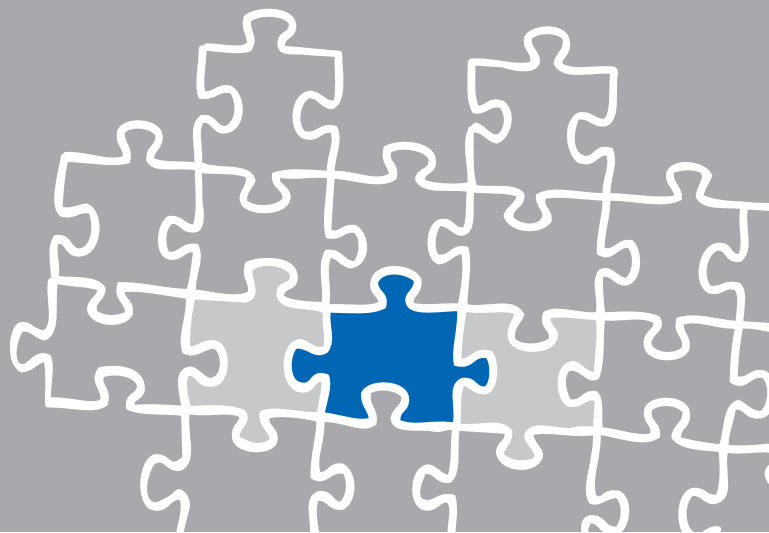


# TechNote

## AudioCodes Mediant Series

August 5, 2013





## Introduction

This document is intended to support you with the integration of the XCAPI, version **3.3.205** or above, into an existing environment of the AudioCodes Mediant gateway. Though being based on version **5.60A.018** of the AudioCodes Mediant 600 it is also applicable to other Mediant series types with other software versioning's.

In the following sections we describe the essential steps of configuration to allow for optimal cooperation of both the XCAPI and the gateway by using the **SIP** protocol stack. At this point we suppose that the AudioCodes Mediant gateway, 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

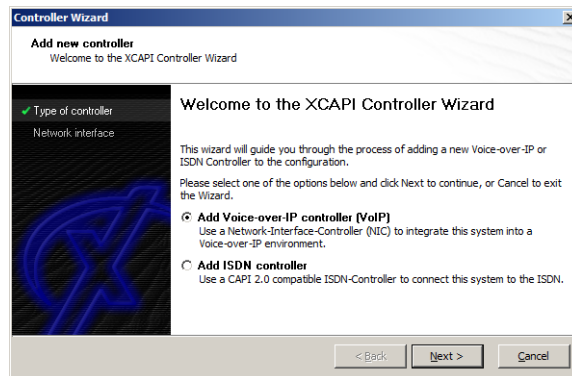
Please start up the XCAPI configuration to create a new controller assigned to the AudioCodes gateway.

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 labeled **Click here to add a controller** on the main page of the XCAPI configuration tool.

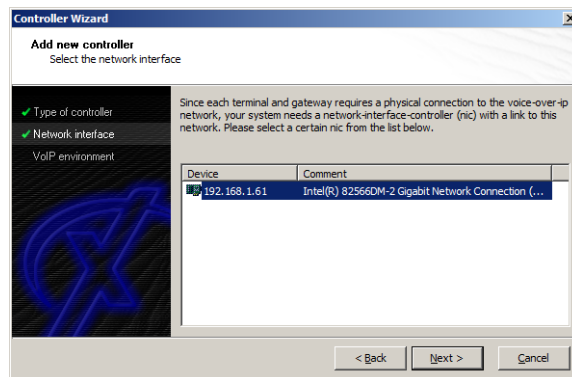


On the first page of the Controller Wizard please select the **Add Voice-over-IP controller (VoIP)** option and continue by clicking on the **Next** button.



## 2.1 Network Interface

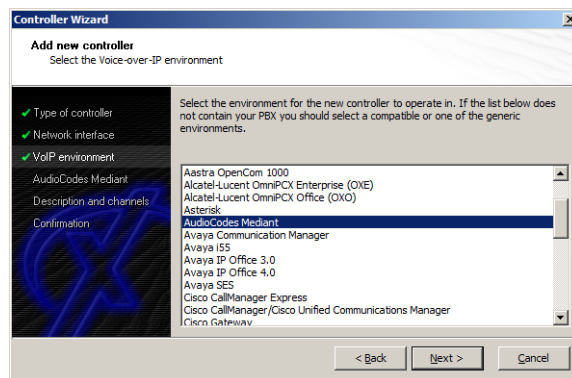
On this page of the XCAPI Controller Wizard you can select the network interface you want to bind to the XCAPI controller.





