Dialogic

Dialogic® Media Gateway Installation and Configuration Integration Note

1. Scope

This document is intended to detail a typical installation and configuration of Dialogic® 2000 Media Gateway Series (DMG2000) when used to interface between PBX and Microsoft® Office Communications Server 2007 (OCS) application.

2. Configuration Details

Listed below are the specific details of the PBX and gateways used in the testing to construct the following documentation.

2.1 PBX

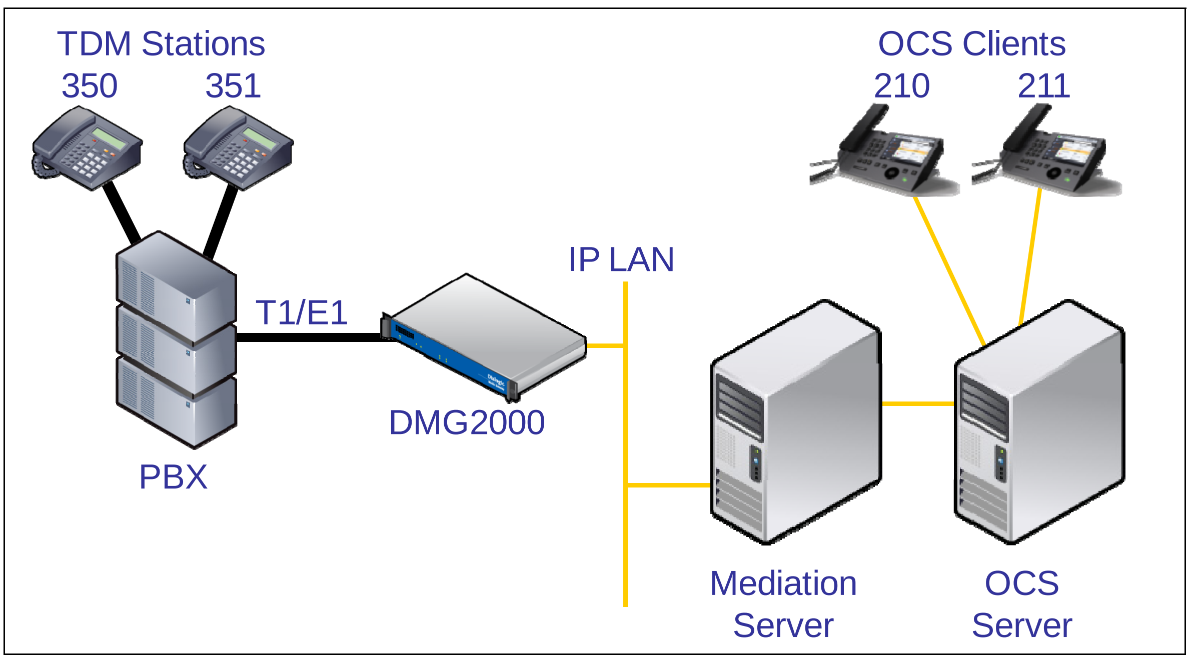
PBX Vendor	Mitel
Model	3300 ICP
Software Version	7.1.6.13
Additional Notes	N/A

2.2 Gateway

Gateway Model	Dialogic® 2000 Media Gateway Series (DMG2000)
Software Version	6.0 (6.0.103)
Protocol	T1 QSIG

2.3 System Diagram

The diagram below details the setup used in the testing and creation of the technical document.



3. Prerequisites

3.1 PBX Prerequisites

PBX must have all supplemental service packages installed for the QSIG protocol to operate properly and provide all advanced supplemental services.

http://www.mitel.com/resources/392_891-51008713RI-EN_FINAL.pdf

3.1.1 PBX Equipment Required

To support the T1 QSIG configuration as documented you need a Mitel 3300 Universal NSU (50001270) with either one or two T1 modules in it. The NSU is external to the 3300 switch.

The 3300 switch must have a fiber optical interface module (typically placed in Module bay number 1).

3.1.2 PBX Cabling Requirements

Cabling for QSIG connections must be CAT5e or better. Standard voice quality cable will not provide optimum signal quality and the gateway will have problems establishing connection on the D-Channel.

Cabling from the NSU to the switch is done with fiber optical cable (FIM link).

3.2 Gateway Prerequisites

The gateway needs to support a T1 QSIG interface.

4. Summary of Limitations

No limitations noted as of the last update to this document.

5. Gateway Setup Notes

Steps for setting up the gateway:

- Parameter Configuration
- Routing Engine Configuration

5.1 Parameter Configuration

To get the gateway connected between the PBX and mediation server there are only a few configuration options that are required.

During the initial setup of the Dialogic gateway using the serial port you must:

- Assign LAN 1 on the gateway a unique IP address, subnet mask and network gateway address (if the latter is required).
- Configure the gateway to use the SIP VoIP protocol.
- Set the Line Mode to T1.
- Set the Protocol to ISDN - QSIG.

During the solution specific setup of the Dialogic gateway using the web interface you must:

- In the IP settings page:

- Set the `BOOTP Enabled` parameter to ‘`No`’. (the default is Yes)

IP Settings, LAN1	
MAC	00-0e-0c-ab-d2-3e
* Client IP Address	192.168.1.2
* Client Subnet Mask	255.255.255.0
* Default Network Gateway Address	192.168.1.250
* BOOTP Enabled	No
* SNTP Server IP Address	

- In the TDM T1/E1 page:
  - Set the Line Encoding and Line Framing as required by your T1 Interface. Typical settings are Encoding = B8ZS and Framing = ESF.

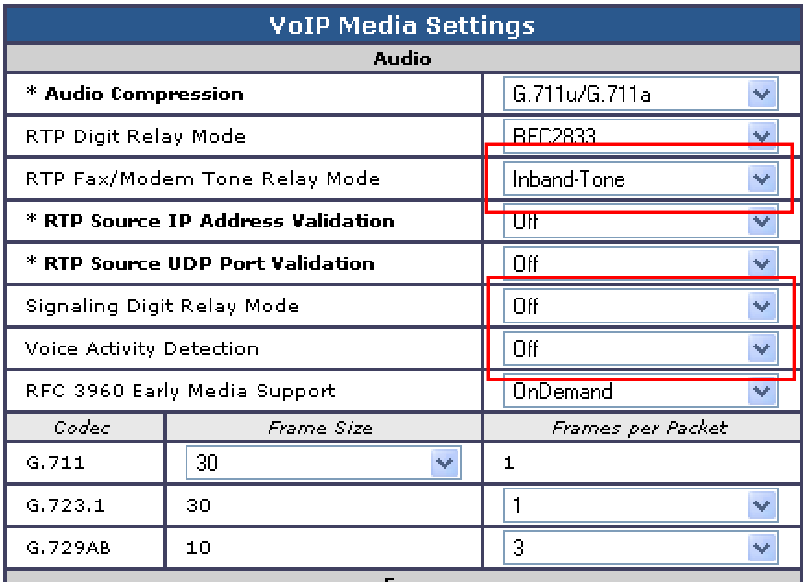
T1/E1 Port Selection	
Select Port to Modify	all ports

T1/E1 Configuration	
Line Settings	
* Line Mode	T1
* Signaling Mode	ISDN
* Telephony Port Interface Side	Terminal
T1 Line	
* Line Encoding	B8ZS
* Framing	ESF
* Selects Transmit Pulse Waveform	Short_Haul_110ft
T1 ISDN protocol	
* ISDN Protocol	QSIG
ISDN Protocol Variant	None
General ISDN Settings	
QSIG Protocol Specification	ISO
Network-Specific Facilities (NSF)	None
ISDN Answer Supervision Enable	Yes
Failover Settings	
* Enable Failover	No

- In the VoIP General page:
  - Set the Transport Type parameter to TCP (the default is UDP)

Voip General Settings	
User-Agent	
* Host and Domain Name	pbxgw.default.com
Transport Type	TCP
Call as Domain Name?	No
SIPS URI Scheme Enabled	No
Invite Expiration (sec)	120

- In the VoIP Media page:
  - Set the `RTP Fax/Modem Tone Relay Mode` parameter to ‘`In band-Tone`’ (the default is RFC2833)
  - Set the `Signaling Digit Relay Mode` parameter to ‘`Off`’ (the default is On)
  - Set the `Voice Activity Detection` parameter to ‘`Off`’ (the default is On)



5.2 Routing Engine Configuration

NOTE: For all the examples in this document going forward the term 'inbound call' refers to a call in the TDM to IP direction and the term 'outbound call' refers to a call in the IP to TDM direction.

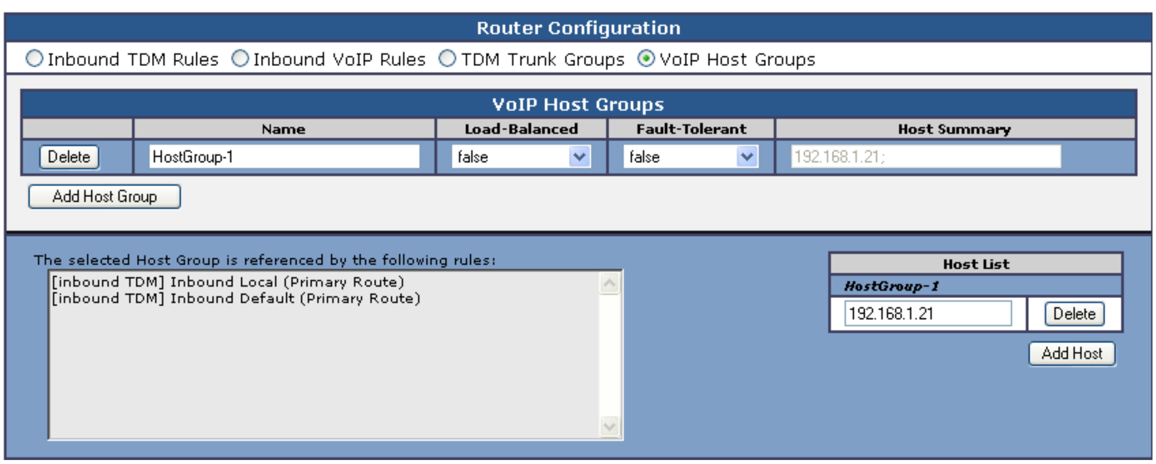
In the example given in the system diagram at the start of this integration guide we see that we have the following dialing plans in the system:

- All TDM side stations have DID numbers assigned in the 3xx extension range.
- All OCS side stations have DID numbers assigned in the 2xx extension range.

We also know that we need to send all inbound calls through to the Mediation Server at a specific IP address.

5.2.1 VoIP Host Group configuration

The first item we should take care of is to set up our IP endpoint to use as our IP destination for all our inbound calls. This is done in the routing table under the section VoIP Host Groups. We define a single host group (using the default group is fine) that includes the IP address of the gateway listening side of the Mediation Server; in our example case we are using the IP address 192.168.1.21 for this.



5.2.2 TDM and VoIP Routing Rule Configuration

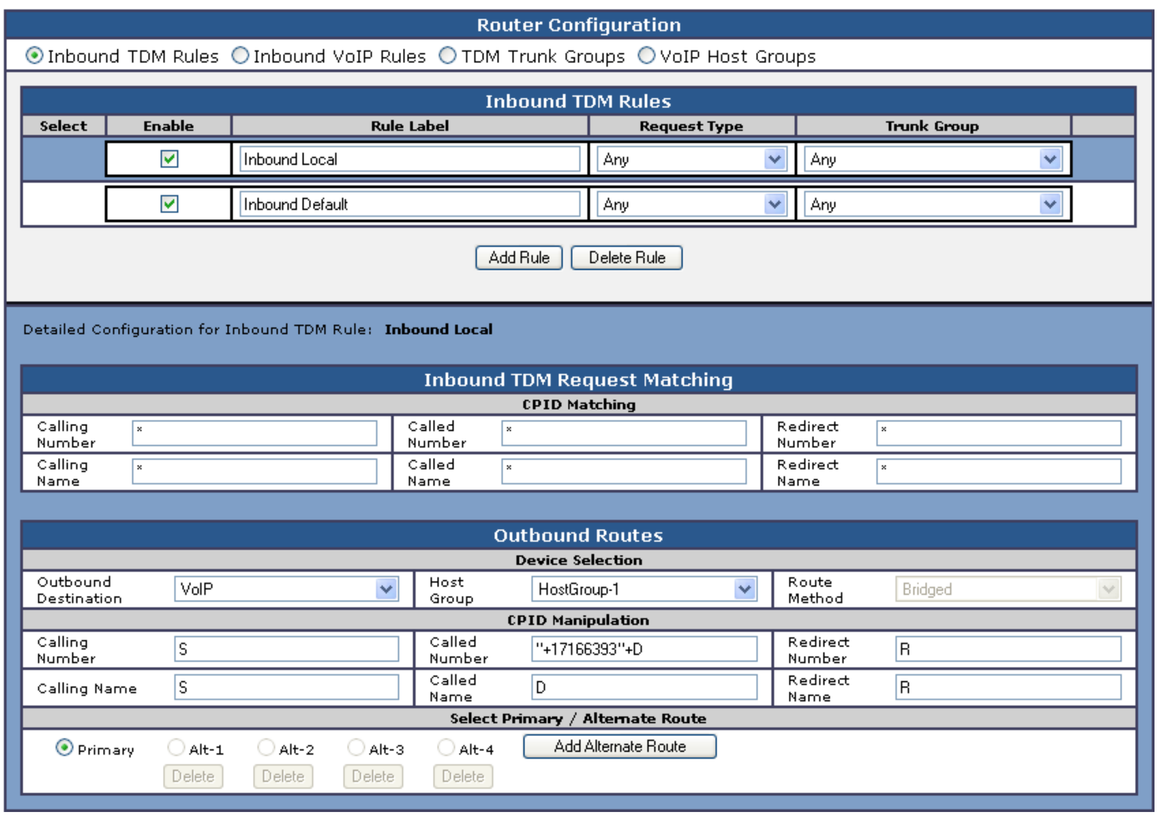
The second item we need to configure are the routing rules that will associate inbound or outbound calls with the proper digit manipulation rules for the type of call they need to service. This will require that the gateway perform some digit manipulation on calls that go from the TDM side to the IP side as well as in the reverse direction, IP to TDM.

The major idea here to remember is that OCS expects to get, and will send out, all addresses in E.164 format. This means that the gateway needs to recognize the need to convert up and down as needed to and from this format as calls pass through. To do this you make use of the Routing engine's CPID manipulation rules.

5.2.2.1 Inbound TDM Rules

When a local user on the PBX picks up their phone and calls one of the extensions on the OCS side within the 2xx range the gateway will receive a call with a calling party of 3 digits. It then needs to convert that number up to full E.164 format and send the call on to OCS.

In our example here we need to take any number that starts in the 2xx range and then convert it into the full E.164 format by concatenating a prefix of '+17166393' onto the front of the number where 716 is the area code and 639 is the local exchange.
Other calls, such as DIDs that arrive over TDM trunks form the PSTN may provide a full 10 digits to the PBX or they may only provide the extension number after the prefix has been stripped off by the PBX. Depending on your site specific requirements you may need to add or build different rules to handle these cases. In our example the inbound rule we use for local PBX users is shown below:



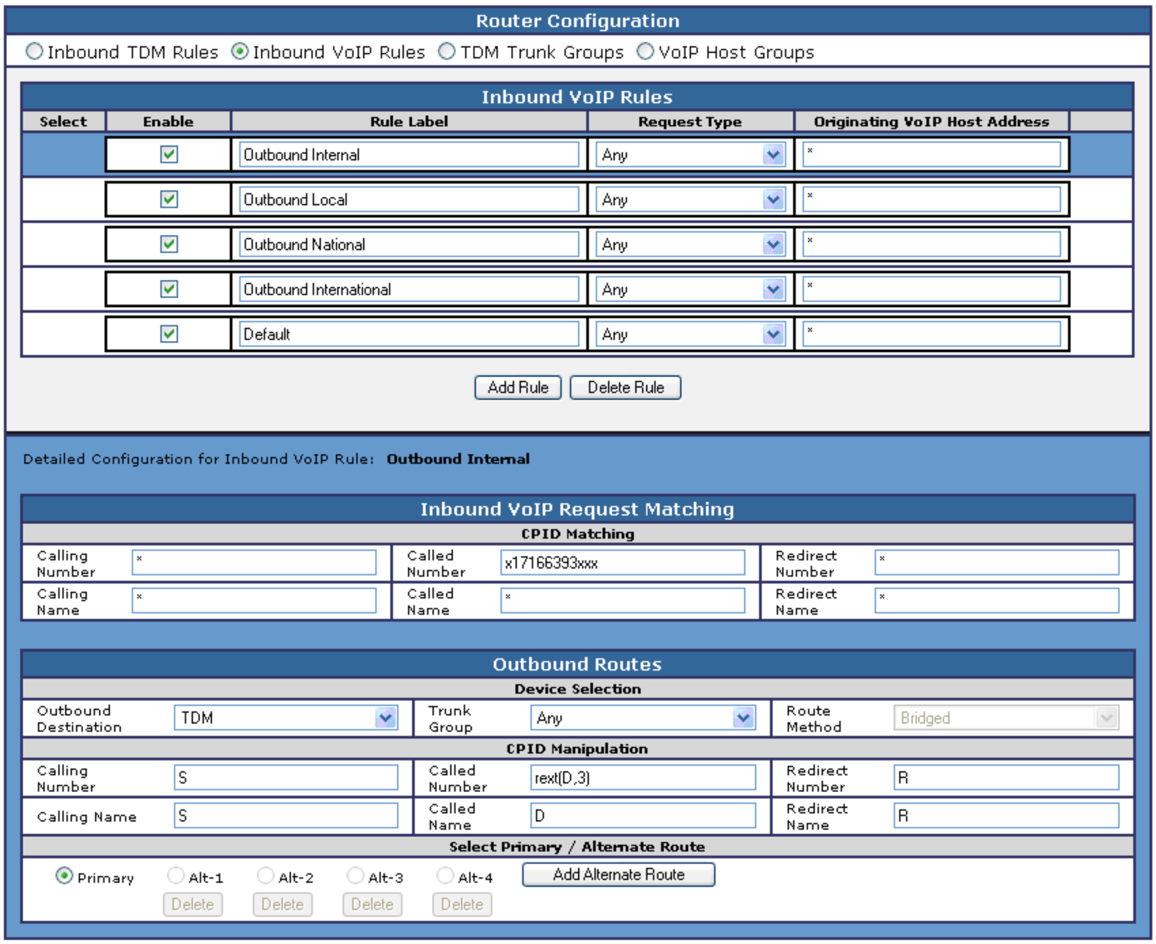
The CPID matching rule is simply a * meaning that any dialed number from a local user presented to this trunk will be seen by this rule. The CPID manipulation rule then uses the digits that are being seen (in this example case it will be a three digit number starting with 2 because that is how the trunk is programmed) and then adds the prefix of "+17166393" onto it to build the full E.164 number that is needed for OCS. This rule also sets the destination to the VoIP Host group we have defined previously that points to the inbound IP address of the Mediation Server.

