

Genesys Cloud

The how to guide on setting up the Genesys Cloud integration with RTT

Pre-requisites

- **Genesys Cloud** account with Admin rights
- **Parloa** account with Admin rights on your tenant

Configuring the SIP Trunk

1. Start by navigating to Genesys Cloud's Admin panel, then Telephony → Trunks.

The screenshot shows the Genesys Cloud Admin dashboard. At the top, there is a navigation bar with links for Activity, Directory, Documents, Clients, Performance, Apps, and Admin. The Admin link is highlighted with a red arrow labeled '1'. Below the navigation bar, the page title is 'Genesys Cloud Admin' with a 'Search' input field. On the left, there is a vertical sidebar with icons for Overview, Activity, Directory, Documents, Clients, Performance, Apps, and Admin. The Admin icon is highlighted with a red arrow labeled '2'. The main content area is divided into several sections: Account Settings (Subscription, Genesys AppFoundry, Organization Settings), People & Permissions (People, Roles / Permissions, Authorized Organizations, Divisions), Directory (Groups, Locations, Profile Fields, External Contacts), Documents (Workspaces), Telephony (Topology, Metrics, Trunks, Sites, Edge Groups, Edges, Phone Management, Certificate Authorities, DID Numbers, Extensions, Global Telephony Settings), and Genesys Cloud Voice (Number Management).

- Click Create New to create a new SIP Trunk

The screenshot shows the RTT Documentation interface with the following details:

- Top Navigation:** Activity, Directory, Documents, Clients, Performance, Apps, Admin.
- Left Sidebar:** Telephony / Trunks, Topology, Metrics, Trunks (selected), Sites, Edge Groups, Edges, Phone Management, Certificate Authorities, DID Numbers.
- Current View:** External Trunks (selected tab).
- Actions Bar:** Status, Usages, Copy, Delete, Limits, Refresh, + Create New (highlighted by a red arrow).
- Data Table:**

	External Trunk Name	State	Listen Port	Protocol	Recording	Trunk Type
<input type="checkbox"/>	In Service	TLS	Disabled	Generic BYOC Car...		
<input type="checkbox"/>	In Service	TCP	Disabled	Generic BYOC Car...		
<input type="checkbox"/>	In Service	TLS	Disabled	Generic BYOC Car...		

- Give it a descriptive name, and under **Type** select **BYOC Carrier**, and **Generic BYOC Carrier**. Under **Protocol**, we strongly suggest **TLS**. Lastly, make sure the **Trunk State** is **In Service**.

The 'Create External Trunk' dialog box contains the following fields:

- External Trunk Name:** My SIP Trunk To Parloa RTT
- Type:** BYOC Carrier (selected) / Generic BYOC Carrier
- Status:** New
- Type:** Generic BYOC Carrier
- Trunk State:** In Service (switch is blue)
- Protocol:** TLS

- Under **Inbound**, select your preferred Site, and type your desired **Inbound SIP Termination Identifier**. This Identifier will be part of the FQDN that will be used for inbound calls into Genesys, so keep that in mind.

Inbound

Number Plan Site *

Parloa RTT Site

This site controls which number plan is used, both for transforms on inbound calls and for subsequent outbound transfers for calls which never flow through another site.

Inbound SIP Termination Identifier ⓘ

parloa-rtt-inbound

Inbound SIP Termination Header ⓘ

DNIS Replacement Routing ⓘ

Disabled

Inbound Request-URI Reference

FQDN Method



INVITE sip:+xxxxxxxxxx@parloa-rtt-inbound.byoc.mypurecloud.de

TGRP Method ⓘ

INVITE sip:+xxxxxxxxxx;tgrp=parloa-rtt-inbound;trunk-context=byoc.mypurecloud.de@lb01.byoc.eu-central-1.mypurecloud.de

- Under Outbound, the only thing you will need to add is a **SIP Server or Proxy**. For RTT, we always use `rtt.voip.parloa.com`, and there is no need to specify the port. The rest can be left blank.

Outbound

Outbound SIP Termination FQDN ⓘ

Outbound SIP TGRP Attribute ⓘ

TGRP Context-ID ⓘ

Outbound SIP DNIS ⓘ

Outbound Request-URI Reference

INVITE sip:+xxxxxxxxxx@rtt.voip.parloa.com

SIP Servers or Proxies ⓘ

rtt.voip.parloa.com	
---------------------	--

Hostname or IP Address

Port



- Scroll down to **SIP Access Control**. Here you have to whitelist Parloa's Signalling IPs. You can find these in our [Public Documentation](#). Make sure you

add the ones tagged as **VoIP Traffic**.

SIP Access Control ⓘ

Allow the Following Addresses ⓘ

20.107.52.237	✖
20.13.27.100	✖
20.86.133.107	✖

7. Scroll down and expand the **Protocol** section. Under **User to User Information (UUI)**, we suggest you enable **UUI Passthrough**.

User to User Information (UUI)

UUI Passthrough ⓘ

Enabled

Dynamic UUI settings apply to UUI added to Architect call flows or agent scripts. All settings in this section apply to outbound only. For more information, [UUI Overview on the Resource Center](#).

Type

User-to-User

Selects the type of User To User header for UUI dynamically set within an IVR Flow or Agent Script. Does not apply to Static UUI.

Encoding Format

Hex

Sets the encoding format for dynamically set UUI.

Protocol Discriminator

00

Describes the protocol or structure used within the dynamically set UUI Data. Must be a two-digit hexadecimal value. Required when using User-To-User header type. Does not apply to Static UUI.

8. Still under **Protocol**, enable **Take Back** and **Transfer**.

Transfer

<p>Take Back and Transfer </p> <p>Enabled </p> <p>Allows for the REFER method and enables transferring of a local party when a transfer request is received.</p>	<p>Release Link Transfer (RLT) </p> <p>Disabled </p> <p>Allows for the REFER method to be sent on the trunk to transfer the remote party to a new destination.</p>
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9. Click Save External Trunk.

This settings should be enough to allow calls from Genesys to Parloa, and vice versa.

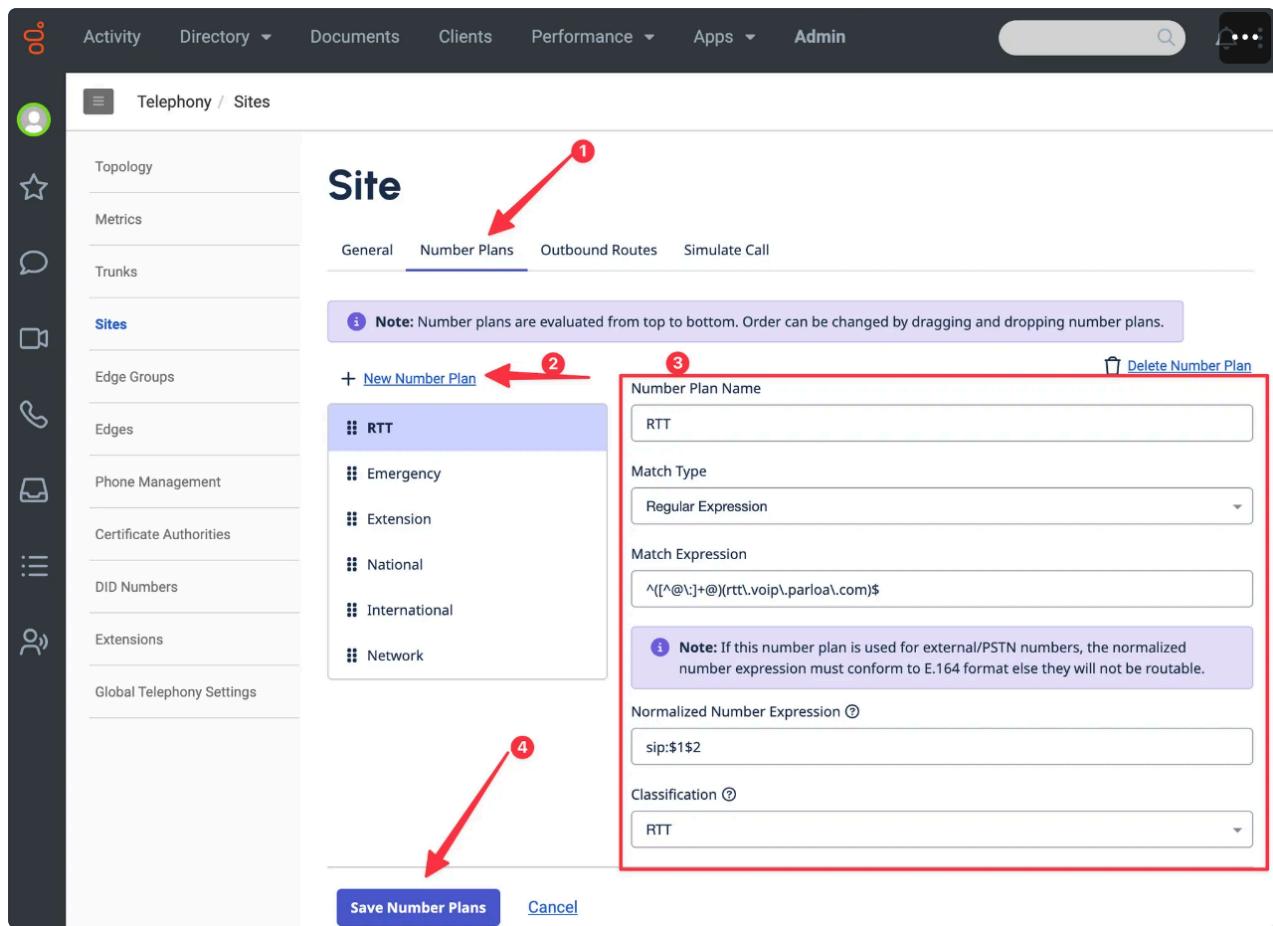
Numbering Plan & Outbound Routes

For every caller language in RTT, you will have a unique URI you need to call from Genesys Cloud. In this section we'll configure Genesys Cloud to route these URIs to the newly created SIP trunk.

