

Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

Mitel 3300 with AudioCodes Mediant 1000/2000

using T1 QSIG

By : AudioCodes

Updated Since : 2007-06-05

READ THIS BEFORE YOU PROCEED

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Content

This document describes the configuration required to setup Mitel 3300 and AudioCodes Mediant 1000/2000 using T1 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
21 March 2007	Version 1

1. Components Information

1.1. PBX or IP-PBX

PBX Vendor	Mitel
Model	3300 ICP MX
Software Version	7
Telephony Signaling	T1 QSIG
Additional Notes	None

1.2. VoIP Gateway

Gateway Vendor	AudioCodes
Model	Mediant 2000
Software Version	5.00A.039.005
VoIP Protocol	SIP
Additional Notes	Tests were conducted with Mediant 2000. However, these tests are also applicable to Mediant 1000.

1.3. Microsoft Exchange Server 2007 Unified Messaging

Version	RTM
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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.
- The Mitel PBX must have the same source number for MWI ON (Lamp on) and MWI OFF (Lamp off). Therefore, it's required to configure the gateway parameter MwiSourceNumber as the voicemail pilot number (i.e., 2690 in our testing environment example). This number is shown on the subscribers phone display when the MWI lamp is turned on.

2.2. PBX Prerequisites

- PBX with T1 dual framer module.
- T1 QSIG option.

2.3. Cabling Requirements

This integration uses cross-over RJ-48c cables (*pairs 1, 2 and 4, 5 crossed*) to connect digital trunks (T1/E1) between and Mediant 2000 trunk interfaces.

3. Summary and Limitations



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

4. Gateway Setup Notes

Step 1: SIP Environment Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Home Search Favorites

Address http://10.15.4.19/ Go Links >>

AudioCodes **TrunkPack 1610**
MG Module 1

Protocol Definition Advanced Parameters Manipulation Tables Routing Tables Profile Definitions Trunk Group Trunk Group Settings Digital Gateway Parameters VXML & RADIUS Parameters

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

Search
SIP

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
SIP Destination Port	5060

Web Server Internet

Step 2: Routing Setup

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.4.19/

AudioCodes TrunkPack 1610 MG Module 1

Protocol Definition | Advanced Parameters | Manipulation Tables | Routing Tables | Profile Definitions | Trunk Group | Trunk Group Settings | Digital Gateway Parameters | VXML & RADIUS Parameters

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

SIP

Search

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 correspond 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

Address: <http://10.15.4.19/>

AudioCodes

TrunkPack 1610
MG Module 1

Protocol Definition | Advanced Parameters | Manipulation Tables | Routing Tables | Profile Definitions | Trunk Group | Trunk Group Settings | Digital Gateway Parameters | VXML & 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.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 Reload Home Search Favorites RSS Print Mail New Tab

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

Address http://10.15.7.99/ Go

AudioCodes **TrunkPack 1610**
MG Module 2

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

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

 Log Off

Search

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;	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	
Default Destination Number	serveduser

Submit

Done Internet

Step 5: Supplementary Services Setting

The screenshot shows the AudioCodes TrunkPack 1610 MG Module 1 web interface. The browser window is titled "AudioCodes - Microsoft Internet Explorer" and the address bar shows "http://10.15.4.19/". The interface has a top navigation bar with the AudioCodes logo and the text "TrunkPack 1610 MG Module 1". Below this is a secondary navigation bar with links: Protocol Definition, Advanced Parameters, Manipulation Tables, Routing Tables, Profile Definitions, Trunk Group, Trunk Group Settings, Digital Gateway Parameters, and VXML & RADIUS Parameters. The main content area is titled "Supplementary Services" and contains a table with the following settings:

Setting	Value
Enable Hold	Enable
Hold Format	0.0.0.0
Enable Transfer	Enable
Transfer Prefix	9989
Enable Call Forward	Enable
Enable Call Waiting	Enable
Hook-Flash Code	

A black arrow points to the "Transfer Prefix" field, which contains the value "9989". Below the table is a "Submit" button. On the left side of the interface, there is a sidebar with a home icon and the following links: Quick Setup, Protocol Management, Advanced Configuration, Status & Diagnostics, Software Update, Maintenance, and Log Off. At the bottom of the sidebar, there is a search bar with the text "SIP" and a "Search" button. The status bar at the bottom of the browser window shows "Done" and "Internet".

Choose any 4-digit number that is not used in the PBX for Transfer Prefix (e.g., 9989).

Step 6: Manipulation Routing Setup

The screenshot shows the AudioCodes TrunkPack 1610 MG Module 1 web interface. The browser is Microsoft Internet Explorer with the address http://10.15.4.19/. The interface has a top navigation bar with the AudioCodes logo and the product name. Below this is a secondary navigation bar with tabs: Protocol Definition, Advanced Parameters, Manipulation Tables (highlighted), Routing Tables, Profile Definitions, Trunk Group, Trunk Group Settings, Digital Gateway Parameters, and VXML & RADIUS Parameters. On the left side, there is a sidebar with a home icon and a list of links: Quick Setup, Protocol Management (highlighted), Advanced Configuration, Status & Diagnostics, Software Update, Maintenance, and Log Off. Below these links is a search bar and the label SIP. The main content area is titled "Destination Phone Number Manipulation Table for Tel -> IP Calls". It features a "Table Index" dropdown set to "1-10". Below this is a table with 10 rows and 5 columns: Destination Prefix, Source Prefix, Number of stripped Digits, Prefix (Suffix) to Add, and Number of Digits to Leave. The first row is pre-filled with "9989" in the Destination Prefix, "*" in the Source Prefix, "4" in the Number of stripped Digits, and "0" in the Number of Digits to Leave. A "Submit" button is located at the bottom right of the table.

	Destination Prefix	Source Prefix	Number of stripped Digits	Prefix (Suffix) to Add	Number of Digits to Leave
1	9989	*	4		
2	*	*	0		0
3					
4					
5					
6					
7					
8					
9					
10					

The first manipulation for the destination prefix will strip the 4-digit number of the Transfer Prefix that was configured in the previous step.

Step 7: Trunk Group Setup

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.4.19/

AudioCodes **TrunkPack 1610**
MG Module 1

Protocol Definition | Advanced Parameters | Manipulation Tables | Routing Tables | Profile Definitions | **Trunk Group** | Trunk Group Settings | Digital Gateway Parameters | VXML & RADIUS Parameters

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

Search
SIP

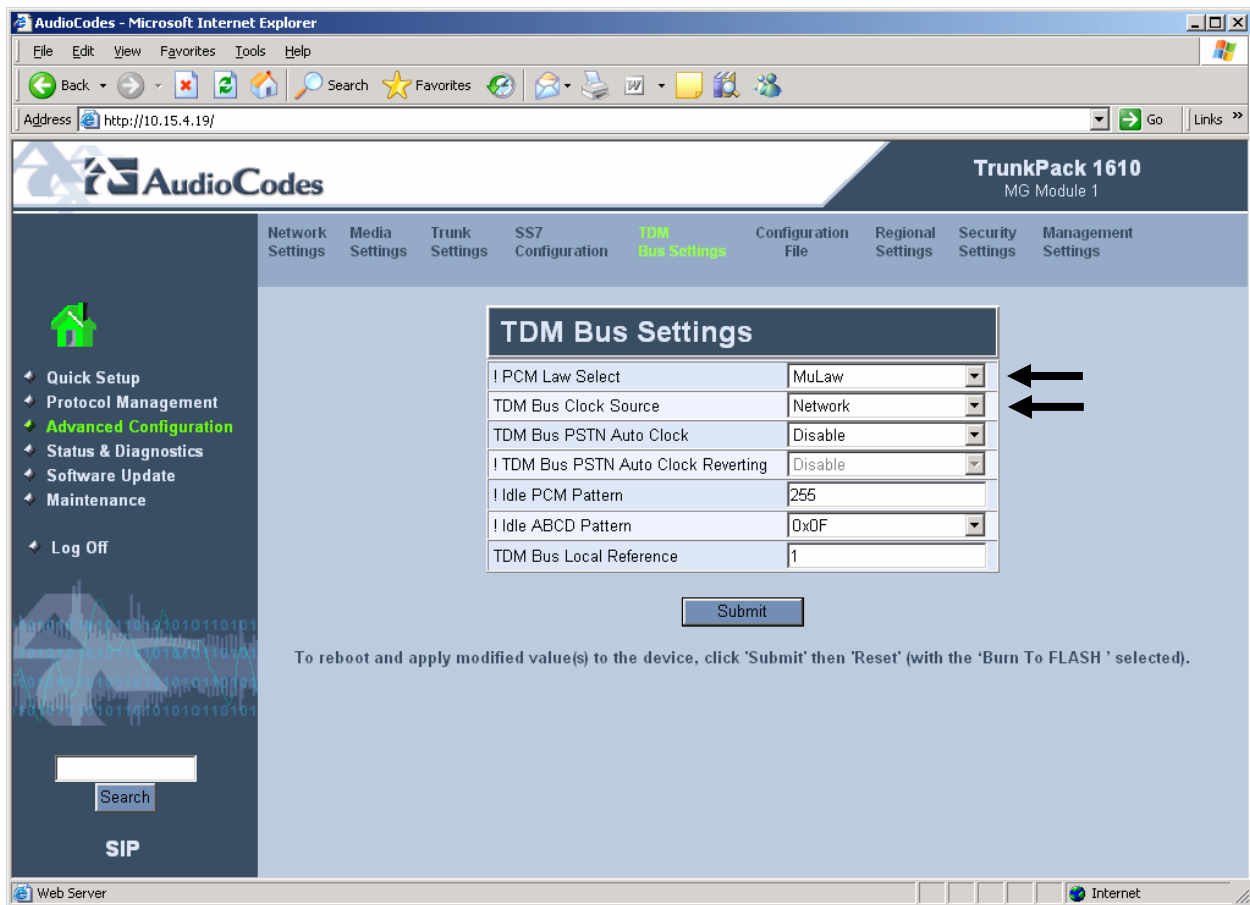
Trunk Group Table

Trunk Group Index: 1-12

Group Index	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	Profile ID
1	1	1	1-24	1000		0
2						
3						
4						
5						
6						
7						
8						
9						
10						
11						
12						