In addition to this rule, there is a default rule left in place that acts as a catch all. This rule does not do CPID manipulation at all and just sends the call to the VoIP host group as dialed.

5.2.2.2 Inbound VoIP Rules

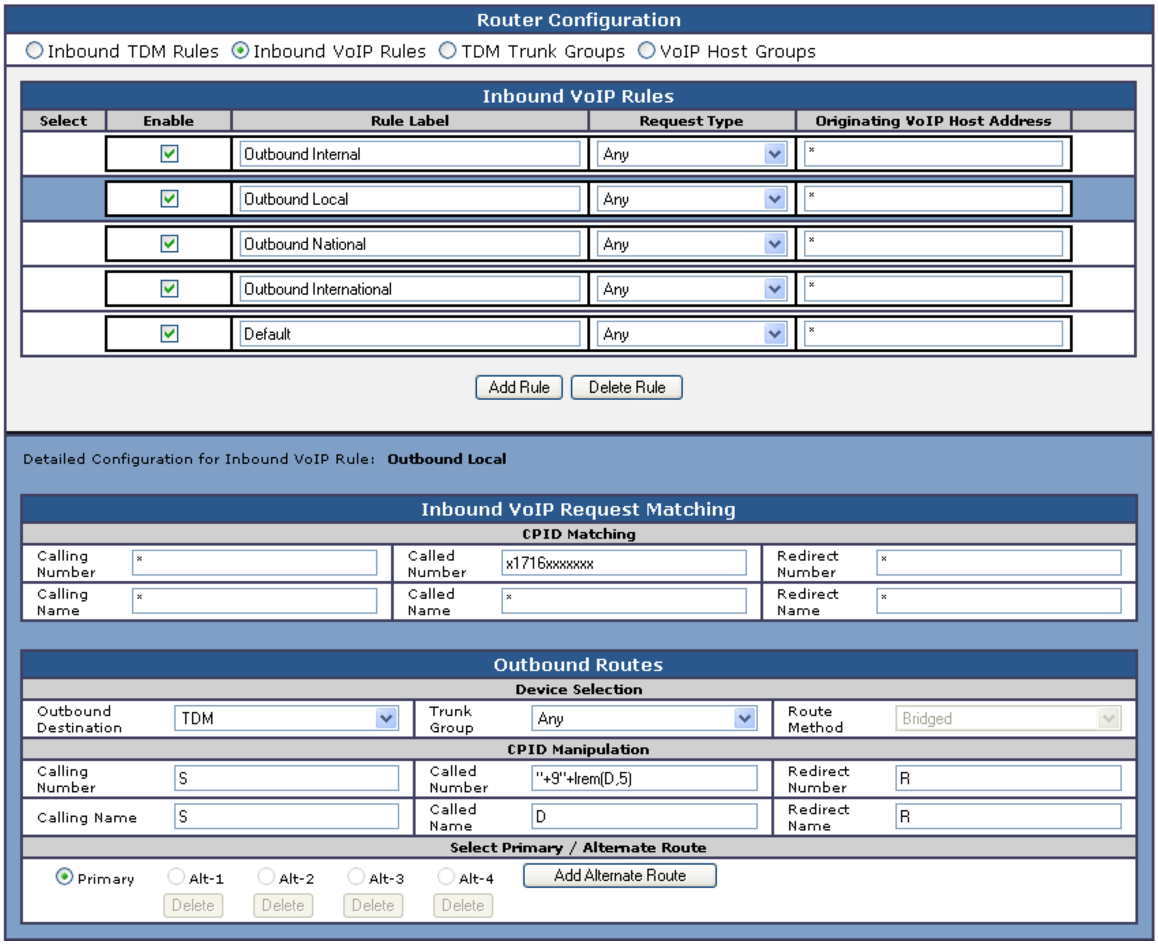
When an OCS user dials a number OCS will, through the use of normalization rules in the Location profile, provide the gateway with a number in full E.164 format so the gateway needs to be able to recognize various number patterns in inbound IP calls and properly manipulate them for the outbound TDM call that results.

In our example here, OCS has been setup (as you will see later) with a route that directs all calls that meet the pattern 5xx to the gateway in full E.164 format. The gateway then needs to know how to identify these numbers as extensions that are local on the PBX and manipulate them accordingly. To do this it needs to simply extract the right 3 digits from the called number provided and removing the prefix of “+17166393” (see the CPID Manipulation section of the next screen shot below).

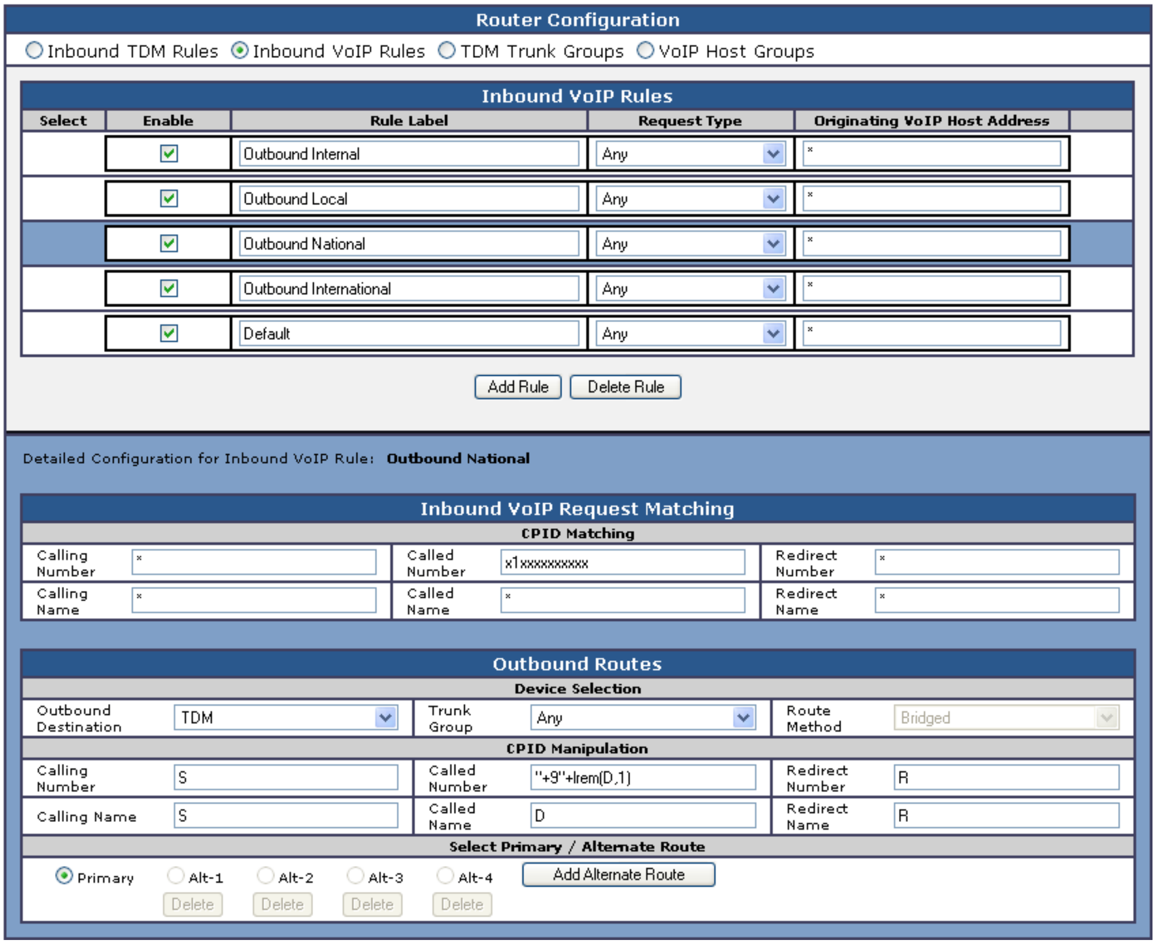


In the screen shot above, the first rule 'Outbound Internal' is selected. Notice that the blue bar near the top of the screen highlights this rule. The lower half of the screen displays the details of the currently selected rule. This rule matches outbound calls (meaning VoIP to TDM) that have a called party number that starts with '+17166393' followed by any three digits. This rule is designed to match the locally defined TDM extensions as shown in the first figure in this document. Calls that match this rule are meant to go to a local user on the PBX. The CPID manipulation section of this rule extracts the last three digits from the called party number. The extracted three digits are then dialed as a local extension on the PBX.

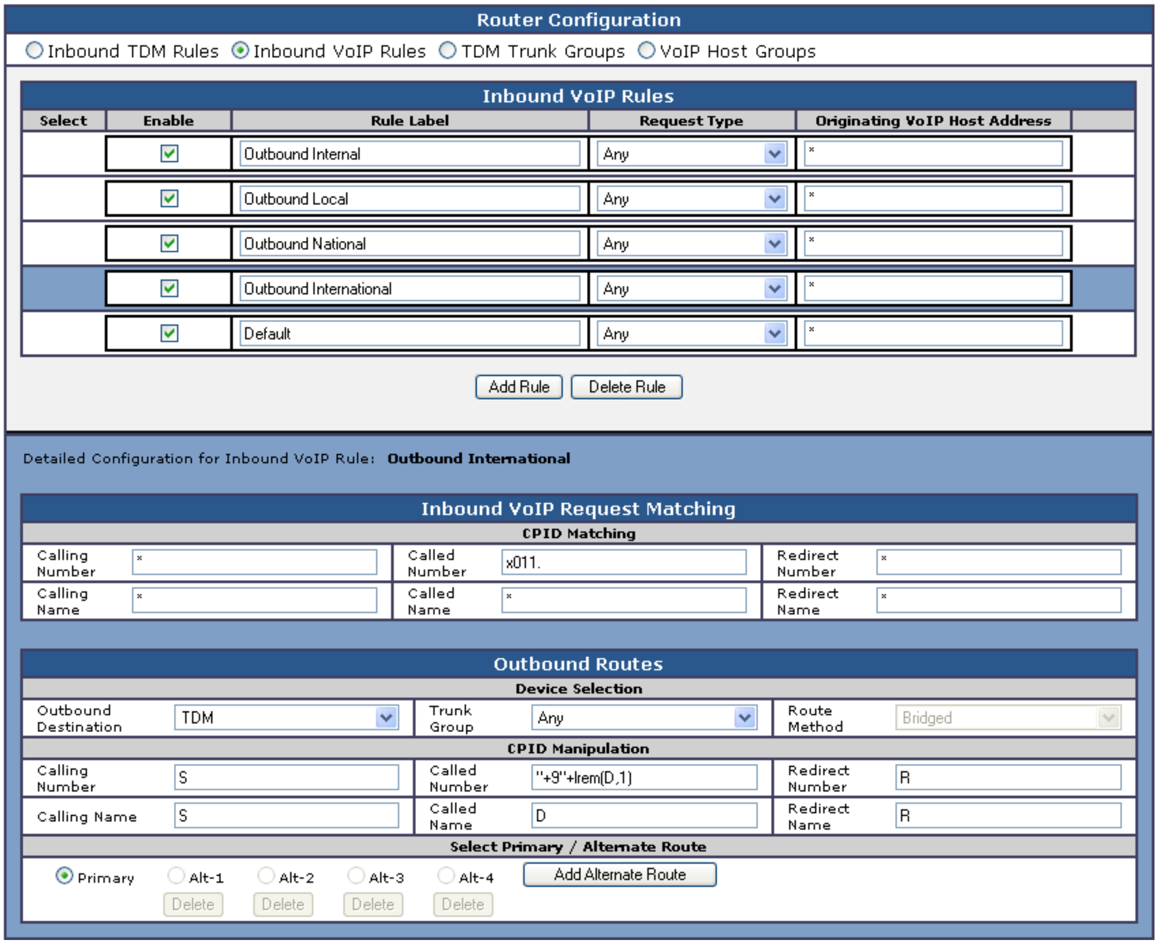
Local, national and international numbers are going to need to be manipulated. At very least they will need a trunk access number, like a 9, prepended onto the front of them in order to dial an outside line. Rules to do this kind of manipulation are shown in the examples below.



This rule, labeled as ‘Outbound Local’, matches any number that starts with ‘+1716’ followed by seven digits. This indicates that it is a local number (outside of the PBX but within the local calling area code of 716) and does not need the area code dialed as part of the number. The CPID manipulation section for the Called Number adds a trunk access code of ‘+9’ to the front of the string and lrem(D,5) strips off the leading five characters (the ‘+1716’). This sends the full string out as ‘+9xxxxxxx’, meaning ‘+9’ followed by the last seven digits of the number from OCS.



This rule, labeled as ‘Outbound National’, matches any number dialed that starts with ‘+1’ followed by ten digits. This will only match numbers not in the local area code because calls to the local area code were handled by the previous rule ‘Outbound Local’. The CPID manipulation section strips off the leading '+', add '+9' to the start of the number creating a result of ‘+91xxxxxxxxxx’.



This rule, labeled as ‘Outbound International’, matches any that starts with ‘+011’ and includes any number of digits after that indicating a number that is not in our local area code and is indeed an international number. In this case the CPID manipulation adds a ‘+9’ to the start of the number and strips off the leading ‘+’ creating a result of ‘+9011xxxxxxxxxx’.

The last rule that you see defined is another default rule that acts as a catch all and simply attempts to dial any number provided that has not matched the previous rules in the list.

Note 1: The last two rules labeled as ‘Outbound National’ and ‘Outbound International’ COULD have been combined into one rule since the CPID manipulation was the same in both. We have split them out here in this example simply for clarity of the example. Also, if your environment uses different trunks for local, national (long Distance) and international calls, breaking these rules out into separate segments allows you to also define trunk groups and direct calls of these specific types to those individual trunks.

Note 2: The rules are evaluated in the order they are listed, top down. The first rule that matches is used so the order is important. Always consider placing your more specific rules at the top of the order and the more general at the bottom.

6. PBX Setup Notes

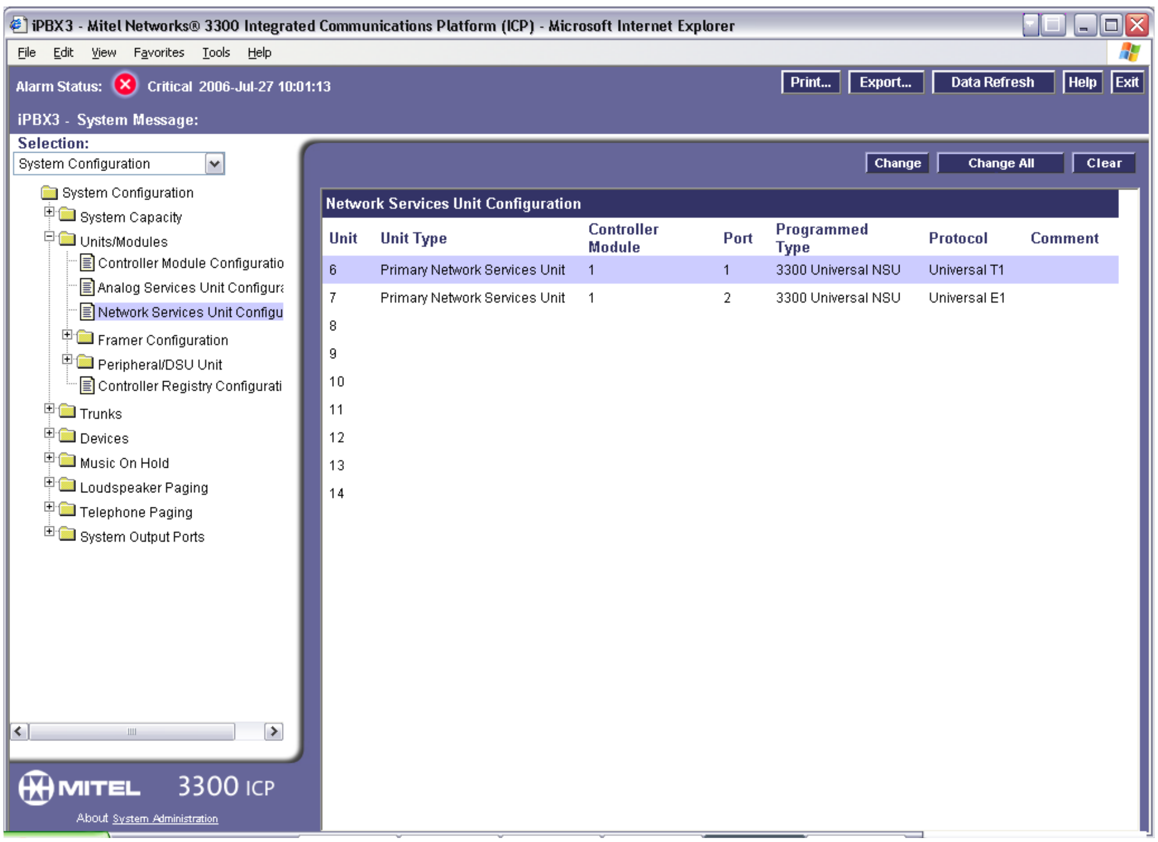
The basic steps of setting up the PBX for use with this gateway and a voice messaging system are as follows:

- Configuring hardware and class of service.
- Configuring ISDN-PRI interface.
- Configuring ARS options.
- Setting up subscriber station sets.