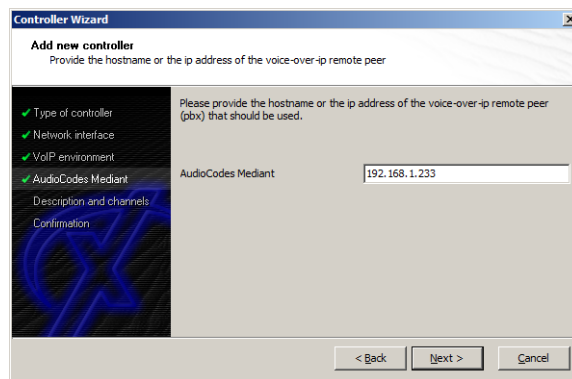
## 2.2 Voice-over-IP Environment

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 IP Address of the AudioCodes Mediant

In the dialog **Network Address** please provide the IP address of your AudioCodes gateway.

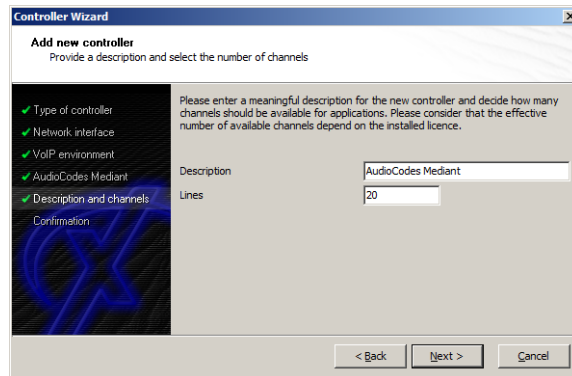




## 2.4 Description and Channels

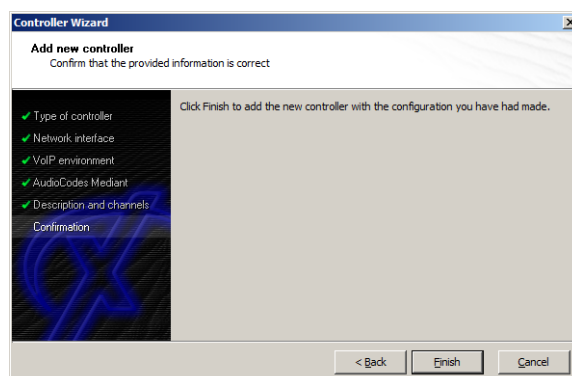
That's about all information that has to be configured with the XCAPI. The next-to-final dialog of the Controller Wizard allows you to configure a meaningful description for the controller you're going to create. This isn't really used anywhere, so you can enter a text of your choice here.

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 AudioCodes gateway.



## 2.5 Confirmation

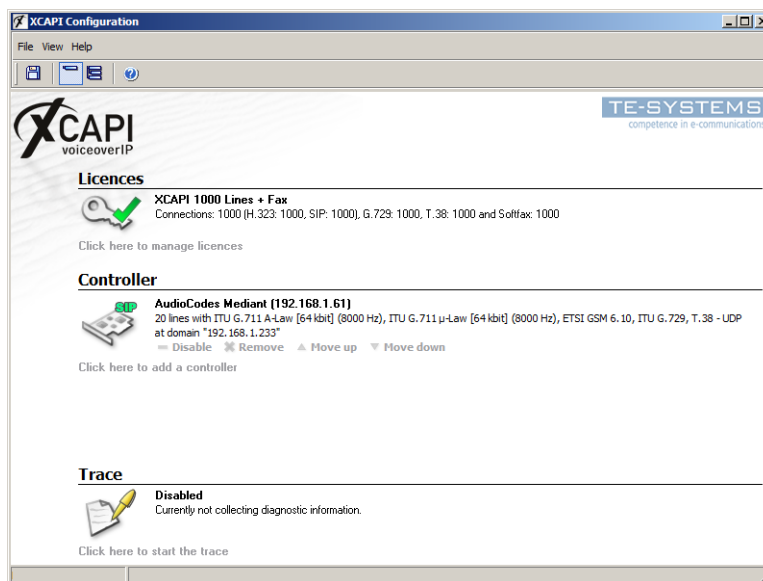
The final dialog of the Controller Wizard performs some checks on the configuration parameters you've made. If any errors are detected here, you can go back to the respective dialogs and correct the necessary input. If everything is correct please use the Finish button in order to finally create the 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 AudioCodes

In order to establish the communication between the XCAPI and the gateway using the SIP protocol, you need to declare the XCAPI as an IP route with its according dial rules within the Mediant configuration.

This example gives you an overview about the necessary configurations.

## Network Settings

### 4.1 IP Settings

The IP settings of the AudioCodes Mediant gateway are used for this example as shown below.

