

Unify OpenScape 4000 V8 & V10



April 20, 2021

## Contents

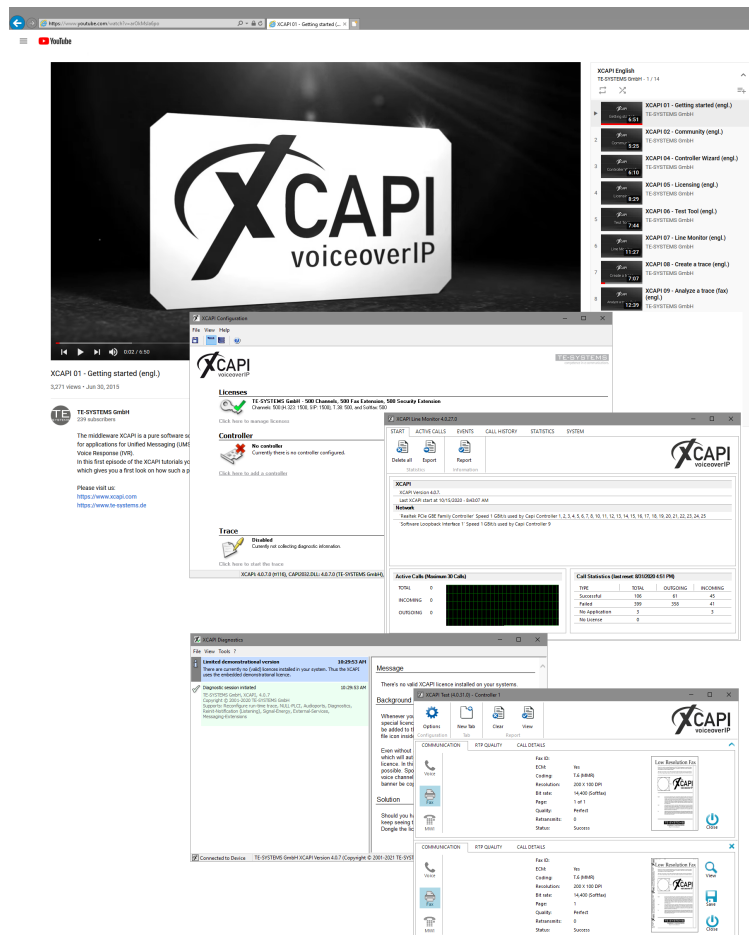
<b>1</b>	<b>Introduction</b>	<b>3</b>
<b>2</b>	<b>XC-API Configuration</b>	<b>4</b>
2.1	Voice-over-IP Environment	5
2.2	Description and Channels	5
2.3	IP Address of the Unify OpenScape 4000	6
2.4	Network Interface	6
2.5	Port Allocation	7
2.6	Confirmation	7
<b>3</b>	<b>Unify OpenScape 4000 Configuration</b>	<b>9</b>
3.1	Configuring Mounting Locations for Modules in the SWU (BCSU)	9
3.2	Configuration of Global- and Feature-Board Data for HG3500 (CGWB)	10
3.3	Trunk Groups (BUEND)	11
3.4	Digital Trunks (TDCSU)	11
3.5	Trunk Routing (RICHT)	12
3.6	LCR Outdial Rule (LODR)	12
3.7	LCR Routes (LDAT)	13
3.8	Digit Analysis (WABE)	13
3.9	LCR Dialplan (LDPLN)	14
3.10	Class of Services	15
3.11	Class of Services (COSSU)	15
3.12	Class of Trunks (COT)	16
3.13	Class of Parameters (COP)	16
3.14	Reference Clock Table (REFTA)	17
3.15	Gateway Properties	17
3.16	SIP Parameters	18
3.17	SIP Trunk Profiles	19
3.18	Payload	21
3.19	Codec Parameters	22
3.20	ISDN Classmarks	22
<b>4</b>	<b>Appendix</b>	<b>23</b>
4.1	Fax Support	23
4.1.1	G.711 Fax Pass Through (Softfax)	24
4.1.2	G.711 Fax Pass Through (Softfax) in Virtual Environments	25
4.1.3	T.38	26
4.1.4	T.38 to G.711 Fax Pass Through Fallback	27
4.2	Codecs	28
4.3	DTMF via RFC2833	29
4.4	Message Waiting Indication	30
4.5	Call Transfer	32
4.6	Diversion Handling	33
4.7	Numbering	34
<b>5</b>	<b>Unify Ready Technology Connectivity Certification</b>	<b>35</b>

## Introduction

This document is intended to support you with the integration of XCAPI into an environment of the Unify OpenScope 4000 series. In the following sections we describe the configuration steps for SIP trunking to allow optimal performance of both, XCAPI and the Unify OpenScope 4000. Though being based on the Unify OpenScope 4000 V8 & V10 and XCAPI 3.6.95 versions, this document is applicable with other versions given a few adjustments.

At this point we assume that the Unify OpenScope 4000 environment and the physical or virtual application server is available and accessible through the network. Application server in this context means, a server with a recent installed 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. Independent of 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\\).

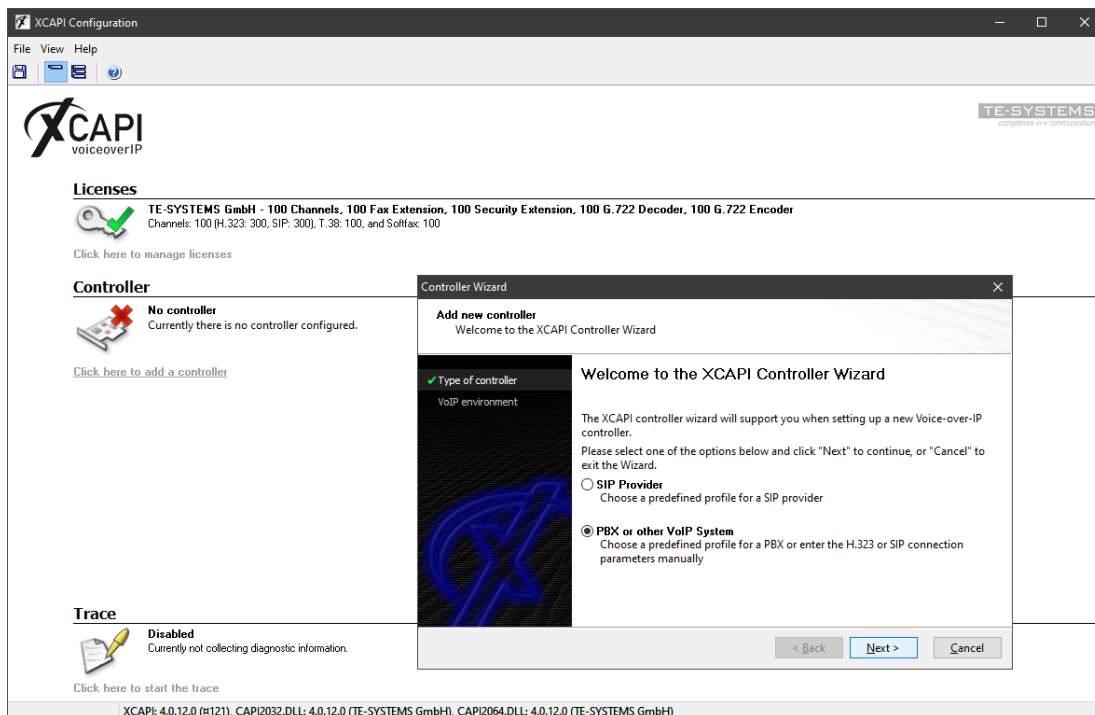
This test tool allows to check inbound and outbound calls, fax and testing several supplementary services. We recommend a visit to our [YouTube channel](#) frequently for XCAPI related tutorials about licensing, the test tool, line monitor, tracing, analyzing and others. Registered [community](#) users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



## XCAPI Configuration

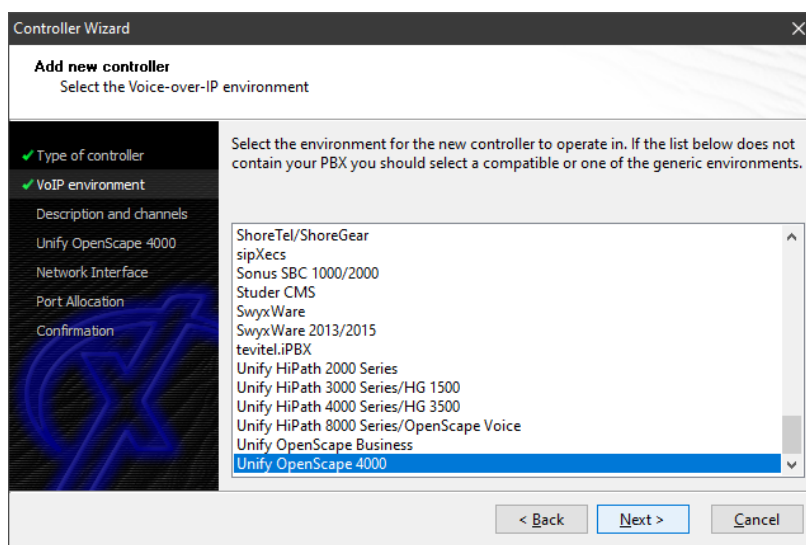
Please start up the XCAPI configuration to create a new controller assigned to the Unify OpenScape 4000.

If you've just installed the XCAPI and start the configuration tool for the first time or no controller is available at all, the XCAPI controller wizard will pop up automatically. To start up the XCAPI controller wizard manually, the hyperlink labeled **Click here to add a controller** on the main page has to be clicked. However, select **PBX or other VoIP System** in the initial **Type of controller** dialog and proceed with the **Next** button.



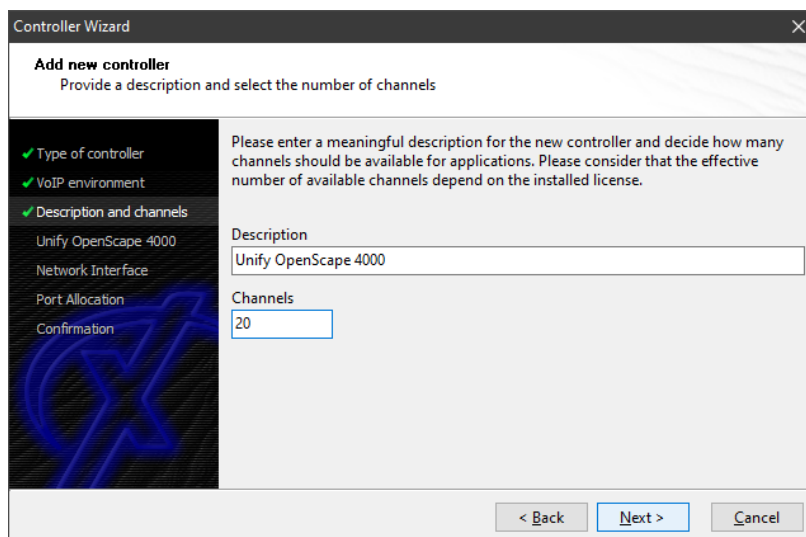
## 2.1 Voice-over-IP Environment

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.



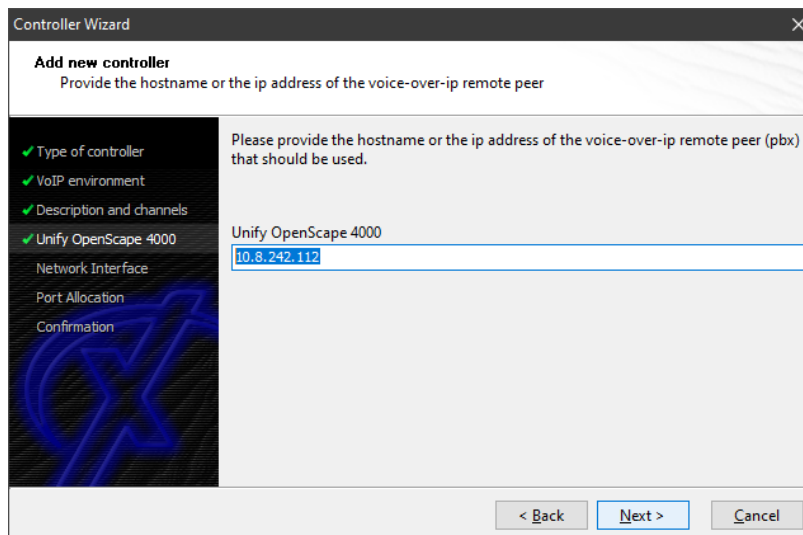
## 2.2 Description and Channels

When the VoIP environment was selected, the next dialog allows setting a name for the controller. Also the number of channels that the new controller will be able to provide can be set. Enter how many licensed simultaneous connections the XCAPI controller should handle when communicating with the Unify OpenScape 4000 and the bound CAPI application.



