Microsoft Exchange Server 2007 Unified Messaging

PBX Configuration Note:

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

By : AudioCodes

Updated Since : 2007-02-09

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Content

This document describes the configuration required to setup Avaya S8300 and AudioCodes MP-11x FXO using analog lines with inband DTMF as the telephony signaling protocol. It also contains the results of the interoperability testing of Microsoft Exchange 2007 Unified Messaging based on this setup.

Intended Audience

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

Technical Support

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

Microsoft Exchange 2007 Unified Messaging (UM) Specialists

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

Version Information

Date of Modification	Details of Modification
21 March 2007	Version 1

1. Components Information

1.1. PBX or IP-PBX

PBX Vendor	Avaya
Model	S8300
Software Version	G3xV11 Communication Manager 1.3
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

2. Prerequisites

2.1. Gateway Prerequisites

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

2.2. PBX Prerequisites

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

2.3. Cabling Requirements

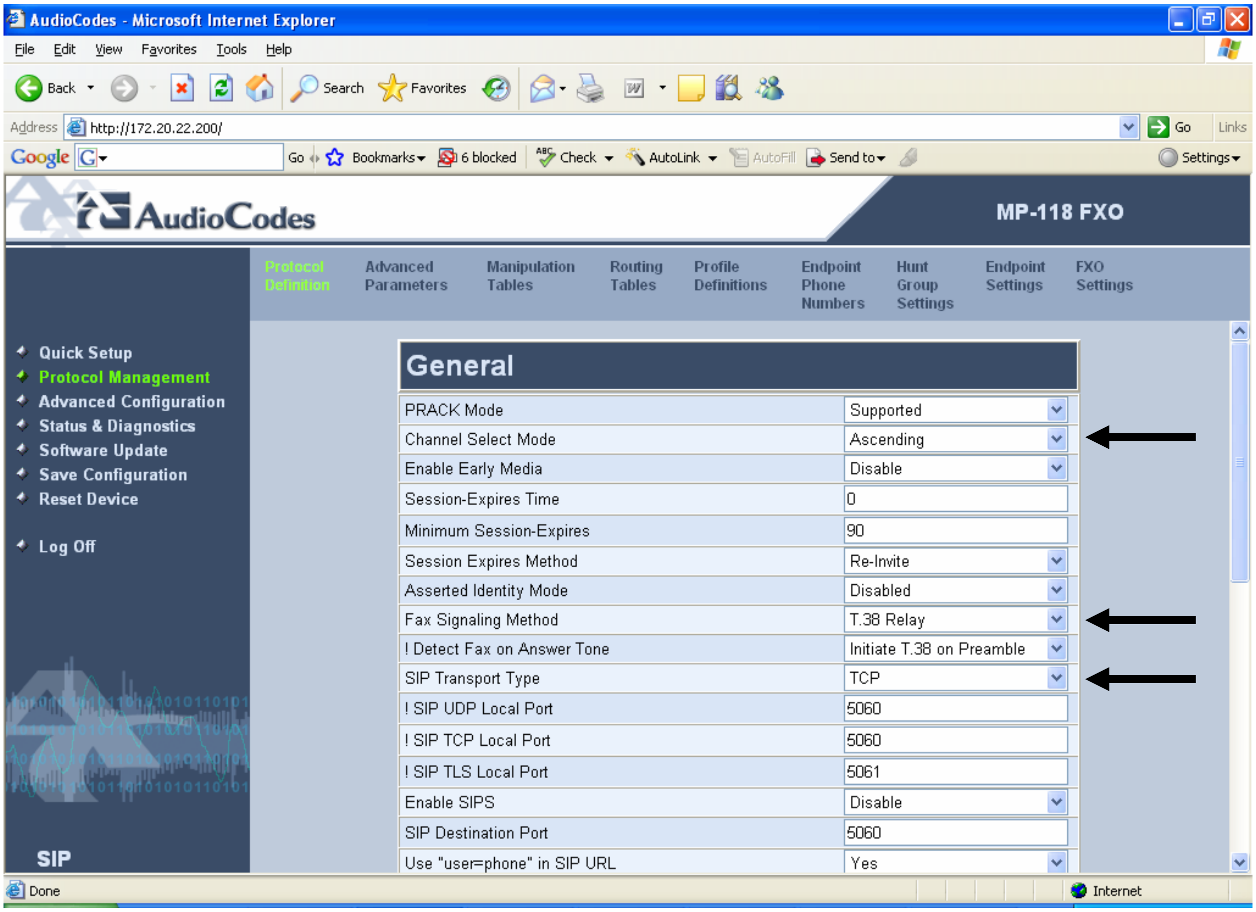
This integration uses standard RJ-11 telephone line cords to connect analog ports between MM711 and MP-11x ports.

3. Summary and Limitations

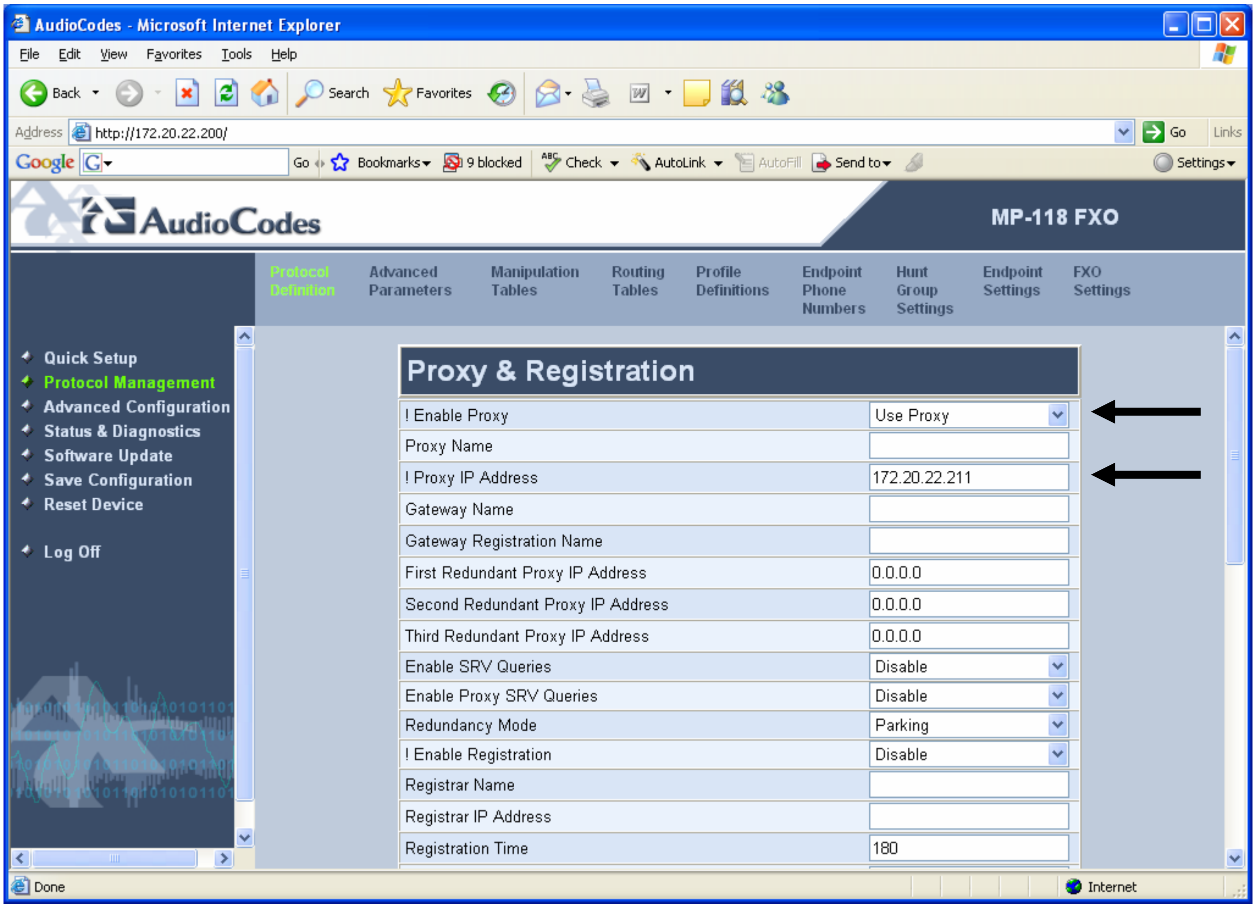
☒ A check in this box indicates the UM feature set is fully functional when using the PBX/gateway in question.

4. Gateway Setup Notes

Step 1: SIP Environment Setup

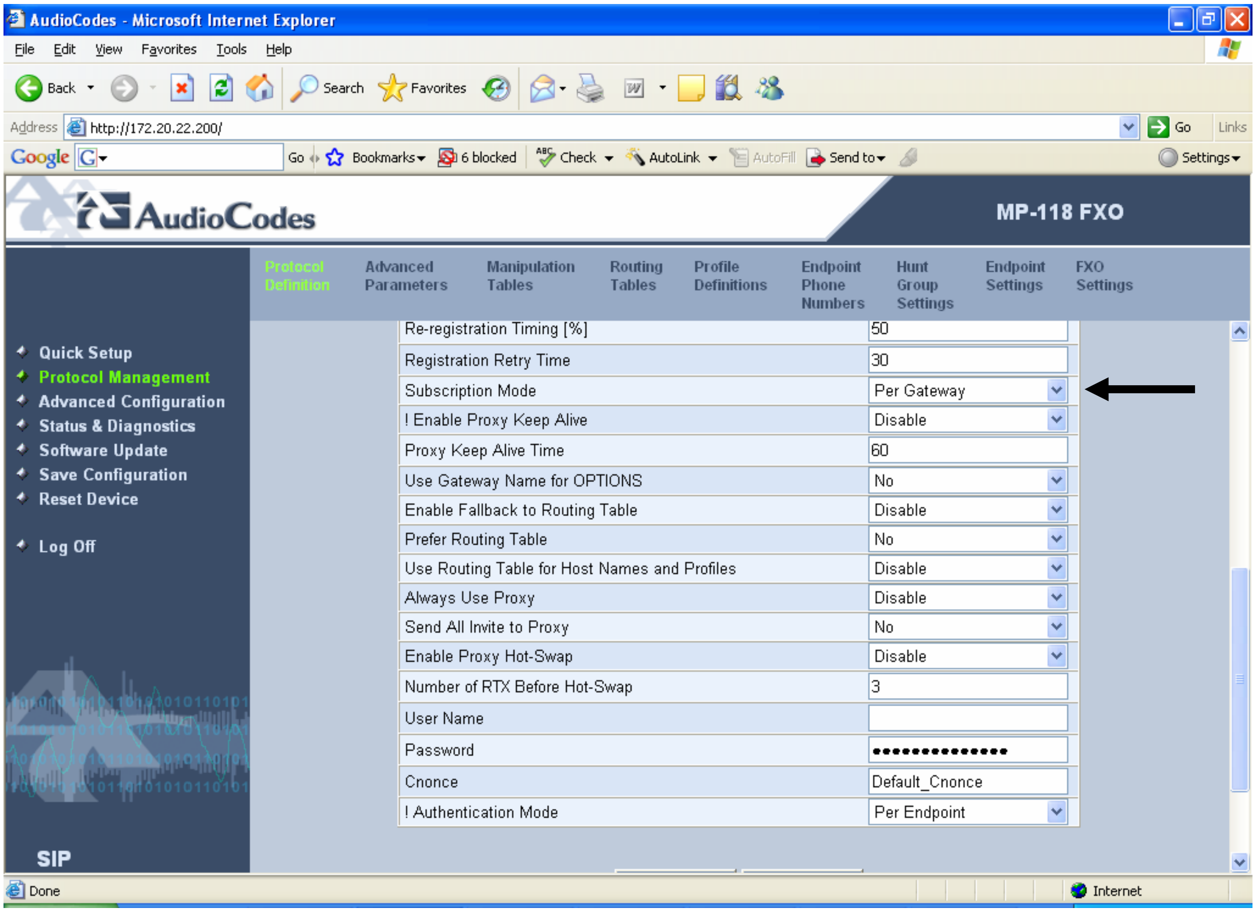


Step 2: Routing Setup

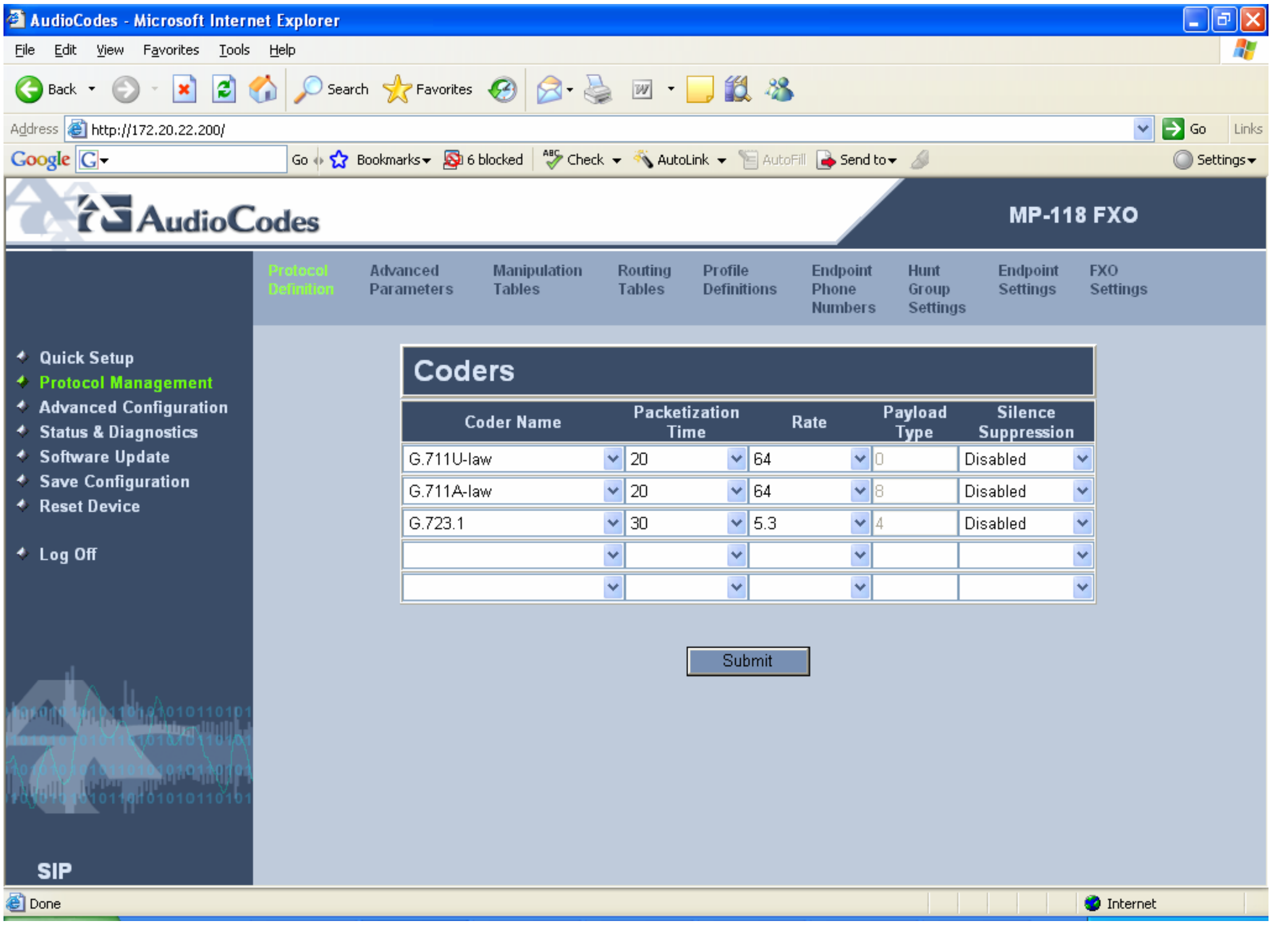


Note: The Proxy IP Address must be one that corresponds to the network environment in which the Microsoft Unified Messaging server is installed (For example, 172.20.22.211 or the FQDN of the Microsoft Unified Messaging host).

