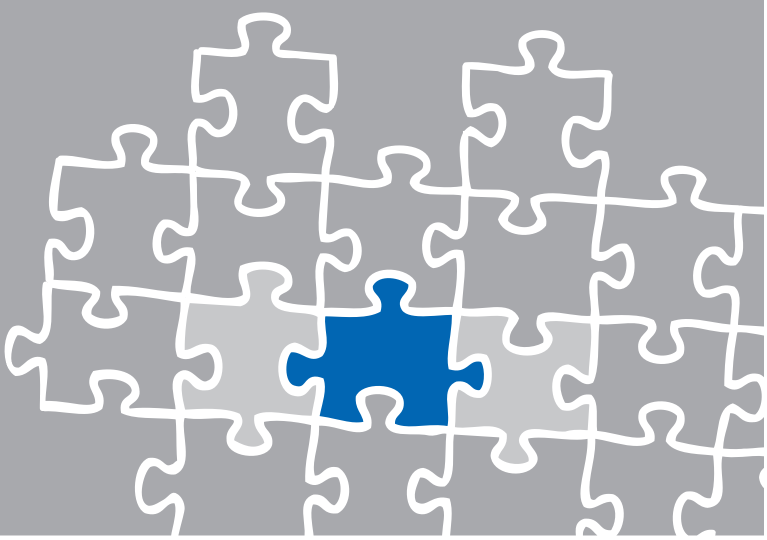
TechNote

AudioCodes Mediant Series

August 5, 2013

voiceoverIP



www.te-systems.de

TE-SYSTEMS
competence in e-communications.

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.

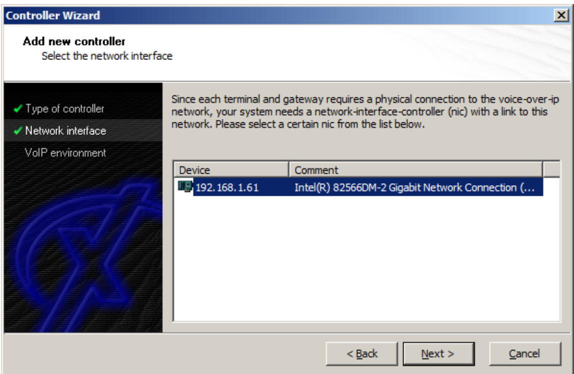
If you've just installed the XCAPI and start the configuration tool for the first time, the **XCAPI Controller Wizard** will pop up automatically. This will also happen if there's no controller configured at all.

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.

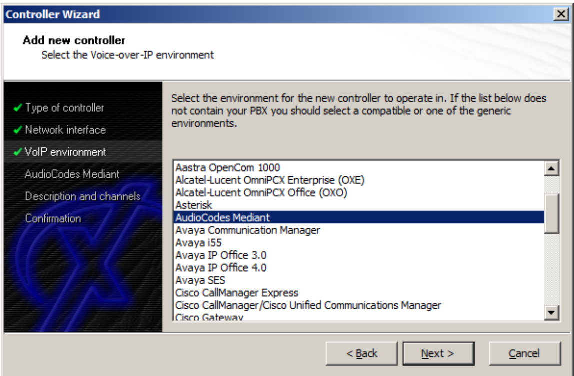
2.1 Network Interface

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



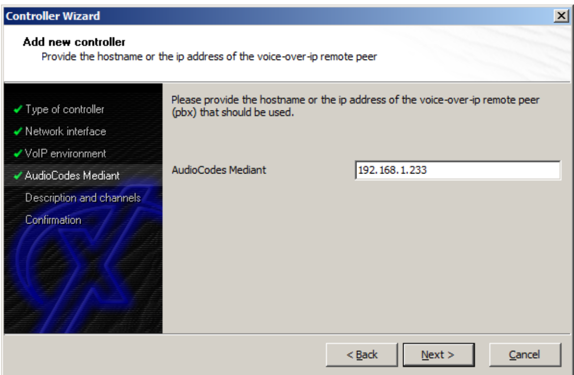
2.2 Voice-over-IP Environment

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

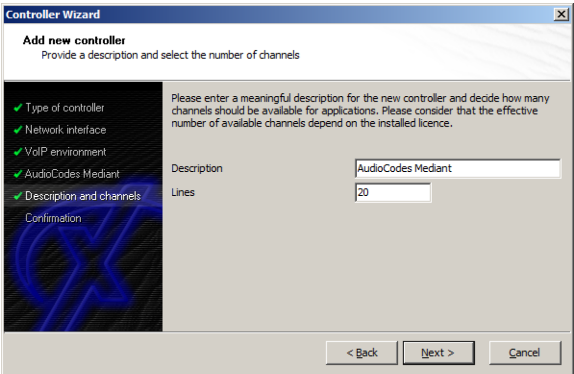
In the dialog **Network Address** please provide the IP address of your AudioCodes gateway.



2.4 Description and Channels

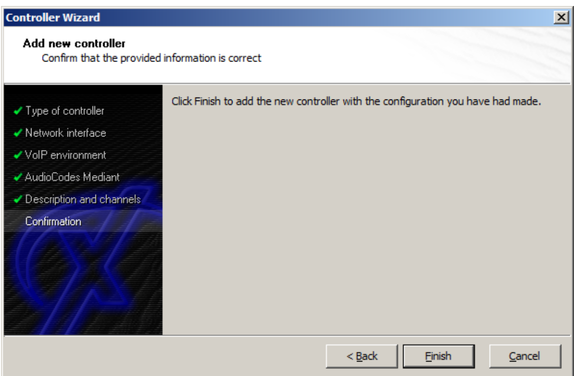
That's about all information that has to be configured with the XCAPI. The next-to-final dialog of the Controller Wizard allows you to configure a meaningful description for the controller you're going to create. This isn't really used anywhere, so you can enter a text of your choice here.

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.



