**Internet Telephony Service Provider (ITSP) Survey**

|  |  |
| --- | --- |
| ITSP Name / Company name |  |

**This survey will help us to understand how we could connect your trunk service best to our OpenScape products:**

* **OpenScape Business**
* **OpenScape 4000**
* **OpenScape Enterprise**
* **OpenScape SBC**
* **OpenScape Cloud**

Therefore please take few minutes to fill in the attached survey and return it to us. The more information we get in advance the better we can support the tests

**Data obtained through this questionnaire will be treated confidential and are used by Unify only.**

We have split the technical questions into the following parts:

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|  |
| --- |
| Questions in “green” cells are mandatory to start the certification |

**Many thanks for your help!**

**Certification type**

|  |  |
| --- | --- |
| This is the first certification *(initial certification)* | Yes  No |
| This is the re- certification *(initial certification was alread passed )*  Have you made any modifications to your interfaces since the last certification date? *(e.g. new platform, new features, new transport)* | Yes  No  Yes  No |

# General ITSP information

|  |  |
| --- | --- |
| ITSP product / Trunk Service Name  *(****this name will be used as profile name in the PBX****)* |  |
| Internet page (URL) *(this link will be published on our Wiki page)* |  |

**Contact Information**

|  |  |
| --- | --- |
| Contact Name(s)  Email  Phone number  Address |  |

**Please select the countries your service supports**

|  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- |
|  | Australia |  | Finland |  | Lithuania |  | Reunion |
|  | Austria |  | France |  | Luxembourg |  | Romania |
|  | Belgium |  | Germany |  | Malta |  | Serbia |
|  | Brazil |  | Greece |  | Martinique |  | Slovakia |
|  | Bulgaria |  | Guadeloupe |  | Mayotte |  | Slovenia |
|  | Canada |  | Hong Kong |  | Mexico |  | Spain |
|  | Croatia |  | Hungary |  | Monaco |  | South Africa |
|  | Chile |  | Ireland |  | Netherlands |  | Sweden |
|  | Cyprus |  | Italy |  | New Zealand |  | Switzerland |
|  | Czech Republic |  | Japan |  | Norway |  | United Kingdom |
|  | Denmark |  | Latvia |  | Poland |  | USA |
|  | Estonia |  | Liechtenstein |  | Portugal |  | |
|  | Others: |  | | | | | |

**General platform information**

|  |  |
| --- | --- |
| Is an interface specification distributed? *(if yes, attach the document or provide the download link )* | Yes  No |
| Is an test guide distributed? (if yes, attach the document or provide the download link ) | Yes  No |
| Used plattform: Please list the manufacturer/model/ version of the relevant products used in your SIP core network (proxy, registrar, media gateways) |  |

# Technical Questions: Transport and configuration

**Transport**

| Question | Answer |  |
| --- | --- | --- |
| 1. Transport protocol? |  |  |
| * 1. Do you support UDP? | Yes | No |
| * + 1. do you require symmetric UPD (sent and receive from same port)? | Yes | No |
| * + 1. do you perform/support automatic upgrade to TCP when MTU size exceeded? | Yes | No |
| * 1. Do you support TCP? | Yes | No |
| * + 1. is there a limit of parallel TCP connections established to your server ? (leave open if no limit exist) |  | |
| * 1. Do you support SIP over TLS? | Yes | No |
| * + 1. what versions of TLS do you support (check all that apply)? | TLS v1.1  TLS v1.2  TLS v1.3 |  |
| * + 1. do you require Basic Authentication:   Mutual Authentication: |  | |
| * + 1. If yes, who is your preferred Certificate Authority / provider (e.g. Comodo) |  | |
| * + 1. If yes, do you support Secure media (SRTP) using SDES? | Yes | No |
| 1. Proxy addresses |  |  |
| * 1. Is a wellknown server address used (PBX is shipped with this preconfigured address, no need to change during trunk configuration e.g. 217.11.12.13 or siptrunk.company.net) |  |  |
| Customer specific server adresses are used (PBX is shipped with a placeholder which needs to be replaced with the address during trunk configuration e.g. please.enter.here) |  |  |
| * 1. SIP Proxy FQDN/address: |  | |
| * + 1. If FQDN is used, do you have an IANA registered domain name which can be resolved using public DNS or DNS-SRV? | DNS-A  DNS-SRV  DNS-NAPTR |  |
| * 1. Is your SIP Proxy listening on the default port (5060/5061) ? | Yes | No |
| * + 1. If no, specify the port to be used (if not determinded DNS-SRV) |  | |
| 1. Connectivity checks |  | |
| * 1. Do you support receiving connectivity checks with OPTIONS? | Yes | No |
| * + 1. If yes what is the recommended keep alice interval |  | |
| * 1. Do you accept “empty” UDP packets on the SIP port (e.g. to keep bindings)? | Yes | No |
| * 1. Do you sent connectivity checks with OPTIONS? | Yes | No |
| * + 1. If yes do you require a SDP-Body in the 200 OK ? | Yes | No |

