

Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

Tadiran Coral FLEXICOM with AudioCodes Mediant 1000 using BRI QSIG

By : AudioCodes

Updated Since : 2007-12-10

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 Tadiran Coral FLEXICOM and AudioCodes Mediant 1000 using BRI QSIG as the telephony signaling protocol. It also contains the results of the interoperability testing of Microsoft Exchange 2007 Unified Messaging based on this setup.

Intended Audience

This document is intended for Systems Integrators with significant telephony knowledge.

Technical Support

The information contained within this document has been provided by Microsoft, 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
10 December 2007	Version 1

1. Components Information

1.1. PBX or IP-PBX

PBX Vendor	Tadiran Telecom
Model	FLEXICOM
Software Version	14.67.49
Telephony Signaling	BRI QSIG
Additional Notes	

1.2. VoIP Gateway

Gateway Vendor	AudioCodes
Model	Mediant 1000
Software Version	5.20AEB.001.005
VoIP Protocol	SIP
Additional Notes	None

1.3. Microsoft Exchange Server 2007 Unified Messaging

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

2. Prerequisites

2.1. Gateway Prerequisites

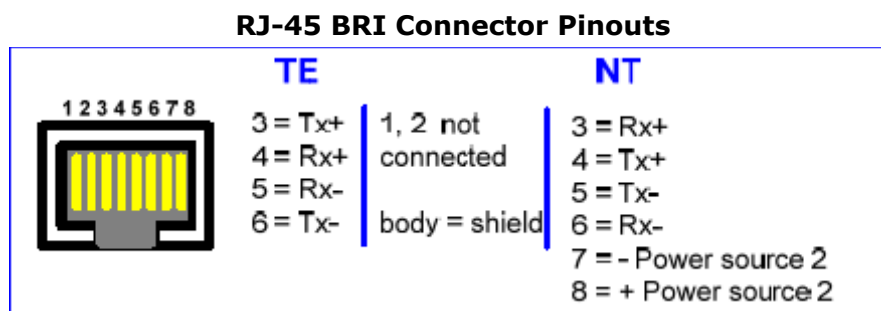
The gateway also supports TLS (in addition to TCP). This provides security by enabling the encryption of SIP packets over the IP network. The gateway supports self-signed certificates as well as Microsoft Windows Certificates Authority (CA) capabilities.

2.2. PBX Prerequisites

The PBX hardware must include an installed Trunk Card Module 8TBRP.

2.3. Cabling Requirements

The Mediant 1000 BRI port can be configured either as TE (Termination Equipment = user side) or NT (Network Termination = network side). The connector pinouts vary according to the configuration, as detailed in the following figure:



3. Summary and Limitations

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

Note: Currently, the Coral FLEXICOM implementing the 8TBRP card with version 5.22 does not support Message Waiting Indicator (MWI) for the QSIG BRI interface.

4. Gateway Setup Notes

Step 1: SIP Environment Setup

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.4.14/

AudioCodes Mediant 1000

Protocol Definition | Advanced Parameters | Manipulation Tables | Routing Tables | Profile Definitions | Trunk Group | Trunk Group Settings | Digital Gateway Parameters | Advanced Applications

General

PRACK Mode	Supported
Channel Select Mode	Cyclic 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
I TCP Timeout	0

Web Server | Internet

Step 2: Routing Setup

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.4.14/

AudioCodes Mediant 1000

Protocol Definition | Advanced Parameters | Manipulation Tables | Routing Tables | Profile Definitions | Trunk Group | Trunk Group Settings | Digital Gateway Parameters | Advanced Applications

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 List Query	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: Coder Setup


AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help


Back Forward Stop Home Search Favorites Print Mail Address http://10.15.4.14/ Go

AudioCodes Mediant 1000

Protocol Definition Advanced Parameters Manipulation Tables Routing Tables Profile Definitions Trunk Group Trunk Group Settings Digital Gateway Parameters Advanced Applications

 **Protocol Management**

- Advanced Configuration
- Status & Diagnostics
- Software Update
- Maintenance
- Log Off

 SIP

Search

Coders

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

Submit

Web Server Internet

Step 4: Digit Collection Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Home Search Favorites

Address http://10.15.4.14/ Go

AudioCodes Mediant 1000

Protocol Definition Advanced Parameters Manipulation Tables Routing Tables Profile Definitions Trunk Group Trunk Group Settings Digital Gateway Parameters Advanced Applications

Protocol Management

- Advanced Configuration
- Status & Diagnostics
- Software Update
- Maintenance
- Log Off

Search

SIP

DTMF & Dialing

Max Digits In Phone Num	30
Inter Digit Timeout [sec]	4
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option;	RFC 2833
2nd Tx DTMF Option;	
3rd Tx DTMF Option;	
4th Tx DTMF Option;	
5th Tx DTMF Option;	
RFC 2833 Payload Type	101
Digit Mapping Rules	
Default Destination Number	serveduser

Submit

Step 5: Trunk Group Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Reload Home Search Favorites