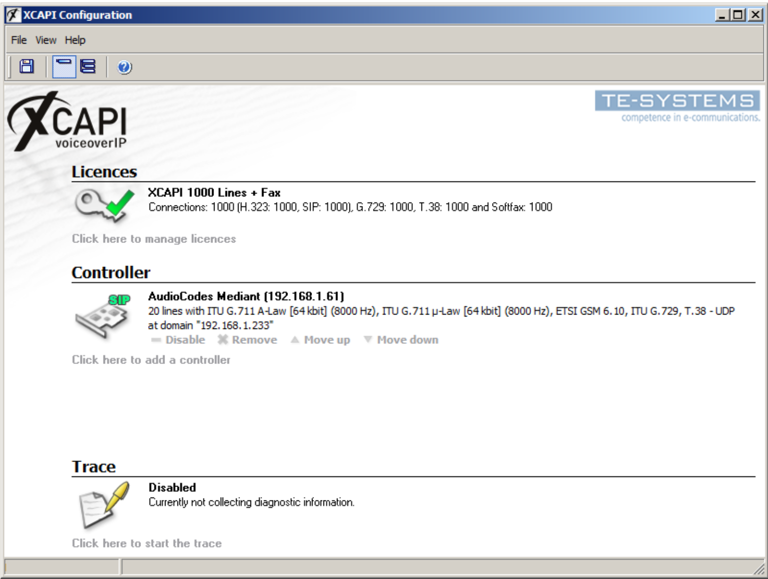
2.5 Confirmation

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

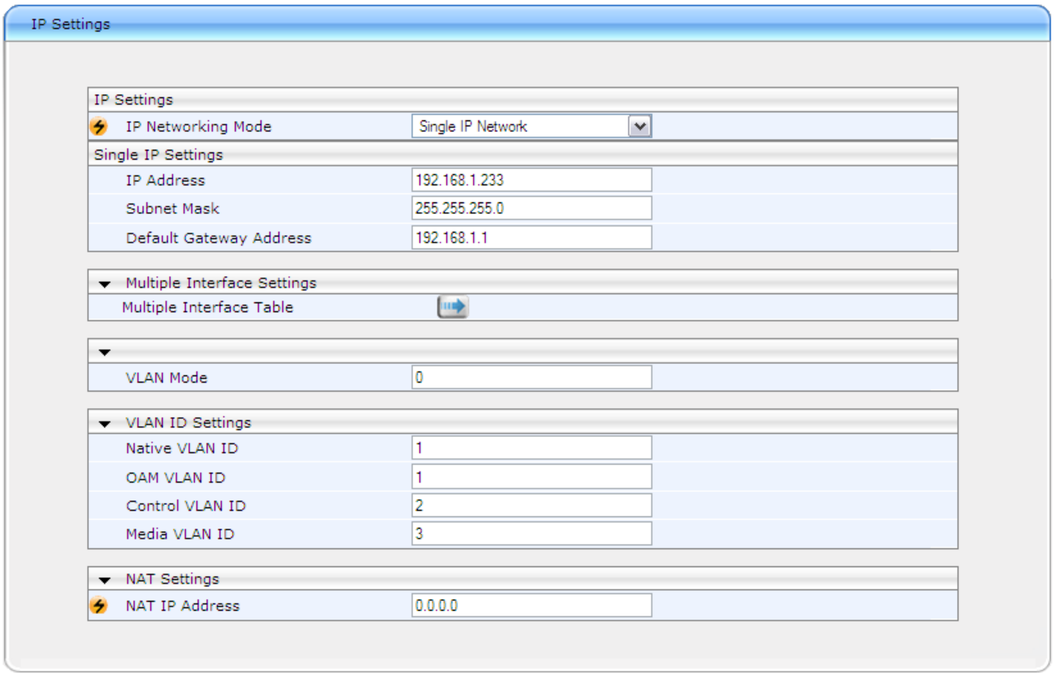
In order to establish the communication between the XCAPI and the gateway using the SIP protocol, you need to declare the XCAPI as an IP route with its according dial rules within the Mediant configuration.

This example gives you an overview about the necessary configurations.

Network Settings

4.1 IP Settings

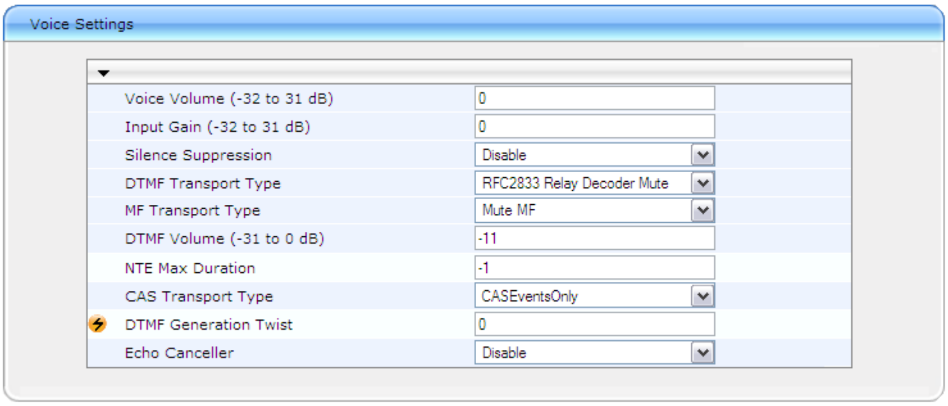
The IP settings of the AudioCodes Mediant gateway are used for this example as shown below.