Web Server | Internet

Step 8: TDM BUS Setting



AudioCodes - Microsoft Internet Explorer

Address: http://10.15.4.19/

AudioCodes TrunkPack 1610 MG Module 1

Network Settings Media Settings Trunk Settings SS7 Configuration **TDM Bus Settings** Configuration File Regional Settings Security Settings Management Settings

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

TDM Bus Settings

! PCM Law Select	MuLaw
TDM Bus Clock Source	Network
TDM Bus PSTN Auto Clock	Disable
! TDM Bus PSTN Auto Clock Reverting	Disable
! Idle PCM Pattern	255
! Idle ABCD Pattern	0x0F
TDM Bus Local Reference	1

Submit

To reboot and apply modified value(s) to the device, click 'Submit' then 'Reset' (with the 'Burn To FLASH' selected).

Search

SIP

Web Server Internet

Step 9: Trunk Setting Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Address <http://172.20.22.201/> Go

AudioCodes

TrunkPack 1610

Network Settings Media Settings **Trunk Settings** SS7 Configuration TDM Bus Settings Configuration File Regional Settings Security Settings Management Settings

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

Search

SIP

Trunk Settings

Trunk Configuration	
Trunk ID	1
Trunk Configuration State	Inactive
Protocol Type	T1 QSIG
Clock Master	Recovered
Auto Clock Trunk Priority	0
Line Code	B8ZS
Line Build Out Loss	0 dB
Trace Level	No Trace
Line Build Out Overwrite	OFF
Framing Method	T1 FRAMING ESF CRC6
ISDN Configuration	
ISDN Termination Side	Network side
Q931 Layer Response Behavior	0x40000000 -->
Outgoing Calls Behavior	0x400 -->
Incoming Calls Behavior	0x0 -->
General Call Control Behavior	0x0 -->

Web Server Internet

Step 10: FAX Setup

The screenshot shows a web browser window with the address bar displaying 'http://10.15.4.19/'. The page title is 'AudioCodes 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 search bar and a list of configuration options: 'Quick Setup', 'Protocol Management', 'Advanced Configuration', 'Status & Diagnostics', 'Software Update', 'Maintenance', and 'Log Off'. The main content area is titled 'Fax/Modem/CID Settings' and contains a table of configuration options. A black arrow points to the 'Events Only' option in the 'CNG Detector Mode' dropdown menu.

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

Step 11: Voice Mail Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

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

Address http://10.15.4.13/ Go

AudioCodes

TrunkPack 1610

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

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

Log Off

Search

SIP

Voice Mail

General	
Voice Mail Interface	QSIG
Digit Patterns	
Forward on Busy Digit Pattern	
Forward on No Answer Digit Pattern	
Forward on Do Not Disturb Digit Pattern	
Forward on No Reason Digit Pattern	
Internal Call Digit Pattern	
External Call Digit Pattern	
Disconnect Call Digit Pattern	
MWI	
MWI Off Digit Pattern	
MWI On Digit Pattern	
SMDI	
Enable SMDI	No SMDI
SMDI Timeout [msec]	2000

Web Server Internet

Step 12: General Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Reload Home Search Favorites RSS Print Mail New Tab

Links Interop Tasks Google G Go Bookmarks 155 blocked Check AutoLink AutoFill Send to Settings

Address http://10.15.4.13/ Go

AudioCodes TrunkPack 1610

Protocol Definition **Advanced Parameters** Manipulation Tables Routing Tables Profile Definitions Trunk Group Trunk Group Settings Digital Gateway Parameters VXML & RADIUS Parameters IPMedia Parameters

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

General Parameters

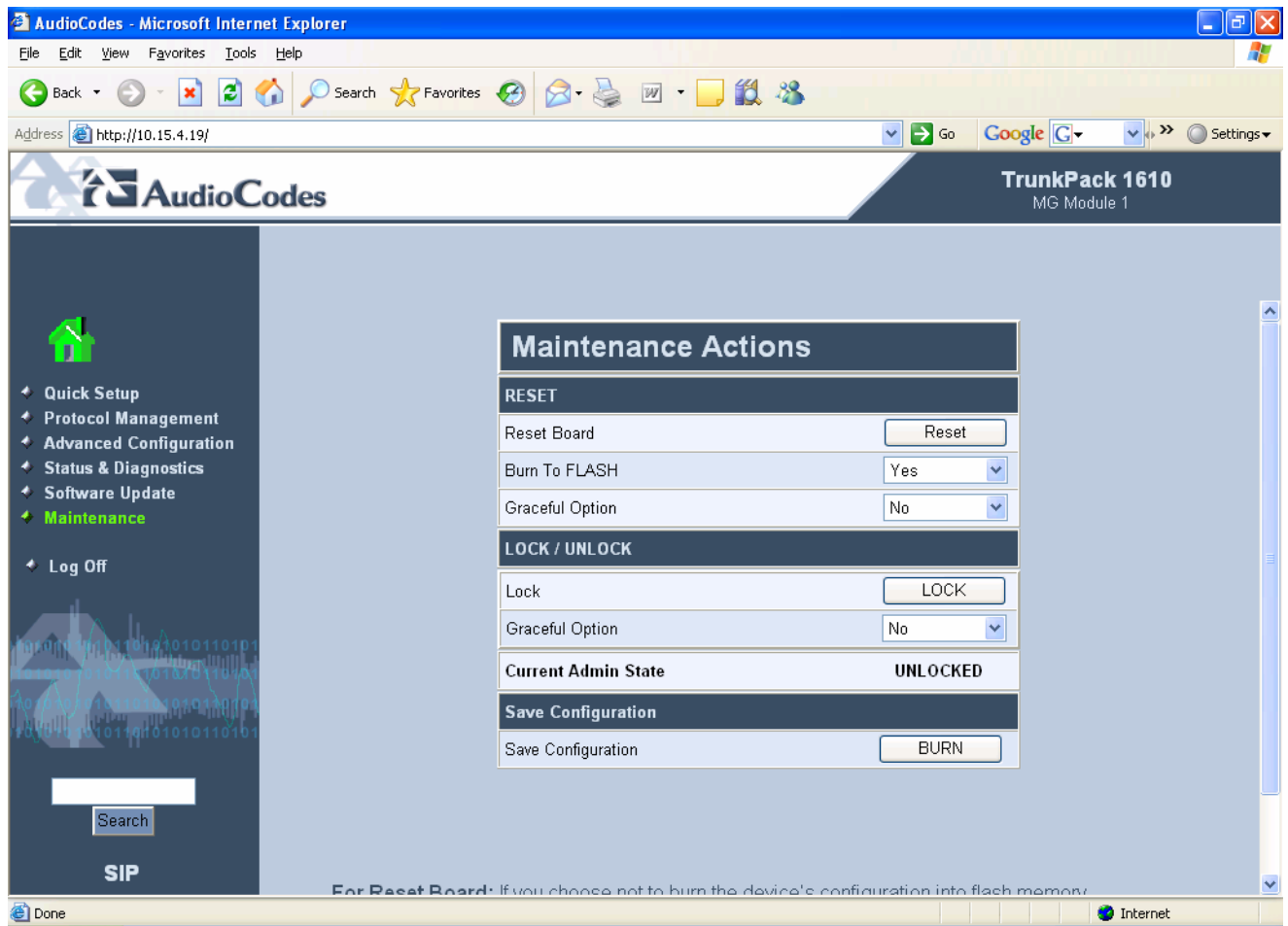
IP Security	Disable
Filter Calls to IP	Don't Filter
I Enable Digit Delivery to Tel	Disable
I Enable Digit Delivery to IP	Disable
Disconnect on Broken Connection	No
Broken Connection Timeout [100 msec]	100
Disconnect Call on Silence Detection	No
Silence Detection Period [sec]	120
Silence Detection Method	Packets Count
CDR and Debug	
CDR Server IP Address	
CDR Report Level	End Call
Debug Level	5
Misc. Parameters	
Progress Indicator to IP	Not Configured
Enable X-Channel Header	Disable

Done Internet

Step 13:

- ISDNIBehavior = 1073741824
 - EnableMWI = 1
 - SubscriptionMode = 1
 - EnableDetectRemoteMACChange = 2
 - ECNLPMODE = 1
 - MwiSourceNumber = 2690
- Note:** This parameter should be set to the Voice Mail pilot number (See Step 9 on PBX Setup).
- TrunkTransferMode_X = 0 (where "X" represents the Trunk number, for example: for the first trunk TrunkTransferMode_0 = 0)

Step 14: Reset Mediant 2000



Click **Reset** to reset the gateway.