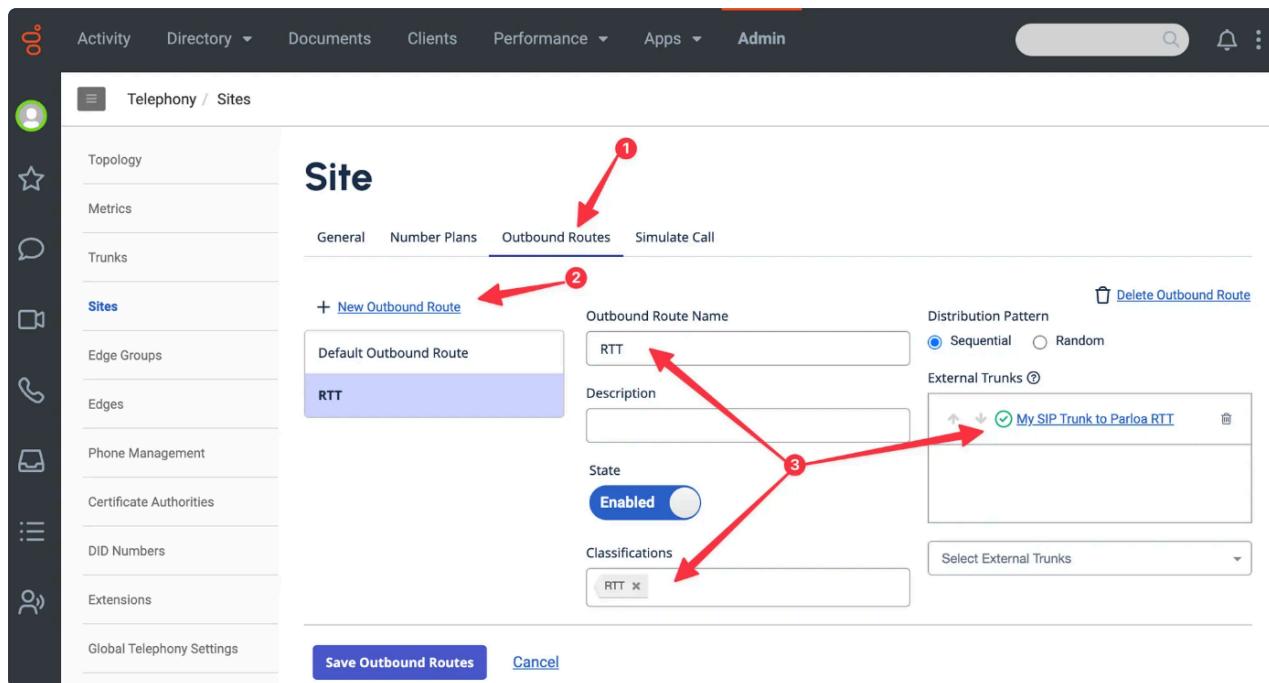
1. Navigate to the Admin panel, then **Telephony → Sites**

The screenshot shows the Genesys Cloud Admin interface. At the top, there is a navigation bar with links: Activity, Directory, Documents, Clients, Performance, Apps, Admin, and a search bar. A red arrow labeled '1' points to the 'Admin' link in the top right. Below the navigation bar, the page title is 'Genesys Cloud Admin' and the breadcrumb path is 'Overview / Admin Home'. On the left, there is a vertical sidebar with icons for Overview, Favorites, Chat, Video, Phone, File, Integrations, and User. The main content area is divided into several sections: Account Settings (Subscription, Genesys AppFoundry, Organization Settings), People & Permissions (People, Roles / Permissions, Authorized Organizations, Divisions), Directory (Groups, Locations, Profile Fields, External Contacts), Integrations (Integrations, Actions, Single Sign-on, OAuth, Authorized Applications), Documents (Workspaces), and Telephony (Topology, Metrics, Trunks, Sites, Edge Groups, Edges, Phone Management, Certificate Authorities, DID Numbers, Extensions, Global Telephony Settings). A red arrow labeled '2' points to the 'Sites' link under the Telephony section.

2. After clicking on your Site (this has to be the same Site the trunk was set up on step #4 of Configuring SIP Trunk), navigate to **Number Plans**, create a New Number Plan for RTT URIs and do the following:
- **Number Plan Name & Classification:** Can be whatever you want; in this case we name both `RTT`.
 - **Match Type: Regular Expression**
 - **Match Expression:** `^([^\@\:\:]+@)(rtt\.\voip\.\parloa\.\com)$`
 - **Normalized Number Expression:** `sip:$1$2`



- After Clicking **Save Number Plans**, navigate to **Outbound Routes**. Here is where we decide which trunk we want to send the calls that match the RTT regex. Click **New Outbound Route**. Give it a name, select the RTT classification you created in the previous step, and select the external trunk you created in the first section of this guide.



We've now told Genesys how to identify RTT URLs, and which trunk should be used whenever we need to call said URLs.

Forwarding calls to RTT

In this section, we'll explain how to forward calls from Genesys to specific languages in RTT. There are of course many ways you can do this, but if you have no idea where to begin, this is a good starting point. This may or may not apply to your setup, but we recommend reviewing it briefly. In this scenario, we have multiple hotlines come into Genesys, and they will be forwarded to different languages using Data Tables in the Architect.

Data Tables

We will use Data Tables to map incoming calls to different hotlines, to different languages in RTT.

1. The first thing we need to do is to create a Data Table. Navigate to the Genesys Admin panel, then under Architect → Data Tables.

Overview / Admin Home

Documents	Telephony	Genesys Cloud Voice	Contact Center
Workspaces	Topology Metrics Trunks Sites Edge Groups Edges Phone Management Certificate Authorities DID Numbers Extensions Global Telephony Settings	Number Management	ACD Skills & Languages Utilization Queues Wrap-Up Codes Email Canned Responses Response Assets Co-browse Widgets Analytics Panel Manager Scripts Script Templates
Message	Architect	Predictive Engagement	Routing
Platforms Platform Configs SMS Number Inventory Threading Timeline Messenger Configurations Messenger Deployments	Architect Data Tables Flow Outcomes Flow Milestones	Live Now Segments Outcomes Action Maps Action Library Global Settings	Operating Schedules Call Routing Message Routing Emergency Groups Disconnect Interactions

- Click + Add to create a new table. Give it a name, and a **Reference Key Label**, which in our case will be the Hotline number (the number that the caller dialed). Click **Save**.

Architect / Data Tables

Data Tables

Create Data Table

Name *	Hotline to RTT URI Mapping
Description	Add Description
Division	Home
Reference Key Label ⓘ *	Hotline

Cancel Save

- Click **Add Field**, then String and name it something like **RTT URI**. Optionally, you can add a third field with the caller language. When you're done, click **Save**.

Data Table

Name *
Hotline to RTT URI Mapping

Description
Add Description

Division
Home

Reference Key Label ⓘ *
Hotline

Custom Fields

- RTT URI**
Default
- Caller Language**
Default

Add Field ▾

- Boolean
- Integer
- Decimal
- String

String Field Options

Field Label *
RTT URI

Default

API Field ID generated after sav...

4. Before we add rows to our table, we must first get our RTT SIP URIs. Log into <https://augment.parloa.com>, go to **Settings → Telephony → Inbound Flow**. Here you will find your Inbound (or caller) languages, each with its own URI. In our example, we will use English and German languages. Copy these URIs.

The screenshot shows the RTT Documentation Settings interface. On the left sidebar, under the 'Languages' section, there is a red arrow labeled '1' pointing to the 'Languages' option. Another red arrow labeled '2' points to the 'Inbound Flow' button at the bottom of the sidebar. The main content area is titled 'Inbound Flow' and contains a table titled 'Inbound Identifiers'. The table has two rows: one for 'English (United States)' with the identifier '671b7adb249e13c93c0c0a57enus@rtt.voip.parloa.com' and another for 'German' with the identifier '671b7adb249e13c93c0c0a57dede@rtt.voip.parloa.com'. There are also sections for 'Options' and 'Intro' configurations.

5. Go back to your Data Table in Genesys and click **+ Add** to add a new row. Input your hotline number with the corresponding URI. Click **Save & Close** and repeat as necessary.

The screenshot shows the Genesys Data Tables interface for 'Hotline to RTT URI Mapping'. On the left, there is a sidebar with 'Data Tables' selected. A red arrow labeled '1' points to the '+ Add' button at the top right of the main content area. A modal window titled 'Create Data Table Row' is open in the center. Inside the modal, there are three fields: 'Hotline' (containing '+49 47'), 'RTT URI' (containing '671b7adb249e13c93c0c0a57enus@rtt.voip.parloa.com'), and 'Caller Language' (containing 'English'). A red box labeled '2' highlights the 'RTT URI' field. A red arrow labeled '3' points to the 'Save & Close' button at the bottom right of the modal.

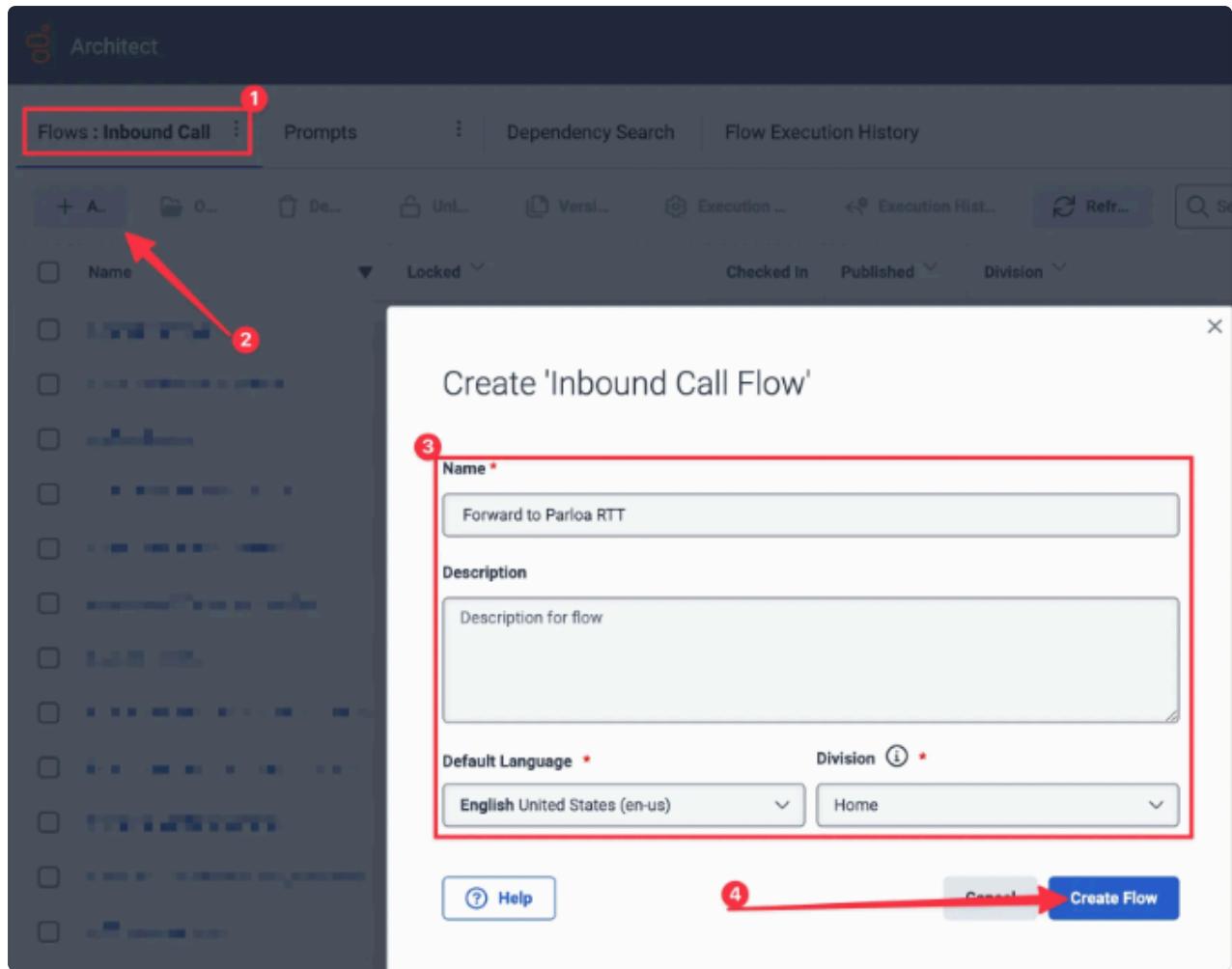
6. Our end result with two languages looks like this:

Hotline to RTT URI Mapping			
<input type="text"/> Find row by Hotline			<input type="button"/>
<input type="checkbox"/> Hotline	▼	RTT URI	Caller Language
<input type="checkbox"/> +49 █ █ 38		671b7adb249e13c93c0c0a57dede@rtt.voip.parloa.com	German
<input type="checkbox"/> +49 █ █ 47		671b7adb249e13c93c0c0a57enus@rtt.voip.parloa.com	English

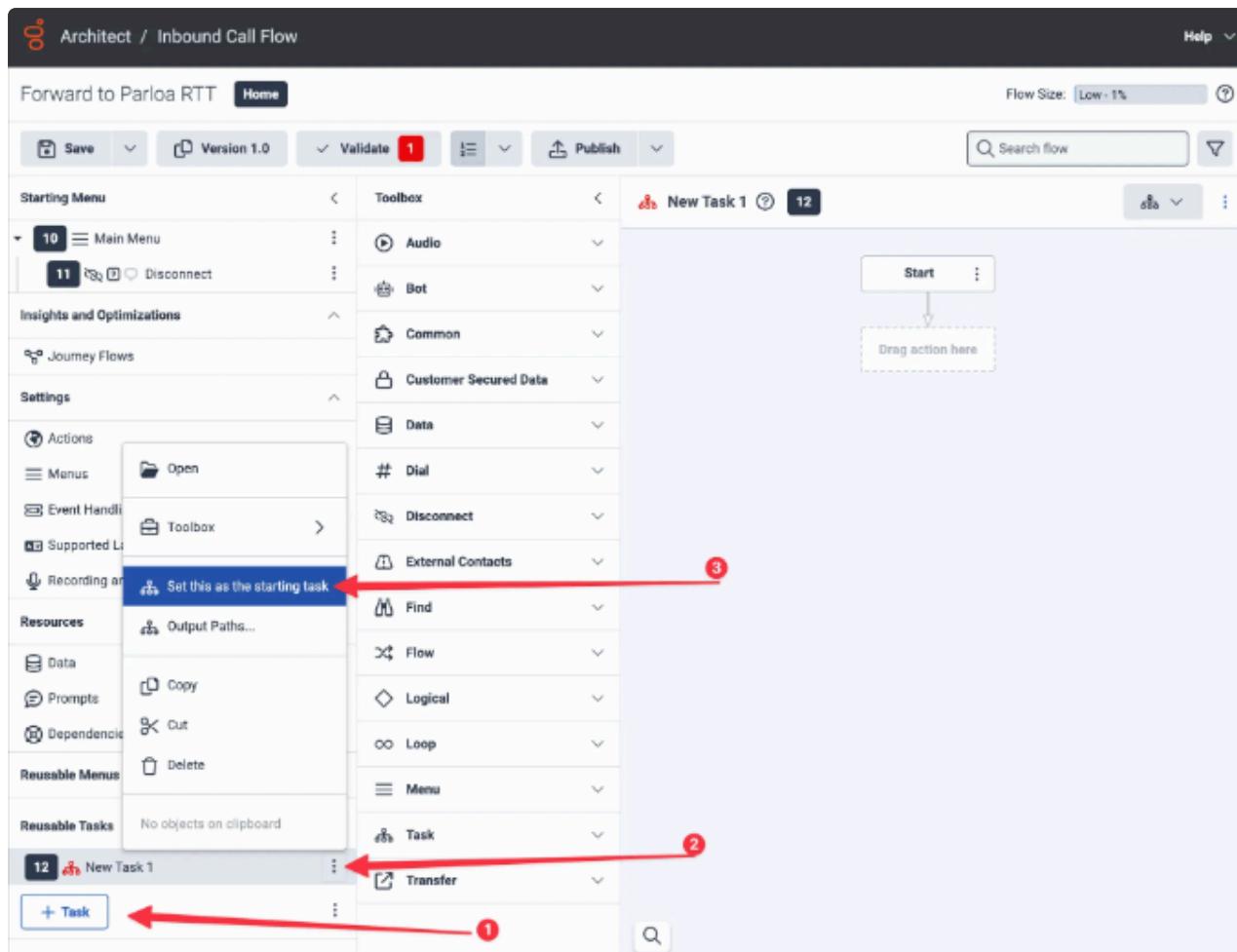
Inbound Call Flow

Now that we have our Data Table, we have to create a new Inbound Call Flow to use said Data Table and forward the call to Parloa.

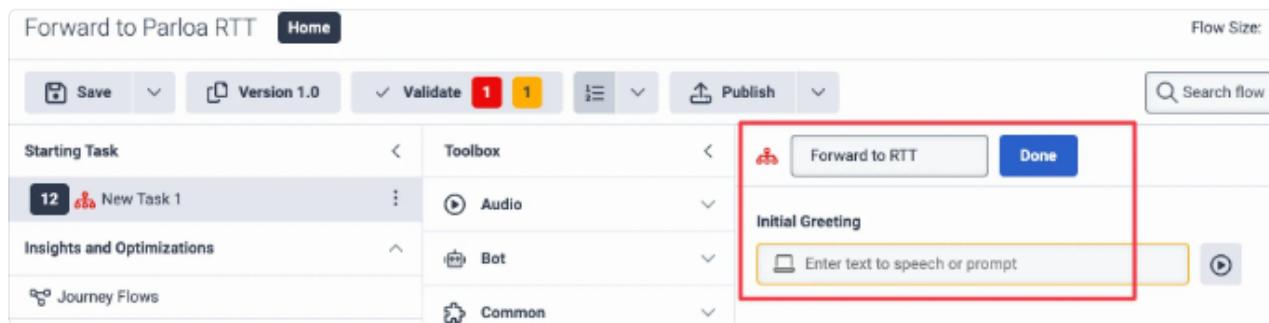
1. Navigate to **Admin** → **Architect** → **Architect**. A new tab will open. Make sure **Inbound Call** is selected under **Flows**. Click **+ Add**, give it a name and click **Create Flow**.



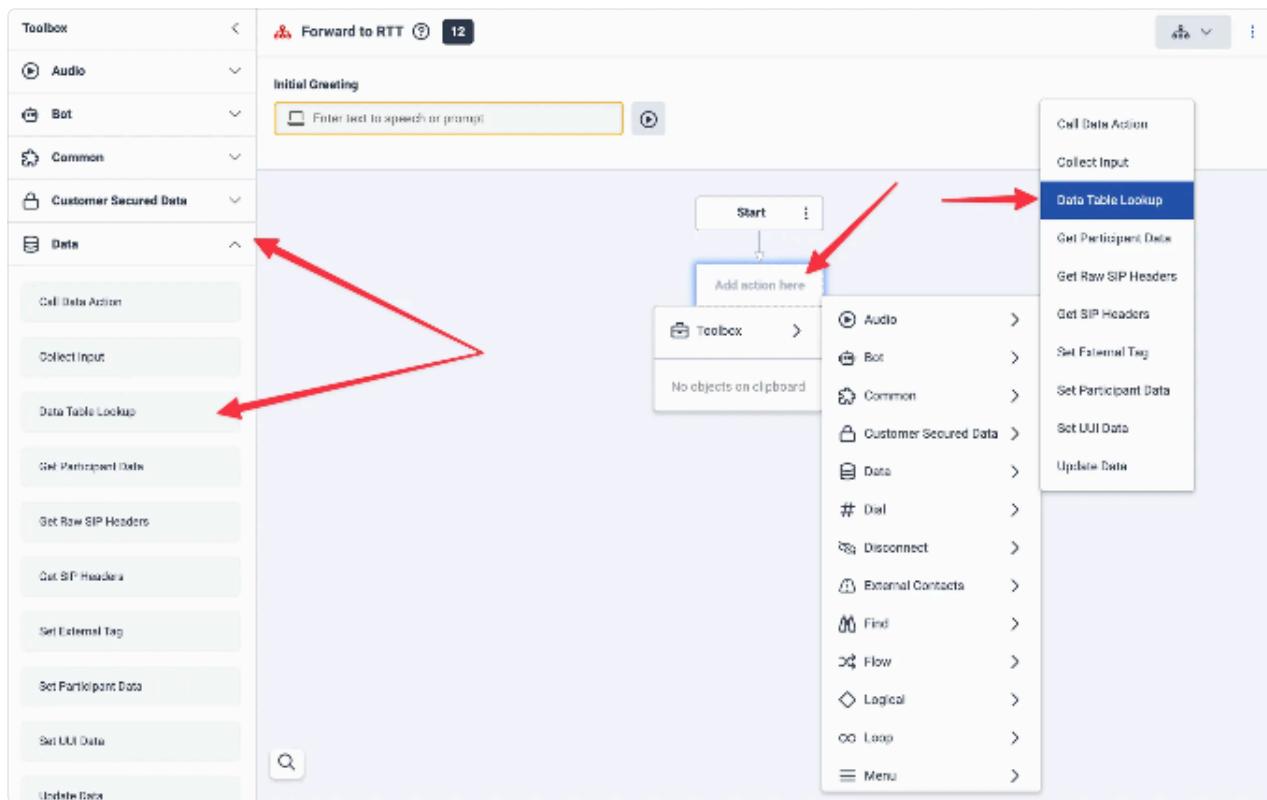
2. First thing we need to do: Add a new **Reusable Task**, and set it as the starting task.