**Registration mode**

|  |  |  |
| --- | --- | --- |
| Question | Answer |  |
| 1. Do you support static mode ? | Yes | No |
| * 1. PBX Address: Do you support FQDN and/or just IP for the static trunk (i.e. you connect to cloud.unify.com rather than to a particular IP)? | FQDN  static IP |  |
| * 1. Do you need Digest authentication for the static trunk? | Yes | No |
| * 1. Are there restrictions for the SIP port used by the PBX ? (e.g. PBX uses port different from 5060/5061 for sending and receiving SIP messages)   If yes, please specify which port must be used by the PBX | Yes | No |
| 1. Do you support registration mode? | Yes | No |
| * 1. Do you support Digest authentication ? | Yes | No |
| * 1. Do you support one single RFC3261 registration for a single number? (also known as MSN type of registration) | Yes | No |
| * 1. Do you support a range of numbers with one single RFC3261 registration? (also known as DID type of registration) | Yes | No |
| * 1. Do you support SIP-connect registration (RFC6140 based) ? | Yes | No |
| * 1. Registrar FQDN/address |  | |
| * + 1. If FQDN is used, do you have an IANA registered domain name which can be resolved using public DNS or DNS-SRV? | DNS-A  DNS-SRV  DNS-NAPTR | |
| * 1. Is your SIP Registrar listening on the default port (5060/5061) ? | Yes  No | |
| * + 1. If no, specify the port to be used (if not determinded DNS-SRV) |  | |
| * 1. Do you require a specific domain name (different from the Registrar domain)      1. If yes, please describe | Yes  No | |
| * 1. Recommended Reregistration interval |  | |
| 1. If both registration and static mode are supported, please specify which is the preferred method to be used for the PBX? | registration mode   static mode | |
| Remarks: | | |

**Multisite / Multiple Customers**

**Multisite / Multiple customers are represented by different call numbers served by a single PBX instance. We need to know if this configuration is supported by the SSP**

| Question | | Answer |  |
| --- | --- | --- | --- |
| 1. Is Multisite / Multiple customers supported ? | No  Yes, on a single trunk  Yes, dedicated trunk for each location area | | |
| If Multisite is supported, are there specific requirements for that feature: | |  |  |
| * 1. If a dedicated trunk is needed, do you support multiple registrations from a single IP address | | Yes | No |
| * 1. do you need a dedicated port per registration from the PBX | | Yes | No |
| * 1. Are you able to provide individual customer billing when multiple registrations attach to/from the same IP address? | | Yes | No |
| Additional notes/remarks: | | | |

# Technical Questions: Call related

**Incoming call SSP -> PBX**

|  |  |  |
| --- | --- | --- |
| Question | Answer | |
| 1. The called party number is presented in   (where do the PBX take the number from) | Request line  To header field  P-Called-Party-Id  other: | |
| 1. The called party is delivered in a | sip: URI  tel: URI  other: | |
| 1. The format of the called party number is:   example for german numbering plan international prefix 00 country code 49 national prefix 0 local area code 89 | international E.164 with + (+498912345678)  international w/o prefix (498912345678)  international w/ prefix (00498912345678)  national w/o prefix (8912345678)  national w/ prefix (08912345678) | |
| 1. The calling party number is presented in (where do the PBX take the number from) | From  [P‑Asserted‑Identity](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html" \l "idx)  [P‑Preferred‑Identity](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html" \l "idx)  other: | |
| 1. The calling party is delivered in a | sip: URI  tel: URI  other: | |
| 1. The format of the calling party number is | international E.164 with +  international w/o prefix  local, national or international number in dialable format including prefixes | |
| 1. Do you support Calling party name within SIP URI display-name field? | Yes | No |
| 1. Is session timer required | Yes | No |
| 1. Is early media supported for incoming calls (18x with SDP supported) ? | Yes | No |
| 1. Do you use initial INVITE without SDP ? | Yes | No |
| 1. Is Call rerouting with302 response supported ? | Yes | No |
| Additional notes/remarks: | | |

