

Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

Alcatel OmniPCX 4400 with AudioCodes MP-11x

FXO using Analog lines (In-band DTMF)

By : AudioCodes

Updated Since : 2007-02-09

READ THIS BEFORE YOU PROCEED

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Content

This document describes the configuration required to setup Alcatel OmniPCX 4400 and AudioCodes MP-11x FXO using analog lines with inband DTMF as the telephony signaling protocol. It also contains the results of the interoperability testing of Microsoft Exchange 2007 Unified Messaging based on this setup.

Intended Audience

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

Technical Support

The information contained within this document has been provided by Microsoft, its partners or equipment manufacturers and is provided AS IS. This document contains information about how to modify the configuration of your PBX or VoIP gateway. Improper configuration may result in the loss of service of the PBX or gateway. Microsoft is unable to provide support or assistance with the configuration or troubleshooting of components described within. Microsoft recommends readers to engage the service of an Microsoft Exchange 2007 Unified Messaging Specialist or the manufacturers of the equipment(s) described within to assist with the planning and deployment of Exchange Unified Messaging.

Microsoft Exchange 2007 Unified Messaging (UM) Specialists

These are Systems Integrators who have attended technical training on Exchange 2007 Unified Messaging conducted by Microsoft Exchange Engineering Team. For contact information, visit [here](#).

Version Information

Date of Modification	Details of Modification
21 March 2007	Version 1

1. Components Information

1.1. PBX or IP-PBX

PBX Vendor	Alacatel
Model	OmniPCX 4400
Software Version	R4.2-d2.304-4-h-il-c6s2
Telephony Signaling	Analog In-band DTMF Tones
Additional Notes	None

1.2. VoIP Gateway

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

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.

2.2. PBX Prerequisites

The PBX hardware must be installed with an Analog Line Interface Card (Z12, Z24, Z32).

2.3. Cabling Requirements

This integration uses standard RJ-11 telephone line cords.

3. Summary and Limitations

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

Alcatel OmniPCX 4400 PBX has several limitations when activating the voice mail functionality. The list below describes these PBX's limitations:

- The PBX doesn't support the sending of the calling user number for direct calls (i.e., user calls UM to retrieve voice message). Therefore, when the user dials directly to the Microsoft Unified Messaging, the user hears the general welcome prompt: "Welcome, you are connected to Microsoft Exchange, to access your mailbox, enter your extension.", at which the user is required to enter the user's extension number in addition to the pin number.
- When performing blind transfer to an invalid number, the PBX doesn't support invalid number notification and the call is routed back to the original transfer user. When an invalid extension number is defined in the Microsoft Unified Messaging for a particular user, and call transfer by Directory Search to this user is requested, the user that requests this transfer is routed back to the Microsoft Unified Messaging welcome prompt.

4. Gateway Setup Notes

Step 1: SIP Environment Setup

AudioCodes - Microsoft Internet Explorer

Address: http://172.20.22.200/

AudioCodes MP-118 FXO

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

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

General

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

Step 2: Routing Setup

The screenshot shows the AudioCodes MP-118 FXO web interface. The browser window is titled 'AudioCodes - Microsoft Internet Explorer' and the address bar shows 'http://172.20.22.200/'. The page has a navigation menu on the left with options like 'Quick Setup', 'Protocol Management', 'Advanced Configuration', 'Status & Diagnostics', 'Software Update', 'Save Configuration', 'Reset Device', and 'Log Off'. The main content area is titled 'Proxy & Registration' and contains a table of configuration parameters. Two black arrows point to the 'I Enable Proxy' dropdown menu and the 'I Proxy IP Address' text field.

Proxy & Registration	
I Enable Proxy	Use Proxy
Proxy Name	
I Proxy IP Address	172.20.22.211
Gateway Name	
Gateway Registration Name	
First Redundant Proxy IP Address	0.0.0.0
Second Redundant Proxy IP Address	0.0.0.0
Third Redundant Proxy IP Address	0.0.0.0
Enable SRV Queries	Disable
Enable Proxy SRV Queries	Disable
Redundancy Mode	Parking
I Enable Registration	Disable
Registrar Name	
Registrar IP Address	
Registration Time	180

Note: The Proxy IP Address must be one that corresponds to the network environment in which the Microsoft Unified Messaging server is installed (For example, 172.20.22.211 or the FQDN of the Microsoft Unified Messaging host).

Step 3: SIP Environment Setup (Cont.)

AudioCodes - Microsoft Internet Explorer

Address: http://172.20.22.200/

AudioCodes MP-118 FXO

Navigation Menu:

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

Protocol Definition Tab Settings:

Setting	Value
Re-registration Timing [%]	50
Registration Retry Time	30
Subscription Mode	Per Gateway
I Enable Proxy Keep Alive	Disable
Proxy Keep Alive Time	60
Use Gateway Name for OPTIONS	No
Enable Fallback to Routing Table	Disable
Prefer Routing Table	No
Use Routing Table for Host Names and Profiles	Disable
Always Use Proxy	Disable
Send All Invite to Proxy	No
Enable Proxy Hot-Swap	Disable
Number of RTX Before Hot-Swap	3
User Name	
Password
Cnonce	Default_Cnonce
I Authentication Mode	Per Endpoint

SIP

Done Internet

Step 4: Coder Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Home Search Favorites RSS Print Mail New Tab

Address http://172.20.22.200/ Go Links

Google Go Bookmarks 6 blocked Check AutoLink AutoFill Send to Settings

AudioCodes MP-118 FXO

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

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

Coders

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

Submit

SIP

Done Internet

Step 5: Digit Collection Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

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

Address http://172.20.22.200/ Go

AudioCodes MP-114 FXO

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

DTMF & Dialing

Max Digits In Phone Num	15
Inter Digit Timeout for Overlap Dialing [sec]	2
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option;	Not Supported
2nd Tx DTMF Option;	Not Supported
3rd Tx DTMF Option;	Not Supported
4th Tx DTMF Option;	Not Supported
5th Tx DTMF Option;	Not Supported
RFC 2833 Payload Type	101
Digit Mapping Rules	
Dial Tone Duration [sec]	16
Hotline Dial Tone Duration [sec]	3
Enable Special Digits	Enable
Hook-Flash Option	Not Supported
Default Destination Number	1000

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

Search

Done Internet

Step 6: Message Waiting Indication Setup

The screenshot shows the AudioCodes MP-118 FXO web interface in a Microsoft Internet Explorer browser. The address bar shows <http://172.20.22.200/>. The interface has a top navigation bar with the AudioCodes logo and the title 'MP-118 FXO'. Below this is a menu bar with the following items: Protocol Definition, **Advanced Parameters**, Manipulation Tables, Routing Tables, Profile Definitions, Endpoint Phone Numbers, Hunt Group Settings, Endpoint Settings, and FXO Settings. On the left side, there is a sidebar with a list of links: Quick Setup, **Protocol Management**, Advanced Configuration, Status & Diagnostics, Software Update, Save Configuration, Reset Device, and Log Off. The main content area is titled 'Supplementary Services' and contains a table of configuration options. A black arrow points to the 'I Enable MWI' option in the 'MWI Parameters' section.

Supplementary Services	
I Enable Hold	Enable
Hold Format	0.0.0.0
I Enable Transfer	Enable
Transfer Prefix	
I Enable Call Forward	Enable
I Enable Call Waiting	Enable
Number of Call Waiting Indications	2
Time Between Call Waiting Indications	10
Time Before Waiting Indication	0
Waiting Beep Duration	300
Enable Caller ID	Disable
Caller ID Type	Bellcore
MWI Parameters	
I Enable MWI	Enable
MWI Analog Lamp	Disable
MWI Display	Disable

Step 7: Manipulation Setup

AudioCodes - Microsoft Internet Explorer

Address: http://172.20.22.200/

AudioCodes MP-118 FXO

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

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

Destination Phone Number Manipulation Table for Tel -> IP Calls

Table Index: 1-10

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

Submit

Step 8: Endpoints Setup

AudioCodes - Microsoft Internet Explorer

Address: http://172.20.22.200/

AudioCodes MP-118 FXO

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

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

Endpoint Phone Number Table

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

Register Un-Register
Submit

SIP

Web Server Internet

Note: The phone numbers must correspond to your network environment as the dial plan pilot number is configured for this PBX in the Microsoft Unified Messaging server (For example, 11111).

Step 9: Hot Line Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

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

Address http://172.20.22.200/ Go Links

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

AudioCodes **MP-118 FXO**

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

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

Automatic Dialing

Gateway Port	Destination Phone Number	Auto Dial Status
Port 1	11111	Hotline
Port 2	11111	Hotline
Port 3		Enable
Port 4		Enable
Port 5		Enable
Port 6		Enable
Port 7		Enable
Port 8		Enable

Submit

Done Internet

Step 10: Voice mail In-Band DTMF Setup

There are two options to setup the voice mail In-Band DTMF:

- If the current Alcatel OnmiPCX 4400 software package supports Caller ID for voice mail extensions (CLIP on VPS SL), follow Step 10 (a).
- If the current Alcatel OnmiPCX 4400 software package does not support Caller ID for voice mail extensions (CLIP on VPS SL), follow Step 10 (b).

To verify whether or not your PBX supports this package, check steps 18 through 21 in Chapter 7.

Step 10 (a): Voice mail In-Band DTMF Setup (PBX software package supports CLIP on VPS SL)

AudioCodes - Microsoft Internet Explorer

Address: http://172.20.22.200

AudioCodes MP-114 FXO

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

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

Voice Mail

General	
Voice Mail Interface	DTMF
Line Transfer Mode	Blind Transfer
Digit Patterns	
Forward on Busy Digit Pattern	A.5R.*S.#
Forward on No Answer Digit Pattern	A.6R.*S.#
Forward on Do Not Disturb Digit Pattern	A.1R.*S.#
Forward on No Reason Digit Pattern	A.2S.#
Internal Call Digit Pattern	A7
External Call Digit Pattern	
Disconnect Call Digit Pattern	
MWI	
MWI Off Digit Pattern	137
MWI On Digit Pattern	136
SMDI	
Enable SMDI	Disable

Step 10 (b): Voice mail In-Band DTMF Setup (PBX software package does not support CLIP on VPS SL)

The screenshot shows a web browser window titled "AudioCodes - Microsoft Internet Explorer" with the address bar displaying "http://172.20.22.200/". The browser's menu bar includes File, Edit, View, Favorites, Tools, and Help. The address bar also shows a search bar with "Google" and a "Go" button. The main content area displays the "AudioCodes" logo and the "MP-118 FXO" title. Below the title is a navigation bar with tabs: Protocol Definition, Advanced Parameters, Manipulation Tables, Routing Tables, Profile Definitions, Endpoint Phone Numbers, Hunt Group Settings, Endpoint Settings, and FXO Settings (highlighted in green). On the left side, there is a sidebar with a list of links: Quick Setup, Protocol Management (highlighted in green), Advanced Configuration, Status & Diagnostics, Software Update, Save Configuration, Reset Device, and Log Off. The main content area is titled "Voice Mail" and contains several sections: General, Digit Patterns, MWI, and SMDI. Each section has a table of settings.

Voice Mail	
General	
Voice Mail Interface	DTMF
Line Transfer Mode	Blind Transfer
Digit Patterns	
Forward on Busy Digit Pattern	A.5R.
Forward on No Answer Digit Pattern	A.6R.
Forward on Do Not Disturb Digit Pattern	A.1R.
Forward on No Reason Digit Pattern	A.2S.
Internal Call Digit Pattern	A7
External Call Digit Pattern	
Disconnect Call Digit Pattern	
MWI	
MWI Off Digit Pattern	137
MWI On Digit Pattern	136
SMDI	
Enable SMDI	Disable
SMDI Timeout [msec]	2000

Step 11: FAX Setup

The screenshot shows the AudioCodes MP-118 FXO configuration page in Microsoft Internet Explorer. The browser window title is "AudioCodes - Microsoft Internet Explorer". The address bar shows "http://172.20.22.200/". The page has a navigation menu with "Network Settings", "Media Settings", "Configuration File", "Regional Settings", "Security Settings", and "Management Settings". The "Media Settings" section is active, showing a sidebar with "Quick Setup", "Protocol Management", "Advanced Configuration", "Status & Diagnostics", "Software Update", "Save Configuration", "Reset Device", 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" dropdown for the "CNG Detector Mode" option.

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

Step 12: FXO General Setup

AudioCodes - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Reload Home Search Favorites RSS Mail Print Wordpad Explorer

Address http://172.20.22.200/ Go Links

Google Go Bookmarks Popups okay Check AutoLink AutoFill Send to Settings

AudioCodes MP-118 FXO

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

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

SIP

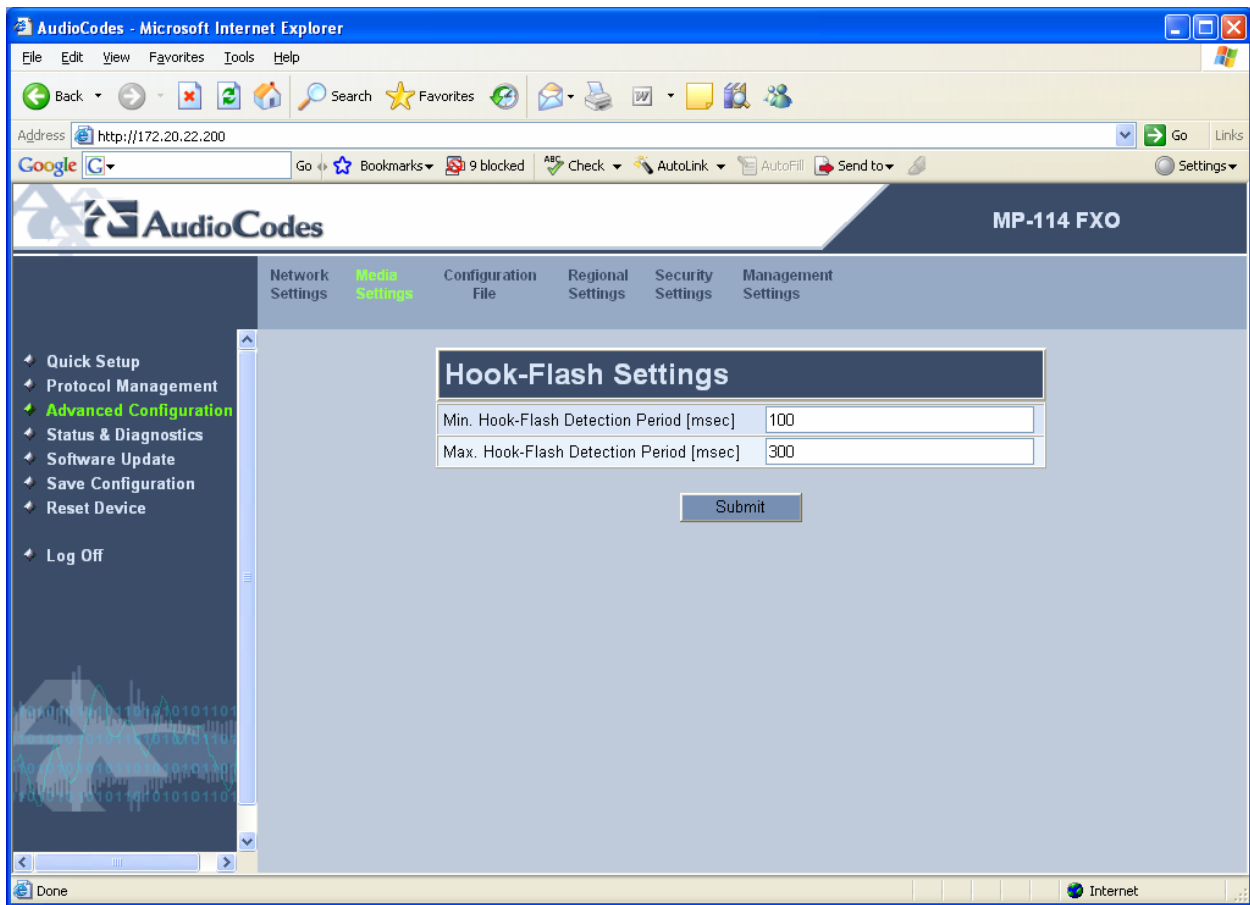
FXO Settings

Dialing Mode	One Stage	←
Waiting for Dial Tone	No	
Time to Wait before Dialing [msec]	1000	
Ring Detection Timeout [sec]	8	
Reorder Tone Duration [sec]	0	
Answer Supervision	Yes	←
Rings before Detecting Caller ID	1	
Send Metering Message to IP	No	
Disconnect on Busy Tone	Yes	
Disconnect On Dial Tone	Disable	
Guard Time Between Calls	1	

Submit

Done Internet

Step 13: FXO General Setup (Cont.)