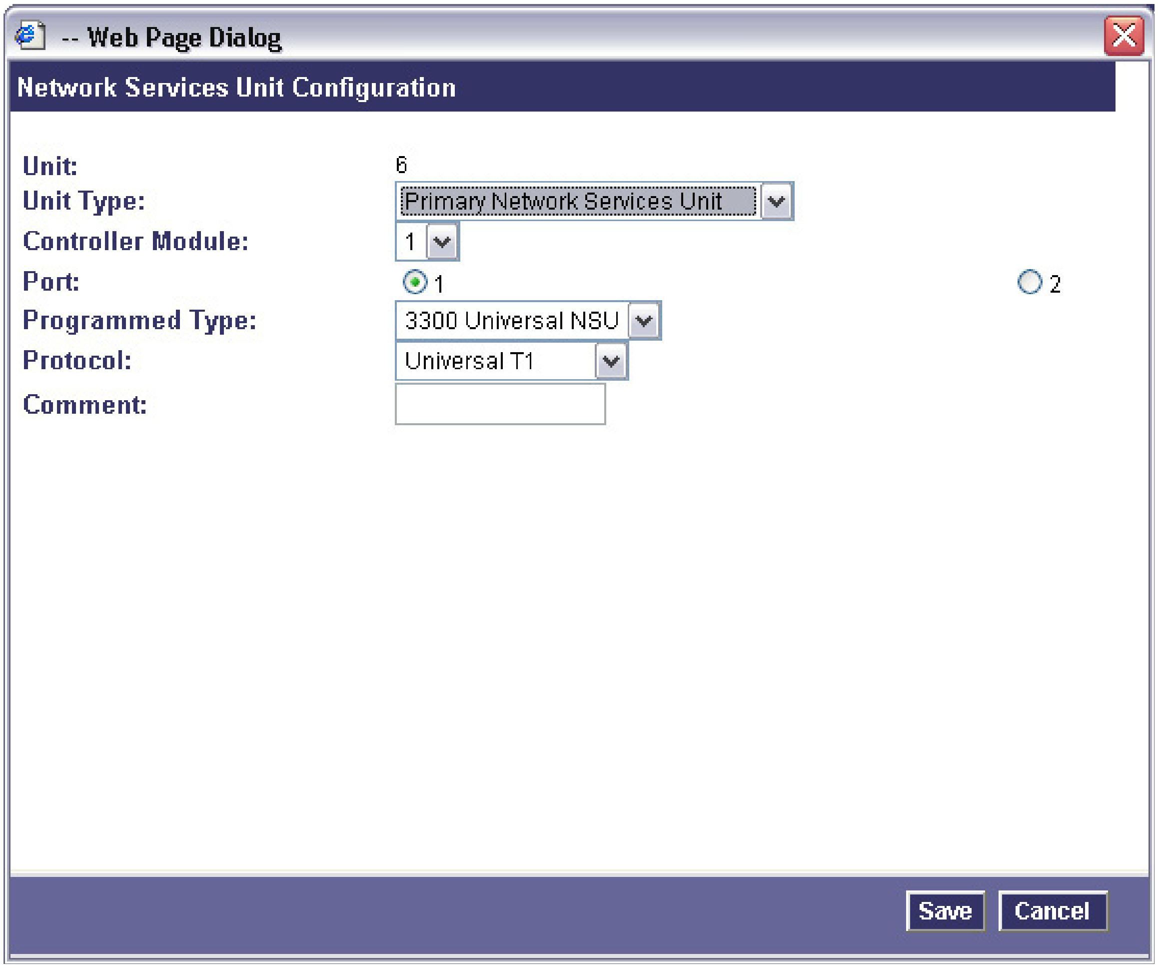
All PBX programming is done via a web browser by connecting to the network port of the PBX.

6.1 Configure Hardware and Class of Service

Use the *Units and Modules Network Services Unit Configuration* menu selection to configure a Network Service Unit. This command sets the options on an installed NSU so it can be configured for a specific type and protocol to be used with the gateway. Select an NSU from the list and click the Change button to configure the options.

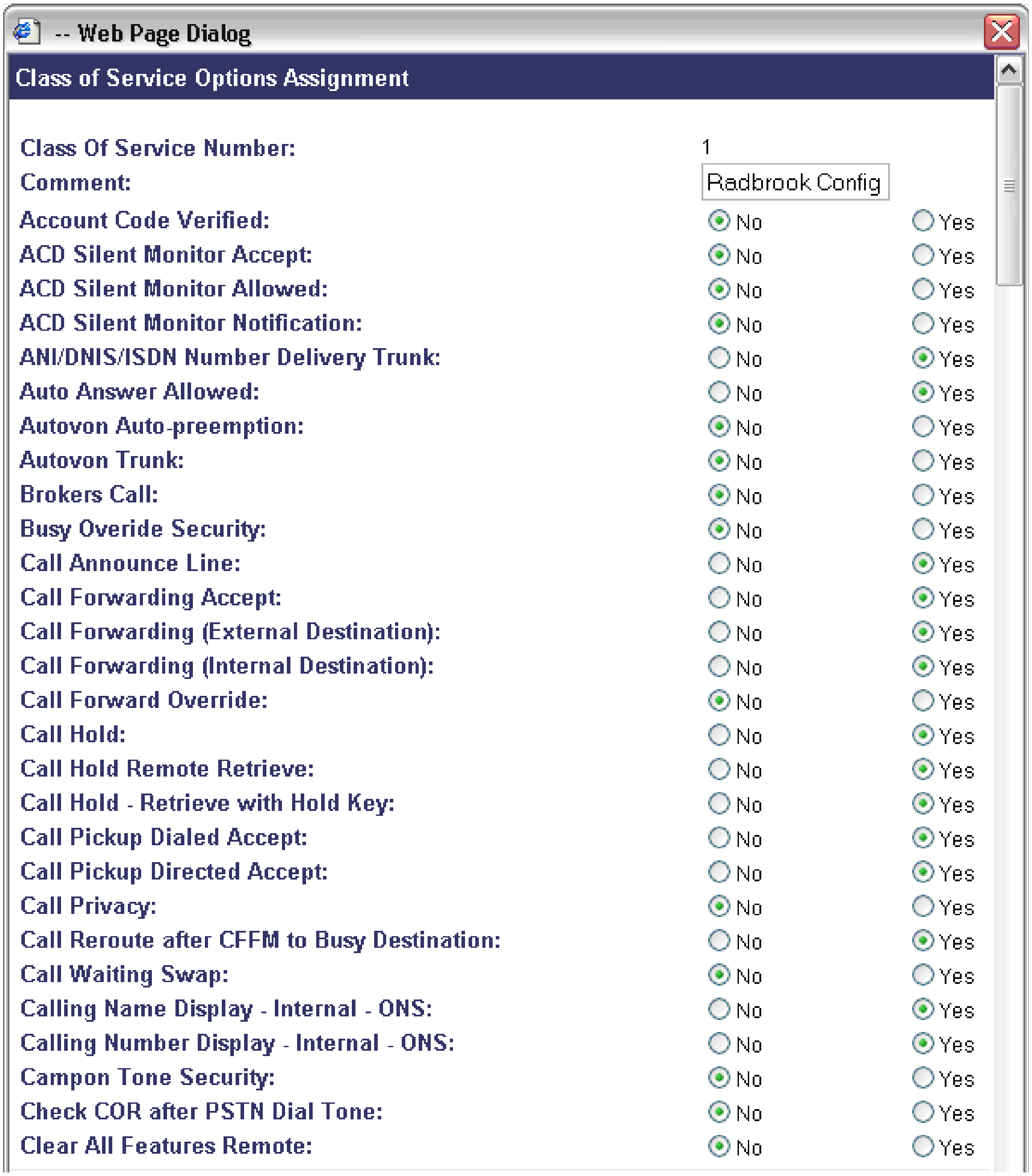


See the screen below for the selected options.



Before moving on to the next task be sure to click the Save button.

Configure a Class of Service Template for the Trunk. Shown below is the listing of all the enabled and disabled classes of service that will be configured on the trunk interface in this example. Yours may very depending on site requirements but keep in mind that disabling certain classes of service will have an effect on certain available features.



-- Web Page Dialog

Class of Service Options Assignment

Class Of Service Number:	1	
Comment:	Radbrook Config	
Account Code Verified:	● No	○ Yes
ACD Silent Monitor Accept:	● No	○ Yes
ACD Silent Monitor Allowed:	● No	○ Yes
ACD Silent Monitor Notification:	● No	○ Yes
ANI/DNIS/ISDN Number Delivery Trunk:	○ No	● Yes
Auto Answer Allowed:	○ No	● Yes
Autovon Auto-preemption:	● No	○ Yes
Autovon Trunk:	● No	○ Yes
Brokers Call:	● No	○ Yes
Busy Overide Security:	● No	○ Yes
Call Announce Line:	○ No	● Yes
Call Forwarding Accept:	○ No	● Yes
Call Forwarding (External Destination):	○ No	● Yes
Call Forwarding (Internal Destination):	○ No	● Yes
Call Forward Override:	● No	○ Yes
Call Hold:	○ No	● Yes
Call Hold Remote Retrieve:	○ No	● Yes
Call Hold - Retrieve with Hold Key:	○ No	● Yes
Call Pickup Dialed Accept:	○ No	● Yes
Call Pickup Directed Accept:	○ No	● Yes
Call Privacy:	● No	○ Yes
Call Reroute after CFFM to Busy Destination:	○ No	● Yes
Call Waiting Swap:	● No	○ Yes
Calling Name Display - Internal - ONS:	○ No	● Yes
Calling Number Display - Internal - ONS:	○ No	● Yes
Campon Tone Security:	● No	○ Yes
Check COR after PSTN Dial Tone:	● No	○ Yes
Clear All Features Remote:	● No	○ Yes

Conference Call:	○ No	◉ Yes
COV/ONS/E&M Voice Mail Port:	◉ No	○ Yes
DASS II OLI/TLI Provided:	◉ No	○ Yes
Dialled Night Service:	○ No	◉ Yes
Disable Send Message:	◉ No	○ Yes
Display ANI/ISDN Calling Number Only:	◉ No	○ Yes
Display ANI/DNIS/ISDN Calling/Called Number:	○ No	◉ Yes
Display Caller ID on multicall/keylines:	○ No	◉ Yes
Display DNIS/Called Number Before Digit Modification:	○ No	◉ Yes
Display Dialed Digits during Outgoing Calls:	○ No	◉ Yes
Display Held Call ID on Transfer:	○ No	◉ Yes
Do Not Disturb:	○ No	◉ Yes
Do Not Disturb - Access to Remote Phones:	○ No	◉ Yes
Do Not Disturb Permanent:	◉ No	○ Yes
Emergency Call Notification - Audio:	◉ No	○ Yes
Emergency Call Notification - Visual:	◉ No	○ Yes
Enable Call Duration Limit on External Calls:	◉ No	○ Yes
Enable Call Duration Limit on Internal Calls:	◉ No	○ Yes
Executive Busy Override:	◉ No	○ Yes
External Trunk Standard Ringback:	◉ No	○ Yes
Flexible Answer Point:	◉ No	○ Yes
Follow 2nd Alternate Reroute for Recall to Busy ACD Agent:	◉ No	○ Yes
Forced Verified Account Code:	◉ No	○ Yes
Forced Non-Verified Account Code:	◉ No	○ Yes
Group Call Forward Follow Me Accept:	◉ No	○ Yes
Group Call Forward Follow Me Allow:	◉ No	○ Yes
Group Page Accept:	◉ No	○ Yes
Group Page Allow:	◉ No	○ Yes
Handset Volume Adjustment Saved:	◉ No	○ Yes
Handsfree AnswerBack Allowed:	◉ No	○ Yes
HCI/CTI/TAPI Call Control Allowed:	◉ No	○ Yes

HCI/CTI/TAPI Monitor Allowed:	(•) No	( ) Yes
Head Set Switch Mute:	(•) No	( ) Yes
Hot Desk Remote Logout Enabled:	(•) No	( ) Yes
Hot Desk Login Accept:	(•) No	( ) Yes
Hotel Room Extension:	(•) No	( ) Yes
Hotel Room Monitor Setup Allowed:	(•) No	( ) Yes
Hotel Room Monitoring Allowed:	(•) No	( ) Yes
Hotel/Motel Room Personal Wakeup Call Allowed:	(•) No	( ) Yes
Hotel/Motel Room Remote Wakeup Call Allowed:	(•) No	( ) Yes
Individual Trunk Access:	( ) No	(•) Yes
Keep TelDir Entry on Check Out:	(•) No	( ) Yes
Local Music On Hold source:	(•) No	( ) Yes
Loudspeaker Pager Override:	( ) No	(•) Yes
Loudspeaker Pager Equivalent Zone Override Security:	(•) No	( ) Yes
Message Waiting:	( ) No	(•) Yes
Message Waiting Audible Tone Notification:	( ) No	(•) Yes
Message Waiting Deactivate On Off-Hook:	( ) No	(•) Yes
Message Waiting Inquire:	( ) No	(•) Yes
Multiline Set Loop Test:	(•) No	( ) Yes
Multiline Set Message Center Remote Read Allowed:	(•) No	( ) Yes
Multiline Set Music:	(•) No	( ) Yes
Multiline Set On-hook Dialing:	( ) No	(•) Yes
Multiline Set Phonebook Allowed:	( ) No	(•) Yes
Multiline Set Voice Mail Callback Message Erasure Allowed:	(•) No	( ) Yes
Name Suppression on outgoing Trunk Call:	(•) No	( ) Yes
Non DID Extension:	(•) No	( ) Yes
Non-Prime Public Network Identity:	(•) No	( ) Yes
Non Verified Account Code:	( ) No	(•) Yes
Off-Hook Voice Announce Allowed:	(•) No	( ) Yes
ONS CLASS/CLIP: Message Waiting Activate/Deactivate:	( ) No	(•) Yes
ONS CLASS/CLIP: Set:	( ) No	(•) Yes
ONS CLASS/CLIP: Visual Call Waiting:	( ) No	(•) Yes
ONS/OPS Internal Ring Cadence for External Callers:	(•) No	( ) Yes

Override Interconnect Restriction on Transfer:	(•) No	( ) Yes
Pager Access All Zones:	( ) No	(•) Yes
Pager Access Individual Zones:	(•) No	( ) Yes
Privacy Released:	(•) No	( ) Yes
Public Network Access via DPNSS:	( ) No	(•) Yes
Public Network Identity Provided:	( ) No	(•) Yes
Public Network To Public Network Connection Allowed:	( ) No	(•) Yes
Public Trunk:	( ) No	(•) Yes
R2 Call Progress Tone:	( ) No	(•) Yes
Record-A-Call Active:	(•) No	( ) Yes
Record-A-Call - Start Recording Automatically:	(•) No	( ) Yes
Record-A-Call - Save Recording on Hang-up:	(•) No	( ) Yes
Recorded Announcement Device:	(•) No	( ) Yes
Recorded Announcement Device - Advanced:	(•) No	( ) Yes
Redial Facilities:	( ) No	(•) Yes
Ringing Line Select:	(•) No	( ) Yes
SC1000 Attendant Basic Function Key:	(•) No	( ) Yes
SMDR External:	(•) No	( ) Yes
SMDR Internal:	(•) No	( ) Yes
Speak@Ease Preferred:	(•) No	( ) Yes
Suite Services Enabled:	(•) No	( ) Yes
Suppress Simulated CCM after ISDN Progress:	(•) No	( ) Yes
Third Party Call Forward Follow Me Accept:	( ) No	(•) Yes
Third Party Call Forward Follow Me Allow:	( ) No	(•) Yes
Timed Reminder Allowed:	( ) No	(•) Yes
Trunk Calling Party Identification:	( ) No	(•) Yes
Trunk Flash Allowed:	( ) No	(•) Yes
Use Held Party Device for Call Re-routing:	( ) No	(•) Yes
Use Called Party Call Hold Timer:	(•) No	( ) Yes
Voice Mail Softkey:	(•) No	( ) Yes
Account Code Length:	12	
After Answer Display Time:		

