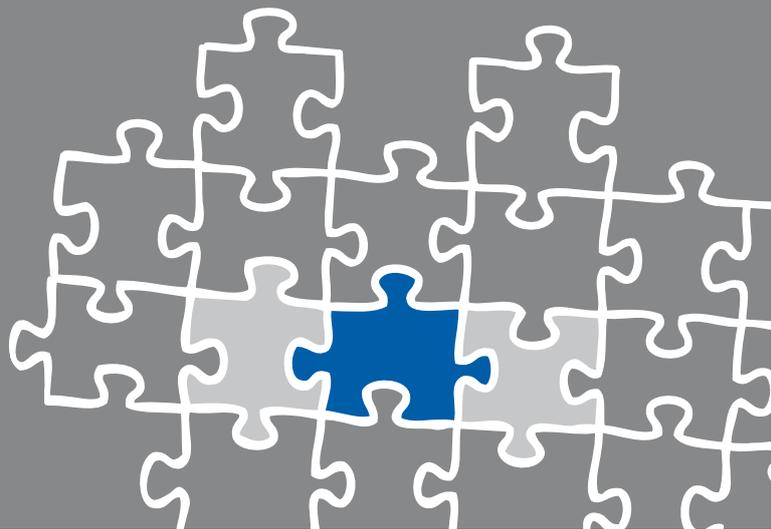


# TechNote

## Patton SmartNode Series

August 20, 2010





## Introduction

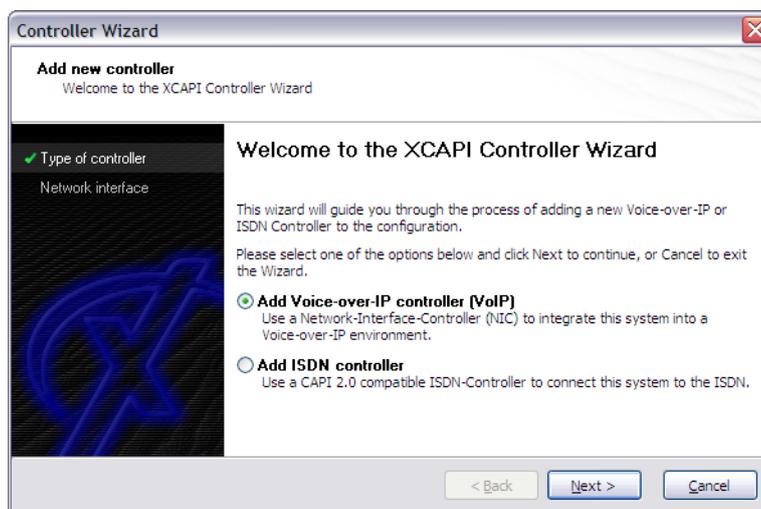
This document is intended to support you with the integration of the XCAPI, Version [3.3.161](#), into an existing environment of the Patton SmartNode. Though being based on version [R5.5](#) of the Patton SmartNode it should be applicable to lower versions, given a few adjustments.

In the following sections we describe the essential steps of configuration to allow for optimal cooperation of both the XCAPI and the Patton SmartNode by using the SIP protocol stack. At this point we suppose that the Patton SmartNode, the hardware the XCAPI is running on and both the XCAPI and your CAPI applications are already installed properly.

For some extended information on installation procedures please refer to the respective manuals. A short installation manual for the XCAPI is available at the [XCAPI Website](#).

## XCAPI Configuration

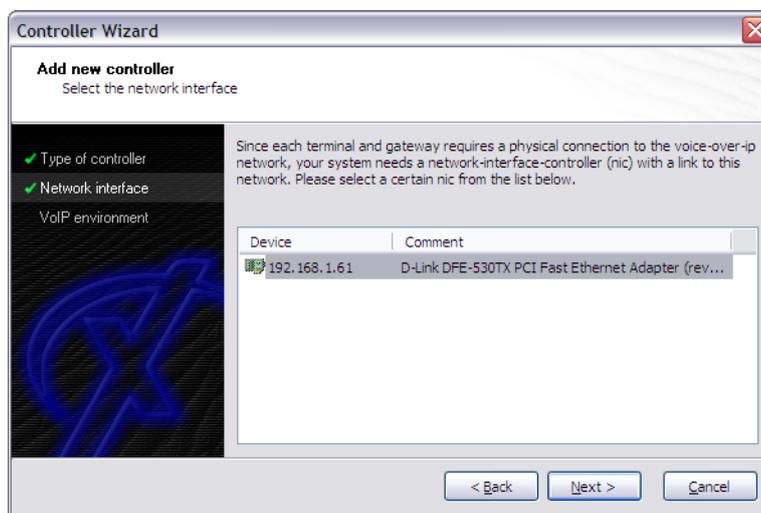
Please start up the XCAPI configuration to create a new controller assigned to the Patton SmartNode. If you've just installed the XCAPI and start the configuration tool for the first time, the [XCAPI Controller Wizard](#) will pop up automatically. This will also happen if there's no controller configured at all. To start up the [XCAPI Controller Wizard](#) on your own, just click the hyperlink labelled [Click here to add a controller](#) on the main page of the XCAPI configuration tool. On the first page of the Controller Wizard please select the [Add Voice-over-IP controller \(VoIP\)](#) option and continue by clicking on the Next button.





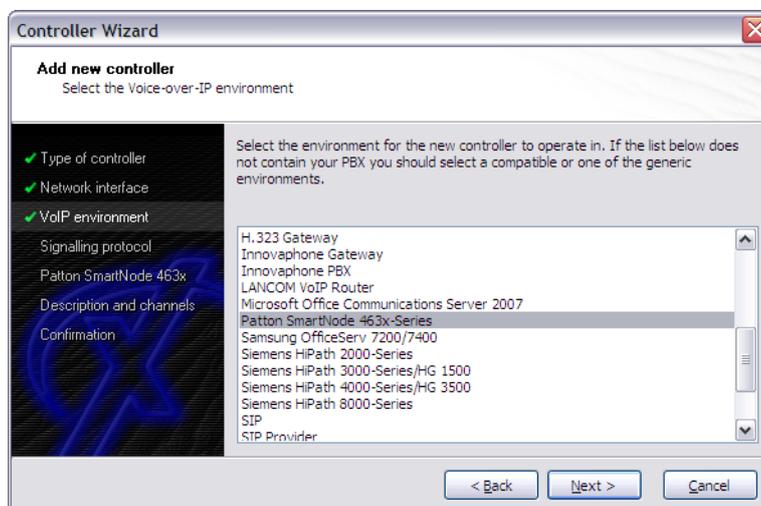
## 2.1 Network Interface

On this page of the XCAPI Controller Wizard you can select the network interface you want to bind to the XCAPI controller.



