Unify OpenScape 4000

Guide for SIP Service Providers

Version 2.5

Document History

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2.0	October 16, 2013	Birgit Stehle, Unify PH DPL H4K 3	Rebranding
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2.2	July 15, 2015	Birgit Stehle, Unify PH DPL H4K 3	Revision of test cases, corrections
2.3	September 2016	Birgit Bammesreiter	Formal corrections
2.4	May 2019	Birgit Bammesreiter	Add test case emergency call
2.5	October 2020	Birgit Bammesreiter	Rebranding

Table of Contents

1. Introduction

This document describes the test requirements and test scenarios to verify interoperability of SIP service provider with OpenScape 4000.

1.1 Purpose

The document covers the following areas:

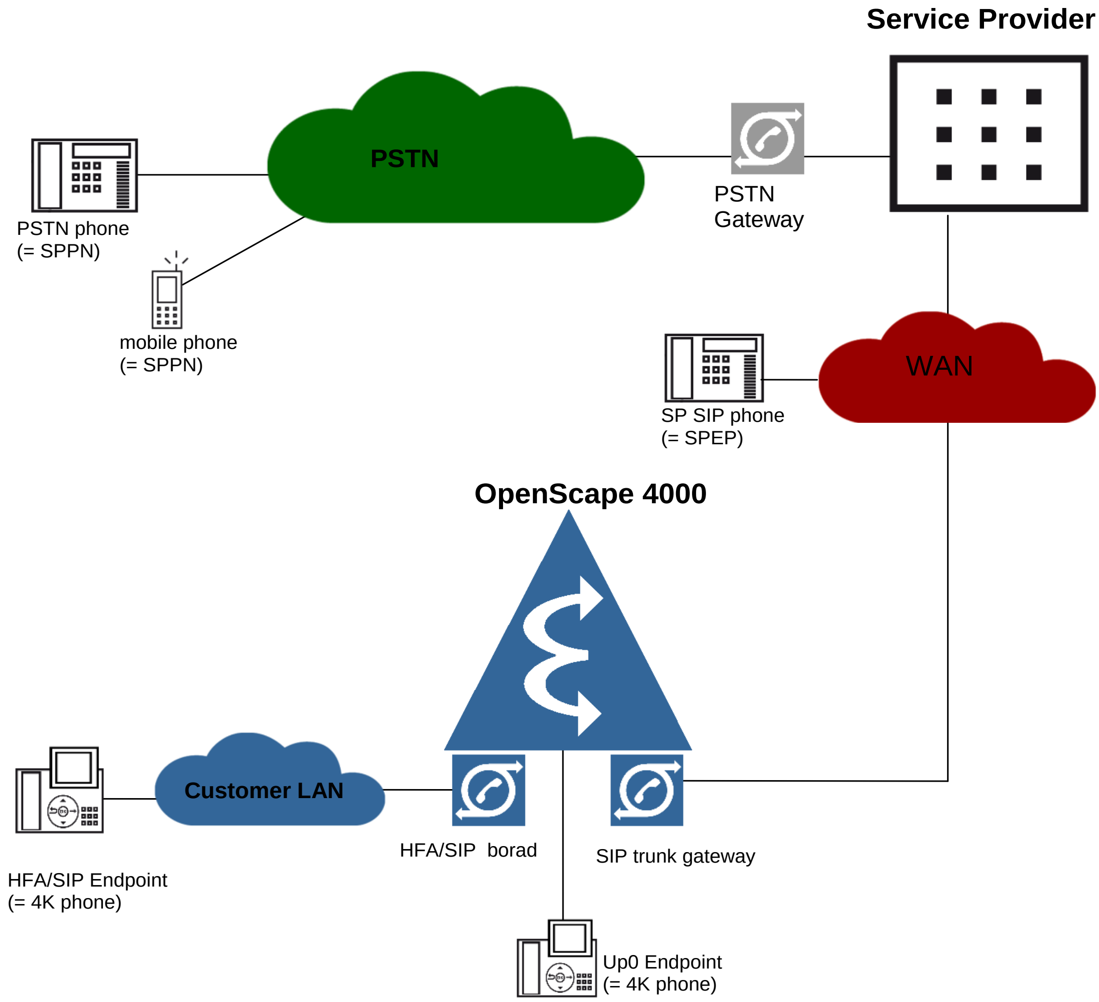
- Test Requirements and Configurations
    - Service Provider Questionnaire
    - Configuring the OpenScape 4000
    - Test Configuration
- Test Procedure
- Test Case Description
- Test Result List

2. Test Requirements and Configurations

2.1 Using CGW as SIP trunk gateway

The test topology consist of a OpenScape 4000 system (OS4K) with a HG3500 SIP trunk gateway (CGW). A set of OpenScape 4000 phones, SIP/HFA/Up0 phones and a fax are connected to the OpenScape 4000.

The native SIP gateway provides connectivity via the WAN to the SIP service provider. The service provider may have a connection to the public PSTN network and to SIP phones (SPEP) which register directly to the service provider.

