SIP Business Trunking Specification
Version 3.0

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1. SIP Business Trunking Specification for SIP Systems

This document describes the subscriber-side interface for SIP end devices with PBX functionality. The SIP trunk can be used to directly reach extensions (DDI) and to signal outgoing calls to extensions and alternative numbers.

2. Overview

The trunk is provided by a highly available SIP server platform that authenticates and authorises calls to the PSTN and other SIP subscribers. All PSTN numbers are also accessible, limited to voice services and selected ISDN data services. Additional features are available depending on the version of the SIP trunk.

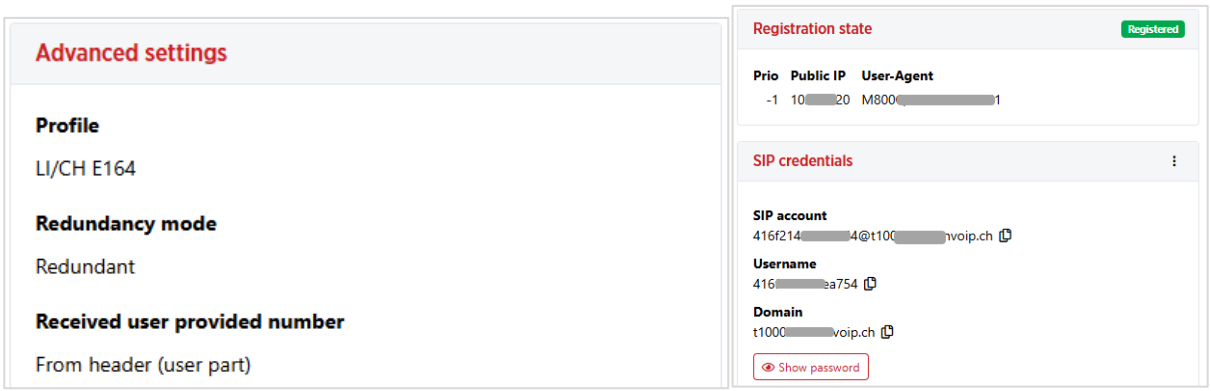
The Business Trunk (BT) described here enables

- multiple number ranges or individual numbers on a Business Trunk
- Redundancy and load balancing through multiple registrations
- Configurable settings such as call number formats for incoming and outgoing calls
- Predefined, expandable profiles (currently three)

2.1. Customer web portal

FL1 Trunk customers can access the customer web portal at https://commplus.fl1.li. There, they can independently activate/deactivate features such as fallback forwarding or change the email address for the fax box.

In addition, the SIP credentials (user name, password, SIP domain) for the SIP trunk (settings on the PBX at the customer's premises) and the activated profile, as well as other technical settings made by Telecom Liechtenstein, can be viewed in the customer web portal.



Notes:

- Please contact Telecom Liechtenstein customer support for any changes to these specifications.
- You have received the access data for the customer web portal in writing from Telecom Liechtenstein.

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2.2. Technical requirements for customer infrastructure

The customer firewall and all other network infrastructure components involved (LAN switches, routers, WLAN components, etc.) should transfer all IP traffic for telephony (UDP + TCP and all protocols) from the internal customer network (LAN) to VoIP infrastructure of Telecom Liechtenstein (WAN). The IP subnet (C-Class) of Telecom Liechtenstein is:

80.66.238.0/24 (TCP + UDP on all protocols/ports).

This IP subnet is used by Telecom Liechtenstein for all current and future VoIP services. By releasing the entire IP subnet, the customer infrastructure is prepared for future expansions and it is also ensured that all IP addresses of the geo-redundant VoIP infrastructure of Telecom Liechtenstein can be reached.

If security restrictions on the customer side do not allow the activation of all IP protocols/ports, the protocols and ports used are listed below.

IP address / net mask	Protokoll:Port	Description
80.66.238.0/24	UDP:5083 (SIP) TCP:5063 (SIPs, TLS) UDP:10000-65535 (RTP und sRTP)	SIP-Outbound-Proxy (SIP signalling) SIP-Outbound-Proxy (SIP signalling secure) SIP-Outbound-Proxy (SIP media-ports)
	TCP:443 (HTTPS)	access to Trunk administration GUI (planned)

However, if any impairments occur, the Telecom Liechtenstein rule listed above must install in order to exclude problems on the firewall or other components.

If the Internet connection comes from Telecom Liechtenstein, the following IP addresses and ports must be enabled for the DNS and NTP services.

DNS name (URL)	IP address	Protocoll:Port	Description
	217.173.235.71 217.173.235.72 217.173.235.73	UDP:53 (DNS)	DNS-server
ntp1.telecom.li ntp2.telecom.li	80.66.224.2 80.66.224.10	UDP:123 (NTP)	NTP-server

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2.3. SIP outbound proxy configuration for settings on the PBX

The following parameters are relevant for the settings on your SIP system (PBX). These parameters apply to FL1 trunk in Liechtenstein (+423) and in Switzerland (+41).

Single registration	
Automatic assignment via SRV lookup [1)]	
SIP outbound proxy:	sbc5-trunk.ipbx.li
Manual configuration	
SIP outbound proxy:	sbc5.ipbx.li
SIP outbound proxy (signalling):	UDP:5083 (SIP)
SIP outbound proxy (secure signalling):	TCP:5063 (SIP, TLS)
SIP outbound proxy (media ports for both cases):	UDP:10000-65535 (RTP and sRTP)

Redundancy/load balancing – 2nd registration	
Automatic assignment via SRV lookup [1]	
SIP outbound proxy 1st registration:	sbc5-trunk.ipbx.li
SIP outbound proxy 2nd registration:	sbc6-trunk.ipbx.li
Manual configuration [2)]	
SIP outbound proxy 1st registration:	sbc5.ipbx.li
SIP outbound proxy 2nd registration:	sbc6.ipbx.li
SIP outbound proxy (signalling):	UDP:5083 (SIP)
SIP outbound proxy (secure signalling):	TCP:5063 (SIP, TLS)
SIP outbound proxy (media ports for both cases):	UDP:10000-65535 (RTP and sRTP)

1) Automatic assignment via SRV lookup must be supported by the PBX, otherwise manual configuration must be used. The advantage of an SRV lookup is that the geo-redundancy provided by FL1 can be fully utilised with a single registration of the VoIP systems. The protocol and port are automatically determined via DNS by the "Transport" (UDP/TLS) setting on the end device. TCP without TLS is no longer offered.

   Note: sbc5-trunk.ipbx.li and sbc6-trunk.ipbx.li do not have a DNS A record!

2) Manual configuration is recommended for redundant registration.

3. SIP Business Trunk variants and differences

There is no standardised method for transmitting information in SIP headers. However, best practices have resulted in three variants that are sufficient in most cases.

The URI scheme in all headers is "sip:". The "tel:" URI scheme is not supported.

The SIP header From, To and the Request URI must always contain the trunk domain in the domain part. The headers P-Preferred-Identity and P-Asserted-Identity should always contain the trunk domain in the domain part.

Important: The **use of IP addresses in the domain parts** is **not permitted** and will no longer be supported in future (**rejected**).

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3.1. Profile "LI/CH E164" (standard profile)

With this profile, the end device must transmit the subscriber number in the P-Preferred-Identity header and in the From header. The subscriber number must be in E.164 format (e.g. **+4233000710** or 004233000710).

Alternatively, the P-Preferred Identity Header can also contain the Authentication User.

A user-provided number ("CLIP No-Screening") can optionally be sent by the terminal device in the From header in E.164 format (e.g. **+4233000710** or 004233000710).

The phone numbers are sent to the end device in international format:

- R-URI and To User with, for example, "+41715000590" (full number including number range)
- From User with, for example, "+4232397409"

If the end device cannot use the P-Preferred-Identity header, the P-Asserted-Identity header can be used as an alternative. The use of both headers is not permitted.

