



NET Unified Communications Configuration Guide

UX2000 SBC, XO Communication SIP Trunk Lync Server 2010

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Contents

1	REFERENCES AND VERSIONS	5
2	PURPOSE	6
3	COMPLETING SIP TRUNK TASK ON UX2000	7
3.1	Prerequisites	7
3.2	Initial Task configuration	7
3.3	Root Certificate	8
3.4	UX Certificate	9
3.5	SIP Signaling Endpoints	10
4	CONFIGURING PUBLIC ETHERNET PORT AND IP ADDRESSING	12
4.1	Configuring the Public Ethernet Port	12
4.2	Selecting Unused Ethernet Port	12
4.3	Configuring the IP Default Route	14
4.4	Configure Internal Private IP Static Route	16
5	LYNC SERVER 2010 CONFIGURATION	18
5.1	Adding UX2000 SBC to Lync Server Topology	18
5.2	Adding the UX2000 into Lync Server Routing	20
5.3	Configuring Assigned DIDs to Lync Users	22
5.4	Creating Analog Contacts with XO SIP Trunk DIDs	23
6	UX2000 TELEPHONY ROUTING PROCESS	25
6.1	Signaling Groups	25
6.2	Call Routing Table	26
6.3	Translation Tables	26
7	MEDIA CONFIGURATION	30

8	ENABLING MEDIA TRANSCODING	33	
9	ANALOG SUPPORT	34	
9.1	Differences Between PSTN and Analog Call Routing	34	
9.2	Analog Configuration	39	
9.2.1	SIP Server Table		39
9.2.2	Analog Translation Table		41
9.2.3	Signaling Group		44
9.2.4	Modify Lync Server 2010 Routing Table to Add ATA Route		45
9.2.5	Add Translation Table for ATA		47
9.2.6	Add Route Table for ATA		48
10	TROUBLESHOOTING	50	
10.1	Visual Analysis	50	
10.1.1	Current SG Status and Call Activity		50
10.1.2	Current Alarms		51
10.2	Viewing Debug Logs	51	
10.3	Sending Debug Logs to a Syslog Collector	51	
	Figure 1 Typical Lync Server 2010 to XO SIP Trunk Deployment		6
	Figure 2 Initial Task Configuration.....		7
	Figure 3 Importing Root Certificate PEM Format.....		8
	Figure 4 Generating UX Certificate Request		9
	Figure 5 Importing UX Certificate PEM Format.....		10
	Figure 6 Setting Up SIP Signalling Endpoints		11
	Figure 7 Node Interfaces		13
	Figure 8 Ethernet Port Configuration		14
	Figure 9 IP Default Route.....		15
	Figure 10 Public IP Default Route.....		15
	Figure 11 Internal Static IP Route 1.....		16
	Figure 12 Internal Static Route 2.....		17
	Figure 13 Adding New IP/PSTN Gateway in Lync Server 2010 Topology		18
	Figure 14 Edit Mediation Server Dialog.....		19
	Figure 15 Lync Server Control Panel Route Selection		20
	Figure 16 Available PSTN Gateways		21
	Figure 17 PSTN Gateway and PSTN Usages		22
	Figure 18 Lync User DID Setting		23
	Figure 19 Adding Analog Contact.....		24
	Figure 20 UX2000 Routing Process.....		25
	Figure 21 Signaling Group.....		25

Figure 22 Call Routing Table	26
Figure 23 Translation Tables	27
Figure 24 Translation Table that Removes + and 1	28
Figure 25 Translation Table that Adds + and 1	29
Figure 26 Media Profiles Showing Enabled Codecs.....	30
Figure 27 Typical G711Alaw Codec Settings	31
Figure 28 Media List Containing G711Alaw and G711Ulaw	31
Figure 29 Media List (cont) Showing Digit Relay Settings.....	32
Figure 30 Signaling Group Showing Media List Selection	32
Figure 31 Media Transcoding Configuration	33
Figure 32 PSTN to Lync Client Call Flow.....	34
Figure 33 PSTN to Analog Call Flow.....	35
Figure 34 Lync Client to PSTN Call Flow.....	36
Figure 35 Lync Client to Analog Call Flow	36
Figure 36 Analog to Analog Call Flow	37
Figure 37 Analog to PSTN Call Flow.....	38
Figure 38 Analog to Lync Client Call Flow	39
Figure 39 ATA SIP Server Table Name	40
Figure 40 SIP Server Configuration Dialog	41
Figure 41 Translation Table Name for ATA Routing	42
Figure 42 Completed Translation Table Dialog for ATA Routing.....	43
Figure 43 SIP SG Configuration Dialog.....	44
Figure 44 Adding ATA Route	46
Figure 45 Route Table Ordering for calls to ATA.....	47
Figure 46 Translation Table to Add + to Called Number	47
Figure 47 Creating a Route Table for the ATA	48
Figure 48 Routing Entry for Calls From ATA to Lync Server 2010	49
Figure 49 XO SIP Trunk Down	50
Figure 50 Alarm Status	51
Figure 51 Debug Logs	51
Figure 52 Syslog Collector Configuration.....	52



1 References and Versions

The following references will provide additional information on NET's Unified Communications Products

Net Product Documentation Hub:

<https://support.net.com/display/ALLDOC/NET+Product+Documentation;jsessionid=BA930EB97FD04F91A5C84BA74B9DAE35>

UX2000 Specific: <https://support.net.com/display/UXDOC/Home>

Tenor Quickstart Guide:

<https://support.net.com/download/attachments/3407913/TenorAFQuickStart.pdf?version=3&modificationDate=1303316655000>

Tested Versions:

UX2000: 1.2.0v39

Lync Server 2010: 4.0.7577.0

2 Purpose

This guide is for configuring the necessary options for interconnecting NET's UX2000 SBC between XO's SIP Trunk offerings and Microsoft's Lync Server 2010 over the public internet. The UX2000 provides a secure method of isolating the internal corporate data network from the publicly reachable SIP Trunk offerings. The UX2000 is located between the Public Internet and the internal WAN/LAN network as shown below:

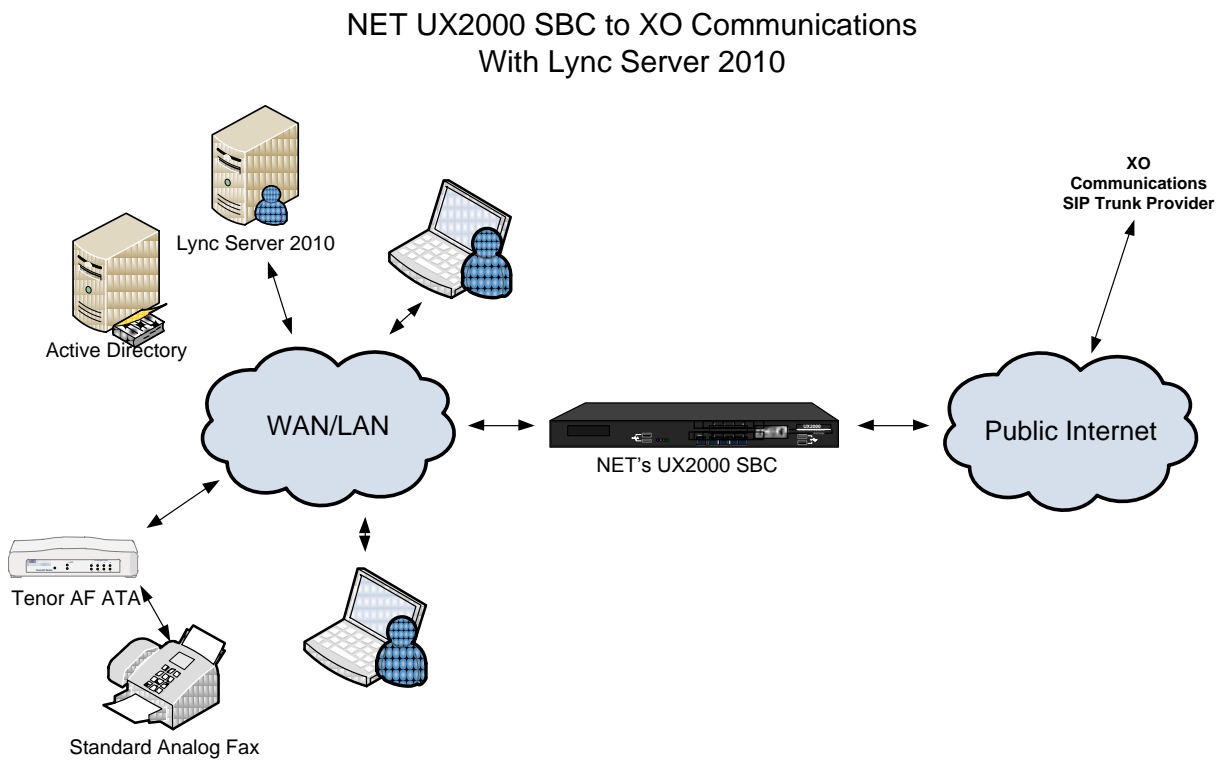


Figure 1 Typical Lync Server 2010 to XO SIP Trunk Deployment

3 Completing SIP Trunk Task on UX2000

Configuring the UX2000 is done through the UX2000s integrated web server. This guide assumes that the operator has already done the initial configuration positioning the UX2000 on the IP network. To start the configuration process, use a standard web browser to connect to the IP or FQDN address of the UX2000. Supply the username and password to complete the login process.

3.1 Prerequisites

The following will be required to complete the configuration of the UX2000 SIP Trunk Task:

- a. Access to the Certificate Authority to download the root certificate and sign and download the user certificate for the proper TLS operation between the UX2000 and the Lync 2010 Server(s)
- b. FQDN of the Lync 2010 Mediation Server or Server pool
- c. IP Address(s) of the XO Communications SIP Signalling destination server
- d. FQDN of the UX2000

3.2 Initial Task configuration

The Initial Task configuration is a step by step process that will complete the steps to position the UX2000 between XO's SIP Trunk and Microsoft's Lync Server 2010. This task will install the necessary TLS Certificates and create SIP components and call routing basics.

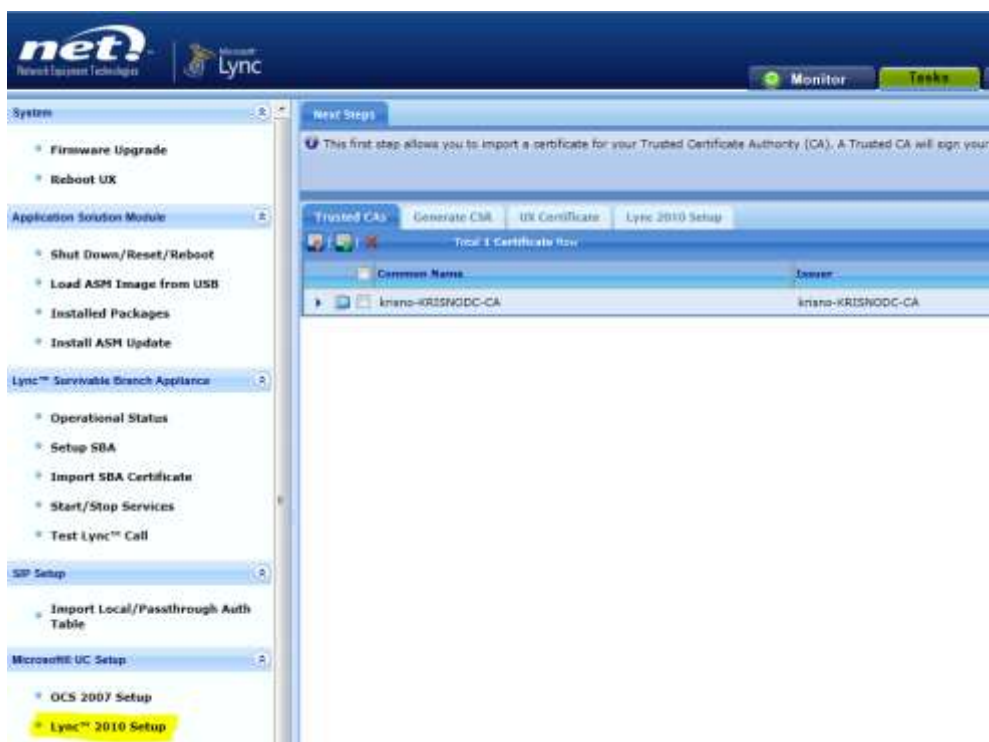


Figure 2 Initial Task Configuration

3.3 Root Certificate

Using the web browser navigate to the Task tab and click on the ‘Microsoft UC Setup/Lync 2010 Setup’ link on the left pane. This will start the task to setup the UX2000 to be configured between the XO SIP Trunk and the Lync Server 2010 components.

The page that loads will have 4 sub tabs located in the right configuration pane. Click on the ‘Trusted CAs’ tab and then the left box with the red up arrow icon. This will prompt for the import of the Root Certificate. There are 2 options for the import type, DER and PEM. DER is used to import a binary type format certificate and will open a file explorer to import the certificate. The PEM type is used to import an ASCII type format. To use the PEM type, first open the certificate file with a text editor such as Microsoft Notepad and then copy and paste into the text box as shown:

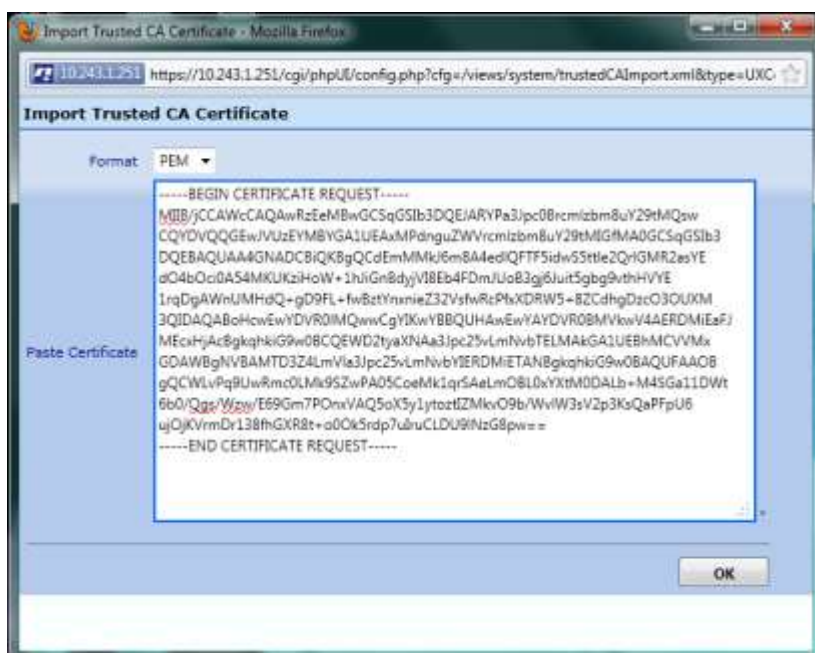


Figure 3 Importing Root Certificate PEM Format

3.4 UX Certificate

Click on the 'Generate CSR' tab. Complete the dialog below with the minimum of the FQDN of the UX2000 in the 'Common Name' field. Once the dialog is populated, click 'OK'. This will create the unsigned certificate that will have to be submitted to the Certificate Authority to be signed. This submission process varies per installation and is out of the scope of this document.



Figure 4 Generating UX Certificate Request

Once the certificate is signed from the Certificate Authority click on the 'UX Certificate' tab, click on the green '+' sign. This will load a dialog similar to the Root Certificate import process. Select either the DER or PEM format in the dropdown to match the file format of the UX Certificate to complete the process. Shown below is the PEM format import process:

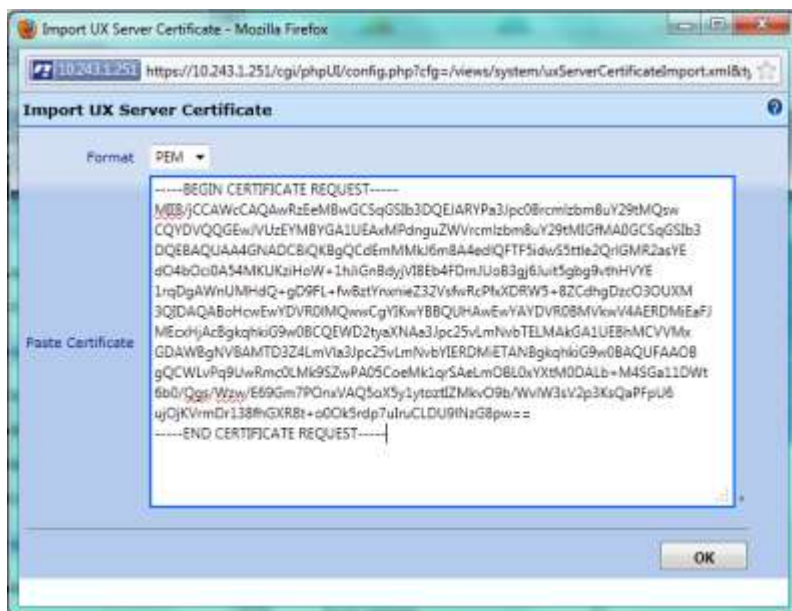


Figure 5 Importing UX Certificate PEM Format

3.5 SIP Signaling Endpoints

Click on the last tab 'Lync 2010 Setup' to configure the SIP Endpoints in UX. This process will setup the UX to interface to the XO Sip Trunk and to the Lync 2010 Server(s). The task will configure the SIP Server tables and the SIP Signaling groups associated with each entity along with the basic routing between each endpoint:

Scenario Information

Scenario Description: Provide a description that will associate to the setup of the SIP Trunk

Gateway Scenario: Select from the dropdown 'SIP Trunking'

SIP Properties

No of Channels: Provide the total number of simultaneous calls expected to the SIP Trunking Provider

Lync Server Pool

Server Pool Host: The FQDN of the Server or Server pool that the UX will communicate with

Port Number: The IP Port that the Server or Server pool listens on for SIP messages from the UX

Border Element Servers

Border Element Server: IP or FQDN Address of the XO server that the UX will communicate with

Port Number: The IP Port that the XO server listens on for SIP messages from the UX

Use Secondary Border Element Server: Select from the dropdown if a second server will be configured

Secondary Border Element Server: IP or FQDN Address of the secondary XO server that the UX will communicate with

Port Number: The IP Port that the secondary XO server listens on for SIP messages from the UX

Once the page has been configured click 'Apply' to finish the dialog.