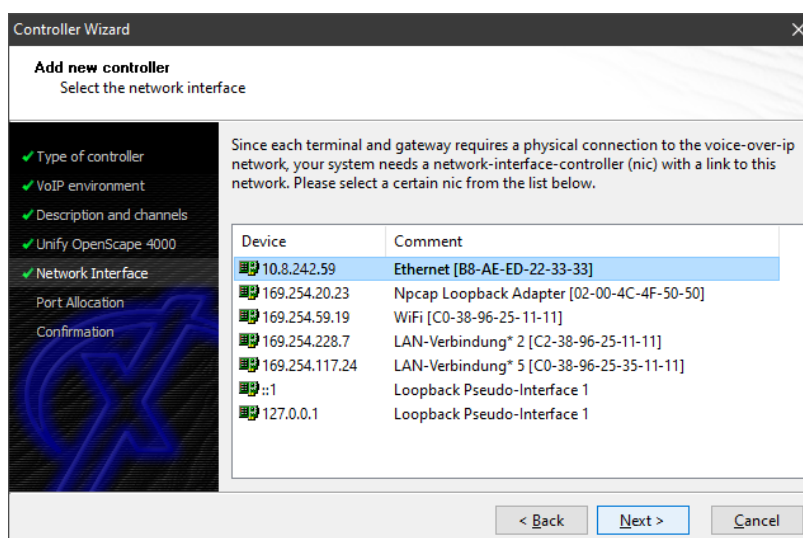
## 2.3 IP Address of the Unify OpenScape 4000

In the next wizard dialog, provide the IP address or host name of the SIP listening Ethernet interface of the Unify OpenScape 4000 gateway. The belonging Unify OpenScape 4000 gateway configurations of this example will be described in the chapter **Unify OpenScape 4000 Configuration** (sub-chapter **Configuring Mounting Locations for Modules in the SWU (BCSU)** and **Configuration of Global- and Feature-Board Data for HG3500 (CGWB)**) from [page 9](#). Please note that both, the XC-API controller and the Unify OpenScape 4000, 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.



Device	Comment
10.8.242.59	Ethernet [B8-AE-ED-22-33-33]
169.254.20.23	Npcap Loopback Adapter [02-00-4C-4F-50-50]
169.254.59.19	WiFi [C0-38-96-25-11-11]
169.254.228.7	LAN-Verbindung* 2 [C2-38-96-25-11-11]
169.254.117.24	LAN-Verbindung* 5 [C0-38-96-25-35-11-11]
:::1	Loopback Pseudo-Interface 1
127.0.0.1	Loopback Pseudo-Interface 1

## 2.5 Port Allocation

When needed and in the case of any router or firewall restrictions for UDP (RTP/T.38) a port range can be set that will be used by the XCAPI controller for addressing the source port. If none is set here, the XCAPI controller uses a random UDP port range between 1024 and 65535.

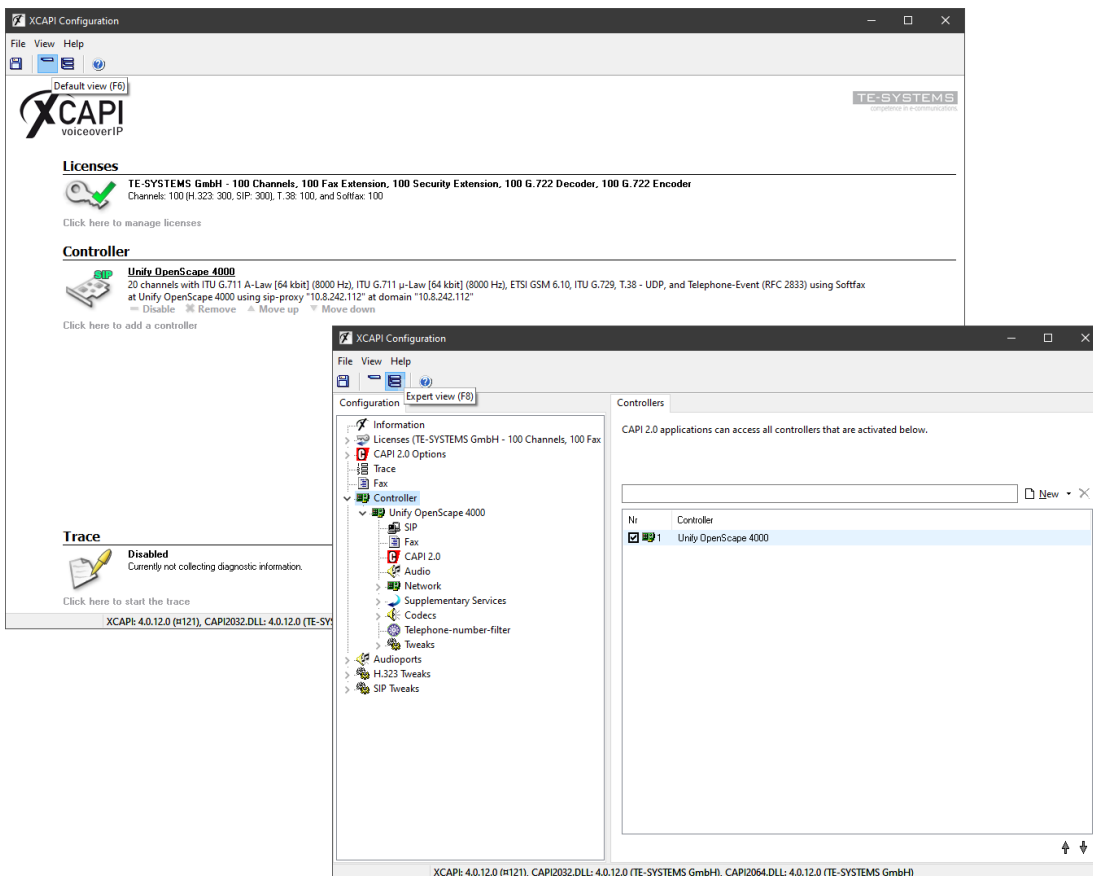
The screenshot shows the 'Controller Wizard' dialog box with the title 'Add new controller' and the subtitle 'Provide information about port allocation'. On the left, a list of steps is shown with green checkmarks: 'Type of controller', 'VoIP environment', 'Description and channels', 'Unify OpenScape 4000', 'Network Interface', 'Port Allocation' (highlighted), and 'Confirmation'. The main area contains the text: 'If you want to operate this system behind a router/gateway it might be necessary to constrain local udp ports to a certain range.' Below this is a checkbox labeled 'Constrain local udp ports to the following range' which is currently unchecked. To the right of the checkbox are two input fields: the first contains '10000' and the second contains '10120', separated by a hyphen. At the bottom right are three buttons: '< Back', 'Next >' (highlighted with a blue border), and 'Cancel'.

## 2.6 Confirmation

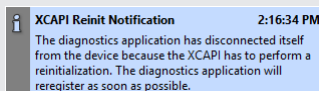
The final wizard dialog performs some checks on the configuration parameters you've made. If errors are detected, use the **Back** button to find the incorrect dialogs and correct them. Push **Finish** in order to finally create the new controller if everything is correct.

The screenshot shows the 'Controller Wizard' dialog box with the title 'Add new controller' and the subtitle 'Confirm that the provided information is correct'. On the left, the same list of steps is shown, but 'Confirmation' is now highlighted with a blue background. The main area contains the text: 'Click Finish to add the new controller with the configuration you have had made.' At the bottom right are three buttons: '< Back', 'Finish' (highlighted with a blue border), and 'Cancel'.

The newly created SIP controller is now listed on the **Default** and **Expert view** of XCAPI configuration tool. Use the **Save** button and exit the tool.



Note that the CAPI 2.0 application with its services must always be restarted to take effect on the XCAPI controller changes. 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 pop-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 OpenScape 4000 Configuration

In order to establish a connection between the XC-API and the Unify OpenScape 4000 gateway, you need to setup the XC-API as a communication gateway and configure the appropriate settings for using line- and direction channels. In this configuration example we'll use the MML (Man Machine Language) interface, its AMO (Administration and Maintenance Order) configuration dialog and the WBM (Web Based Management) for accessing and modifying the Unify OpenScape 4000 device.

For both we assume some general familiarity with. If there are doubts, please refer to the relevant documentation for further information.

### 3.1 Configuring Mounting Locations for Modules in the SWU (BCSU)

The AMO **BCSU** (Board Configuration in the Switching Unit) shows that for this example a Q2330-X module type vHG3500 with the related IP address **10.8.242.112** is used as LTU (Line Trunk Unit).

```

DISPLAY-BCSU:TYPE=TBL,LTG=1,LTU=99,SLOT=5;
H500: AMO BCSU STARTED

```

LTG	1	LTU	99	SRCGRP	1	ALARMNO-LTU	0	SOFTGATE1000	20	PORTS USED
PEN	ASSIGNED	MODULE	FCT	E W ARM P	INSERTED	HW-	MODULE	STATE	INFO	STATUS
5	Q2330-X	vHG3500	1	0	Q2330-X	1	-07	-	READY	
IP ADDRESS : 10. 8.242.112      B-CHANNELS : 120      BCHLCNT : 30 IP MODE : IPV4      DHCP V4 : NO      DHCP V6 : NO BLOCK NO : 6      PRERESERVED LINES ASSIGNED : NO 1. FUNCT : HG3550      4 LINES      B-CHANNELS : 120      BCHLCNT : 30										
NO SECURITY STATUS AVAILABLE, SINCE FEATURE SPE IS NOT ACTIVATED										

```

AMO-BCSU -111      BOARD CONFIGURATION, SWITCHING UNIT

```

### 3.2 Configuration of Global- and Feature-Board Data for HG3500 (CGWB)

The AMO CGWB (Common Gateway Board) is used for configuring device specific parameters. The **GLOBIF**, **ASC** and **JB** parameters are used as shown below. On demand check with the appropriate Codec, DTMF and Fax chapters of this TechNote.

```

DISPLAY-CGWB:LTU=99,SLOT=5;
H500: AMO CGWB STARTED
-----
|  CGW BOARD DATA  |
-----
|  HG3550  |
-----
|  LTU = 99      SLOT = 5      SMODE = NORMAL      POOLNO: 0  |
-----

GLOBAL DATA AND ETHERNET INTERFACE DATA - CONFIGURABLE VALUES:
-----
IPADR      = 10 .8 .242.112      TCP      =      (4060)
NETMASK    = 255.255.255.0      VLAN      = NO      (NO)
DEFRT      = 10 .8 .242.1      (0.0.0.0 = NOT CONFIGURED)
BTRATE     =      (CHECK OS)    VLANID    = 0      (0)
PATTERN    = 213 (213)          TLSP      = 4061 (4061)
TRPRSIP    = 60 (0)             TRPRH323 = 0 (0)
TRPRSIPQ   = 0 (0)             TPRH323A = 0 (0)
DNSIPADR   = 10 .8 .251.103     SIPTCPP   = 5060 (5060)
DNSIPADR2  = 0 .0 .0 .0        SIPTLSP   = 5061 (5061)
USEWANIF   = NO (NO)
WPUBIP     = 0 .0 .0 .0

ASC DATA - CONFIGURABLE VALUES:
-----
TOSPL      = 184 (184)          TOSSIGNL = 104 (104)
UDPPRTLO   = 29100 (29100)     UDPPRTHI = 30099 (30099)
T38FAX     = YES (YES)         REDRFCTN  = NO (YES)
RFCFMOIP   = YES (NO)         RFCDTMF   = YES (YES)

PRI01 : CODEC = G711A  VAD = NO      RTP-SIZE = 20
PRI02 : CODEC = G711U  VAD = NO      RTP-SIZE = 20
PRI03 : CODEC = NONE   VAD = NO      RTP-SIZE = 30
PRI04 : CODEC = G729   VAD = NO      RTP-SIZE = 30
PRI05 : CODEC = G729A  VAD = NO      RTP-SIZE = 30
PRI06 : CODEC = NONE   VAD = NO      RTP-SIZE = 20
PRI07 : CODEC = G729AB VAD = YES     RTP-SIZE = 20
PRI08 : CODEC = G722   VAD = NO      RTP-SIZE = 20
PRI09 : CODEC = OPUS   VAD = NO      RTP-SIZE = 20

DSP CONFIGURATION DATA
-----
JITBUFD    = 60 (60) (PARAMETER IS NOT USED ANYMORE)

JB DATA - CONFIGURABLE VALUES:
-----
JBMODE     = 2
AVGDLYV    = 40 (40)
MAXDLYV    = 120 (120) MINDLYV = 20 (20)
PACKLOSS   = 4 (4)
AVGDLYD    = 60 (60) MAXDLYD = 200 (200)

AMO-CGWB -111      CONFIGURATION OF HG3500 BOARD