**Outgoing call PBX -> SSP**

| Question | Answer | | | |
| --- | --- | --- | --- | --- |
| 1. Do you require a specific domain name (different from the SIP proxy domain)    1. If yes, please describe | Yes  No domain: | | | |
| 1. The calling party must be present in the |  | | |  |
| * 1. From header field | as call number (format specified below)  [SIP](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html" \l "idx) User name (valid only in case of registration) | | | |
| * 1. [P‑Asserted‑Identity](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html" \l "idx) header field | as call number (format specified below)  [SIP](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html" \l "idx) User name(valid only in case of registration) | | | |
| * 1. [P‑Preferred‑Identity](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html" \l "idx) header field | as call number (format specified below)  [SIP](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html" \l "idx) User name(valid only in case of registration) | | | |
| * 1. The expected format of the calling party number is   example shown for german numbering plan: international prefix 00 country code 49 national prefix 0 local area code 8 | international E.164 with + (+498912345678)  international w/o prefix (498912345678)  international w/ prefix (00498912345678)  national w/o prefix (8912345678)  national w/ prefix (08912345678) | | | |
| 1. The format of the called party number in Request-line and To: header is | international E.164 with +  international w/o prefix  national or international number in dialable format including prefixes | | | |
| 1. The following header is used for billing / validation check | From  [P‑Asserted‑Identity](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html#idx) (PAI)  [P‑Preferred‑Identity](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html" \l "idx) (PPI)  [Diversion header](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html#idx)  [IP](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html#idx) address  other: | | | |
| * 1. if different headers are used, please specify the priority (e.g.1. Div 2. PAI 3.From) |  | | | |
| 1. Do you support Calling party name within SIP URI display-name field? | Yes | No | | |
| * 1. The SIP URI display-name field must contain the calling party numberin the following headers: | From  [P‑Asserted‑Identity](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html#idx) (PAI)  [P‑Preferred‑Identity](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html" \l "idx) (PPI)  [Diversion header](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html#idx)  other: | | | |
| 1. The user part of the contact URI | is empty (SIP-Connect)  [contains](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html#idx) the calling party number  [contains](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html" \l "idx) the registration AOR  other: | | | |
| 1. Call diversion with 2nd call : |  | | | |
| * 1. Do you support the presentation of the original calling party (A-Number) for diverted calls? | Yes | | No | |
| * 1. If yes: | is available in SSP default configuration  needs to be booked / configured on customer request  needs payed  other: | | | |
| * 1. If yes: A-Number must be present in | From  [P‑Asserted‑Identity](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html" \l "idx)  [P‑Preferred‑Identity](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html#idx)  other: | | | |
| * 1. Do you support Diversion header ? | Yes | | No | |
| 1. Do you support Anonymous calls (CLIR) ? | Yes | | No | |
| * 1. Format of anonymous From: | Anonymous <sip:anonymous@anonymous.invalid>  sip:anonymous@anonymous.invalid  sip:anonymous@itsp.net  Anonymous <sip:number@ itsp.net >  no special formatting for CLIR  other: | | | |
| * 1. Privacy header field: | Privacy: id  Privacy: user, id  other: | | | |
| 1. Do you support CLIP no Screening (calling party number outside the customer's number range known by the SSP) | Yes | | No | |
| * 1. CLIP no Screening | is available in SSP default configuration  needs to be booked / configured on customer request  needs payed  other: | | | |
| * 1. CLIP no Screening number is sent in From: and needs a trusted number in | [no](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html#idx) additional trusted information needed  [P‑Asserted‑Identity](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html#idx)  [P‑Preferred‑Identity](http://www.tech-invite.com/fo-abnf/tinv-fo-abnf-sip-p-assertid.html#idx)  other: | | | |
| 1. Is session timer required  include: supported: timer | Yes | | No | |
| * 1. if yes: is a specific value for session expires required  Session-Expires: 1800 | no, default (1800) is accepted  Session expires       required | | | |
| 1. Is PRACK required  include: supported: 100rel | Yes | | No | |
| 1. Is P-Early-Media required  include: P-Early-Media: supported | Yes | | No | |
| 1. Is COLP/TIP supported, 200 OK has a PAI header containing the connected B-Party | Yes | | No | |