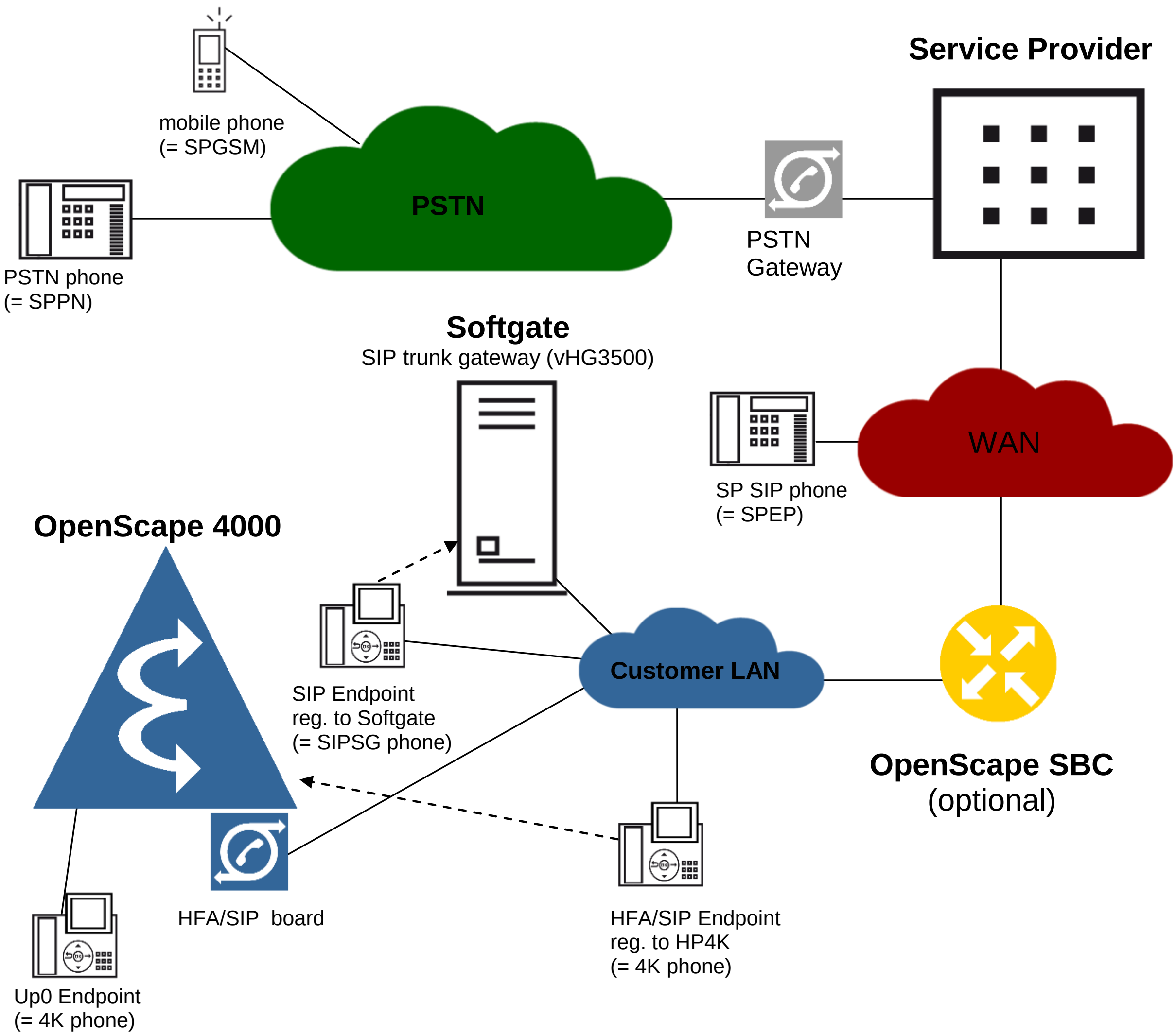

Service Provider
PSTN
PSTN phone
(= SPPN)
mobile phone
(= SPPN)
PSTN
Gateway
WAN
SP SIP phone
(= SPEP)
OpenScape 4000
Customer LAN
HFA/SIP borad
SIP trunk gateway
HFA/SIP Endpoint
(= 4K phone)
Up0 Endpoint
(= 4K phone)

2.2 Using Softgate with vHG3500 as SIP trunk gateway

The test topology consist of a OpenScape 4000 system (OS4K) with a virtual HG3500 SIP trunk gateway (Softgate). A set of OpenScape 4000 phones, SIP/ HFA/Up0 phones and a fax are connected to the OpenScape 4000 or Softgate.

The native SIP trunk gateway of the Softgate (vHG3500) provides connectivity to the SIP service provider. The service provider may have a connection to the public PSTN network and to SIP phones (SPEP) which register directly to the service provider.

With enabled DMC direct payload is tested with SIP endpoints registered to Softgate. The use of an SBC is recommened.



2.3 Service Provider Questionnaire

The Service Provider Questionnaire is provided as separate excel list. Please fill it out. This questionnaire intends to identify the basic requirements for interconnecting a OpenScape 4000 with a Service Provider. The answers are the basis for further detailed interoperability investigations.

2.4 Test Result List

The test result list is provided as separate excel list.

2.5 Configuration Overview

Via the SIP trunk gateway the OpenScape 4000 is connected to the service provider using DNS or fixed IP addresses.

Be aware that the IP address range of the OpenScape 4000 system (Assistant, HFA and SIP phones) are located in the customer LAN and the SIP trunk gateway IP address has an IP address which is routable through the internet. Appropriate protection (Firewall) is necessary! For the WBM administration it is necessary to have an https connection to the SIP trunk gateway.

NOTE:

- The SIP transport protocol and security mode are editable. If TLS is supported besides TCP and/or UDP the main tests should be executed with TLS. If TCP and UDP are supported the main test should be executed with TCP (see chapter 5.4.4 SIP Transport Protocol and chapter 5.4.5 SIP Trunk Security Mode).
- Supported Numbering Plans:
  - Explicit format: Digitstring with leading '+' specifying an international number, e.g. +498972233333. This is mapped to NPI= ISDN, TON= INTERNATIONAL (explicit format).
  - Implicit format: Pure digitstring, which is mapped to NPI= ISDN, TON= UNKNOWN. Prefixes might be neccessary to distinguish between extension, national or international numbers, e.g. 004989722333333 for international.

The preferred numbering plan from our side is the explicit format with NPI= ISDN, TON= INTERNATIONAL. Implicit numbering plan may cause more configuration depending on in which format the SIP service provider expects the called and calling party.

- CLIP/COLP, Session Timer

  All involved party have CLIP and COLP configured and the session timer of the SIP trunk gateway is on (default value), if not indicated differently.
- DMC

  DMC is always on (default value).
- Display Name

  Configure a display name for each used subscriber. Every name should include at least one special character like ä,á,æ,….
- NAT traversal, STUN and PPPoE to service provider are not supported by the SIP trunk gateway.

3. Testing

3.1 Test Procedure

The following steps should be performed BEFORE starting the tests:

- Verify the Service Provider Questionnaire has been filled out
- Verify the OpenScape 4000 has a service provider connectivity with an activated account with the correct SIP transport protocol
- Write down all used hardware components
- Write down all used software versions

- Provide an overview of the test scenario with IP addresses and telephone numbers
- Provide a complete AMO regen, add-ons and changes for SIP Provider connectivity. This should include all AMOS that are involved in configuration of trunk, LCR and maybe 4K subscriber (BFDAT, BCSU, BUEND, TDCSU, COT, COP, CGWB, GKREG, RICHT, LDPLN, LDAT, LODR, KNDEF, KNFOR, KNMAT, SBSCU, SDAT).

The following steps should be performed for ALL tests:

- Record any configuration changes required and not already described in the test case description
- A wireshark trace capture for every performed test case (named as the performed abbreviated test case) and take a note of the displays. If it is not specifically stated in the test scenario description, the traces shall include the call release. For unsuccessfull test cases, additional traces may be necessary.
- Take always attention to the following points:
  - Number display
  - ringback tone and PDD. If the PDD is longer than 5s please note.
  - Payload in both directions and speech quality
  - Cut through delay. If one or more syllable is missing, please write it down.

The following steps should be performed AFTER tests are completed:

- Please send the following to the to the SIP Service Provider Team
  - the Service Provider Questionnaire,
  - the overview of the test scenario and the AMOs,
  - the filled out test configuration and test result list,
  - collected wireshark traces.