## 2.2 Voice-over-IP Environment

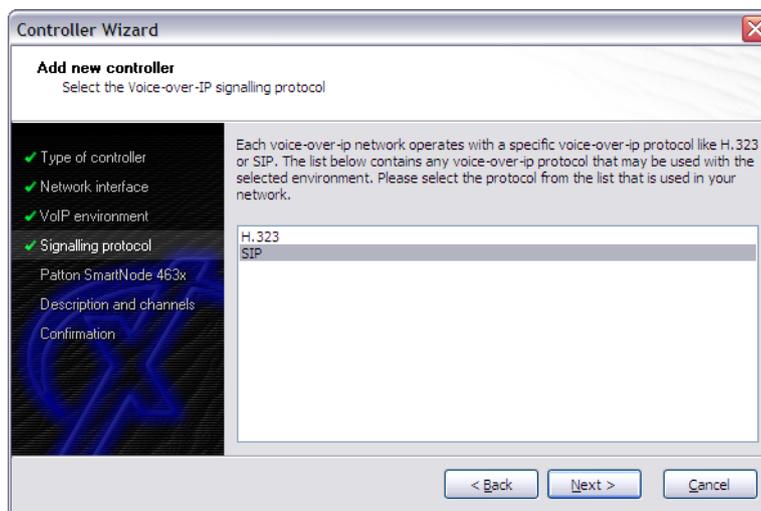
The next dialog of the configuration tool shows a list of some common Voice-over-IP environments. Selecting one of those will configure the XCAPI with a selection of near-optimal presets for the kind of environment you have, sparing you quite a lot of manual configuration.





## 2.3 Signalling Protocol

The next dialog shows a list of signalling protocols which are supported for the given Voice-over-IP environment. According to this example, SIP is selected.



## 2.4 IP Address of the Patton SmartNode

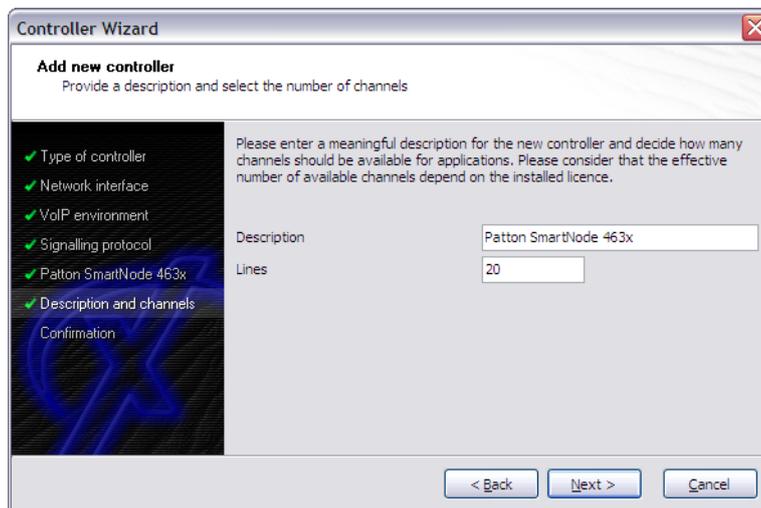
In the dialog **Network Address** please provide the IP address of your Patton SmartNode.





## 2.5 Description and Channels

That's about all information that has to be configured with the XCAPI. The next-to-final dialog of the Controller Wizard allows you to configure a meaningful description for the controller you're going to create. This isn't really used anywhere, so you can enter a text of your choice here. This dialog, however, also allows configuring the number of channels that the new controller will be able to provide. Please enter how many simultaneous connections the XCAPI should handle when communicating with the Patton SmartNode.

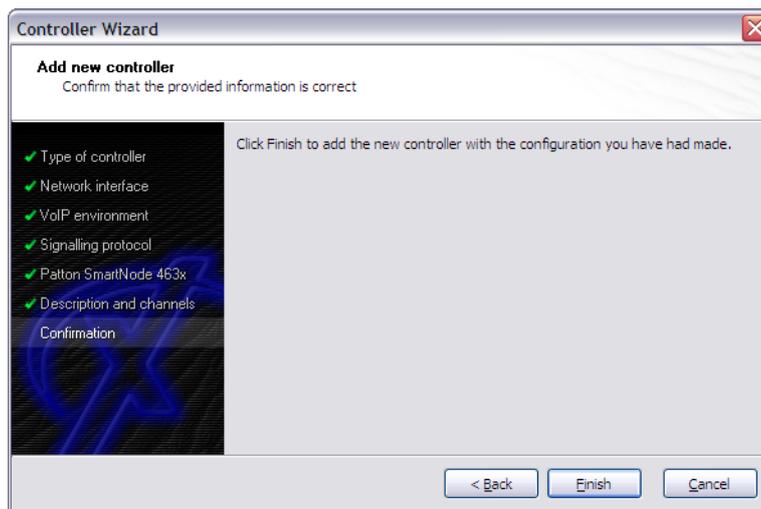


## 2.6 Confirmation

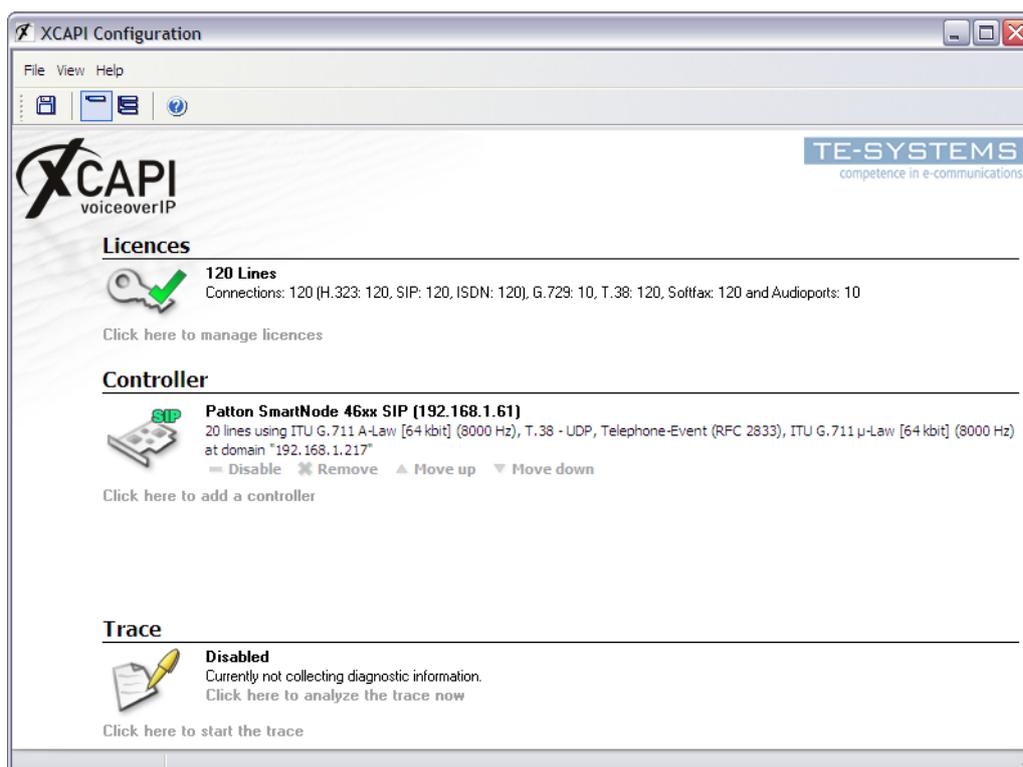
The final dialog of the Controller Wizard performs some checks on the configuration parameters you've made. If any errors are detected here, you can go back to the respective dialogs and correct the necessary input.



If everything is correct please use the Finish button in order to finally create the new controller.



The controller you've just created now will appear on the main page of the XCAPI configuration. As we're now finished with all XCAPI-related configuration tasks, please save the changes you've made and exit the configuration tool.





## Configuring the SmartNode Gateway

In order to establish the communication between the XCAPI and the Patton SmartNode using the SIP protocol, you need to create the XCAPI as a SIP interface with all its according configurations. In common, the easiest way is to customize the running configuration (running.cfg) at your local machine and uploading the modifications back again to the SmartNode gateway.

