

# LiveKit Cloud + Frejun SIP Trunk Integration Guide

Comprehensive setup guide for integrating LiveKit Cloud with Frejun Teler SIP trunking for both inbound and outbound voice calls.

## 1. Prerequisites

- Active LiveKit Cloud account
- Active Frejun account with SIP trunk enabled
- Purchased channel capacity in Frejun
- Access to LiveKit outbound [GitHub repository](#) .

## 2. Creating a SIP Trunk in Frejun

Basic Configuration:

The screenshot shows a progress bar at the top with three steps: 'Basic Information' (green), 'Advanced Settings' (grey), and 'Review & Create' (grey). Below the progress bar is a section titled 'Basic Information'. It contains fields for 'Name' (with placeholder 'Enter trunk name') and 'Domain name'. Under 'Domain name', there are two input fields: 'Enter subdomain' (empty) and '.sip.frejun.ai'. A red error message 'Please enter a valid subdomain' is displayed below the second field. Further down, there is a 'Channel limit' field (empty), a 'Recording' toggle switch (off), and a 'Secure' toggle switch (off).

Basic Information

Name ⓘ

Enter trunk name

Domain name ⓘ

Enter subdomain .sip.frejun.ai

Please enter a valid subdomain

Channel limit ⓘ

Enter channel limit

Recording ⓘ

Secure ⓘ

### Name:

- ➔ Assign a name to your trunk.
- ➔ Any name can be assigned.

### Domain name:

- ➔ The domain name of your trunk, all the calls will be routed to this domain, also known as Termination URL (you will provide this to your AI platforming further step )
- ➔ Name should not include special characters.

### **Channel limit:**

- ➔ Number of concurrent calls the trunk can handle.
- ➔ You will need to purchase channels. By default, 1.

### **Recording:**

- ➔ If enabled, recordings will be available.

### **Secure:**

- ➔ If enabled, communication will be encrypted and happen over SIP/TLS (Session Initiation Protocol over Transport Layer Security) for call control and media is streamed over SRTP (Secure Real Time Protocol).

Outbound Authentication

Outbound Authentication

IP Authentication      Credential Authentication

IP Addresses ⓘ  
Enter IP address 1  
+ Add IP Address (1/3)

Username ⓘ  
Enter username

Password ⓘ  
Enter password      Show

Outbound calls are when AI agent initiates call to a customer number. (AI Agent -> Customer)

For Outbound calls authentication, any one of the 2 options can be selected depending on the platforms:

1. **IP Authentication:** If AI platform supports static IP addresses to initiate call always, assign those IPs (max 3). That means every time a call is initiated from those IPs, Teler will know the calls are authenticated.
2. **Credential Authentication:** Provide a username and password and remember these credentials as it will be required to set up in your AI platform Trunk.

Once filled, proceed next.

