated with different types of calls.

Use "display system-parameters mode-code" command

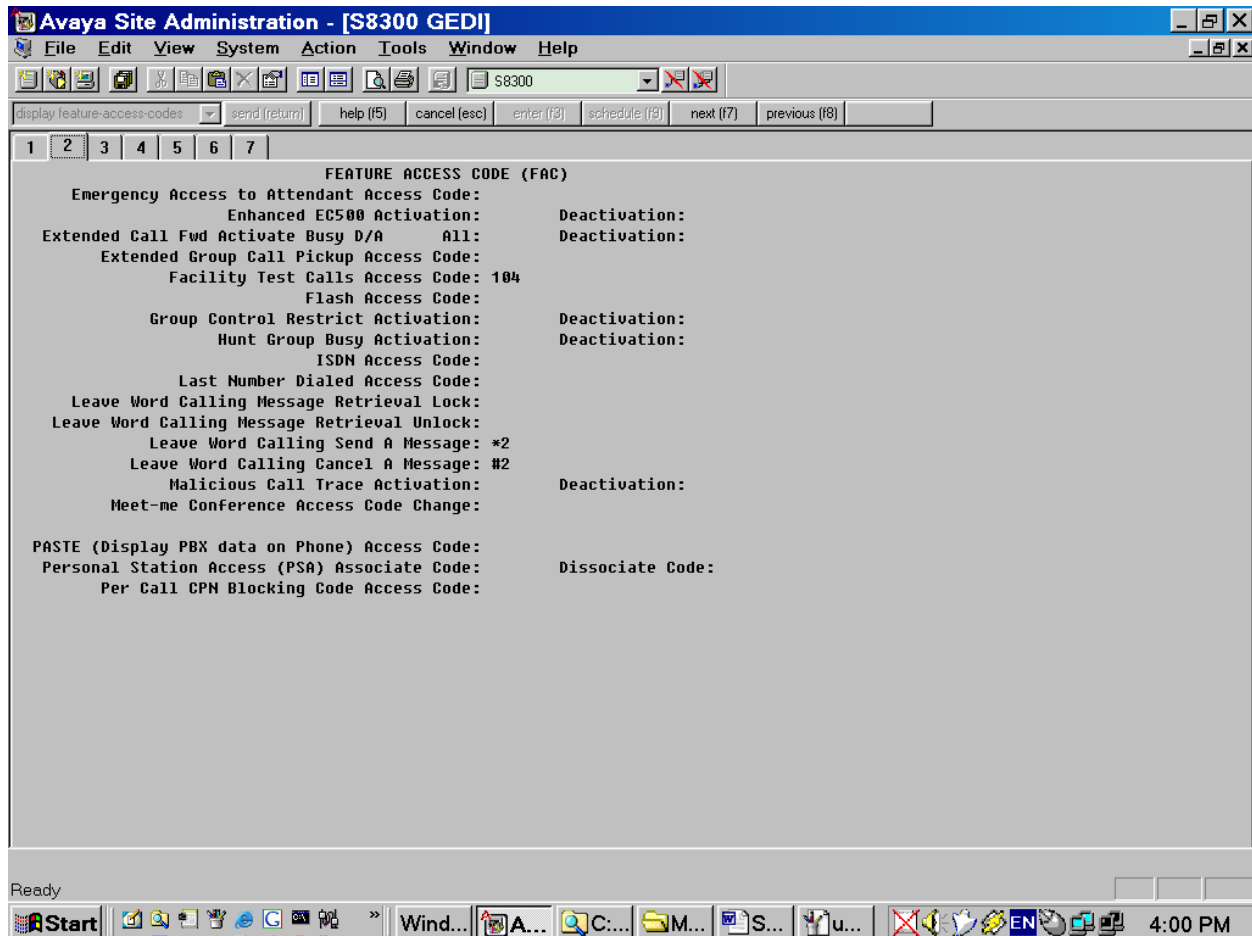


**Step 6:** Verify and change if needed MWI on & off codes.

Use "display feature-access-codes" command

Leave Word Calling Send A Message: \*2 (MWI ON)

Leave Word Calling Cancel A Message: #2 (MWI OFF)



### **5.1. TLS Setup**

- N/A.

### **5.2. Fail-Over Configuration**

- N/A.

### **5.3. Tested Phones**

Avaya 2420 Digital phone.

### **5.4. Other Comments**

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).	P	

	Dial the extension for the AA and confirm the AA answers the call.		
6	Call Transfer by Directory Search.		
6a	Call Transfer by Directory Search and have the called party answer.  Confirm the correct called party answers the phone.	P	
6b	Call Transfer by Directory Search when the called party's phone is busy.  Confirm the call is routed to the called party's voicemail.	P	
6c	Call Transfer by Directory Search when the called party does not answer.  Confirm the call is routed to the called party's voicemail.	P	
6d	Setup an invalid extension number for a particular user. Call Transfer by Directory Search to this user.  Confirm the number is reported as invalid.	P	
7	Outlook Web Access (OWA) Play-On-Phone Feature.		
7a	Listen to voicemail using OWA's Play-On-Phone feature to a user's extension.	P	
7b	Listen to voicemail using OWA's Play-On-Phone feature to an external number.	P	
8	Configure a button on the phone of a UM-enabled user to forward the user to the pilot number. Press the voicemail button.  Confirm you are sent to the prompt: "Welcome, you are connected to Microsoft Exchange. <User>. Please enter your pin and press the pound key."	P	
9	Send a test FAX message to user extension.	P	