Media Settings

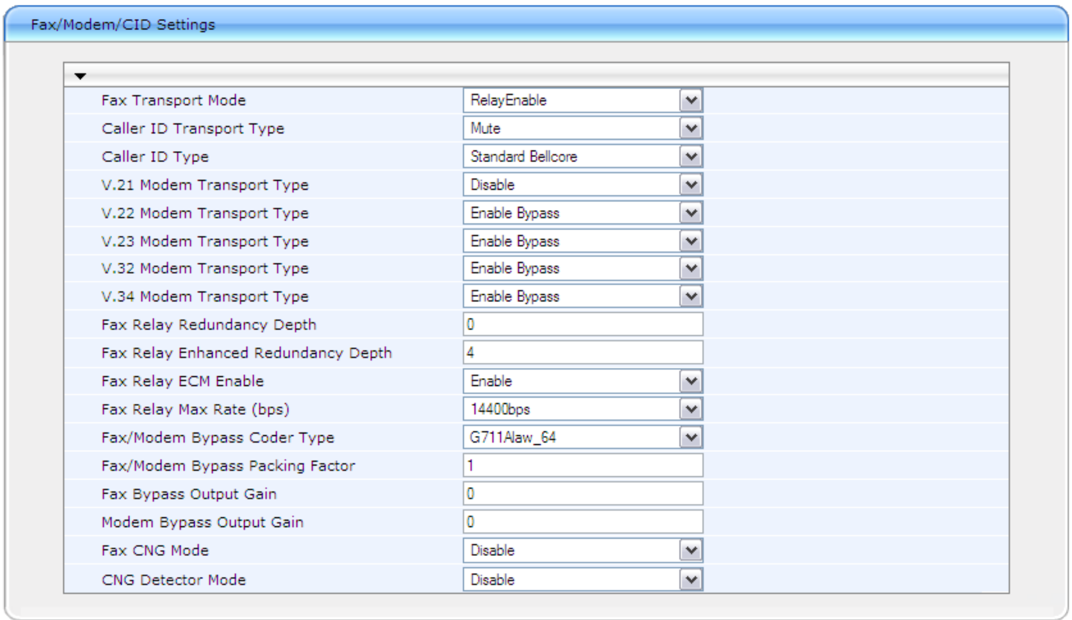
5.1 Voice Settings

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



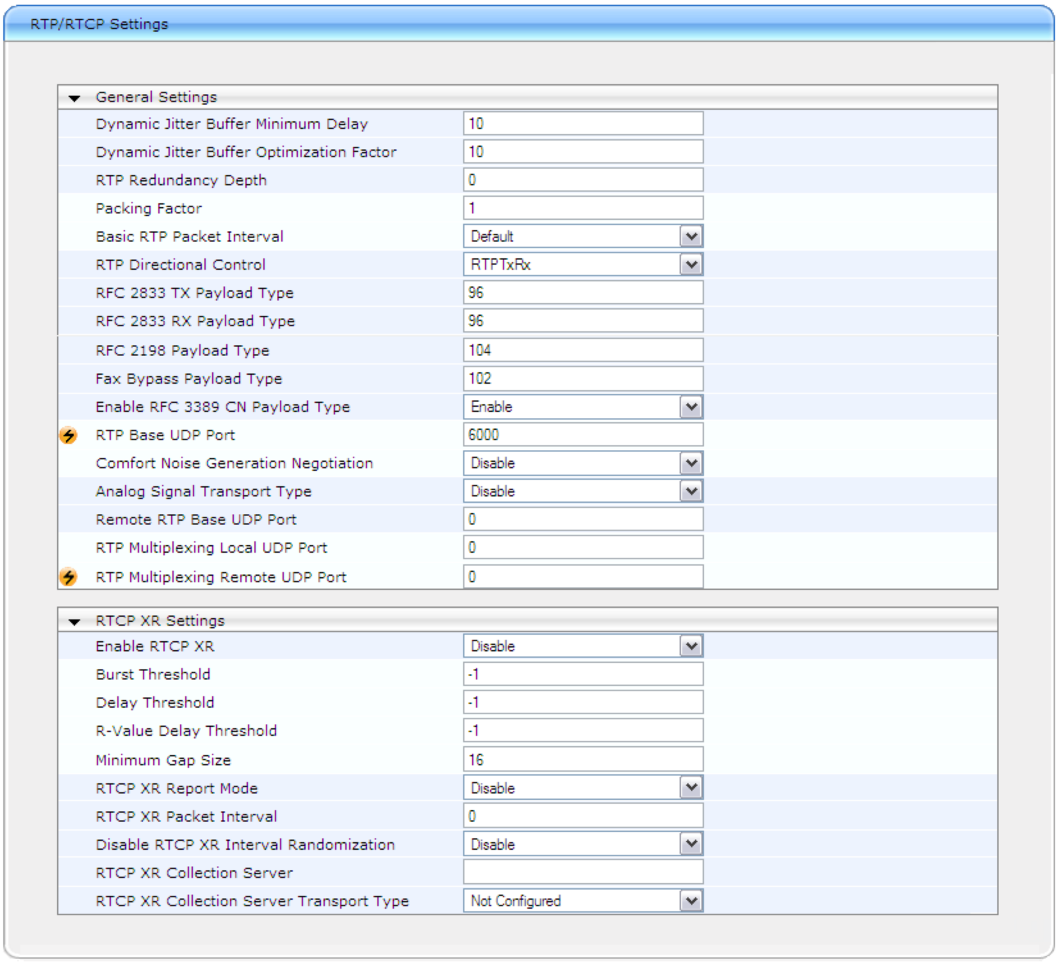
5.2 Fax/Modem/CID Settings

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.



5.3 RTP/RTPC Settings

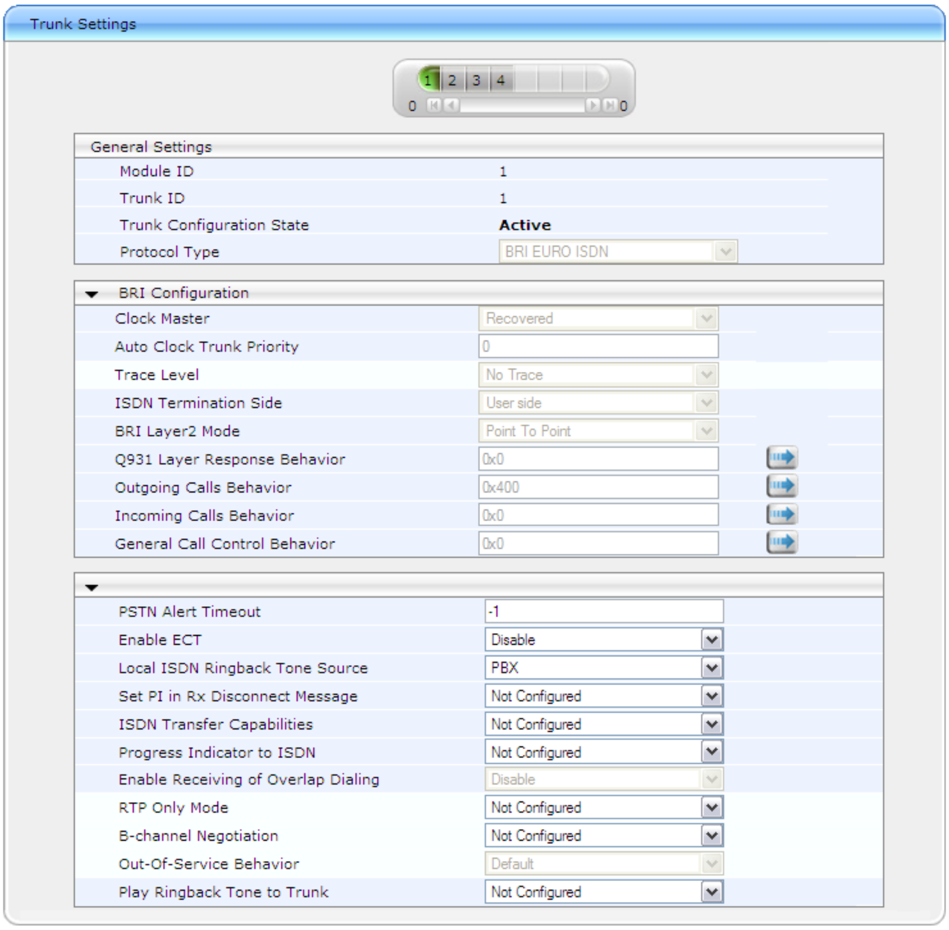
The RTP/RTPC settings are used for this configuration example with their default values.



PSTN Settings

6.1 Trunk Settings

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



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

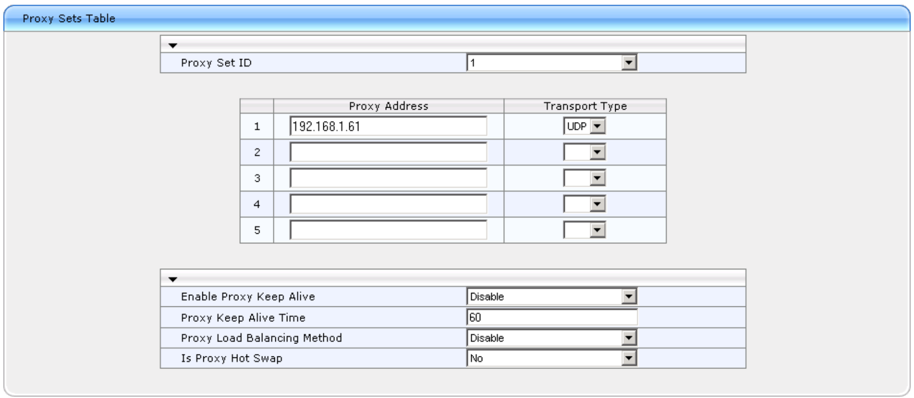
PRACK Mode	Supported
Channel Select Mode	Cyclic Ascending
Enable Early Media	Enable
183 Message Behavior	Progress
Session-Expires Time	0
Minimum Session-Expires	90
Session Expires Method	Re-INVITE
Asserted Identity Mode	Disabled
Fax Signaling Method	T.38 Relay
Detect Fax on Answer Tone	Initiate T.38 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

