Test report & configuration guide

Patton Gateway / eSBC

Deutsche Telekom – CompanyFlex SIP-Trunk

(Technical Specification 1TR119)

Contents

Introduction

Patton SmartNode products integrate IP and TDM communications for Enterprise and Carrier access networks, offering VoIP gateways combined with IP access routing, WAN transmission, and transcoding functionality. SmartNode scales from 1 to 32,768 VoIP or fax calls with various telephony interfaces including analog FXS/FXO and digital ISDN BRI, PRI, DS3 and STM-1.

This document includes a general overview of the requirements and provides the necessary configuration to interconnect Patton SmartNode VoIP Gateway / eSBC devices with the NGN Platform of Deutsche Telekom through the SIP interface CompanyFlex SIP-Trunk (Technical Specification 1TR119).

CompanyFlex SIP-Trunk is a SIP-Trunking service of Deutsche Telekom AG, which is available to Business customers. It offers the possibility of distributing phone number ranges over several locations, splitting locations, combining several DDI-ranges or single numbers on a central access line, redundant access, as well as parallel call processing over several locations.

Product information related to **CompanyFlex SIP-Trunk**:

https://geschaeftskunden.telekom.de/internet-dsl/tarife/festnetz-internet-dsl/companyflex

Patton SmartNode overview

In scope of these tests, Patton SmartNode VoIP Gateways & eSBC's have been tested in the 3rd-Party Lab of Deutsche Telekom in Bonn.

Following Patton equipment has been tested:

- Analog Gateways interconnecting analog devices (phone, fax) to SIP-Trunk

Analog	SN200	SN4140	SN5540	SN5550	SN4740
Product Photo					
Embedded Software	Trinity™	Trinity™	Trinity™	Trinity™	Trinity™
Patton Cloud Connect	Yes	Yes	Yes	Yes	Yes
Telephony Interfaces	FXS (or 1FXS+1FXO)	FXS & FXO	FXS & FXO	FXS/FXO & BRI S0/T0	FXS (FXO soon)
Number of Telephony Ports	1,2 or 4	2,4 or 8	2,4 or 8	2,4,8 or 16	16,24 or 32
Call Capacity	Up to 4 Calls	Up to 8 Calls	Up to 8 Calls	Up to 16 Calls	Up to 32 Calls
VoIP Gateway (Converts TDM to SIP) Converts TDM to IP	✓	✓	✓	✓	✓
USB Support (WiFi, Cellular Modem etc.) IP Routing, QoS, VPN, etc.		✓	✓	✓	✓
IP Router IP Routing, QoS, VPN, etc.		Optional	✓	✓	Optional
Number of Ethernet Ports	1 10/100	1 or 2 10/100/1000	2 10/100/1000	2 10/100/1000	2 10/100/1000
WAN Access			Fiber SFP G.SHDSL-EFM/ATM ADSL/VDSL Gigabit Ethernet	Fiber SFP G.SHDSL-EFM/ATM ADSL/VDSL Gigabit Ethernet	
Industrial version available		✓			

- ISDN Gateways interconnecting Legacy ISDN PBX (BRI, PRI) to SIP-Trunk

    - PRI (S2M) models

T1 / E1 / PRI	SN4170	SN4970	SN5570	SN4980	SN4990
Product Photo					
Embedded Software	Trinity™	Trinity™	Trinity™	Trinity™	Trinity™
Patton Cloud Connect	Yes	Yes	Yes	Yes	Yes
Telephony Interfaces	T1/E1/PRI	T1/E1/PRI	T1/E1/PRI	T1/E1/PRI	T1/E1/PRI
Number of Telephony Ports	1 (2nd Fallback)	1 or 4	1 (2nd Fallback)	1 or 4	1 or 4
Call Capacity	15 to 30	15, 30, 60 or 120	15 to 30	15, 30, 60 or 120	15, 30, 60 or 120
VoIP Gateway Converts TDM to IP	✓	✓	✓	✓	✓
USB Support WiFi, Cellular Modem etc.	✓		✓		
IP Router IP Routing, QoS, VPN, etc.			✓	✓	✓
Number of Ethernet Ports	1 or 2 10/100/1000	1 10/100/1000	2 10/100/1000	2 10/100/1000	2 10/100/1000
Transcoding Interconnect multiple VoIP networks	Yes	Yes	Yes	Optional	Optional
WAN Access			Fiber SFP G.SHDSL-EFM/ATM ADSL/VDSL Gigabit Ethernet		Fiber SFP G.SHDSL-EFM/ATM ADSL/VDSL Gigabit Ethernet

    - BRI models

ISDN BRI	SN4130	SN4150
Product Photo		
Embedded Software	Trinity™	Trinity™
Telephony Interfaces	BRI	BRI with FXS/FXO
Number of Telephony Ports	2,4 or 8	4 or 8
Call Capacity	4, 8 or 16	4 or 8
VoIP Gateway Converts TDM to IP	✓	✓
IP Router IP Routing, QoS, VPN, etc.		
Number of Ethernet Ports	1 10/100	1 10/100/1000
WAN Access		

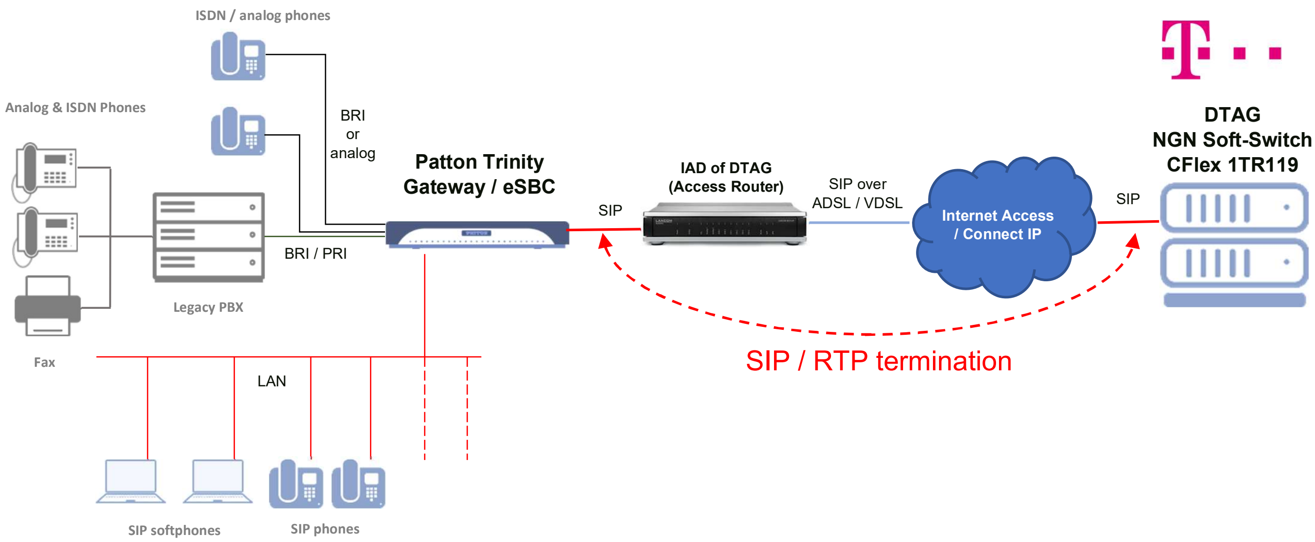
- eSBC (Enterprise Session Border Controller) for SIP-SIP connectivity