The screenshot shows the 'Lync 2010 Setup' dialog box with the 'SIP Properties' tab selected. The 'Scenario Information' section has 'Scenario Description' set to 'XO Lync' and 'Gateway Scenario' set to 'SIP Trunking'. The 'SIP Properties' section includes 'No. of Channels' set to 30. The 'Lync Server Pool' section has 'Server Pool Host' set to 'mskrisno.krisno.com' and 'Port Number' set to 5067. The 'Border Element Servers' section has 'Border Element Server' set to '224.4.5.6' and 'Port Number' set to 5060. The 'Use Secondary Border Element Server' dropdown is set to 'Disabled'. An 'Apply' button is located at the bottom right.

Figure 6 Setting Up SIP Signalling Endpoints

4 Configuring Public Ethernet Port and IP Addressing

The UX2000 SBC mediates between a company's internal IP network for SIP signalling and SIP media and the external public IP network to the XO SIP signalling and media servers. The internal IP addressing is setup during the initial installation of the UX2000 SBC and is outside the scope of this Configuration Guide.

4.1 Configuring the Public Ethernet Port

There are 5 Ethernet ports on the front of the UX2000 SBC. The far left (Admin) is reserved for administrative use during the initial configuration. The 4 other ports can be used for normal SIP signalling and media. Port 1 will be configured during the initial setup of the UX2000 SBC and will normally be used for the company's internal IP network.



Prerequisites:

- a. Select unused Ethernet Port
- b. Public IP Address for UX2000 SBC
- c. Public IP Subnet Mask for UX2000 SBC
- d. Public IP Default Route for UX2000 SBC
- e. Private IP Static Route if Needed

4.2 Selecting Unused Ethernet Port

From the Settings Tab/Node Interfaces/Ports display select an Ethernet that is not currently used. These are shown below:

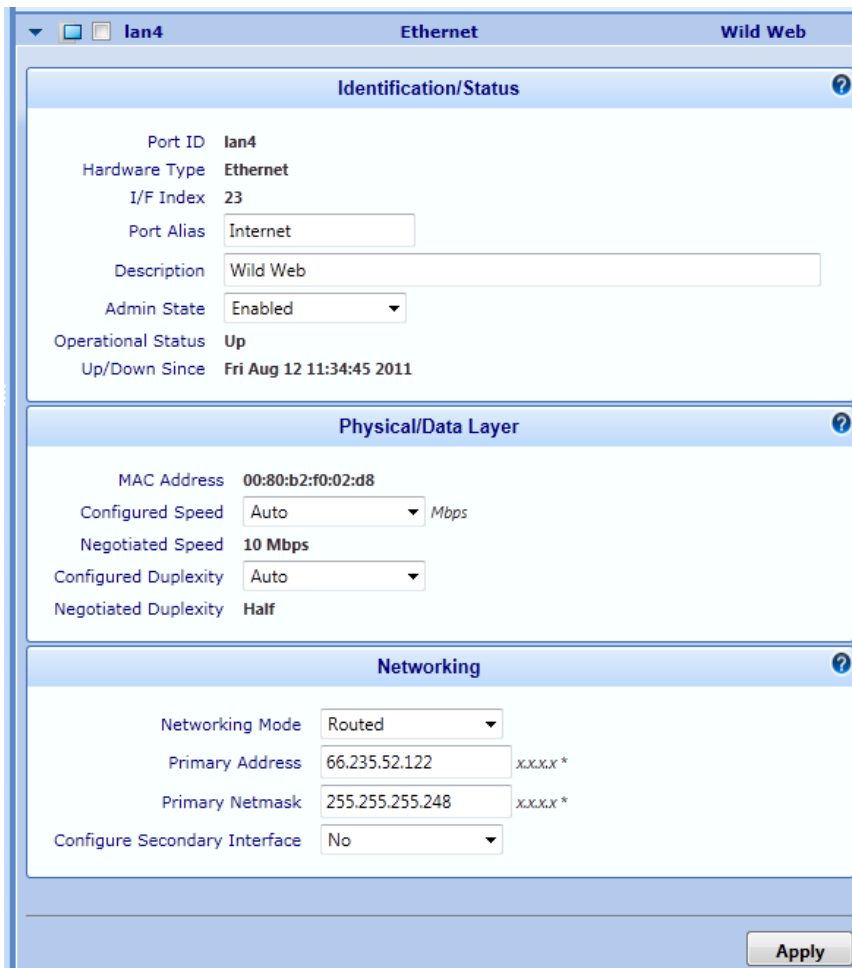


Figure 7 Node Interfaces

In Figure 7 Ethernet port 4 will be used for the Public IP connection. Click the Right Arrow next to Lan4 to open the Configuration Dialog. Supply the following information:

- Port Alias: Name for the Ethernet Port
- Description: Provide a description of the Ethernet Port
- Configured Speed: Leave this at 'Auto' unless the Ethernet port will be connected to a device that can not automatically negotiate the highest speed
- Configured Duplexity: Leave this at 'Auto' unless the Ethernet port will be connected to a device that can not automatically negotiate the port duplex settings
- Networking Mode: Select 'Routed' from the drop down
- Primary Address: Supply the IP Address assigned to the UX2000 SBC
- Primary Netmask: Supply the Netmask associated with the IP Address of the UX2000 SBC
- Configure Secondary Address: Ensure this is set to 'No'

Click 'Apply' to complete the configuration dialog.



Identification/Status	
Port ID	lan4
Hardware Type	Ethernet
I/F Index	23
Port Alias	Internet
Description	Wild Web
Admin State	Enabled
Operational Status	Up
Up/Down Since	Fri Aug 12 11:34:45 2011

Physical/Data Layer	
MAC Address	00:80:b2:f0:02:d8
Configured Speed	Auto Mbps
Negotiated Speed	10 Mbps
Configured Duplexity	Auto
Negotiated Duplexity	Half

Networking	
Networking Mode	Routed
Primary Address	66.235.52.122 x.x.x.x *
Primary Netmask	255.255.255.248 x.x.x.x *
Configure Secondary Interface	No

Apply

Figure 8 Ethernet Port Configuration

4.3 Configuring the IP Default Route

The IP Default route is required so that the UX2000 SBC will know where to send IP packets into the public IP network. From the Settings/Protocols/IP/Static Routes click on the green '+' sign to add the route. This will open the dialog for the IP Route. Complete the dialog with the following information:



Figure 9 IP Default Route

Destination IP: Enter '0.0.0.0'
 Mask: Enter '0.0.0.0'
 Gateway: Enter the IP Address of the IP Default Router
 Metric: Enter '1'

Once configured click 'OK' to apply the configuration.

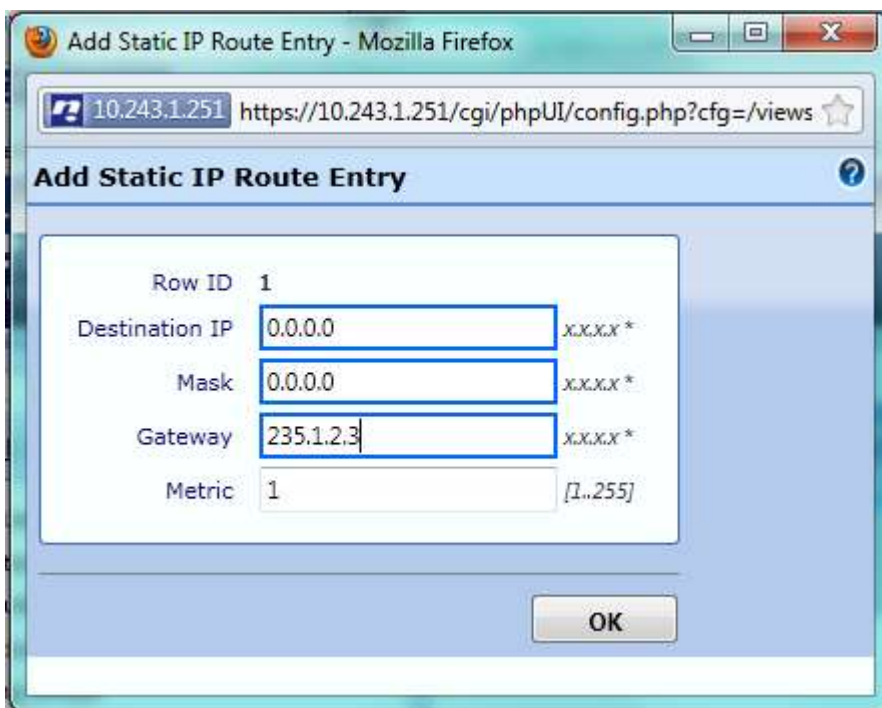


Figure 10 Public IP Default Route

4.4 Configure Internal Private IP Static Route

Internal IP Static Routes may be required if the UX2000 SBC needs to communicate with multiple internal private IP addresses. These static routes have to be deterministic because there can be only one default route and this is used on the public side. If there are multiple internal networks then multiple static routes will be required. The examples below show static routes for a private 10.0.0.0 network and a private 192.168.0.0 network:

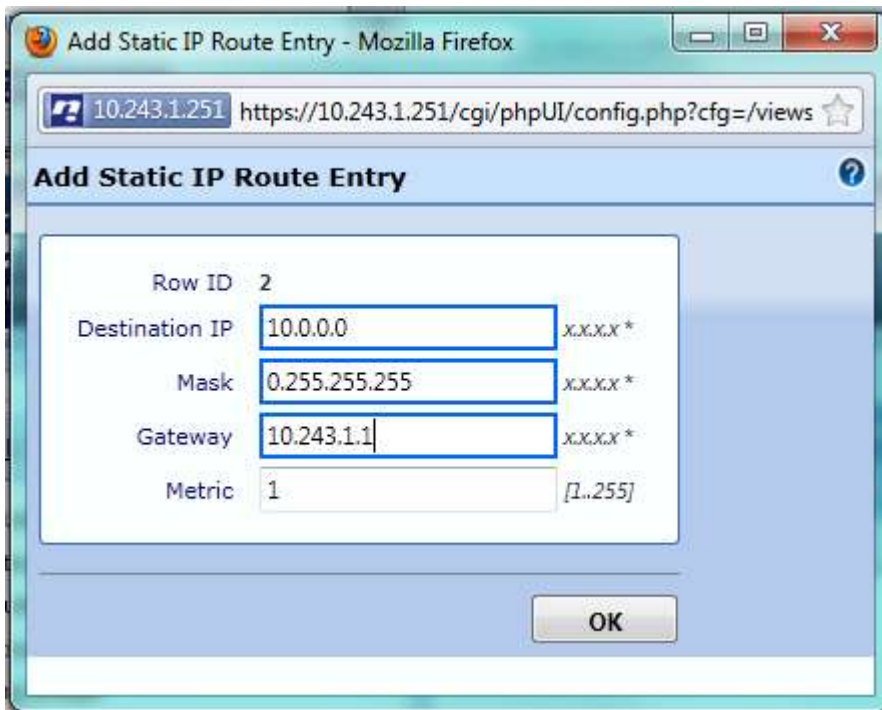


Figure 11 Internal Static IP Route 1

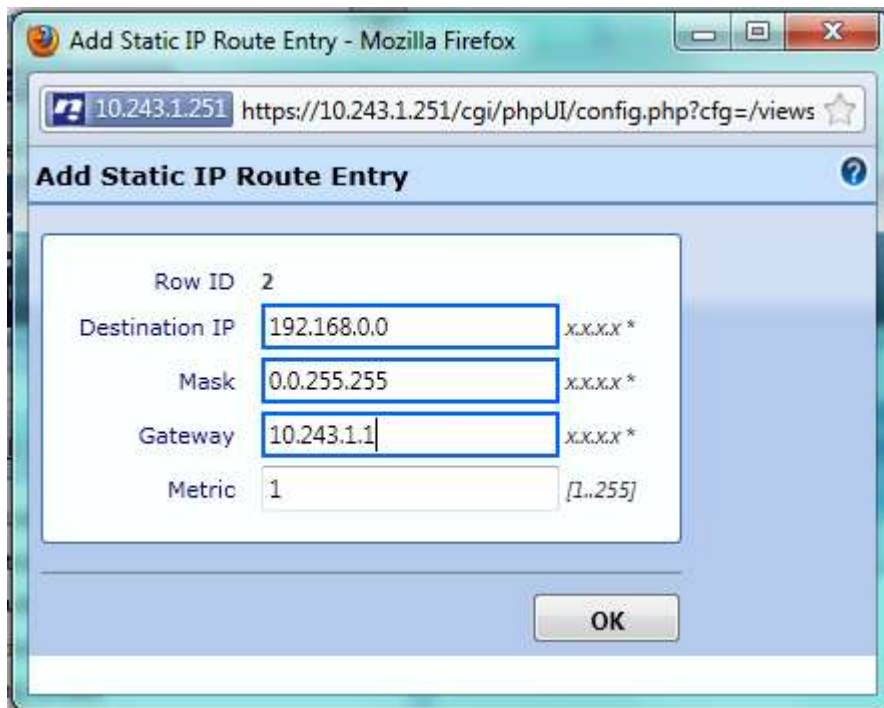


Figure 12 Internal Static Route 2

5 Lync Server 2010 Configuration

Lync Server 2010 will need to be configured to support the UX2000 SBC in order to communicate with the XO SIP Trunk. This section covers the addition of the UX2000 into the Lync Server topology and adding the UX2000 to the Lync Server 2010 routing. XO SIP Trunk DIDs will be assigned to a Lync users account and the commands will be shown how to add an analog contact with a XO SIP Trunk DID.

This section assumes that the Lync Server components have been installed along with Lync users. The user should be familiar with Lync Server Topology Builder, Lync Server Control Panel and Lync Server management Shell. This section does not cover the basic installation of Lync Server 2010.

5.1 Adding UX2000 SBC to Lync Server Topology

The Lync Server topology needs to be modified by adding the UX2000 SBC as a Gateway device. The Gateway device will be the interface to the XO SIP Trunk.

Open Lync Server Topology builder and load the current topology. Expand the topology and click on the 'PSTN Gateways' link in the left hand pane. This will provide the link in the right hand pane 'New IP/PSTN Gateway'. Click on this to open the dialog below:

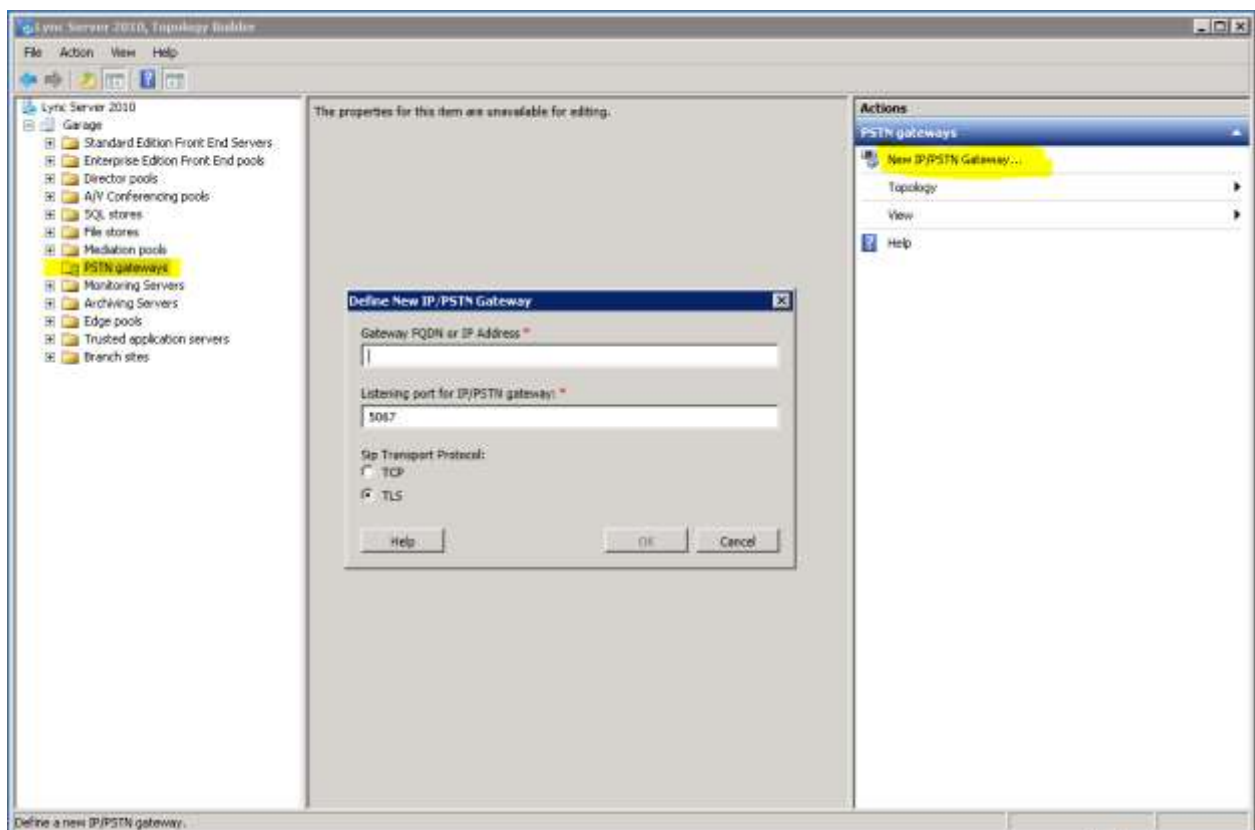


Figure 13 Adding New IP/PSTN Gateway in Lync Server 2010 Topology

Populate the following into the dialog:

- Gateway FQDN or IP Address: For this config enter the FQDN of the UX2000 SBC since TLS is going to be used between the Lync Server 2010 and the UX2000 SBC 'ux.krisno.com'
- Listening port for the IP/PSTN gateway: Enter '5067'
- SIP Transport Protocol: Ensure the radio button for TLS is selected

Once the dialog is complete click 'OK'.