Note: The aim of this profile is to always transmit all phone numbers in the +E.164 format (e.g. "+4237910001").

3.2. Profile "LI National" (Liechtenstein +423 only)

With this profile, the terminal must transmit the subscriber number in the From header. The subscriber number must be in E.164 format (e.g.+ 4233000710 or 004233000710 or, alternatively, nationally as 3000710). The From header can also contain the SIP Authentication User.

A user-provided number ("CLIP No-Screening") can optionally be sent by the end device in the P-Preferred-Identity header (fallback: in the P-Asserted-Identity header) in E.164 format.

The phone numbers are sent to the end device in **national** format:

- R-URI and To User with, for example, "3000710" (full number including number range)
- From User with, for example, "2000409" (national) or "0041315000998" (international)

Note: This profile still exists for compatibility reasons and is **no longer offered or supported for new connections**. If problems occur, switch to the "LI/CH E164" profile.

3.3. Profile "CH International" (Switzerland +41 only)

With this profile, the end device must transfer the subscriber number in the From header. The subscriber number must be in E.164 format (e.g.+ 41715000590 or 0041715000590 or 41715000590). Alternatively, the From header can also contain the authentication user.

A user-provided number ("CLIP No-Screening") can optionally be sent by the end device in the P-Preferred-Identity header (fallback: in the P-Asserted-Identity header) in E.164 format.

The phone numbers are sent to the end device in international format:

- R-URI and To User with, for example, "0041715000590" (full number including number range)
- From User with, for example, "004232397409"

Note: "National" call number signalling is not provided for Swiss call numbers.

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4. Security

The following section defines the requirements for using a SIP Business Trunking Service.

4.1. Registration

Registration is mandatory in order to make and receive calls. Each SIP Business Trunk must therefore be registered; this ensures that the phone number can be traced back and is therefore required by law. Digest Authentication (MD5) with QOP is used for this purpose.

For redundancy or load balancing, a Q-value is added to the contact by the CPE. The Q-value is a value between 0 (lowest priority) and 1 (highest priority).

Example: Load balancing is achieved by registering two end devices on the same business trunk with the same Q-value:

CPE#1: Contact: <sip:2000409@192.168.41.180>**;q=0.5**

CPE#2: Contact: <sip:2000409@192.168.41.181>**;q=0.5**

In redundancy mode load balancing or failover, the "Replace registered contacts" function must be deactivated in any case!

The "Replace registered contacts" function is intended for cases where the contact information changes regularly. For example, due to UDP timeouts in NAT devices (changing port) or router restart.

4.2. Additional security measures

- Only domain names provided by the provider can be used in the domain part of the Request/To/From URI in the SIP header. IP addresses are not accepted.
- IP filter: The SIP Business Trunk user can only register with the IP address or IP network reported to Telecom Liechtenstein. Registration attempts from other IP addresses will be rejected. The stored IP addresses can be viewed in the customer portal under Settings > IP Filter.
- The authentication user (under "SIP access data") is displayed in the customer portal at https://commplus.fl1.li after login and must be used as specified.
- Encryption using TLS/SRTP (see chapter 8)

4.3. Security for originating calls

The following conditions must be met in order to make outgoing calls:

1. The user must be registered (see chapter 4.1).
2. Each call must be authenticated. Digest Authentication (MD5) with QOP is also used for this purpose. Authentication is always performed with the user account. Nonce Count and Responses must match the nonce used. The Nonce Count must be reset for each new nonce.

If all conditions are met, the call is routed.

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5. Calls, firewall, SIP, PBX

5.1. Outgoing calls (from the customer connection to the network)

Outgoing calls are always authenticated, authorised and then routed.

The following functions are provided:

- Authentication of the call (see chapter 4.3).
- Implementation (mapping) of SIP parameters such as outgoing phone number or additional phone number (CLIP no screening) into the network.
- Special handling of emergency calls.
- Fax support with T.38 (udptl) and G.711 A-Law fallback.

If a fax is detected, the PBX must respond to the codec change to T.38 in the signalling. If a change to T.38 is not possible, the PBX must reject a change request from the remote station (e.g. another SIP subscriber) with 488 and **continue the call with the previously negotiated codec**.

5.2. Terminating calls (from the network to the customer connection)

Calls from the Telecom Liechtenstein network to the SIP trunk are always routed to the registered contact. To reach a subscriber, they must first provide their IP address to the registrar via SIP REGISTER. Static routing to any fixed IP addresses is not supported.

The destination number is transmitted in the **Request URI**; the format of the To: header field is not defined. The use of the To: header for routing purposes in the customer's system is at your own risk and is not supported.

The SIP Business Trunk is preset to +E.164 (profile "LI/CH E164") but allows individual customisation of the format.

Routing to PBX supported:

- Only block dialling (en-bloc dialling)
- Geographic, nomadic and ported phone numbers.
- Fax support with T.38 (udptl) and G.711 A-Law fallback

The suggested codecs vary depending on the source of the call. However, the G.711 A-Law codec is always available as a fallback.

If a fax is detected, the PBX must **initiate** the codec change to T.38 **and signal** this if desired. If a change to T.38 is not possible, the PBX must reject a change request from the remote station (e.g. another SIP subscriber) with 488 **and continue the call with the previously negotiated codec**.

5.3. NAT / Firewall support

NAT and firewalls on the customer side are supported. However, full functionality can only be guaranteed after checking the conditions. This requires that no additional measures for NAT support are taken on the customer side. If NAT support is to function on the provider side, the device performing NAT must pass on all information in the SIP protocol unchanged.

If a SIP NAT ALG (Application Layer Gateway) is in use, this ALG must perform all functions of SIP NAT support accordingly so that the end device appears on the provider side as if it were not behind NAT.

If NAT is detected, the REGISTER interval is reduced to approx. 65 seconds on the provider side to prevent any firewalls and NAT end devices from interrupting the connection after a certain time (UDP timeout). The customer's end device can also send "keep alives".

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The user agent client is forced to adjust the register interval accordingly by means of Expires: Header in the 200 OK of the REGISTER response. **The CPE must adhere to the expires timeout specified by the server** (NAT refresh by adjusting the register interval).

5.4. IP fragmentation

The PBX and any upstream firewall or router must be able to accept and process fragmented IP packets. Out-of-order packets must also be able to be processed.

SIP messages can be larger than 1500 bytes. These are then transmitted via fragmented IP/UDP packets. A firewall or NAT device must recognise and allow the fragments accordingly.

5.5. Multiple LAN-side network segments

If several LAN segments (e.g. a separate segment for WLAN infrastructure) are connected via routers/firewalls/Layer 3 switches on the customer side, it must be ensured that all ports (especially the RTP/sRTP ports) are also interconnected between the LAN segments.

5.6. Quality of Service (QoS) functions in the LAN

There are two methods for ensuring the required quality in the LAN:

- Sufficient bandwidth: If the entire infrastructure at the customer's site is implemented with Gigabit LAN throughout, no further QoS support is required, but it is still strongly recommended.
- QoS support and clean configuration of PC clients: The LAN infrastructure and the Layer 3 switches used must trust DiffServ Code Points (DSCP) from end devices (DSCP trusted) and transfer them to Class-of-Service (COS). SIP signalling messages are marked with the DSCP value AF41. (s)RTP packets are marked with the DSCP value EF.

5.7. Unsupported functions

The following functions are currently not supported:

- Modems: All types of modem transmissions are not supported. If the G.711 A-Law codec allows it, modem transmissions can be carried out at your own risk.
- Overlap Sending: If an initial INVITE is sent, this is the final destination. The PBX is responsible for collecting the digits and sending them as a "block".
- SMS: No SMS service is currently offered.
- Super-G3 fax or fax transmission at a speed higher than 14.4 kb/s.
- RFC2543 support. All calls must be sent "loose routed"; "strict routing" is no longer supported (only RFC3261). This applies in particular to reInvites after session timers and codec changes.
- Unsupported SIP methods are REFER, UPDATE, SUBSCRIBE, NOTIFY.