4.1. Configuration Files

The ZIP file includes the following files:

1. AudioCodes Mediant 1000 / Mediant 2000 configuration for TCP environment (.ini file extension).
2. AudioCodes Mediant 1000 / Mediant 2000 configuration for TLS environment (.ini file extension).



Mitel 3300 T1 QSIG Audiocodes Mediant 1000 & 2000 TLS.zip

4.2. TLS Setup

Step 1: PBX to IP Routing Setup

Proxy & Registration	
Enable Proxy	Use Proxy
Proxy Name	exchange2007.server2003.c
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 Not Supp	Disable

Note: The Proxy IP Address and Name must correspond 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).

The screenshot shows the AudioCodes TrunkPack 1610 configuration interface. The browser is Microsoft Internet Explorer. The address bar shows <http://10.15.4.13/>. The page title is "AudioCodes TrunkPack 1610". The navigation menu on the left includes: Quick Setup, Protocol Management (highlighted), Advanced Configuration, Status & Diagnostics, Software Update, Maintenance, and Log Off. The main content area has tabs for: Protocol Definition, Advanced Parameters, Manipulation Tables, Routing Tables, Profile Definitions, Trunk Group, Trunk Group Settings (active), Digital Gateway Parameters, and Advanced Applications. The "Trunk Group Settings" tab displays a list of configuration parameters:

Parameter	Value
Enable fallback to routing table	Disable
Prefer Routing Table	No
Use Routing Table for Host Names and Profiles	Disable
Always Use Proxy	Disable
Send All Invite to Proxy	No
Enable Proxy Hot-Swap	Disable
Enable Registration	Disable
Gateway Name	gw1.m2k.audiocodes.com
Gateway Registration Name	
DNS Query Type	A-Record
Proxy DNS Query Type	A-Record
Use Gateway Name for OPTIONS	No
Number of RTX Before Hot-Swap	3
User Name	
Password	
Cnonce	Default_Cnonce

A black arrow points to the "Gateway Name" field. At the bottom of the page, there are buttons for "Register", "Un-Register", and "Submit". The status bar at the bottom indicates "Web Server" and "Internet".

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Step 3: SIP Environment Setup (Cont.)

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.4.13/

AudioCodes TrunkPack 1610

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	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
Use "user-agent" in SIP URI	Yes

Web Server | Internet

Step 4: DNS Servers Setup

AudioCodes - Microsoft Internet Explorer

Address: <http://10.15.4.13/>

AudioCodes TrunkPack 1610

Network Settings Media Settings Trunk Settings SS7 Configuration TDM Bus Settings Configuration File Regional Settings Security Settings Management Settings

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

SIP

IP Settings

IP Networking Mode	Single IP Network
IP Address	10.15.4.13
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

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 TrunkPack 1610 web interface in a Microsoft Internet Explorer browser. The address bar shows <http://10.15.4.13/>. The interface has a top navigation bar with the AudioCodes logo and the title "TrunkPack 1610". 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 a list of links: Quick Setup, **Protocol Management** (highlighted in green), 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

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.4.13/

AudioCodes TrunkPack 1610

Network Settings Media Settings Trunk Settings SS7 Configuration TDM Bus Settings Configuration File Regional Settings Security Settings Management Settings

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

SIP

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

STUN Settings

Enable STUN Disable

STUN Server Primary IP 0.0.0.0

STUN Server Secondary IP 0.0.0.0

NFS Settings

NFS Table -->

Submit

Web Server Internet

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

Step 7: Generate Certificate Setup

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

The screenshot shows the AudioCodes TrunkPack 1610 web interface in Microsoft Internet Explorer. The browser's address bar shows `http://10.15.4.13/`. The page has a navigation menu with options: Network Settings, Media Settings, Trunk Settings, SS7 Configuration, TDM Bus Settings, Configuration File, Regional Settings, Security Settings (highlighted), and Management Settings. On the left, there is a sidebar with a home icon and links: Quick Setup, 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 'Certificate Signing Request'. It contains a form with a 'Subject Name' field set to 'gw1.m2k.audiocodes.com' and a 'Generate CSR' button. Below the button, it says 'Copy the certificate signing request and send it to your Certification Authority for signing.' and displays a long base64-encoded string for the CSR. At the bottom, there is a small note: 'Press the button "Generate self-signed" to create a self-signed certificate using the subject name provided above.'

AudioCodes TrunkPack 1610

Network Settings Media Settings Trunk Settings SS7 Configuration TDM Bus Settings Configuration File Regional Settings **Security Settings** Management Settings

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

SIP

Certificate Signing Request

Subject Name

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

```
-----BEGIN CERTIFICATE REQUEST-----
MIIBYDCBygIBADAhMR8wHQYDVQOExZndzEubTJrLmF1ZGlvY29kZXMuY29tMIGf
MAOGCSqGSIb3DQEBAQUAA4GNADCBiQKBQC2AZdeiLwVBBni+bI638fpEWIn4nuh
cc6U1LhAMPuhtjoaqlYeNeFMfT6i92wEX0gehQbxAZyA0P0jqT0bbmrROMugs8pt
hghJTbajgg1INFppccm181pW7FDz4rZyMZtm5pC4PgYT6W2QF4MyxhE3h2qmLk2f
UEH5XCaOMFI8CQIDAQABoAAwDQYJKoZIhvcNAQEEBQADgYEAkwv9R2XqUEqCRUXb
wV8LVp+7OzkzCMQkc3FUKqL6uqs9IW5B2h+MyO/nxBcWV355sT/zksb0d6hW13E9
rCa+ar0OM1yOmUDcTh1TqobakBN53PKLMDElkG1QcFRfTFs9UggsBDCg9nHYPrBP
1Mow1QrUvnp711Fhuedt/h2xUUo=
-----END CERTIFICATE REQUEST-----
```

Press the button "Generate self-signed" to create a self-signed certificate using the subject name provided above.

