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

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

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

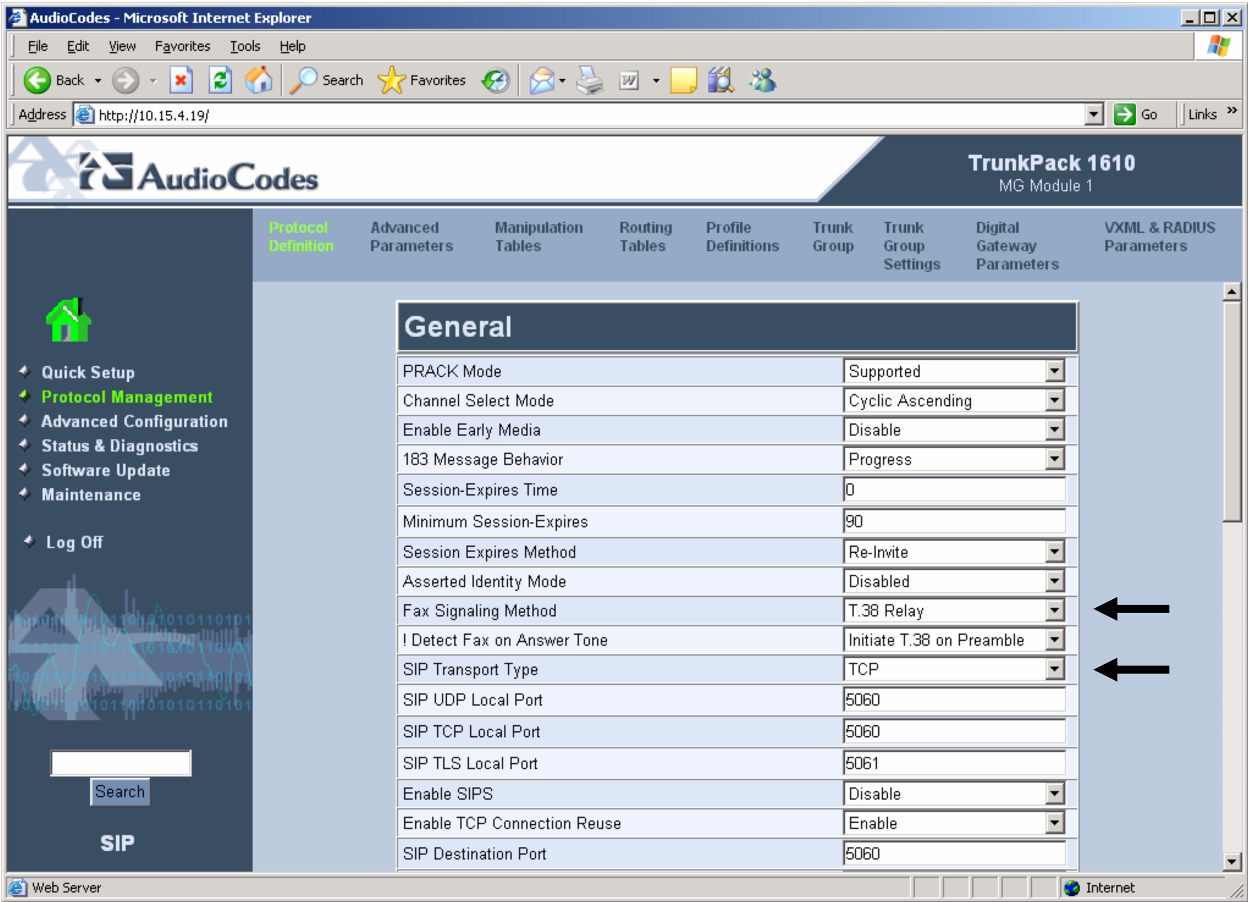
This integration uses cross-over RJ-48c cables *(pairs 1, 2 and 4, 5 crossed)* to connect digital trunks (T1/E1) between and Mediant 2000 trunk interfaces.

3. Summary and Limitations

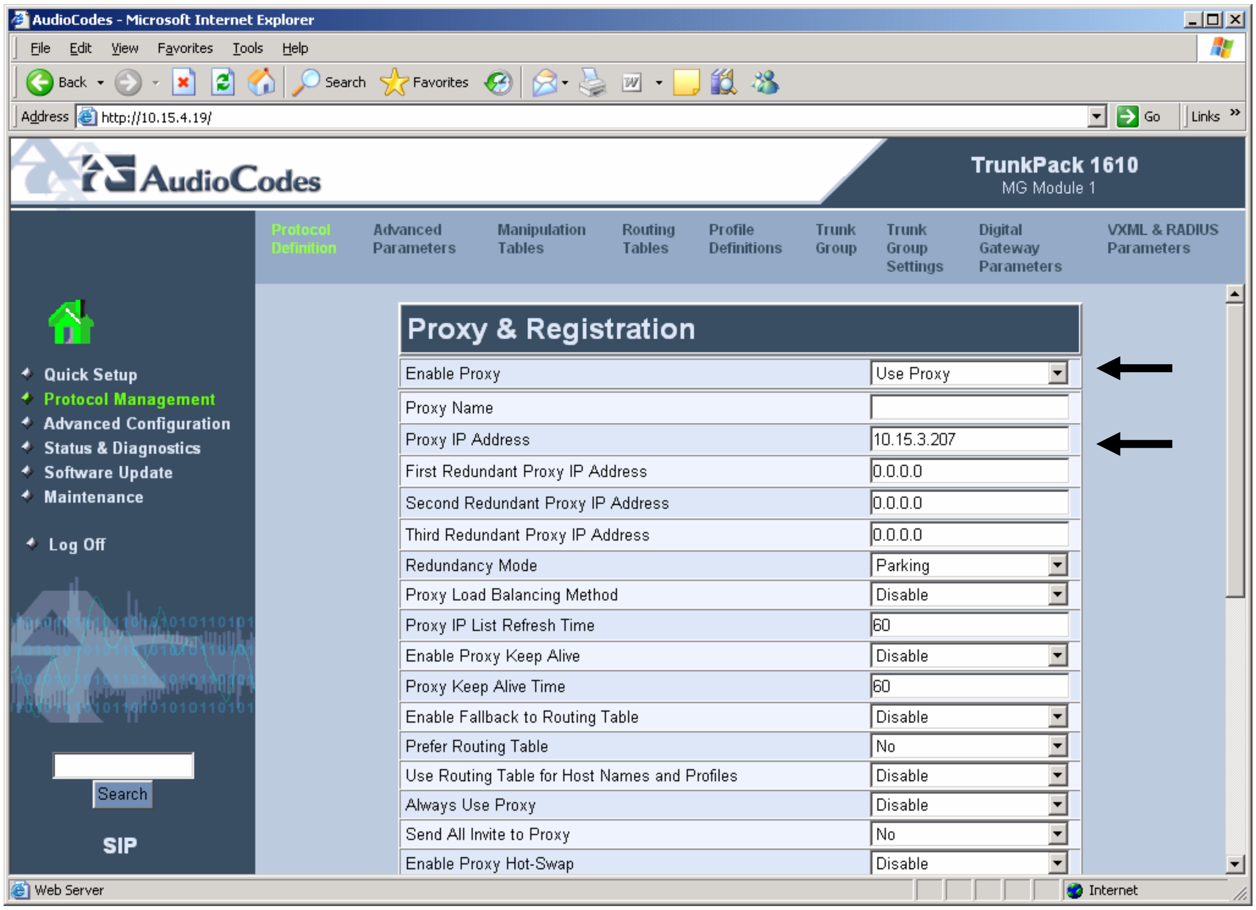
☒ A check in this box indicates the UM feature set is fully functional when using the PBX/gateway in question.

4. Gateway Setup Notes

Step 1: SIP Environment Setup

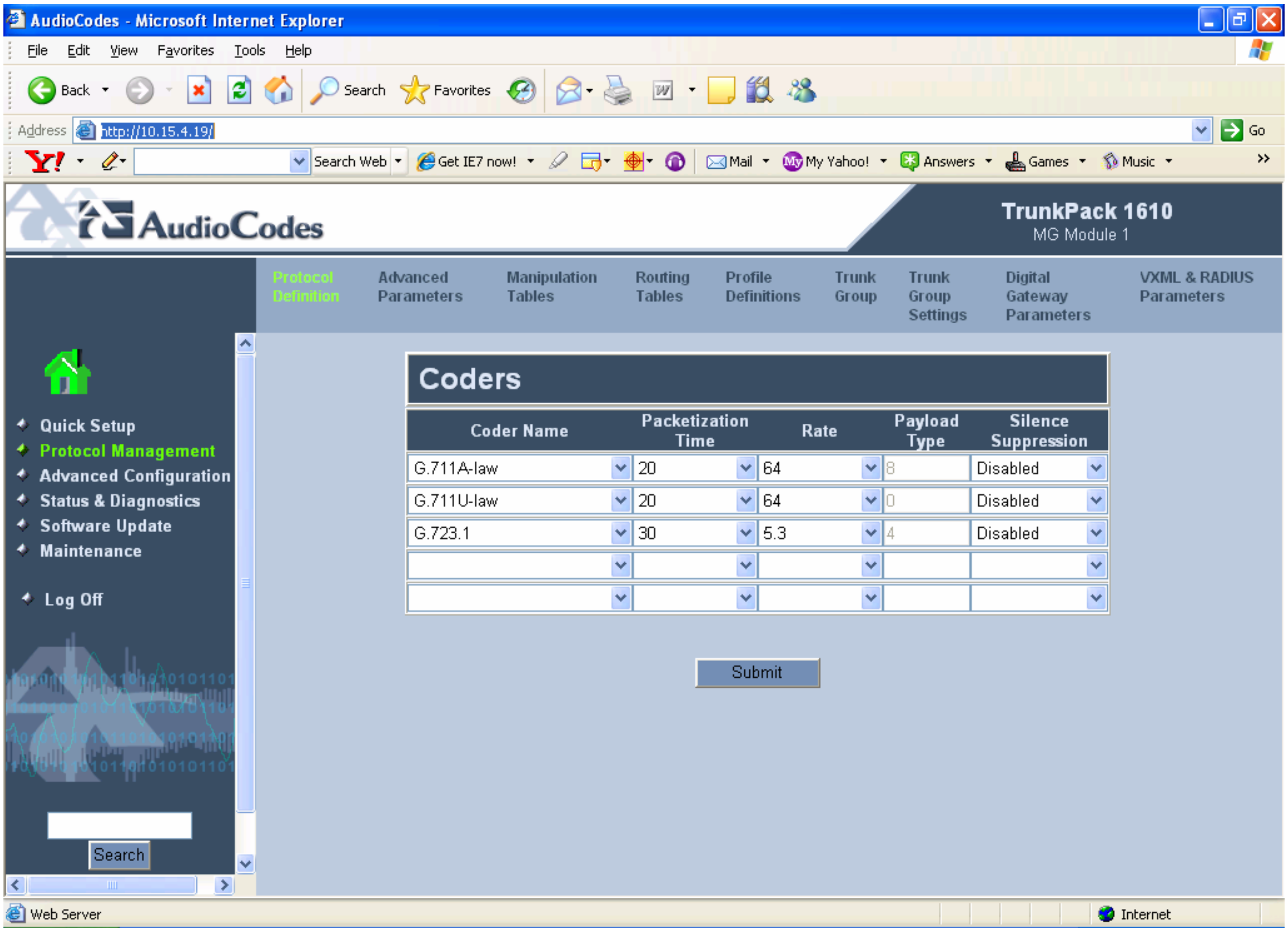


Step 2: Routing Setup

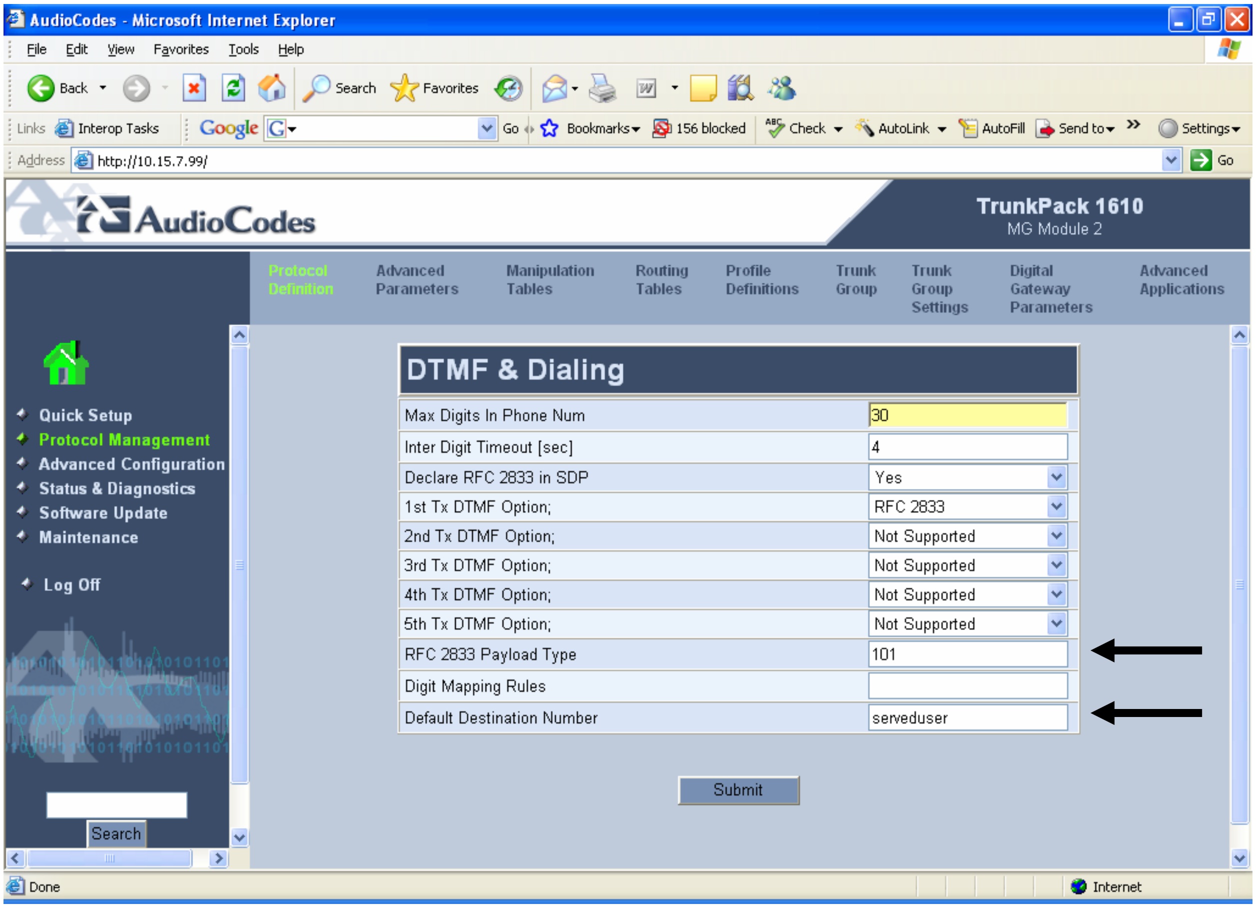


Note: The Proxy IP Address must correspond to the network environment in which the Microsoft Unified Messaging server is installed (For example, 10.15.3.207 or the FQDN of the Microsoft Unified Messaging host).