5.8. Provisional responses

Provisional responses according to RFC 3262 are not supported. Provisional responses are not actively requested. See also the unsupported SIP method UPDATE.

5.9. Session timers

Session timers according to RFC 4028 are supported if they are requested during call setup. However, session timers are not actively requested.

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5.10. Private header extensions

For technical reasons, so-called private SIP headers beginning with "X-" may appear in SIP header names. The PBX must ignore these fields. "X-" headers coming from the PBX are ignored.

5.11. ISDN Clear Channel (64 kb/s unrestricted)

The ISDN service "64 kb/s Unrestricted" or the so-called "Clear Channel" is supported for data transmission. The PBX must use the X-CCD or CLEARMODE codec. A URI parameter **;x-sin=700** can be added in the P-Preferred-Identity (private option for "Service Indicator").

5.12. ISDN Hold/Unhold

ISDN Hold/Unhold (e.g. when switching in ISDN PBX systems) is not forwarded to the PBX but intercepted in the network. A SIP subscriber thus receives the subscriber's hold music from the ISDN.

A reInvite of a session with SDP attribute "a=sendonly" is transmitted to the network as "Hold". A subsequent reInvite with the attribute "a=sendrecv" is transmitted to the network with "Unhold".

The PBX can transmit the hold music accordingly with "a=sendonly" if no hold music is configured in the trunk domain.

If "sendonly" (SIP) and subsequently "hold" (PSTN) is signalled, hold music can be played from the external network. Telecom Liechtenstein has no control over this function in external networks.

Note: It is recommended to play hold music for PBX-internal transfers without further signalling in the SIP in the existing audio stream ("a=sendrecv"). This means that external subscribers should not receive any signalling information for PBX-internal processes if possible.

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6. SIP header specification

General requirements:

- RFC3261 support: The PBX must support "loose routing" according to RFC3261. The following must also be supported: Multiple record route, route and via headers, as well as correct request URI and route set operations for in-dialogue requests.
- No RFC2543 support.
- A complete contact URI in the request URI must be used for session updates in reInvites. All route headers of the current route set must be included, as well as the complete list of via headers. The contact URI that must be used for reInvites must also use all URI parameters (contact URI is everything between <...> in the contact header. See URI vs URL).
- Supported SIP methods are INVITE, CANCEL, BYE, ACK, INFO, OPTIONS.
- Active conversations are periodically checked using in-dialog INFO messages. The PBX must respond to these in some form, whereby the conversation is terminated by the network in the event of 481 or 480 responses or no response (important for firewalls that block these SIP messages after a certain period of time).
- Domain names (and not IP addresses) are required in the Request URI (REGISTER and initial INVITE), as well as in the domain part of the From: and To: headers (see chapter 4.2).
- An outbound proxy only determines the route to the first hop. Under no circumstances should the outbound proxy be used as a domain name in SIP headers (except Route:) or in the Request URI (see RFC3261).
- Request URI, To URI and From URI should not contain any port information if possible (see RFC3261).
- The Contact header in the initial INVITE or REGISTER must contain the (UDP or TCP/TLS) port if the standard port 5060 is not used (see RFC3261).

6.1. VoIP to PSTN number mapping

For PBX to PSTN calls, the SIP header information is translated as follows. <domainname> is neither the outbound proxy nor any domain, but **the domain of the SIP account** (see chapter 4.2).

6.1.1. Request URI

The Request URI in the initial INVITE must contain the destination number in the following format

sip:<destination>@<domainname>

The destination number is taken exclusively from the Request URI. The destination number can be selected in the following formats:

- Emergency number (without area code) - see chapter 6.1.5
- Phone number (local dialling, only for geographic phone numbers)
- 0 area code phone number (national call – only in Switzerland): The destination is in the same country code as the connection.
- 00 country code area code phone number (international dialling)
- + Country code Area code Phone number (international dialling) (Default = Profile Setting)

6.1.2. From, P-Preferred-Identity and P-Asserted-Identity

The SIP header From, P-Preferred-Identity and P-Asserted-Identity for transmitting the phone number to be displayed are interpreted depending on the SIP trunk type and profile (see chapter 10).

Notes on "User Provided Number": We cannot guarantee that the phone number will be displayed to the called party, as this depends in part on the end device or network operator of the called party. Furthermore, the transmission of the "User Provided Number" abroad cannot be guaranteed; as a rule, this number is

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still displayed within Europe. However, there are increasing regulatory requirements in various countries that require such calls from abroad with CLI/CLIP to be displayed as "anonymous" to the called party.

6.1.3. Privacy header (suppression of caller ID)

The suppression of the caller ID per call (CLIR-T) can be carried out using the privacy header in accordance with RFC3323. The supported tokens are "none" and "id". If privacy is set to "id" (Privacy: id), all phone numbers are sent to the PSTN "suppressed" and replaced with "anonymous@anonymous.invalid" for calls to other SIP subscribers.

In general, if call display is to be suppressed, all transmitted phone numbers are suppressed in the same way.

6.1.4. System-supported call forwarding in the network (partial call forwarding)

The PBX can divert one call per call by sending a 302 Moved Temporary with the new destination phone number in the Contact Header before the Final Response. The call is then forwarded in the network, retaining the caller's phone number. The PBX phone number is added as the redirecting number (diversion) and the forwarding is charged to the PBX account in accordance with the tariff provisions for the destination phone number.

Although redirection saves bandwidth to the customer's connection, it counts as one of the permitted simultaneous calls and increases this counter. The connection is therefore still "busy" after redirection has taken place when the maximum number of simultaneous calls is reached.

6.1.5. Emergency calls

Emergency calls are routed to the emergency call centres assigned to the account based on its header number. Emergency calls are handled separately on the Telecom Liechtenstein network and given priority over conventional calls.

An emergency call is only treated as such if no area code is dialled before the emergency number.

If an emergency call is made with an area code, it is delivered to the emergency call centre assigned to that area code.

Emergency calls are defined on the regulatory authority's website at:

- For Switzerland (+41):
  https://www.bakom.admin.ch/de/notrufdienste
- For Liechtenstein (+423):
  https://www.llv.li/de/landesverwaltung/amt-fuer-kommunikation/elektronische-kommunikation/nummerierung

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6.2. PSTN to VoIP number mapping

For PSTN to PBX calls, the SIP header information is translated as follows.

6.2.1. Calling Party ID and Additional Calling Party ID (CLI)

The format is defined by the configuration of the SIP business trunk and is "+E.164" by default (e.g. "+4232000409").

If another phone number ("User Provided Number" or "Additional Calling Party ID") is available, it is transmitted in the From field instead of the network-provided phone number. The SIP Business Trunk allows the additional transmission of the network-provided phone number in the P-Asserted-Identity header in exceptional cases.

6.2.2. Privacy (suppression of phone number display)

If the caller wishes to suppress the display of their phone number, no phone number is sent. "anonymous@anonymous.invalid" is transmitted in the From field.

6.2.3. Diverted calls

If the subscriber is called via a diversion (indirectly) and this formation is set by the diverting network, the SIP header "Diversion" is added in accordance with draft-levy-sip-diversion-08.txt.

6.3. Response code mapping

Response code mapping is not part of this specification. Telecom Liechtenstein endeavours to carry out mapping that is as meaningful as possible in accordance with the specifications.

Since few special response codes are available in SIP, 5xx response codes (500 and 503) are mapped to a large number of ISDN response codes. In particular, "busy" should be mentioned here, which, unlike "subscriber busy", is not answered with 486 but also with 503.