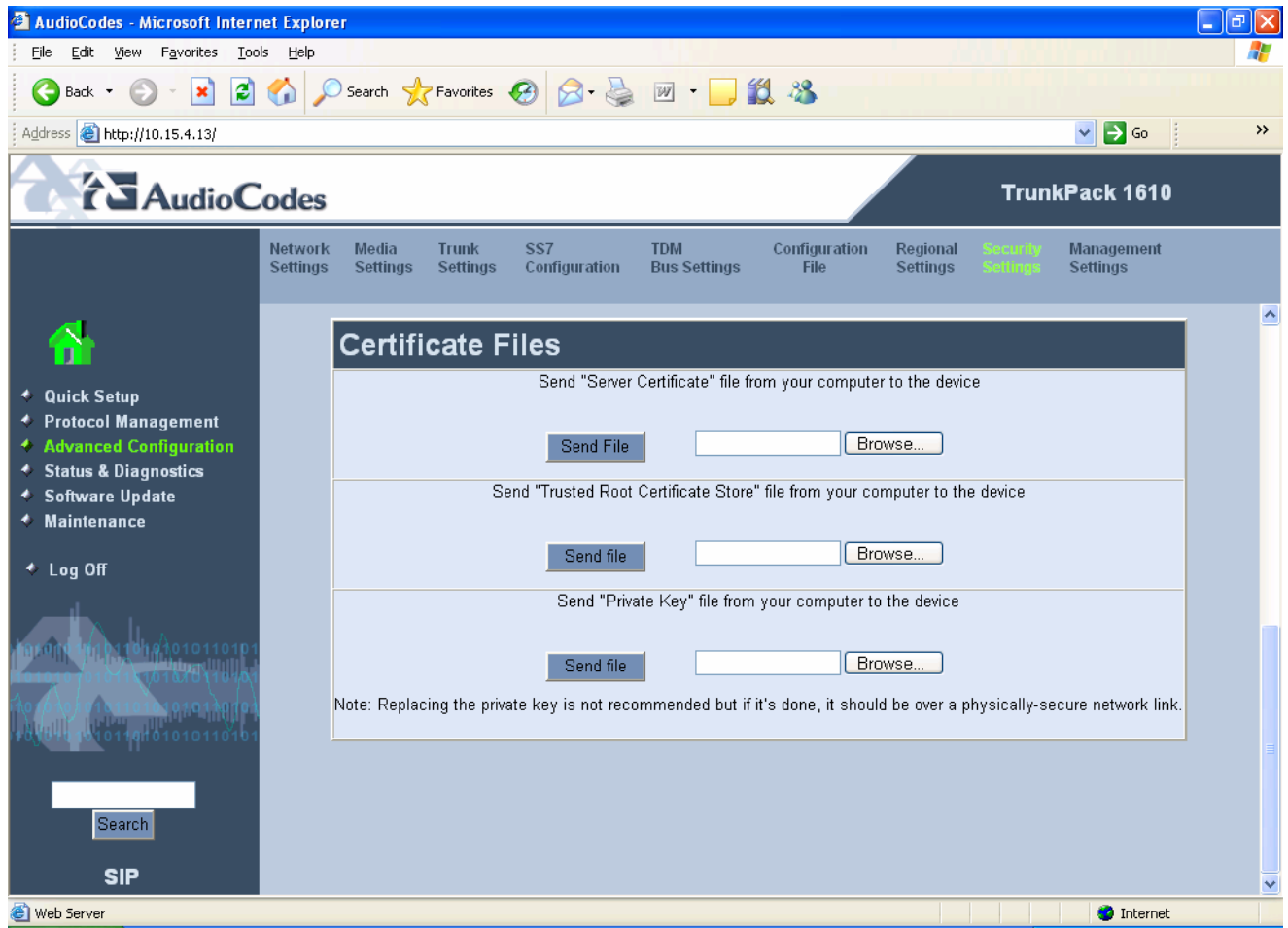
Web Server Internet

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:

- **Digital VoiceMail ports:** T1 QSIG trunk
- **VoiceMail Hunt Group Pilot:** 2690
- **VoiceMail User Phone:** ext. 2608 and ext. 8999

Step 1: Ensure PBX is equipped with T1 PRI Module to Support T1 QSIG

Use the following path to ensure that the PBX is equipped with a T1 PRI module:

System Configuration / UnitsModules / Units Configuration Display.

The screenshot shows the Mitel Networks 3300 ICP web interface in a Microsoft Internet Explorer browser. The top status bar indicates an alarm: "Alarm Status: Major 2007-Apr-25 14:50:31". The main navigation pane on the left shows the "System Configuration" tree, with "Units/Modules" expanded and "Unit Configuration Display" selected. The main content area displays a table titled "Unit Configuration Display" with columns "Unit", "Unit Type", and "Comment". The table lists 16 units, with unit 6 highlighted as "Dual T1/E1 Framer" with a "PRI" comment.

Unit	Unit Type	Comment
1	ICP	
2		
3		
4	Analog Service Unit	
5		
6	Dual T1/E1 Framer	PRI
7	Peripheral/DSU Unit	FD Per
8		
9		
10		
11		
12		
13		
14		
15		
16		

MITEL 3300 ICP
About System Administration

Step 2: Create Class of Service for QSIG Trunks

Use the following path to create CoS for QSIG trunk:

System Configuration / Trunks / Class of Service Options Assignment.

Alarm Status: ! Major 2007-Apr-25 14:50:31

Audiocode - System Message: Requested record changed successfully.

Selection: System Configuration

- System Configuration
 - System Capacity
 - IP Network Configuration
 - Voice Network Configuration
 - Units/Modules
 - Trunks
 - Class of Service Options**
 - Analog Trunks
 - Digital Trunks
 - DPNSS/DASS2
 - ISDN-BRI
 - ISDN-PRI
 - Link Descriptor Assi
 - Digital Link Assignm
 - MSDN-DPNSS-DAS
 - Trunk Service Assi
 - Digital Trunk Assignr
 - Network Synchroniz
 - E1
 - T1
 - Protocol Assignm
 - Outgoing Call Cha
 - T1/D4
 - IP Networking/XNET
 - Direct Trunk Select
 - Automatic Route Selecti
 - Devices
 - Music On Hold
 - Loudspeaker Paging
 - Telephone Paging
 - System Output Ports

Class of Service Options Assignment Search:

Find a field named: that has a value of:

Class of Service Options Assignment	
Class Of Service Number	Comment
4	QSIG
5	SS's w/ind Trk
6	
7	Console
8	ISDN Trk's

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Change the following options to 'Yes':

- ANI/DNIS/ISDN Number Delivery Trunk
- Call Announce Line
- Call Forwarding (External Destination)
- Call Hold - Retrieve with Hold Key
- Call Reroute after CFFM to busy Destination

-- Web Page Dialog

Class of Service Options Assignment

Class Of Service Number: 4

Comment: Superset&QSIG

Account Code Verified:	☐ No	☒ Yes
ACD Make Busy on Login:	☒ No	☐ Yes
ACD Silent Monitor Accept:	☒ No	☐ Yes
ACD Silent Monitor Allowed:	☒ No	☐ Yes
ACD Silent Monitor Notification:	☒ No	☐ Yes
ANI/DNIS/ISDN Number Delivery Trunk:	☐ No	☒ Yes
Auto Answer Allowed:	☐ No	☒ Yes
Brokers Call:	☒ No	☐ Yes
Busy Override Security:	☒ No	☐ Yes
Call Announce Line:	☐ No	☒ Yes
Call Forwarding Accept:	☐ No	☒ Yes
Call Forwarding (External Destination):	☐ No	☒ Yes
Call Forwarding (Internal Destination):	☐ No	☒ Yes
Call Forward Override:	☒ No	☐ Yes
Call Hold:	☐ No	☒ Yes
Call Hold Remote Retrieve:	☐ No	☒ Yes
Call Hold - Retrieve with Hold Key:	☐ No	☒ Yes
Call Park-Allowed To Park:	☒ No	☐ Yes
Call Pickup Dialed Accept:	☐ No	☒ Yes
Call Pickup Directed Accept:	☐ No	☒ Yes
Call Privacy:	☐ No	☒ Yes
Call Reroute after CFFM to Busy Destination:	☐ No	☒ Yes
Call Waiting Swap:	☒ No	☐ Yes
Calling Name Display - Internal - ONS:	☐ No	☒ Yes
Calling Number Display - Internal - ONS:	☐ No	☒ Yes
Calling Party Name Substitution:	☒ No	☐ Yes
Campon Tone Security / FAX Machine:	☒ No	☐ Yes
Check COR after PSTN Dial Tone:	☒ No	☐ Yes
Clear All Features Remote:	☐ No	☒ Yes
Conference Call:	☐ No	☒ Yes
COV/ONS/E&M Voice Mail Port:	☒ No	☐ Yes
DASS II OLI/TLI Provided:	☒ No	☐ Yes
Dialled Night Service:	☐ No	☒ Yes
Disable Call Reroute Chaining On Diversion:	☒ No	☐ Yes
Disable Conference Join Tone:	☐ No	☒ Yes
Disable Executive Busy Override Tone:	☒ No	☐ Yes

Save Cancel

Change the following options to 'Yes':

- Display ANI/DNIS/ISDN Calling/Called Number
- Display DNIS/Called Number Before Digit Modification
- Display Dialed Digits During Outgoing Calls
- Display Held Call ID on Transfer

The screenshot shows a 'Web Page Dialog' window with a list of configuration options. Each option has two radio buttons, one for 'No' and one for 'Yes'. The options are as follows:

Option	No	Yes
DASS II OLI/TLI Provided:	☒	☐
Dialled Night Service:	☒	☐
Disable Call Reroute Chaining On Diversion:	☒	☐
Disable Conference Join Tone:	☒	☐
Disable Executive Busy Override Tone:	☒	☐
Disable Send Message:	☒	☐
Display ANI/ISDN Calling Number Only:	☒	☐
Display ANI/DNIS/ISDN Calling/Called Number:	☐	☒
Display Caller ID on multicall/keylines:	☐	☒
Display DNIS/Called Number Before Digit Modification:	☐	☒
Display Dialed Digits during Outgoing Calls:	☐	☒
Display Held Call ID on Transfer:	☐	☒
Do Not Disturb:	☐	☒
Do Not Disturb - Access to Remote Phones:	☐	☒
Do Not Disturb Permanent:	☐	☒
Emergency Call Notification - Audio:	☒	☐
Emergency Call Notification - Visual:	☒	☐
Enable Call Duration Limit on External Calls:	☒	☐
Enable Call Duration Limit on Internal Calls:	☒	☐
Executive Busy Override:	☒	☐
External Trunk Standard Ringback:	☒	☐
Flexible Answer Point:	☒	☐
Follow 2nd Alternate Reroute for Recall to Busy ACD Agent:	☒	☐
Forced Verified Account Code:	☒	☐
Forced Non-Verified Account Code:	☒	☐
Group Call Forward Follow Me Accept:	☐	☒
Group Call Forward Follow Me Allow:	☐	☒
Group Page Accept:	☐	☒
Group Page Allow:	☐	☒
Handset Volume Adjustment Saved:	☐	☒
Handsfree AnswerBack Allowed:	☒	☐
HCI/CTI/TAPI Call Control Allowed:	☒	☐
HCI/CTI/TAPI Monitor Allowed:	☒	☐
Head Set Switch Mute:	☒	☐
Hot Desk Remote Logout Enabled:	☒	☐
Hot Desk Login Accept:	☒	☐
Hotel Room Extension:	☒	☐
Hotel Room Monitor Setup Allowed:	☒	☐
Hotel Room Monitoring Allowed:	☒	☐
Hotel/Motel Room Personal Wakeup Call Allowed:	☒	☐
Hotel/Motel Room Remote Wakeup Call Allowed:	☒	☐
Individual Trunk Access:	☒	☐

At the bottom right of the dialog are 'Save' and 'Cancel' buttons.