Click on the '+' sign next to the Mediation Pools to expand and then click on the configured Mediation Server. Click on the 'Edit Properties' link in the right hand pane to open the edit dialog as shown below:

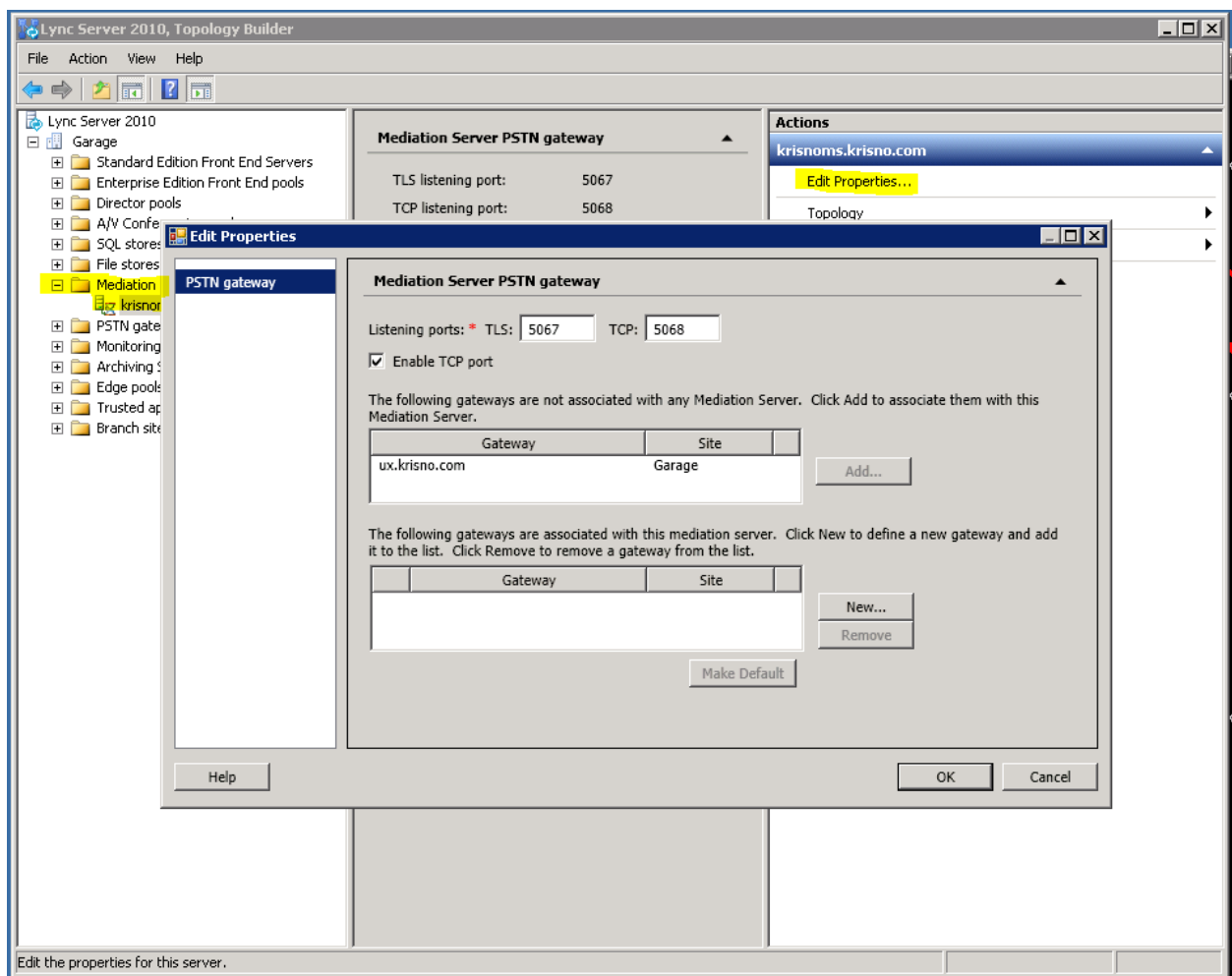


Figure 14 Edit Mediation Server Dialog

Click on the gateway FQDN 'ux.krisno.com' in the none associated gateway table. Once selected, click the 'Add' button. This will move the 'ux.krisno.com' FQDN to the bottom table for associated gateways. Click 'OK' to complete the dialog.

The topology has been modified with the gateway added and associated with the Mediation Server.

Publish the topology to complete the task.

5.2 Adding the UX2000 into Lync Server Routing

In order for Lync Server 2010 to send calls to the XO SIP Trunk the UX2000 SBC will have to be added to the Routing. Open Lync Server Control Panel and click on the Voice Routing link on the left hand pane. Click on the Route tab on top of the right hand pane to show the dialog below:

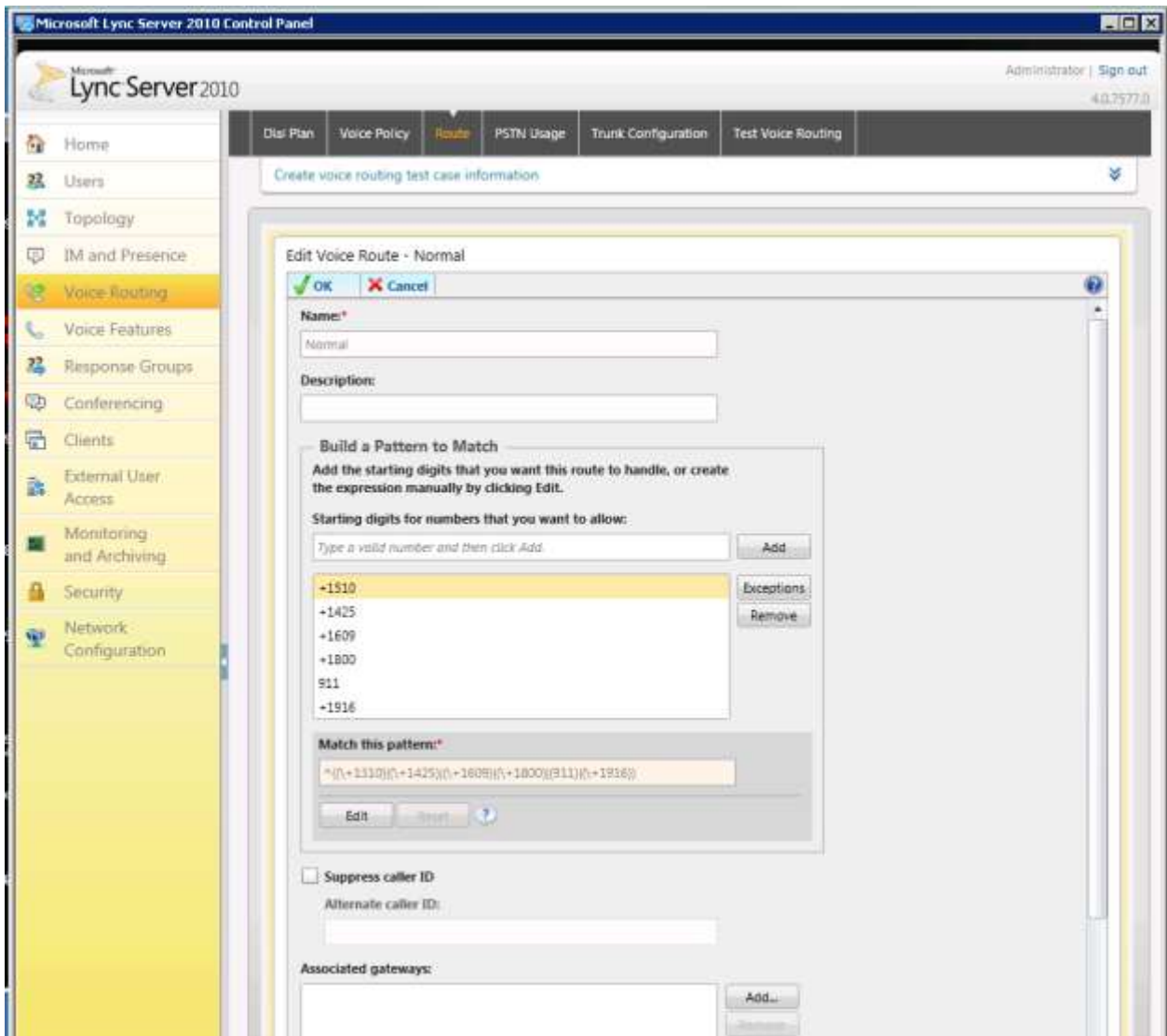


Figure 15 Lync Server Control Panel Route Selection

Click on the 'Add' button next to 'Associated gateways' table. This will bring up a list of available gateways and select the FQDN for the UX2000 SBC as shown below:

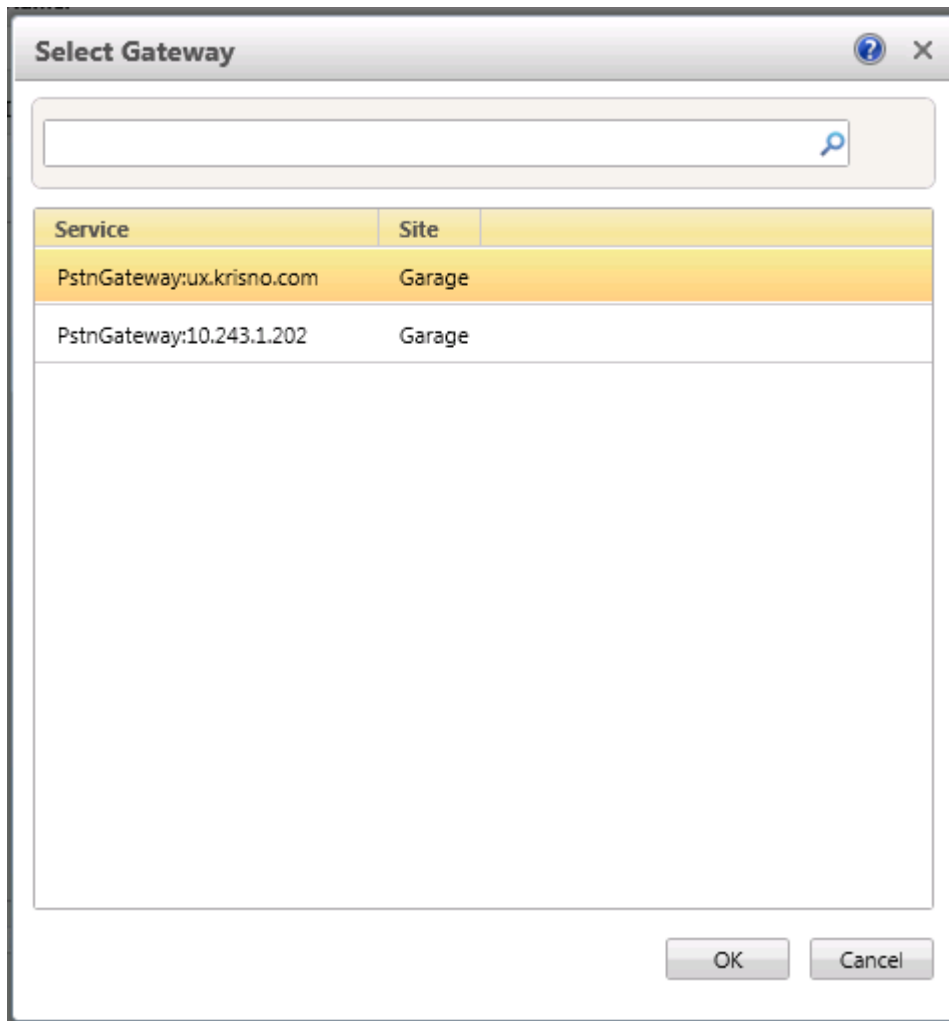


Figure 16 Available PSTN Gateways

Ensure the UX2000 SBC is highlighted in the dialog and click 'OK'.

Scroll down to the 'Associated PSTN Usages'. Click on the 'Select' button to bring up the list of available PSTN Usages. Select the proper PSTN Usages from the list and click 'OK'.

With the PSTN Gateway selected and the PSTN Usages selected the dialog should be similar to below:

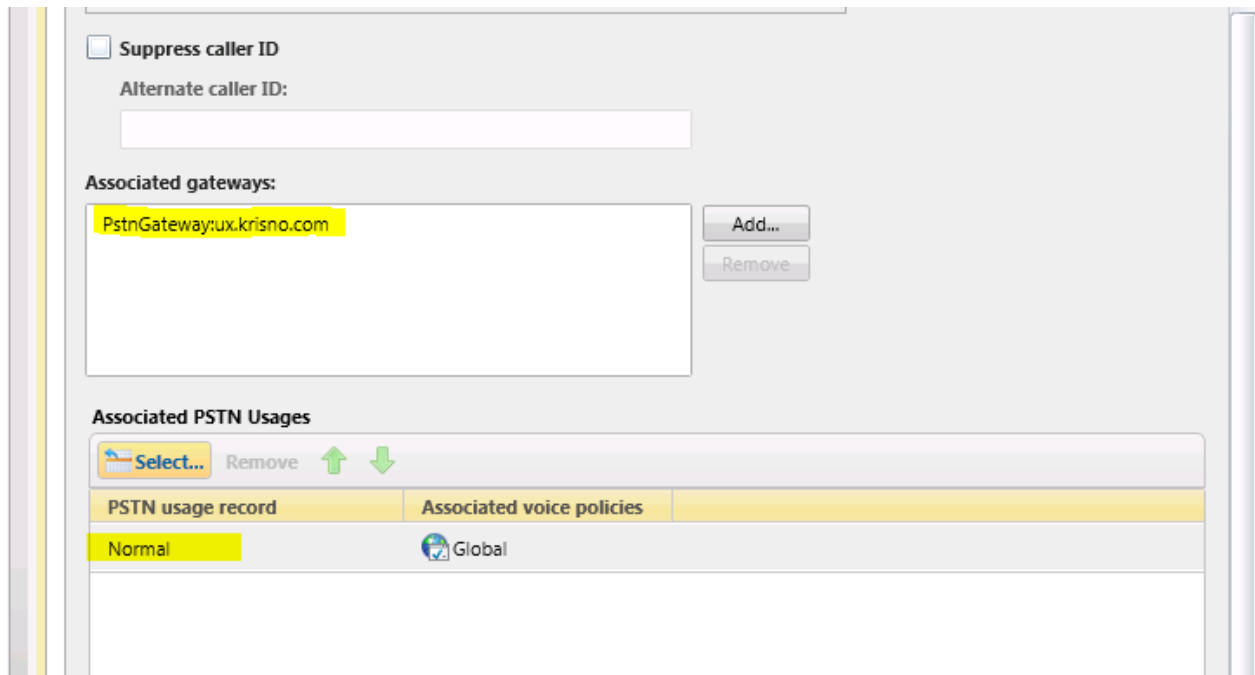


Figure 17 PSTN Gateway and PSTN Usages

At this point commit these changes to the topology.

5.3 Configuring Assigned DID to Lync Users

In order for Lync users to be called directly from the PSTN, the Lync users will have to have the XO SIP Trunk DID numbers assigned. This is done in the User section of Lync Server Control Panel.

Open the specific Lync user in Lync Server Control panel that will have a DID assigned to it. Configure the DID in the Line URI configuration as shown below:

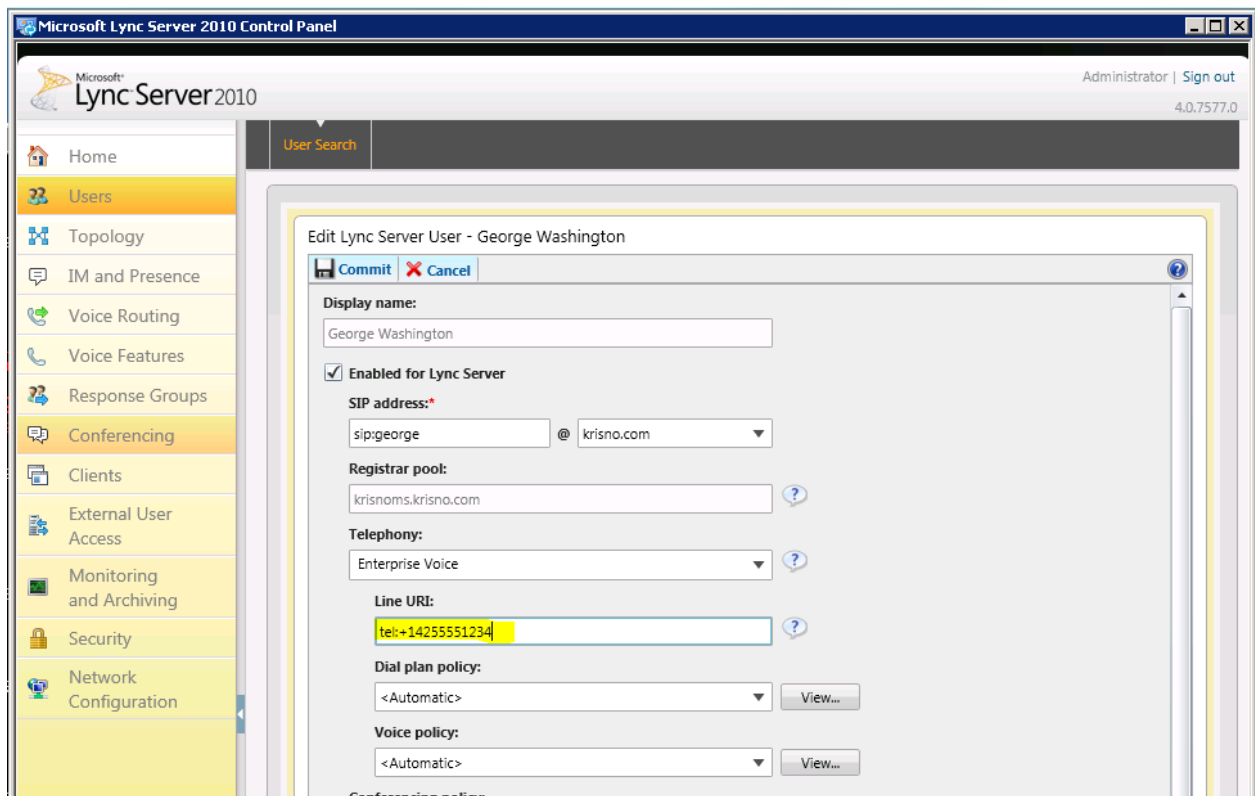


Figure 18 Lync User DID Setting

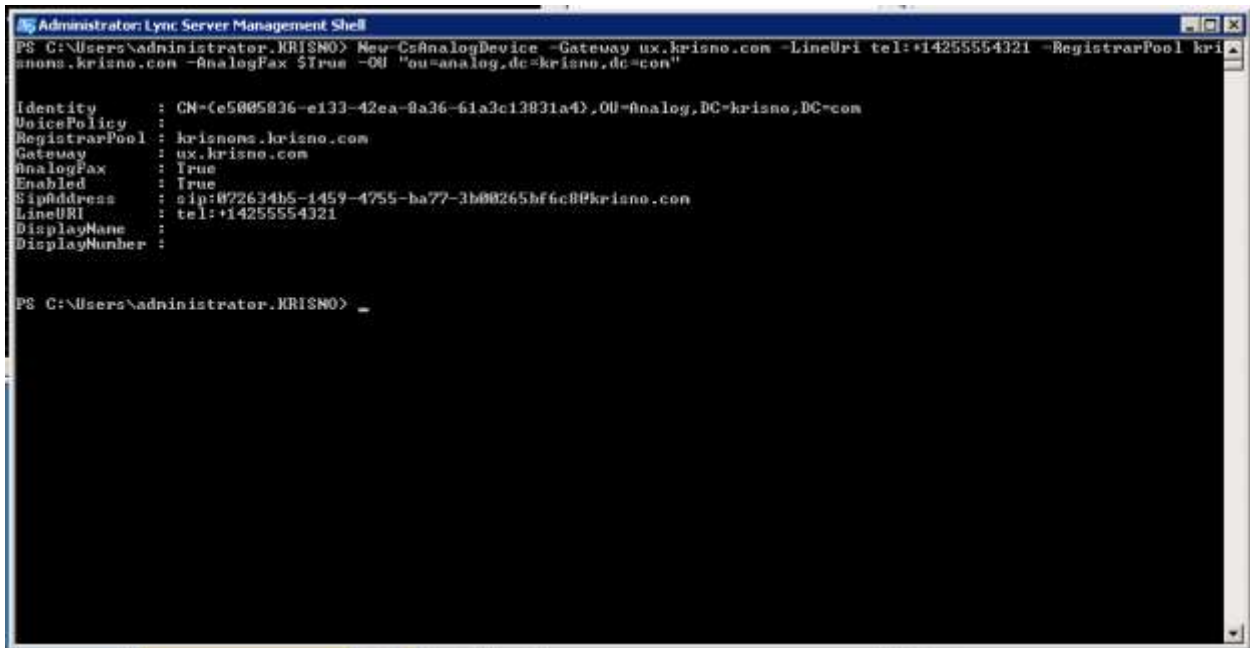
Click 'Commit' to complete the changes.

