



Patton SN4960 and Elastix Server

Setup Guide

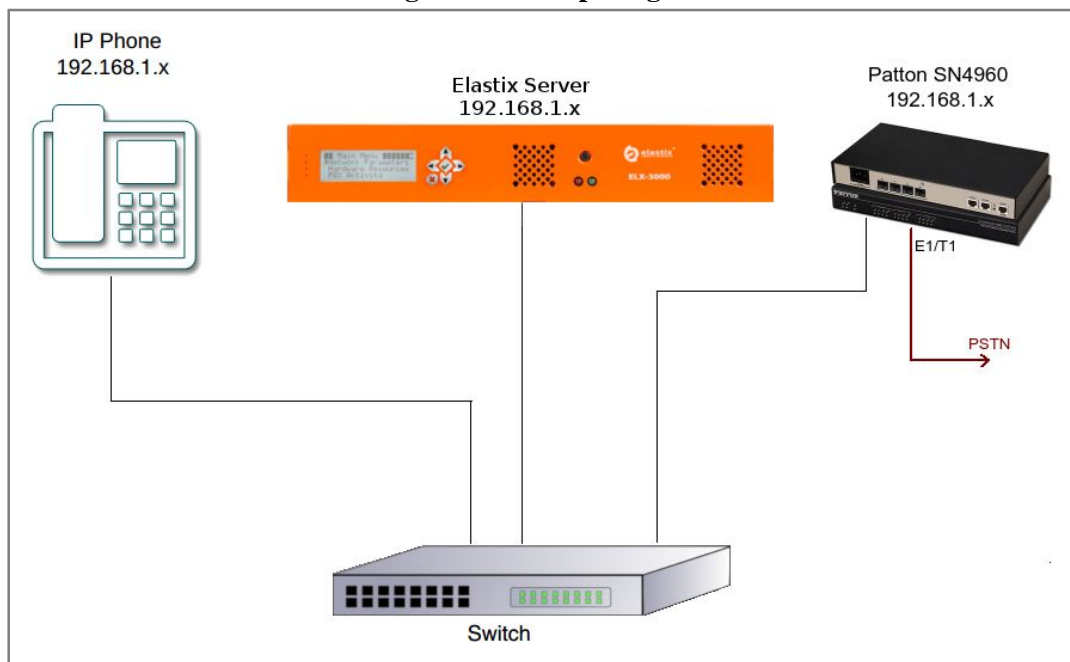




1.0 Setup Diagram

Figure 1-1 is a setup diagram for a single Patton SN4960 gateway configuration. We're going to configure a SIP Trunk for communication between the Analog Phone and IP Phone.

Figure 1-1. Setup Diagram



2.0 Host PC Environment

Table 2-1. Host Server Environment Details

	Description
Hardware Type	Elastix Appliance ELX-Series
Hardware Version	ELX-3000
Software Type	Elastix
Software Version	2.3

3.0 Test Setup Equipment

Table 3-1. Test Setup Equipment

Equipment	Model	Version
IP (SIP) Phone	N/A	N/A
Analog Phone	N/A	N/A
Patton	SN4960	FW: R6.2 (HW: 4.1)
Switch	N/A	N/A