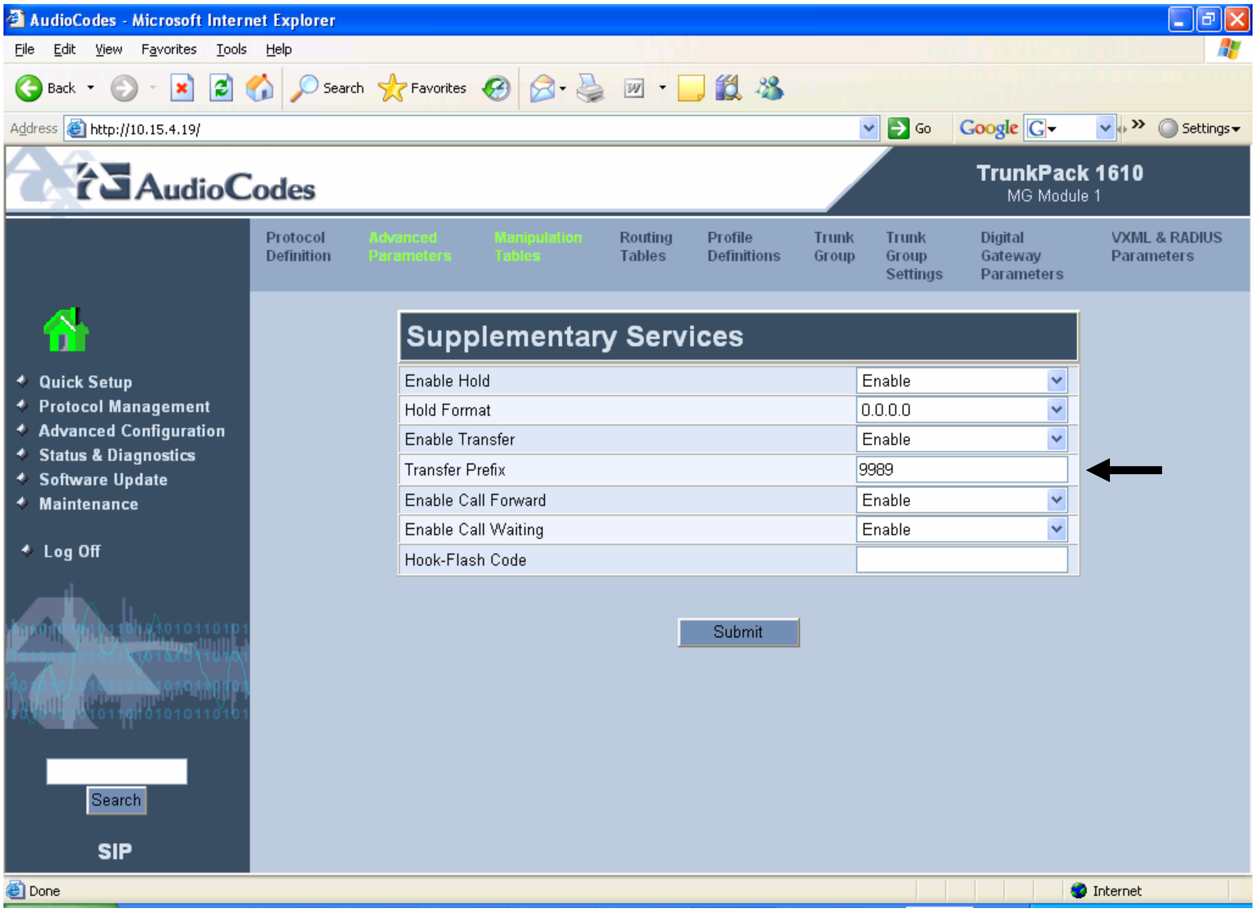
Step 3: Coder Setup



Step 4: Digit Collection Setup

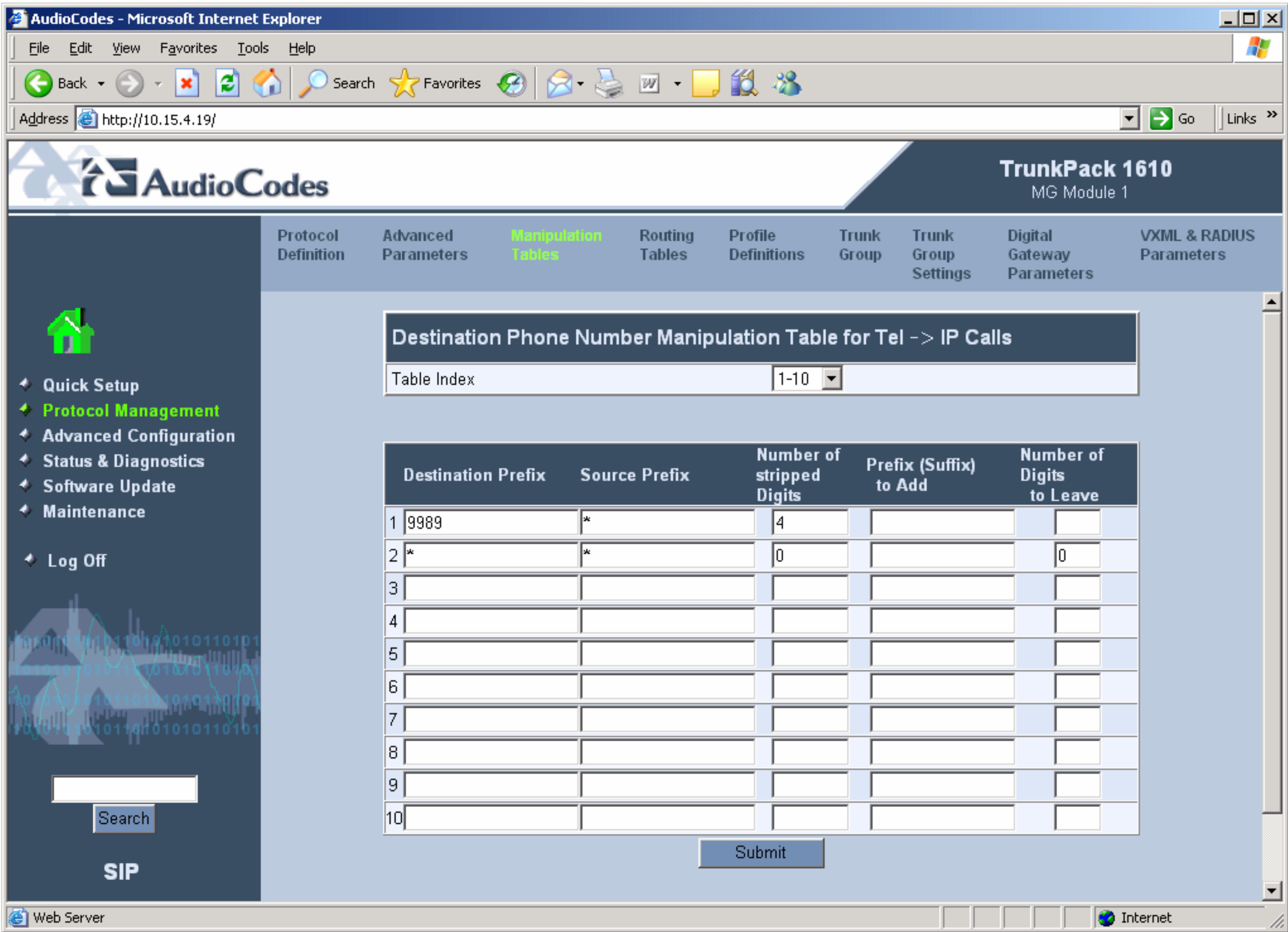


Step 5: Supplementary Services Setting



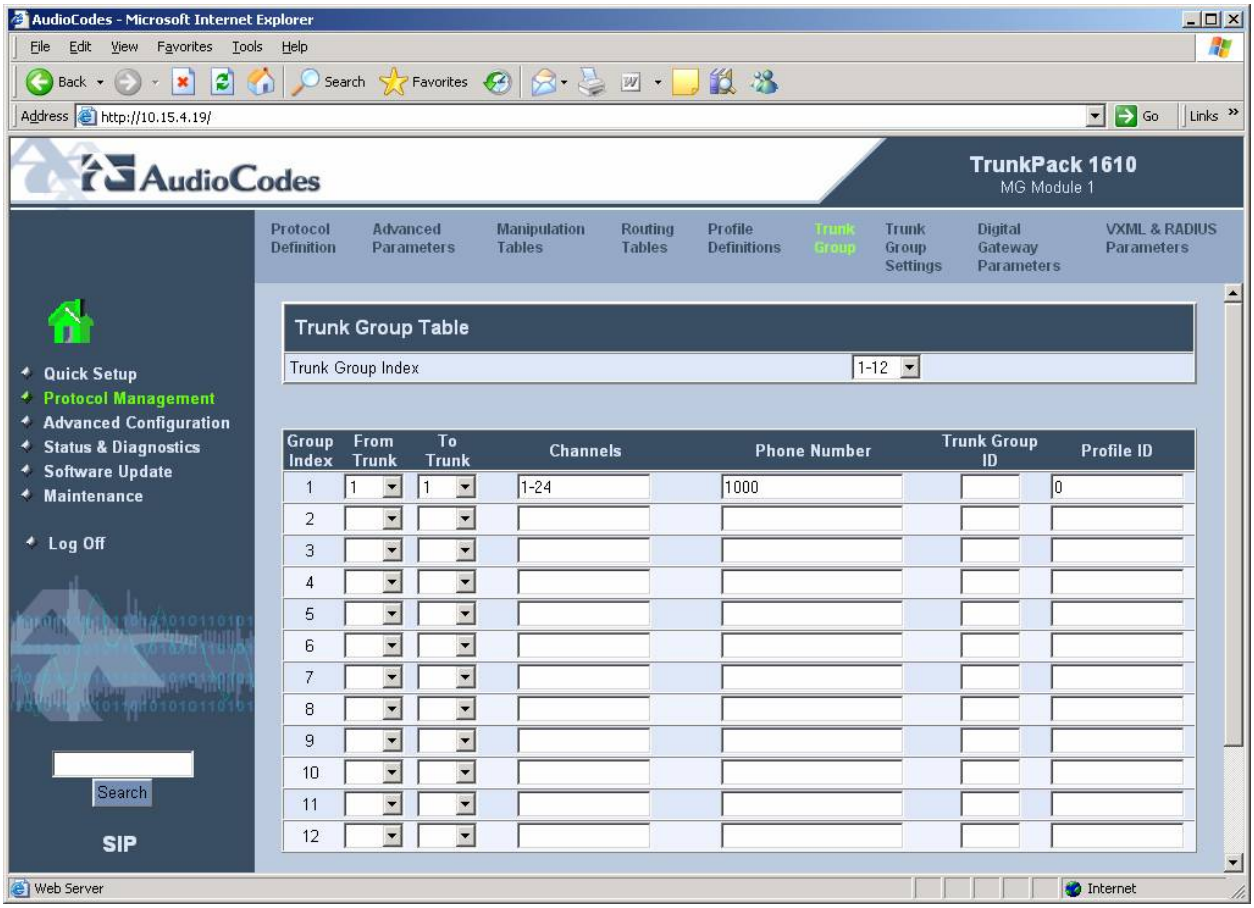
Choose any 4-digit number that is not used in the PBX for Transfer Prefix (e.g., 9989).

Step 6: Manipulation Routing Setup

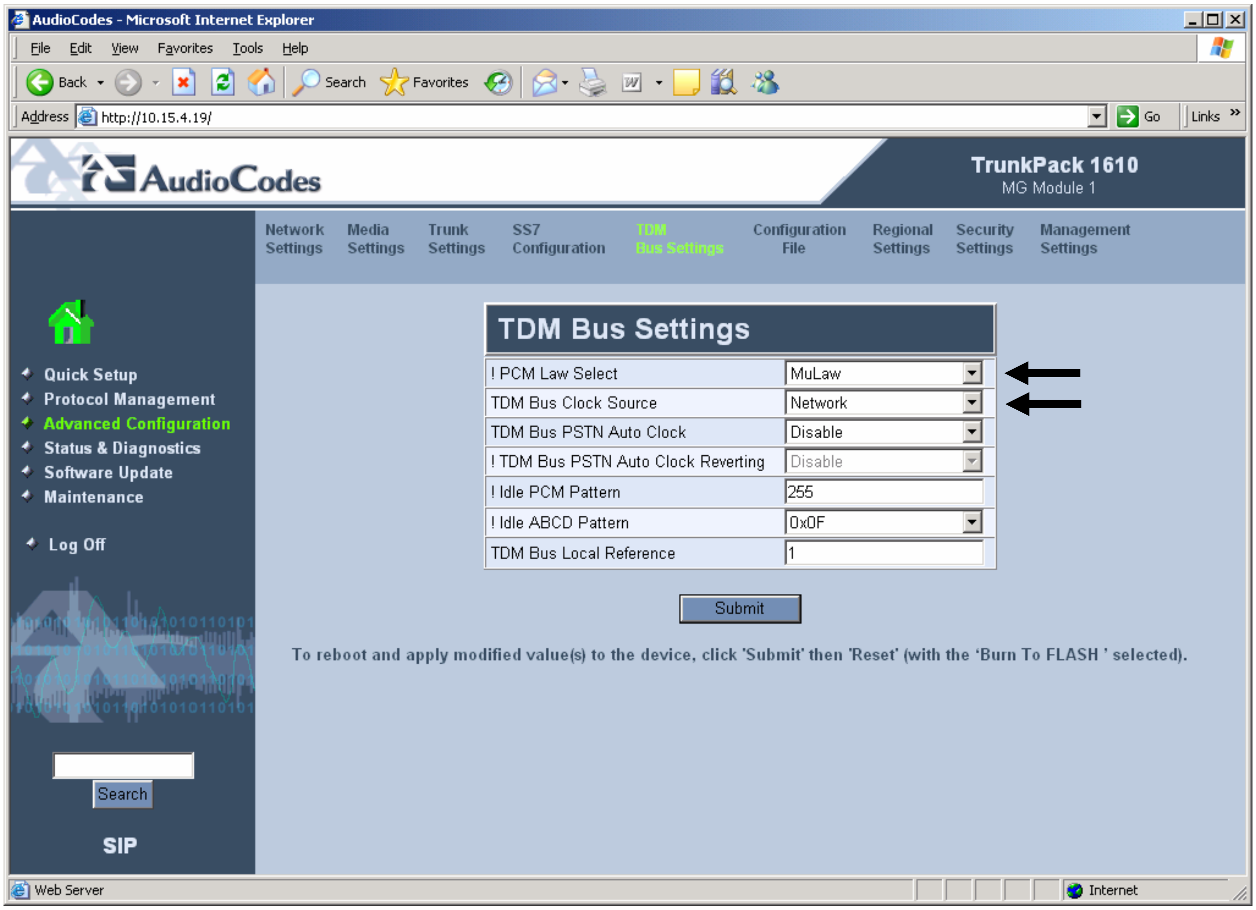


The first manipulation for the destination prefix will strip the 4-digit number of the Transfer Prefix that was configured in the previous step.