5.4 Creating Analog Contacts with XO SIP Trunk DIDs

In order for analog devices to be called directly from the PSTN, the analog devices will have to have the proper XO SIP Trunk DID assigned to it.

This section does not go into details on configuring analog devices. Please see the Lync Server 2010 documentation for further details.

Open Lync Server Management Shell. Below is an example of how to create an analog contact as a fax machine and a XO SIP Trunk DID number of 14255554321:



```
Administrator: Lync Server Management Shell
PS C:\Users\administrator.KRISNO> New-CsAnalogDevice -Gateway ux.krisno.com -LineUri tel:+14255554321 -RegistrarPool krisnons.krisno.com -AnalogFax $True -OU "ou=analog,dc=krisno,dc=com"

Identity       : CN=(c5005036-e133-42ea-8a36-61a3c13831a4),OU=analog,DC=krisno,DC=com
VoicePolicy    :
RegistrarPool  : krisnons.krisno.com
Gateway        : ux.krisno.com
AnalogFax      : True
Enabled        : True
SipAddress     : sip:0722634h5-1459-4755-ha77-3b00265bf6c0@krisno.com
LineURI        : tel:+14255554321
DisplayName    :
DisplayNumber  :
```

PS C:\Users\administrator.KRISNO> _

Figure 19 Adding Analog Contact

6 UX2000 Telephony Routing Process

The telephone routing process in the UX2000 moves calls received on one Signaling Group (SG) to an outbound SG. While the call is moving the number or name may be translated and additional lookups can be done to ensure the call is routed properly.

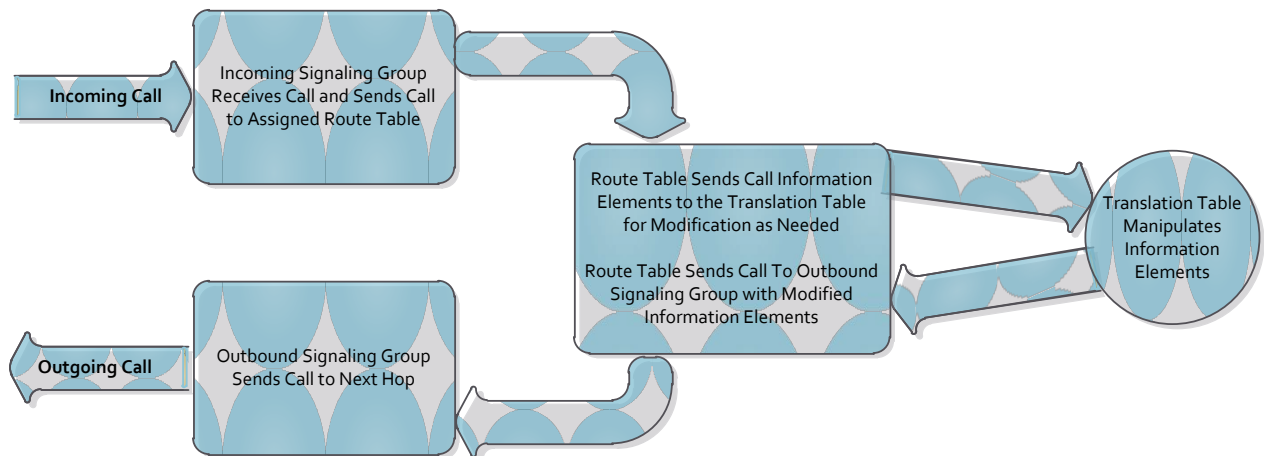


Figure 20 UX2000 Routing Process

6.1 Signaling Groups

The SG is the first place an incoming call will reach the UX2000. The Signaling group will direct the call to the Routing Table as shown below:

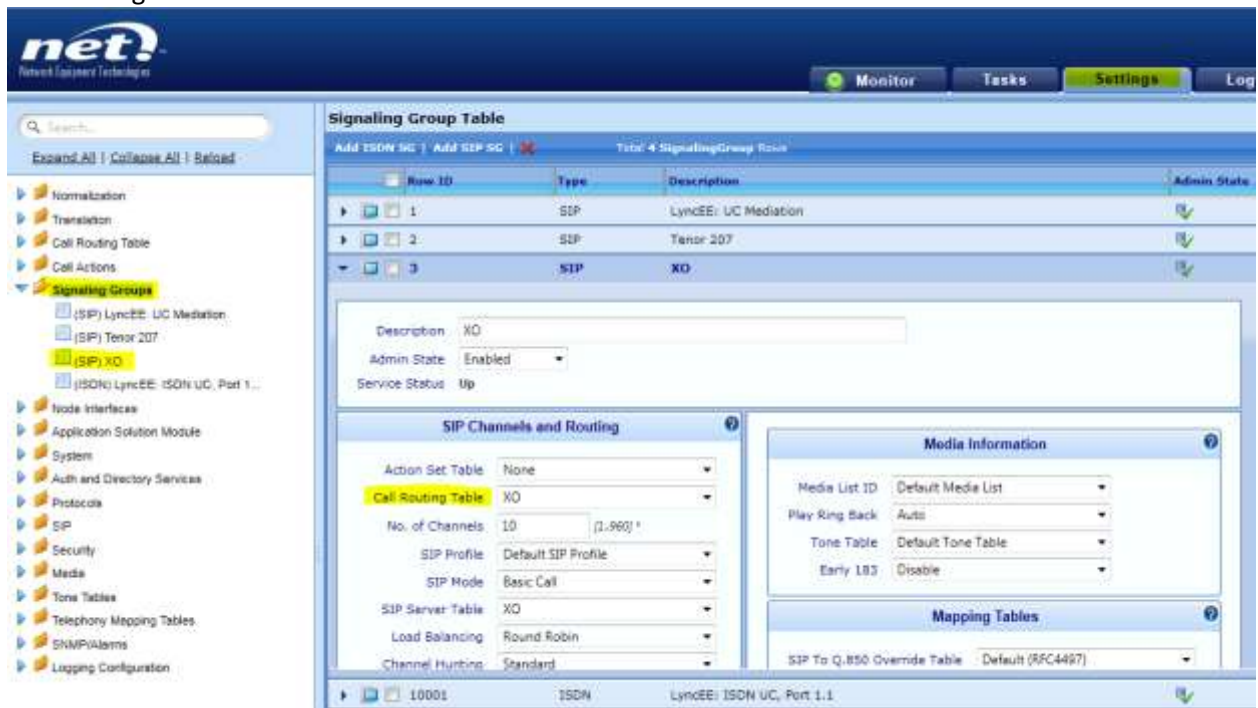


Figure 21 Signaling Group

6.2 Call Routing Table

The Call Routing Table will send the Called and Calling Party information elements to a Translation Table so that they can be modified if required to match the destination systems format. Once the translations have been completed the Route Table will then route the call to the applicable SG for the destination system.

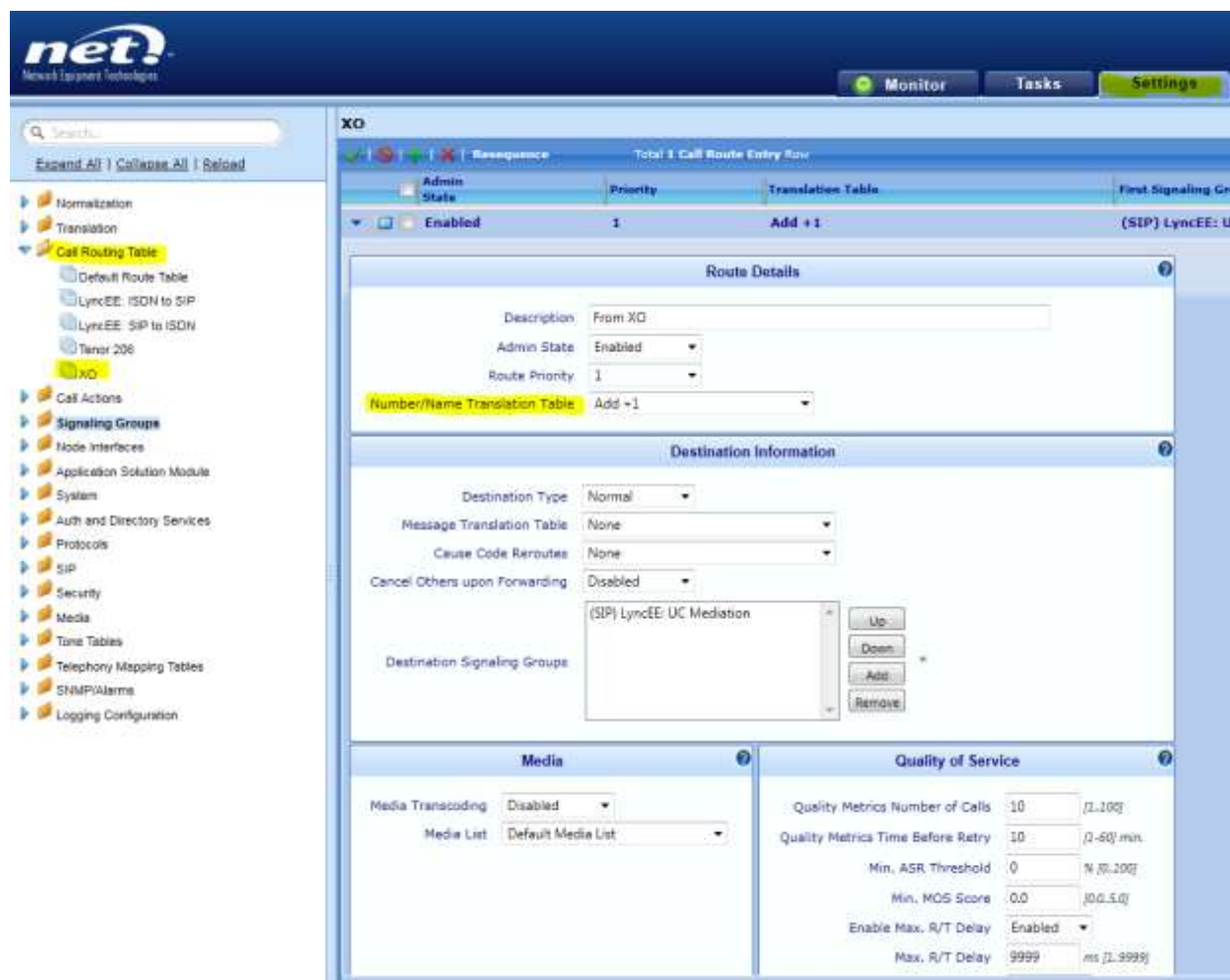


Figure 22 Call Routing Table

6.3 Translation Tables

Translation tables are used to modify the called and calling party information elements among other elements that are pertinent to proper call handling between the origination and destination servers. The basic translations rely on the use of Regular Expressions as the numbers/names are changed from what is received to what is sent to the destination. Translation tables must have a matching entry format in order for the call to be routed to that route. If there are no matching Translation table entries the call will be returned back to the Routing table so the next Route entry can be tried for a match. This will happen as

long as there are Routes in the Route table to try. If all Routes have been tried with no matches in the Translation table(s) the call will fail.



Figure 23 Translation Tables

Most translation table entries for a SIP Trunking application will be used for matching Lync Server 2010 requirement for called and calling numbers to be in E.164 format. Below are examples to match XO number format to Lync Server 2010 number format.

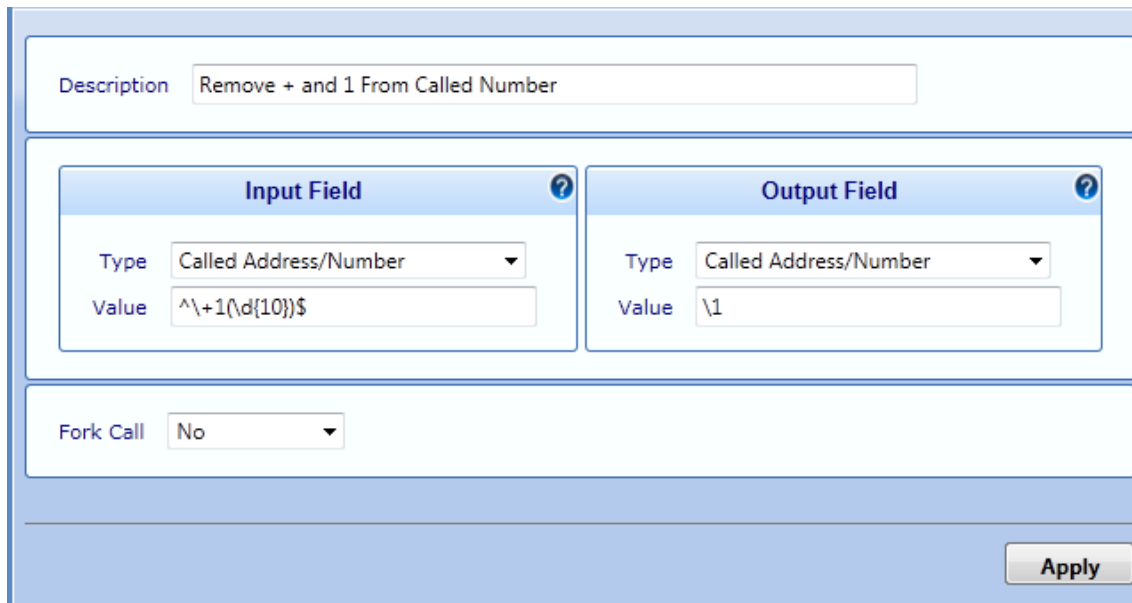
This example is used to translate a called number from Lync 2010 Server in E.164 format to a US 10 digit number for routing into the XO SIP Trunk:

Input Field

Type: Called Address/Number – Used to capture the numeric portion of the Dialed Number
 Value: `^\+1(\d{10})$` - Regular Expression that looks for the +1 at the start of the Dialed Number and then captures the 10 digits that follows between the open and close parenthesis

Output Field

Type: Called Address Number – Identifies the Information Element that the Input Field Type should be translated to
 Value: `\1` – Identifies the open and close parenthesis that should be carried over from the Input Value Field



Description

Input Field	Output Field
Type <input type="text" value="Called Address/Number"/>	Type <input type="text" value="Called Address/Number"/>
Value <input type="text" value="^\+1(\d{10})\$"/>	Value <input type="text" value="\1"/>

Fork Call

Figure 24 Translation Table that Removes + and 1

This example is used to modify a called number from the XO SIP Trunk into an E.164 format that Lync 2010 Server will accept:

Input Field

Type: Called Address/Number – Used to capture the numeric portion of the Dialed Number

Value: `^\+1(\d{10})$` - Regular Expression that looks for the +1 at the start of the Dialed Number and then captures the 10 digits that follows between the open and close parenthesis

Output Field

Type: Called Address Number – Identifies the Information Element that the Input Field Type should be translated to

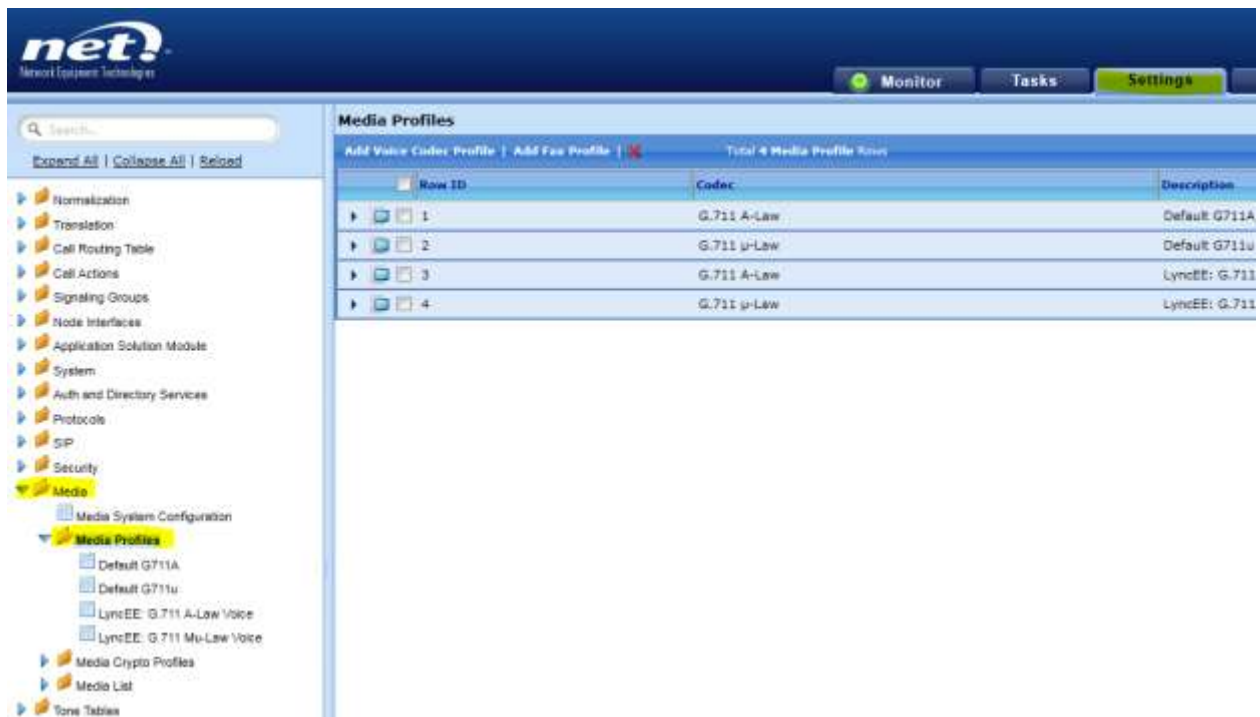
Value: `+1\1` – Prepends '+1' and Identifies the open and close parenthesis that should be carried over from the Input Value Field