eSBC	vSN	SN5300	SN5480	SN5490	SN5500
Product Photo					
Embedded Software	Trinity™	Trinity™	SmartWare™ or Trinity™	SmartWare™ or Trinity™	Trinity™
Telephony Interfaces	N/A	N/A	N/A	N/A	N/A
SIP Sessions	1 to 1,000's	4 to 60	Up to 80	Up to 80	4 to 200
Transcoded Calls	N/A	N/A	Up to 64	Up to 64	Up to 16
IP router IP Routing, QoS, VPN, etc.	✓	✓	✓	✓	✓
Number of Ethernet Ports	HW dependent	4 10/100	2 10/100/1000	2 10/100/1000	2 10/100/1000
Transcoding	Roadmap		✓	✓	✓
WAN Access	N/A	G.SHDSL.bis (ATM or EFM)		Fiber G.SHDSL EFM X.21	G.SHDSL EFM&ATM, ADSL-VDSL

eSBC	SN5530	SN5540	SN5550	SN5570	SN5600
Product Photo					
Embedded Software	Trinity™	Trinity™	Trinity™	Trinity™	Trinity™
Telephony Interfaces	2, 4 or 8 BRI	2, 4 or 8 FXS/FXO	BRI with FXS/FXO	1 or 2 PRI	N/A
SIP Sessions	4 to 200	4 to 200	4 to 200	4 to 200	4 to 1,000
Transcoded Calls	Up to 8	Up to 4	Up to 4	Up to 15	N/A
IP router IP Routing, QoS, VPN, etc.	✓	✓	✓	✓	✓
Number of Ethernet Ports	2 10/100/1000	2 10/100/1000	2 10/100/1000	2 10/100/1000	2 10/100/1000
Transcoding	✓	✓	✓	✓	N/A
WAN Access	VDSL/ADSL G.SHDSL.bis	VDSL/ADSL G.SHDSL.bis		VDSL/ADSL G.SHDSL.bis	N/A

Remark: Limitation concerning the virtual SN (vSN) and SN5600 -> no transcoding and no SRTP available (models without DSPs). Support of SRTP planned in a future Firmware release.

Test environment



The tests performed in the 3rd-Party Lab involved Patton SmartNode Gateways / eSBC's with analog & ISDN devices connected through FXS & BRI interfaces. Additionally, specific tests with SIP hard- and softphones on LAN side of SBC have been used to test SIP-SIP interworking (Yealink T41S as SIP phone, and MicroSIP as SIP softphone).

As displayed in the figure above, SIP and RTP termination is realized between Patton Gateway / eSBC and the NGN-Softswitch, meaning that the SIP-Trunking functionality on the IAD shall be disabled for this scenario.

Recommended Firmware version : Trinity 3.18 (August 2020 Release) or higher

SIP-Trunk Access Data:

SIP-Domain:	tel.t-online.de
Outbound-Proxy:	55XXXXXXXXXX.primary.companyflex.de
	55XXXXXXXXXX.secondary.companyflex.de
Registrar:	tel.t-online.de
Registration / pilot number:	+4919929600000000XXXXX
Telephony username:	+4919929600000000XXXXX@tel.t-online.de
Telephony password:	XXXXXXXXX
Assigned phone number ranges:	+49 228 4220839 0*
	+49 228 4220839 1*
	+49 228 18449677
	+49 228 40386765

Registration mode

The Technical Specification 1TR119 describes the supported registration mode in the chapter 5.1., based on RFC 3261. The registering device (SmartNode) shall send only one initial REGISTER to the NGN using the provided pilot number for the SIP-Trunk. After a successful authentication, all numbers related to the SIP trunk are implicitly registered.

The pilot number corresponds to the Private Identity (registration phone number) provided along with the Password for every Trunk Group of a SIP-Trunk of Deutsche Telekom as part of the customer contract.

Example of registration request sent by Patton's eSBC / GW with configuration for 1TR119 without encryption (SIP/TCP & RTP):

```
>>> Sent 979 bytes to 217.0.21.70:5060:
    REGISTER sip:tel.t-online.de SIP/2.0
    Via: SIP/2.0/TCP 192.168.1.1:5060;branch=z9hG4bK8a52af72cfa431579.727f8ec1ebf94ac89;rport
    Route: <sip:551134275265.primary.companyflex.de;lr>, <sip:551134275265.secondary.companyflex.de;lr>
    Max-Forwards: 70
    From: <sip:+491992960000000XXXXX@tel.t-online.de>;tag=ec16c46402
    To: <sip:+491992960000000XXXXX@tel.t-online.de>
    Call-ID: d14e5a42f9bcd765
    CSeq: 668352851 REGISTER
    Authorization: Digest username="+491992960000000XXXXX",realm="tel.t-online.de",nonce="C12C...",
uri="sip:tel.t-online.de",response="...",algorithm=MD5,qop=auth,cnonce="...",nc=00000001
    Contact: <sip:+491992960000000XXXXX@192.168.1.1:5060;transport=tcp>;reg-id=1;+sip.instance=
"<urn:00A0BA0E8D0B:GW_SIP_DTAG>"
    Expires: 1800
    Supported: path, outbound
    User-Agent: Patton SN4131/2ETH4BIS8VHP 00A0BA0E8D0B 3.18.T8267-2 1.7 M5T SIP Stack/4.2.28.153
    Content-Length: 0
```

Additionally, for media encryption (SRTP), 1TR119 / chapter 10.3 describes the required Registration Mechanism based on RFC 3329 and Sipcore Mediasec Parameter.

Example of registration request sent by Patton's eSBC / GW with configuration for 1TR119 with encryption (SIP/TLS & SRTP) enabled, logged by the device internal SIP Trace:

```
>>> Sent 1179 bytes to 217.0.137.2:5061:
     REGISTER sip:tel.t-online.de SIP/2.0
     Via: SIP/2.0/TLS 192.168.0.206:5061;branch= …;rport
     Route: <sip:551134275265.primary.companyflex.de;lr>, <sip:551134275265.secondary.companyflex.de;lr>
     Proxy-Require: mediasec
     Max-Forwards: 70
     From: <sip:+491992960000000XXXXX@tel.t-online.de>;tag=19bfdc9198
     To: <sip:+491992960000000XXXXX@tel.t-online.de>
     Call-ID: 564eac129cf084af
     CSeq: 1448212192 REGISTER
     Authorization: Digest username="+491992960000000XXXXX",realm="tel.t-online.de",
         nonce="F48…",uri="sip:tel.t-online.de",response="abc…",algorithm=MD5,qop=auth,cnonce="…",nc=00000001
     Contact: <sip:+491992960000000XXXXX@192.168.0.206:5061;transport=tls>;reg-
         id=1;+sip.instance="<urn:00A0BA0D8A8C:GW_SIP_DTAG_TLS>"
     Expires: 1800
     Require: mediasec
     Security-Client: sdes-srtp;mediasec
     Security-Verify: msrp-tls;mediasec
     Security-Verify: sdes-srtp;mediasec
     Security-Verify: dtls-srtp;mediasec
     Supported: path, outbound
     User-Agent: Patton SN4131/2ETH2BIS4VHP 00A0BA0D8A8C 3.18.T8267-2 1.6 M5T SIP Stack/4.2.28.153
     Content-Length: 0
```

Configuration guide

The example below is based on Patton SN4131 (ISDN BRI) test configuration file SN_Config_CFLEX_19.cfg with Trinity Firmware Version 3.18.
If an analog or an ISDN PRI Gateway model is used, adapt the port settings accordingly.
The configuration contains both the unencrypted SIP/TCP + RTP and the encrypted part SIP/TLS + SRTP. Only three very minor configuration changes must be done to switch from one to the other operation mode (see red highlighting).