Step 14: FXO General Setup (Cont.)

General Parameters	
IP Security	Disable
Filter Calls to IP	Don't Filter
! Enable Digit Delivery to Tel	Disable
! Enable Digit Delivery to IP	Disable
RTP Only Mode	Disable
Enable DID Wink	Disable
Disconnect and Answer Supervision	
Enable Polarity Reversal	Disable
Enable Current Disconnect	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	Voice/Energy Detectors
CDR and Debug	
CDR Server IP Address	

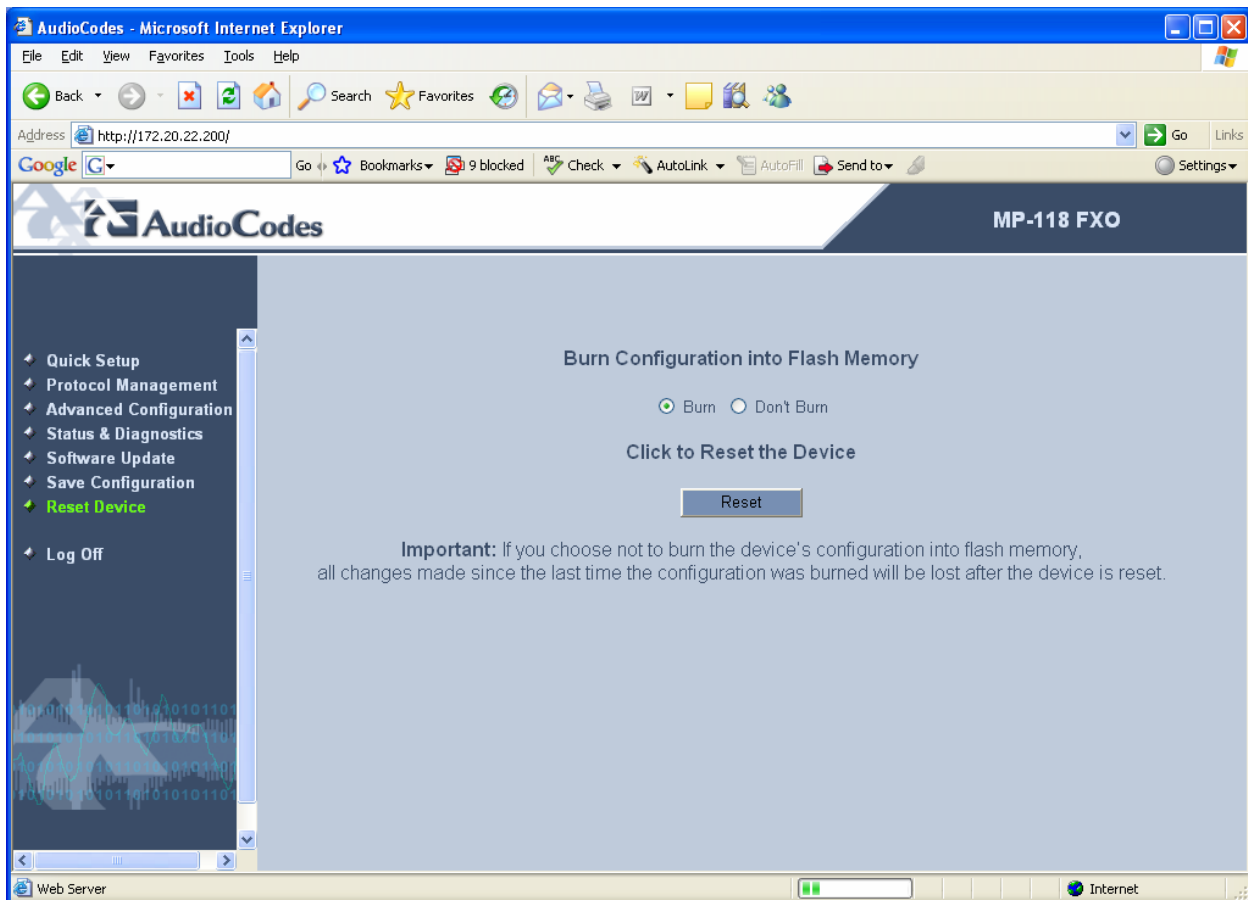
Step 15: FXO General Setup (Cont.)

CallProgressTonesFilename = 'CPT_alcatel_omnipcx.dat'

EnableDetectRemoteMACChange = 2

ECNLPMMode = 1

Step 16: Reset FXO

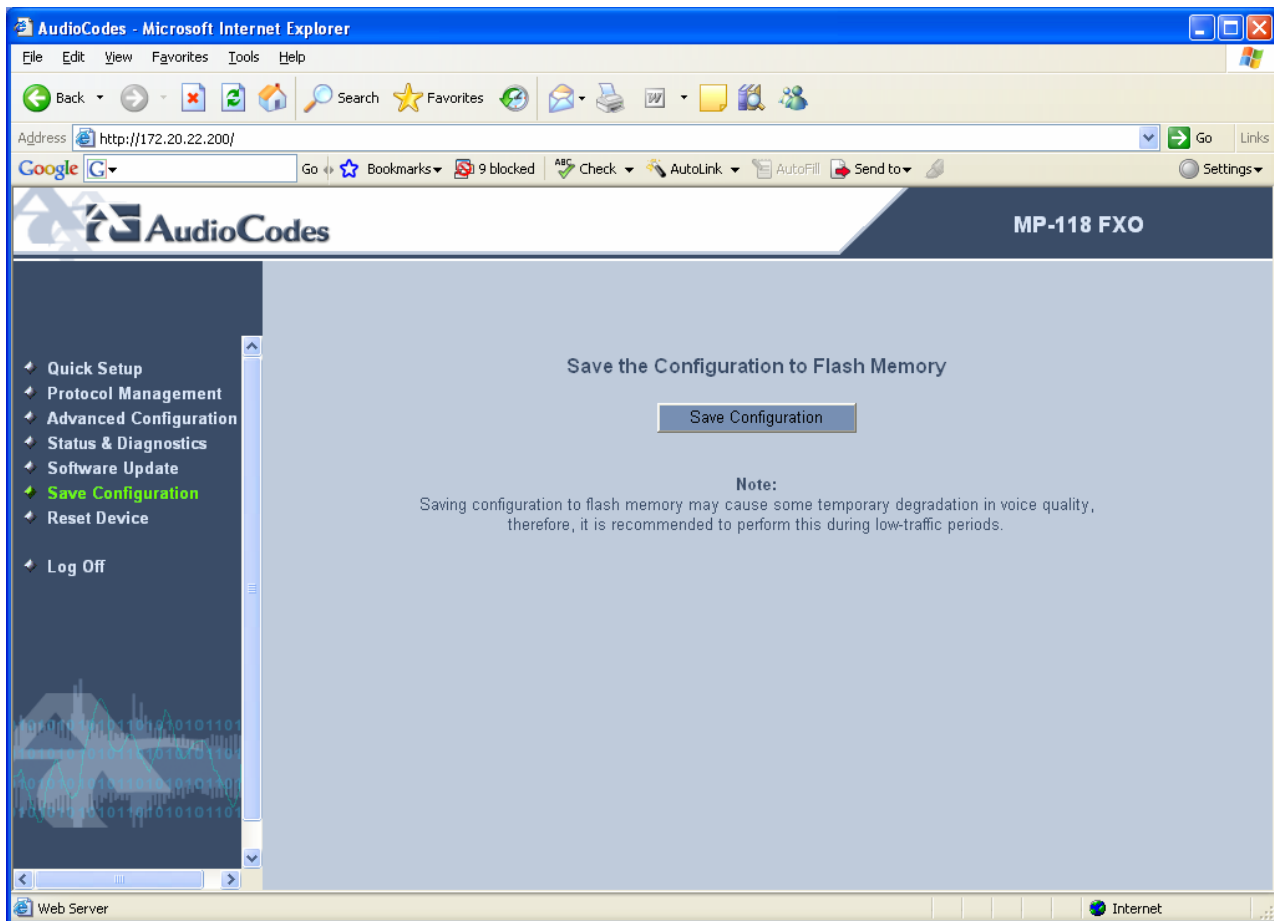


Note: Steps 1 and 7 involve core configuration changes (versus default settings):

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

These changes require a gateway reset (by default, when performing a gateway reset, the configuration is burnt to flash memory). If no change is made to these two core configuration parameters, skip to Step 17.

Step 17: Save Gateway Configuration



Note: This step is optional and is not required if you performed Step 16.

4.1. Configuration Files

The ZIP file includes the following files:

- Audiocodes configuration ini file for PBXs that support Caller ID for voice mail extensions (INI Alcatel OmniPCX4400 FXO DTMF - PBX Caller id enabled.ini).
- Audiocodes configuration ini file for PBXs that do not support Caller ID for voice mail extensions (INI Alcatel OmniPCX4400 FXO DTMF - PBX Caller id disabled.ini).
- Audiocodes Call Progress Tones file for Alcatel OmniPCX 4400 PBX (.dat file extension).



Alcatel OmniPCX4400
FXO DTMF

4.2. TLS Setup

Step 1: PBX to IP Routing Setup

Proxy & Registration	
Enable Proxy	Use Proxy
Proxy Name	exchange2007.com
Proxy IP Address	10.15.3.207
First Redundant Proxy IP Address	0.0.0.0
Second Redundant Proxy IP Address	0.0.0.0
Third Redundant Proxy IP Address	0.0.0.0
Redundancy Mode	Parking
Proxy Load Balancing Method	Disable
Proxy IP List Refresh Time	60
Enable Proxy Keep Alive	Disable
Proxy Keep Alive Time	60
Enable Fallback to Routing Table	Disable
Prefer Routing Table	No
Use Routing Table for Host Names and Profiles	Disable
Always Use Proxy	Disable
Send All Invites to Proxy	No

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

Step 2: SIP Environment and Gateway Name Setup

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

Setting	Value
Send All Invite to Proxy	No
Enable Proxy Hot-Swap	Disable
Enable Registration	Disable
Gateway Name	gw2.fxo.audiocodes.com
Gateway Registration Name	
DNS Query Type	A-Record
Proxy DNS Query Type	A-Record
Subscription Mode	Per Gateway
Use Gateway Name for OPTIONS	No
Number of RTX Before Hot-Swap	3
User Name	
Password
Cnonce	Default_Cnonce
Authentication Mode	Per Endpoint

Buttons: Register, Un-Register, Submit

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

Step 3: SIP Environment Setup (Cont.)

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.6.2/

AudioCodes MP-114 FXO

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

General

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

Arrows pointing to configuration items:

- PRACK Mode
- Channel Select Mode
- Fax Signaling Method
- SIP Transport Type
- SIP UDP Local Port
- SIP TCP Local Port
- SIP TLS Local Port
- SIP Destination Port

Step 4: DNS Servers Setup

AudioCodes - Microsoft Internet Explorer

Address: http://10.15.6.2/

AudioCodes MP-114 FXO

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

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

Search

IP Settings

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

DNS Settings

DNS Primary Server IP	10.1.1.11
DNS Secondary Server IP	10.1.1.10

DHCP Settings

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

NAT Settings

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

Differential Services

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

Done Internet

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

Step 5: Internal DNS Setup

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

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

Submit

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

Step 6: NTP Server Setup

The screenshot shows the AudioCodes MP-114 FXO web interface in Microsoft Internet Explorer. The browser address bar shows <http://10.15.6.2/>. The page title is "AudioCodes" and the model is "MP-114 FXO". The navigation menu includes: Network Settings, Media Settings, Configuration File, Regional Settings, Security Settings, and Management Settings. The left sidebar contains: Quick Setup, Protocol Management, Advanced Configuration (highlighted), Status & Diagnostics, Software Update, Maintenance, and Log Off. The main content area is titled "Application Settings" and contains the following sections:

- NTP Settings**
 - NTP Server IP Address: 10.15.6.50 (indicated by a black arrow)
 - NTP UTC Offset: Hours 0, Minutes 0
 - NTP Update Interval: Hours 24, Minutes 0
- Telnet Settings**
 - I Embedded Telnet Server: Disable
 - I Telnet Server TCP Port: 23
 - I Telnet Server Idle Timeout: 0
- STUN Settings**
 - Enable STUN: Disable
 - STUN Server Primary IP: 0.0.0.0
 - STUN Server Secondary IP: 0.0.0.0
- NFS Settings**
 - NFS Table: -->

A "Submit" button is located at the bottom right of the settings area.

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 it's internal clock).