Address http://10.15.4.14/ Go

AudioCodes **Mediant 1000**

Protocol Definition Advanced Parameters Manipulation Tables Routing Tables Profile Definitions **Trunk Group** Trunk Group Settings Digital Gateway Parameters Advanced Applications

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

SIP

Search

Web Server Internet

Trunk Group Table

Trunk Group Index 1-12

Group Index	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	Profile ID
1	1	1	1-2	1000		0
2						
3						
4						
5						
6						
7						
8						
9						
10						
11						
12						

Step 6: TDM BUS Settings

The screenshot shows the AudioCodes Mediant 1000 web interface. The browser is Microsoft Internet Explorer, displaying the address <http://10.15.4.14/>. The page title is "AudioCodes Mediant 1000". The navigation menu includes: Network Settings, Media Settings, PSTN Settings, **TDM Bus Settings** (highlighted), Configuration File, Regional Settings, Security Settings, and Management Settings. The left sidebar contains: Protocol Management, **Advanced Configuration** (highlighted), Status & Diagnostics, Software Update, Maintenance, and Log Off. The main content area is titled "TDM Bus Settings" and contains the following configuration table:

I PCM Law Select	ALaw	▼
TDM Bus Type	Framers	▼
I Idle PCM Pattern	255	
I Idle ABCD Pattern	0x0F	▼
TDM Bus Local Reference	1	
TDM Bus PSTN Auto Clock	Disable	▼
TDM Bus PSTN Auto Clock Reverting	Disable	▼
TDM Bus Clock Source	Network	▼

Two black arrows point to the dropdown menus for "I PCM Law Select" and "TDM Bus Clock Source". Below the table is a "Submit" button. At the bottom of the main content area, a note reads: "To reboot and apply modified value(s) to the device, click 'Submit' then 'Reset' (with the 'Burn To FLASH' selected)." The bottom status bar shows "Web Server" and "Internet".

Step 7: Trunk Setting Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Reload Home Search Favorites Print Mail New Tab

Address http://10.15.4.14/ Go





AudioCodes **Mediant 1000**

Network Settings Media Settings **PSTN Settings** TDM Bus Settings Configuration File Regional Settings Security Settings Management Settings

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

Search


SIP

Trunk Number 1 2 3 4
Trunk Status    

Trunk Settings

Trunk Configuration	
Module ID	1
Trunk ID	1
Trunk Configuration State	Inactive
Protocol Type	BRI QSIG
Clock Master	Recovered
Auto Clock Trunk Priority	0
Trace Level	No Trace
ISDN Configuration	
ISDN Termination Side	User side
BRI Layer2 Mode	BRI L2 MODE P2P
Q931 Layer Response Behavior	0x40000000 -->
Outgoing Calls Behavior	0x400 -->
Incoming Calls Behavior	0x0 -->

Web Server Internet



Step 8: FAX Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Reload Home Search Favorites

Address http://10.15.4.14/ Go

AudioCodes

Mediant 1000

Network Settings **Media Settings** PSTN Settings TDM Bus Settings Configuration File Regional Settings Security Settings Management Settings

- Protocol Management
- Advanced Configuration**
- Status & Diagnostics
- Software Update
- Maintenance
- Log Off

SIP

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 Codec Type	G711Alaw
Fax/Modem Bypass Packing Factor	1
CNG Detector Mode	Events Only

Submit

Web Server Internet

Step 9: General Setup


AudioCodes - Microsoft Internet Explorer


File Edit View Favorites Tools Help

Back Forward Stop Reload Home Search Favorites RSS Print Mail Address http://10.15.4.14/ Go

AudioCodes **Mediant 1000**

Protocol Definition **Advanced Parameters** Manipulation Tables Routing Tables Profile Definitions Trunk Group Trunk Group Settings Digital Gateway Parameters Advanced Applications