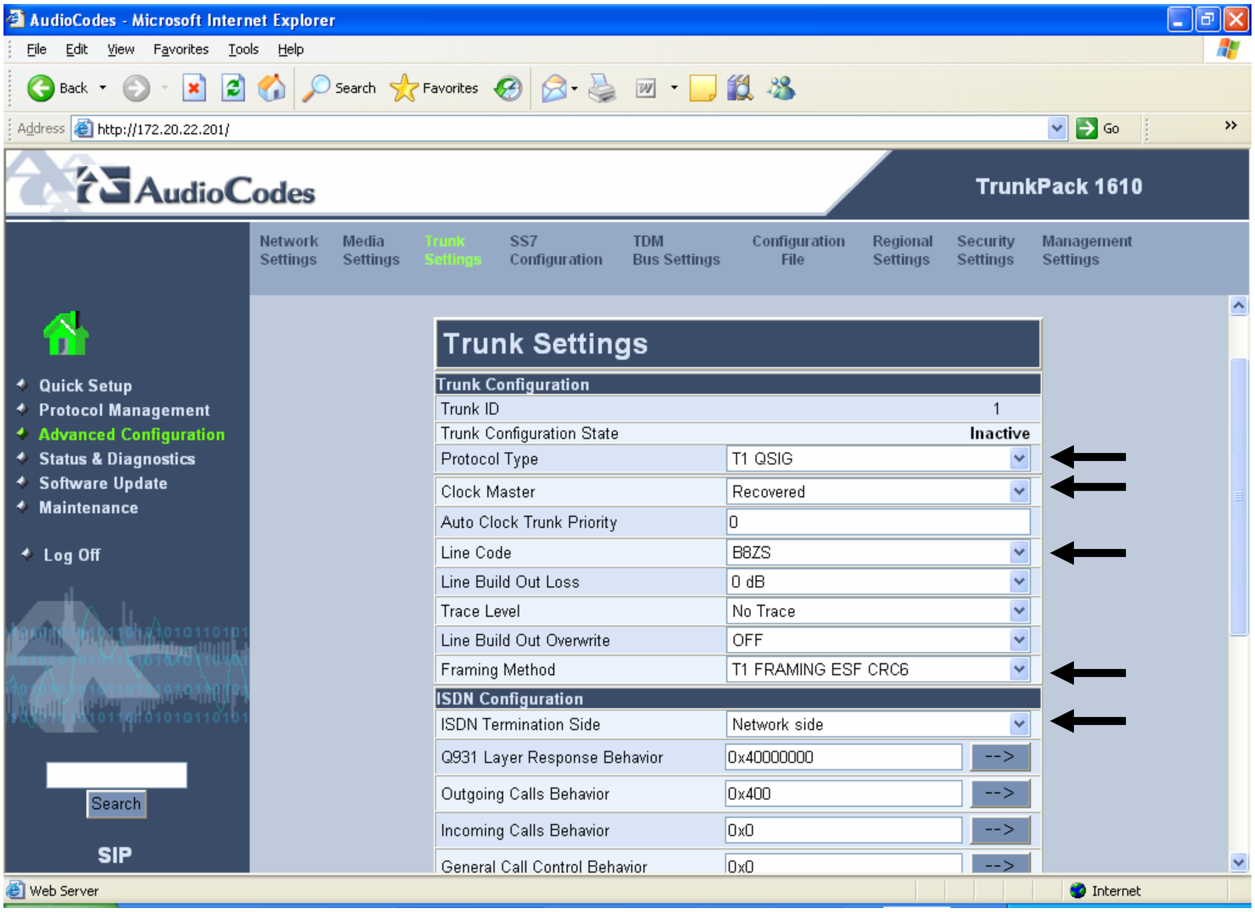
Step 7: Trunk Group Setup



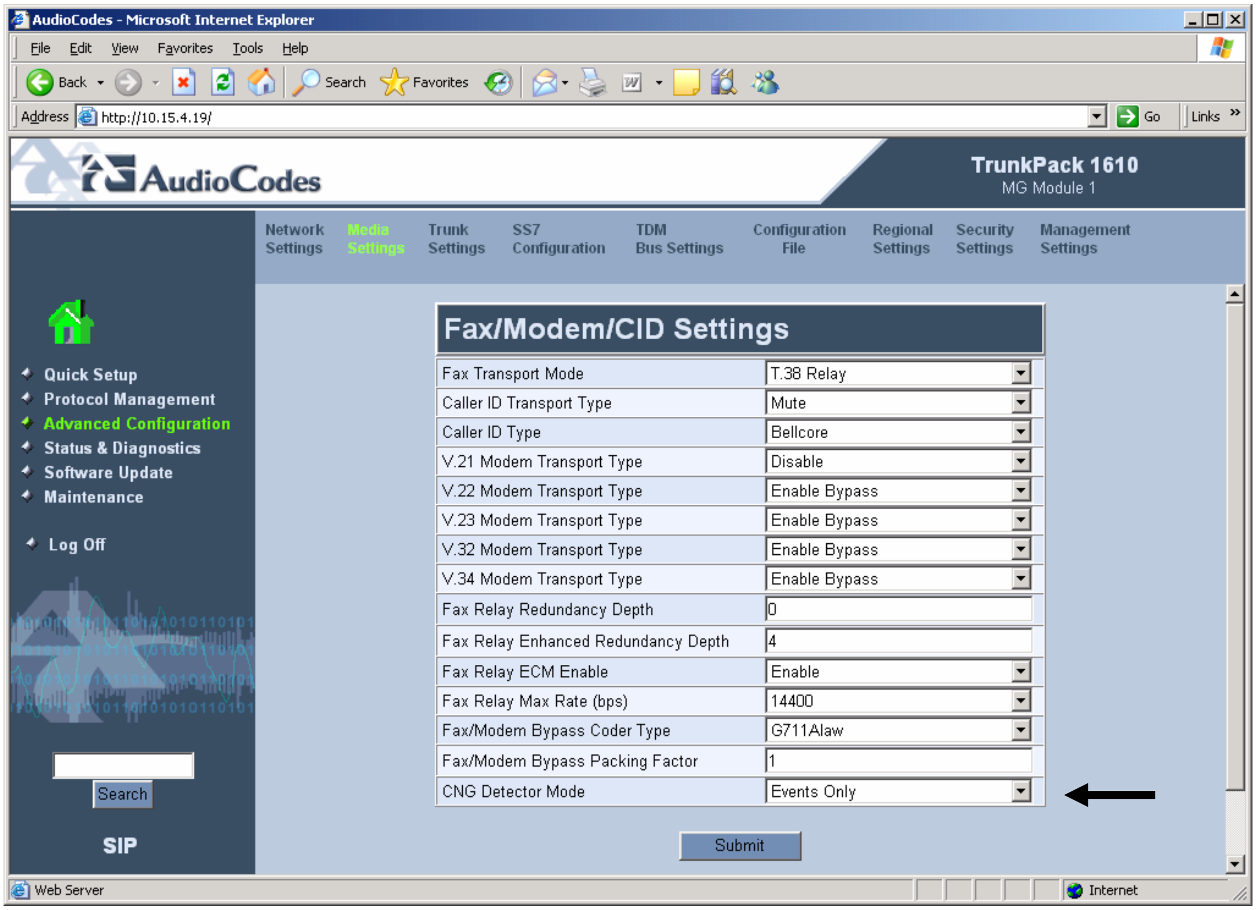
Step 8: TDM BUS Setting



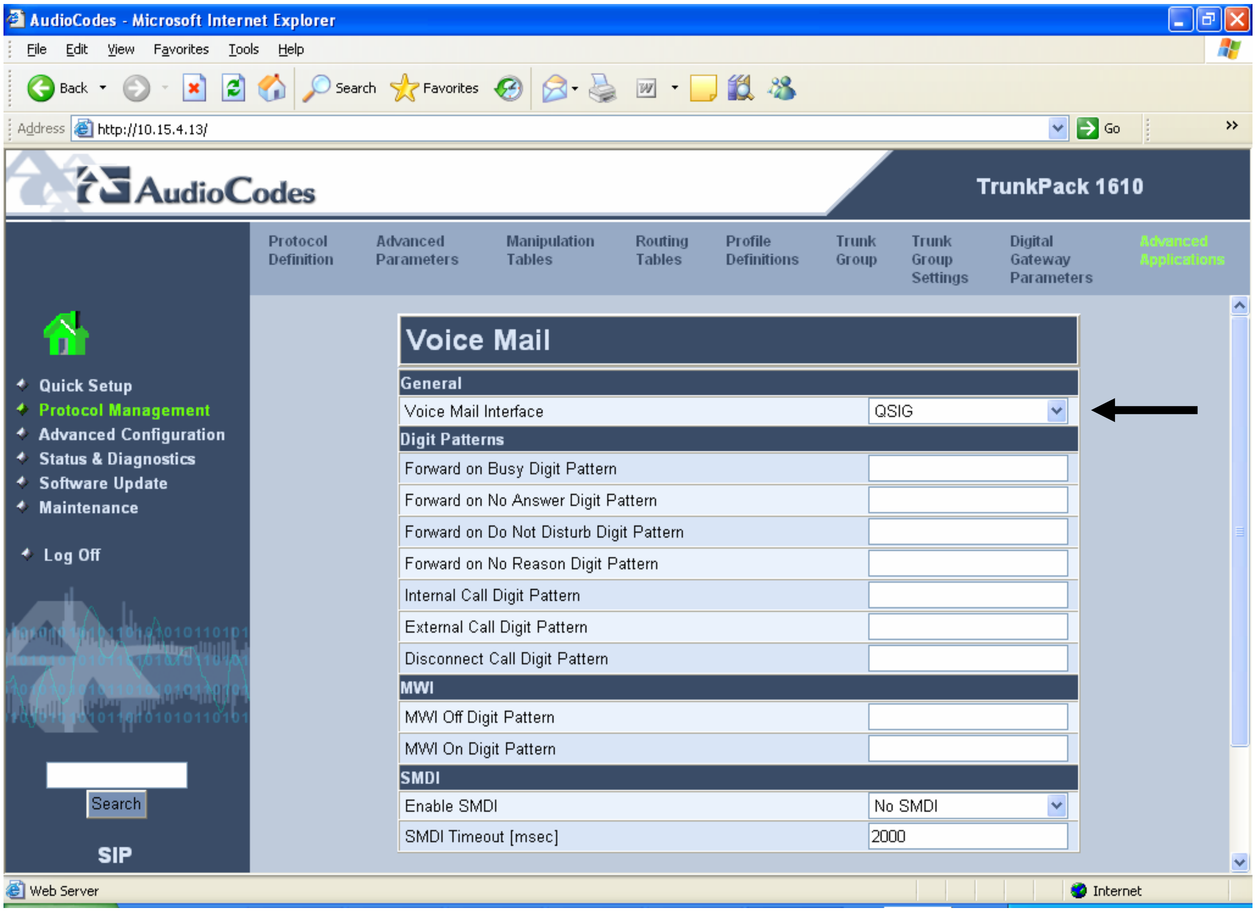
Step 9: Trunk Setting Setup



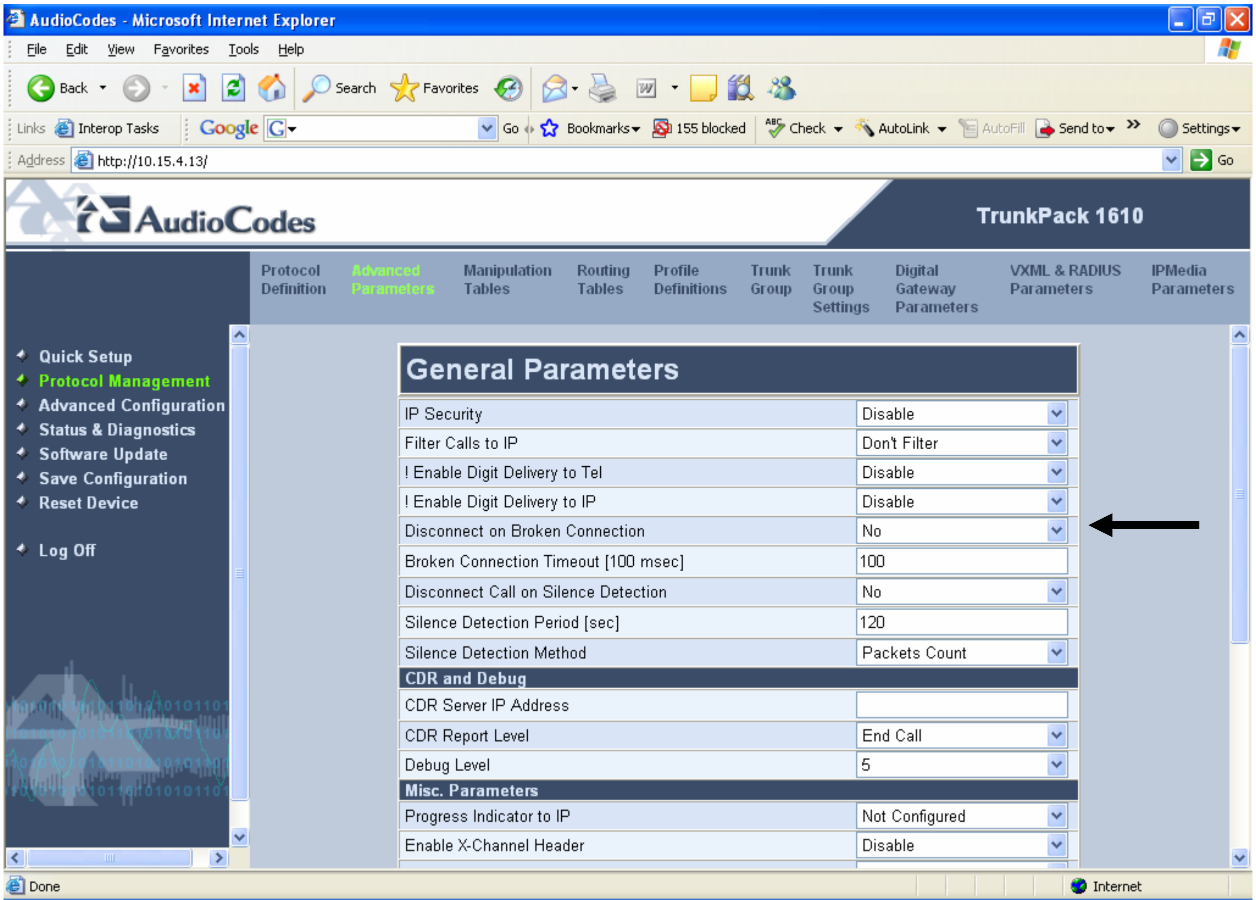
Step 10: FAX Setup



Step 11: Voice Mail Setup



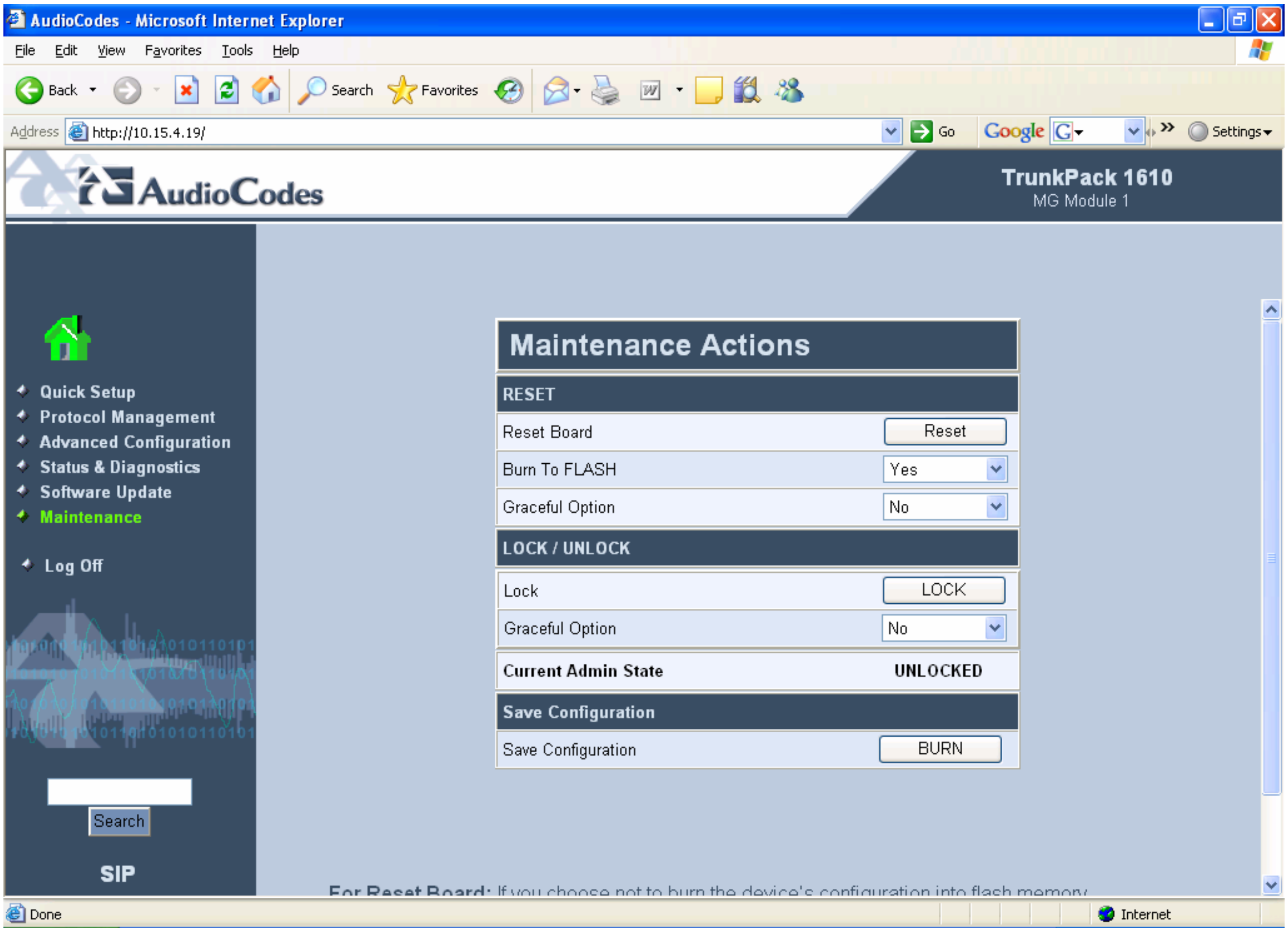
Step 12: General Setup



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

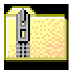


Click **Reset** to reset the gateway.

4.1. Configuration Files

The ZIP file includes the following files:

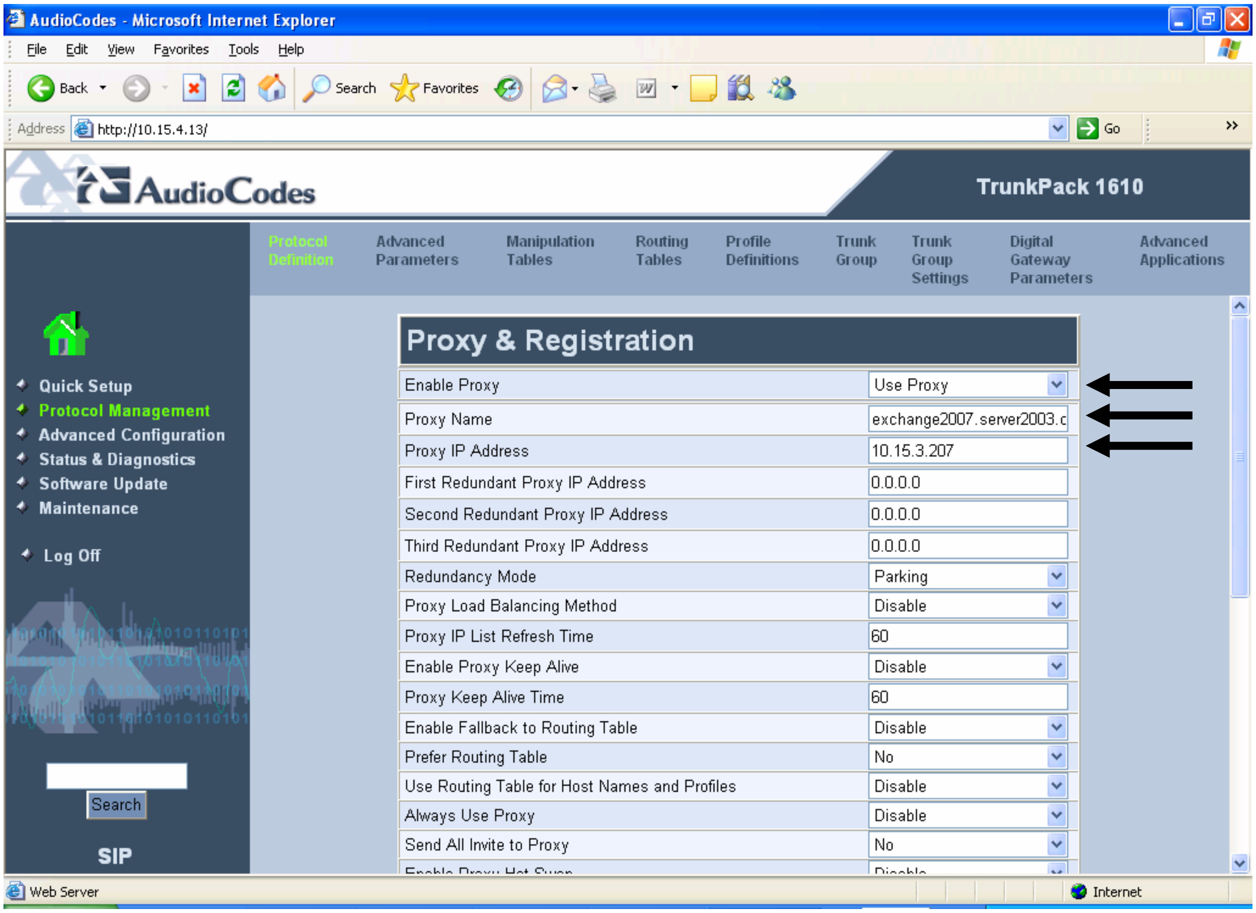
1. AudioCodes Mediant 1000 / Mediant 2000 configuration for TCP environnent (.ini file extension).
2. AudioCodes Mediant 1000 / Mediant 2000 configuration for TLS environnent (.ini file extension).



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

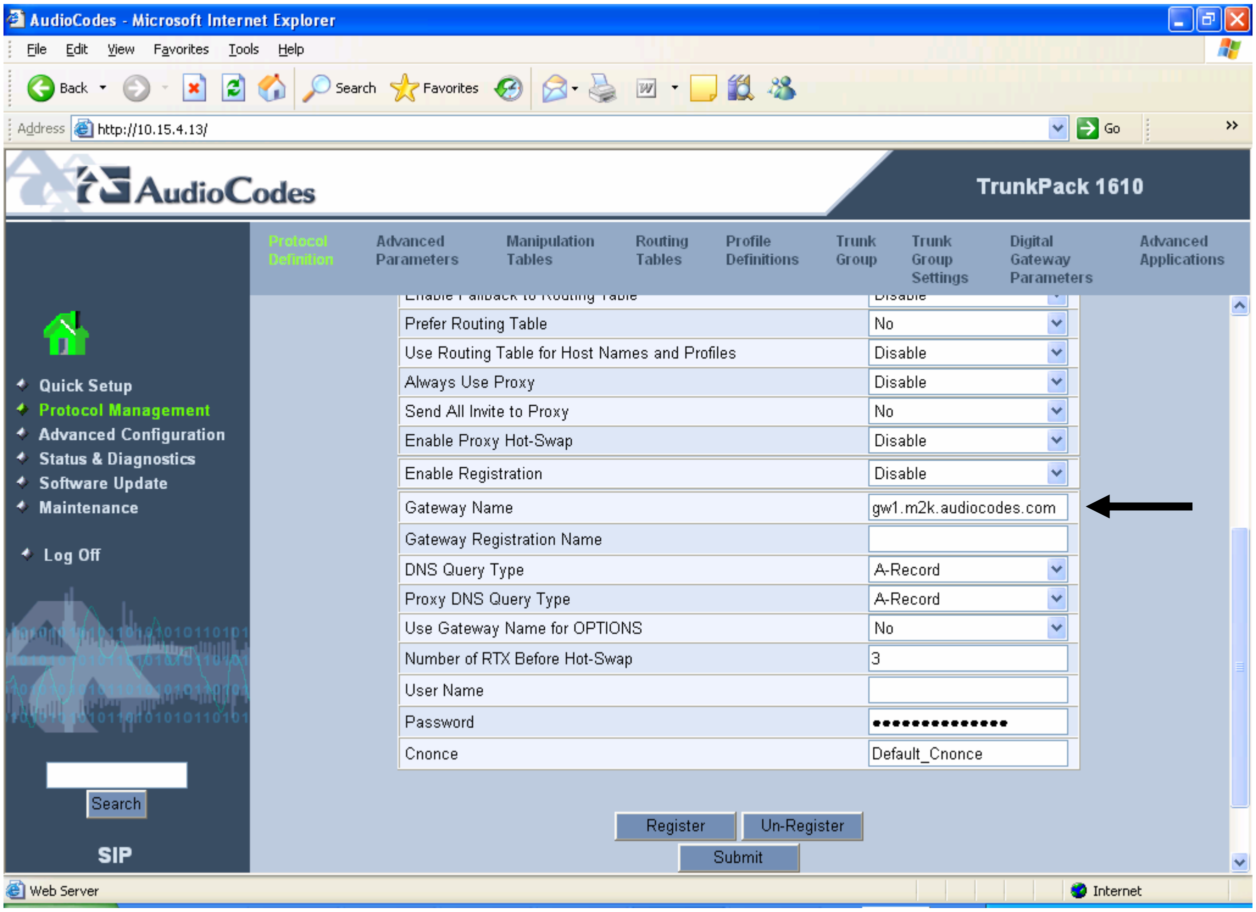
4.2. TLS Setup

Step 1: PBX to IP Routing Setup



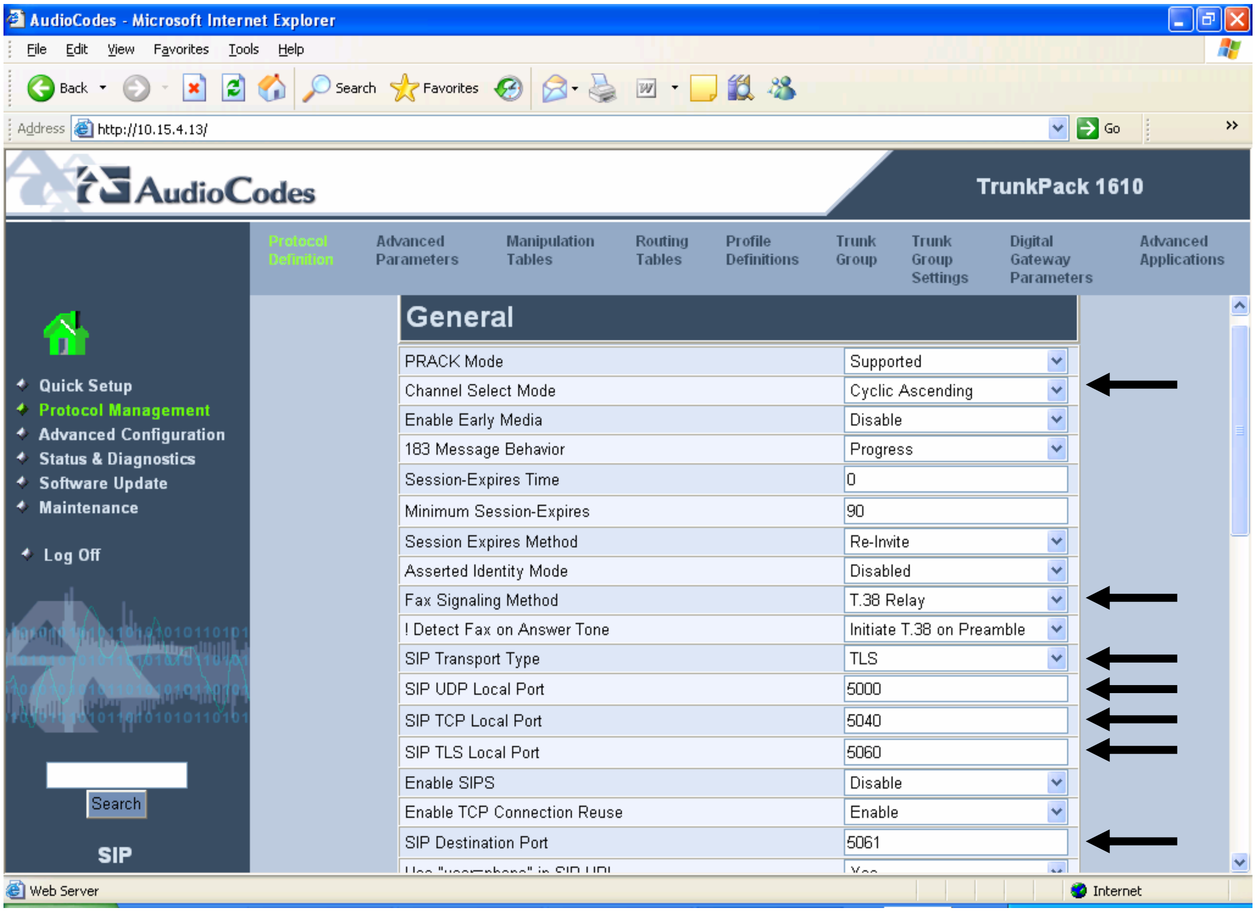
Note: The Proxy IP Address and Name must correspond to the network environment in which the Microsoft Unified Messaging server is installed (For example, 10.15.3.207 for IP Address and exchaneg2007.server2003.com for the FQDN of the Microsoft Unified Messaging host).