The screenshot shows the 'IP Settings' configuration window. It contains the following sections and fields:

- IP Settings**
  - IP Networking Mode: Single IP Network
- Single IP Settings**
  - IP Address: 192.168.1.233
  - Subnet Mask: 255.255.255.0
  - Default Gateway Address: 192.168.1.1
- Multiple Interface Settings**
  - Multiple Interface Table: [Icon]
- VLAN Mode**
  - VLAN Mode: 0
- VLAN ID Settings**
  - Native VLAN ID: 1
  - OAM VLAN ID: 1
  - Control VLAN ID: 2
  - Media VLAN ID: 3
- NAT Settings**
  - NAT IP Address: 0.0.0.0



## Media Settings

### 5.1 Voice Settings

The voice settings, especially the DTMF parameters, need to be conforming to your VoIP environment.

Voice Settings	
Voice Volume (-32 to 31 dB)	0
Input Gain (-32 to 31 dB)	0
Silence Suppression	Disable
DTMF Transport Type	RFC2833 Relay Decoder Mute
MF Transport Type	Mute MF
DTMF Volume (-31 to 0 dB)	-11
NTE Max Duration	-1
CAS Transport Type	CASEventsOnly
DTMF Generation Twist	0
Echo Canceller	Disable

### 5.2 Fax/Modem/CID Settings

The configuration of the Fax/Modem/CID Settings are rather complex. Misconfiguration could easily cause some interoperability issues. However, for this example the settings are used as shown below.

Fax/Modem/CID Settings	
Fax Transport Mode	RelayEnable
Caller ID Transport Type	Mute
Caller ID Type	Standard Bellcore
V.21 Modem Transport Type	Disable
V.22 Modem Transport Type	Enable Bypass
V.23 Modem Transport Type	Enable Bypass
V.32 Modem Transport Type	Enable Bypass
V.34 Modem Transport Type	Enable Bypass
Fax Relay Redundancy Depth	0
Fax Relay Enhanced Redundancy Depth	4
Fax Relay ECM Enable	Enable
Fax Relay Max Rate (bps)	14400bps
Fax/Modem Bypass Coder Type	G711Alaw_64
Fax/Modem Bypass Packing Factor	1
Fax Bypass Output Gain	0
Modem Bypass Output Gain	0
Fax CNG Mode	Disable
CNG Detector Mode	Disable





### 5.3 RTP/RTCP Settings

The RTP/RTCP settings are used for this configuration example with their default values.

**RTP/RTCP Settings**

General Settings	
Dynamic Jitter Buffer Minimum Delay	10
Dynamic Jitter Buffer Optimization Factor	10
RTP Redundancy Depth	0
Packing Factor	1
Basic RTP Packet Interval	Default
RTP Directional Control	RTPTxRx
RFC 2833 TX Payload Type	96
RFC 2833 RX Payload Type	96
RFC 2198 Payload Type	104
Fax Bypass Payload Type	102
Enable RFC 3389 CN Payload Type	Enable
RTP Base UDP Port	6000
Comfort Noise Generation Negotiation	Disable
Analog Signal Transport Type	Disable
Remote RTP Base UDP Port	0
RTP Multiplexing Local UDP Port	0
RTP Multiplexing Remote UDP Port	0

RTCP XR Settings	
Enable RTCP XR	Disable
Burst Threshold	-1
Delay Threshold	-1
R-Value Delay Threshold	-1
Minimum Gap Size	16
RTCP XR Report Mode	Disable
RTCP XR Packet Interval	0
Disable RTCP XR Interval Randomization	Disable
RTCP XR Collection Server	
RTCP XR Collection Server Transport Type	Not Configured



## PSTN Settings

### 6.1 Trunk Settings

The trunk settings are used for this example as shown on the next screenshot.

General Settings	
Module ID	1
Trunk ID	1
Trunk Configuration State	Active
Protocol Type	BRI EURO ISDN

BRI Configuration	
Clock Master	Recovered
Auto Clock Trunk Priority	0
Trace Level	No Trace
ISDN Termination Side	User side
BRI Layer2 Mode	Point To Point
Q931 Layer Response Behavior	0x0
Outgoing Calls Behavior	0x400
Incoming Calls Behavior	0x0
General Call Control Behavior	0x0

PSTN Alert Timeout	
PSTN Alert Timeout	-1
Enable ECT	Disable
Local ISDN Ringback Tone Source	PBX
Set PI in Rx Disconnect Message	Not Configured
ISDN Transfer Capabilities	Not Configured
Progress Indicator to ISDN	Not Configured
Enable Receiving of Overlap Dialing	Disable
RTP Only Mode	Not Configured
B-channel Negotiation	Not Configured
Out-Of-Service Behavior	Default
Play Ringback Tone to Trunk	Not Configured

## Protocol Configuration

### 7.1 SIP General Parameters

The configuration of SIP general parameters are rather complex. Misconfiguration can easily cause interoperability issues. You need to ensure to configure the conform fax signaling method, SIP transport type and UDP/TCP ports.

