

Unify OpenScape Business V3



April 20, 2021

Contents

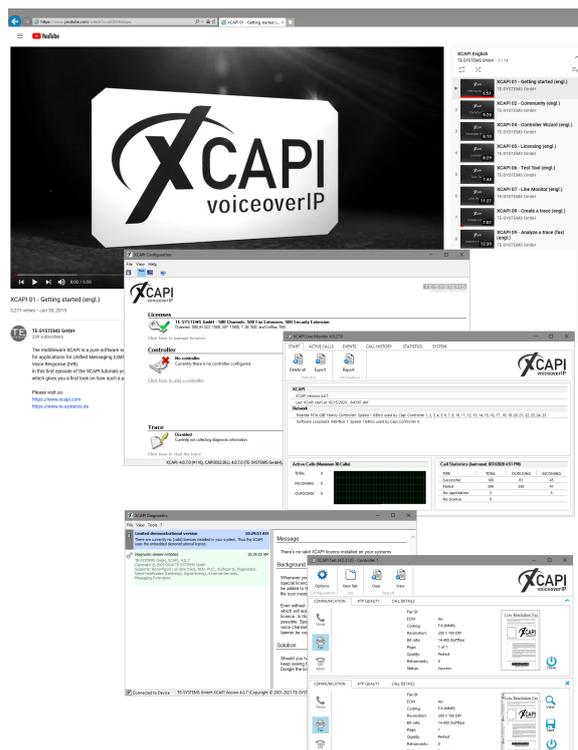
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|----------|--|-----------|
| 1 | Introduction | 3 |
| 2 | XC-API Configuration | 4 |
| 2.1 | Voice over IP Environment | 5 |
| 2.2 | Description and Channels | 5 |
| 2.3 | Gateway Address | 6 |
| 2.4 | Network Interface | 6 |
| 2.5 | Port Allocation | 7 |
| 2.6 | Confirmation | 7 |
| 3 | Unify OpenScope Business Configuration | 9 |
| 3.1 | SIP Parameters | 9 |
| 3.2 | Native SIP Server Trunk | 10 |
| 3.3 | Codec Parameters | 11 |
| 3.4 | LCR | 12 |
| 3.5 | Route | 12 |
| 3.6 | Routing Parameters | 13 |
| 3.7 | Dial Plan | 14 |
| 3.8 | Routing Table | 15 |
| 3.9 | Dial Rule | 15 |
| 3.10 | Trunk Lines | 16 |
| 3.11 | System Parameter Flags | 17 |
| 4 | Appendix | 18 |
| 4.1 | Codecs | 18 |
| 4.2 | RFC2833 Payload | 19 |
| 4.3 | Fax Support | 20 |
| 4.3.1 | G.711 Fax Pass Through (Softfax) | 20 |
| 4.3.2 | T.38 | 22 |
| 4.3.3 | T.38 to G.711 Fax Pass Through Fallback | 23 |
| 4.4 | Simulated Call Transfer | 24 |
| 4.5 | Codecs | 25 |
| 4.6 | Timer | 26 |
| 4.7 | Diversion Handling | 27 |
| 4.8 | Numbering | 28 |
| 5 | Unify Ready Technology Connectivity Certification | 29 |

Introduction

This document is intended to support you with the integration of XCAPI into an existing environment of the Unify OpenScope Business. In the following sections we describe the configuration steps for SIP trunking to allow optimal integration of XCAPI and the Unify OpenScope Business. Even though this is based on the Unify OpenScope Business using firmware V3 and XCAPI version 4.0.12, this document is applicable to other versions given a few adjustments.

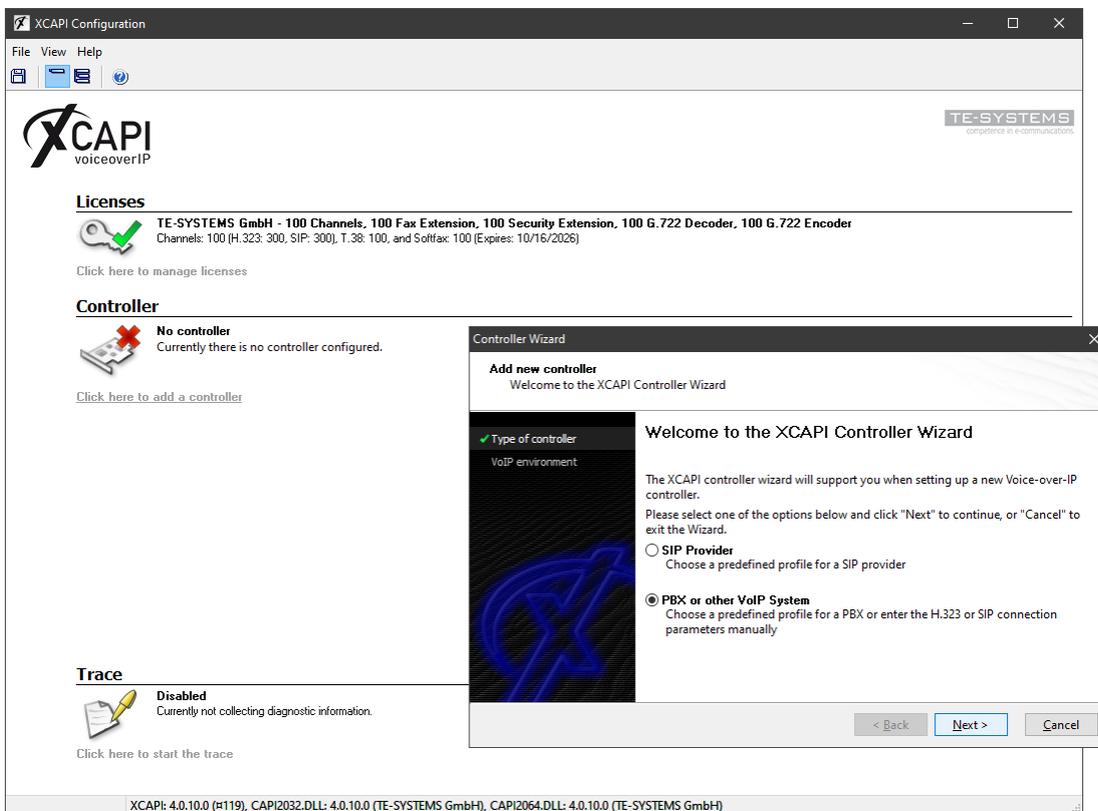
At this point we assume that the Unify OpenScope Business environment and the physical or virtual application server is available and accessible through the network. Application server is defined as a server with a recent installation Microsoft Windows operating system with all updates and patches included. It is important that XCAPI and the CAPI 2.0 voice or fax application is properly installed. It is also assumed that the public network access via ISDN and/or SIP is working properly, functioning correctly with any custom and country dependent numberings and call routes. The same goes for the networking (LAN, WAN, DMZ, NAT, Firewall...) itself, as such those are beyond the scope of this document and thus not shown here at all. Please refer to the respective manufacturer's documentation, manuals and examples. Without using the deployed CAPI 2.0 application, the SIP trunk configuration can initially be tested with the **xtest.exe** application which is located in the XCAPI installation folder (by default \\Program Files (x86)TE-SYSTEMS\XCAP). This test tool allows you to check inbound and outbound calls, fax and testing several supplementary services.

We recommend a visit to our [YouTube channel](#) frequently for XCAPI related tutorials about licensing, the test tool, line monitor, tracing, analyzing and others. Registered [community](#) users can check about latest XCAPI documents, TechNotes and versions.



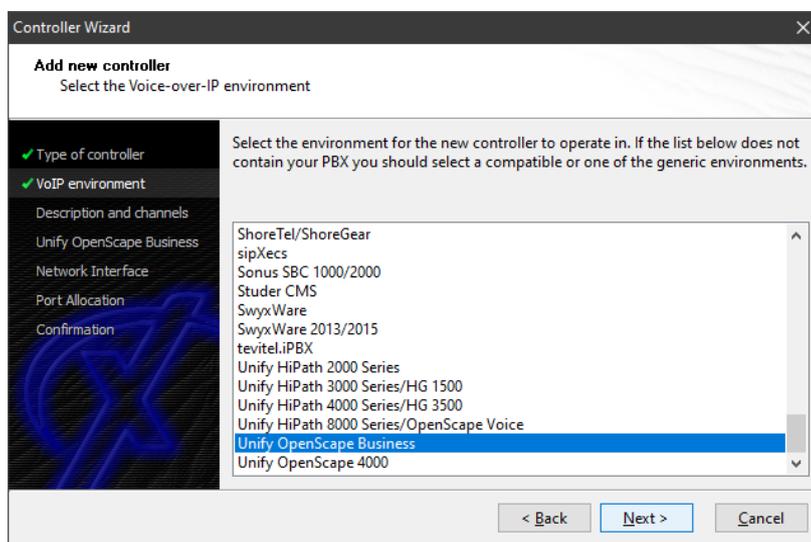
XCAPI Configuration

Please start up the XCAPI configuration to create a new controller assigned to the Unify OpenScope Business. On the first dialog of the XCAPI **Controller Wizard**, select **PBX or other VoIP System** and proceed with the **Next** button.



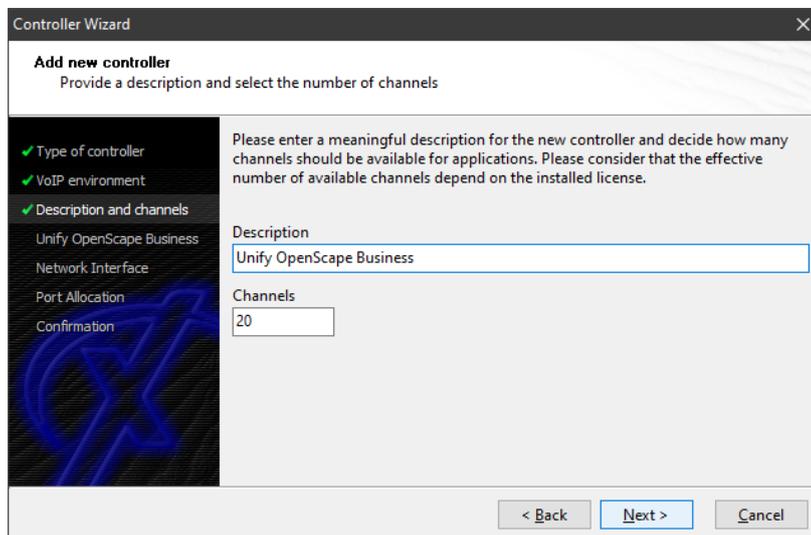
2.1 Voice over IP Environment

The next dialog lists some common Voice-over-IP environments. Selecting one of those will set up the XCAPI controller with a selection of near-optimal presets thus sparing you a lot of manual configuration.



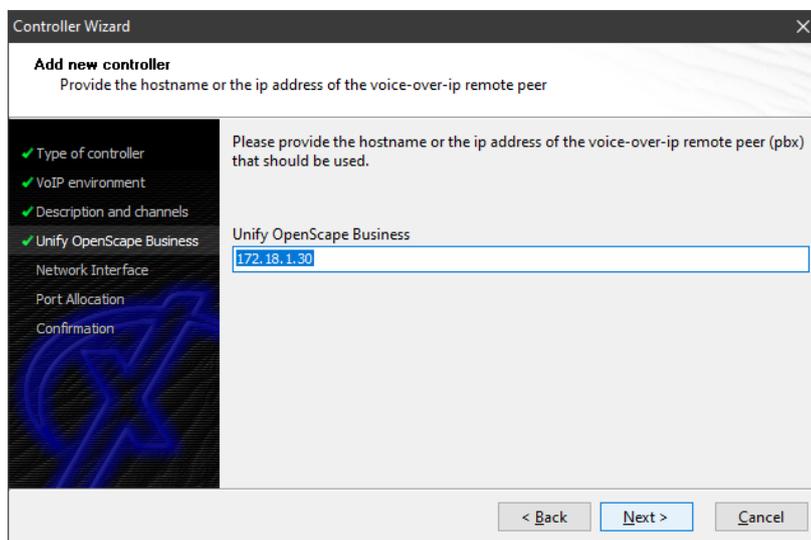
2.2 Description and Channels

When the VoIP environment was selected, the next dialog allows to setting a name for the controller. Also the number of channels that the new controller will be able to provide can be set. Here you enter how many simultaneous connections the XCAPI controller should handle when communicating with the VoIP gateway and the CAPI 2.0 application.



