

Twilio Elastic SIP Trunking – Asterisk Configuration Guide

This configuration guide is intended to help you provision your Twilio Elastic SIP Trunk to communicate with Asterisk, an open source communication server.

Please be aware, due to different versions of Asterisk, the settings you see may differ from those shown in this configuration guide. This configuration guide was created using Asterisk 15.50 with `chan_pjsip`. This document is intended to provide general guidelines for setting up your Twilio Elastic SIP Trunk and not as a comprehensive configuration template for Asterisk.

Twilio does not provide direct support for third-party products; you should reach out to the applicable vendor support options for assistance configuring their products.

Twilio Elastic SIP Trunk Provisioning

We begin by provisioning the Twilio Elastic SIP Trunk through the Twilio Console following the steps below. These steps assume there are no existing SIP Trunks provisioned.

Our examples will use the Twilio US1 (Virginia) data center as the primary data center for call Termination and Origination and the Twilio US2 (Oregon) data center as the secondary data center for call Termination and Origination. You should modify the Termination and Origination URI's based on the data centers that are physically closest to your location. A list of the regional termination and origination URI's can be found on our SIP Trunking landing page:

Termination: <https://www.twilio.com/docs/sip-trunking#termination>

Origination: <https://www.twilio.com/docs/sip-trunking#origination>

Once you decide on the Twilio datacenters and are logged into the Twilio Console, you can visit the URL below to determine what signaling and media IP addresses and ports will be used, so you can restrict your network and/or host running Asterisk to just those IP addresses/ports:

Networking Information

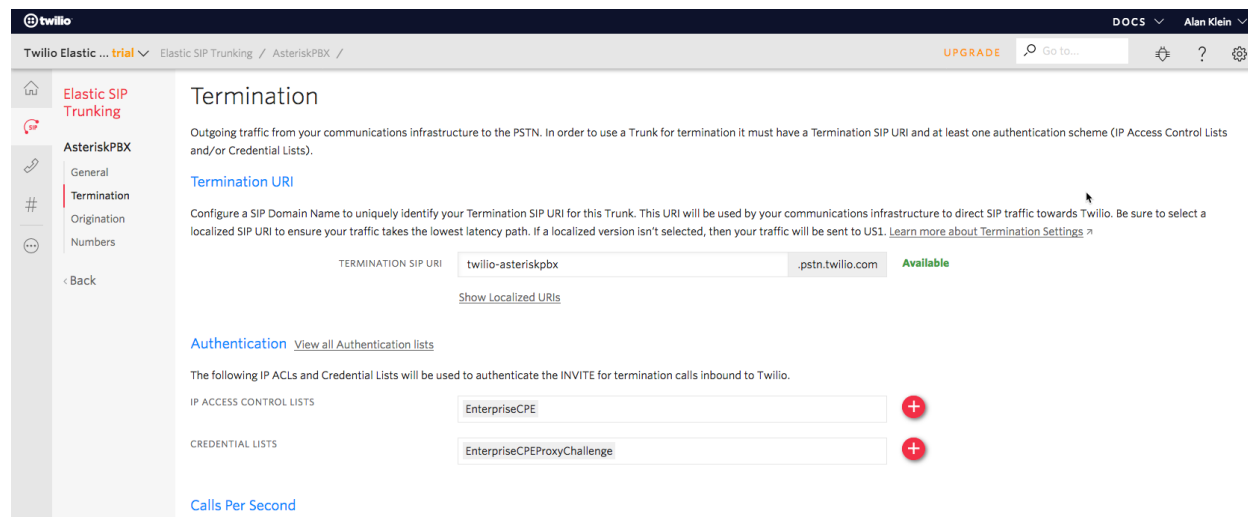
<https://www.twilio.com/console/sip-trunking/your-network>

Note: Media Ports 10000-20000 use UDP

1. Log into the Twilio Console - <https://www.twilio.com/login>.
2. From the main Dashboard on the left side of the screen, select (...) for All Products & Services.
3. Scroll down to Elastic SIP Trunking and click it.
4. From the Elastic SIP Trunking Dashboard, click the "Getting Started" button.
5. From the Getting Started with Elastic SIP Trunking page, Click the "Create a SIP Trunk".
6. Enter a Friendly Name. For the configuration guide, I used "AsteriskPBX". Click Create.
7. On the "AsteriskPBX" General Settings Screen, note the settings that impact this specific trunk.