If you have any doubts, please review the SmartNode configuration guides.

However, you can find some configuration examples within our community download section based on a SmartNode 4634 using BRI ports. Convey the necessary configuration parts to your running config. Take care of the IP addresses, SIP ports, BRI/PRI ports and call routes. Keep always in mind that there might be slight differences between the CLI versions, the SmartNode models and the firmware you are using currently.



Do not import, export or edit any configuration files without having any knowledge about! Any configuration import is at your own risk!

### 3.1 Configuration Import

This chapter describes a customized running configuration (running.cfg) import, which can be done via the **Import/Export** configuration dialog. Please take notice of each dialog's instructions.

The screenshot shows the Patton SmartNode web interface. On the left is a vertical red navigation menu labeled 'CONFIGURATION MENU' with categories: Network, Telephony, Ports, and Various. The main content area is titled '192.168.1.217 / Import/Export' and has tabs for 'Import Firmware', 'Import Configuration', 'Import Licenses', and 'Export Configuration'. The 'Import Configuration' tab is active, displaying instructions: 'If you have previously exported the system configuration to a file then you can submit that file below and the system will update its startup configuration from the data saved in the file. After this operation the system should be reloaded to activate the new settings. The configuration is loaded directly into the flash and so does NOT immediately modify any settings.' Below this is a warning icon and text: 'This operation will erase whatever startup settings you currently have in the system.' The process is divided into two steps: '(1) Download Configuration' and '(2) Reload Device'. Under step 1, there is a text input field for 'Select file containing new startup configuration:', a 'Browse...' button, and an 'Import' button. Under step 2, there is a note: 'To activate the new configuration the device needs to be reloaded. We suggest you to immediately reload the device.'



The adapted configuration file needs to be selected within **Import Configuration** dialog. Use the Import button for proceeding the process.

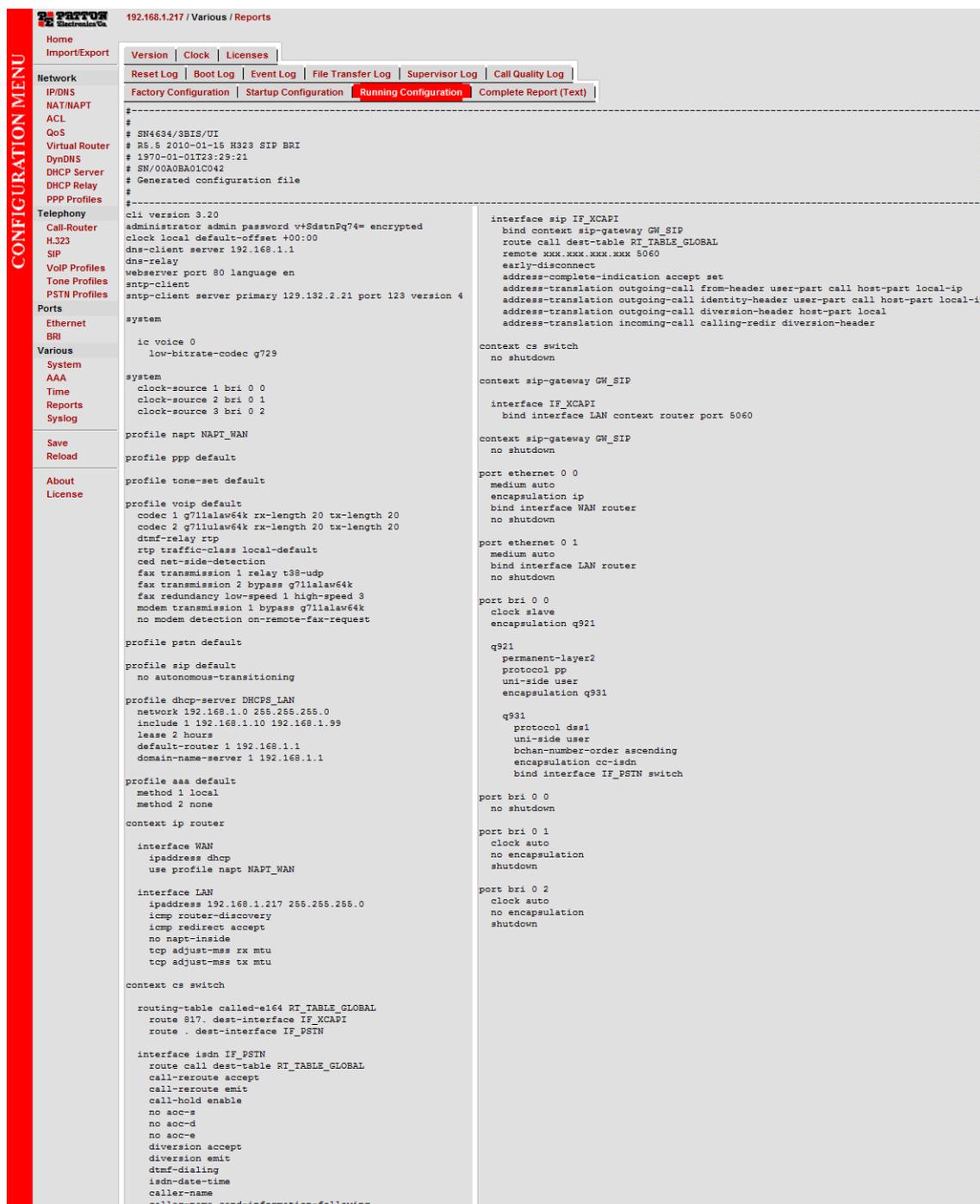
When the startup configuration has been imported successfully you can proceed with reloading the SmartNode device.

Finally you will be informed about the SmartNode saving mechanism. Invoke the reload process for running the new configuration.



### 3.2 Running Configuration

Review the running configuration to ensure that your import was successful. The following screenshot is only a summary of the most relevant settings.



The screenshot shows the configuration page for a Patton SmartNode. The 'Running Configuration' tab is selected, displaying the following configuration snippets:

```

#
# SN4634/3BIS/UI
# R5.5 2010-01-15 H323 SIP BRI
# 1970-01-01T23:29:21
# SN/00A0BA01C042
# Generated configuration file
#
-----
cli version 3.20
administrator admin password v+SdatnPg74= encrypted
clock local default-offset +00:00
dns-client server 192.168.1.1
dns-relay
webserver port 80 language en
ntp-client
ntp-client server primary 129.132.2.21 port 123 version 4

system
ic voice 0
  low-bitrate-codec g729

context cs switch
  no shutdown

context sip-gateway GW_SIP

interface IF_XCAPi
  bind context sip-gateway GW_SIP
  route call dest-table RT_TABLE_GLOBAL
  remote xxx.xxx.xxx.xxx 5060
  early-disconnect
  address-complete-indication accept set
  address-translation outgoing-call from-header user-part call host-part local-ip
  address-translation outgoing-call identity-header user-part call host-part local-ip
  address-translation outgoing-call diversion-header host-part local
  address-translation incoming-call calling-redir diversion-header

context cs switch
  no shutdown

context sip-gateway GW_SIP

interface IF_XCAPi
  bind interface LAN context router port 5060

context sip-gateway GW_SIP
  no shutdown

port ethernet 0 0
  medium auto
  encapsulation ip
  bind interface WAN router
  no shutdown

port ethernet 0 1
  medium auto
  bind interface LAN router
  no shutdown

port bri 0 0
  clock slave
  encapsulation q921

q921
  permanent-layer2
  protocol pp
  uni-side user
  encapsulation q931

q931
  protocol ds1
  uni-side user
  bchan-number-order ascending
  encapsulation cc-isdn
  bind interface IF_PSTN switch

port bri 0 0
  no shutdown

port bri 0 1
  clock auto
  no encapsulation
  shutdown

port bri 0 2
  clock auto
  no encapsulation
  shutdown

routing-table called-rl64 RT_TABLE_GLOBAL
  route 817. dest-interface IF_XCAPi
  route . dest-interface IF_PSTN

interface isdn IF_PSTN
  route call dest-table RT_TABLE_GLOBAL
  call-reroute accept
  call-reroute emit
  call-hold enable
  no acc-s
  no acc-d
  no acc-e
  diversion accept
  diversion emit
  dtmf-dialing
  isdn-date-time
  caller-name
  caller-name send-information-following
  