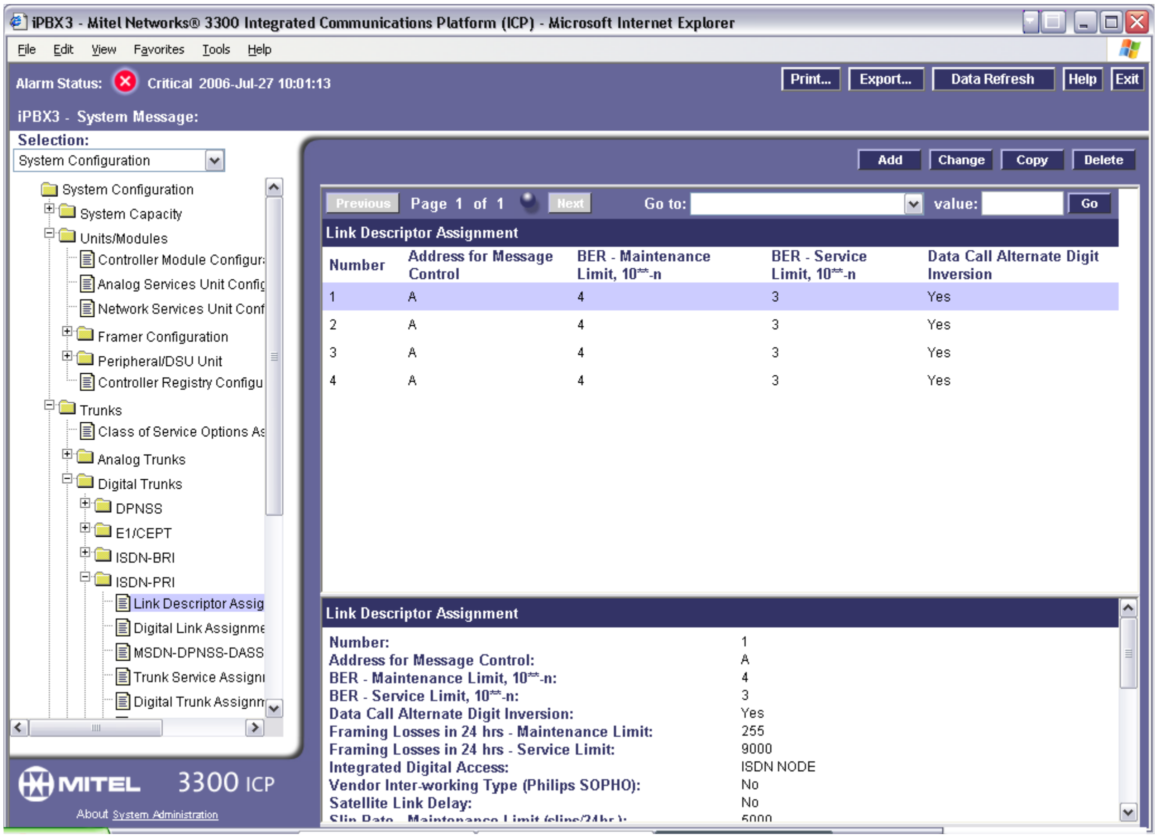
Answer Plus Delay To Message Timer:	20
Answer Plus Expected Off-hook Timer:	30
Answer Plus Message Length Timer:	10
Answer Plus System Reroute Timer:	0
Attendant Busy Out Timer:	10
Auto Campon Timer:	10
Autovon Precedence:	4
Busy Tone Timer:	30
Call Duration:	10
Call Duration Forced Cleardown Timer:	0
Call Forward - Delay:	0
Call Forward No Answer Timer:	15
Call Hold Timer:	30
Campon Recall Timer:	10
Delay Ring Timer:	10
Dialing Conflict Timer:	3
Display Caller ID On Multicall/Keylines Timer:	5
Emergency Call - Audio Level for Set:	Ringer
First Digit Timer:	15
Inter Digit Timer:	10
Lockout Timer:	45
ACD 2000 Logout Agent No Answer Timer:	15
Message Waiting Ringing Start Time Hour:	
Message Waiting Ringing Start Time Minute:	
Message Waiting Ringing Stop Time Hour:	
Message Waiting Ringing Stop Time Minute:	
No Answer Recall Timer:	17
ONS VMail-Delay Dial Tone Timer:	5
Ringing Timer:	180
Work Timer:	0
Key A:	
Key B:	
Key C:	
Key D:	

Save Cancel

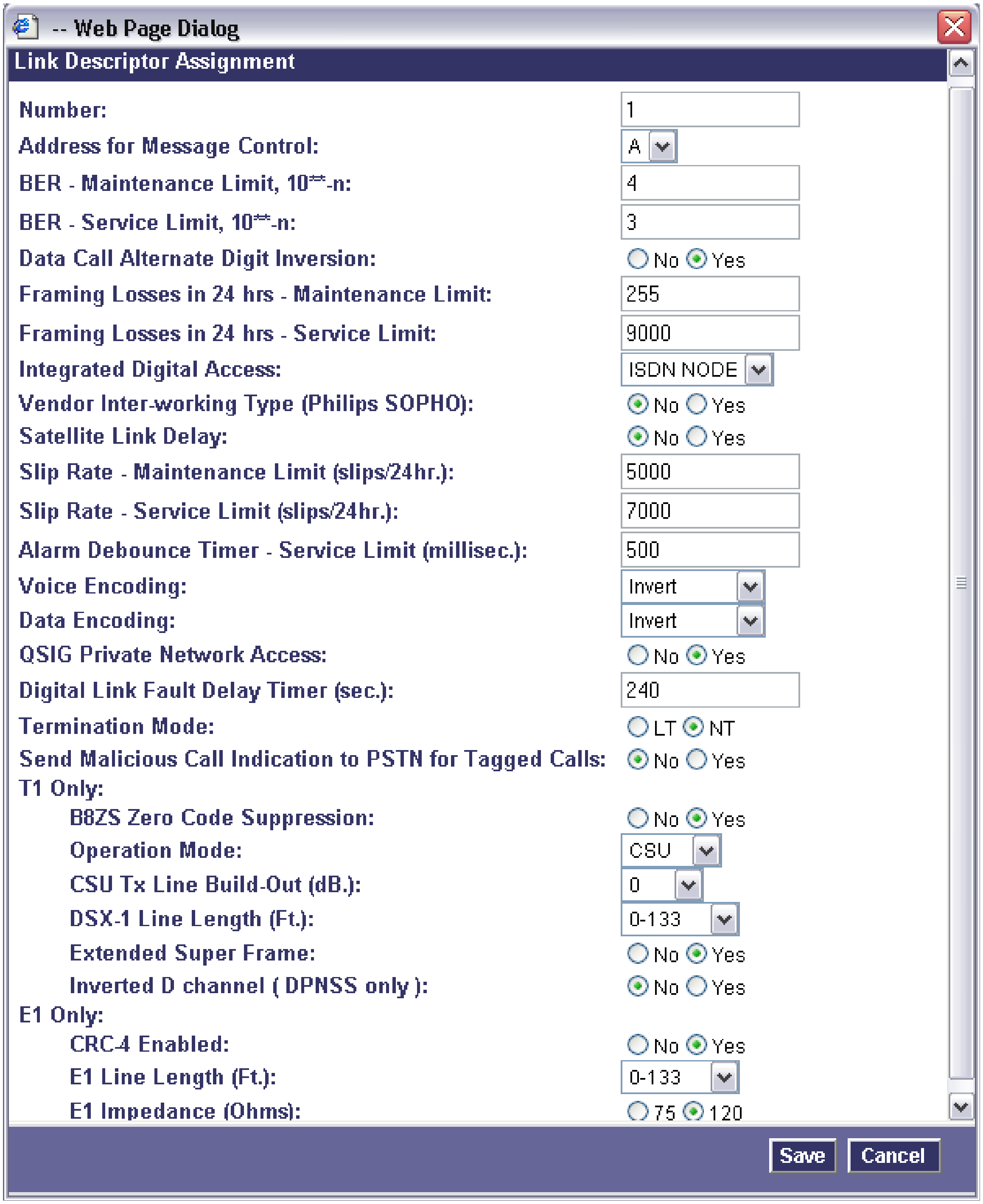
Before moving on to the next task be sure to click the Save button.

6.2 Configuring ISDN-PRI Interface

Use the *ISDN PRI Link Descriptor Assignment* menu selection to build a template that sets the various options for a trunk interface such as framing, coding and interface type. Use the *Add* or *Change* buttons to either build a new Link Descriptor or modify an existing Link Descriptor.

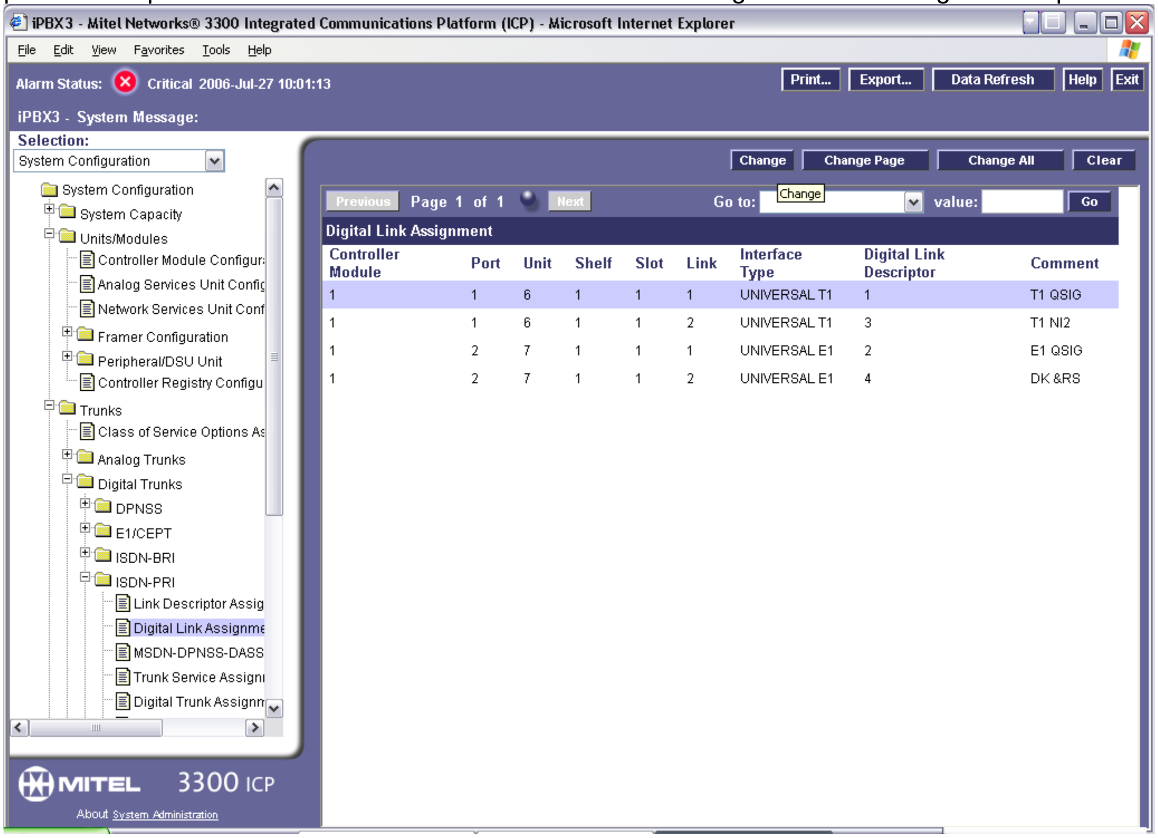


See the screen below for the selected options.

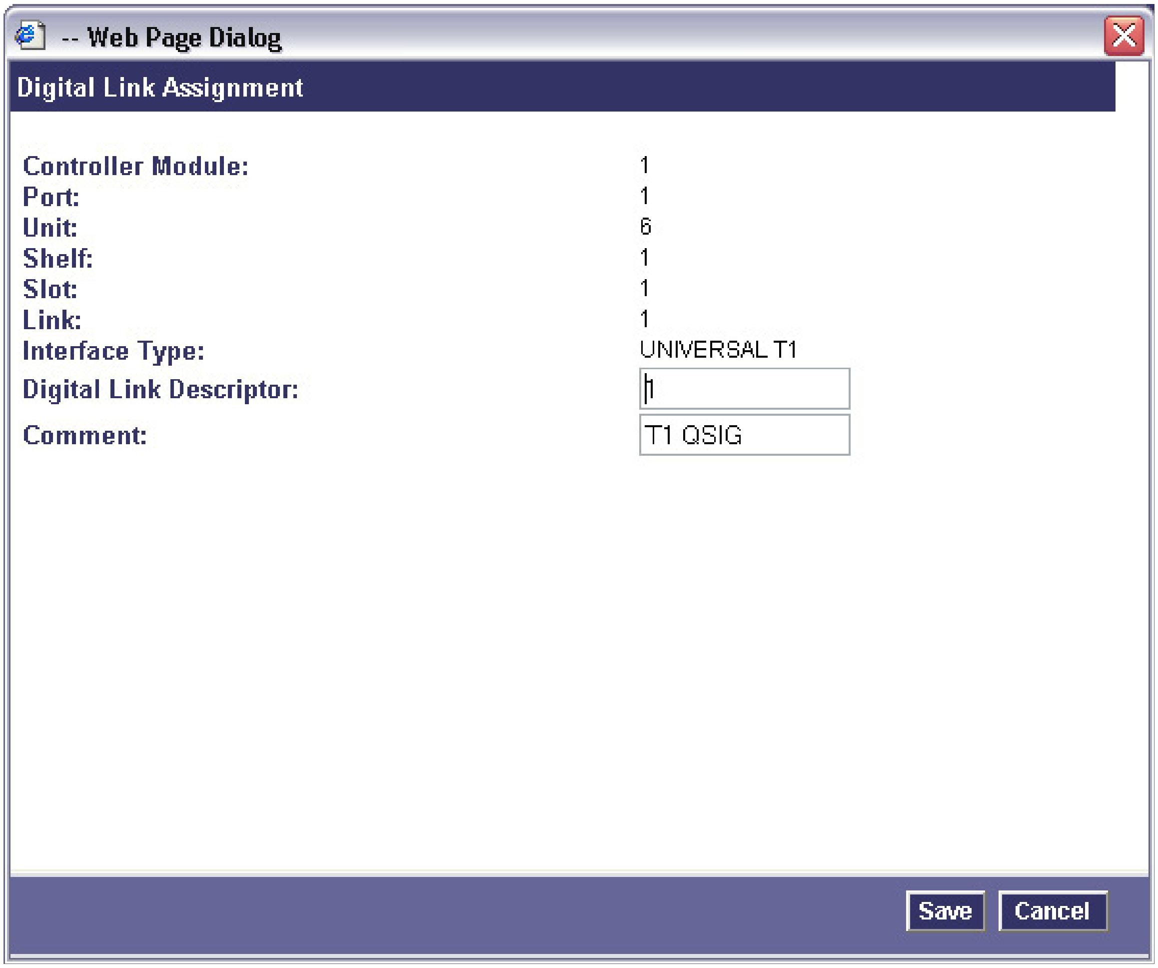


Before moving on to the next task be sure to click the Save button.

Use the *ISDN-PRI Digital Link Assignment* menu selection to connect a physical location (port, unit, shelf, slot and link) within the PBX with the NSU and Link Descriptor configured in the previous steps. Select a *Controller Module* and click the Change button to configure the options.

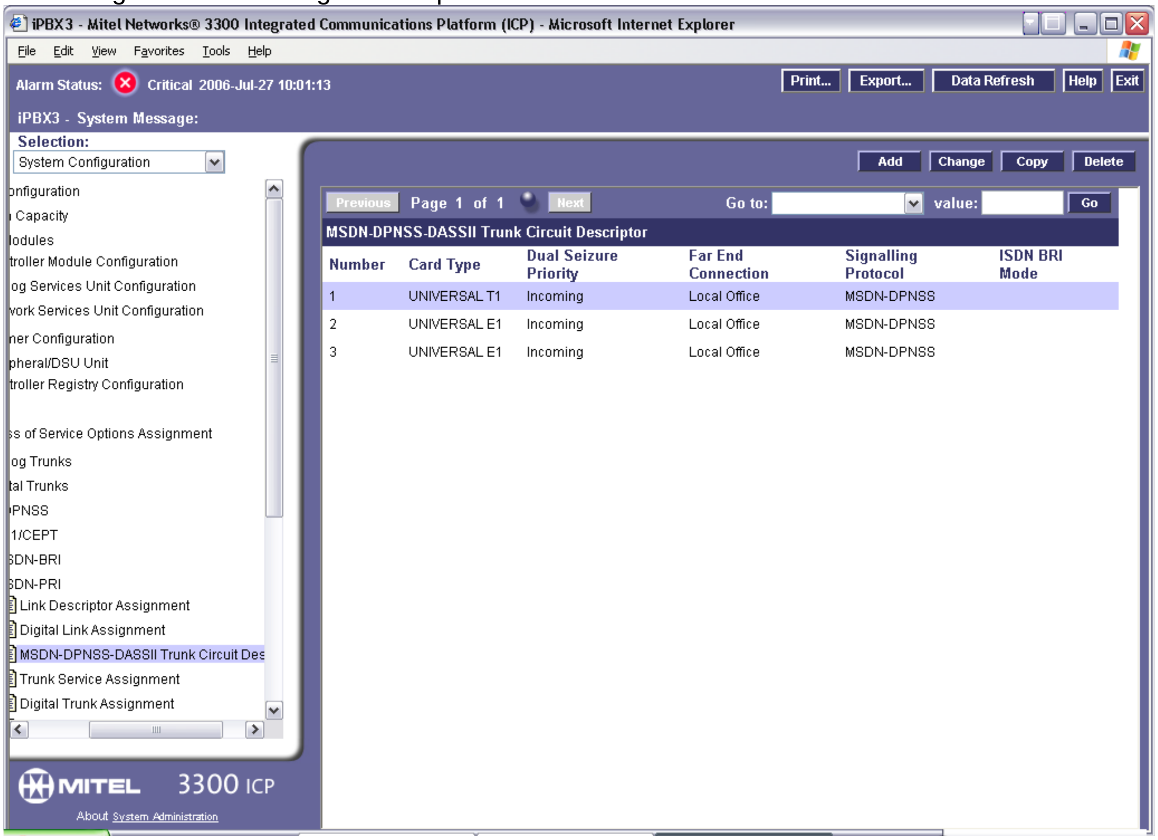


See the screen below for the selected options.