Meaning of the highlighted configuration parts:

- Recommended general configuration (non-variable part)
- Project / customer specific configuration (variable part): should be adapted/replaced accordingly by considering your customer's environment (IP addresses, DDI range, credentials etc.)
- Switch between [SIP/TCP + RTP] and [SIP/TLS + SRTP]
- (Comments are inserted between # and do not affect the configuration file)

```
cli version 4.00
superuser admin password xxxxxx
system hostname DTAG_CompanyFlex
system description DTAG_CompanyFlex
system location "Inalp Networks - Bern CH"
system provider DTAG
clock local default-offset +01:00
clock local dst-rule SUMMERTIME +1:00 from mar last sunday 02:00 2019 until oct last sunday 03:00
2036

profile aaa DEFAULT
  method 1 nodems continue-on-reject
  method 2 local
  method 3 none

console
  use profile aaa DEFAULT

telnet-server
  use profile aaa DEFAULT
  no shutdown

ssh-server
  use profile aaa DEFAULT
  no shutdown

snmp-server
  shutdown

web-server
  protocol http port 80
  protocol https port 443
  use profile aaa DEFAULT
  no shutdown

ntp
  server 0.de.pool.ntp.org
  server 1.de.pool.ntp.org
  server 2.de.pool.ntp.org
  server 3.de.pool.ntp.org
  no shutdown

system
  clock-source 1 bri 0 0
  clock-source 2 bri 0 1

profile napt NAPT_WAN

# DNS server config. used for SIP devices and other IP clients on LAN side #
# Ignore this part otherwise #
dns-server
  host 172.30.74.1 smartnode.local
```

```
  relay dns-client
  no shutdown

# In necessary, configure one or more DNS servers.
# Mandatory if static IP addressing is used on WAN interface.
dns-client
  name-server 9.9.9.11

# Enable DHCP for SIP devices and other IP clients on LAN side #
# Ignore this part otherwise #
profile dhcp-server DHCPS_LAN
  network 172.30.74.0/24
  lease 24 hours
  default-router 172.30.74.1
  domain-name-server 172.30.74.1
  include 172.30.74.2 172.30.74.100

profile tls DEFAULT
  authentication incoming
  authentication outgoing
  private-key pki:private-key/DEFAULT
  own-certificate 1 pki:certificate/DEFAULT
  diffie-hellman-parameters pki:diffie-hellman-parameters/DEFAULT-2048

profile tls PF_TLS_DTAG
  no protocol tls-v1.0
  no protocol tls-v1.1
  no authentication incoming
  authentication outgoing
  private-key pki:private-key/DEFAULT
  own-certificate 1 pki:certificate/DEFAULT
  diffie-hellman-parameters pki:diffie-hellman-parameters/DEFAULT-2048

profile tone-set DEFAULT

profile voip DEFAULT
  codec 1 g711alaw64k rx-length 20 tx-length 20
  codec 2 g722-64k rx-length 20 tx-length 20
  codec 3 g711ulaw64k rx-length 20 tx-length 20
  codec 4 g729 rx-length 20 tx-length 20
  codec 5 transparent-clearmode rx-length 20 tx-length 20
  dtmf-relay rtp
  response-preferred-codec g711alaw64k
  fax transmission 1 relay t38-udp
  fax transmission 2 bypass g711alaw64k rx-length 20 tx-length 20
  fax max-bit-rate 9600

profile voip SRTP
  codec 1 g711alaw64k rx-length 20 tx-length 20
  codec 2 g722-64k rx-length 20 tx-length 20
  codec 3 g711ulaw64k rx-length 20 tx-length 20
  codec 4 transparent-clearmode rx-length 20 tx-length 20
  srtp transmission forced
  response-preferred-codec g711alaw64k
  fax transmission 1 relay t38-udp
  fax transmission 2 bypass g711alaw64k rx-length 20 tx-length 20
  fax max-bit-rate 9600

profile pstn DEFAULT

profile rip DEFAULT

profile sip DEFAULT

context ip ROUTER

# Depending on the customer network administration #
# set dynamic WAN IP through DHCP like here or, if required, set a static IP + routing #
  interface WAN
    ipaddress DHCP dhcp

# In our setup, the LAN Itf is used on private side for the SIP-SIP SBC setup  #
  interface LAN
    ipaddress LAN 172.30.74.1/24

# In case of static settings, insert the corresponding routes #
```

```
# No routes in this example, as the WAN Itf gets them through DHCP #
  routing-table DEFAULT

  bgp
    shutdown

  rip
    shutdown

context ip ROUTER
  use profile dhcp-server DHCPS_LAN

nodems-client
  organization-key XXXXXXXX         # configure your specific Org.Key for Patton Cloud #
  resource any
  no shutdown

profile ppp DEFAULT

cwmp-client
  session-retry-maximum 1
  shutdown

  stun
    shutdown

context cs SWITCH

# internal numbering plan 20x           #
# ISDN MSNs : 200,201 on BRI01 and BRI02 #
# SIP phones (LAN side) : 205, 206       #
# DDI translation used :                 #
# (pubic <---> internal)
# +49 228 422 0839000 <---> 200          #
# +49 228 422 0839001 <---> 201          #
# ...                        ...         #
# +49 228 422 0839009 <---> 209          #

  mapping-table calling-e164 to calling-e164 MT_INTERNAL_to_SIP_DTAG_CNPN
    map 20(.) to \+49228422083900\1

  mapping-table called-e164 to called-e164 MT_INTERNAL_to_SIP_DTAG_CDPN
    map 0(.%) to \+49\1
    map 00(.%) to +\1

  mapping-table called-e164 to called-e164 MT_SIP_DTAG_to_INTERNAL_CDPN
    map 0228422083900(.) to 20\1
    map 49228422083900(.) to 20\1

  mapping-table calling-numbering-plan to calling-numbering-plan MT_SIP_DTAG_to_INTERNAL_NPI
    map default to isdn-telephony

  mapping-table calling-e164 to calling-e164 MT_SIP_DTAG_to_INTERNAL_CNPN
    map 49(.%) to \1

  mapping-table calling-e164 to calling-type-of-number MT_SIP_DTAG_to_INTERNAL_CNPN_DE
    map 49(.%) to national

  routing-table called-e164 RT_FROM_INTERNAL
    route T dest-interface IF_SIP_DTAG CF_INTERNAL_to_SIP_DTAG
# Valid for SIP/TCP & RTP. In case of SIP/TLS + SRTP, set dest-interface to IF_SIP_DTAG_TLS instead #

# In the RT below, set the PBX DDI range(s) to be routed to the Hunt Group SRV_HG, by using RegEx #
# In our test case, only these two ISDN phone numbers were used: 49228422083900[01] #
# In our test case, only these two SIP phone numbers were used: 49228422083900[56] #
  routing-table called-e164 RT_FROM_SIP_DTAG
    route 49228422083900[01] dest-service SRV_HG CF_SIP_DTAG_to_INTERNAL
    route 49228422083900[56] dest-service SRV_SIP_LOC MT_SIP_DTAG_to_INTERNAL_CDPN

  complex-function CF_INTERNAL_to_SIP_DTAG
    execute 1 MT_INTERNAL_to_SIP_DTAG_CNPN
    execute 2 MT_INTERNAL_to_SIP_DTAG_CDPN

  complex-function CF_SIP_DTAG_to_INTERNAL
    execute 1 MT_SIP_DTAG_to_INTERNAL_CDPN
    execute 2 MT_SIP_DTAG_to_INTERNAL_NPI
```

```
    execute 3 MT_SIP_DTAG_to_INTERNAL_CNPN_DE
    execute 4 MT_SIP_DTAG_to_INTERNAL_CNPN