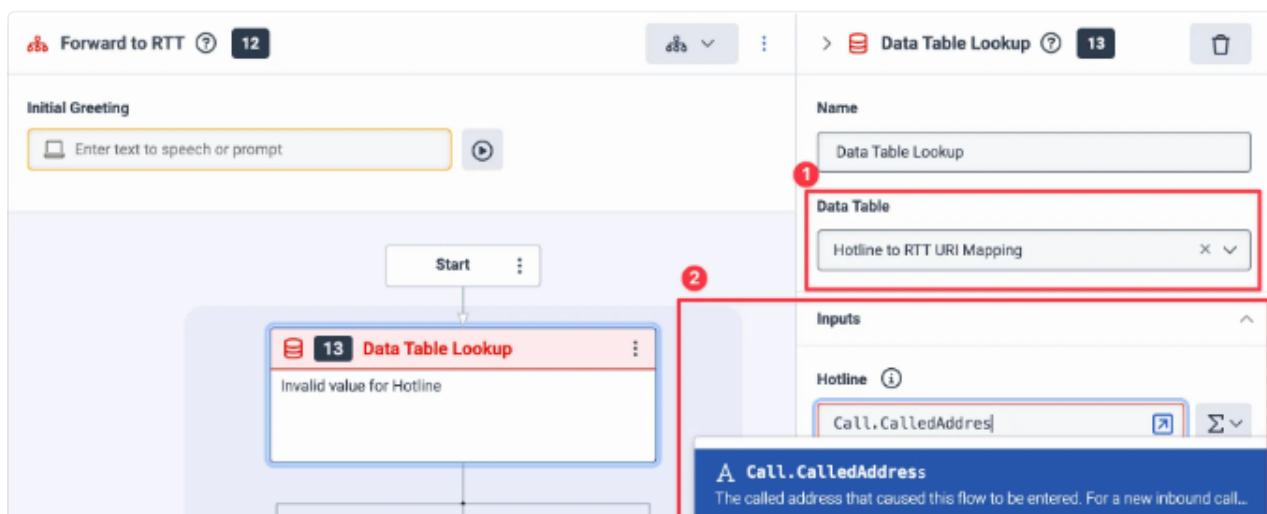
3. Delete the **Initial Greeting**, and feel free to rename the task. This is so that no announcements will be played. This is mandatory to avoid that sentences that are not spoken by the CSA are recognised by RTT and sent to the caller.



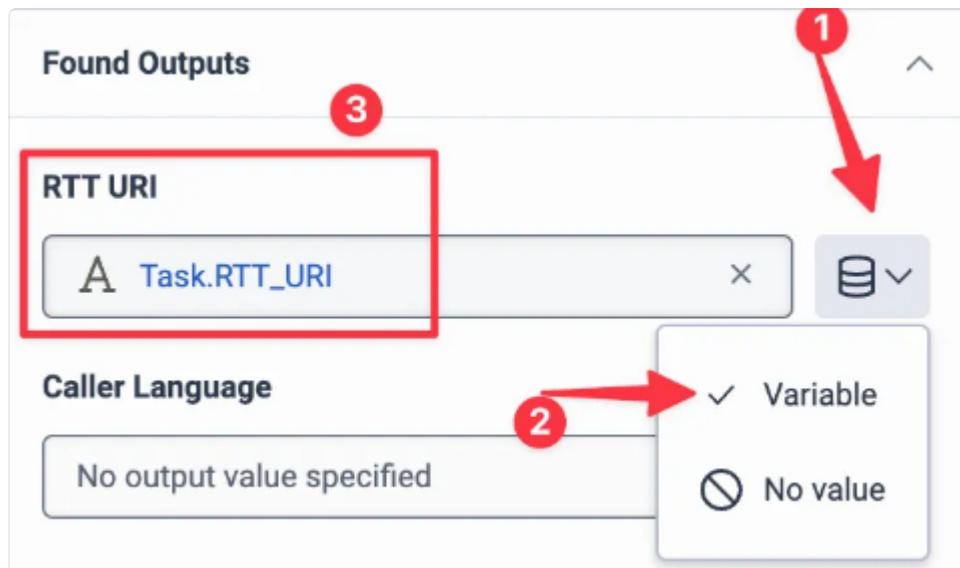
4. Now we'll focus on the flowchart. Let's add a **Data Table Lookup** action. You can add it by either dragging from the toolbox to the left, or by clicking the "Drag action here" box.



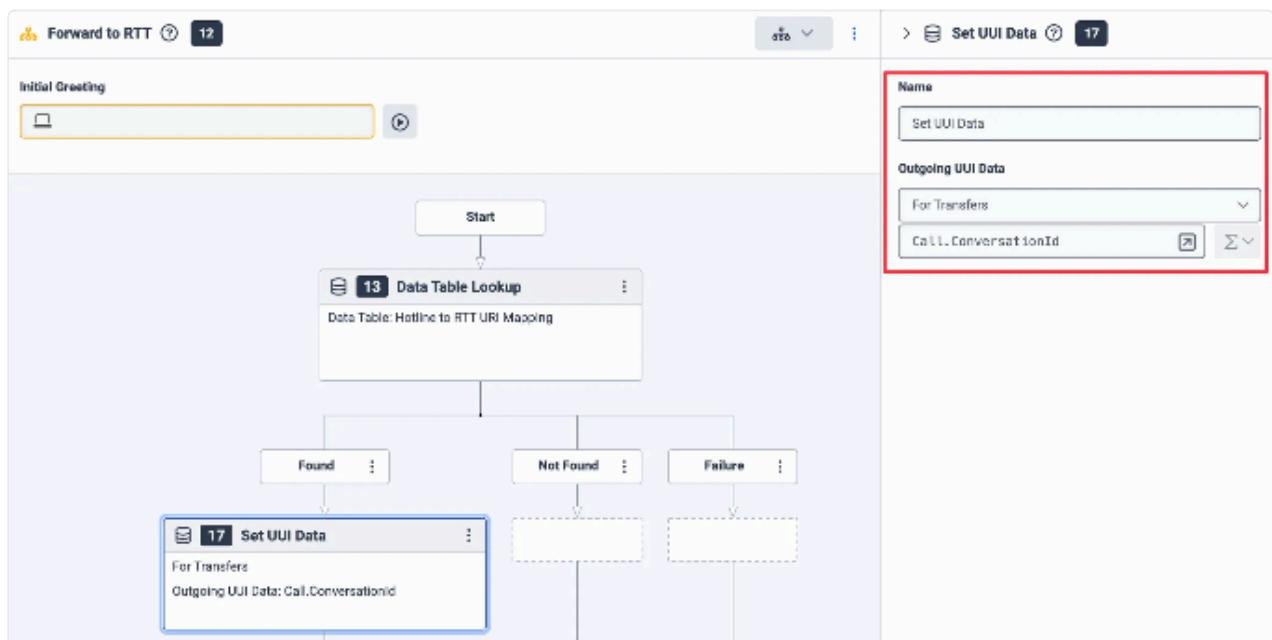
- On the right hand side, choose the Data Table we created. Under Inputs, there should be a field for the Hotline. Here we need to pass the hotline number that was dialed to reach this Inbound Flow, which are also present in the Data Table. For this, you can simply input `Call.CalledAddress` (Genesys will autocomplete if done correctly)



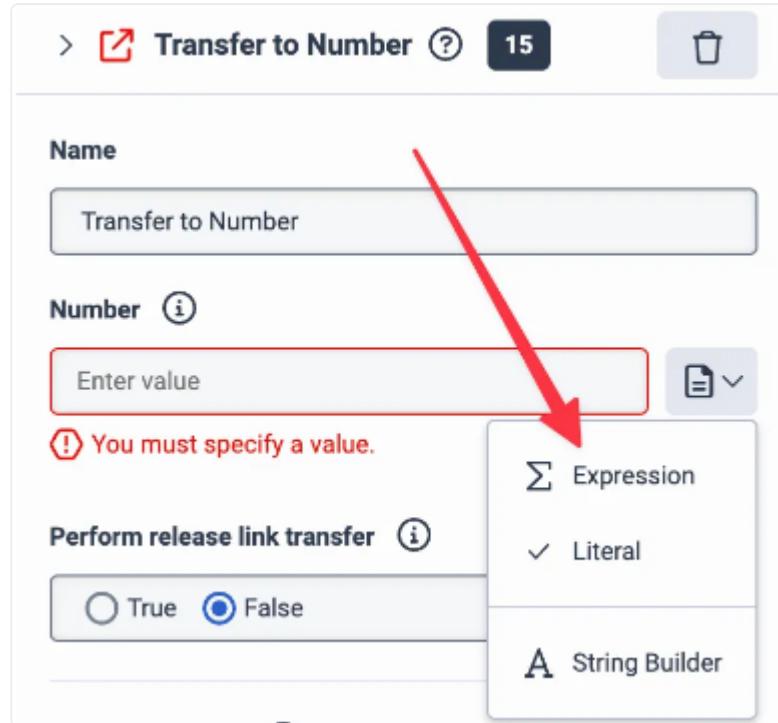
- Genesys will do a lookup in the table for the hotline number, and will return the rest of the data. In this example, we only care about the RTT URI. Under Found Outputs, create a new variable to store the returned value. Note: there is no need to add `Task`. It will be done automatically by Genesys.



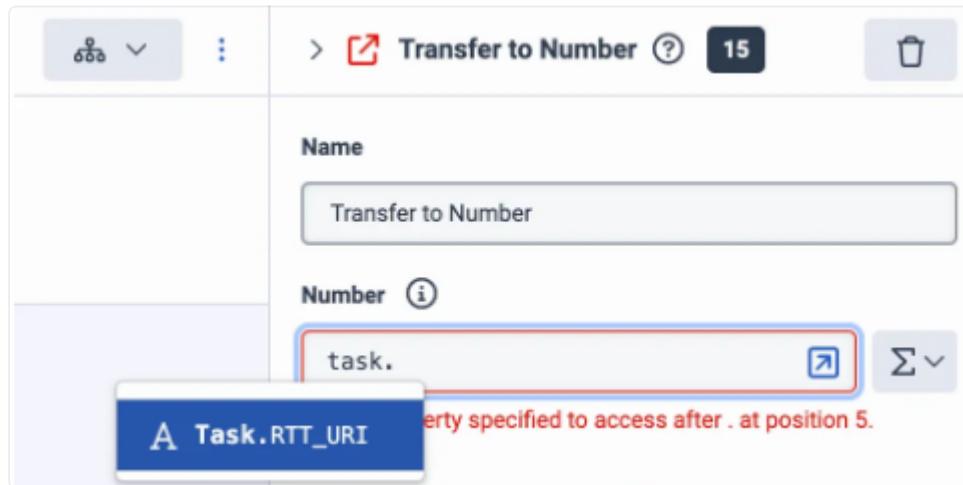
- Now that we have our RTT URI, we need to forward the call. Before we forward the call, we need to add User-to-User info (UUI) to the call, this will help with the Agent Pop Up later on. Under the Found branch, add a **Set UUI Data block**. Select For Transfers, type in `Call.ConversationId`. This ensures we send a unique identifier on each call.