```



## SmartNode Configuration details

Next we show, based on the imported running.cfg file, some configuration details. Some enhanced configurations, such as mapping tables, might be used.

### 4.1 ISDN Interface

The ISDN interface configuration is used with the default values and profiles. The here described ISDN interface named IF\_PSTN is bound to port bri 0 0, as described in the chapter BRI Port Configuration starting on [page 16](#).

Interfaces			
ISDN SIP			
Name	Bound Port	Routing Destination	
IF_PSTN	bri 0 0	RT_TABLE_GLOBAL (Table)	✘
			☰

The Call-Routing Destination is related to the RT\_TABLE\_GLOBAL table as described in the chapter Routing and Mapping Tables starting on [page 12](#).

Configuration		Status
Call-Routing Destination	<input type="radio"/> Interface (none) <input checked="" type="radio"/> Table RT_TABLE_GLOBAL <input type="radio"/> Service (none)	
DTMF Dialing	<input checked="" type="checkbox"/> Allows a user to dial using DTMF signals	
Early Proceeding	<input checked="" type="checkbox"/> 12 seconds Sends a CALL PROCEEDING to the remote terminal after the specified timeout if routed to a slow destination network that does not change to the PROCEEDING state before that time	
Call-Waiting	<input checked="" type="checkbox"/>	
Call-Transfer	Accept: <input checked="" type="checkbox"/> Accepts ECT invocations from the connected phone Emit: <input checked="" type="checkbox"/> Sends ECT invocations for internally looped calls	
Call-Reroute	Accept: <input checked="" type="checkbox"/> Accepts Reroute invocations from the connected PBX Emit: <input checked="" type="checkbox"/> Sends Reroute invocations for internally looped calls	
Diversion	Accept: <input checked="" type="checkbox"/> Accepts DivertingLegInformation2 invocations Emit: <input checked="" type="checkbox"/> Sends DivertingLegInformation2 invocations	
Advice of Charge	AOC-S (Tariff Info) <input type="checkbox"/> transparent AOC-D (Charge During The Call) <input type="checkbox"/> transparent AOC-E (Charge At The End) <input type="checkbox"/> transparent <small>Transparent transparently passes AOC information; automatic always sends AOC information; explicit only sends AOC information if requested (network), or explicitly requests AOC information (user).</small>	
Address-Complete Indication	Accept: <input type="checkbox"/> transparent Emit: <input type="checkbox"/> transparent <small>Transparent transparently converts a Sending-Complete IE to an address-complete indication; set always sets the address-complete indication (accept), always sends a Sending-Complete IE (emit), and clear never sets the address-complete indication (accept), never sends a Sending-Complete IE (emit).</small>	
Publish Date/Time	<input checked="" type="checkbox"/> Sends the system date/time with every CONNECT message to the connected phone	
Caller Name	Accept/Emit: <input checked="" type="checkbox"/> Enables transmission of calling name in User to User information elements of DSS1 setup messages or in Facility information elements of other protocol setup messages. Early Alerting: <input checked="" type="checkbox"/> 2000 milliseconds When checked, for an incoming ISDN call, sends an ALERTING to the remote ISDN terminal when there is no name within the timeout. Set to 0 if an ALERTING should be sent immediately (only applicable for non-DSS1 networks) Ignore Name Absence: <input checked="" type="checkbox"/> 4000 milliseconds When checked, for an incoming ISDN call, handles the call even if there is no name after the timeout. Set to 0 to handle the call immediately (only applicable for non-DSS1 networks) Send Information Following: <input type="checkbox"/> When checked, for an outgoing ISDN call, sends an informationFollowing invoke with the SETUP when there is no name yet (only applicable for non-DSS1 networks) Presentation: <input type="checkbox"/> simple Configures the format in which the calling name is sent	
Inband Info	Accept Transparent <input checked="" type="checkbox"/> Select Progress Descriptors <input checked="" type="checkbox"/> 1. CallsNotIsdnTolsdn <input type="checkbox"/> 2. DestinationNotIsdn <input type="checkbox"/> 3. OriginationNotIsdn <input type="checkbox"/> 4. ReturnedTolsdn <input checked="" type="checkbox"/> 8. InbandInfoAvailable Force Call-Setup <input type="checkbox"/> Select Trigger Message alerting Force Call-Release <input type="checkbox"/>	
PSTN Profile	<input type="text" value="default"/>	
Tone Profile	<input type="text" value="default"/>	

Apply ✓



## 4.2 SIP Interface

The SIP interface, in this example named IF\_XCAPI, is assigned to the IP address 192.168.1.61 by using the default SIP port 5060, which represents the application server and its properly running XCAPI. This interface is related to the SIP gateway named GW\_SIP, as described in the chapter SIP Gateway on [page 14](#). The SIP interface is also assigned to the call routing destination table RT\_TABLE\_GLOBAL as described in chapter Routing and Mapping Tables starting on [page 12](#).

Interfaces			
ISDN			
SIP			
Name	Remote	Routing Destination	
IF_XCAPI	192.168.1.61 /5060	RT_TABLE_GLOBAL (Table)	✘
			📺

The tone set and SIP profile are used with their default values whilst the VoIP profile is related to PR\_XCAPI, see chapter VoIP Profiles on [page 14](#). The call transfer options must be configured by your needs. For this example these options are enabled. All other parameters are not set or used with their default values.