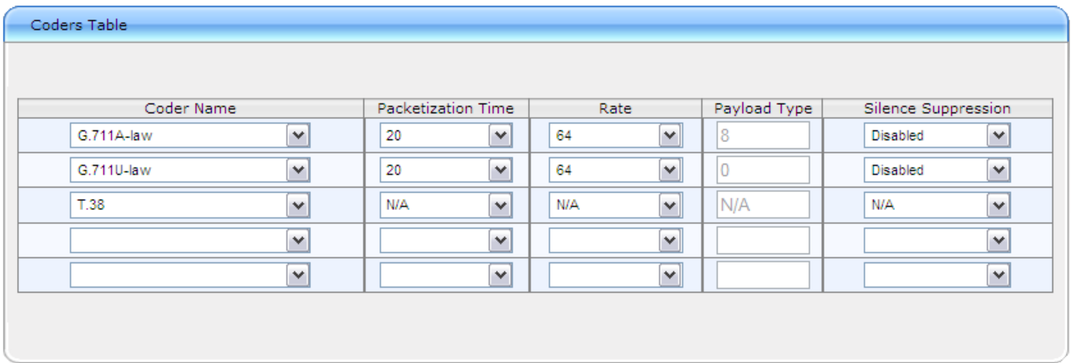
7.2 Proxy Sets Table

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



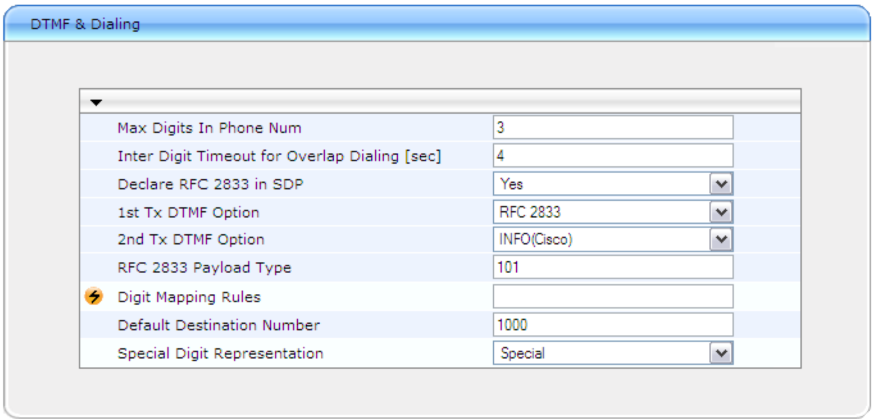
7.3 Coders

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



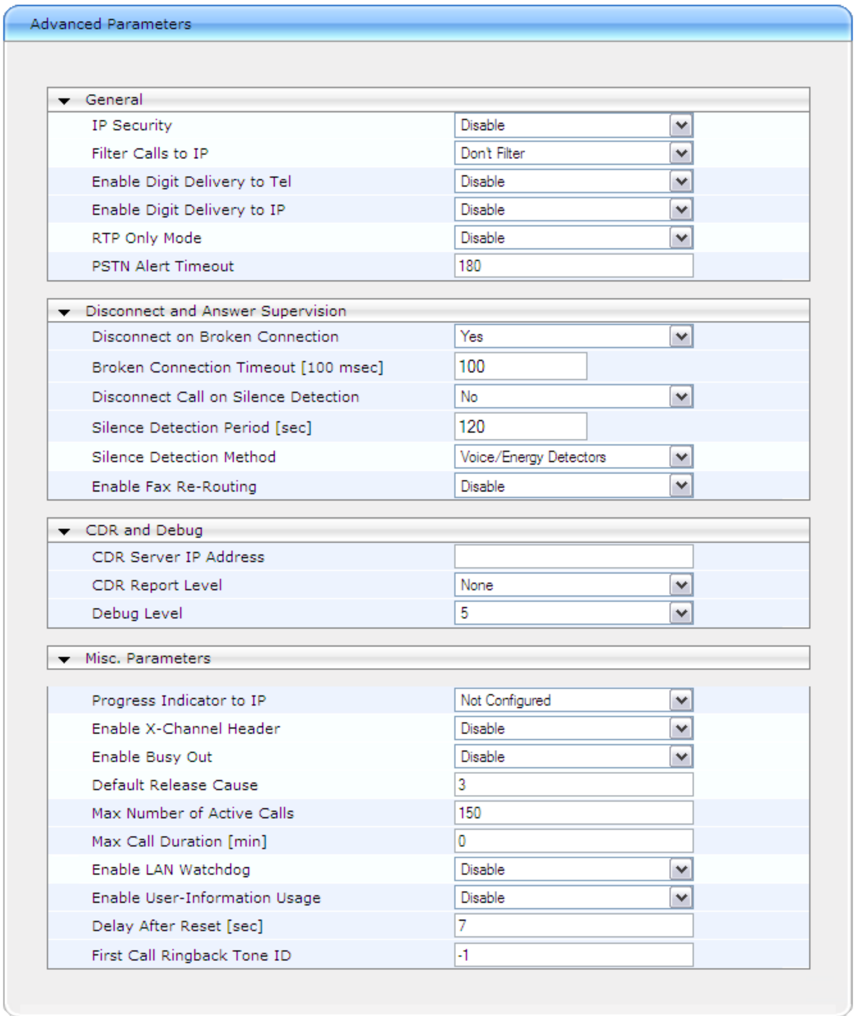
7.4 DTMF & Dialing

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.