Step 7: Generate Certificate Setup

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

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

The main content area is titled "Certificate Signing Request". It contains a form with a "Subject Name" field set to `gw2.fxo.audiocodes.com` and a "Generate CSR" button. Below the button, there is a text prompt: "Copy the certificate signing request and send it to your Certification Authority for signing." followed by a block of text representing the CSR:

```
-----BEGIN CERTIFICATE REQUEST-----
MIIBYDCBygIBADAhMR8wHQYDVQDEx2ndzIubTJrLmF1ZG1vY29kZXMuY29tMIGf
MA0GCSqGSIb3DQEBAQUAA4GNADCB1QKBgQCw7x/nSvJJzUOKsSRvYoGS9LguLjGJ
oK4td5mMRE9isdMVy4iogiMEJLG51BSQ5Jta9J8Sd/d7munYt3utpUnIjbbfG5OG
K1br4pR3jG9jot2oWjqhemH34hTouhjc1dJQLY21e12RrOXcL6Qa1AQ1pZPrMnLM
eBFdwSPJ8bGAQwIDAQABoAAwDQYJKoZIhvcNAQEEBQADgYEAHxJZM48IXV+MjiPe
Qnn580m9Xs1ht2S1E075G6O/5su+tZuV10X1ss1hvlorqiOJxWJRgH4bBP+G8g8F
sSLG2Bkq08QghFTT2PXp1CiqqTyI3Ru6Ge4UCOTWtb/DRmUU1qZtMfuHy7DqcmHd
gcdLo3ncyAgVoOMLOy8LPreKqPc=
-----END CERTIFICATE REQUEST-----
```

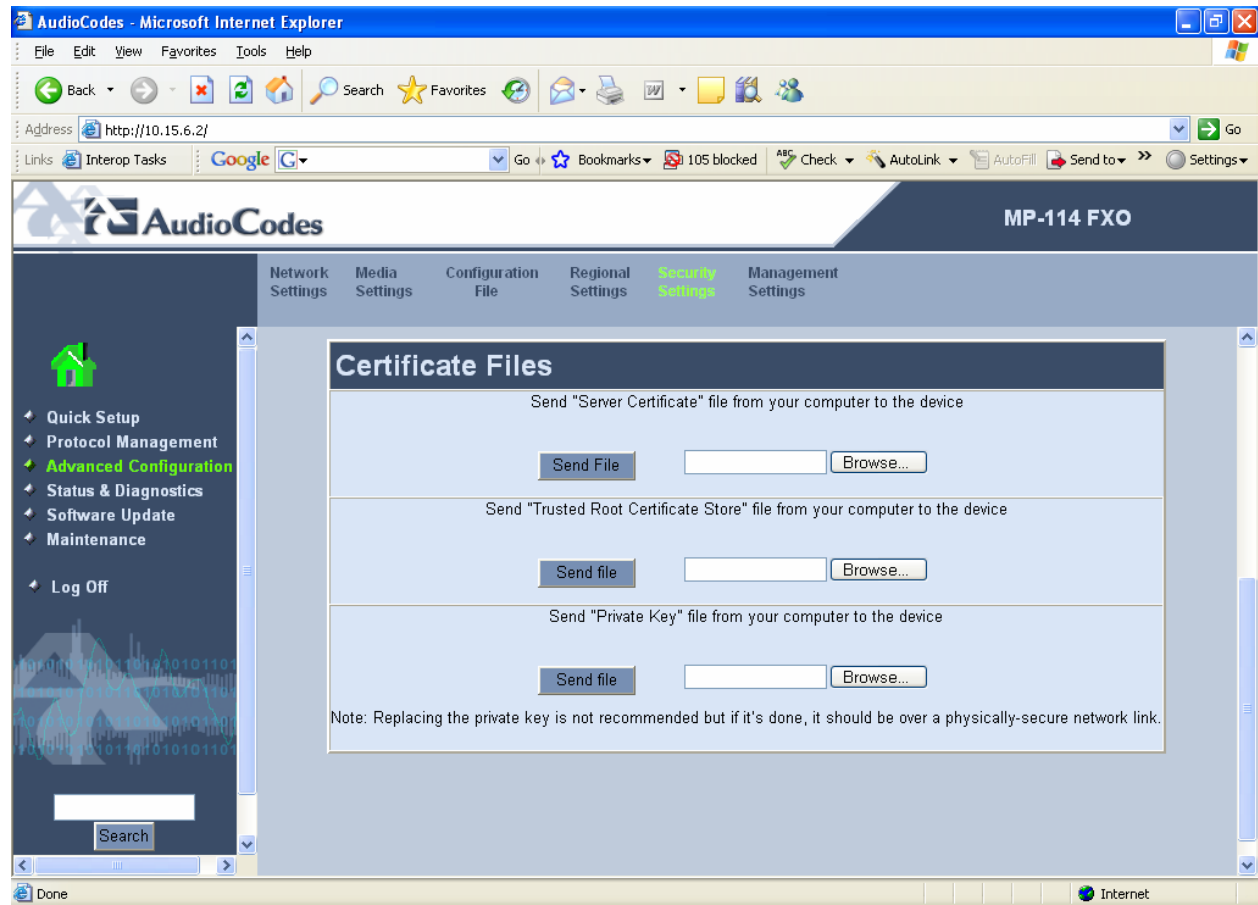
At the bottom of the browser window, the status bar shows "Done" and "Internet".

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.

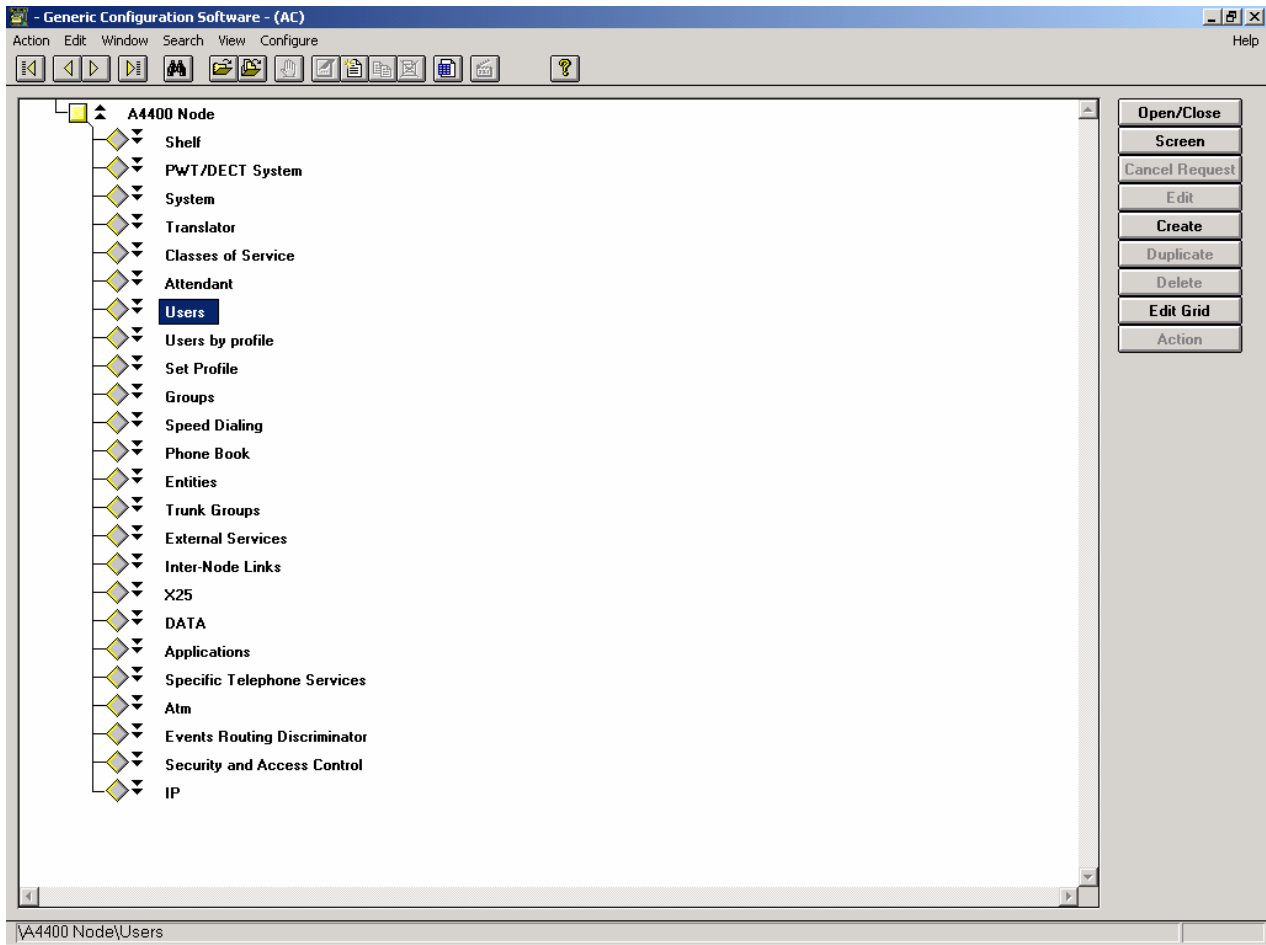
In the "Trusted Root Certificate Store" area, upload the CA certificate.



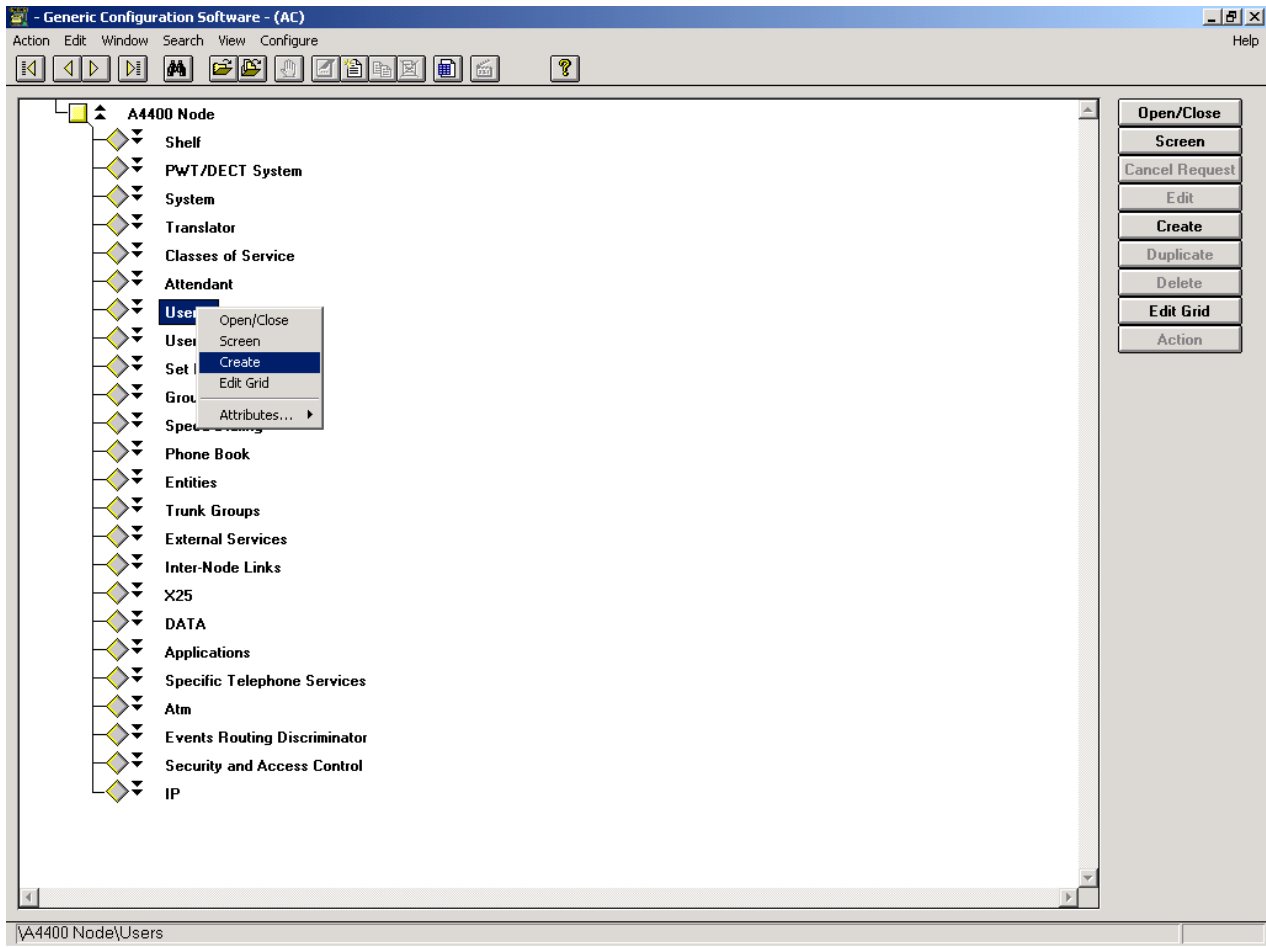
5. PBX Setup Notes

Configuring Voice Mail extensions

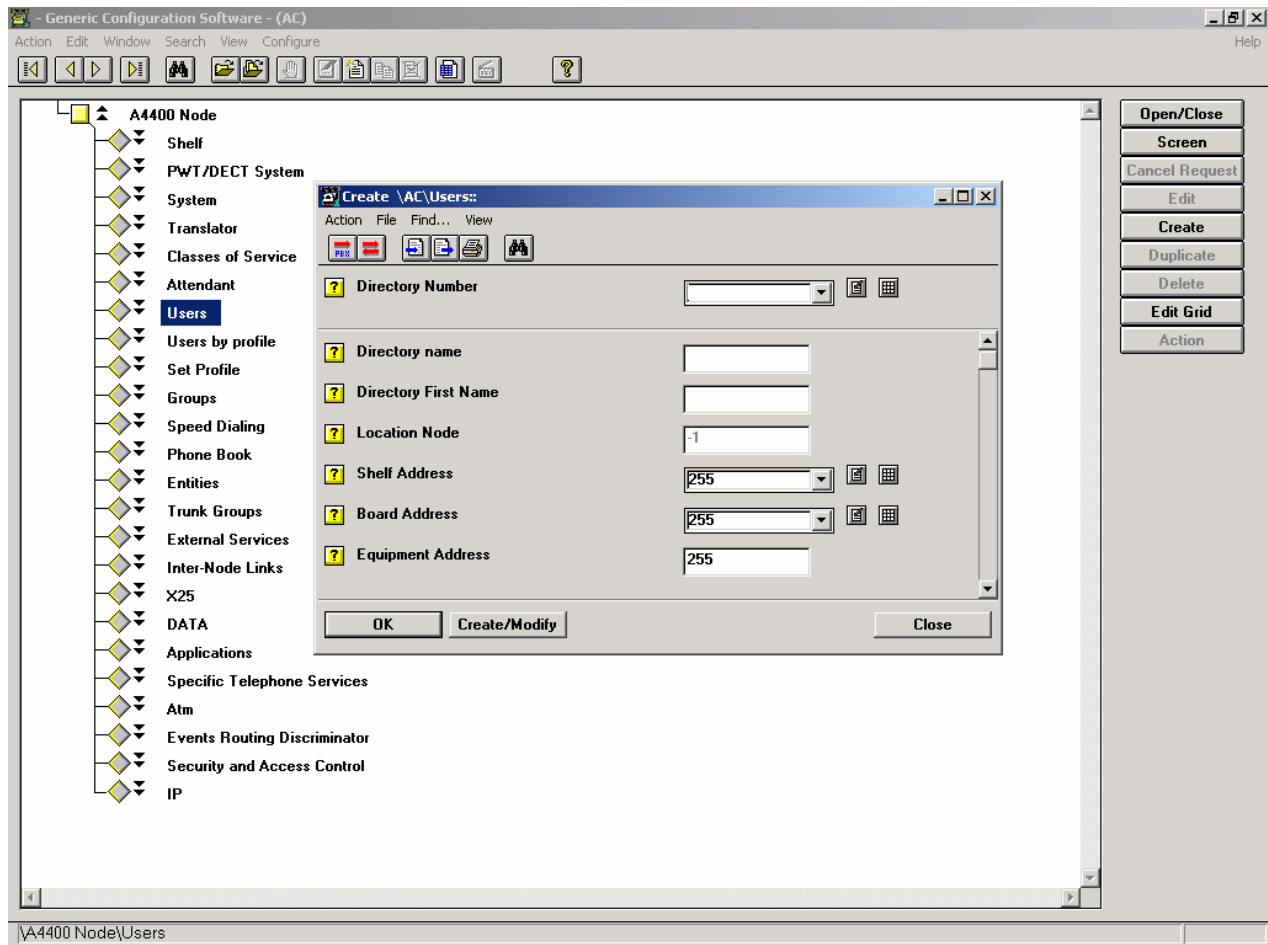
Step 1: Select the Users object.