```

### 3.3 Trunk Groups (BUEND)

A trunk group consists of several trunks and is identified with its unique trunk group number, see option **TGRP NUMBER**. The associated XCAPI trunk group is used as shown next.

```

DISPLAY-BUEND:TGRP=6;
H500: AMO BUEND STARTED
+-----+
| TGRP NUMBER :      6   TGRP NAME   : XCAPI           MAXIMUM NO. :   30 |
|                   :      CHARCON   : NEUTRAL          |
| SUBGROUP NO.:      8   DEVICE TYPE : HG3550IP         TRACENO    :    0 |
| SEARCH MODE : DESCENDING                                     ACD THRESHOLD : * |
| NUMBER OF ASSOCIATED ROUTES : 1                             PRIORITY    :    2 |
| TDDRFLAG    : OFF   TDDRTHRESHOLD:                     SOURCEGROUPIDX :    5 |
| GDTRRULE    : 0     ACDPMGRP : 0                         |
| THE FOLLOWING TRUNKS (LTG-LTU-SLOT-CCT) HAVE BEEN ALLOCATED: |
+-----+
| 1-99- 5-0      1 | 1-99- 5-0      2 | 1-99- 5-0      3 |
| 1-99- 5-0      4 | 1-99- 5-0      5 | 1-99- 5-0      6 |
| 1-99- 5-0      7 | 1-99- 5-0      8 | 1-99- 5-0      9 |
| 1-99- 5-0     10 | 1-99- 5-0     11 | 1-99- 5-0     12 |
| 1-99- 5-0     13 | 1-99- 5-0     14 | 1-99- 5-0     15 |
| 1-99- 5-0     16 | 1-99- 5-0     17 | 1-99- 5-0     18 |
| 1-99- 5-0     19 | 1-99- 5-0     20 | 1-99- 5-0     21 |
| 1-99- 5-0     22 | 1-99- 5-0     23 | 1-99- 5-0     24 |
| 1-99- 5-0     25 | 1-99- 5-0     26 | 1-99- 5-0     27 |
| 1-99- 5-0     28 | 1-99- 5-0     29 | 1-99- 5-0     30 |
+-----+

```

### 3.4 Digital Trunks (TDCSU)

The AMO TDCSU (Digital Trunk Circuits in the Switching Unit) is used for the digital trunk configuration, in meaning of an ISDN Basic Rate Interface (BRI), ISDN Primary Rate Interface (PRI) and the external gateways for IP Trunking. The TDCSU configurations of the referenced SIP gateway used as follows:

```

DISPLAY-TDCSU:PEN1=1-99-005-0,FORMAT=L;
H500: AMO TDCSU STARTED
+-----+
| DEV      = HG3550IP   PEN      = 1-99-005-0   TGRP      = 6 |
+-----+
| PROTVAR  = ECMAV2     INS       = Y           SRCHMODE  = DSC |
| COTNO    = 3          COPNO    = 3           DPLN       = 0 |
| ITR      = 0          COS      = 4           LCOSV      = 1 |
| LCOSD    = 1          CCT      = XCAPI        DESTNO    = 0 |
| SEGMENT  = 8          DEDSCC   =              DEDSVC    = NONE |
| FACILITY =            DITIDX   =              SRTIDX    = |
| TRTBL    = GDTR       SIDANI   = N           ATNTYP    = TIE |
| CBMATTR  = NONE      NWMUXTIM = 10          TCHARG     = N |
| SUPPRESS = 0          DGTPR    =              CHIMAP    = N |
| ISDNIP   = 00         ISDNNP   = 0           PNPAC      = |
| PNPL2P   =           PNPL1P   =              NNO       = |
| TRACOUNT = 30         SATCOUNT = MANY       CARRIER   = 1 |
| ALARMNO  = 0          FIDX     = 1           FWDX       = 5 |
| ZONE     = EMPTY     COTX     = 3           TPROFNO    = |
| DOMTYPE  =           DOMAINNO =              CCHDL     = |
| INIGHT   =           UUSCCY   = 8           FNIDX      = 0 |
| UUSCCX   = 16         CLASSMRK = EC & G711 & G729AOPT SRCGRP   = ( 5 ) |
| TCCID    =           TCCID    =              SECLEVEL  = TRADITIO |
| HMUSIC   = 0          CALLTIM  = 60          WARNTIM   = 60 |
+-----+
| BCNEG    = N          BCGR     = 1           LWPARG     = |
| LWPP     = 0          LWLT     = 0           LWPS      = 0 |
| LWR1     = 0          LWR2     = 1           BCHAN      = 1 && 30 |
| DMCALLWD = Y          SVCDOM   = |
+-----+

```

### 3.5 Trunk Routing (RICHT)

The AMO **RICHT** dialog is used for allocating access codes which are determined by the digit analysis group, the call progress state and the involved communication services (Voice, FAX and DATA). For this example the trunk routing is used as shown below.

```

DISPLAY-RICHT:MODE=LRTE,LRTE=15;
H500:  AMO RICHT STARTED
+-----+
| LRTE = 15      NAME = XCAPI              (NEUTRAL)  LSVC = ALL |
| DNNO =1 -1   -124 PDNNO =                0          |
| ROUTOPT = NO   REROUT = YES   PLB = YES   FWDBL = NO  |
| DTMFCNV = FIX   DTMFDSP = DIGITS DTMFTEXT = DTMF     |
| DTMFPULS = PP300 BUGS = LIN  ROUTATT = NO   MAINGRP = 13 |
| EMCYRTT = NO    CONFONE = NO  RERINGRP = NO  RTENO = 13 |
| INFO =                                                 |
| NOPRCFWD = NO                                         |
| NITO = NO                                              |
| CLNAMEDL = NO                                         |
| FWDSWTC = NO                                         |
| LINFEMER = NO                                         |
| NOINTRTE = NO                                         |
+-----+
| TGRP = 6  LDAT  XCAPI              (NEUTRAL)  SUBGROUP = 8 |
+-----+
H23:  SVC FAX IS NOT USED FOR RICHT BRANCH CD.  VOICE IS USED
      FOR G3 FAX AND DATA FOR G4 FAX.
  
```

### 3.6 LCR Outdial Rule (LODR)

The AMO **LODR** is used to declare LCR Outdial rules (LODR). Those rules will be identified over the ODR numbers according to the LDAT configuration described in the chapter **LCR Routes (LDAT)** on [page 13](#). The LCR Outdial for this example is used as shown next.

```

DISPLAY-LODR:ODR=15;
H500:  AMO LODR STARTED
+-----+
| ODR   POSITION  CMD      PARAMETER |
+-----+
| 15    1       NPI      ISDN       |
|        2       TON      UNKNOWN    |
|        3       ECHO      1         |
|        4       ECHO      2         |
|        5       END       |
+-----+
| INFO: XCAPI |
+-----+
  
```

### 3.7 LCR Routes (LDAT)

The administration of LCR routes (LROUTE) are configured within the AMO LDAT including its LCR route elements (LRTEL). Those LCR route elements specify the accumulation of the trunk groups. The LCR routes are related to a trunk group, an LCR out dial rule and a LCR authorization.

Please note that only trunks can be used which therefore have been assigned within the AMO RICHT.

```
DISPLAY-LDAT:TYPE=LCR,LROUTE=15;
H500: AMO LDAT STARTED
```

-----										
LROUTE = 15		LDPLN		NAME = XCAPI				SERVICE = ALL		
TYPE = LCR		DNNO OF ROUTE = 1 -1 -124								
SERVICE INFO =										
-----										
LRTEL	LVAL	TGRP	ODR	LAUTH	SCHEDULE	CARRIER	ZONE	LATTR	LDSRT	COTIDX
-----										
1	1	6	15	1	*****	1	EMPTY	NONE		0
-----										

### 3.8 Digit Analysis (WABE)

The digit analysis (WABE), which assignments depends on the call progress states and the digit analysis group or dial plan, refers to the digit analysis results of a dialed digit sequence or dialed code.

```
DISPLAY-WABE:TYPE=GEN,CD=64;
H500: AMO WABE STARTED
```

-----										
DIGIT INTERPRETATION					VALID FOR ALL DIAL PLANS					
-----										
CODE		CALL PROGRESS STATE				NODE/DIGIT		RESERVED/CONVERT		
		1 1111 11112 22				ANALYSIS		DNI/ADD-INFO/UFIP		
		0 12345 67890 12345 67890 12				RESULT		*=OWN NODE		
-----										
64		. . **** *... .. *				TIE				
-----										

### 3.9 LCR Dialplan (LDPLN)

The AMO LCR dialing plan (LDPLN) specifies the LCR digit patterns (LDP). Each pattern is assigned to an LCR route pattern number (LROUTE) and an LCR authorization number (LAUTH) and group-wise related to the WABE digit analysis. This example use LDP **64-xxxx**. Adjust the dialplan to the local requirements (for example **9-W-1-XXX-XXX-XXXX** for a North American dialplan).

```
DISPLAY-LDPLN:TYPE=LDP,LDP="64"- "XXXX";
H500: AMO LDPLN STARTED
```

DIPLNUM:	0		
LDPNO :15		LDP : 64-XXXX	
		SPC : 22	
		FDSFIELD : 0	SDSFIELD : 0 PINDP : N
DPLN	LROUTE	LAUTH	
0	15	1	
1	15	1	
2	15	1	
3	15	1	
4	15	1	
5	15	1	
6	15	1	
7	15	1	
8	15	1	
9	15	1	
10	15	1	
11	15	1	
12	15	1	
13	15	1	
14	15	1	
15	15	1	

### 3.10 Class of Services

The class of services, in meaning of the AMOs COT (Class of Trunks), COP (Class of Parameters) and COSSU (Class of Services), has to be configured in accordance with the TDCSU, **Digital Trunks (TDCSU)** and **Trunk Routing (RICHT)** as shown in the same named chapters starting on [page 12](#). For this example those classes of services (COP, COT, COSSU) are used as described next.

