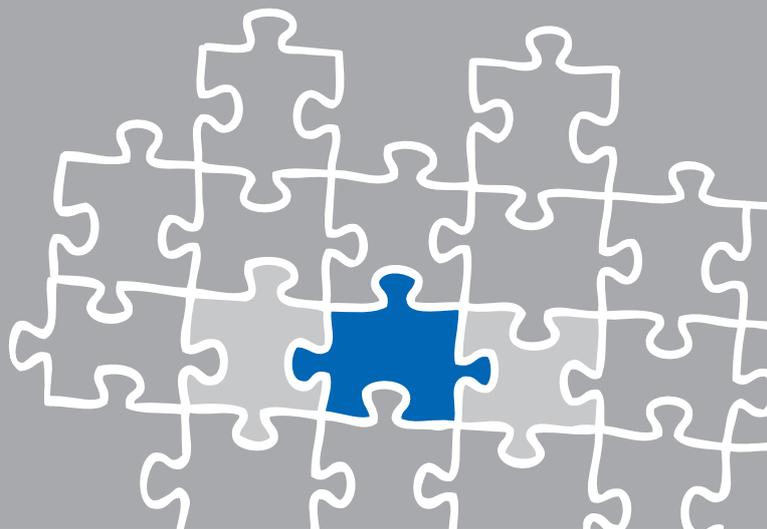


# TechNote

## Unify OpenScape Business V1

September 26, 2014





## Introduction

This document is intended to support you with the integration of the XCAPI, Version 3.5.0, into an existing environment of the Unify OpenScape Business series. The configurations and screenshots are here based on Unify OpenScape Business version osbiz\_v1\_R3.1.0\_470. In the following sections we describe the essential steps of configuration to allow for optimal cooperation of both the XCAPI and the Unify OpenScape Business. At this point we suppose that the Unify OpenScape Business, 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.

For XCAPI a quick starter guide (**XCAPI TechNote (en) - Quick Start Guide.pdf**) is available within our community download section at <http://www.xcapi.de>.

## XCAPI Configuration

Please start up the XCAPI configuration to create a new controller assigned to the Unify OpenScape Business.

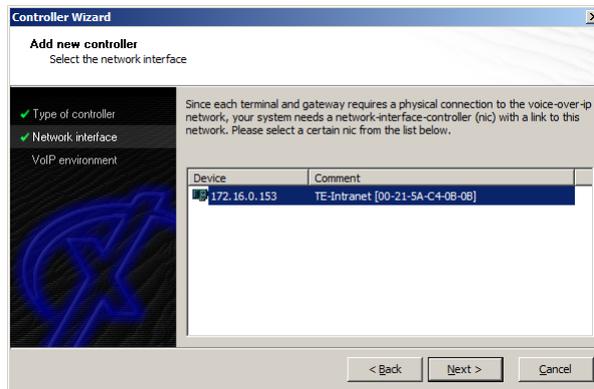
On the first dialog of the Controller Wizard please select the **Add Voice-over-IP controller (VoIP)** option.





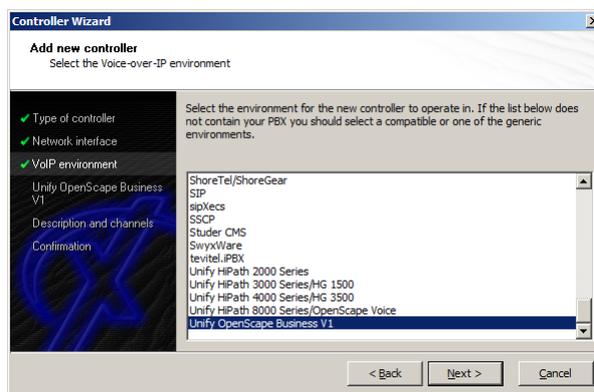
## 2.1 Network Interface

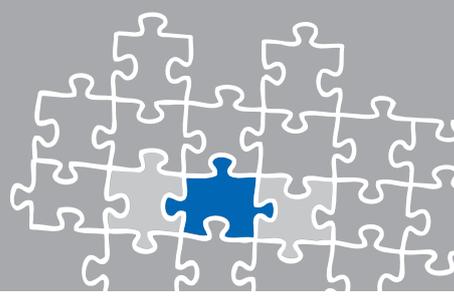
In the wizards **Network interface** dialog the Ethernet interface used by the XCAPI controller has to be determined.



## 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. Please select **Unify OpenScope Business**, selectable from XCAPI version 3.5.0.