Step 2: Right-click the Users object, and then from the shortcut menu, choose Create.

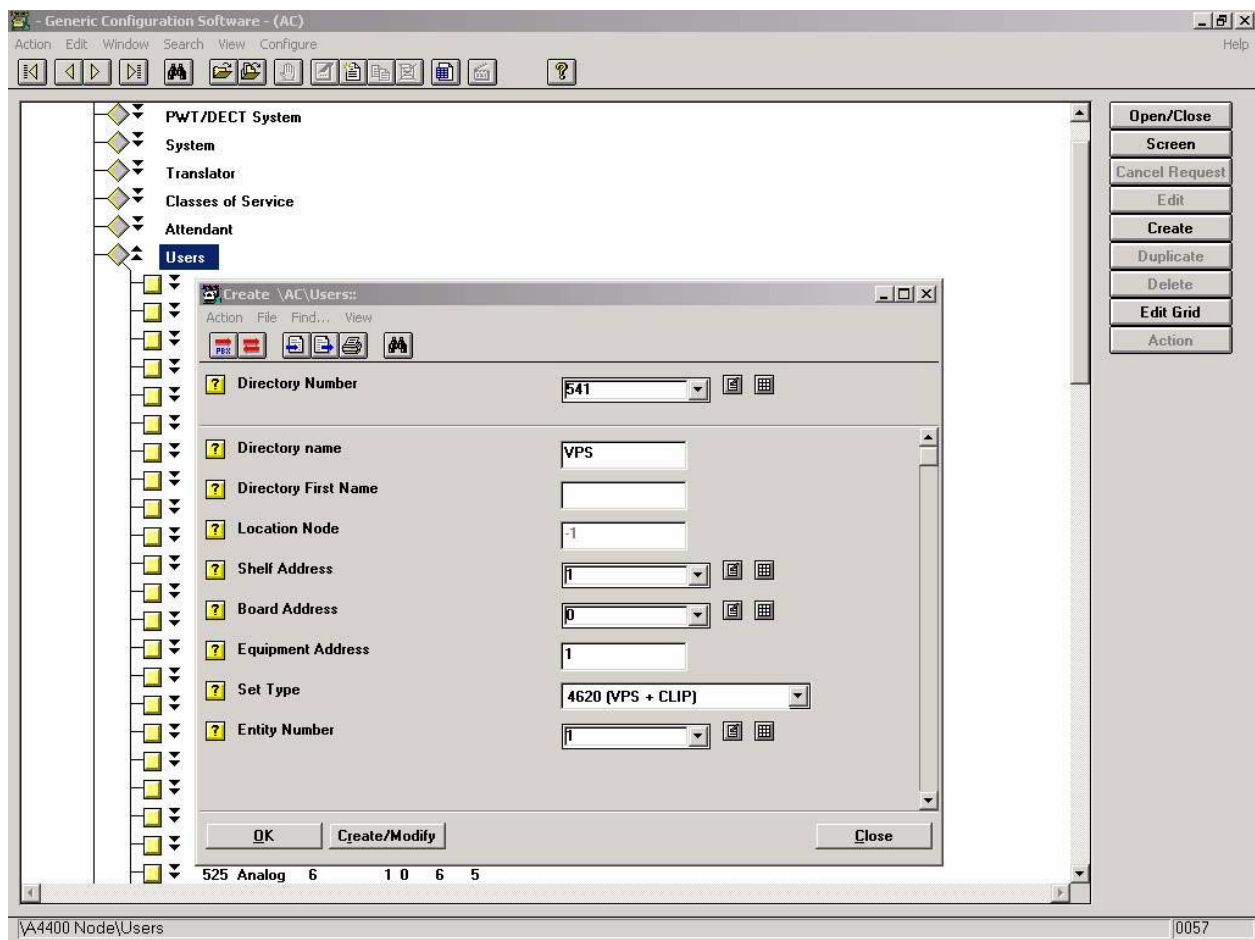


Step 3: The Create window opens.



Step 4: In the Create window, fill in the following fields:

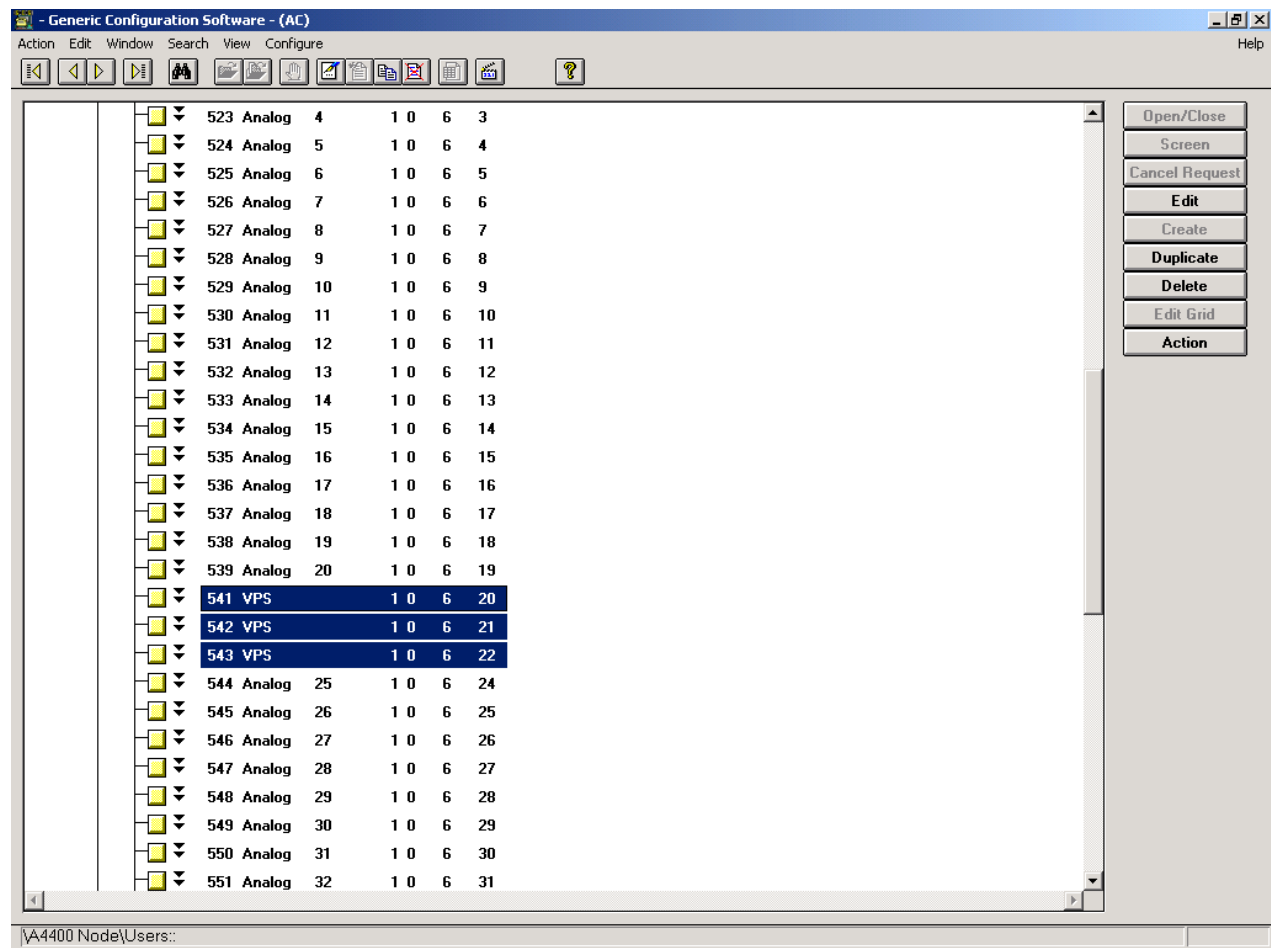
- Directory Number: Free extension number in the PBX, for example, 541,542.
- Directory Name: Unique name for this group in the PBX, for example, VPS.
- Shelf Address, Board Address, and Equipment Address: Fill in the physical location of the extension in the PBX.
- Set Type: must be 4620 (VPS + CLIP).



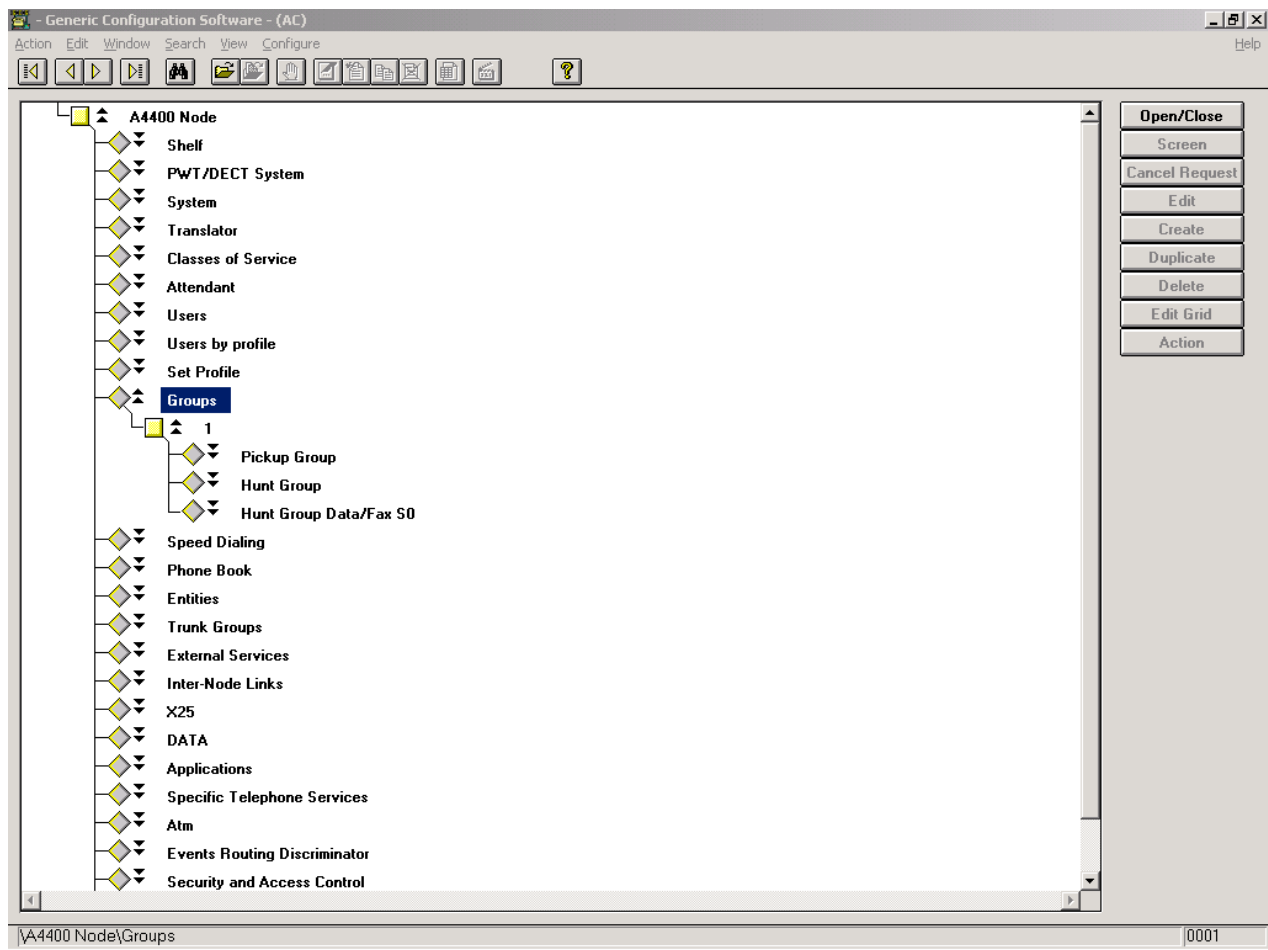
Step 5: Click the **Create/Modify** button.

Step 6: Repeat steps 2 through 5 for as many Voice-Mail access extension that are needed.

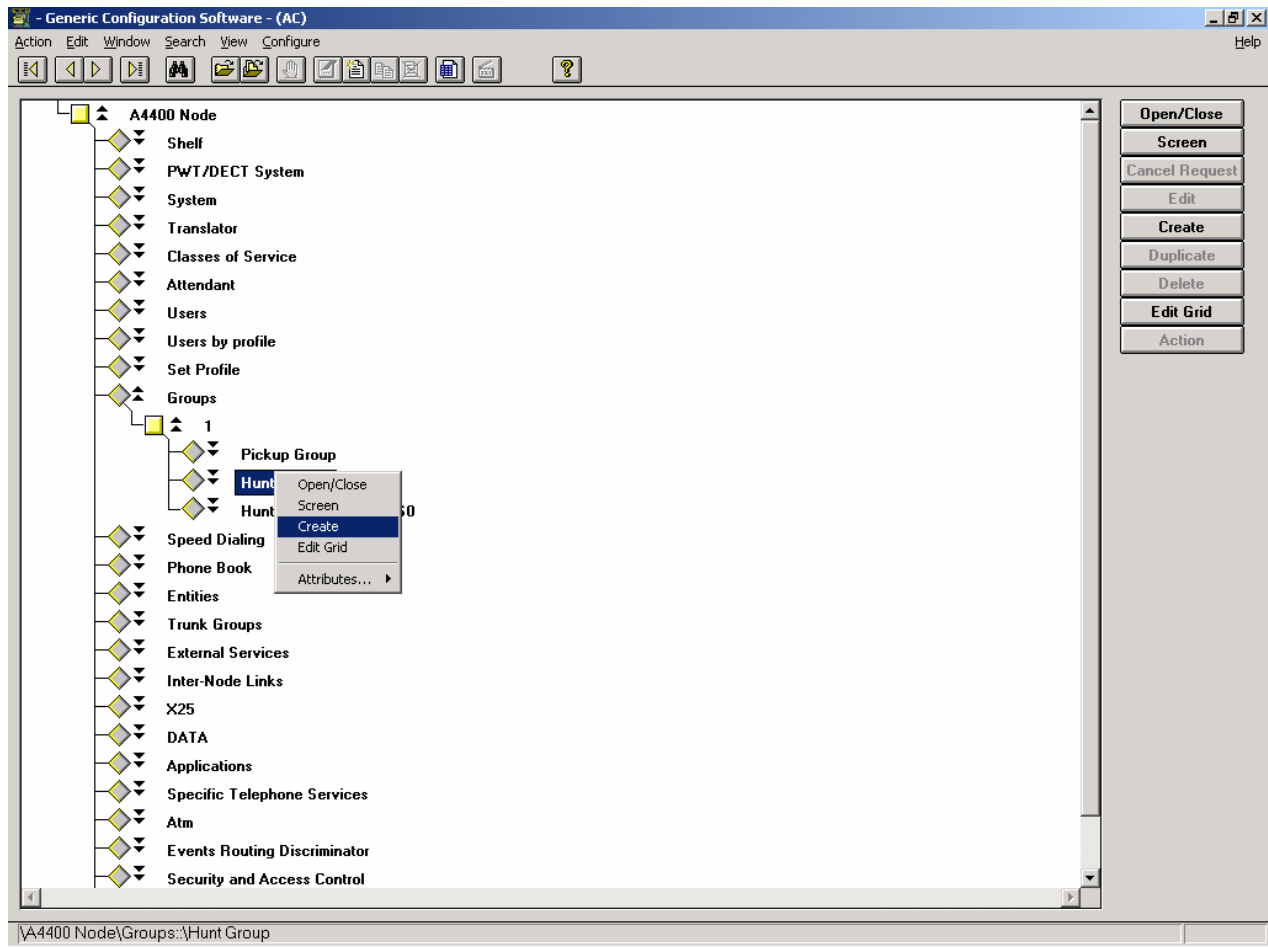
Step 7: After you have finished creating all the required Voice-Mail access extensions, double-click the Users object. The full users/extensions list of the PBX appears. Scroll down the list until you see the users that you defined.