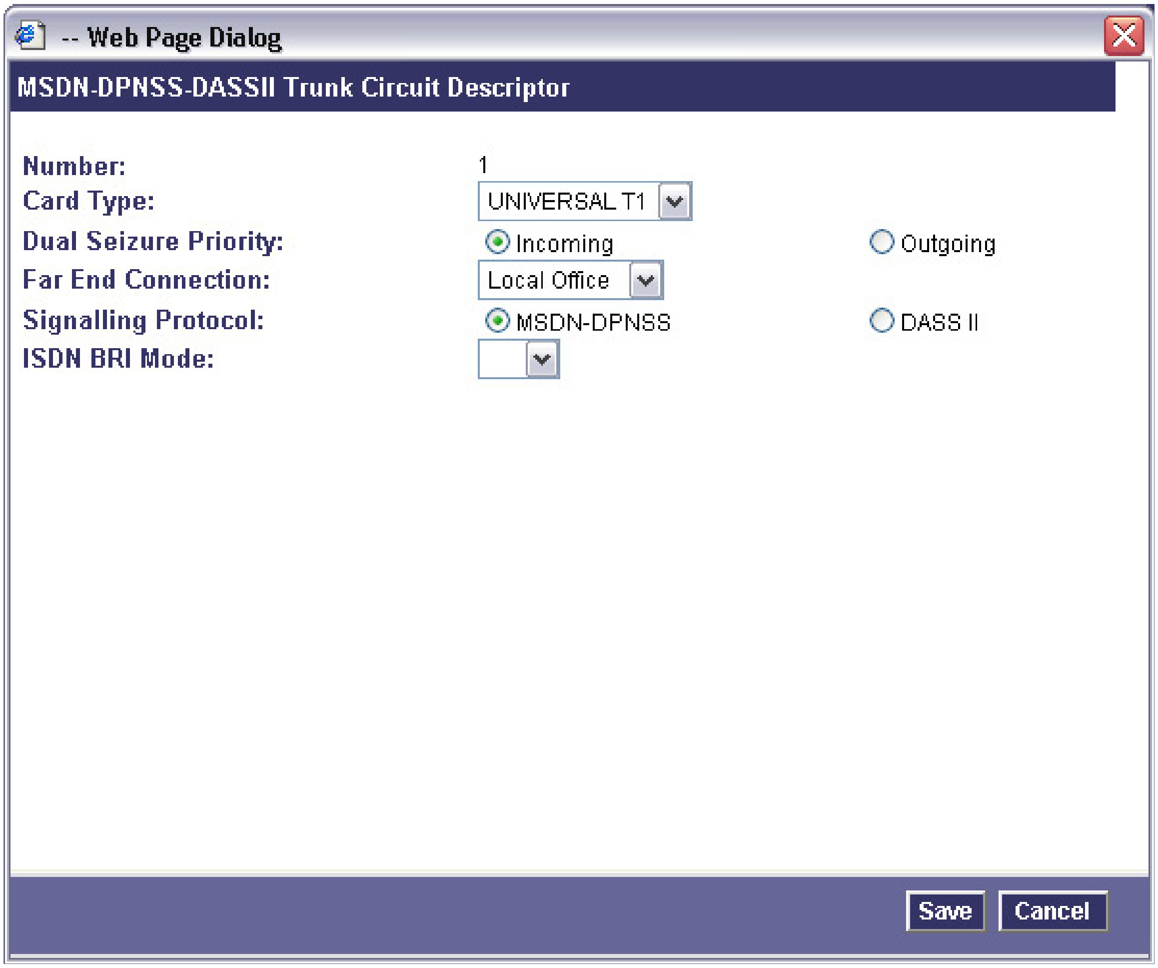


Before moving on to the next task be sure to click the Save button.

Use the *ISDN-PRI MSDN-DPNSS-DASSII Trunk Circuit Descriptor* menu selection to assign direction and protocols to the individual trunk cards in the PBX. Select a trunk number and click the Change button to configure the options.

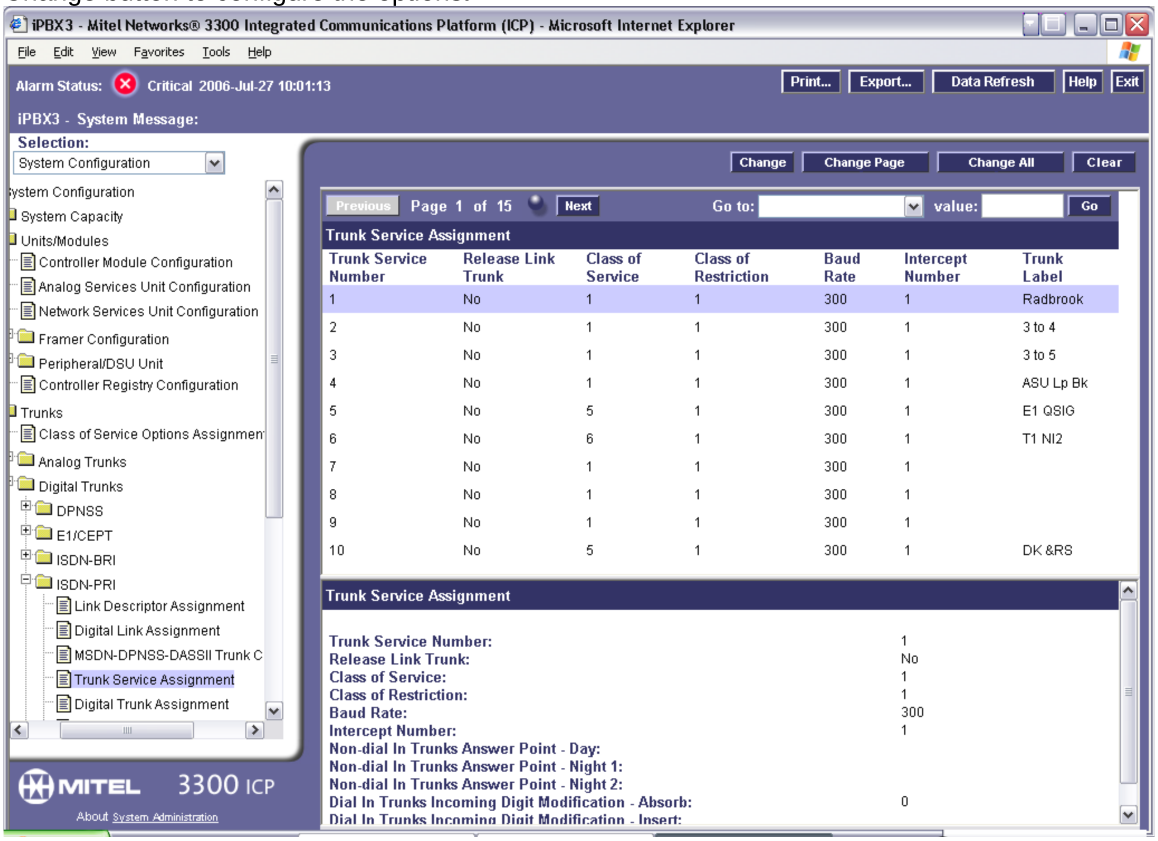


See the screen below for the selected options.

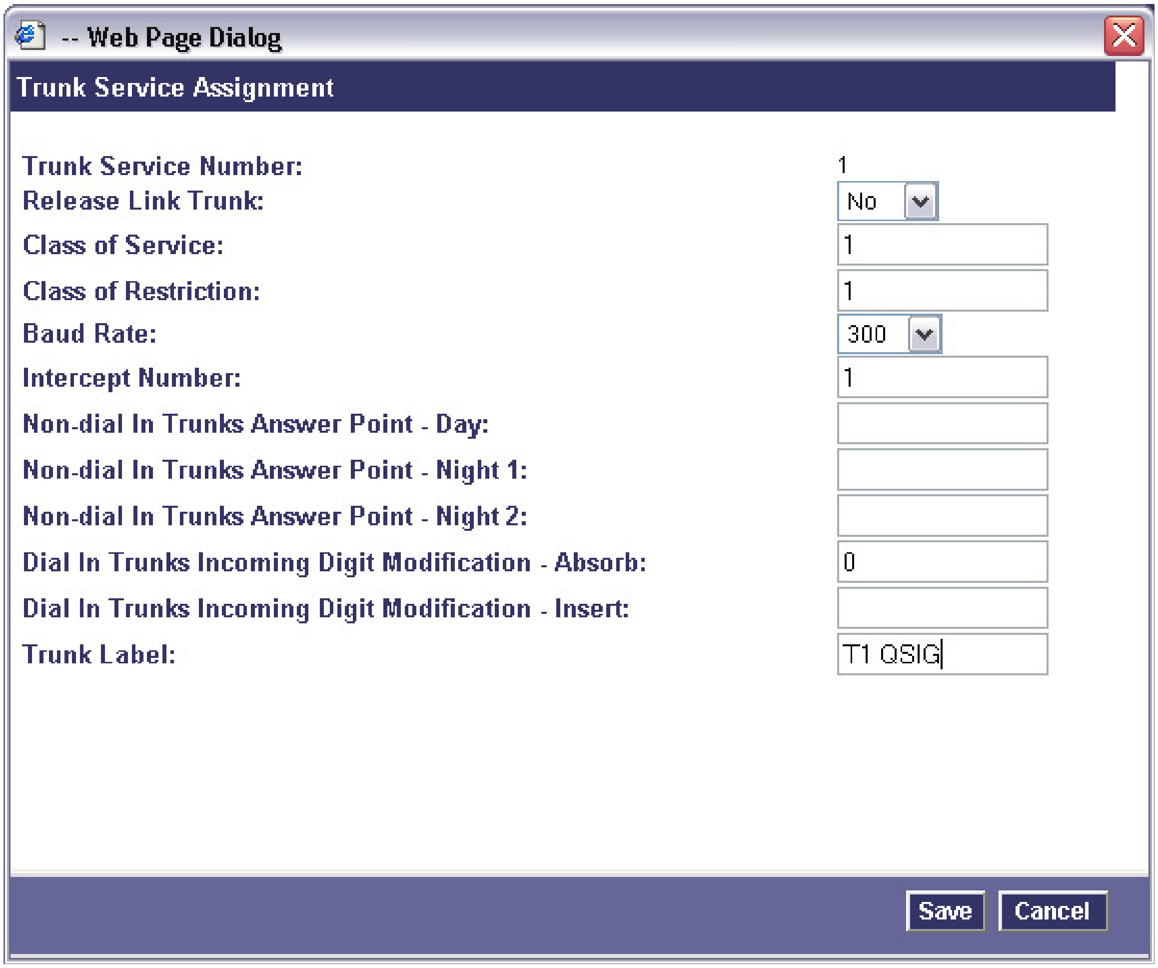


Before moving on to the next task be sure to click the Save button.

Use the *ISDN-PRI Trunk Service Assignment* menu selection to set up a template that contains various service levels, for example the trunks class of service that was previously defined, into a template that will latter on be assigned to a trunk. Select the trunk service number and click the Change button to configure the options.

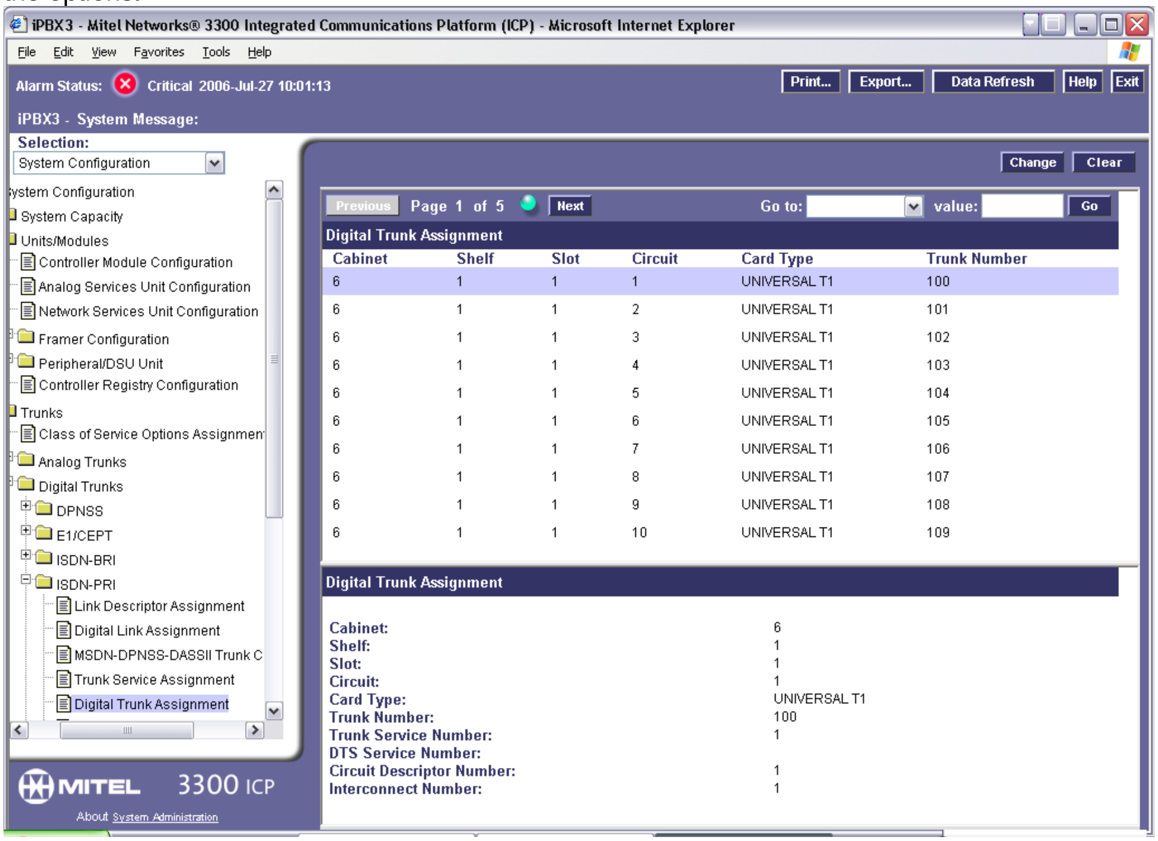


See the screen below for the selected options.

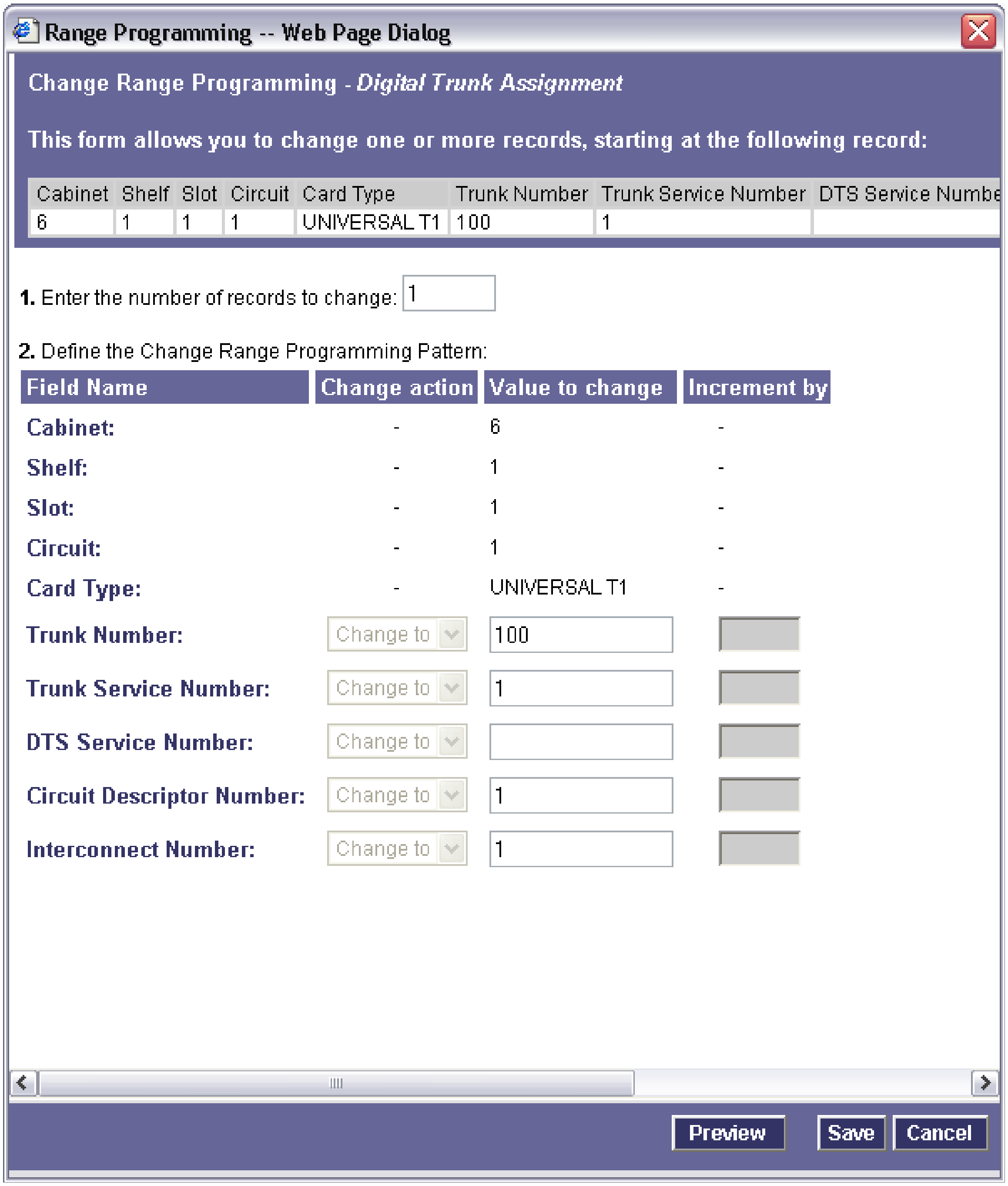


Before moving on to the next task be sure to click the Save button.