Configuration		Incoming Call Address Translation	Outgoing Call Address Translation	Status
SIP Gateway	<input checked="" type="checkbox"/>	GW_SIP		
Call-Routing Destination	<input checked="" type="checkbox"/>	Interface: (none)	Table: RT_TABLE_GLOBAL	
Remote User Agent Host Name / Port	<input checked="" type="checkbox"/>	192.168.1.61	5060	
Local User Agent Host Name / Port	<input type="checkbox"/>			
Early Connect	<input type="checkbox"/>	Connect call when local terminal plays precall announcement		
Early Disconnect	<input checked="" type="checkbox"/>	Release call when local terminal hangs up		
Hold-Method		zero-ip		
Call-Transfer		<input checked="" type="checkbox"/> Accepts REFER messages from the connected user agent <input checked="" type="checkbox"/> Sends REFER messages to transfer internally looped calls <input checked="" type="checkbox"/> Detects external call loops and connects intern through		
Call-Reroute		<input type="checkbox"/> Sends 302 moved temporarily messages to reroute internally looped calls		
Address-Complete Indication	Accept:	set Set always sets the address-complete indication; and clear never sets the address-complete indication.		
Advice of Charge	AOC-D (Charge During The Call)	<input type="checkbox"/> Accept (receive AOC-D from the remote SIP terminal and pass them to ISDN) <input type="checkbox"/> Emit (send AOC-D messages received from ISDN to the remote SIP terminal)		
Privacy		<input type="checkbox"/> Use the Identity-header for the Calling Party Number in addition to the From header. The handling of this header can be configured for incoming and outgoing direction separately.		
Accept Address Update	<input type="checkbox"/>	wait-for-name	Proceeding Timeout [ms] 4000	Alerting Timeout [ms] 0
Overlap dialing	With new transaction	<input type="checkbox"/> Accept (receive INVITE with updated called-user information from the remote SIP terminal and forward them) <input type="checkbox"/> Emit (send INVITE with updated called-user information received to the remote SIP terminal)		
Penalty Box	<input type="checkbox"/>			
Use new session after redirect	<input type="checkbox"/>			
Session Timer	<input type="checkbox"/>	1800	seconds	
VoIP Profile		PR_XCAPI		
Tone Set Profile		default		
Sip Profile		default		

Apply ✓



### 4.3 Routing and Mapping Tables

The routing and mapping tables for this configuration example are used as described next. The routing table, named RT\_TABLE\_GLOBAL, looks up for the called-e164 number for routing calls based on the called party E.164 number.

Routing Tables		
Name	Looks up for	
RT_TABLE_GLOBAL	called-e164	✘
<input type="text"/>	called-e164	☰

For this example, all matching numbers starting with 817 will be routed to the SIP interface destination named IF\_XCAPI.

The prefix 0 is used for routing all inbound calls to the PSTN BRI interface named IF\_PSTN. The related Mapping Table function, named MAP\_TABLE\_GLOBAL, is configured as follows.

Looks Up For called-e164 Of	Destination	Execute Function (Optional)	
817...	IF_XCAPI (SIP Interface)		✘
0.	IF_PSTN (ISDN Interface)	MAP_TABLE_GLOBAL (Mapping Table)	✘
called-e164 value or default <input type="text"/>	<input type="radio"/> Interface (none) <input type="radio"/> Table (none) <input type="radio"/> Service (none) <input type="radio"/> none	Optional function to execute (none)	☰

(To change an entry, enter the value of an existing entry)



The mapping table, named MAP\_TABLE\_GLOBAL, looks up the called-e164 number.

Mapping Tables			
Name	Looks up for	Modifies	
MAP_TABLE_GLOBAL	called-e164	called-e164	✖
<input type="text"/>	called-e164	called-e164	🗑️

Complex Functions	
Name	
<input type="text"/>	🗑️

The mapping table, named MAP\_TABLE\_GLOBAL, is used for deleting the matching PSTN prefix number 0.

Configuration		
Looks Up For called-e164 Of	Modifies called-e164 To	
0.(%)	\1	✖
called-e164 value or <i>default</i>	called-e164 value	🗑️
<input type="text"/>	<input type="text"/>	
<small>(To change an entry, enter the value of an existing entry)</small>		

The according configuration entries are clearly arranged.

```

Version | Clock | Licenses
Reset Log | Boot Log | Event Log | File Transfer Log | Supervisor Log | Call Quality Log
Factory Configuration | Startup Configuration | Running Configuration | Complete Report

#-----#
# SN4634/3BIS/UI                               #
# R5.5 2010-01-16 H323 SIP BRI                   #
# Generated configuration file                   #
#-----#

context cs switch

  routing-table called-e164 PT_TABLE_GLOBAL
  route 0. dest-interface IF_PSTN MAP_TABLE_GLOBAL
  route 8179.. dest-interface IF_XCAPI

  mapping-table called-e164 to called-e164 MAP_TABLE_GLOBAL
  map 0.(%) to \1
  
```



## 4.4 SIP Gateway

The SIP gateway must be enabled and related to the according XCAPI SIP interface, as described in the chapter SIP Interface on [page 11](#).

Gateways		
Name	State	
GW_SIP	Enabled	✗
<input type="text"/>		☰

It's bound to the LAN IP interface, see chapter LAN Interface on [page 18](#) and related to the default SIP port 5060.

Configuration		Status	
SIP Gateway		Enabled ▾	
		Apply ✓	
Sip Interface			
Name	Binding		Settings
	Bind	IP Interface	Port
IF_XCAPI	<input checked="" type="checkbox"/>	LAN ▾	5060
			0
			Contact
			☑
			✗
			☰
Location Services bound			
		☰	

## 4.5 VoIP Profiles

The [PR\\_XCAPI VoIP Profile](#) is related to the XCAPI SIP interface. Beside of the default values, the voice codecs g711alaw64k and g711ulaw64k are used with a frame size of 20ms.

Profiles		
VoIP Profiles		
Name		
default		
PR_XCAPI		✗
<input type="text"/>		☰
Import From File		
Select VoIP Profile File:	<input type="text"/>	Durchsuchen... Import
VoIP Profile Users		
Interface	Used VoIP Profile	
IF_XCAPI (SIP Interface)	PR_XCAPI ▾	☑



Please ensure that beside of the selected voice codecs and the configuration of the additional voice parameters, the fax transmission methods and the configuration of the additional fax parameters are in accordance with the XCAPI controller configuration.

Voice Codescs					
Position	Codec	Rx Length [ms]	Tx Length [ms]	Silence Suppression	
1	g711alaw64k	20	20	<input checked="" type="radio"/> default <input type="radio"/> yes <input type="radio"/> no	✓ ✗
2	g711ulaw64k	20	20	<input checked="" type="radio"/> default <input type="radio"/> yes <input type="radio"/> no	✓ ✗
	transparent			<input checked="" type="radio"/> default <input type="radio"/> yes <input type="radio"/> no	✗

Additional Voice Parameters	
Default Silence Suppression	<input checked="" type="checkbox"/> If not specified by the codec
Highpass Filter	<input checked="" type="checkbox"/> Voice input filter for A/D conversion
Post Filter	<input checked="" type="checkbox"/> Voice output filter for D/A conversion
DTMF Relay	<input checked="" type="radio"/> default <input checked="" type="radio"/> rtp <input type="radio"/> signaling <input type="text" value="default"/>
Flash-hook Relay	<input type="radio"/> default <input type="radio"/> rtp <input type="radio"/> signaling <input type="text" value="default"/>
RTP Payload Type For Tone Events (NTE)	<input type="text" value="101"/>
RTP Payload Type For Signaling Events (NSE)	<input type="text" value="100"/>
RTP Payload Type For Transparent Clearmode	<input type="text" value="97"/>
RTP Payload Type For G.726-32	<input type="text" value="2"/>
RTP Payload Type For G.726-32 Cisco Compatible	<input type="text" value="2"/>
RTP Traffic Class	<input type="text" value="local-default"/>

Apply ✓



In this example, the t38-udp protocol is used for facsimile transmissions, which is also set as default fax codec within the XCAPI controller configuration. To avoid any T.38 interoperability problems, please move the t38-udp codec to the first position. Additional information can be found in the T.38 chapter starting on [page 20](#).