 **Protocol Management**
◆ Advanced Configuration
◆ Status & Diagnostics
◆ Software Update
◆ Maintenance
◆ Log Off

 SIP

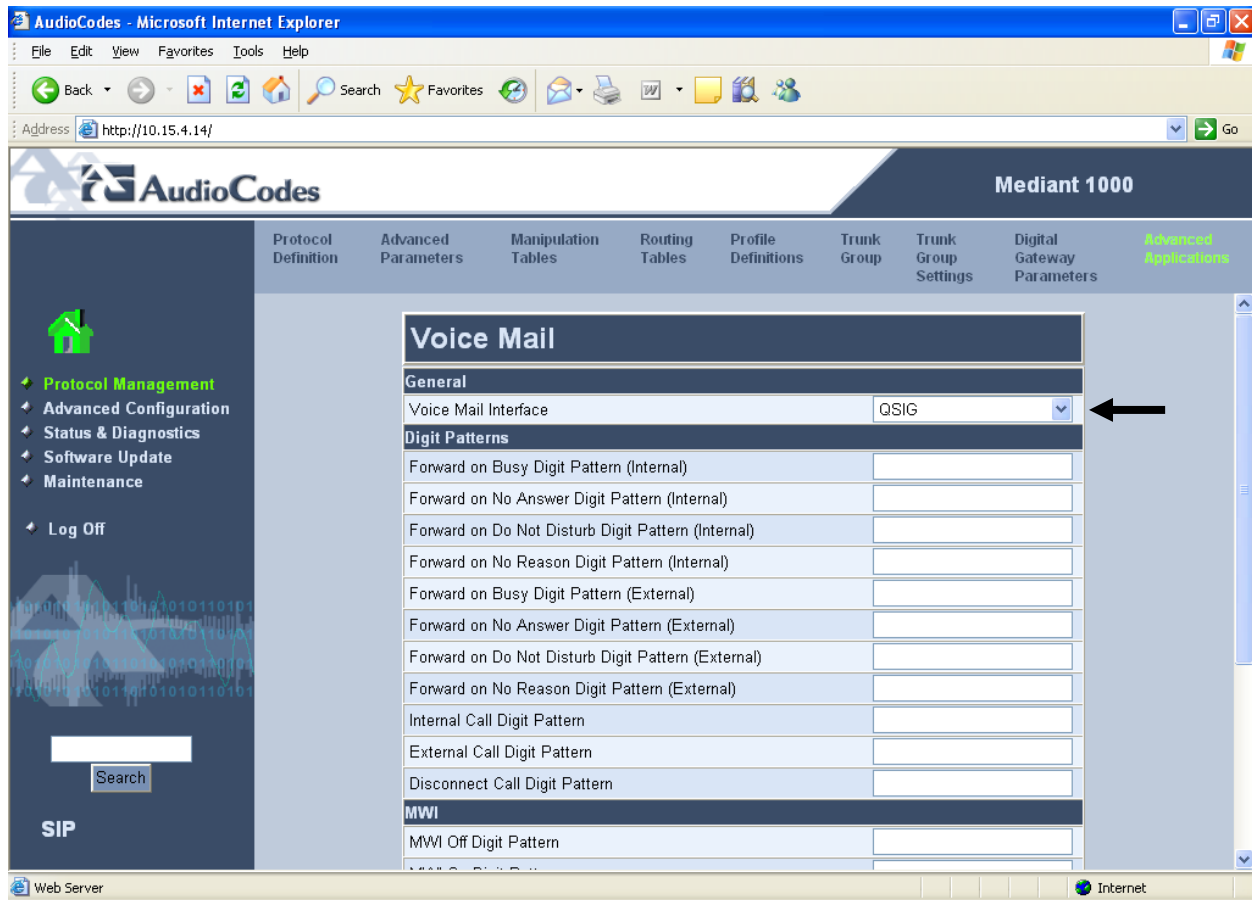
Search

General Parameters

IP Security	Disable
Filter Calls to IP	Don't Filter
! Enable Digit Delivery to Tel	Disable
! Enable Digit Delivery to IP	Disable
RTP Only Mode	Disable
PSTN Alert Timeout	180
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
Enable Fax Re-Routing	Disable
CDR and Debug	
CDR Server IP Address	
CDR Report Level	None
Debug Level	0
Misc. Parameters	

Web Server Internet

Step 10: Voice Mail Setup



Step 11:

- ISDNIBehavior = 1073741824
- EnableMWI = 1
- SubscriptionMode = 1
- MWISourceNumber = 400
- ECNLPMODE = 1
- TrunkTransferMode_X = 0 (where x refers to the Trunk number, for example, for the first trunk TrunkTransferMode_0 = 0)

Step 12: Reset Mediant 1000

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Reload Home Search Favorites

Address http://10.15.4.14/ Go

AudioCodes Mediant 1000

Home

- Protocol Management
- Advanced Configuration
- Status & Diagnostics
- Software Update
- Maintenance**
- Log Off

Search

SIP

Maintenance Actions

RESET

Reset Board

Burn To FLASH Yes

Graceful Option No

LOCK / UNLOCK

Lock

Graceful Option No

Current Admin State **UNLOCKED**

Save Configuration

Save Configuration

Web Server Internet

Click **Reset** to reset the gateway.

4.1. Configuration Files

- AudioCodes Mediant 1000 configuration ini file (.ini file extension).



Coral FLEXICOM BRI QSIG AudioCodes Mediant 1000.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

Proxy & Registration	
Enable Proxy	Use Proxy
Proxy Name	Exchange2007.server2003.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 Invite to Proxy	No
Enable Proxy List Cleanup	Disable

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.server2003.com for the FQDN of the Microsoft Unified Messaging host).

[illegible]

18

Step 3: SIP Environment Setup (Cont.)