4.0 Setup Procedure

To set up the Elastix Server for the Patton SN4960

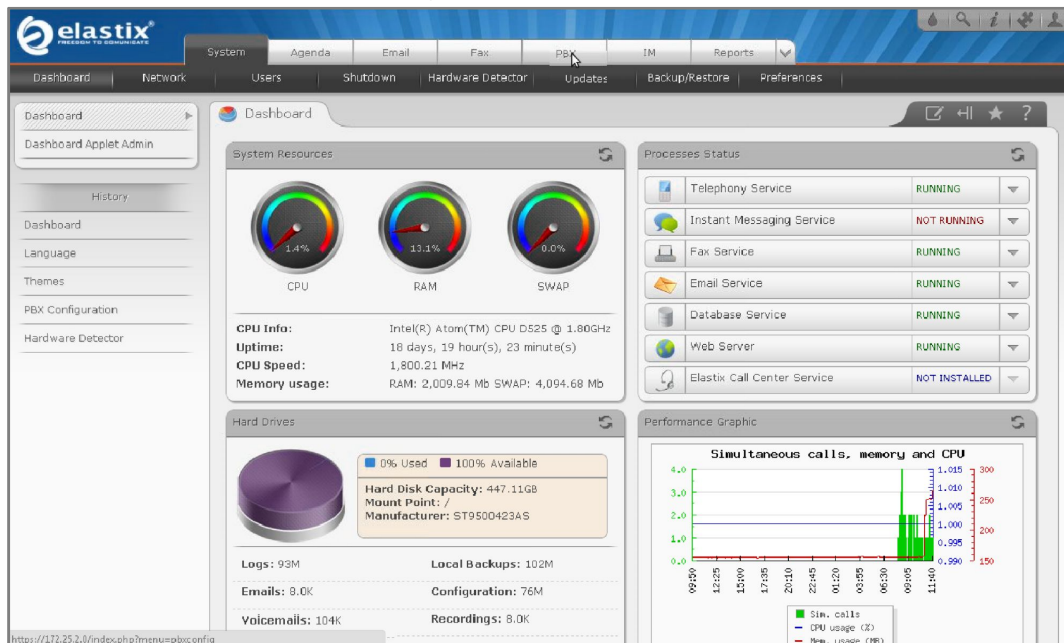
1. Go to the web address of the Elastix Server Login page. The web address is determined by the customer, for this guide we have used the IP address 192.168.1.75
2. On the Login page, type the username and password for an administrative user into the Username and Password fields, see Figure 4-1. The username and password are determined by the customer.

Figure 4-1. Login



3. Press Enter or click on the Submit button to go to Elastix's Dashboard
4. Once inside, click on the PBX tab on the menu at the top of the screen

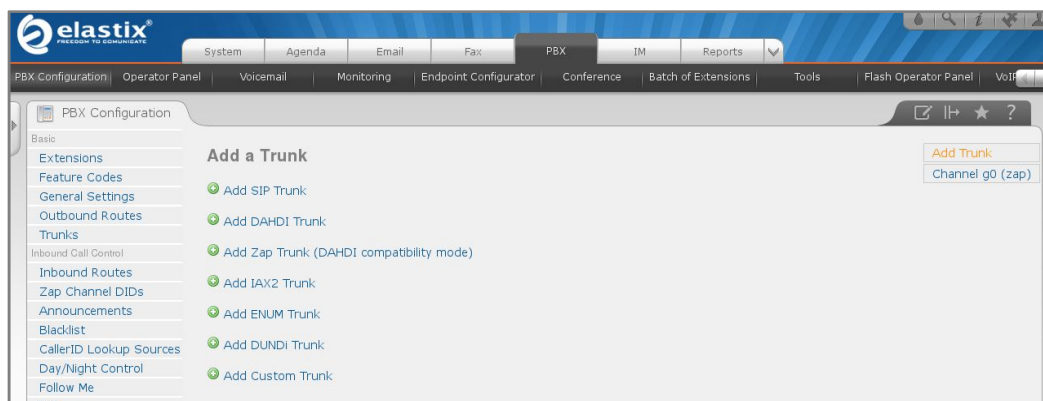
Figure 4-2. Dashboard



5. Go to "PBX Configuration => Trunks => Add SIP Trunk", see Figure 4-3. This will take you to configure a SIP Trunk.



Figure 4-3. Add a SIP Trunk



6. On the “Add SIP Trunk” page (Figure 4-4), fill in the following information:

General Settings

- **Trunk Name:** (PattonSN4960Trunk in this example)

Outgoing Settings

- **Trunk Name:** (Patton4960 in this example)
- **Peer Details:**
 - host=dynamic
 - username=(Patton4960 in this example)
 - secret=(jx8FkOU13sv6 in this example)
 - qualify=yes
 - type=peer
 - insecure=very

Figure 4-4. Add SIP Trunk

Add SIP Trunk

General Settings

Trunk Name:

Outbound Caller ID:

Outgoing Settings

Trunk Name:

PEER Details:

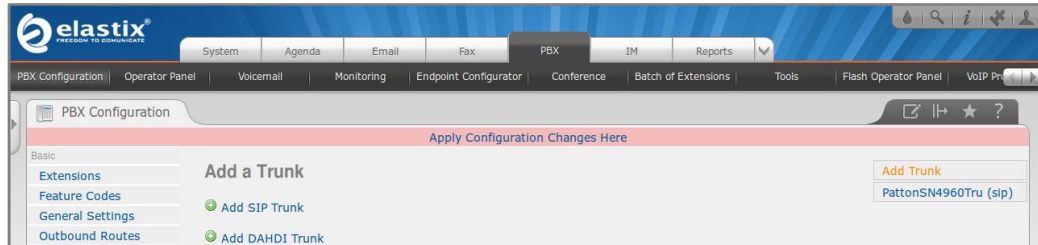
```
host=dynamic
username=Patton4960
secret=jx8FkOU13sv6
qualify=yes
type=peer
insecure=very
```



7. Click on the ‘Submit’ button at the end of the page. The SIP Trunk will be created and you will see the page on Figure 4-5 displaying the “Apply Configuration Changes Here” pink ribbon on top of the screen.