  interface isdn IF_BRI_00
    route call dest-table RT_FROM_INTERNAL
    call-reroute emit
    diversion emit

  interface isdn IF_BRI_01
    route call dest-table RT_FROM_INTERNAL
    call-reroute emit
    diversion emit

  interface sip IF_SIP_DTAG
    bind context sip-gateway GW_SIP_DTAG
    route call dest-table RT_FROM_SIP_DTAG
    remote tel.t-online.de
    local tel.t-online.de
    hold-method direction-attribute sendonly
    no call-transfer emit
    call-reroute emit
    privacy
    uri-scheme sip
    address-translation outgoing-call contact-header user-part fix +491992960000000XXXXX
    session-timer 1800

  interface sip IF_SIP_PHONES
    bind context sip-gateway GW_SIP_PHONES
    route call dest-table RT_FROM_INTERNAL
    local 172.30.74.1
    call-reroute accept
    privacy

  interface sip IF_SIP_DTAG_TLS
    bind context sip-gateway GW_SIP_DTAG_TLS
    route call dest-table RT_FROM_SIP_DTAG
    remote tel.t-online.de
    local tel.t-online.de
    hold-method direction-attribute sendonly
    no call-transfer emit
    call-reroute emit
    privacy
    uri-scheme sip
    address-translation outgoing-call contact-header user-part fix +491992960000000XXXXX
    use profile voip SRTP
    session-timer 1800

  service hunt-group SRV_HG
    drop-cause normal-unspecified
    drop-cause no-circuit-channel-available
    drop-cause network-out-of-order
    drop-cause temporary-failure
    drop-cause switching-equipment-congestion
    drop-cause access-info-discarded
    drop-cause circuit-channel-not-available
    drop-cause resources-unavailable
    route call 1 dest-interface IF_BRI_00
    route call 2 dest-interface IF_BRI_01

# SIP location service necessary if SIP devices are used #
  service sip-location-service SRV_SIP_LOC
    bind location-service LS_SIP_PHONES

context cs SWITCH
  no shutdown

# Authentication service : use here the SIP-Trunk credentials provided by DTAG #
authentication-service AS_DTAG_CFLEX
  realm 1 tel.t-online.de
  username +491992960000000XXXXX password XXXXXX

authentication-service AS_SIP_PHONES
  realm 1 GROUP1
  username 205 password XXXXX
  username 206 password XXXXX
```

```
location-service LS_SIP_DTAG
  domain 1 tel.t-online.de

  identity-group DEFAULT
    alias expression \+4922842208390.*
    alias expression \+4922842208391.*
    alias expression \+4922818449677
    alias expression \+4922840386765
    user phone

    authentication outbound
      authenticate 1 authentication-service AS_DTAG_CFLEX username +491992960000000XXXXX

    registration outbound
      registrar tel.t-online.de
      uri-scheme sip
      transport-protocol force tcp
      proxy 1 551134275265.primary.companyflex.de
      proxy 2 551134275265.secondary.companyflex.de
      lifetime 1800
      register auto
      flows
      keep-alive options 50

    call outbound
      force-destination registrar address
      proxy 1 551134275265.primary.companyflex.de
      proxy 2 551134275265.secondary.companyflex.de
      transport-protocol force tcp
      use profile voip DEFAULT
      flows

  identity +491992960000000XXXXX inherits DEFAULT

location-service LS_SIP_DTAG_TLS
  domain 1 tel.t-online.de

  identity-group DEFAULT
    alias expression \+4922842208390.*
    alias expression \+4922842208391.*
    alias expression \+4922818449677
    alias expression \+4922840386765
    user phone

    authentication outbound
      authenticate 1 authentication-service AS_DTAG_CFLEX username +491992960000000XXXXX

    registration outbound
      registrar tel.t-online.de
      uri-scheme sip
      transport-protocol force tls
      proxy 1 551134275265.primary.companyflex.de
      proxy 2 551134275265.secondary.companyflex.de
      lifetime 1800
      register auto
      flows
      keep-alive options 50

    call outbound
      force-destination registrar address
      proxy 1 551134275265.primary.companyflex.de
      proxy 2 551134275265.secondary.companyflex.de
      transport-protocol force tls
      use profile voip SRTP
      flows

    call inbound
      use profile voip SRTP

  identity +491992960000000XXXXX inherits DEFAULT

location-service LS_SIP_PHONES
  domain 1 172.30.74.1
  match-any-domain

  identity-group ID_GROUP_PHONES
```

```
    authentication inbound
      authenticate authentication-service AS_SIP_PHONES

    registration inbound
      lifetime default 600 min 1 max 800

  identity 205 inherits ID_GROUP_PHONES

    authentication inbound

  identity 206 inherits ID_GROUP_PHONES

    authentication inbound

sip
  no lock-dns-record

context sip-gateway GW_SIP_DTAG
  bind location-service LS_SIP_DTAG

  interface IF_GW_SIP_DTAG
    transport-protocol udp+tcp 5060
    no transport-protocol tls
    bind ipaddress ROUTER WAN DHCP

context sip-gateway GW_SIP_DTAG
  no shutdown
# Valid for SIP/TCP & RTP. In case of SIP/TLS + SRTP, set state to shutdown #

context sip-gateway GW_SIP_DTAG_TLS
  use profile tls PF_TLS_DTAG
  bind location-service LS_SIP_DTAG_TLS

  interface IF_GW_SIP_DTAG_TLS
    no transport-protocol udp+tcp
    transport-protocol tls 5061
    bind ipaddress ROUTER WAN DHCP

context sip-gateway GW_SIP_DTAG_TLS
  security-agreement mediasec
  shutdown
# Valid for SIP/TCP & RTP. In case of SIP/TLS + SRTP, set state to no shutdown #

context sip-gateway GW_SIP_PHONES
  bind location-service LS_SIP_PHONES

  interface IF_GW_SIP_PHONES
    transport-protocol udp+tcp 5060
    no transport-protocol tls
    bind ipaddress ROUTER LAN LAN

context sip-gateway GW_SIP_PHONES
  no shutdown

sip-survivability
  shutdown

port ethernet 0 0
  bind interface ROUTER WAN
  no shutdown

port ethernet 0 1
  bind interface ROUTER LAN
  no shutdown

port bri 0 0
  power-feed

  q921
    permanent-layer2
    protocol pmp
    uni-side auto

    q931
      protocol dss1
```

```
      uni-side net
      max-calls 2
      channel-range 0 1
      bind interface SWITCH IF_BRI_00

port bri 0 0
  no shutdown

port bri 0 1
  power-feed

  q921
    permanent-layer2
    protocol pmp
    uni-side auto

    q931
      protocol dss1
      uni-side net
      max-calls 2
      channel-range 0 1
      bind interface SWITCH IF_BRI_01

port bri 0 1
  no shutdown
```

Test plan – Standard calls

Following calls and telephony features have been tested with the configuration described in this guide.