## Inbound Routing

Name ⓘ

Enter routing name

SIP URL ⓘ

Enter SIP URL

### Name:

→ Provide any name for the inbound route

### SIP URL:

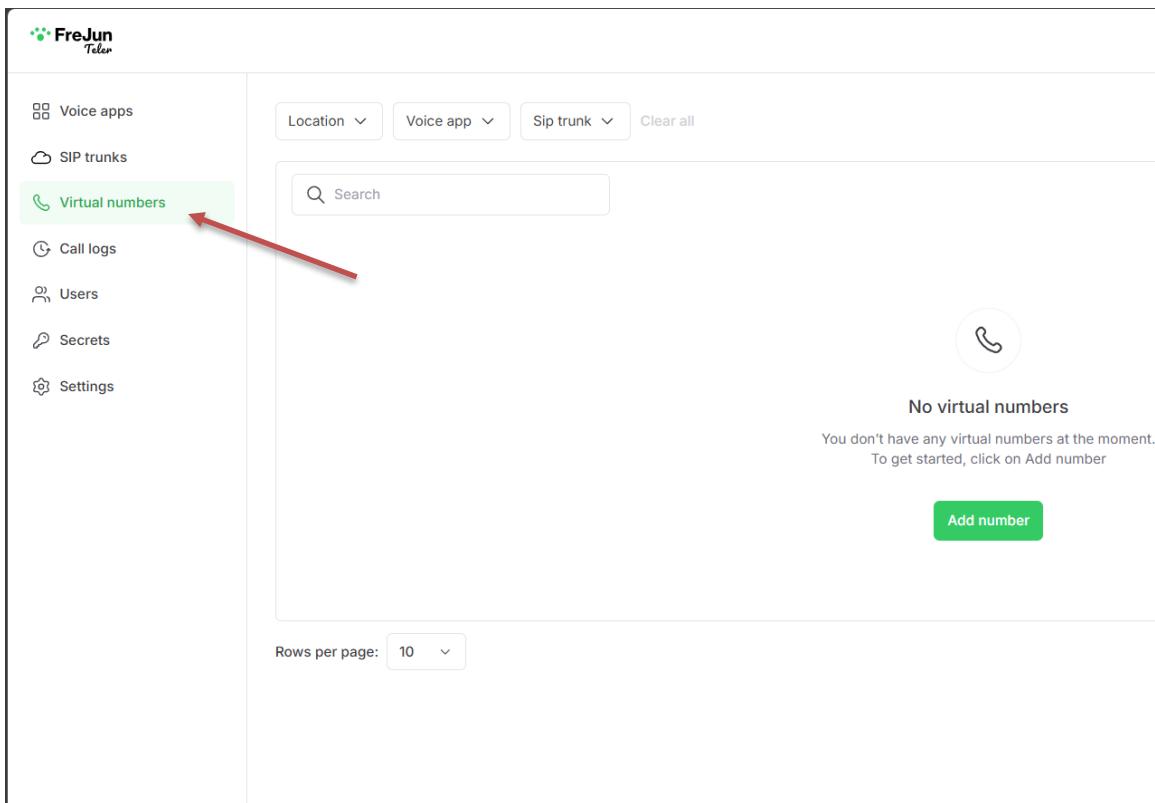
- URL where Teler should deliver inbound calls also known as Origination URL (your AI provider will provide this). When a call comes to the Teler number this is where Teler will route the call, to the AI Agent.
- In case of LiveKit you can find your SIP URL inside ‘SIP Trunks’ under the ‘Telephony’ sections of livekit dashboard

The screenshot shows the LiveKit dashboard with the sidebar navigation open. Under the 'Telephony' section, 'SIP trunks' is selected. The main view displays two sections: 'Inbound' and 'Outbound'. The 'Inbound' section shows one trunk named 'test inbound' with a SIP URI of 'sip:toggobnnv93.sip.livekit.cloud'. The 'Outbound' section shows one trunk named 'testTrunk' with a SIP URI of 'livekit.dev.sip.frejun.ai'. A red arrow points to the 'SIP URI' field for the 'test inbound' trunk.

## 3. Assigning a Virtual number to you sip Trunk

Steps:

You can find your allotted virtual numbers under the ‘Virtual Numbers’ section in the frejun website.



From there you can select the number, click on Actions and select assign

Now you can assign that number to your sip trunk.

And keep track of the numbers that you are assigning, since you will be needing them to setup inbound and outbound for the AI-provider too.

#### 4. LiveKit Cloud SIP Setup

Steps:

1. Navigate to [LiveKit Cloud Dashboard](#)
2. Head to the Agent section and create an Agent and deploy it

The screenshot shows the LiveKit dashboard with the 'Agents' tab selected. At the top, there's a summary card with metrics: 'AGENTS DEPLOYED' (1), 'CONCURRENT AGENT SESSIONS' (0), and 'AGENT SESSION MINUTES THIS BILLING PERIOD' (72/1,000 mins). Below this is a chart titled 'AGENT SESSIONS SERVED' showing session counts over time from Feb 13 to Feb 20. The main area is titled 'Your agents' and lists one entry: 'Umbrella' (ab\_16m29gxp2) with 'CONCURRENT SESSIONS' (0) and status 'PENDING'. A deployment log entry is shown: 'Deployed V20240220064823 3 hours ago'. The bottom left shows a search bar and support links for 'LiveKit DevDay' and 'Learn more'.

This is a modal window titled '+ Deploy new agent'. It contains two main sections: 'Build agent in browser' and 'Deploy agent with code'. The 'Build agent in browser' section includes a sub-section 'Deploy agent with code'.