Use the *ISDN-PRI Digital Trunk Assignment* menu selection to configure the individual trunk members and assign them the defined *Trunk Service Assignment* template and *Trunk Circuit Descriptor* template that were configured in the previous steps. This gets done for each member of a trunk interface. Select each individual trunk member and click the Change button to configure the options.



See the screen below for the selected options.



Range Programming -- Web Page Dialog

Change Range Programming - *Digital Trunk Assignment*

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 Numbe
6	1	1	1	UNIVERSAL T1	100	1	

1. Enter the number of records to change: 1

2. Define the Change Range Programming Pattern:

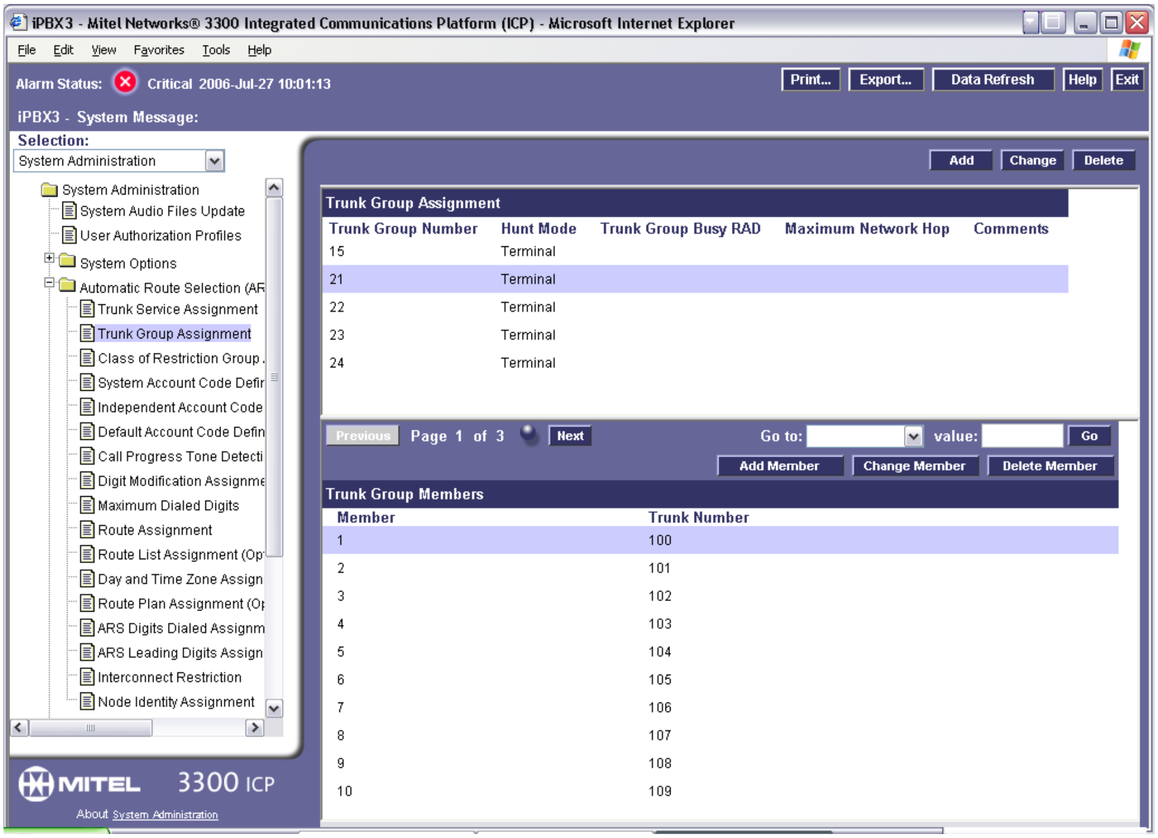
Field Name	Change action	Value to change	Increment by
Cabinet:	-	6	-
Shelf:	-	1	-
Slot:	-	1	-
Circuit:	-	1	-
Card Type:	-	UNIVERSAL T1	-
Trunk Number:	Change to	100	
Trunk Service Number:	Change to	1	
DTS Service Number:	Change to		
Circuit Descriptor Number:	Change to	1	
Interconnect Number:	Change to	1	

Preview Save Cancel

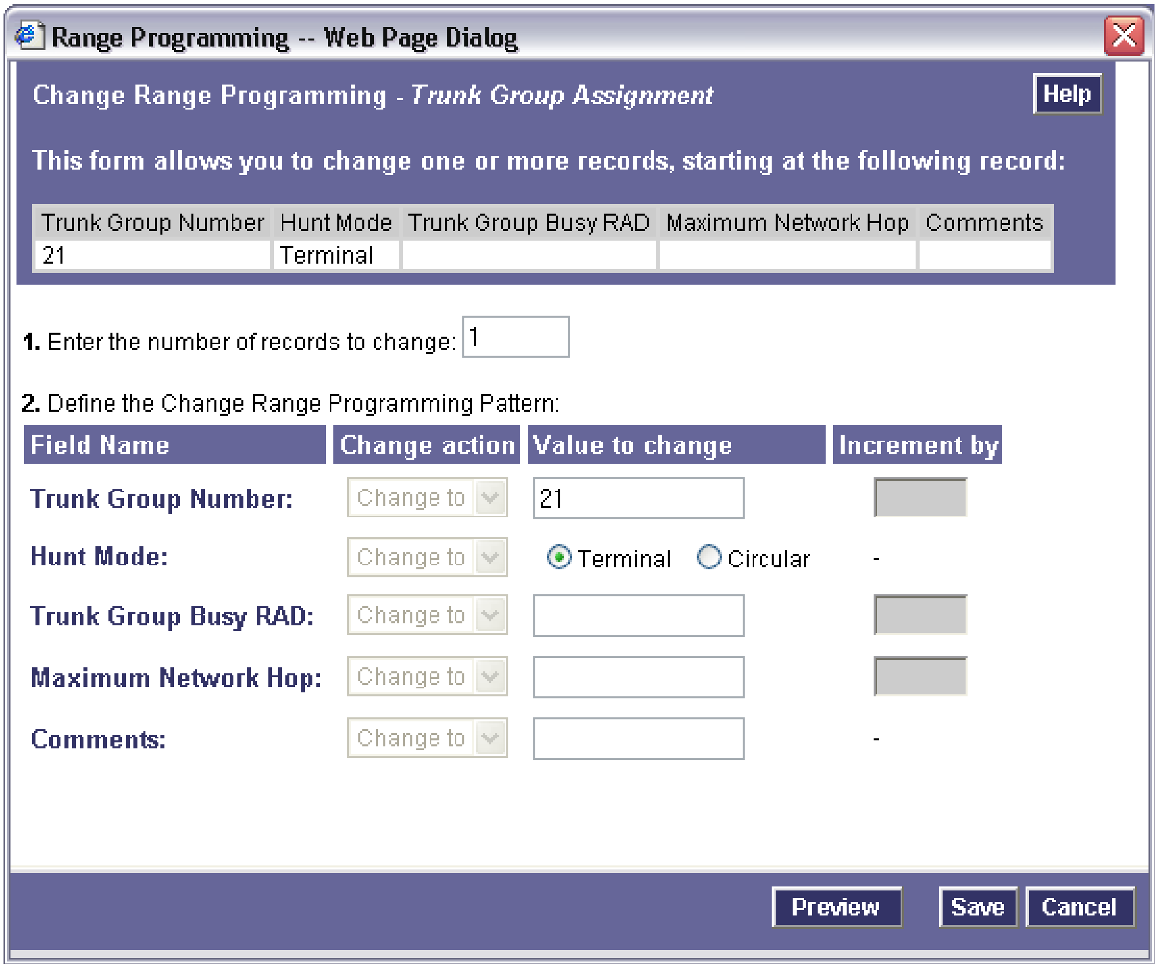
Before moving on to the next task be sure to click the Save button.

6.3 Configuring ARS Options

Use the *ARS Trunk Group Assignment* menu selection to edit your trunk group configuration. Use the Add or Change buttons to either build a new trunk group or modify an existing group.

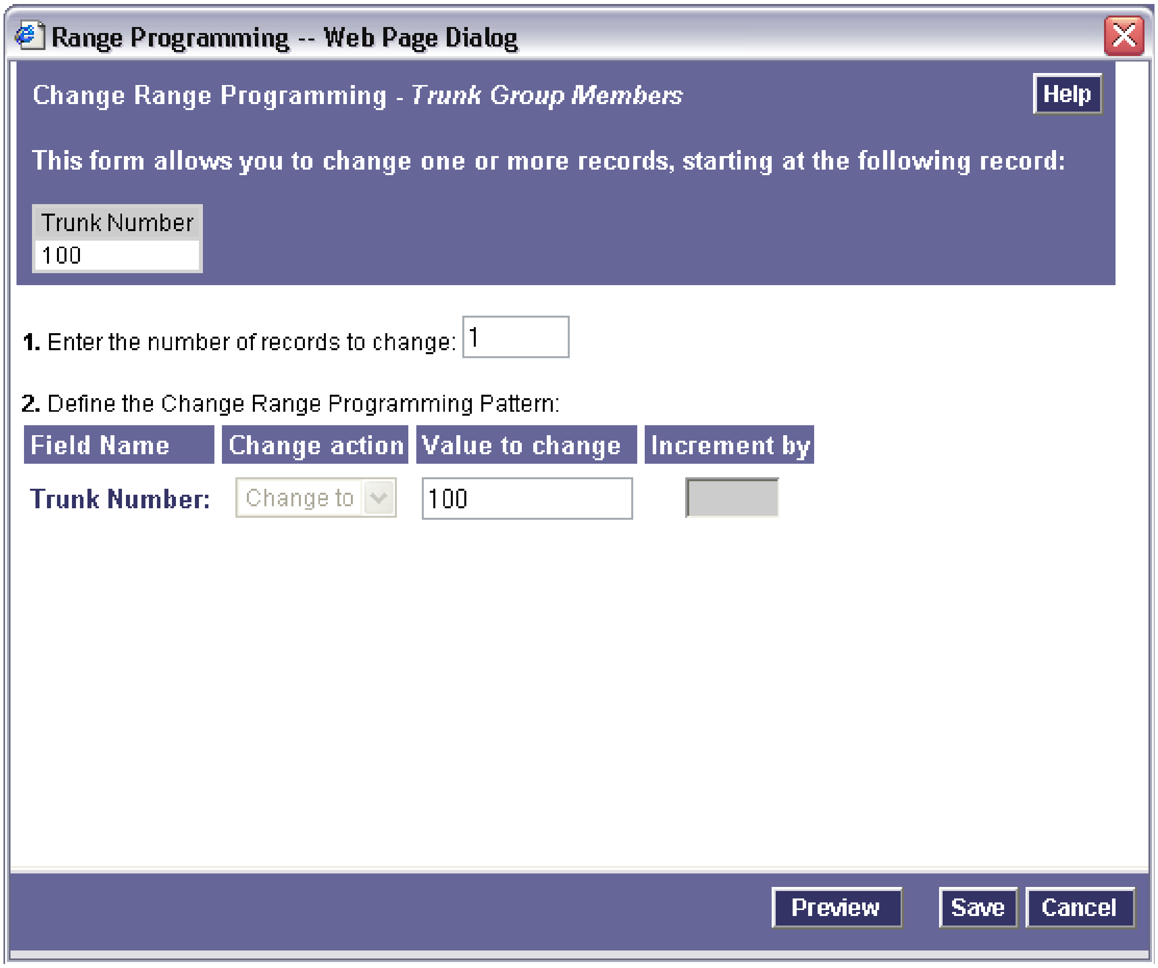


See the screen below for the selected options.



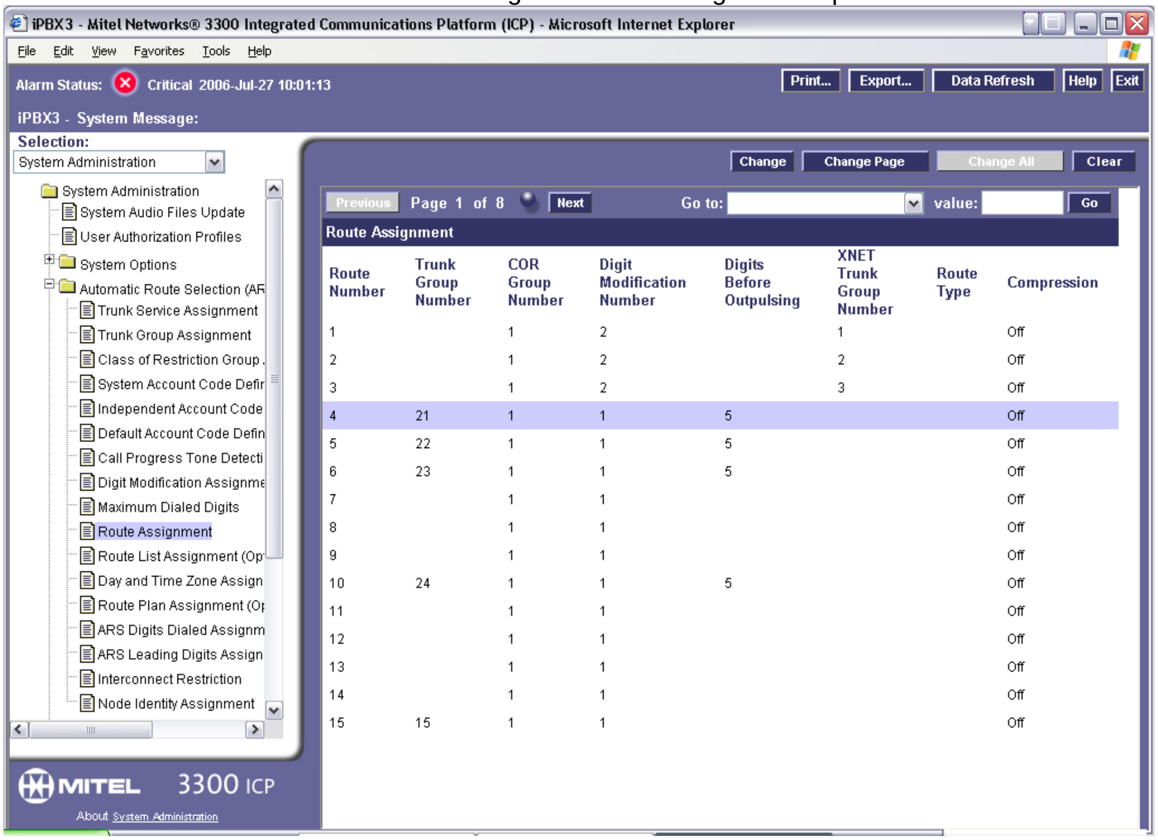
Click the Save button when you are finished editing the options and wish to save your configuration.

When you have configured a trunk group you use the Add Member button to add individual trunk members to the group.