Change the following options to 'Yes':

- Message Waiting - Audible Tone Notification
- ONS CLASS/CLIP: Message Waiting Activate/Deactivate
- ONS CLASS/CLIP Set
- Public Network Access via DPNSS
- Public Network Identity Provided
- Public Network to Public Network Connection Allowed
- Public Trunk
- R2 Call Progress Tone

-- Web Page Dialog		
Hotel Room Extension:	☐ No	☐ Yes
Hotel Room Monitor Setup Allowed:	☐ No	☐ Yes
Hotel Room Monitoring Allowed:	☐ No	☐ Yes
Hotel/Motel Room Personal Wakeup Call Allowed:	☐ No	☐ Yes
Hotel/Motel Room Remote Wakeup Call Allowed:	☐ No	☐ Yes
Individual Trunk Access:	☐ No	☐ Yes
Keep TelDir Entry on Check Out:	☐ No	☐ Yes
Local Music On Hold source:	☐ No	☐ Yes
Loudspeaker Pager Override:	☐ No	☐ Yes
Loudspeaker Pager Equivalent Zone Override Security:	☐ No	☐ Yes
Message Waiting:	☐ No	☐ Yes
Message Waiting Audible Tone Notification:	☐ No	☐ Yes
Message Waiting Deactivate On Off-Hook:	☐ No	☐ Yes
Message Waiting Inquire:	☐ No	☐ Yes
Multiline Set Loop Test:	☐ No	☐ Yes
Multiline Set Message Center Remote Read Allowed:	☐ No	☐ Yes
Multiline Set Music:	☐ No	☐ Yes
Multiline Set On-hook Dialing:	☐ No	☐ Yes
Multiline Set Phonebook Allowed:	☐ No	☐ Yes
Multiline Set Voice Mail Callback Message Erasure Allowed:	☐ No	☐ Yes
Name Suppression on outgoing Trunk Call:	☐ No	☐ Yes
Non DID Extension:	☐ No	☐ Yes
Non-Prime Public Network Identity:	☐ No	☐ Yes
Non Verified Account Code:	☐ No	☐ Yes
Off-Hook Voice Announce Allowed:	☐ No	☐ Yes
ONS CLASS/CLIP: Message Waiting Activate/Deactivate:	☐ No	☐ Yes
ONS CLASS/CLIP: Set:	☐ No	☐ Yes
ONS CLASS/CLIP: Visual Call Waiting:	☐ No	☐ Yes
ONS/OPS Internal Ring Cadence for External Callers:	☐ No	☐ Yes
Override Interconnect Restriction on Transfer:	☐ No	☐ Yes
Pager Access All Zones:	☐ No	☐ Yes
Pager Access Individual Zones:	☐ No	☐ Yes
Privacy Released:	☐ No	☐ Yes
Public Network Access via DPNSS:	☐ No	☐ Yes
Public Network Identity Provided:	☐ No	☐ Yes
Public Network To Public Network Connection Allowed:	☐ No	☐ Yes
Public Trunk:	☐ No	☐ Yes
R2 Call Progress Tone:	☐ No	☐ Yes
Record-A-Call Active:	☐ No	☐ Yes
Record-A-Call - Start Recording Automatically:	☐ No	☐ Yes
Record-A-Call - Save Recording on Hang-up:	☐ No	☐ Yes
Recorded Announcement Device:	☐ No	☐ Yes

Save Cancel

Change the following options to 'Yes':

- Third Party Call Forward Follow Me Accept
- Third Party Call Forward Follow Me Allow
- Trunk Flash Allowed

-- Web Page Dialog

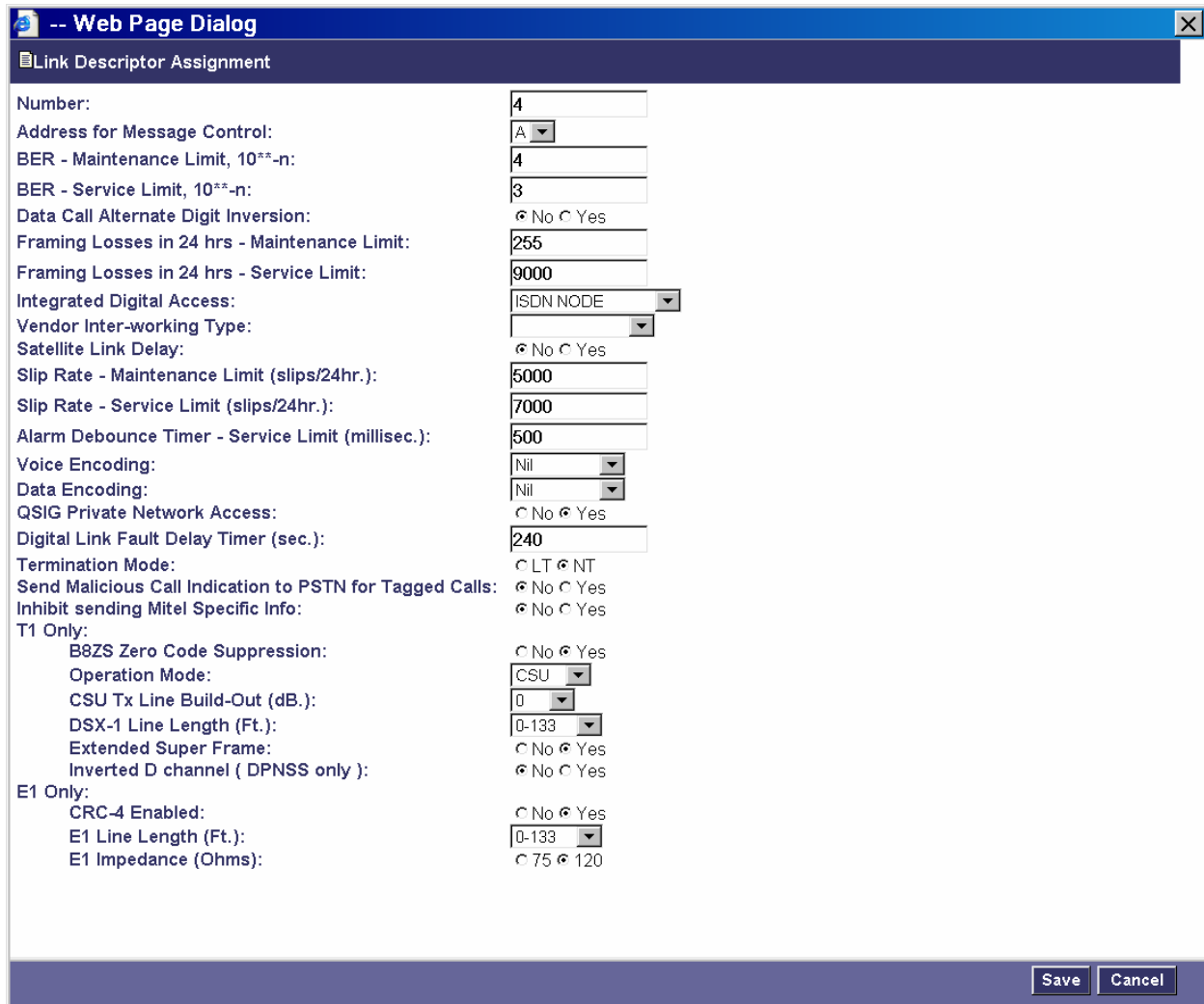
R2 Call Progress Tone:	☒ No	☐ Yes
Record-A-Call Active:	☒ No	☐ Yes
Record-A-Call - Start Recording Automatically:	☒ No	☐ Yes
Record-A-Call - Save Recording on Hang-up:	☒ No	☐ Yes
Recorded Announcement Device:	☒ No	☐ Yes
Recorded Announcement Device - Advanced:	☒ No	☐ Yes
Redial Facilities:	☐ No	☒ Yes
Ringing Line Select:	☐ No	☒ Yes
SC1000 Attendant Basic Function Key:	☒ No	☐ Yes
SMDR External:	☐ No	☒ Yes
SMDR Internal:	☒ No	☐ Yes
Speak@Ease Preferred:	☒ No	☐ Yes
Suite Services Enabled:	☒ No	☐ Yes
Suppress Simulated CCM after ISDN Progress:	☒ No	☐ Yes
Third Party Call Forward Follow Me Accept:	☐ No	☒ Yes
Third Party Call Forward Follow Me Allow:	☐ No	☒ Yes
Timed Reminder Allowed:	☐ No	☒ Yes
Trunk Calling Party Identification:	☐ No	☒ Yes
Trunk Flash Allowed:	☒ No	☐ Yes
Two B-Channel Transfer Allowed:	☒ No	☐ Yes
Use Held Party Device for Call Re-routing:	☒ No	☐ Yes
Use Called Party Call Hold Timer:	☒ No	☐ Yes
Voice Mail Softkey:	☒ No	☐ Yes
Account Code Length:	4	
After Answer Display Time:		
Answer Plus Delay To Message Timer:	20	
Answer Plus Expected Off-hook Timer:	30	
Answer Plus Message Length Timer:	10	
Answer Plus System Reroute Timer:	0	
Attendant Busy Out Timer:	1440	
Auto Campon Timer:	10	
Busy Tone Timer:	30	
Call Duration:	10	
Call Duration Forced Cleardown Timer:	0	
Call Forward - Delay:	0	
Call Forward No Answer Timer:	15	

Save Cancel

Step 3: Create Link Descriptor Assignment

Use the following path to create Link Descriptor Assignment:

System Configuration / Trunks / Digital Trunks / ISDN-PRI / Link Descriptor Assignment.



