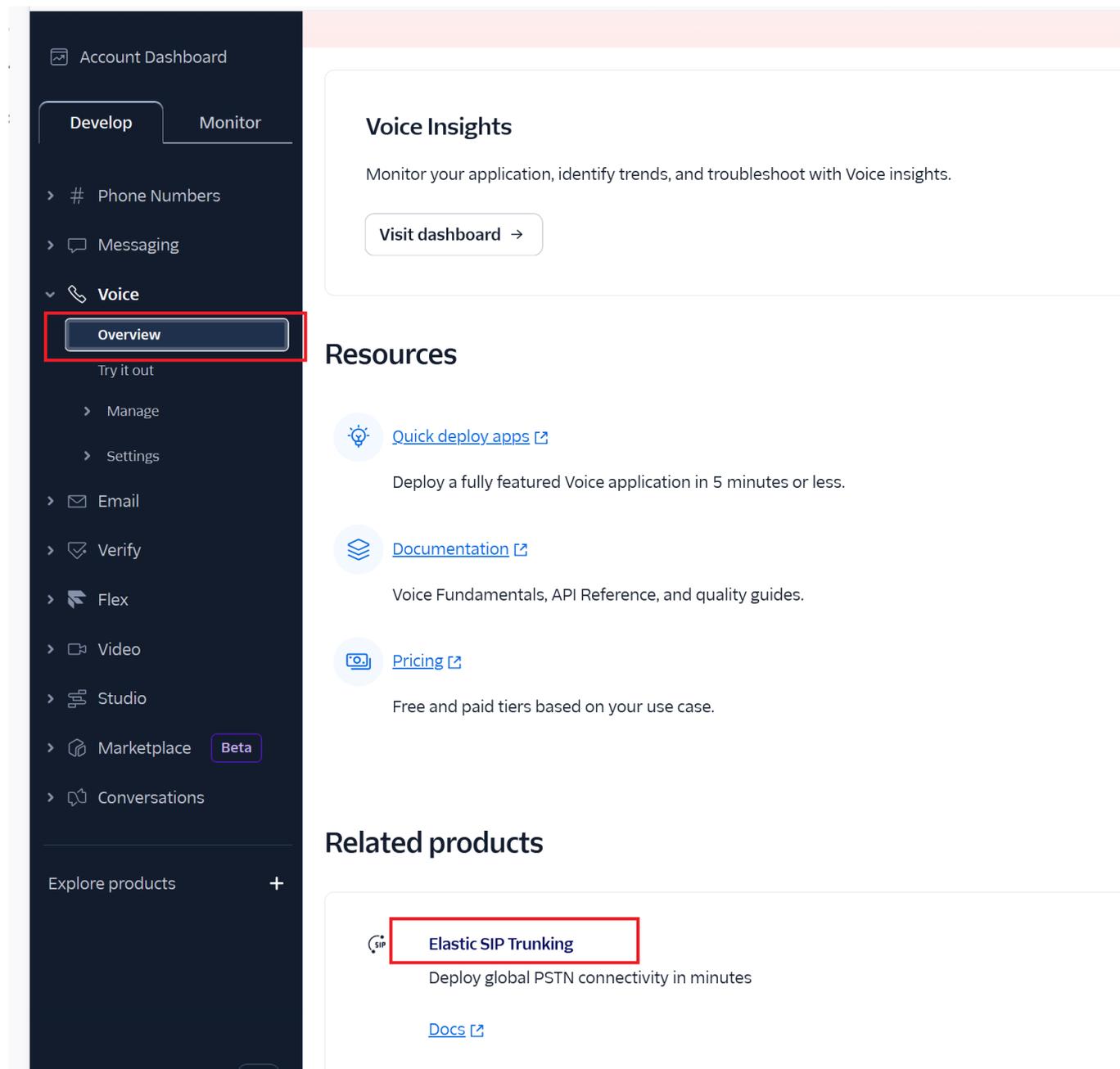


# Use SIP Trunk / VoIP Phone Service

## How to config Ecsow Dialer / Voice Broadcast to use Twilio?

A. You can follow the below steps if you want to use the twilio **Elastic SIP