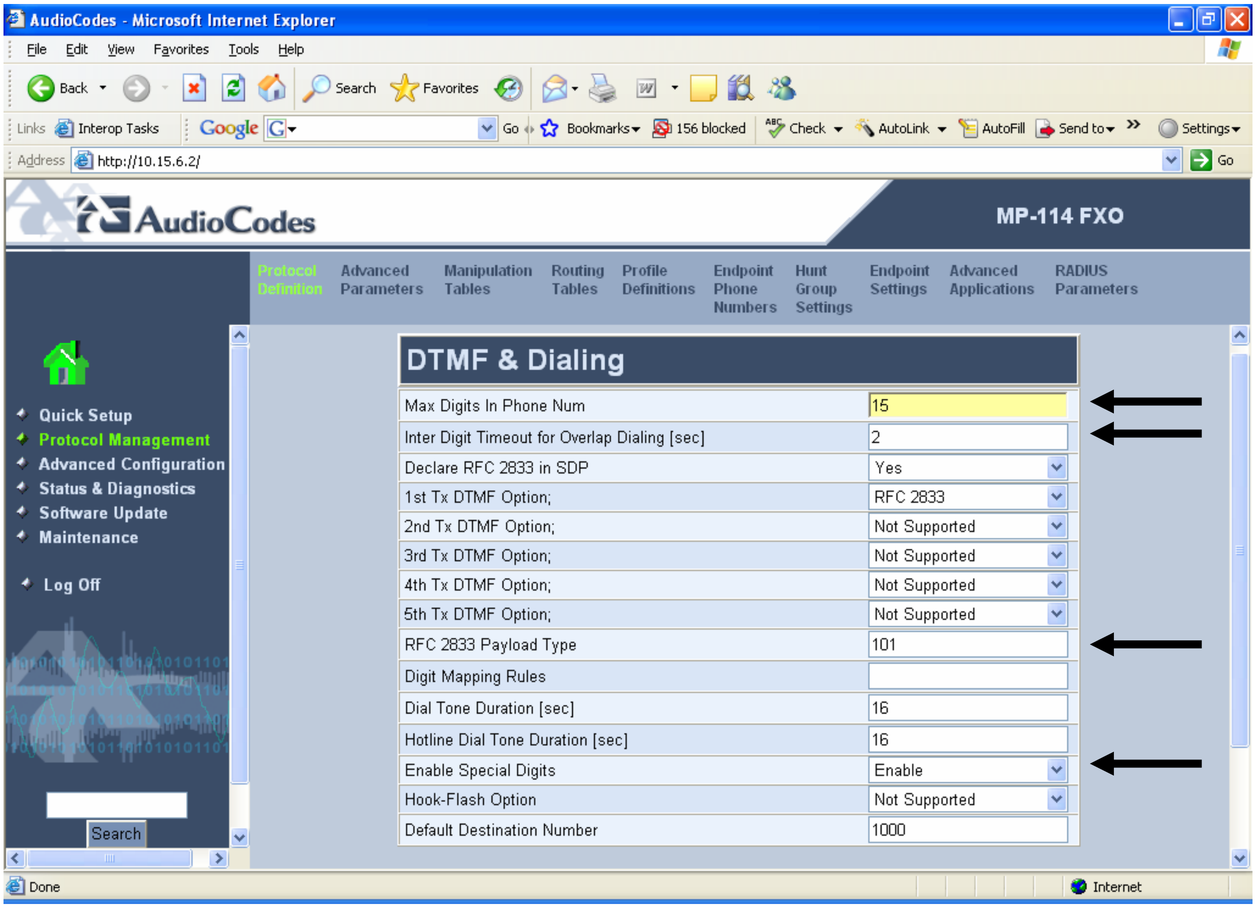
Step 3: SIP Environment Setup (Cont.)