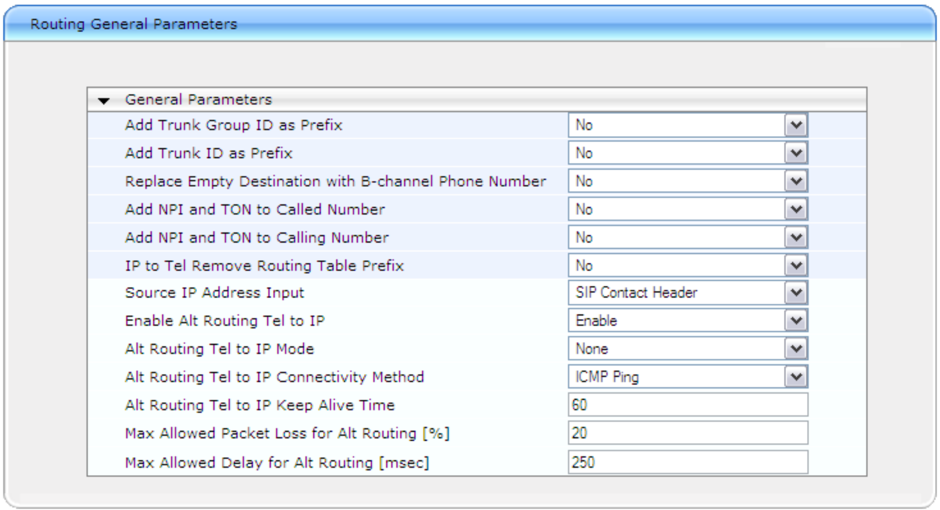
7.5 SIP Advanced Parameters

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



7.6 Routing General Parameters

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

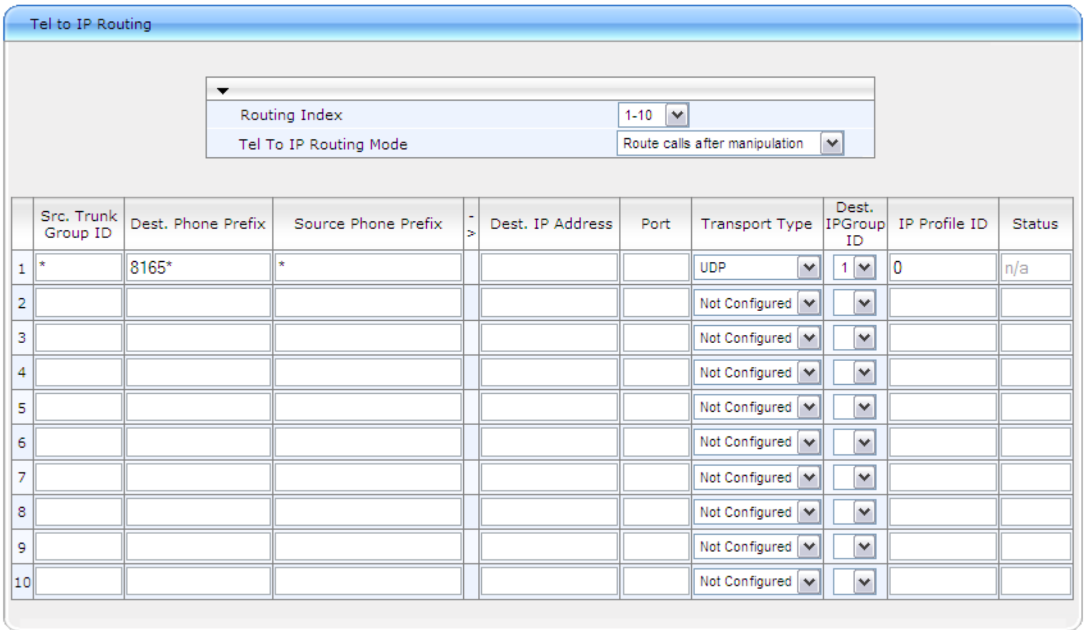

Routing General Parameters

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/XCAPI.
- 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.

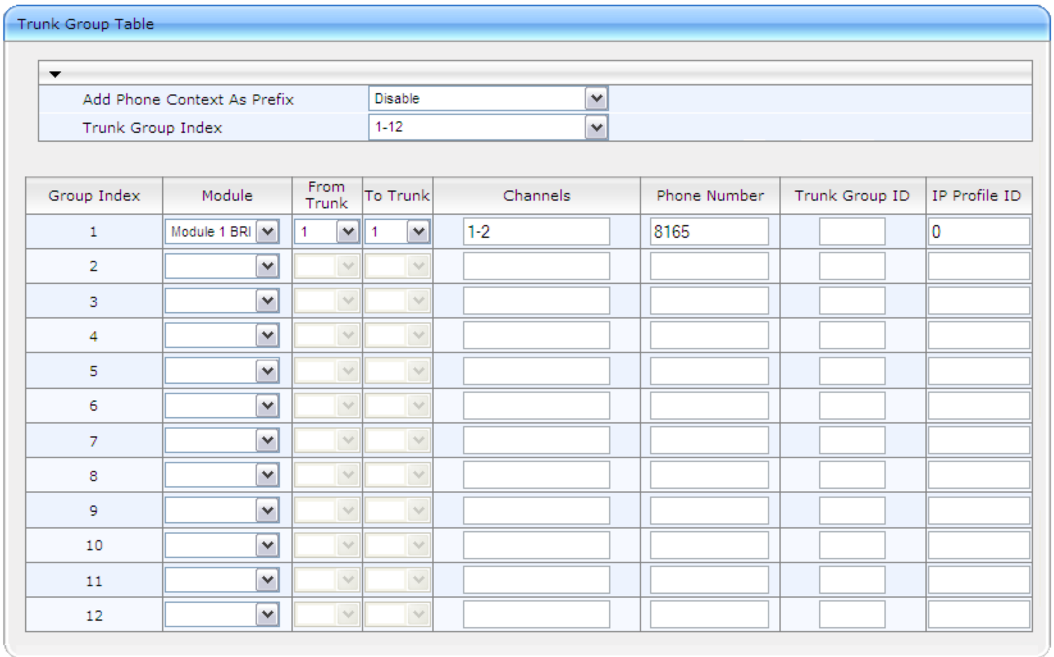

Tel to IP Routing

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			

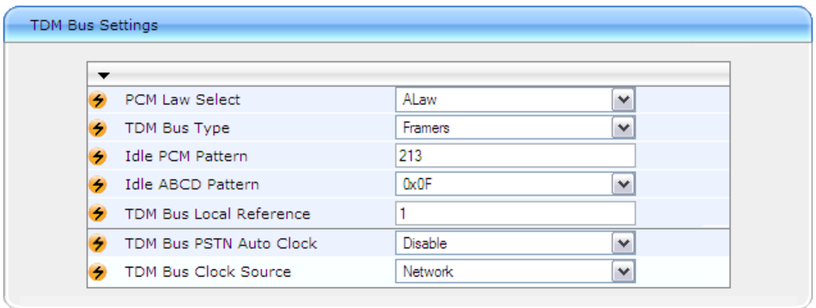
7.8 Trunk Group

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.