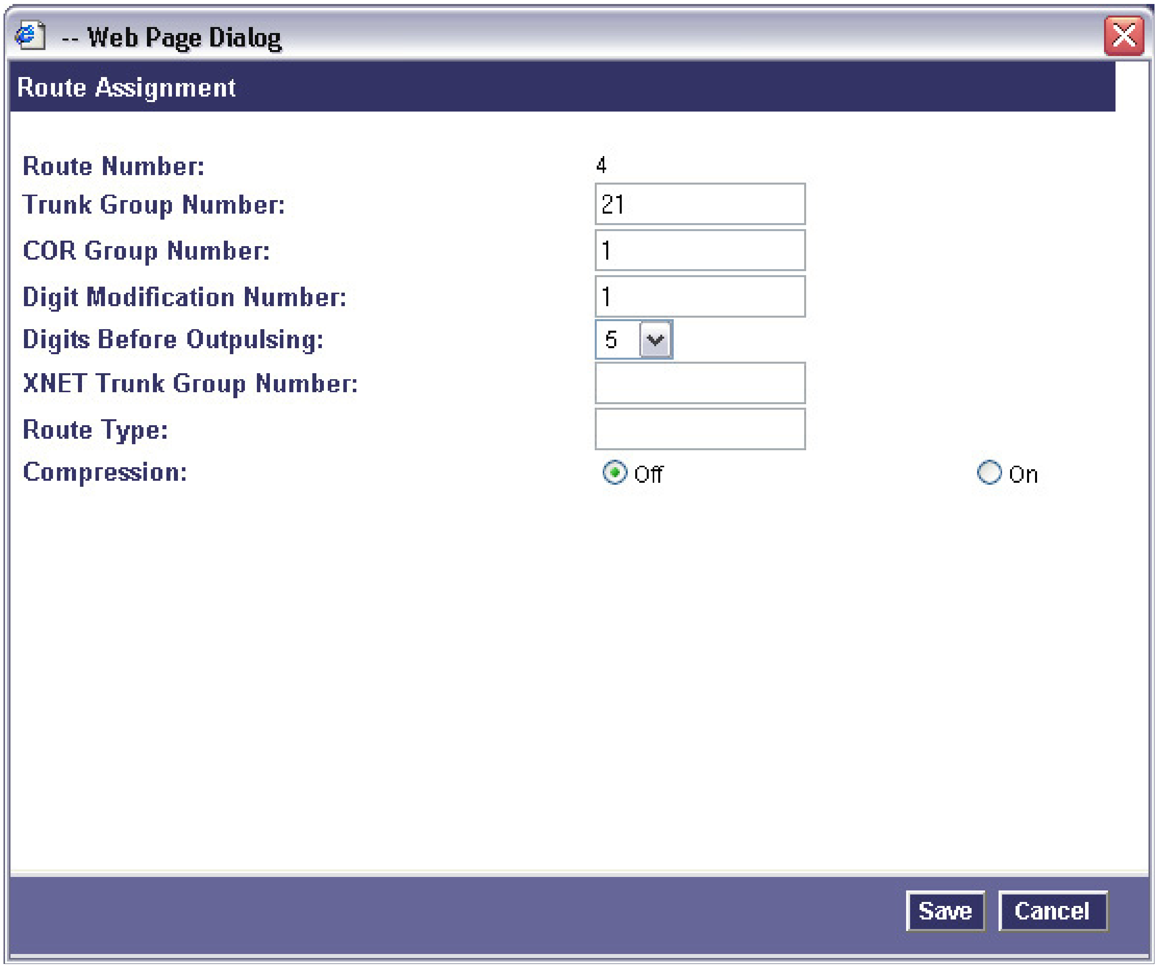


Before moving on to the next task be sure to click the Save button.

Use the *ARS Route Assignment* menu selection to define a route to direct calls to a specific trunk. Select a Route Number and click the Change button to configure the options.

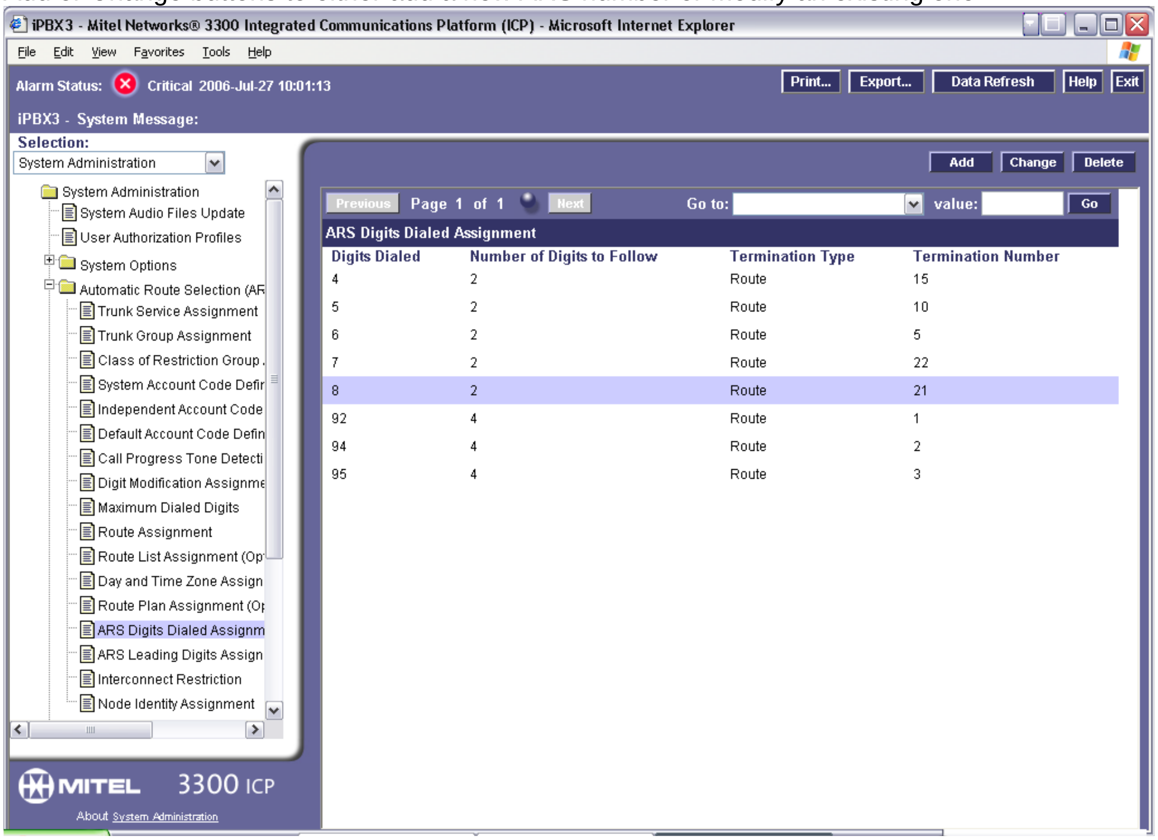


See the screen below for the selected options.



Before moving on to the next task be sure to click the Save button.

Use the *ARS Digits Dialed Assignment* menu selection to configure an ARS number to use to place and forward calls to a specific trunk group. The ARS number is used as the forwarding target for subscriber station sets and the inbound entry point for direct calls to the server. Use the Add or Change buttons to either add a new ARS number or modify an existing one.

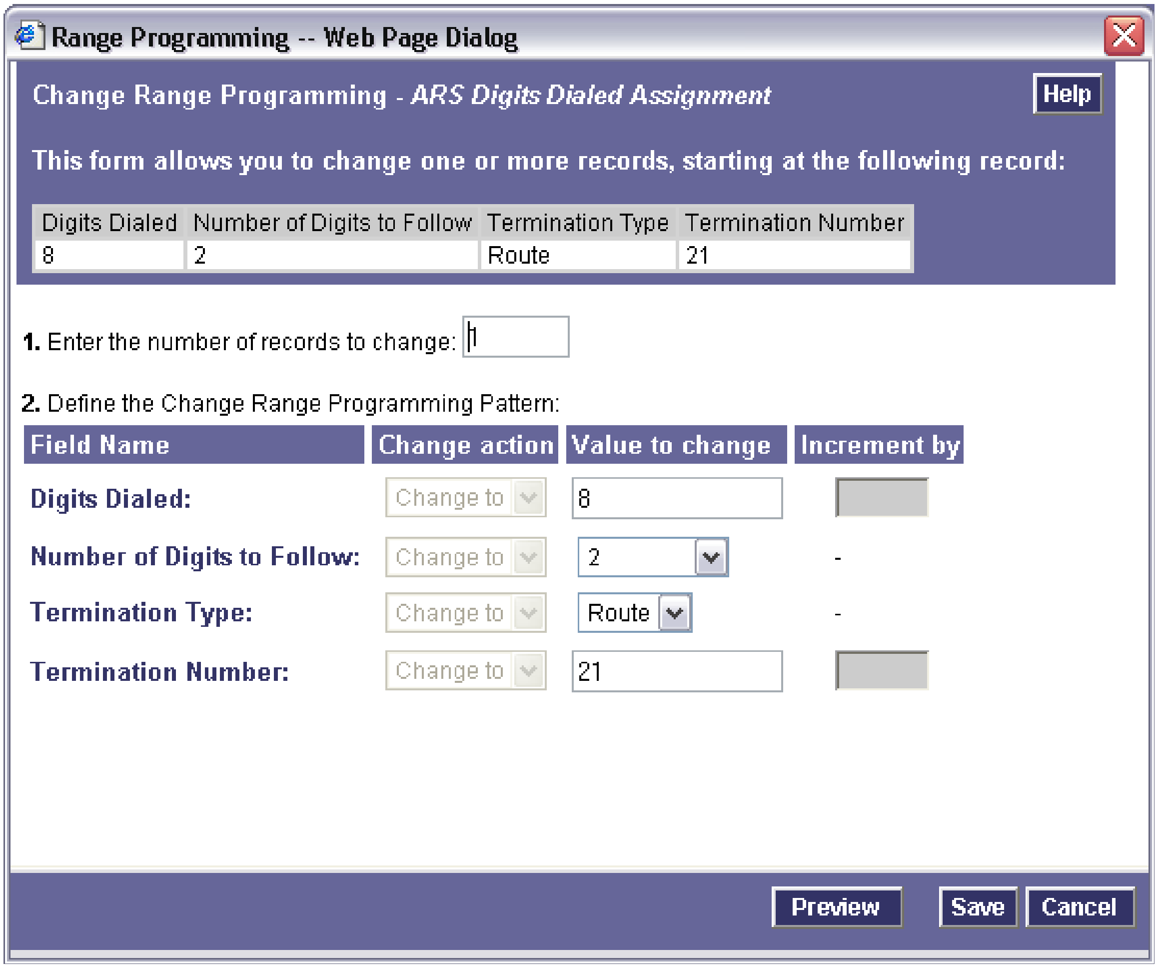


This example shows setting up ARS to except any 3 digit number that starts with an 8 as a dial-able number. The ASR table will then take the call and route it to trunk group 21. Useable numbers in this example would be any number between 800 and 899 all inclusive.

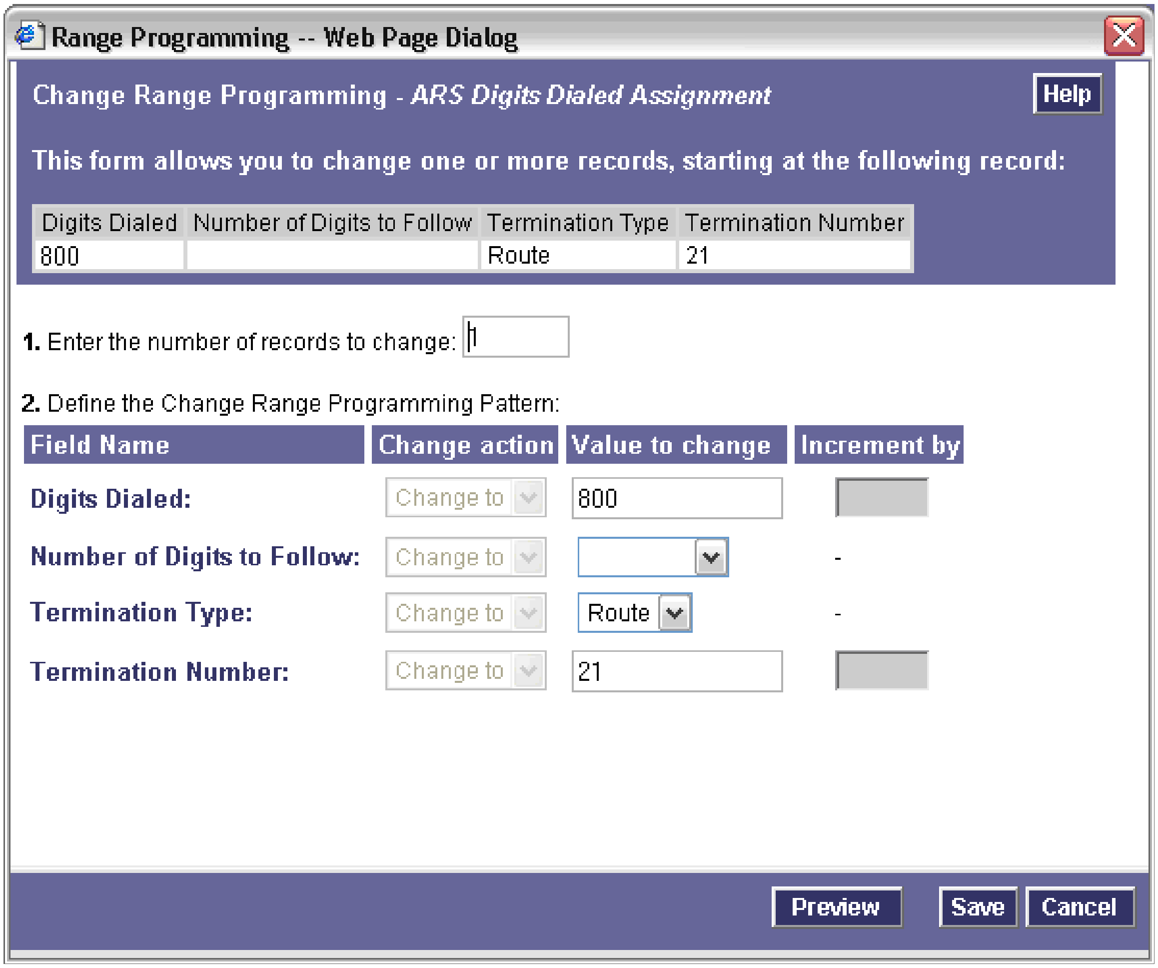
An alternate method of configuration would be to define a very specific number, for example 800, not an entire range, and not define any following digits.

The method you choose is up to what your sites configuration will support.

See the screen below for the selected options.



Below is an example of the configuration using the described alternative method.



Before moving on to the next task be sure to click the Save button.

6.4 Setting Up Subscriber Station Sets

There is no PBX-side programming for setting up the subscriber station sets. All the forwarding of the subscriber station sets is defined directly on subscriber station set using the phone's soft menu keys. The subscriber should be directed to set their internal and external ring no answer and busy forwarding conditions to the Pilot Number setting defined in the hunt group configuration.

6.5 Additional Comments

Ensure that the Node Identity assignment has not been entered. If this is entered, it will append extra digits onto extensions as they pass across the trunk to the gateway.

7. Microsoft OCS setup

7.1 Steps for configuring OCS

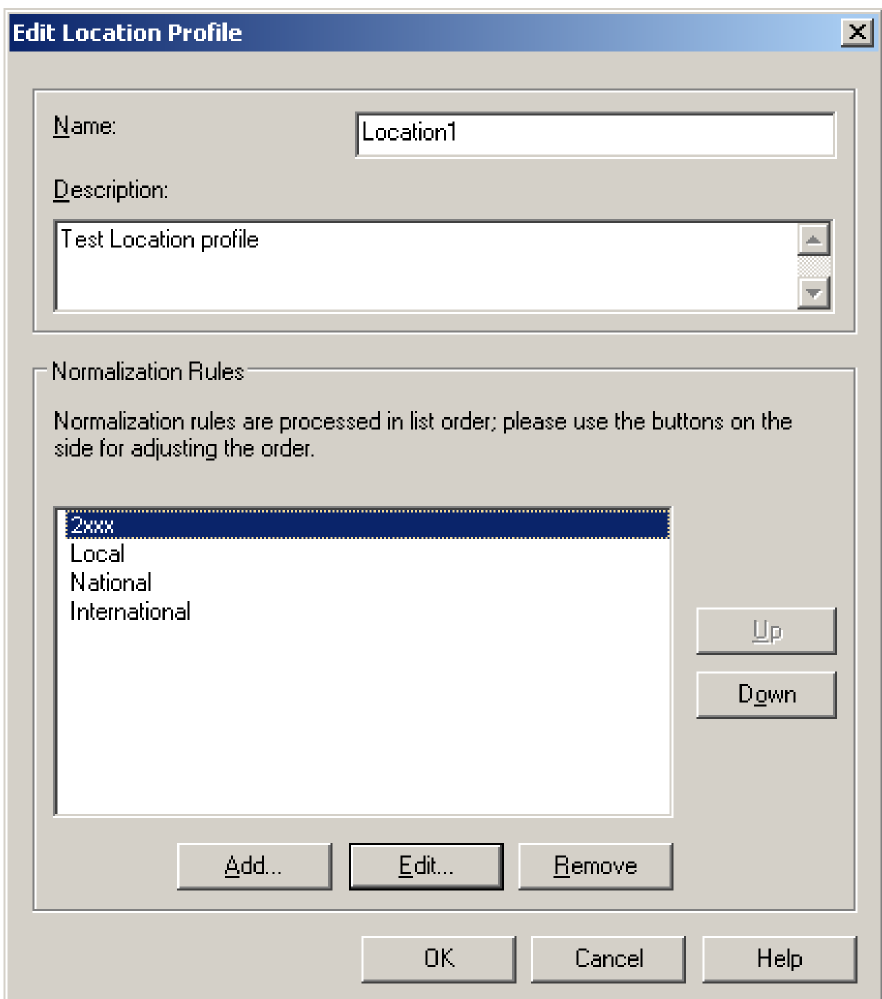
Normalization rules are used to convert all possible dial numbers into full E.164 formatted numbers. Microsoft OCS uses the standard E.164 format to search for all users listed in Active Directory (AD).