Outgoing calls

Test scenario	Result ×/✓	Comments
Outgoing call to PSTN : call to a fix network number in Germany, call pick up, dialog et hang up by calling party		Calls performed with ISDN and SIP devices behind Patton. In ISDN case, mostly ISDN devices connected through BRI, but also through a BRI/PRI PBX.
The ring back tone is received and displayed by the calling terminal.	✓	
The called device is ringing.	✓	
The caller's number / identity is correctly presented.	✓	
The call is being connected after pick up.	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by calling party.	✓	
Call hang up by calling party : call correctly released on both sides.	✓	
Outgoing call to PSTN : call to a fix network number in Germany, call pick up, dialog et hang up by called party		
Call hang up by called party : call correctly released on both sides.	✓	
Outgoing call to PSTN : call to a fix network number in Germany, hang up before pick up		
The device of the called party is ringing.	✓	
Hang up before pick up: called device stops ringing.	✓	

Incoming calls

Test scenario	Result ×/✓	Comments
Incoming call from PSTN : call from a fix network number in Germany, call pick up, dialog et hang up by calling party		Calls performed with ISDN and SIP devices behind Patton. In ISDN case, mostly ISDN devices connected through BRI, but also through a BRI/PRI PBX.
The ring back tone is received and displayed by the calling terminal.	✓	
The called device is ringing.	✓	
The caller's number / identity is correctly presented.	✓	
The call is being connected after pick up.	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by calling party.	✓	
Call hang up by calling party : call correctly released on both sides.	✓	
Incoming call from PSTN : call from a fix network number in Germany, call pick up, dialog et hang up by called party		
Call hang up by called party : call correctly released on both sides.	✓	
Outgoing call to PSTN : call to a fix network number in Germany, hang up before pick up		
The phone device of the called party is ringing.	✓	
Hang up before pick up: called device stops ringing.	✓	

DTMF codes (RFC2833 / RFC4733)

Test scenario	Result ×/✓	Comments
Patton's eSBC / Gateway uses RFC 2833 / RFC4733 by sending DTMF events.	✓	
The payload can be set to 101.	✓	
Call a public IVR (for ex. Vodaphone 0800 724 2640 or health care service 116117) and check the correct menu navigation.	✓	
Call a DTAG IVR (for ex. customer service 0800 33 01000 or 0800 33 02202) and check the correct menu navigation.	✓	

Call hold (RFC3264)

Test scenario	Result ×/✓	Comments
Put a call Patton (ISDN BRI/PRI) <-> DTAG/NGN/PSTN on hold		
Use of the method described in RFC 3264 (ReINVITE offer SDP parameter 'media attribute' set to recvonly / sendonly / inactive).	✓	
The remote party of the call is correctly put on hold, according to chapter "13.9 Call Hold and Announcements (Music-on-Hold)" of 1TR119.	✓	The MoH itself is not played by Patton device, but by the (IP)PBX of the remote connected party.
Put call off hold : the communication gets reestablished.	✓	
Put a call Patton (SIP) <-> DTAG/NGN/PSTN on hold		
Use of the method described in RFC 3264 (ReINVITE offer SDP parameter 'media attribute' set to recvonly / sendonly / inactive).	✓	
The remote party of the call is correctly put on hold, according to chapter "13.9 Call Hold and Announcements (Music-on-Hold)" of 1TR119.	✓	The MoH itself is not played by Patton device, but by the (IP)PBX of the remote connected party.
Put call off hold : the communication gets reestablished.	✓	

Emergency calls

These are calls setup from customer's (IP)PBX through Patton SBC to destination numbers 110, 112, 115, 116 (or 11x in general) in R-URI.

The most relevant emergency calls have been tested to check the correct routing by the NGN based on the origin. Also the correct call setup has been verified.

Test scenario	Result ×/✓	Comments
Call the police: 110		
Dial 110 from a device behind Patton SBC. Call setup without delay.	✓	
The call is correctly setup.	✓	
The public-safety point answering the call is responsible for the region of the calling party.	✓	
Call the EU-Emergency: 112		
Dial 112 from a device behind Patton SBC. Call setup without delay.	✓	
The call is correctly setup.	✓	

The public-safety point answering the call is responsible for the region of the calling party.	✓	
Call the Government Service number: 115		
Dial 115 from a device behind Patton SBC. Call setup without delay.	✓	
The call is correctly setup.	✓	
The public-safety point answering the call is responsible for the region of the calling party.	✓	

Unsuccessful calls / wrong dialled numbers

Test scenario	**Result ×/✓**	**Comments**
Call to a busy PSTN subscriber		
Setup a call from a PSTN subscriber line having no Call-Waiting feature enabled. Setup a call from a device behind Patton eSBC / Gateway to this busy PSTN line. The calling terminal informs the user by the display and busy tone that the called party is busy.	✓	
Call a number not belonging to the national numbering plan		
Setup a call from a device behind Patton eSBC / Gateway to a phone number not belonging to the national numbering plan. Either a fast-busy tone is signaled or a corresponding announcement of the provider is played.	✓	
Call a non-existing international phone number		
Setup a call from a device behind Patton eSBC / Gateway to a non-existing / invalid international phone number. Either a fast-busy tone is signaled, or a corresponding announcement of the provider is played.		Could not be tested due to the traffic class limitation in the IP-Test center to only national numbers, but this should also work the same way as the previous test (NGN configuration)

Telephony codecs

Test scenario	**Result ×/✓**	**Comments**
Patton eSBC / Gateway shall support G.711a	✓	
The codec G.711a has the highest priority in the offer of outgoing calls. Incoming calls with G.711a codec are correctly answered and G.711a is supported.		
Patton eSBC / Gateway should support G.722	✓	

The codec G.722 is defined with 2nd priority in the offer of outgoing calls. Incoming calls with G.722 codec are correctly answered and G.722 is supported.		
Patton eSBC / Gateway should support G.711µ, G.729 and clear channel		
The codecs G.711µ, G.729 and clear channel (RFC 4040) are not modified in offers for calls via the NGN. They can be used if all involved elements (the B-party's end device as well as e.g. other carrier's nodes) agree in negotiating them The codecs G.711 µ , G.729 and clear channel (RFC 4040) are not modified in offers for calls via the NGN.	✓	

DDI Management (belonging to the SIP-Trunk account DTAG 1TR119)