8. Click in the “Apply Configuration Changes Here” link

Figure 4-5. Apply Configuration Changes Here



9. With this you have finished creating a SIP Trunk that will be used by the Patton SN4960 to register with the Elastix Server. Now, go to “PBX => PBX Configuration => Outbound Routes” to configure the outbound route to the Patton SN4960 Gateway. Fill in the following information: (Figure 4-6)

Route Settings

- **Route Name:** (“7_Patton” in this example)

Dial patterns

- **Prefix:** (“7” in this example) | **Match pattern:** (“.” in this example)

Trunk Sequence for Matched Routes

- **0:** (“PattonSN4960Trunk” in this example)

Figure 4-6. Add Route

Add Route

Route Settings

Route Name:

Route CID: ☐ Override Extension

Dial Patterns that will use this Route

(prepend) + | / CallerId

+ Add More Dial Pattern Fields

Dial patterns wizards: (pick one)

Trunk Sequence for Matched Routes

0

1

2

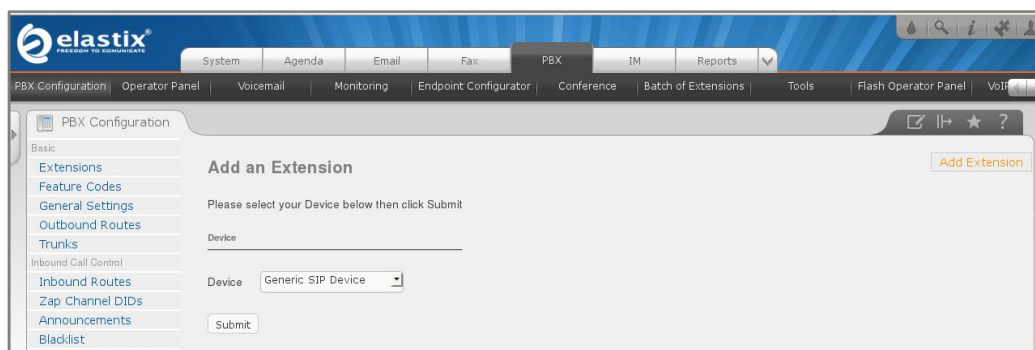
Submit Changes

10. Click on “Submit” at the end of the page and Apply changes. Once we have configured the main parameters for communication to Patton SN4960 in Elastix, we’ll create an extension for an IP Phone. Go to “PBX => PBX Configuration =>



Extensions” and click on “Submit” having the “Generic SIP Device” option selected. (Figure 4-7)

Figure 4-7. Add SIP Extension



11. Fill in the following information on the Add SIP Extension page:

- **User Extension** (302 in this example)
- **Display Name** ('IPPhone' in this example)
- **secret** ('f7pkg0Wf2nt1s' in this example)

Figure 4-8. Add SIP Extension

Add SIP Extension

Add Extension

User Extension	302
Display Name	IPPhone
CID Num Alias	
SIP Alias	

Device Options

This device uses sip technology.

secret	f7pkg0Wf2nt1s
dtmfmode	rfc2833

12. Click on “Submit” at the end of the page and Apply changes. To register the Patton SN4960 with Elastix, you will need to enter the information from the trunk created on the Elastix Server into the Patton gateway by logging into the WebUI.

Go to the Patton SN4960’s WebUI by pointing your browser to the Patton’s IP address (Figure 4-9).



Figure 4-9. Patton SN4960's WebUI



For the initial configuration, refer to the Patton SN4960 User Manual found at: <http://www.patton.com/manuals/SN4960.pdf>

Factory default settings

ETH0 IP Address	DHCP
ETH1 IP Address	192.168.1.1/24
Web Access Administrator User	administrator
Web Access Administrator Password	(none)

13. When the WebUI is loaded, go to “Telephony => SIP” located on the left side of the page and go to “Authentication Services” tab. Write a name for your authentication service and click on “Add authentication service” to add it into the list. (“My_Authentication” in this example). Figure 4-10.