Step 8: In the main menu, double-click the Groups object. A list of the available groups in the PBX are shown.



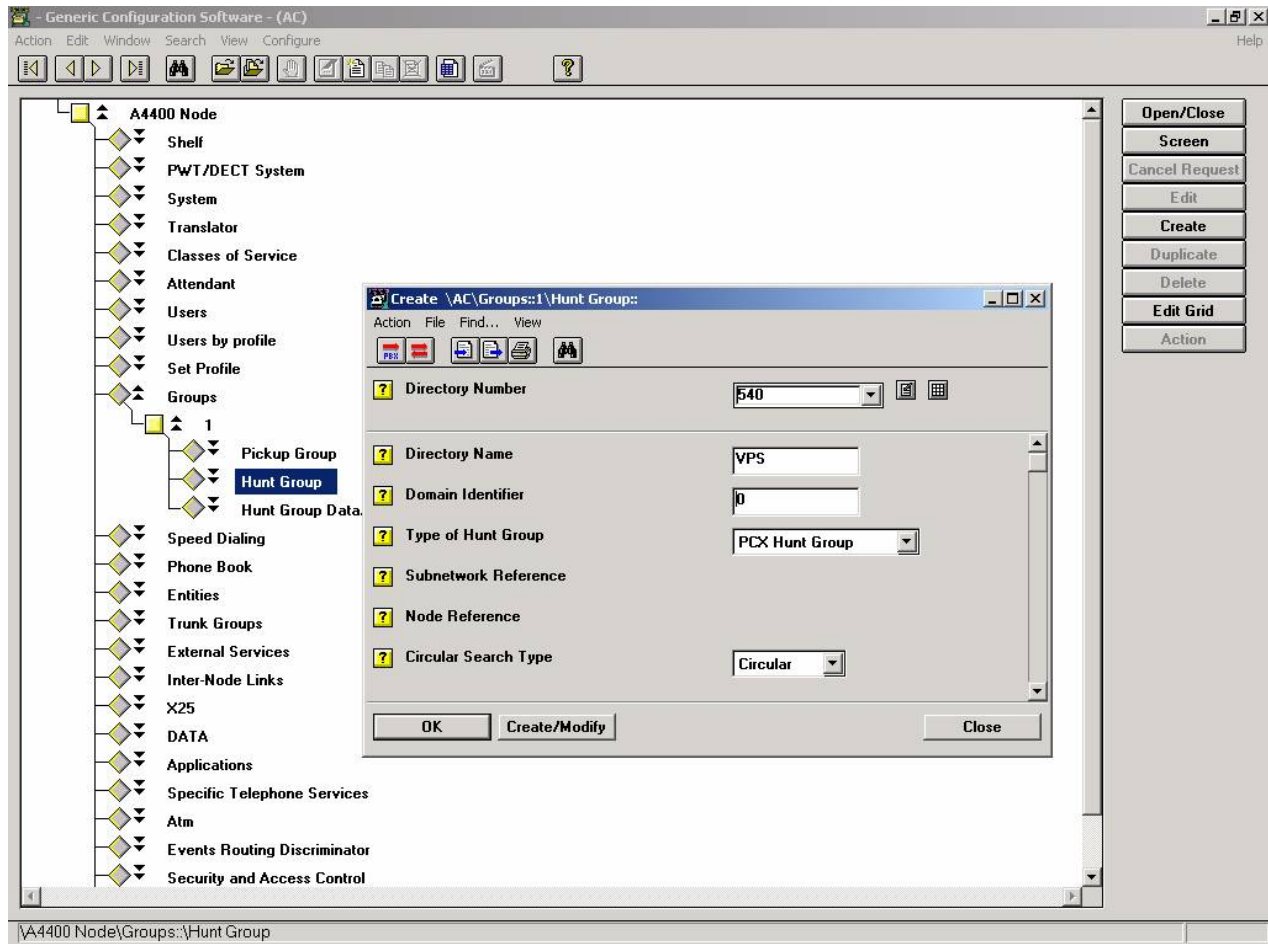
Step 9: Right-click the Hunt Group object, and then from the shortcut menu, choose **Create**.



Step 10: In the opened Create window, enter the following:

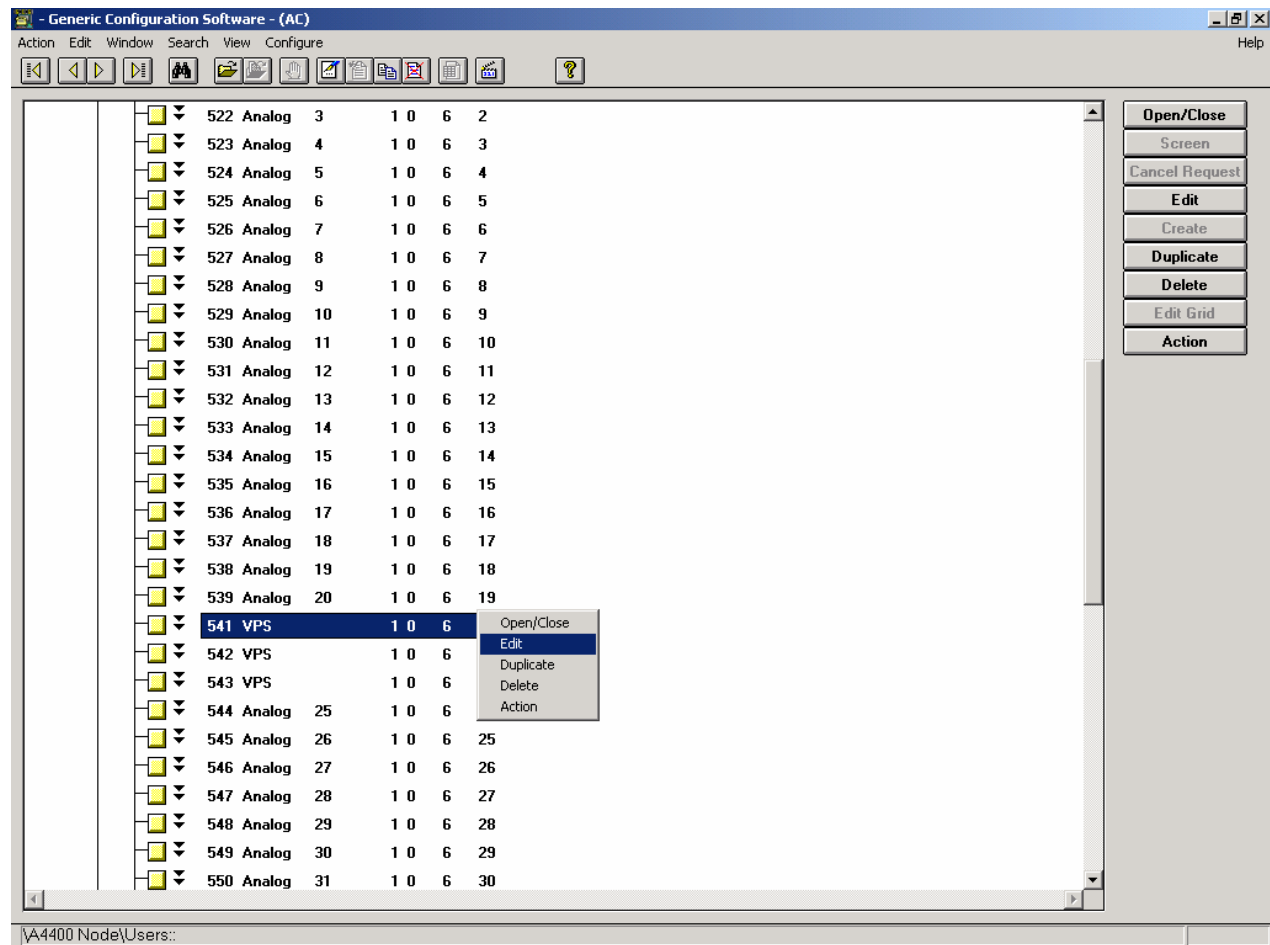
- Directory Number: Unique hunt group number in the PBX, for example, 540.
- Directory Name: Unique hunt group name in the PBX, for example, VPS.

Verify that the Type of Hunt Group is PBX Hunt Group.

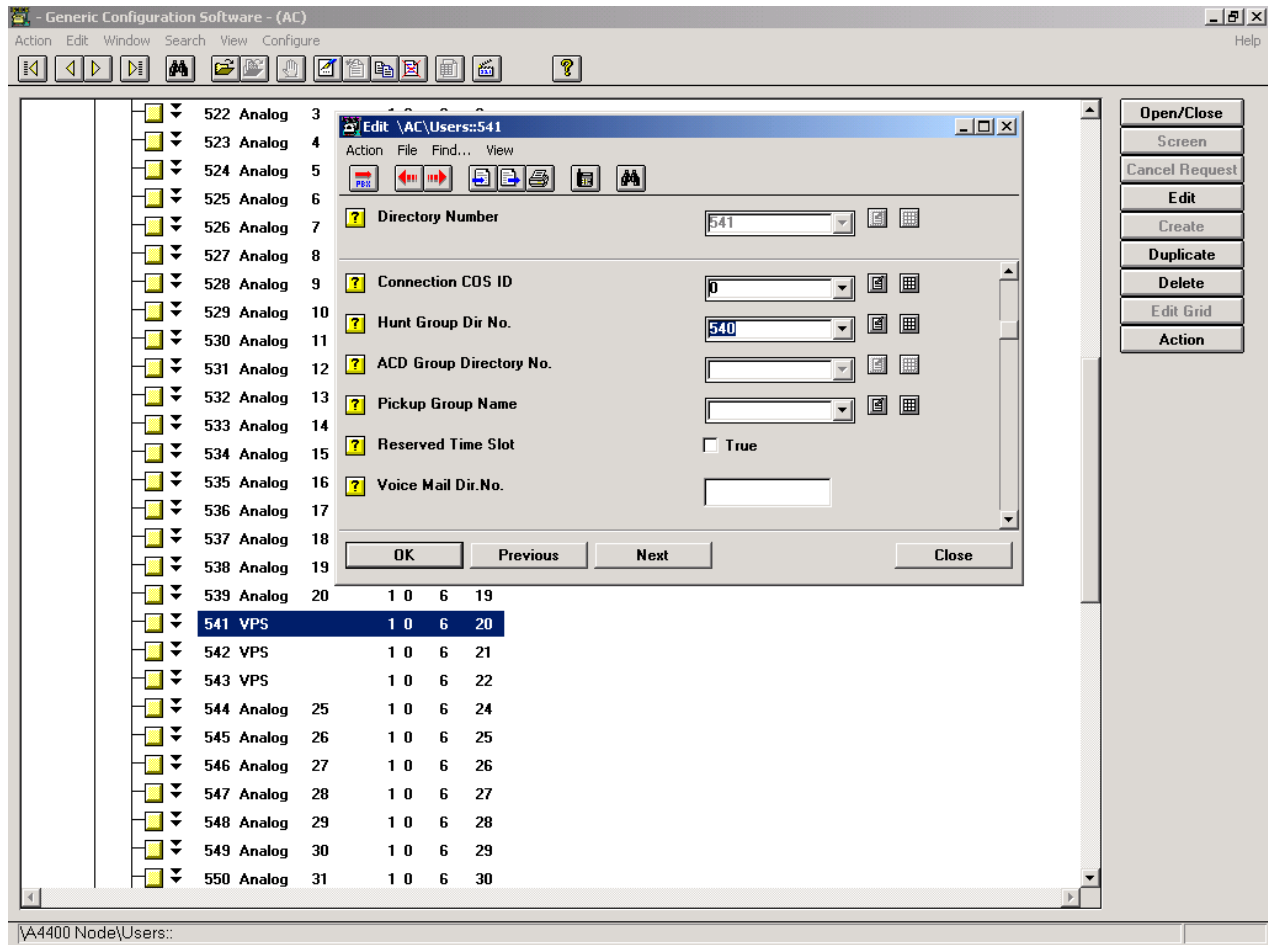


Step 11: When you have completed defining the Hunt Group, click **Create/Modify**.

Step 12: Click the Users object. The full users/extensions list of the PBX appears. Scroll down the list until you see the users that you defined. Access each of the users by right-clicking the user, and then from the shortcut menu, choosing **Edit**.

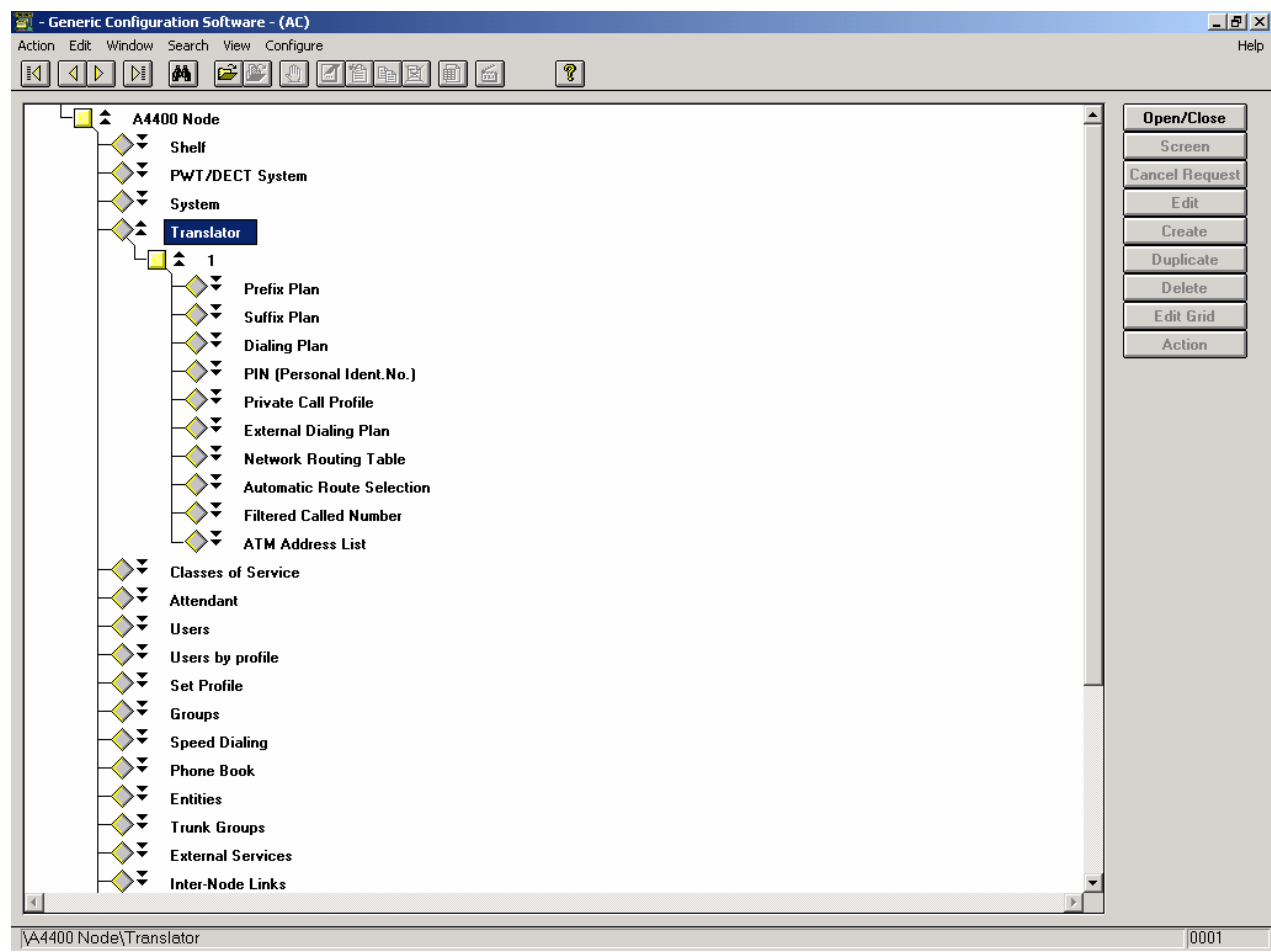


Step 13: In the Edit window, scroll down until the Hunt Group Dir No. field is visible. In this field, enter the Directory Number of the hunt group that you defined in Step 10, for example, 540. Click **OK** to apply the settings. Repeat this step for each voice mail user you defined in steps 2 through 5.