****Configuring Your Termination URI (From AsteriskPBX > Twilio)****

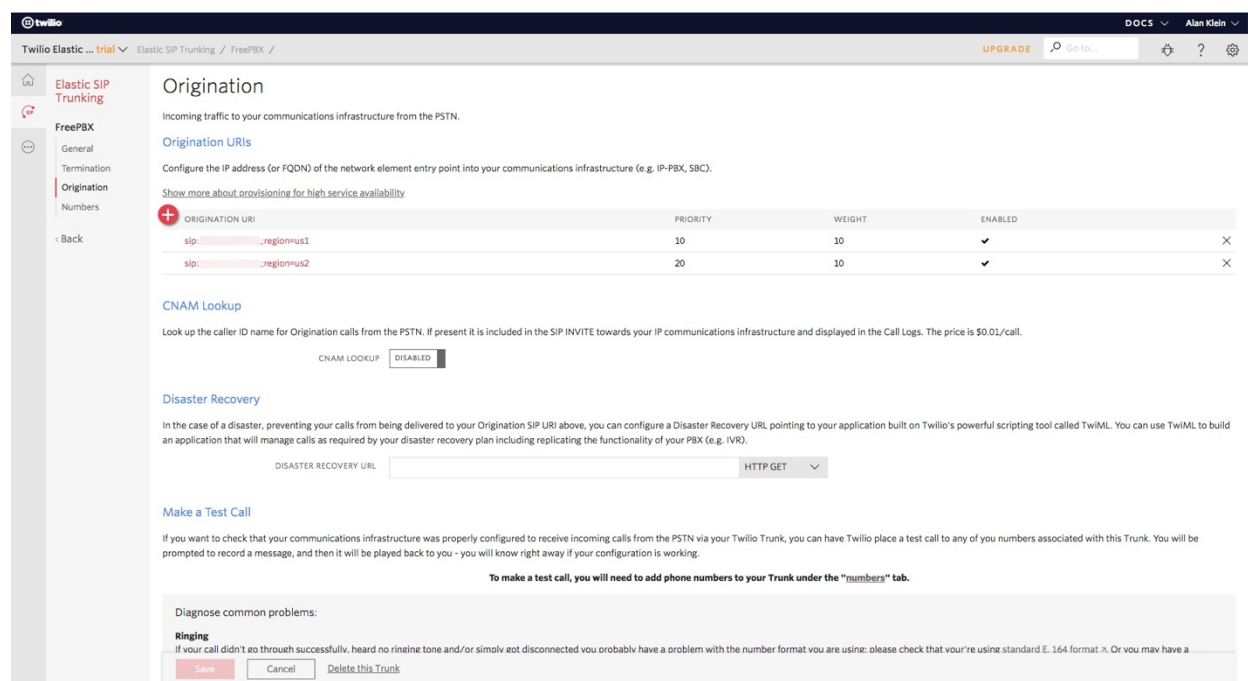
8. On the left side of the screen under your "AsteriskPBX" Trunk Name, Choose Termination.
9. Note, that in order to use a Trunk for termination (AsteriskPBX > Twilio) it must have a Termination SIP URI and at least one authentication scheme (IP Access Control Lists and/or Credential Lists). We will provision both authentication schemes.
10. In the Termination SIP URI edit box, enter a unique termination URI (we use "twilio-asteriskpbx" aka twilio-asteriskpbx.pstn.twilio.com), on the far-right, you will be notified if the name is unique and available when you click outside the edit box.
11. Click the red (+) next to IP Access Control List to create a new IP Access Control List that will restrict requests to your Termination SIP URI.
12. In the "New Access Control List" (ACL) provide a Friendly Name for the ACL, I used "EnterpriseCPE". Under this list, you can have multiple IP addresses you allow to communicate with your SIP termination URI. Enter an IP address and a Friendly Name for that IP address, in my case "AtlantaAsteriskPBX" and click Create ACL.
13. You should receive confirmation the ACL was successfully created and see the Friendly Name, in my case "EnterpriseCPE", selected in the IP Access Control Lists.
14. Click the red (+) next to Credential List to create a new Credential List that will challenge requests to your Termination SIP URI.
15. In the "New Credential List" provide a Friendly Name for the Credential List, I used "EnterpriseCPEProxyChallenge". Under this list, you can have multiple credentials that allow communications with your SIP termination URI. Enter a username, in my case "myasteriskpbx" and a password, in my case "myasteriskpbxzx11%VzX" (pick a unique username and complex password, It must have a minimum length of 12 characters, at least one number, uppercase character and lowercase character) and click Create.
16. You should receive confirmation you have successfully created a new Credential List and see the Credential List name, in my case, "EnterpriseCPEProxyChallenge" selected in the Credential Lists.
17. Scroll down to the bottom, and Click Save.
18. You should receive confirmation you have successfully updated your SIP Trunk.