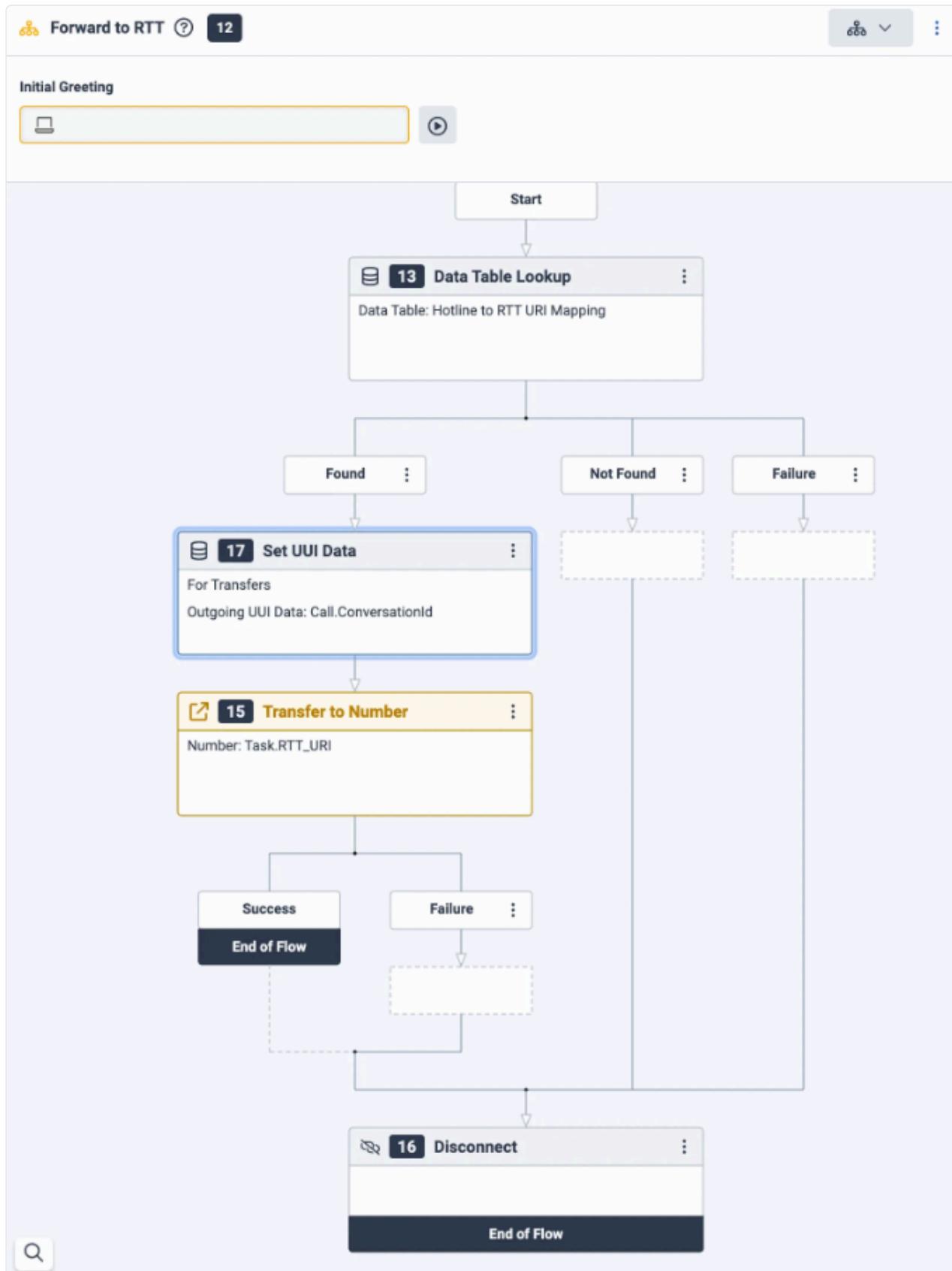
- Under the **Set UUI Data**, add a new block: **Transfer to Number**. Change the Number field to Expression



- Enter the variable created in step 6. Leave the rest of the settings unchanged. Number in this case means destination, so also includes URLs.



- Lastly, add a Disconnect action at the end of the flow. (You can add additional steps and/or error handling, but we are doing the very basics.) In the end, it should look like this:

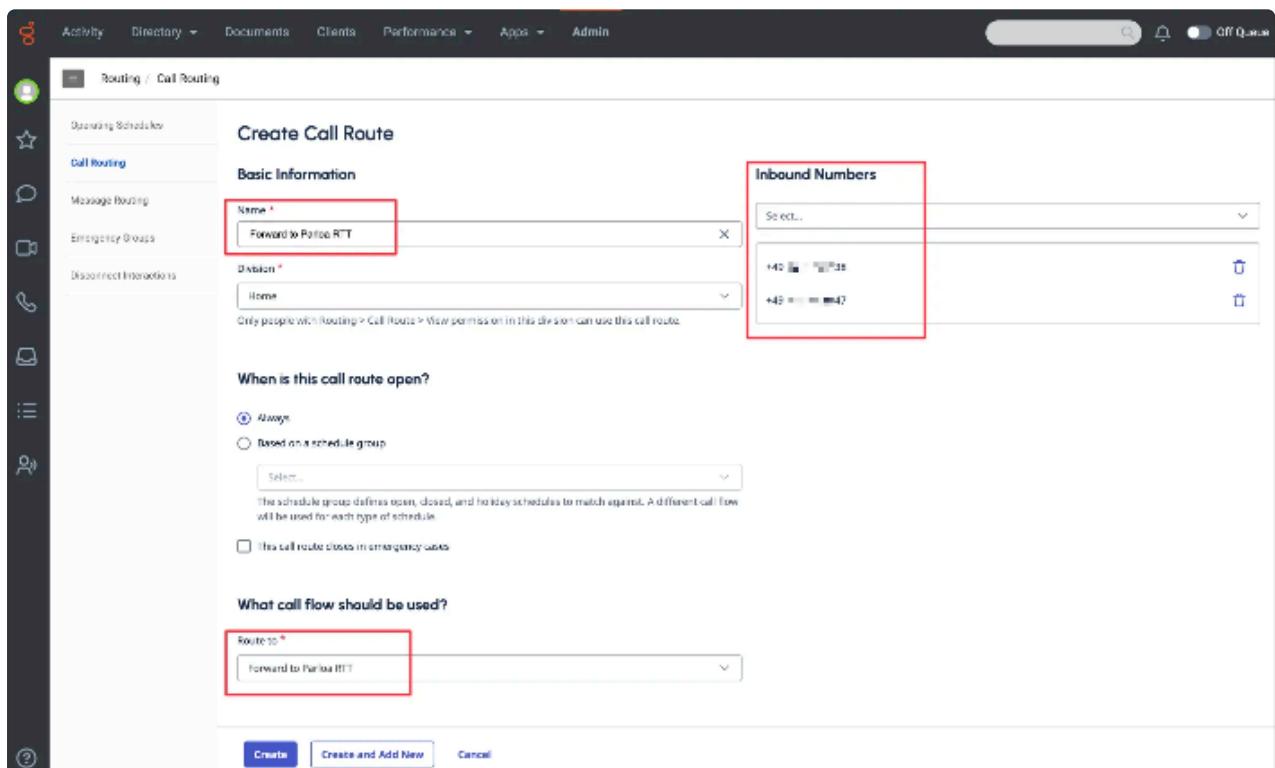


11. Click **Save** and then **Publish**.

Call Routing

Now that we have our Inbound Call Flow, let's configure a route to trigger this Inbound Flow.

Navigate to **Admin → Routing → Call Routing**, and click **+ Add** to create a new route. Give it a name, select the **Inbound Call Flow** we just created in the previous section, and select the hotline numbers (or inbound numbers) that will trigger this flow. These numbers should be the ones added in the Data Table we created earlier.



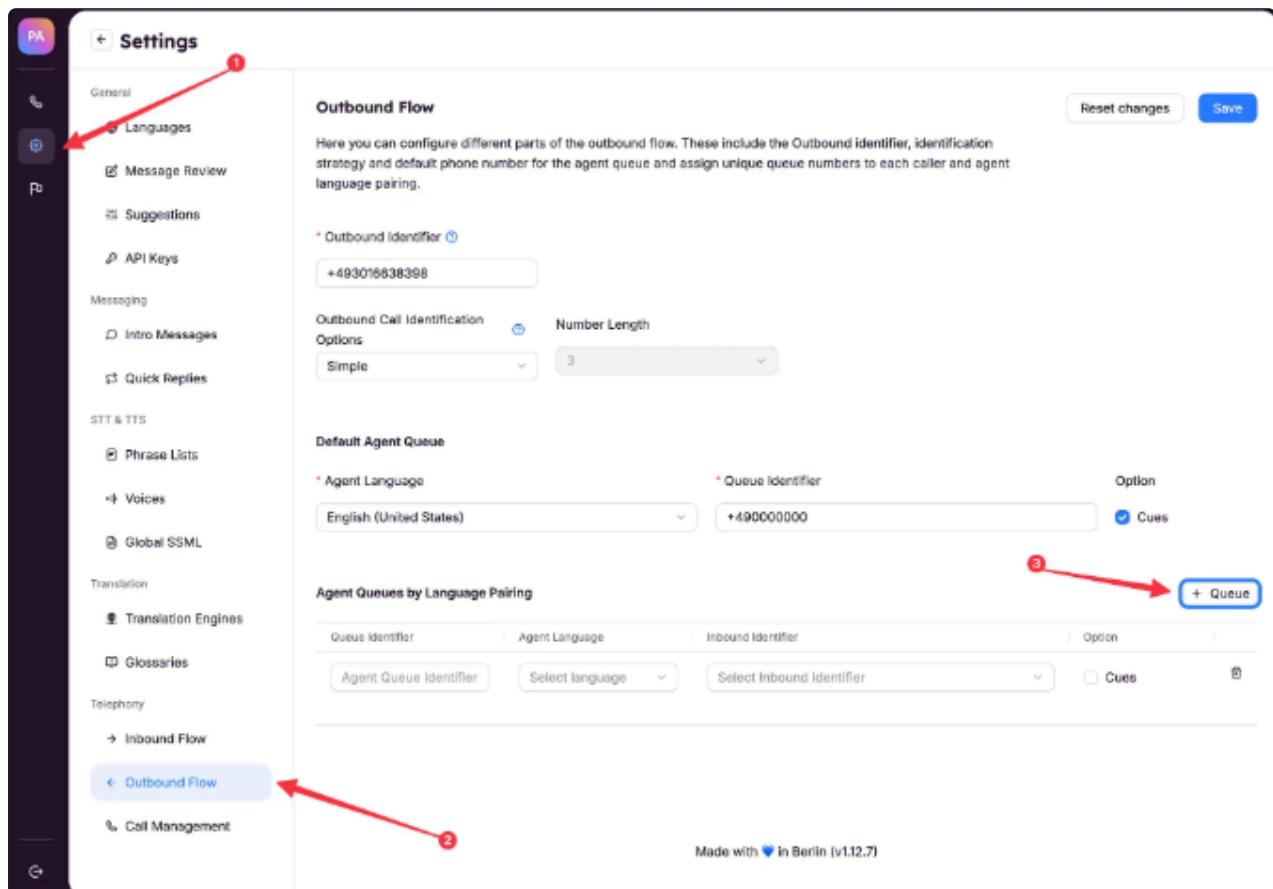
And you're done! If you followed these steps properly, you should be able to reach Parloa RTT's different languages based on the number dialed.