The screenshot shows the configuration page for the 'Umbrella' agent. The top navigation bar includes 'Auto-refresh off', a search bar, and support links. The main area has tabs for 'Instructions', 'Models & Voice', 'Actions', and 'Advanced'. Under 'Name', the value 'Umbrella' is highlighted with a red box. The 'Instructions' section contains guidelines for personality, tone, and behavior. The 'Welcome message' section includes a checkbox for 'Allow users to interrupt the greeting'. On the right, there's a 'Live preview' section with a waveform visualization and a 'START CALL' button, and a 'Code' section. A red arrow points to the 'Deploy agent' button at the top right. A warning message at the bottom right states: 'Error creating agent builder [internal] twirp error resource\_exhausted: maximum number of agents reached (1/1)'.

Make sure you remember the Agent name, as you would require the agent name to dispatch it both in case of inbound and outbound.

You can also view all the agents from the Agent-dashboard.

### 3. Create outbound SIP trunk at livekit

The screenshot shows the LiveKit SIP trunks dashboard. On the left, there's a sidebar with options like Overview, Sessions, Agents, and Telephone. Under Telephone, the SIP trunks section is selected. The main area displays two trunks:

| TOTAL INBOUND TRUNKS | TOTAL OUTBOUND TRUNKS | SIP URI                           |
|----------------------|-----------------------|-----------------------------------|
| 1                    | 1                     | sip:taggo6nnv93.sip.livekit.cloud |

Below this, there are two sections: Inbound and Outbound. The Outbound section shows a single trunk:

| TRUNK ID        | TRUNK NAME   | NUMBERS          | CREATED AT              |
|-----------------|--------------|------------------|-------------------------|
| ST_N5XtoAsJDUXc | test Inbound | +918065193777 +1 | 13 Feb 2026, 8:26:25 pm |

At the top right of the main content area, there's a blue button labeled "Create new trunk" with a red arrow pointing towards it. The bottom of the screen has a search bar, support links, and a project dropdown.

You will see a form similar to this : -----

**Create a new trunk**

**TRUNK DETAILS** JSON EDITOR

**Trunk name**

**Trunk direction**

Inbound     Outbound

**Numbers** ⓘ  
List of provider phone numbers this trunk accepts calls for. If none are specified, it accepts calls to any number.

**Allowed addresses** ⓘ  
For better security, only allow IP addresses you trust. If left empty or set to "0.0.0.0/0", all IP addresses will be allowed.

**> Optional settings**

[Learn more in the docs](#)   [Cancel](#) [Create](#)

**Create a new trunk**

**TRUNK DETAILS** JSON EDITOR

**Trunk name**

**Trunk direction**

Inbound     Outbound

**Address** ⓘ

**Transport** ⓘ

**Numbers** ⓘ

**Optional settings**

**Media encryption (SRTP)** ⓘ

Select media encryption

**Username** ⓘ

**Password** ⓘ

For the numbers field you can put in the numbers that you assigned to your sip trunk in frejun Teler.

And for the address field in case of outbound type in the complete domain name the you got after creation of SIP trunk in frejun teler

| Name       | Domain Name               | Status | Date Created |
|------------|---------------------------|--------|--------------|
| testTrunk  | livekit.dev.sip.frejun.ai | Active | Feb 13, 2026 |
| test11labs | e1abs.dev.sip.frejun.ai   | Active | Feb 16, 2026 |
| test1      | test1.dev.sip.frejun.ai   | Active | Jan 26, 2026 |

Now for the authentication, if TLS then use the same username and password that you had configured for the frejun sip trunk.

And then click create.

## 5. Linking SIP Trunk to LiveKit Agent

In LiveKit:

- Create or select an Agent
- Assign inbound number to agent
- Configure dispatch rule to route incoming calls
- Ensure outbound trunk is mapped for agent call initiation

## 6. Using the Outbound Repository

Repository: [www.github.com/shantanubindhani/livekit-outbound](https://www.github.com/shantanubindhani/livekit-outbound)

Typical Steps:

- Clone repository
- Configure LiveKit API key & secret
- Configure SIP trunk ID
- Run outbound call script