Figure 4-10. Authentication services



14. Once you have added the authentication service, click on the just created name to go to its properties and fill in the following information related to the previous step 6. When finish click on “Add user” (Figure 4-11).

Authentication Users

- **User Name:** (“Patton4960” in this example)
- **Password:** (“jx8FkOU13sv6” in this example)



Figure 4-11. Authentication Service Configuration

15. Now go to “Telephony => SIP => Location Services”. Write a name for your location service and click on “Add location service” to add it into the list (“My_Location” in this example. Figure 4-12).

Figure 4-12. Location Services

16. Once you have added the location service, click on the just created name to go to its properties and fill in the following information. (Figure 4-13).

Domain

- **Name:** Elastix Server’s IP Address (“192.168.1.75 in this example”)

Identities

- **Name:** Name of the SIP Trunk (“Patton4960” in this example)

Figure 4-13. Location Service Configuration



17. Click on the name of the just created identity to access to its properties, and add the following information (Figure 4-14):

Authentication Outbound

- **Authentication Services:** Previously created authentication service (“My_Authentication” in this example).
- **Username:** Username added to the authentication service (“Patton4960” in this example).

Registration Outbound

- **Registrar:** Elastix Server’s IP Address (“192.168.1.75” in this example).
- **Port:** 5060
- **Registrar Mode:** Enable (Checked)

Figure 4-14.Configuration

The screenshot shows the Elastix configuration interface. On the left is a sidebar menu with options: Ethernet, E1/T1, Various, System, AAA, Time, Reports, Syslog, Save, Reload, About, and License. The main area is titled 'Authentication Outbound' and contains a table with columns: Authenticate, Position, Authentication Services, and Username. The table has three rows: 'authenticate' with position 1 and service 'My_Authentication' (username 'Patton4960'), 'authenticate' with a dropdown for 'My_Authentication' and a dropdown for 'Patton4960', and 'none' with a description 'Disable authentication, even from base identity-group'. Below this is the 'Authentication Inbound' section. The 'Registration Outbound' section is also visible, with fields for Registrar (192.168.1.75), Host (5060), Port, Lifetime (3600), Register Mode (checked), Preferred Transport Protocol (tcp), System Error Retry Timeout (10), Client Error Retry Timeout (10), Server Error Retry Timeout (10), NAT-traversal (unchecked), Dns Security Check (unchecked), Force destination (unchecked), and Priority (in promille). An 'Apply' button is at the bottom right.

18. Now, go to “Telephony => SIP => Gateways”. Write a name to the SIP gateway and click on “Create a SIP gateway” to add it into the list (“My_Gateway” in this example”. Figure 4-15).

Figure 4-15. SIP Gateway

The screenshot shows the Elastix configuration interface for creating a SIP gateway. The top bar indicates the current location: '192.168.1.75 / Telephony / SIP'. The left sidebar menu is the same as in Figure 4-14. The main area has tabs for 'Gateways', 'Interfaces', 'Location Services', 'Authentication Services', 'Profiles', and 'General'. The 'Gateways' tab is active, showing a form with fields for 'Name' (containing 'My_Gateway') and 'State'. A 'Create SIP gateway' button is at the bottom right.

19. Click on the name of the just created SIP gateway to access to its settings, and add the following information (Figure 4-16):



SIP Gateway

- Enable (Click on Apply)

SIP Interface

- **Name:** A name for gateway interface (“GW_SIP” in this example) (Click on apply to enable the next options).
- **Bind:** Checked
- **IP Interface:** (WAN in this example)
- **Port:** 5060

Location Services Bound

- (“My_Location” in this example)

Figure 4-16

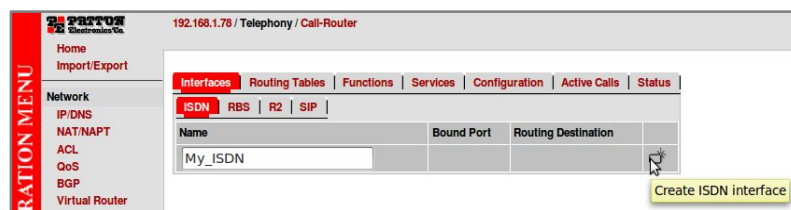
20. To check if the Patton SN4960 is successfully registered, go to “Status” tab on the same page of the previous step. You should see “Registered” on the State field of SIP Registration (Figure 4-17).