Configuring the Call Flow in Parloa

As described in the [Call Flow diagram](#), RTT orchestrates two calls. Now that you configured the first call to Parloa RTT, we need to configure Parloa to generate the

outbound call (the agent or the translated call) to Genesys. Before you continue, make sure you have an inbound number (can be a virtual number) that we can use.

- First, we will pair a caller language with an agent language. Log into <https://augment.parloa.com>, go to Settings → Telephony → Outbound Flows. Click + Queue to add a new Agent Queue.



- Once you add an Agent Queue:

- Provide a Queue Identifier. This is the endpoint where Parloa will send the agent call to. This is where the inbound identifier of Genesys comes in, the one you created in Step 4. The Queue Identifier should have the following format:

```
sip:<inbound number>@<inbound identifier>.byoc.mypurecloud.de;transport=tls
```

In our example, our inbound Genesys identifier is `parloa-rtt-inbound`, and my inbound number is `+4999900000`. So this means our **Queue Identifier** in Parloa should be:

```
sip:+4999900000@parloa-rtt-
inbound.byoc.mypurecloud.de;transport=tls
```

You can also set this as your Default Agent Queue

- b. Chose the **Agent Language**. This will be the language in which the Agent in **Genesys** will receive the call. In our example, our Agent will speak Spanish.
- c. Add the **Inbound Identifiers**: These are the caller languages that will be routed to that particular Agent Queue. In our example, both English and German callers will be routed to a Spanish speaking agent. Don't forget to click **Save!**

Outbound Flow

Here you can configure different parts of the outbound flow. These include the Outbound identifier, identification strategy and default phone number for the agent queue and assign unique queue numbers to each caller and agent language pairing.

* Outbound Identifier [?](#)
+493016638398

Outbound Call Identification Options [?](#) Number Length
Simple 3

Default Agent Queue

* Agent Language Spanish * Queue Identifier sip:+4999900000@parloa-rtt-inbound.byoc.mypurecloud.de;transport=tls Option Cues

Agent Queues by Language Pairing		
Queue Identifier	Agent Language	Inbound Identifier
sip:+4999900000@parloa-rtt-inl	Spanish	671b7adb249e13c93c0c0a57enus@rtt.voip.parloa.com English (United States) <input type="checkbox"/> Cues
		671b7adb249e13c93c0c0a57dede@rtt.voip.parloa.com German <input type="checkbox"/>

That's all you have to do in Parloa! We now configured that both English and German speaking callers will be routed to Spanish speaking Agents in Genesys.

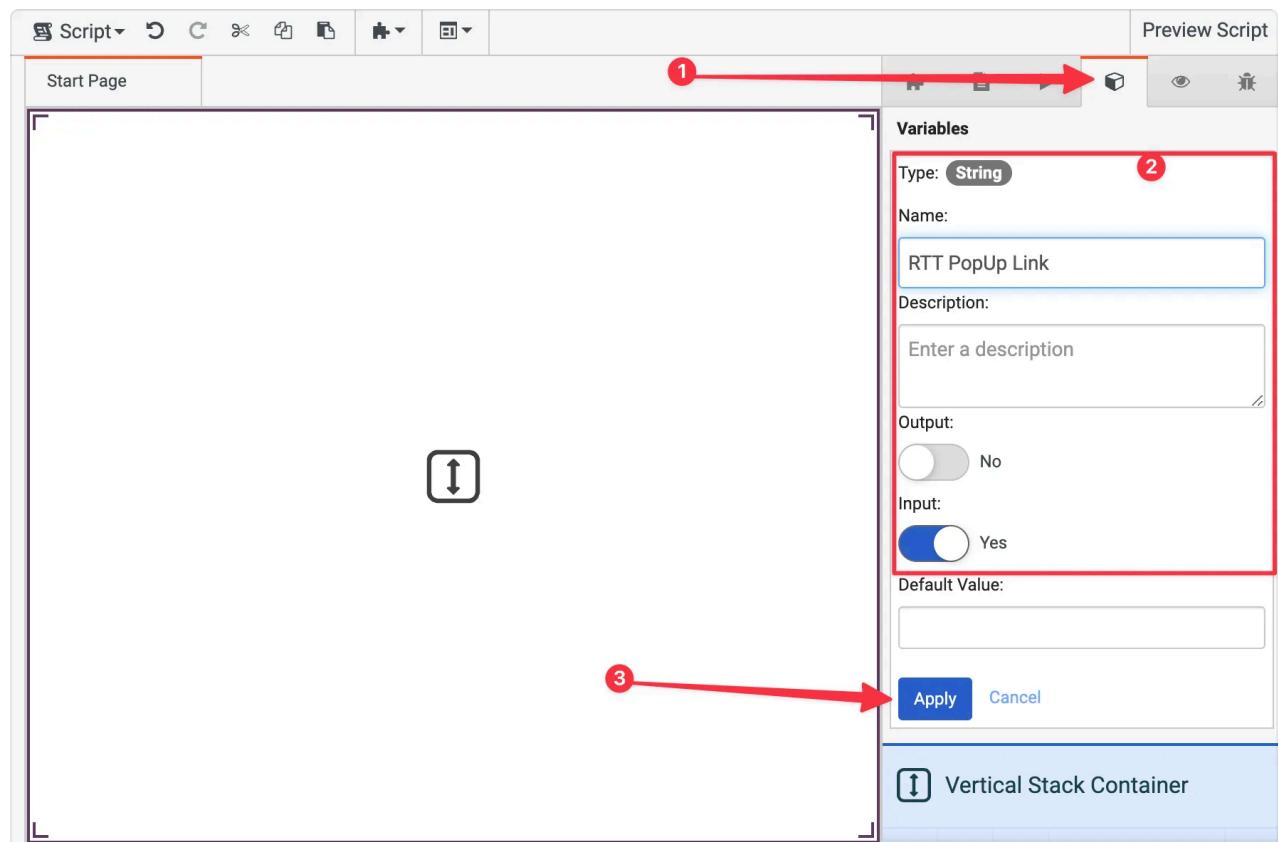
Configuring the Pop Up Screen in Genesys

One of the benefits of using Genesys Cloud and connecting via SIP to Parloa, is that we can send call metadata in the headers. In this section, we will use the metadata to automatically open the correct call when the Agent receives the call in Genesys.

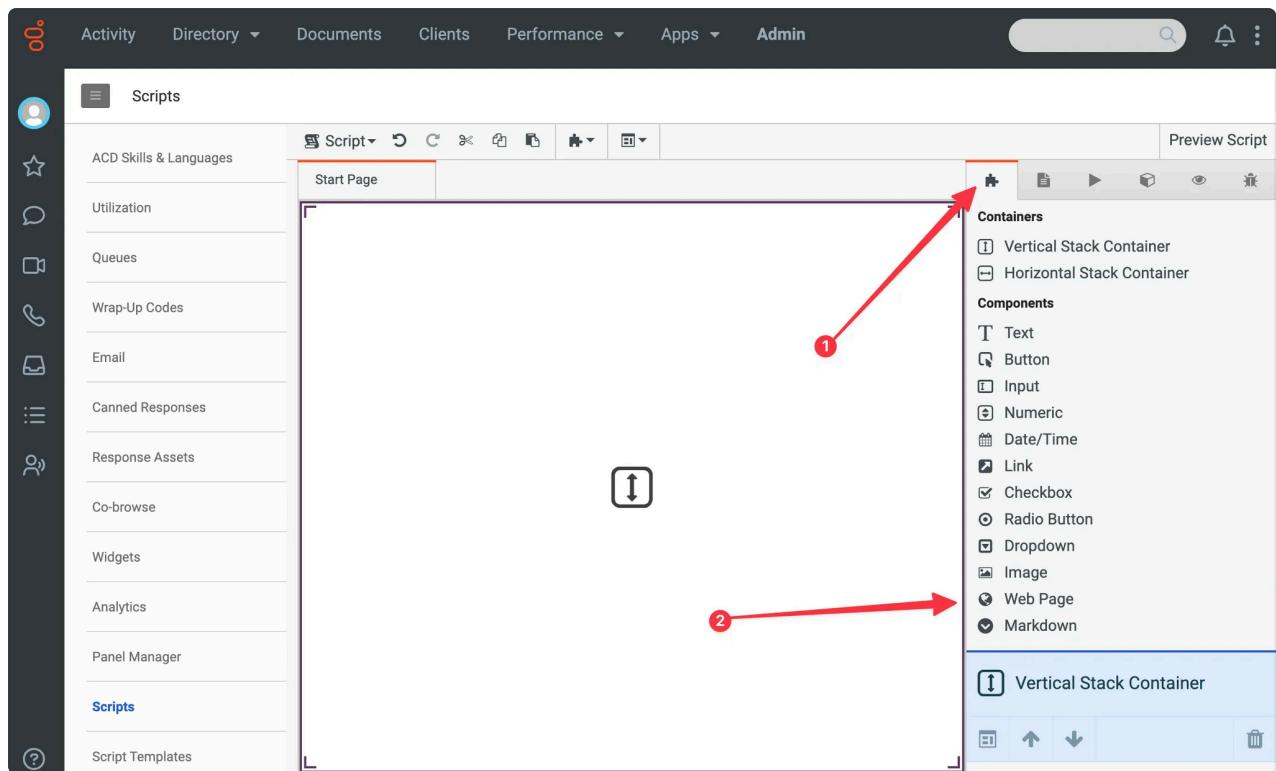
Agent Script

When the agent receives a call, we want to make it easy for the agent to open the same call in the RTT interface, so we will configure a script to automatically open the correct conversation.