Fax Transmission Methods				
Position	Method	Protocol		
1	relay	t38-udp		✗
2	bypass	g711alaw64k		✗
	relay	t38-udp		☒
	bypass	g711alaw64k		☒

Additional Fax Parameters	
Fax Detection	ced-tone
Error Correction	<input checked="" type="checkbox"/>
Max Bitrate	14400 bps
HDLCL Image Transfer	<input checked="" type="checkbox"/>
T.38 Redundancy	Low Speed 1 additional packets High Speed 3 additional packets
T.38 CED Retransmission	<input checked="" type="checkbox"/> Number of additional packets 2
T.38 No-Signal Retransmission	Number of packets (1 - 5) 3
T.38 Output Volume	-9.5
Dejitter Buffer Max Delay	200 milliseconds
Bypass Method	default
CED-Tone Network Side Detection	<input type="checkbox"/> Allow detection of fax/modem answer tones on the network (RTP) side

Apply ✓

## 4.6 BRI Port Configuration

The **BRI Port Configuration** is not relevant for setting up any SIP trunk configuration at all and it is just shown to see this configuration example as a whole. The clock, the Q.921 and Q.931 settings must be adapted to your BRI/PRI connection. You can verify the link state within the status tab.

Configuration		Status
Clock	slave	
Line Power	Off	
Encapsulation	<input checked="" type="checkbox"/> q921	
Port State	Enabled	
Apply ✓		
Q.921 (ISDN Layer 2)		
Permanent Activity	Enabled	
Endpoint Type	user	
Protocol	pp	
Encapsulation	<input checked="" type="checkbox"/> q931	
Apply ✓		
Q.931 (ISDN Layer 3)		
Signaling Protocol	dss1	
Endpoint Type	user	
B-Channel Allocation	ascending	
B-Channel Range	0 to 1	
Maximum Calls	2	
Bind	<input checked="" type="checkbox"/> IF_PSTN	
Apply ✓		



## 4.7 Call-Router Configuration

The Patton SmartNode call-router configuration is also used with the default settings.

Interfaces	Routing Tables	Functions	Services	Configuration	Active Calls	Status
<b>State</b>						
Call-Router	Enabled <input type="button" value="v"/>		When the Call-Router is disabled all calls routed to a table (route call dest-table table) are dropped			
Apply <input checked="" type="checkbox"/>						
<b>Digit-Collection Timeout</b>						
The digit-collection timeout starts running when a called-party number matches a called-e164 routing-table entry that ends with a T. The Call-Router then collects overlap-dialed digits sent within the timeout. The timeout restarts whenever another digit arrives. When the timeout elapses the call is placed to the destination configured with the T-entry. In this section the timeout duration can be configured. Additionally you can configure the actions that shall be performed when the timeout elapses, for example, appending the terminating character to the called-party number or setting the address-complete indication.						
Enabled	<input checked="" type="checkbox"/>	Enables digit collection on T-entries.				
Default Timeout	5	seconds	Default digit-collection timeout. <b>Note:</b> This timeout can be overridden on a <b>per-rule basis</b> adding the timeout in seconds after the T, for example <b>0T3</b> , to use a timeout of 3 seconds for this entry.			
Append Terminating Character	<input type="checkbox"/>	When the timeout elapses, appends a terminating character to the called-party number as configured below.				
Set Address-Complete Indication	<input type="checkbox"/>	When the timeout elapses, sets the address-complete indication, for example generating an ISDN Sending-Complete IE.				
Apply <input checked="" type="checkbox"/>						
<b>Digit-Collection Terminating Character</b>						
The digit-collection terminating character immediately terminates the digit-collection timeout and is an indication that the called-party number is complete. The Call-Router normally removes the terminating character from the called-party number. In this section the terminating character can be configured. Additionally you can configure the actions that shall be performed when the terminating character is detected, for example, re-appending the terminating character to the called-party number or setting the address-complete indication.						
Enabled	<input checked="" type="checkbox"/>	Enables the immediate termination of the digit-collection timeout by reception of the terminating character.				
Default Character	#	Default Digit-Collection Terminating Character.				
Append Terminating Character	<input type="checkbox"/>	When the terminating character is detected, re-appends the terminating character to the called-party number as configured above.				
Set Address-Complete Indication	<input type="checkbox"/>	When the terminating character is detected, sets the address-complete indication, for example generating an ISDN Sending-Complete IE.				
Apply <input checked="" type="checkbox"/>						
<b>Digit-Collection Full Match</b>						
A full match happens when a called-party number matches a called-e164 entry that ends with a S. In this section you can configure the actions that shall be performed when a full match is detected, for example, appending the terminating character to the called-party number or setting the address-complete indication.						
Append Terminating Character	<input type="checkbox"/>	When a full match is detected, appends the terminating character to the called-party number as configured above.				
Set Address-Complete Indication	<input type="checkbox"/>	When a full match is detected, sets the address-complete indication, for example generating an ISDN Sending-Complete IE.				
Apply <input checked="" type="checkbox"/>						
<b>Address-Completion</b>						
The address-completion timeout starts running when a called-party number is incomplete to match one of the called-e164 routing-table entries. The Call-Router then collects overlap-dialed digits sent within the timeout. The timeout restarts whenever another digit arrives. When the timeout elapses, the call is dropped.						
Timeout	<input checked="" type="checkbox"/>	12	seconds	Address-Completion timeout		
Apply <input checked="" type="checkbox"/>						
<b>E.164 Number Prefixes</b>						
National Prefix	<input type="checkbox"/>	0	Prefix that is prepended to national E.164 numbers at the ingress interface			
International Prefix	<input type="checkbox"/>	00	Prefix that is prepended to national E.164 numbers at the ingress interface			
Apply <input checked="" type="checkbox"/>						



## 4.8 LAN Interface

This SmartNode gateway is running with the user defined IP address 192.168.1.217 and configured as shown on the next screenshot.

The screenshot shows the configuration page for the LAN interface. The tabs at the top are Configuration, Link Supervision, and Status. The Configuration tab is active. The settings are as follows:

- IP Address:** DHCP is unselected, and User Defined IP Address is selected with the value 192.168.1.217. The IP Mask is 255.255.255.0.
- Point-to-Point:** Checked.
- NAPT-Outside:** Checked, with Profile set to NAPT\_WAN.
- RTP Encryption:** Unchecked. A note states: "(If enabled, local RTP streams traverse the ACL and Service Profiles below; IPsec may be applied to RTP streams)".
- ACL Profile:** Inbound and Outbound are both set to (none).
- Service Profile:** Inbound and Outbound are both set to (none).
- TCP MSS Adjust (Limits TCP segment size in the opposite direction; used on access links with reduced MTU, e.g. PPPoE):**
  - Inbound: Checked, Auto, MSS: MTU - 40 Bytes.
  - Outbound: Checked, Auto, MSS: MTU - 40 Bytes.
- MTU:** 1500 Bytes.
- IGMP interface Type:** (none).
- ICMP Redirect Messages:** Send is checked, Accept is unchecked.
- ICMP Router Discovery:** Checked.

At the bottom right of the configuration area is an "Apply" button with a green checkmark. Below the configuration area is a warning message: "Changing IP interface settings may disconnect your browser from the webserver on the device. The changes are immediately applied when you click to the Apply button. For example when you change the IP address of the IP interface over which you are connected with your browser you have to change the URL in your browser manually before you can continue using the Web-GUI to configure the device."