3.2 Test Results

As test result you can chose between

- open

  This test case has not yet been executed.

- ok

  This test case has been tested successfully. If the display is not as expected please mark it, it is not a reason for the failure of the test case.

- fail

  This test case has been tested unsuccessfully. Please mark as detailed as possible the reason for the failure like: no payload, halfway payload: party B hears nothing, no music on hold, failed to put party B on hold, etc.

- not supported by SP (nsbSP)

  This test case is not supported by service provider.

- not relevant (nr)

4. Test Case Description

The test cases are named using abbreviations which are explained in the following table:

Abbreviation	
4K	CGW test topology: Phone configured in the OpenScape 4000 (TDM/Up0E, HFA, SIP subscriber) Softgate test topology: SIP subscriber phone configured at and registered to Softgate.
SP	It can be either SPEP or SPPN or both. It depends on the service provider platform. Are the interfaces different?
SPEP	External SIP phone directly connected to the service provider
SPPN	PSTN phone which is called through the service provider.In some test scenarios it can be also a fax machine or GSM phone.
REG	Registration
BC	Basic Call
HR	Hold/ Retrieve

TO	Toggle
AT	Attended Transfer
ST	Semi-Attended Transfer
BT	Blind Transfer
CFU	Call Forward Unconditional
CFB	Call Forward Busy
CFNR	Call Forward No Reply
CNF	Conference
MGP	Multiple Gateways to Provider
(rel)	This party releases the call
(hr)	This party activates hold/ retrieve
(t)	This party activates transfer
(to)	This party toggles between the other parties
(c)	This party activates conference
(cancel)	This party cancels the call, before it is established
(busy)	This party is busy
(VM)	This party has its voice mail activated
(reject)	This party rejects the call
(CLIR)	This party has CLIR activated
(COLR)	This party has COLR activated
(STon)	Session Timer is on

(SToff)	Session Timer is off

The calls are always established from left to right. For example TO_SP_4K(to)_SP: SP calls 4K, 4K makes a call back, to a second SP, which places first SP on hold. Then 4K toggles between first SP and second SP party.

Normally it does not matter which party releases the call. If it does, this is indicated by an (rel) like in BC_SPEP(rel)_4K; in this example SPEP releases the call.

Registration

There are several possibilities. Please make which one(s) is (are) supported by the tested provider:

- No registration, but a fixed IP address (= proxy IP address) is needed
- Registration without authentification
- Registration with authentification

1-1 REG_4K_SP

Activate the service provider and wait for the registration request of the gateway at the service provider.

4.1 Basic Call

Description

BC_A_B

- A calls B
- B rings
- B goes offhook
- A or B releases

2-1 BC_SPEP(rel)_4K

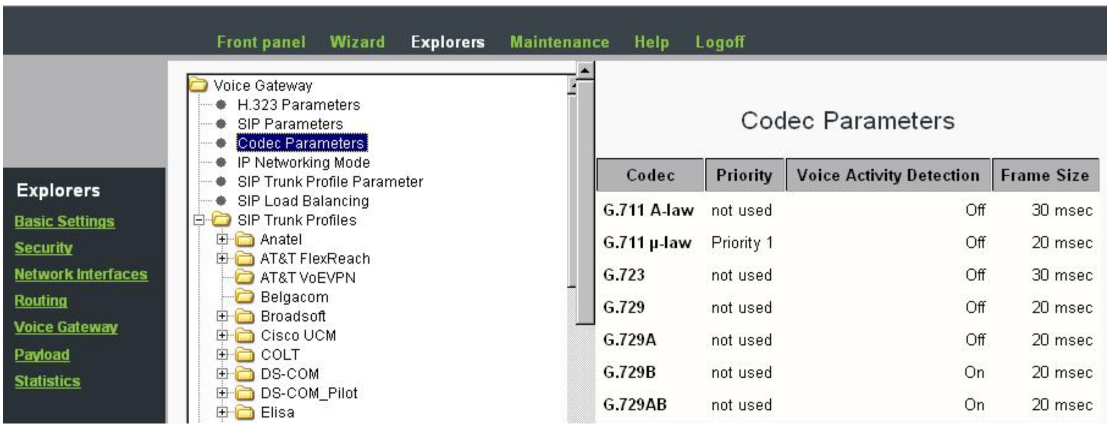
Verify correct numbers/names are displayed. Verify ringback tone and payload in both directions. Verify the call is released properly.

2-2 BC_4K(rel)_SPEP_g711pref

Similar to BC_SPEP(rel)_4K. Set codec G.711 as preferred codec (with highest priority).

SP gateway:	codec: G.711 only or G.711 with the highest priority

The codec settings for the SIP service provider gateway are done via AMO and displayed by WBM: explorer > voice gateway > codec parameters:



2-3 BC_SPPN_4K(rel)

Similar to BC_SPEP(rel)_4K.

2-4 BC_4K_SPPN(rel)_g711pref

Similar to BC_4K(rel)_SPEP_g711pref.

2-5 BC_Loop

A 4K subscriber dials the external provider number of another 4K subscriber.

Verify a **display name** has been configured in the OpenScape 4000 to both subscribers as well as CLIP and COLP.

Verify correct numbers/names are displayed. Verify ringback tone and payload in both directions. Party B releases the call. Verify the call is released properly.

2-6 BC_SPEP_4K(VM)

The activated voice mail system takes the call immediately. Keep attention to the cut through delay. Is the first word heard correctly?

2-7 BC_4K_SPEP(VM)

Similar to BC_SPEP_4K(VM).

2-8 BC_4K_SPGSM

Party B is a national GSM phone. Verify correct numbers/names are displayed. Verify ringback tone and payload in both directions. Keep attention to the PDD (Post Dial Delay)!

2-9 BC_national4K_SP_g729pref

Set codec G.729 as preferred codec (with highest priority). Party A makes a national call, using the national prefix and city code. Verify correct numbers/names are displayed.

Similar to BC_4K(rel)_SPEP_g711pref.

SP gateway:	codec: G.729 only or G.729 with the highest priority

2-10 BC_nationalSP_4K

Party A makes a national call, using the national prefix and city code. Verify correct numbers/names are displayed.

2-11 BC_international4K_SP

Party A makes an international call, using the international prefix and country code. Verify correct numbers/names are displayed.

2-12 BC_internationalSP_4K

Party A makes an international call, using the international prefix and country code. Verify correct numbers/names are displayed.

2-13 BC_ Emergency Call

Party A calls emergency number (i.e. 112 for EU or 911 for the US)

Call center answers, speech path in both directions

Call center agent has correct location information

Party A clears call

4.2 Unsuccessful Basic Calls

3-1 BC_SP_4K_noAnswer

Verify correct numbers/names are displayed.

Party B does not answer the call. Verify ringback tone. Verify the call is released after time expiry.