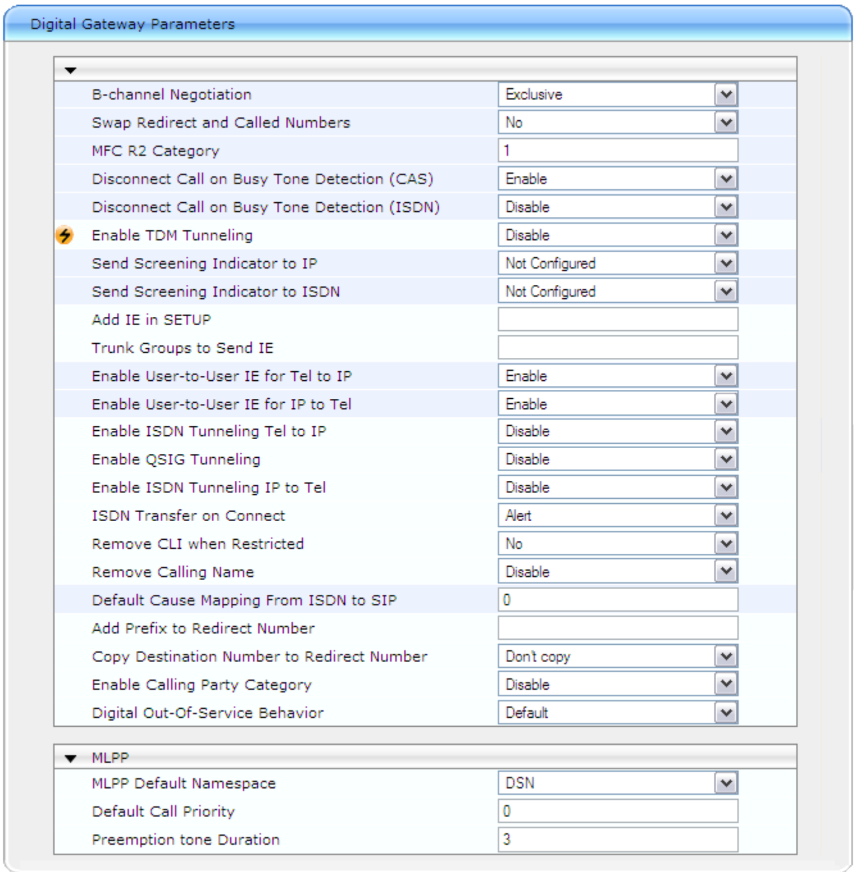
TDM Bus Settings

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



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

Parameter	Value
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

MLPP

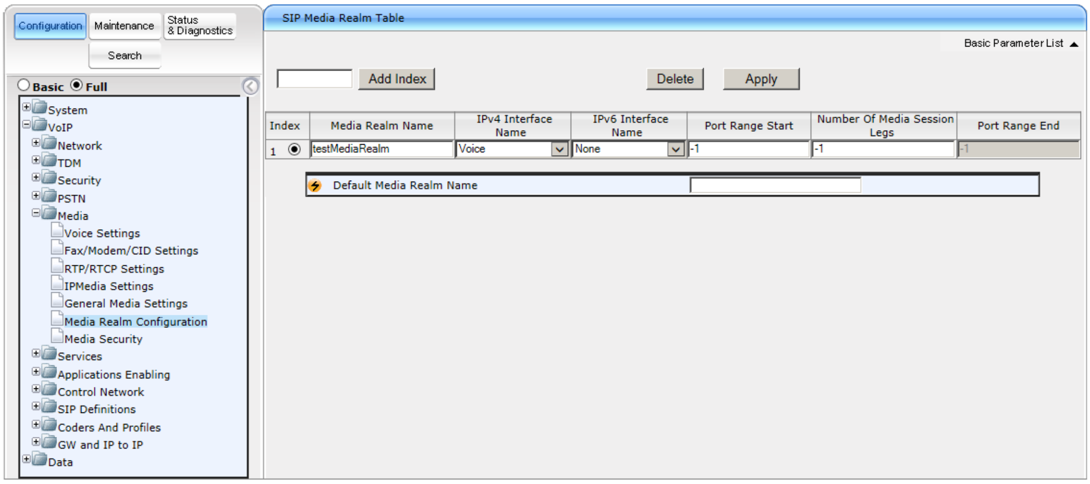
Parameter	Value
MLPP Default Namespace	DSN
Default Call Priority	0
Preemption tone Duration	3

Supplementary Services

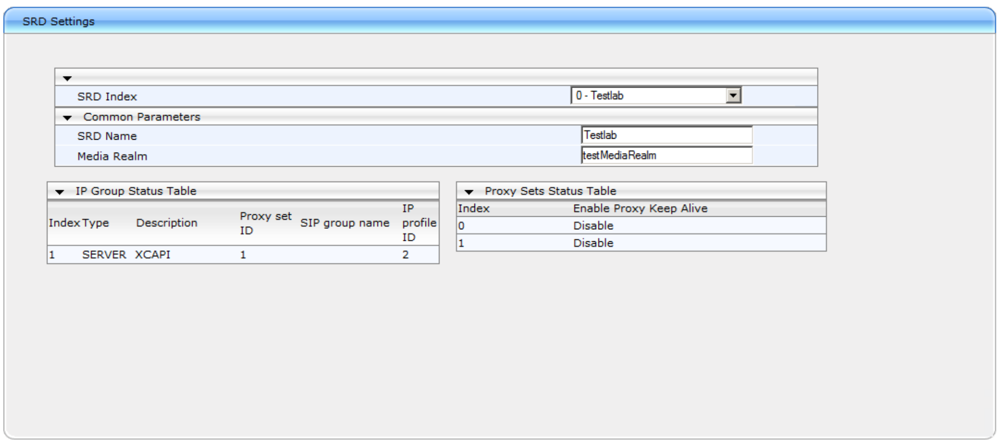
Please review the following chapters for some information on optimal supplementary services configuration.

9.1 Call Transfer

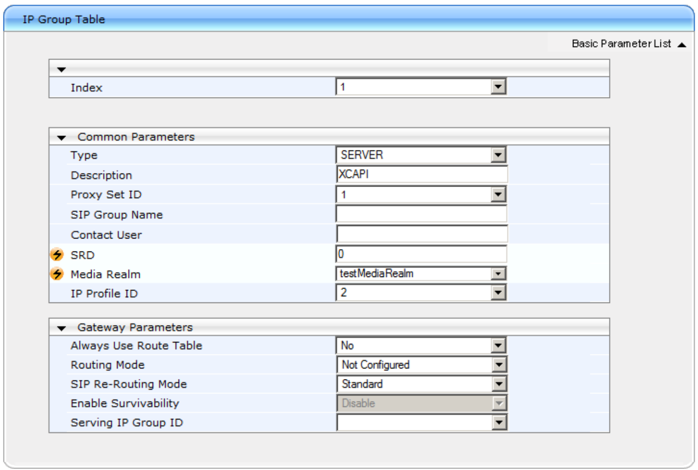
For appropriate call transfer interworking a **Media Realm** has to be specified.



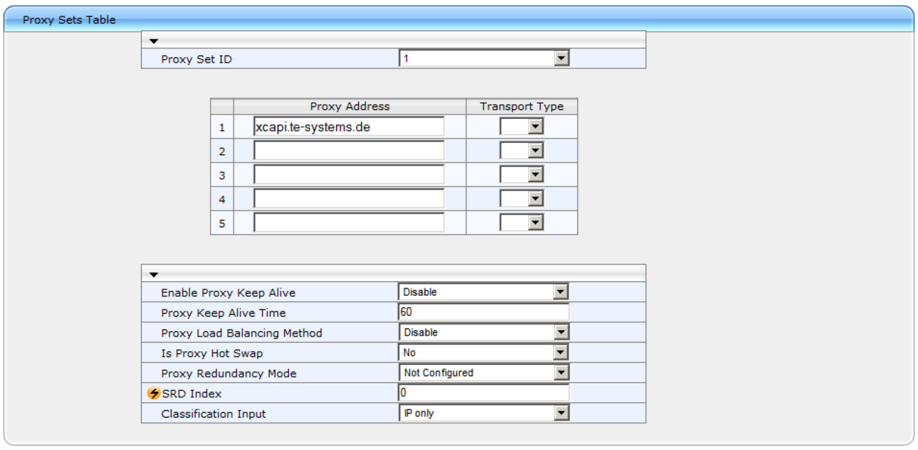
This **Media Realm** has to be indexed to the according **SRD Table**.



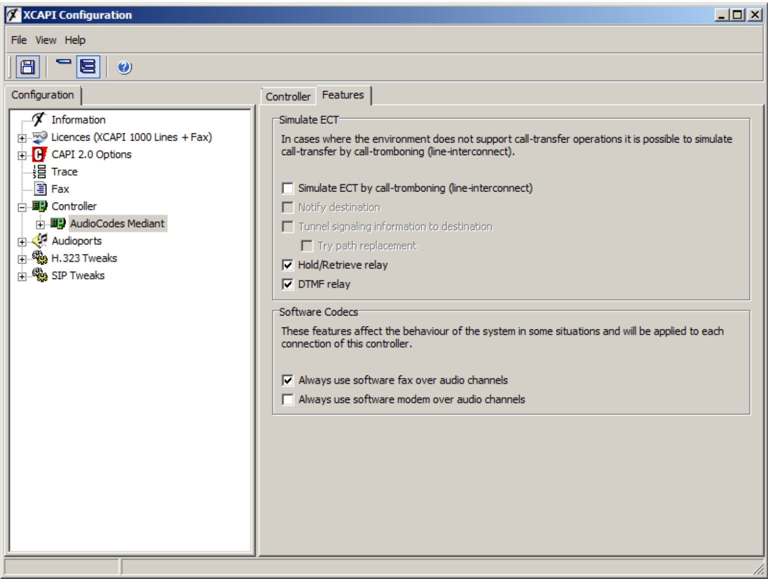
The SRD index and media realm must be related to the IP group table.