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.4.14/

AudioCodes Mediant 1000

Protocol Definition Advanced Parameters Manipulation Tables Routing Tables Profile Definitions Trunk Group Trunk Group Settings Digital Gateway Parameters Advanced Applications

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

SIP

PRACK Mode	Supported	
Channel Select Mode	Cyclic Ascending	←
Enable Early Media	Disable	
183 Message Behavior	Progress	
Session-Expires Time	0	
Minimum Session-Expires	90	
Session Expires Method	Re-Invite	
Asserted Identity Mode	Disabled	
Fax Signaling Method	T.38 Relay	←
! Detect Fax on Answer Tone	Initiate T.38 on Preamble	
SIP Transport Type	TLS	←
SIP UDP Local Port	5000	←
SIP TCP Local Port	5040	←
SIP TLS Local Port	5060	←
Enable SIPS	Disable	
Enable TCP Connection Reuse	Enable	
! TCP Timeout	0	
SIP Destination Port	5061	←

Web Server Internet

Step 4: DNS Servers Setup

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The browser window is titled 'AudioCodes - Microsoft Internet Explorer' and the address bar shows 'http://10.15.4.14/'. The page has a navigation bar with tabs: Network Settings, Media Settings, PSTN Settings, TDM Bus Settings, Configuration File, Regional Settings, Security Settings, and Management Settings. The 'Network Settings' tab is active. On the left, there is a sidebar with a home icon, a search bar, and a list of links: 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, DNS Settings, DHCP Settings, NAT Settings, and Differential Services. The 'DNS Settings' section has two arrows pointing to the 'DNS Primary Server IP' and 'DNS Secondary Server IP' fields, which are currently set to 10.1.1.11 and 10.1.1.10 respectively.

IP Settings	
IP Networking Mode	Single IP Network
IP Address	10.15.4.14
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	26

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 Mediant 1000 web interface in a Microsoft Internet Explorer browser window. The address bar shows <http://10.15.4.14/>. The interface has a top navigation bar with the AudioCodes logo and the title "Mediant 1000". Below this is a menu bar with the following items: Protocol Definition, Advanced Parameters, Manipulation Tables, Routing Tables (highlighted in green), Profile Definitions, Trunk Group, Trunk Group Settings, Digital Gateway Parameters, and Advanced Applications. On the left side, there is a sidebar with a green house icon and the following links: Protocol Management, Advanced Configuration, Status & Diagnostics, Software Update, Maintenance, and Log Off. Below these links is a search bar with the text "SIP" and a "Search" button. The main content area is titled "Internal DNS Table" and contains a table with 10 rows. The first row is pre-filled with "exchange2007.server20" in the Domain Name column and "10.15.3.207" in the First IP Address column. The other columns are empty. A black arrow points to the first row of the table. Below the table is a "Submit" button.

	Domain Name	First IP Address	Second IP Address
1	exchange2007.server20	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 Mediant 1000 configuration web interface in a Microsoft Internet Explorer browser window. The address bar shows `http://10.15.4.14/`. The interface has a top navigation bar with tabs: Network Settings, Media Settings, PSTN Settings, TDM Bus Settings, Configuration File, Regional Settings, Security Settings, and Management Settings. On the left is a sidebar with a home icon and links: Protocol Management, Advanced Configuration (highlighted), Status & Diagnostics, Software Update, Maintenance, and Log Off. Below these is a search bar and the label 'SIP'. The main content area is titled 'Application Settings' and contains several sections: NTP Settings, Telnet Settings, STUN Settings, and NFS Settings. In the NTP Settings section, the 'NTP Server IP Address' field is set to '10.15.6.50' and is highlighted with a black arrow. Other fields in 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), 'Telnet Server Idle Timeout' (0), 'SSH Server Enable' (Disable), and 'SSH Server Port' (22). 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. The bottom status bar shows 'Web Server' and 'Internet'.

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	
! Embedded Telnet Server	Disable
! Telnet Server TCP Port	23
! Telnet Server Idle Timeout	0
SSH Server Enable	Disable
SSH Server Port	22
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.

The screenshot shows the AudioCodes Mediant 1000 web interface in Microsoft Internet Explorer. The browser's address bar shows the URL `http://10.15.4.14/`. The page has a navigation menu with tabs: Network Settings, Media Settings, PSTN Settings, TDM Bus Settings, Configuration File, Regional Settings, Security Settings (highlighted), and Management Settings. On the left, there is a sidebar with links: Protocol Management, Advanced Configuration (highlighted), Status & Diagnostics, Software Update, Maintenance, and Log Off. The main content area is titled "Certificate Signing Request". It contains a form with a "Subject Name" field set to `gw1.m1k.audiocodes.com` and a "Generate CSR" button. Below the button, there is a text box containing the following text:

Copy the certificate signing request and send it to your Certification Authority for signing.

```
-----BEGIN CERTIFICATE REQUEST-----
MIIBYDCBygIBADAhMR8wHQYDVQQDEx2ndzEubTFRLnM1ZG1vY29kZXMuY29tMIGf
MAOGCSqGSIB3DQEBAAQUAA4GNADCB1QKBgQDSDM422rz0/duxZpN8LORsywQZ+4Oy
x0Aw8oZFpz0vOSiHgC9F7mZLvz68wRTxKGB0ONGxWqeEOaKxkgLbtUMzbp+rsCIt
UpPTHc3vuns1vpzpbzz14J2T7K/Cy6fmjjgqZT4mFpCtI1Lv591kbwMQj2fJNMB3
UIwiD9g2OV+uOwIDAQABoAAwDQYJKoZIhvcNAQEEBQADgYEAZbtqLKdHLXVY/xTC
O6W7xdeGK+xPn6j2PpIVLcAFy5392x15JpWgSfhsZqd6rZm71YoKZA1B7M7NuQJ1
76pqvb1+EScpEunfeYvkhJpRLvME6SaotoLrD9BgTuqd3Mjhr1EoOxykiG9OmR9p
CNJKBV6ohJDCVHv9+cV1ZOpBQ3A=
-----END CERTIFICATE REQUEST-----
```