**Call signaling PBX -> SSP (SSP is supporting the following in receiving direction)**

|  |  |  |
| --- | --- | --- |
| Question | Answer | |
| 1. Do you support receiving of Emergency Location Information? | Yes | No |
| * 1. If yes, in which SIP fields do you expect this information ? |  | |
| * 1. Is the use of PIDF-LO supported ? | Yes | No |
| 1. Is mid-session 3PCC call retargeting using re-INVITE without SDP supported? | Yes | No |
| 1. Do you support receiving UPDATE? | Yes | No |
| Additional notes/remarks: | | |

**Call signaling SSP -> PBX (SSP is supporting the following in sending direction)**

|  |  |  |
| --- | --- | --- |
| Question | Answer | |
| 1. Is REFER used by the SSP to initate a transfer? | Yes | No |
| 1. Is 302 used by the SSP to redirect a call | Yes | No |
| 1. Do you support sending UPDATE? | Yes | No |
| Additional notes/remarks: | | |

**Media related**

| Question | Answer | |
| --- | --- | --- |
| 1. Codec support |  | |
| * 1. Do you support G.711A Media Attribute (a): rtpmap:8 PCMA/8000 | Yes | No |
| * 1. Do you support G.711U Media Attribute (a): rtpmap:0 PCMU/8000 | Yes | No |
| * 1. Do you support G.729AB Media Attribute (a): rtpmap:18 G729/8000 Media Attribute (a): fmtp:18 annexb=yes (default, if not send formally) | Yes | No |
| * 1. Do you support G.729A Media Attribute (a): rtpmap:18 G729/8000 Media Attribute (a): fmtp:18 annexb=no | Yes | No |
| * 1. Is ISDN data codec/ clearmode/ clearchannel supported as shown? Media Attribute (a): rtpmap:96 CLEARMODE/8000 | Yes | No |
| * 1. Are there other mandatory codecs ? |  | |
| 1. Is silence suppression for G.711 supported? | Yes | No |
| * 1. silence suppression is signalled as  Media Attribute (a): silenceSupp:on | Yes | No |
| * 1. silence suppression is signalled as  Media Attribute (a): rtpmap:13 CN/8000 | Yes | No |
| * 1. if silence suppresion is not supported, is the SDP attribute supported?  Media Attribute (a): silenceSupp:off | Yes | No |
| 1. RFC2833/4733 telefone events |  |  |
| * 1. Are DTMF tones supported Media Attribute (a): fmtp:0-15 | Yes | No |
| * 1. Are Fax tones supported Media Attribute (a): fmtp:98 32-36,49 | Yes | No |
| * 1. is payload type (96-127) negotiated and supported as shown: Media Attribute (a): fmtp:98 0-15 | Yes | No |
| * + 1. if no please specify which payload type MUST be used (e.g. 101) | Payload type: | |
| 1. Is T38 fax supported ? | Yes | No |
| 1. Is media inactivity detection used ? | Yes | No |
| Additional notes/remarks: | | |