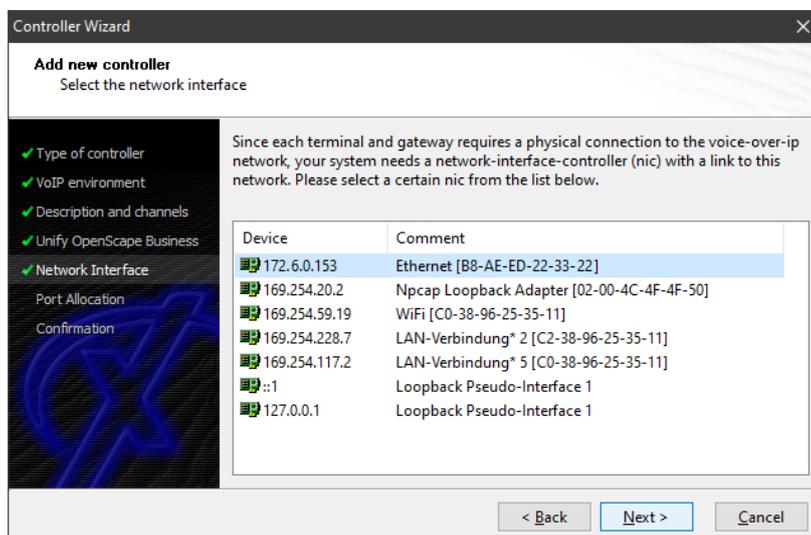
2.3 Gateway Address

Next, the host name or the IP address of the SIP listening Unify OpenScope Business Ethernet interface must be provided. In this example IP address **172.18.1.30** is used. Please note that both, the XCAPI controller and the Unify OpenScope Business, use by default the UDP port 5060 for SIP signaling.



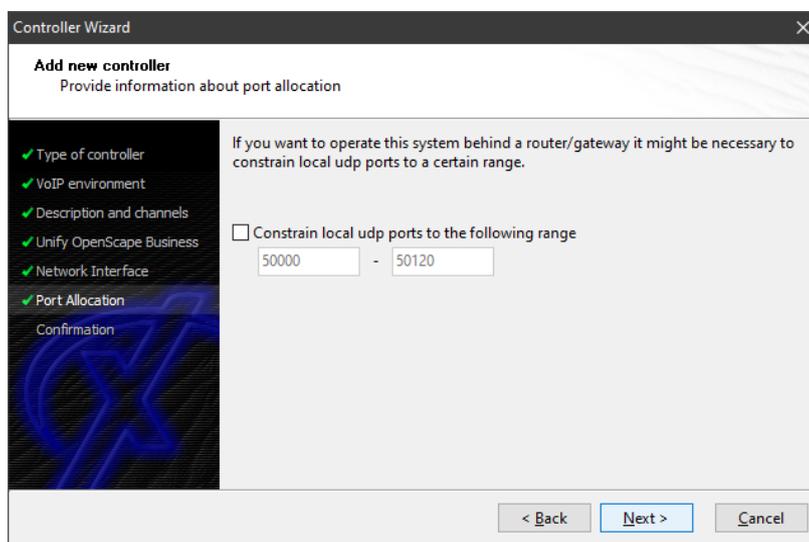
2.4 Network Interface

Afterwards, select the network interface that will be used for the inbound and outbound VoIP communications for this SIP controller.



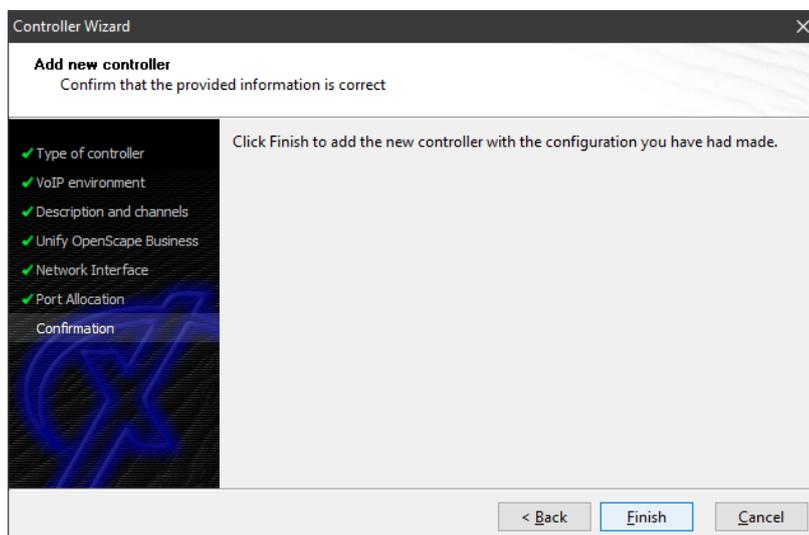
2.5 Port Allocation

On demand and in the case of any router or firewall restrictions for UDP (RTP/T.38) a port range can be specified. In this example no range will be set which allows the XCAPI controller to use a random port range between 1024 and 65535.

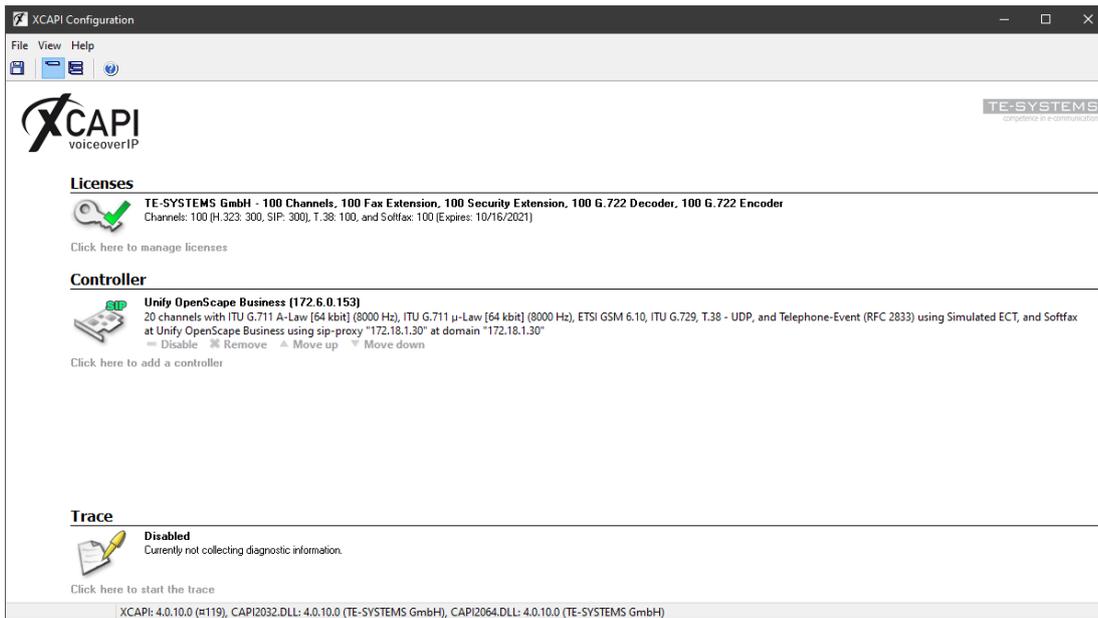


2.6 Confirmation

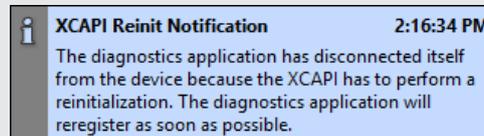
The final wizard dialog performs some checks on the configuration parameters you've made. If errors are detected, use the **Back** button to find the incorrect settings and correct them. Use the **Finish** button in order to create the new XCAPI controller.



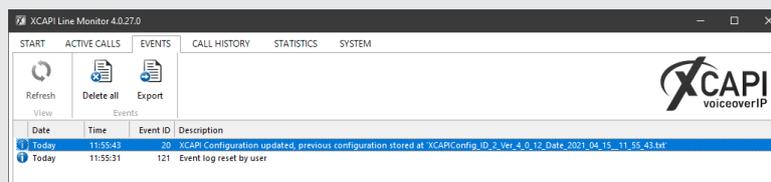
Now, the newly created controller is listed on the main page of XCAPI configuration tool. Use the **Save** button and exit the tool.



Please note that the bound CAPI 2.0 application with its services must be completely stopped and restarted for the XCAPI controller changes to take effect. Restarting any of the XCAPI services won't help at all. Alternatively the Server where XCAPI is running on can be restarted. If enabled, the XCAPI diagnostic monitor pops-up with a re-initialization notification on success:



Alternatively check with the **Events** tab of the **XCAPI Line Monitor** about a configuration update notification (Event ID 20):



Unify OpenScope Business Configuration

In order to establish a connection between XCAPI and the Unify OpenScope Business gateway, the XCAPI SIP controller must be setup as **native SIP Server Trunk** with all appropriate configurations. Please note, native SIP Trunking requires a Unify OpenScope Business **Networking** license.

3.1 SIP Parameters

The global SIP parameters of the OpenScope Business gateway are commonly used with their default values. If some values have to be adjusted for local reasons or specific ITSP (SIP carrier) or SBC (Session Border Controller) bindings, then they should also be set conform with the XCAPI SIP controller to keep functionality through all participating SIP bindings of the Unify OpenScope Business environment. For e.g.: if disabling SIP via UDP for the gateway, the XCAPI SIP controller should also be changed to the **Prefer TCP over TCP** method. This can be done in the XCAPI expert configuration view (SIP controller -> SIP settings -> Protocol tab -> **Preferred Transport**). The same goes for the **Session Expires** and **Minimal SE** timer values.

If required, the timer values can be adjusted for the XCAPI SIP controller (SIP controller -> SIP settings -> Timer tab).

The **Timer** chapter on [page 26](#) shows an example of the XCAPI SIP controller timer defaults.

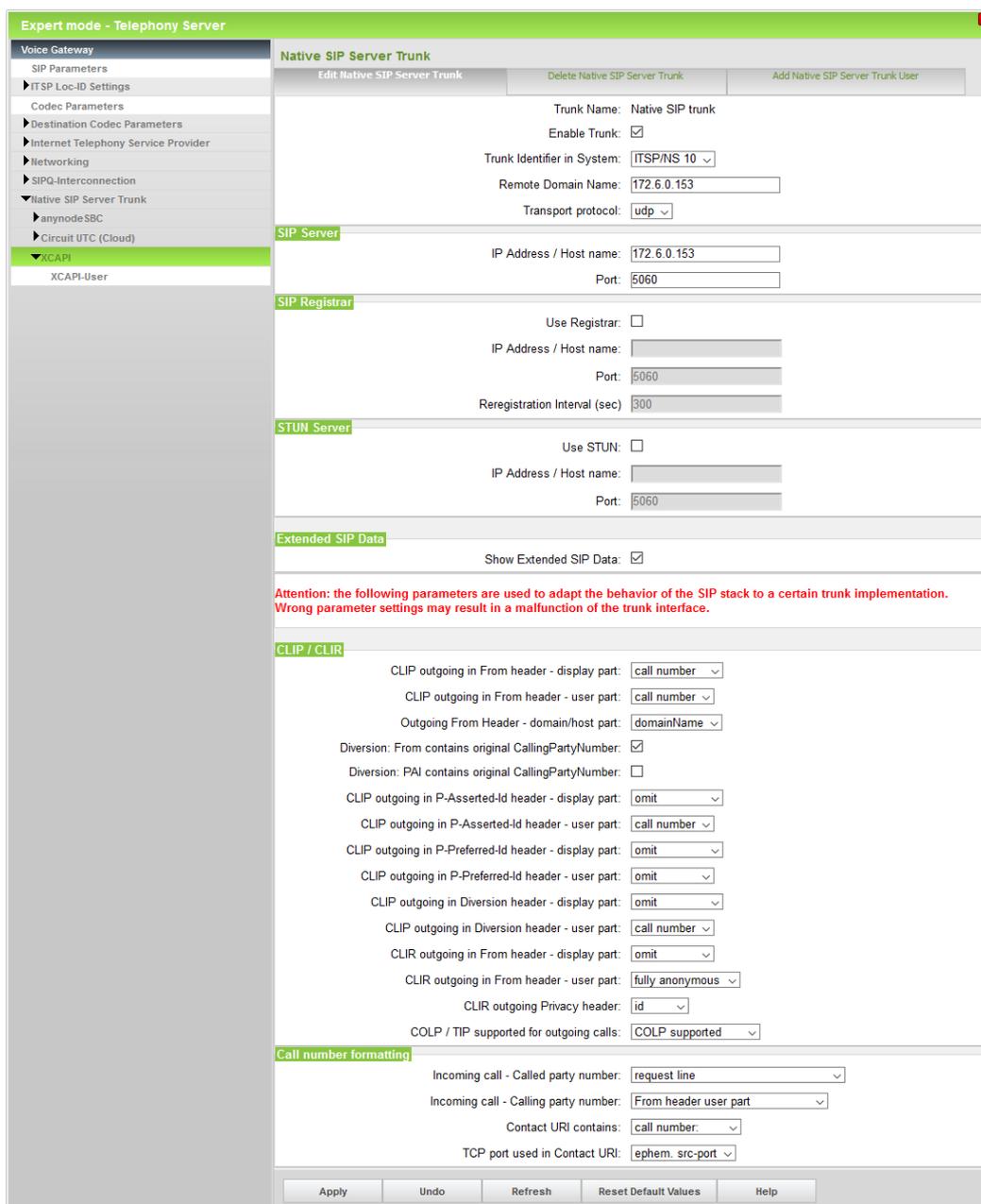
The screenshot displays the 'Expert mode - Telephony Server' configuration window. The left sidebar shows a tree view with 'SIP Parameters' selected. The main area is titled 'SIP Parameters' and contains several sections with adjustable settings:

- 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
- DNS Records:** Blocking time for unreachable destination(sec): 60
- Provider Calls:** Maximum possible Provider Calls: 10

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

3.2 Native SIP Server Trunk

The Unify OpenScope Business V3 XCAPI SIP trunk must be enabled as **Native SIP Trunk**. The **Remote Domain Name** and **IP Address / Host name** must be bound to the Ethernet interface of the XCAPI controller, here **172.6.0.153**. The **Transport protocol** and **Port** is used with the default **UDP** and **5060**. Ensure that **Native SIP trunk** is enabled and connected to the appropriate **Trunk Identifier in System**, in this example **ITSP/NS 10**. The **Extended SIP Data** are here set as shown on the next screenshot.



Expert mode - Telephony Server

Voice Gateway

- SIP Parameters
 - ITSP Loc-ID Settings
 - Codec Parameters
 - Destination Codec Parameters
 - Internet Telephony Service Provider
 - Networking
 - SIPQ-Interconnection
 - Native SIP Server Trunk
 - anynodeSBC
 - Circuit UTC (Cloud)
 - XCAPI
 - XCAPI-User

Native SIP Server Trunk

Edit Native SIP Server Trunk Delete Native SIP Server Trunk Add Native SIP Server Trunk User

Trunk Name: Native SIP trunk
 Enable Trunk:
 Trunk Identifier in System: ITSP/NS 10
 Remote Domain Name: 172.6.0.153
 Transport protocol: udp

SIP Server

IP Address / Host name: 172.6.0.153
 Port: 5060

SIP Registrar

Use Registrar:
 IP Address / Host name:
 Port: 5060
 Reregistration Interval (sec): 300

STUN Server

Use STUN:
 IP Address / Host name:
 Port: 5060

Extended SIP Data

Show Extended SIP Data:

Attention: the following parameters are used to adapt the behavior of the SIP stack to a certain trunk implementation. Wrong parameter settings may result in a malfunction of the trunk interface.

CLIP / CLIR

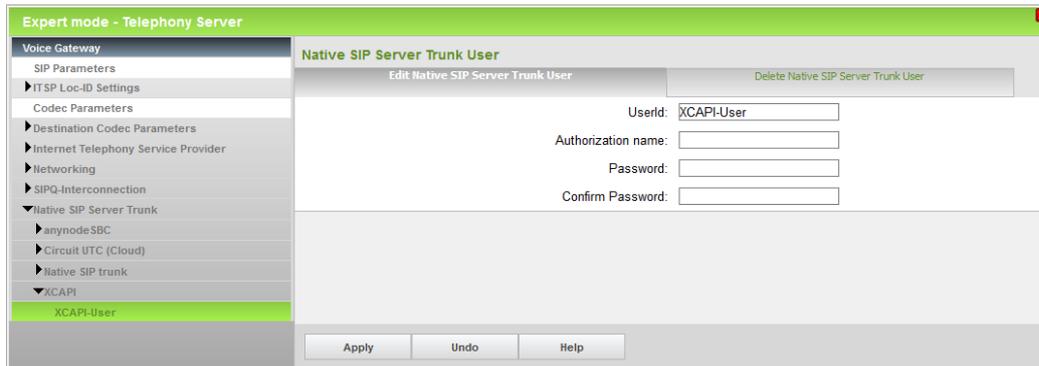
CLIP outgoing in From header - display part: call number
 CLIP outgoing in From header - user part: call number
 Outgoing From Header - domain/host part: domainName
 Diversion: From contains original CallingPartyNumber:
 Diversion: PAI contains original CallingPartyNumber:
 CLIP outgoing in P-Asserted-Id header - display part: omit
 CLIP outgoing in P-Asserted-Id header - user part: call number
 CLIP outgoing in P-Preferred-Id header - display part: omit
 CLIP outgoing in P-Preferred-Id header - user part: omit
 CLIP outgoing in Diversion header - display part: omit
 CLIP outgoing in Diversion header - user part: call number
 CLIR outgoing in From header - display part: omit
 CLIR outgoing in From header - user part: fully anonymous
 CLIR outgoing Privacy header: id
 COLP / TIP supported for outgoing calls: COLP supported

Call number formatting

Incoming call - Called party number: request line
 Incoming call - Calling party number: From header user part
 Contact URI contains: call number:
 TCP port used in Contact URI: ephem. src-port

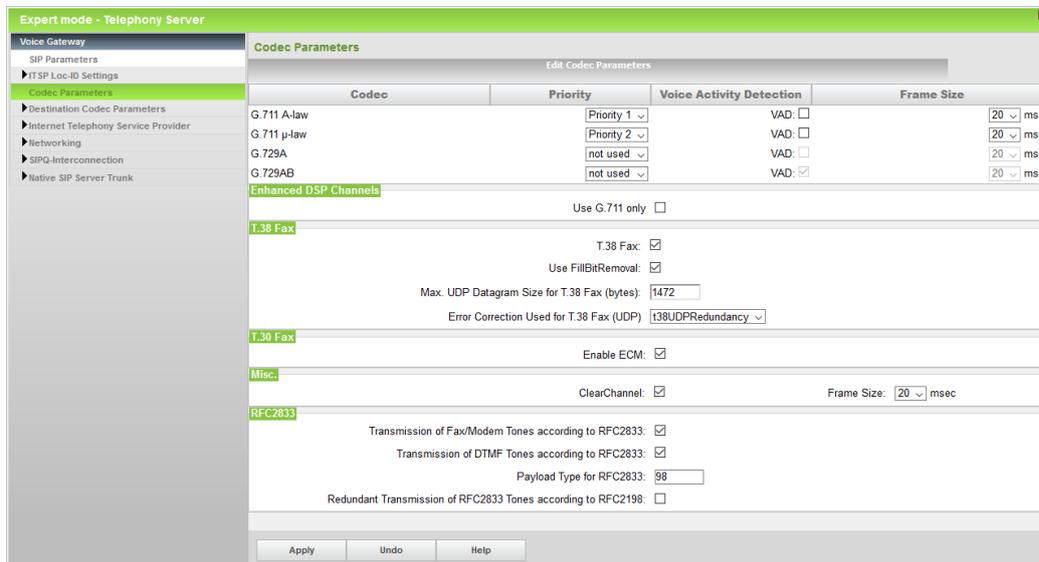
Apply Undo Refresh Reset Default Values Help

The auto-created Native SIP Server Trunk User is used without any **Authorization** credentials.



3.3 Codec Parameters

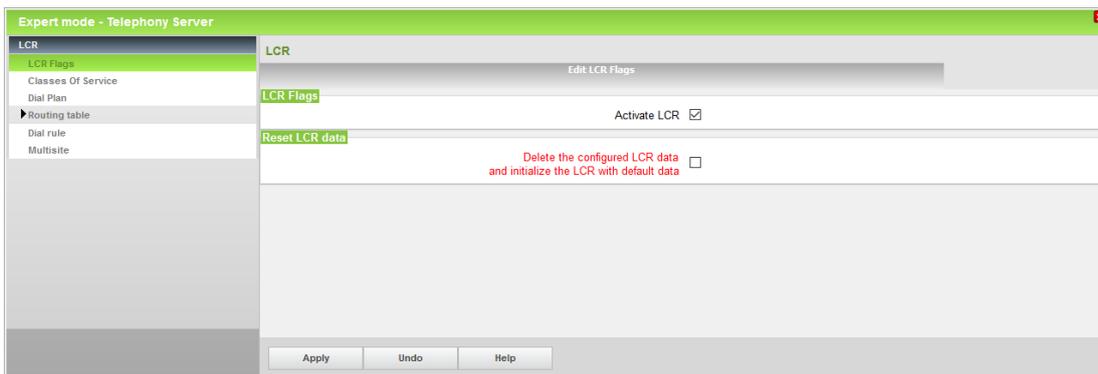
It is recommended using conform codec settings for the VoIP environment. If the gateway defaults were changed for any ITSP and/or SBC bindings, the XC-API SIP controller settings must be adjusted as well. An example of the XC-API controller codec settings can be reviewed in the chapter **Codecs** from [page 25](#).



Important: for fax interoperability **Transmission of DTMF and Fax/Modem Tones according to RFC2833** must be enabled and **Redundant Transmission of RFC2833 Tones according to RFC2198** must be disabled. To avoid potential non-conformities due to media transcoding malfunction through the gateway with its bindings (ITSP's, SBC's), the **G.711** codecs should be used with a **Frame Size** of **20 msec**. The **Payload Type for RFC2833** is by default used with value **98**. If changed for the PBX, it must also be adjusted for the XC-API SIP controller. For details check with the **RFC2833 Payload** chapter on [page 19](#).

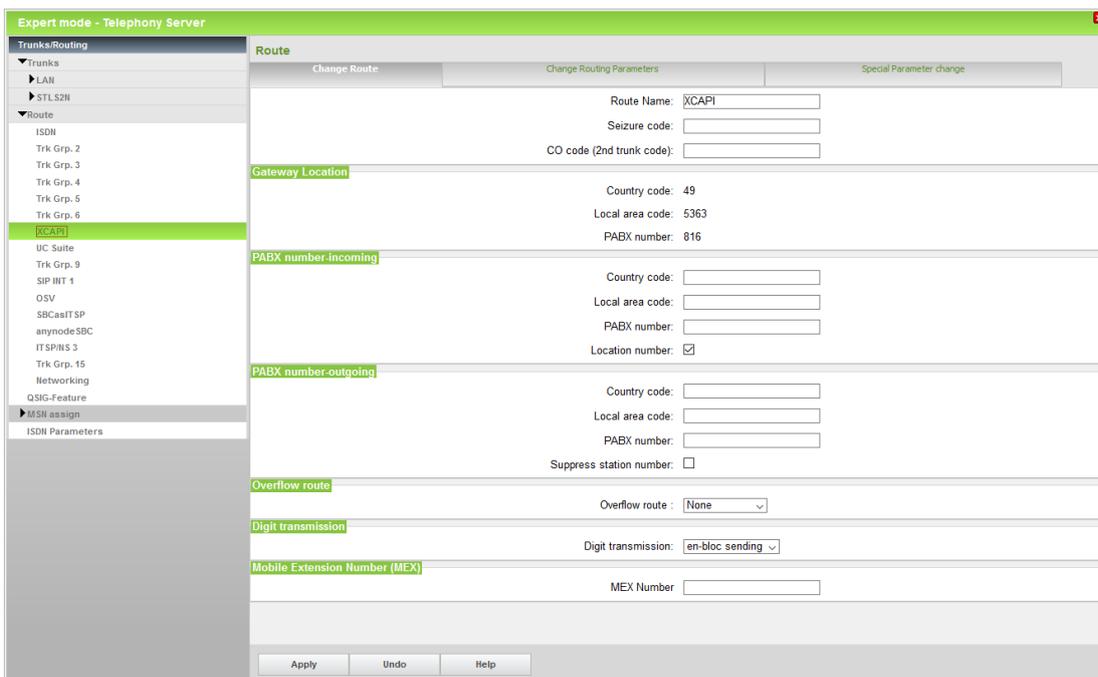
3.4 LCR

The test environment used in this example is based on the LCR (Automatic Least Cost Routing).