3-2 BC_4K_SP_noAnswer

Similar to 3-1 BC_SP_4K_noAnswer.

3-3 BC_SP(cancel)_4K

Before party B answers the call, party A releases it. Verify ringback tone. Verify the call is released properly with request CANCEL.

3-4 BC_4K(cancel)_SP

Similar to 3-3 BC_SP(cancel)_4K.

3-5 BC_SP_4K(busy)

Verify busy tone. Verify the call is released properly with the correct response (486 Busy Here).

3-6 BC_4K_SP(busy)

Similar to 3-4 BC_SP_4K(busy).

3-7 BC_4K_SP(reject)

Party B presses 'refuse call' (reject). Verify the call is released properly with the correct response (603 Decline).

3-8 BC_SP_4K_invNbr

Party A calls an invalid OS4K number. Verify the call is released properly with the correct response (404 Not Found or 484 Address incomplete).

3-9 BC_4K_SP_invNbr

Party A calls an invalid number via service provider. Verify the call is released properly with the correct response (404 Not Found).

4.3 Display Restriction (CLIR, COLR)

To configure CLIR (Calling Line Identification Presentation Restriction) set AMO SBCSU parameter SSTNO (secret station number)=yes.

4-1 BC_4K(CLIR)_SP

Party A: CLIR

Party B: COLP

Verify ringback tone and payload in both directions.

In ringing state verify display of party B shows "Anonymous" or similar.

After call has been established, verify display of party B shows "Anonymous" or similar. Verify display at party A shows number of party B.

4-2 BC_4K(CLIR)_SP_emergency

Party A: CLIR

Party B: emergency center

After call has been established, verify display of party B shows the number of party A: ask emergency center exactly which number is shown in their display and write it down.

4-3 BC_SP(CLIR)_4K

Similar to BC_4K(CLIR)_SP.

4-4 BC_SP_4K(COLR)

Party A: CLIP

Party B: COLR

Verify ringback tone and payload in both directions.

After call has been established, verify display of party A shows "Anonymous" or similar. Verify display party B shows number of party A.

4-5 BC_4K_SP(COLR)

Similar to BC_SP_4K(COLR).

4.4 Session Monitoring

Session monitoring is provider dependent. It may be not supported by the SIP service provider.

If supported it can be signaled with reINVITEs or UPDATE requests and one part plays the active and one the passive role.

If supported by SP the "Session-Expires" time and so the call duration can be decreased. Verify the "Session-Expires" time is accepted by SP:

```
>>> INVITE
        Message Header
            Min-SE: 90
            Session-Expires: 90
<<< 422 Session Interval Too Small
        Message Header
            Min-SE: 180
>>> INVITE
        Message Header
            Min-SE: 180
            Session-Expires: 180
<<< 200 OK
        Message Header
            Session-Expires: 180;refresher=uac
```

BC_4K(STon)_SP_1h+

Session Timer: OS4K=on, Sessions-Expires: 1800 sec (30 min) or less

The call remains active for 1h or longer (session timer dependent). Twice a refresh from the active part must be send.

5-2 BC_SP_4K(STon)_1h+

Similar to BC_4K(STon)_SP_1h+.

4.5 Hold / Retrieve

Description

HR_A_B

- A calls B, basic call has been successfully established
- If A(hr): A puts B on hold (A: hold party, B: held party)
  - B receives MOH (Music on Hold)
  - A retrieves B
  - A or B releases the call
- If B(hr): B puts A on hold (B: hold party, A: held party)
  - A receives MOH (Music on Hold)
  - B retrieves A
- A or B releases the call

6-1 HR_4K(hr)_SP(hr)

Verify MOH is heard by held party. Is the held status displayed? Write down exactly what is shown in the display of party A and B.

Verify the payload in both directions after party B has been retrieved from hold and the display is updated.

6-2 HR_SP(hr)_4K(hr)

Similar to HR_4K(hr)_SP(hr).

4.6 Toggle

Description

TO_A_B_C

Basic call between A and B has been successfully established

- Consultation call is made to C, which puts the other party on hold

  Or in case of call waiting:

  C calls B. When B accepts the call, A is automatically put on hold

- The toggling party toggles **five times** between the other two. One is always put on hold and receives MOH.

7-1 TO_SP_4K(to)_SP

Verify MOH is heard by held party and the hold/ held status is displayed correctly.

After the held party has been retrieved, verify the payload in both directions and if the displays are updated correctly.

7-2 TO_SP(to)_4K_SP

Similar to TO_SP_4K(to)_SP.

4.7 Call Forward Unconditional

Description

CFU A_B_C

- A calls B
- B has configured call forward unconditional
- C rings
- C goes offhook
- A and C are connected
- A or C releases the call

4.8 8-1 CFU_4K_SP_4K

Write down what is shown in the display of party A and C. Verify ringback tone and payload in both directions.

8-2 CFU_SP_4K_SP

Similar to CFU_4K_SP_4K.

8-3 CFU_SP(CLIR)_4K_SP

Similar to CFU_4K_SP_4K.

4.9 Call Forward On Busy

Description

CFB_A_B_C

- A calls B
- B is busy (B has configured forward on busy)
- C rings
- C goes offhook
- A and C are connected
- A or C releases the call

9-1 CFB_SP_SP_4K

Write down exactly what is shown in the display of party A and C. Verify ringback tone and payload in both directions. Keep attention to the PDD.

9-2 CFB_4K_4K_SP

Similar to CFB_SP_SP_4K.

4.10 Call Forward On No Reply

<u>Description</u>

CFNR A_B_C

- A calls B
- B rings (B has configured forward on no reply)
- call forward timer of C expires
- C rings
- C goes offhook
- A and C are connected
- A or C releases the call

10-1 CFNR_4K_SP_4K

Write down exactly what is shown in the display of party A and C. Verify ringback tone and payload in both directions. Keep attention to the PDD.

10-2 CFNR_SPPN_4K_SPPN

Similar to CFNR_4K_SP_4K.

10-3 CFNR_SPPN_4K_SPEP

Similar to CFNR_4K_SP_4K.

10-4 CFNR_SPEP_4K_SPPN

Similar to CFNR_4K_SP_4K.

10-5 CFNR_SPEP_4K_SPEP

Similar to CFNR_4K_SP_4K.

10-6 CFNR_4K_4K_SP

Similar to CFNR_4K_SP_4K.

4.11 Conference

<u>Description</u>

CNF_A_B_C

- A basic call is established between A and B
- B estabilishes a consulting call to C
- A is automatically put on hold
- B makes a three party conference
- A, B and C are connected
- A, B or C releases the call

11-1 CNF_SP_4K(c-rel)_SP

Write down exactly what is shown in the displays of all parties. Verify speech path and quality in all directions.

