

Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

Siemens HiPath 4000 with AudioCodes MP-11x using Analog lines (In-band DTMF)

By : AudioCodes

Updated Since : 2007-02-09

READ THIS BEFORE YOU PROCEED

This document is for informational purposes only and is provided "AS IS". Microsoft, its partners and vendors cannot verify the accuracy of this information and take no responsibility for the content of this document. **MICROSOFT, ITS PARTNERS AND VENDORS MAKE NO WARRANTIES, EXPRESS, IMPLIED OR STATUTORY, AS TO THE INFORMATION IN THIS DOCUMENT.**

Content

This document describes the configuration required to setup Siemens HiPath 4000 and AudioCodes MP-11x 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

PBX Vendor	Siemens
Model	HiPath 4000
Software Version	Ver 3.0 SMR5 SMP4
Telephony Signaling	Analog with inband DTMF
Additional Notes	None

1.2. VoIP Gateway

Gateway Vendor	AudioCodes
Model	MP-11x FXO (MP-114 / MP-118)
Software Version	5.00AV.023.002
VoIP Protocol	SIP

1.3. Microsoft Exchange Server 2007 Unified Messaging

Version	RTM
----------------	-----

2. Prerequisites

2.1. Gateway Prerequisites

- 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

PBX with installed Analog card module Q2246-X SLMA24.

2.3. Cabling Requirements

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

The screenshot shows the AudioCodes MP-118 FXO configuration page. The browser title is "AudioCodes - Microsoft Internet Explorer" and the address bar shows "http://10.15.6.1/". The page has a navigation menu with tabs: Protocol Definitions, Advanced Parameters, Manipulation Tables, Routing Tables, Profile Definitions, Endpoint Phone Numbers, Hunt Group Settings, Endpoint Settings, FXO Settings, and RADIUS Parameters. The "General" tab is active, displaying a table of configuration parameters. Three black arrows point to the dropdown menus for "Channel Select Mode", "Fax Signaling Method", and "SIP Transport Type".

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	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

Step 2: Routing Setup

The screenshot shows the AudioCodes MP-118 FXO configuration page. The browser window title is "AudioCodes - Microsoft Internet Explorer" and the address bar shows "http://10.15.6.1/". The page has a navigation menu with options like "Protocol Definition", "Advanced Parameters", "Manipulation Tables", "Routing Tables", "Profile Definitions", "Endpoint Phone Numbers", "Hunt Group Settings", "Endpoint Settings", "FXO Settings", and "RADIUS Parameters". The "Proxy & Registration" section is active, displaying a table of configuration parameters. Two black arrows point to the "Use Proxy" dropdown menu and the "Proxy IP Address" text input field.

Proxy & Registration	
Enable Proxy	Use Proxy
Proxy Name	
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 Invite to Proxy	No
Enable Proxy Hot Swap	Disable

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, 10.15.3.207 or the FQDN of the Microsoft Unified Messaging host).

Step 3: SIP Environment Setup (Cont.)

The screenshot shows the AudioCodes MP-118 FXO configuration interface. The browser window title is "AudioCodes - Microsoft Internet Explorer" and the address bar shows "http://10.15.6.1/". The page has a navigation menu with the following items: Protocol Definition (highlighted), Advanced Parameters, Manipulation Tables, Routing Tables, Profile Definitions, Endpoint Phone Numbers, Hunt Group Settings, Endpoint Settings, FXO Settings, and RADIUS Parameters. The main configuration area contains the following settings:

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

At the bottom of the configuration area, there are three buttons: "Register", "Un-Register", and "Submit". A black arrow points to the "Subscription Mode" dropdown menu.

On the left side of the page, there is a sidebar with a search bar and the following menu items: Quick Setup, Protocol Management (highlighted), Advanced Configuration, Status & Diagnostics, Software Update, Maintenance, and Log Off. The sidebar also features a "SIP" logo and a "Web Server" indicator at the bottom.

Step 4: Coder Setup

AudioCodes MP-118 FXO

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

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

Search

SIP

Done Internet

Step 5: Digit Collection Setup

The screenshot shows the 'DTMF & Dialing' configuration page for the MP-114 FXO device. The browser is 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 content area with a table of settings. Black arrows point to the input fields for 'Max Digits In Phone Num', 'Inter Digit Timeout for Overlap Dialing [sec]', 'RFC 2833 Payload Type', and 'Enable Special Digits'.

DTMF & Dialing	
Max Digits In Phone Num	20
Inter Digit Timeout for Overlap Dialing [sec]	3
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option;	Not Supported
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

Step 6: Disconnect Supervision 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://10.15.6.2/". 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 is titled "General Parameters" and contains a table of configuration options. The "Disconnect and Answer Supervision" section is highlighted, and the "Enable Current Disconnect" option is set to "Enable". A double-headed arrow points to this option.

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
Disconnect and Answer Supervision	
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
CDR and Debug	
CDR Server IP Address	
CDR Report Level	End Call

Step 7: Message Waiting Indication Setup

The screenshot shows the AudioCodes MP-118 FXO web interface in Microsoft Internet Explorer. The browser address bar shows `http://10.15.6.1/`. The interface 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 is titled "Supplementary Services" and contains a table of configuration options. A black arrow points to the "Enable MWI" dropdown menu.