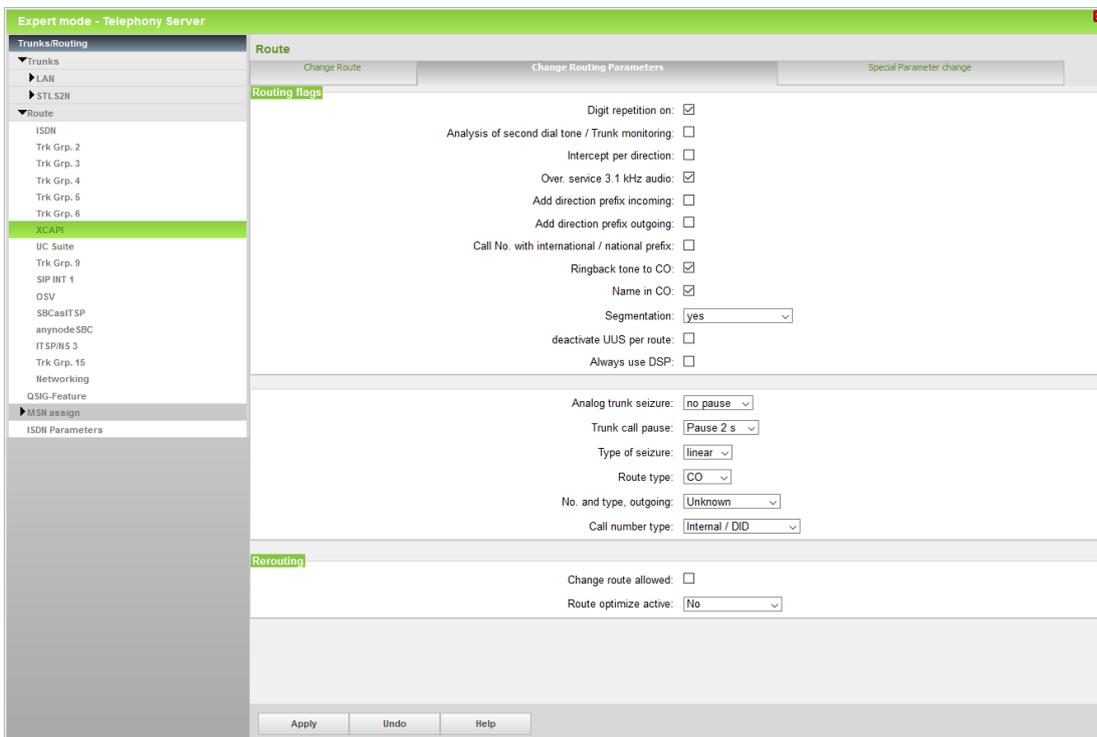
3.5 Route

For this test environment the XC-API route is used as shown below. The local VoIP environment needs additional configurations and adjustments for a closed or open numbering scheme. The same goes for the **Routing Parameters**, **Special Parameters** and **Dial Plan** configurations.

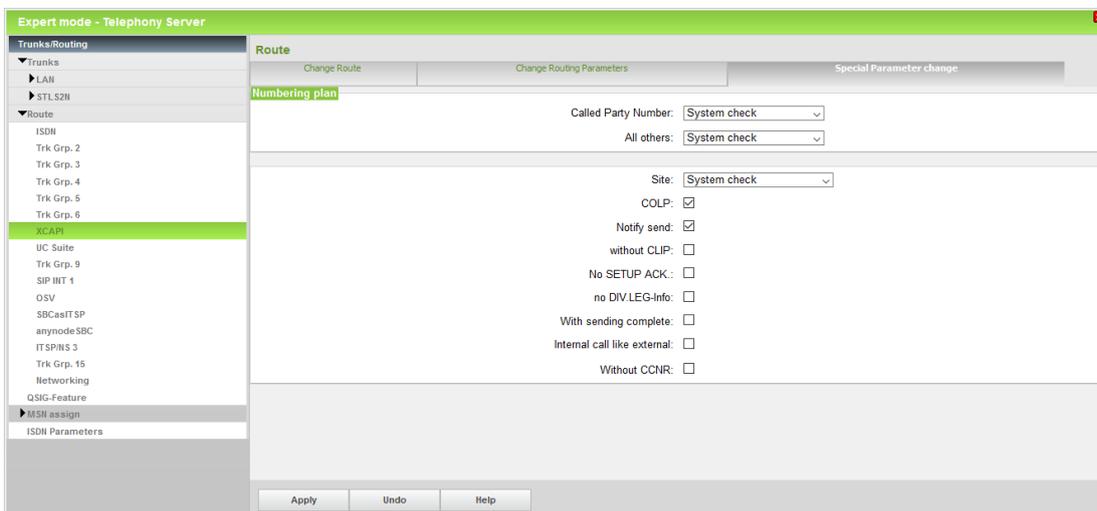


3.6 Routing Parameters

The **Routing** and **Special Parameters** of the XC-API route are used as shown next. Again, we want to mention that all the numbering related configurations must be set according to local requirements.



The **Special Parameters** are used as shown below.



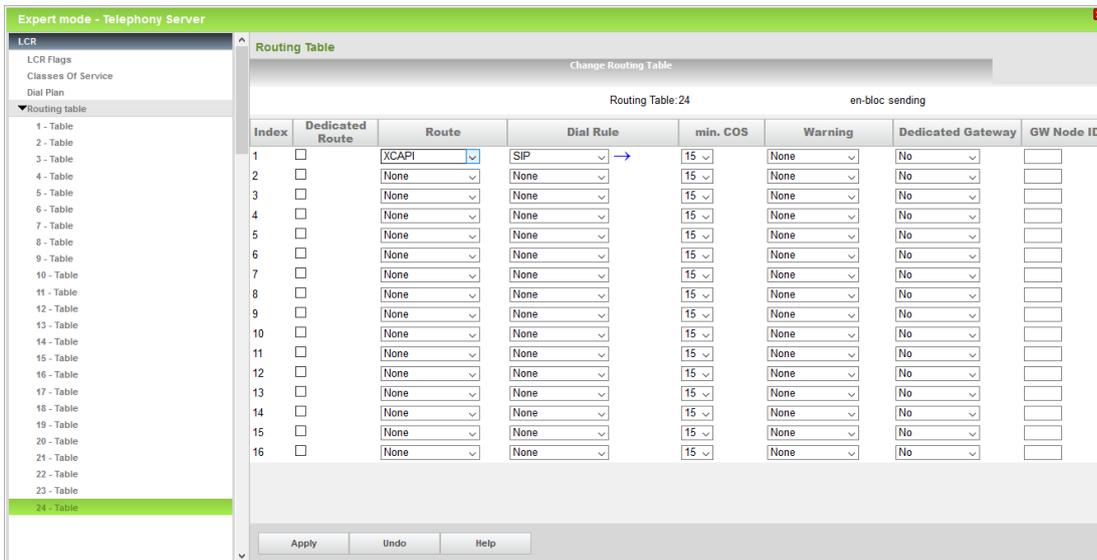
3.7 Dial Plan

The XCAPI related dial plan entries are basically used for an internal numbering range, here with the dialed digit string **92CZ**. However there are also some variations shown for matching local, national, and international dial strings as well. All **Dial Plan** entries used the XCAPI native SIP trunk are associated to **Routing Table** number **24**. The **Routing Table** details will be shown on the next page.

| Dial Plan | Name | Dialed digits | Routing Table | Acc. code | Classes of service | Emergency |
|-----------|------------|----------------------|---------------|-----------|-------------------------------------|--------------------------|
| 46 | anynodeSBC | 0C8195816-Z | 46 | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 47 | anynodeSBC | 0C0-53638195816-Z | 47 | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 48 | anynodeSBC | 0C00-4953638195816-Z | 48 | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 49 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 50 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 51 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 52 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 53 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 54 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 55 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 56 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 57 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 58 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 59 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 60 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 61 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 62 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 63 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 64 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 65 | Standard | | * | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 66 | XCAPI | 92CZ | 24 | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 67 | XCAPI | 92C0-Z | 24 | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 68 | XCAPI | 92C1Z | 24 | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 69 | XCAPI | 92C0Z | 24 | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 70 | XCAPI | 92C00-Z | 24 | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 71 | XCAPI | 819581692CZ | 24 | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
| 72 | XCAPI | 5363819581692CZ | 24 | | <input checked="" type="checkbox"/> | <input type="checkbox"/> |

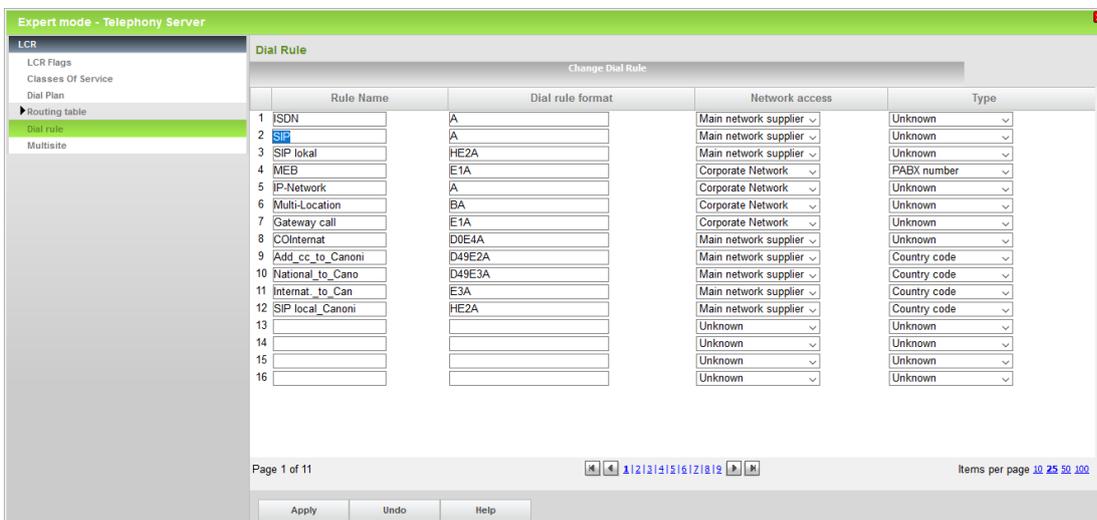
3.8 Routing Table

The **Routing Table** is set to the intended **Route** and **Dial Rule** for the XC-API native SIP trunk. Some of the other settings might be enabled on demand.