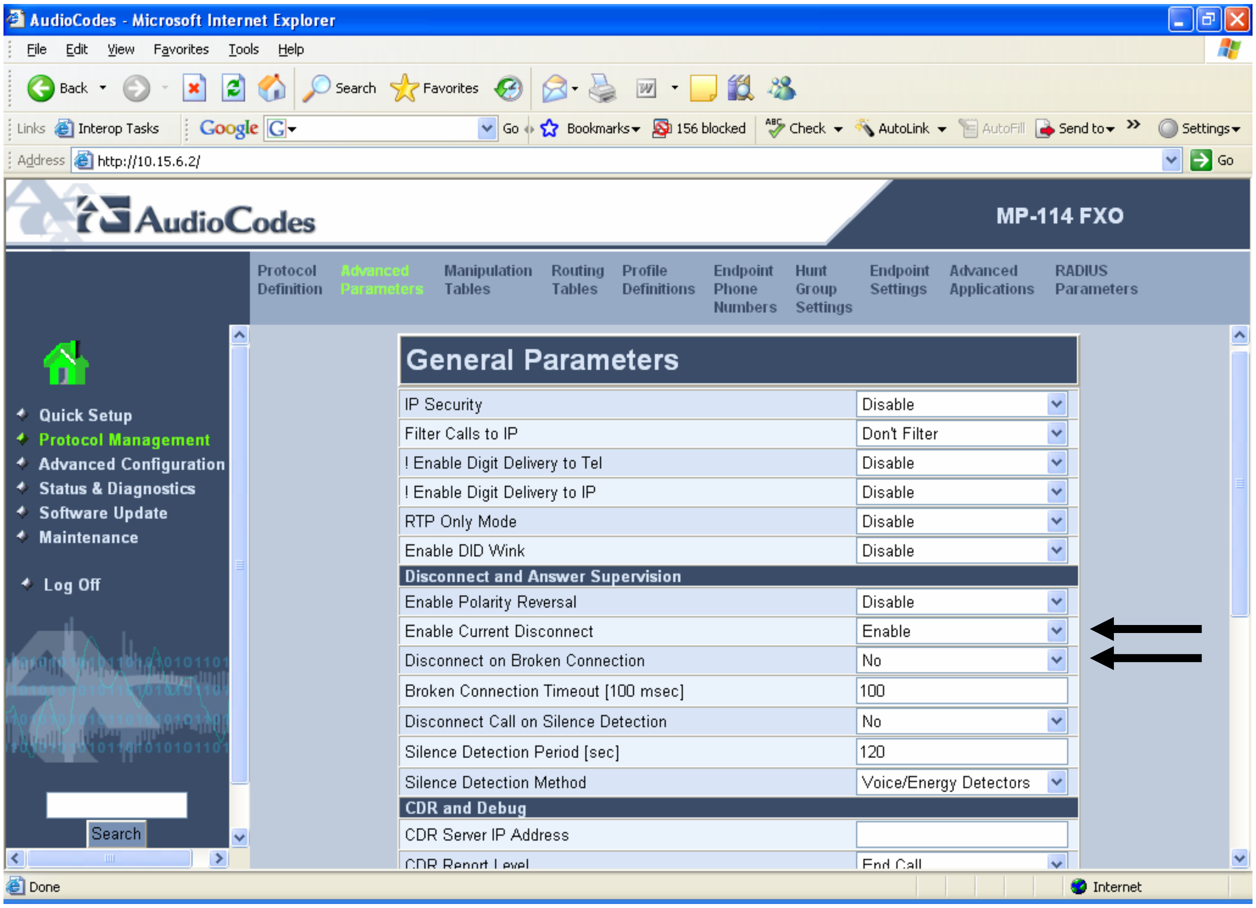
Step 4: Coder Setup



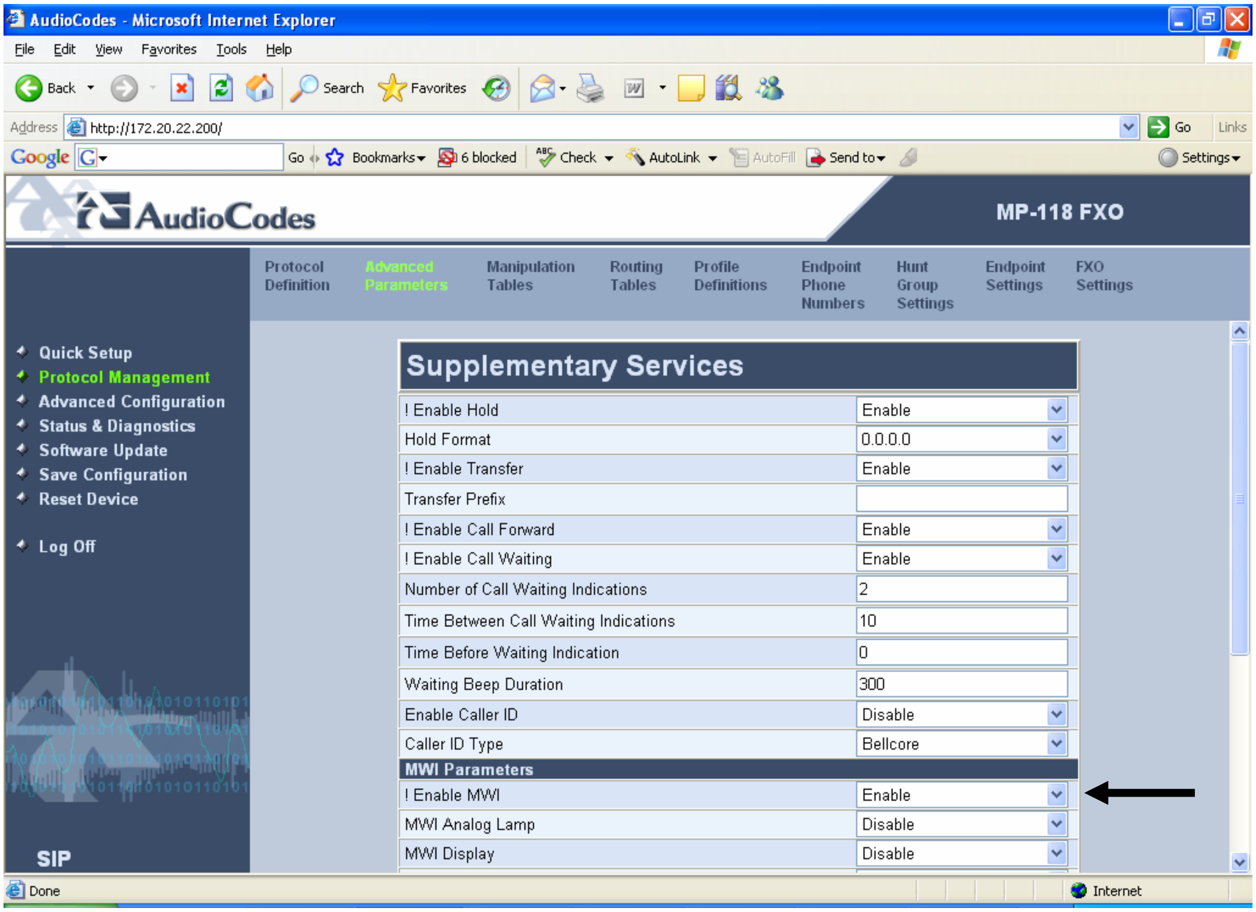
Step 5: Digit Collection Setup



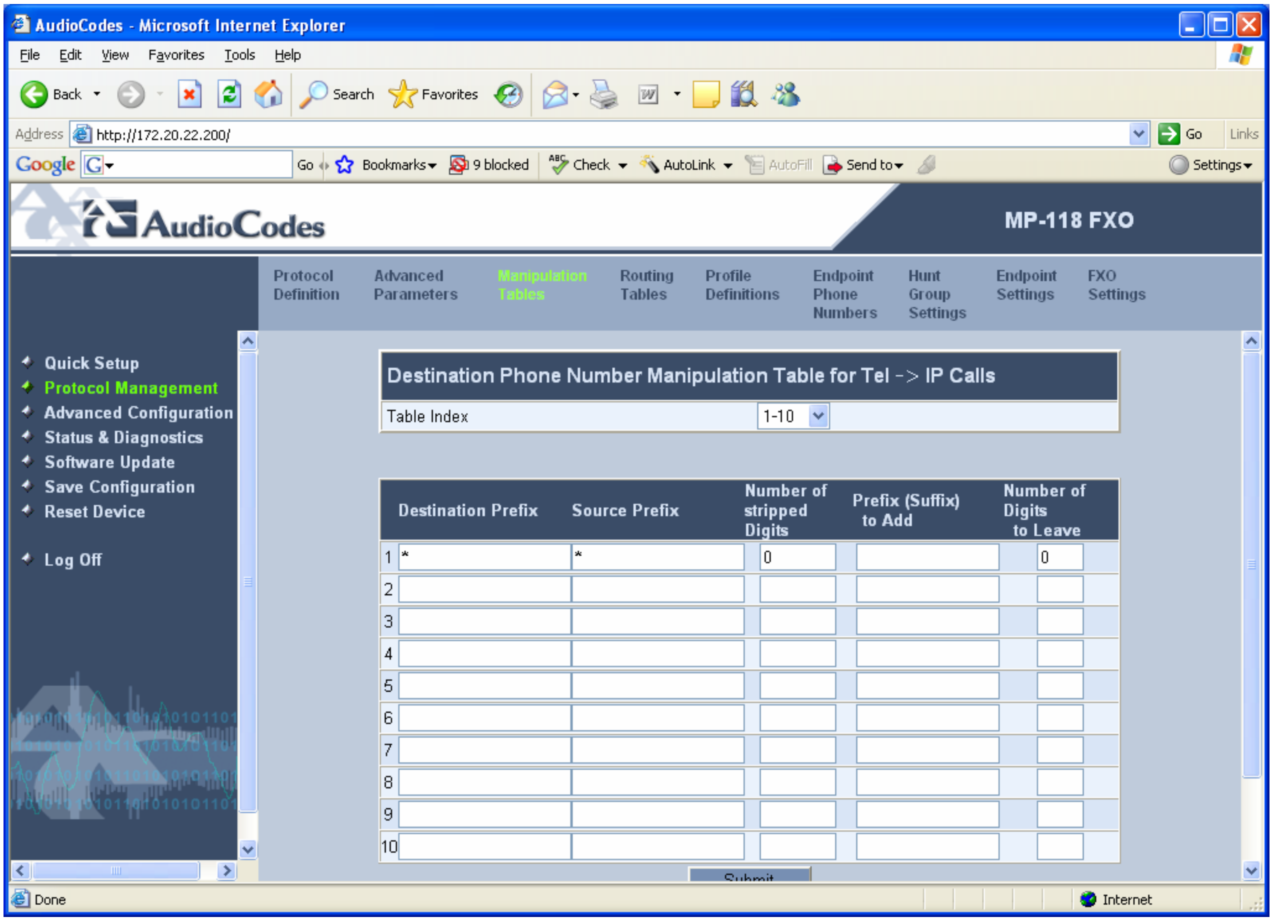
Step 6: Disconnect Supervision Setup



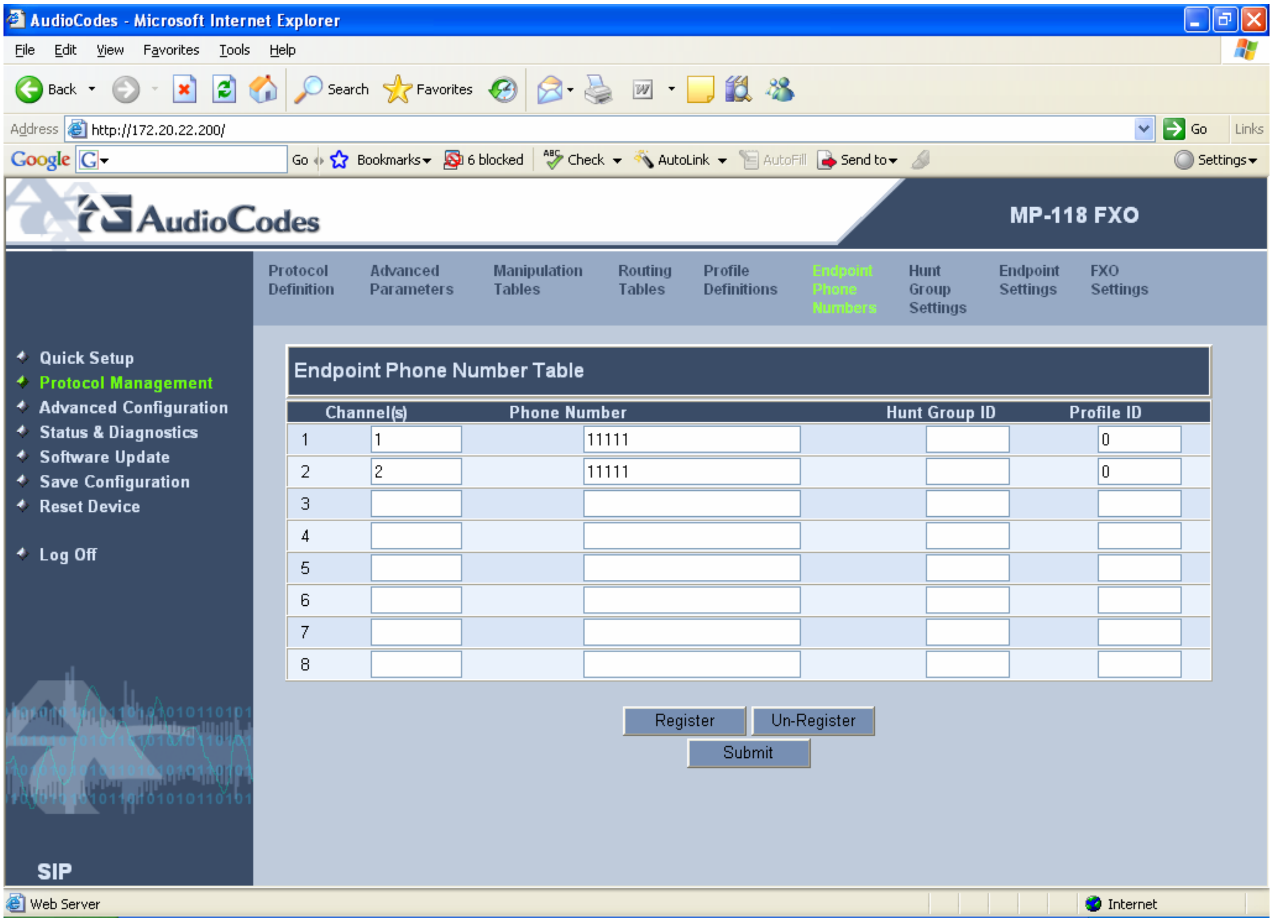
Step 7: Message Waiting Indication Setup



Step 8: Manipulation Routing Setup

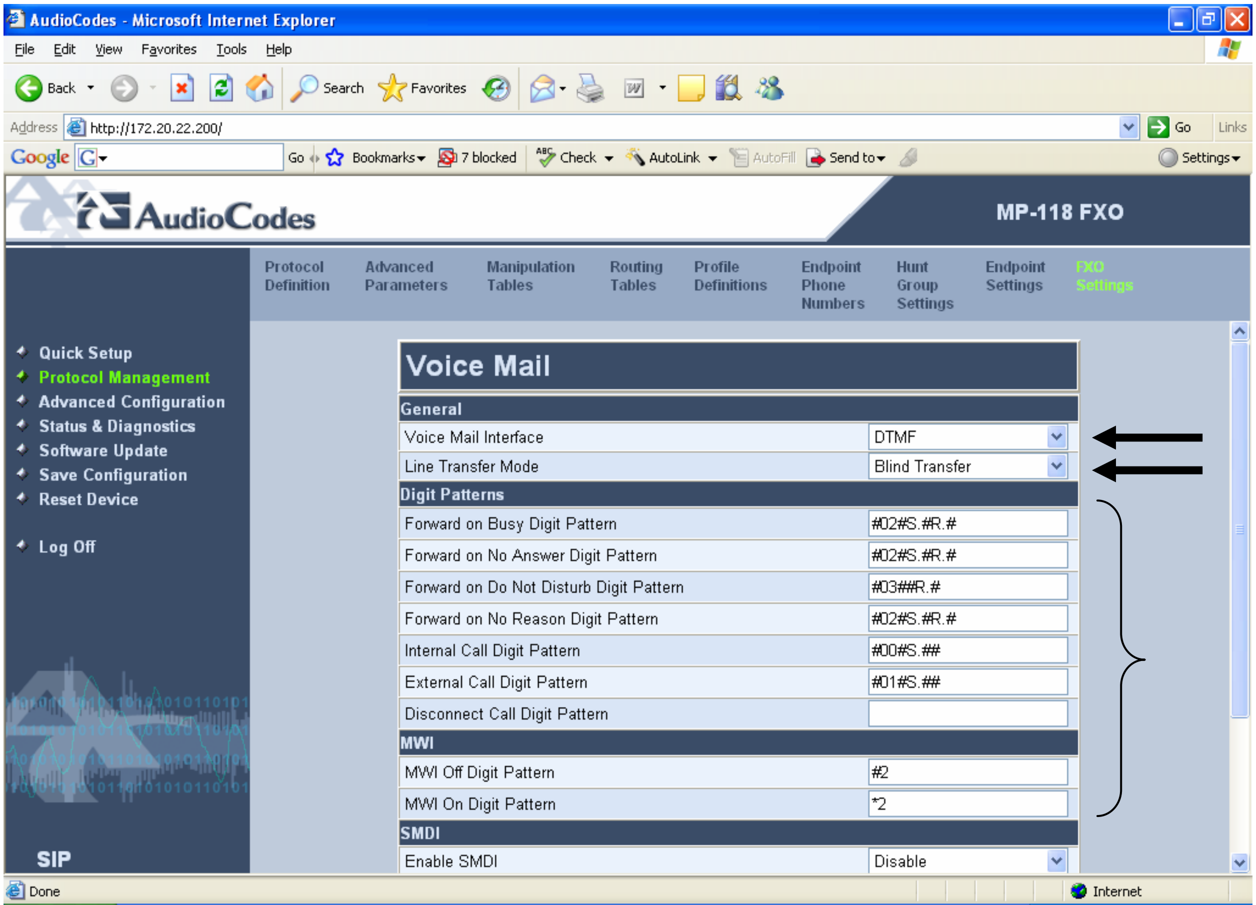


Step 9: Endpoints Setup

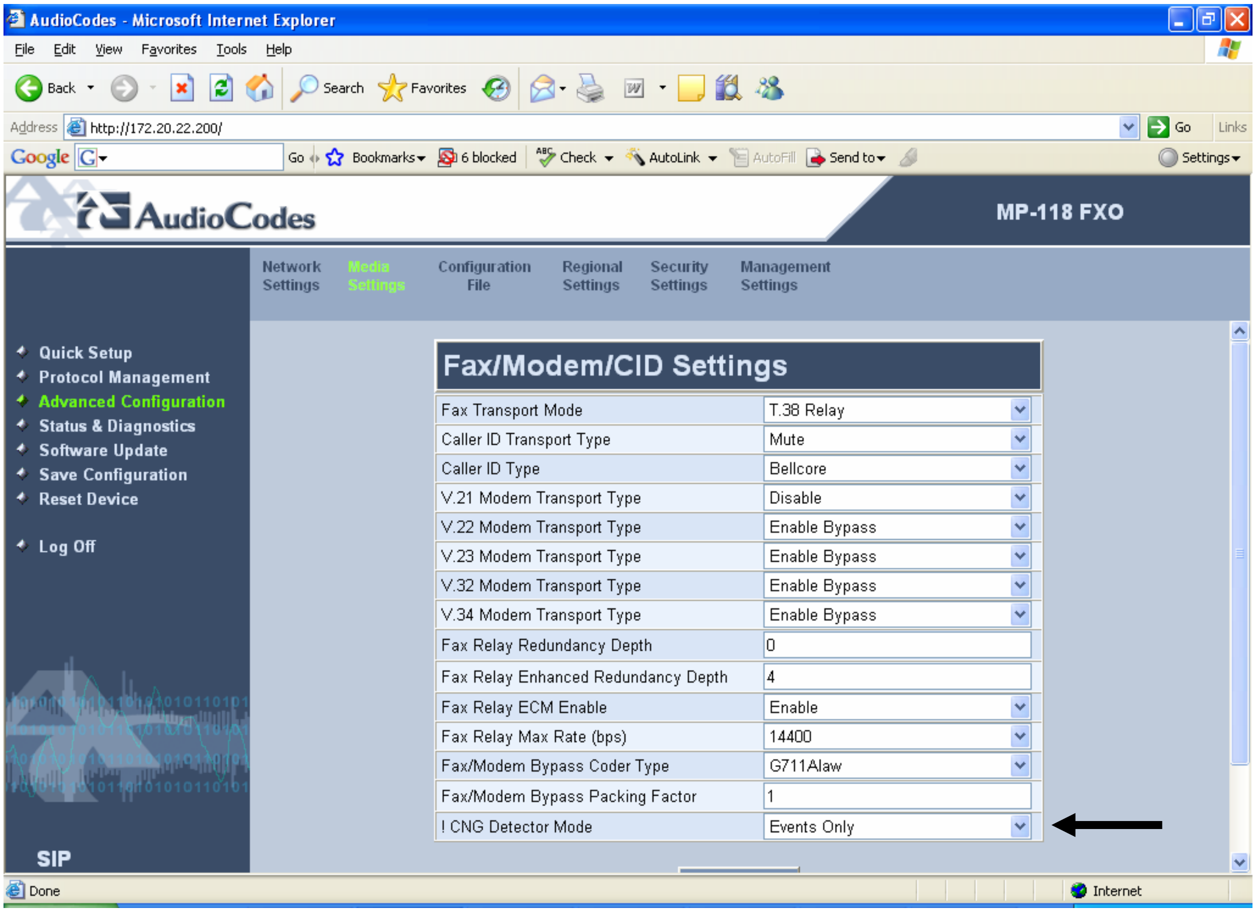


Note: The phone numbers must correspond to your network environment as the dial plan pilot number is configured for this PBX in the Microsoft Unified Messaging server (For example, 11111).