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	Bellcore
Hook-Flash Code	
MWI Parameters	
Enable MWI	Enable
MWI Analog Lamp	Disable
MWI Display	Disable

Step 8: Manipulation Routing Setup

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.6.1/

MP-118 FXO

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

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

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

SIP

Done Internet

Step 9: Endpoints Setup

The screenshot shows the AudioCodes MP-118 FXO web interface in a Microsoft Internet Explorer browser. The browser's address bar shows the URL `http://172.20.22.200/`. The interface has a navigation menu at the top with the following items: Protocol Definition, Advanced Parameters, Manipulation Tables, Routing Tables, Profile Definitions, **Endpoint Phone Numbers** (highlighted), Hunt Group Settings, Endpoint Settings, Advanced Applications, and RADIUS Parameters. On the left side, there is a sidebar menu with options: Quick Setup, **Protocol Management** (highlighted), Advanced Configuration, Status & Diagnostics, Software Update, Maintenance, and Log Off. The main content area is titled "Endpoint Phone Number Table" and contains a table with 8 rows and 4 columns: Channel(s), Phone Number, Hunt Group ID, and Profile ID. The first four rows have the phone number "1111" and Profile ID "0". Below the table are three buttons: Register, Un-Register, and Submit. The browser's status bar at the bottom shows "Web Server" and "Internet".

	Channel(s)	Phone Number	Hunt Group ID	Profile ID
1	<input type="text" value="1"/>	<input type="text" value="1111"/>	<input type="text"/>	<input type="text" value="0"/>
2	<input type="text" value="2"/>	<input type="text" value="1111"/>	<input type="text"/>	<input type="text" value="0"/>
3	<input type="text" value="3"/>	<input type="text" value="1111"/>	<input type="text"/>	<input type="text" value="0"/>
4	<input type="text" value="4"/>	<input type="text" value="1111"/>	<input type="text"/>	<input type="text" value="0"/>
5	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text" value="0"/>
6	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
7	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
8	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>

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: Voice Mail In-Band DTMF Setup

The screenshot displays the AudioCodes MP-118 FXO web interface for configuring Voice Mail. The configuration is organized into several sections:

- General:**
 - Voice Mail Interface: DTMF
 - Line Transfer Mode: Blind Transfer
- Digit Patterns:**
 - Forward on Busy Digit Pattern: ****SSSSSSSR.**
 - Forward on No Answer Digit Pattern: [Empty]
 - Forward on Do Not Disturb Digit Pattern: [Empty]
 - Forward on No Reason Digit Pattern: [Empty]
 - Internal Call Digit Pattern: SSSSSSSS
 - External Call Digit Pattern: [Empty]
 - Disconnect Call Digit Pattern: [Empty]
- MWI:**
 - MWI Off Digit Pattern: *530
 - MWI On Digit Pattern: *532
- SMDI:**
 - Enable SMDI: No SMDI
 - SMDI Timeout [msec]: 2000

Annotations in the image include two black arrows pointing to the 'Voice Mail Interface' and 'Line Transfer Mode' dropdown menus, and a large curly bracket on the right side of the 'Digit Patterns' section.

Note: In the 'Forward on Busy Digit Pattern' field, ensure that the number of 'S' characters (which represent the calling party's telephone number) equals the ALEN value (defined in Step 6 of Section 5).

Step 11: FAX Setup

The screenshot shows the AudioCodes MP-118 FXO web interface in Microsoft Internet Explorer. The browser address bar shows `http://10.15.6.1/`. The page title is "MP-118 FXO". The navigation menu includes: Network Settings, **Media Settings**, Configuration File, Regional Settings, Security Settings, and Management Settings. The left sidebar contains: Home icon, Quick Setup, Protocol Management, **Advanced Configuration**, Status & Diagnostics, Software Update, Maintenance, and Log Off. Below the sidebar is a search box with a "Search" button and the text "SIP".

The main content area is titled "Fax/Modem/CID Settings" and contains the following configuration table:

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

A black arrow points to the "Events Only" dropdown menu in the CNG Detector Mode row. At the bottom of the settings table is a "Submit" button. The browser status bar at the bottom shows "Done" and "Internet".

Step 12: FXO General Setup

The screenshot shows the AudioCodes MP-118 FXO configuration interface. The browser window title is "AudioCodes - Microsoft Internet Explorer" and the address bar shows "http://172.20.22.200/". The page has a navigation menu with options like "Protocol Definition", "Advanced Parameters", "Manipulation Tables", "Routing Tables", "Profile Definitions", "Endpoint Phone Numbers", "Hunt Group Settings", "Endpoint Settings", "Advanced Applications", and "RADIUS Parameters".

The main content area is titled "FXO Settings" and contains a table of configuration parameters. Three black arrows point to the following rows in the table:

FXO Settings	
Dialing Mode	One Stage
Waiting for Dial Tone	No
Time to Wait before Dialing [msec]	2000
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

Below the table is a "Submit" button. The left sidebar contains a navigation menu with options: "Quick Setup", "Protocol Management", "Advanced Configuration", "Status & Diagnostics", "Software Update", "Maintenance", and "Log Off".

Step 13: FXO General Setup (Cont.)

- CurrentDisconnectDuration = 450
- TimeToSampleAnalogLineVoltage = 100
- DigitPatternDigitToIgnore = *
- EnableDetectRemoteMACChange = 2
- ECNLPMODE = 1