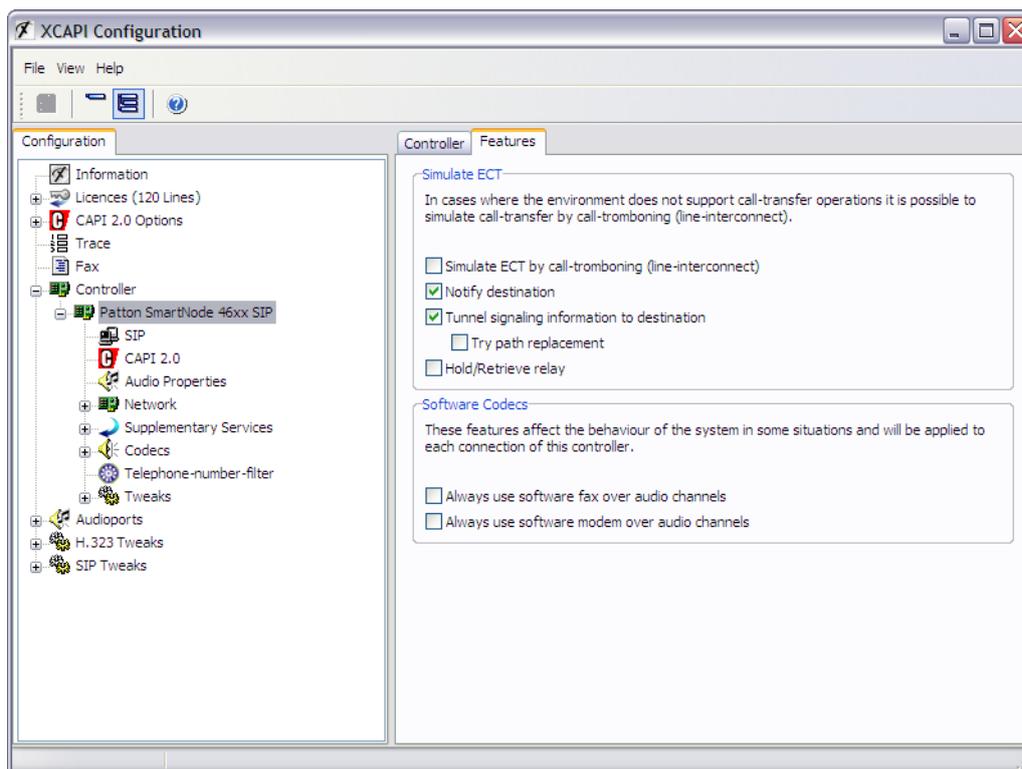


## Supplementary Services

Please review the following chapters for some information on optimal supplementary services configuration.

### 5.1 Call Transfer

Please ensure that the **Simulated ECT by call-tromboning (line-interconnect)** parameter of the XCAPI controller **Features** dialog is not activated for supporting call transfers via the SIP refer method. You may have to adapt the according call transfer parameters of the SmartNode ISDN and SIP interface configuration, as described in the chapters ISDN and SIP Interface starting on [page 10](#).

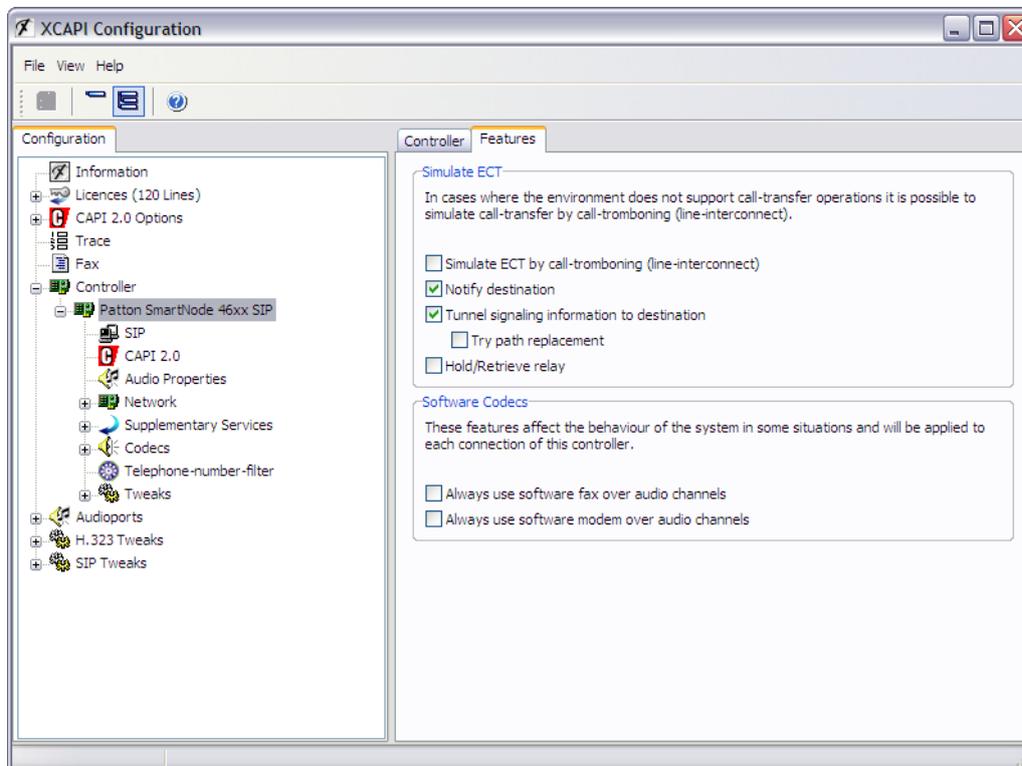




## 5.2 T.38

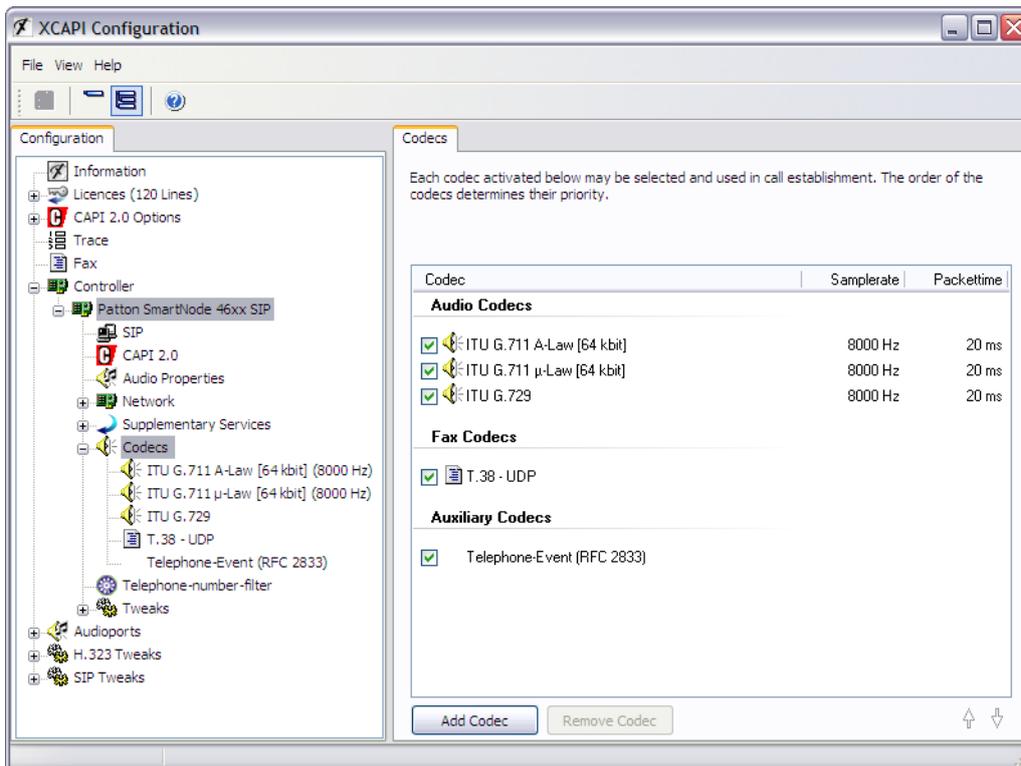
When using the T.38 protocol you have to enable the t38-udp method, as already described in the chapter VoIP Profiles on [page 14](#), in the related XCAPI VoIP profile.