If an ISUP release cause is available, it is transferred in the reason header. Example:

Reason: Q.850; cause= 1; text="Unallocated (unassigned) number"

Conversely, the PBX can also set a Reason header. This is transmitted transparently and is preferred over SIP response causes.

6.3.1. Transferring announcements from the PSTN

A status indication "no indication" is translated to 183 Progress.

A status indication "Subscriber free" is translated to 180 Ringing.

If "In-band information is now available" is also set by the PSTN, the 1xx status information also has SDP parameters.

In general, 183/SDP is sent, but it may happen that 180/SDP is transmitted for "Subscriber free"/"In-band information is now available". An explicit example of this is tariff announcements for mobile numbers when porting numbers. The PBX is responsible for passing on the appropriate announcement to the subscriber even with 180/SDP.

We accept no liability for tariff announcements that are not connected and the resulting costs due to "ignorance" during calls to value-added numbers or mobile subscribers.

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6.3.2. Transmission of announcements to the PSTN

No guarantee is given for the transmission of 183/SDP ("text before ringing") to the PSTN. To ensure that announcements are transmitted to the PSTN, the PBX must establish the connection (200 OK, Connect) ("text after ringing").

6.3.3. Ringing timeout

Connections in state 180 Ringing or 183 Progress/SDP are disconnected in our network after 120 seconds. This disconnection may also occur earlier in external networks.

6.4. DTMF (Dual Tone Multi Frequency)

DTMF is transmitted inband using RFC2833 or as RTP audio in accordance with the negotiation via RFC 3264. The default DTMF payload type is 101 but can also be another type in the range 96 to 127.

6.5. Fax support

Fax transmissions are supported either using G.711 A-law (bypass or fallback) or T.38 (udptl).

A prerequisite for error-free fax transmission is an appropriate IP connection with IP Quality of Service with a packet loss < 0.1% and a jitter < 20 ms.

If T.38 is used for fax transmission, the CPE provided by the customer or the T.38 software stack (e.g. in modern fax server solutions) must guarantee compatibility with the fax devices used. In this case, Telecom Liechtenstein cannot guarantee error-free transmission to all fax devices.

A fax transmission must always begin with a voice codec and switch to T.38 using reInvite. Direct connection establishment with transmission via T.38 is not supported.

The UPDATE method cannot be used.

The maximum supported speed is 14400 bps. Super G3 or V.34 fax machines (33600 bps) are not supported. These devices must be set to a maximum speed of 14400 bps, as a successful handshake at 33600 bps at the beginning of the connection (before switching to T.38) will result in disconnections.

The system must support a fallback to G.711 A-law if the remote station rejects a codec change to T.38 with SIP reply code 488 or 415.

6.5.1. Fax transmission with T.38 on the PBX side

Fax transmission via T.38 to the PSTN is supported in the following way:

1. T.38 passive mode: This (default) mode allows switching to T.38 within a session if the remote station initiates the codec change.
2. T.38 active mode: The PBX can initiate the codec change. In most cases, however, successful T.30 negotiation is questionable. This mode may be useful in exceptional cases, but is not recommended.

Towards other SIP participants, successful T.38 transmission depends on the terminal used on the other side. If the PBX wants to switch to T.38 and the remote station rejects this with response code 488, the PBX must continue the call with the previously used codec or perform a fallback to another codec to be negotiated (preferably G.711 A-Law fallback).

An attempt is always made to establish an end-to-end connection with T.38. Repeated transcoding in the network usually results in unsuccessful fax transmission.

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6.5.2. Fax transmission with T.38 on the provider side and G.711 A-law bypass on the PBX side

Fax transmission from the PSTN to the PBX using T.38 is supported in the following way:

1. T.38 passive mode: This (default) mode allows switching to T.38 within a session, but the PBX must switch the codec itself. The gateway will never actively suggest switching to T.38.

If the call comes from another SIP subscriber and they want to switch to T.38, the PBX must reject this with a response code 488 or 415, if desired. The remote station and the PBX must then continue the call with the previously negotiated codec or negotiate a new common codec (preferably G.711 A-Law) (fallback).

7. Codec support

The following codec list is supported. The PBX must be able to handle the list of codecs in the INVITE in accordance with RFC3264. The default value of the packetisation interval is 20 ms (ptime 20), where possible.

- G.711 A-law (only packetization interval <= 40ms)
- OPUS (end-to-end between VoIP endpoints)
- G.722 (end2end between VoIP endpoints)

Defined codec payload types (PT) are transmitted in accordance with RFC3551.

G.711 μ-law is no longer supported.

G.729 (all variants) are no longer supported.

8. Encryption (SIPS, SDES)

The voice connection and associated signalling can be encrypted. TLS (analogous to the HTTPS protocol) is used for signalling (SIPS) and SDES is used for the media path for symmetric encryption of the RTP streams (SRTP).

The same header specifications apply as for UDP or TCP.

If TLS is used as the transport protocol, SRTP must be used. TLS signalling without SRTP will be rejected.

Please refer to the document "FL1 Trunk Domain Concept and Proxy Settings" available on the FL1 homepage https://www.fl1.li/service/download-center/ and subsequently in the customer portal for the TLS outbound proxy settings.

9. Supported SIP systems (PBX)

There are a number of SIP systems that have been tested and/or certified by the manufacturer and/or Telecom Liechtenstein. For a list of SIP systems, please refer to the document "FL1 Trunk PBX _ Compatibility List", which can be found on the FL1 homepage https://www.fl1.li/service/download-center/ and in the customer portal under Documents.

The appropriate configuration settings for the tested or certified SIP systems are available on the FL1 homepage at https://www.fl1.li/service/download-center/.