However, for this example the settings are used mainly with their default values.



SIP General Parameters	
▼ SIP General	
PRACK Mode	Supported
Channel Select Mode	Cyclic Ascending
Enable Early Media	Enable
183 Message Behavior	Progress
Session-Expires Time	0
Minimum Session-Expires	90
Session Expires Method	Re-INVITE
Asserted Identity Mode	Disabled
Fax Signaling Method	T.38 Relay
⚡ Detect Fax on Answer Tone	Initiate T.38 on Preamble
SIP Transport Type	UDP
SIP UDP Local Port	5060
SIP TCP Local Port	5060
SIP TLS Local Port	5061
Enable SIPS	Disable
Enable TCP Connection Reuse	Enable
TCP Timeout	0
SIP Destination Port	5060
Use user=phone in SIP URL	No
Use user=phone in From Header	No
Use Tel URI for Asserted Identity	Disable
Tel to IP No Answer Timeout	180
Enable Remote Party ID	Disable
Add Number Plan and Type to RPI Header	Yes
Enable History-Info Header	Disable
Use Source Number as Display Name	No
Use Display Name as Source Number	No
Enable Contact Restriction	Disable
Play Ringback Tone to IP	Don't Play
Play Ringback Tone to Tel	Play According to Early Media
Use Tgrp information	Disable
Enable GRUU	Disable
User-Agent Information	
SDP Session Owner	AudiocodesGW
Play Busy Tone to Tel	Don't Play
Subject	
Multiple Packetization Time Format	None
Enable Semi-Attended Transfer	Disable
3xx Behavior	Forward
Enable P-Charging Vector	Disable
Enable VoiceMail URI	Disable
Retry-After Time	0
Enable P-Associated-URI Header	Disable
Source Number Preference	
Forking Handling Mode	Sequential handling
Enable Reason Header	Enable
▼ Retransmission Parameters	
SIP T1 Retransmission Timer [msec]	500
SIP T2 Retransmission Timer [msec]	4000
SIP Maximum RTX	7



## 7.2 Proxy Sets Table

The proxy address within the Proxy Set Table dialog is related to the application server with the XCAPI.

Proxy Sets Table

Proxy Set ID: 1

	Proxy Address	Transport Type
1	192.168.1.61	UDP
2		
3		
4		
5		

Enable Proxy Keep Alive: Disable  
 Proxy Keep Alive Time: 60  
 Proxy Load Balancing Method: Disable  
 Is Proxy Hot Swap: No

## 7.3 Coders

The codecs of the coders table are used as shown on the next screenshot.

Coders Table

Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression
G.711A-law	20	64	8	Disabled
G.711U-law	20	64	0	Disabled
T.38	N/A	N/A	N/A	N/A



## 7.4 DTMF & Dialing

The DTMF & Dialing configuration needs to conform to your VoIP environment. Misconfiguration can easily cause interoperability issues. For this example the settings are used as shown below.

DTMF & Dialing	
Max Digits In Phone Num	3
Inter Digit Timeout for Overlap Dialing [sec]	4
Declare RFC 2833 in SDP	Yes
1st Tx DTMF Option	RFC 2833
2nd Tx DTMF Option	INFO(Cisco)
RFC 2833 Payload Type	101
⚡ Digit Mapping Rules	
Default Destination Number	1000
Special Digit Representation	Special



## 7.5 SIP Advanced Parameters

The SIP advanced parameters are used as shown on the next screenshot.

**Advanced Parameters**

<b>General</b>	
IP Security	Disable
Filter Calls to IP	Don't Filter
Enable Digit Delivery to Tel	Disable
Enable Digit Delivery to IP	Disable
RTP Only Mode	Disable
PSTN Alert Timeout	180
<b>Disconnect and Answer Supervision</b>	
Disconnect on Broken Connection	Yes
Broken Connection Timeout [100 msec]	100
Disconnect Call on Silence Detection	No
Silence Detection Period [sec]	120
Silence Detection Method	Voice/Energy Detectors
Enable Fax Re-Routing	Disable
<b>CDR and Debug</b>	
CDR Server IP Address	
CDR Report Level	None
Debug Level	5
<b>Misc. Parameters</b>	
Progress Indicator to IP	Not Configured
Enable X-Channel Header	Disable
Enable Busy Out	Disable
Default Release Cause	3
Max Number of Active Calls	150
Max Call Duration [min]	0
Enable LAN Watchdog	Disable
Enable User-Information Usage	Disable
Delay After Reset [sec]	7
First Call Ringback Tone ID	-1



## 7.6 Routing General Parameters

The routing general parameters are used as shown on the next screenshot.