After the release of a party are the left parties still connected? What does the display show?

4.12 Attended Transfer

The OpenScape 4000 executes the transfer itself and does not send any REFERs. ReInvites are sent to update the display with help of the P-Asserted-Identity header field.

The OpenScape 4000 does not accept REFER requests from any service provider.

The display is updated using the P-Asserted-Identity header field. This may not be supported by the service provider.

Description

AT_A_B_C

- A calls B, basic call has been successfully established
- If A(t): A estabilishes a consulting call to C
  - B is automatically put on hold
  - A joins the calls
  - The call to A is automatically released
  - B and C are connected
  - B or C releases the call
- If B(t): B estabilishes a consulting call to C
  - A is automatically put on hold
  - B joins the calls
  - The call to B is automatically released
  - A and C are connected
  - A or C releases the call

12-1 AT_SP_4K(t)_SP

Verify correct numbers/names are displayed. After transfer verify payload in both directions. Write down exactly what is shown in the display of all parties after the transfer. Has the display been updated after the transfer?

12-2 AT_4K(t)_SP_4K

Similar to AT_SP_4K(t)_SP.

12-3 AT_4K_SP(t)_4K

Verify correct numbers/names are displayed. After transfer verify payload in both directions. Write down exactly what is shown in the display of all parties after the transfer. Has the display been updated after the transfer?

Verify a transfer signaled by SP with REFER it is rejected. Select as result "not supported by SP". A transfer signaled by SP with reINVITEs is accepted.

4.13 Semi-Attended Transfer

The OpenScape 4000 executes the transfer itself and does not send any REFERs. ReInvites are sent to update the display with help of the P-Asserted-Identity header field.

The display is updated using the P-Asserted-Identity header field. This may not be supported by the service provider.

Description

ST_A_B_C

- A calls B, basic call has been successfully established
- If A(t): A estabilishes a consulting call to C
    - B is automatically put on hold
    - C is ringing
    - A hangs up
    - C goes offhook
    - B and C are connected
    - B or C releases the call
- If B(t): B estabilishes a consulting call to C
    - A is automatically put on hold
    - C is ringing
    - B hangs up
    - C goes offhook
    - A and C are connected
    - A or C releases the call

13-1 ST_SP(t)_4K_4K

Verify correct numbers/names are displayed. Verify ringback tone and payload in both directions. Write down exactly what is shown in the display of all parties after the transfer. Has the display been updated after the transfer?

13-2 ST_SP_4K(t)_SP

Similar to ST_SP_4K(t)_4K.

13-3 ST_4K(t)_4K_SP

Similar to ST_SP_4K(t)_4K.

4.14 Blind Transfer

Blind transfer is not executable from OpenScape4000 side.

The display is updated using the P-Asserted-Identity header field. This may not be supported by the service provider.

Description

BT_A_B_C

- A calls B, basic call has been successfully established
- If A(t): A transfers the call to C
  - The call to A is automatically released
  - B and C are connected
  - B or C releases the call
- If B(t): B transfers the call to C
  - The call to A is automatically released
  - A and C are connected
  - A or C releases the call

14-1 BT_SPEP(t)_4K_SP

Verify correct numbers/names are displayed. Verify ringback tone and payload in both directions. Write down exactly what is shown in the display of all parties after the transfer. Has the display been updated after the transfer?

If the transfer is signaled by SP with a REFER, it is rejected. Select as result "not supported by SP".

A transfer signaled by SP with reINVITEs is accepted.

4.15 DTMF

Description

DTMF_A_B

- A calls B
- B rings
- B goes offhook
- Party A or B presses digits
- A or B releases

15-1 DTMF_SP_4K_RFC2833/4733

Verify that DTMF is sent as own payload type.

Use wireshark filter: “rtpevent” to identify the DTMF packets:

Real-Time Transport Protocol

RFC 2833 RTP Event

Event ID: DTMF One 1 (1)

15-2 DTMF_4K_SP_RFC2833/4733

Similar to DTMF_ SP_4K_RFC2833/4733.

15-3 DTMF_SP_4K_inband

Codec settings:

Party A: Codec: g711

Party B: Codec: g711

Verify that DTMF is sent inband by calling a telephone service or mailbox system.

15-4 DTMF_4K_SP_inband

Similar to DTMF_SP_4K_inband.

4.16 FAX

- Verify that the fax devices support T.38 fax. To be able to switch to T.38 the gateway has to receive CNG and CED tones
- Each test case should be repeated four (17-1 to 17-4), respectively six (T.38 test cases: 17-5 to 17-8) times.
- For each test case 10 fax pages should be sent in order to have a session timer refresh during the connection.
- Set value of "Session timer expires" to 90 sec. Verify that this time has been exepted. Otherwise increase the number of fax pages.

```
>>> INVITE
    Message Header
        Min-SE: 90
        Session-Expires: 90
<<< 422 Session Interval Too Small
    Message Header
        Min-SE: 180
>>> INVITE
    Message Header
        Min-SE: 180
        Session-Expires: 180
<<< 200 OK
    Message Header
        Session-Expires: 180;refresher=uac
```

Description

FAX_A_B

- A calls B
- B answers automatically and starts sending the fax pages
- A or B releases

16-1 FAX_SP_4K _RFC2833/4733g711

Disable in AMO CGWB T38.

GW settings:	G711 as preferred codec RFC2833/4733: on

> **RFC2833/4733 is offered, but if the partner does not support it, the fax tones will be sent inband and fax is sent with g711. Please just note that RFC2833/4733 is not supported, but select ok as test result, if the fax has been sent successfully.**

Verify that several pages are sent within g711 stream. Maybe own payload type (RFC2833/4733) is used:

Use wireshark filter: "rtpevent" to identify the RFC2833/4733 packets:

Real-Time Transport Protocol

RFC 2833 RTP Event

Event ID: ….

16-2 FAX_4K_SP_RFC2833/4733g711

Similar to FAX_SP_4K_RFC2833/4733g711.

16-3 FAX_SP_4K_g729fallbackg711

Disable in AMO CGWB T38.

GW settings:	G729 as preferred codec RFC2833/4733: on

Verify the fallback to g711 is done properly and several pages are sent within g711 stream. This is only possible if both sides have g711 as additional codec set. In case only g729 is provided the call is released by OS4K.

16-4 FAX_4K_SP_g729fallbackg711

Similar to FAX_SP_4K_g729fallbackg711.

16-5 FAXT38_SPEP_4K

Enable in AMO CGWB T38.

GW settings:	G729 as preferred codec Enable T38 fax

Device setting:

- 4K: low speed

Verify that pages are sent within T.38 stream. Use wireshark filter: “t38” to identify the T.38 packets:

```
No.     Time     Source          Destination      Protocol    Info
1230 41.087894  218.1.53.171    218.1.53.176     T.38        UDP: UDPTLPacket Seq=00000  t30ind: ced
ITU-T Recommendation T.38
    [Stream setup by SDP (frame 1213)]
        …
```

16-6 FAXT38_4K_SPEP

Similar to FAXT38_SP_4Klow.