### 3.11 Class of Services (COSSU)

The AMO COSSU (Classes of Services) specifies the authorizations and features assigned to stations (AMO-ACSU, AMO-SBCSU, AMO-SCSU and AMO-SSCSU) and trunks (AMO-TACSU, AMO-TDCSU and AMO-TSCSU). For this example the class of services are used as follows:

```
DISPLAY-COSSU:TYPE=COS,COS=4;
H500: AMO COSSU STARTED
```

COS	VOICE	FAX	DTE
4	>NATIVE SIP - XCAPI TA TNOTCR TTT	NOCO NOTIE	TA TNOTCR BASIC MULTRA

```
DISP-COSSU:LAUTH,1;
H500: AMO COSSU STARTED
```

LAUTH	1
LCOSV	1 2 5 7
LCOSD	1

```
DISPLAY-COSSU:TYPE=LCOSV,LCOSV=1;
H500: AMO COSSU STARTED
```

LCOS	1	2	3	4	5	6	COPIN
V	1234567890123456789012345678901234567890123456789012345678901234						
	>SERVICE INFORMATION						
1	XX						0
	>LCR ATTENDANT FOR VOICE						

```
DISPLAY-COSSU:TYPE=LCOSD,LCOSD=1;
H500: AMO COSSU STARTED
```

LCOS	1	2	3	4	5	6	COPIN
D	1234567890123456789012345678901234567890123456789012345678901234						
	>SERVICE INFORMATION						
1	XX						.
	>LCR ATTENDANT FOR DATA						

### 3.12 Class of Trunks (COT)

The class of trunk for call processing (**COT**) is used to specify the switching technology parameters. Each trunk has an assigned COT which is here used as shown below.

```

DISPLAY-COT:COTNO=3;
H500: AMO COT STARTED
COT: 3 INFO: XC-API
DEVICE: INDEP SOURCE: DB
PARAMETER:
RECALL IF USER HANGS UP IN CONSULTATION CALL RCL
TRUNK CALL TRANSFER XFER
TRUNK SIGNALING ANSWER ANS
CALL EXTEND FOR BUSY, RING OR CALL STATE CEBC
NETWORKWIDE AUTOMATIC CALLBACK ON BUSY CBBN
NETWORKWIDE AUTOMATIC CALLBACK ON FREE CBFN
NETWORKWIDE CALL FORWARDING PERMITTED FWDN
NETWORKWIDE FORWARDING NO-ANSWER FNAN
DON'T RELEASE CALL TO BUSY HUNT GROUP BSHT
END-OF-DIAL FOR BLOCK IS SET BLOC
SEND NO NODE NUMBER TO PARTNER LWNC
CONNECTION TO ROUTE OPTIMIZATION NODE ROPT
INCOMING CIRCUIT FROM SYSTEM WITHOUT LCR NLCR
TSC-SIGNALING FOR NETWORKWIDE FEATURES (MANDATORY) TSCS
USE DEFAULT NODE NUMBER OF LINE DFNN
INCOMING CIRCUIT FROM SYSTEM WITHOUT LCR (DATA) NLRD
PIN NETWORKWIDE POSSIBLE PINR
SEND NO BILLINGELEMENTS SNBE
SUPPRESS SIGNALING OF CDR-E SPECIFIC MESSAGES NCDR
LAST REDIRECTING NUMBER IS SEND TO PHONEMAIL LRPM
NO FLAG TRACE NOFT
NPI ISDN, OUTGOING CALL NPIS
INTERNAL DIAL TONE DTNI

```

### 3.13 Class of Parameters (COP)

The class of parameters (COP) is used to specify the line parameters for signaling control of the device handler. Each trunk has also an assigned COP. For this example the CO TRUNK ACCESS and TOLL ACCESS parameter for the COP number **99** is used with the **TRUNK ACCESS (TA)** option.

```

DISPLAY-COP:COPNO=3;
H500: AMO COP STARTED

COP: 3 INFO: NATIVE SIP - XC-API
DEVICE: INDEP SOURCE: DB
PARAMETER:

CO TRUNK ACCESS:
TRUNK ACCESS TA

TOLL ACCESS:
TRUNK ACCESS TA

```



### 3.14 Reference Clock Table (REFTA)

Please ensure that the Reference Clock Table (AMO REFTA) is handled in the right way. It is necessary that all systems are synchronized. Wrong settings/priorities may lead to facsimile abruption.

```
DISP-REFTA: ;
H500: AMO REFTA STARTED
```

REFERENCE CLOCK CIRCUITS									
PEN	MODULE	DEVICE	PRI	ERROR	BLOCK	SUPP.	READY BUT ASYN.	SRCGRP	
1-17-	7- 0	DIU-N2	S2COD	90	64000	N		N	1

### 3.15 Gateway Properties

Please note: The **SIP Protocol Variant** and **Signaling Protocol for IP Networking** must be used with **SIP** and **Native SIP**. SIP-Q is **not** supported!

**vHG 3500**

Configuration Maintenance Help Logoff

**Configuration**  
Basic Settings  
Security  
Network & Routing  
Voice Gateway

**Gateway Properties**

General

Board Name: SoftGate-SIP

Physical Node Number (4K): 1-1-100

Gateway Location: SG99

Contact Address:

System Country Code: 49 (Germany)

Global Gateway of Type G.711: A-law

Supported IP Version: IPV4 only

Gateway IP Address: 10.8.242.112

Gateway Subnet Mask: 255.255.255.0

Public WAN IP Address:

Additional Features

Conference Improvement: ☒

Support Dispatch Application: ☐ only for Native SIP Trunking GW

Allow SIP Register for Trunking: ☐ only for Native SIP Trunking with profile

Enable SMP: ☒

Maximum number of DMC connections: 0

Use Early Media for Disconnect to SIP: ☐ only for Native SIP Trunking GW

Enable SMP for SIPQ proxy: ☐

Signaling Protocol for IP Networking: SIP

SIP Protocol Variant for IP Networking: Native SIP

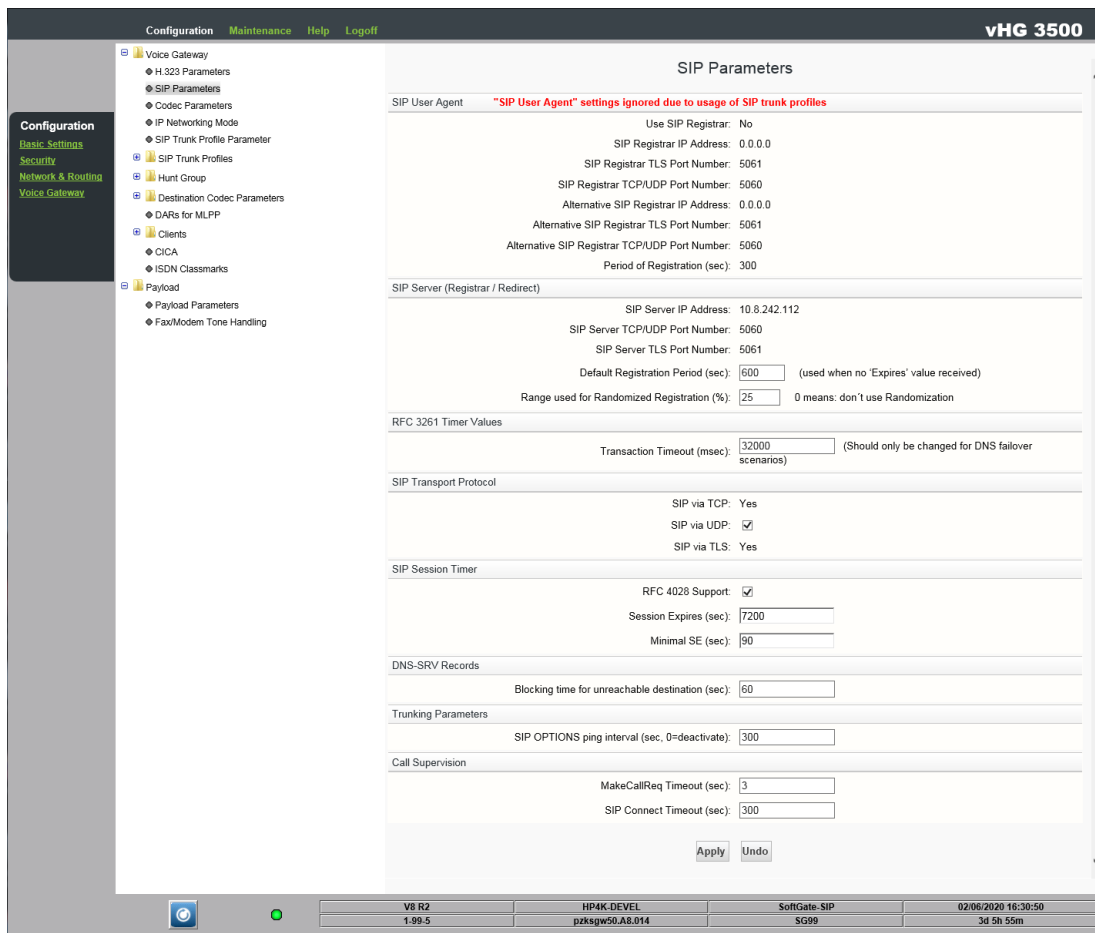
DisplayName Character Code Set:

Apply Undo

V8 R2 HP4K-DEVEL SoftGate-SIP 02/06/2020 16:37:20  
1.99.5 pzksqw50.A8.014 SG99 3d 6h 1m

### 3.16 SIP Parameters

The SIP Parameters are used as shown next.



**Configuration Maintenance Help Logoff** **vHG 3500**

**Configuration**  
 Basic Settings  
 Security  
 Network & Routing  
 Voice Gateway

**Voice Gateway**  
 H.323 Parameters  
 SIP Parameters  
 Codec Parameters  
 IP Networking Mode  
 SIP Trunk Profile Parameter  
 SIP Trunk Profiles  
 Hunt Group  
 Destination Codec Parameters  
 DARs for MLPP  
 Clients  
 CICA  
 ISDN Classmarks  
 Payload  
 Payload Parameters  
 Fax/Modem Tone Handling

### SIP Parameters

**SIP User Agent** "SIP User Agent" settings ignored due to usage of SIP trunk profiles

Use SIP Registrar: No  
 SIP Registrar IP Address: 0.0.0.0  
 SIP Registrar TLS Port Number: 5061  
 SIP Registrar TCP/UDP Port Number: 5060  
 Alternative SIP Registrar IP Address: 0.0.0.0  
 Alternative SIP Registrar TLS Port Number: 5061  
 Alternative SIP Registrar TCP/UDP Port Number: 5060  
 Period of Registration (sec): 300

**SIP Server (Registrar / Redirect)**

SIP Server IP Address: 10.8.242.112  
 SIP Server TCP/UDP Port Number: 5060  
 SIP Server TLS Port Number: 5061  
 Default Registration Period (sec): 600 (used when no 'Expires' value received)  
 Range used for Randomized Registration (%): 25 0 means: don't use Randomization

**RFC 3261 Timer Values**

Transaction Timeout (msec): 32000 (Should only be changed for DNS failover scenarios)

**SIP Transport Protocol**

SIP via TCP: Yes  
 SIP via UDP: ☒  
 SIP via TLS: Yes

**SIP Session Timer**

RFC 4028 Support: ☒  
 Session Expires (sec): 7200  
 Minimal SE (sec): 90

**DNS-SRV Records**

Blocking time for unreachable destination (sec): 60

**Trunking Parameters**

SIP OPTIONS ping interval (sec, 0=deactivate): 300

**Call Supervision**

MakeCallReq Timeout (sec): 3  
 SIP Connect Timeout (sec): 300

**Apply Undo**

V8 R2	HP4K-DEVEL	SoftGate-SIP	02/06/2020 16:30:50
1.99.5	pksgw50.A8.014	SG99	3d 5h 55m

## 3.17 SIP Trunk Profiles

From Unify OpenScape 4000 V8 & V10 the **NatTrkEnterprise** profile can be used for appropriate interworking and call transfer support via SIP refer method. Nevertheless, an own SIP trunk profile should be created for using/enabling required services/settings such as call transfer and numbering behavior.

The SIP trunk for this example uses the proxy address 10.8.242.59. This is the IP address used by the XCAPI controllers' related Ethernet interface, as shown in the chapter **Network Interface** starting on [page 6](#). The **SIP Transport Protocol** is used by default (also from XCAPI) with **UDP** and no **Registrar** is set at all.

The SIP profile has to be enabled. If it's not in operation, just try disabling and enabling the profile once again. If required, the HG board must be rebooted as well.

The screenshot displays the configuration interface for a vHG 3500 device. The left sidebar shows the navigation menu with 'Configuration' selected, and 'SIP Trunk Profiles' highlighted under the 'Voice Gateway' section. The main area is titled 'SIP Trunk Profile' and shows the configuration for the 'XCAPI' profile.