The screenshot shows the Twilio Elastic SIP Trunking configuration interface. The page title is "Termination". Below the title, there is a descriptive paragraph: "Outgoing traffic from your communications infrastructure to the PSTN. In order to use a Trunk for termination it must have a Termination SIP URI and at least one authentication scheme (IP Access Control Lists and/or Credential Lists)." The "Termination URI" section contains a text input field with the value "twilio-asteriskpbx.pstn.twilio.com" and a green "Available" status indicator. Below this is a "Show Localized URIs" link. The "Authentication" section includes a link to "View all Authentication lists" and a note: "The following IP ACLs and Credential Lists will be used to authenticate the INVITE for termination calls inbound to Twilio." Underneath, there are two lists: "IP ACCESS CONTROL LISTS" with one entry "EnterpriseCPE" and "CREDENTIAL LISTS" with one entry "EnterpriseCPEProxyChallenge". Each entry has a red plus sign icon to its right. At the bottom left, there is a "Calls Per Second" section.

****Configuring Your Origination URI's (From Twilio > AsteriskPBX)****

19. On the left side of the screen under your "AsteriskPBX" Trunk Name, Choose Origination.
20. Click the "Add new Origination URI" button, where we will define how calls are sent from Twilio to your AsteriskPBX.
21. For the Origination SIP URI edit box, enter the format (without quotes and with your unique public IP address): "sip:203.0.113.254;region=us1" with a priority of 10 and a weight of 10. This will originate all SIP Traffic from the Twilio US1 (Virginia) data center to your AsteriskPBX. Click Add.
22. Click the (+) next to Origination URI, to add a secondary Origination URI, should the primary encounter issues.
23. For the Origination SIP URI edit box, enter the format (without quotes and with your unique public IP address): "sip:203.0.113.254;region=us2" with a priority of 20 and a weight of 10. This will originate SIP Traffic from the Twilio US2 (Oregon) data center to your AsteriskPBX, only if the US1 Virginia data center is unable to deliver the call. Click Add.



****Assigning Telephone Numbers To Your Elastic SIP Trunk****

24. On the left side of the screen under your "AsteriskPBX" Trunk Name, Choose Numbers.
25. Click the "Buy a Number" button.
26. Search for a number that meets your needs.
27. Click Buy next to the number.
28. Click Buy This Number to confirm the purchase.
29. You will receive a confirmation dialog the number was purchased, click close.
30. Your purchased number is now associated with your SIP Trunk.

**** This completes all the Twilio Specific Provisioning Steps ****

Asterisk Provisioning

This configuration guide was created using:
Asterisk 15.50

Example Configuration files (located in the default directory: /etc/asterisk).