Description <input type="text" value="Called Party Add + 1"/>	
Input Field ?	Output Field ?
Type <input type="text" value="Called Address/Number"/>	Type <input type="text" value="Called Address/Number"/>
Value <input type="text" value="^\d{10}\$"/>	Value <input type="text" value="+1\1"/>
Fork Call <input type="text" value="No"/>	
<input type="button" value="Apply"/>	

Figure 25 Translation Table that Adds + and 1

7 Media Configuration

The UX2000 initial configuration task from Section 1 of this guide will setup the required media settings for both the Lync 2010 Server and the XO SIP Trunk. Media lists will be created and applied to the appropriate SGs. Below are examples of a typical Media configuration:



The screenshot shows the 'Media Profiles' configuration page in the net! interface. The left sidebar contains a navigation tree with 'Media Profiles' selected. The main content area displays a table with the following data:

Row ID	Codec	Description
1	G.711 A-Law	Default G711A
2	G.711 μ-Law	Default G711u
3	G.711 A-Law	LyncEE: G.711
4	G.711 μ-Law	LyncEE: G.711

Figure 26 Media Profiles Showing Enabled Codecs



Figure 27 Typical G711Alaw Codec Settings

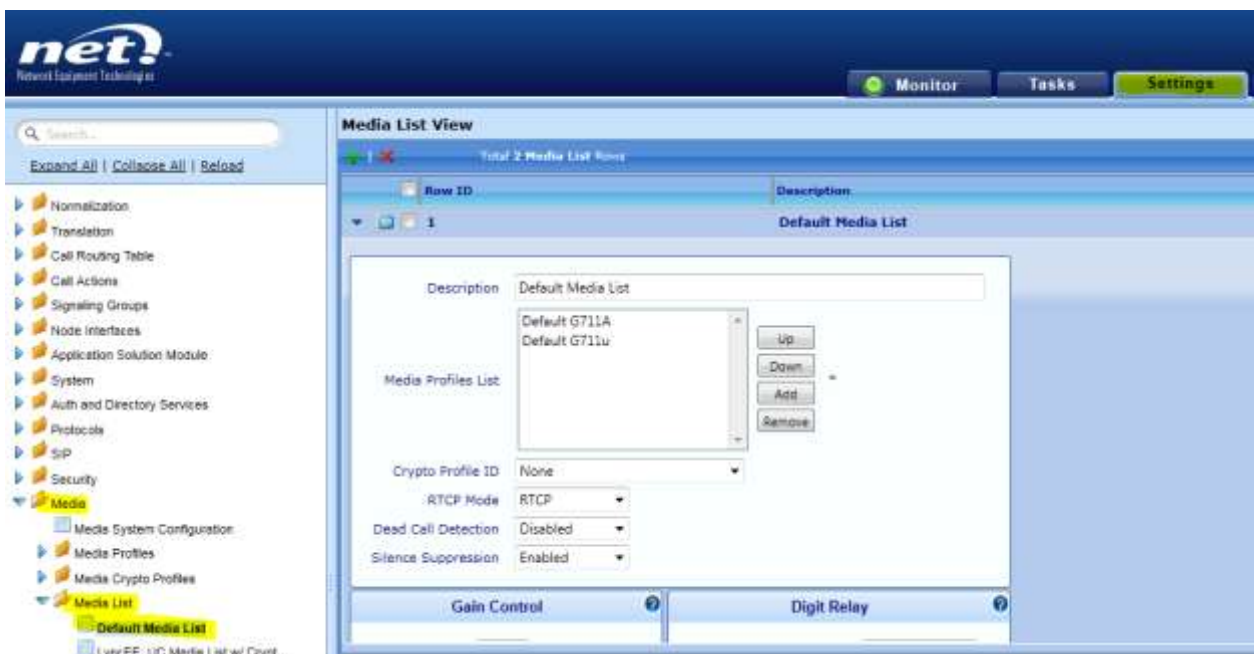


Figure 28 Media List Containing G711Alaw and G711Ulaw

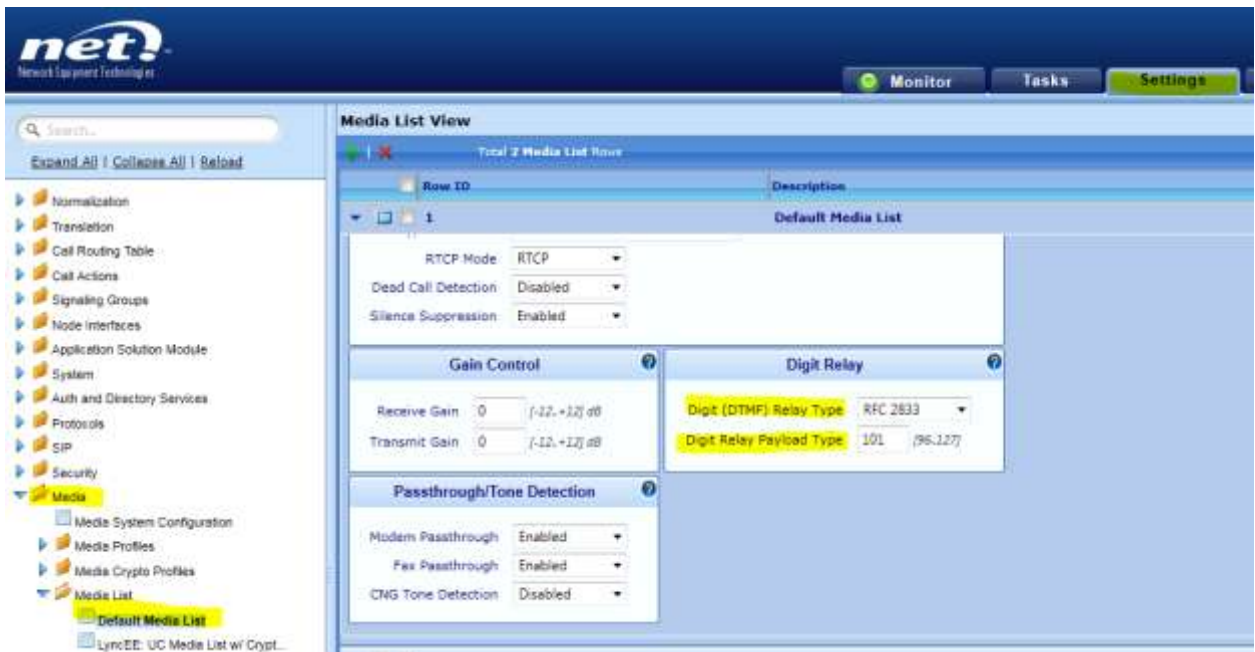


Figure 29 Media List (cont) Showing Digit Relay Settings

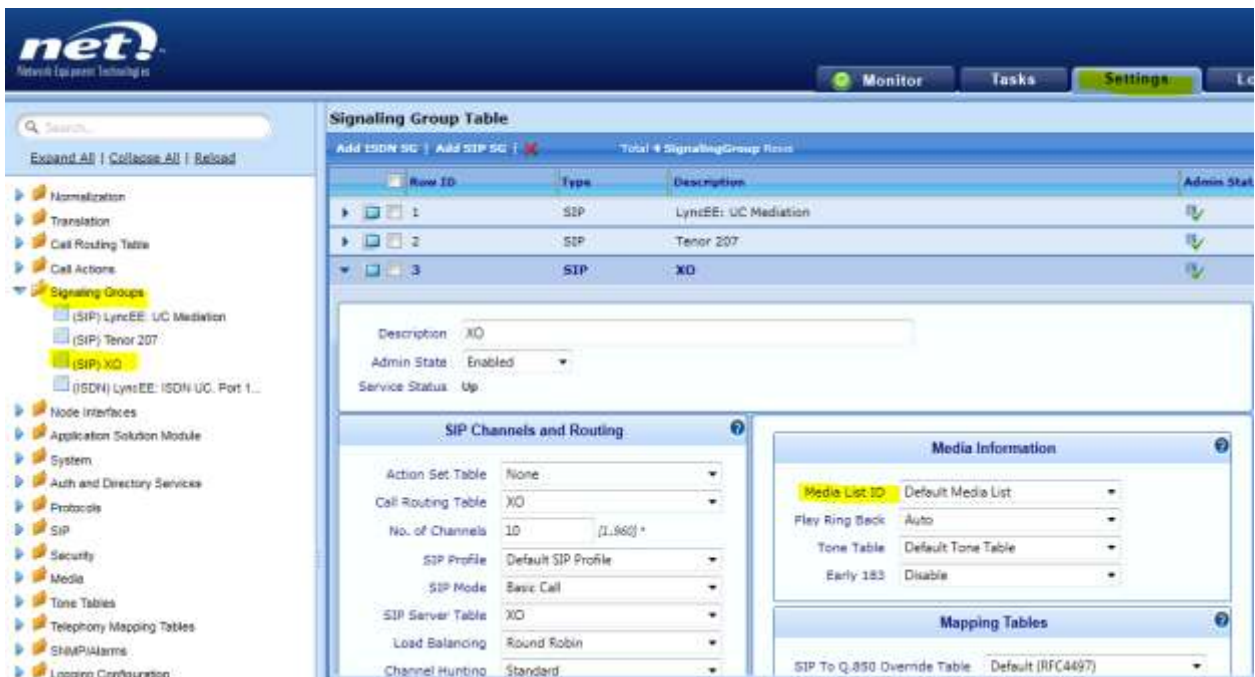
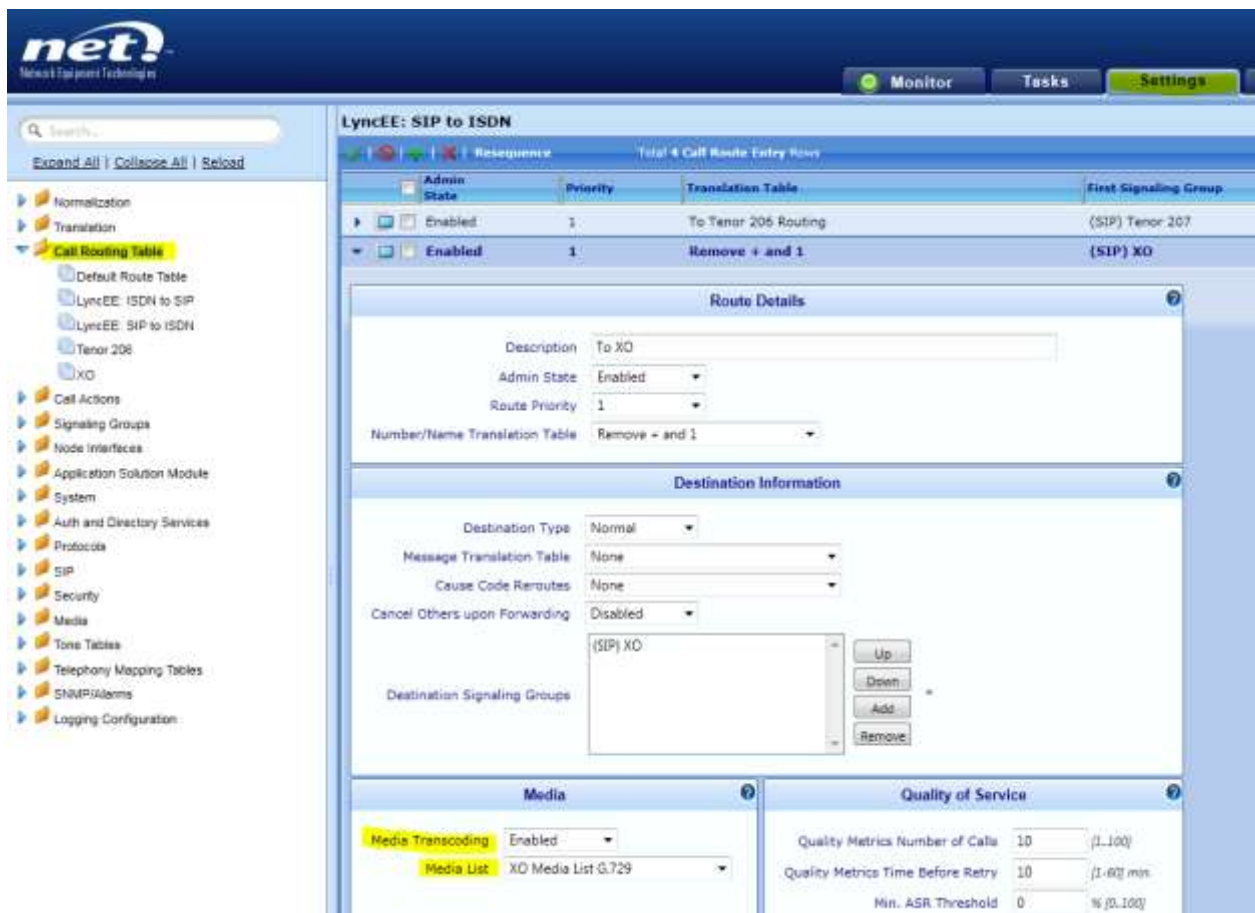


Figure 30 Signaling Group Showing Media List Selection

8 Enabling Media Transcoding

Media Transcoding can be used for changing both codec and DTMF digit transfer methods. Since Lync 2010 Server does not support both G.729 and inband DTMF digit transfer methods the UX2000 can be configured to transcode one list to another list between the XO SIP Trunk and Lync 2010 Server. In order for this to work properly there has to be a Media List for the XO SIP Trunk with G.729 and/or Inband DMF digit transfer settings and another Media List with G711 (A or U law as required) with RFC2833 digit transfer.

The actual Transcoding configuration setting is done at the Route Table. The example below is for a call originating on the Lync 2010 Server using G711 and the matching Route table entry for XO has Transcoding enabled and calls out the Media list that is used for the XO SIP Trunk:



The screenshot shows the configuration page for 'LyncEE: SIP to ISDN'. The left sidebar lists various configuration categories, with 'Call Routing Table' expanded. The main area displays a table of route entries:

Admin State	Priority	Translation Table	First Signaling Group
Enabled	1	To Tenor 206 Routing	(SIP) Tenor 207
Enabled	1	Remove + and 1	(SIP) XO

The 'Route Details' section for the selected entry shows:

- Description: To XO
- Admin State: Enabled
- Route Priority: 1
- Number/Name Translation Table: Remove + and 1

The 'Destination Information' section shows:

- Destination Type: Normal
- Message Translation Table: None
- Cause Code Reroutes: None
- Cancel Others upon Forwarding: Disabled
- Destination Signaling Groups: (SIP) XO

The 'Media' section shows:

- Media Transcoding: Enabled
- Media List: XO Media List G.729

The 'Quality of Service' section shows:

- Quality Metrics Number of Calls: 10 (1..100)
- Quality Metrics Time Before Retry: 10 (1..60) min
- Min. ASR Threshold: 0 % (0..100)

Figure 31 Media Transcoding Configuration

9 Analog Support

Lync Server 2010 supports analog devices such as phones and fax machines in its topology. The devices can be configured with DID numbers so that they can be dialed directly from the PSTN. The analog devices are configured to home to a configured SBC or gateway in the Lync Server 2010 topology. This means that the configuration on the homed device needs to be able to route calls to the analog devices from Lync Server 2010 while also routing calls to the SIP Trunk.

9.1 Differences Between PSTN and Analog Call Routing

When analog devices are part of the Lync Server 2010 topology the UX2000 will have to be able to recognize a call to an analog ATA from Lync Server 2010 and route these differently than calls to the SIP Trunk. Calls from the SIP Trunk will all go to the Lync 2010 Server. The Route table in the UX2000 will have to have knowledge of all analog numbers so these calls can be routed to an ATA instead of to the SIP Trunk. All other numbers that are not recognized as analog devices will then be routed to the PSTN through the XO SIP Trunk.