Test scenario	Result ×/✓	Comments
Authentication		
Outgoing calls from assigned DDI range correctly use the authentication credentials belonging to main number of the SIP-Trunk account.	✓	
From header field		
The From header field of outgoing calls from Patton device contains the DDI of the calling party. Patton GW / eSBC sends an E.164 phone number from the phone numbers assigned to the SIP-Trunk in the P-Preferred-Identity header field.	✓	
Correct connection of incoming calls		
Patton GW / eSBC displays the identity provided in the From Header, since a P-Asserted-Identity header field may not be available in all cases.	✓	
OIR of incoming calls		
Incoming calls with OIR (Originating Identification Restriction): Patton device correctly interprets the From header field of NGN (set to anonymous@anonymous.invalid) and restricts the presentation of the calling party number.	✓	
OIR of outgoing calls		
Outgoing calls with OIR (Originating Identification Restriction): Patton device sets the Privacy header field to "id" or "user" and the calling party number in the P-Preferred-Identity header field.	✓	

Test plan - Telephony features

Call-Waiting

Test scenario	Result ×/✓	Comments
Setup a first call to a device behind Patton eSBC, then setup a second call from PSTN to the same device and do not pick-up.		
Ring-back tone played on the PSTN calling device.	✓	
The call-waiting signal is played on the busy device.	✓	
Setup a first call to a device behind Patton eSBC, then setup a second call from PSTN to the same device. Hang-up the first call during the call-waiting of the second.		
After hang-up, the first device starts ringing.	✓	
Pick-up the call : the second call is correctly connected.	✓	
Setup a first call to a device behind Patton eSBC, then setup a second call from PSTN to the same device. Put the first call on hold and pick-up the second.		
After hearing the call-waiting signal on the first device, press R-key (or line key). The second call is connected.	✓	
The first call is put on hold	✓	
Pressing R-key (or Swap) allows swapping from one call to the other.	✓	

Blind call transfer (unattended)

As there are many different call scenarios, only the most common use cases have been tested, which are listed below. Furthermore, call transfer tests enable the validation of call-waiting / swapping. Each time we refer to a device behind Patton, we mean an extension belonging to the DDI of the SIP-Trunk. It can be a device on a PBX connected to Patton SmartNode, or an ISDN device directly connected through BRI, or a SIP device registered on SmartNode.

Incoming call transferred to an outgoing call

Test scenario	Result ×/✓	Comments
Call transfer from device behind Patton to a PSTN device: call from a 1st to a 2nd device behind Patton. From the 2nd device, call setup to a PSTN device.		

Setup a call from a 1st to a 2nd device behind Patton. Pick up the call on the 2nd device and put the call on hold. The 1st device is put on hold.	✓	
Proceed to a call transfer to a PSTN phone number. The call is being transferred.	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by calling party.	✓	
Call transfer from one to another device behind Patton: call from a 1st to a 2nd device behind Patton. From the 2nd device, call setup to a 3rd device behind Patton.		
Setup a call from a 1st to a 2nd device behind Patton. Pick up the call on the 2nd device and put the call on hold. The 1st device is put on hold.	✓	Fully internal call scenario, not involving the SIP-Trunk
Proceed to a call transfer to a 3rd device behind Patton. The call is being transferred.	✓	Fully internal call scenario, not involving the SIP-Trunk
Voice correctly heard by called party.	✓	Fully internal call scenario, not involving the SIP-Trunk
Voice correctly heard by calling party.	✓	Fully internal call scenario, not involving the SIP-Trunk
Call transfer from a PSTN device to another PSTN device: call from a 1st PSTN device to a device behind Patton. From the 2nd device, call setup to a PSTN device.		
From a PSTN device call a device behind Patton. Pick up the call and put it on hold. The PSTN device is put on hold.	✓	
Proceed to a blind transfer to a 2nd PSTN device. The call is being transferred.	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by calling party.	✓	
Call transfer of an incoming PSTN call to an internal device: call from a PSTN device to a device behind Patton. From this device, call setup to another device behind Patton.		
From a PSTN device call a device behind Patton. Pick up the call and put it on hold. The PSTN device is put on hold.	✓	
Proceed to a blind transfer to a 2nd device behind Patton. The call is being transferred.	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by calling party.	✓	

Transfer from outgoing call to outgoing call

Test scenario	Result ✗/✓	Comments
Call transfer of an outgoing call to a PSTN device: call from a 1st to a 2nd device behind Patton. From the 1st device, call setup to a PSTN device.		
Setup a call from a 1st to a 2nd device behind Patton. Pick up the call on the 2nd device and put the call on hold. The 2nd device is put on hold.	✓	
From the 1st device proceed to a call transfer to a PSTN phone number. The call is being transferred.	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by calling party.	✓	
Call transfer of an outgoing call to a PSTN device: call from a device behind Patton to a PSTN device. From the same device behind Patton, call setup to a 2nd PSTN device.		
Setup a call from a device behind Patton to a PSTN device. Pick up the call and put it on hold. The PSTN device is put on hold.	✓	
From the device behind Patton proceed to a call transfer to another PSTN phone number. The call is being transferred.	✓	
Voice correctly heard by the 1st PSTN party.	✓	
Voice correctly heard by the 2nd PSTN party.	✓	
Call transfer of an internal outgoing call to another internal device: call from a 1st to a 2nd device behind Patton. From the 1st device behind Patton, call setup to a 3rd device behind Patton.		
Setup a call from a 1st to a 2nd device behind Patton. Pick up the call and put it on hold. The 2nd device is put on hold.	✓	Fully internal call scenario, not involving the SIP-Trunk
From the 1st device proceed to a call transfer to a 3rd device behind Patton. The call is being transferred.	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by transferred party.	✓	

Attended call transfer

As there are many different call scenarios also in this case, only the most common use cases have been tested, which are listed below.

Incoming call transferred to an outgoing call

Test scenario	Result ×/✓	Comments
Call transfer of an internal incoming call to PSTN device: call from a 1st to a 2nd device behind Patton. From the 2nd device behind Patton, call setup to a PSTN device.		
Setup a call from a 1st to a 2nd device behind Patton. Pick up the call on the 2nd device and put the call on hold. The 1st device is put on hold.	✓	
From the second internal device, call a PSTN device. The call is correctly setup.	✓	
Transfer the call (from 2nd device). The call is correctly transferred	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by transferred party.	✓	
Call transfer from one to another internal device behind Patton : call from a 1st to a 2nd device behind Patton. From the 2nd, call to 1 3rd internal device.		
From a 1st, call a 2nd device behind Patton. On the 2nd device pick up the call and put it on hold. The 1st internal device is put on hold.	✓	
From the 2nd internal device, call a 3rd internal device. The call is correctly setup.	✓	Fully internal call scenario, not involving the SIP-Trunk
Proceed to call transfer. The call is correctly transferred.	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by transferred party.	✓	
Transfer of a PSTN call to a PSTN device : call from a 1st PSTN device to an internal device. From the internal device, call a 2nd PSTN device.		
From a 1st PSTN device, call an internal device behind Patton. On the internal device, pick up the call and put it on hold. The 1st PSTN device is correctly put on hold.	✓	
From the internal device, call a 2nd PSTN device. The call is correctly setup.	✓	
Proceed to call transfer. The call is correctly transferred.	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by transferred party.	✓	
Transfer of a PSTN call to an internal device : call from a PSTN device to a 1st internal device. From this 1st internal device, call 1 2nd internal device.		