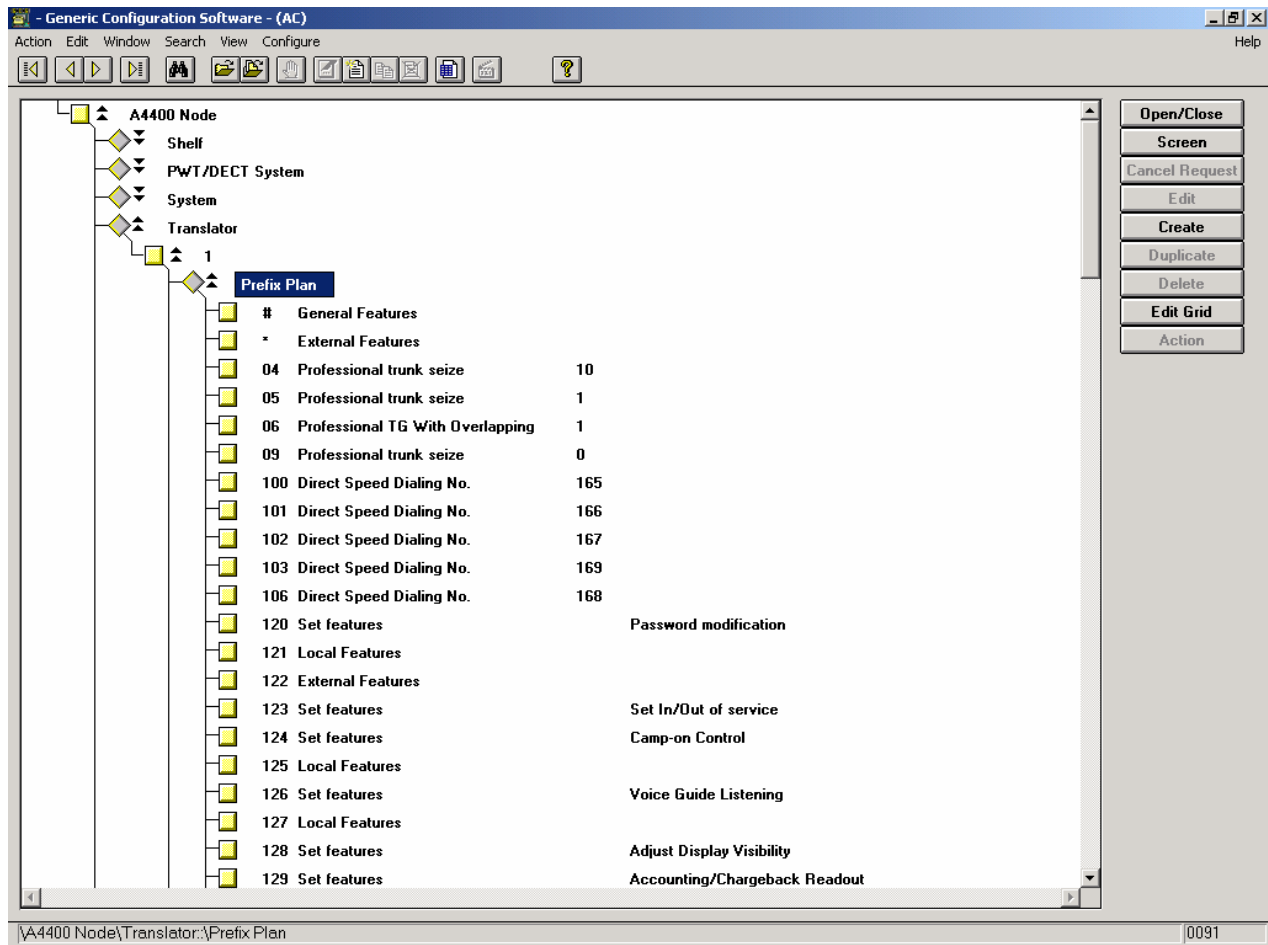


Configuring Message Waiting Indication

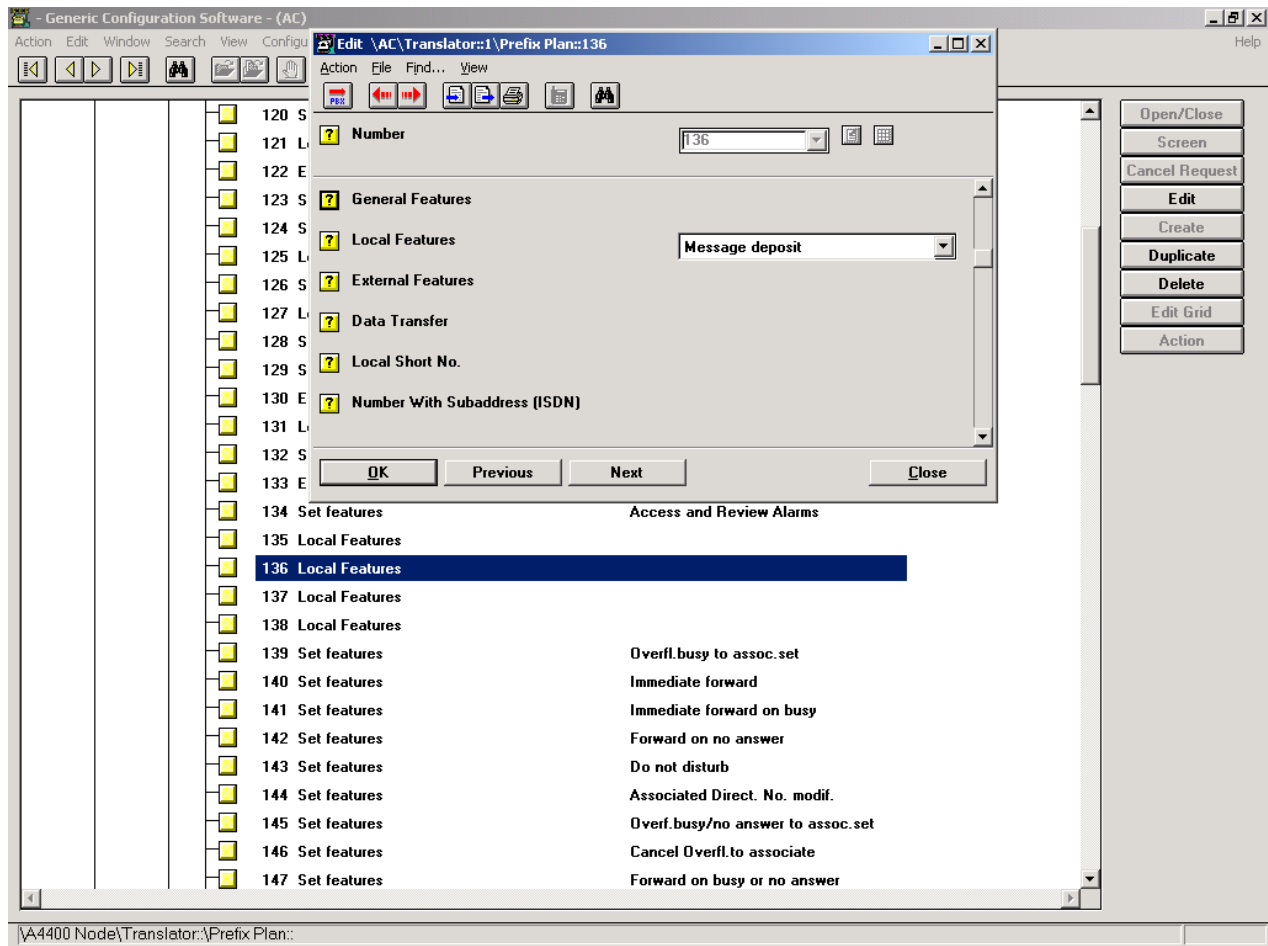
Step 14: In the main menu, double-click the Translator object.



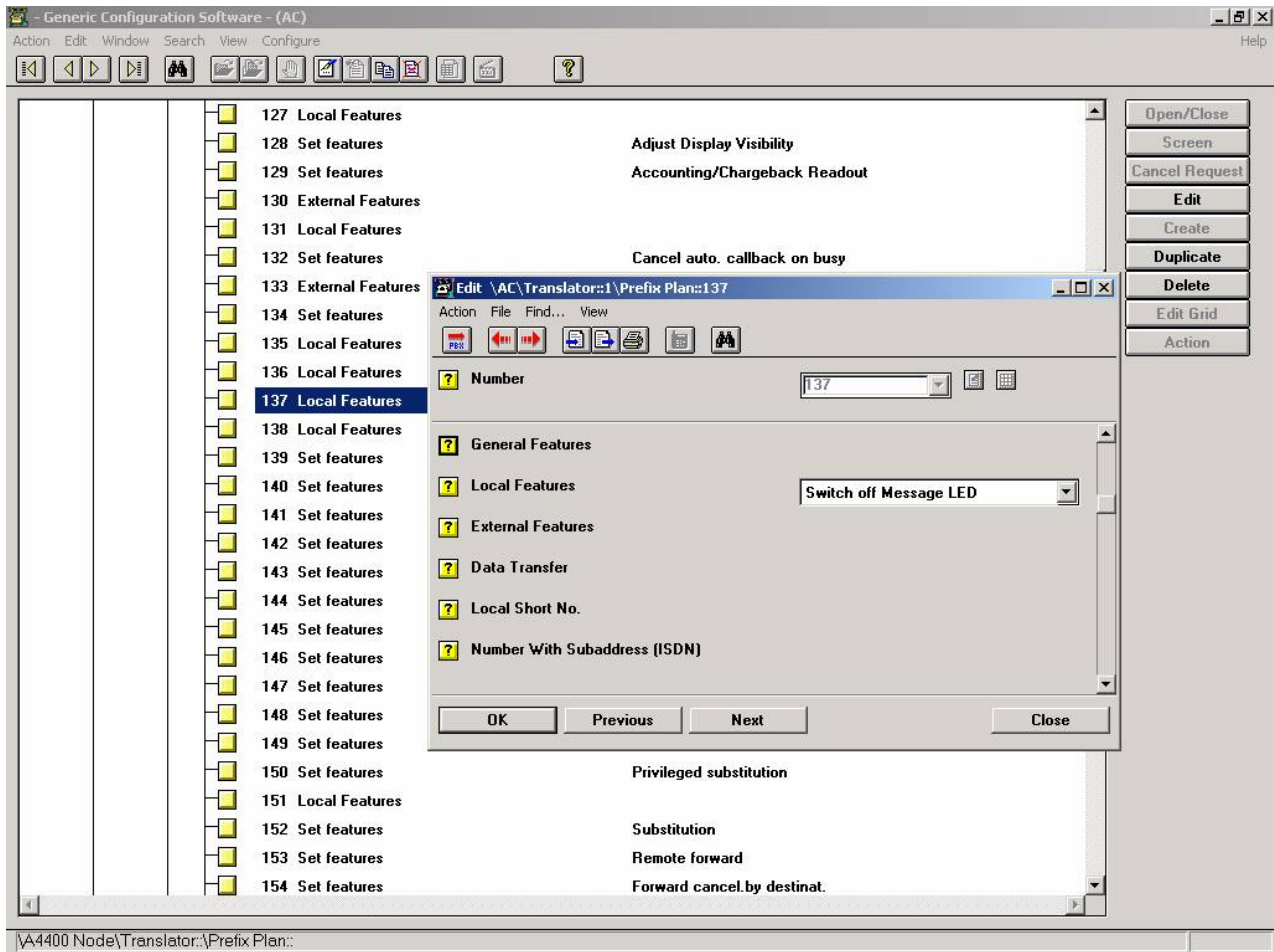
Step 15: Double-click the Prefix Plan object. A list of all the feature access codes appears.



Step 16: Verify the feature access code (Number) for the Local Feature of the type Message Deposit. Perform this by right-clicking each feature in the list, and then from the shortcut menu, choosing **Edit**. For example, the feature access code number is 136 (as shown below).