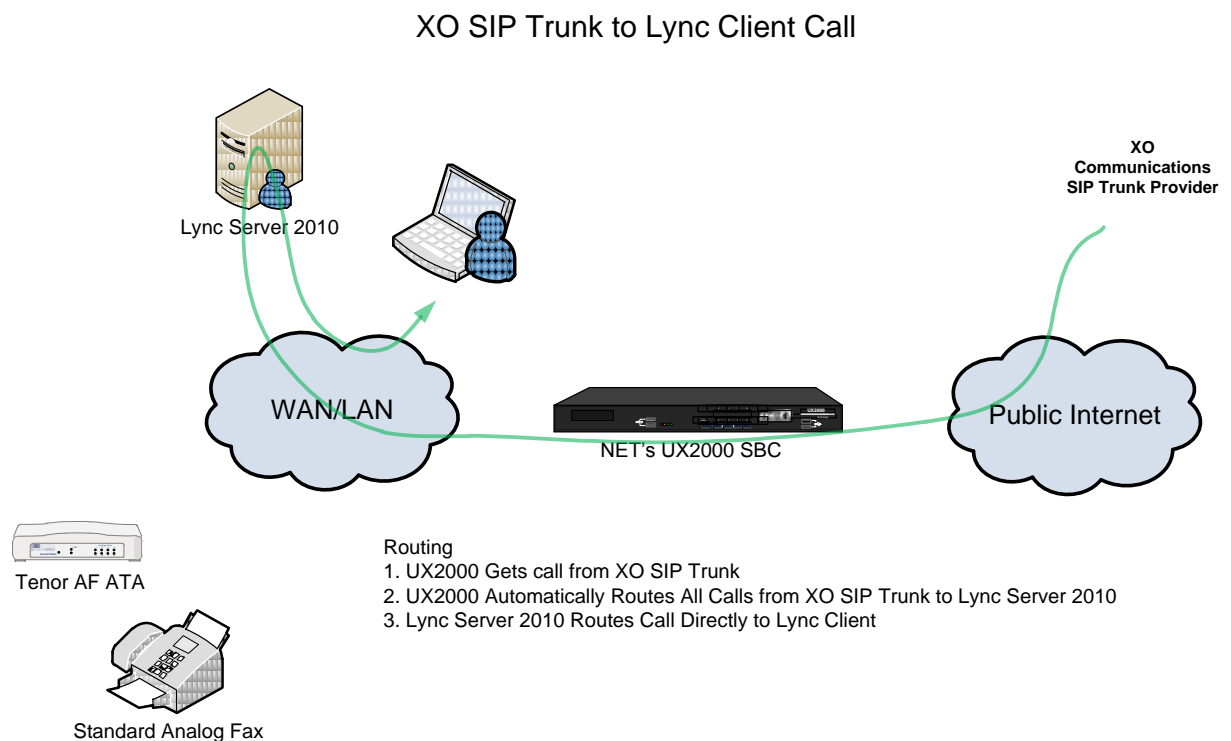


Figure 32 PSTN to Lync Client Call Flow

XO SIP Trunk to Analog Fax Routing

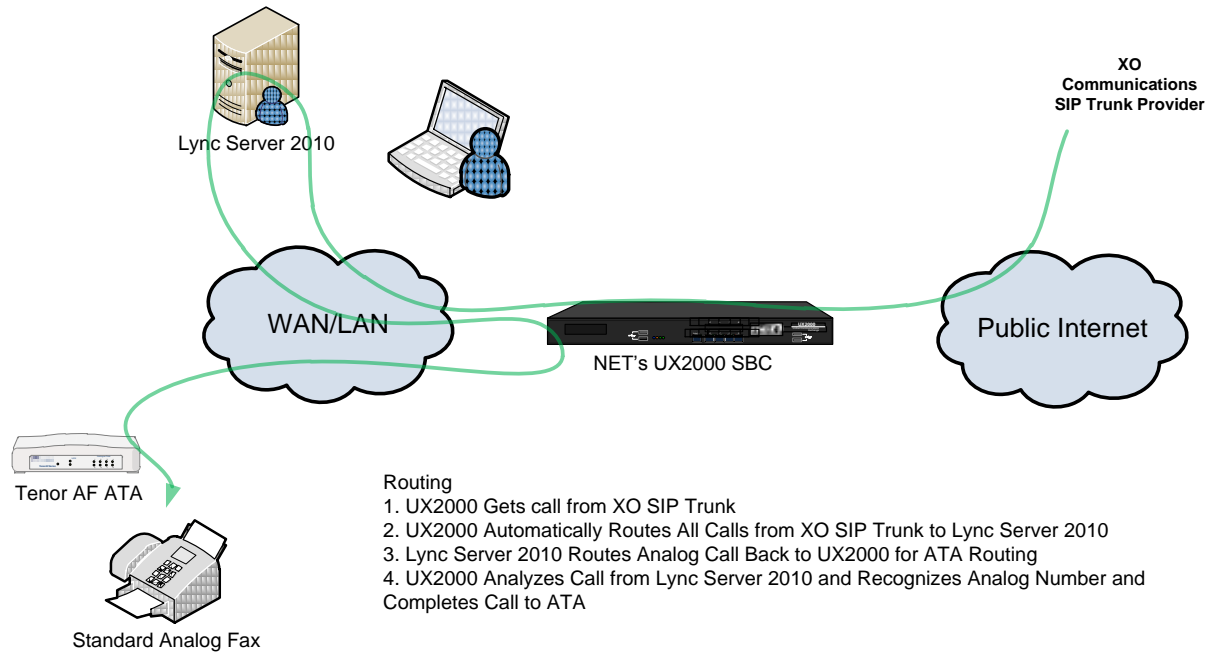


Figure 33 PSTN to Analog Call Flow

Lync Client to PSTN

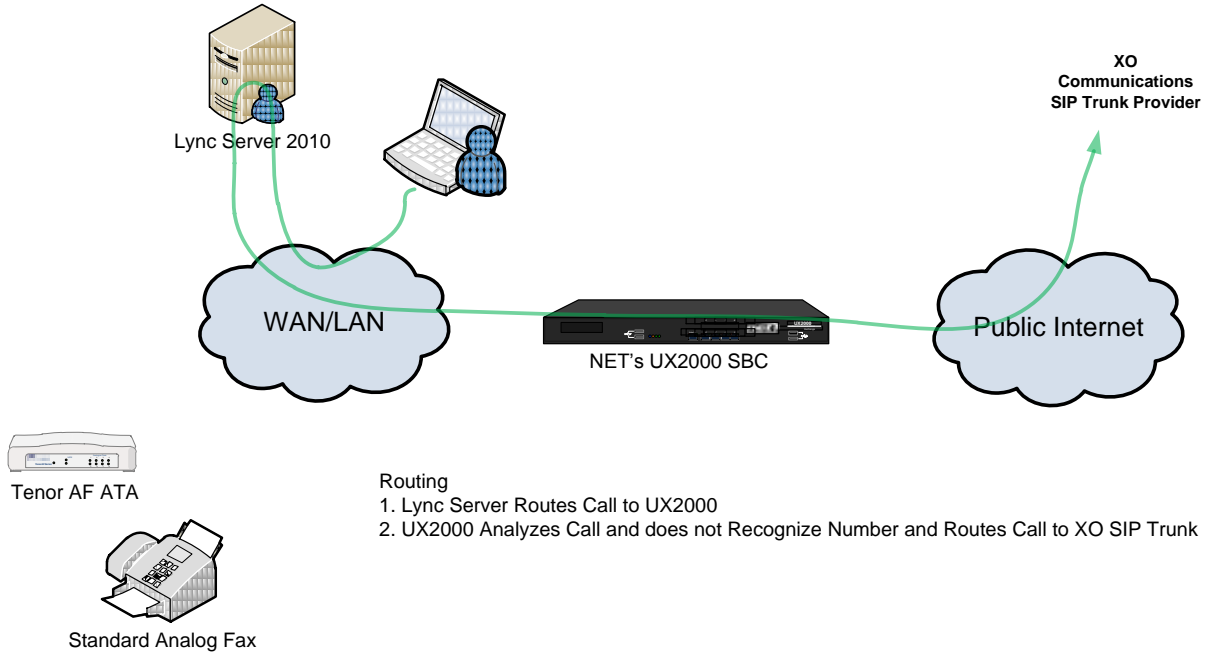


Figure 34 Lync Client to PSTN Call Flow

Lync Client to Analog

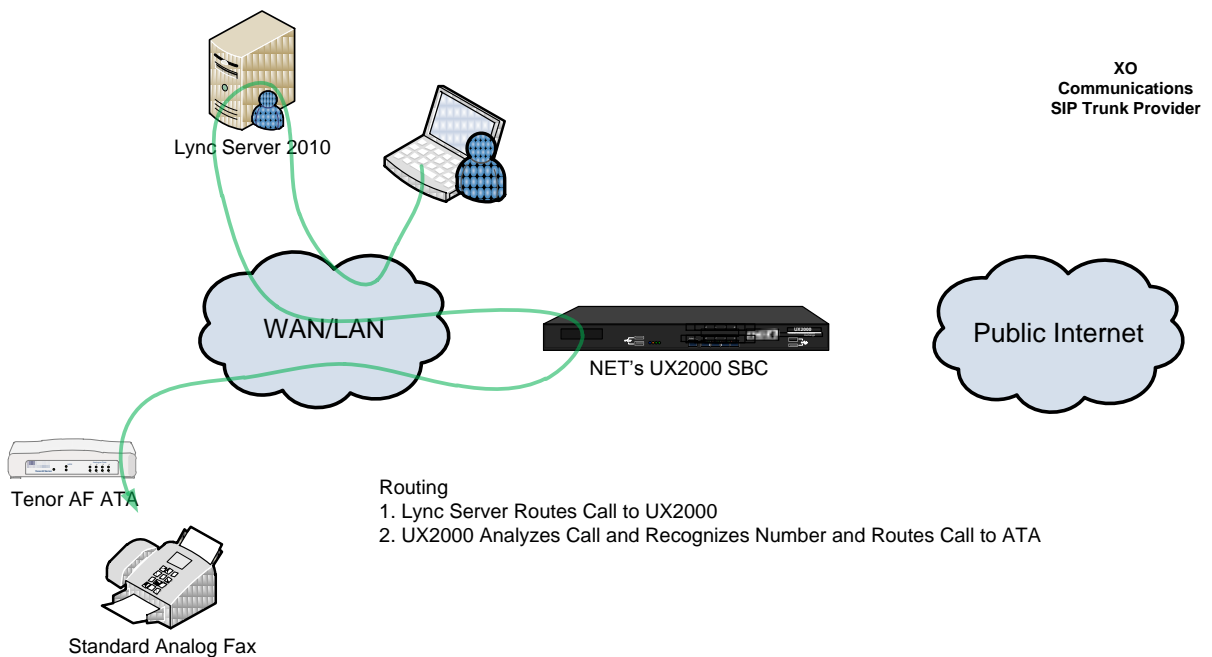


Figure 35 Lync Client to Analog Call Flow

Analog to Analog

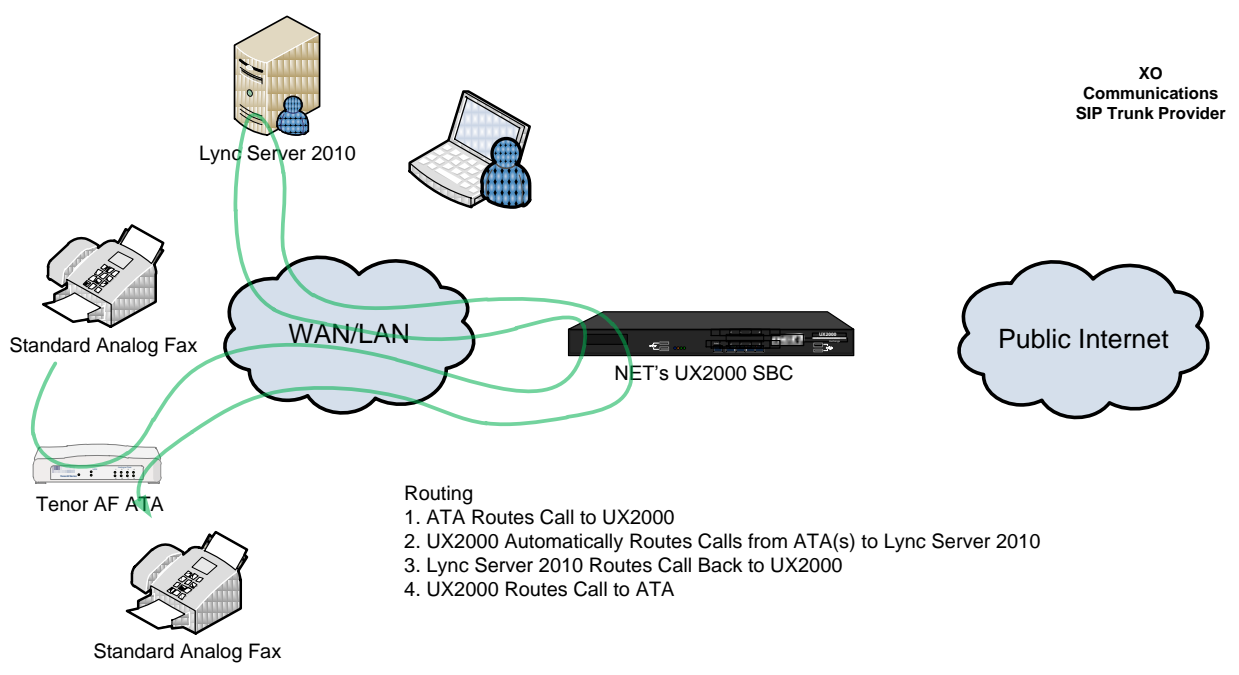


Figure 36 Analog to Analog Call Flow

Analog to PSTN

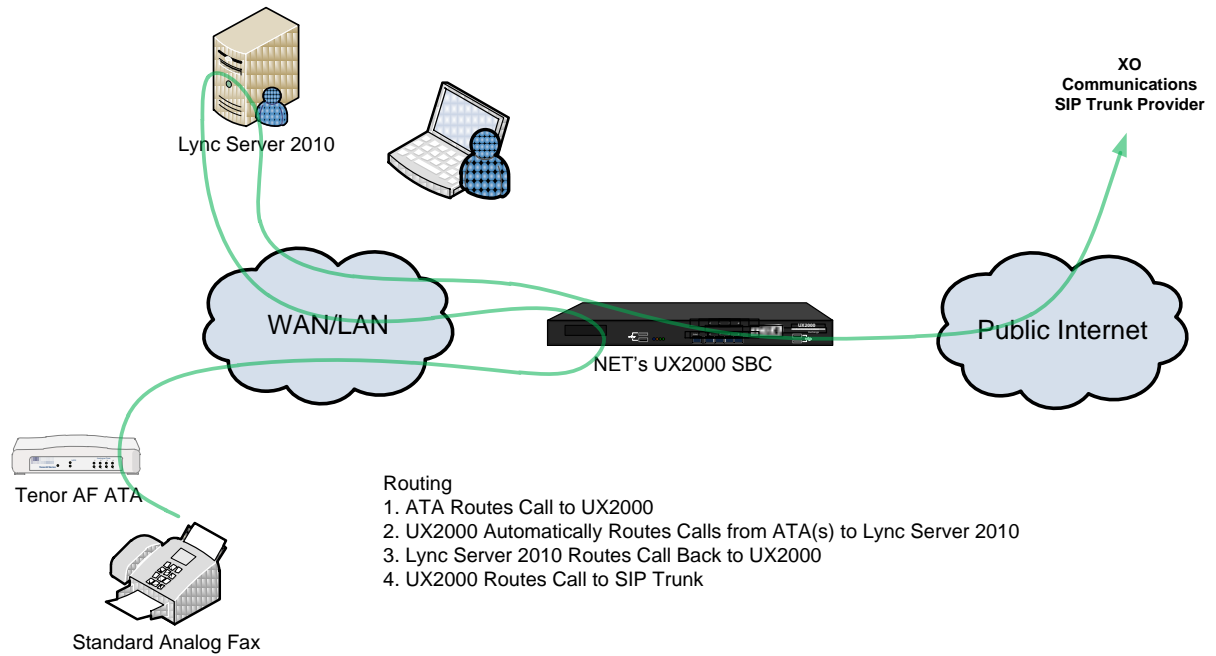


Figure 37 Analog to PSTN Call Flow

Analog to Lync Client

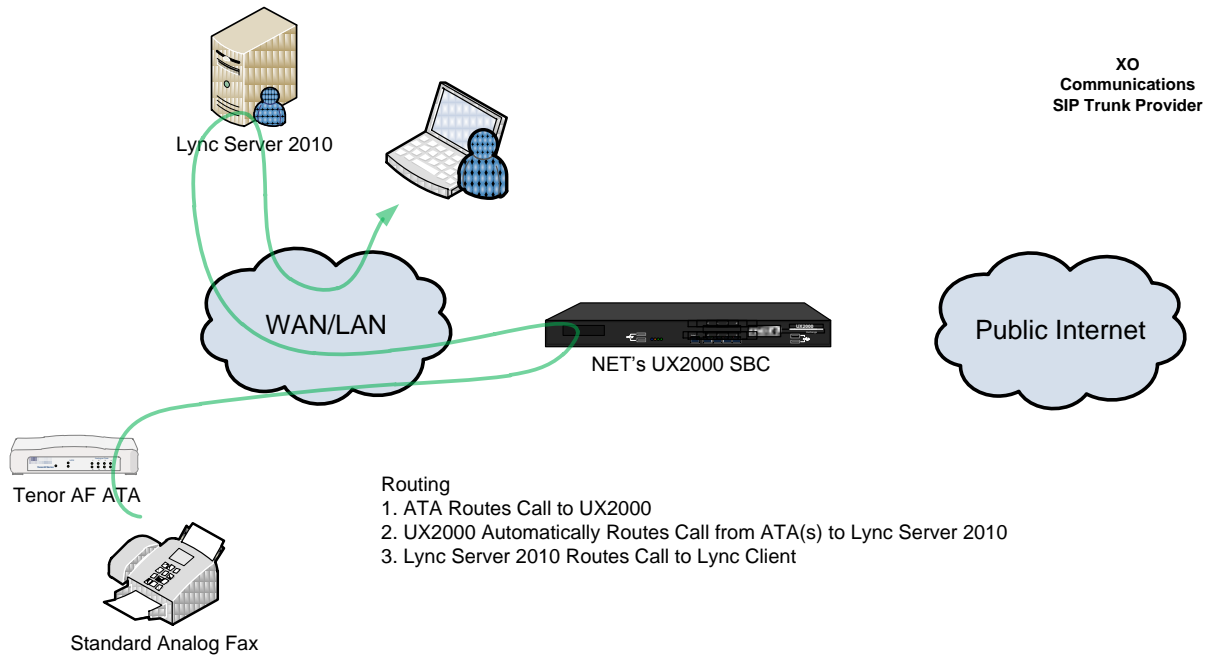


Figure 38 Analog to Lync Client Call Flow

9.2 Analog Configuration

To support Analog devices additional configuration will be required. For each ATA an additional SIP Server table and Signaling Group and Route table will be needed. The routing will be to send all calls from the Analog devices to the Lync Server 2010. Also since the UX2000 does not currently support Analog connections a separate ATA will be required.