## 2.3 IP Address of the Gateway

Next, please provide the Unify gateway related Ethernet IP address, in this example **172.18.1.30**

The screenshot shows the 'Controller Wizard' dialog box with the title 'Add new controller'. The subtitle reads 'Provide the hostname or the ip address of the voice-over-ip remote peer'. On the left, a progress bar shows the following steps: 'Type of controller', 'Network interface', 'VoIP environment', 'Unify OpenScape Business V1', 'Description and channels', and 'Confirmation'. The 'Unify OpenScape Business V1' step is currently active. The main area contains the text 'Please provide the hostname or the ip address of the voice-over-ip remote peer (pbx) that should be used.' Below this, there is a text input field with the value '172.18.1.30'. At the bottom, there are three buttons: '< Back', 'Next >', and 'Cancel'.

## 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 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 Unify OpenScape Business.

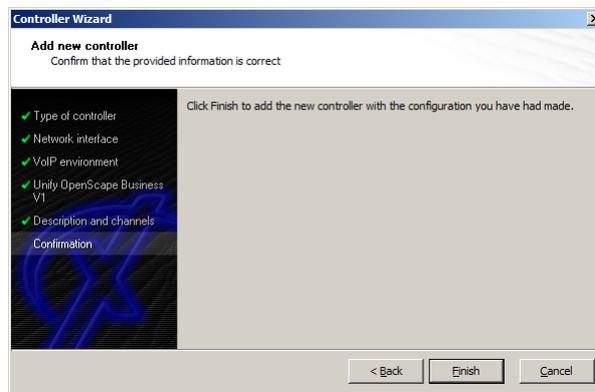
The screenshot shows the 'Controller Wizard' dialog box with the title 'Add new controller'. The subtitle reads 'Provide a description and select the number of channels'. On the left, the progress bar shows the following steps: 'Type of controller', 'Network interface', 'VoIP environment', 'Unify OpenScape Business V1', 'Description and channels', and 'Confirmation'. The 'Description and channels' step is currently active. The main area contains the text 'Please enter a meaningful description for the new controller and decide how many channels should be available for applications. Please consider that the effective number of available channels depend on the installed license.' Below this, there are two text input fields: 'Description' with the value 'Unify OpenScape Business V1' and 'Channels' with the value '10'. At the bottom, there are three buttons: '< Back', 'Next >', and 'Cancel'.



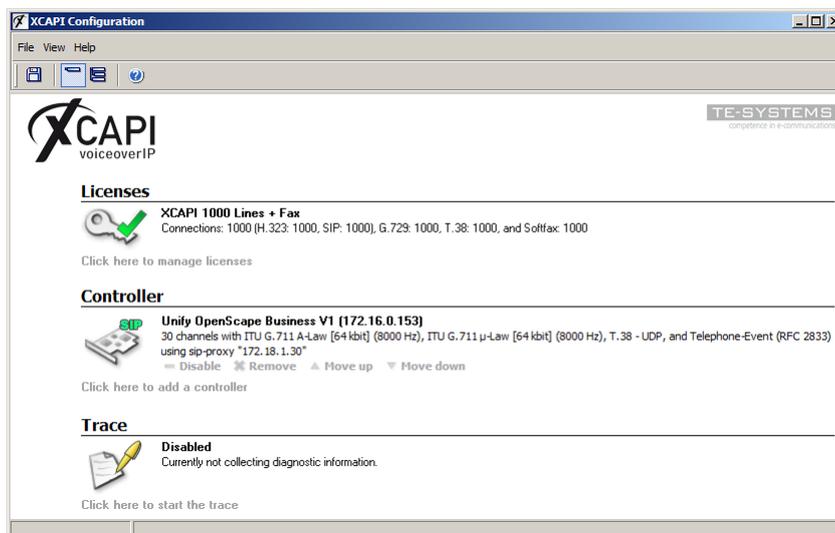
## 2.5 Confirmation

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.

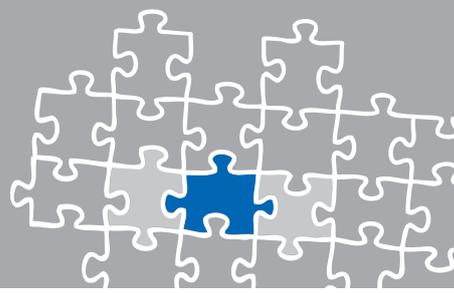
If everything is correct please use the **Finish** button in order to finally create the new controller.



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.



You need to restart the bound CAPI applications, in meaning of its service, to take effect on any controller changes.



## Unify OpenScape Business Configuration

In order to establish a connection between the XCAPI and the Unify OpenScape Business gateway, you need to setup the XCAPI as **native SIP Server Trunk** with all its appropriate configurations.

### 3.1 SIP Parameters

The SIP parameters are used with their default values.