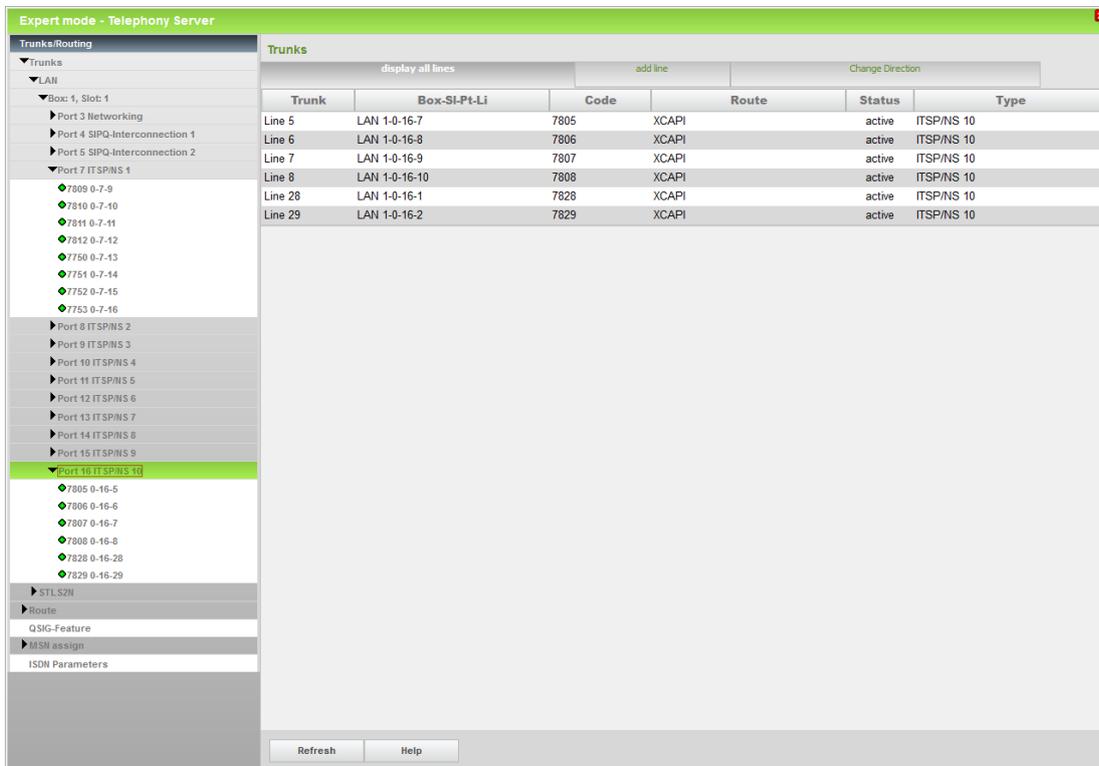
3.9 Dial Rule

The associated **Dial Rule**, here the second entry with the dial rule name **SIP**, is set to the **Dial rule format A** as shown below. When needed, use a rule and dial rule format that matches the local numbering requirements.



3.10 Trunk Lines

The required amount of trunk lines must be added and associated to the appropriate XCAPI route, in this example **Type ITSP/NS10**. Ensure that those trunk lines are properly licensed and available, otherwise their states won't become active (please check with the according **Status** column!).



The screenshot shows the 'Trunks' configuration page in the 'Expert mode - Telephony Server' interface. The left sidebar shows a tree view of the configuration, with 'Port 16 ITSP/NS 10' selected. The main area displays a table of trunk lines with the following data:

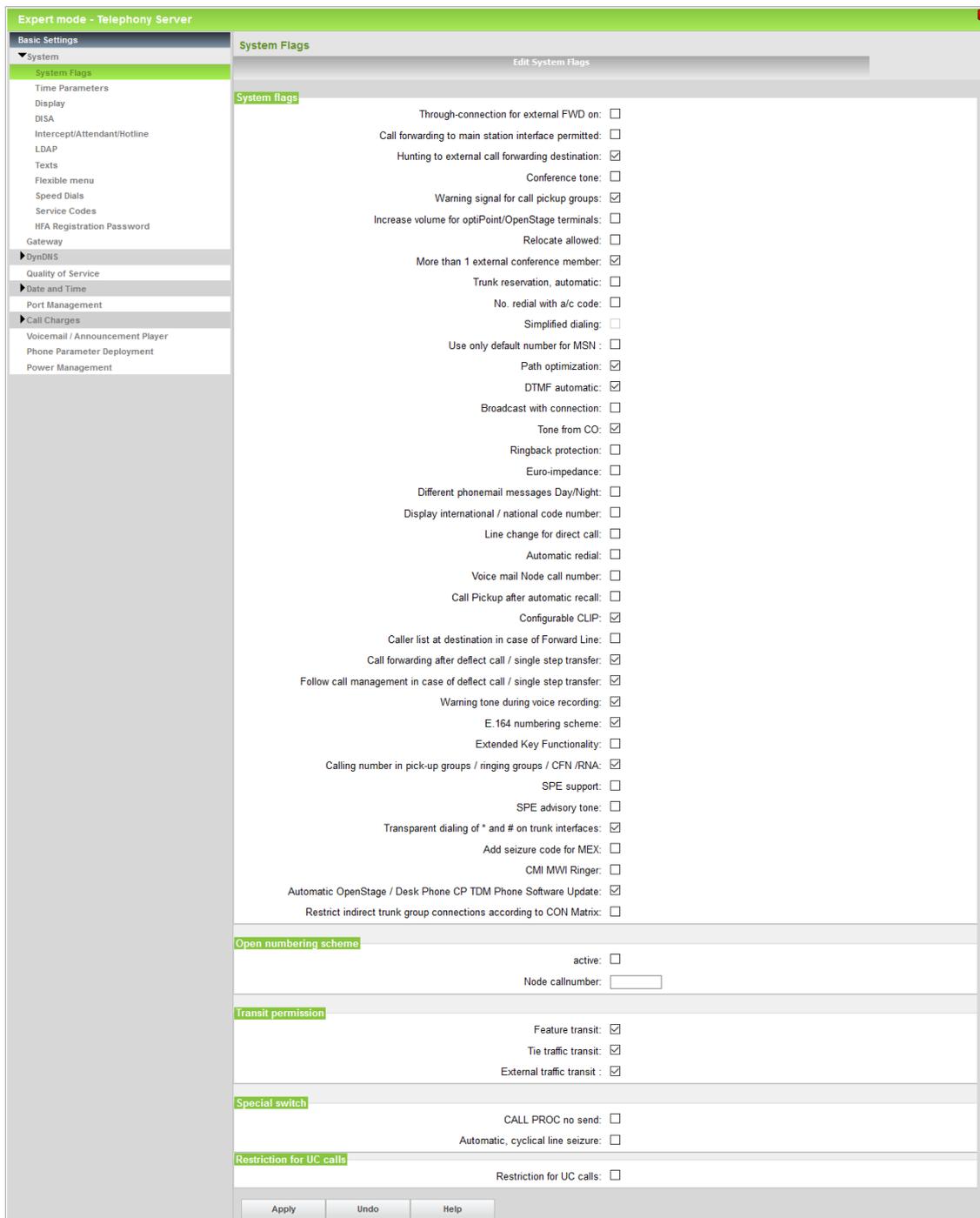
| Trunk | Box-SI-Pt-Li | Code | Route | Status | Type |
|---------|---------------|------|-------|--------|------------|
| Line 5 | LAN 1-0-16-7 | 7805 | XCAPI | active | ITSP/NS 10 |
| Line 6 | LAN 1-0-16-8 | 7806 | XCAPI | active | ITSP/NS 10 |
| Line 7 | LAN 1-0-16-9 | 7807 | XCAPI | active | ITSP/NS 10 |
| Line 8 | LAN 1-0-16-10 | 7808 | XCAPI | active | ITSP/NS 10 |
| Line 28 | LAN 1-0-16-1 | 7828 | XCAPI | active | ITSP/NS 10 |
| Line 29 | LAN 1-0-16-2 | 7829 | XCAPI | active | ITSP/NS 10 |



Important: a newly created Native SIP trunk may require a Unify OpenScape Business gateway reboot to become active.

3.11 System Parameter Flags

The Transit permission flags must be enabled, others upon local requirements.



The screenshot shows the 'Expert mode - Telephony Server' configuration window. The left sidebar contains a tree view with 'System Flags' selected. The main area is titled 'System Flags' and 'Edit System Flags'. It contains a list of system flags with checkboxes, many of which are checked. Below the list are sections for 'Open numbering scheme', 'Transit permission', 'Special switch', and 'Restriction for UC calls', each with its own set of checkboxes. At the bottom, there are 'Apply', 'Undo', and 'Help' buttons.

| System Flag | Checked |
|--|-------------------------------------|
| Through-connection for external FWD on: | <input type="checkbox"/> |
| Call forwarding to main station interface permitted: | <input type="checkbox"/> |
| Hunting to external call forwarding destination: | <input checked="" type="checkbox"/> |
| Conference tone: | <input type="checkbox"/> |
| Warning signal for call pickup groups: | <input checked="" type="checkbox"/> |
| Increase volume for optiPoint/OpenStage terminals: | <input type="checkbox"/> |
| Relocate allowed: | <input type="checkbox"/> |
| More than 1 external conference member: | <input checked="" type="checkbox"/> |
| Trunk reservation, automatic: | <input type="checkbox"/> |
| No. redial with a/c code: | <input type="checkbox"/> |
| Simplified dialing: | <input type="checkbox"/> |
| Use only default number for MSN: | <input type="checkbox"/> |
| Path optimization: | <input checked="" type="checkbox"/> |
| DTMF automatic: | <input checked="" type="checkbox"/> |
| Broadcast with connection: | <input type="checkbox"/> |
| Tone from CO: | <input checked="" type="checkbox"/> |
| Ringback protection: | <input type="checkbox"/> |
| Euro-impedance: | <input type="checkbox"/> |
| Different phonemail messages Day/Night: | <input type="checkbox"/> |
| Display international / national code number: | <input type="checkbox"/> |
| Line change for direct call: | <input type="checkbox"/> |
| Automatic redial: | <input type="checkbox"/> |
| Voice mail Node call number: | <input type="checkbox"/> |
| Call Pickup after automatic recall: | <input type="checkbox"/> |
| Configurable CLIP: | <input checked="" type="checkbox"/> |
| Caller list at destination in case of Forward Line: | <input type="checkbox"/> |
| Call forwarding after deflect call / single step transfer: | <input checked="" type="checkbox"/> |
| Follow call management in case of deflect call / single step transfer: | <input checked="" type="checkbox"/> |
| Warning tone during voice recording: | <input checked="" type="checkbox"/> |
| E. 164 numbering scheme: | <input checked="" type="checkbox"/> |
| Extended Key Functionality: | <input type="checkbox"/> |
| Calling number in pick-up groups / ringing groups / CFN /RNA: | <input checked="" type="checkbox"/> |
| SPE support: | <input type="checkbox"/> |
| SPE advisory tone: | <input type="checkbox"/> |
| Transparent dialing of * and # on trunk interfaces: | <input checked="" type="checkbox"/> |
| Add seizure code for MEX: | <input type="checkbox"/> |
| CMI MWI Ringer: | <input type="checkbox"/> |
| Automatic OpenStage / Desk Phone CP TDM Phone Software Update: | <input checked="" type="checkbox"/> |
| Restrict indirect trunk group connections according to CON Matrix: | <input type="checkbox"/> |

| Open numbering scheme | Checked |
|-----------------------|--------------------------|
| active: | <input type="checkbox"/> |
| Node callnumber: | <input type="text"/> |

| Transit permission | Checked |
|---------------------------|-------------------------------------|
| Feature transit: | <input checked="" type="checkbox"/> |
| Tie traffic transit: | <input checked="" type="checkbox"/> |
| External traffic transit: | <input checked="" type="checkbox"/> |

| Special switch | Checked |
|-----------------------------------|--------------------------|
| CALL PROC no send: | <input type="checkbox"/> |
| Automatic, cyclical line seizure: | <input type="checkbox"/> |

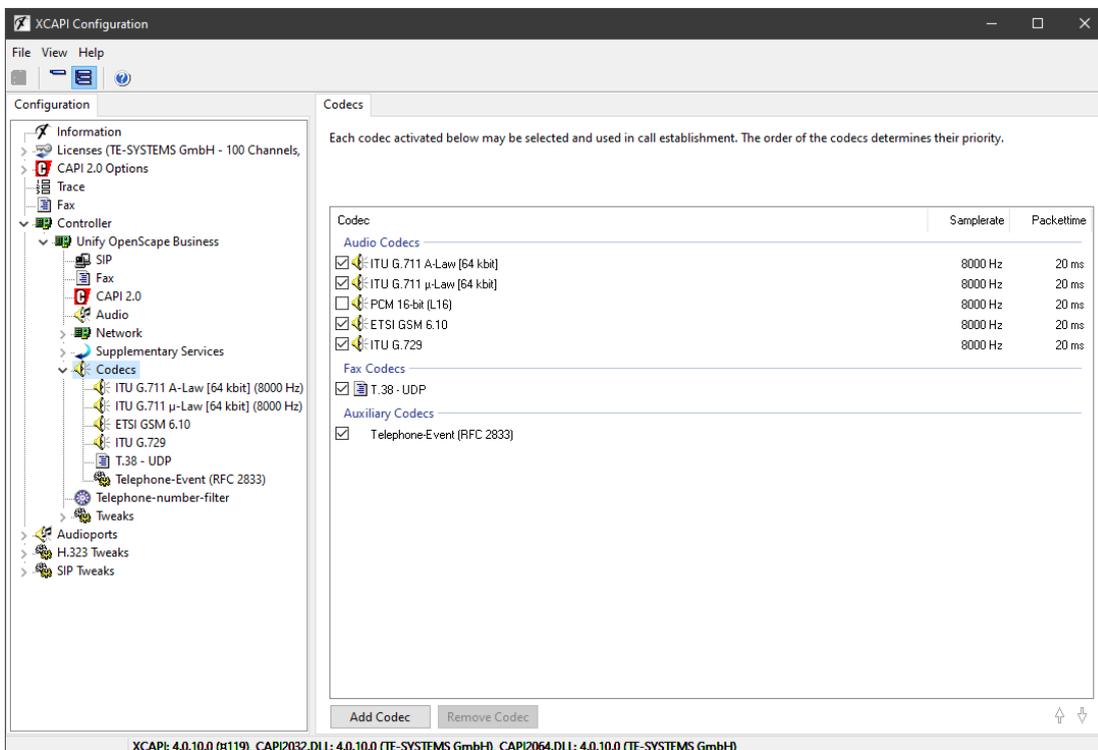
| Restriction for UC calls | Checked |
|---------------------------|--------------------------|
| Restriction for UC calls: | <input type="checkbox"/> |

Appendix

The **Appendix** chapter gives information and configuration hints as well as other considerations. If using the XCAPI controller wizard with its Unify OpenScope Business template, most of the shown configurations are set by default. The following topics and the shown configurations should still be reviewed, checked, and tested, especially with the participating stations in combination with the other trunk bindings of the Unify OpenScope Business environment. Some of the features, codecs and fax methods might not be supported in conjunction with specific Unify OpenScope Business trunk bindings.