Step 14: Reset FXO

AudioCodes MP-118 FXO

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

Search

SIP

Maintenance Actions

RESET

Reset Board	Reset
Burn To FLASH	Yes
Graceful Option	No

LOCK / UNLOCK

Lock	LOCK
Graceful Option	No

Current Admin State UNLOCKED

Save Configuration

Save Configuration	BURN
--------------------	------

Done Internet

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

4.1. Configuration Files

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



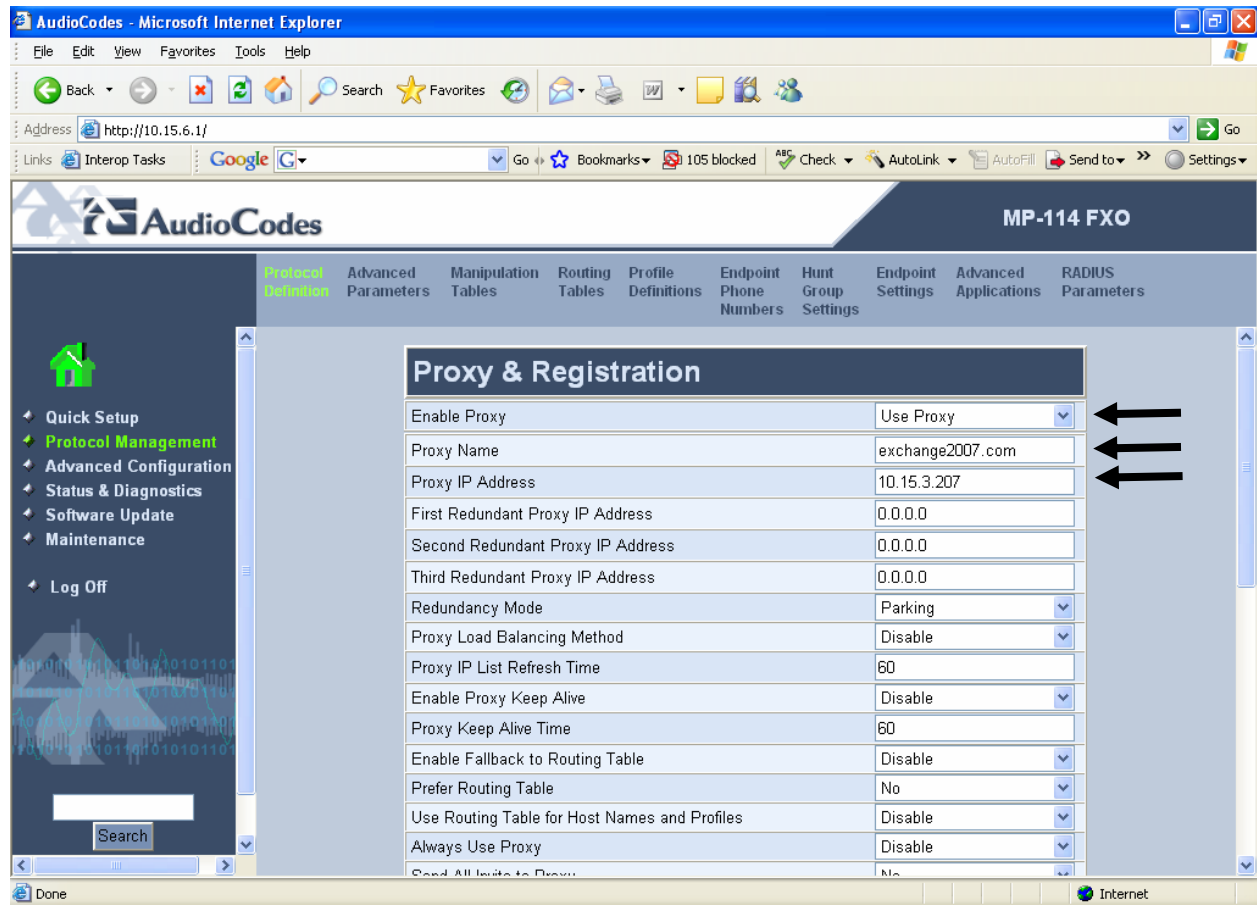
INI Seimens HiPath 4000 AudioCodes FXO DTMF.zip

4.2. TLS Setup

The specific gateway software version used in this PBX Configuration Guide was not tested for TLS. However, TLS was tested successfully for other gateway software versions operating with Microsoft Exchange 2007 TLS capabilities.

Refer to the procedure below for TLS setup.

Step 1: PBX to IP Routing Setup



The screenshot shows the AudioCodes MP-114 FXO web interface. The main content area is titled "Proxy & Registration" and contains a table of configuration settings. The settings are as follows:

Setting	Value
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 Ports to Proxy	No

Three black arrows on the right side of the table point to the "Use Proxy" dropdown menu, the "Proxy Name" text field, and the "Proxy IP Address" text field.

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

Parameter	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

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 title is "AudioCodes - Microsoft Internet Explorer" and the address bar shows "http://10.15.6.2/". The page has a navigation menu with tabs: Protocol Definition, Advanced Parameters, Manipulation Tables, Routing Tables, Profile Definitions, Endpoint Phone Numbers, Hunt Group Settings, Endpoint Settings, Advanced Applications, and RADIUS Parameters. The "General" tab is selected, displaying a table of configuration parameters. A search bar is visible on the left side of the configuration area.