The screenshot shows the 'Expert mode - Telephony Server' configuration window. The left sidebar lists the configuration tree: Voice Gateway, SIP Parameters (selected), Codec Parameters, Destination Codec Parameters, Internet Telephony Service Provider, Networking, and SIP Interconnection. The main area is titled 'SIP Parameters' and contains the following settings:

- SIP Transport Protocol:** SIP via TCP: Yes, SIP via UDP: , SIP via TLS: Yes
- SIP Registrar:** Period of registration (sec): 120
- RFC 3261 Timer Values:** Transaction Timeout (msec): 32000
- SIP Session Timer:** RFC 4028 support: , Session Expires (sec): 1800, Minimal SE (sec): 90
- Provider Calls:** Maximum possible Provider Calls: 0

At the bottom of the window are buttons for 'Apply', 'Undo', and 'Help'.



### 3.2 Codec Parameters

Ensure that the codec settings are conforming to those of the XC-API controller configuration. Please review the codec-related chapters **DTMF**, on [page 19](#), and **Fax Services** starting on [page 16](#).

The parameter **Redundant Transmission of RFC2833 Tones according to RFC2198** must be disabled.

The screenshot shows the 'Expert mode - Telephony Server' configuration window. The left sidebar contains a tree view with 'Codec Parameters' selected. The main area is titled 'Edit Codec Parameters' and contains a table of codec settings.

Codec	Priority	Voice Activity Detection	Frame Size
G.711 A-law	Priority 1	VAD: <input type="checkbox"/>	20 msec
G.711 μ-law	Priority 2	VAD: <input type="checkbox"/>	20 msec
G.729A	not used	VAD: <input type="checkbox"/>	20 msec
G.729AB	not used	VAD: <input checked="" type="checkbox"/>	20 msec

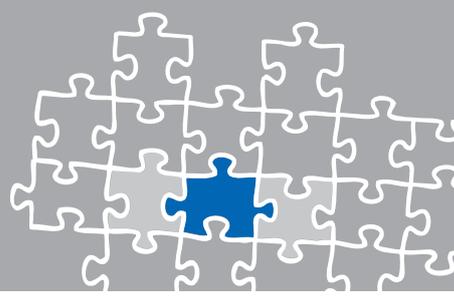
Below the table, there are sections for 'Enhanced DSP Channels', 'T.38 Fax', 'Misc.', and 'RFC2833'. The 'RFC2833' section includes the following settings:

- Transmission of Fax/Modem Tones according to RFC2833:
- Transmission of DTMF Tones according to RFC2833:
- Payload Type for RFC2833: 98
- Redundant Transmission of RFC2833 Tones according to RFC2198:

Buttons for 'Apply', 'Undo', and 'Help' are located at the bottom of the configuration window.



Please note that the **Transmission of Fax/Modem Tones according to RFC2833** parameter has to be enabled for facsimile interoperability.



### 3.3 SIP Interconnection

The **IP Address / Host name** has also to be related the XCAPI controllers bound Ethernet interface and the controllers used local listening port for SIP (by default 5060). Ensure that this **Native SIP Server Trunk** is enabled and the according **Trunk Identifier in System** is selected up on requirements (IP Trunks and Route relations).

The screenshot shows the 'Expert mode - Telephony Server' configuration window. The left sidebar contains a tree view with the following items: Voice Gateway, SIP Parameters, Codec Parameters, Destination Codec Parameters, Internet Telephony Service Provider, Networking, SIP Interconnection, Application Suite, HiPath 4000, Native SIP Server trunk (selected), Native\_SIP\_Server\_trunk-User, OpenScapeVoice, and SIPQ Server trunk. The main area is titled 'SIP Interconnection' and contains the following fields:

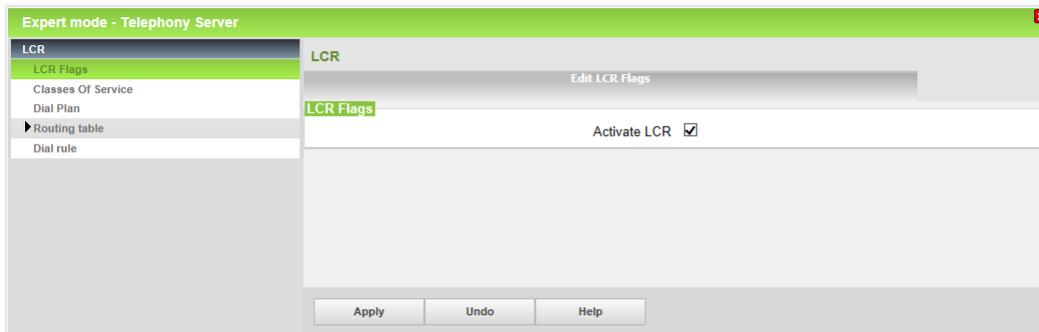
- Name:** Native SIP Server trunk
- Enable Trunk:**
- Trunk Identifier in System:** SIP-Interconnection1
- Remote Domain Name:** (empty)
- SIP Server:**
  - IP Address / Host name:** 172.16.0.153
  - Port:** 5060
  - Secure Transport:**
- SIP Registrar:**
  - Use Registrar:**
  - IP Address / Host name:** (empty)
  - Port:** 5060
  - Reregistration Interval (sec):** 300
- Outbound Proxy:**
  - Use Outbound Proxy:**
  - IP Address / Host name:** 0.0.0.0
  - Port:** 0
- Inbound Proxy:**
  - Use Inbound Proxy:**
  - IP Address / Host name:** 0.0.0.0
  - Port:** 0

At the bottom of the window, there are buttons for 'Apply', 'Undo', 'Refresh', 'Reset Default Values', and 'Help'.