Figure 4-17. Status

21. Once it’s registered, we’ll add the interfaces to get it ready for calls. Go to “Telephony => Call-Router”. Go to “Interfaces => ISDN” tab and write a name for an ISDN interface (“My_ISDN” in this example). Click on “Create ISDN interface” when finish. (Figure 4-18)



Figure 4-18. ISDN Interface



22. Now we have to enable the E1 Port in the gateway. For this, in the main menu, go to “Ports => E1/T1” (Figure 4-19). You have to choose the port you want to enable. In this example we only have one. Click on the port and it’ll take you to a configuration page.

Figure 4-19. Ports



23. In the configuration page set the following (Figure 4-20):

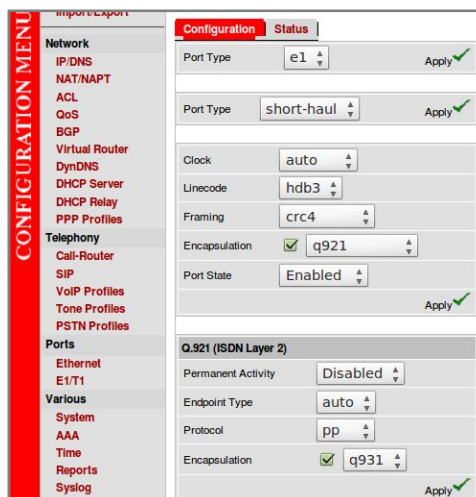
- **Port type:** E1
- **Clock:** auto
- **Encapsulation:** q921 (Checked)
- **Port State:** Enable

Apply changes. Now set the following:

- **Bind:** Checked (Select the created ISDN Interface. “My_ISDN” in this example)

Note: All this values may vary depending on your ISDN provider.

Figure 4-20. E1 Configuration





24. Apply changes when finish. Now we'll create a SIP interface. Go to "Telephony => Call-Router". Next select "SIP" located on "Interface" tab. Write a name of SIP interface for inbound calls and click on "Create SIP interface" (Figure 4-21).

Figure 4-21. SIP Interfaces

25. Once added the interface click on it to go to the configuration page. Set the following information (Figure 4-22):

- **SIP Gateway:** Checked (Select the name of the gateway created on step 19)
- **Remote User Agent Host Name / Port:** Checked. Elastix Server's IP Address, and port ("192.168.1.75" "5060" in this example)

Figure 4-22. SIP Interface configuration

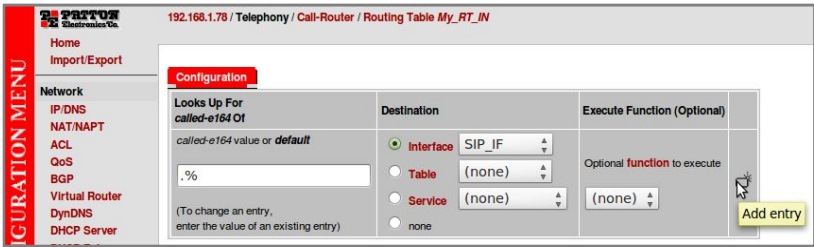
26. Apply changes clicking on the link at the end of the page. Now create a routing table for calls from ISDN port to SIP interface. Go to "Telephony => Call-Router". Select "Routing Tables" tab and write a name for the route ("My_RT_IN" in this example). In the "Looks up for" field select "called-e164" option (Figure 4-23).

Figure 4-23. Routing tables

27. Once created the route, click on it and it'll take you to a configuration page. Set the following (Figure 4-24):

- Looks up for:** Here you have to enter the dialplan (“.%” in this example that means any character repeated zero or more times).
- Destination:** Interface. Select outbound SIP interface created on step 27. (“SIP_IF” in this example).

Figure 4-24. Configuration of routing table



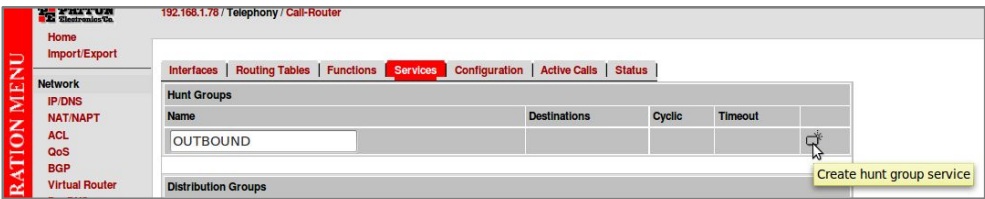
28. Add the entry by clicking on the link located on the right. Go back and create another Routing table (“My_RT_OUT” in this example). See figure 4-25.