BOLDED items will be unique settings for your environment.

extension.conf

```
[from-pstn]
exten => _+1NXXXXXXXXX,1,Dial(PJSIP/233)

[from-internal]
exten => _NXXNXXXXXXXX,1,Set(CALLERID(all)="Alan" <+13213513261>)
same => n,Dial(PJSIP/+1${EXTEN}@twilio-na-us)
same => n(end),Hangup()
```

pjsip.conf

```
[global]
type=global

[0.0.0.0-udp]
type=transport
protocol=udp
bind=0.0.0.0:5060
external_media_address=x.x.x.x
external_signaling_address= x.x.x.x
allow_reload=no
tos=cs3
cos=3
local_net=x.x.x.x/24

[0.0.0.0-tls]
type=transport
protocol=tls
bind=0.0.0.0:5061
external_media_address=x.x.x.x
external_signaling_address=x.x.x.x
ca_list_file=/etc/asterisk/keys/ca.crt
cert_file=/etc/asterisk/keys/server.crt
priv_key_file=/etc/asterisk/keys/server.key
method=tlsv1
verify_client=no
```

```
verify_server=no
allow_reload=no
tos=cs3
cos=3
local_net=x.x.x.x/24
```

pjsip_wizard.conf

<https://wiki.asterisk.org/wiki/display/AST/PJSIP+Configuration+Wizard>

```
[user_defaults](!)
type = wizard
accepts_registrations = yes
sends_registrations = no
accepts_auth = yes
sends_auth = no
endpoint/context = from-internal
endpoint/tos_audio=ef
endpoint/tos_video=af41
endpoint/cos_audio=5
endpoint/cos_video=4
endpoint/allow = !all,ulaw
endpoint/dtmf_mode = rfc4733
endpoint/aggregate_mwi = yes
endpoint/use_avpf = no
endpoint/rtcp_mux = no
endpoint/bundle = no
endpoint/ice_support = no
endpoint/media_use_received_transport = no
endpoint/trust_id_inbound = yes
endpoint/media_encryption = no
endpoint/timers = yes
endpoint/media_encryption_optimistic = no
endpoint/send_pai = yes
endpoint/rtp_symmetric = yes
endpoint/rewrite_contact = yes
endpoint/force_rport = yes
endpoint/language = en
```

```
[233] (user_defaults)
aor/max_contacts = 2
endpoint/callerid = Alan Klein <233>
inbound_auth/username = 233
inbound_auth/password = aef9f02938bfd5a5321ad83b9599f25d
```

```
[trunk_defaults](!)
type = wizard
endpoint/transport=0.0.0.0-udp
endpoint/allow = !all,ulaw
endpoint/t38_udptl=no
endpoint/t38_udptl_ec=none
endpoint/fax_detect=no
endpoint/trust_id_inbound=no
endpoint/t38_udptl_nat=no
endpoint/direct_media=no
endpoint/rewrite_contact=yes
endpoint/rtp_symmetric=yes
endpoint/dtmf_mode=rfc4733
endpoint/allow_subscribe = no
aor/qualify_frequency = 60
```

```
[twilio-na-us](trunk_defaults)
sends_auth = yes
sends_registrations = no
remote_hosts = twilio-asteriskpbx.pstn.us1.twilio.com,twilio-
asteriskpbx.pstn.us2.twilio.com
outbound_auth/username = myasteriskpbx
outbound_auth/password = myasteriskpbx11%VzX
endpoint/context = from-pstn
aor/qualify_frequency = 60
```

Troubleshooting

Twilio Troubleshooting

The most comprehensive page for Twilio specific Elastic SIP Trunk Troubleshooting is our Troubleshooting landing page, <https://www.twilio.com/docs/sip-trunking/troubleshooting>.

You can look at the packet captures between Twilio and Asterisk by going to your Elastic SIP Trunking call logs, <https://www.twilio.com/console/sip-trunking/logs/calls>, clicking on a specific call date, and then on the right, clicking the download link next to SIP PCAP Log. This packet capture is in the common .pcap format. Note that for SIP/TLS .pcap files will be blank.

Some of the common issues encountered:

1. The IP addresses and ports used for SIP Signaling and Media must be whitelisted so they can pass through to Asterisk. The most up to date list of IP addresses and ports can be found on the Twilio Console Elastic SIP Trunking Networking Information - <https://www.twilio.com/console/sip-trunking/your-network>. The ;region parameter for the Origination URI restricts the IP addresses to those regions defined for your origination URI's. A common symptom of a firewall issue is viewable in the .pcap file where Twilio sends SIP INVITES to your origination URI's with no responses from Asterisk.
2. The From/To/Request-URI/P-Asserted Identity should be in E.164 Format. What is E.164? <https://www.twilio.com/docs/glossary/what-e164>
3. Twilio Trial accounts have limits which may impact how Elastic SIP Trunking is used.