**SIP Trunk Profile Configuration:**

- Profile Name:** XCAPI
- Activate Trunk Profile:** ☒
- Account/Authentication Required:** ☐
- Remote Domain Name:**
- IP Transport Protocol:** UDP (used for O/G call establishment)
- PAI for anonymous:**
- Security:** Released Security Level: No Security
- Registrar:**
  - Use Registrar:** ☐
  - IP Address / Host name:**
  - Specify Port:** ☐
  - Reregistration Interval (sec):** 0
- Proxy:**
  - IP Address / Host name:** 10.8.242.59
  - Specify Port:** ☐
- Outbound Proxy:**
  - Use Outbound Proxy:** ☐
  - IP Address / Host name:**
  - Specify Port:** ☐
- Inbound Proxy:**
  - Use Inbound Proxy:** ☐
  - IP Address / Host name:**
  - Specify Port:** ☐
- Miscellaneous (read only):**
  - Emergency Call Behavior:** no modification, same as normal call
  - Active Hold Mode:** inactive
  - Ignore 100 Rel:** No
  - Use rport from Via header:** No
  - Call Diversion:** No Screening with diversion header
  - Direct Payload:** Yes
  - UPDATE Allowed:** Yes
  - REFER Allowed:** Yes
  - Rerouting after receiving 404 message from trunk partner:** No
  - Direct Blind Transfer (RFC 3892 Referred-By) support:** No
  - Registration for Multiple Phone Numbers (RFC 6140):** No
  - Silence Suppression Support:** Yes
  - TCP Connection Reuse:** enables ConnReuse
  - Disable SDP in 1xx unreliable responses:** No
  - Send response to OPTIONS request:** with full SDP content

At the bottom of the configuration area are buttons for **Apply**, **Undo**, and **Delete**.

The status bar at the very bottom shows system information: V8 R2, 1.99.5, HP4K-DEVEL, prksqw50.A8.014, SoftGate-SIP, SG99, and the date/time 02/06/2020 16:35:07.

This example uses an own profile which can be added via **Maintenance, Appl Diagnostics** and **Developer-Settings** requires the appropriate user administration rights for the WBM.

**vHG 3500**

Configuration Maintenance Help Logoff

**Maintenance**  
 Config & Update  
 Job List  
 Traces & Events  
 Appl Diagnostics

Application Diagnostics  
 COCP: COCP functions  
 OAM Downloads: OAM download functions  
 OAM Stimuli: OAM stimulus functions  
 OAM: OAM functions  
 Trace: Trace functions  
 RAM Trace: RAM Trace functions  
 Dep: Dep functions  
 SystemIpAddress: SystemIpAddress functions  
 SSA: SSA functions  
 SIP: SIP functions  
 H323: H323 functions  
 LocServ: LocServHttpMenu functions  
 DVMGR Diagnose: DVMGR Diagnose functions  
 DVMGR Test: DVMGR Test functions  
 DMC: DMC functions  
 SIP: SIP functions  
 H323: H323 functions  
 LocServ: LocServHttpMenu functions  
 DVMGR Diagnose: DVMGR Diagnose functions  
 DVMGR Test: DVMGR Test functions  
 DMC: DMC functions  
 Developer-Settings  
 SIP Provider Profiles

**Extended SIP Provider Data**

Provider Name: XCAPI

**CLIP / CLIR**

CLIP outgoing in From header - display part: call number  
 CLIP outgoing in From header - user part: call number  
 CLIP outgoing in P-Asserted-Id header - display part: omit  
 CLIP outgoing in P-Asserted-Id header - user part: omit  
 CLIP outgoing in P-Preferred-Id header - display part: omit  
 CLIP outgoing in P-Preferred-Id header - user part: omit  
 CLIR outgoing in From header - display part: anonymous  
 CLIR outgoing in From header - user part: call number  
 CLIR outgoing From Header - domain/host part: IP Address  
 outgoing Privacy header: none

**Call number formatting**

Incoming call - Called party number: request line  
 Incoming call - Calling party number: From header user part

**Security**

Released Security Level: no security released  
 Media Remote Address Validation: AddPortVal

**Miscellaneous**

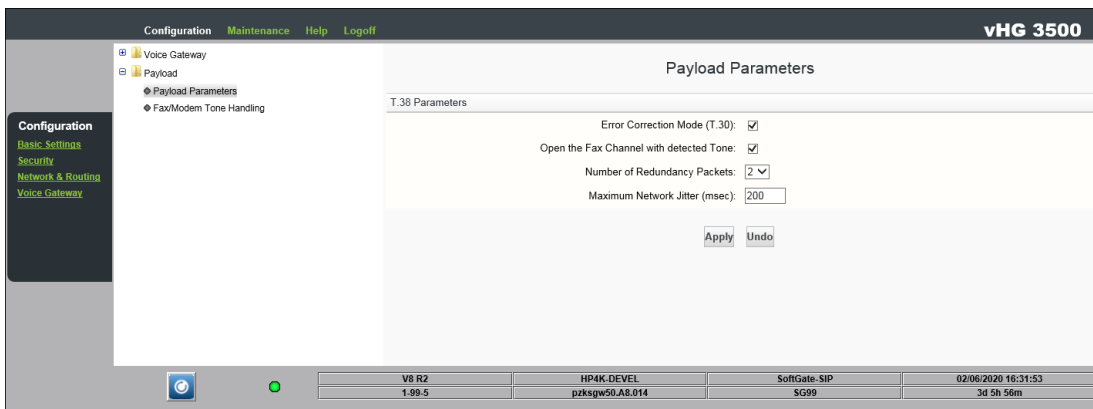
Emergency Call Behavior: No modification, same as normal call  
 Active Hold Mode: inactive  
 Ignore 100 Rel: ☐  
 ContactUriContains: the URI composed of calling number and home domain as the From header - default  
 ContactUriParams: w/ any Transport and OwnSecurityCapString  
 DirectPayload: ☒  
 Use rport from Via header: ☐  
 CallDiversion: No Screening with diversion header  
 UPDATE Allowed: ☒  
 REFER Allowed: ☒  
 Rerouting after receiving 404 message from trunk partner: ☐  
 Blind Transfer directly back to SIP provider with transparent Referred-By Header: ☐  
 SIP transport protocol configurable: ☒  
 Registration for Multiple Phone Numbers (RFC 6140): ☐  
 Silence Suppression Support: ☒  
 TCP Connection Reuse: enables ConnReuse  
 Show hint "PSR Authorisation required": ☐  
 Disable SDP in 1xx unreliable responses: ☐  
 Send response to OPTIONS request: with full SDP context  
 Enable P-Early-Media: ☒

Back OK & Next

V8 R2	HP4K.DEVEL	SoftGate: SIP	02/06/2020 16:40:43
1.99.5	pksgw50.A8.014	SG99	3d 6h 5m

### 3.18 Payload

The payload parameters are used as shown next. Ensure that **Error Correction Mode** is enabled for T.38 usage.



**Configuration** Maintenance Help Logoff **vHG 3500**

Voice Gateway  
Payload  
Payload Parameters  
Fax/Modem Tone Handling

**Configuration**  
Basic Settings  
Security  
Network & Routing  
Voice Gateway

**Payload Parameters**

T.38 Parameters

Error Correction Mode (T.38): ☒

Open the Fax Channel with detected Tone: ☒

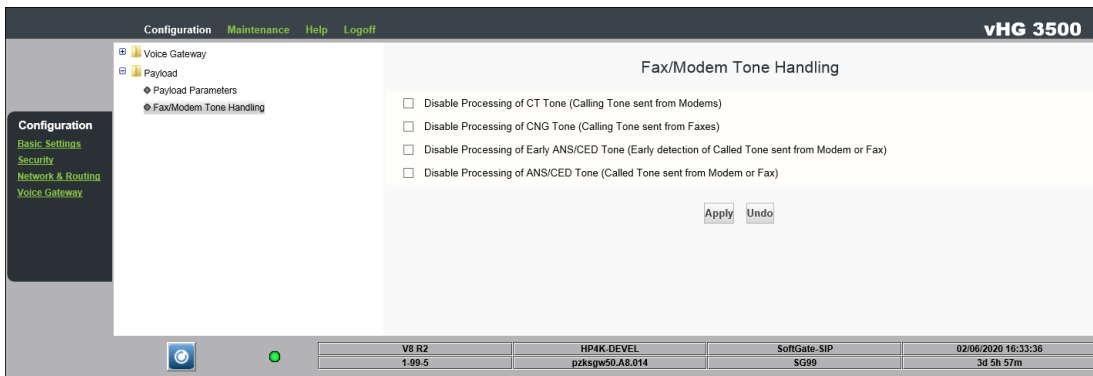
Number of Redundancy Packets:

Maximum Network Jitter (msec):

Apply Undo

V8 R2 HP4K.DEVEL SoftGate-SIP 02/06/2020 16:31:53  
1.99.5 pzksqw50.A8.014 SG99 3d 5h 56m

The **Fax/Modem Tone Handling** parameters are used with their defaults.



**Configuration** Maintenance Help Logoff **vHG 3500**

Voice Gateway  
Payload  
Payload Parameters  
Fax/Modem Tone Handling

**Configuration**  
Basic Settings  
Security  
Network & Routing  
Voice Gateway

**Fax/Modem Tone Handling**

☐ Disable Processing of CT Tone (Calling Tone sent from Modems)

☐ Disable Processing of CNG Tone (Calling Tone sent from Faxes)

☐ Disable Processing of Early ANS/CED Tone (Early detection of Called Tone sent from Modem or Fax)

☐ Disable Processing of ANS/CED Tone (Called Tone sent from Modem or Fax)

Apply Undo

V8 R2 HP4K.DEVEL SoftGate-SIP 02/06/2020 16:33:36  
1.99.5 pzksqw50.A8.014 SG99 3d 5h 57m

## 3.19 Codec Parameters

The **Codec Parameters** are used as shown next. Most of the parameters must be configured through the CGWB-AMO as shown in the chapter **Configuration of Global- and Feature-Board Data for HG3500 (CGWB)** starting on [page 10](#).

The defaults usually work fine, but it is recommended to use the G.711 codecs with a frame size of 20 msec. This is a common default for VoIP, also for XCAPI, and prevents possible transcoding issues of the gateway and other participating instances. You can review the facsimile related chapters (**Fax Support** starting on [page 23](#)) for additional configuration hints. The XCAPI controller codec defaults can be reviewed in the chapter **Codecs** on [page 28](#).

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.729	Priority 4	VAD: <input type="checkbox"/>	20 msec
G.729A	Priority 5	VAD: <input type="checkbox"/>	20 msec
G.729B	not used	VAD: <input checked="" type="checkbox"/>	20 msec
G.729AB	Priority 7	VAD: <input checked="" type="checkbox"/>	20 msec
G.722	Priority 8	VAD: <input type="checkbox"/>	20 msec
Opus	Priority 9	VAD: <input type="checkbox"/>	20 msec

**Opus-Parameter**

Use Inband Forward Error Correction (FEC): ☒

Use Constant Bitrate: ☐

Low Delay: ☐

Payload Type for Opus: 124

Max. Playback Sample Rate (Hz): 16000

Complexity: 1

**T.38 Fax**

T.38 Fax: ☒

Max. UDP Datagram Size for T.38 Fax (bytes): 375

Error Correction Used for T.38 Fax (UDF): 03UDFRedundancy

Time Range for Immediate Switch to T.38 Fax (s): 0 0 means: No Immediate Switching

**Misc.**

ClearMode (ClearChannelData): ☒ Frame Size: 20 msec

**RFC2833**

Transmission of Fax/Modem Tones according to RFC2833: ☒

Transmission of DTMF Tones according to RFC2833: ☒

Payload Type for ClearChannel: 96

Payload Type for RFC2833: 98

Payload Type for RFC2198: 99 (= 'Payload Type for RFC2833' + 1)

Redundant Transmission of RFC2833 Tones according to RFC2198: ☐

Payload Type for RFC4733 WideBand: 100 (= 'Payload Type for RFC2833' + 2)

Apply Undo

## 3.20 ISDN Classmarks

The **ISDN Classmarks** are used as follows.