16-7 FAXT38_4K_SPPN

Similar to FAXT38_SP_4Klow, but with different fax device settings:

4K: high speed

16-8 FAXT38_SPPN_4K

Similar to FAXT38_4Khigh_SP.

4.17 Message Waiting Indication

OpenScape 4000 supports MWI as consumer without SUBSCRIBE.

A NOTIFY is usually sent when a voice message is left in a voice mailbox. Configure a voice mailbox for the 4K subscriber at the SIP service provider, if supported.

Description

MWI_A_B

- A sends B a NOTIFY message

17-1 MWI_SP_4K on

A NOTIFY is sent to 4K subscriber with the following message body:

Message body

Messages-Waiting: yes\r\n

Voice-Message: 1/0\r\n

Verify that the message waiting indication light of 4K subscriber turns on and the NOTIFY is answered with a 200 OK.

17-2 MWI_SP_4K off

A NOTIFY is sent to 4K subscriber with the following message body:

Message body

Messages-Waiting: no\r\n

\r\n

Verify that the message waiting indication light of 4K subscriber turns off and the NOTIFY is answered with a 200 OK.

5. Supported Transport Protocols

The SIP transport protocol is configurable if UDP and TCP are supported by the SIP Service Provider. If TCP and UDP are supported all test cases must be executed with TCP. With the following additional tests cases the functionality of the UDP is ensured.

Modify the configuration and repeat the test cases 19-x1, 19-x2 and 19-x3.

18-x1 TPROT_REG_4K_SP

RESTRICTION: CGW does not support SDES!

Activate the service provider and wait for the registration request of the gateway at the service provider.

18-x2 TPROT_BC_4K_SP

RESTRICTION: CGW does not support SDES!

Verify ringback tone and payload in both directions.

18-x3 TPROT_BC_SP_4K

RESTRICTION: CGW does not support SDES!

Verify ringback tone and payload in both directions.

6. Appendix: OpenScape 4000 Configuration

The OpenScape 4000 configuration guidelines are described under the assumption that enough Comscendo Licenses for the SIP Trunk ports are available and a slot for the STMI2/4-Board is reserved in the OpenScape 4000 System.

The configuration covers specific SIP provider specific data.

6.1 TRUNK configuration

AMO TDCSU: configure tdcsu with DEV=HG3550CO for OpenScape 4000

Besides BFDAT, BCSU, BUEND TDCSU (ECMAV2 protocol) COT and COP is of interest, see following example:

```
ADD-COT:COTNO=80,PAR=ANS&CEBC&BSHT&BLOC&LWNC&NLCR&TSCS&DFNN&NLRD&NOFT&NTON;
ADD-COP:COPNO=80,TRK=TA,TOLL=TA;
```

Routing and numbering plan specific issues see below (TDCSU,COT,COP).

6.2 GW Board configuration

HG3500 supports DNS/SRV and uses SIP trunking profiles , though all provider specific data has to be configured over WBM (see next chapter HG3540 Configuration)

AMO CGWB (in the following example with IPADR=212.202.240.122) is used to configure, the used trunk protocol (Native SIP), DNS/SRV server (used by profile) and the codec list (example has VAD set to yes, maybe the provider doesn’t support VAD), etc.

The GW is configured as local GW w/ GKREG only for INTGW w/o any provider specific data (there is no difference at AMO CGWB/LDAT configuration regarding NON-registration and registration)

Example of service provider configuration:

```
ADD-CGWB:LTU=1,SLOT=8,SMODE=NORMAL,IPADR=212.202.240.122,NETMASK=255.255.255.248;
CHANGE-CGWB:MTYPE=CGW,LTU=1,SLOT=8,TYPE=GLOBIF, DEFRT=212.202.240.121,TRPRSIP=60,
TLSP=4061,DNSIPADR=212.202.240.121;
CHANGE-CGWB:MTYPE=CGW,LTU=1,SLOT=8,TYPE=ASC,REDRFCTN=NO;
CHANGE-CGWB:MTYPE=CGW,LTU=1,SLOT=8,TYPE=LEGKDATA,GWNO=11;
ADD-GKREG:GWNO=11,GWATTR=INTGW&HG3550V2&SIP,DIPLNUM=0,DPLN=0,LAUTH=1,INFO="LOCAL GW";
ADD-LDAT:LROUTE=300,LSVC=ALL,LVAL=1,TGRP=1,ODR=300,LAUTH=1;
```

6.3 Routing and numbering plan configuration

For outgoing calls block dialing has to be configured for SIP provider trunk, dialing pattern has to end with e.g. '-Z':

```
ADD-LDPLN:LCRCONF=LCRPATT,DIPLNUM=0,LDP="0"-"Z",LROUTE=1,LAUTH=1;
```

LCR configuration rules depend on the numbering plan, that is used between OS4K and SIP service provider.

The numbers over SIP can be sent in two different ways:

- Digitstring with leading '+' specifying an international number, e.g. +498972233333. This is mapped to NPI= ISDN, TON= INTERNATIONAL (explicit format).
- Pure digitstring, which is mapped to NPI= ISDN, TON= UNKNOWN. Prefixes might be neccessary to distinguish between extension, national or international numbers, e.g.00498972233333 for international.

The preferred numbering plan from our side is NPI= ISDN, TON= INTERNATIONAL (explicit format) Implicit numbering plan may cause more configuration depending on in which format the SIP service provider expects the called and calling party.

AMO hints:

AMO-COT: Parameter LINC is added to define calling number as an implicit ISDN number without access codes.

AMO-COP: Parameter IMEX is added to define called number as an implicit ISDN number.

AMO-KNFOR: specifies calling party modification (national or international).

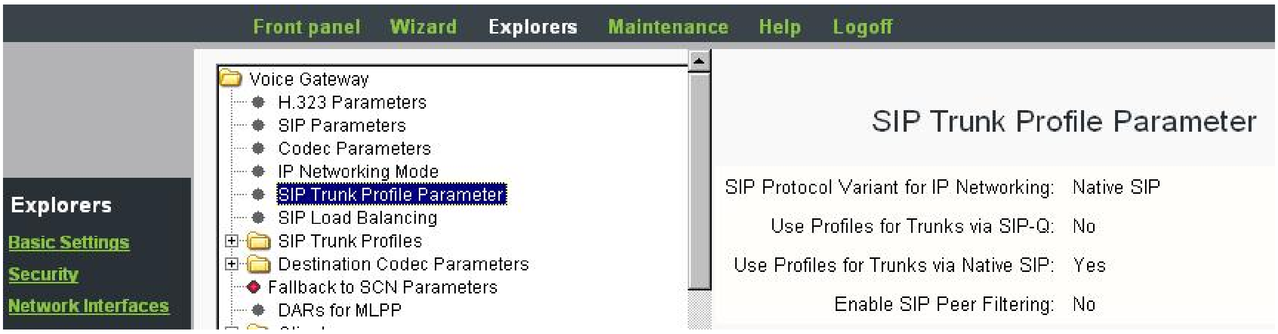
AMO-TDCSU: ISDNIP=00,ISDNNP=0 specifies national and international prefix for implicit ISDN number for CO trunking.

