SIPTeleport FAQ

1. What is Subspace?

Subspace is a parallel internet, built alongside the traditional public internet, designed and optimized for real-time performance. It is secure, and dedicated to accelerating real-time applications by delivering lower latency, less jitter, and better overall customer experiences.

We currently serve more than 400M users and some of the largest companies in the world. Learn more about Subspace by visiting https://www.subspace.com.

2. What is SIPTeleport?

SIPTelepot enables superior voice quality for remote call center agents, while improving customer support experiences. It is a globally deployed, stateful SIP proxy that sits between a company's telecom infrastructure and the user (or contact center agent). SIPTeleport takes advantage of all of Subspace's inherent capabilities: active weather-mapping for optimal path selection, rerouting mid-call to avoid congested paths, inline DDOS protection without affecting performance.

These capabilities allow SIPTeleport to securely move traffic via Subspace to/from the user and the PBX, while maintaining the most stable and reliable performance.

Learn more about SIPTeleport by visiting https://subspace.com/product/sipteleport.

3. What are the benefits of using SIPTeleport?

SIPTeleport offers a global, fast and stable network for RTP, RTCP, SRTP, and SRTCP media traffic allowing you to maximize network (internet) performance. The service has built-in security that protects your traffic from attacks, such as DDoS. With these benefits, you can easily grow your contact center with at-home agents, while lowering your overall cost of operations.

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4. Do I need to have a Session Border Controller to enable SIPTeleport?

No, however NAT traversal for media will need to be configured and handled by your voice server.

5. Can I allow my SIP client to handle NAT Traversal when using SIPTeleport?

No. Client side NAT traversal should be disabled (i.e. STUN/TURN).

6. What SIP clients are supported by SIPTeleport?

Most open source SIP clients will work (Linphone, Zoiper, etc.). Our team continues to test with various SIP clients (both soft and hard). If you would like to have a specific SIP client tested, please contact us via https://subspace.com/contact.

7. Can I use SIPTeleport if I am using a cloud UCaaS and/or CCaaS provider?

Yes, SIPTeleport has been tested with cloud providers such as Vonage and Flowroute. If you would like to have SIPTeleport tested with a specific cloud UC or CC provider, please contact us via https://subspace.com/contact.

8. Will SIPTeleport support calls going through a VPN?

No. Calls must NOT go through a VPN. If a VPN is being utilized by your users, SIP and RTP traffic will need to be filtered to not go through the VPN.

9. Does Subspace leverage anycast?

Subspace has improved anycast's resiliency. Under normal anycast, if a server fails, and the route to it is withdrawn from BGP, then BGP will reconverge to the next-geographically closest server. This next server lacks any context about the pre-existing connection, causing a reset and requiring the client to establish a new connection.

In the rare event of a Subspace PoP failure, all Subspace PoPs share network state information with each other, so a PoP failure cannot stop even a pre-existing connection. Instead, the moment a PoP fails and its BGP announcements are withdrawn, BGP reconverges to send traffic to the next-geographically closest PoP as expected.

That PoP, since it already knows the appropriate IP and port and forwarding information, accepts the next packet in the stream, and forwards it on appropriately. So it is as though there was only a brief pause in the connection stream.

10. Which transport protocols are supported?

TCP, UDP, and TLS are supported.

11. Are both voice and video supported?

Yes.

12. Does my traffic need specific transcoding to work with SIPTeleport?

SIPTeleport has no impact on audio codecs and therefore, any pre-existing transcoding is supported.

13. How many concurrent calls does SIPTeleport support?

It depends on the subscribed service plan. Our free plan is limited to two concurrent calls; however, our Developer, Business and Enterprise tiers all offer unlimited concurrent calls.

Learn more at https://www.subspace.com/pricing/sipteleport.

14. Where can I learn more about Subspace and/or SIPTeleport?

Here are some useful links:

- https://www.subspace.com
- https://subspace.com/resources
- https://subspace.com/api
- https://subspace.com/product/sipteleport
- https://subspace.com/pricing/sipteleport

15. Can I try SIPTeleport for free?

Yes, sign up at https://console.subspace.com/try or visit https://www.subspace.com to get started.