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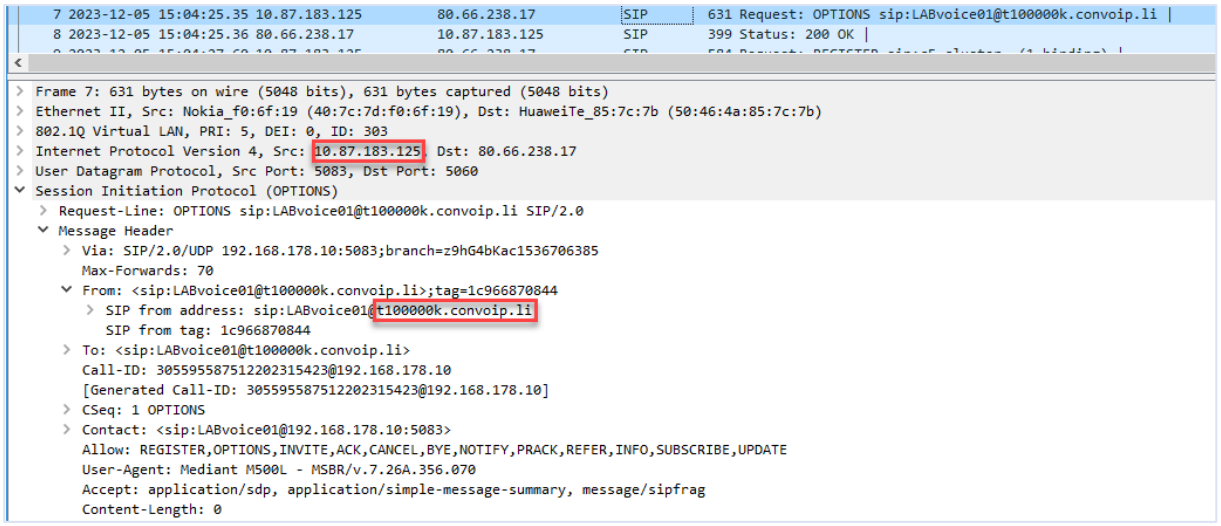
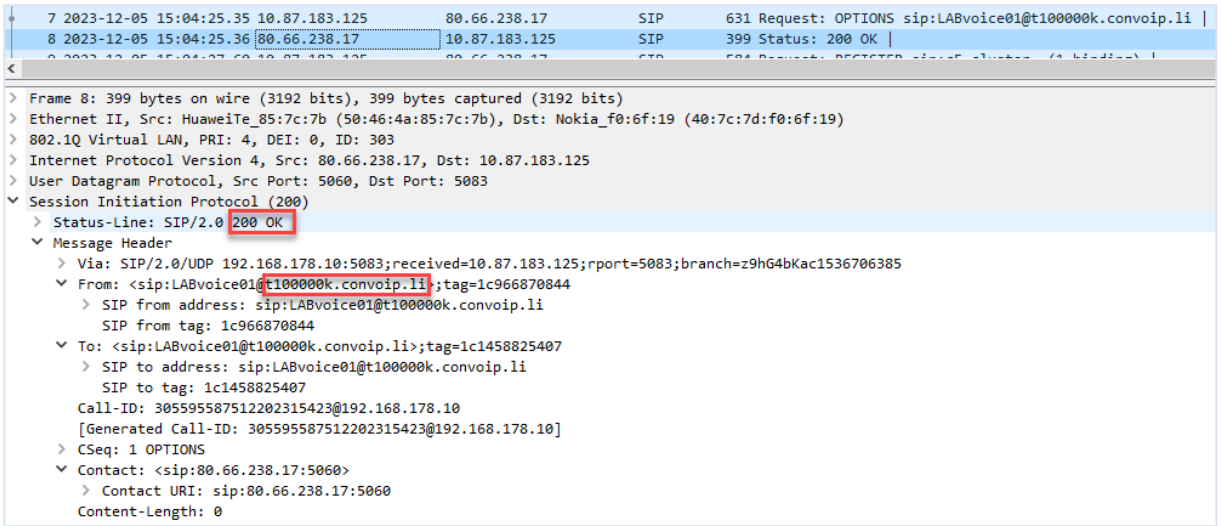
10. SIP options Monitoring

In general, SIP options can be used by the PBX to monitor a SIP trunk. If SIP options are not answered by FL1, an alarm can be triggered on the customer side so that the transport connection to FL1 can first be checked for functionality.

SIP options that have IP addresses set in the From header are not answered. For a successful response, the SIP trunk must be registered and the domain name must be entered in the host part of the From header (see the following example.)



Below is the successful response from FL1:



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11. SIP trunk e examples

Predefined SIP trunk profiles are available to ensure broad support for various PBX systems.

The following applies to all SIP trunk variants:

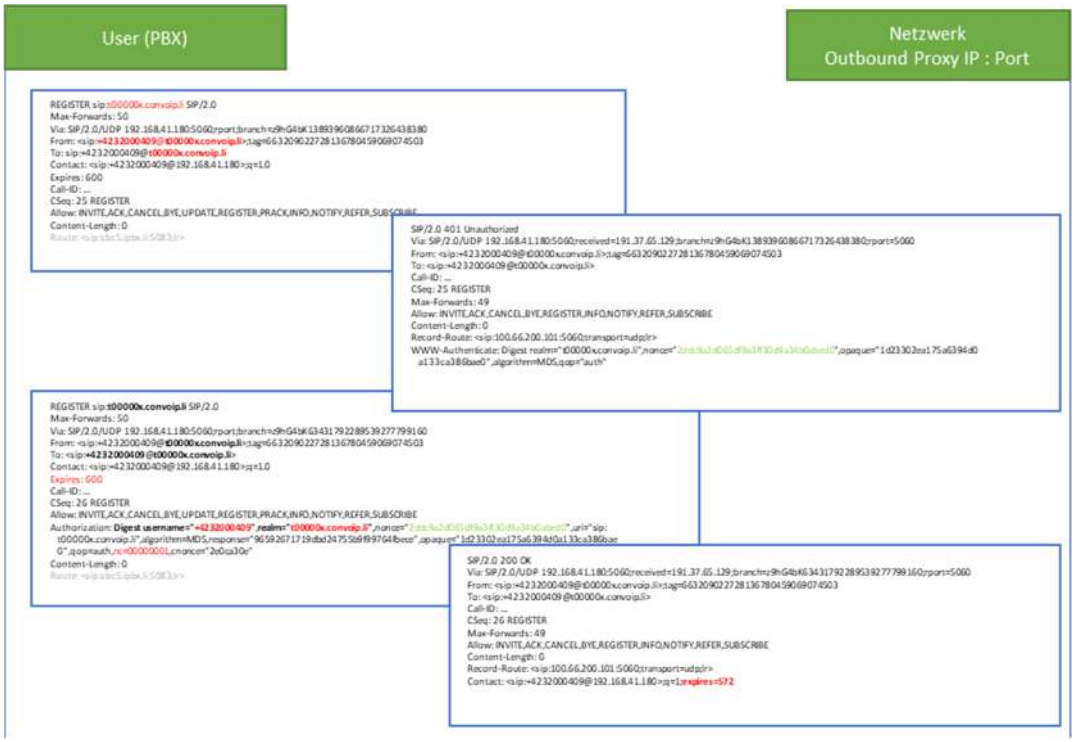
- The authentication user name is the value displayed in the customer portal.
- The authentication user name does not contain an extension or a domain.

The Authentication Realm is the domain of the SIP account.

To authenticate the SIP trunk, the SIP account must appear either in *the From* or *P-Asserted-Identity* header. A request without this information (e.g. in an anonymous case where only anonymous@anonymous.invalid appears) will be answered with 404.

All variants require a valid registration; without this, calls will not be allowed (rejection with response cause 403).

Example of a registration:



The client must respond to the expires in the 200 OK response and must re-register in good time, but not before half the time has elapsed.

A route header is optional; the CPE must send the REGISTER to the outbound proxy. The outbound proxy is not necessarily the DNS A record of the SIP domain and is communicated in the follow-up sheet and subsequently in the customer portal.

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11.1. Business Trunk Profile "LI/CH E164" (see chapter 3.1)

The connection number is transmitted in the *P-Preferred-Identity* (PPID) header. Alternatively, the *P-Asserted-Identity* header (PAID) can be used.

The PAID or PPID can also contain any extension at the end.

This profile allows the transmission of "anonymous@anonymous.invalid" in the From header.

The domain in the *From* header must always be the domain of the SIP account or anonymous.invalid.

- From header user part in +E.164 format
  - Own phone number in the number range
  - Other phone number to be displayed to the recipient
    This is transmitted as a so-called "User Provided Number" (Additional Calling Party ID, user provided, not verified).
- PPID in +E.164 format (recommended) or 00E.164
  - Your own phone number (!)
- PAID in +E.164 format (recommended) or 00E.164
  - Your own phone number (!)

The CLIP No Screening function is an option that must be activated on the Business Trunk in order to be used.