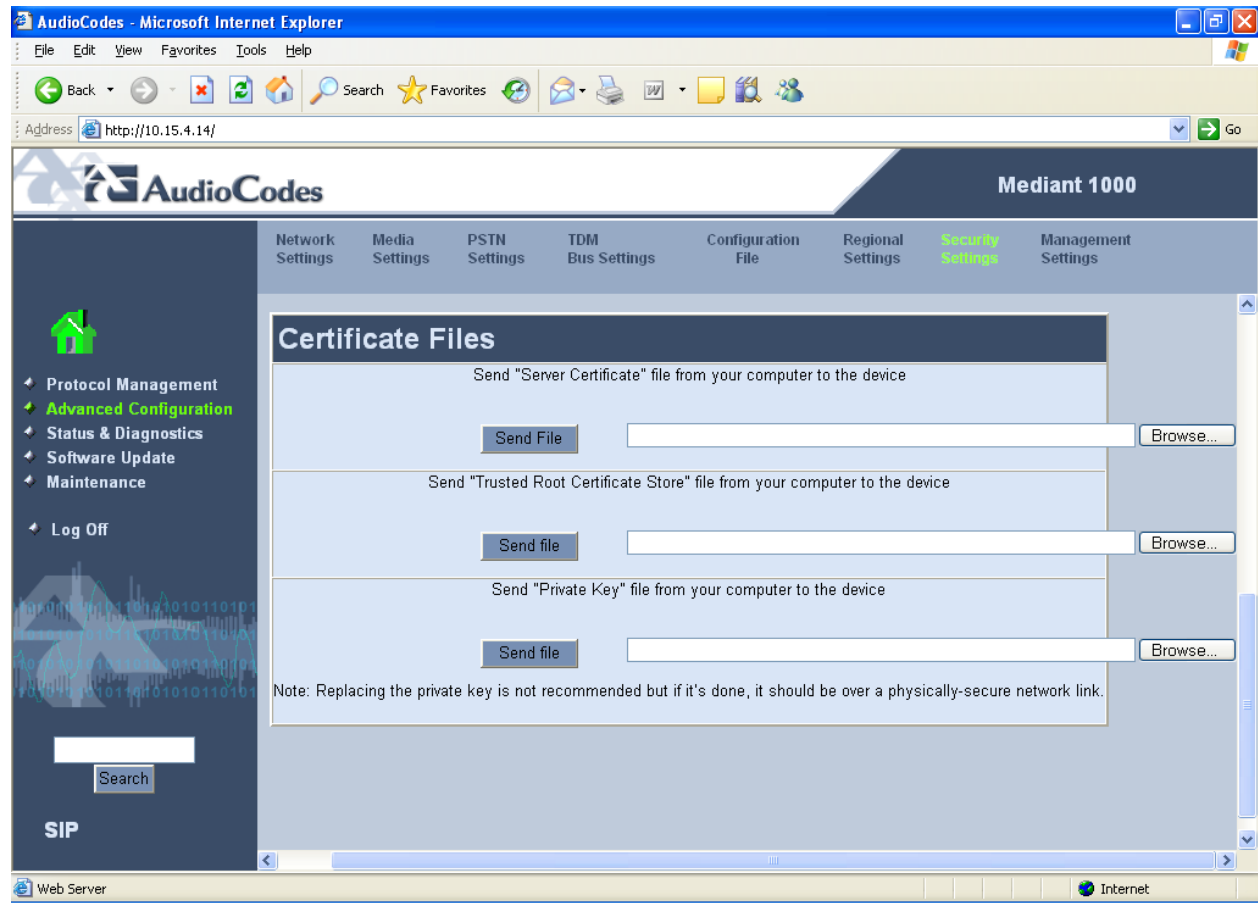
At the bottom of the page, there is a "Web Server" status bar and an "Internet" icon.

Step 8: Uploading Certificates Setup

The screen below is used to upload the signed 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:

- **Digital Voice Mail Ports:** BRI QSIG trunk
- **Voice Mail Hunt Group Pilot:** 340
- **Voice Mail User Phone:** ext. 310 and ext. 311

Step 1: Verify BRI Card (8TRB) Version

The BRI card (8TRB) must be version 5.22 or higher.