Trunking** to work with Ecsow dialer.

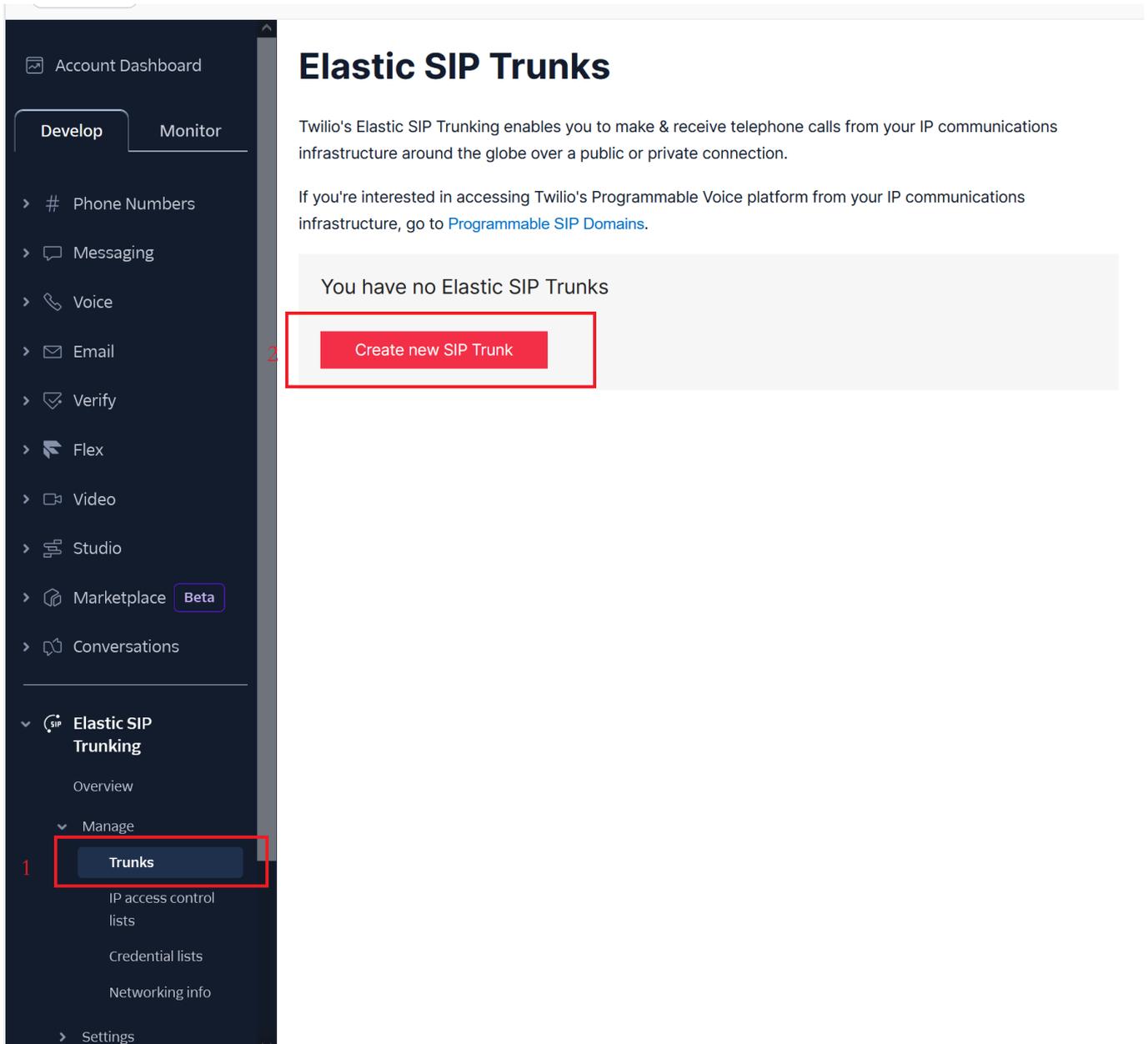
1, Click the “overview” under the “Voice” menu after you login your twilio account. Then click the “Elastic SIP Trunking” link as below.