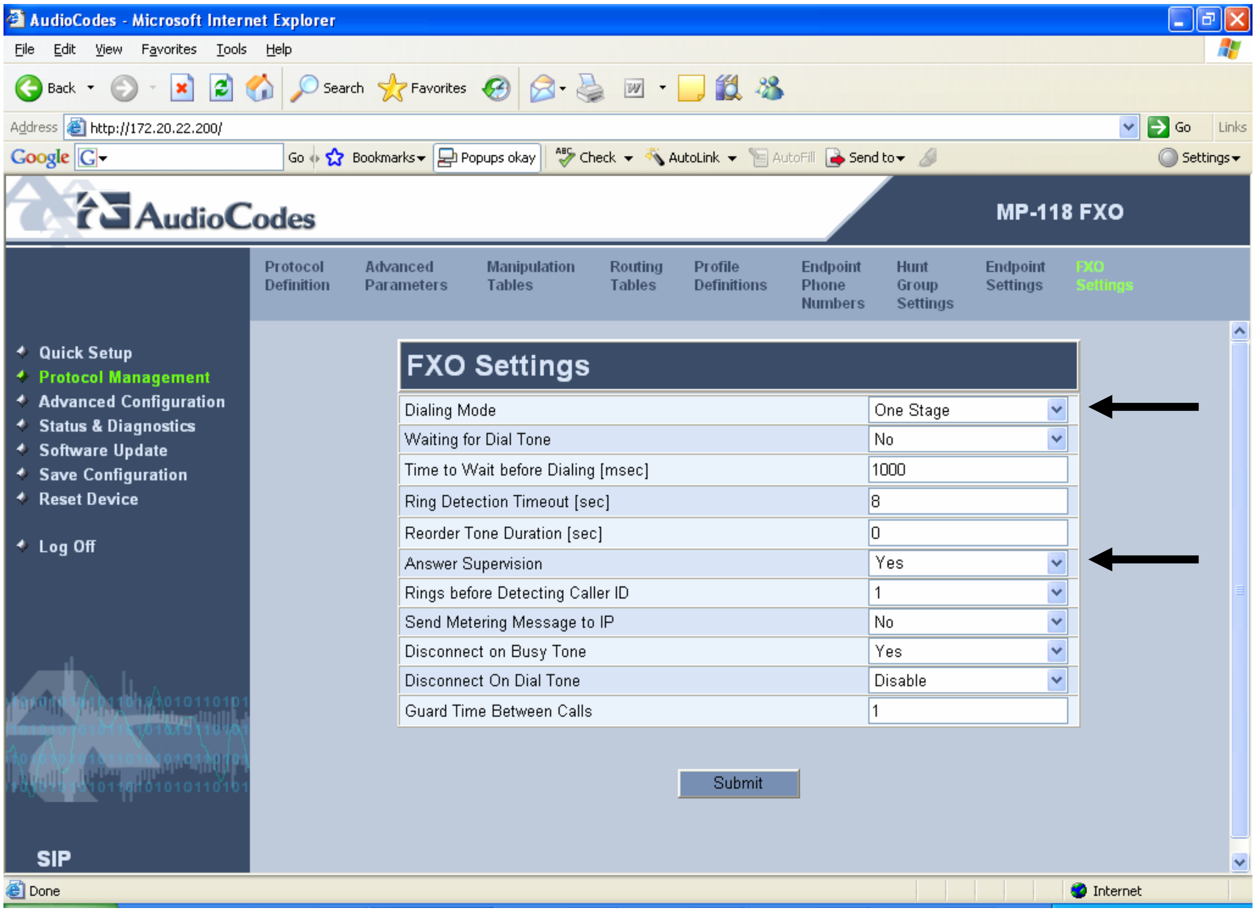
Step 10: Voicemail In-Band DTMF Setup



Step 11: FAX Setup



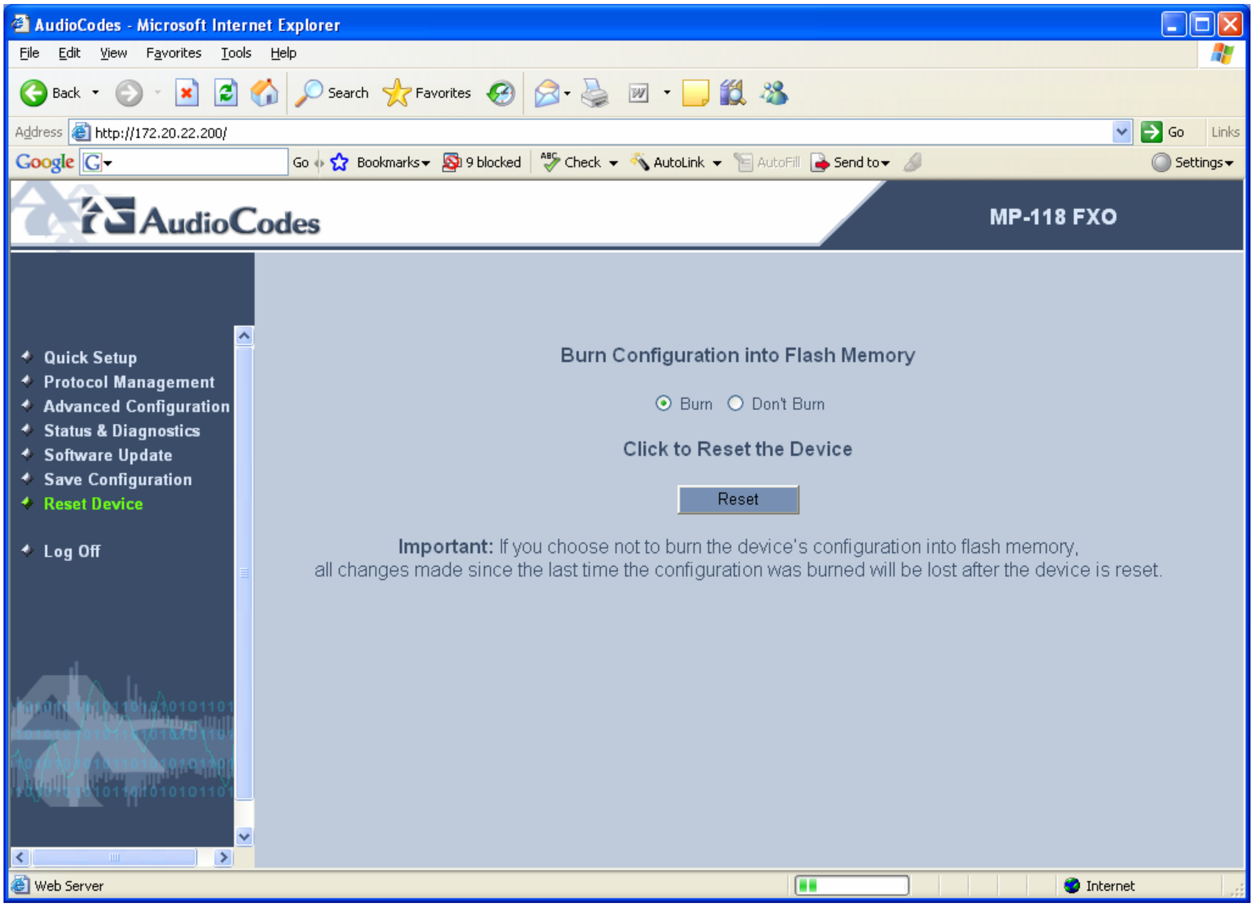
Step 12: FXO General Setup



Step 13: FXO General Setup (Cont.)

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

Step 14: Reset FXO

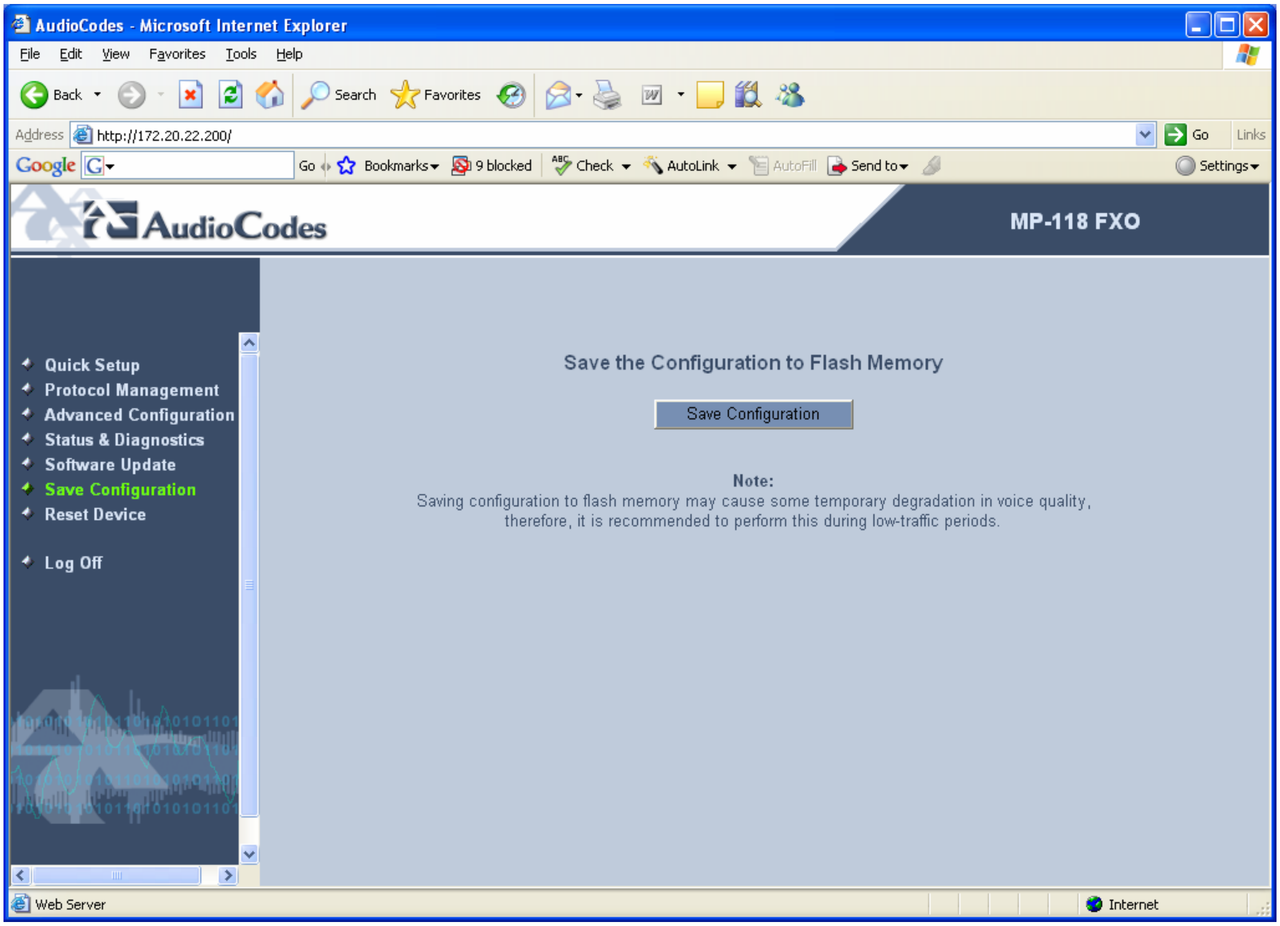


Note: Steps 1 and 7 involve core configuration changes (versus default settings):

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

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

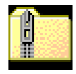
Step 15: Save Gateway Configuration



Note: This step is optional and is note required if you performed Step 14.

4.1. Configuration Files

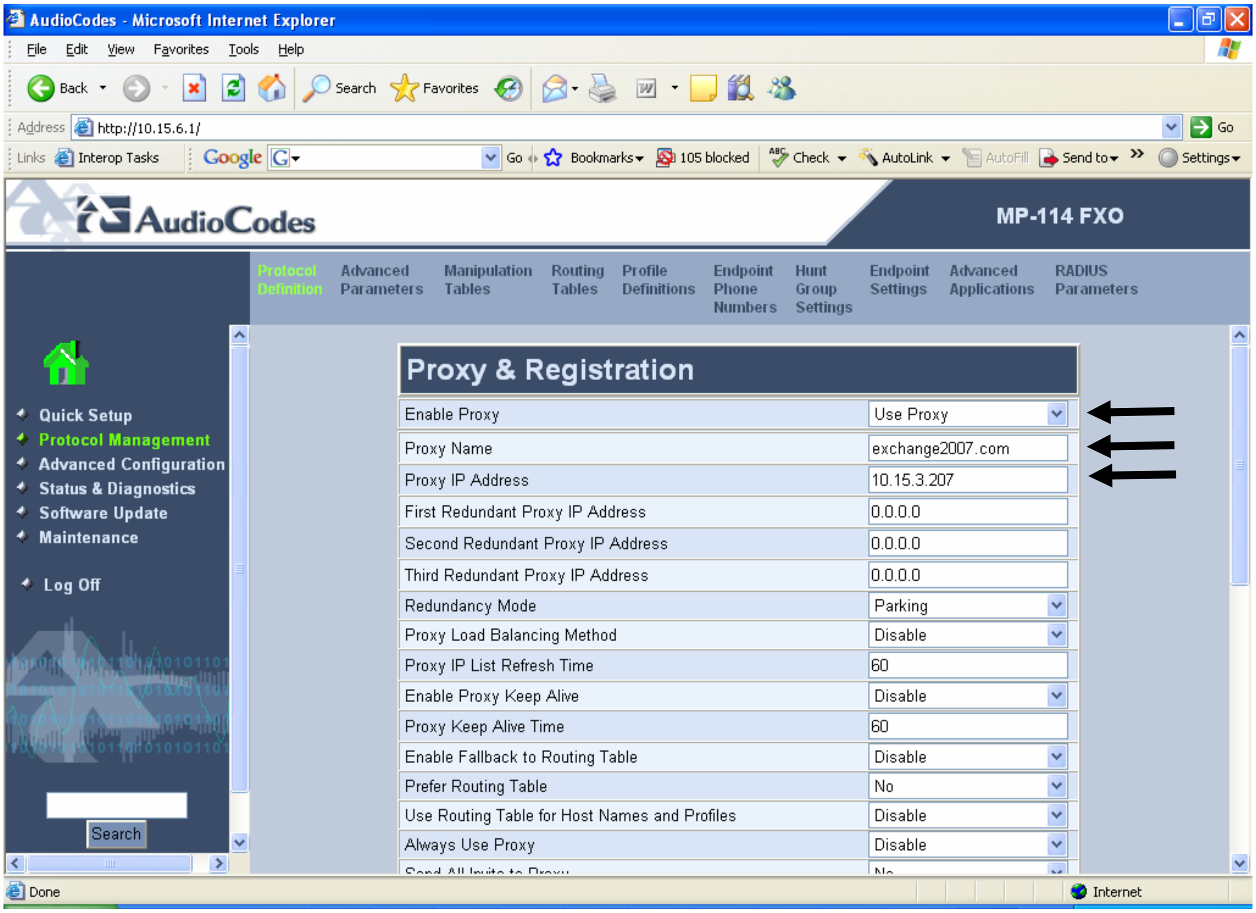
- AudioCodes configuration ini file (.ini file extension).