Step 2: SIP Environment and Gateway Name Setup

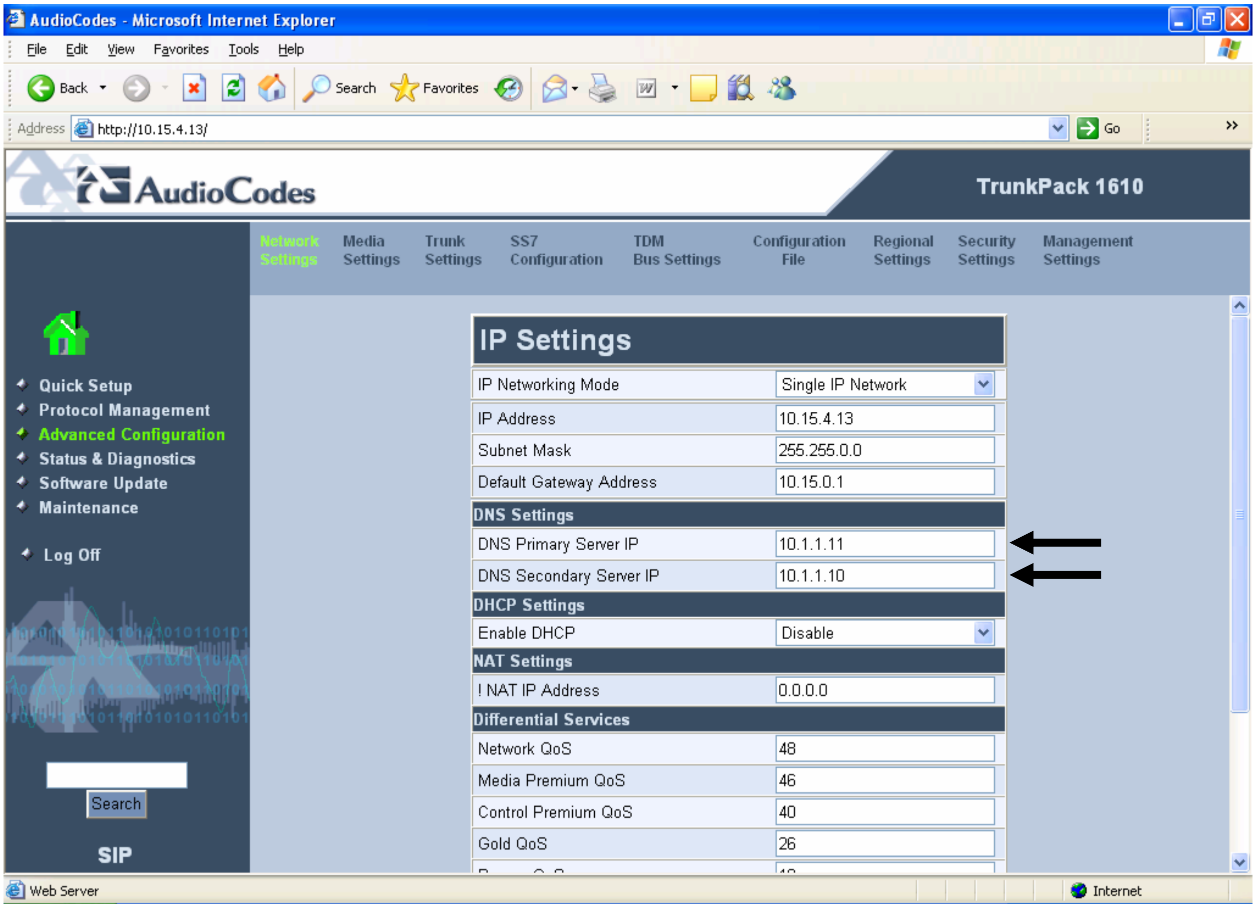


Note: Assign an FQDN name to the gateway (for example, gw1.m2k.audiocodes.com). Any gateway name that corresponds to your network environment is applicable; the only limitation is not to include underscores in the name (Windows Certification server limitation).

Step 3: SIP Environment Setup (Cont.)

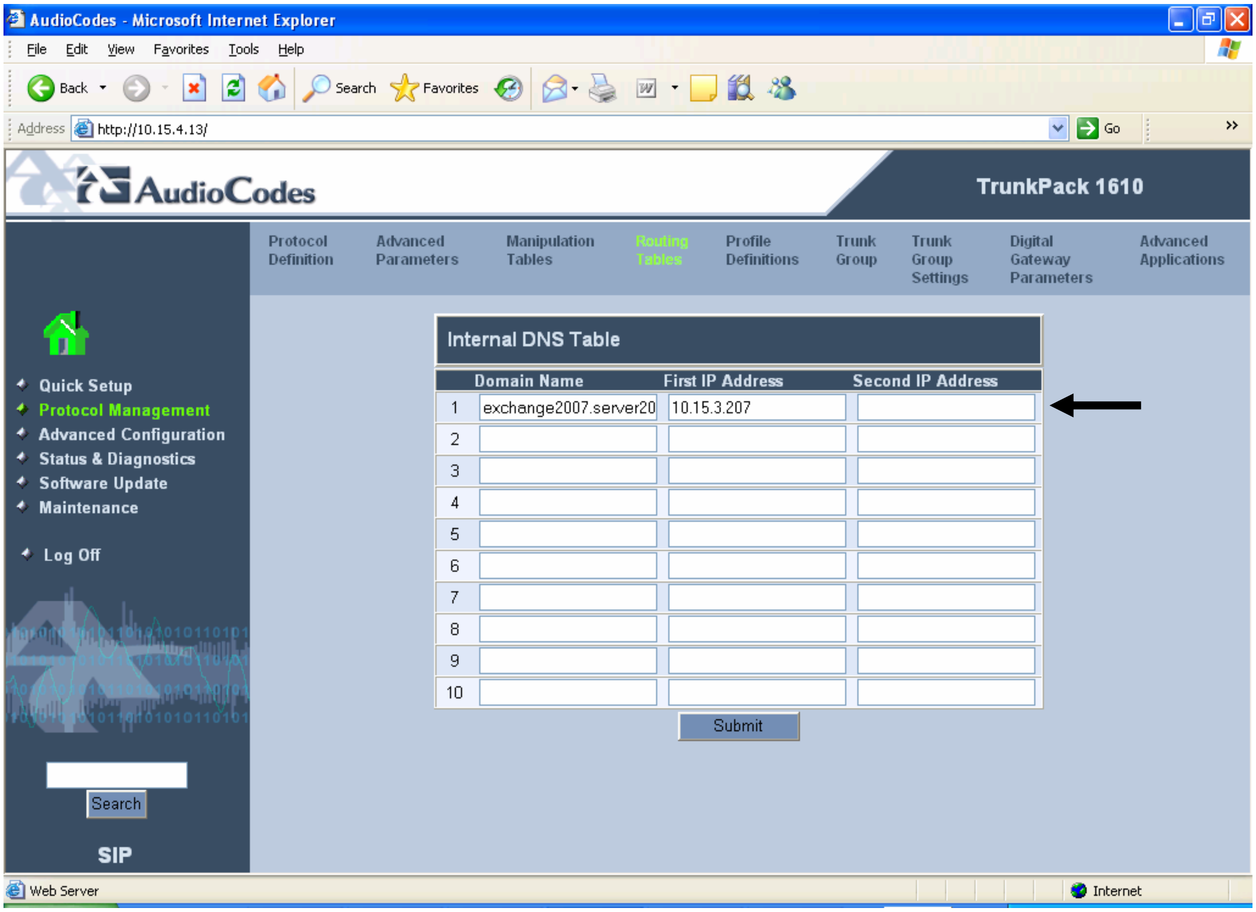


Step 4: DNS Servers Setup



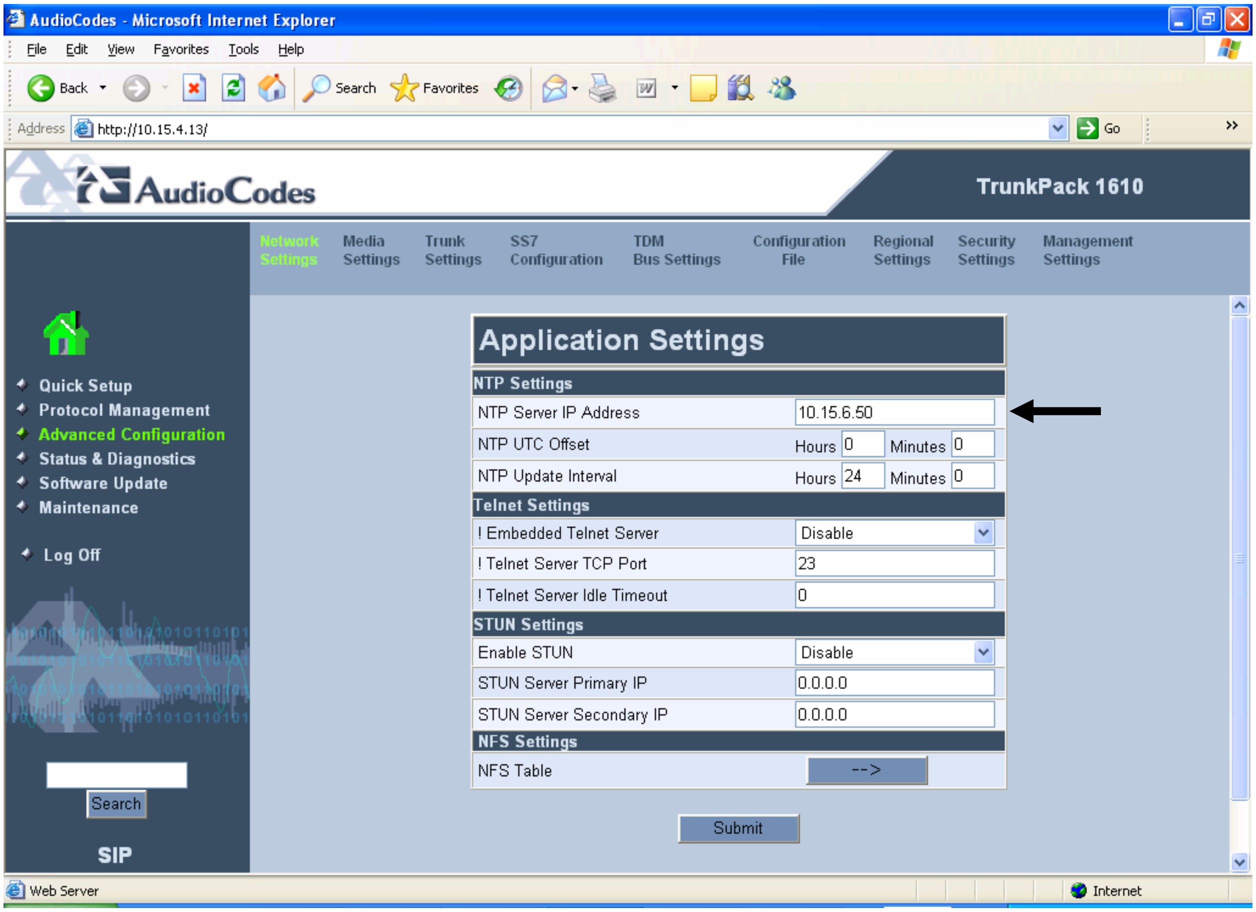
Note: Define the primary and secondary DNS servers' IP addresses so that they correspond to your network environment (for example, 10.1.1.11 and 10.1.1.10). If no DNS server is available in the network, then skip this step.

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).

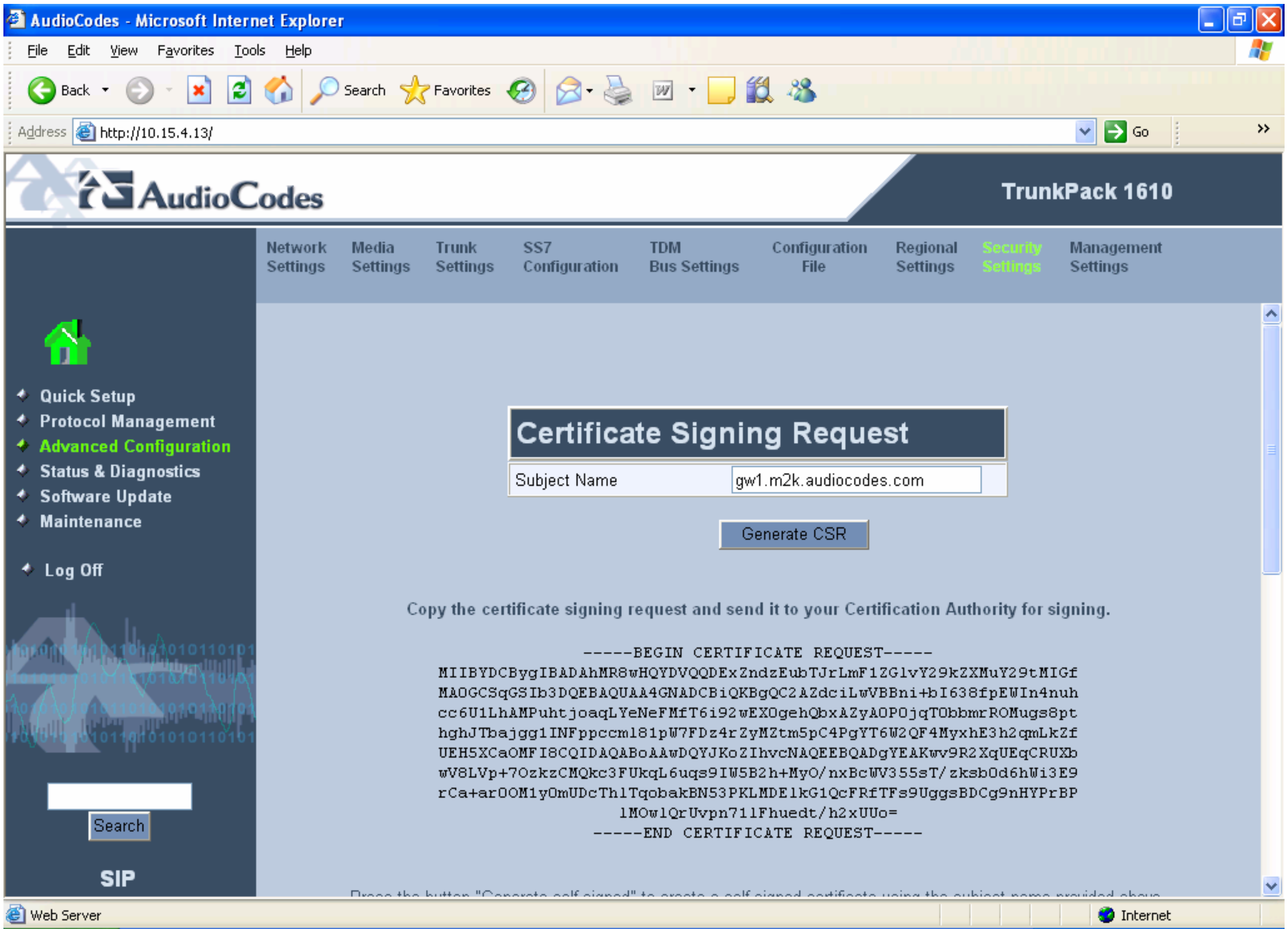
Step 6: NTP Server Setup



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.

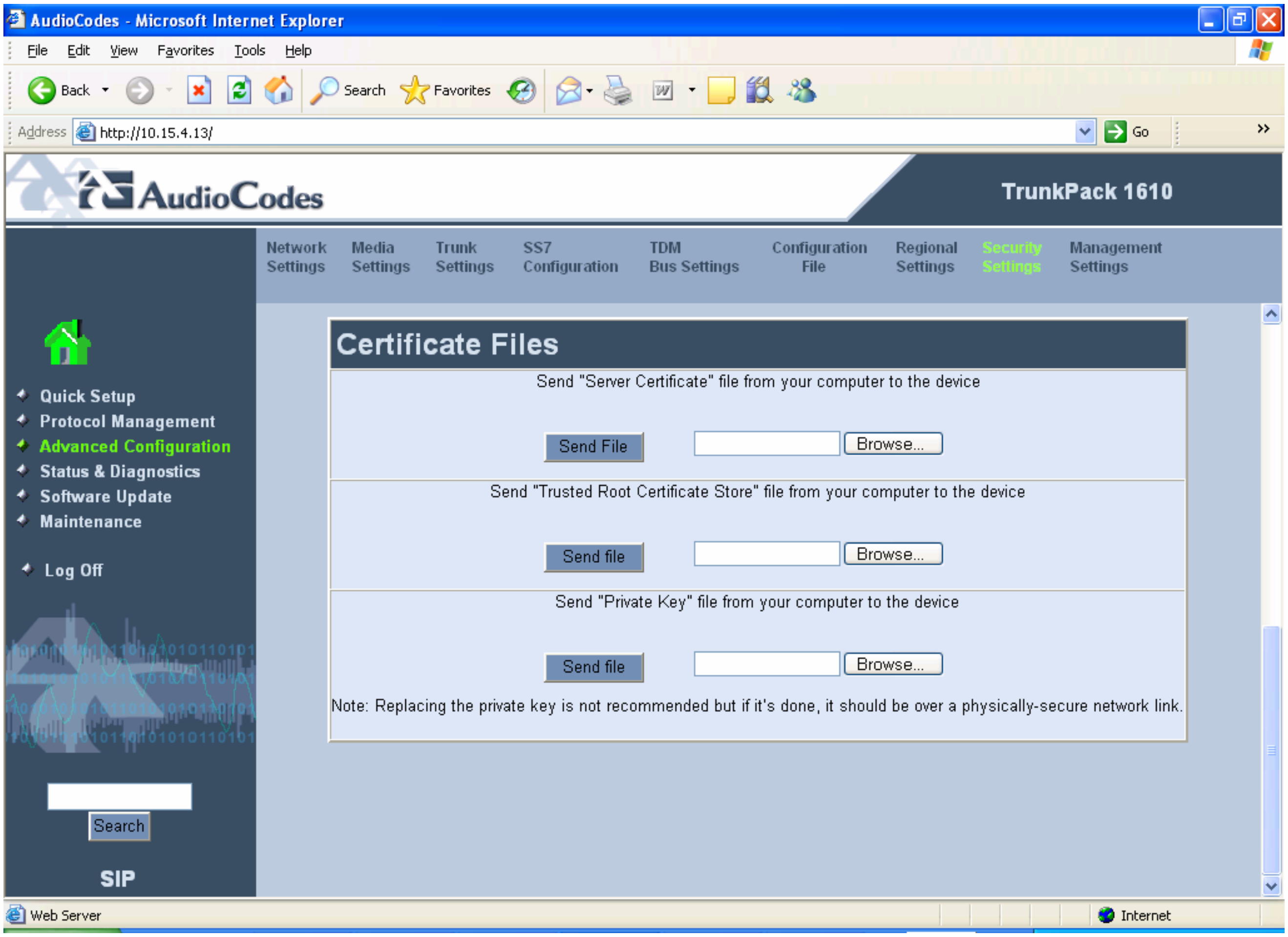


Step 8: Uploading Certificates Setup

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.



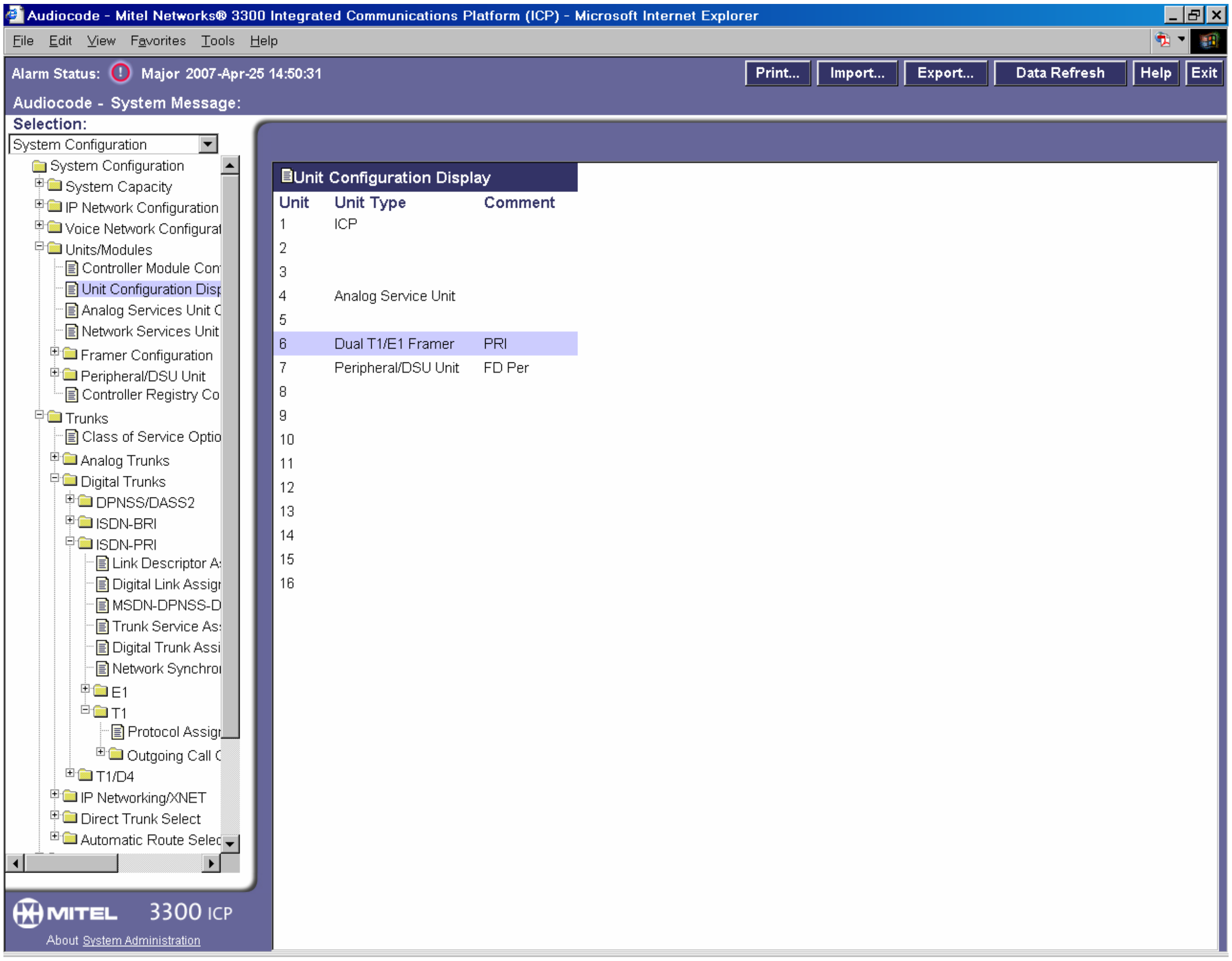
5. PBX Setup Notes

Information used for this test case:

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

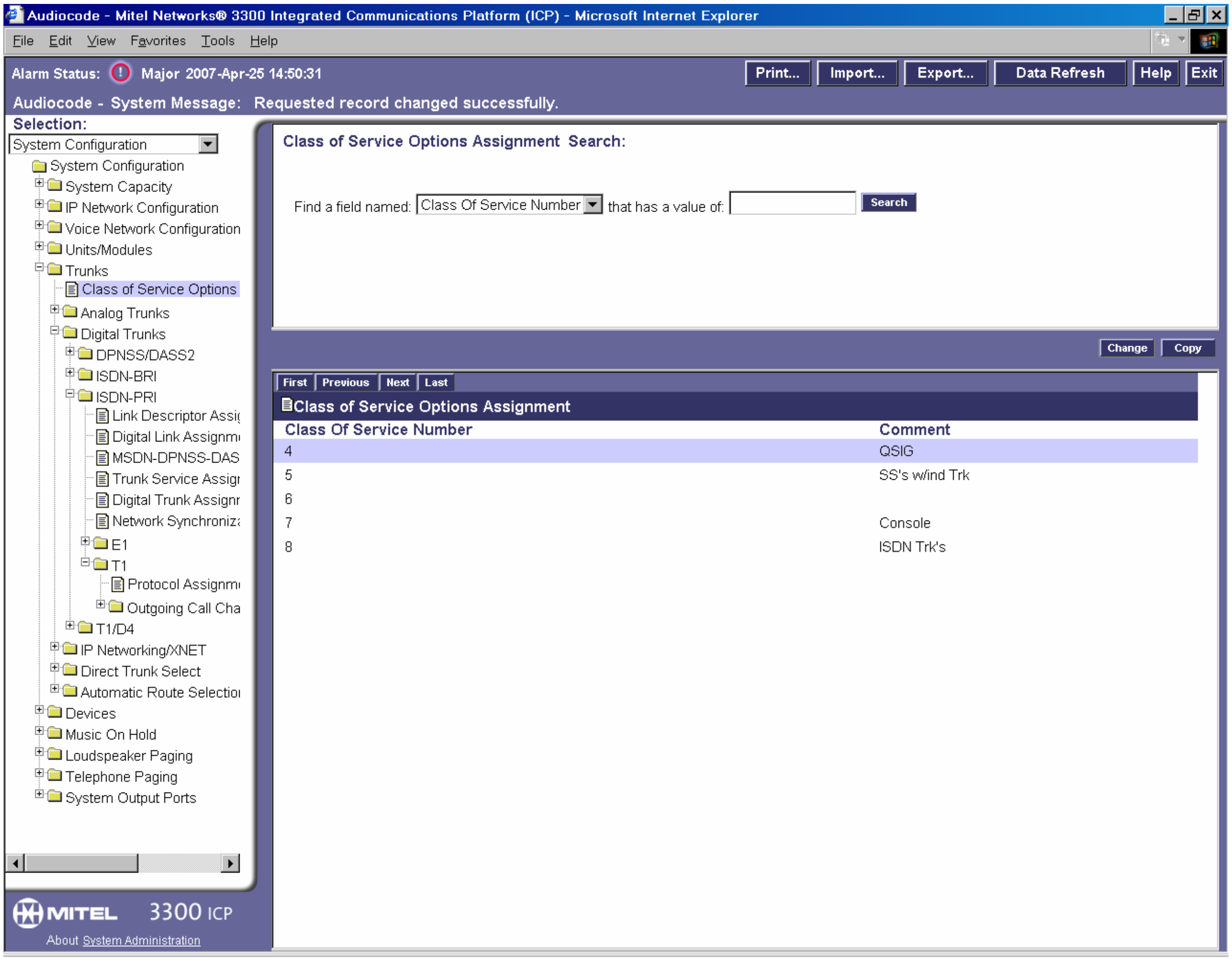
Step 1: Ensure PBX is equipped with T1 PRI Module to Support T1 QSIG

Use the following path to ensure that the PBX is equipped with a T1 PRI module:
System Configuration / UnitsModules / Units Configuration Display.



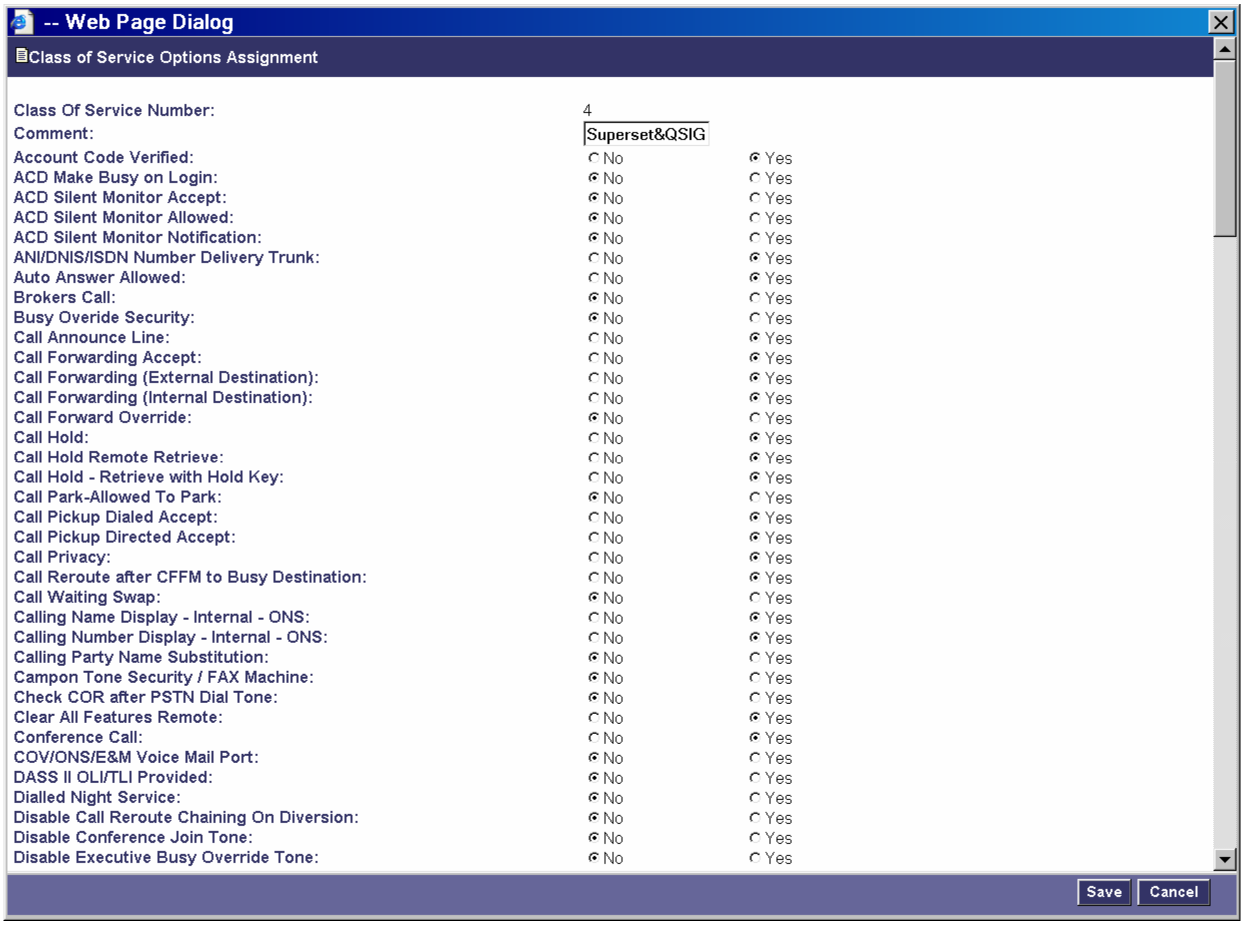
Step 2: Create Class of Service for QSIG Trunks

Use the following path to create CoS for QSIG trunk:
System Configuration / Trunks / Class of Service Options Assignment.



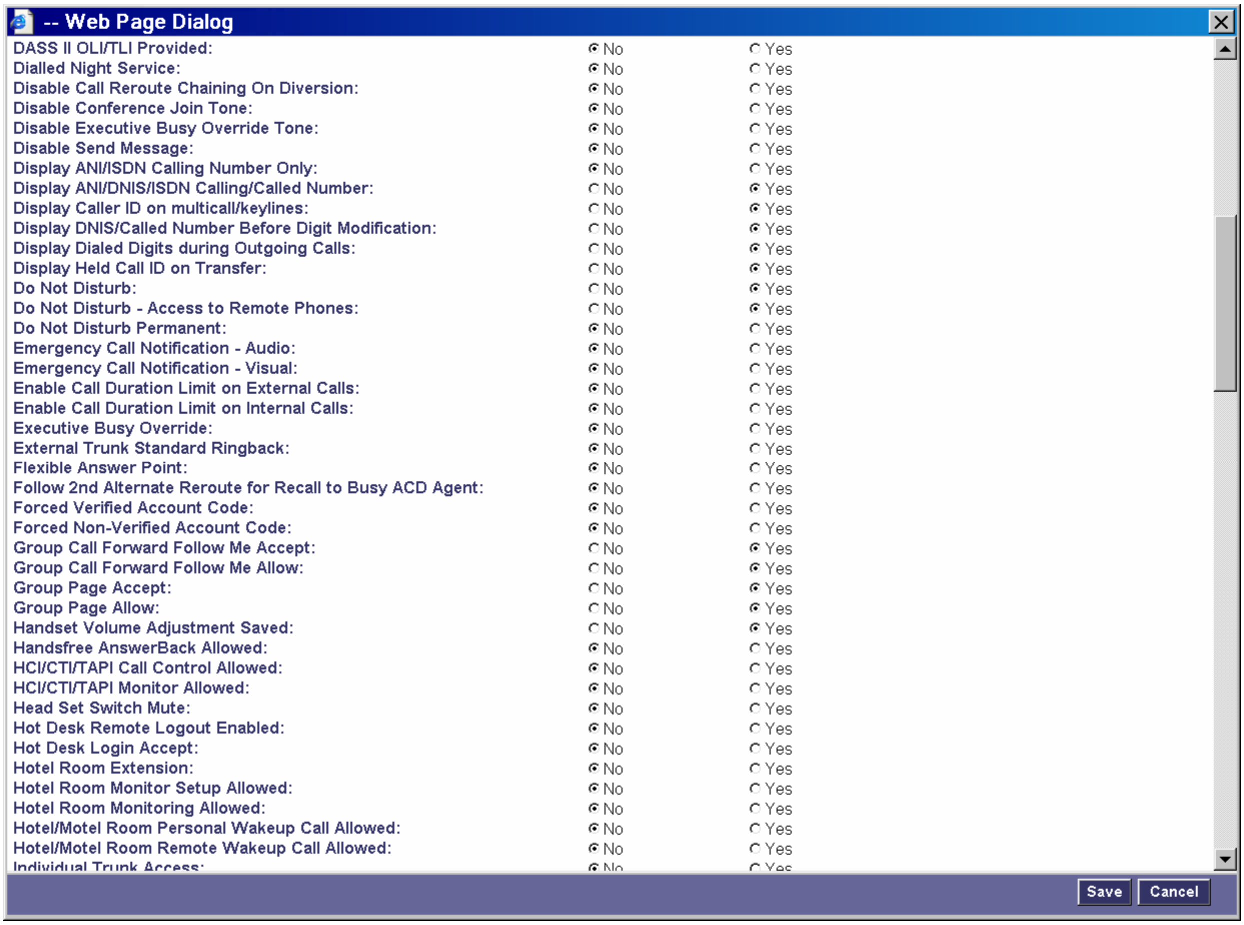
Change the following options to 'Yes':

- 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



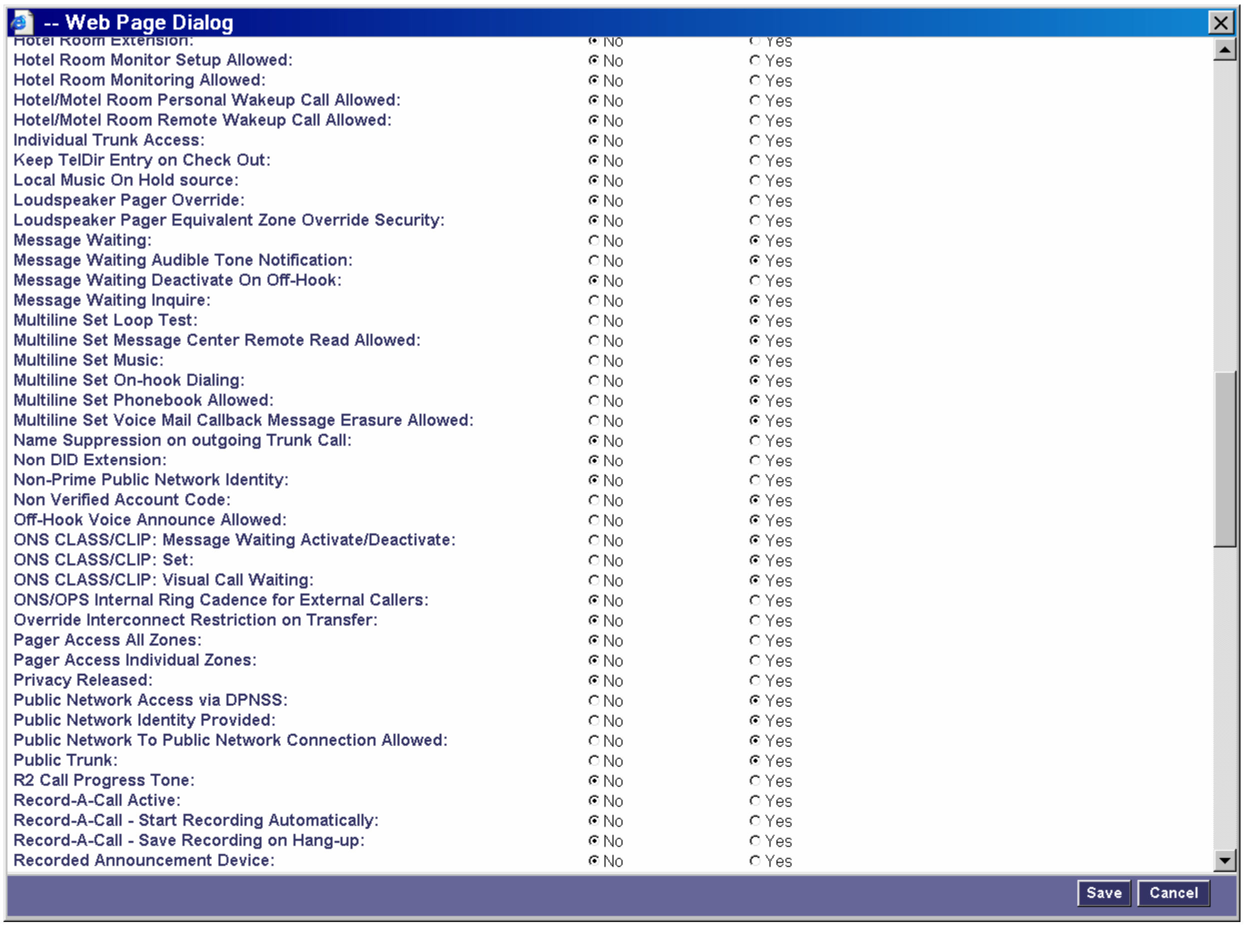
Change the following options to 'Yes':

- 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



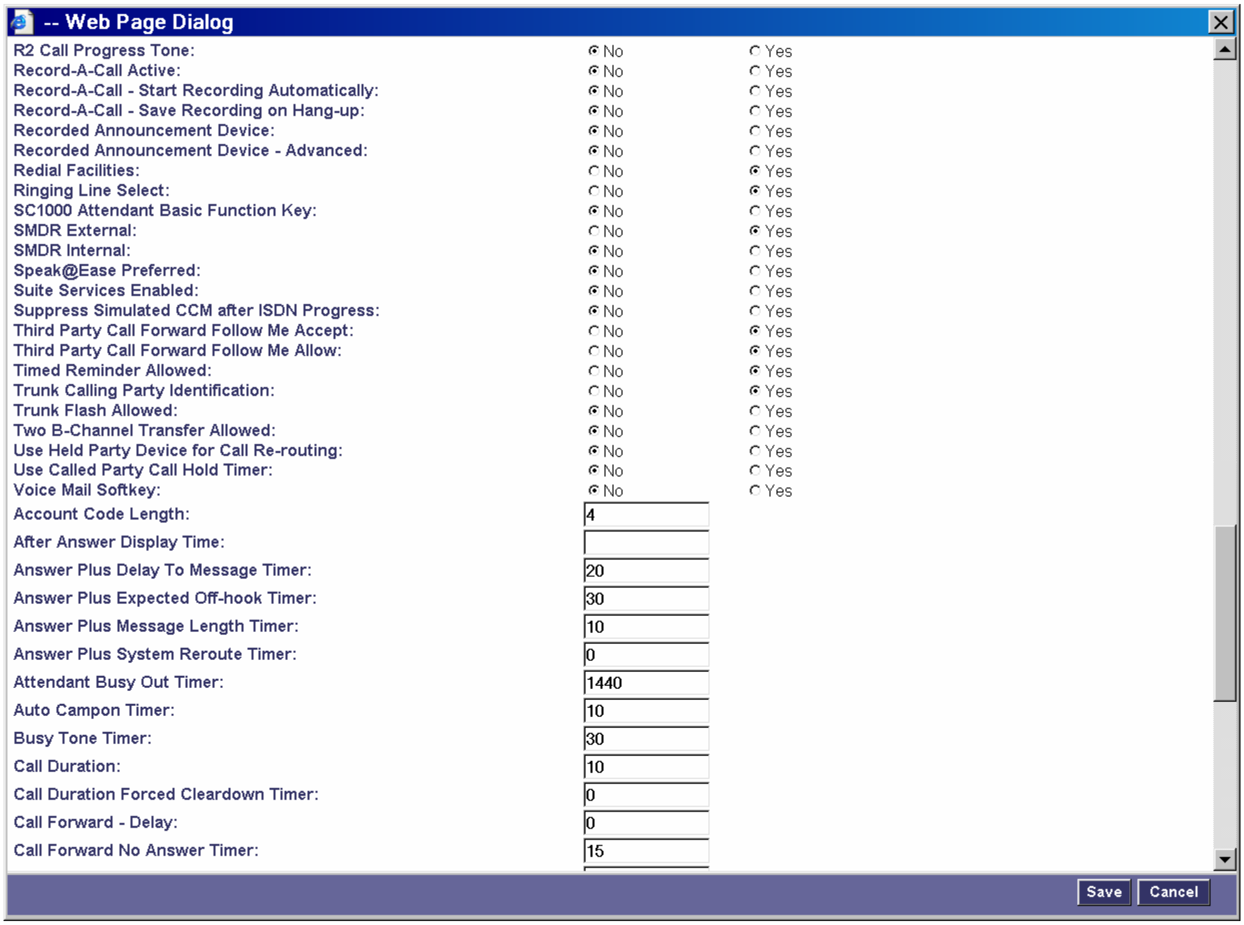
Change the following options to 'Yes':

- 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



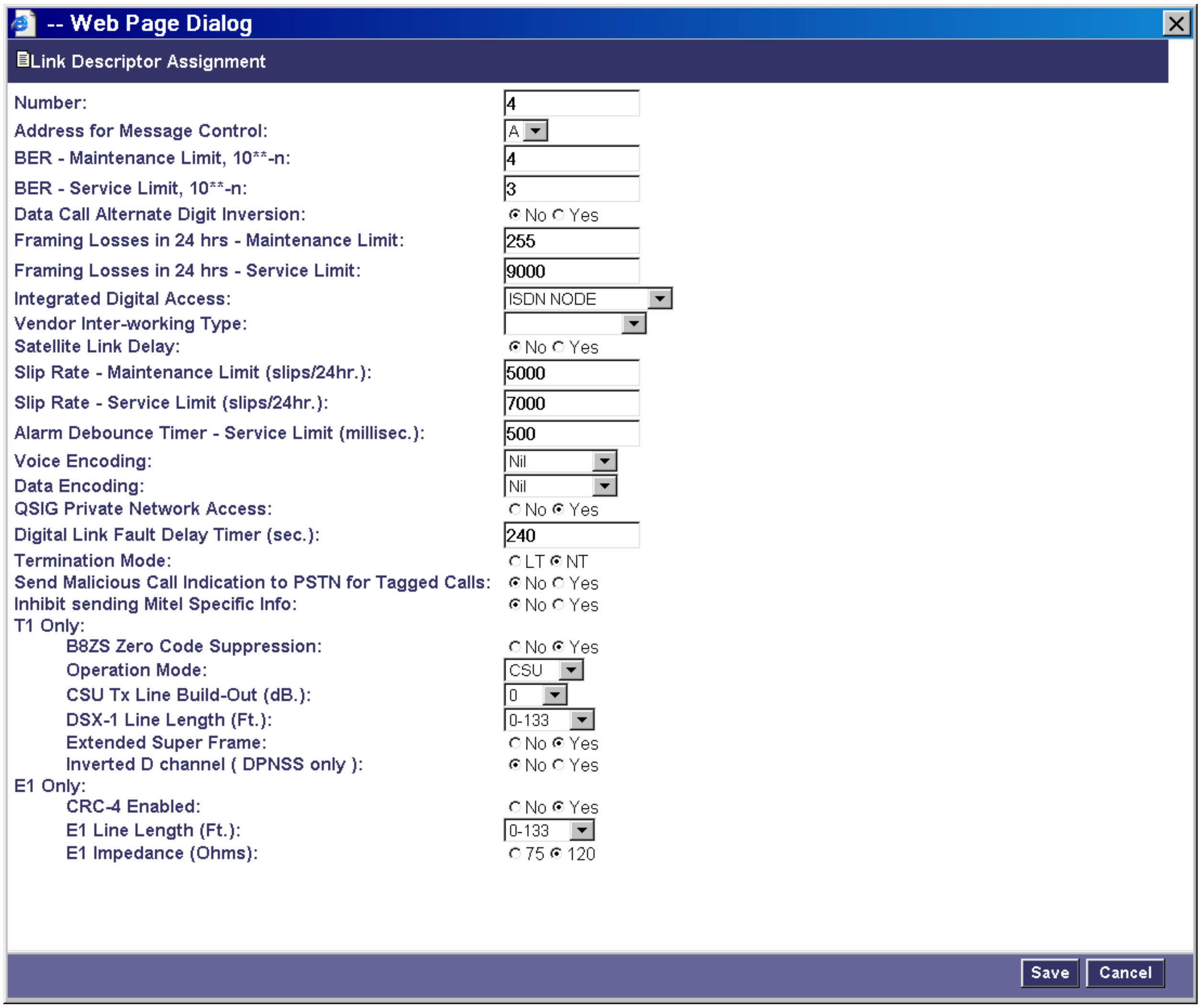
Change the following options to 'Yes':

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



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.