(CLIS)

choose mode

0 - UPDATE

1 - DISPLAY

4 - SNAP

*: 1

FROM SHELF#- 0 0

TO SHELF#- 0 0

FROM SLOT# - 1 3

TO SLOT# - 3 3

Any specific data field (type ? for help)

CARDS LIST

shelf#/slot#	p_type	i_type	card_db#	vers/subver	status
0 / 3	8TBR/P	8TBR	0	5 22	ACTIVE

Step 2: Check the Access Codes of the Associated Two Ports of the 8TRB Card

Use the **PHYS_LOC** command in **PLIS** level to view the two ports of 8TRB card.
For example: ports 7265-7266 are associated with 8TRB card allocated in shelf 0 slot 3.

(PLIS)

PORTS_LIST

0-PHYS_LOC

1-DIAL_NUM

*: 0

choose mode

0 - UPDATE

1 - DISPLAY

*: 1

FROM SHELF#- 0 0 0 3 3 0 1

Any specific data field (type ? for help)

PORTS LIST

SHELF/SLOT/CKT	TYPE	DIAL#	PORT_DB#	VERS	SHORT & FULL NAMES
0 / 3 / 0	8TBR	7265 0	---	BLANK	:BLANK
0 / 3 / 1	8TBR	7266 0	---	BLANK	:BLANK

Step 3: Digital Trunk Setting

Use the DTDB 4 level (ISDN_SIG.CHANNEL(D-CHANNEL)) to define the Digital trunk as a QSIG Trunk.

Verify that the PROTOCOL_ID is defined as QSIG.

(DTDB)

0-CONFIG

1-CARD_DB

2-PORT_DB

3-SYNC

4-ISDN_SIG.CHANNEL(D-CHANNEL)

*: 4

choose access method:

0-SIGNALING_CHANNEL

1-CARD_LOCATION

*: 0

choose mode

0 - UPDATE

1 - DISPLAY

*: 1

FROM SIGNALING_CHANNEL#- 0 3

TO SIGNALING_CHANNEL#- 3 3

Any specific data field (type ? for help)

signaling_channel

3

NAME- BLANK

SIGNALING_CHANNEL- 3

B_CHANNEL_NEGOTIATION:

(Exclusive/Preferred)- Preferred

PROTOCOL_ID (At&t/Etsi/aUstralia/Qsig)- Qsig

PROTOCOL_SIDE: U(User or slave)/N(Network or master)- Network

END_OF_DIAL_DIGIT- NONE

SENDING_COMPLETE for outgoing calls (Y/N)- Y

SENDING_COMPLETE for Enblock Incoming calls (Y/N)- N

Send Connected Number to Public Network (Y/N)- Y

CONNECT_WHEN_DEST_IS_NOT_ISDN (Y/N)- N

DTMF_WHEN_CALL_PROC (OVERLAP ONLY) (Y/N)- Y

MLPP_SUPPORT (Y/N)- N

Adjacent Entity Number - 0

SYNC_CHANNEL (Y/N)- N

PERMANENT_ACTIVE_CHANNEL (Y/N)- Y

POWER SUPPLY (8TBR/P Card) (Y/N)- Y

TEI ASSIGNMENT (Fixed/Auto)- Fixed

EXTERNAL_LINE_IS_PHYSICALLY_CONNECTED (Y/N)- Y

QSIG DEFINITIONS :

Support Call Independent Signalling Connection (CISC) (Y/N)- Y

Transit Counter in CISC calls (Y/N)- Y

NET DIVERSION (Y/N)- Y

TRANSIT_COUNTER_CODING (Ecma/Iso)- Iso

PROTOCOL_PROFILE (Ecma/Iso)- Ecma

Path Replacement re-use of connection element (Y/N)- Y

NUMBER_OF_TERMINALS- 1

Step 4: Trunk Group Setting

Use the TGDEF level to define the trunk B-channels (members) and the Trunk Access Code (i.e., Trunk Access Code is 817 and the B-channels are 7265-7266).

Follow the below setting:

(TGDEF)

choose mode

0 - UPDATE

1 - DISPLAY

*: 1

FROM TK_GRP# - 9 817

TO TK_GRP# - 817 817

Any specific member (CR/NUM) -
817

NAME:

SHORT(5) - __irb

FULL(16) - BLANK

IP_ZONE (0-7) - 0

ISDN ONLY (Y/N) - N

QSIG (Y/N) - Y

DTMF_DIGITS_BEFORE_ANSWER - Y

ANI_SCREENING_SEND(Unavailable,Site_Idn,Transparent,Omit) - T

SEARCH TYPE (0-circ 1-term) - 1

DTD OVERRIDE - N

OGR_OVERRIDE - N

COLLECT_TONE_OVERRIDE - Y

PAGING - N

TK_TK_CONNECT_OVERRIDE - Y

BCCOS - 0

ROUTING ACCESS - 7081

LAR_MAX_ASYNCHRONOUS_FAILS (0-10) - 2

LAR_SYSTEM_PREFERENCE (Cost/Performance) - P

LAR_TRIGGERS_SET - 0

TRANSIT ALI - NONE

DIALING METHOD (Enblock/Overlap) - E

DIAL IN FILTER -

DIAL IN/CALLER OUT OFFSET- NONE

CALLER # OUT FILTER -

INCOMING ANI FILTERS (Y/N) - N

METERING_UNIT_CHARGE (xxxxx.yy) - N

INCOMING_CLI_REQUEST (Y/N) - N

NUMBER OF DIGITS EXPECTED -

DISABLE_DTMF_SUPERVISION (Y/N) - N

JOIN GROUP CALL IN MUTE (Y/N) - Y

MEM# 1 - 7265

MEM# 2 - 7266

Step 5: Define Feature Access Codes

To access the voice mail as a user, several private libraries need to be created. This takes the user directly to the personal greeting and then a prompt to leave or retrieve voice mail messages. The libraries are used to dial up to Auto Attendants applications and various call forwarding conditions. This provides the voice mail with various greetings such as standard greeting, out of office, ring no answer, busy or message retrieve.