4.1 Codecs

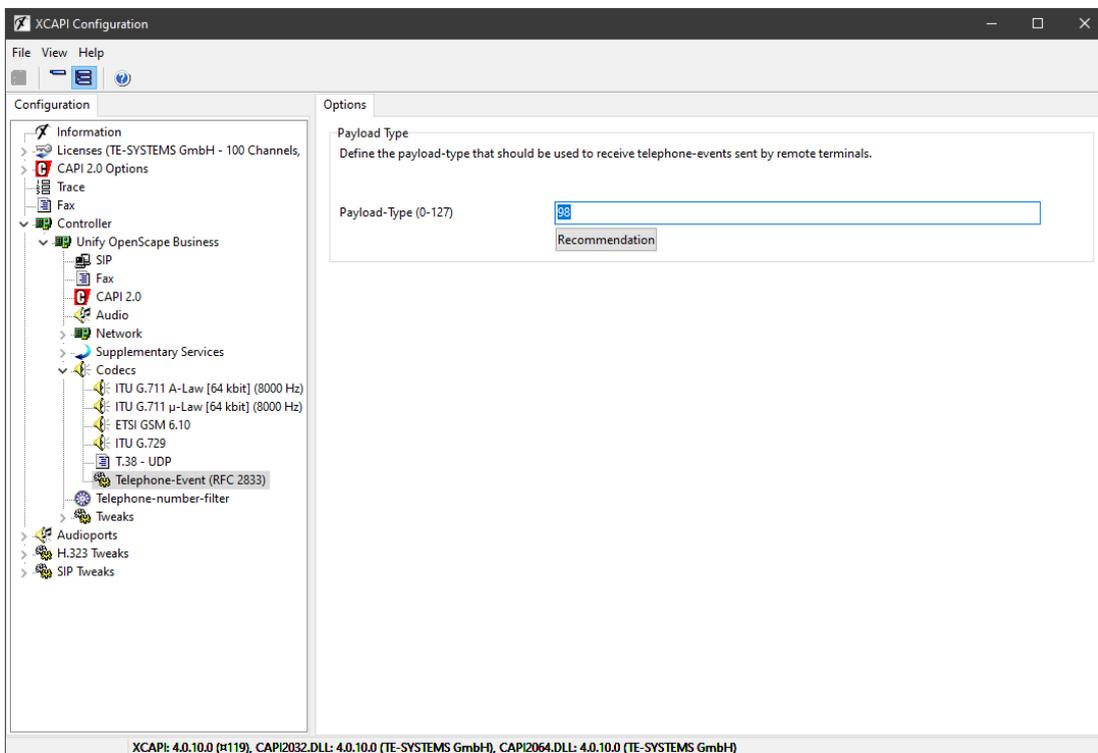
The codec defaults of the XCAPI SIP controller usually work fine and should only be changed if requested by our support team. When needed, please review with the codec related chapters **Codecs** on [page 18](#), **RFC2833 Payload** on [page 19](#), and **Fax Support** from [page 20](#). The Unify OpenScope Business **Codec Parameters** chapter is described from [page 11](#).



Important: Incorrect configurations can result in codec negotiation errors and thus may lead into fax and voice malfunction of the SIP trunk.

4.2 RFC2833 Payload

As mentioned in the **Codec Parameters** chapter of the Unify OpenScope Business, those RFC2833 options must be enabled. The RFC2833 payload value must be set conform for the XCAPI SIP controller and the PBX. By default both should be set to value **98**. If required, the payload value can be changed as shown below.



4.3 Fax Support

This chapter refers to Fax related topics about leveraging T.38, Softfax (G.711) and T.38 to Softfax fallback.

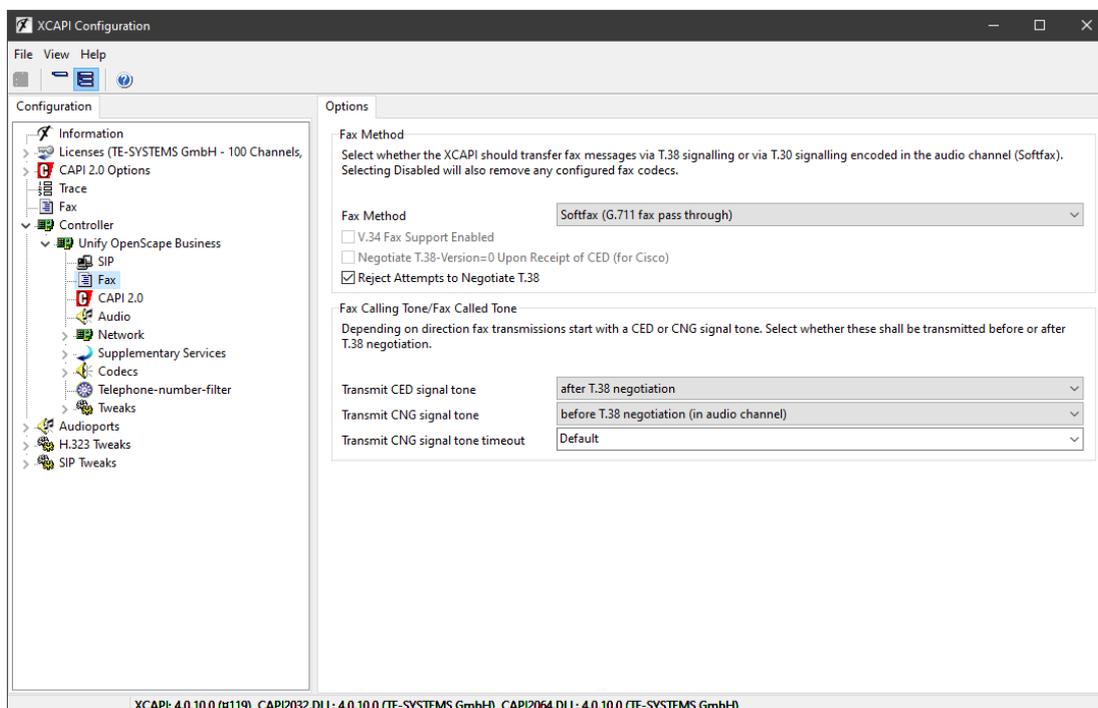
It must be ensured that the fax method, codecs, framing, bandwidth and DTMF settings are properly set and conform to the ones of the XCAPI SIP controller and other participating instances (SBC's, ITSP's etc.) connected to the Unify OpenScope Business. We always recommend using the latest XCAPI version and manufacturer releases.



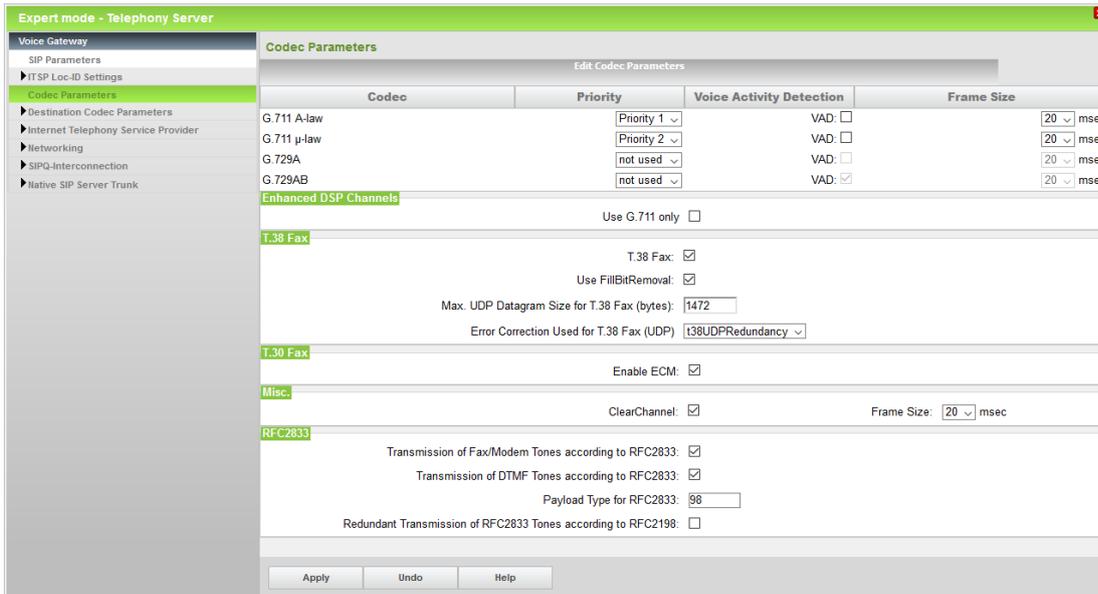
Always use the recommended and supported Fax method between the Unify OpenScope Business and the SIP carrier for the XCAPI controller. For most known SIP trunk scenarios G.711 Fax Pass Through should be the first choice!

4.3.1 G.711 Fax Pass Through (Softfax)

With the Softfax mode, the XCAPI simulates an analogue Fax device by transmitting modulated Fax-signals modem-like through the established G.711 audio channels. For enabling the **Softfax (G.711 Fax Pass Through)** method, it must be selected as shown in the next screenshot.



The **Codec Parameters** of the Unify OpenScape Business gateway should be set as shown below.