11.1.1. Examples

- The number range is 423/200040x and the extension 9 should be displayed:

```
From: <sip:+4232000409@t100000x.convoip.li;user=phone>
P-Asserted-Identity: <sip:+4232000409@t100000x.convoip.li;user=phone>
```

or

```
From: <sip:+4232000409@t100000x.convoip.li;user=phone>
P-Preferred-Identity: <sip:+4232000409@t100000x.convoip.li;user=phone>
```

or

```
From: <sip:+4232000409@t100000x.convoip.li;user=phone>
```

- Trunk phone number is 423/2000409 and the (external) phone number 7910001 should be displayed:

```
From: <sip:+4237910001@t100000x.convoip.li>
P-Preferred-Identity: <sip:+4232000409@t100000x.convoip.li>
```

or

```
From: <sip:+4237910001@t100000x.convoip.li>
P-Asserted-Identity: <sip:+4232000409@t100000x.convoip.li>
```

- Business Trunk Authentication User is "LIFL1Gech01" and the (external) phone number 7910001 should be displayed:

```
From: <sip:+4237910001@t100000x.convoip.li;user=phone>;
P-Asserted-Identity: sip: LIFL1Gech01@t100000x.convoip.li;user=phone
```


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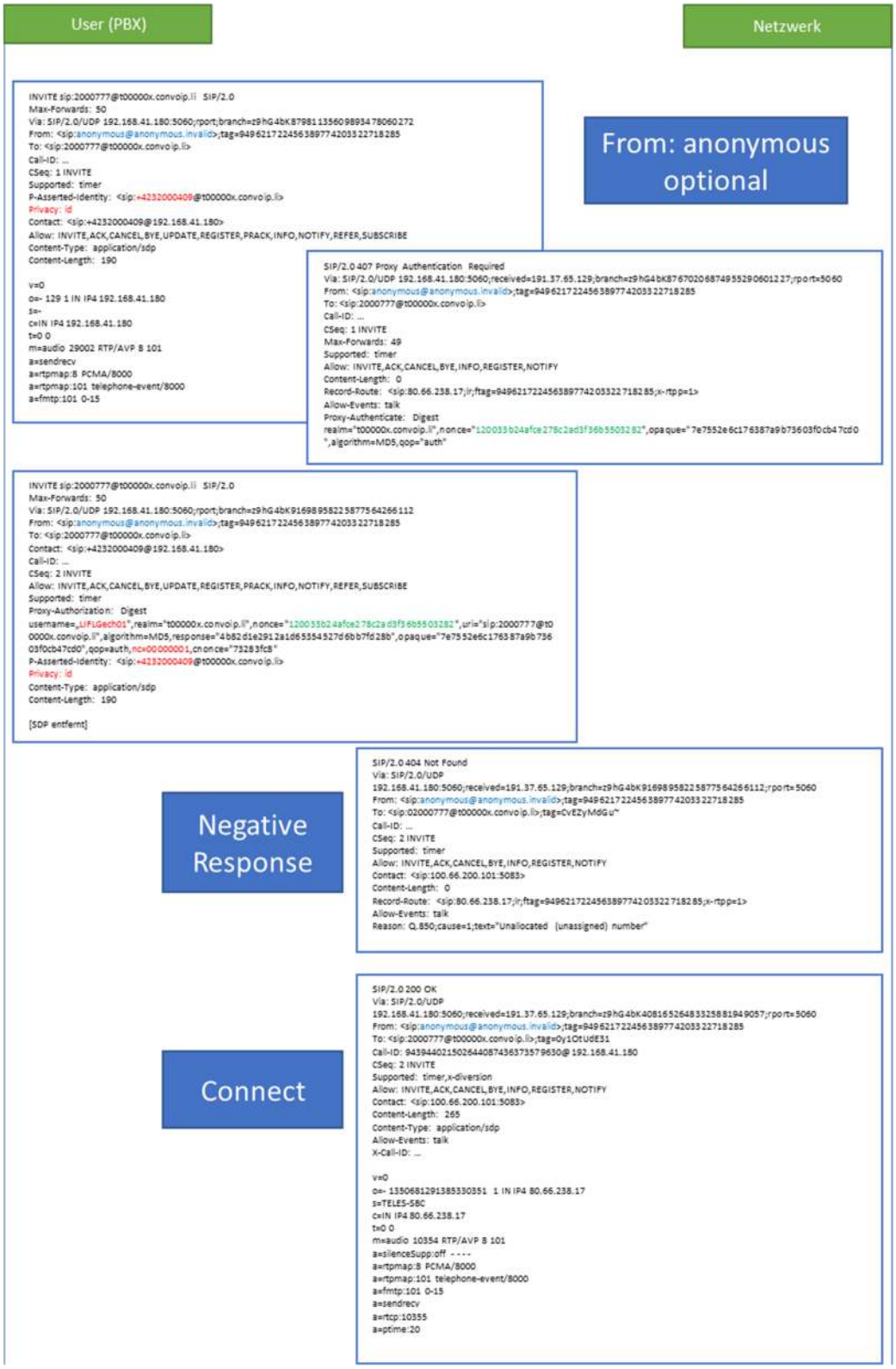
11.1.2. Call Flows

- Without CLIP screening, anonymous call:



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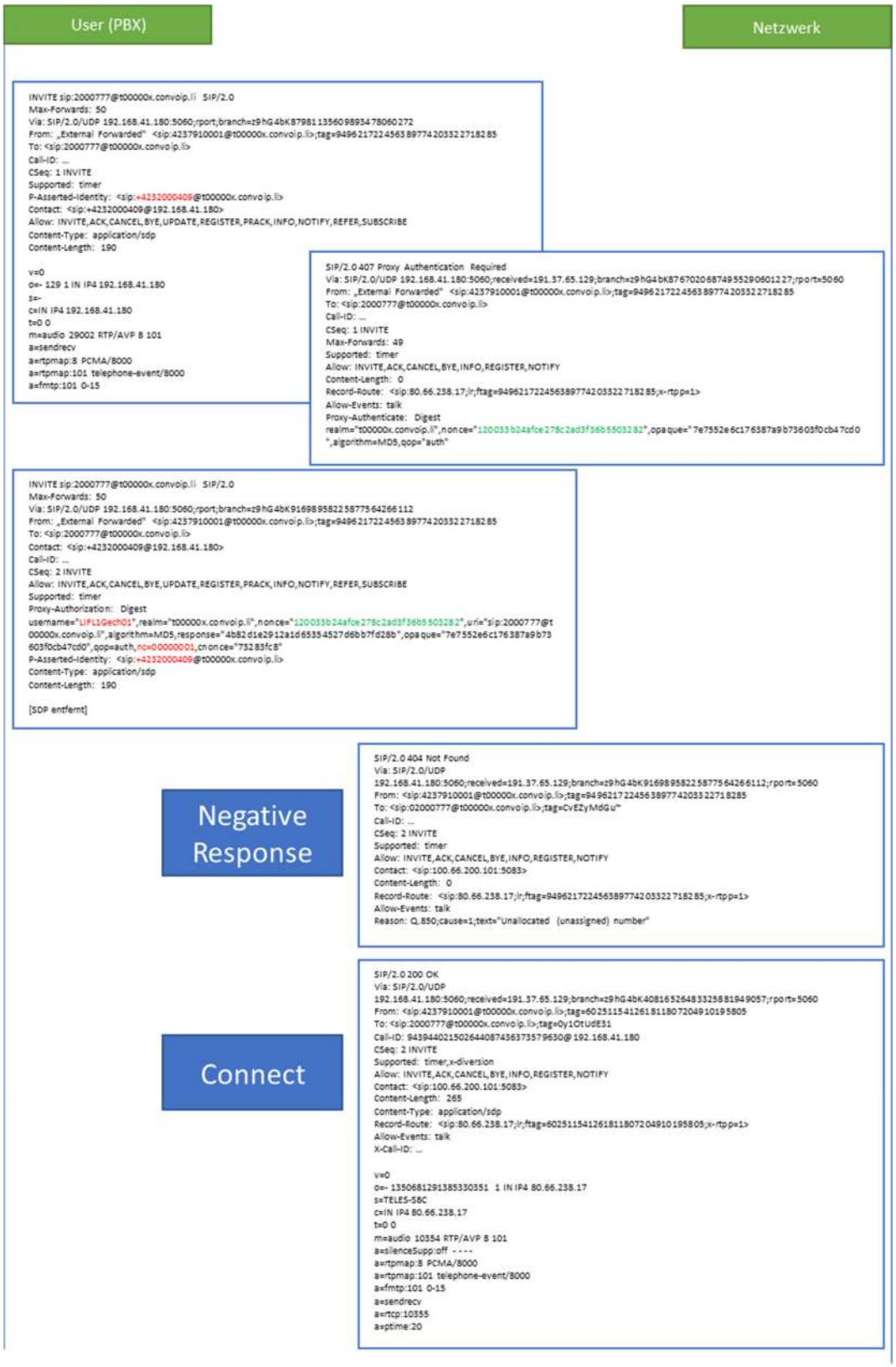
- With CLIP No Screening (user-provided number):