AMO-SDAT: Be aware of the used numbering plan when using PUBNUM.

6.4 vHG3500/CGW Configuration

5.4.1. SIP Protocol Variant

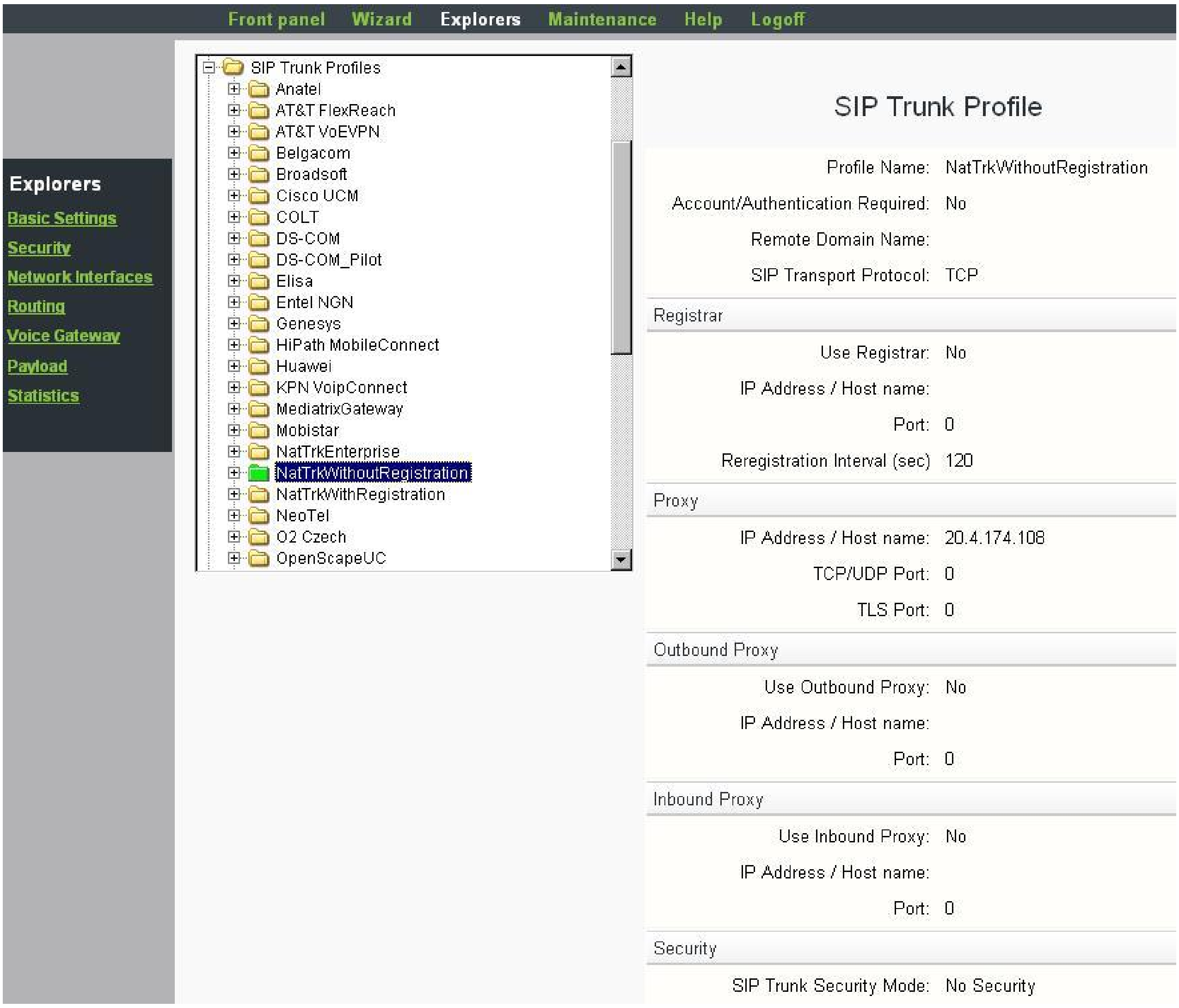
Verifiy that 'SIP Protocol Variant for IP Networking' is set to "Native SIP" (AMO CGWB) and 'Use Profiles for Trunks via Native SIP' flag is set (profile usage is activated) on the following screen:



5.4.2. SIP Trunk Profile

The offered NatTrkWithoutRegistration profile can be used for provider configuration without registration. The provider IP Address/Host name (e.g. IP =1.2.3.4) has to be configured under Proxy and the profile has to be activated (shown in green).

The following screen shot shows an active profile w/o registration using IP address/port.