Use the LIB level to define the Access Code (i.e., 340 seizes a member of trunk group 817 and send the E0-end off digit) for the voice mail via BRI QSIG Trunk.

Verify that the **OUT TK** is the trunk group defined in Step 2.

(LIB)

0-PUB.LIB

1-PRIV.LIB

2-SER.LIB

3-LARGE_PUB

4-DIRECTORY

*: 0

choose mode

0 - UPDATE

1 - DISPLAY

*: 1

FROM PUB_LIB# - 340 340

TO PUB_LIB# - 340 340

LIB 340

NAME:

SHORT(5) - __ MV

FULL(16) - __ MV

TOLL_OVERRIDE- Y

NAME_RETENTION_OVERRIDE- N

PROTECTED- N

USER_CANNED_MESSAGE# (0-15/N) - NONE

TNNT_GRP - 0

PRIORITY_PREEMPTION_CALL- N

DIAL NUM = 340E0

OUT TK = 817

Step 6: Verify the Numbering Plan for Forward

Use the NPL level to verify that the dialing code 140 to 143 is assigned to feature number 50 (forwarding feature).

NPL

NUM_PLAN

0-GENERAL

1-SPECIAL FEATURE CODES

*: 0

GENERAL

0 - UPDATE

1 - DISPLAY

2 - ADD

3 - REMOVE

5 - SHOW

7 - ERASE

*: 1

FROM DIAL#	TO DIAL#	TYPE	INDEX#/SHELF,SLOT,CKT/NODE#
140	143	FEATURE	50

To set Call Forward feature for each extension:

1. Lift the handset.
2. Dial the required Call Forward feature code (see the table below).
3. Listen for a dial tone.
4. Dial the destination number to where calls are to be forwarded. (see the table below).
5. Listen for the confirmation tone.
6. Hang up.

To cancel Call Forward for each extension:

1. Lift the handset.
2. Dial the required Call Forward feature code (see the table below).
3. Listen for a dial tone.
4. Dial the cancellation code (default is 10).
5. Listen for the confirmation tone.
6. Hang up.

Call Forward Type	Feature Code	Where calls are to be forwarded
unconditional	141	340
no answer	142	340
busy	140	340

Step 7: Add Numbering Plan for MWI

Use the NPL level to add numbering plan for MWI. This number must correspond to the gateway parameter MWISourceNumber, as defined in Section 4, Step 11.

(NPL)

NUM_PLAN

0-GENERAL

1-SPECIAL FEATURE CODES

2-FLEXICALL/IRSS FEATURE CODES

3-SHOW TYPE OF NUMBER

*: 0

GENERAL

choose mode

0 - UPDATE

1 - DISPLAY

2 - ADD

3 - REMOVE

5 - SHOW

7 - ERASE

*: 1

FROM DIAL#	TO DIAL#	TYPE	INDEX#/SHELF,SLOT,CKT/NODE#/index,ckt
------------	----------	------	---------------------------------------

400	400	NETWORK	1
-----	-----	---------	---

5.1. TLS Setup

- N/A.

5.2. Fail-Over Configuration

- N/A.

5.3. Tested Phones

- Analog phones.

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	Microsoft UM informs the following: "The call can't be transferred. Return to main menu".
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	<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>	P	
10b	<p>Dial a user extension and leave a voicemail.</p> <p>Confirm the user receives the voicemail.</p>	P	
10c	<p>Send a test FAX message to user extension.</p> <p>Confirm the FAX is received in the user's inbox.</p>	P	
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>	F	Currently, Coral FLEXICOM using the 8TBRP card with version 5.22 does not support Message Waiting Indicator for the QSIG BRI interface.
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	Coral FLEXICOM does not support Message Waiting Indicator (MWI) when using the QSIG BRI card.
Phone type (if phone-specific)	Any
Call scenarios(s) associated with failure point	12
List of UM features affected by failure point	Message Waiting Indicator (MWI).
Additional Comments Currently, Coral FLEXICOM using the 8TBRP card with version 5.22 does not support Message Waiting Indicator for the QSIG BRI interface.	

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.