From a PSTN device, setup a call to a 1st internal device. On this 1st internal device, pick up the call and put it on hold. The PSTN device is put on hold.	✓	
From the 1st internal device, call a 2nd internal device. The call is correctly setup.	✓	
Proceed to call transfer. The call is correctly transferred.	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by transferred party.	✓	

Transfer of outgoing calls : outgoing call transferred to outgoing call

Test scenario	Result ×/✓	Comments
Transfer of an internal outgoing call to a PSTN device: call from a 1st internal device to a 2nd one. From the 1st internal device, call a PSTN device.		
Setup a call from a 1st internal device to a 2nd one. From the 1st internal device, put the call on hold. The 2nd internal device is correctly put on hold.	✓	
From the 1st internal device, call a PSTN device. The call is correctly setup.	✓	
Proceed to call transfer. The call is correctly transferred.	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by transferred party.	✓	
Transfer of an outgoing PSTN call to a PSTN device : Call from an internal to a 1st PSTN device. From the internal device, call a 2nd PSTN device.		
From an internal device, call a 1st PSTN device. From the internal device, put the call on hold. The 1st PSTN device is correctly put on hold.	✓	
From the internal device, call a 2nd PSTN device. The call is correctly setup.	✓	
Proceed to call transfer. The call is correctly transferred.	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by transferred party.	✓	
Transfer of an internal outgoing call to another internal device : call from a 1st to a 2nd internal device. From the 1st internal device, call a 3rd internal device.		
From a 1st internal device, call a 2nd internal device. From the 1st device, put the call on hold. The 2nd device is put on hold.	✓	Fully internal call scenario, not involving the SIP-Trunk

From the 1st internal device, call a 3rd internal device. The call is correctly setup.	✓	
Proceed to call transfer. The call is correctly transferred.	✓	
Voice correctly heard by called party.	✓	
Voice correctly heard by transferred party.	✓	

3-party conferences

Test scenario	Result ✗/✓	Comments
Setup a 1st call from one to another internal device. From the 1st one, call a PSTN device and establish a 3-party conference.		
From an internal device, call another internal device. From the 1st internal device, put the call on hold and setup a 2nd call to a PSTN device. The 2nd internal device is correctly put on hold and the communication with the PSTN device is established.	✓	The conference feature was tested from either a SIP device registered on Patton eSBC or an ISDN PBX with its internal conference feature, but not from a simple ISDN phone connected to eSBC through BRI port (ISDN phones not supporting 3-P conf. without a PBX)
Establish the conference. The conference is properly established.	✓	
The first device hears the two other devices.	✓	
The two other devices hear the 1st device.	✓	
The two other devices can hear each other.	✓	
Setup a 1st call from one to another internal device. From the 1st one, call a 3rd internal device and establish a 3-party conference.		
From an internal device, call another internal device. From the 1st internal device, put the call on hold and setup another call to a 3rd internal device. The 2nd internal device is correctly put on hold and the communication with the 3rd internal device device is established.	✓	Completely internal call scenario.
Establish the conference. The conference is properly established.	✓	
The first device hears the two other devices.	✓	
The two other devices hear the 1st device.	✓	
The two other devices can hear each other.	✓	
Setup a 1st call from an internal device to a 1st PSTN device. From the internal device, call a 2nd PSTN device and establish a 3-party conference.		
From an internal device, call a 1st PSTN device. From the internal device, put the call on hold and setup another call to a	✓	The conference feature was tested from either a SIP device registered on Patton eSBC or an

2nd PSTN device. The 1st PSTN device is correctly put on hold and the communication with the 2nd PSTN device is established.		ISDN PBX with its internal conference feature, but not from a simple ISDN phone connected to eSBC through BRI port (ISDN phones not supporting 3-P conf. without a PBX)
Establish the conference. The conference is properly established.	✓	
The first device hears the two other devices.	✓	
The two other devices hear the 1st device.	✓	
The two other devices can hear each other.	✓	

Caller Identity Presentation / CLIP

Test scenario	Result ×/✓	Comments
CLIP towards a PSTN device		
From an internal device, setup a call to a PSTN having the CLIP feature available. The calling party number presentation is correct. The call is being connected.	✓	
CLIP towards a device behind Patton		
From a PSTN device, setup a call to an internal device (over SIP-Trunk). The calling party number presentation is correct.	✓	
The calling party name (if any) is correctly displayed. The call is correctly connected.	✓	
CLIR / OIR (Identity restriction)		
On an internal device, activate the Identity Restriction (on SIP devices : enable anonymous calls). From this device, setup a call to a PSTN having the CLIP feature available. The calling party number is NOT presented. The call is being connected.	✓	CLIR call from SIP device with public phone number +492284220839006; Patton eSBC sets the **Privacy: id** header field and the From + PPI header fields are set to Anonymous <sip:+492284220839006@tel.t-online.de;user=phone> **Result OK** CLIR from ISDN phone / PBX, Patton eSBC sets Privacy: id, and From + PPI headers still contain whole user identity in the user part; **Result OK**
CLIR / OIR towards a device behind Patton		
Setup an external call from PSTN to an internal device supporting the CLIP feature. The calling party number is NOT presented. The call is being connected.	✓	
Correct presentation of DDI		
On outgoing calls from a device belonging to the assigned DDI-Range of the SIP-Trunk, the correct DDI number is presented on the PSTN called device.	✓	

Swapping between two hold calls

Hold a PSTN call	**Result ×/✓**	**Comments**
Setup a call with a PSTN device from/to an internal one, then put the call on hold from the internal device. The music on hold is heard correctly on the PSTN device.	✓	MoH dependent on the remote party PBX capabilities.
Swapping from one held call to the other reconnects to the waiting party without issues.	✓	