Figure 4-25. Routing tables



29. Now go to “Telephony => Call-Router => Services” and add a *Hunt Group*. See figure 4-26.

Figure 4-26. Services

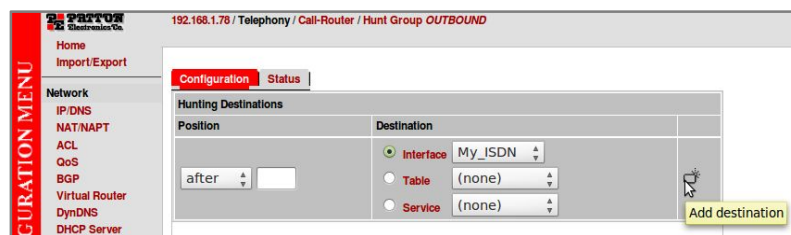


30. Then click on the created group to go to the configuration page. Set the following (figure 4-27).

- Destination:** Interface. Select ISDN interface created on step 21. (“My_ISDN” in this example).



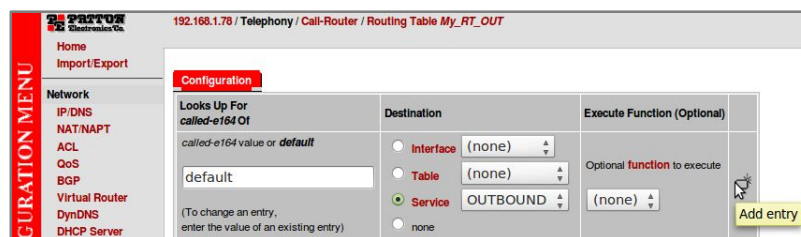
Figure 4-27. Hunt Groups



31. Now go back to “Telephony => Call Router => Routing Tables” and click on the second Routing table we created (“My_RT_OUT”). Set the following parameters (Figure 4-28).

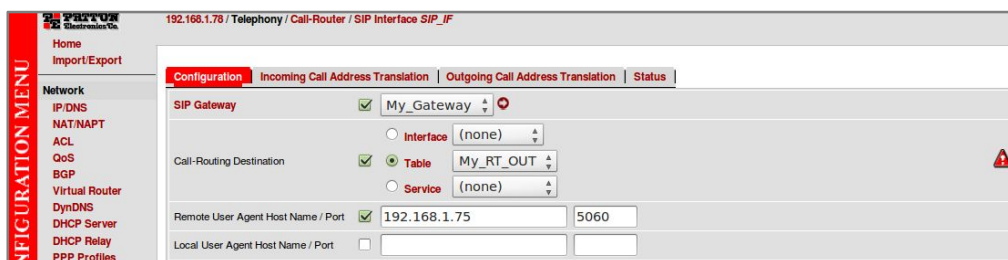
- **Looks up for:** default.
- **Destination:** Service. Select the Hunt Group created on step 31. (“OUTBOUND” in this example).

Figure 4-28. Configuration Routing table



32. Click on Add entry. Now go to “Telephony => Call-Router => Interfaces => SIP” and click on the SIP interface to modify it. Set the Call-Routing Destination with the routing table that use the hunt group . (Figure 4-29)

Figure 4-29. SIP Interface



33. Click on Apply link at the end of the page. Now we have to configure the ISDN interface we created in step 21 to use the routing table. Go to “Telephony => Call-Router”. Select “ISDN” from “Interfaces” tab (Figure 4-30).

Figure 4-30. ISDN Interfaces



34. Click on the interface to go to the configuration page. Set the following (Figure 4-31):
- Call-Routing Destination:** Checked. Select “Table” and choose the table created on step 26. (“My_RT_IN” in this example).

Figure 4-31



35. Apply changes by clicking the link at the end of the page.
36. Now go to “Telephony => Call-router => Configuration” and set the national and international prefix with a point “.” (Figure 4-32):

Figure 4-32. Call router - Configuration

E.164 Number Prefixes		
National Prefix	<input checked="" type="checkbox"/>	<input type="text" value="."/>
International Prefix	<input checked="" type="checkbox"/>	<input type="text" value="."/>

36. We have finished setting the Patton SN4960 Gateway. Configure the other IP (SIP) Phone with the correct parameters.
37. To test the Patton SN4960, pick up the previously configured IP (SIP) Phone and call to any number through the PSTN with the prefix 7. The call should be established.