INI Avaya S8300.zip

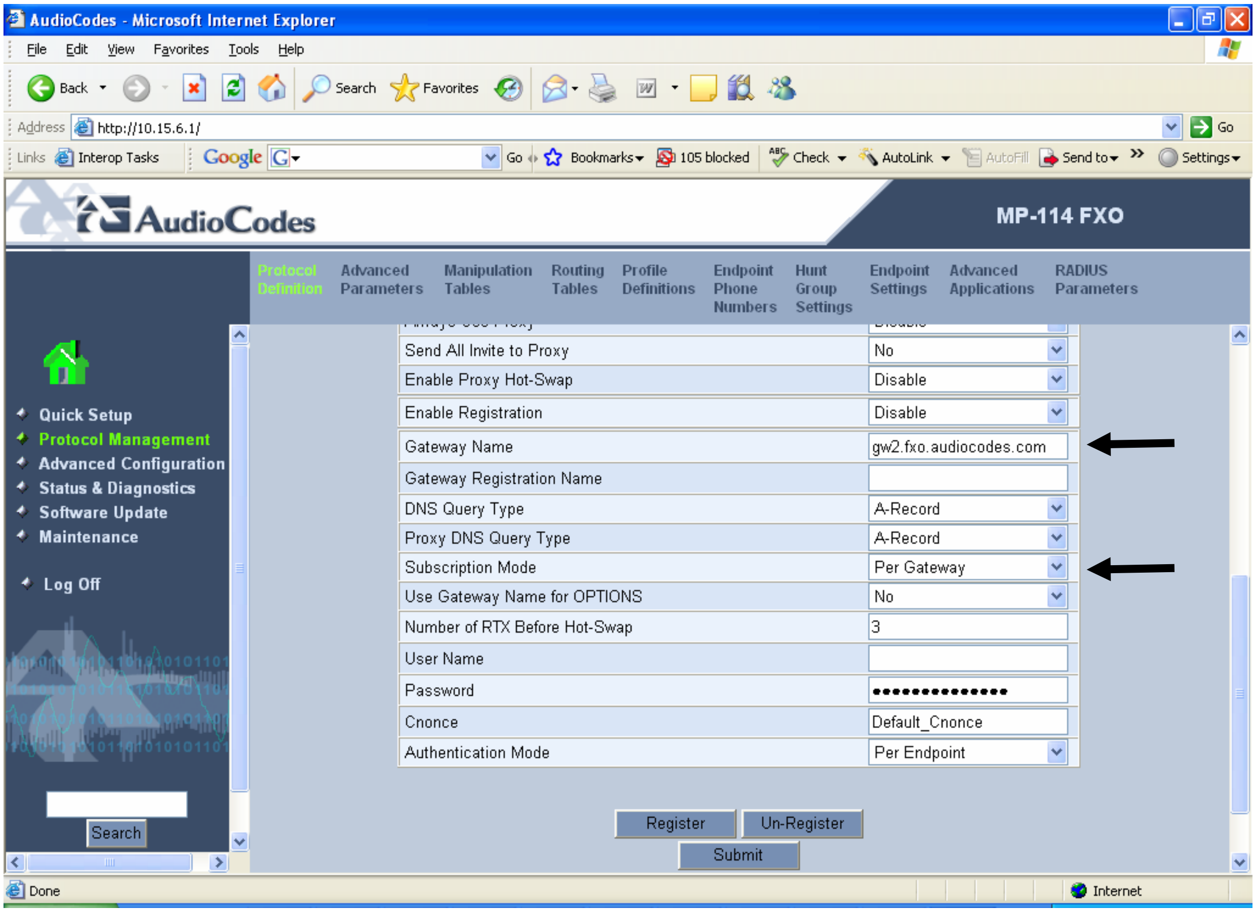
4.2. TLS Setup

Step 1: PBX to IP Routing Setup



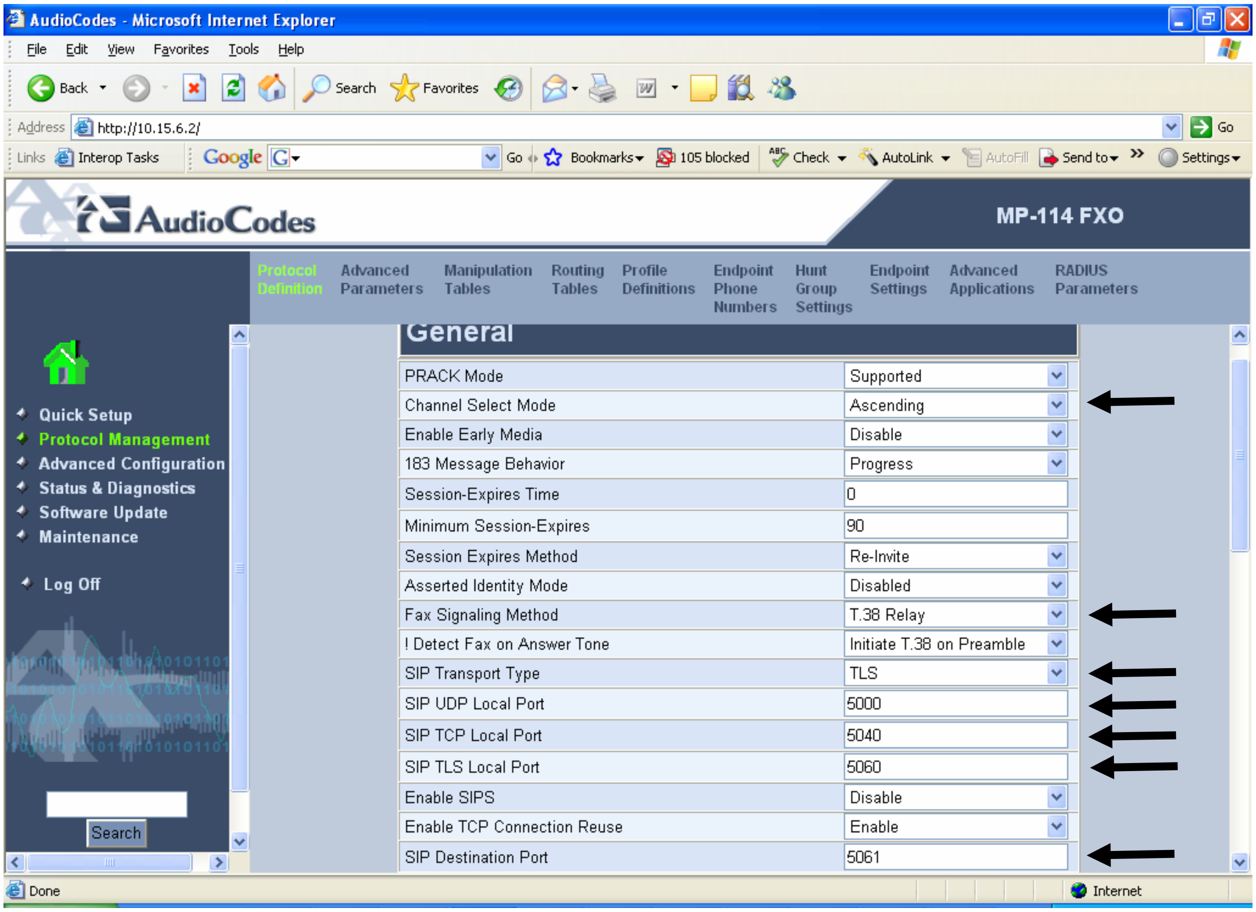
Note: The Proxy IP Address and Name must be one that corresponds to the network environment in which the Microsoft Unified Messaging server is installed (For example, 10.15.3.207 for IP Address and exchaneg2007.com for the FQDN of the Microsoft Unified Messaging host).

Step 2: SIP Environment and Gateway Name Setup

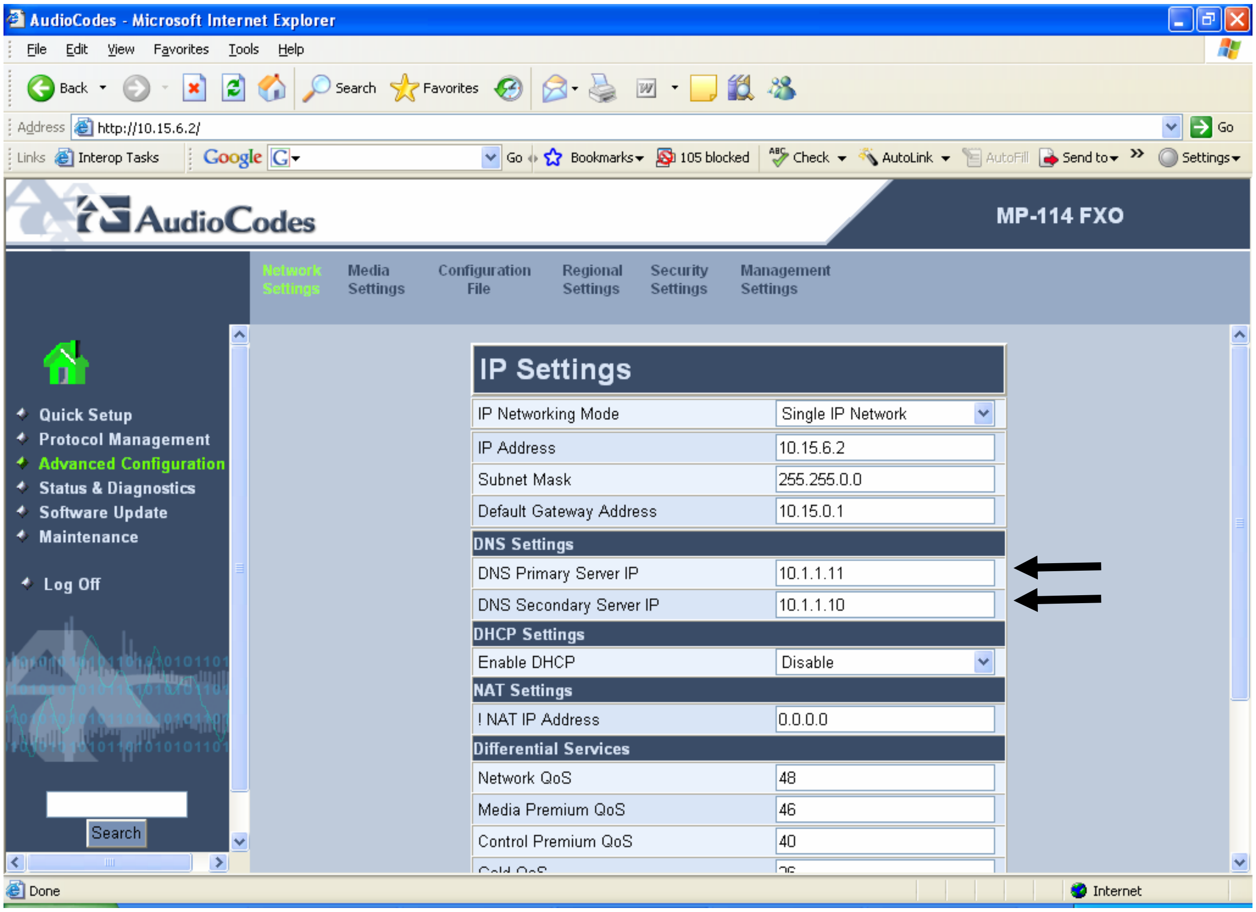


Note: Assign an FQDN name to the gateway (for example, gw2.fxoaudiocodes.com). Any gateway name that corresponds to your network environment is applicable; the only limitation is not to include underscores in the name (Windows Certification server limitation).

Step 3: SIP Environment Setup (Cont.)

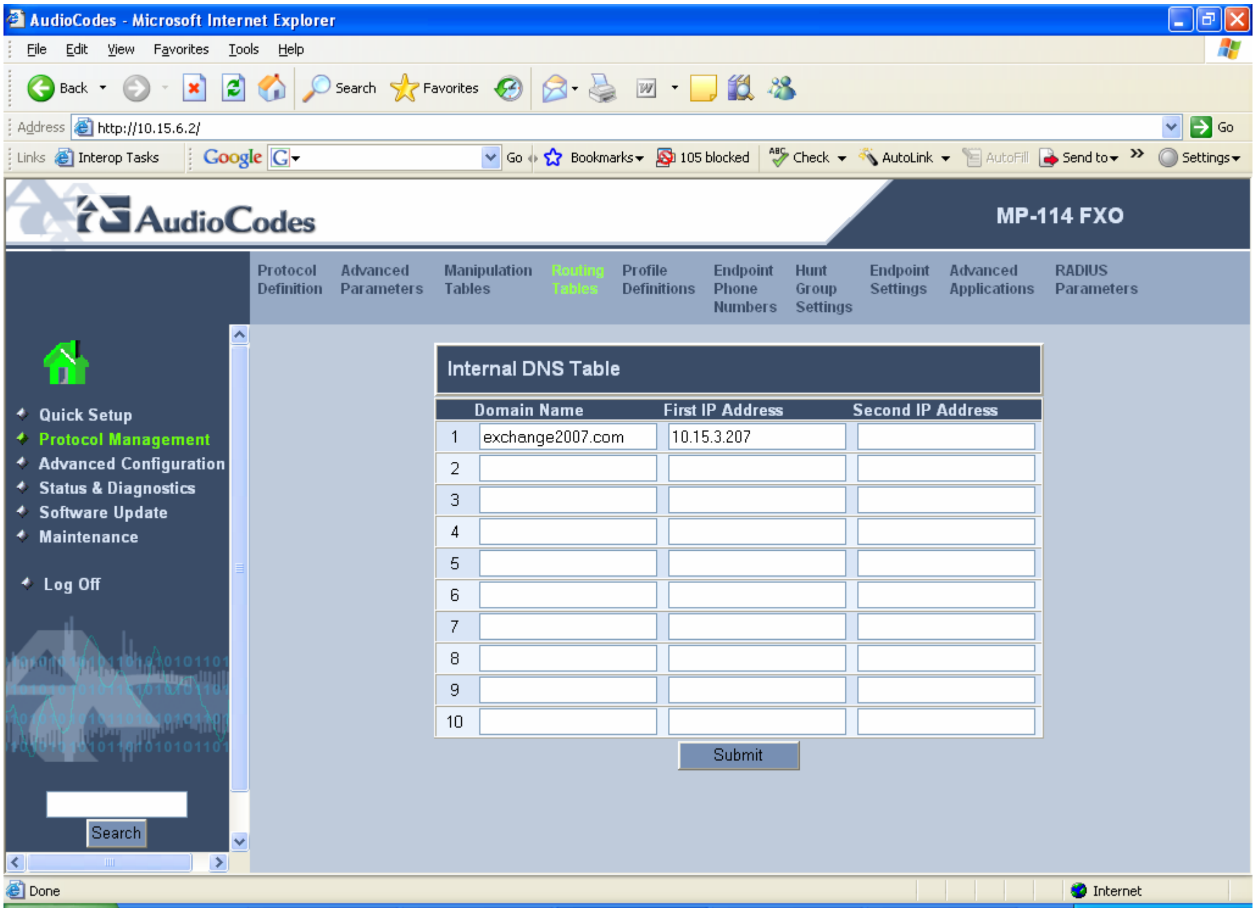


Step 4: DNS Servers Setup



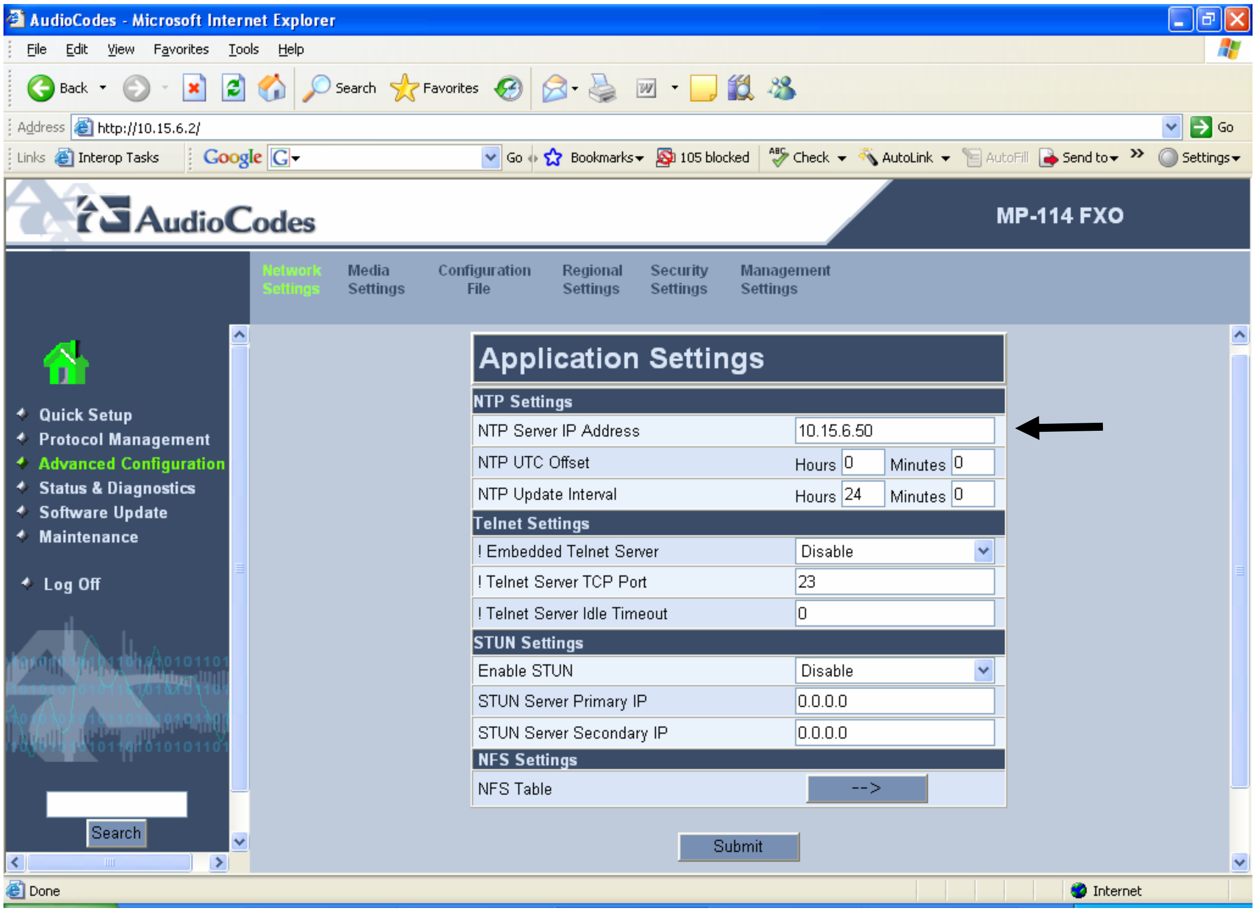
Note: Define the primary and secondary DNS servers' IP addresses so that they correspond to your network environment (for example, 10.1.1.11 and 10.1.1.10). If no DNS server is available in the network, then skip this step.

Step 5: Internal DNS Setup



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).