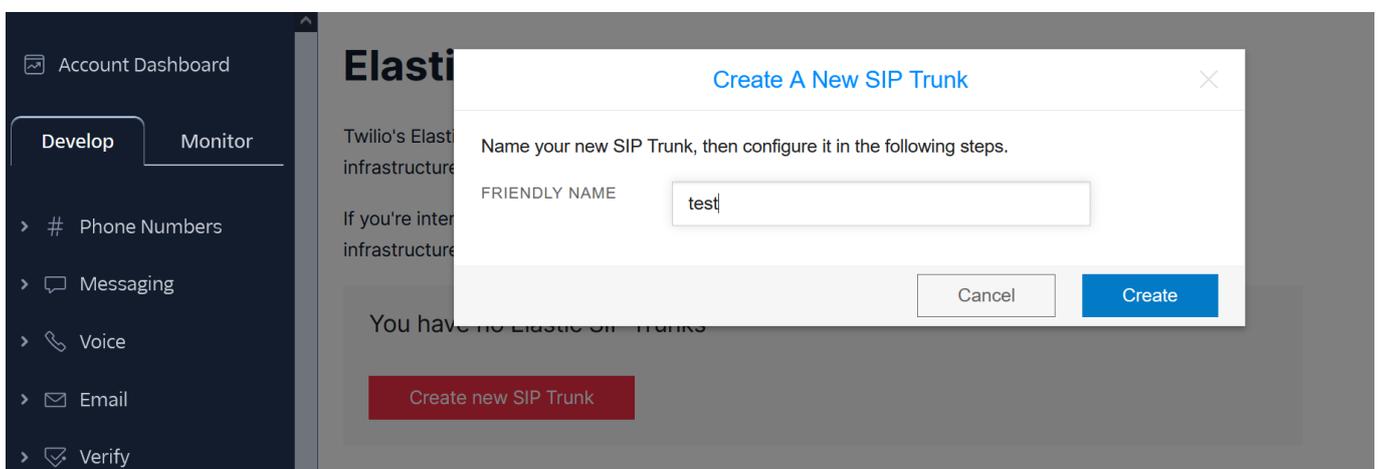
The screenshot displays the Twilio Account Dashboard interface. On the left, a dark sidebar contains a navigation menu with categories 'Develop' and 'Monitor'. Under the 'Voice' category, the 'Overview' option is highlighted with a red rectangular box. The main content area on the right features a 'Voice Insights' section with a 'Visit dashboard' button, a 'Resources' section with links for 'Quick deploy apps', 'Documentation', and 'Pricing', and a 'Related products' section. In the 'Related products' section, the 'Elastic SIP Trunking' product card is highlighted with a red rectangular box. The card includes a SIP icon, the product name, a brief description 'Deploy global PSTN connectivity in minutes', and a 'Docs' link.

2, It will show up the “Elastic SIP Trunking” page as below, then click the “Create new SIP Trunk” button from the “Trunks” page.

# Use SIP Trunk / VoIP Phone Service



3, You need to input a name.



4, By default, it do not need to change anything on General page. Just click the Save button as below.

# Use SIP Trunk / VoIP Phone Service

Account Dashboard

Develop Monitor

Elastic SIP Trunking (US1)

← test

General

Termination

Origination

Numbers

## General Settings

• **Friendly name**

test

A human readable descriptive text, up to 64 characters long.

**Trunk SID**

TKa8c6e58860e374ee3f4d25f85dd25cda

## Features

To learn more about SIP Trunking features, please [see our user documentation](#).

**Call Recording**

Disabled Calls will not be recorded.

**Call Recording**

Record from ringing

**Recording Trim**

Disabled Silence will not be trimmed from recording

**Secure Trunking**

Disabled RTP must be used for media packets. SIP messages may be sent unencrypted or encrypted using SRTP encrypted calls will be rejected

**Call Transfer (SIP REFER)**

Disabled Twilio will reject any incoming SIP REFERs from your communications infrastructure

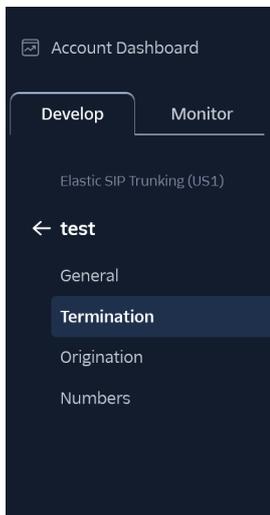
**Caller ID for Transfer Target**

Set caller ID as Transferee

Save Cancel Delete Trunk

5, Click the "Termination" menu to config it. You need to input the Termination SIP URI with any name. This is the domain name for Ecsow dialer.

# Use SIP Trunk / VoIP Phone Service



## Termination

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).