The screenshot shows the 'Expert mode - Telephony Server' configuration window. The left sidebar lists navigation options: Voice Gateway, SIP Parameters, ITSP Loc-ID Settings, Codec Parameters (selected), Destination Codec Parameters, Internet Telephony Service Provider, Networking, SIPQ-Interconnection, and Native SIP Server Trunk. The main area is titled 'Codec Parameters' and 'Edit Codec Parameters'. It contains a table with columns for Codec, Priority, Voice Activity Detection, and Frame Size. Below the table are sections for 'Enhanced DSP Channels', 'T.38 Fax', 'T.30 Fax', 'Misc.', and 'RFC2833'.

| 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 FII/BitRemoval:
Max. UDP Datagram Size for T.38 Fax (bytes): 1472
Error Correction Used for T.38 Fax (UDP): 138UDPRedundancy

T.30 Fax
Enable ECM:

Misc.
ClearChannel: Frame Size: 20 msec

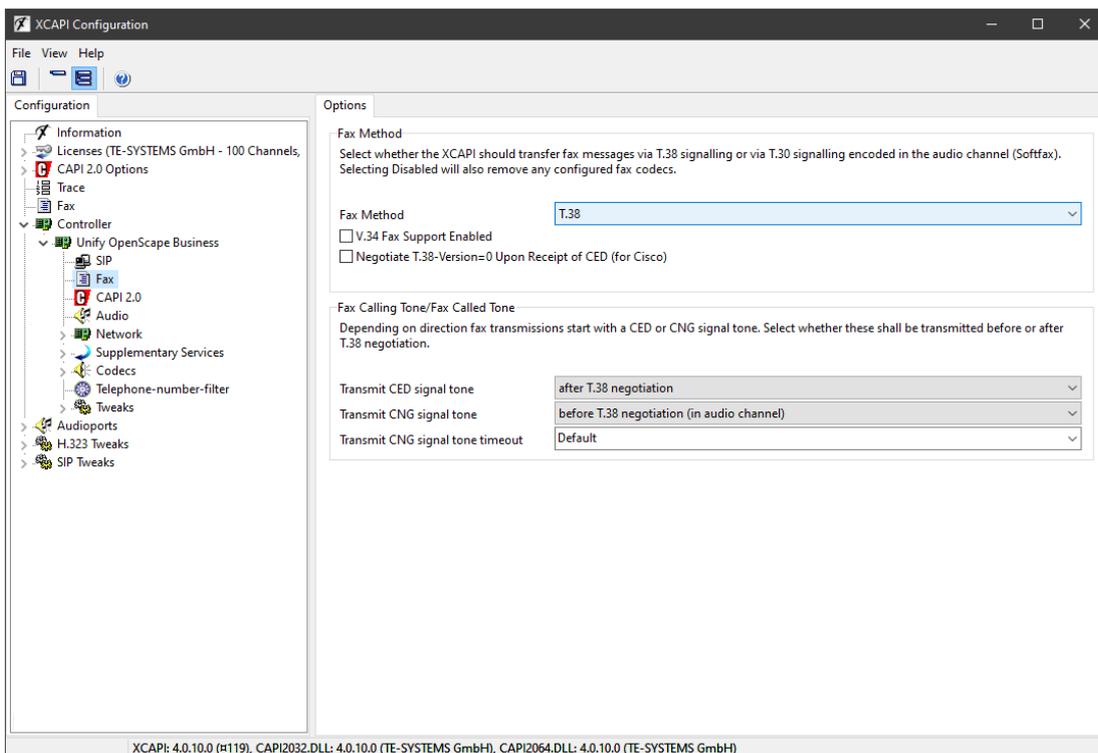
RFC2833
Transmission of Fax/Modem Tones according to RFC2833:
Transmission of DTMF Tones according to RFC2833:
Payload Type for RFC2833: 98
Redundant Transmission of RFC2833 Tones according to RFC2198:



For G.711 Fax Pass Through (Softfax) integration, the gateways **T.38 Fax** flag can usually left enabled. Under certain conditions and in case of specific Fax interoperability issues between the Unify OpenScape Business gateway with its other ITSP and SBC bindings, the **T.38 Flag** might have to be disabled. In any case, the **Transmission of Fax/Modem Tones according to RFC2833** flag must be enabled and **Redundant Transmission of RFC2833 Tones according to RFC2198** must be disabled.

4.3.2 T.38

In the case of T.38 usage, this protocol must also be the recommended and supported fax method between the Unify OpenScope Business and its carrier trunk binding. It is recommended to avoid transcoding (G.711 to T.38 or vice versa) and using matching fax methods for all participating VoIP instances. Please note that it is mandatory that the **T.38 - UDP** and at least one voice codec (for the initial call establishment) is enabled for the XCAPI controller, what it is by default. An example of the default codec setup can be reviewed in the chapter from [page 25](#). For XCAPI, T.38 has to be enabled as shown on the next screenshot. For the Unify OpenScope Business, the **T.38 Fax** attribute must be enabled in the **Codec Parameters** as shown in the same named chapter from [page 11](#).

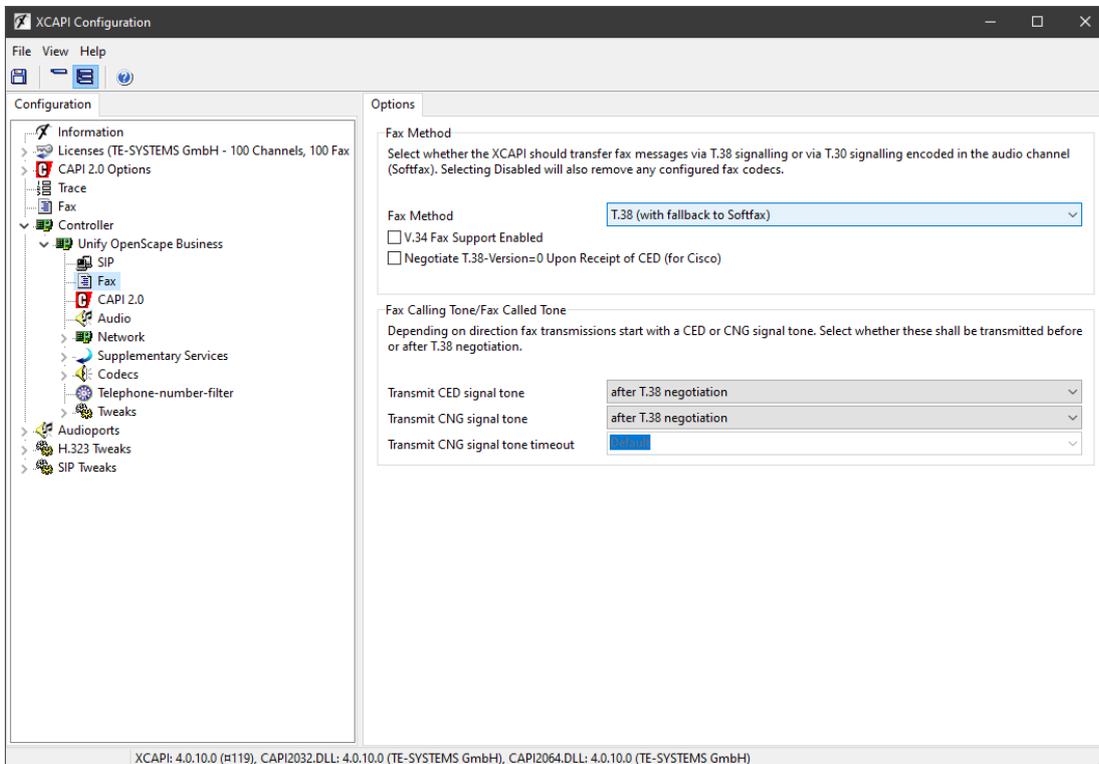


Please also check with the Unify OpenScope Business documentation for T.38 related limitations and recommendations for fax support, especially in conjunction with other ITSP, SBC bindings.

Important: **Never** enable V.34 as this protocol is not supported by the Unify OpenScope Business gateway nor change any of the T.38 related codec settings (Rate Management, T.38 Version) in the XCAPI SIP controller. In any case, this will predominantly force fax handshake and transmission failures.

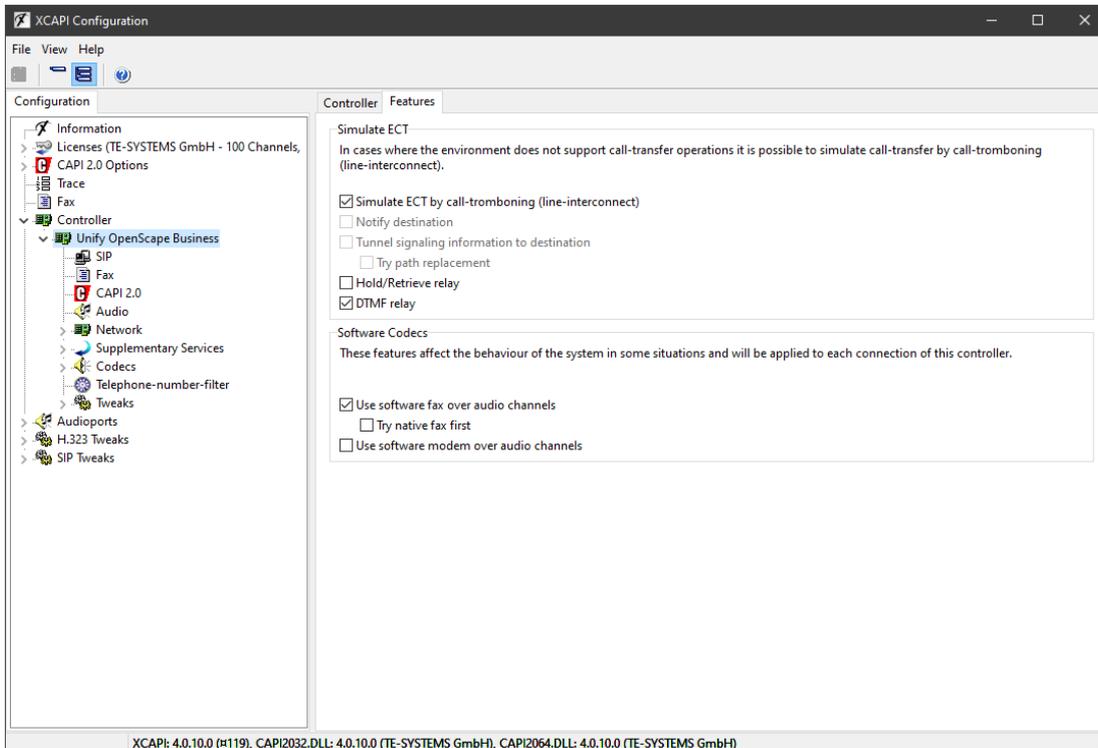
4.3.3 T.38 to G.711 Fax Pass Through Fallback

The fax fallback can be enabled as shown on the screenshot below. It is important to check if this fax mode is supported by all participating VoIP instances, especially in case of cross-compatibility due to participating SBC's or connected ITSP's. Depending on the VoIP environment with its participating instances, additional configurations and adjustments might be required.



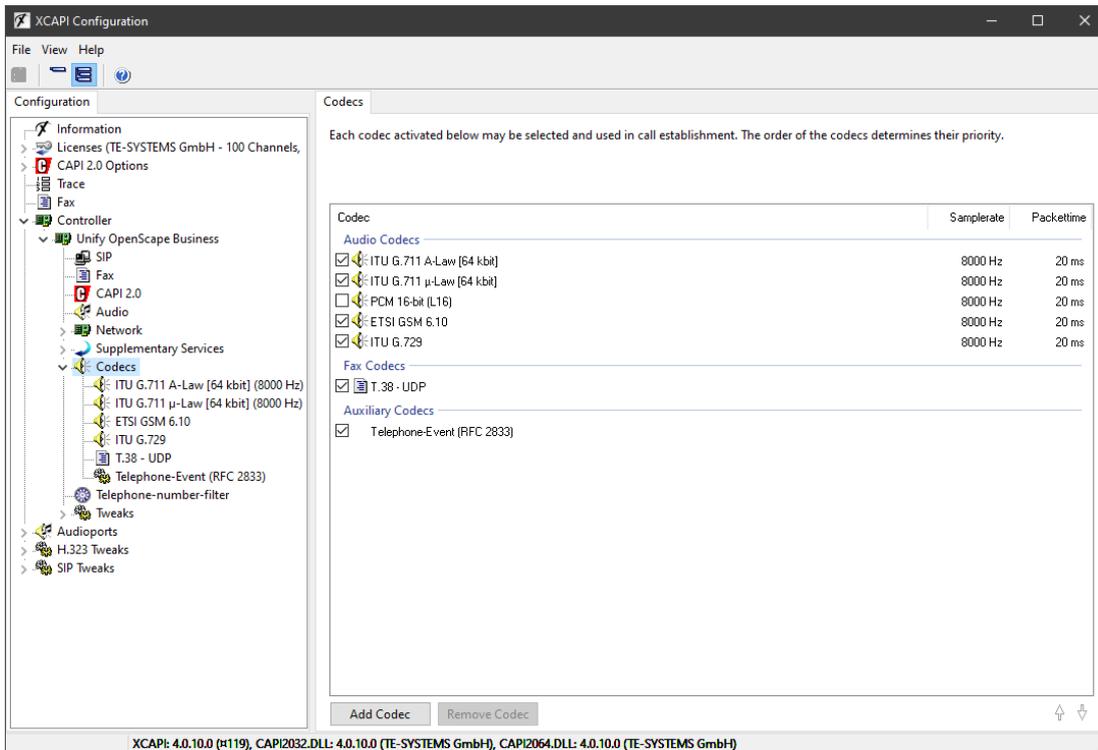
4.4 Simulated Call Transfer

Even though it is recommended to use the previously described call transfer via SIP refer, in some application specific cases the **Simulated Call Transfer** has to be used. 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. When needed, check the **Features** tab of the XCAPI controller and ensure that the **Simulate ECT by call-tromboning (line-interconnect)** parameter is set.