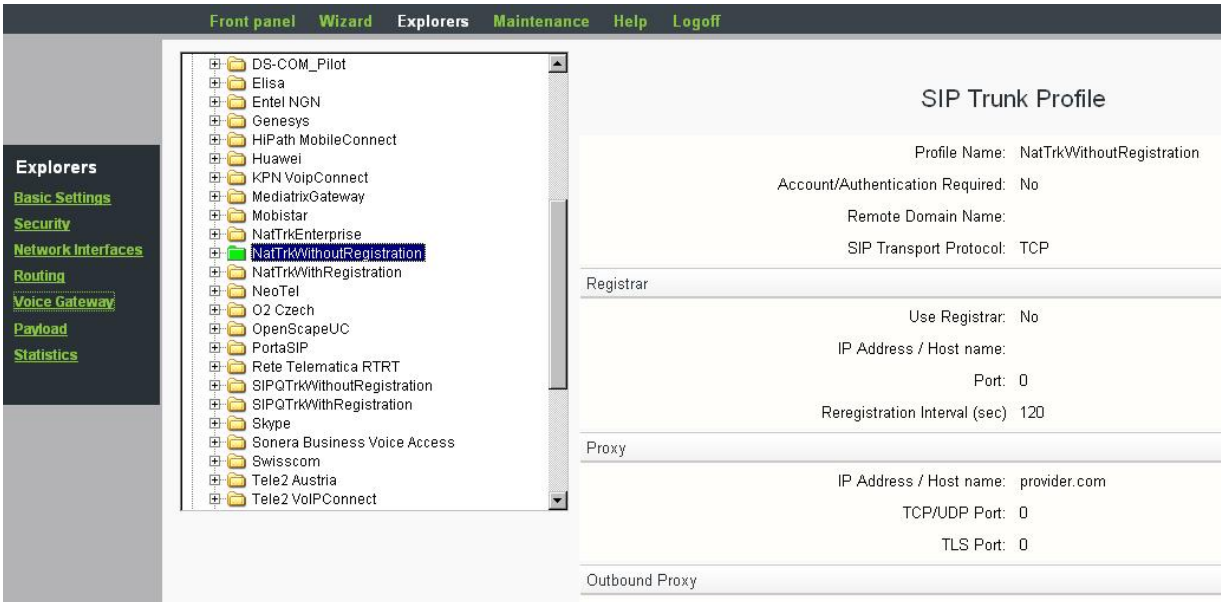


The offered NatTrkWithRegistration profile can be used for provider configuration with registration. The account used for registration and if required authentication data are configured as shown in the following screen shot: (‘Account/Authentication required’ has to be enabled (default, see next screen shot).

5.4.3. Proxy

After selecting the suitable profile the provider IP Address/Host name (e.g. DNS name = provider.com) has to be configured under Registrar and under Proxy. Domain Name has normally the same value as IP address/Host name of registrar or proxy and is used as host part of URI in SIP messages.

The following screen shot shows an active profile with registration using Host name.

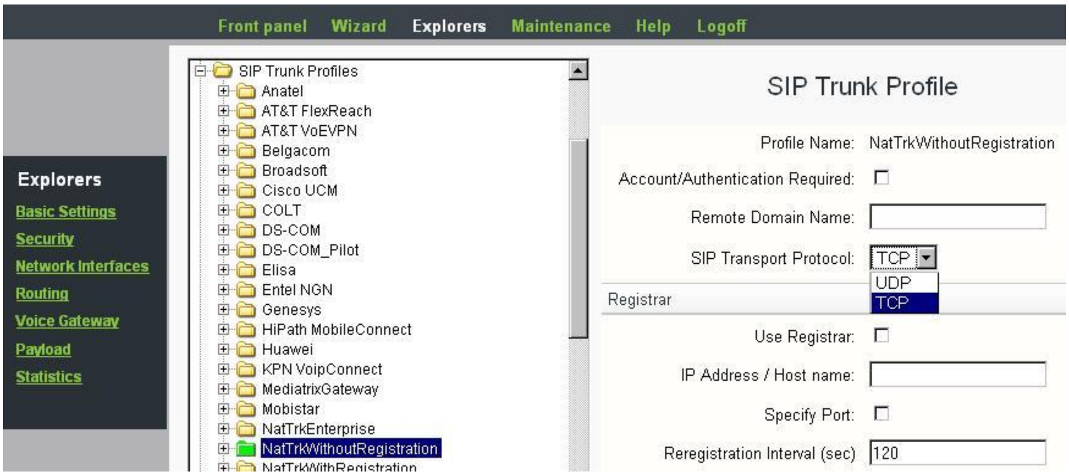


Only one profile can be activated. Outbound Proxy has to be configured when SBC e.g. OpenScape SBC (LAN IP) is used for LAN/WAN Nat traversal.

5.4.4. SIP Transport Protocol

The SIP transport protocol and security mode are editable. If TLS is supported besides TCP and/or UDP the main tests should to be executed with TLS. If TCP and UDP are supported the main test should be executed with TCP (see chapter 5 Supported Transport Protocols).

Modify the SIP transport protocol: WBM > Explorers > Voice gateway > SIP Trunk Profiles > select a native SIP trunk partner > right mouse button: edit



5.4.5. SIP Trunk Security Mode

Signaling encryption

OpenScape 4000 supports TLS. Either the server-provided TLS certificates model is supported as well as the Mutual TLS certificates model (MTLS). For the server-provided TLS certificates model it is necessary that the gateway registers by the service provider.

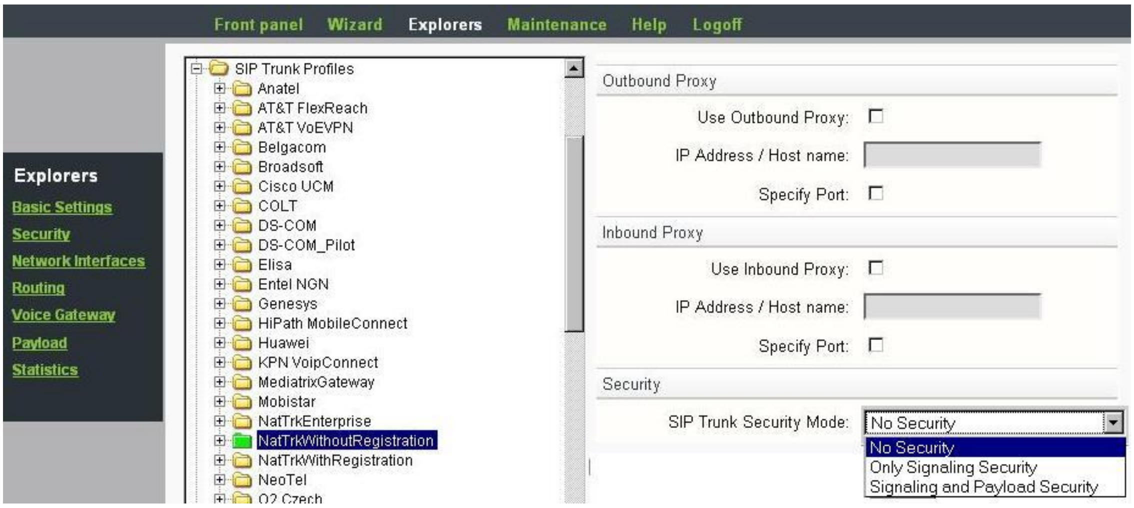
Payload encryption

vHG3500 supports SDES, but not CGW!

NOTE:

The pulldown menu offers security options only if these have been released for the native SIP trunk partner.

Modify the SIP trunk security mode: WBM > Explorers > Voice gateway > SIP Trunk Profiles > select a native SIP trunk partner > right mouse button: edit



7. Abbreviations and Terms

CGW	Common Gateway (running on STMI board)
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Presentation Restriction
COLP	Connected Line Presentation
COLR	Connected Line Restriction
DTMF	Dual Tone Multiple Frequency
GSM	Global System for Mobile Communications
GW	Gateway
MOH	Music On Hold
MWI	Message Waiting Indication
OS4K	OpenScape 4000
PDD	Post Dial Delay (time after dialling until ringing tone is heard)
SP	Service Provider
VAD	Voice Activity Detection (silence suppression)
VM	Voicemail
WBM	Web Based Management