-- Web Page Dialog

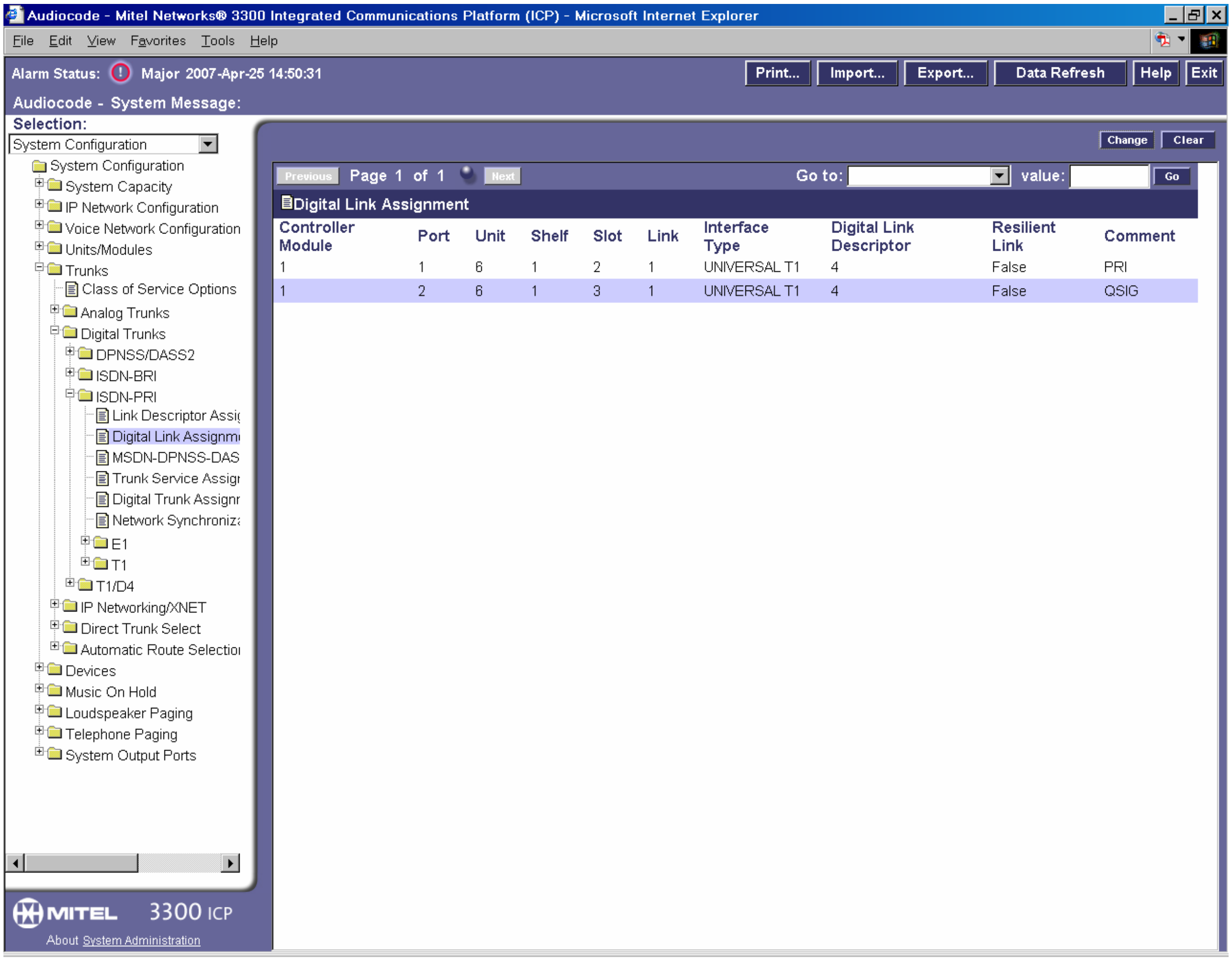
Link Descriptor Assignment

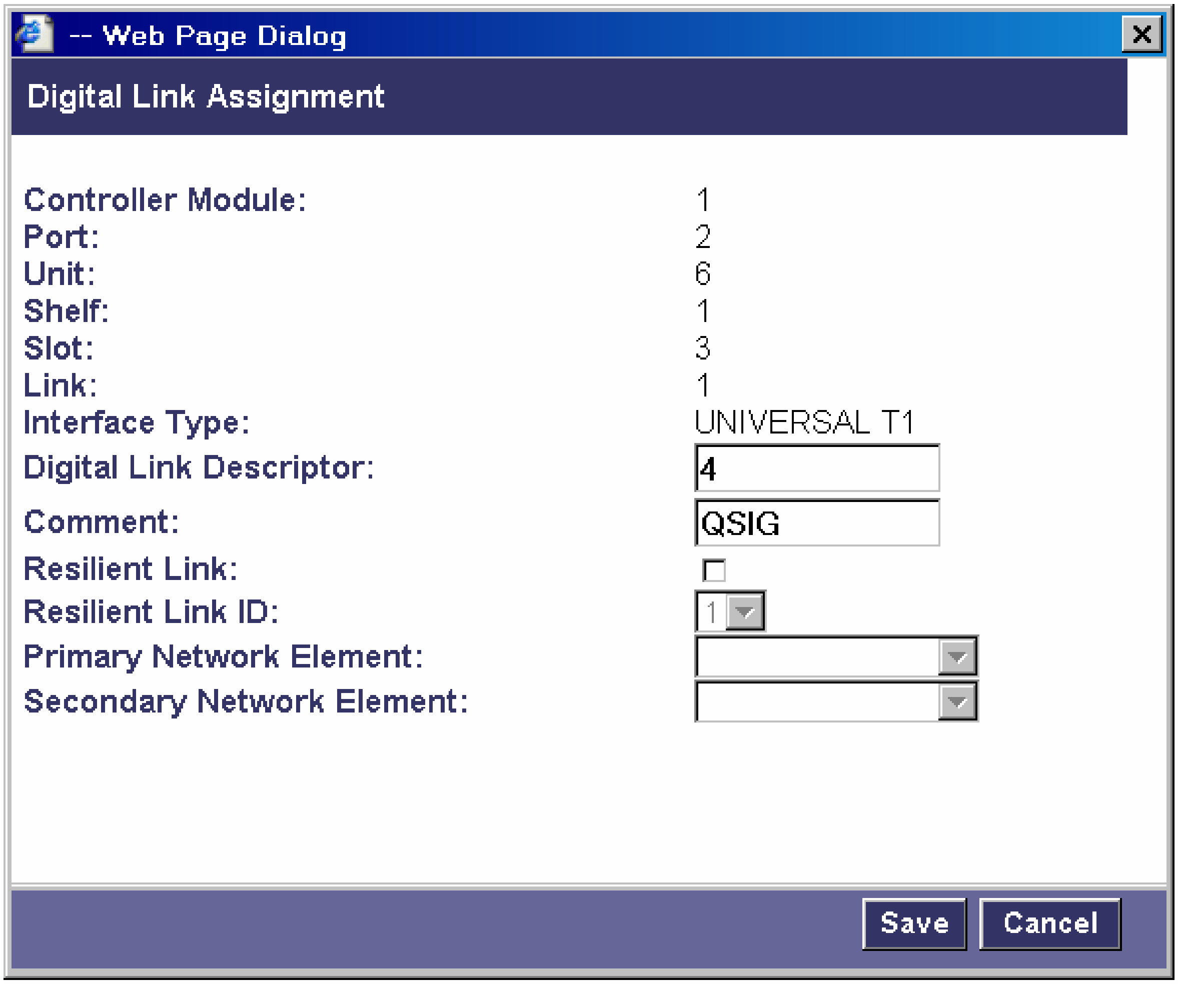
Number:	4
Address for Message Control:	A
BER - Maintenance Limit, 10**-n:	4
BER - Service Limit, 10**-n:	3
Data Call Alternate Digit Inversion:	(•) No ( ) Yes
Framing Losses in 24 hrs - Maintenance Limit:	255
Framing Losses in 24 hrs - Service Limit:	9000
Integrated Digital Access:	ISDN NODE
Vendor Inter-working 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

Save Cancel

Step 4: Create a Digital Link Assignment

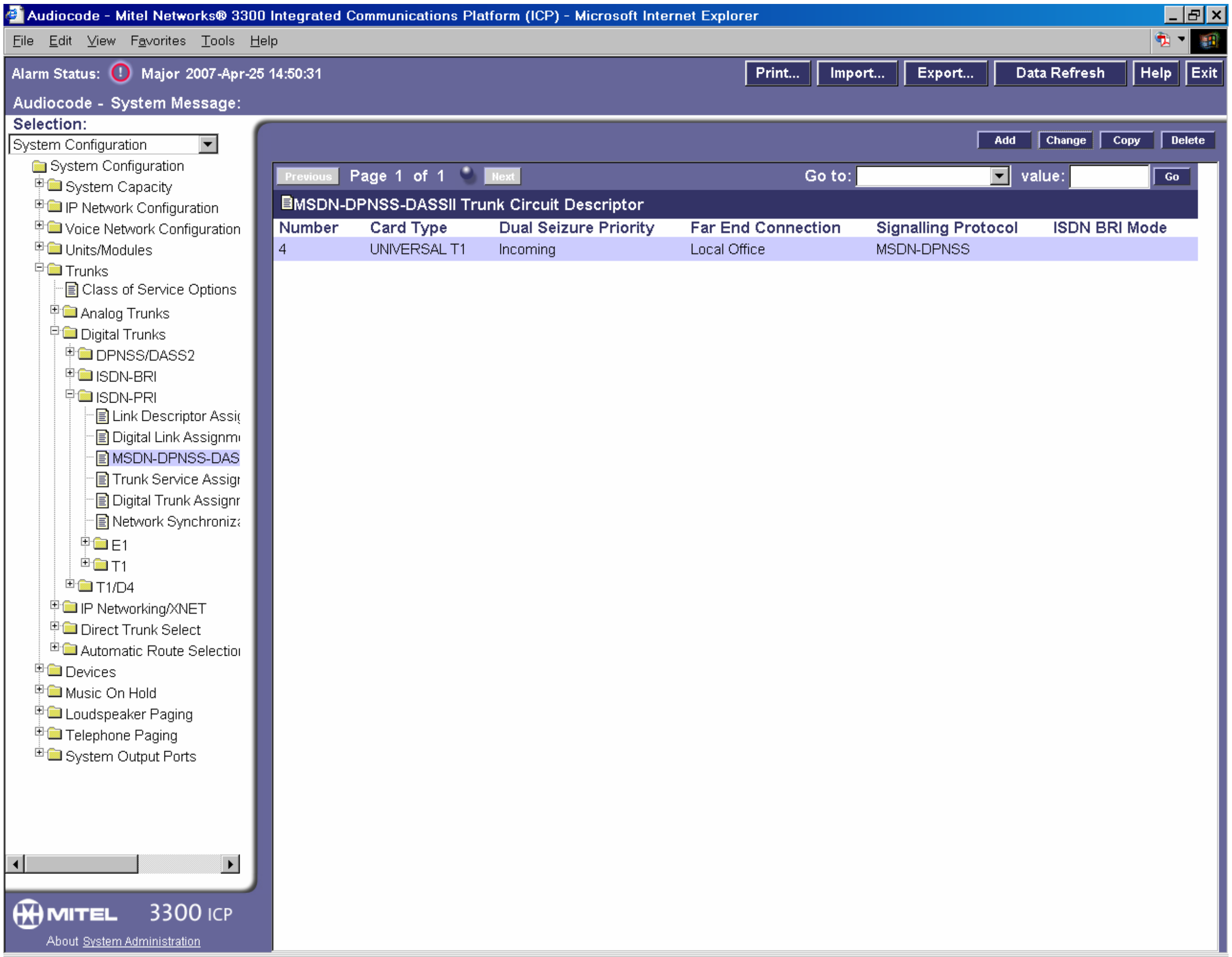
Use the following path to create a Digital Link Assignment:
System Configuration / Trunks / Digital Trunks / ISDN-PRI / Digital Link Assignment.

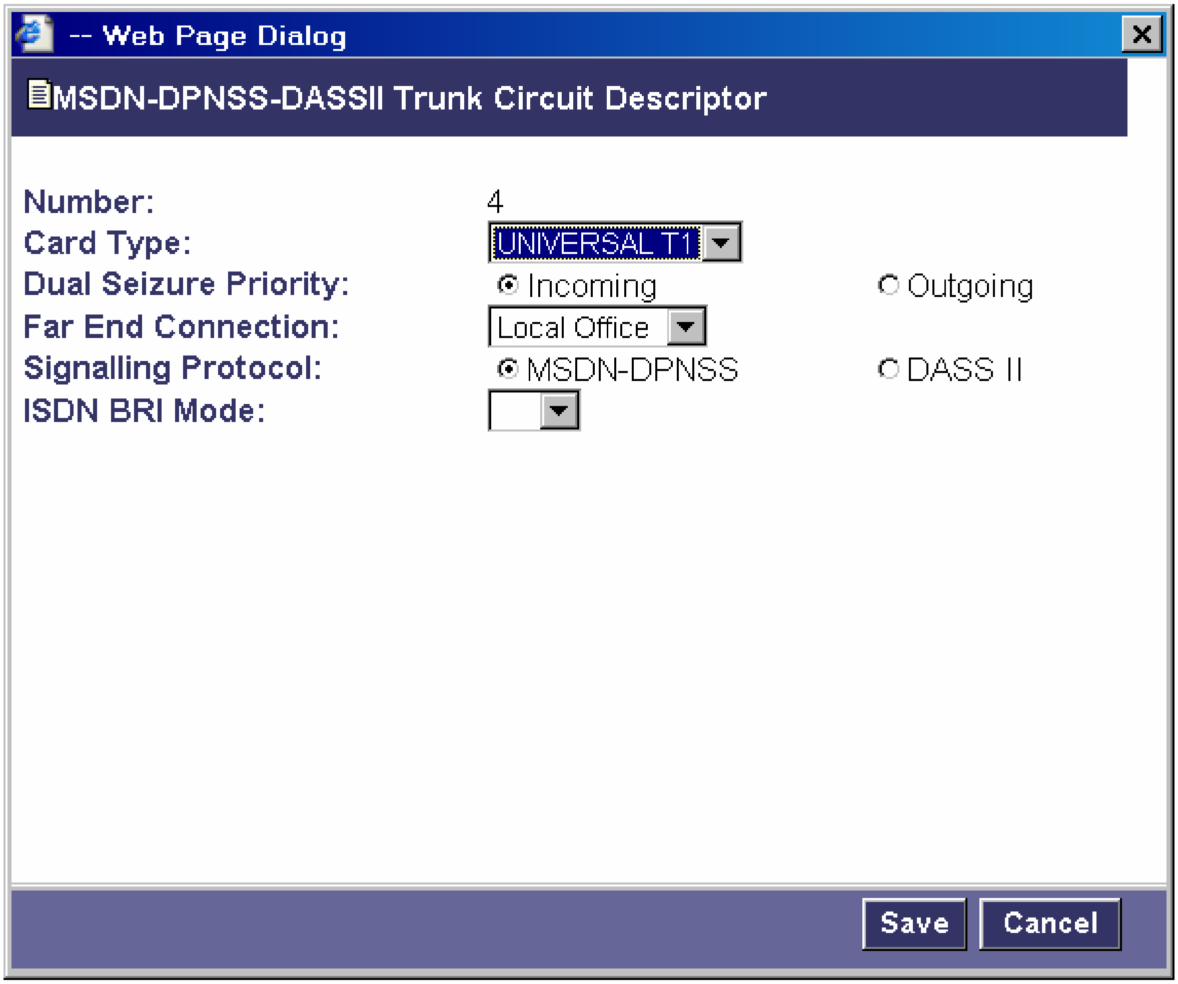



-- Web Page Dialog
Digital Link Assignment
Controller Module: 1
Port: 2
Unit: 6
Shelf: 1
Slot: 3
Link: 1
Interface Type: UNIVERSAL T1
Digital Link Descriptor: 4
Comment: QSIG
Resilient Link:
Resilient Link ID: 1
Primary Network Element:
Secondary Network Element:
Save
Cancel

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

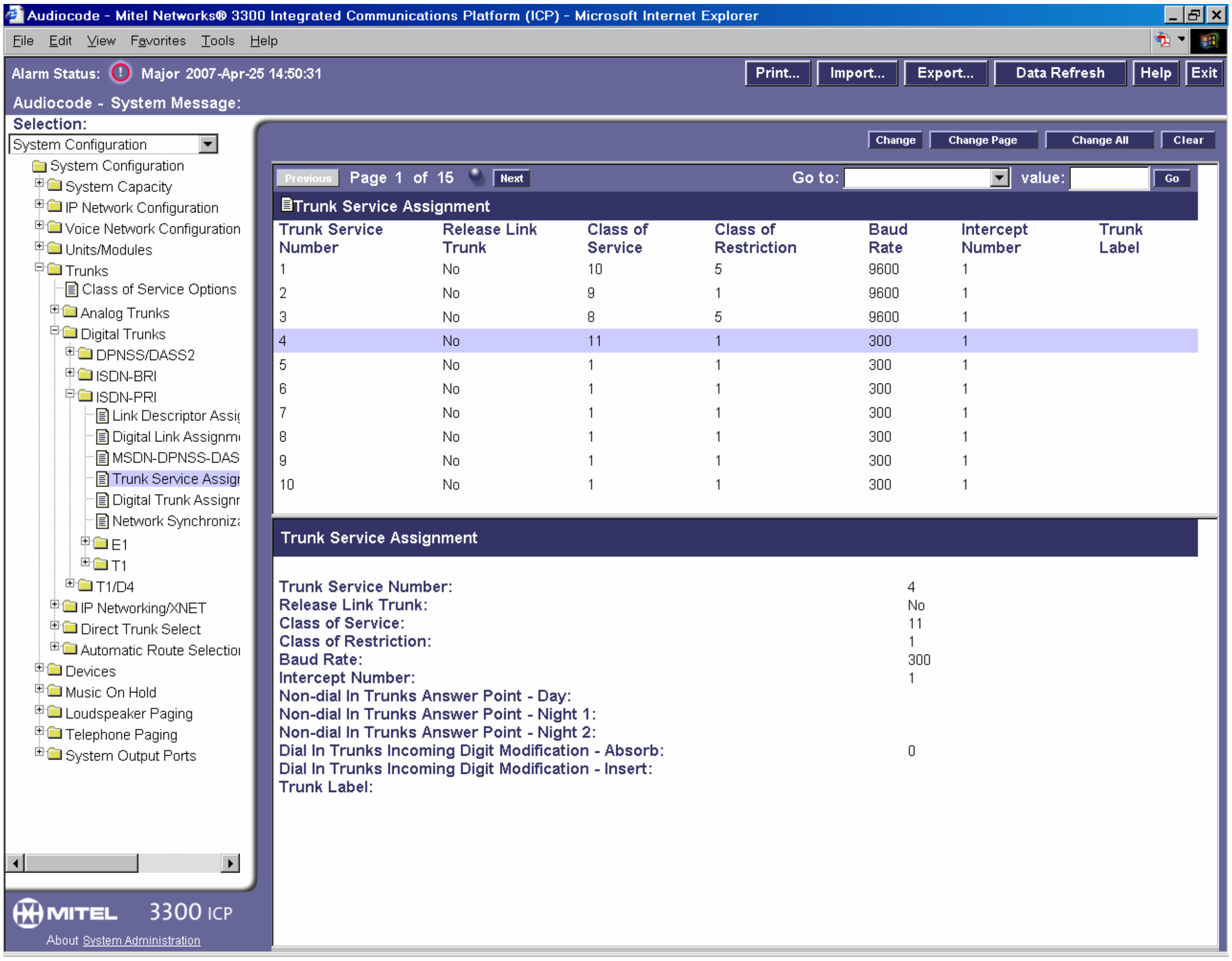
Use the following path to Add MSDN-DPNSS-DASSII Trunk Circuit Descriptor:
System Configuration / Trunks / Digital Trunks / ISDN-PRI/MSDN-DPNSS-DASSII Trunk Circuit Descriptor.