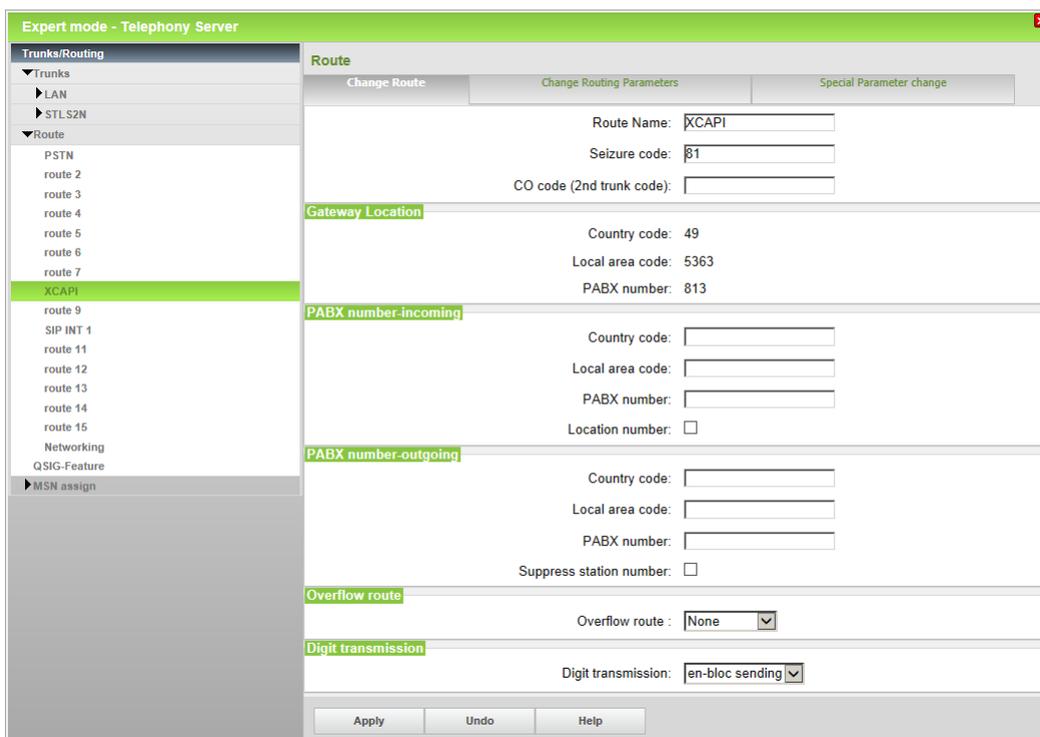
### 3.4 LCR

This environment make use of the **LCR** (Automatic Least Cost Routing).



### 3.5 Routes

The XCAPI route is used as shown below. If required one of the non-reserved routes should be used. Of course the local VoIP environment needs additional custom configurations and adjustments for a closed or open numbering scheme. The same goes for the **Routing Parameters**, **Special Parameters** and **Dial Plan** configurations.





### 3.6 Dial Plan

The according dial plan configuration for the XC-API is here used with dialed digits **-81XZ** and related to the **Routing Table** number **36**.

Dial Plan	Name	Dialed digits	Routing Table	Acc. code	Classes of service	Emergency
36	XC-API	-81XZ	36	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
37			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
38			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
39			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
40			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
41			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
42			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
43			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
44			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>
45			-	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>

### 3.7 Dial Rule

The XC-API related Dial Rule entry is here used as shown next.

Rule Name	Dial rule format	Network access	Type
36	XC-API	E1A	Unknown
37		Unknown	Unknown
38		Unknown	Unknown
39		Unknown	Unknown
40		Unknown	Unknown
41		Unknown	Unknown
42		Unknown	Unknown
43		Unknown	Unknown
44		Unknown	Unknown
45		Unknown	Unknown
46		Unknown	Unknown
47		Unknown	Unknown
48		Unknown	Unknown
49		Unknown	Unknown
50		Unknown	Unknown



### 3.8 Routing Table

The XCAPI's Routing Table is here set to its own Route and Dial Plan definitions. Additional configurations for COS, Warning, Dedicated Gateway and the GW Node ID must be set on demand.