When an OCS user dials an internal extension number (normally 3-5 digits), the normalization rules convert it into full E.164 format. These normalization rules should cover dialed digits that are for internal extensions, local numbers, long distance numbers, and international numbers.

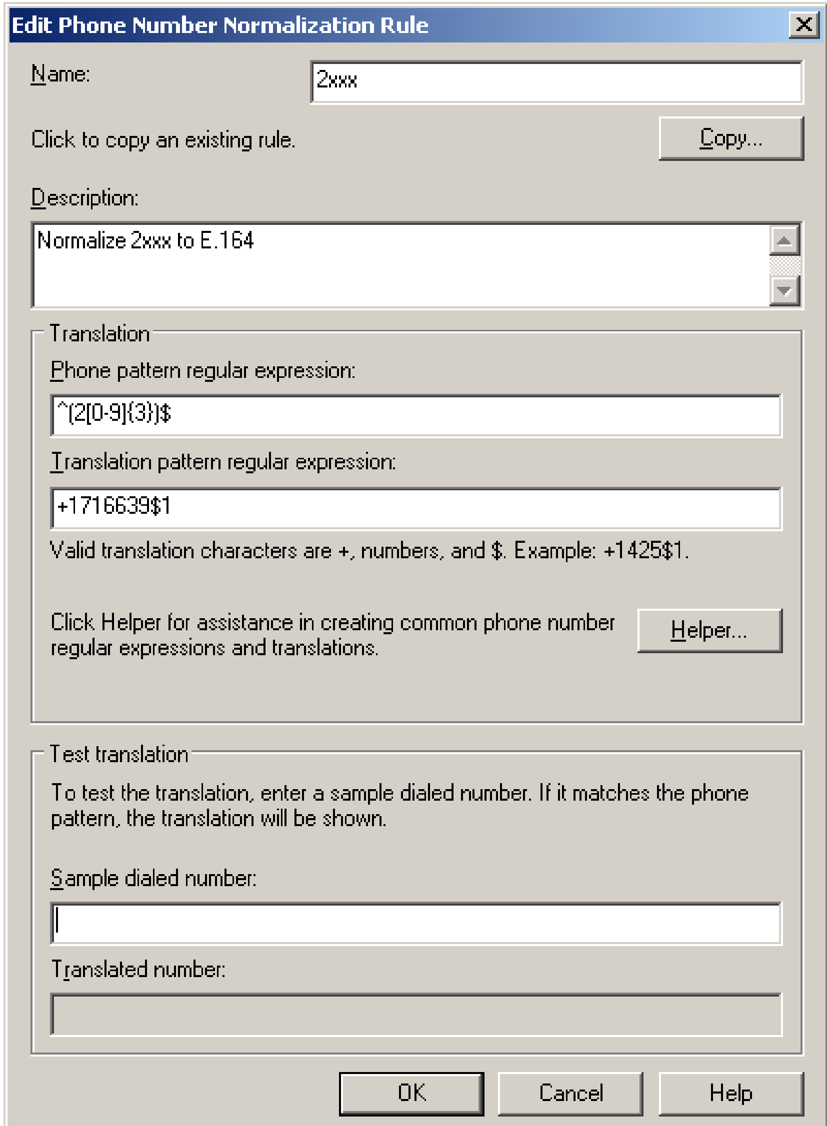
From the Start menu select the following to configure the OCS server:

- Programs
- Administrative Tools
- OCS 2007

On the tree presented in the configuration window right click on Forest then select `Properties` and then `Voice Properties` form the menu provided. Edit a location profile as shown in the example below.



Click `Add` or `Edit` to create or change a particular rule.



In this example, when a user dials any 4-digit number starting with 2, it will be converted to its E.164 equivalent of +1716639xxxx and then that number will be searched for in AD.

More examples are shown in the following table:

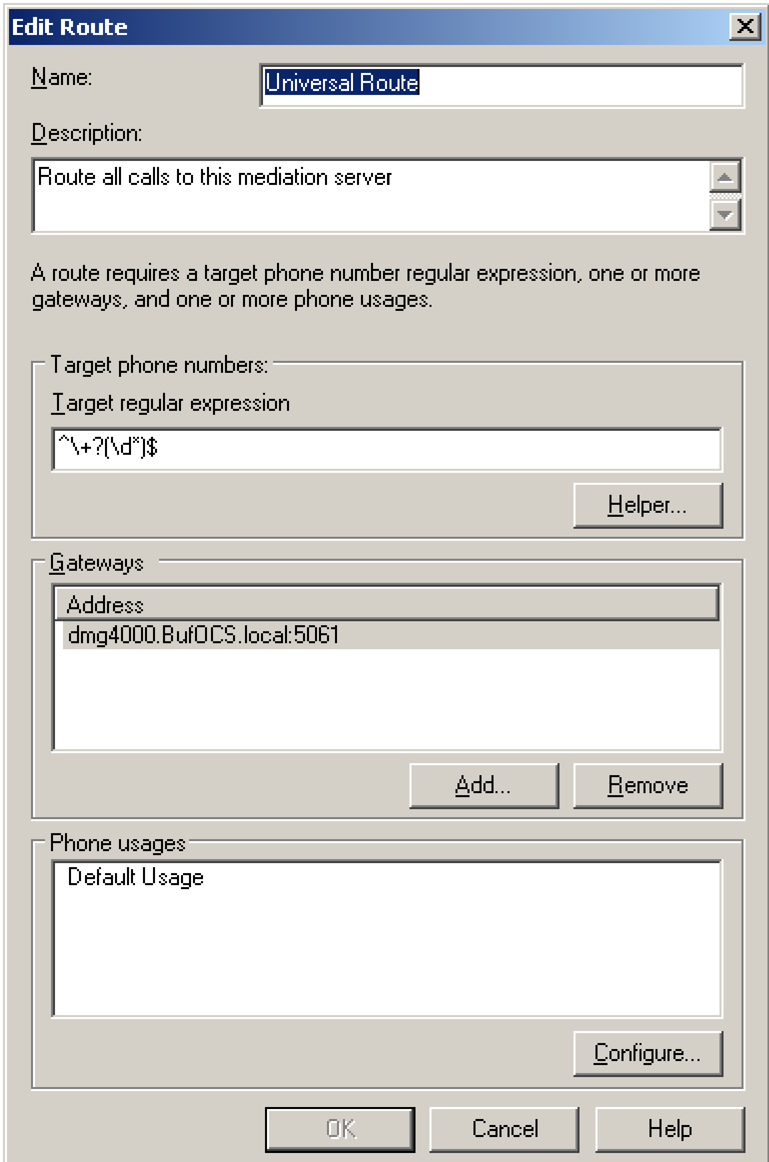
Name	Phone Pattern	Translation Pattern	Descriptions
2xxx	^(2[0-9]{3})$	+1716639$1	Normalize 2xxx to E.164
Local	^(\d{7})$	+1716$1	Local number
National	^1(\d*)$	+1$1	Long distance number
International	^011(\d*)	+011$1	International number

A default route is used to route all calls to the Mediation server. If you need to route some calls to a different Mediation server, configure the Target phone numbers field accordingly.

From the Start menu select the following to configure the OCS server:

- Programs
- Administrative Tools
- OCS 2007

On the tree presented in the configuration window right click on Forest then select `Properties` and then `Voice Properties` form the menu provided. Edit a route as shown in the example below.



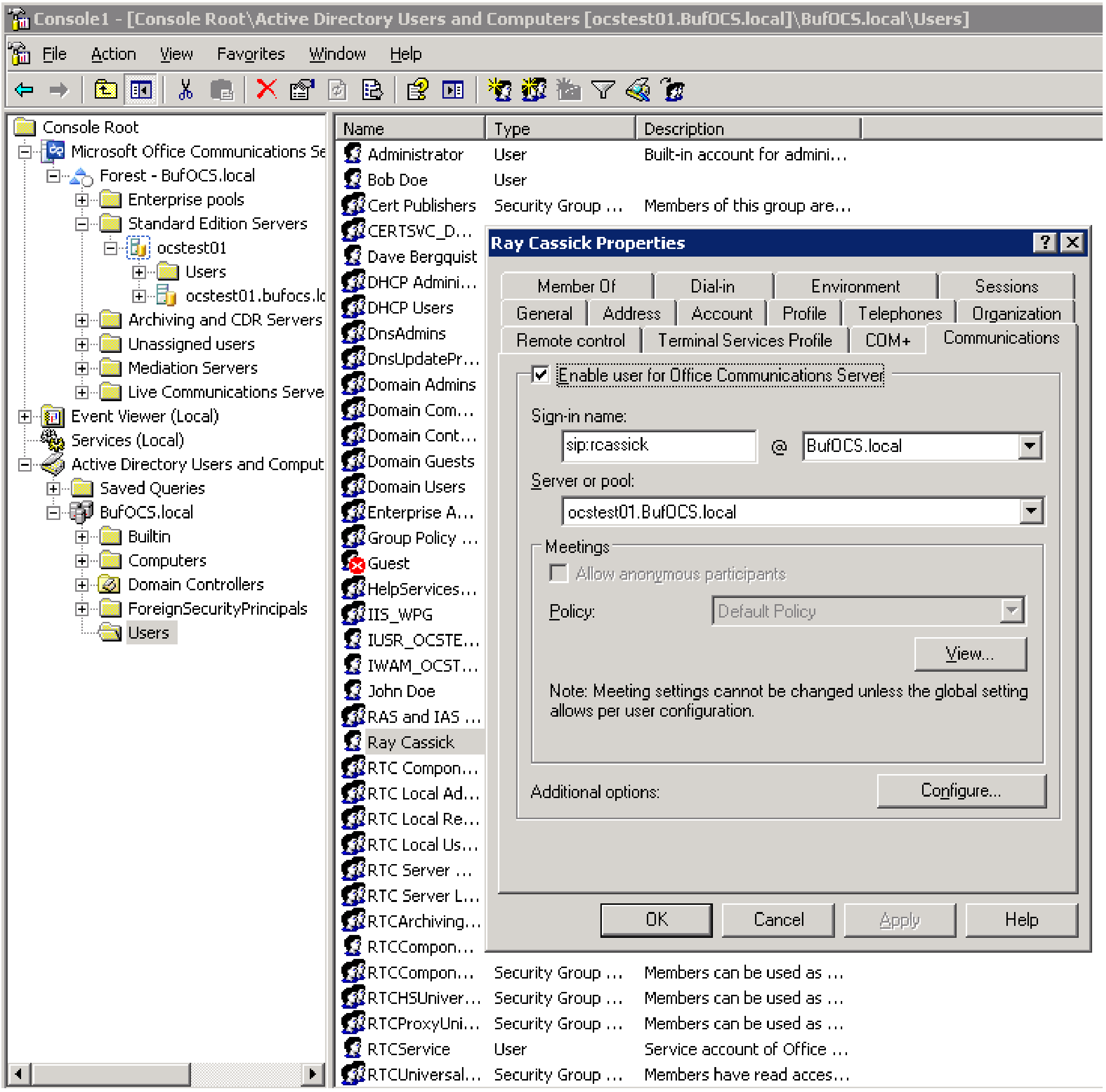
This entry routes any number with or without '+' prefix followed by any digits to Mediation server dmg4000.bufocs.local

Restart the Front End Services for the above changes to take effect, including all Normalization rules. This can be done from Window Services.

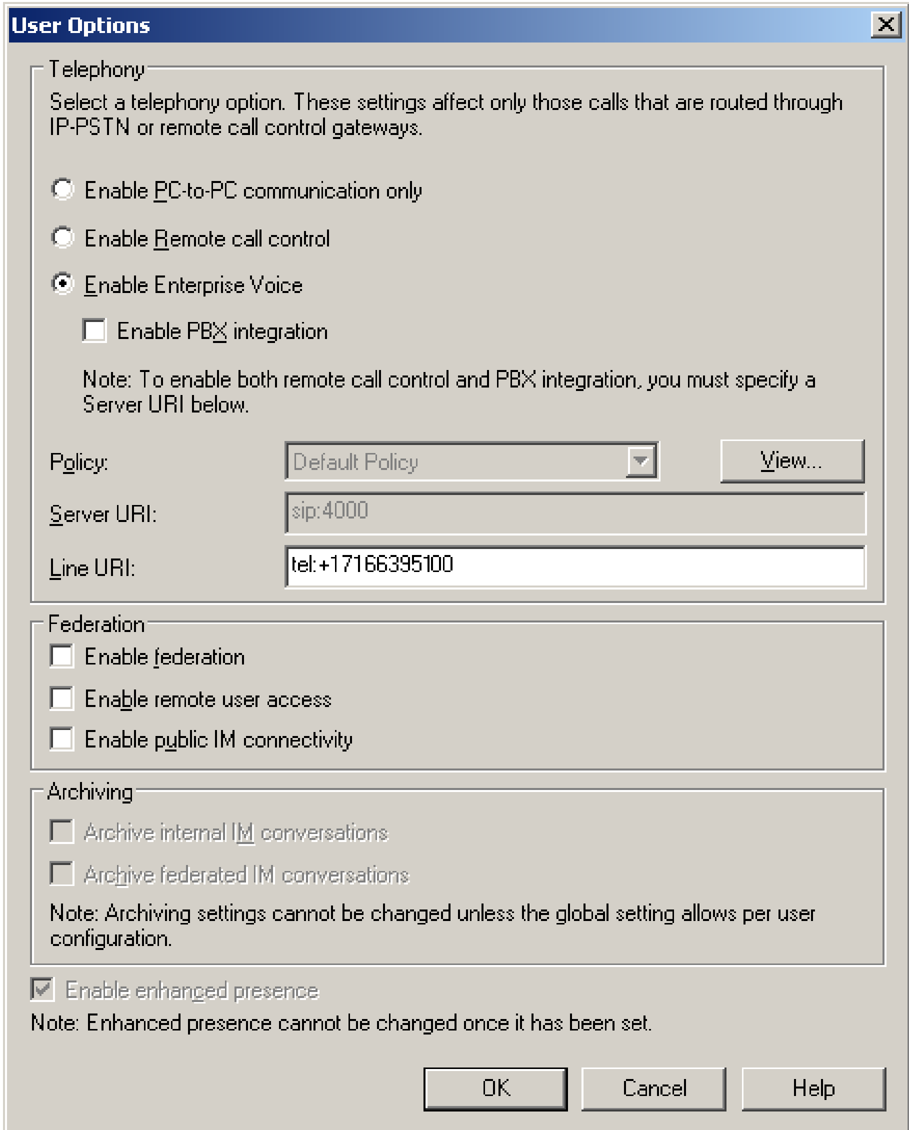
Note: Unless the dialed number from OCS client (such as Office Communicator) is in E.164 format, OCS must find a normalization rule to convert the dialed number to E.164.

7.2 Steps for configuring OCS clients

The domain users need to be enabled for making calls through OCS server.



Under `Communications` tab, check the Enable user for `Office Communications Server` option and then click the `Configure` button.



In the above configuration for user Ray Cassick, when an inbound PSTN call for 5100, it will be converted by the gateway CPID manipulation and routing rules into +17166395100. OCS will match that number provided by the gateway to the `Line URI` parameter for this user and ring Ray Cassick if he is logged on to OCS from Office Communicator or any OCS supported device.

8. Testing Validation Matrix

The table below shows various test scenarios that are run as typical validation scenarios when the gateway is used in a voice messaging situation. The notes column specifies any notable parts of the test.

The test scenarios below assume that all gateway configuration parameters are at their default values. For a complete sample showing call flows and states please consult the Gateway SIP Compatibility Guide.

Test Number	Call Scenario Description	Notes
Inbound call scenarios		
1	Direct call from TDM station set to OCS client.	
2	Direct call from OCS client to TDM station set.	

9. Troubleshooting

9.1 Important Debugging Tools

- Ethereal/Wireshark – Used to view and analyze the network captures provided by the Dialogic gateway diagnostic firmware.
- Adobe Audition -- Used to review and analyze the audio extracted from the network captures to troubleshoot any audio related issues.

9.2 Important Gateway Trace Masks

These keys are helpful during all troubleshooting scenarios and should be considered keys to activate by default fro all troubleshooting cases.

- `voip prot` and `voip code` – this allows the collection of all SIP related messages as they are sent from and received by the gateway. This data is important in cases where you feel that the gateway is not able to communicate properly with the messaging server.
- `tel event` and `tel code` – This allows the collection of all circuit side activity of the emulated station set such as display updates, key presses, light transitions and hook state changes. This data is very important in the following scenarios:
  - Call control problems (dropped calls, failing transfers, etc…)
  - Integration problems (incorrect mailbox placement, missed auto-attendant greetings etc…)
- `teldrv prot` – This allows the collection of all ISDN messages both transmitted and received on the gateways front end interface. This data is very important in the following scenarios:
  - Call control problems (dropped calls, failing transfers, etc…)
  - Integration problems (incorrect mailbox placement, missed auto-attendant greetings etc…)
- `Routingtable (all keys)` – This allows you to look inside the routing table engine and see how matching rules and CPID manipulation rules work with respect to your call. This data is very important in the following scenarios:

- Call routing problem (reaching the incorrect OCS client or no client at all, etc…)

NOTE: Turning on all traces is not recommended. Doing this floods the debug stream with significant amounts of information that can cause delays in determining the root cause of a problem.

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05-2655-001
September 2008