General Parameters	
Add Trunk Group ID as Prefix	No
Add Trunk ID as Prefix	No
Replace Empty Destination with B-channel Phone Number	No
Add NPI and TON to Called Number	No
Add NPI and TON to Calling Number	No
IP to Tel Remove Routing Table Prefix	No
Source IP Address Input	SIP Contact Header
Enable Alt Routing Tel to IP	Enable
Alt Routing Tel to IP Mode	None
Alt Routing Tel to IP Connectivity Method	ICMP Ping
Alt Routing Tel to IP Keep Alive Time	60
Max Allowed Packet Loss for Alt Routing [%]	20
Max Allowed Delay for Alt Routing [msec]	250



## 7.7 Tel to IP Routing

The **Tel to IP Routing** for this example is used as described below.

- **SRC Trunk Group ID** and the **Source Phone Prefix** are set to **\***.
- The **Dest. Phone Prefix** is used for this example with the value **8165\***. The prefix **816** is the local BRI access number in this example. The following digits **5\*** allow to route all calling numbers starting with **8165** and any following suffix to the application/XC-API.
- The **Transport Type** is set to **UDP**.
- The **Dest. IPGroup ID** is set to the IP group number one.
- The **IP Profile ID** is set to zero.

Tel to IP Routing

Routing Index: 1-10

Tel To IP Routing Mode: Route calls after manipulation

	Src. Trunk Group ID	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Port	Transport Type	Dest. IPGroup ID	IP Profile ID	Status
1	*	8165*	*			UDP	1	0	n/a
2						Not Configured			
3						Not Configured			
4						Not Configured			
5						Not Configured			
6						Not Configured			
7						Not Configured			
8						Not Configured			
9						Not Configured			
10						Not Configured			





## 7.8 Trunk Group

The trunk group table must be related to the according modules, trunk relations, channels, trunk group IDs and IP profile IDs. For this example the trunk group configuration is used as shown on the next screenshot.

Trunk Group Table

Add Phone Context As Prefix: Disable

Trunk Group Index: 1-12

Group Index	Module	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	IP Profile ID
1	Module 1 BRI	1	1	1-2	8165		0
2							
3							
4							
5							
6							
7							
8							
9							
10							
11							
12							

## TDM Bus Settings

The TDM bus settings needs to be conform to your PSTN/provider configurations. This example uses the settings as shown next.

TDM Bus Settings

PCM Law Select: ALaw

TDM Bus Type: Framers

Idle PCM Pattern: 213

Idle ABCD Pattern: 0x0F

TDM Bus Local Reference: 1

TDM Bus PSTN Auto Clock: Disable

TDM Bus Clock Source: Network



## 8.1 Digital Gateway Parameters

The configuration of digital gateway parameters is rather complex. Misconfiguration can easily cause interoperability issues. For this example the parameters are used as shown next.

Digital Gateway Parameters	
B-channel Negotiation	Exclusive
Swap Redirect and Called Numbers	No
MFC R2 Category	1
Disconnect Call on Busy Tone Detection (CAS)	Enable
Disconnect Call on Busy Tone Detection (ISDN)	Disable
Enable TDM Tunneling	Disable
Send Screening Indicator to IP	Not Configured
Send Screening Indicator to ISDN	Not Configured
Add IE in SETUP	
Trunk Groups to Send IE	
Enable User-to-User IE for Tel to IP	Enable
Enable User-to-User IE for IP to Tel	Enable
Enable ISDN Tunneling Tel to IP	Disable
Enable QSIG Tunneling	Disable
Enable ISDN Tunneling IP to Tel	Disable
ISDN Transfer on Connect	Alert
Remove CLI when Restricted	No
Remove Calling Name	Disable
Default Cause Mapping From ISDN to SIP	0
Add Prefix to Redirect Number	
Copy Destination Number to Redirect Number	Don't copy
Enable Calling Party Category	Disable
Digital Out-Of-Service Behavior	Default
<b>MLPP</b>	
MLPP Default Namespace	DSN
Default Call Priority	0
Preemption tone Duration	3



## Supplementary Services

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

### 9.1 Call Transfer

For appropriate call transfer interworking a **Media Realm** has to be specified.

Index	Media Realm Name	IPv4 Interface Name	IPv6 Interface Name	Port Range Start	Number Of Media Session Legs	Port Range End
1	testMediaRealm	Voice	None	-1	-1	-1

Default Media Realm Name:

This **Media Realm** has to be indexed to the according **SRD Table**.

Index	Type	Description	Proxy set ID	SIP group name	IP profile ID
1	SERVER	XCAPI	1		2

Index	Enable Proxy Keep Alive
0	Disable
1	Disable



The SRD index and media realm must be related to the IP group table.

