SIPTeleport with SIP Client Quickstart

The [SIPTeleport] Quickstart is here to help you get started and create your first accelerator for your real-time application. This guide illustrates how to enable Subspace with your SIP client using the 'Telephone' SIP Client by *64 Characters* as an integration example.

SIPTeleport works both by acting as a SIP proxy between your SIP client and SIP server as well as accelerating the calls between your SIP clients by sending their traffic over the accelerated Subspace network.

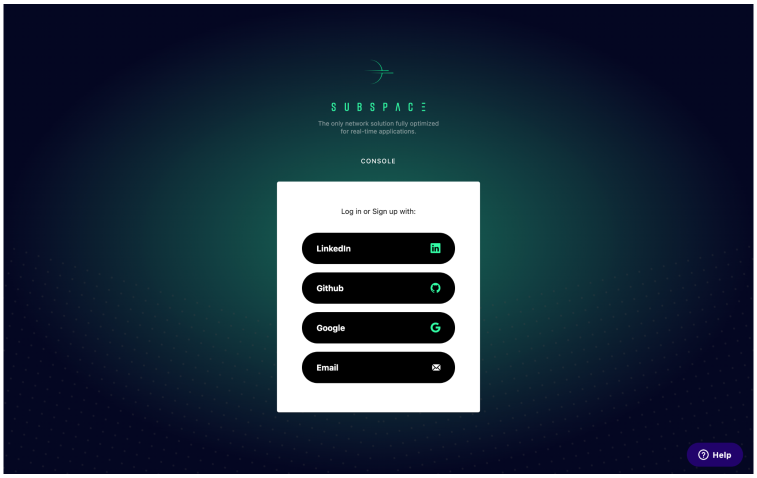
Learn more about SIPTeleport and Subspace by visiting: https://www.subspace.com.

Quickstart with SIP Client Setup

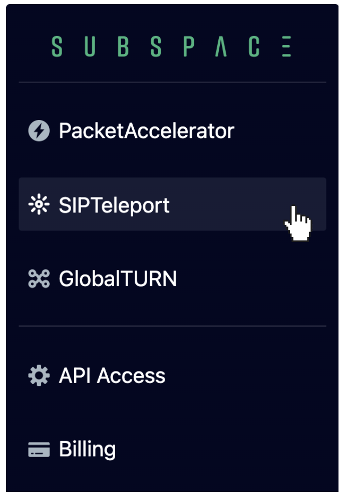
1. Gather the following information from your SIP provider:

   a. Name
   b. FQDN (this determines your destination SIP)
   c. User Name
   d. Password
   e. SIP port, if known (if unknown, use default of '5060' for UDP or TCP connections and '5061' for TLS)

2. Log in or Sign up on the Subspace console at: https://console.subspace.com.



3. In the navigation menu on the left, click on the link for “SIPTeleport.”

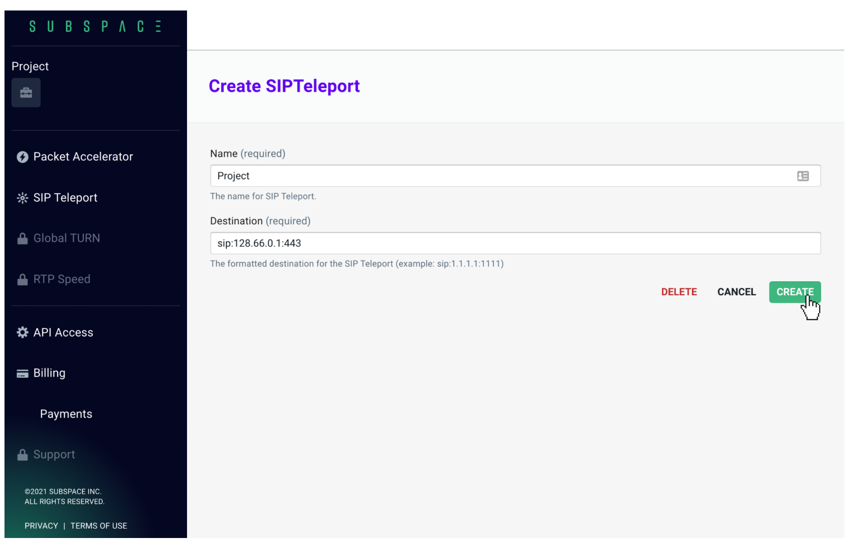


4. In the “SIPTeleport” screen, click on the + ADD button.

5. Fill in the needed details to create a new SIPTeleport. Give it a unique **name**, and the **destination SIP address** and **port** your sip provider gave you (if known).

   The SIP address may be in the form of a domain name, such as “subspace.com”. If the port was not given (and you connect to your SIP provider via a UDP or TCP connection, use port ‘5060,’ otherwise leave the field blank. If your SIP provider uses TLS, use port ‘5061.’