Expert mode - Telephony Server

Routing Table

Change Routing Table

Routing Table: 36 en-bloc sending

Index	Route	Dial Rule	min. COS	Warning	Dedicated Gateway	GW Node ID
1	XCAPI	XCAPI	15	None	No	
2	None	None	15	None	No	
3	None	None	15	None	No	
4	None	None	15	None	No	
5	None	None	15	None	No	
6	None	None	15	None	No	
7	None	None	15	None	No	
8	None	None	15	None	No	
9	None	None	15	None	No	
10	None	None	15	None	No	
11	None	None	15	None	No	
12	None	None	15	None	No	
13	None	None	15	None	No	
14	None	None	15	None	No	
15	None	None	15	None	No	
16	None	None	15	None	No	

Apply Undo Help



### 3.9 Routing Parameters

The **Routing** and **Special Parameters** are used as shown next.

**Expert mode - Telephony Server**

**Trunks/Routing**

- Trunks
- LAN
- STLS2N
- Route
  - PSTN
  - route 2
  - route 3
  - route 4
  - route 5
  - route 6
  - route 7
  - XCAPI**
  - route 9
  - SIP INT 1
  - route 11
  - route 12
  - route 13
  - route 14
  - route 15
  - Networking
  - QSIG-Feature
  - MSN assign

**Route**

Change Route | Change Routing Parameters | Special Parameter change

**Routing flags**

- Digit repetition on:
- Analysis of second dial tone / Trunk monitoring:
- Intercept per direction:
- Over. service 3.1 kHz audio:
- Add direction prefix incoming:
- Add direction prefix outgoing:
- Ringback tone to CO:
- Segmentation:
- deactivate UUS per route:
- Always use DSP:

**Analog trunk seizure:**

**Trunk call pause:**

**Type of seizure:**

**Route type:**

**No. and type, outgoing:**

**Call number type:**

**Rerouting**

- Change route allowed:
- Route optimize active:

Apply | Undo | Help

Those **Special Parameters** are used with their defaults.

**Expert mode - Telephony Server**

**Trunks/Routing**

- Trunks
- LAN
- STLS2N
- Route
  - PSTN
  - route 2
  - route 3
  - route 4
  - route 5
  - route 6
  - route 7
  - XCAPI**
  - route 9
  - SIP INT 1
  - route 11
  - route 12
  - route 13
  - route 14
  - route 15
  - Networking
  - QSIG-Feature
  - MSN assign

**Route**

Change Route | Change Routing Parameters | Special Parameter change

**Numbering plan**

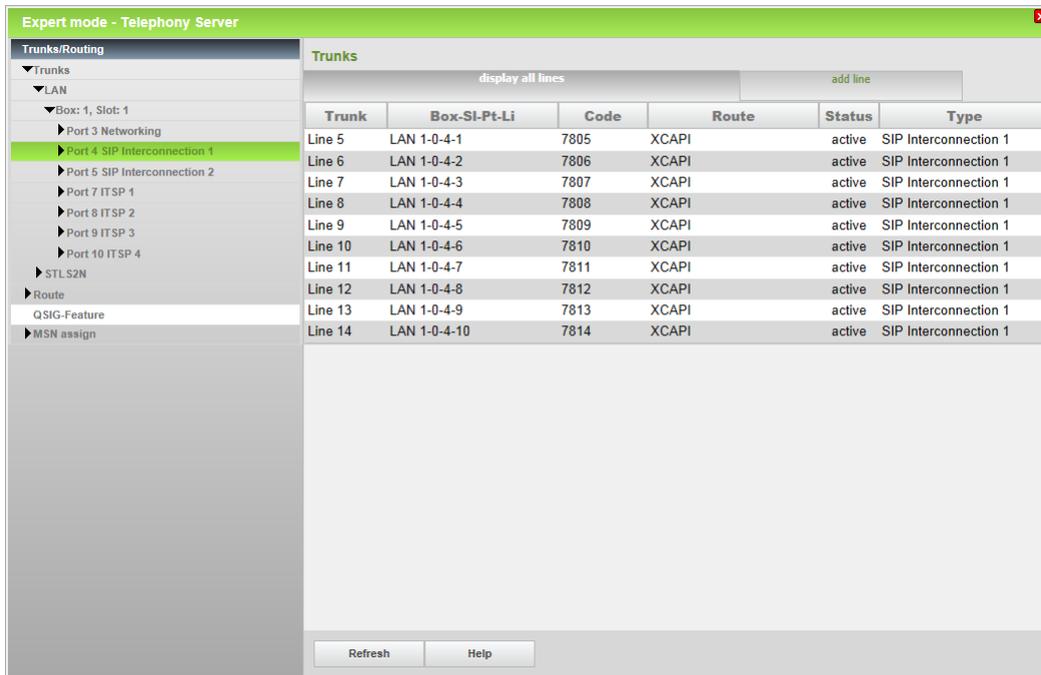
- Called Party Number:
- All others:
- Site:
- COLP:
- Notify send:
- without CLIP:
- No SETUP ACK.:
- no DIV.LEG-Info:
- With sending complete:

Apply | Undo | Help



### 3.10 IP Trunks

Don't forget to add the IP trunks which will be related to the XC-API route, as shown in the next chapter **Trunks**. The declared XC-API route needs to be assigned to each added trunk line.