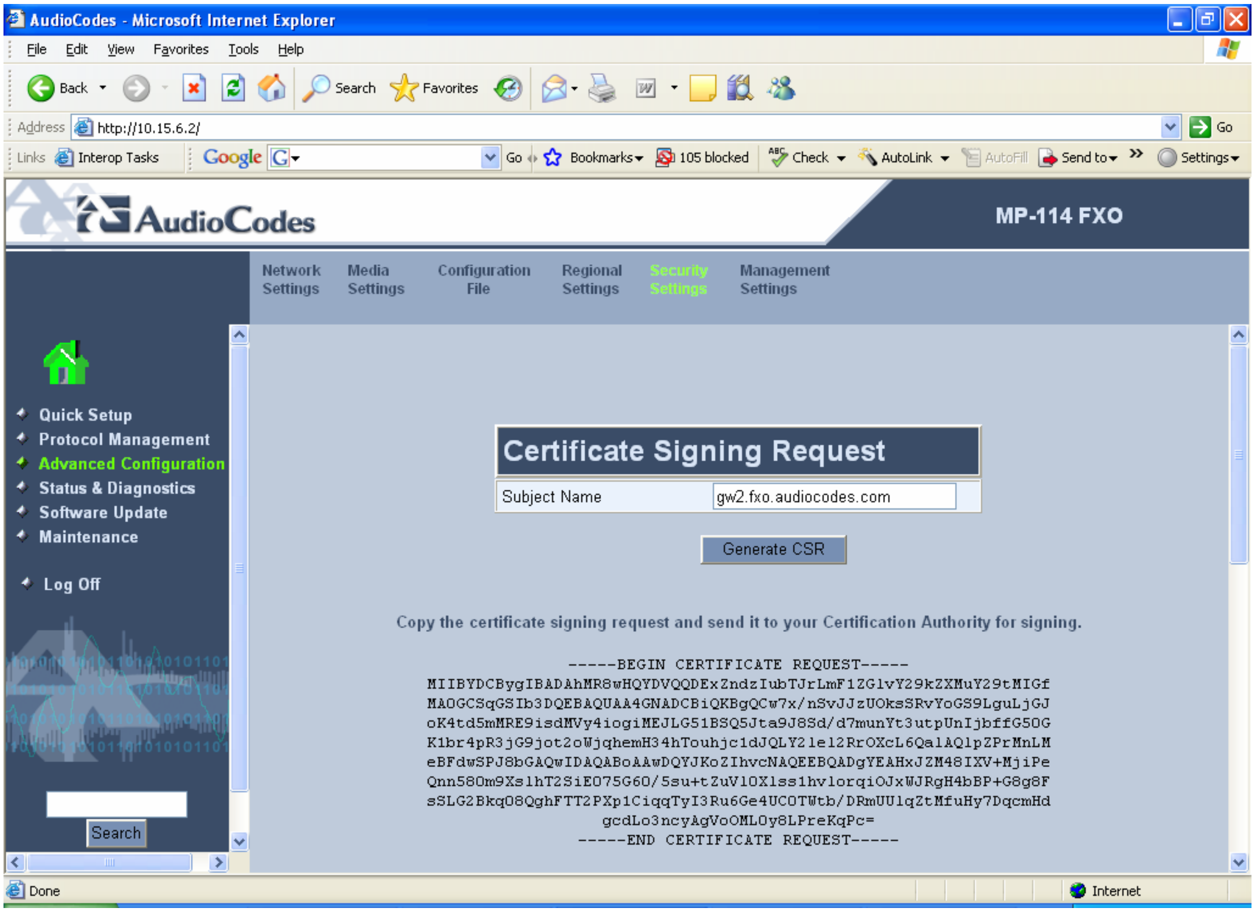
Step 6: NTP Server Setup



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.

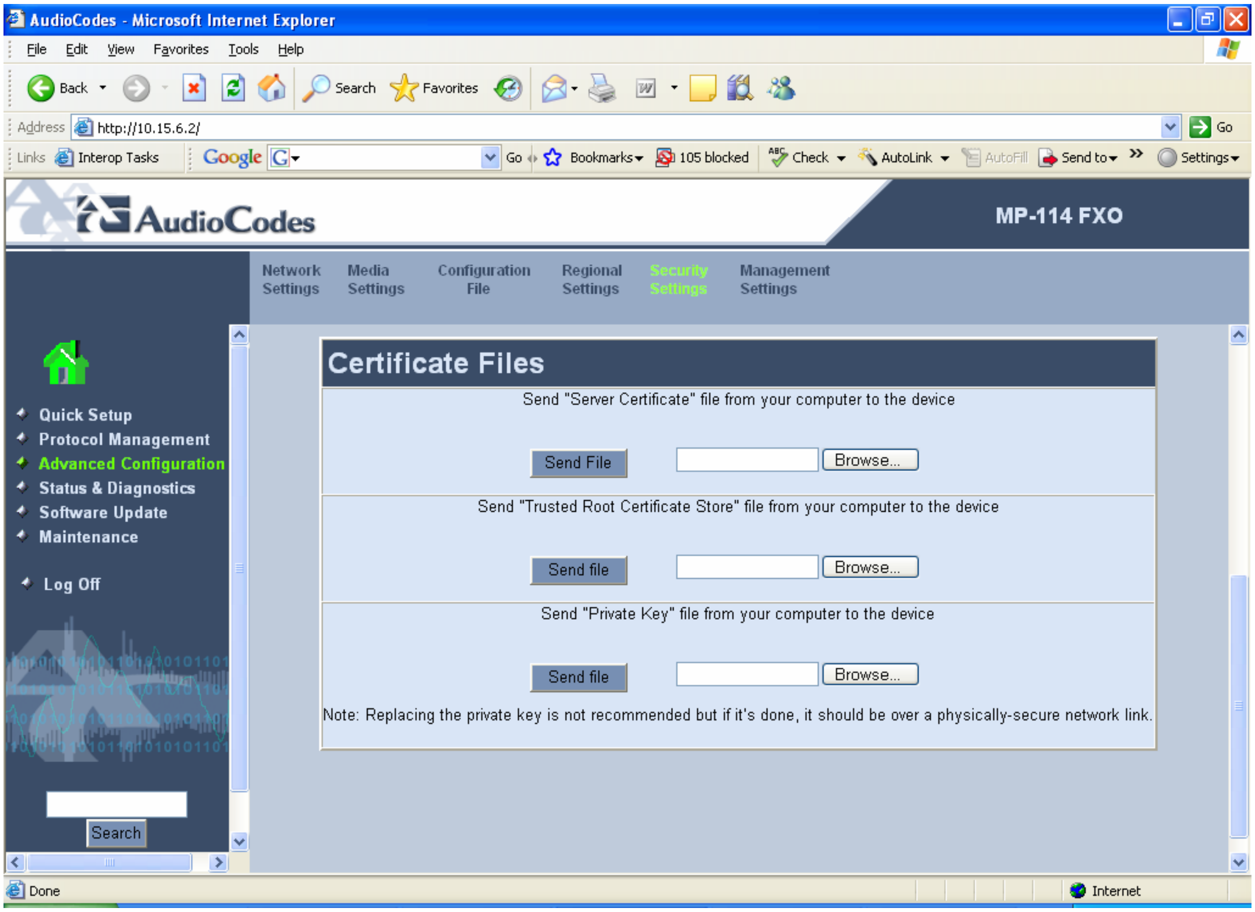


Step 8: Uploading Certificates Setup

The screen below is used to upload the sign certificates.

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

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



5. PBX Setup Notes

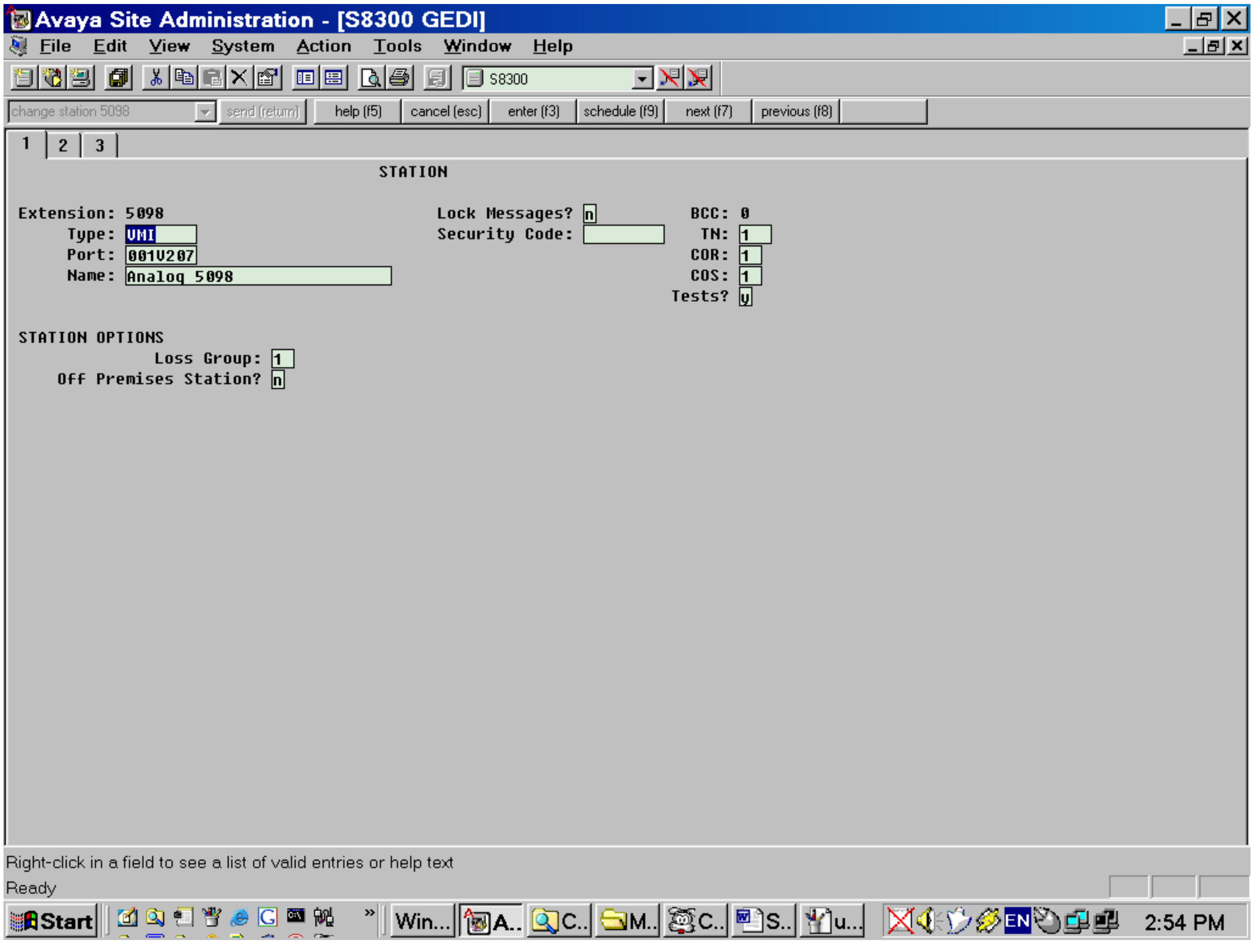
Information used for this test case:

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

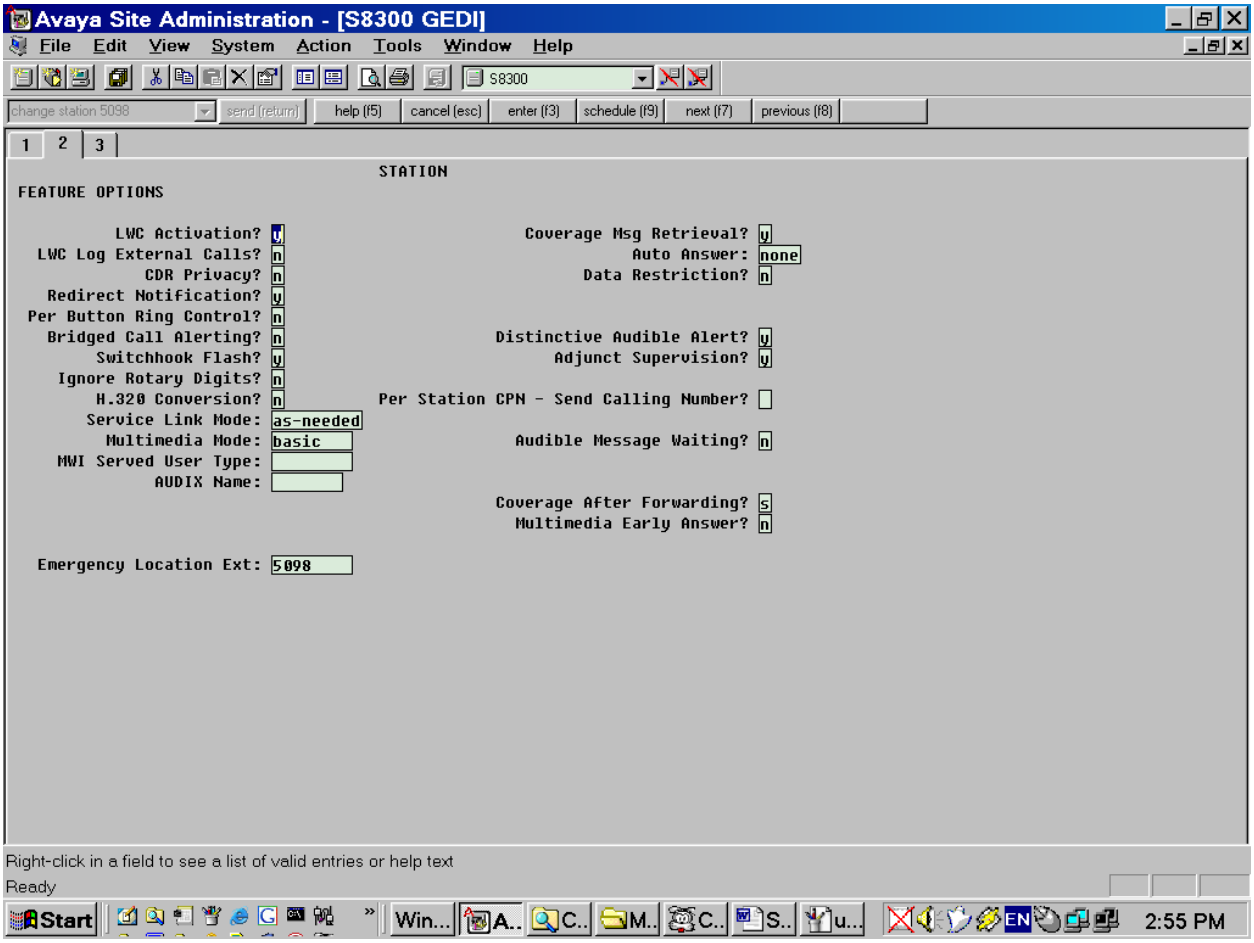
Step 1: Define analog voice mail ports:

"add station" or "cha station" command:

Page 1: Set port type as VMI



Page 2: Set adjunct supervision to Y

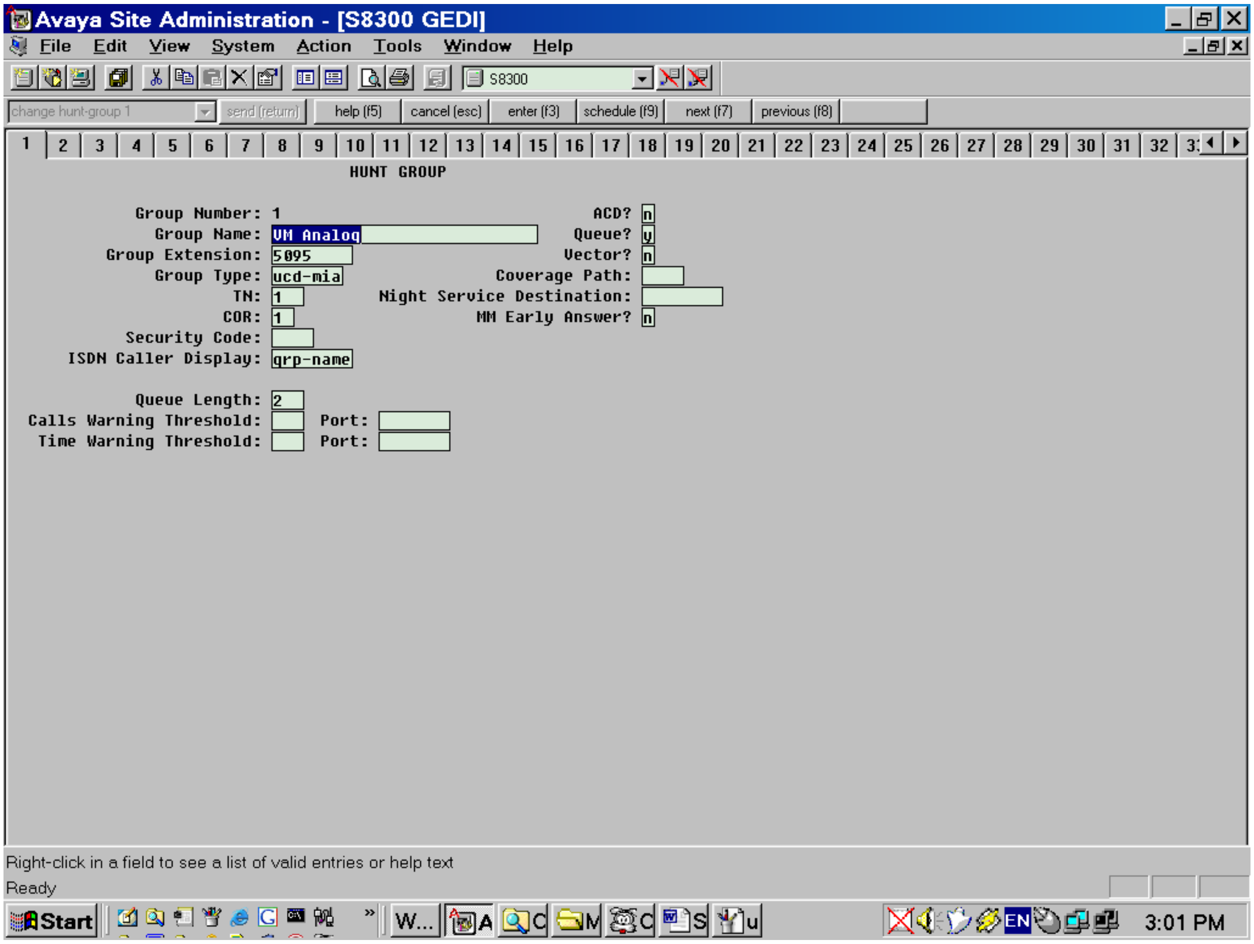


Repeat this process for all analog VoiceMail ports.