External Call	Hold/Call Transfer	Call Forwarding	Callback
<input checked="" type="checkbox"/> (Public Net instead of Private Net)	<input checked="" type="checkbox"/> (Octets 3a bit7, 3b bit7 and 3c bit3)	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>

Apply Undo

## Appendix

The appendix gives several information and configuration hints as well as other considerations. If using the XC-API controller wizard with its Unify OpenScape 4000 template, most of the shown configurations are set by default. The shown configurations must be reviewed, checked and tested, especially with the participating stations and public trunk bindings of the Unify OpenScape 4000 environment. Some of the features, codecs and fax methods might not being supported in conjunction with specific Unify OpenScape 4000 trunk bindings.

### 4.1 Fax Support

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

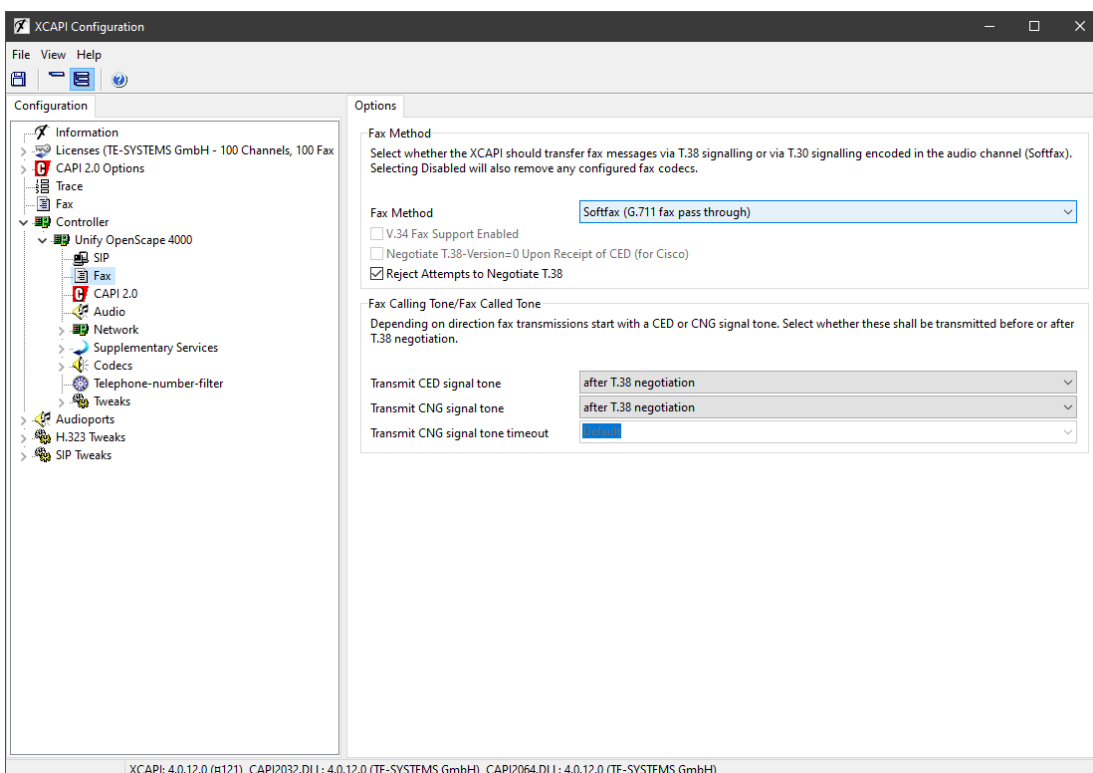
It must be ensured that the fax method, codecs, framing, bandwidth and DTMF settings are properly set, and conform to the ones of the XC-API SIP controller and other participating VoIP instances (SBC's, ITSP's etc.) connected to the Unify OpenScape 4000. We always recommend using the latest XC-API version and manufacturer releases.



Always use the recommended and supported Fax method between the Unify OpenScape Business and the SIP carrier for the XC-API controller. For most known VoIP scenarios G.711 Fax Passthrough should be the first choice!

## 4.1.1 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 **Softfax (G.711 fax pass through)** must be set as **Fax Method** within the labeled **Fax** tab.





#### 4.1.2 G.711 Fax Pass Through (Softfax) in Virtual Environments

Apart of the given suggestions for virtual environments in our **XCAPI TechNote (en) - VMware Virtual Machines** and **XCAPI TechNote (en) - Microsoft Hyper-V** documentation, facsimile transmission (which is real-time based) might be improved by adjusting the jitter buffer. A gateway jitter buffer operating with more tolerance might improve the stability for facsimile transmissions. Thus the timing difference between single RTP packets, caused by a virtual environment, could be absorbed by the XCAPI and the gateway. Set the **Jitter Buffer Type** to **Static**, the **Average Delay for Voice (msec)** to **200** and **Maximum Delay for Voice (msec)** to **300**.

```
DISP-CGWB:99,5,JB;
H500: AMO CGWB STARTED

-----
| CGW BOARD DATA                                     |
-----
| SIP          HG3550                                |
-----
| LTU = 99      SLOT = 5      SMODE = NORMAL          POOLNO: 0      |
-----

JB DATA - CONFIGURABLE VALUES:
-----
JBMODE      = 1
AVGDLYV     = 200 (40)
MAXDLYV     = 300 (120) MINDLYV = 20 (20)
PACKLOSS    = 4 (4)
AVGDLYD     = 60 (60) MAXDLYD = 200 (200)
```

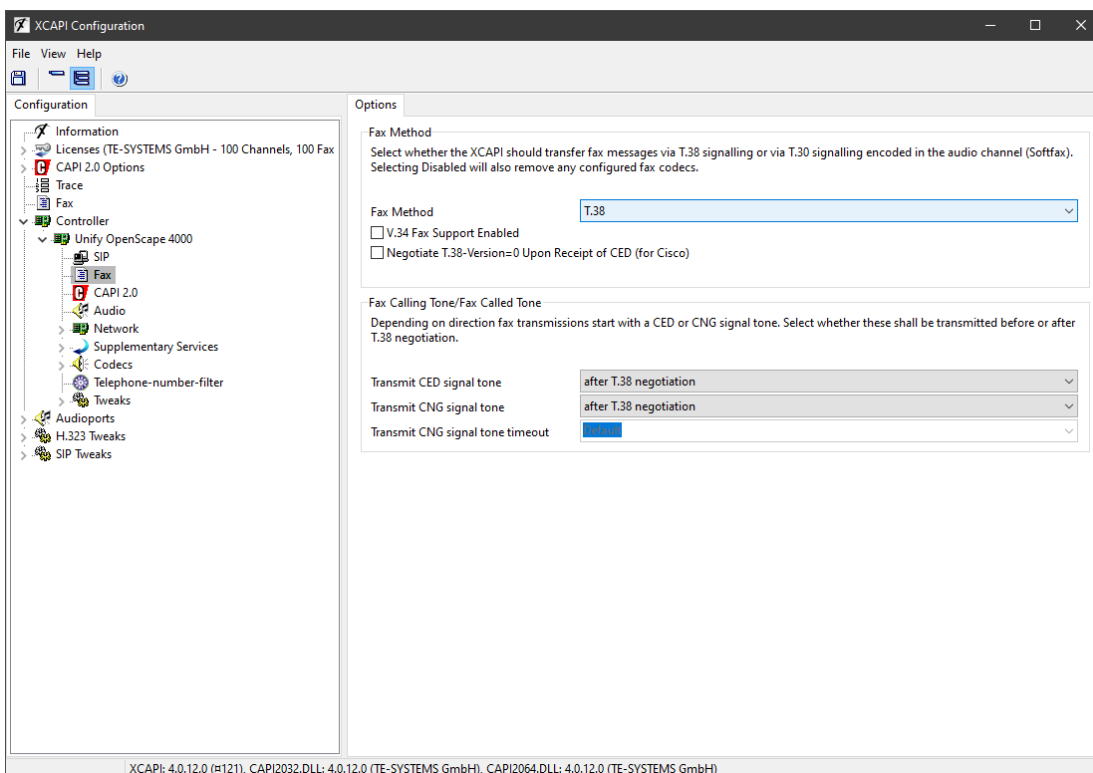


Please note, that this kind of adjustment should only be done for environments that are using the gateway for plain facsimile transmission and only done by experienced engineers. This chapter was originally intended for physical HG3500 gateways and the settings should only be adjusted after consultation with the Atos Unify support. All adjustments of the DSP subsequently require performance, stability and quality tests.

### 4.1.3 T.38

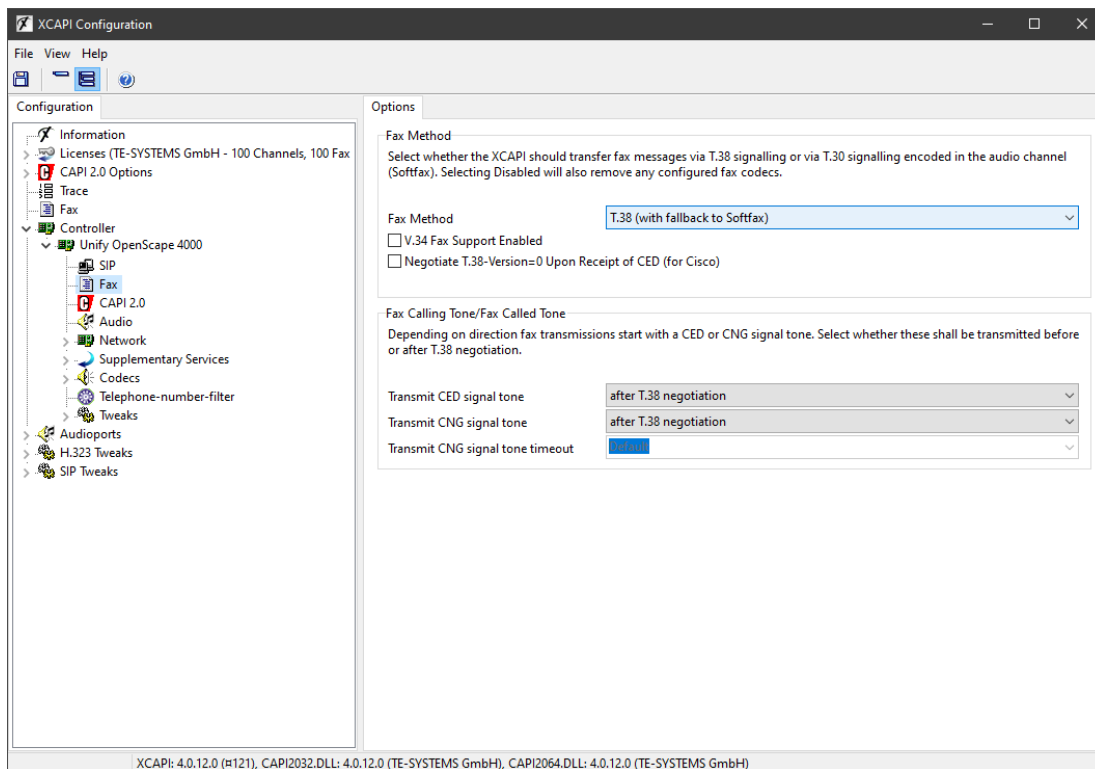
When using the T.38 Fax-over-IP protocol ensure that the **T38FAX** parameter in the AMO CGWB has to be enabled and those RFC2833 parameters are set as shown in the chapter **Configuration of Global- and Feature-Board Data for HG3500 (CGWB)** starting on [page 10](#). In the case of T.38 usage, this protocol must also be the recommended and supported fax method between the Unify OpenScape 4000 and its carrier trunk binding. It is recommended to avoid unnecessary transcoding (G.711 to T.38 or vice versa) and use conforming fax methods for all participating 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 referring chapter from [page 28](#). For XCAPI, T.38 has to be enabled as shown on the next screenshot. For the Unify OpenScape 4000, **T.38 (Fax)** must be enabled in the gateways **Codec Parameters**.