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



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.



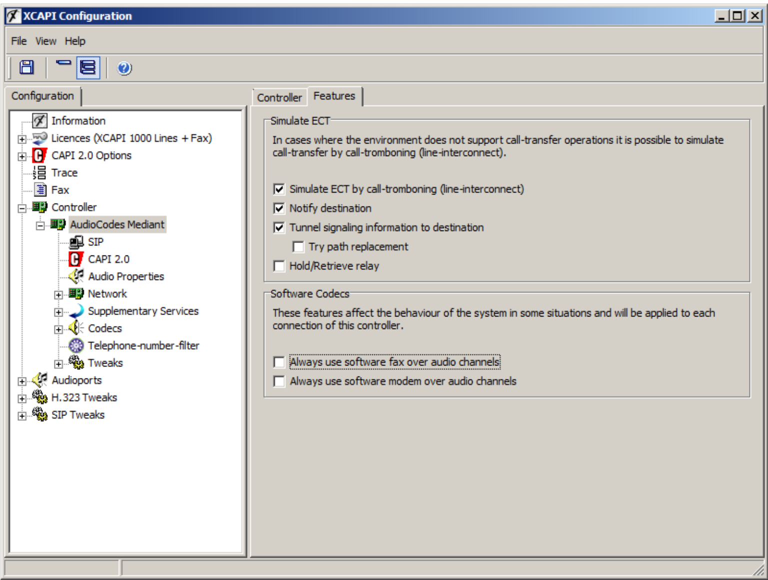
Fax Services

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

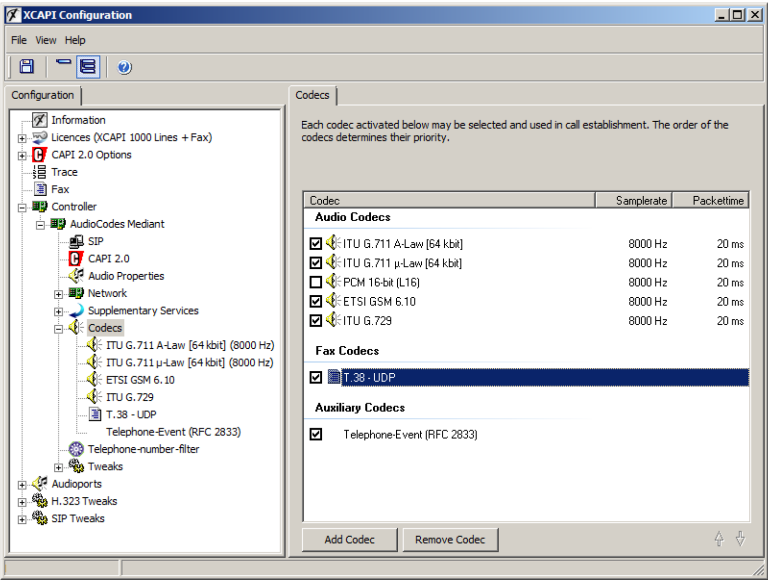
10.1 T.38

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.

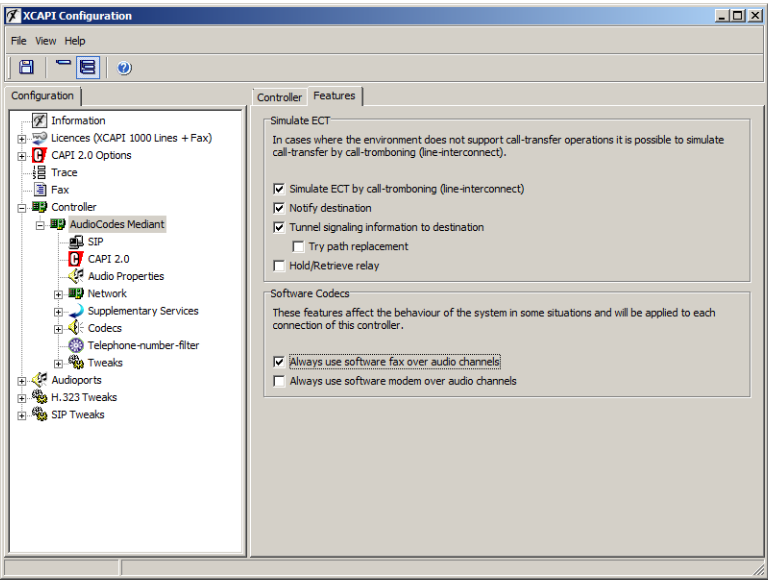


Ensure that **T.38 - UDP** codec, which should be set by default when using the XCAPI controller wizard, is set and enabled within the **Codecs** configuration dialog of the XCAPI controller.



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.