### Termination URI

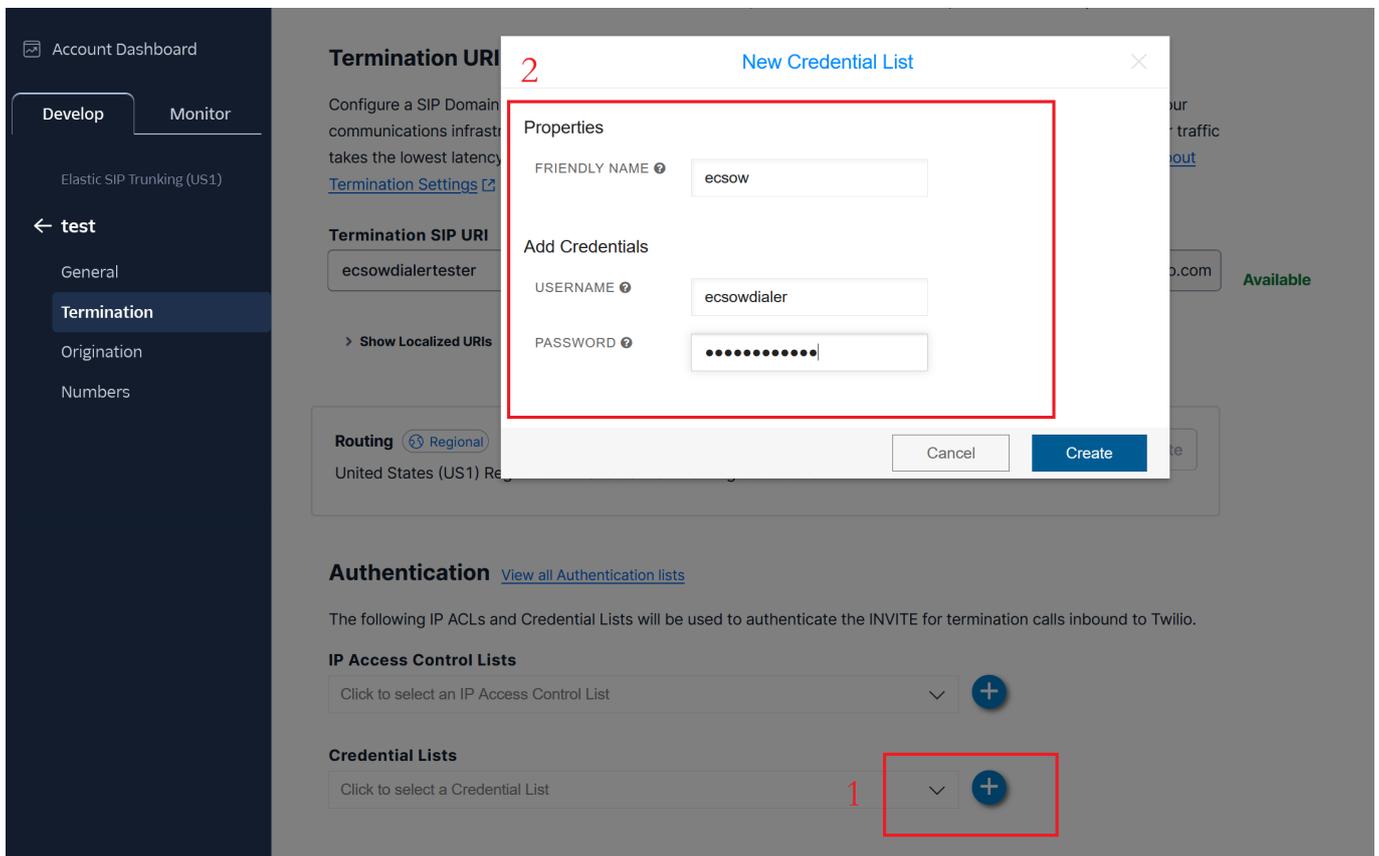
Configure a SIP Domain Name to uniquely identify your Termination SIP URI for this Trunk. This URI will be used by your communications infrastructure to direct SIP traffic towards Twilio. Be sure to select a localized SIP URI to ensure your traffic takes the lowest latency path. If a localized version isn't selected, then your traffic will be sent to US1. [Learn more about Termination Settings](#)

### Termination SIP URI

ecsowdialertester .pstn.twilio.com Available

> Show Localized URIs

6, You need to add Authentication. Click the + icon from Credential Lists option, then input the username and password. Please remember the username and password, it will be used on Ecsow dialer later.



7, Select the Credential you just added as the below image step 1 shows. Then click the "Save" button to save your settings.