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

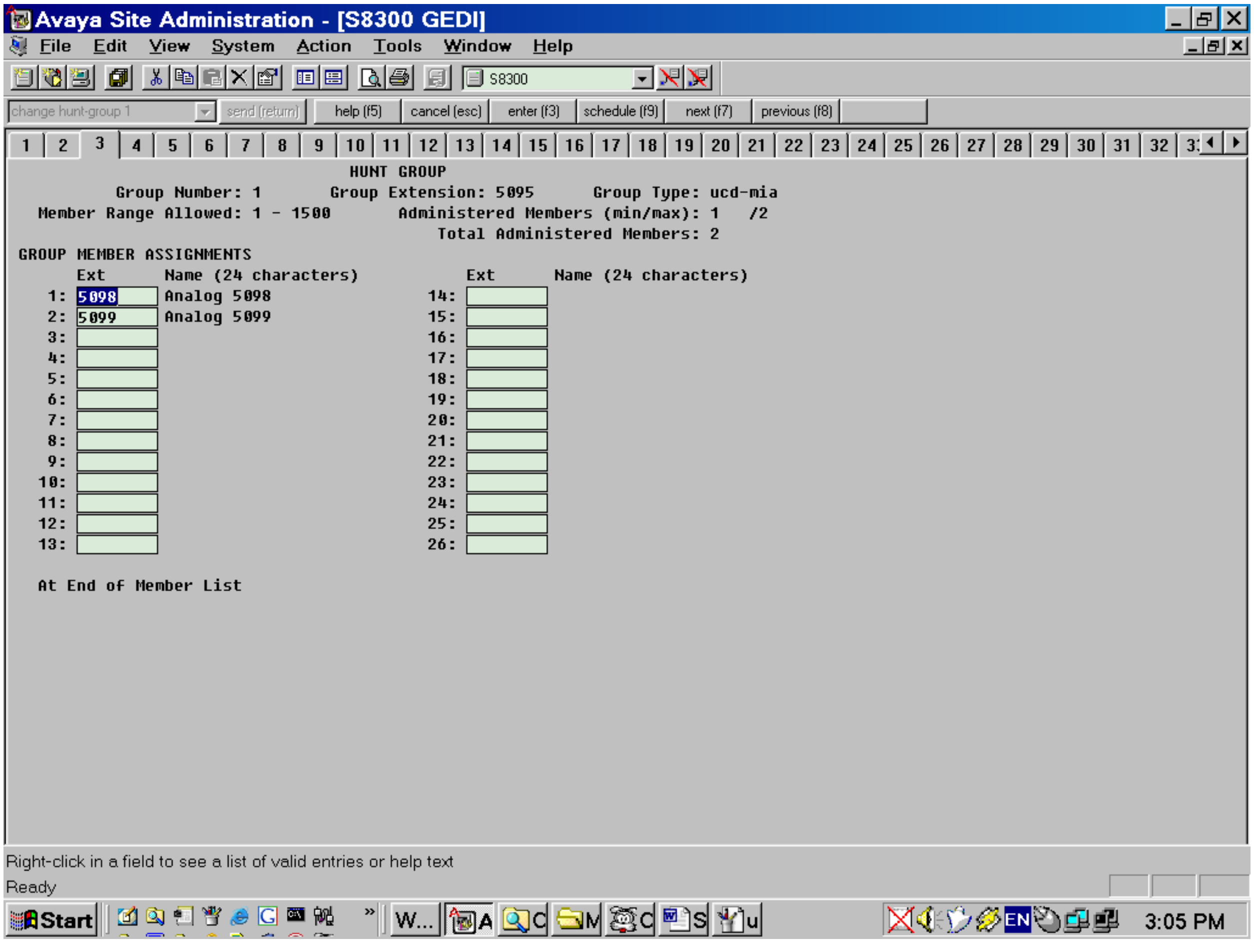
"add hunt-group" or "cha hunt-group" command

Page 1: Set group type as UCD-MIA



Add analog voice mail ports to this hunt group.

Page 3.

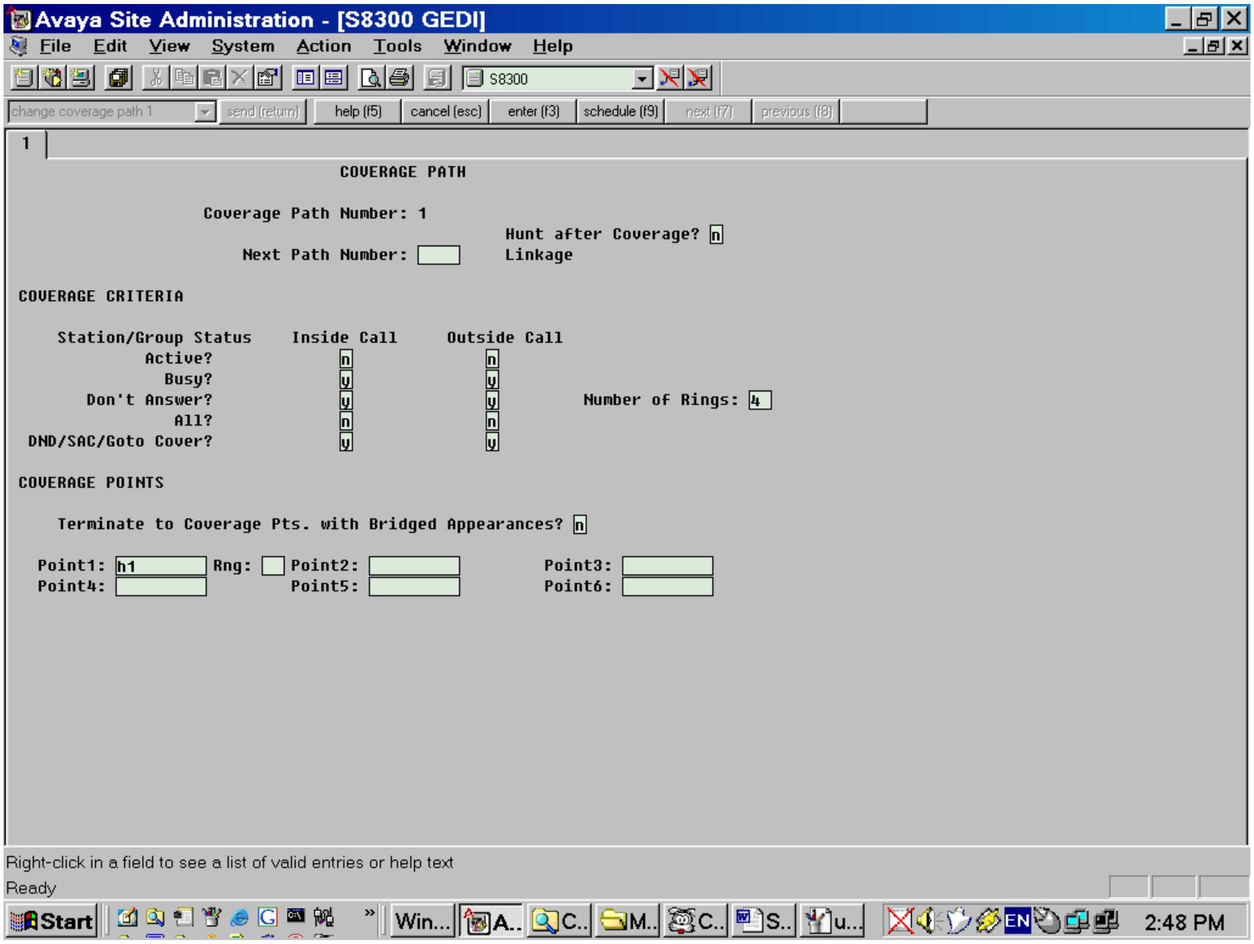


Step 3: Set the coverage path

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

Point to the analog voice mail hunt group 5095

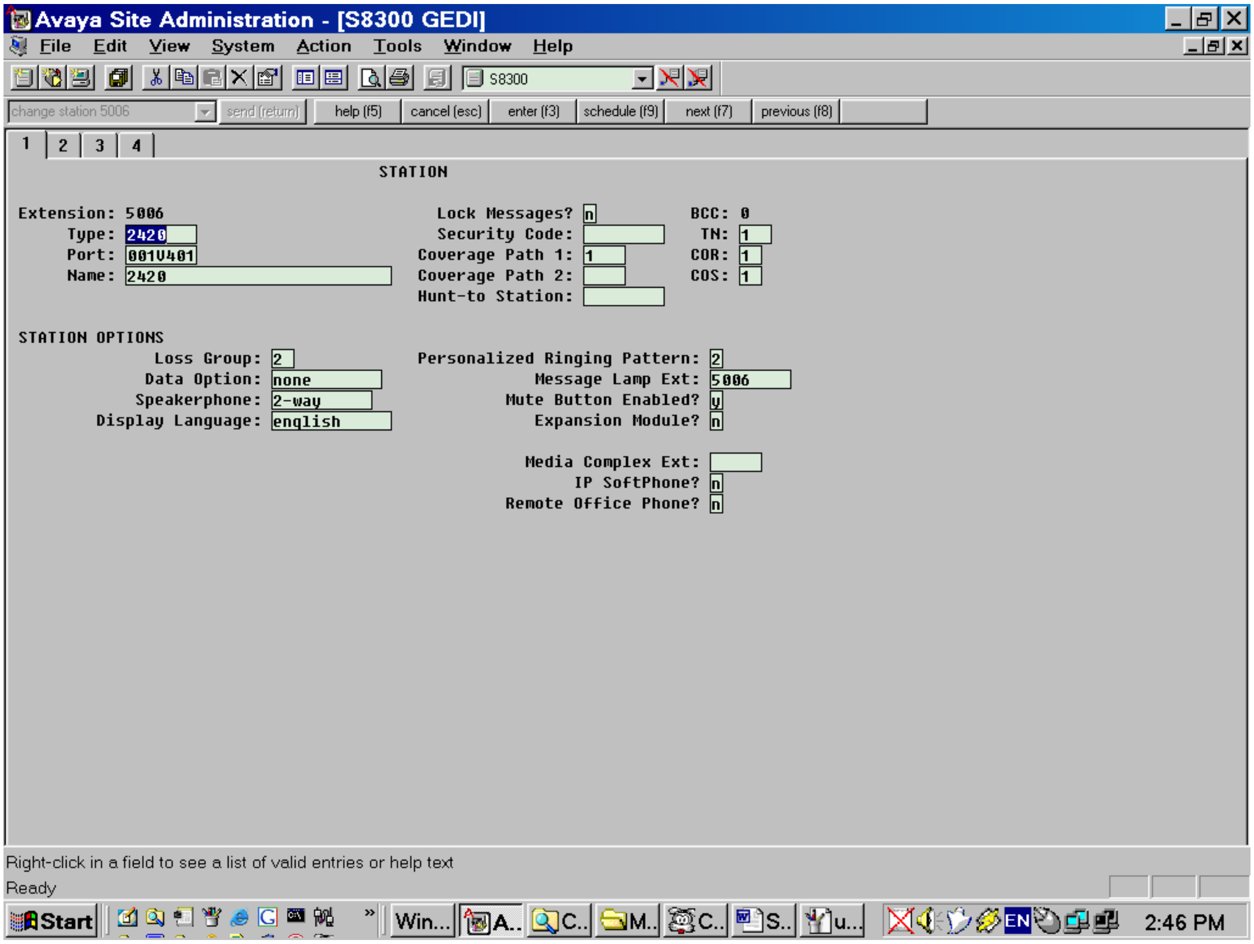
H1 in point1



Step 4: Add or change station coverage path:

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

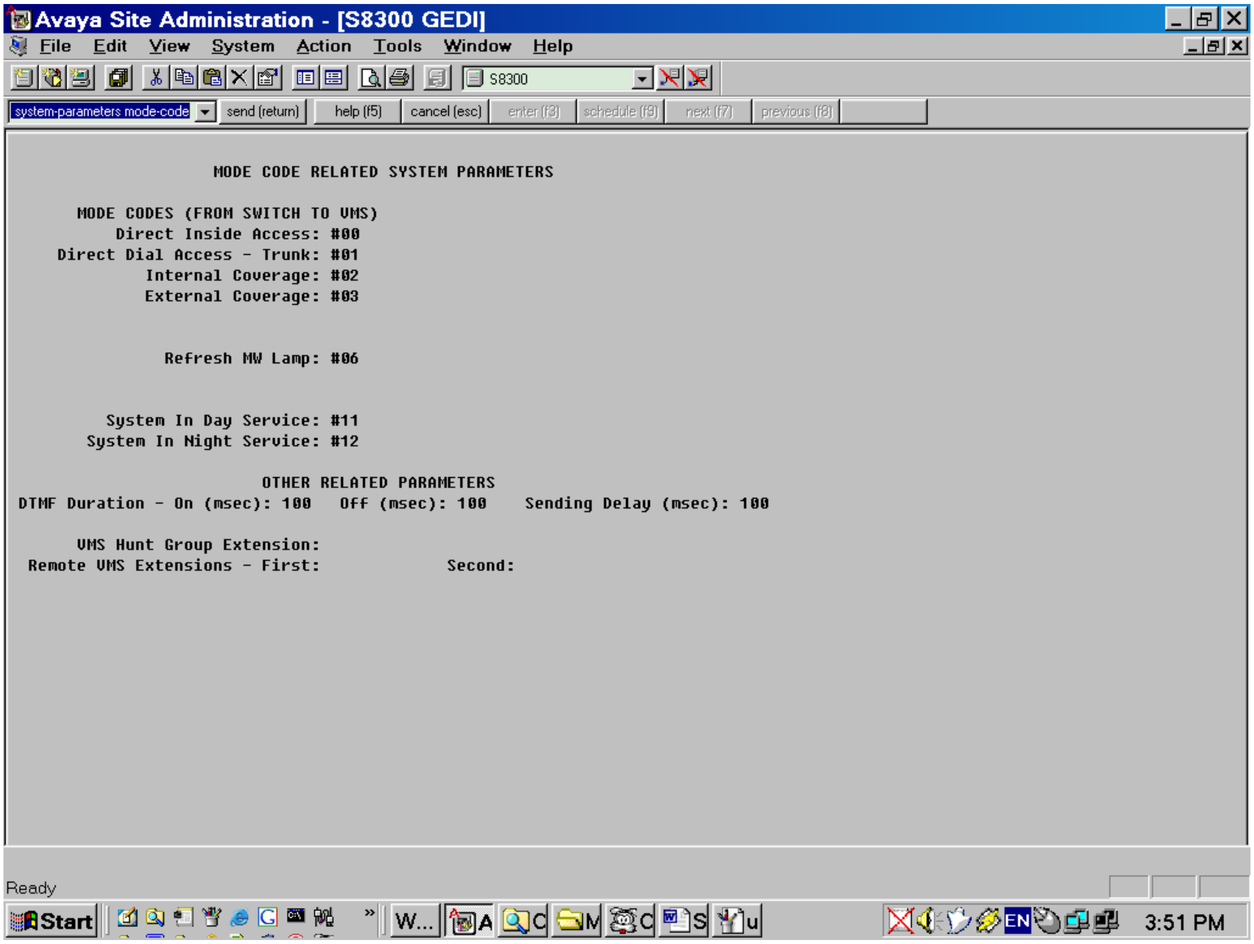
Use "add station" or "change station" command



Step 5: Display mode-code information.