9.2.1 SIP Server Table

The SIP Server table for the ATA is configured so that the UX2000 will know where to send analog calls to. The SIP Server table is configured under Settings/SIP/SIP Server Tables. Click on the green '+' to add the name of the ATA. In the example below an ATA with the name of 'Tenor 206' has been added:

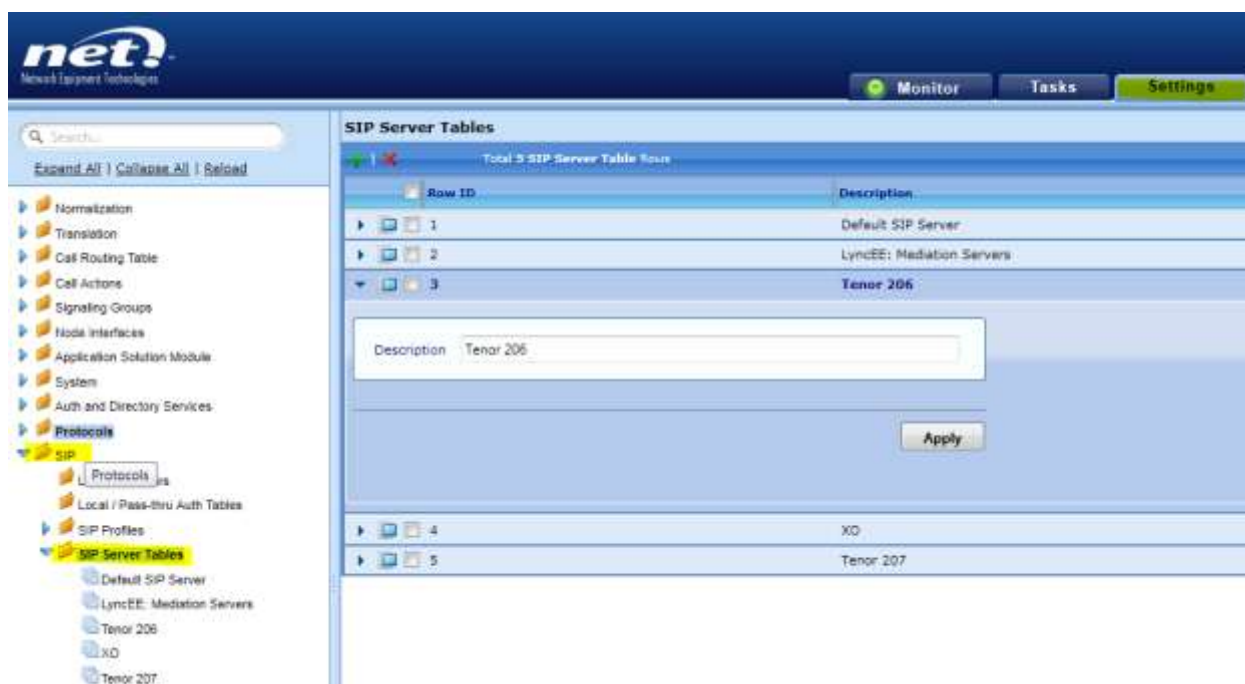


Figure 39 ATA SIP Server Table Name

Once the SIP Server Table Name has been created for the ATA, click on the name in the list on the left pane. From here click the right facing arrow (right arrow will turn down as dialog opens) next to the SIP Server name in the right hand pane to open the configuration dialog as shown below:

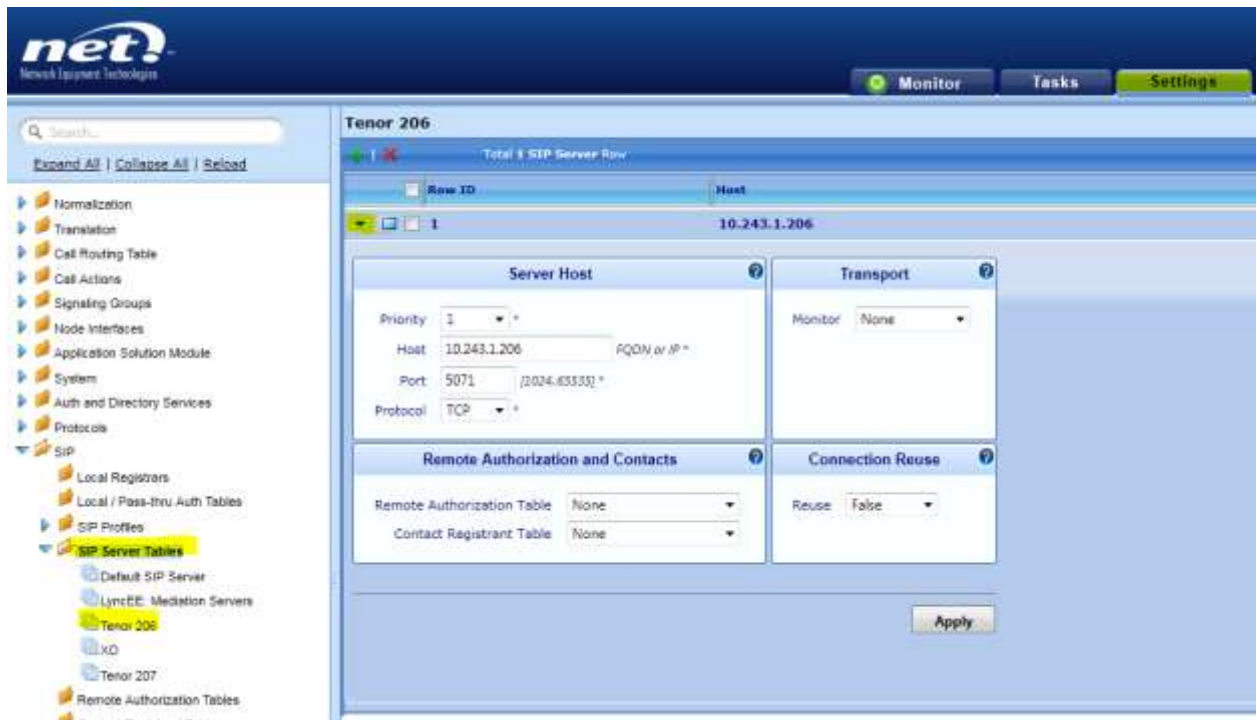


Figure 40 SIP Server Configuration Dialog

Provide the following information to complete the dialog:

- Priority: Leave default
- Host: FQDN or IP Address of ATA
- Port: SIP Signaling Listen Port of the ATA

Protocol: IP Protocol that will be used to reach the ATA

The rest of the settings can stay default for a basic install.

9.2.2 Analog Translation Table

Add a Translation table that will be used to route to an ATA with Settings/Translation Table. The example below is adding a Translation table entry for a DID number assigned to the Tenor 206 ATA:



Figure 41 Translation Table Name for ATA Routing

Click on the entry in the Translation table listing and then on the green '+' sign in the right hand pane to open the configuration dialog for this DID entry::

Enter the following information in the dialog:

Description: Enter a description so this will be recognized as the ATA route
 Input Field

Type: Select 'Called Address/Name' from the dropdown

Value: Enter the number to route to this ATA as

^+{(14255551234)}\$

Output Field

Type: Select 'Called Address/Name' from the dropdown

Value: Enter \1 to route the contents of the Input Field

When the dialog is populated it should look similar to below:

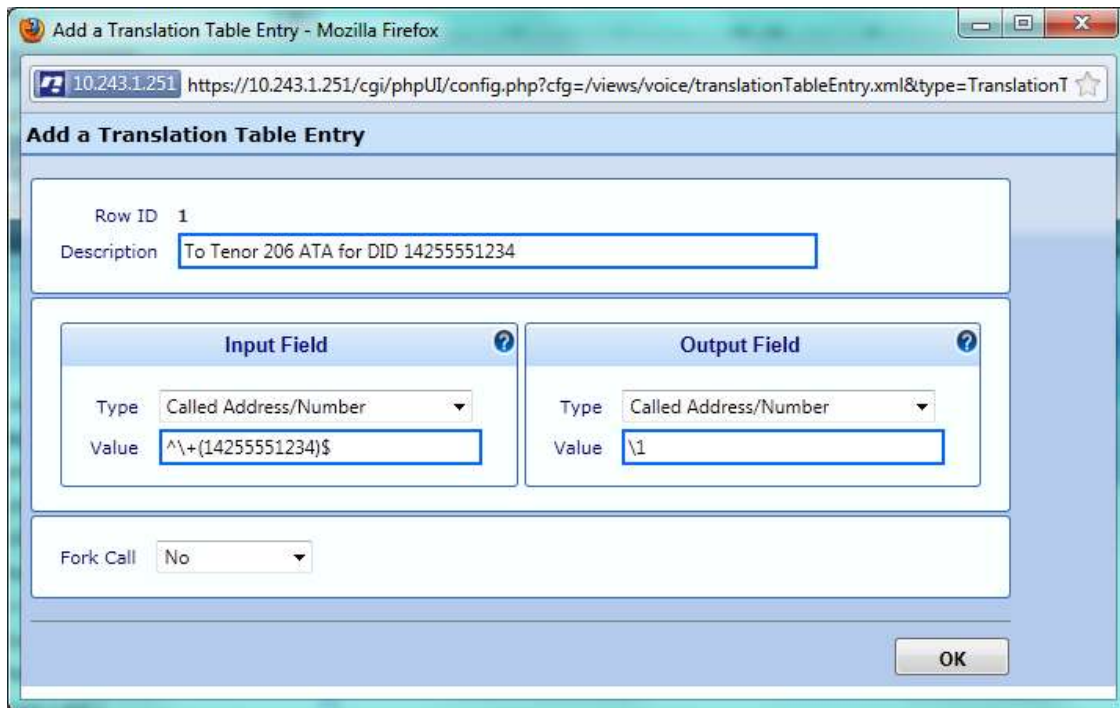


Figure 42 Completed Translation Table Dialog for ATA Routing

9.2.3 Signaling Group

The Signaling Group for the ATA is configured at Settings/Signaling Groups. Click on the 'Add SIP SG' link in the top of the right hand pane to add the name of the SIP SG. The example bellows shows a SG with the name of 'Tenor 206' with the basic configuration changes made from the default dialog:

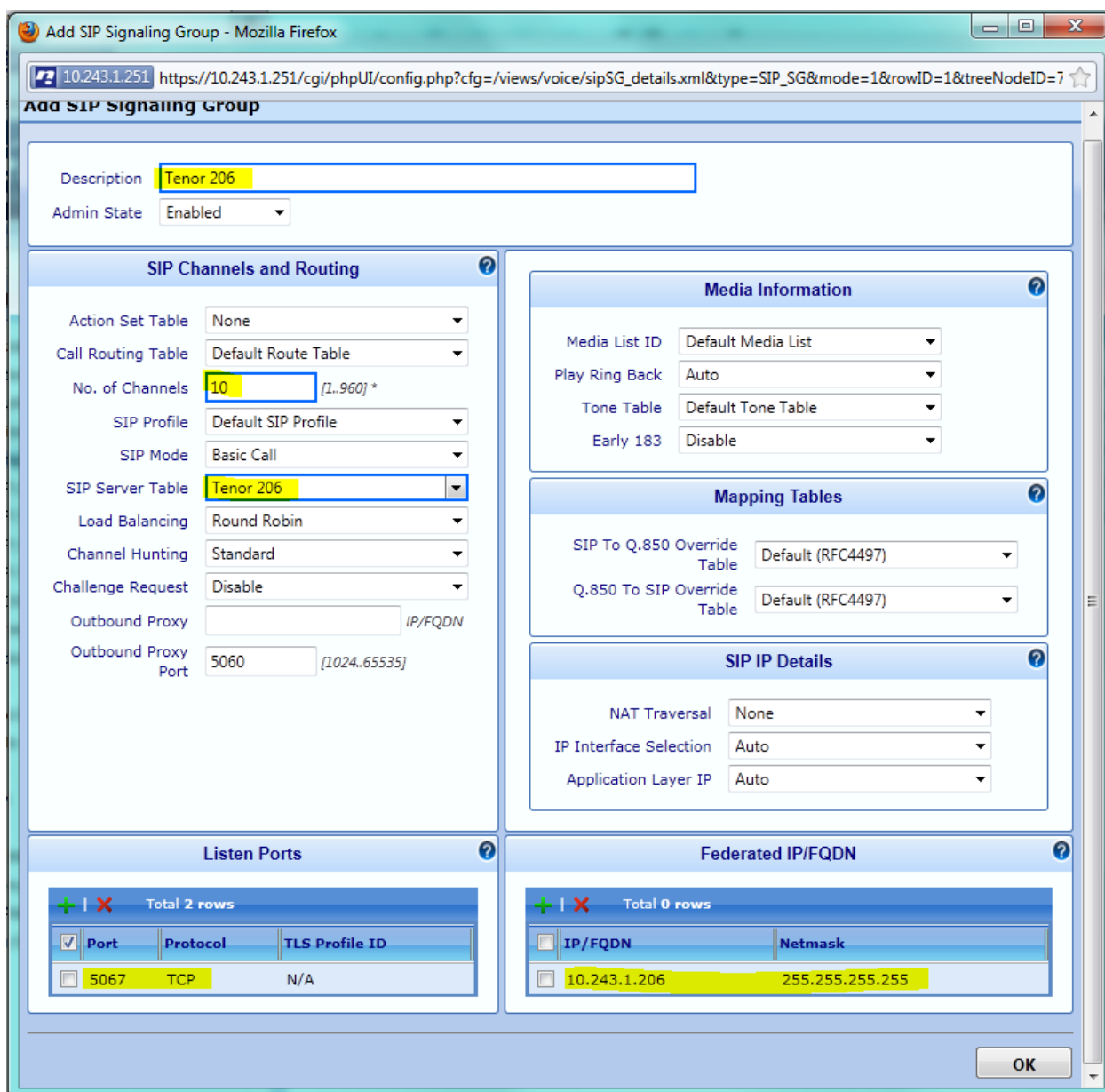


Figure 43 SIP SG Configuration Dialog

Description: Tenor 206

No. of Channels: This is related to the number of calls the ATA can support

SIP Server Table: Select from the dropdown the name of the SIP Server table configured in the previous step



Listen Ports: The port(s) that the SG will listen to for SIP Signaling from the ATA
Federated IP/FQDN: The IP Address of the ATA or Network Address of the ATA IP Network. This is used to ensure the SG listens only to the originating ATA for SIP signalling

Click 'OK' to complete the dialog.

9.2.4 Modify Lync Server 2010 Routing Table to Add ATA Route

With the SIP Server, Translation Table and SG configured, the Routing table that routes calls from Lync Server 2010 needs to be modified so that the calls for the ATA will be routed properly. This is done by adding the ATA routes into the table.

Open the Lync Server 2010 Routing table by clicking on Settings/Call Routing Table. Click on the Call Routing table for calls coming from Lync Server 2010, in this configuration it is the 'Lync Server 2010 Routing' entry. Once the table is opened click on the green '+' sign to add the route to the table. Configure the following to complete the dialog:

Description: Add a description to associate with this route
Number/Name Translation Table: Select the proper Translation Table from the dropdown, in this case select the Translation configured above
Destination Signaling Group: Click 'Add' and then select the SG for the ATA

The completed dialog is shown below:

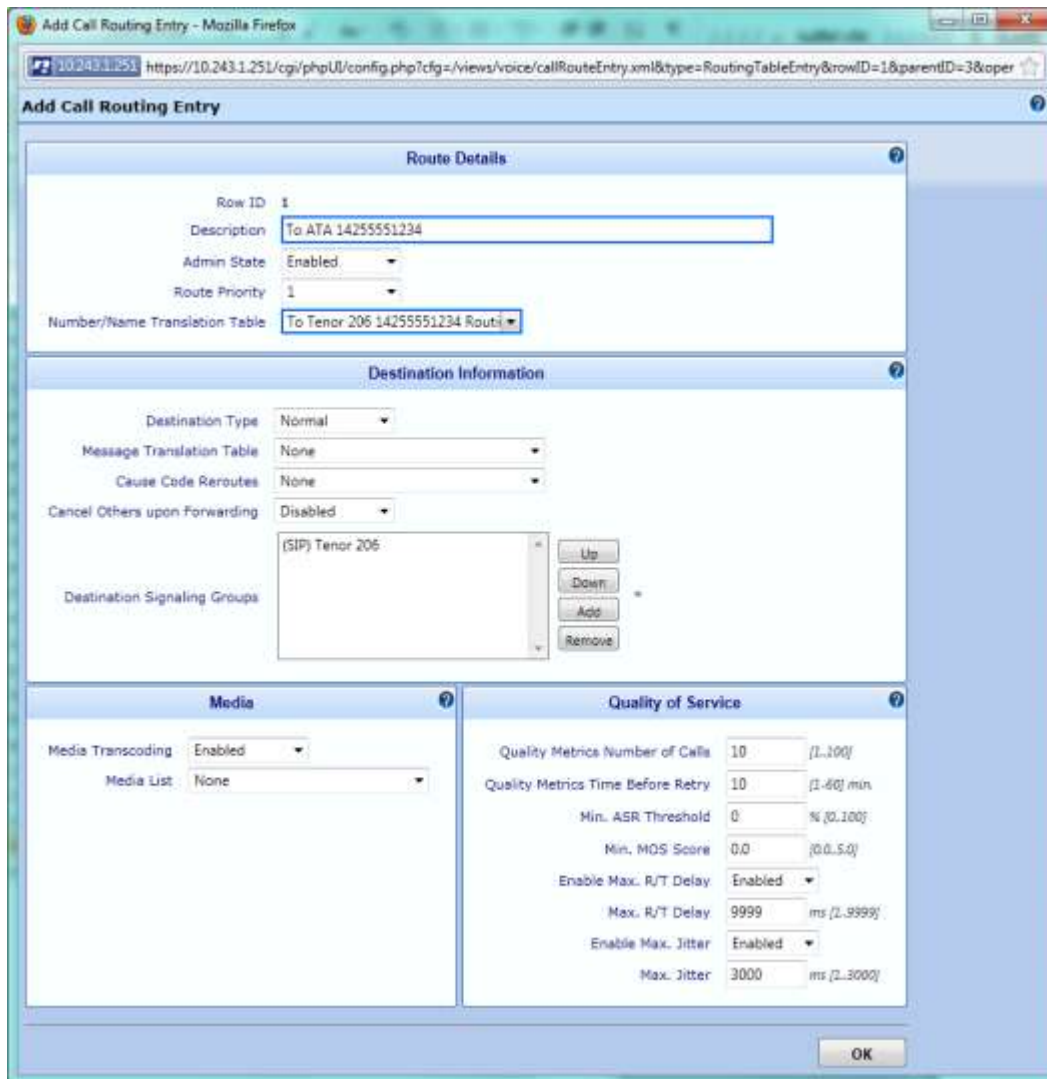


Figure 44 Adding ATA Route

Click 'OK' to complete the dialog and then click 'Resequence' at the top of the Route Table entries. In the dialog that opens click on the route for the ATA and then click 'Up' to move this route to the top of the table. This will ensure that the Route table will analyse the routes for the ATA first before the call is routed to the XO SIP Trunk:



Figure 45 Route Table Ordering for calls to ATA

9.2.5 Add Translation Table for ATA

All Calls from the ATA will have to be routed to Lync Server 2010. A Translation table may be required to match the default number format on the Lync Server 2010 (E.164). The Translation table below is used to add a '+' to the called number from the ATA:

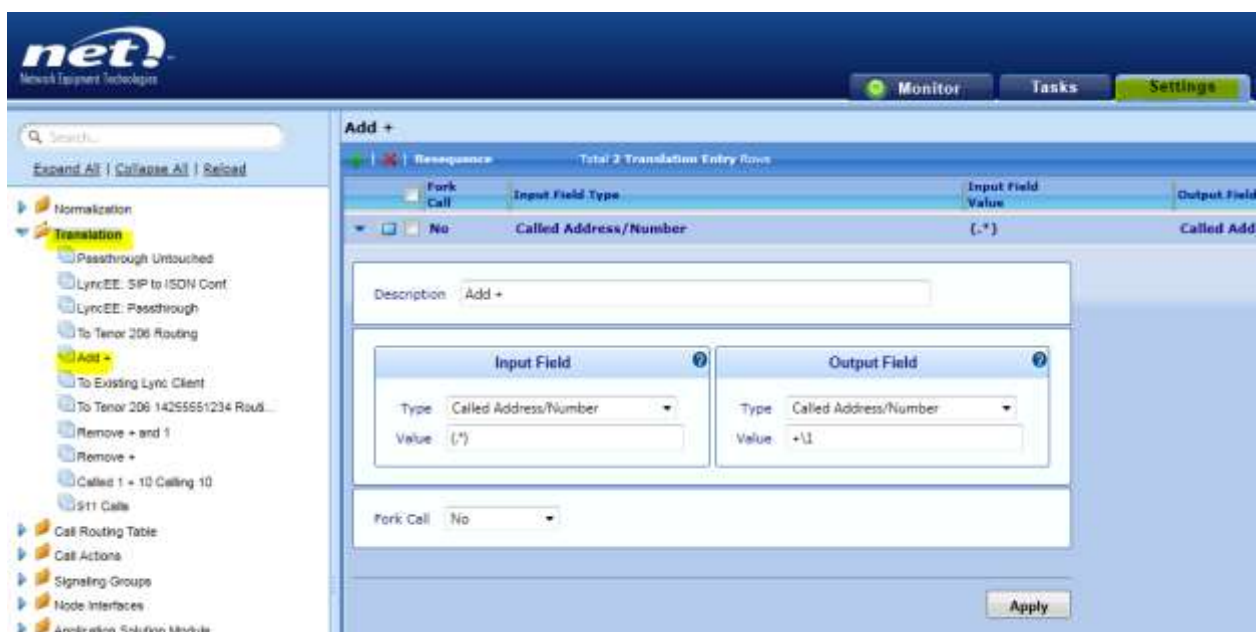


Figure 46 Translation Table to Add + to Called Number

9.2.6 Add Route Table for ATA

In order for calls to route from the ATA to Lync Server 2010 an ATA Route table needs to be added. This will allow the selection of the proper Translation table and route from the UX2000 to Lync Server 2010.