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

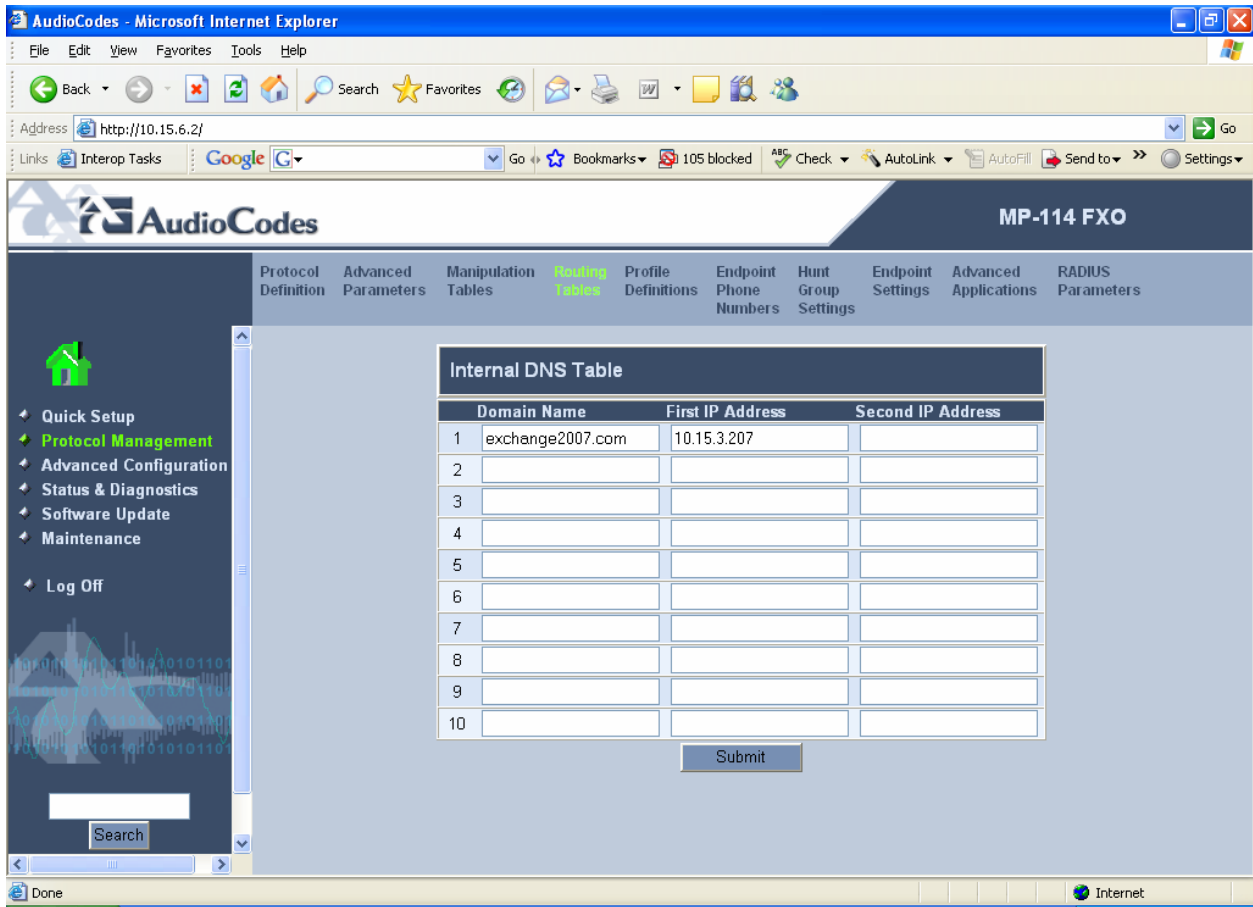
The screenshot shows the AudioCodes MP-114 FXO configuration interface. The browser window is titled "AudioCodes - Microsoft Internet Explorer" and the address bar shows "http://10.15.6.2/". The page has a navigation menu with categories: Network Settings, Media Settings, Configuration File, Regional Settings, Security Settings, and Management Settings. A 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 "IP Settings" and contains several sections:

- 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:**
 - ! NAT IP Address: 0.0.0.0
- Differential Services:**
 - Network QoS: 48
 - Media Premium QoS: 46
 - Control Premium QoS: 40
 - Gold QoS: 26

Two black arrows point to the "DNS Primary Server IP" and "DNS Secondary Server IP" fields, indicating they are the focus of the current step.

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



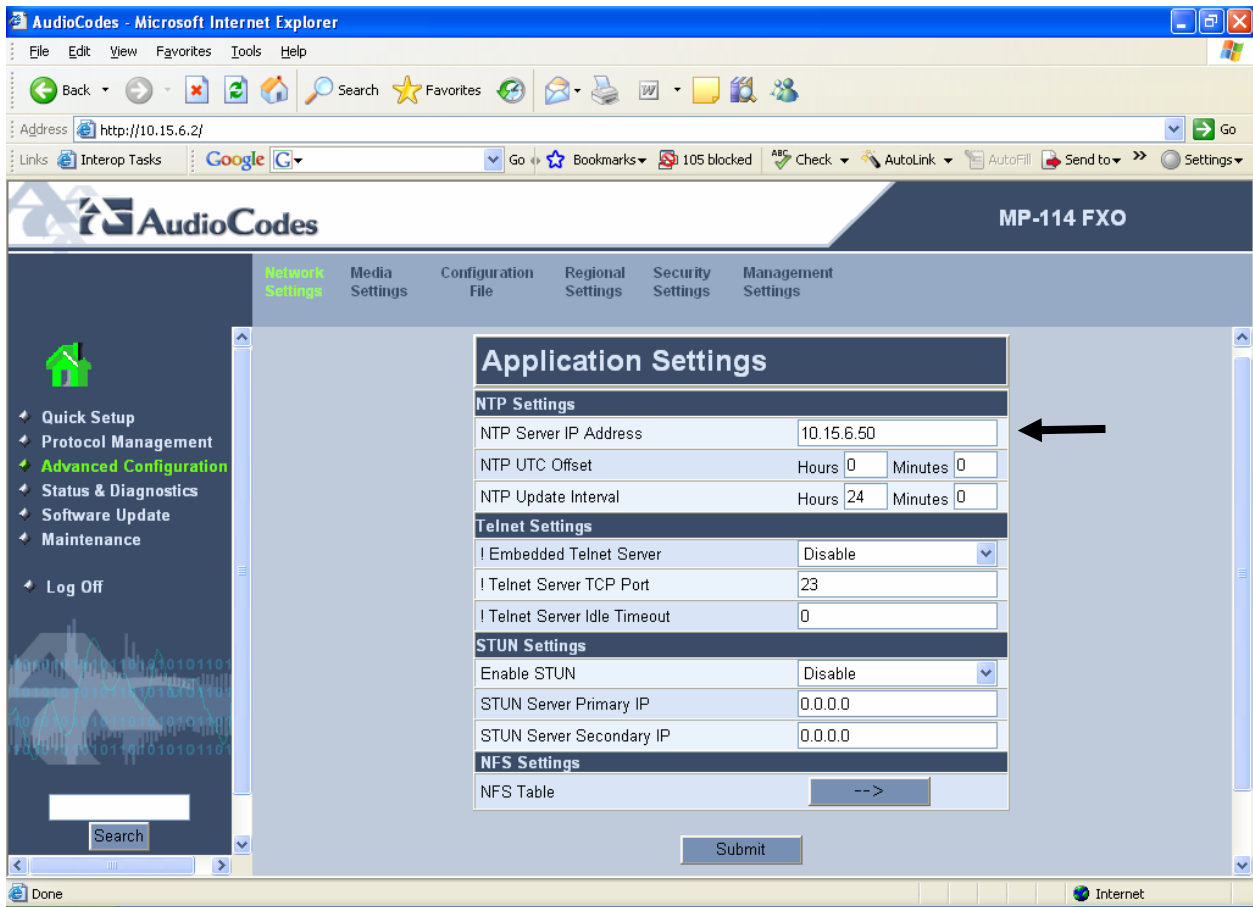
The screenshot shows the AudioCodes MP-114 FXO web interface in Microsoft Internet Explorer. The browser address bar shows <http://10.15.6.2/>. The page title is "AudioCodes" and the sub-header is "MP-114 FXO". The navigation menu includes: Protocol Definition, Advanced Parameters, Manipulation Tables, **Routing Tables**, Profile Definitions, Endpoint Phone Numbers, Hunt Group Settings, Endpoint Settings, Advanced Applications, and RADIUS Parameters. The left sidebar contains: Quick Setup, **Protocol Management**, Advanced Configuration, Status & Diagnostics, Software Update, Maintenance, and Log Off. The main content area is titled "Internal DNS Table" and contains a table with 10 rows. The first row is pre-filled with "exchange2007.com" in the Domain Name column and "10.15.3.207" in the First IP Address column. A "Submit" button is located below the table.