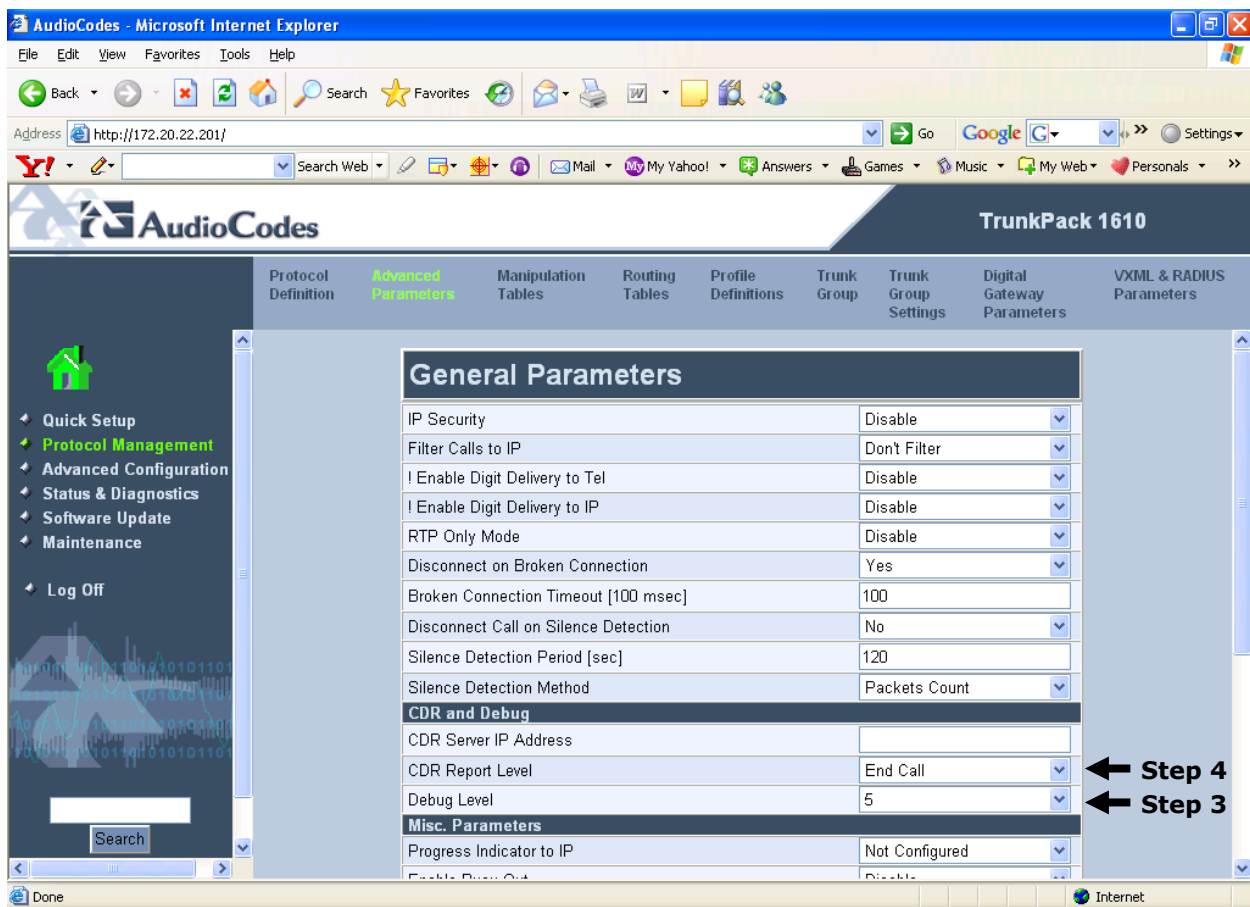
The screenshot shows the AudioCodes Mediant 1000 web interface. The browser window is titled 'AudioCodes - Microsoft Internet Explorer' and the address bar shows 'http://10.15.4.14/'. The page has a navigation menu on the left with options like 'Protocol Management', 'Advanced Configuration', 'Status & Diagnostics', 'Software Update', 'Maintenance', and 'Log Off'. The main content area is titled 'Management Settings' and contains several sections: 'Syslog Settings', 'SNMP Settings', and 'Activity Types to Report via 'Activity Log' Messages'. The 'Syslog Settings' section includes the following fields:

Syslog Settings	
Syslog Server IP Address	10.15.2.5
Syslog Server Port	514
Enable Syslog	Enable
Trunks Filter	-1

Arrows from the text 'Step 2' and 'Step 1' point to the 'Syslog Server IP Address' and 'Enable Syslog' fields respectively. The 'SNMP Settings' section includes fields for 'SNMP Trap Destinations', 'SNMP Community String', 'SNMP V3 Table', 'SNMP Trusted Managers Table', 'Enable SNMP' (set to 'Enable'), and 'Trap Manager Host Name'. The 'Activity Types to Report via 'Activity Log' Messages' section includes checkboxes for 'Parameters Value Change' and 'Auxiliary Files Loading'.

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:

- **PSTN Trace:** PSTN Trace is a procedure used to monitor and trace the PSTN elements (E1/T1) in AudioCodes digital gateways (Mediant 1000 & Mediant 2000). These utilities are designed to convert PSTN trace binary files to textual form.
- **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.