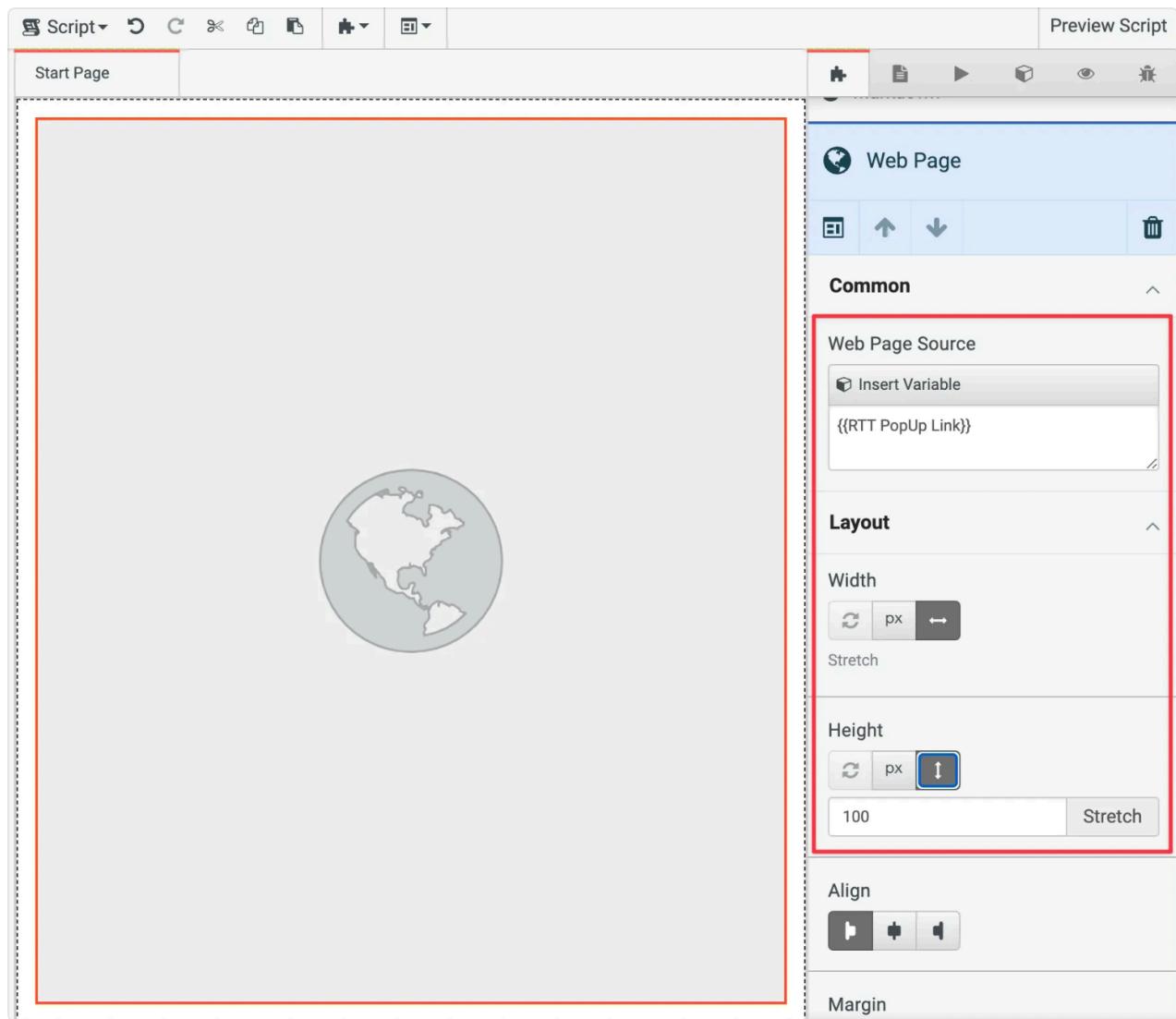
1. Navigate to **Admin → Contact Center → Scripts** and create a new blank script. First we need to create a new variable for the Pop Up link. Click the **Variables** tab and create a new **String** type **variable**, and enable it as an **Input**. Click **Apply**.



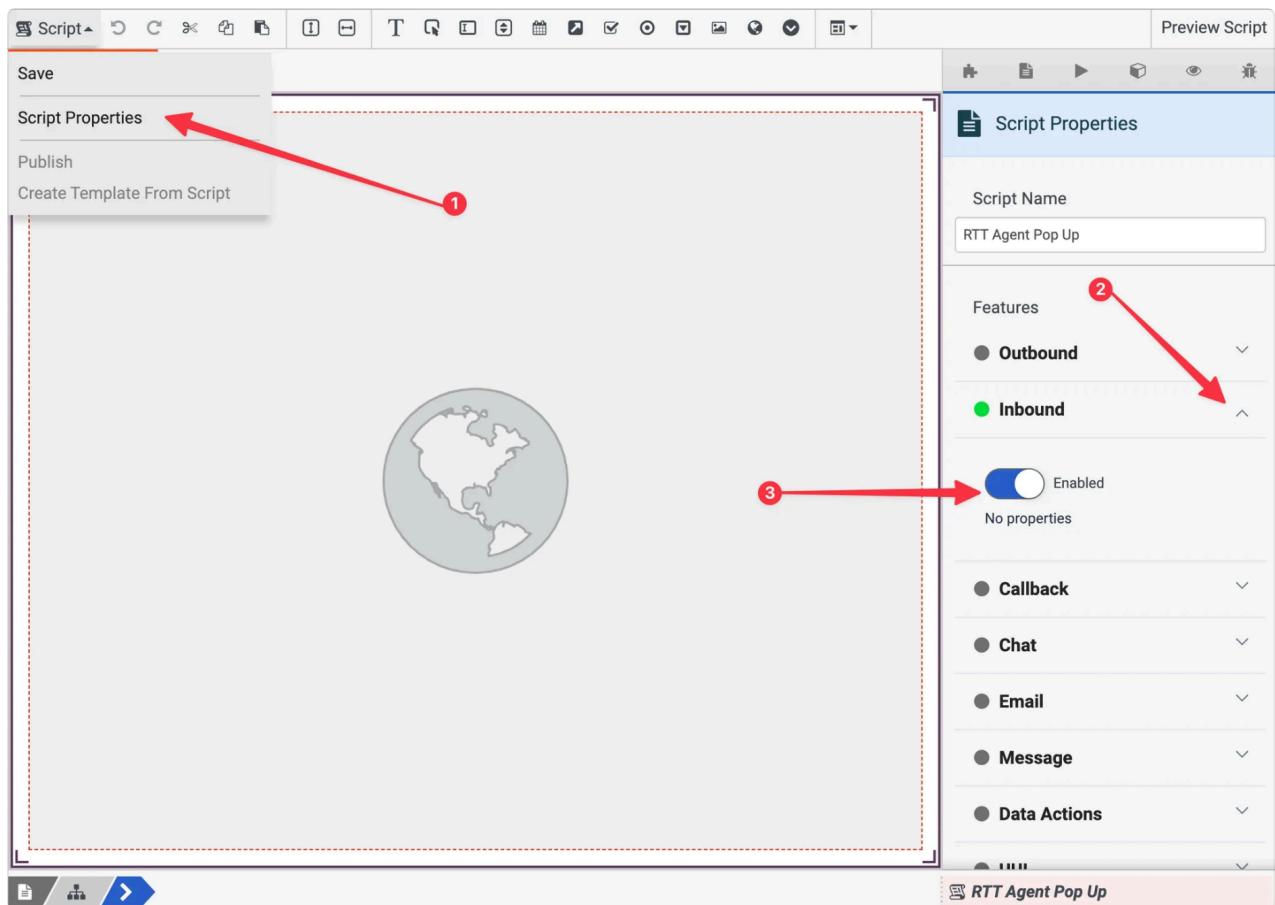
2. Go back to the containers tab, and click Web Page.



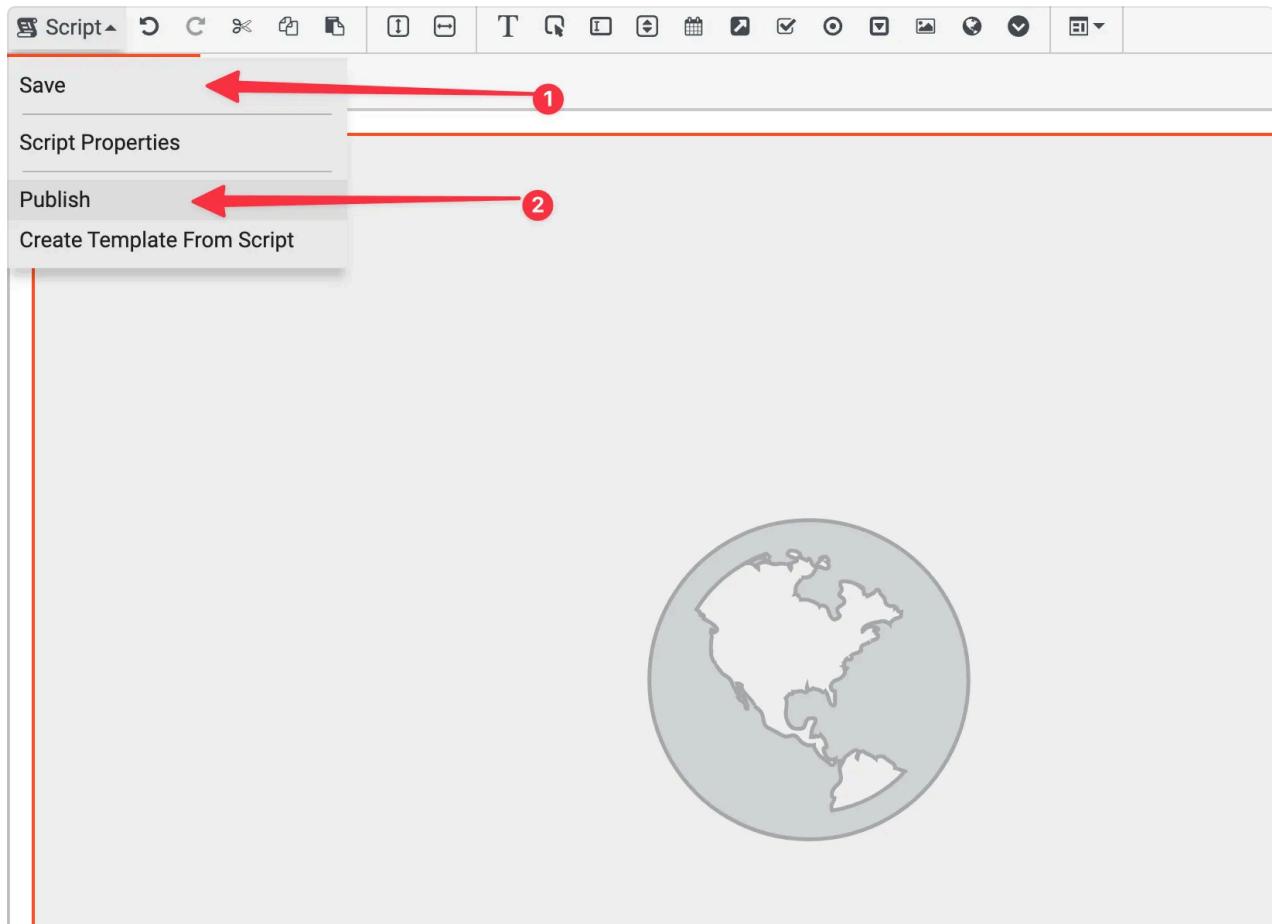
3. On the Right hand side, a few options will appear for our Web Page component.
 - a. Under Common → Web Page Source, click Insert Variable and select the variable you just created in the step before.
 - b. Under Layout, change both Width and Height to Stretch.