	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



The screenshot shows the AudioCodes MP-114 FXO configuration interface in a Microsoft Internet Explorer browser. The browser's address bar shows the URL `http://10.15.6.2/`. The page title is "AudioCodes" and the sub-header is "MP-114 FXO". The navigation menu includes "Network Settings", "Media Settings", "Configuration File", "Regional Settings", "Security Settings", and "Management Settings". The "Network Settings" menu is expanded, showing options like "Quick Setup", "Protocol Management", "Advanced Configuration", "Status & Diagnostics", "Software Update", "Maintenance", and "Log Off". The main content area is titled "Application Settings" and contains several sections: "NTP Settings", "Telnet Settings", "STUN Settings", and "NFS Settings". The "NTP Settings" section is highlighted, and a black arrow points to the "NTP Server IP Address" field, which contains the value "10.15.6.50". Other NTP settings include "NTP UTC Offset" (Hours: 0, Minutes: 0) and "NTP Update Interval" (Hours: 24, Minutes: 0). The "Telnet Settings" section includes "Embedded Telnet Server" (Disable), "Telnet Server TCP Port" (23), and "Telnet Server Idle Timeout" (0). The "STUN Settings" section includes "Enable STUN" (Disable), "STUN Server Primary IP" (0.0.0.0), and "STUN Server Secondary IP" (0.0.0.0). The "NFS Settings" section includes an "NFS Table" field with a "-->" button. A "Submit" button is located at the bottom of the form.

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	
Embedded Telnet Server	Disable
Telnet Server TCP Port	23
Telnet Server Idle Timeout	0

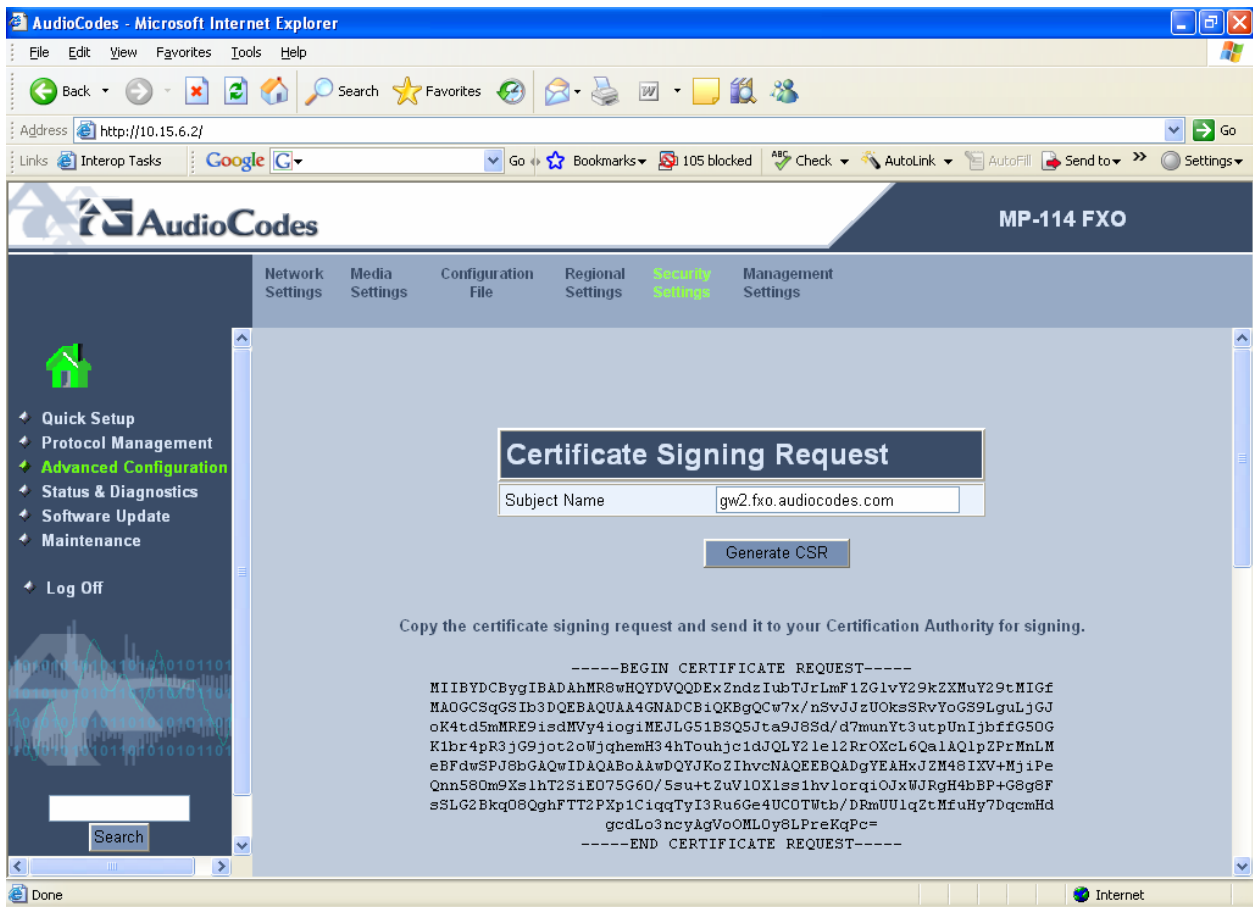
STUN Settings	
Enable STUN	Disable
STUN Server Primary IP	0.0.0.0
STUN Server Secondary IP	0.0.0.0