# Use SIP Trunk / VoIP Phone Service

Elastic SIP Trunking (US1)

← test

General

**Termination**

Origination

Numbers

### Termination URI

Configure a SIP Domain Name to uniquely identify your Termination SIP URI for this Trunk. This URI will be used by your communications infrastructure to direct SIP traffic towards Twilio. Be sure to select a localized SIP URI to ensure your traffic takes the lowest latency path. If a localized version isn't selected, then your traffic will be sent to US1. [Learn more about Termination Settings](#)

Termination SIP URI

ecsowdialertester .pstn.twilio.com Available

Show Localized URIs

Routing Regional

United States (US1) Region Termination SIP URI routing is: **Inactive** Re-route

### Authentication [View all Authentication lists](#)

The following IP ACLs and Credential Lists will be used to authenticate the INVITE for termination calls inbound to Twilio.

#### IP Access Control Lists

Click to select an IP Access Control List +

#### Credential Lists

ecsow x 1 +

Save Cancel Delete this Trunk

8, You can input your SIP Trunk information on Ecsow dialer as the below image shows. Please enable the “Mark the above SIP account as SIP Trunking” option.

Termination

Origination

Numbers

Account Dashboard

Develop Monitor

Elastic SIP Trunking (US1)

test

General

**Termination**

Origination

Numbers

### Termination URI

Configure a SIP Domain Name to uniquely identify your Termination SIP URI for this Trunk. This URI will be used by your communications infrastructure to direct SIP traffic towards Twilio. Be sure to select a localized SIP URI to ensure your traffic takes the lowest latency path. If a localized version isn't selected, then your traffic will be sent to US1. [Learn more about Termination Settings](#)

Termination SIP URI

ecsowdialertester .pstn.twilio.com Available

Show Localized URIs

Routing Regional

United States (US1) Region Termination SIP URI routing is: **Inactive** Re-route

### Authentication [View all Authentication lists](#)

The following IP ACLs and Credential Lists will be used to authenticate the INVITE for termination calls inbound to Twilio.

#### IP Access Control Lists

Click to select an IP Access Control List +

#### Credential Lists

ecsow x 1 +

### New Credential List

Properties

FRIENDLY NAME ecsow

Add Credentials

USERNAME ecsowdialer

PASSWORD \*\*\*\*\*

Cancel Create

### Options

General SIP Account Do-Not-Call Voice Activity Detection Call Transfer Preset Call Script Advanced

Account information from your SIP VoIP service provider

Display Name twilio

SIP ID/User Name ecsowdialer

Domain ecsowdialertester.pstn.twilio.com

Authorization User ecsowdialer

Password \*\*\*\*\*

Outbound Proxy

Send outbound calls via:

Domain

Proxy

Concurrent Calls

Number of concurrent calls 1

\* SIP service must support multiple channels if you want to run concurrent calls

Mark the above SIP account as SIP Trunking (Such as twilio SIP Trunking)

Advanced Settings

OK Cancel

9, The twilio SIP trunk request the + symbol before the Country code. Please remember to add it.

# Use SIP Trunk / VoIP Phone Service

Options

General SIP Account Do-Not-Call Voice Activity Detection Call Transfer Preset Call Script Advanced

Specify the phone number Country/Region code

Country/Region code:  For example: 1 for USA

Specify the number of seconds to wait for an answer

Wait for answer (seconds):

If the call is failed it can be postponed(redial)

Maximum attempts:

Specify a pause time between calls

Pause seconds between calls:

OK Cancel

Now, you can make a test call to twilio test number (650)489-4546) or (415)475-8378.

**B.** You can follow the below steps if you use the **twilio voice sevice**.

Once your active your sip account on Twilio.com. You will got the following credentials information as the below shows.

Please use the following credentials to setup your Ecsow Dialer System.

Domain/Host: justcallv1.sip.us1.twilio.com

Username: 191287

Password: Oc1544519871

# Use SIP Trunk / VoIP Phone Service

Input your SIP information at Ecsow dialer -> SIP Account tab as the below image shows.

Domain/Host: justcallv1.sip.us1.twilio.com  
Username: 191287  
Password: Oc1544519871

Options

General SIP Account Do-Not-Call Voice Activity Detection Call Distribution Call Transfer Preset

Account information from your SIP VoIP service provider

Display Name twilio

SIP ID/User Name 191287

Domain justcallv1.sip.us1.twilio.com

Authorization User 191287

Password ●●●●●●

Outbound Proxy

Send outbound calls via:

Domain

Proxy

Concurrent Calls

Number of concurrent calls 6

\* SIP service must support multiple channels if you want to run concurrent calls

Advanced Settings

OK Cancel

Click the "OK" button to apply your settings.

Unique solution ID: #1047

Author: eva

Last update: 2025-10-18 07:33