The image shows a web-based configuration dialog titled "Link Descriptor Assignment". It contains various settings for a link descriptor, organized into sections. The settings are as follows:

Parameter	Value
Number:	4
Address for Message Control:	A
BER - Maintenance Limit, 10 ^{xx} -n:	4
BER - Service Limit, 10 ^{xx} -n:	3
Data Call Alternate Digit Inversion:	☒ No ☐ Yes
Framing Losses in 24 hrs - Maintenance Limit:	255
Framing Losses in 24 hrs - Service Limit:	9000
Integrated Digital Access:	ISDN NODE
Vendor Inter-working Type:	
Satellite Link Delay:	☒ No ☐ Yes
Slip Rate - Maintenance Limit (slips/24hr.):	5000
Slip Rate - Service Limit (slips/24hr.):	7000
Alarm Debounce Timer - Service Limit (millisec.):	500
Voice Encoding:	Nil
Data Encoding:	Nil
QSIG Private Network Access:	☐ No ☒ Yes
Digital Link Fault Delay Timer (sec.):	240
Termination Mode:	☐ LT ☒ NT
Send Malicious Call Indication to PSTN for Tagged Calls:	☒ No ☐ Yes
Inhibit sending Mitel Specific Info:	☒ No ☐ Yes
T1 Only:	
B8ZS Zero Code Suppression:	☐ No ☒ Yes
Operation Mode:	CSU
CSU Tx Line Build-Out (dB.):	0
DSX-1 Line Length (Ft.):	0-133
Extended Super Frame:	☐ No ☒ Yes
Inverted D channel (DPNSS only):	☒ No ☐ Yes
E1 Only:	
CRC-4 Enabled:	☐ No ☒ Yes
E1 Line Length (Ft.):	0-133
E1 Impedance (Ohms):	☐ 75 ☒ 120

At the bottom right of the dialog are "Save" and "Cancel" buttons.

Step 4: Create a Digital Link Assignment

Use the following path to create a Digital Link Assignment:

System Configuration / Trunks / Digital Trunks / ISDN-PRI / Digital Link Assignment.

The screenshot shows the Mitel Networks 3300 Integrated Communications Platform (ICP) web interface in Microsoft Internet Explorer. The browser title is "Audiocode - Mitel Networks® 3300 Integrated Communications Platform (ICP) - Microsoft Internet Explorer". The interface includes a menu bar (File, Edit, View, Favorites, Tools, Help) and a status bar at the bottom with the Mitel logo and "3300 ICP About System Administration".

The main content area is titled "Audiocode - System Message:" and displays "Alarm Status: Major 2007-Apr-25 14:50:31". Below this, there are buttons for "Print...", "Import...", "Export...", "Data Refresh", "Help", and "Exit".

The left sidebar shows a tree view of the system configuration. The "Trunks" folder is expanded, and the "Digital Trunks" folder is selected. The "ISDN-PRI" folder is also expanded, showing sub-items like "Link Descriptor Assi", "Digital Link Assignm", "MSDN-DPNSS-DAS", "Trunk Service Assigr", "Digital Trunk Assignr", and "Network Synchroniz".

The main content area displays the "Digital Link Assignment" page. It includes a "Previous" button, "Page 1 of 1", a "Next" button, and a "Go to:" field with a "value:" field and a "Go" button. The page title is "Digital Link Assignment".

Controller Module	Port	Unit	Shelf	Slot	Link	Interface Type	Digital Link Descriptor	Resilient Link	Comment
1	1	6	1	2	1	UNIVERSAL T1	4	False	PRI
1	2	6	1	3	1	UNIVERSAL T1	4	False	QSIG

-- Web Page Dialog

Digital Link Assignment

Controller Module:	1
Port:	2
Unit:	6
Shelf:	1
Slot:	3
Link:	1
Interface Type:	UNIVERSAL T1
Digital Link Descriptor:	4
Comment:	QSIG
Resilient Link:	☐
Resilient Link ID:	1
Primary Network Element:	
Secondary Network Element:	

Save Cancel

Step 5: Add MSDN-DPNSS-DASSII Trunk Circuit Descriptor

Use the following path to Add MSDN-DPNSS-DASSII Trunk Circuit Descriptor:

System Configuration / Trunks / Digital Trunks / ISDN-PRI/MSDN-DPNSS-DASSII Trunk Circuit Descriptor.

The screenshot shows the Mitel Networks 3300 ICP web interface in Microsoft Internet Explorer. The top status bar indicates an alarm status of 'Major' on 2007-Apr-25 at 14:50:31. The left navigation pane shows the 'System Configuration' tree with the following structure:

- System Configuration
 - System Capacity
 - IP Network Configuration
 - Voice Network Configuration
 - Units/Modules
 - Trunks
 - Class of Service Options
 - Analog Trunks
 - Digital Trunks
 - DPNSS/DASS2
 - ISDN-BRI
 - ISDN-PRI
 - Link Descriptor Assig
 - Digital Link Assignm
 - MSDN-DPNSS-DAS**
 - Trunk Service Assigt
 - Digital Trunk Assignr
 - Network Synchroniz
 - E1
 - T1
 - T1/D4
 - IP Networking/XNET
 - Direct Trunk Select
 - Automatic Route Selection
 - Devices
 - Music On Hold
 - Loudspeaker Paging
 - Telephone Paging
 - System Output Ports

The main content area displays the 'MSDN-DPNSS-DASSII Trunk Circuit Descriptor' configuration page. It includes a table with the following data:

Number	Card Type	Dual Seizure Priority	Far End Connection	Signalling Protocol	ISDN BRI Mode
4	UNIVERSAL T1	Incoming	Local Office	MSDN-DPNSS	

The bottom of the interface features the Mitel logo and the text '3300 ICP About System Administration'.

-- Web Page Dialog

MSDN-DPNSS-DASSII Trunk Circuit Descriptor

Number: 4

Card Type: UNIVERSAL T1

Dual Seizure Priority: ☒ Incoming ☐ Outgoing

Far End Connection: Local Office

Signalling Protocol: ☒ MSDN-DPNSS ☐ DASS II

ISDN BRI Mode:

Save Cancel

Step 6: Create a Trunk Service Assignment

Use the following path to create a Trunk Service Assignment:

System Configuration / Trunks / Digital Trunks / ISDN-PRI / Trunk Service Assignment.

The screenshot displays the Mitel Networks 3300 Integrated Communications Platform (ICP) web interface. The left sidebar shows the navigation tree with 'Trunk Service Assignment' selected under 'ISDN-PRI'. The main content area shows a table of trunk service assignments and a configuration summary.

Alarm Status: ! Major 2007-Apr-25 14:50:31

Audiocode - System Message:

Selection: System Configuration

Change Change Page Change All Clear

Previous Page 1 of 15 Next Go to: value: Go

Trunk Service Number	Release Link Trunk	Class of Service	Class of Restriction	Baud Rate	Intercept Number	Trunk Label
1	No	10	5	9600	1	
2	No	9	1	9600	1	
3	No	8	5	9600	1	
4	No	11	1	300	1	
5	No	1	1	300	1	
6	No	1	1	300	1	
7	No	1	1	300	1	
8	No	1	1	300	1	
9	No	1	1	300	1	
10	No	1	1	300	1	

Trunk Service Assignment

Trunk Service Number: 4
Release Link Trunk: No
Class of Service: 11
Class of Restriction: 1
Baud Rate: 300
Intercept Number: 1
Non-dial In Trunks Answer Point - Day:
Non-dial In Trunks Answer Point - Night 1:
Non-dial In Trunks Answer Point - Night 2:
Dial In Trunks Incoming Digit Modification - Absorb: 0
Dial In Trunks Incoming Digit Modification - Insert:
Trunk Label:

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About System Administration

Step 7: Assign Digital Trunks

Use the following path to assign Digital Trunks:

System Configuration / Trunks / Digital Trunks / ISDN-PRI / Digital Trunk Assignment.