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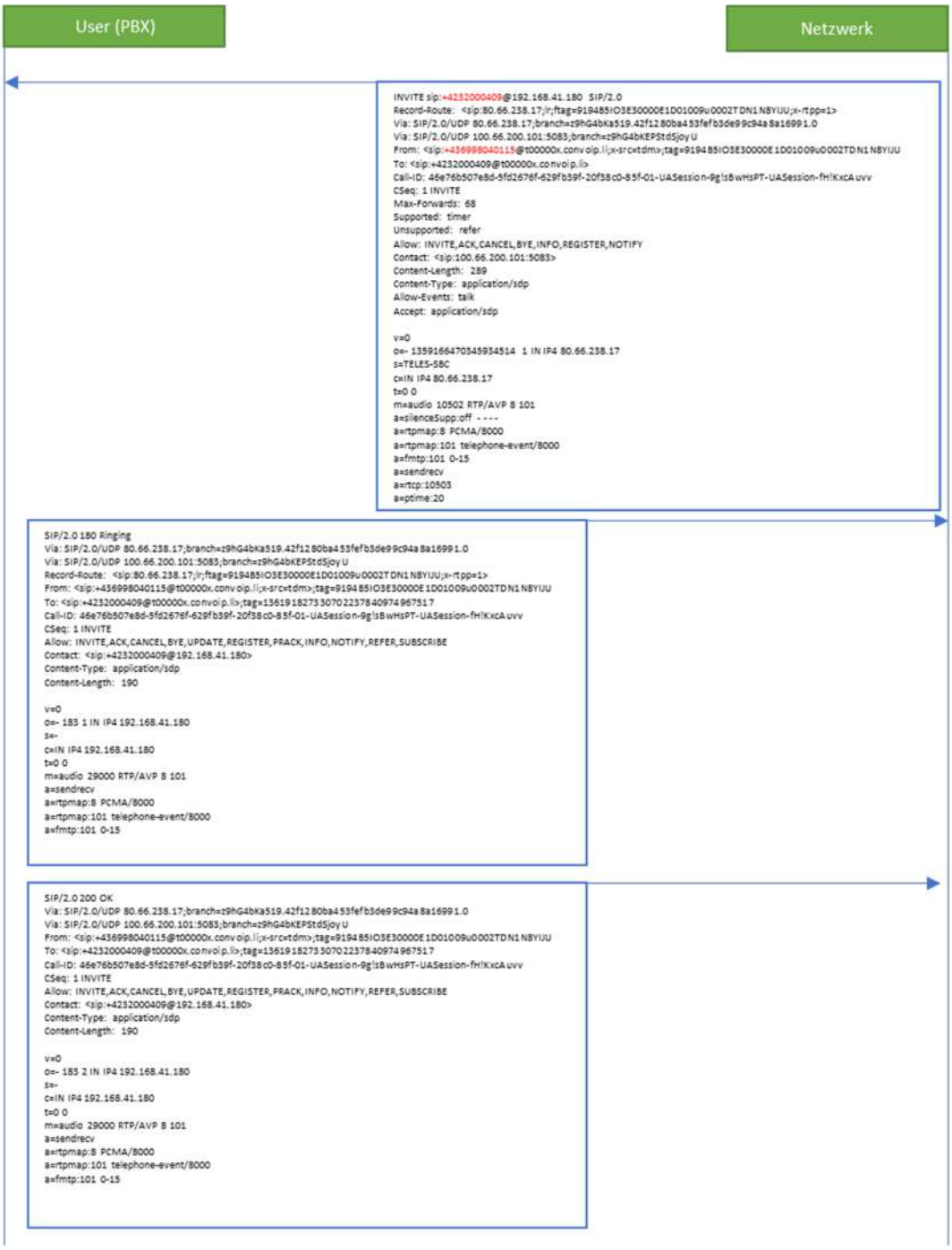
- Terminating call:



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11.2. Business Trunk Profile "LI National" (see chapter 3.2)

The connection number is transmitted in the *From* header.

The domain in the *From* header must always be the domain of the SIP trunk.

The behaviour is essentially the same as in the Business Trunk Profile "LI/CH E164", but with the *From* and *P-Preferred-Identity* (PPID) (alternatively PAID) headers reversed:

- From header user part in the format +E.164 (recommended), 00E.164 or also national
  - own phone number
- PAID in format +E.164
  - own phone number
- PPID in +E.164 format
  - own phone number
  - Other phone number to be displayed to the recipient

Transmitted as a so-called "User Provided Number" or "CLIP-No-Screening" (Additional Calling Party ID, user provided, not verified).

The CLIP No Screening function is an option that must be activated on the Business Trunk in order to be used.

11.2.1. Examples

- Business Trunk phone number is 2000409 and the (external) phone number 7910001 should be displayed:

```
From: <sip:2000409@t100000x.convoip.li>
P-Asserted-Identity: <sip:+4237910001@t100000x.convoip.li>
```

or

```
From: <sip:+4232000409@t100000x.convoip.li>
P-Preferred-Identity: <sip:+4237910001@t100000x.convoip.li>
```

or

```
From: <sip:+4232000409@t100000x.convoip.li>
P-Preferred-Identity: sip:004237910001@t100000x.convoip.li
```

- Business Trunk Authentication User is "LIFL1Gech01" and the (external) phone number 7910001 should be displayed:

```
From: <sip:LIFL1Gech01@t100000x.convoip.li>
P- Preferred -Identity: <sip:+4237910001@t100000x.convoip.li>
```