NFS Settings	
NFS Table	-->

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.

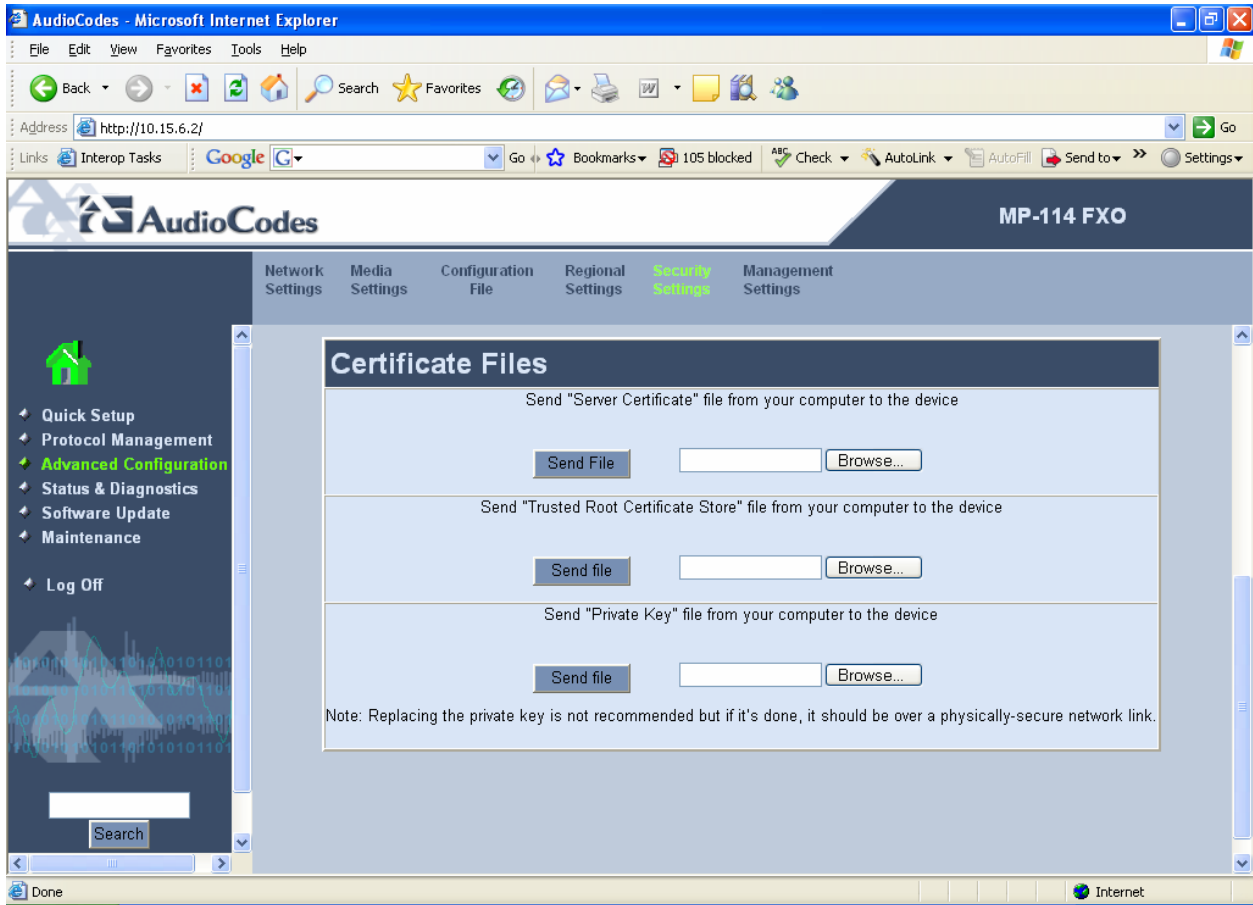


Step 8: Uploading Certificates Setup

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:

Our test configuration uses analog extensions 2464 and 2465 as voice mail ports. Siemens Optiset phone ext. 2417 is a test phone that forwards to a voice mail hunt group. The hunt group pilot number is ext. 2473.