The screenshot displays the Mitel Networks 3300 ICP web interface in a Microsoft Internet Explorer browser. The top status bar shows an alarm status and the date/time. The left sidebar contains a tree view of the system configuration, with 'Digital Trunk Assignment' selected under the 'ISDN-PRI' section. The main content area shows a table of digital trunk assignments, with the fourth row (Cabinet 6, Shelf 1, Slot 3, Circuit 1) highlighted. Below the table, a summary section lists the configuration details for the selected row.

Alarm Status: ! Major 2007-Apr-25 14:50:31

Print... Import... Export... Data Refresh Help Exit

Audiocode - System Message:

Selection: System Configuration

System Configuration

- System Capacity
- IP Network Configuration
- Voice Network Configuration
- Units/Modules
- Trunks
 - Class of Service Options
 - Analog Trunks
 - Digital Trunks
 - DPNSS/DASS2
 - ISDN-BRI
 - ISDN-PRI
 - Link Descriptor Assig
 - Digital Link Assignm
 - MSDN-DPNSS-DAS
 - Trunk Service Assig
 - Digital Trunk Assig
 - Network Synchroniz
 - E1
 - T1
 - T1/D4
 - IP Networking/XNET
 - Direct Trunk Select
 - Automatic Route Selectio
- Devices
- Music On Hold
- Loudspeaker Paging
- Telephone Paging
- System Output Ports

Change Clear

Previous Page 3 of 5 Next Go to: value: Go

Cabinet	Shelf	Slot	Circuit	Card Type	Trunk Number
6	1	2	21	UNIVERSAL T1	1121
6	1	2	22	UNIVERSAL T1	1122
6	1	2	23	UNIVERSAL T1	1123
6	1	3	1	UNIVERSAL T1	1201
6	1	3	2	UNIVERSAL T1	1202
6	1	3	3	UNIVERSAL T1	1203
6	1	3	4	UNIVERSAL T1	1204
6	1	3	5	UNIVERSAL T1	1205
6	1	3	6	UNIVERSAL T1	1206
6	1	3	7	UNIVERSAL T1	1207

Digital Trunk Assignment

Cabinet: 6
Shelf: 1
Slot: 3
Circuit: 1
Card Type: UNIVERSAL T1
Trunk Number: 1201
Trunk Service Number: 4
DTS Service Number:
Circuit Descriptor Number: 4
Interconnect Number: 11
Tenant Number: 1

MITEL 3300 ICP
About System Administration

Range Programming -- Web Page Dialog

Change Range Programming - Digital Trunk Assignment
Help

This form allows you to change one or more records, starting at the following record:

Cabinet	Shelf	Slot	Circuit	Card Type	Trunk Number	Trunk Service Number	DTS Service Number	Circuit Descriptor Number	Interconnect Number	Tenant Number
6	1	3	1	UNIVERSAL T1	1201	4		4	11	1

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Cabinet:	-	6	-
Shelf:	-	1	-
Slot:	-	3	-
Circuit:	-	1	-
Card Type:	-	UNIVERSAL T1	-
Trunk Number:	Change to	1201	
Trunk Service Number:	Change to	4	
DTS Service Number:	Change to		
Circuit Descriptor Number:	Change to	4	
Interconnect Number:	Change to	11	
Tenant Number:	Change to	1	

Preview
Save
Cancel

Step 8: Add Trunk Group for QSIG

Use the following path to assign ARS for call routing:

System Administration / Automatic Route Selection / Trunk Group Assignment.

Add a new trunk group for QSIG and add members to this group.

Alarm Status: Major 2007-Apr-25 14:50:31

Audiocode - System Message:

Selection:

System Administration

- System Administration
 - System Audio Files Update
 - User Authorization Profiles
 - System Options
 - Automatic Route Selection (ARS)
 - Trunk Service Assignment
 - Trunk Group Assignment**
 - Class of Restriction Group
 - System Account Code Default
 - Independent Account Code Default
 - Default Account Code Default
 - Call Progress Tone Detection
 - Digit Modification Assignment
 - Maximum Dialed Digits
 - Route Assignment
 - Route List Assignment (Call)
 - Day and Time Zone Assignment
 - Route Plan Assignment (Call)
 - ARS Digits Dialed Assignment
 - ARS Leading Digits Assignment
 - Interconnect Restriction
 - Node Identity Assignment
 - Automatic Call Distribution (ACD)
 - Call Handling
 - Emergency Services Management
 - Telephone Management
 - Telephone Directory Management
 - Property Management System
 - Voice Mail

Trunk Group Assignment

Trunk Group Number	Hunt Mode	Trunk Group Busy RAD	Maximum Network Hop	Comments
1	Terminal			Copper
2	Terminal			PRI
3	Terminal			QSIG

Page 1 of 3

Go to: value: Go

Trunk Group Members

Member	Trunk Number
1	1201
2	1202
3	1203
4	1204
5	1205
6	1206
7	1207
8	1208
9	1209
10	1210

MITEL 3300 ICP

About System Administration

Step 9: Add Route Assignment

Use the following path to add Route Assignment:

System Administration / Automatic Route Selection / Route Assignment.

Alarm Status: ! Major 2007-Apr-25 14:50:31

Audiocode - System Message:

Selection:

System Administration

- System Administration
 - System Audio Files Update
 - User Authorization Profiles
 - System Options
 - Automatic Route Selection ()
 - Trunk Service Assignmer
 - Trunk Group Assignment
 - Class of Restriction Grou
 - System Account Code De
 - Independent Account Coc
 - Default Account Code De
 - Call Progress Tone Detec
 - Digit Modification Assignn
 - Maximum Dialed Digits
 - Route Assignment**
 - Route List Assignment (C
 - Day and Time Zone Assig
 - Route Plan Assignment (t
 - ARS Digits Dialed Assign
 - ARS Leading Digits Assig
 - Interconnect Restriction
 - Node Identity Assignment
 - Automatic Call Distribution ()
 - Call Handling
 - Emergency Services Manag
 - Telephone Management
 - Telephone Directory Manage
 - Property Management Syste
 - Voice Mail

Change Change Page Change All Clear

Previous Page 1 of 14 Next Go to: value: Go

Route Number	Routing Medium	Trunk Group Number	SIP Peer Profile	COR Group Number	Digit Modification Number	Digits Before Outpulsing	Route Type	Compression
1	TDM Trunk Group	1		1	1			Off
2	TDM Trunk Group	1		2	1			Off
3	TDM Trunk Group	1		3	1			Off
4	TDM Trunk Group	1		4	1			Off
5	TDM Trunk Group	1		5	1			Off
6	TDM Trunk Group	2		1	2			Off
7	TDM Trunk Group	2		2	2			Off
8	TDM Trunk Group	2		3	2			Off
9	TDM Trunk Group	2		4	2			Off
10	TDM Trunk Group	2		5	2			Off
11	TDM Trunk Group	3		1	3	5		Off
12				1	1			Off
13				1	1			Off
14				1	1			Off
15				1	1			Off

MITEL 3300 ICP
About System Administration

-- Web Page Dialog

Route Assignment

Route Number: 11

Routing Medium: TDM Trunk Group

Trunk Group Number: 3

SIP Peer Profile:

COR Group Number: 1

Digit Modification Number: 3

Digits Before Outpulsing: 5

Route Type:

Compression: ☒ Off ☐ On

Save Cancel

Step 10: Add ARS Digits Dialed Assignment

Use the following path to add ARS Digits Dialed Assignment:

System Administration / Automatic Route Selection / ARS Digits Dialed Assignment.

Alarm Status: ! Major 2007-Apr-25 14:50:31

Audiocode - System Message:

Selection:

System Administration

- System Administration
 - System Audio Files Update
 - User Authorization Profiles
 - System Options
 - Automatic Route Selection ()
 - Trunk Service Assignment
 - Trunk Group Assignment
 - Class of Restriction Group
 - System Account Code De
 - Independent Account Coc
 - Default Account Code De
 - Call Progress Tone Deter
 - Digit Modification Assignn
 - Maximum Dialed Digits
 - Route Assignment
 - Route List Assignment (C
 - Day and Time Zone Assig
 - Route Plan Assignment (t
 - ARS Digits Dialed Assign
 - ARS Leading Digits Assig
 - Interconnect Restriction
 - Node Identity Assignment
 - Automatic Call Distribution ()
 - Call Handling
 - Emergency Services Manag
 - Telephone Management
 - Telephone Directory Manage
 - Property Management Syste
 - Voice Mail

ARS Digits Dialed Assignment

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
269	1	Route	11
90	Unknown	List	1
9011	Unknown	List	1
912	9	List	2
91201	7	List	3
91203	7	List	3
91212540	4	Route	20
91212550	4	Route	20
91212970	4	Route	20
91212976	4	Route	20
913	9	List	2
91347	7	List	4
914	9	List	2
91518	7	List	4
91518970	4	Route	20
91518976	4	Route	20
916	9	List	2
91609	7	List	3
91631	7	List	4
91646	7	List	4

MITEL 3300 ICP
About System Administration

Range Programming -- Web Page Dialog

Change Range Programming - ARS Digits Dialed Assignment
Help

This form allows you to change one or more records, starting at the following record:

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
269	1	Route	11

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Digits Dialed:	Change to	269	
Number of Digits to Follow:	Change to	1	-
Termination Type:	Change to	Route	-
Termination Number:	Change to	11	

Preview
Save
Cancel

Step 11: Option Assignment

In System Option Assignment, change the following fields:

- Route Optimization Network ID: change to any unique number.
- DPNSS/QSIG Diversion Enabled: change to 'Yes'.