IP Group Table Basic Parameter List ▲

Index: 1

**Common Parameters**

Type	SERVER
Description	XCAPI
Proxy Set ID	1
SIP Group Name	
Contact User	
SRD	0
Media Realm	testMediaRealm
IP Profile ID	2

**Gateway Parameters**

Always Use Route Table	No
Routing Mode	Not Configured
SIP Re-Routing Mode	Standard
Enable Survivability	Disable
Serving IP Group ID	

Also the referring proxy set tables has to be SRD indexed.

Proxy Sets Table

Proxy Set ID: 1

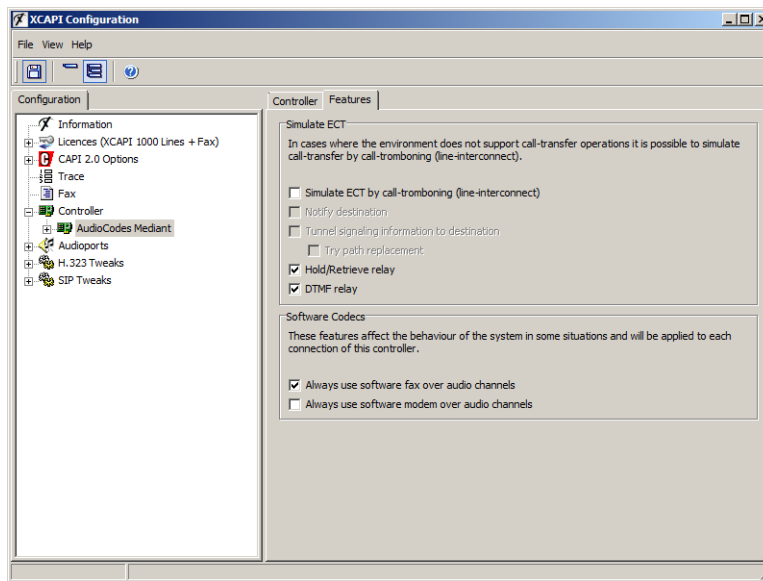
	Proxy Address	Transport Type
1	xcapi.te-systems.de	
2		
3		
4		
5		

**Parameters**

Enable Proxy Keep Alive	Disable
Proxy Keep Alive Time	60
Proxy Load Balancing Method	Disable
Is Proxy Hot Swap	No
Proxy Redundancy Mode	Not Configured
SRD Index	0
Classification Input	IP only



For the XCAPI configuration, please ensure that the **Simulated ECT by call-tromboning (line-interconnect)** parameter of the XCAPI controller **Features** dialog is disabled for supporting call transfer via SIP refer.





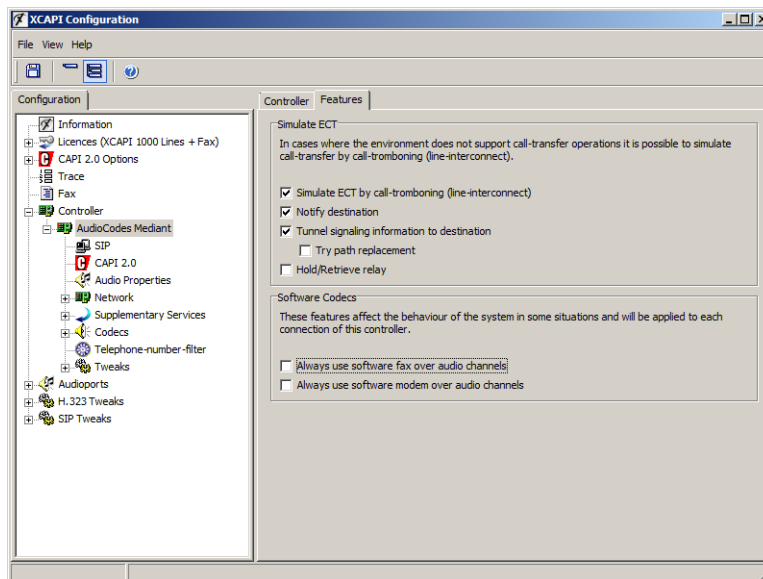
## Fax Services

This chapter shows the necessary settings when using Fax services via the T.38 protocol stack or the SoftFax method.