4.5 Codecs

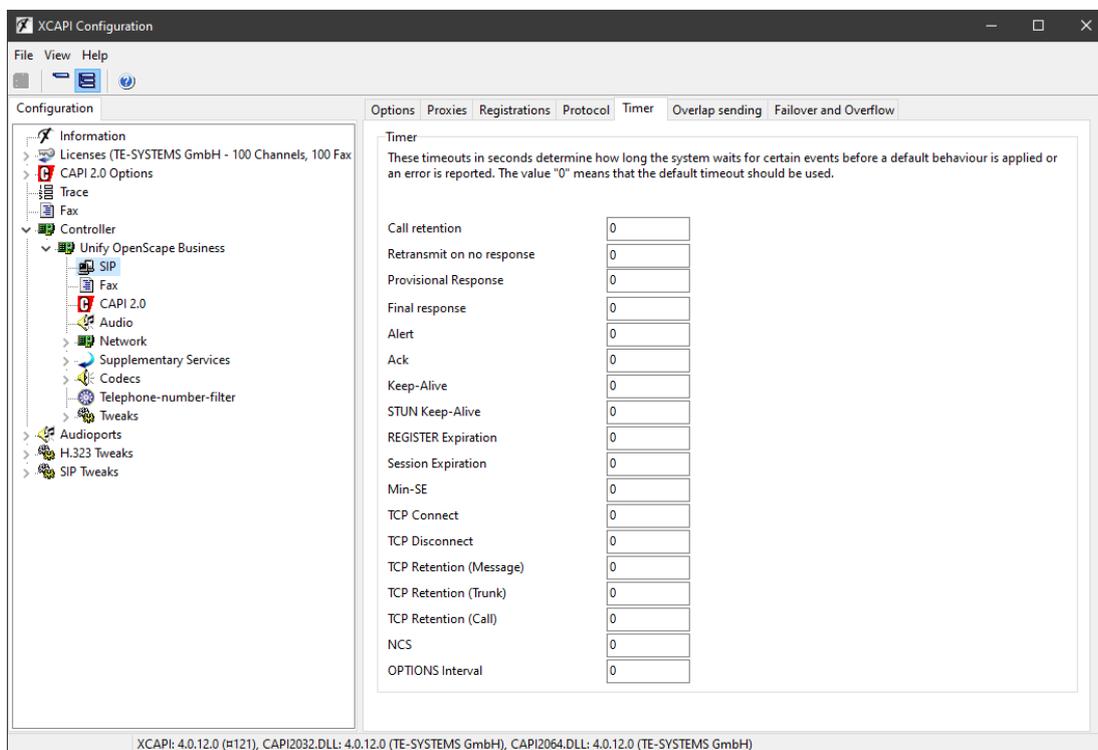
The screenshot below shows the default codec settings of the XCAPI SIP controller. As a general rule, there is no need to change anything here.



4.6 Timer

The XCAPI SIP controller timer values should only be adjusted if needed. The displayed timer value **0** always refers to a constant timer value based on standard RFC and protocol regulations. For example, with **MIN-SE** and **Session Expiration** default value **0**, the XCAPI SIP controller uses **Min-SE: 90** and **Session-Expires: 300** as SIP timer defaults.

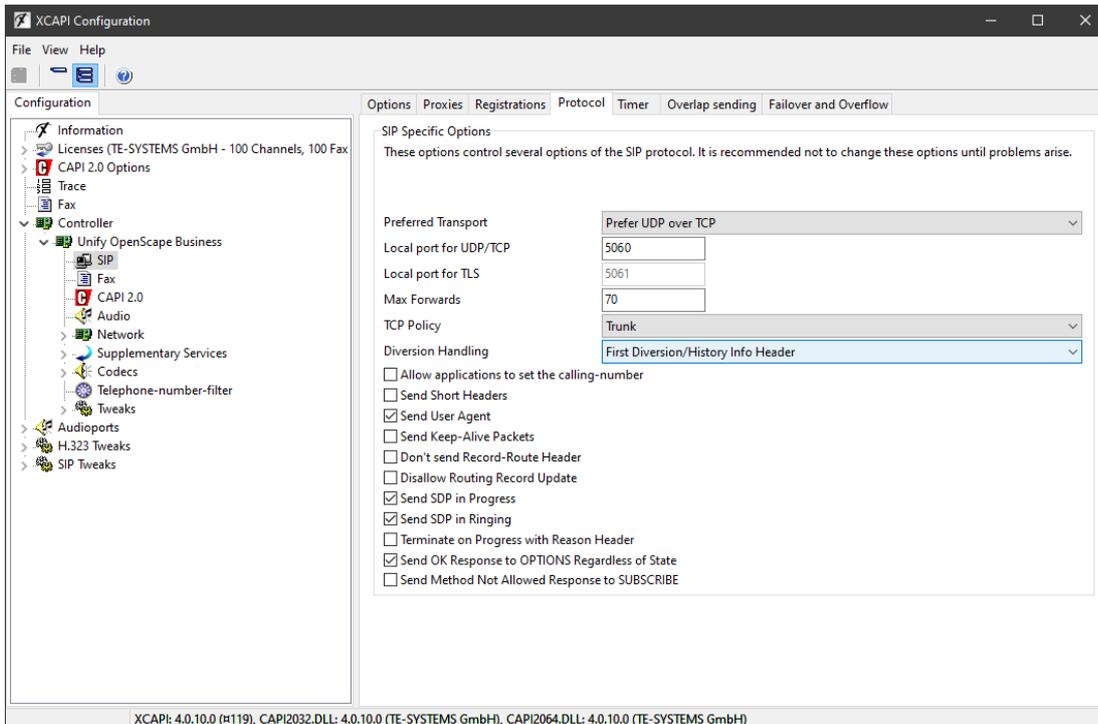
As previously mentioned in the **SIP Parameters** chapter on [page 9](#), if the timer defaults of the Unify OpenScope Business were changed, value **0** should also be adjusted to the conform values in seconds.



4.7 Diversion Handling

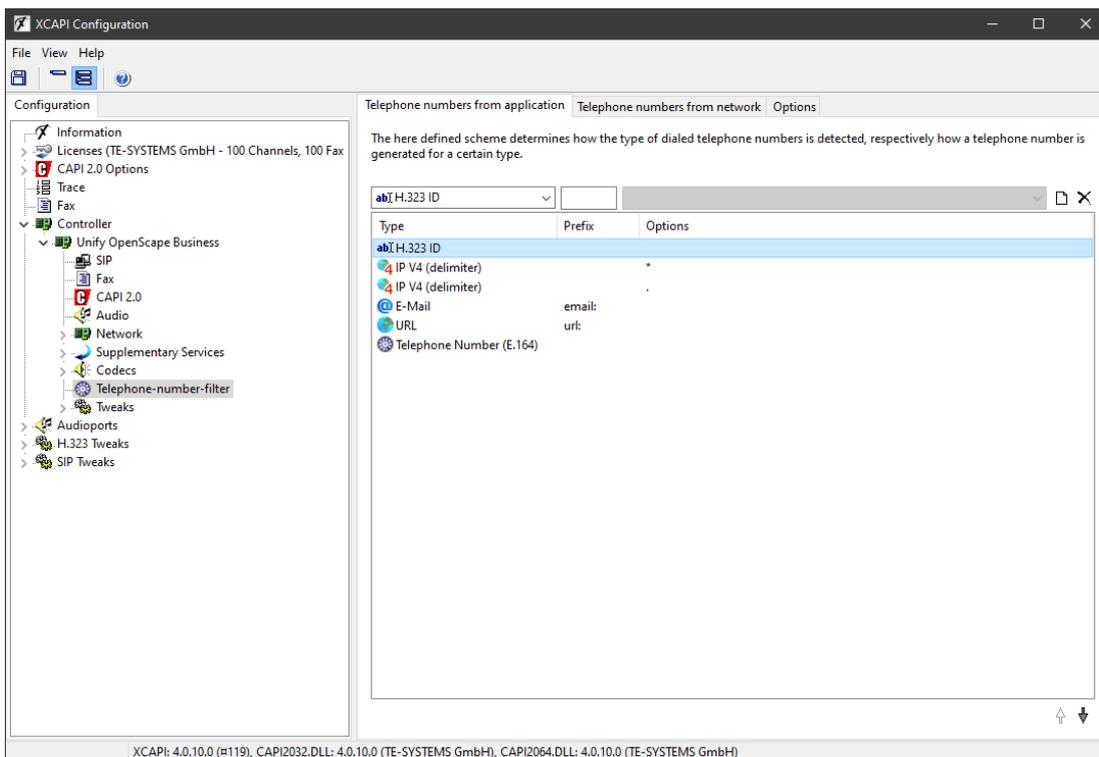
For some call scenarios, the CAPI 2.0 application might be aware of the first or last provided redirection number of the PBX. If you desire, this behavior can be changed via the **Diversion Handling** option in XCAPI's SIP controller protocol tab as shown on the screenshot below. This implements that the Unify OpenScope Business gateway delivers this information via the SIP **Diversion** header towards XCAPI.

For this, the **CLIP outgoing in Diversion header** must be enabled for XCAPI's native SIP trunk. This header has to be specified in the CLIP/CLIR section of the **Extended SIP Data** as shown in the screenshot of the **Native SIP Server Trunk** chapter from [page 10](#).



4.8 Numbering

Please note, apart from very few exceptions, XCAPI can't do any numbering manipulations. It must be ensured that the PBX or VoIP instance and the CAPI 2.0 application is based on a uniform dial plan. That means, that the CAPI application and the Unify OpenScope Business must provide proper calling and called numbers. If required, additional numbering manipulations have to be done within the PBX. AS the CAPI 2.0 ISDN layer is basically only aware of dialed digits (from 0 to 9) and some numbering attributes (NPI/TON), other dial strings formats (for e.g. +49536381950) are not passed through XCAPI by default. There are also some CAPI 2.0 applications which are not even able to generate other dial strings. If it's required that XCAPI pass dial strings like +49536381950, the controller has to be adjusted through the **Telephone-number-filter** settings. In the **Telephone numbers from application** tab, select the **H.323 ID** type and delete its **h323id:** related **Prefix** and move this entry to the top of the list as shown on the next screenshot.



Important: There is no validation check of the provided dial string. XCAPI would even pass non-valid dial strings which may lead to malfunctions and call failures.

Unify Ready Technology Connectivity Certification

The official certificate and test report is available on the [Unify Technology Partners Extranet](#).



The image shows a certificate for Unify Ready Technology Connectivity Certification. It features the Unify logo at the top left. The main title is 'Atos Unify Ready Technology connectivity certification'. Below this, it states 'The connectivity of XCAPI V4 developed by TE-Systems GmbH has been certified at the SIP-Interface of Atos Unify OpenScope Business V3 in accordance with the respective test report, dated April 16th, 2021'. A disclaimer follows: 'The test was conducted conforming to DIN EN ISO 9001. This certificate is only valid in conjunction with the full test report and the notes contained therein. Please consider that the test report only covers the functionality of the interface. The certificate and test report are not good for a statement of end-to-end functionality.' The date 'Munich, April 19th, 2021' is listed, along with a signature and the name 'Andre Bergmann, Director Technology Partner Program'. At the bottom left is a 'Unify Ready' badge, and at the bottom right is the 'Atos' logo. The footer of the certificate reads 'Trusted partner for your Digital Journey'.

 Unify

Atos Unify Ready Technology connectivity certification

The connectivity of
XCAPI V4

developed by TE-Systems GmbH has been certified at the SIP-Interface of Atos Unify OpenScope Business V3 in accordance with the respective test report, dated April 16th, 2021

The test was conducted conforming to DIN EN ISO 9001. This certificate is only valid in conjunction with the full test report and the notes contained therein. **Please consider that the test report only covers the functionality of the interface. The certificate and test report are not good for a statement of end-to-end functionality.**

Munich, April 19th, 2021


Andre Bergmann
Director Technology Partner Program



Trusted partner for your **Digital Journey**



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