Step 1: Create Analog Extensions for a Voice Mail Hunt Group

DISP-SCSU:2464,ALL;
H500: AMO SCSU STARTED

```

----- USER DATA -----
STNO      =2464          COS1       =5          DPLN      =0          SPDI      =0
PEN       = 1- 1- 3- 4   COS2       =5          ITR       =0          SPDC1     =
DVCFIG    =ANATE        LCOSV1    =25        COSX      =0          SPDC2     =
INS       =YES          LCOSV2    =25        RCBKB     =NO
PMIDX     =0           ALARMNO   =0          LCOSD1    =1          HMUSIC    =0          RCBKNA    =NO
SSTNO     =NO          COTRACE   =NO        LCOSD2    =1          SPEC      =SSTN    &SUFDIAL
COFIDX    =0           DIAL      =VAR      PULSTYPE  =          PULSLEV   =
CCTIDX    =           ACKST     =          TEXTSEL   =
DHPAR     =DTMFST &PREDIA
CONN      =DIR         FLASH     =YES      DTMFBLK   =NO          WITKEY    :NO
----- ACTIVATION IDENTIFIERS FOR FEATURES -----
DND       :NO          CWT       :NO
HTOS      :NO          HTOF      :NO      HTOD      :NO      VCP       :NO
----- FEATURES AND GROUP MEMBERSHIPS -----
PUGR      :           HUNTING GROUP : YES      SERVICE(S) : VCE
NIGHT OPTION : NO
----- STATION ATTRIBUTES (AMO SDAT) -----
NONE
-----

```


Step 5: Add Mailbox Key to a Test Phone

```
<disp-tapro
TYPE = stn
STNO = 2417
STD = ;
DISP-TAPRO:STN,2417,,;
H500: AMO TAPRO STARTED
+-----+-----+-----+-----+-----+
| STATION | STD | DIGTYP | FUNCTION KEYS WHICH DIFFER FROM STANDARD |
+-----+-----+-----+-----+-----+
| 2417    | 52 | OPTISET | 1 MB          7 DND                        |
+-----+-----+-----+-----+-----+

AMO-TAPRO-111          PROGRAMMABLE KEY DEFINITION FOR DIGITAL TERMINALS
DISPLAY COMPLETED;
```

Note: Use either system or station forwarding and DND to send calls to the Voice Mail Hunt group pilot number 2473.

Step 6: Defined source and redirect number length.

```
<DISP-ZAND
TYPE = VMI;
DISP-ZAND:VMI;
H500: AMO ZAND STARTED
      VOICE MAIL INTERFACE
      =====
      MWIOPEN  = NO ,           ALEN    = 8,           BLEN    = 6,
      DTMFCTRL = YES ,         PICKUP  = NO ,           NARELOUT = NO ,
      PIN      = NO ;
```

Note: The ALEN parameter defines the maximum number of digits of the calling party's telephone number. If the calling party's number surpasses the ALEN value, the gateway doesn't send this number for caller ID to the VM; instead, the gateway sends the digits 0 and [ALEN -1]. For example, If ALEN equals 8 and the calling party's number is longer than 8 digits, the gateway sends 01111111.

The BLEN parameter defines the maximum number of digits of the called party's telephone number. Ensure that the BLEN value includes the length of the PBX digit extension number (typically three to four digits) plus two additional digits.

5.1. TLS Setup

- N/A.

5.2. Fail-Over Configuration

- N/A.

5.3. Tested Phones

- Analog Phones
- Siemens Optiset 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). 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	The PBX indicates an invalid number by playing the user error tone.
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	P	

	<p>extension.</p> <p>Confirm the FAX is received in the user's inbox.</p>		
10	<p>Setup TLS between gateway/IP-PBX and Exchange UM.</p> <p>Replace this italicized text with your TLS configuration: self-signed certificates or Windows Certificate Authority (CA).</p>		
10a	<p>Dial the pilot number and logon to a user's mailbox.</p> <p>Confirm UM answers the call and confirm UM responds to DTMF input.</p>		The gateway supports TLS. However, TLS implementation is currently being tested.
10b	<p>Dial a user extension and leave a voicemail.</p> <p>Confirm the user receives the voicemail.</p>		The gateway supports TLS. However, TLS implementation is currently being tested.
10c	<p>Send a test FAX message to user extension.</p> <p>Confirm the FAX is received in the user's inbox.</p>		The gateway supports TLS. However, TLS implementation is currently being 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>	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

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

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.

The screenshot shows the AudioCodes TrunkPack 1610 Management Settings page. The page is displayed in a Microsoft Internet Explorer browser window. The address bar shows the URL http://10.15.4.19/. The page title is "TrunkPack 1610 MG Module 1". The navigation menu includes Network Settings, Media Settings, Trunk Settings, SS7 Configuration, TDM Bus Settings, Configuration File, Regional Settings, Security Settings, and Management Settings. The left sidebar contains a navigation menu with options: Quick Setup, Protocol Management, Advanced Configuration (highlighted), Status & Diagnostics, Software Update, Maintenance, and Log Off. The main content area is titled "Management Settings" and contains the following configuration options:

Syslog Settings	
Syslog Server IP Address	10.15.2.5
Syslog Server Port	514
Enable Syslog	Enable