Trial Accounts have a maximum of:

- 1 unique SIP trunk
- 1 origination phone number
- 4 concurrent calls
- 5 verified CallerID's

Trial accounts may only place calls TO and FROM verified numbers. A purchased Twilio phone number also qualifies as a verified number you can use as the caller ID.

4. Check the Twilio Debugger, <https://www.twilio.com/console/runtime/debugger?quickDate=24>, from the Twilio Console, to see if a specific error message appears which may be preventing the call.

Asterisk Troubleshooting

Helpful Asterisk CLI commands

```
core show help pjsip
pjsip show settings
pjsip show version
pjsip show identifies
pjsip show endpoints
pjsip show contacts
pjsip show transports
pjsip show auths
pjsip show aors
pjsip show contacts
pjsip show channels
```

Asterisk PJSIP Troubleshooting (bold text enables SIP messaging in Asterisk CLI)

Asterisk CLI (from the Bash Shell):

```
asterisk -vvvr
```

Enable:

- core set verbose 4
- core set debug 4
- **pjsip set logger on**

Disable:

- core set verbose 3
- core set debug 0
- pjsip set logger off

When using SIP/TLS with pjsip, the

`priv_key_file=/etc/asterisk/keys/server.key` entry in `pjsip.conf` is critical for proper SIP/TLS initialization. The private key must match the certificate created with `ast_tls_cert` or the alternate method you use to create the certificate signing request.

SIP TLS/SRTP – Addendum

Caution: These changes will impact call origination and termination until the full set-up is completed and verified.

****SETTING UP TWILIO SIDE****

1. Log into the Twilio Console - <https://www.twilio.com/login>
2. From the main Dashboard on the left side of the screen, select (...) for All Products & Services
3. Scroll down to Elastic SIP Trunking and click it
4. Select Trunks
5. Click on your SIP Trunk (my Example used: AsteriskPBX)
6. Under Secure Trunking, Toggle the Slider to Enabled and Click Save
7. Click Origination, and modify the origination URI's to use TLS by clicking on each entry and making the following change (adjusting the IP address to match your configuration):

Current: sip:203.0.113.254;region=us1
New: sip:203.0.113.254;transport=tls;region=us1

Current: sip:203.0.113.254;region=us2
New: sip:203.0.113.254;transport=tls;region=us2

****SETTING UP Asterisk SIDE****

The **BLUE** text below details the changes to the initial pjsip_wizard.conf file to enable SIP/TLS with SRTP.

For this configuration guide, we used the process detailed at: <https://wiki.asterisk.org/wiki/display/AST/Secure+Calling+Tutorial>, to create the necessary certificate files to enable SIP/TLS. Note the SIP/TLS comment in the troubleshooting section above.

Download the Twilio Root CA Certificates:

Importing Twilio's Root CA Certificates

<https://www.twilio.com/docs/sip-trunking#rootCA>

Right Click Save As Link: Download Twilio's CA certificate (name: cacert.pem)

Save the file to: /etc/asterisk/keys on your FreePBX server

- Note: This configuration note uses the ca.crt file created in the Secure+Calling+Tutorial above, however this file is not immediately used as verify_server=no in pjsip.conf.

Example Configuration files (located in the default directory: /etc/asterisk)

pjsip.conf

```
[global]
type=global

[0.0.0.0-udp]
type=transport
protocol=udp
bind=0.0.0.0:5060
external_media_address=x.x.x.x
external_signaling_address= x.x.x.x
allow_reload=no
tos=cs3
cos=3
local_net=x.x.x.x/24

[0.0.0.0-tls]
type=transport
protocol=tls
bind=0.0.0.0:5061
external_media_address=x.x.x.x
external_signaling_address=x.x.x.x
ca_list_file=/etc/asterisk/keys/ca.crt
cert_file=/etc/asterisk/keys/server.crt
priv_key_file=/etc/asterisk/keys/server.key
method=tlsv1
verify_client=no
verify_server=no
allow_reload=no
tos=cs3
cos=3
local_net=x.x.x.x/24
```