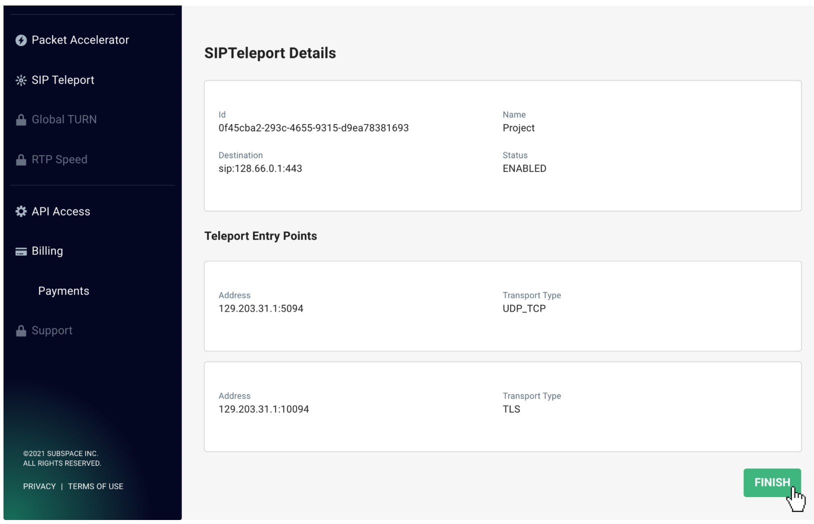
   Then click CREATE to create a new SIPTeleport.



6. On creation, Subspace will return all the details you need for your new SIPTeleport. SIPTeleport instantiation is immediate, it's set up and ready to use right away.

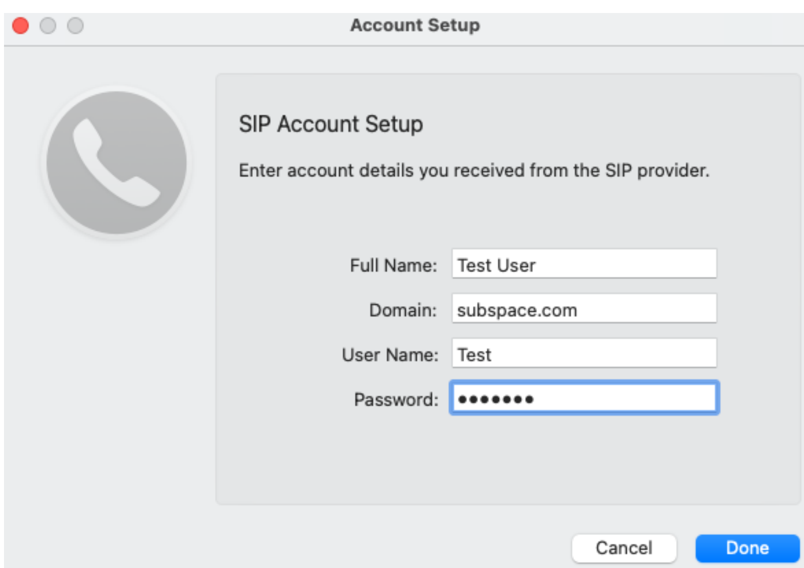
   Be sure to note the Teleport Entry Points, you will need them to set up your SIP client.

   Then, click **FINISH** .

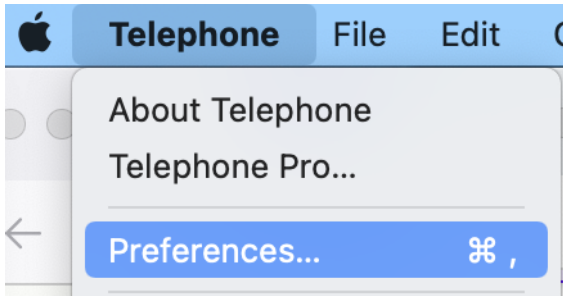


7. Next, open up your SIP client. For this quickstart, we'll be detailing an integration with the 'Telephone' SIP client by *64 Characters*.

   Enter the account details obtained from your SIP provider. This includes a **user name**, a **domain name**, a **password**, and your own **name**.



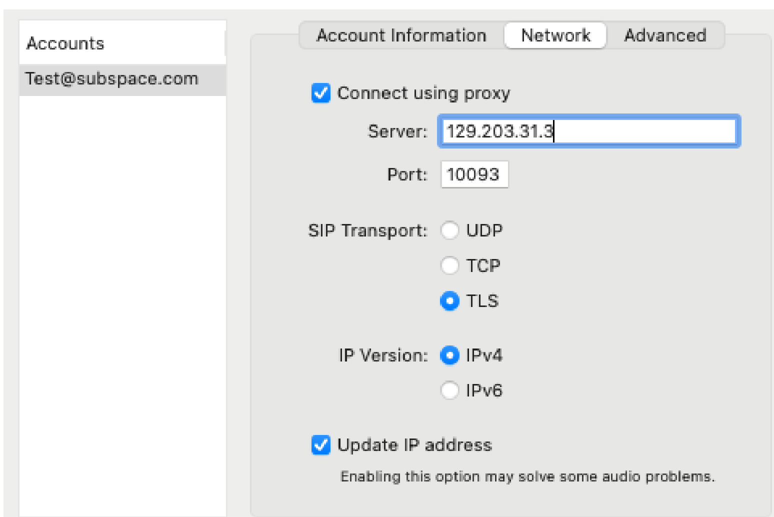
8. Select your client's "preferences" or "setting" options.



9. Using the Teleport Entry Points returned when the SIPTeleport was created, set your client to connect using a proxy.

Subspace recommends connecting using TLS. To do this, copy the IP address and port from the Teleport Entry Point whose Transport Type is "TLS" as the Server. Then select "TLS."

If your SIP provider only supports UDP or TCP, please select those instead, and use the Teleport Entry Point associated with that protocol.



10. Now, any call made via your SIP client will first contact Subspace SIPTeleport, and utilize it as an outbound proxy to talk with your SIP service's server, including sending RTP connections over Subspace's network for lower latency, lower jitter, and simply a better and more reliable call from anywhere in the world.

If you have additional questions, feel free to reach out via support@subspace.com.