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

Use "display system-parameters mode-code" command

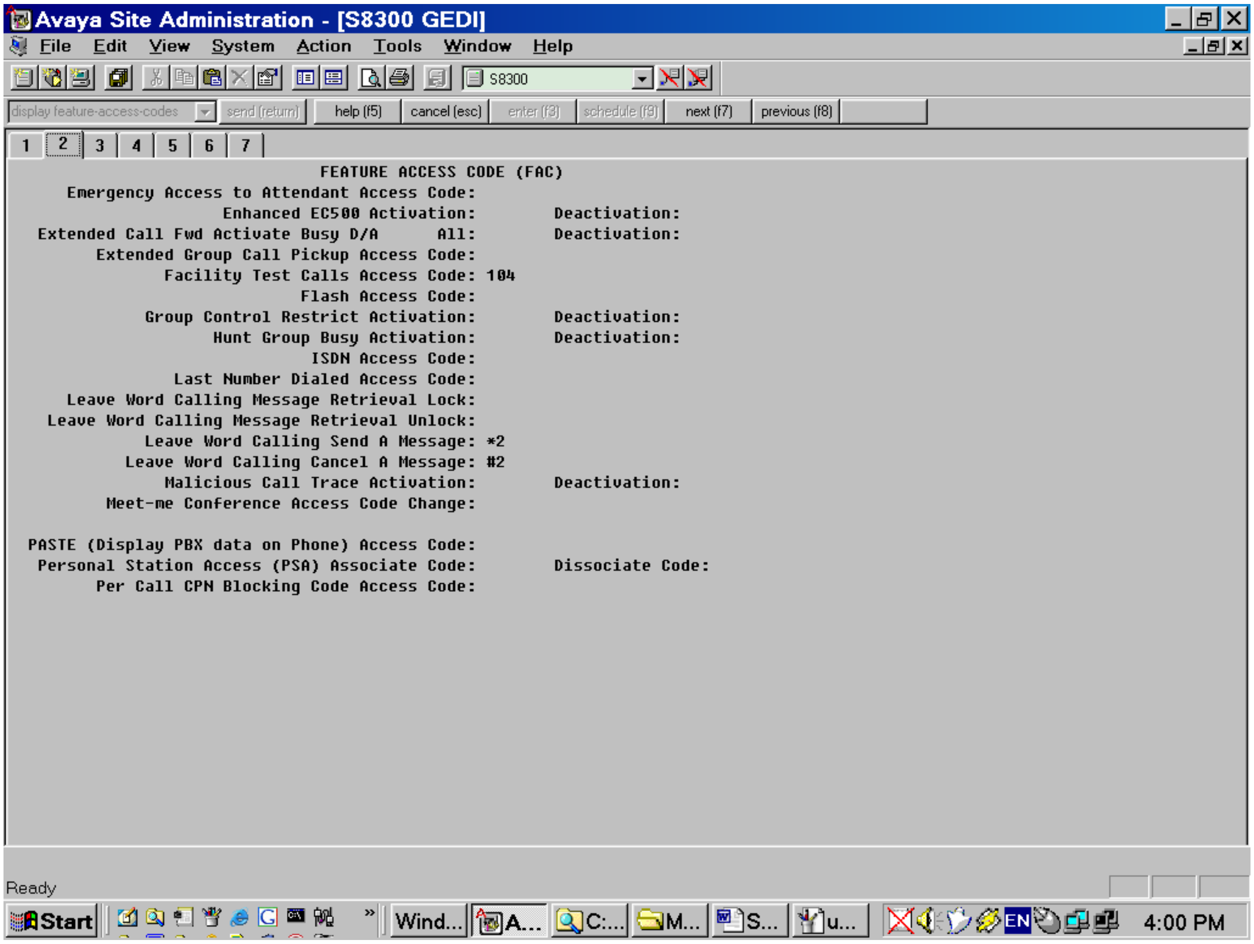


Step 6: Verify and change if needed MWI on & off codes.

Use "display feature-access-codes" command

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

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



5.1. TLS Setup

- N/A.

5.2. Fail-Over Configuration

- N/A.

5.3. Tested Phones

Avaya 2420 Digital phone.

5.4. Other Comments

None.

6. Exchange 2007 UM Validation Test Matrix

The following table contains a set of tests for assessing the functionality of the UM core feature set. The results are recorded as either:

- Pass (**P**)
- Conditional Pass (**CP**)
- Fail (**F**)
- Not Tested (**NT**)
- Not Applicable (**NA**)

Refer to:

- Appendix for a more detailed description of how to perform each call scenario.
- Section 6.1 for detailed descriptions of call scenario failures, if any.

No.	**Call Scenarios (see appendix for more detailed instructions)**	(**P**/**CP**/**F**/**NT**)	**Reason for Failure (see 6.1 for more detailed descriptions)**
1	Dial the pilot number from a phone extension that is NOT enabled for Unified Messaging and logon to a user's mailbox. Confirm hearing the prompt: "Welcome, you are connected to Microsoft Exchange. To access your mailbox, enter your extension…"	**P**	
2	Navigate mailbox using the Voice User Interface (VUI).	**P**	
3	Navigate mailbox using the Telephony User Interface (TUI).	**P**	
4	Dial user extension and leave a voicemail.		
4a	Dial user extension and leave a voicemail from an internal extension. Confirm the Active Directory name of the calling party is displayed in the sender field of the voicemail message.	**P**	
4b	Dial user extension and leave a voicemail from an external phone. Confirm the correct phone number of the calling party is displayed in the sender field of the voicemail message.	**P**	
5	Dial Auto Attendant (AA).	**P**	

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

	Confirm the FAX is received in the user's inbox.		
10	Setup TLS between gateway/IP-PBX and Exchange UM. Windows Certificate Authority (CA).		
10a	Dial the pilot number and logon to a user's mailbox. Confirm UM answers the call and confirm UM responds to DTMF input.	**P**	
10b	Dial a user extension and leave a voicemail. Confirm the user receives the voicemail.	**P**	
10c	Send a test FAX message to user extension. Confirm the FAX is received in the user's inbox.	**P**	
11	Setup G.723.1 on the gateway. (If already using G.723.1, setup G.711 A Law or G.711 Mu Law for this step). Dial the pilot number and confirm the UM system answers the call.	**P**	
12	Setup Message Waiting Indicator (MWI). Geomant offers a third party solution: MWI 2007. Installation files and product documentation can be found on Geomant's 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	

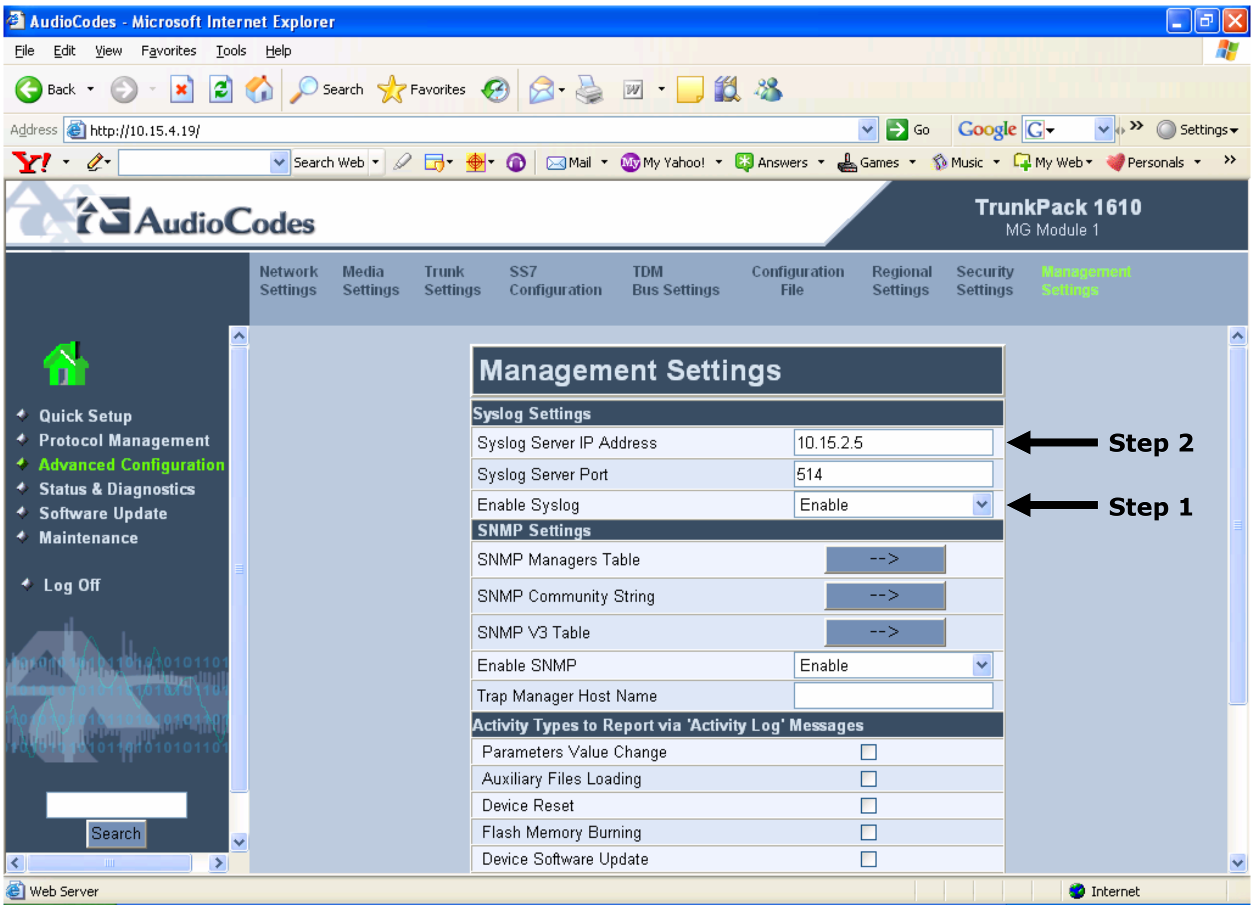
7. Troubleshooting

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.

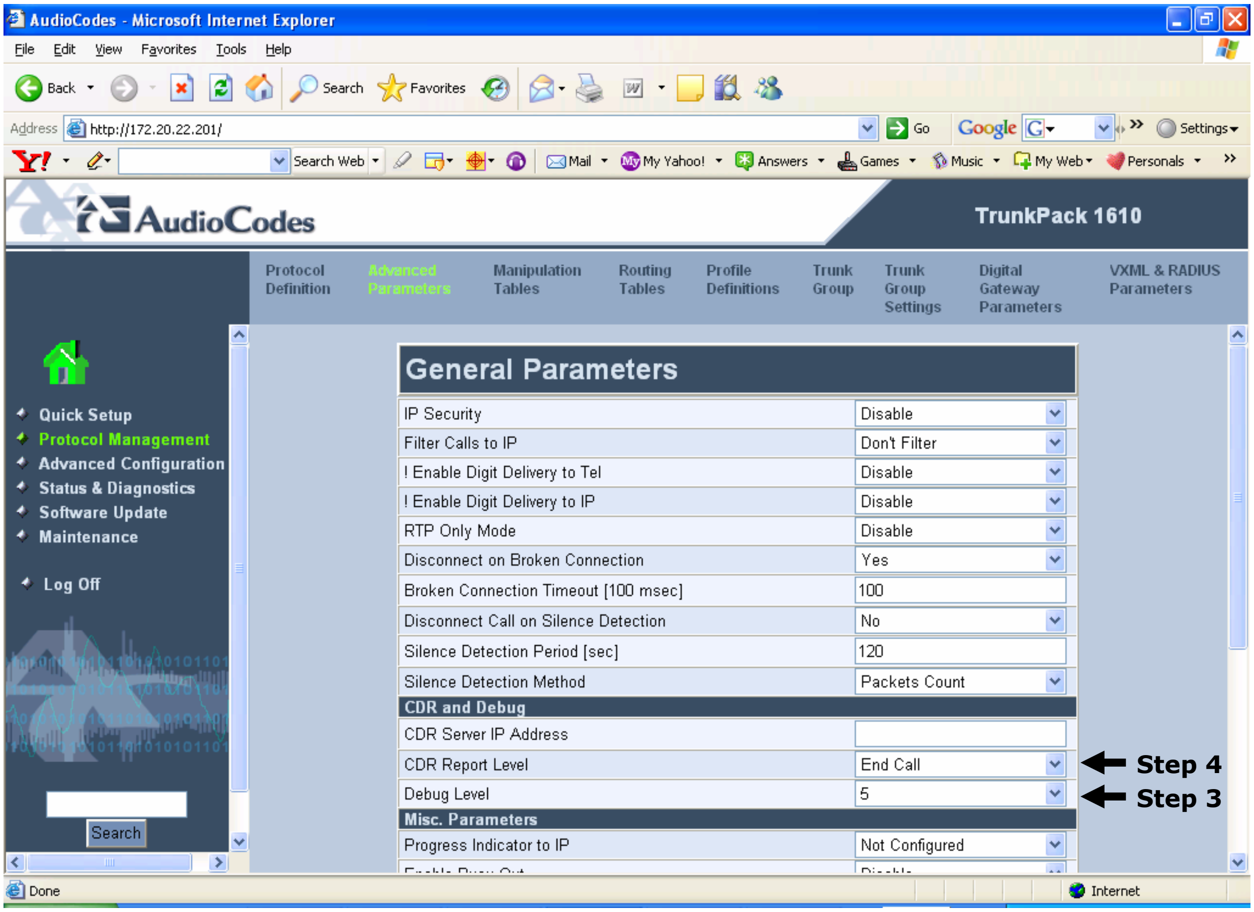
To activate the Syslog client on the AudioCodes gateways:

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.



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.