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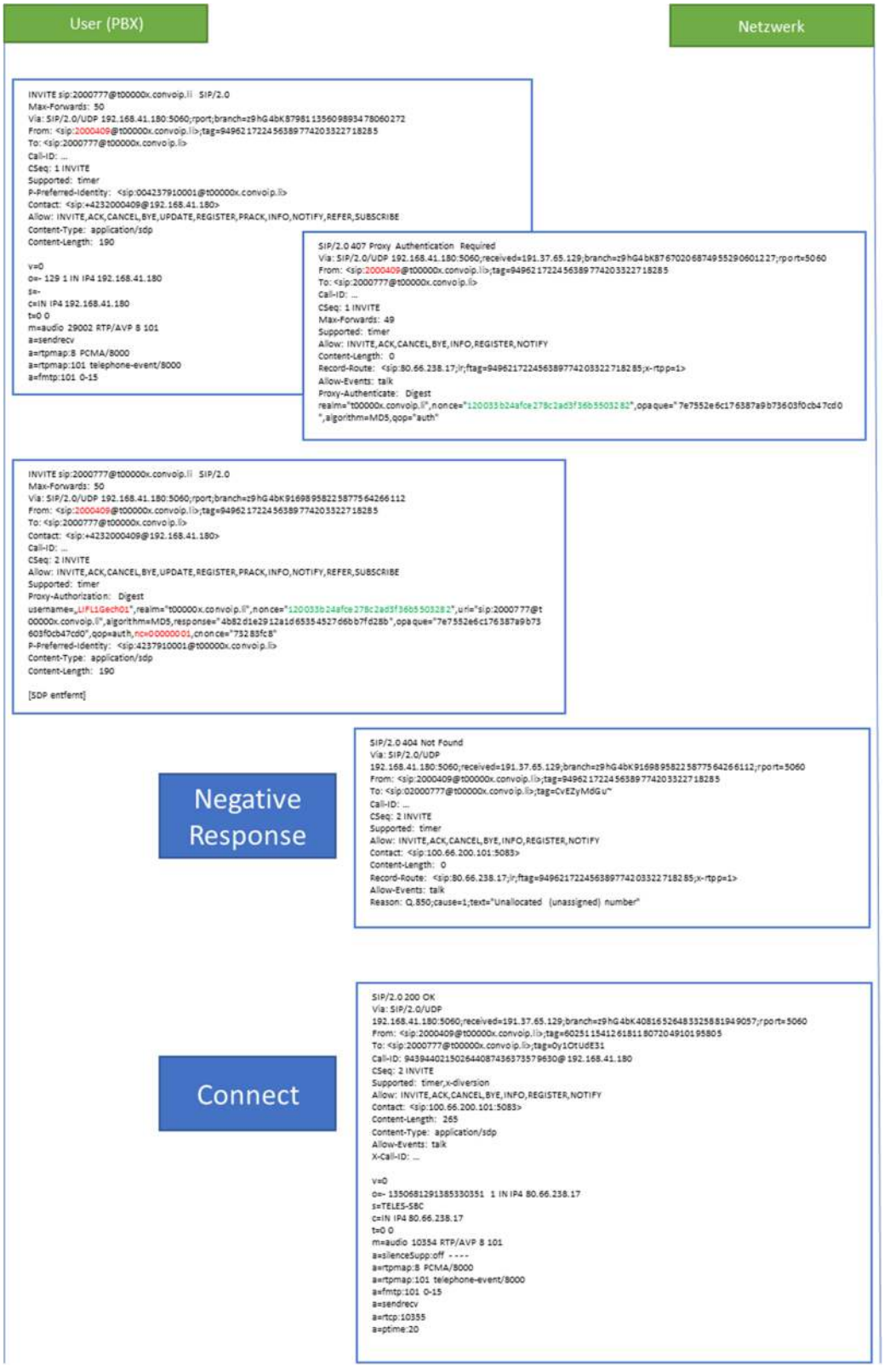
11.2.2. Call Flows

- With CLIP No Screening:



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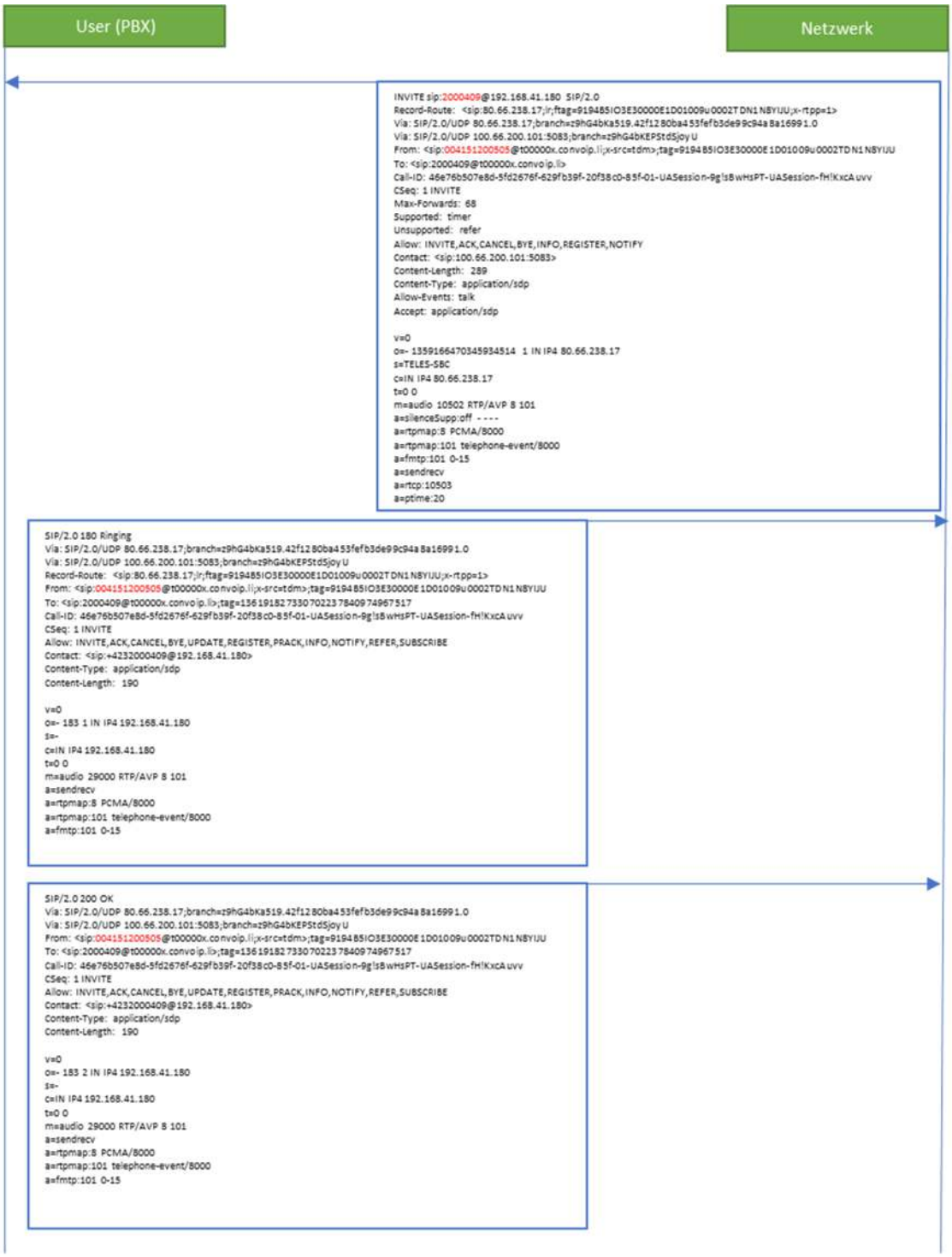
- Terminating Call:



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13. Gloss

- Calling Party ID – Number of the caller
- cc – Country code, e.g.: 43 for Austria
- DDI – direct dial in – direct dialling capability of a phone number
- Domain name – SIP domain name provided by the provider for a SIP business trunk
- Domain part – Domain name in a SIP header <sip:01234567@t100000x.convoip.li>
- E.164 – phone number format, e.g.: 4231234567 for the phone number 1234567 in Liechtenstein
- ext – Extension
- nc – National code (area code)
- num – phone number without area code
- PAID – SIP header P-Asserted-Identity
- PBX – Private Branch Exchange – Telephone system
- PPID – SIP header P-Preferred Identity
- User-Part – User name in a SIP header < sip:1234567@t100000x.convoip.li >

14. Standards and RFCs

The following standards and RFCs are supported and used:

RFC3261	SIP: Session Initiation Protocol
RFC3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
RFC3263	Session Initiation Protocol (SIP): Locating SIP Servers
RFC3264	An Offer/Answer Model with the Session Description Protocol (SDP)
RFC3323	A Privacy Mechanism for the Session Initiation Protocol (SIP)
RFC3325	Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks
RFC2833	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
RFC3550	RTP: A Transport Protocol for Real-Time Applications
RFC3551	RTP Profile for Audio and Video Conferences with Minimal Control
RFC4028	Session Timers in the Session Initiation Protocol (SIP)
draft-levy-sip-diversion-08.txt	Diversion Indication in SIP
RFC4244	An Extension to the Session Initiation Protocol (SIP) for Request History Information
RFC4317	Session Description Protocol (SDP) Offer/Answer Examples
RFC4566	SDP: Session Description Protocol
ITU-T Q.1912.5	Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control Protocol or ISDN User Part

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