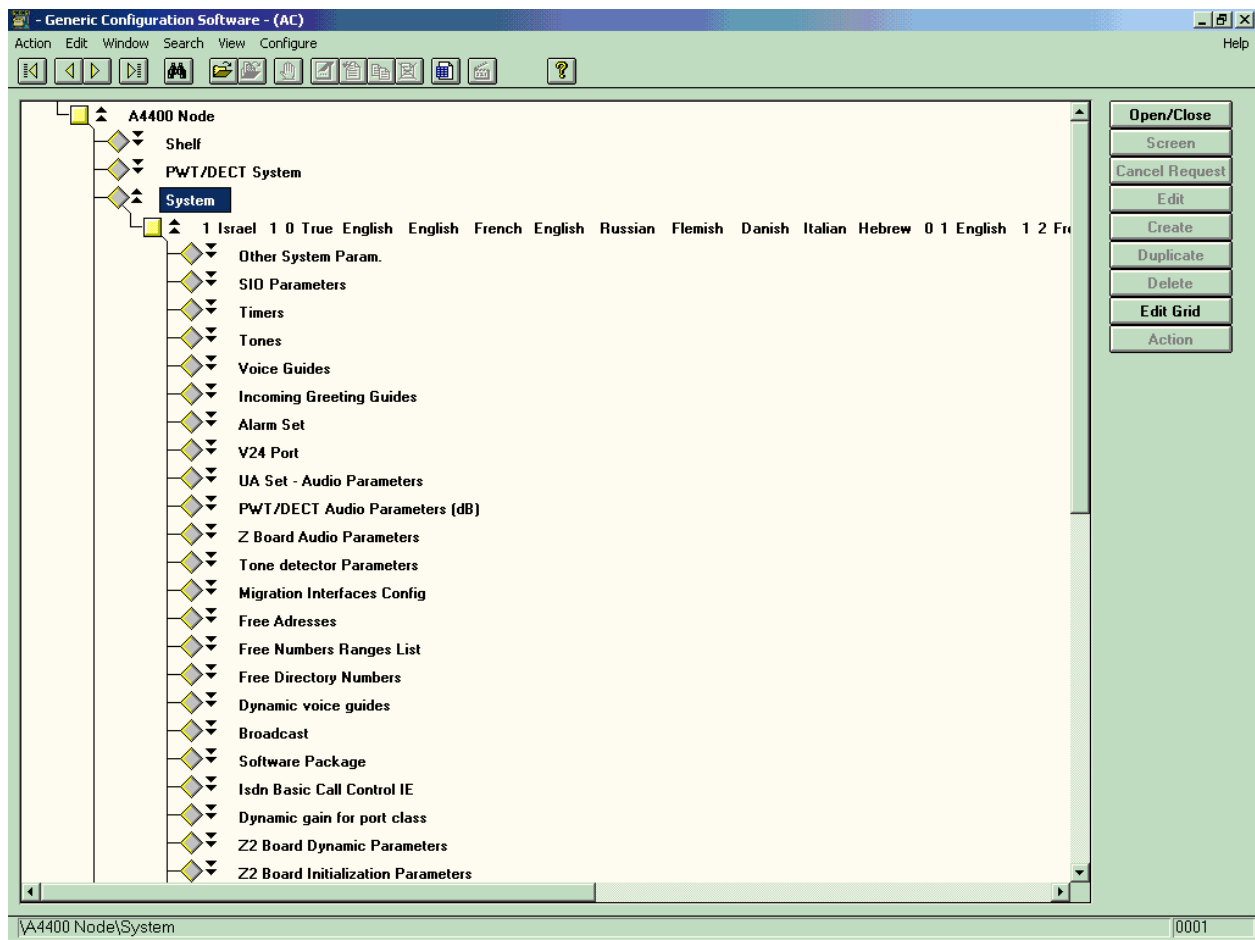


Step 17: In the same list of feature access codes, verify the feature access code (Number) for the Local Feature of the type Switch off Message LED. Perform this by right-clicking each feature in the list, and then from the shortcut menu, choosing **Edit**. For example, the feature access code number is 137 (as shown below).

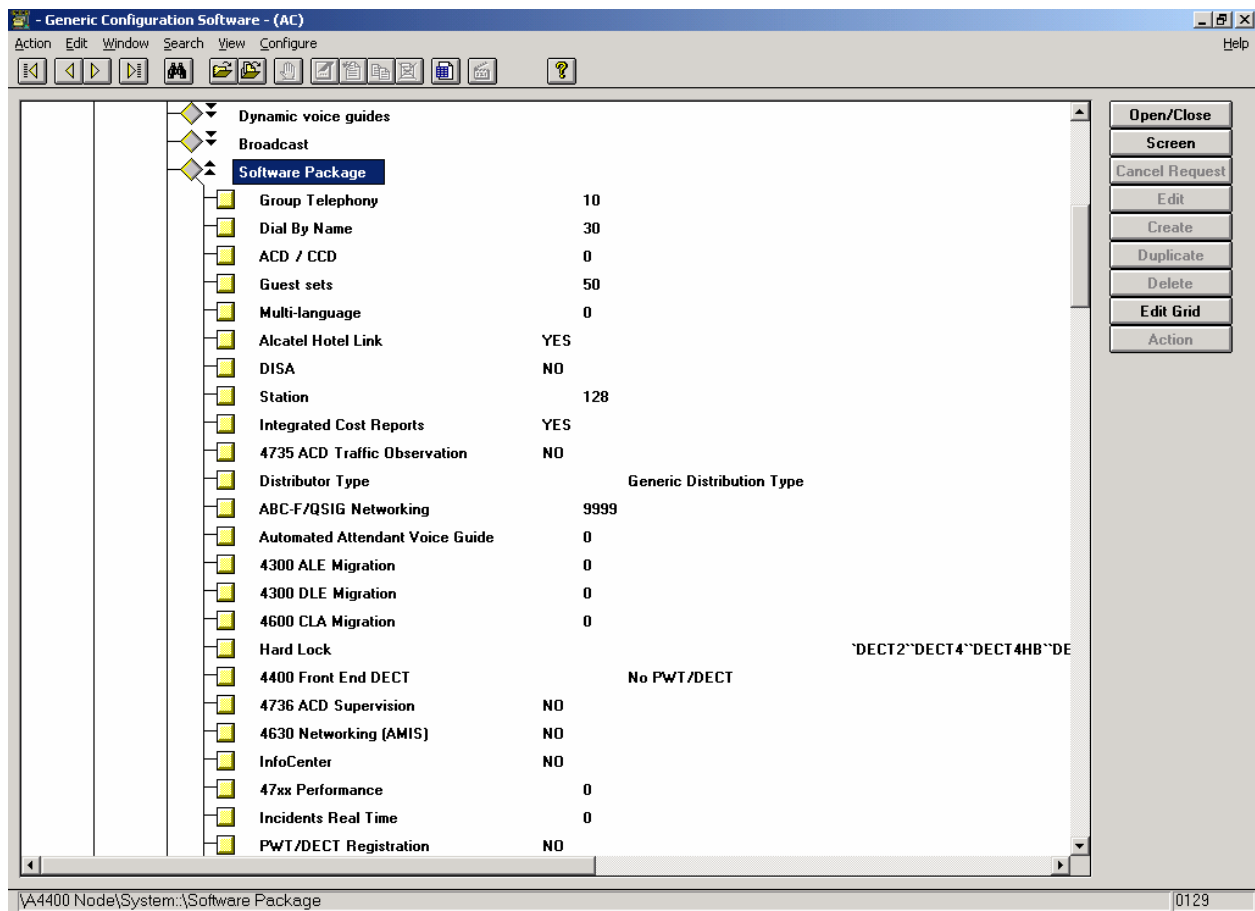


Verify whether or not the PBX supports Caller ID for Voice Mail extensions

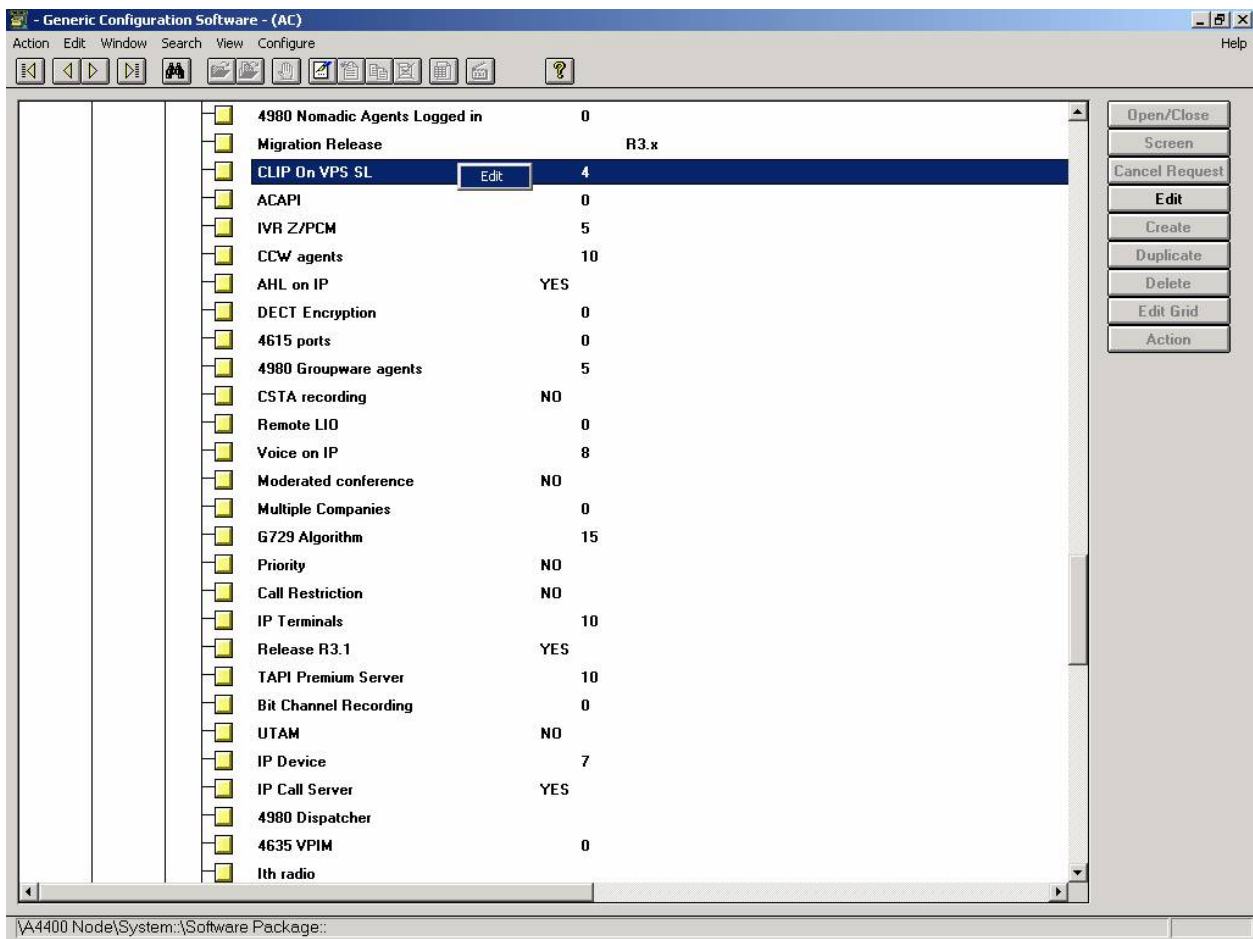
Step 18: In the main menu, double-click the System object. A list of the available system parameters in the PBX is shown.



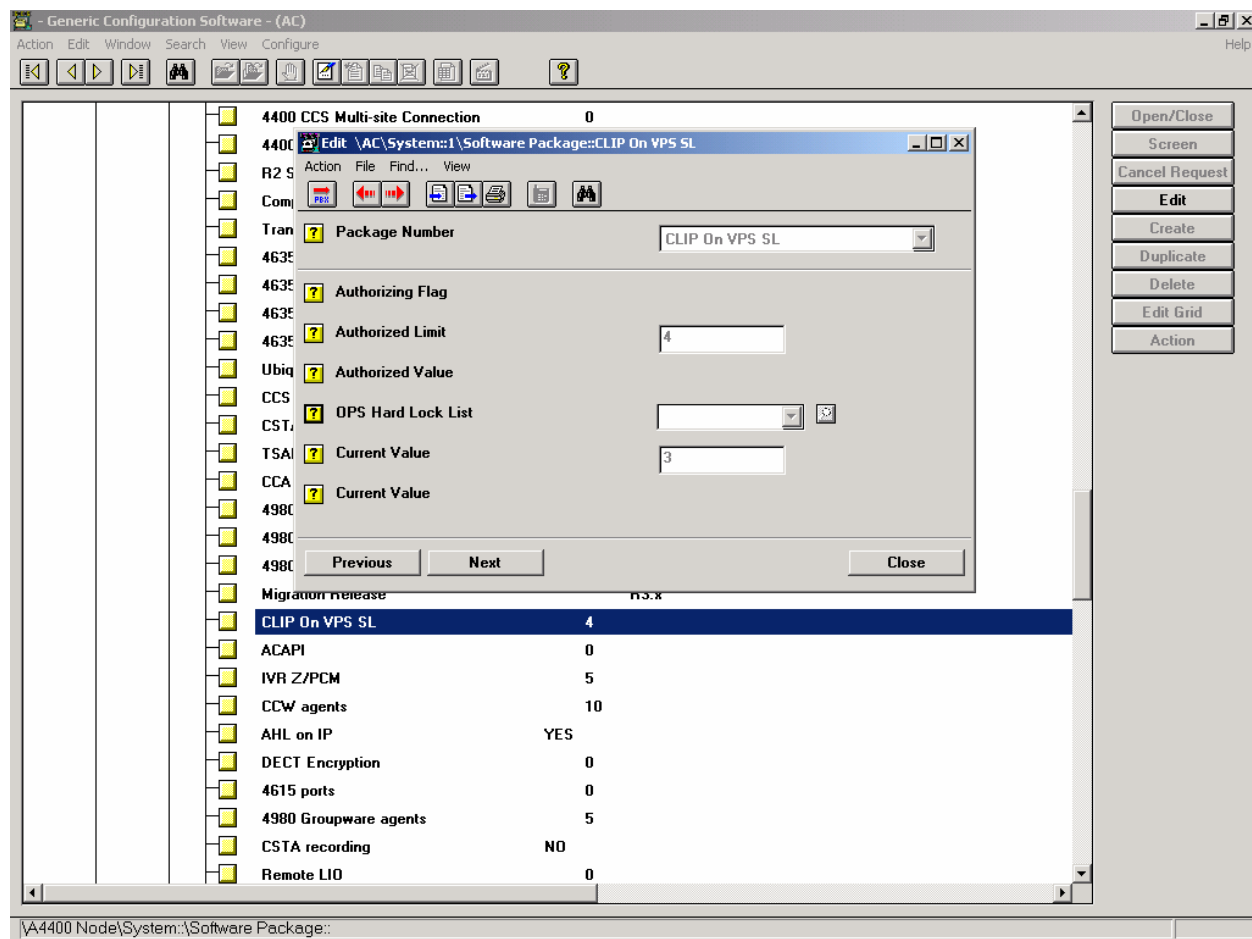
Step 19: Scroll down the list until you see the Software Package object. Double-click the Software Package object. A list of all software packages appears.



Step 20: Scroll down the list until you see the CLIP on VPS SL object. To verify the availability of Caller ID for the voice mail extensions, right-click the object, and then from the shortcut menu, choose **Edit**.



Step 21: In the Edit window, verify the value in the Authorized Limit field. This value displays the total amount of voice mail extensions that are authorized to have the Caller ID feature. If the value of this field is zero (0), the PBX's software package does not support Caller ID for the voice mail extension.



5.1. TLS Setup

- N/A.

5.2. Fail-Over Configuration

- N/A.

5.3. Tested Phones

Alcatel Advanced Reflexes.

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.	CP	The Alcatel OmniPCX doesn't support invalid number notification and the call is routed back to the Microsoft Unified Messaging welcome prompt.
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."	CP	Due to PBX limitation, the user is requested to not only enter the pin number, but also the extension number.
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	The Alcatel OmniPCX 4400 PBX doesn't send the calling extension number for direct calls.
Phone type (if phone-specific)	All types
Call scenarios(s) associated with failure point	8
List of UM features affected by failure point	The user is requested to not only enter the pin number, but also the extension number.
Additional Comments The PBX doesn't support the sending of the calling user number for direct calls (i.e., user calls UM to retrieve voice message). Therefore, when the user dials directly to the Microsoft Unified Messaging, the user hears the general welcome prompt: "Welcome, you are connected to Microsoft Exchange, to access your mailbox, enter your extension.", at which the user is required to enter the user's extension number in addition to the pin number.	