Expert mode - Telephony Server

Trunks/Routing

- Trunks
  - LAN
    - Box: 1, Slot: 1
      - Port 3 Networking
      - Port 4 SIP Interconnection 1
      - Port 5 SIP Interconnection 2
      - Port 7 ITSP 1
      - Port 8 ITSP 2
      - Port 9 ITSP 3
      - Port 10 ITSP 4
    - STLS2N
    - Route
    - QSIG-Feature
    - MSN assign

Trunks

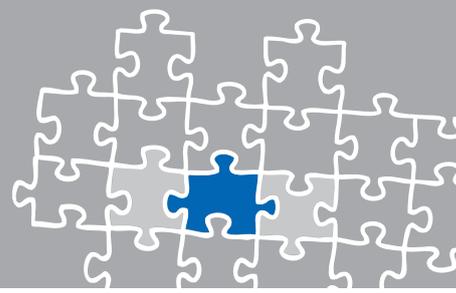
display all lines      add line

Trunk	Box-SI-Pt-Li	Code	Route	Status	Type
Line 5	LAN 1-0-4-1	7805	XC-API	active	SIP Interconnection 1
Line 6	LAN 1-0-4-2	7806	XC-API	active	SIP Interconnection 1
Line 7	LAN 1-0-4-3	7807	XC-API	active	SIP Interconnection 1
Line 8	LAN 1-0-4-4	7808	XC-API	active	SIP Interconnection 1
Line 9	LAN 1-0-4-5	7809	XC-API	active	SIP Interconnection 1
Line 10	LAN 1-0-4-6	7810	XC-API	active	SIP Interconnection 1
Line 11	LAN 1-0-4-7	7811	XC-API	active	SIP Interconnection 1
Line 12	LAN 1-0-4-8	7812	XC-API	active	SIP Interconnection 1
Line 13	LAN 1-0-4-9	7813	XC-API	active	SIP Interconnection 1
Line 14	LAN 1-0-4-10	7814	XC-API	active	SIP Interconnection 1

Refresh      Help



Please note that enabling a newly created SIP trunk requires a gateway reboot.



### 3.11 System Parameter Flags

If required, some specific flags such as the **External traffic transit** or the **SIP Prov. to SIP Prov. transit** flag, must be enabled.

**Expert mode - Telephony Server**

**Basic Settings**

- System
  - System Flags**
  - Time Parameters
  - Display
  - DISA
  - Intercept/Attendant/Hotline
  - LDAP
  - Texts
  - Flexible menu
  - Speed Dials
  - Service Codes
- Gateway
- DynDNS
  - AF/EF Codepoints
  - Quality of Service
- Date and Time
- Port Management
- Call Charges
  - Voice mail / Announcement Player

**System Flags**

Edit System Flags

**System flags**

- Through-connection for external FWD on:
- Call forwarding to main station interface permitted:
- Hunting to external call forwarding destination: 
  - Conference tone:
- Warning signal for call pickup groups:
- Increase volume for optiPoint/OpenStage terminals: 
  - Relocate allowed:
- More than 1 external conference member: 
  - Trunk reservation, automatic:
  - No. redial with a/c code:
- Use only default number for MSN: 
  - Path optimization:
  - DTMF automatic:
  - Broadcast with connection:
  - Tone from CO:
  - Ringback protection:
  - Euro-impedance:
- Different phonemail messages Day/Night:
- Display international / national code number: 
  - Line change for direct call:
  - Automatic redial:
  - Voice mail Node call number:
- Call Pickup after automatic recall: 
  - Configurable CLIP:
- Caller list at destination in case of Forward Line:
- Call forwarding after deflect call / single step transfer:
- Follow call management in case of deflect call / single step transfer:
- Calling number in pick-up groups / ringing groups / CFN /RNA: 
  - SPE support:
  - SPE advisory tone:
- SIP Prov. to SIP Prov. transit:
- Transparent dialing of \* and # on trunk interfaces:

**Open numbering scheme**

- active:
- Node callnumber:

**Transit permission**

- Feature transit:
- Tie traffic transit:
- External traffic transit:

**Restriction for UC calls**

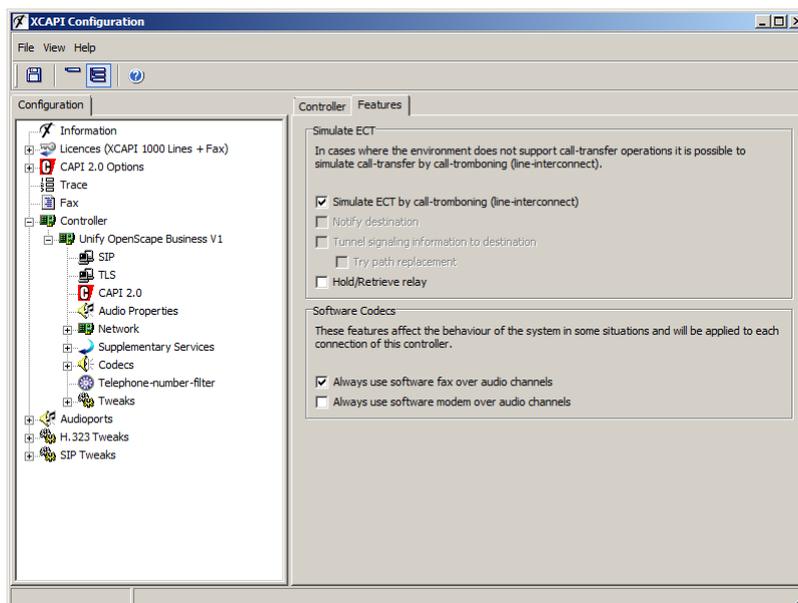
- Restriction for UC calls:



## Call Transfer

The Unify OpenScape Business series does not support call transfer scenarios via SIP refer yet.

For enabling **Call Transfer** this service can be simulated by the XCAPI. Whenever the CAPI application initiates a call transfer between two active participants, the XCAPI starts triggering the call transfer simulation. During this simulation two b-channels are occupied, but from application side the calls are released such as in a real call transfer scenario. Please review the **Features** tab of the respective XCAPI controller and ensure that the **Simulate ECT by call-tromboning (line-interconnect)** parameter is enabled.

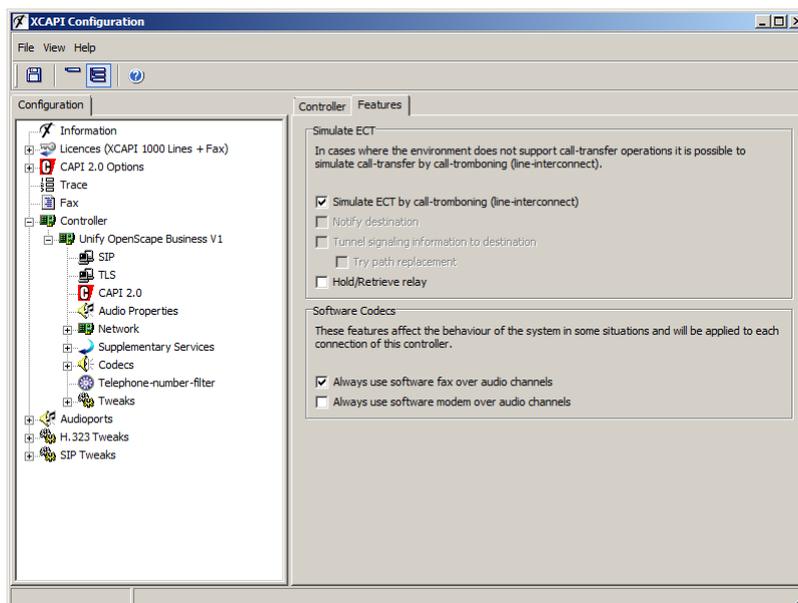




## SoftFax

In the **SoftFax** mode, the XCAPI simulates an analog Fax device by transmitting modulated facsimile signals modem-like via audio channels. To configure the SoftFax mode, please open the XCAPI configuration utility and select in the advanced configuration mode the SIP controller assigned to the Unify OpenScape Business.

Open the configuration tab labeled **Features**. Enable the **SoftFax** mode by setting the **Always use software fax over audio channels** option and save the changes to the XCAPI controller configuration.





The Routing parameters, described already in chapter **PBX Routing** on [page 7](#), are the same as for voice transmission. The configuration flags of the **Codec Parameter** dialog should be set as follows:

Codec	Priority	Voice Activity Detection	Frame Size
G.711 A-law	Priority 1	VAD: <input type="checkbox"/>	20 msec
G.711 $\mu$ -law	Priority 2	VAD: <input type="checkbox"/>	20 msec
G.729A	not used	VAD: <input type="checkbox"/>	20 msec
G.729AB	not used	VAD: <input checked="" type="checkbox"/>	20 msec

**Enhanced DSP Channels**  
Use G.711 only:

**T.38 Fax**  
T.38 Fax:   
Use FillBitRemoval:   
Max. UDP Datagram Size for T.38 Fax (bytes): 1472  
Error Correction Used for T.38 Fax (UDP): i38UDPRedundancy

**Misc.**  
ClearChannel:  Frame Size: 20 msec

**RFC2833**  
Transmission of Fax/Modem Tones according to RFC2833:   
Transmission of DTMF Tones according to RFC2833:   
Payload Type for RFC2833: 98  
Redundant Transmission of RFC2833 Tones according to RFC2198:

Buttons: Apply, Undo, Help



For SoftFax please ensure that **T.38 Fax** is disabled for appropriate interworking. Further the option **Transmission of Fax/Modem Tones according to RFC2833** is enabled and **Redundant Transmission of RFC2833 Tones according to RFC2198** is disabled.