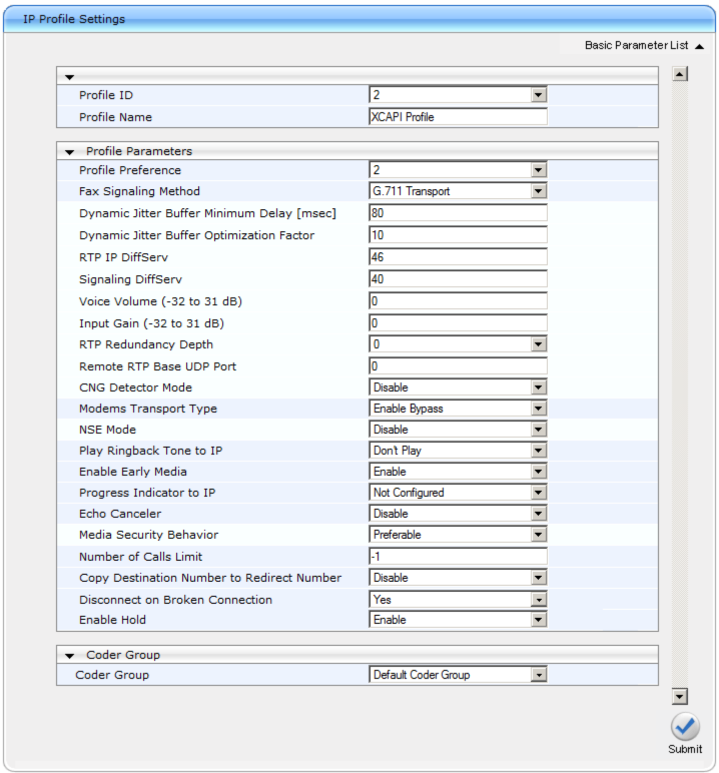


10.3 SoftFax in virtual environments

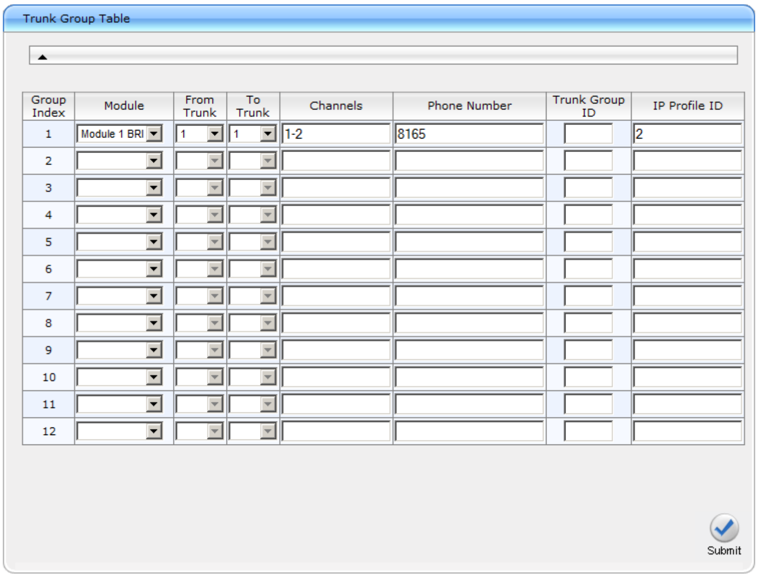
Within virtual environments you might improve the jitter buffer behavior within the AudioCodes gateway configuration by adapting the **Dynamic Jitter Buffer Minimum Delay**.

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**.



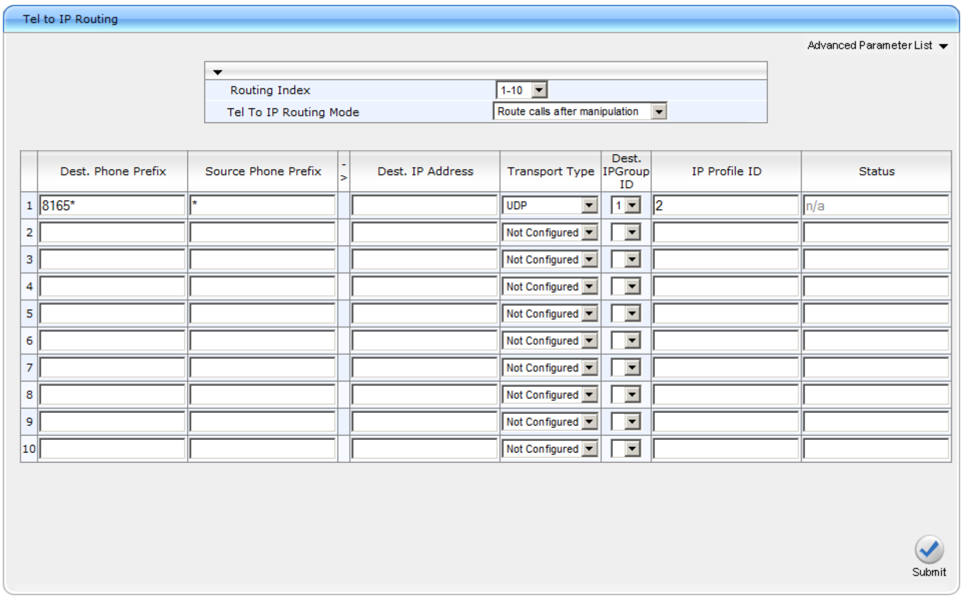
The according configuration dialogs **Tel to IP**, **IP to Trunk Group Routings** and **Trunk Group Tables** have to be in relation with the according **IP Profile ID**.



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

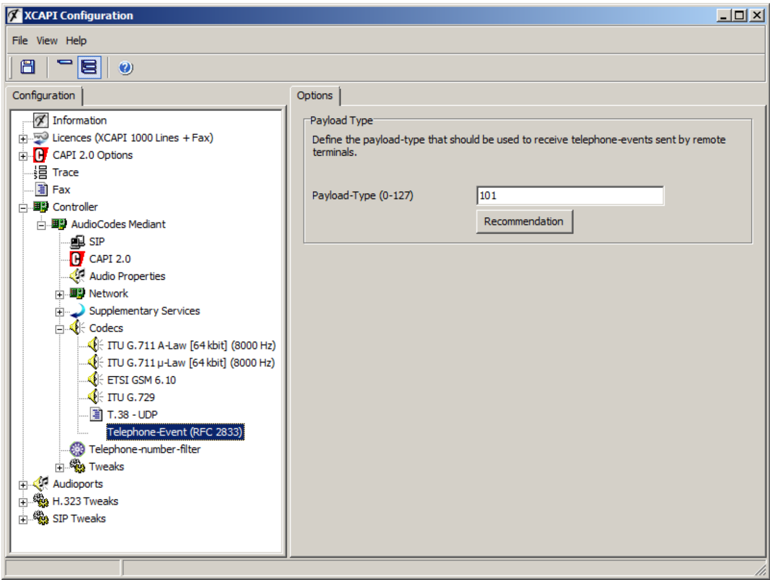
Routing Index	1-10
Tel To IP Routing Mode	Route calls after manipulation

	Dest. Phone Prefix	Source Phone Prefix	->	Dest. IP Address	Transport Type	Dest. IPGroup 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

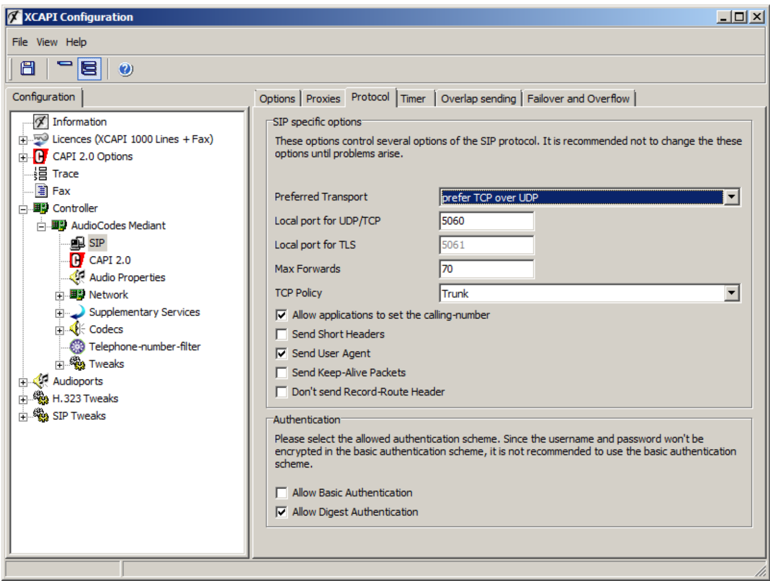
DTMF

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.



Exclusion of Liability

Trademarks

All names of products or services used are trademarks or registered trademarks (also without specified indication) of the respective private or legal persons and are therefore subject to legal regulations.

Third Party Disclaimer and Limitations

This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit. (http://www.openssl.org/)

This product includes cryptographic software written by Eric Young (eay@cryptsoft.com).

This product includes software written by Tim Hudson (tjh@cryptsoft.com).

This product includes source code derived from the RSA Data Security, Inc. MD2, MD4 and MD5 Message Digest Algorithms.

This product includes source code derived from the RFC 4634 Secure Hash Algorithm software.

Copyright-Notices

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
Internet www.te-systems.de
www.xcapi.de