### 10.1 T.38

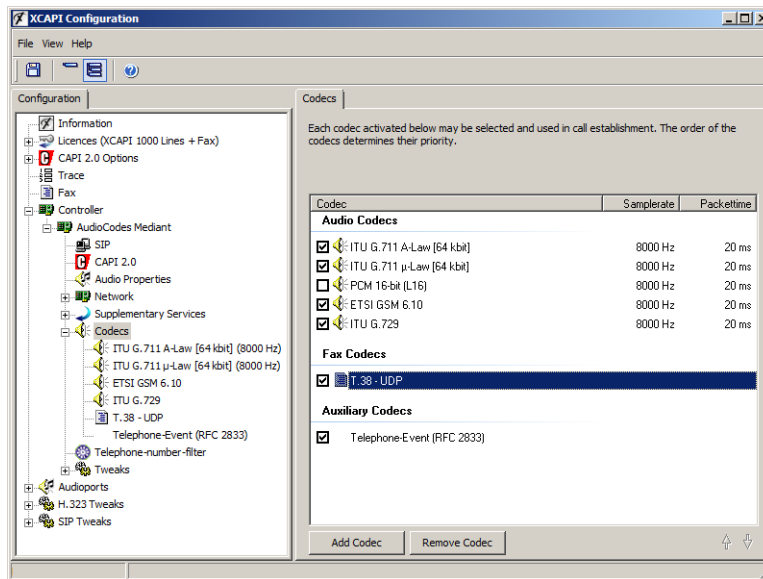
When using the T.38 protocol you have to enable the **T.38** method, as shown in the Coders chapter starting on [page 12](#) and the SIP General Parameters chapter on [page 10](#).

It is also required to disable the **Always use software fax over audio channels** option within the XCAPI controller **Features** dialog.





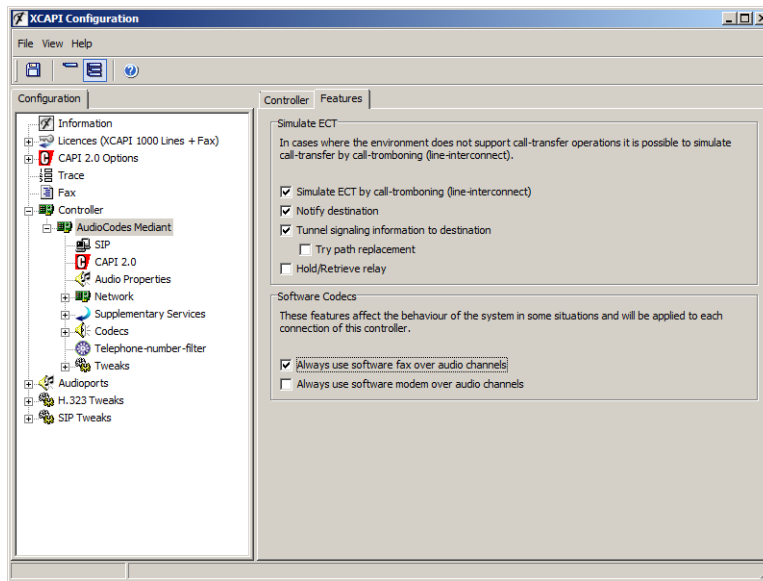
Ensure that **T.38 - UDP** codec, which should be set by default when using the XCAPI controller wizard, is set and enabled within the **Codecs** configuration dialog of the XCAPI controller.





## 10.2 SoftFax

In the **SoftFax** mode, the XCAPI simulates an analog Fax device by transmitting modulated facsimile signals modem-like via audio channels. For this you have to enable the **Always use software fax over audio channels** option within the **Features** configuration dialog.







### 10.3 SoftFax in virtual environments

Within virtual environments you might improve the jitter buffer behavior within the AudioCodes gateway configuration by adapting the **Dynamic Jitter Buffer Minimum Delay**.

You should consider using different profile definitions instead of the general RTP/RTPc settings for voice and fax services in order to prevent unwanted delay in voice communications.

The next screenshot shows the **IP Profile Settings** with using a **Dynamic Jitter Buffer Minimum Delay** [msec] valued with **80**.

The screenshot shows the 'IP Profile Settings' window with the following configuration:

Basic Parameter List	
Profile ID	2
Profile Name	XCAPI Profile
▼ Profile Parameters	
Profile Preference	2
Fax Signaling Method	G.711 Transport
Dynamic Jitter Buffer Minimum Delay [msec]	80
Dynamic Jitter Buffer Optimization Factor	10
RTP IP DiffServ	46
Signaling DiffServ	40
Voice Volume (-32 to 31 dB)	0
Input Gain (-32 to 31 dB)	0
RTP Redundancy Depth	0
Remote RTP Base UDP Port	0
CNG Detector Mode	Disable
Modems Transport Type	Enable Bypass
NSE Mode	Disable
Play Ringback Tone to IP	Don't Play
Enable Early Media	Enable
Progress Indicator to IP	Not Configured
Echo Canceler	Disable
Media Security Behavior	Preferable
Number of Calls Limit	.1
Copy Destination Number to Redirect Number	Disable
Disconnect on Broken Connection	Yes
Enable Hold	Enable
▼ Coder Group	
Coder Group	Default Coder Group

Submit



The according configuration dialogs **Tel to IP**, **IP to Trunk Group Routings** and **Trunk Group Tables** have to be in relation with the according **IP Profile ID**.