For XCAPI, please ensure that **T.38** is selected as **Fax Method** within the XCAPI controllers **Fax** dialog.



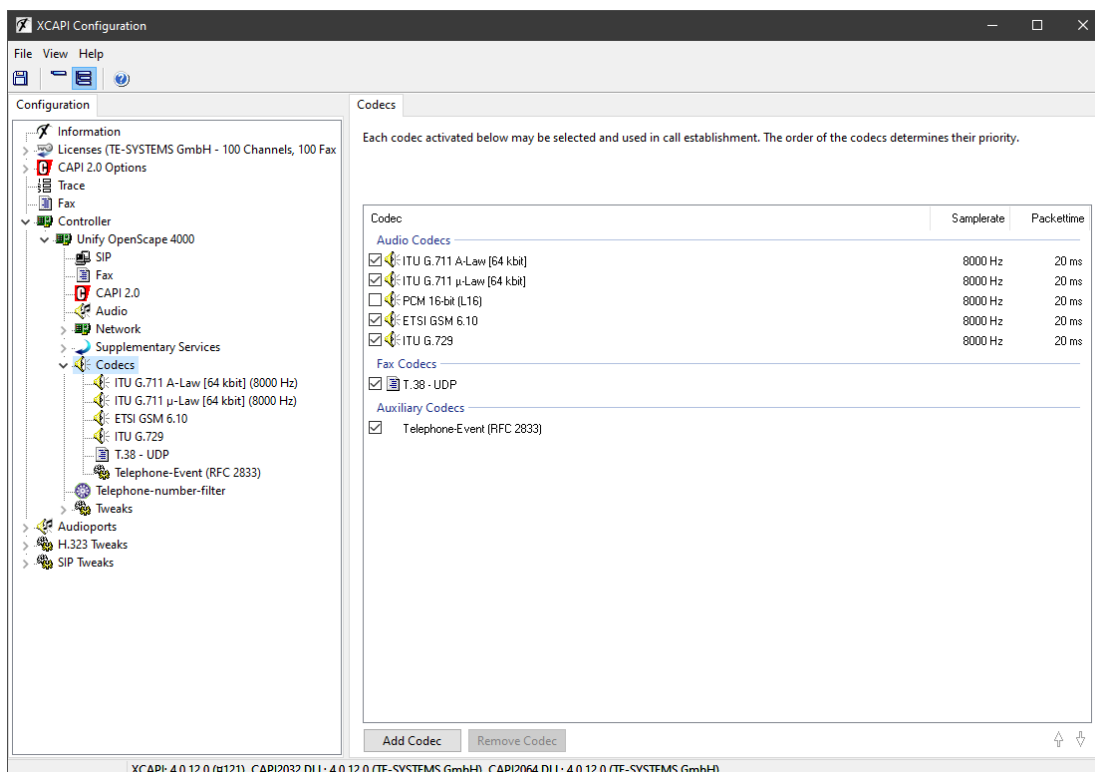
## 4.1.4 T.38 to G.711 Fax Pass Through Fallback

The fax fallback can be enabled as shown on the screenshot below. Even though this seems to be one of the most reliable options for fax transmissions, it can cause more issues in some of the VoIP infrastructures. So 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, within the Unify OpenScape 4000 environment. Depending on the VoIP environment with its participating instances, additional configurations and adjustments might be required. However, in practice and for most instances plain G.711 Fax Pass Through (Softfax) should be the first choice for faxing.



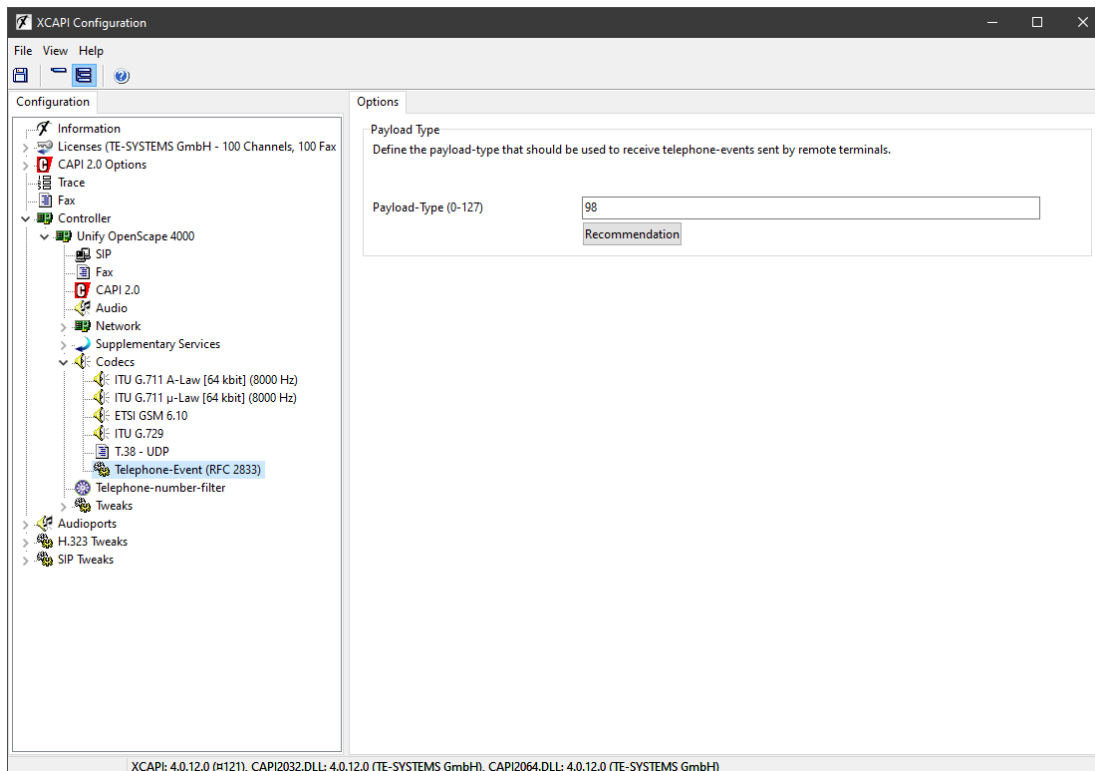
## 4.2 Codecs

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.



## 4.3 DTMF via RFC2833

Please ensure that the **RFC2833** parameters **RFCMOIP** and **RFCDTMF** are set. Further, the parameter **Redundant Transmission of RFC2833 Tones according to RFC2198** (CGWB-Parameter **REDRFC2833**) must be disabled for appropriate interworking. For interworking, the value for the **Payload Type for RFC2833** (shown in the chapter **Codec Parameters** starting on [page 22](#)) must match the XCAPI controller settings. Please review the appropriate configurations.



## 4.4 Message Waiting Indication

Signaling message waiting indications must be specified through the **PM** mode parameter of the AMO RICHT.

```
H500: AMO RICHT STARTED
+-----+-----+-----+-----+
| IDX |      SAN      |      NAME      |  TYPE  |
+-----+-----+-----+-----+
| 1|492119598559 | MWI            | OTHER  |
+-----+-----+-----+-----+
H23: SVC FAX IS NOT USED FOR RICHT BRANCH CD. VOICE IS USED
FOR G3 FAX AND DATA FOR G4 FAX.
```

The **PMIDX** (Phone Mail Index) must be enabled for the according subscribers.

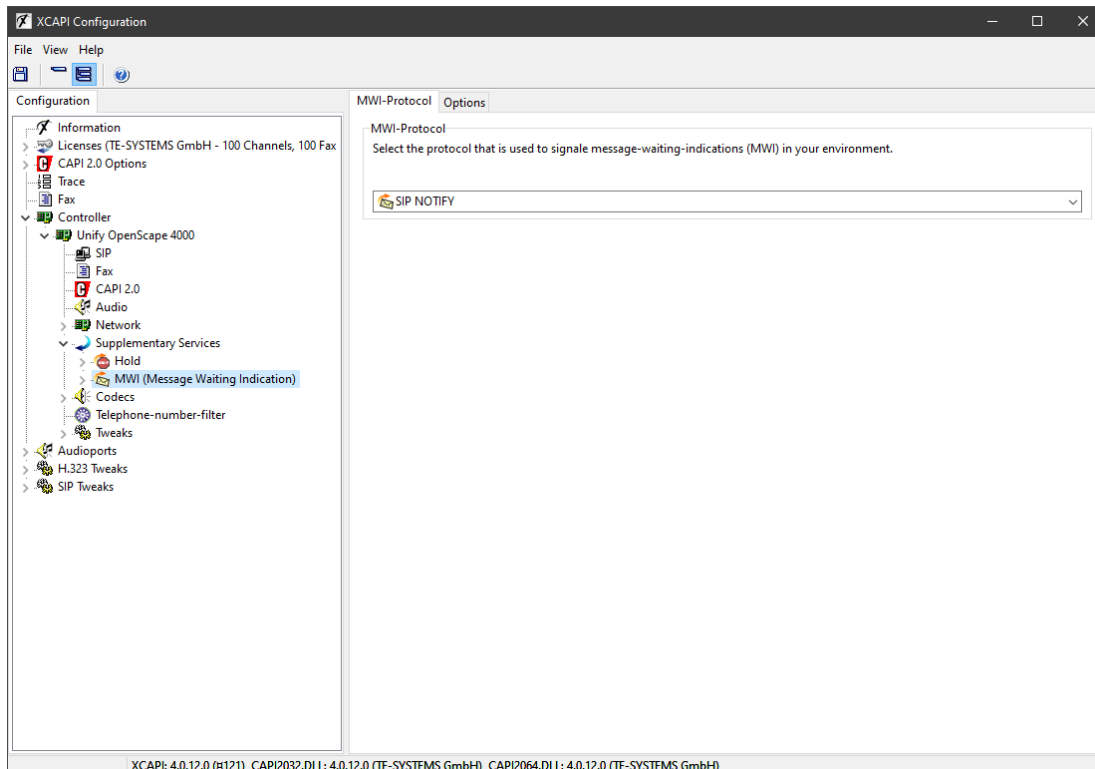
```
DISP-SBCSU:4001;
H500: AMO SBCSU STARTED

----- USER DATA -----
STNO    =4001      OPT    =OPTI    COS1    =1        DPLN    =0
MAIN0   =4001      CONN    =IP2     COS2    =1        ITR     =0
PEN     =1-17- 2- 41  LCOSV1   =1        COSX    =0
INS     =Y         ASYNCT  =500     LCOSV2   =1
SSTNO   =N         PERMACT  =        LCOSD1   =1        CBKBMAX  =5
TRACE   =N         EXTBUS   =        LCOSD2   =1        RCBKB    =N
ALARMNO =0         DFSVCANA=        SPDI     =0        RCBKNA   =N
HMUSIC  =0         FLASH    =        SPDC1    =        CBKNAMB  =N
PMIDX   =1         DIGNODIS=N  DSSTNA   =N        COMGRP   =0
SECR    =N         CALLOG   =NONE    DSSTNB   =Y        TEXTSEL  =ENGLISH
STD     =110

REP      =0         OPTICOM  =N        OPTIUSB  :        VPI      =
IDCR     =N         OPTICA   =        OPTISOA  :0       VCI      =
APPM     =          OPTIDA   =        OPTISPA  :0       PATTERN  =
BLF      =N         OPTIUPOE:0    OPTIABA  :0

DCFWBUSY=N          HEADSET  =N        APMOBUSR=N        APICLASS=
DNIDSP   =N          HSKEY    =NORMAL  IPCODEC  =G711P    SECAPPL  =0
DTMFBLK  =N
DTMFCTRD=N          BASICSVC=
DVCFIG   =OPTIIP  *  TSI      =1        SPROT    =        SOPTIDX  =
                                DPROT    =        DOPTIDX  =
                                FPROT    =        FOPTIDX  =
```