The screenshot displays the Audiocode - Mitel Networks 3300 Integrated Communications Platform (ICP) web interface. The interface includes a top navigation bar with buttons for Print, Import, Export, Data Refresh, Help, and Exit. Below this, there is a status bar showing 'Alarm Status: Major 2007-Apr-25 14:50:31'. The main content area is titled 'Audiocode - System Message: Selection:' and features a left-hand navigation menu with a list of configuration options. The main content area displays a list of system parameters and their current values.

Parameter	Value
Last Number Redial Source:	All Trunks
Route Optimization Establishment Timer:	10
Route Optimization Attempts:	3
Route Optimization Network Id:	4321
Route Optimization Trailing Digits:	2
Dialed Number Editing For Trunks:	Yes
Number Of Forward Hops:	4
Multiline Set Display 24 Hour Format:	No
Att Cancel-All Feature Access:	Both
ACD 2000-Auto Logout Last Agent On No Answer:	No
Call Forwarding Always - Line Status Indicator ON:	Yes
Night Answer Prompt for Network Configuration:	No
DISA Number Lock-Out Timer:	15
DISA Failed Attempts before Lock-Out:	3
Speed Call Pause Duration:	3
Set Registration Access Code:	***
Set Replacement Access Code:	###
Set Registration Security:	
Resource Tuning Threshold:	0
DPNSS/QSIG Diversion Enabled:	Yes
BLF - Busy Indication based on set enabled:	No
ACD Make Busy Walk Away Codes:	No
ACD Real Time Events Feature Level:	0
Advice of Charge Feature Active:	No
Advice of Charge - Surcharge:	0
Advice of Charge - Multiplier:	0
Campan Repetitive Tone Timer:	0
Conference/Call Intrusion Repetitive Tone Timer:	0
Feature Active Dial Tone - Call Forwarding:	No
Outgoing External Call Prefix For Applications:	
Call History - Disable Record Generation:	No
Default Language:	English
Site Preference for Hot Desk Device:	5020 IP
Email Server:	
Sender's E-mail Address:	
Voice Encryption Enabled:	Yes
Call History - Default Call History Records:	20
System Data Synchronization:	No
Alpha Tagging Enabled:	No
ACD Make Busy On Login Reason Code:	0
Remote Help Server:	

5.1. TLS Setup

- N/A.

5.2. Fail-Over Configuration

- N/A.

5.3. Tested Phones

- Mitel SuperSet 4025

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	Setup TLS between gateway/IP-PBX and Exchange UM. Windows Certificate Authority (CA).		
10a	Dial the pilot number and logon to a user's mailbox. Confirm UM answers the call and confirm UM responds to DTMF input.	P	
10b	Dial a user extension and leave a voicemail. Confirm the user receives the voicemail.	P	
10c	Send a test FAX message to user extension. Confirm the FAX is received in the user's inbox.	P	
11	Setup G.723.1 on the gateway. (If already using G.723.1, setup G.711 A Law or G.711 Mu Law for this step). Dial the pilot number and confirm the UM system answers the call.	P	
12	Setup Message Waiting Indicator (MWI). Geomant offers a third party solution: MWI 2007. Installation files and product documentation can be found on Geomant's MWI 2007 website .	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	None
Phone type (if phone-specific)	
Call scenarios(s) associated with failure point	
List of UM features affected by failure point	
Additional Comments	

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:

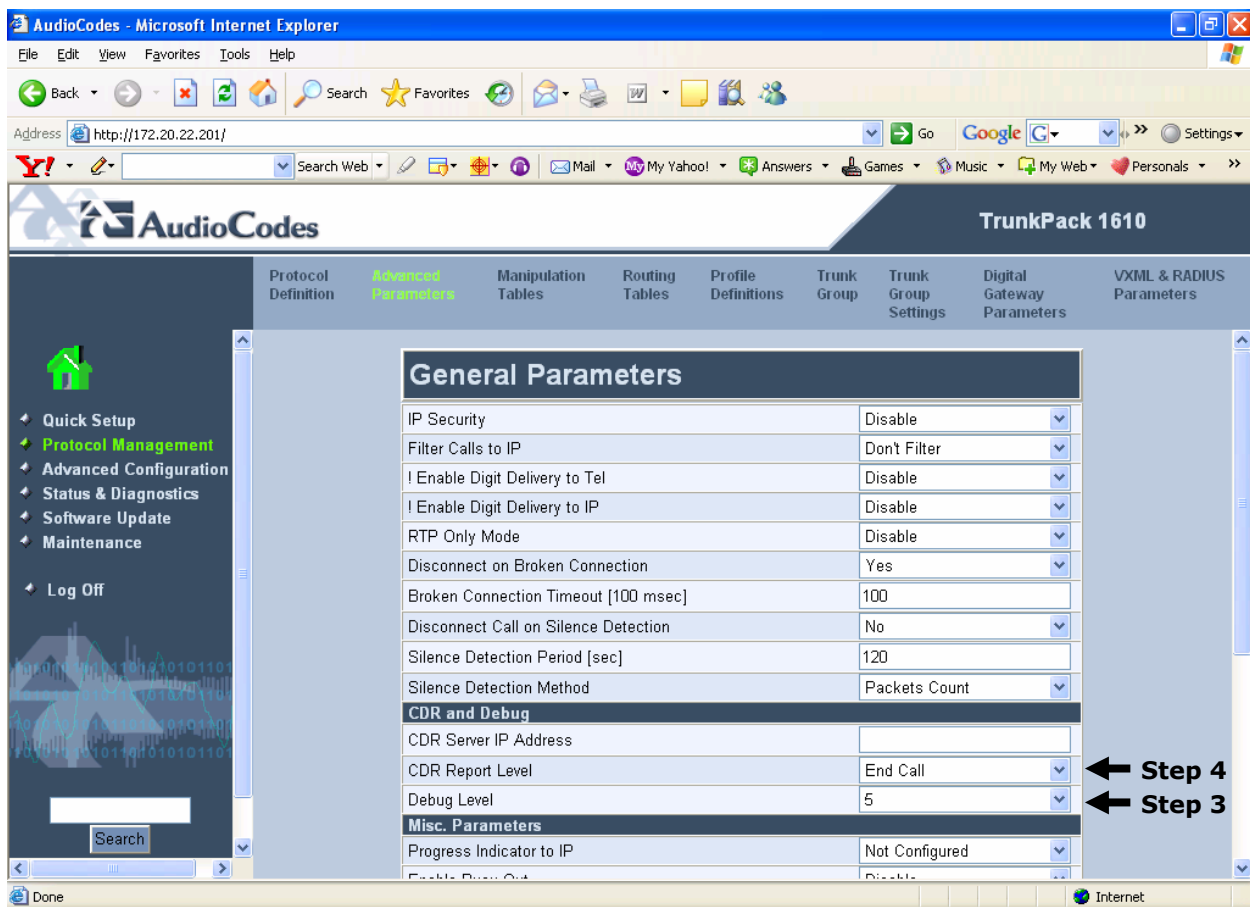
1. Set the parameter **Enable Syslog** to 'Enable'.
2. Use the parameter **Syslog Server IP Address** to define the IP address of the Syslog server you use.

The screenshot shows the AudioCodes TrunkPack 1610 MG Module 1 Management Settings page. The browser window is titled 'AudioCodes - Microsoft Internet Explorer' and the address bar shows 'http://10.15.4.19/'. 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 'Management Settings' and contains two sections: 'Syslog Settings' and 'SNMP Settings'. The 'Syslog Settings' section includes fields for 'Syslog Server IP Address' (10.15.2.5), 'Syslog Server Port' (514), and 'Enable Syslog' (a dropdown menu set to 'Enable'). The 'SNMP Settings' section includes fields for 'SNMP Managers Table', 'SNMP Community String', 'SNMP V3 Table', 'Enable SNMP' (a dropdown menu set to 'Enable'), and 'Trap Manager Host Name'. Below these sections is a table titled 'Activity Types to Report via 'Activity Log' Messages' with columns for the activity type and a checkbox. The activities listed are Parameters Value Change, Auxiliary Files Loading, Device Reset, Flash Memory Burning, and Device Software Update. Arrows from the text 'Step 1' and 'Step 2' point to the 'Enable Syslog' dropdown and the 'Syslog Server IP Address' field, respectively.

Management Settings	
Syslog Settings	
Syslog Server IP Address	10.15.2.5
Syslog Server Port	514
Enable Syslog	Enable
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	☐
Auxiliary Files Loading	☐
Device Reset	☐
Flash Memory Burning	☐
Device Software Update	☐

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.