-- 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

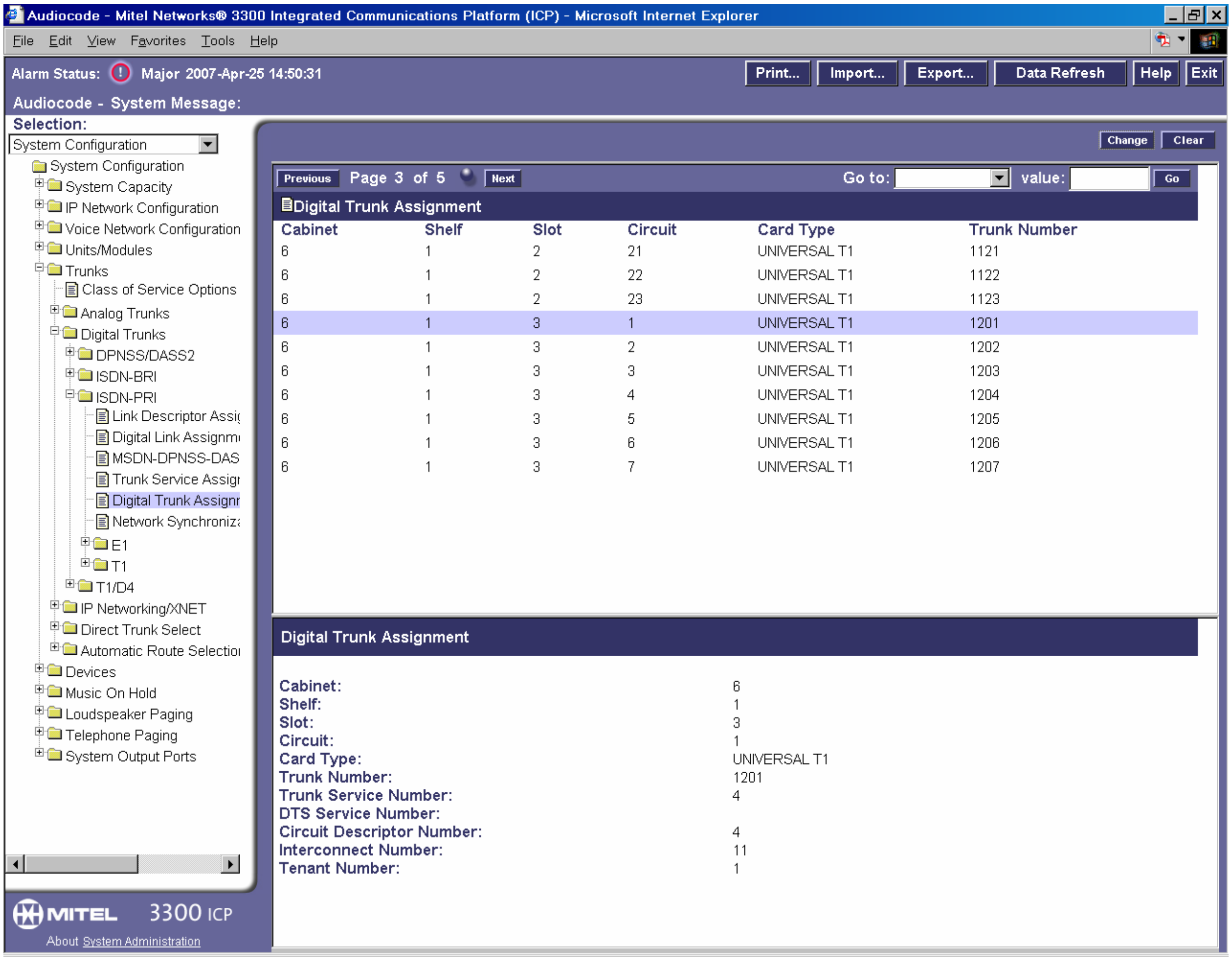
Step 6: Create a Trunk Service Assignment

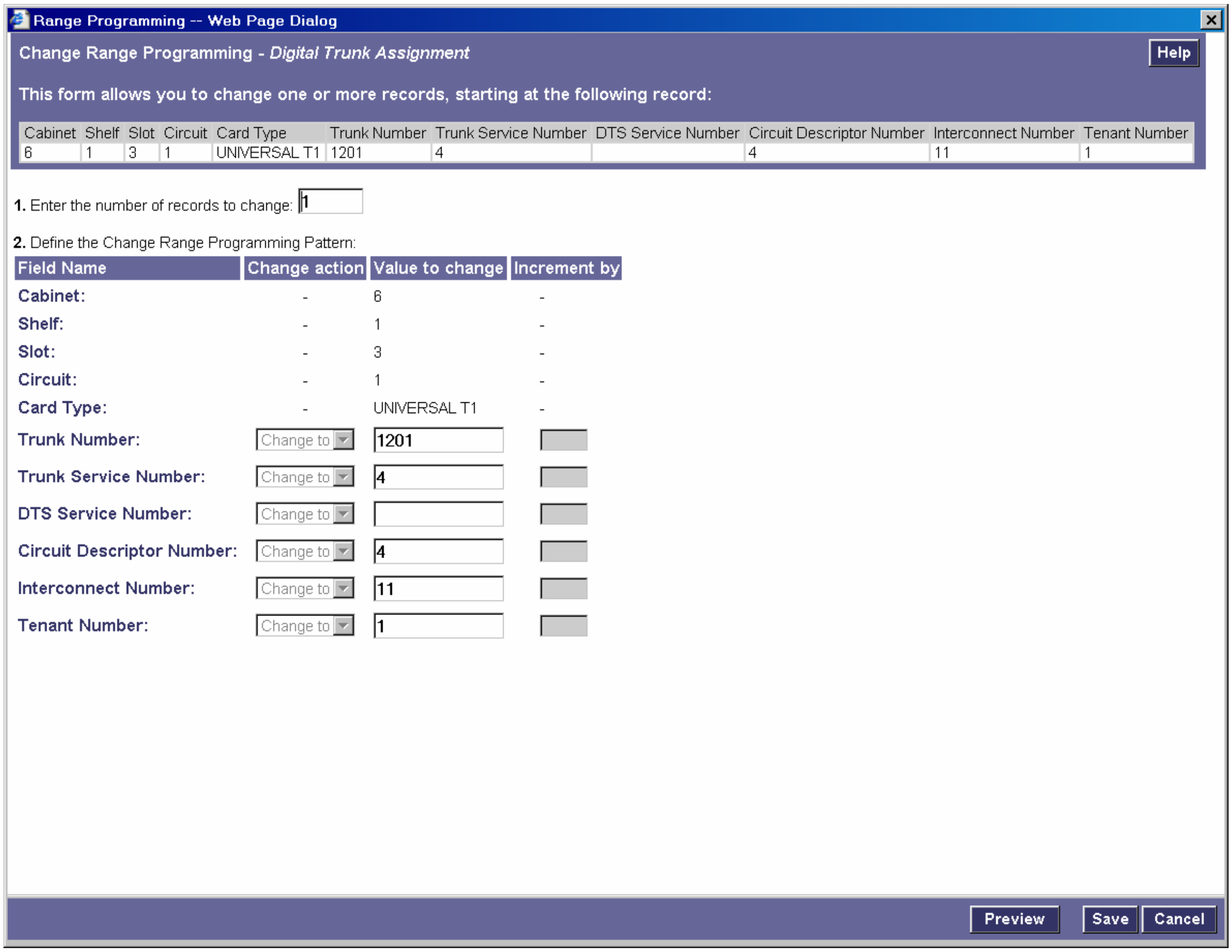
Use the following path to create a Trunk Service Assignment:
System Configuration / Trunks / Digital Trunks / ISDN-PRI / Trunk Service Assignment.



Step 7: Assign Digital Trunks

Use the following path to assign Digital Trunks:
System Configuration / Trunks / Digital Trunks / ISDN-PRI / Digital Trunk Assignment.

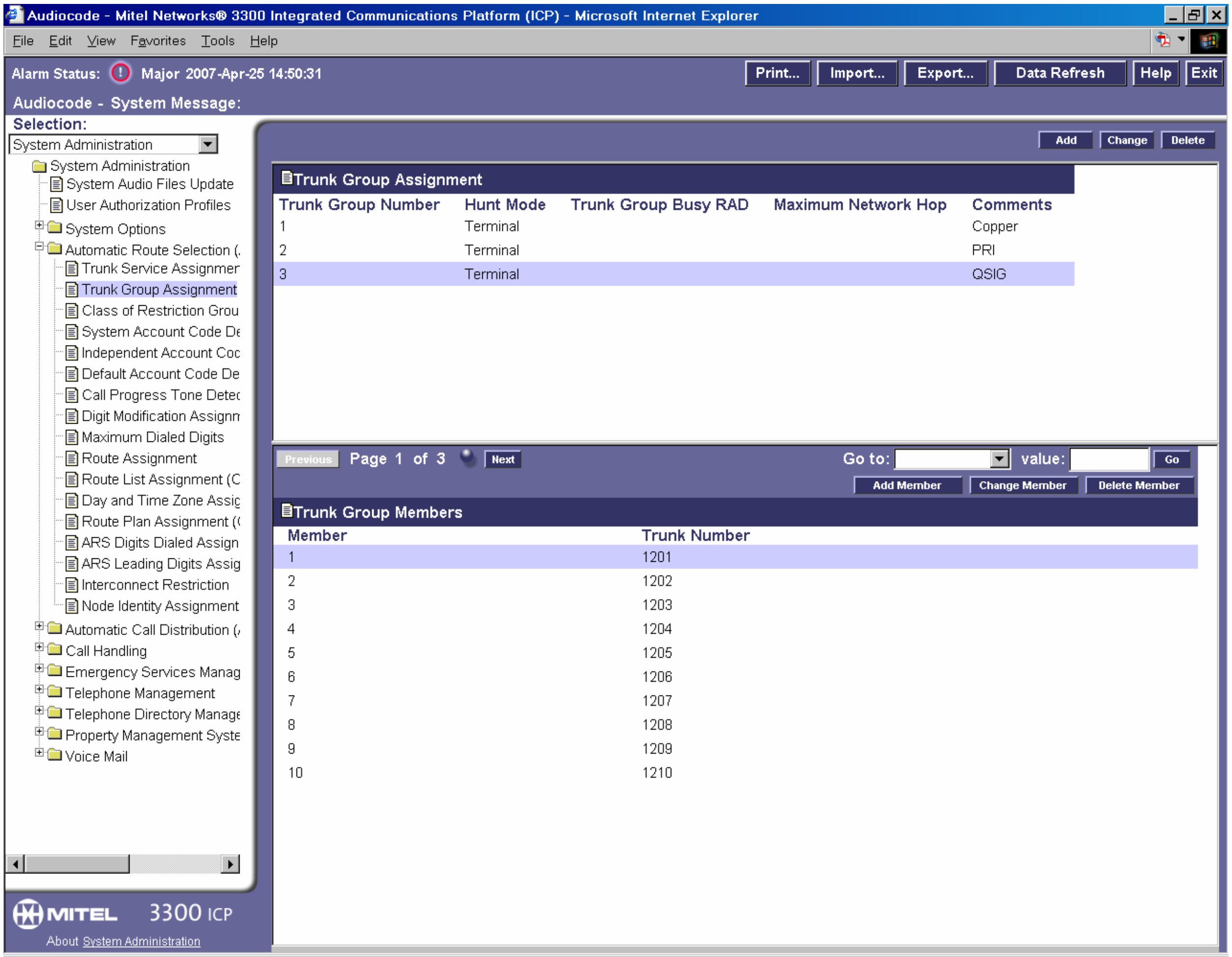



Range Programming -- Web Page Dialog
Change Range Programming - Digital Trunk Assignment
Help
This form allows you to change one or more records, starting at the following record:
Cabinet Shelf Slot Circuit Card Type Trunk Number Trunk Service Number DTS Service Number Circuit Descriptor Number Interconnect Number Tenant Number
6 1 3 1 UNIVERSAL T1 1201 4 4 11 1
1. Enter the number of records to change: 1
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

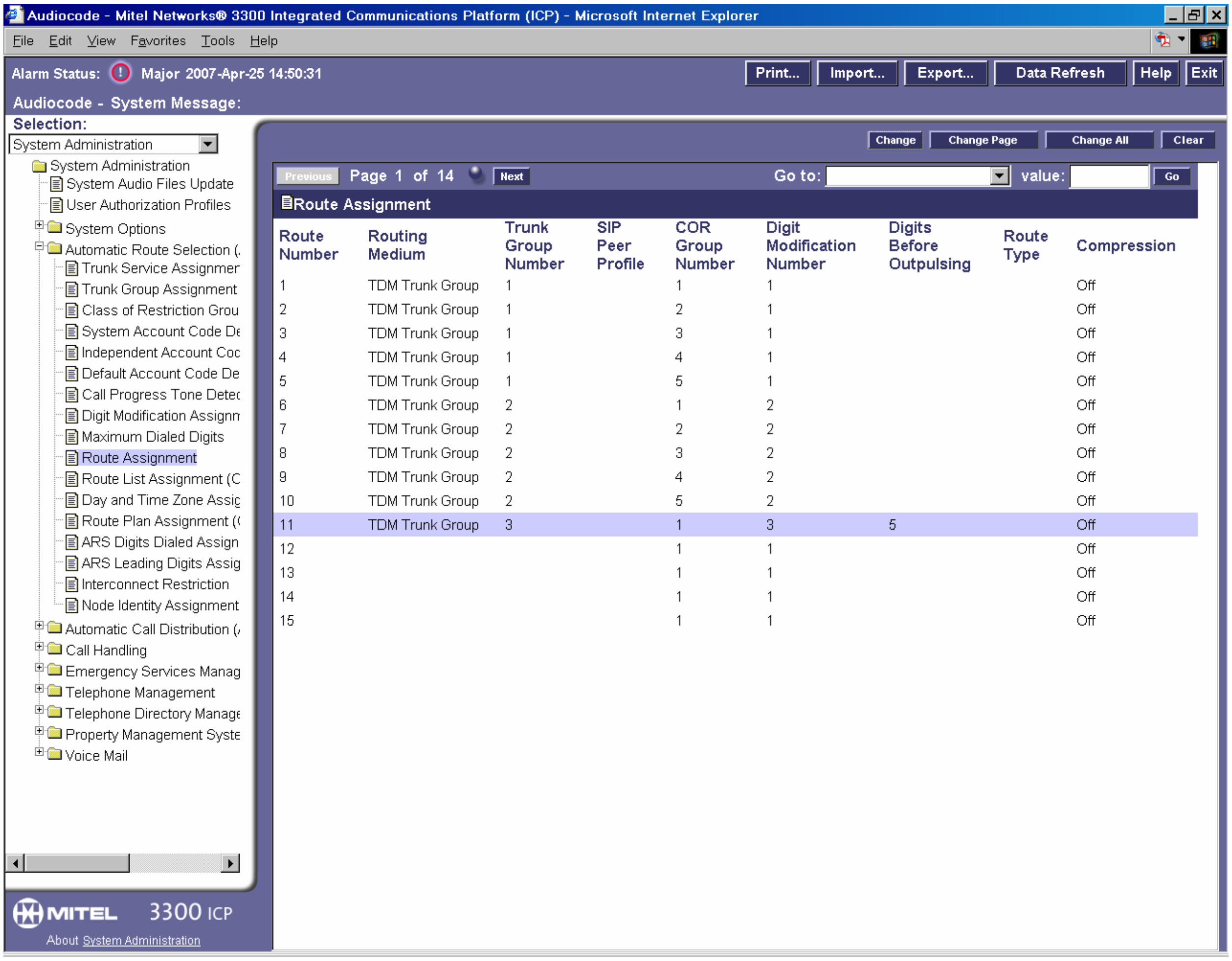
Use the following path to assign ARS for call routing:
System Administration / **Automatic Route Selection** / **Trunk Group Assignment**.

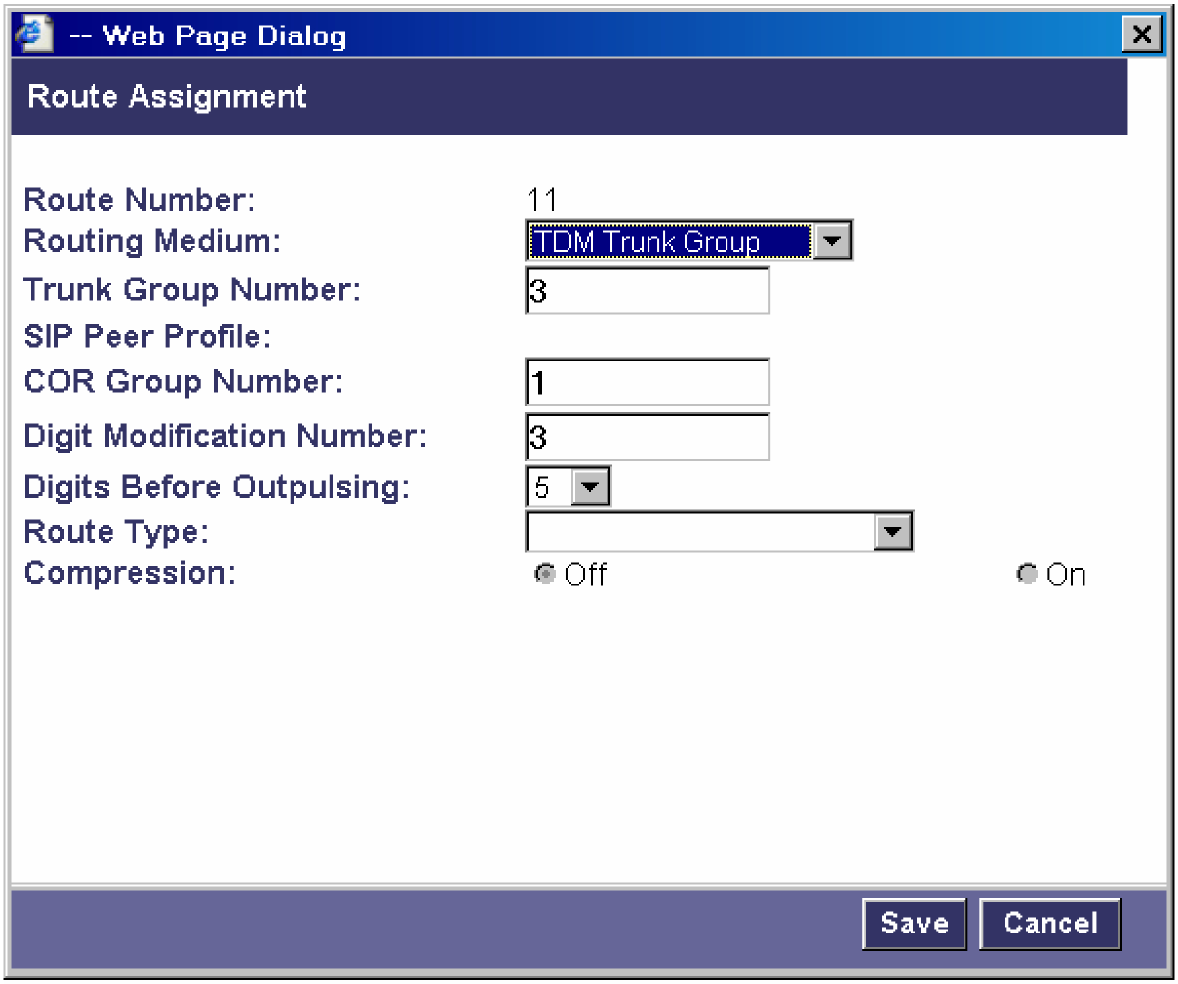
Add a new trunk group for QSIG and add members to this group.