Failure Point	The Alcatel OmniPCX 4400 PBX doesn't support invalid number notification.
Phone type (if phone-specific)	All types
Call scenarios(s) associated with failure point	6d
List of UM features affected by failure point	Transfer to a Microsoft Unified Messaging server user's voicemail after transfer to an invalid number of the user.
Additional Comments When performing blind transfer to an invalid number, the PBX doesn't support invalid number notification and the call is routed back to the original transfer user. When an invalid extension number that's defined in the Microsoft Unified Messaging for a particular user, and a call transfer by Directory Search to this user is requested, the user that requests this transfer is routed back to the Microsoft Unified Messaging welcome prompt.	

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:

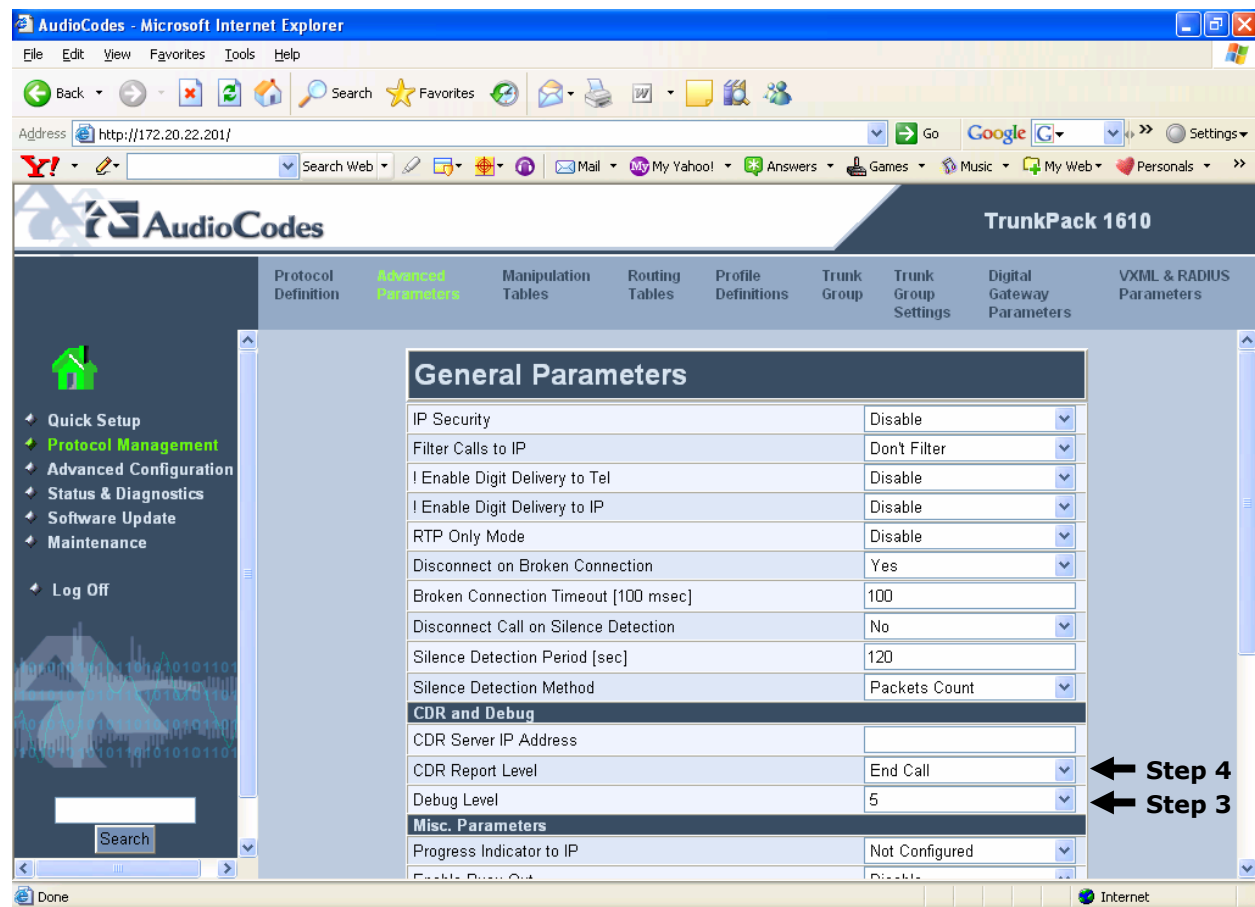
1. Set the parameter **Enable Syslog** to 'Enable'.
2. Use the parameter **Syslog Server IP Address** to define the IP address of the Syslog server you use.

The screenshot shows the AudioCodes TrunkPack 1610 MG Module 1 web interface. The left sidebar contains a navigation menu with options: Quick Setup, Protocol Management, Advanced Configuration (highlighted), Status & Diagnostics, Software Update, Maintenance, and Log Off. The main content area is titled 'Management Settings' and contains several sections: Syslog Settings, SNMP Settings, and Activity Types to Report via 'Activity Log' Messages. The Syslog Settings section includes fields for Syslog Server IP Address (10.15.2.5), Syslog Server Port (514), and a dropdown for Enable Syslog (set to 'Enable'). The SNMP Settings section includes fields for SNMP Managers Table, SNMP Community String, and SNMP V3 Table, each with a '-->' button, and a dropdown for Enable SNMP (set to 'Enable'). The Activity Types to Report via 'Activity Log' Messages section includes checkboxes for Parameters Value Change, Auxiliary Files Loading, Device Reset, Flash Memory Burning, and Device Software Update. Arrows point from the text 'Step 1' to the 'Enable Syslog' dropdown and from 'Step 2' to the 'Syslog Server IP Address' field.

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

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

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



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

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

Appendix

1. Dial Pilot Number and Mailbox Login

- Dial the pilot number of the UM server from an extension that is NOT enabled for UM.
- Confirm hearing the greeting prompt: "Welcome, you are connected to Microsoft Exchange. To access your mailbox, enter your extension..."
- Enter the extension, followed by the mailbox PIN of an UM-enabled user.
- Confirm successful logon to the user's mailbox.

2. Navigate Mailbox using Voice User Interface (VUI)

- Logon to a user's UM mailbox.
- If the user preference has been set to DTMF tones, activate the Voice User Interface (VUI) under personal options.
- Navigate through the mailbox and try out various voice commands to confirm that the VUI is working properly.
- This test confirms that the RTP is flowing in both directions and speech recognition is working properly.

3. Navigate Mailbox using Telephony User Interface (TUI)

- Logon to a user's UM mailbox.
- If the user preference has been set to voice, press "#0" to activate the Telephony User Interface (TUI).
- Navigate through the mailbox and try out the various key commands to confirm that the TUI is working properly.
- This test confirms that both the voice RTP and DTMF RTP (RFC 2833) are flowing in both directions.

4. Dial User Extension and Leave Voicemail

- Note: If you are having difficulty reaching the user's UM voicemail, verify that the coverage path for the UM-enabled user's phone is set to the pilot number of the UM server.

a. From an Internal Extension

- a. From an internal extension, dial the extension for a UM-enabled user and leave a voicemail message.
- b. Confirm the voicemail message arrives in the called user's inbox.
- c. Confirm this message displays a valid Active Directory name as the sender of this voicemail.

b. From an External Phone

- a. From an external phone, dial the extension for a UM-enabled user and leave a voicemail message.
- b. Confirm the voicemail message arrives in the called user's inbox.
- c. Confirm this message displays the phone number as the sender of this voicemail.

5. Dial Auto Attendant(AA)

- Create an Auto Attendant using the Exchange Management Console:
 - a. Under the Exchange Management Console, expand "Organizational Configuration" and then click on "Unified Messaging".
 - b. Go to the Auto Attendant tab under the results pane.
 - c. Click on the "New Auto Attendant..." under the action pane to invoke the AA wizard.
 - d. Associate the AA with the appropriate dial plan and assign an extension for the AA.
 - e. Create PBX dialing rules to always forward calls for the AA extension to the UM server.
 - f. Confirm the AA extension is displayed in the diversion information of the SIP Invite.
- Dial the extension of Auto Attendant.
- Confirm the AA answers the call.

6. Call Transfer by Directory Search

- Method one: Pilot Number Access
 - Dial the pilot number for the UM server from a phone that is NOT enabled for UM.
 - To search for a user by name:
 - Press # to be transferred to name Directory Search.
 - Call Transfer by Directory Search by entering the name of a user in the same Dial Plan using the telephone keypad, last name first.
 - To search for a user by email alias:
 - Press "# " to be transferred to name Directory Search
 - Press "# #" to be transferred to email alias Directory Search
 - Call Transfer by Directory Search by entering the email alias of a user in the same Dial Plan using the telephone keypad, last name first.
- Method two: Auto Attendant
 - Follow the instructions in appendix section 5 to setup the AA.
 - Call Transfer by Directory Search by speaking the name of a user in the same Dial Plan. If the AA is not speech enabled, type in the name using the telephone keypad.

- Note: Even though some keys are associated with three or four numbers, for each letter, each key only needs to be pressed once regardless of the letter you want. Ignore spaces and symbols when spelling the name or email alias.

a. Called Party Answers

- Call Transfer by Directory Search to a user in the same dial plan and have the called party answer.
- Confirm the call is transferred successfully.

b. Called Party is Busy

- Call Transfer by Directory Search to a user in the same dial plan when the called party is busy.
- Confirm the calling user is routed to the correct voicemail.

c. Called Party does not Answer

- Call Transfer by Directory Search to a user in the same dial plan and have the called party not answer the call.
- Confirm the calling user is routed to the correct voicemail.

d. The Extension is Invalid

- Assign an invalid extension to a user in the same dial plan. An invalid extension has the same number of digits as the user's dial plan and has not been mapped on the PBX to any user or device.
 - a. UM Enable a user by invoking the "Enable-UMMailbox" wizard.
 - b. Assign an unused extension to the user.
 - c. Do not map the extension on the PBX to any user or device.
 - d. Call Transfer by Directory Search to this user.
 - e. Confirm the call fails and the caller is prompted with appropriate messages.

7. Play-On-Phone

- To access play-on-phone:
 - a. Logon to Outlook Web Access (OWA) by going to URL <https://<server name>/owa>.
 - b. After receiving a voicemail in the OWA inbox, open this voicemail message.
 - c. At the top of this message, look for the Play-On-Phone field (Play on Phone...).
 - d. Click this field to access the Play-On-Phone feature.

a. To an Internal Extension

- Dial the extension for a UM-enabled user and leave a voicemail message.
- Logon to this called user's mailbox in OWA.

- Once it is received in the user's inbox, use OWA's Play-On-Phone to dial an internal extension.
- Confirm the voicemail is delivered to the correct internal extension.

b. To an External Phone number

- Dial the extension for a UM-enabled user and leave a voicemail message.
- Logon to the UM-enabled user's mailbox in OWA.
- Confirm the voicemail is received in the user's mailbox.
- Use OWA's Play-On-Phone to dial an external phone number.
- Confirm the voicemail is delivered to the correct external phone number.
- Troubleshooting:
 - a. Make sure the appropriate UMMailboxPolicy dialing rule is configured to make this call. As an example, open an Exchange Management Shell and type in the following commands:
 - b. `$dp = get-umdialplan -id <dial plan ID>`
 - c. `$dp.ConfiguredInCountryOrRegionGroups.Clear()`
 - d. `$dp.ConfiguredInCountryOrRegionGroups.Add("anywhere,*,*,")`
 - e. `$dp.AllowedInCountryOrRegionGroups.Clear()`
 - f. `$dp.AllowedInCountryOrRegionGroups.Add("anywhere")`
 - g. `$dp|set-umdialplan`
 - h. `$mp = get-ummailboxpolicy -id <mailbox policy ID>`
 - i. `$mp.AllowedInCountryGroups.Clear()`
 - j. `$mp.AllowedInCountryGroups.Add("anywhere")`
 - k. `$mp|set-ummailboxpolicy`
 - l. The user must be enabled for external dialing on the PBX.
 - m. Depending on how the PBX is configured, you may need to prepend the trunk access code (e.g. 9) to the external phone number.

8. Voicemail Button

- Configure a button on the phone of a UM-enabled user to route the user to the pilot number of the UM server.
- Press this voicemail button on the phone of an UM-enabled user.
- Confirm you are sent to the prompt: "Welcome, you are connected to Microsoft Exchange. <User Name>. Please enter your pin and press the pound key."
- Note: If you are not hearing this prompt, verify that the button configured on the phone passes the user's extension as the redirect number. This means that the user extension should appear in the diversion information of the SIP invite.

9. FAX

- Use the Management Console or the Management Shell to FAX-enable a user.
- Management Console:
 - a. Double click on a user's mailbox and go to Mailbox Features tab.
 - b. Click Unified Messaging and then click the properties button.
 - c. Check the box "Allow faxes to be received".
- Management Shell - execute the following command:
 - a. Set-UMMailbox -identity UMUser -FaxEnabled:\$true
- To test fax functionality:
 - a. Dial the extension for this fax-enabled UM user from a fax machine.
 - b. Confirm the fax message is received in the user's inbox.
 - c. Note: You may notice that the UM server answers the call as though it is a voice call (i.e. you will hear: "Please leave a message for..."). When the UM server detects the fax CNG tones, it switches into fax receiving mode, and the voice prompts terminate.
 - d. Note: UM only support T.38 for sending fax.

10.TRANSPORT SECURITY LAYER (TLS)

- Setup TLS on the gateway/IP-PBX and Exchange 2007 UM.
- Import/Export all the appropriate certificates.

a. Dial Pilot Number and Mailbox Login

- Execute the steps in scenario 1 (above) with TLS turned on.

b. Dial User Extension and Leave a Voicemail

- Execute the steps in scenario 4 (above) with TLS turned on.

c. FAX

- Execute the steps in scenario 9 (above) with TLS turned on.

11.G.723.1

- Configure the gateway to use the G.723.1 codec for sending audio to the UM server.
- If already using G.723.1 for the previous set of tests, use this step to test G.711 A Law or G.711 Mu Law instead.
- Call the pilot number and verify the UM server answers the call.
- Note: If the gateway is configured to use multiple codecs, the UM server, by default, will use the G.723.1 codec if it is available.

12.Message Waiting Indicator (MWI)

- Although Exchange 2007 UM does not natively support MWI, Geomant has created a 3rd party solution - MWI2007. This product also supports SMS message notification.
- Installation files and product documentation can be found on Geomant's [MWI 2007 website](#).

13.Test-UMConnectivity

- Run the Test-UMConnectivity diagnostic cmdlet by executing the following command in Exchange Management Shell:
- Test-UMConnectivity -UMIPGateway:<Gateway> -Phone:<Phone> |fl
- <Gateway> is the name (or IP address) of the gateway which is connected to UM, and through which you want to check the connectivity to the UM server. Make sure the gateway is configured to route calls to UM.
- <Phone> is a valid UM extension. First, try using the UM pilot number for the hunt-group linked to the gateway. Next, try using a CFNA number configured for the gateway. Please ensure that a user or an AA is present on the UM server with that number.
- The output shows the latency and reports if it was successful or there were any errors.

14.Test Fail-Over Configuration on IP-PBX with Two UM Servers

- This is only required for direct SIP integration with IP-PBX. If the IP-PBX supports fail-over configuration (e.g., round-robin calls between two or more UM servers):
 - a. Provide the configuration steps in Section 5.
 - b. Configure the IP-PBX to work with two UM servers.
 - c. Simulate a failure in one UM server.
 - d. Confirm the IP-PBX transfers new calls to the other UM server successfully.