It is also required to disable the [Always use software fax over audio channels](#) option within the XCAPI controller Features dialog.





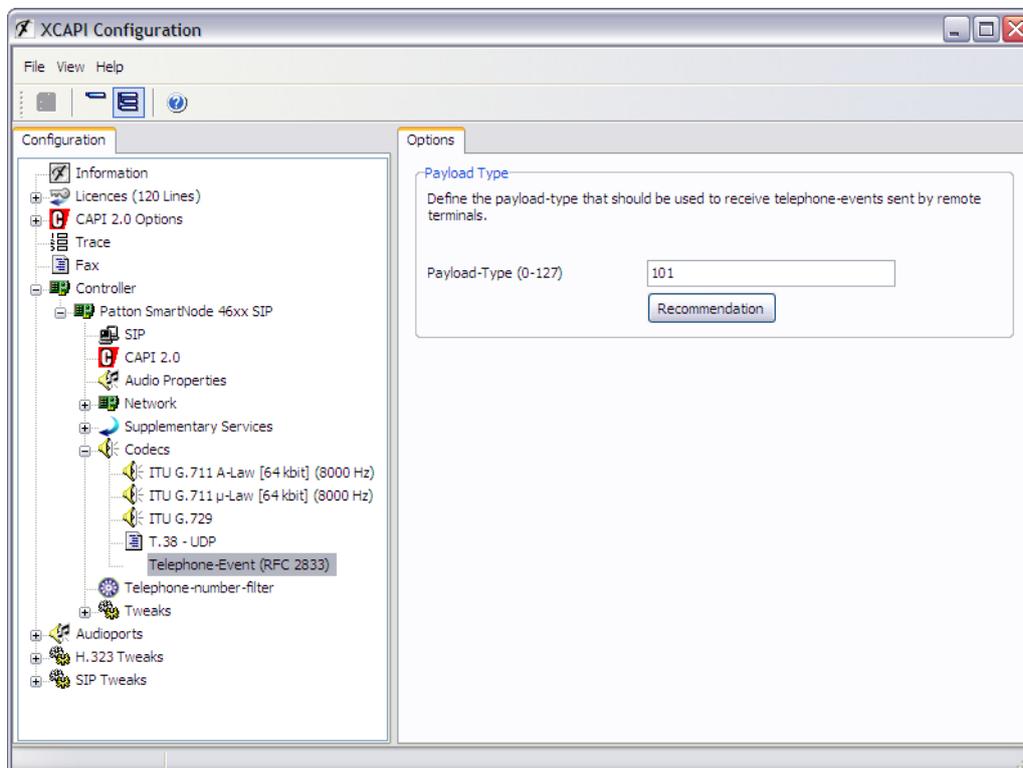
Ensure that the T.38 - UDP codec, which should be set by default when using the XCAPI controller wizard, is set and enabled.





## 5.3 DTMF

The payload type for signalling Telephone Events via RFC 2833 is by default set to value 101. For DTMF interoperability please ensure that the rtp option, as shown in the chapter VoIP Profiles starting on [page 14](#), is selected as DTMF relay method within the [Additional Voice Parameters](#) configuration dialog, and that the RTP Payload Type For Tone Events (NTE) is also set to 101.





## Troubleshooting

Use the CLI interface via telnet connection for appropriate debugging. Please refer to the SmartWare Software Configuration Guide for detailed debug information.

### 6.1 Debug Call Routes

Next, we give you a short example about debugging your call routes. For this please login and enable the configuration mode.

```
SmartNode>enable
SmartNode#configure
SmartNode(cfg)#
```

Use the debug call-control or debug call router detail 5 command for enabling the debug output. It's recommended to verify the inbound, outbound and all other configured call routes. The following debug output shows that the dialed number 00536381950 completely matches the prefix 0. of the RT\_TABLE\_GLOBAL table. Afterwards the dialed number becomes mapped by the MAP\_TABLE\_GLOBAL definition and the call will be placed to the IF\_PSTN interface which is bound to the BRI port.

```
SmartNode(cfg)#debug call-router detail 5

17:28:51 CR > [switch] Routing-Lookup:
17:28:51 CR > Execute all entries in table IF_XCAPI-precallservice
17:28:51 CR > Find best-matching called- entry in table RT_TABLE_GLOBAL
17:28:51 CR > 00: Prefix Timeout Expression: called-e164 of 00536381950 completely matches ^(?:0.)
17:28:51 CR > 01: Prefix Timeout Expression: called-e164 of 00536381950 does not match ^(?:8179..)
17:28:51 CR > 02: Prefix Timeout Expression: called-e164 of 00536381950 does not match ^(?:8178..)
17:28:51 CR > 03: Prefix Timeout Expression: called-e164 of 00536381950 does not match ^(?:8172..)
17:28:51 CR > Selecting entry 0
17:28:51 CR > Find best-matching called- entry in table MAP_TABLE_GLOBAL
17:28:51 CR > 00: Prefix Timeout Expression: called-e164 of 00536381950 completely matches ^(?:0(.*))
17:28:51 CR > Selecting entry 0
17:28:51 CR > Execute Expression: called-e164 changed to '0536381950'
17:28:51 CR > Execute all entries in table IF_PSTN-dest
17:28:51 CR > Execute all entries in table route-found-place-call
17:28:51 CR > Lookup result: Route found; place call (timeout=0)
```

```
SmartNode(cfg)#no debug call-router
```

The next debug output shows that an incoming call of the IF\_PSTN interface completely matches the prefix 8179.. of the RT\_TABLE\_GLOBAL table. Afterwards the call of the found route becomes placed to the IF\_XCAPI interface.

```
SmartNode(cfg)#debug call-router detail 5

17:39:35 CR > [switch] Routing-Lookup:
17:39:35 CR > Execute all entries in table IF_PSTN-precallservice
17:39:35 CR > Find best-matching called- entry in table RT_TABLE_GLOBAL
17:39:35 CR > 00: Prefix Timeout Expression: called-e164 of 817900 does not match ^(?:0.)
17:39:35 CR > 01: Prefix Timeout Expression: called-e164 of 817900 completely matches ^(?:8179..)
17:39:35 CR > 02: Prefix Timeout Expression: called-e164 of 817900 does not match ^(?:8178..)
17:39:35 CR > 03: Prefix Timeout Expression: called-e164 of 817900 does not match ^(?:8172..)
17:39:35 CR > Selecting entry 1
17:39:35 CR > Execute all entries in table IF_XCAPI-dest
17:39:35 CR > Execute all entries in table route-found-place-call
17:39:35 CR > Lookup result: Route found; place call (timeout=0)
```

```
SmartNode(cfg)#no debug call-router
```



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