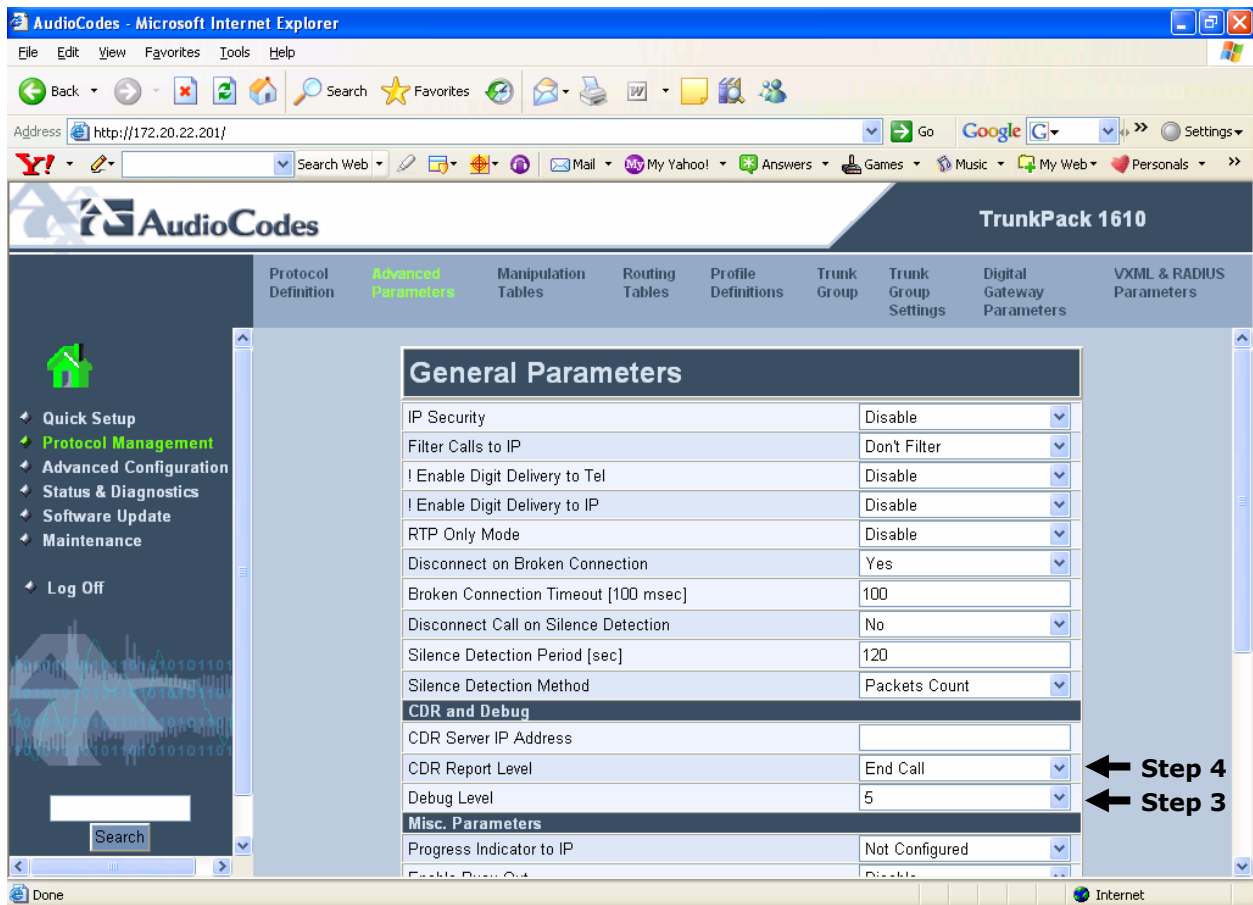
Two arrows point to the "Syslog Server IP Address" field (labeled "Step 2") and the "Enable Syslog" dropdown menu (labeled "Step 1").

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

- d. From an external phone, dial the extension for a UM-enabled user and leave a voicemail message.
- e. Confirm the voicemail message arrives in the called user's inbox.
- f. 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:
 - g. Under the Exchange Management Console, expand "Organizational Configuration" and then click on "Unified Messaging".
 - h. Go to the Auto Attendant tab under the results pane.
 - i. Click on the "New Auto Attendant..." under the action pane to invoke the AA wizard.
 - j. Associate the AA with the appropriate dial plan and assign an extension for the AA.
 - k. Create PBX dialing rules to always forward calls for the AA extension to the UM server.
 - l. 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.
 - m. UM Enable a user by invoking the "Enable-UMMailbox" wizard.
 - n. Assign an unused extension to the user.
 - o. Do not map the extension on the PBX to any user or device.
 - p. Call Transfer by Directory Search to this user.
 - q. Confirm the call fails and the caller is prompted with appropriate messages.

7. Play-On-Phone

- To access play-on-phone:
 - r. Logon to Outlook Web Access (OWA) by going to URL <https://<server name>/owa>.
 - s. After receiving a voicemail in the OWA inbox, open this voicemail message.
 - t. At the top of this message, look for the Play-On-Phone field (Play on Phone...).
 - u. 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:
 - v. 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:
 - w. `$dp = get-umdialplan -id <dial plan ID>`
 - x. `$dp.ConfiguredInCountryOrRegionGroups.Clear()`
 - y. `$dp.ConfiguredInCountryOrRegionGroups.Add("anywhere,*,*,")`
 - z. `$dp.AllowedInCountryOrRegionGroups.Clear()`
 - aa. `$dp.AllowedInCountryOrRegionGroups.Add("anywhere")`
 - bb. `$dp|set-umdialplan`
 - cc. `$mp = get-ummailboxpolicy -id <mailbox policy ID>`
 - dd. `$mp.AllowedInCountryGroups.Clear()`
 - ee. `$mp.AllowedInCountryGroups.Add("anywhere")`
 - ff. `$mp|set-ummailboxpolicy`
 - gg. The user must be enabled for external dialing on the PBX.
 - hh. 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:
 - ii. Double click on a user's mailbox and go to Mailbox Features tab.
 - jj. Click Unified Messaging and then click the properties button.
 - kk. Check the box "Allow faxes to be received".
- Management Shell - execute the following command:
 - ll. `Set-UMMailbox -identity UMUser -FaxEnabled:$true`
- To test fax functionality:
 - mm. Dial the extension for this fax-enabled UM user from a fax machine.
 - nn. Confirm the fax message is received in the user's inbox.
 - oo. 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.
 - pp. 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):
 - qq. Provide the configuration steps in Section 5.
 - rr. Configure the IP-PBX to work with two UM servers.
 - ss. Simulate a failure in one UM server.
 - tt. Confirm the IP-PBX transfers new calls to the other UM server successfully.