	Confirm the FAX is received in the user's inbox.		
10	Setup TLS between gateway/IP-PBX and Exchange UM.  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.	P	
10b	Dial a user extension and leave a voicemail.  Confirm the user receives the voicemail.	P	
10c	Send a test FAX message to user extension.  Confirm the FAX is received in the user's inbox.	P	
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 <a href="#">MWI 2007 website</a> .	P	
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

<b>Failure Point</b>	None
<b>Phone type (if phone-specific)</b>	
<b>Call scenarios(s) associated with failure point</b>	
<b>List of UM features affected by failure point</b>	
<b>Additional Comments</b>	

## 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.

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.4.19/

AudioCodes TrunkPack 1610 MG Module 1

Management Settings

**Syslog Settings**

Syslog Server IP Address	10.15.2.5	← Step 2
Syslog Server Port	514	
Enable Syslog	Enable	← Step 1

**SNMP Settings**

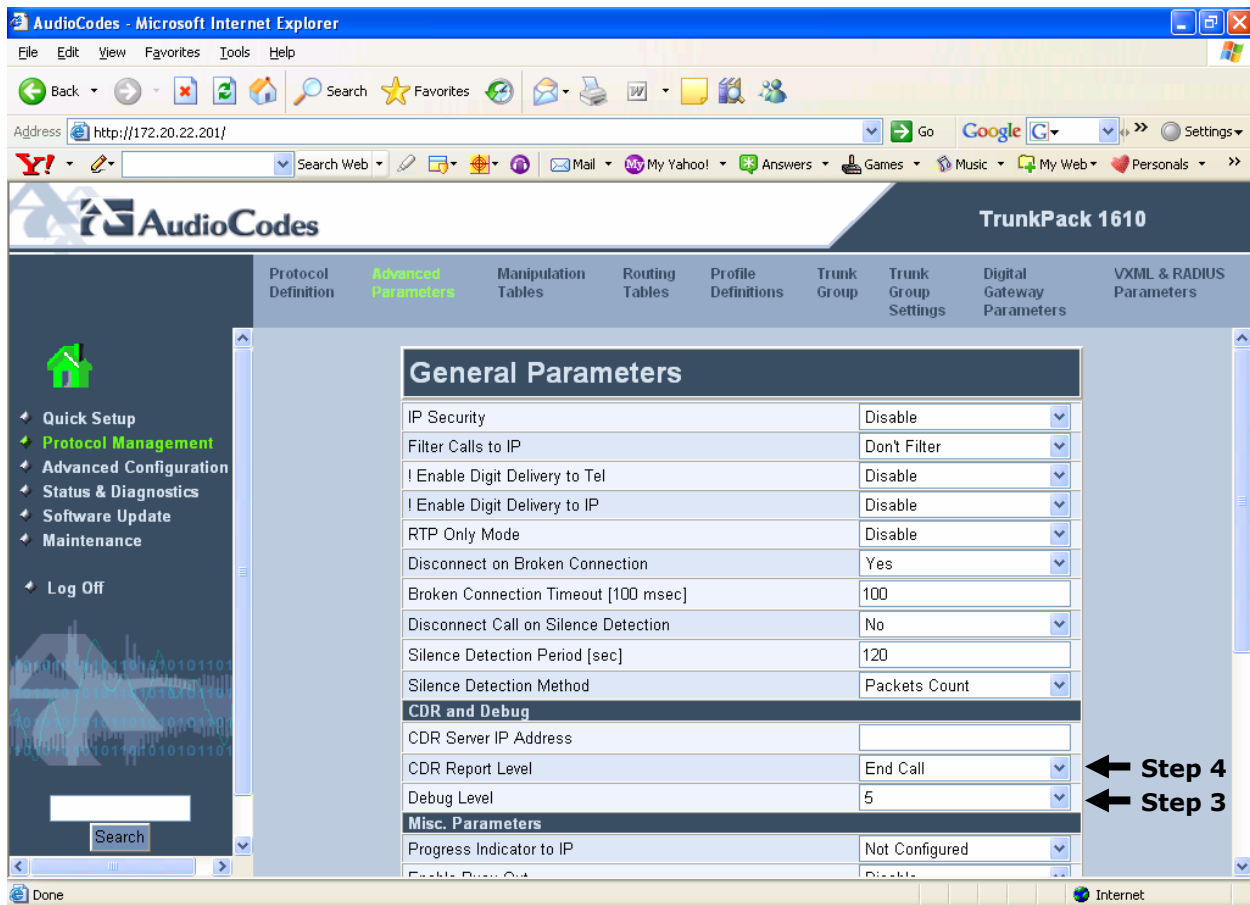
SNMP Managers Table	-->
SNMP Community String	-->
SNMP V3 Table	-->
Enable SNMP	Enable
Trap Manager Host Name	

**Activity Types to Report via 'Activity Log' Messages**

Parameters Value Change	<input type="checkbox"/>
Auxiliary Files Loading	<input type="checkbox"/>
Device Reset	<input type="checkbox"/>
Flash Memory Burning	<input type="checkbox"/>
Device Software Update	<input type="checkbox"/>

**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.