## T.38

When using the T.38 protocol you have to enable the **T.38 Fax** option within the **Codec Parameters** dialog of the **Voice Gateway**.

Codec	Priority	Voice Activity Detection	Frame Size
G.711 A-law	Priority 1	VAD: <input type="checkbox"/>	20 msec
G.711 μ-law	Priority 2	VAD: <input type="checkbox"/>	20 msec
G.729A	not used	VAD: <input type="checkbox"/>	20 msec
G.729AB	not used	VAD: <input checked="" type="checkbox"/>	20 msec

**Enhanced DSP Channels**  
Use G.711 only

**T.38 Fax**  
T.38 Fax:   
Use FillBitRemoval:   
Max. UDP Datagram Size for T.38 Fax (bytes): 1472  
Error Correction Used for T.38 Fax (UDP): 38UDPRedundancy

**Misc.**  
ClearChannel:  Frame Size: 20 msec

**RFC2833**  
Transmission of Fax/Modem Tones according to RFC2833:   
Transmission of DTMF Tones according to RFC2833:   
Payload Type for RFC2833: 98  
Redundant Transmission of RFC2833 Tones according to RFC2198:

You also need to disable the option **Always use software fax over audio channels** within the XCAPI controllers **Features** dialog.

**Controller Features**

**Simulate ECT**  
In cases where the environment does not support call-transfer operations it is possible to simulate call-transfer by call-tromboning (line-interconnect).

Simulate ECT by call-tromboning (line-interconnect)  
 Notify destination  
 Tunnel signaling information to destination  
 Try path replacement  
 Hold/Retrieve relay

**Software Codecs**  
These features affect the behaviour of the system in some situations and will be applied to each connection of this controller.

Always use software fax over audio channels  
 Always use software modem over audio channels



Please note that the amount of T.38 fax channels are limited by the gateway.



## DTMF

The Parameter **Transmission of DTMF Tones according to RFC2833** must be enabled.

The value for the **Payload Type for RFC2833** must be equivalent to the XCAPI controller settings.

**Expert mode - Telephony Server**

**Codec Parameters**

Edit Codec Parameters

Codec	Priority	Voice Activity Detection	Frame Size
G.711 A-law	Priority 1	VAD: <input type="checkbox"/>	20 msec
G.711 μ-law	Priority 2	VAD: <input type="checkbox"/>	20 msec
G.729A	not used	VAD: <input type="checkbox"/>	20 msec
G.729AB	not used	VAD: <input checked="" type="checkbox"/>	20 msec

**Enhanced DSP Channels**

Use G.711 only

**T.38 Fax**

T.38 Fax:

Use FillBitRemoval:

Max. UDP Datagram Size for T.38 Fax (bytes): 1472

Error Correction Used for T.38 Fax (UDP): 38UDPRedundancy

**Misc.**

ClearChannel:  Frame Size: 20 msec

**RFC2833**

Transmission of Fax/Modem Tones according to RFC2833:

Transmission of DTMF Tones according to RFC2833:

Payload Type for RFC2833: 98

Redundant Transmission of RFC2833 Tones according to RFC2198:

Apply Undo Help

Please review the according codec within the XCAPI controller configuration.

**XCAPI Configuration**

File View Help

Configuration

- Information
- Licences (XCAPI 1000 Lines + Fax)
- CAPI 2.0 Options
- Trace
- Fax
- Controller
  - Unify OpenScape Business V1
    - SIP
    - TLS
    - CAPI 2.0
    - Audio Properties
    - Network
    - Supplementary Services
    - Codecs
      - ITU G.711 A-Law [64 kbit] (8000 Hz)
      - ITU G.711 μ-Law [64 kbit] (8000 Hz)
      - ETSI GSM 6.10
      - ITU G.729
      - T.38 - LUDP
      - Telephone-Event (RFC 2833)
    - Telephone-number-filter
  - Tweaks
    - Audioports
    - H.323 Tweaks
    - SIP Tweaks

Options

Payload Type

Define the payload-type that should be used to receive telephone-events sent by remote terminals.

Payload-Type (0-127): 98

Recommendation



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