Trunk Group Table

Group Index	Module	From Trunk	To Trunk	Channels	Phone Number	Trunk Group ID	IP Profile ID
1	Module 1 BRI	1	1	1-2	8165		2
2							
3							
4							
5							
6							
7							
8							
9							
10							
11							
12							

Submit

Tel to IP Routing

Advanced Parameter List

Routing Index: 1-10

Tel To IP Routing Mode: Route calls after manipulation

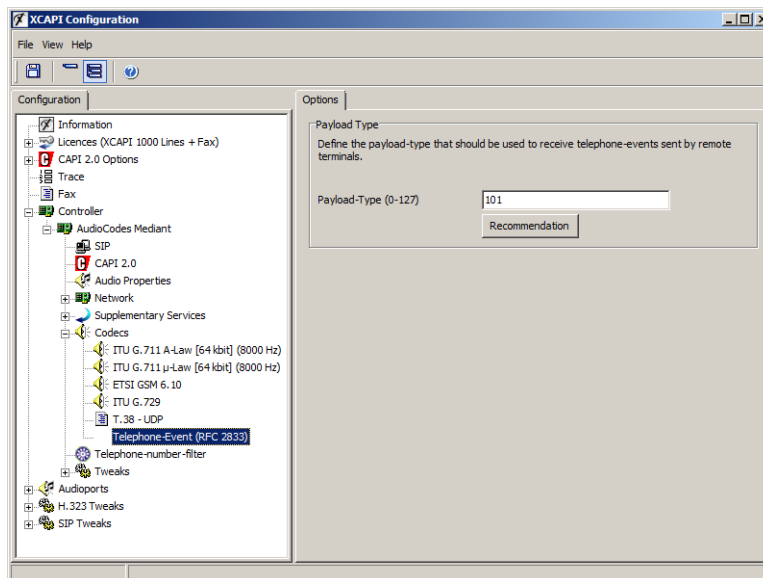
	Dest. Phone Prefix	Source Phone Prefix	Dest. IP Address	Transport Type	Dest. IP Group ID	IP Profile ID	Status
1	8165*			UDP	1	2	n/a
2				Not Configured			
3				Not Configured			
4				Not Configured			
5				Not Configured			
6				Not Configured			
7				Not Configured			
8				Not Configured			
9				Not Configured			
10				Not Configured			

Submit



## DTMF

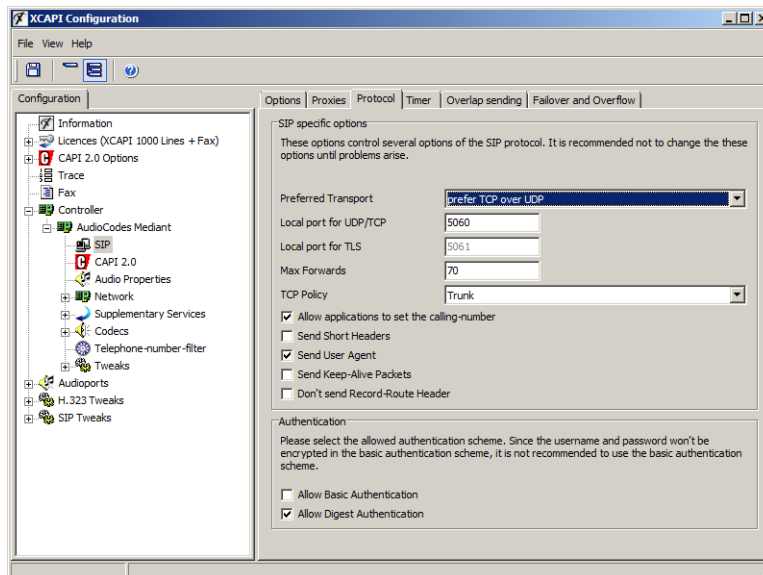
The payload type for signaling Telephone Events via RFC 2833 is by default set to value 101. For DTMF interoperability please ensure that the DTMF Transport Method, as shown in the chapter **Coders** on [page 12](#), is selected as **Out-of-Band using RTP** method and that the Payload Type is also set to 101.





## SIP via TCP

If you need XCAPI-sided SIP trunking via TCP you have to ensure that the preferred transport protocol is set to **prefer TCP over UDP**. The **TCP Policy** parameter must be set to **Trunk**. Ensure that the AudioCodes gateway configuration is conform to the XCAPI controller settings, in meaning of the correct port and transport protocol settings.





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TE-SYSTEMS GmbH

**Managing Directors** Andreas Geiger  
Oliver Körber

**Address** Max-von-Laue-Weg 19  
D-38448 Wolfsburg  
Germany

**Tel.** +49 5363 8195-0  
**Fax** +49 5363 8195-999

**E-Mail** [info@te-systems.de](mailto:info@te-systems.de)  
**Internet** [www.te-systems.de](http://www.te-systems.de)  
[www.xcapi.de](http://www.xcapi.de)