# Optional Technical Questions

|  |  |  |
| --- | --- | --- |
| Question | Answer | |
| Transport |  | |
| 1. Do you deliver an access device? | Yes | No |
| * 1. If yes, specify the device |  | |
| * 1. If yes, access device is acting as a router | Yes | No |
| * 1. If yes, access device is acting as a SBC (terminating the SIP/media traffic) | Yes | No |
| 1. NAT Traversal |  |  |
| * 1. Do you expect public routable addresses in SIP/SDP ? | Yes | No |
| * 1. If yes, do provide a STUN server ? | Yes | No |
| * 1. If yes, please specify the FQDN/address to be used: |  | |
| 1. QOS |  |  |
| * 1. Do you require a specific QOS tagging for Signalling | Yes |  |
| * 1. Do you require a specific QOS tagging for Media (RTP) | Yes |  |
| * 1. What kind of E2E QoS Management do you have in place |  | |
| 1. Do you support IPV6? | Yes | No |
| 1. Do you have any limitation in regard of port range you support? | Yes | No |
| * 1. If yes, please listen the port range Media:   Signalling: |  | |
| 1. Is media routed to the same address as the SIP signalling | Yes | No |
| 1. Do you terminate the payload over your media gateway? | Yes | No |
| 1. Do you support SIP trunk connectivity directly over the public Internet (vs managed circuit such as MPLS and metro ethernet).    1. Which other types of access presentation do you support (e.ge MPLS, Ethernet)? | Yes | No |
| Additional notes/remarks: | | | |

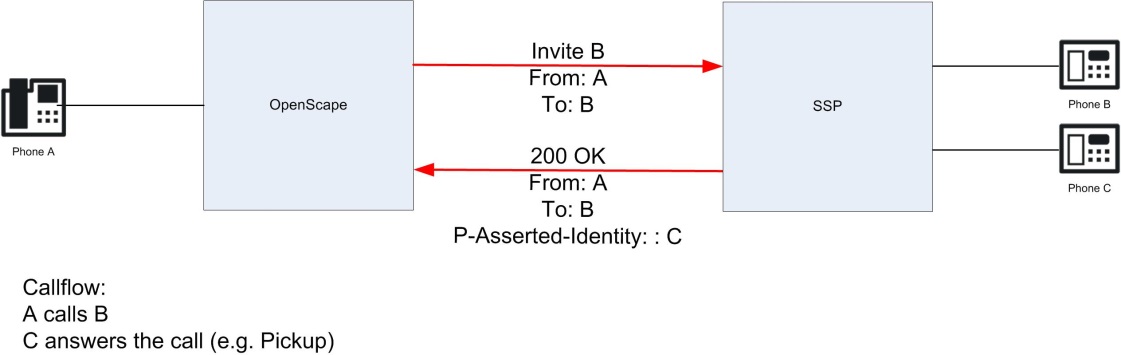
|  |  |  |
| --- | --- | --- |
| Question | Answer |  |
| Multi customer / Multi Site |  | |
| 1. Do you support peering through your network with multiple customer domains as a SIP Proxy? | Yes | No |
| 1. If multiple customer domain peering is not supported, does your access gateway support an SBC terminating signaling and media flows? | Yes | No |
| 1. If multiple registrations from a single IP address are not supported: Please prove us with the information for which purpose a dedicated IP-address (or port) is needed (e,g. CDR, emergency location….) |  | |
| Additional notes/remarks: | | | |

|  |  |  |
| --- | --- | --- |
| Question | Answer |  |
| Redundancy / Failover |  | |
| 1. Redundancy / Fail-over How do you support fail-over of SIP trunks from one Data Center to a redundant Data Center, which in general cannot be connected via L2. | DNS-SRV  Active-Standby SIP Trunks | |
| 1. Does your interface support redundancy? If so how is it supported | transparent VIP failover  peer failover detection | |
| 1. Does your interface support high availability (HA)? | multiple servers addressed  SIP load balancer | |
| * 1. If HA SIP load balancer used, is it transparent or operate at Layer 7 If Layer 7, is it using SIP 3xx redirection | Yes | No |
| Security |  | |
| 1. Do you take measures to prohibit or restrict **toll fraud**: e.g. monitor for abuse, limit the costs over a period, restrict premium lines ...    1. If yes, please describe | Yes | No |
| Additional notes/remarks: | | | |

# Background information / Feature descripiton

**COLP / TIP supported**

In ISDN the feature COLP (Connect Line Identification Presentation) was introduced. In SIP this feature is sometimes referred to as TIP (Termination Identification Presentation).



RFC3324(section 5.) defines a mechanism to transport the identity of the accepting party (C) in the P-Asserted-Identity header field of the 200 OK response.

# History of Change

|  |  |  |
| --- | --- | --- |
| Date | Author |  |
|  |  |  |
|  |  |  |
|  |  |  |
|  |  |  |
|  |  |  |