Step 9: Add Route Assignment

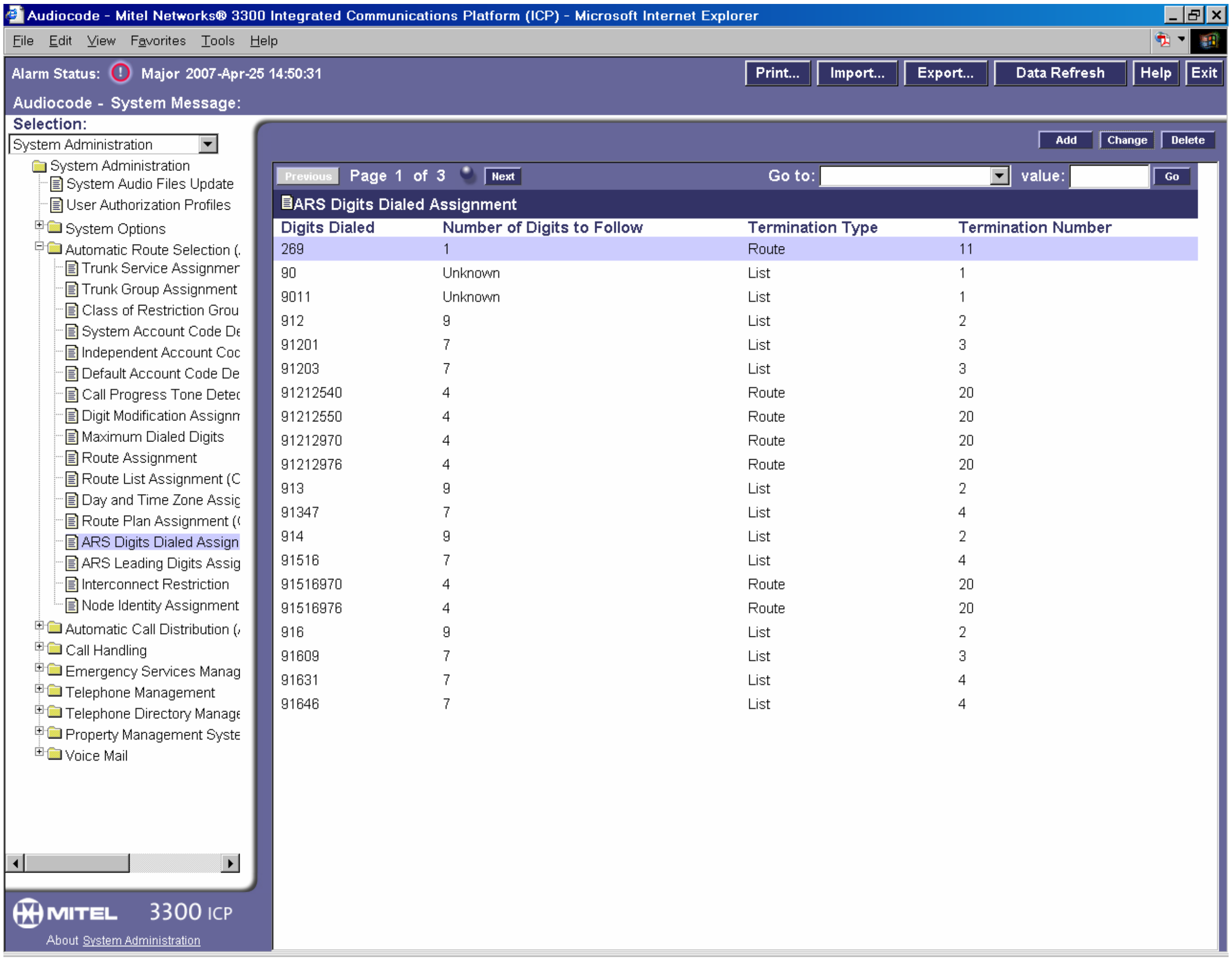
Use the following path to add Route Assignment:
System Administration / **Automatic Route Selection** / **Route Assignment**.

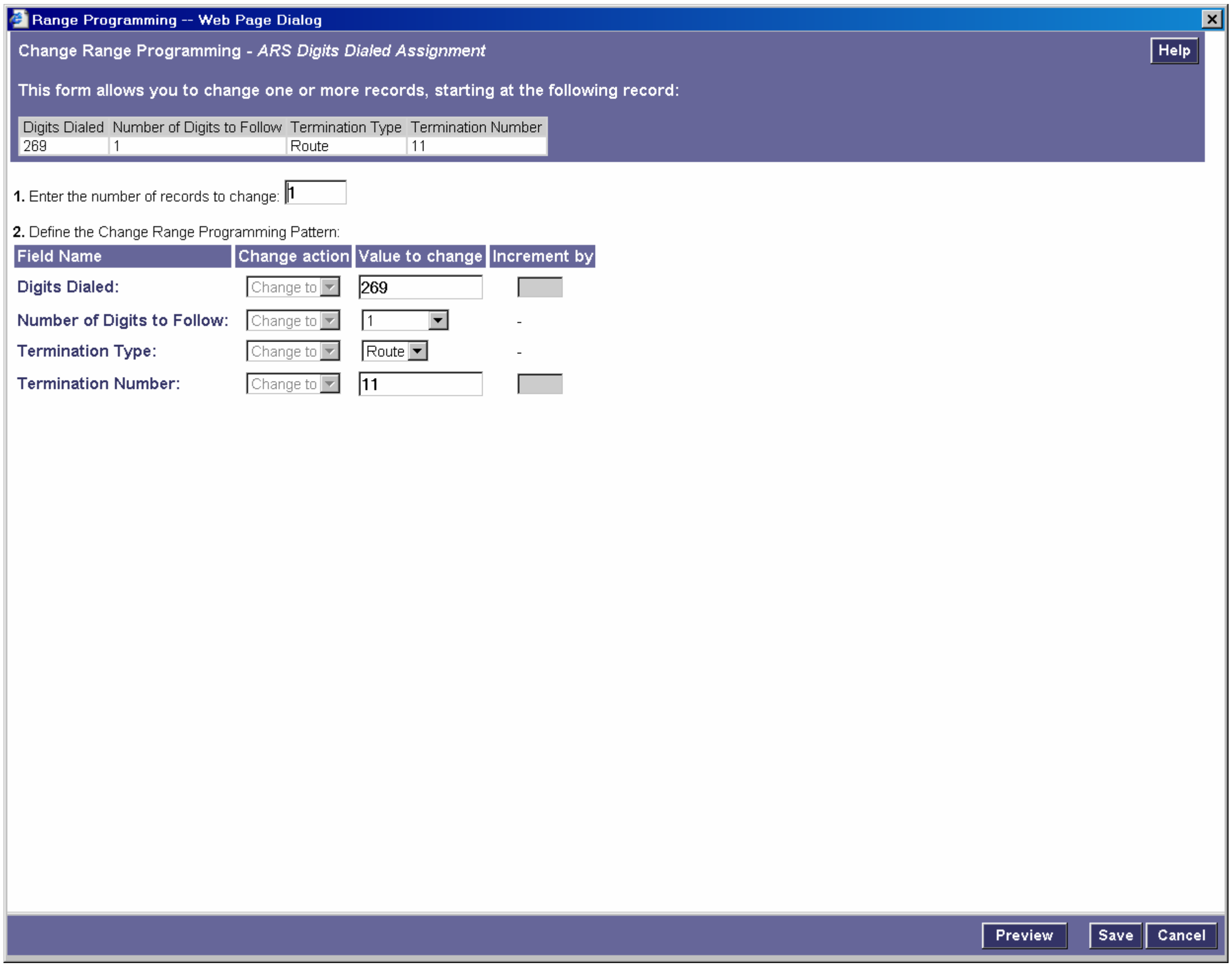



-- Web Page Dialog
Route Assignment
Route Number:
11
Routing Medium:
TDM Trunk Group
Trunk Group Number:
3
SIP Peer Profile:
COR Group Number:
1
Digit Modification Number:
3
Digits Before Outpulsing:
5
Route Type:
Compression:
Off
On
Save
Cancel

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**.

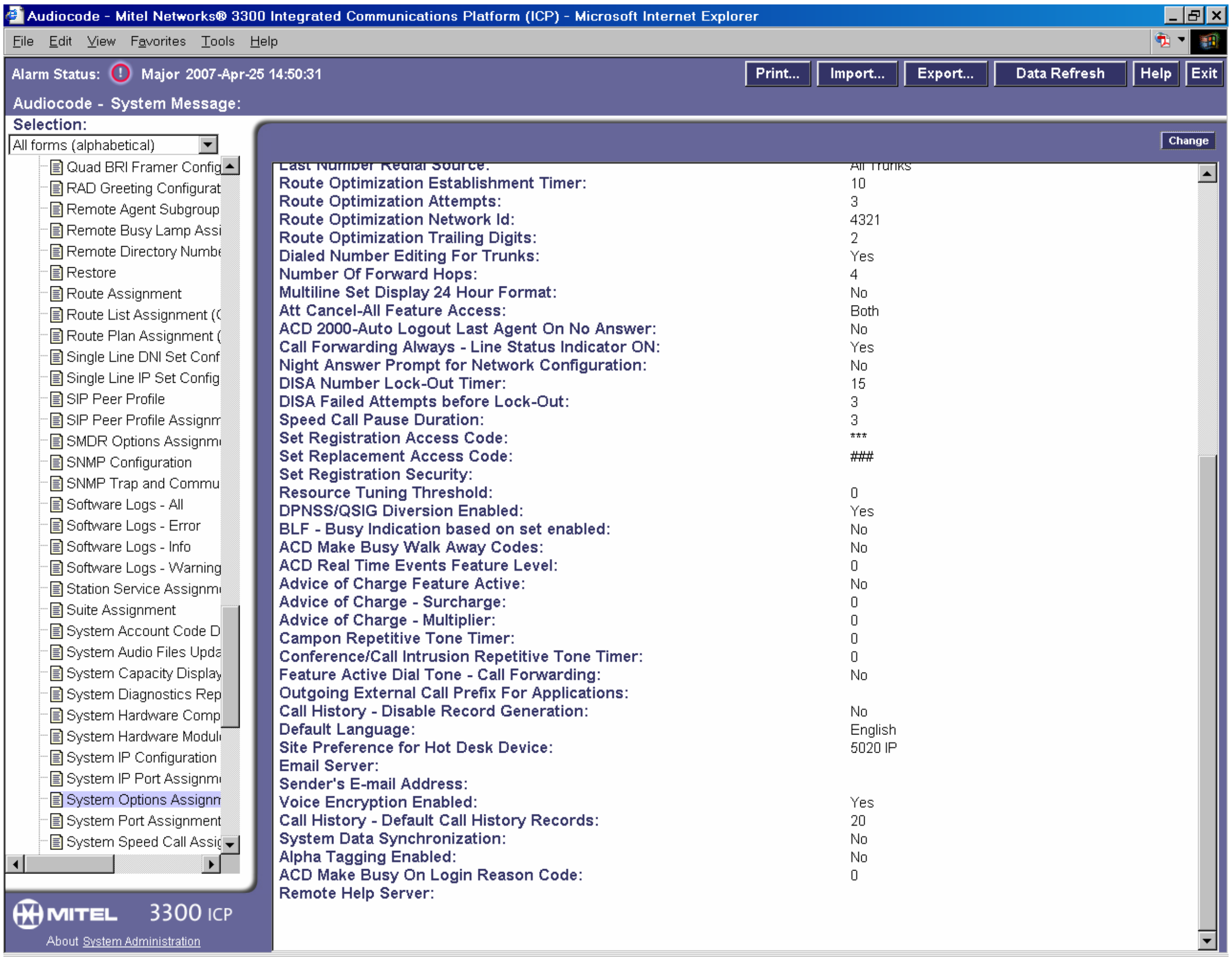



Range Programming -- Web Page Dialog
Change Range Programming - ARS Digits Dialed Assignment
Help
This form allows you to change one or more records, starting at the following record:
Digits Dialed Number of Digits to Follow Termination Type Termination Number
269 1 Route 11
1. Enter the number of records to change: 1
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'.



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	

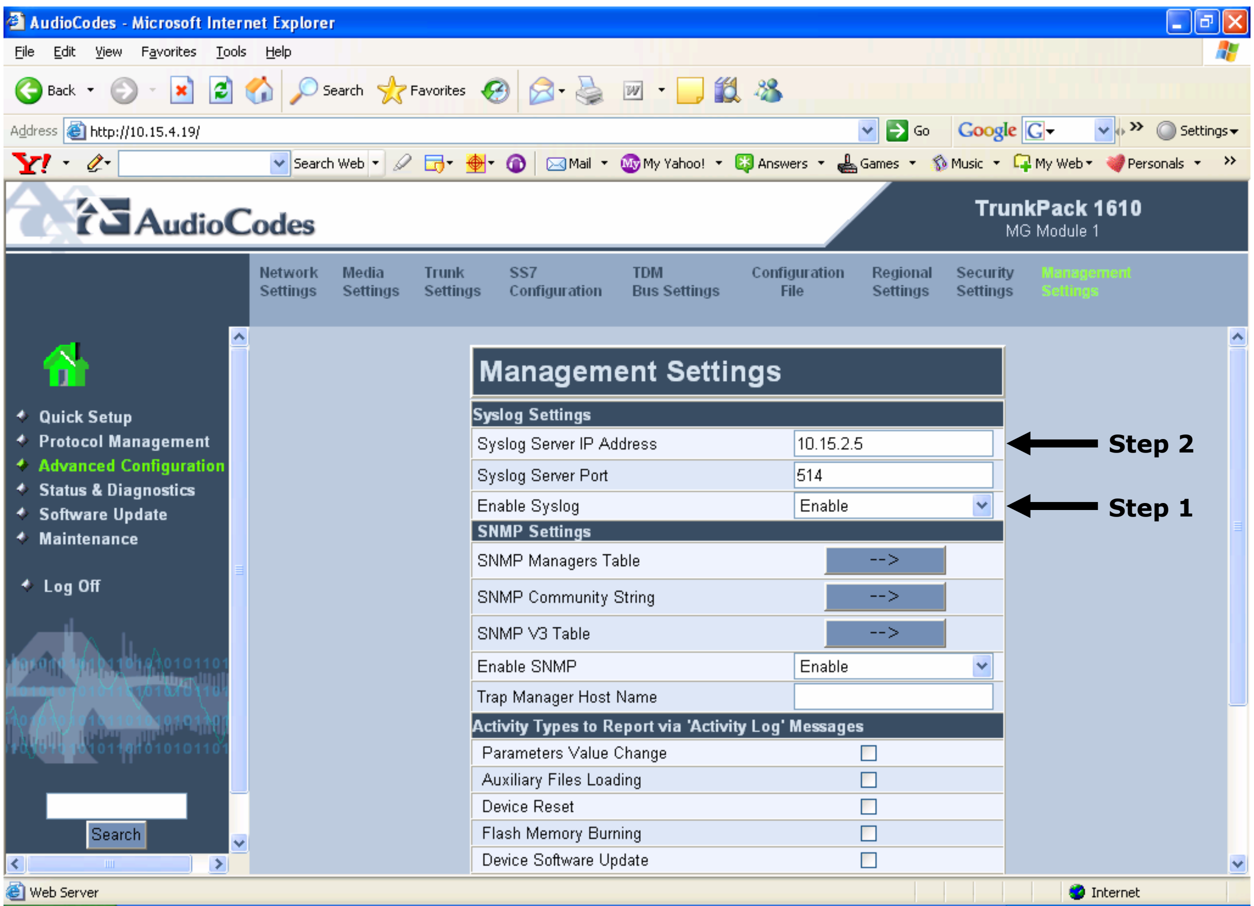
7. Troubleshooting

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.

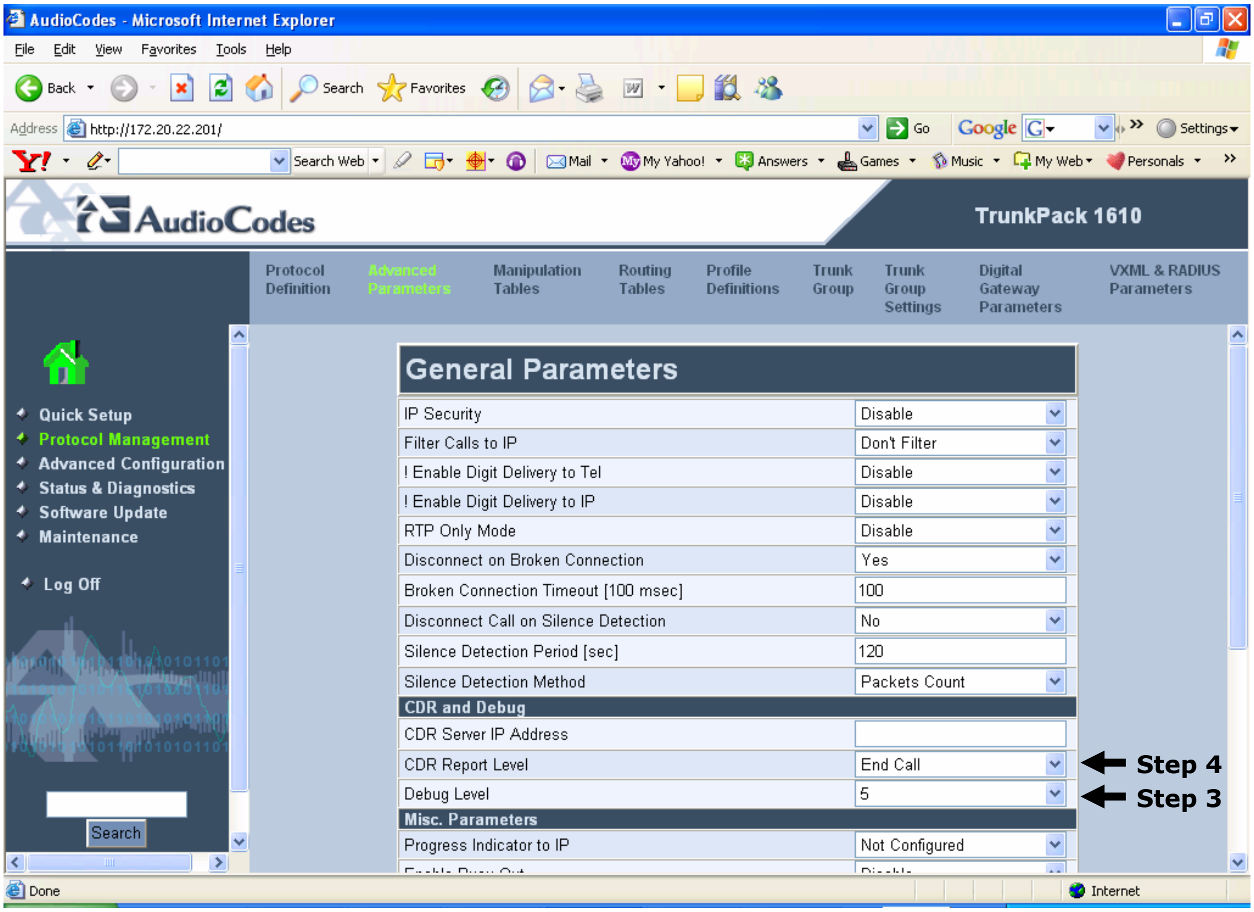
To activate the Syslog client on the AudioCodes gateways:

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.



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.