Click on Settings/Call Routing Table and then on the green '+' sign in the right hand pane. Enter a name for the route table dialog as shown below:

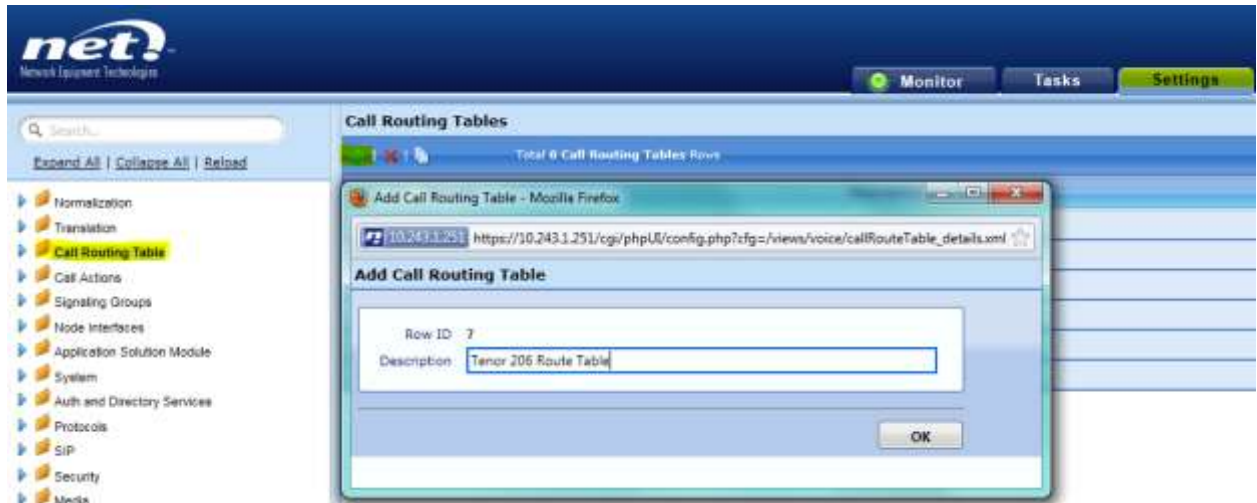


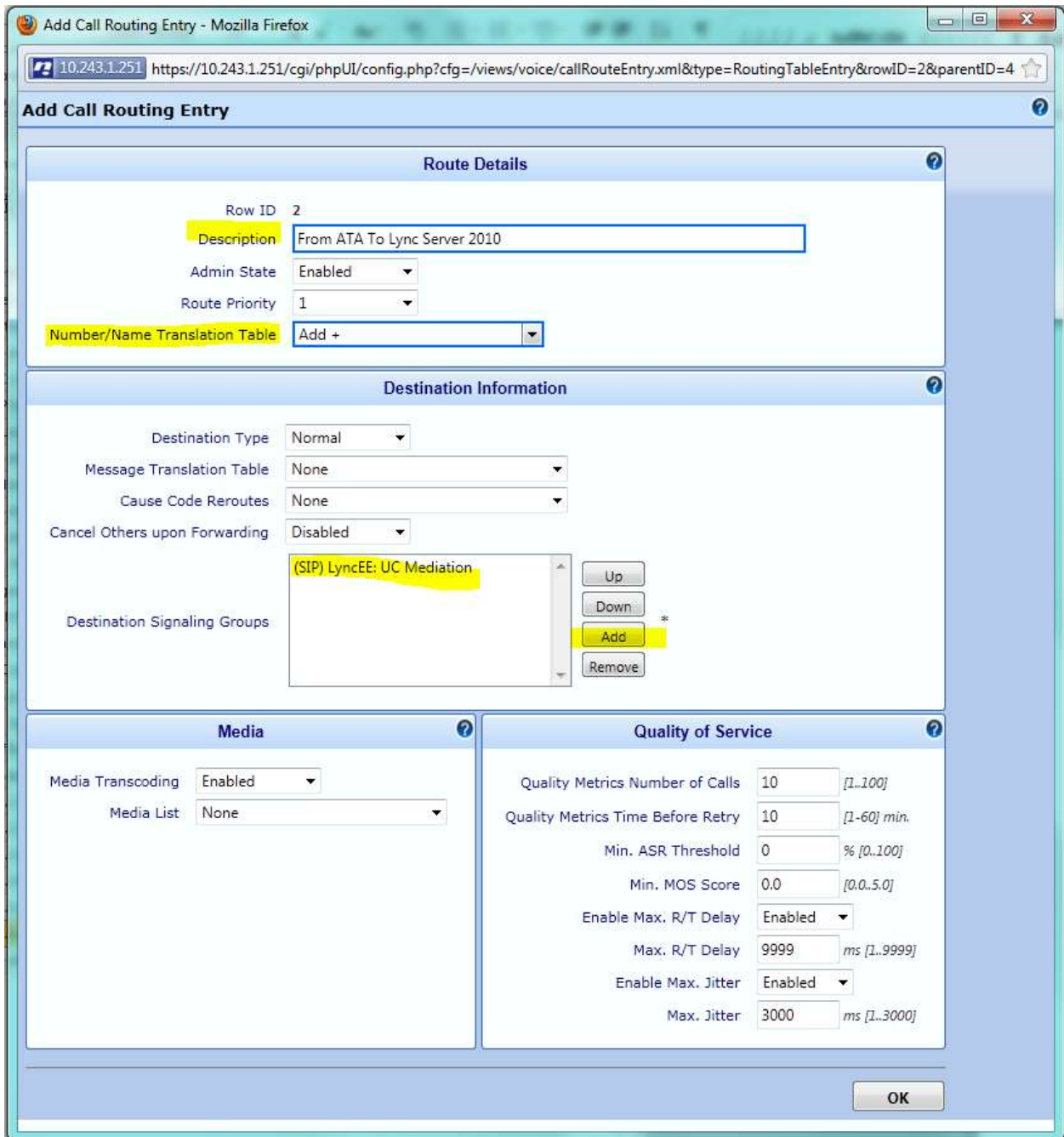
Figure 47 Creating a Route Table for the ATA

Click 'OK' to complete the dialog.

Once Route Table is added, complete the configuration dialog with the following information:

- Description: Enter description to associate with the route
- Number/Name Translation Table: Select the Translation Table from above
- Destination Signaling Group: Click 'Add' and select the SG for the Lync Server 2010 Server

The completed dialog is shown below:



Add Call Routing Entry

Row ID: 2

Description: From ATA To Lync Server 2010

Admin State: Enabled

Route Priority: 1

Number/Name Translation Table: Add +

Destination Information

Destination Type: Normal

Message Translation Table: None

Cause Code Reroutes: None

Cancel Others upon Forwarding: Disabled

Destination Signaling Groups: (SIP) LyncEE: UC Mediation

Up, Down, Add, Remove buttons

Media

Media Transcoding: Enabled

Media List: None

Quality of Service

Quality Metrics Number of Calls: 10 [1..100]

Quality Metrics Time Before Retry: 10 [1-60] min.

Min. ASR Threshold: 0 % [0..100]

Min. MOS Score: 0.0 [0.0..5.0]

Enable Max. R/T Delay: Enabled

Max. R/T Delay: 9999 ms [1..9999]

Enable Max. Jitter: Enabled

Max. Jitter: 3000 ms [1..3000]

OK

Figure 48 Routing Entry for Calls From ATA to Lync Server 2010

Click 'OK' to complete the dialog.

10 Troubleshooting

The UX2000 provides various ways to troubleshoot call issues. The first is through visual analysis of configured systems and active and certain archived alarms through the Monitor page. And the UX2000 can generate debug logs to be viewed locally, downloaded to a text editor or the debug logs can be sent to a Syslog collector.

10.1 Visual Analysis

Clicking on the 'Monitor' tab will bring up a real time view of the current status of the configured SGs and also a real time view of call status. Also, the current alarms will be shown in the 'Alarm Window' at the bottom of the screen and the highest unacknowledged alarm color will show in the 'Monitor' tab.

10.1.1 Current SG Status and Call Activity

The top portion of the 'Monitor' display will show the current status of the SGs and their associated channels. If a SG is down the SG will be colored Red as shown below for the XO SG:

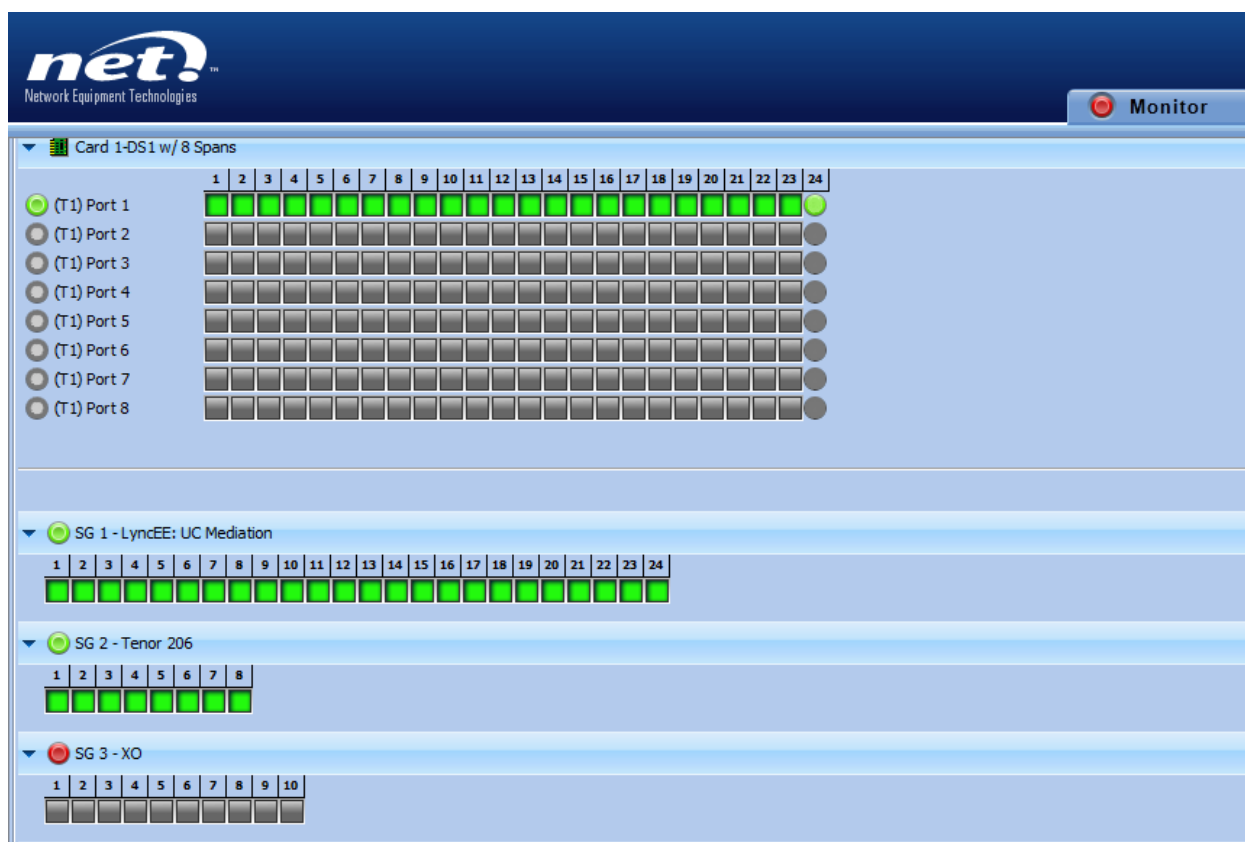
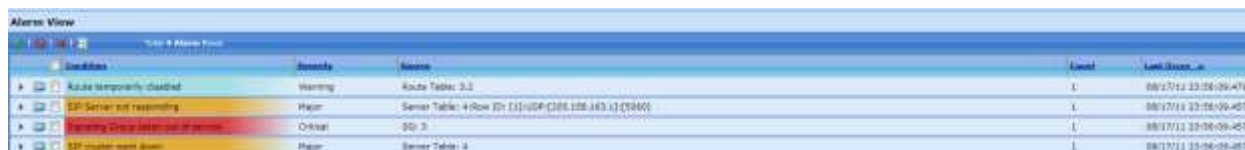


Figure 49 XO SIP Trunk Down

There are many reasons a SG may be down. The far end equipment may be off line, there could be a routing issue or there could be a configuration problem. The visual Monitor will provide a quick view on where to start trouble shooting.

10.1.2 Current Alarms

The current alarms will also indicate problems. The alarms shown below posted as the XO SG went out of service:



Severity	Message	Source	Level	Last Occur. in
Warning	Route temporarily disabled	Route Table: 3.2	L	08/17/12 23:08:09-478
Major	SIP Server not responding	Server Table: 4 Row ID: [1]UDP(108.183.13)SRV0	L	08/17/12 23:08:09-457
Minor	Outgoing traffic interrupted by network	SIP: 3	L	08/17/12 23:08:09-457
Major	SIP cluster went down	Server Table: 4	L	08/17/12 23:08:09-457

Figure 50 Alarm Status

10.2 Viewing Debug Logs

The UX2000 will continually record the last hour and 15 minutes of debug logs. These are split into 15 minute logs and can be viewed by clicking on the 'Logs' tab as shown below:



Log File Name	First Message Time	Last Message Time	Size	Action
netlog	2012-08-17 23:00:00	Current File	602.84 kilobytes	Download / Delete
private_log_20120817-234501	2012-08-17 23:45:01	2012-08-17 23:45:01	3.70 Megabytes	Download
private_log_20120817-234001	2012-08-17 23:40:01	2012-08-17 23:40:01	1.31 Megabytes	Download
private_log_20120817-233501	2012-08-17 23:35:01	2012-08-17 23:35:01	1.80 Megabytes	Download
private_log_20120817-233001	2012-08-17 23:30:01	2012-08-17 23:30:01	2.55 Megabytes	Download
private_log_20120817-232501	2012-08-17 23:25:01	2012-08-17 23:25:01	1.30 Megabytes	Download
private_log_20120817-232001	2012-08-17 23:20:01	2012-08-17 23:20:01	1.38 Megabytes	Download

Figure 51 Debug Logs

The logs are ordered with the current log being written on top. When the current log reaches the end of the recording period it will be rotated down from the top and a new log will start to write.

To view one of these logs click the blue box to the left of the file name. This will open a web based text box that can be scrolled through to find certain debug information.

To download the log to either save or open in a text editor, click on the 'Download' link to the right of the file information. This will open a dialog to either save to a file or open.

10.3 Sending Debug Logs to a Syslog Collector

To send debug information to a Syslog server, click 'Settings/Logging Configuration/Remote Log Servers'. Click on the green '+' sign to add a Syslog collector. Complete the dialog with the following information:

Global Log Level: Set to the level required

Log Destination: Enter the FQDN or IP Address of the Syslog Collector

Click 'OK' to complete the dialog. This will start sending the debug log information to the Syslog collector. Different levels of collection may be required to show applicable trouble shooting information

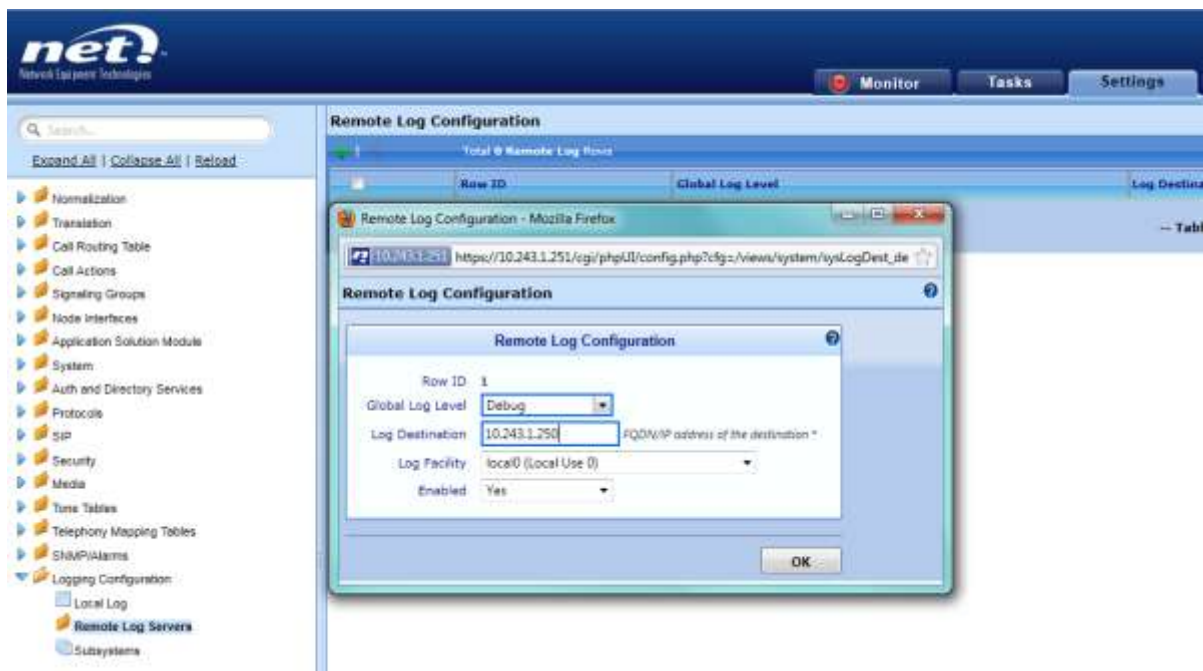


Figure 52 Syslog Collector Configuration