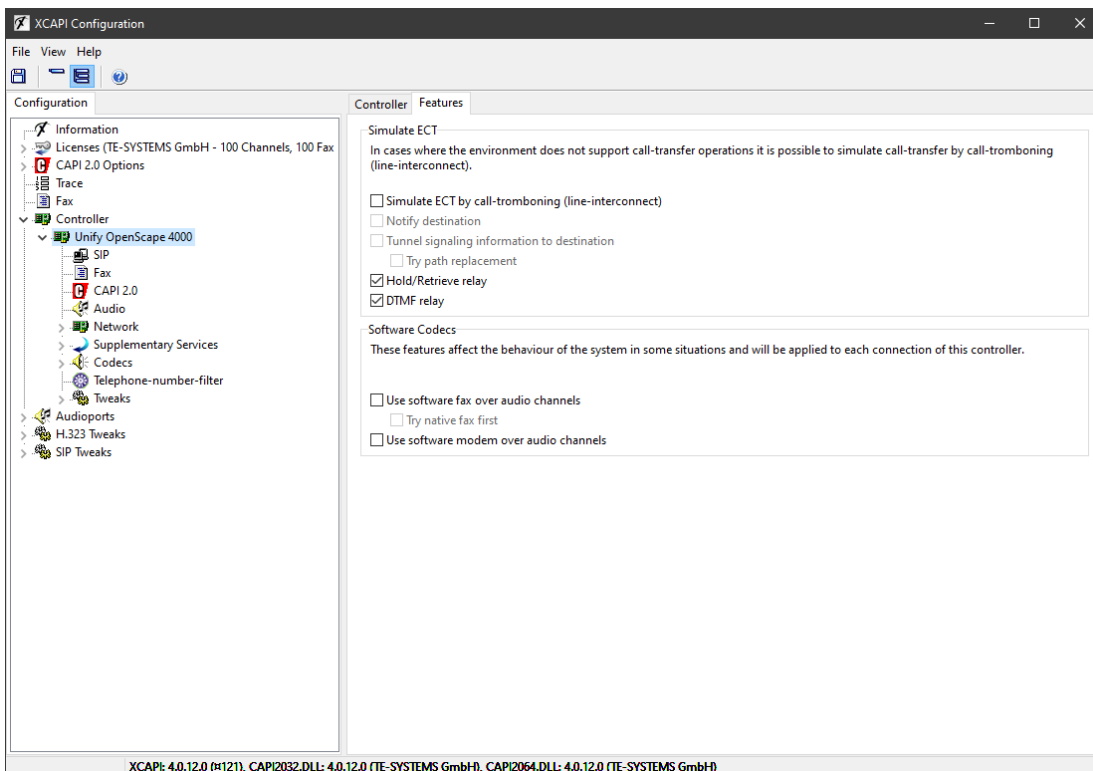
Ensure that the default MWI method for the XCAPI controller is set to **SIP Notify**.



## 4.5 Call Transfer

Ensure that **REFER Allowed** is enabled for the XC-API related **SIP Trunk Profile**. as shown in the same named chapter for the **Extended SIP Provider Data** configurations starting on [page 19](#).

Within the XC-API controller **Features** tab, the **Simulated ECT by call-tromboning (line-interconnect)** parameter must be disabled for supporting call transfer via SIP REFER.

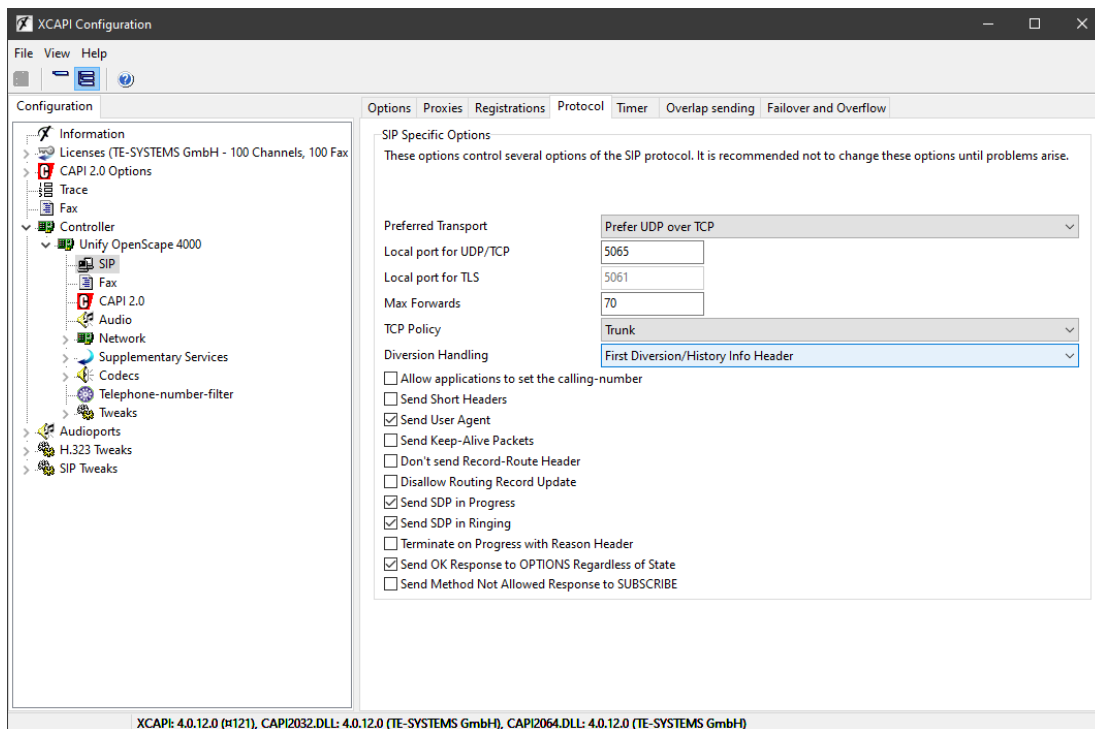




## 4.6 Diversion Handling

For some call scenarios, the CAPI 2.0 application might be aware about the first or last provided redirection number of the PBX. If needed, this behavior can be changed via the **Diversion Handling** option in XCAPi's SIP controller protocol tab as shown on the screenshot below. The Unify OpenScape 4000 gateway delivers this information via the SIP **Diversion** header towards XCAPi.

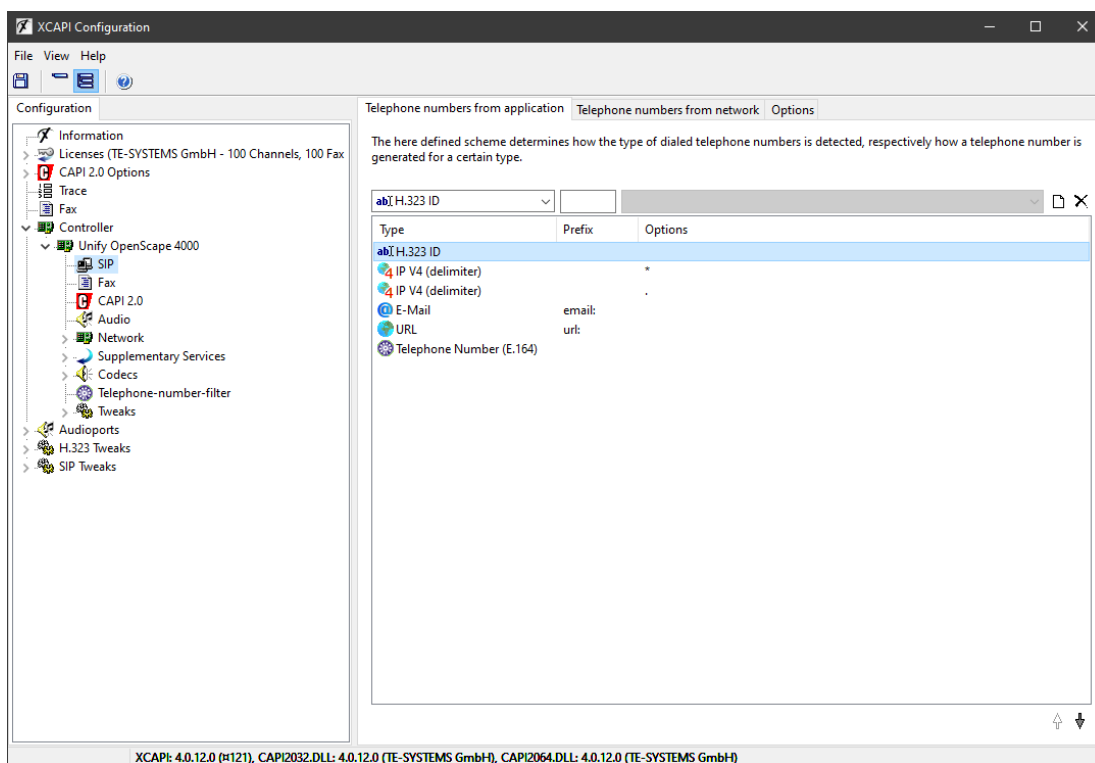
For this, the **Call Diversion** header must be enabled for the associated SIP trunk of XCAPi. This header has to be specified in the **Miscellaneous** section of the **Extended SIP Provider Data** as shown in the screenshot of the **SIP Trunk Profiles** chapter from [page 19](#).



## 4.7 Numbering

Please note, with very few exceptions, XCAPI can't do any numbering manipulations. The numbering or digits from 0 to 9, of the VoIP binding or the CAPI applications will be passed through by XCAPI. 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 OpenScape 4000 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 about dialed digits and some numbering attributes, other dial strings formats (for e.g. +49536381950) are not passed through XCAPI by default. Apart of that, there are also some CAPI 2.0 applications which are not even able to generate such dial strings.

If it's required that XCAPI pass dial strings like +49536381950, the SIP controller has to be adjusted. Go to the **Telephone-number-filter** dialog of the XCAPI controller. 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: In this configuration, XCAPI would even pass non-valid dial strings which may lead to numbering malfunction.

## Unify Ready Technology Connectivity Certification

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



## Exclusion of Liability

### Copyright © 2021 TE-SYSTEMS GmbH

All rights reserved

This document, in part or in its entirety, may not be reproduced in any form without the prior consent of TE-SYSTEMS GmbH.

The information contained in this document was correct at the time of writing. TE-SYSTEMS GmbH reserves the right to make any alterations without prior notice.

The utmost care was applied during the compilation of texts and images, as well as during the creation of the software. Nevertheless, no responsibility can be taken for the content being accurate, up to date or complete, nor for the efficient or error-free operation of the software for a particular purpose. Therefore, TE-SYSTEMS GmbH cannot be held liable for any damages resulting directly or indirectly from the use of this document.

### 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](mailto:eay@cryptsoft.com)).

This product includes software written by Tim Hudson ([tjh@cryptsoft.com](mailto: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

All files included in this sample are copyrighted by TE-SYSTEMS GmbH.

All samples and the SDK may only be used in combination with the XC-API-product.

The SDK contains code from libtiff with the following copyright-notice:

Copyright (c) 1988-1997 Sam Leffler

Copyright (c) 1991-1997 Silicon Graphics, Inc.

Permission to use, copy, modify, distribute, and sell this software and its documentation for any purpose is hereby granted without fee, provided that (i) the above copyright notices and this permission notice appear in all copies of the software and related documentation, and (ii) the names of Sam Leffler and Silicon Graphics may not be used in any advertising or publicity relating to the software without the specific, prior written permission of Sam Leffler and Silicon Graphics.

THE SOFTWARE IS PROVIDED „AS-IS“ AND WITHOUT WARRANTY OF ANY KIND, EXPRESS, IMPLIED OR OTHERWISE, INCLUDING WITHOUT LIMITATION, ANY WARRANTY OF MERCHANTABILITY OR FITNESS FOR A PARTICULAR PURPOSE.

IN NO EVENT SHALL SAM LEFFLER OR SILICON GRAPHICS BE LIABLE FOR ANY SPECIAL, INCIDENTAL, INDIRECT OR CONSEQUENTIAL DAMAGES OF ANY KIND, OR ANY DAMAGES WHATSOEVER RESULTING FROM LOSS OF USE, DATA OR PROFITS, WHETHER OR NOT ADVISED OF THE POSSIBILITY OF DAMAGE, AND ON ANY THEORY OF LIABILITY, ARISING OUT OF OR IN CONNECTION WITH THE USE OR PERFORMANCE OF THIS SOFTWARE.