pjsip_wizard.conf

```
[user_defaults](!)
type = wizard
accepts_registrations = yes
sends_registrations = no
accepts_auth = yes
sends_auth = no
endpoint/context = from-internal
endpoint/tos_audio=ef
endpoint/tos_video=af41
```

```
endpoint/cos_audio=5
endpoint/cos_video=4
endpoint/allow = !all,ulaw
endpoint/dtmf_mode = rfc4733
endpoint/aggregate_mwi = yes
endpoint/use_avpf = no
endpoint/rtcp_mux = no
endpoint/bundle = no
endpoint/ice_support = no
endpoint/media_use_received_transport = no
endpoint/trust_id_inbound = yes
endpoint/media_encryption = no
endpoint/timers = yes
endpoint/media_encryption_optimistic = no
endpoint/send_pai = yes
endpoint/rtp_symmetric = yes
endpoint/rewrite_contact = yes
endpoint/force_rport = yes
endpoint/language = en
```

[233] (user_defaults)

```
aor/max_contacts = 2
endpoint/callerid = Alan Klein <233>
inbound_auth/username = 233
inbound_auth/password = aef9f02938bfd5a5321ad83b9599f25d
```

[trunk_defaults] (!)

```
type = wizard
endpoint/transport=0.0.0.0-tls
endpoint/allow = !all,ulaw
endpoint/t38_udptl=no
endpoint/t38_udptl_ec=none
endpoint/fax_detect=no
endpoint/trust_id_inbound=no
endpoint/t38_udptl_nat=no
endpoint/direct_media=no
endpoint/rewrite_contact=yes
endpoint/rtp_symmetric=yes
endpoint/dtmf_mode=rfc4733
endpoint/allow_subscribe = no
aor/qualify_frequency = 60
endpoint/media_encryption = sdes
```

```
[twilio-na-us] (trunk_defaults)
sends_auth = yes
sends_registrations = no
remote_hosts = twilio-
asteriskpbx.pstn.us1.twilio.com:5061, twilio-
asteriskpbx.pstn.us2.twilio.com:5061
outbound_auth/username = myasteriskpbx
outbound_auth/password = myfreepbxzx11%VzX
endpoint/context = from-pstn
aor/qualify_frequency = 60
```

Note, you will see the following entries in your log file and the Asterisk CLI. Twilio uses wild card certificates. Even though this log entry appears, it will not impact call processing if verify server is set to no.

```
ERROR[3857]: pjproject:0 <?>:      tlsc0x7f5b6033cd38 RFC 5922 (section 7.2) does not
allow TLS wildcard certificates. Advise your SIP provider, please!
```

Relevant Twilio Documentation

Twilio Elastic SIP Trunking

<https://www.twilio.com/docs/sip-trunking>

SIP Trunking Configuration Guides

<https://www.twilio.com/docs/sip-trunking/sample-configuration>

Troubleshooting Your SIP Trunk

<https://www.twilio.com/docs/sip-trunking/troubleshooting>

Twilio SIP Trunking Scale and Limits

<https://www.twilio.com/docs/sip-trunking/scale-and-limits>

Elastic SIP Trunking Emergency Calling

<https://www.twilio.com/docs/sip-trunking/emergency-calling>

Calls Per Second (CPS) - Trunking Termination

<https://www.twilio.com/docs/sip-trunking/cps-trunk-termination>

Configuring Your Trunking Using Twilio Interconnect

<https://www.twilio.com/docs/sip-trunking/configure-with-interconnect>

Elastic SIP Trunking: Region selection for origination traffic

<https://www.twilio.com/blog/2017/07/sip-trunking-origination-traffic.html>

On-demand pcaps – Greater Visibility into Elastic SIP Trunking

<https://www.twilio.com/blog/2015/12/pcaps.html>

Announcing Secure Trunking

<https://www.twilio.com/blog/2015/09/secure-trunks.html>