Call forwarding

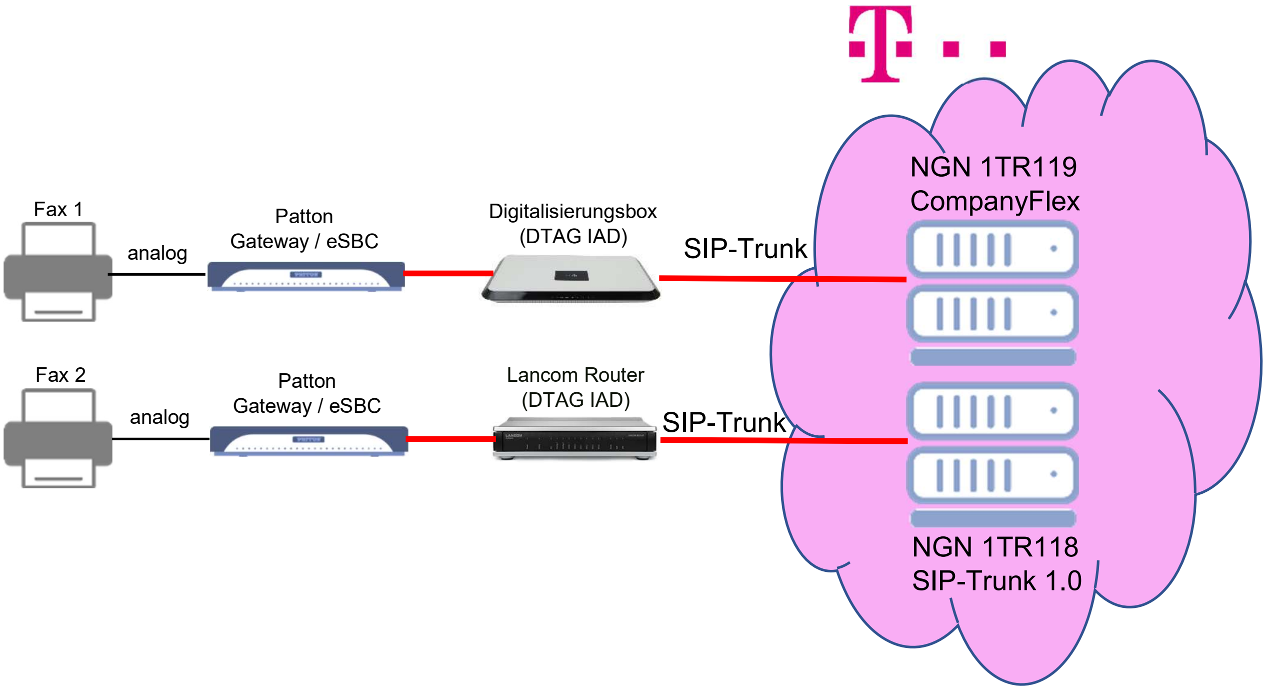
Test scenario	**Result ×/✓**	**Comments**
Call Forwarding Unconditional (CFU) of a PSTN call to another PSTN device		
Set a CFU from an internal device to a 1st PSTN device. From a 2nd PSTN device, call this internal device with activated CFU. The 1st PSTN device starts ringing immediately.	✓	"call-reroute emit" configured on Patton SIP interface towards DTAG → 302 moved temporarily response to DTAG, working OK (RTP does not flow through SIP-Trunk)
The call is established between the two devices and both parties hear each other.	✓	
Call Forwarding Unconditional (CFU) of an internal call to a PSTN device		
Keep the same CFU activated before. From another internal device, call the internal device on which CFU is active. The PSTN device (CFU target number) starts ringing immediately.	✓	
The call is established between the two devices and both parties hear each other.	✓	
Call Forwarding Unconditional (CFU) of a PSTN call to another internal device		
Set a CFU from an internal device to another internal device. From a PSTN device, call this internal device with activated CFU. The 2nd internal device (CFU target number) starts ringing immediately.	✓	
The call is established between the two devices and both parties hear each other.	✓	
Call Forwarding Unconditional (CFU) of an internal call to another internal device		
Keep the same CFU activated before. From another (3rd) internal device, call the internal device on which CFU is active. The 2nd internal device (CFU target number) starts ringing immediately.	✓	Internal call scenario without use of SIP-Trunk

The call is established between the two devices and both parties hear each other.	✓	
Call Forwarding No Reply (CFNR) of a PSTN call to another PSTN device		
Deactivate all call forwardings. Activate a CFNR to a PSTN device, with for example ringing duration of 20s before forwarding. From another PSTN device, call the internal device on which CFNR was activated and do not pick up the call. The internal device is ringing during 20s, then stops ringing.	✓	"call-reroute emit" configured on Patton SIP interface towards DTAG → 302 moved temporarily response to DTAG, working OK (RTP does not flow through SIP-Trunk)
The target PSTN device starts ringing. Pick up the call.	✓	
The call is established between the two devices and both parties hear each other.	✓	
Call Forwarding No Reply (CFNR) of an internal call to a PSTN device		
Keep the same CFNR activated before. From another internal device, call the internal device on which CFNR was activated. The internal device is ringing during the configured delay (20s), then stops ringing.	✓	
The target PSTN device starts ringing. Pick up the call.	✓	
The call is established between the two devices and both parties hear each other.	✓	
Call Forwarding No Reply (CFNR) of a PSTN call to another internal device		
Deactivate all call forwardings. Activate a CFNR to another internal device, with for example ringing duration of 20s before forwarding. From another PSTN device, call the internal device on which CFNR was activated and do not pick up the call. The internal device is ringing during 20s, then stops ringing.	✓	
The other internal device (CFNR target number) starts ringing. Pick up the call.	✓	
The call is established between the two devices and both parties hear each other.	✓	
Call Forwarding No Reply (CFNR) of an internal call to another internal device		
Keep the same CFNR activated before. From a 3rd internal device, call the internal device on which CFNR was activated and do not pick up. The 1st device starts is ringing during 20s, then stops ringing.	✓	
The other internal device (CFNR target number) starts ringing. Pick up the call.	✓	
The call is established between the two devices and both parties hear each other.	✓	

Call Forwarding Busy (CFB) of a PSTN call to another PSTN device		
Deactivate all call forwardings. On an internal device, activate a CFB to a PSTN device. Let this internal device be busy. From another PSTN device, call this internal device. The PSTN device, which was configured as CFB target number, starts ringing immediately. Pick up the call.	✓	
The call is established between the two devices and both parties hear each other.	✓	
Call Forwarding Busy (CFB) of an internal call to a PSTN device		
Keep the CFB previously activated. Let the internal device with activated CFB be busy. From another internal device, call this internal device. The PSTN device (CFB target number) starts ringing immediately.	✓	
The call is established between the two devices and both parties hear each other.	✓	
Call Forwarding Busy (CFB) of a PSTN call to another internal device		
Deactivate all call forwardings. Activate a CFB to another internal device. Let the internal device with activated CFB be busy. From a PSTN device, call this 1st internal device. The 2nd internal device (CFB target number) starts ringing immediately. Pick up the call.	✓	
The call is established between the two devices and both parties hear each other.	✓	
Call Forwarding Busy (CFB) of an internal call to another internal device		
Keep the CFB previously activated. Let the internal device with activated CFB be busy. From another (3rd) internal device, call this 1st internal device. The 2nd internal device (CFB target number) starts ringing immediately. Pick up the call.	✓	
The call is established between the two devices and both parties hear each other.	✓	

Fax calls

Fax calls have been tested within the following network topology in the 3rd Party Lab of DTAG:



Outgoing Fax calls:

Fax 1 -> Patton GW1-> IAD -> NGN 1TR119 -> NGN 1TR118 -> IAD -> Patton GW2 -> Fax 2

- T.38 Fax calls tested successfully with provided configuration
- Fallback to G.711 tested successfully

Incoming Fax calls:

Fax 2 -> Patton GW2-> IAD -> NGN 1TR118 -> NGN 1TR119 -> IAD -> Patton GW1 -> Fax 1

- T.38 Fax calls tested successfully with provided configuration
- Fallback to G.711 tested successfully

Signalling and Media Security (TLS & SRTP)

SIP/TLS and SRTP have been successfully tested with the provided configuration guide, which is fully prepared for both configurations.

In order for SIP/TLS and SRTP to work with this Patton Configuration, only the three following minimal changes must be done, as already highlighted in the configuration guide above:

- Deactivate the SIP Gateway for SIP/TCP (shutdown)
- Activate the SIP Gateway for TLS (no shutdown)
- In the Routing Table from internal side towards SIP-Trunk, replace dest-interface IF_SIP_DTAG by dest-interface IF_SIP_DTAG_TLS.

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