4. Lastly, we need to enable the Script for inbound use. Click on Script → Script Properties, on the right hand side expand Inbound and enable it.

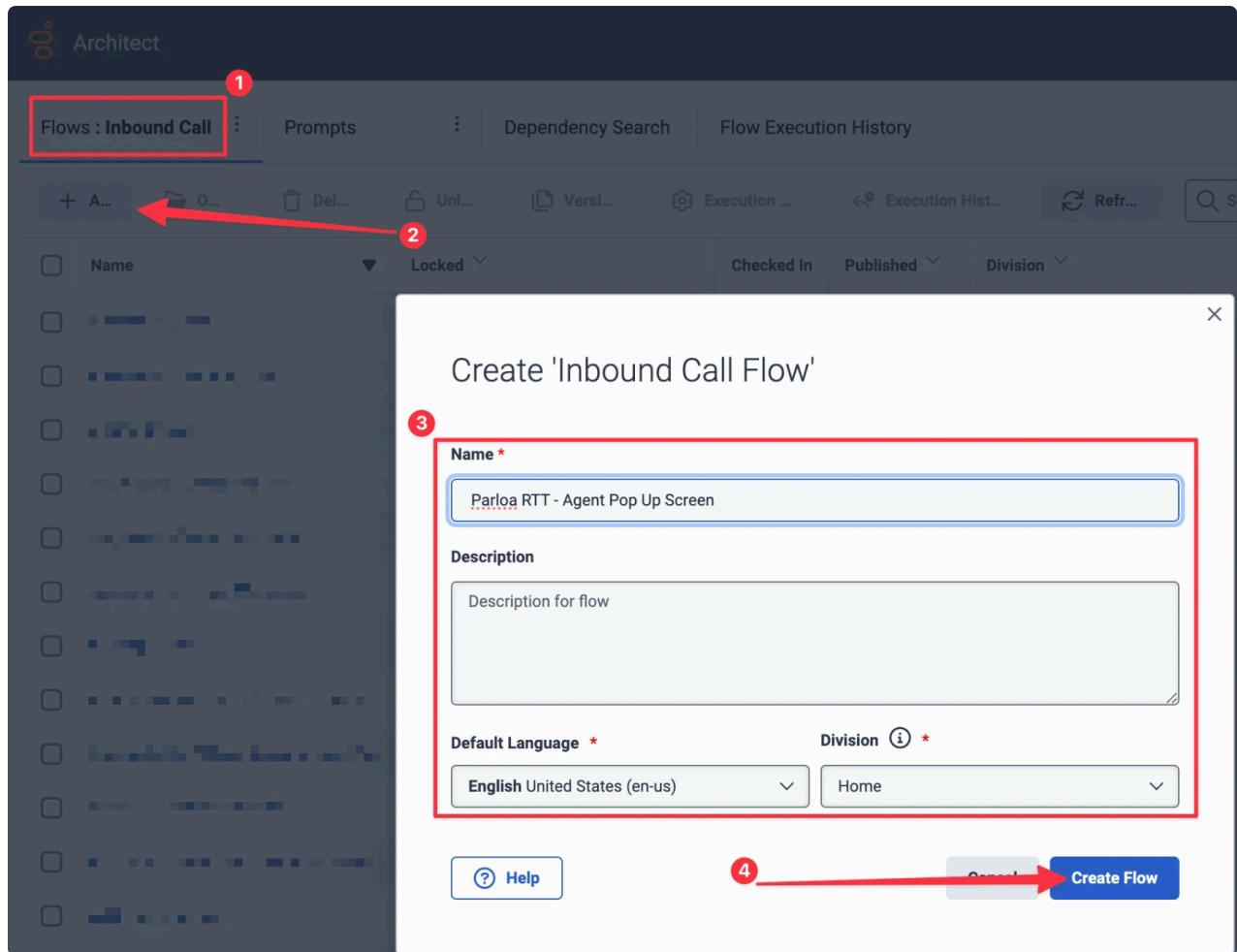


5. When you're done, click Save and Publish.

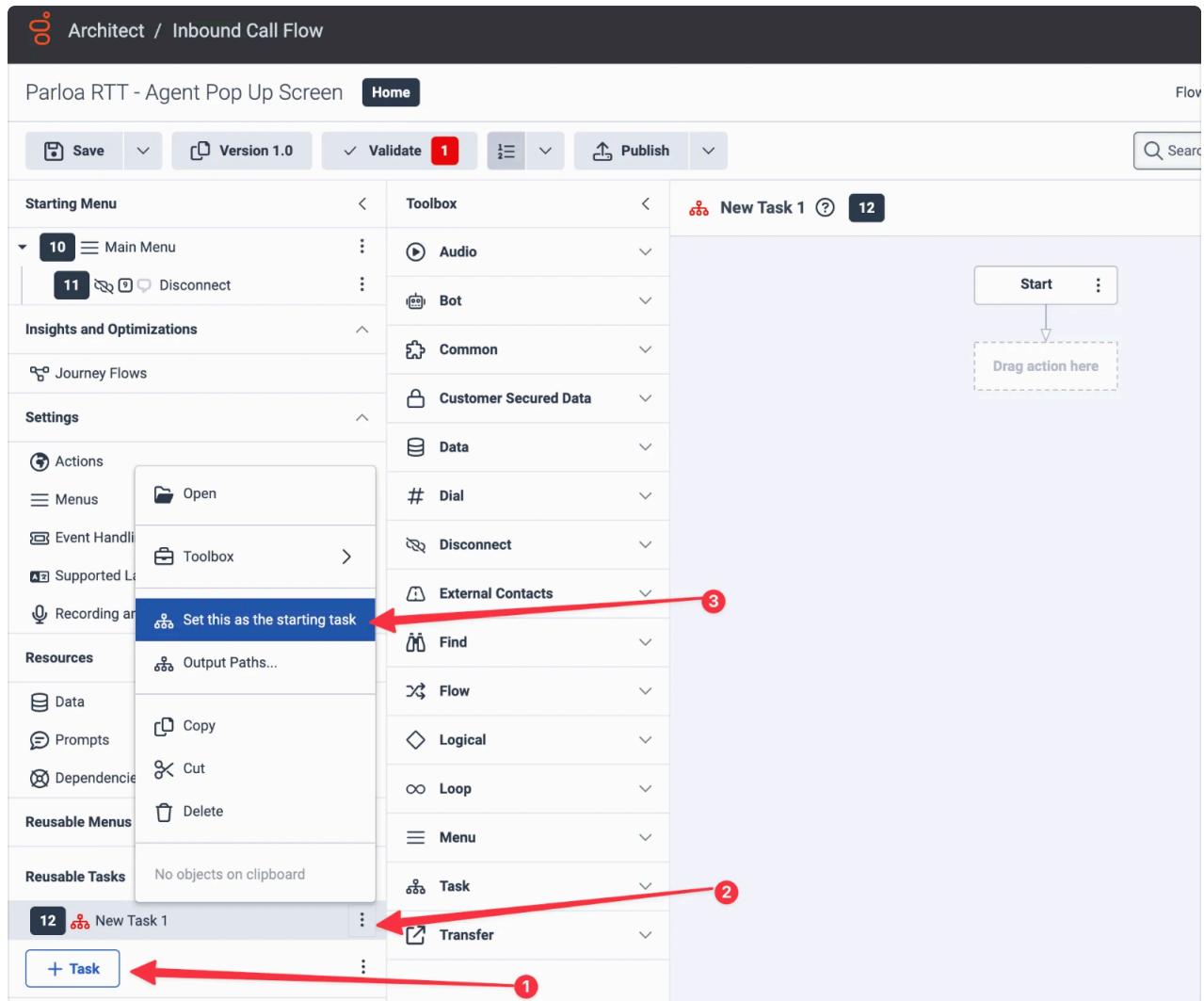


Agent Inbound Call Flow

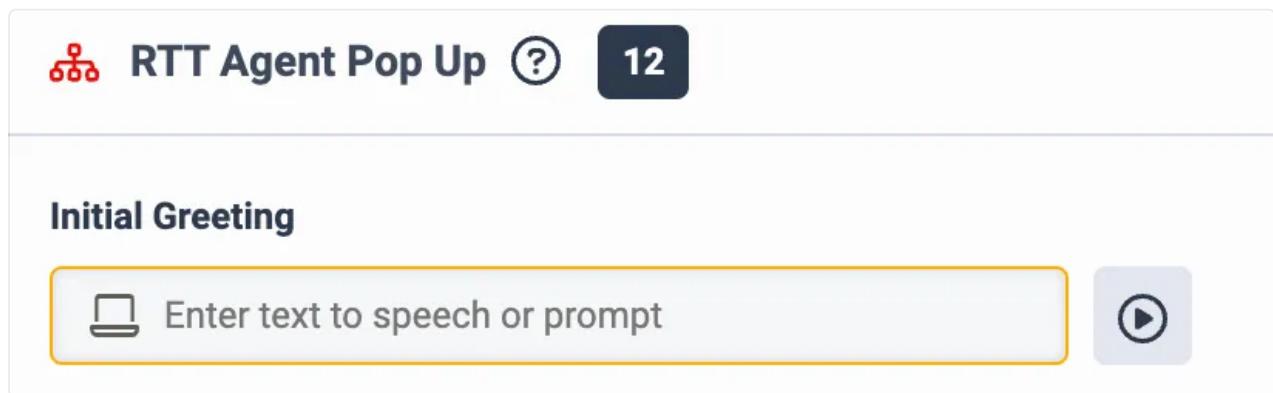
1. Now that we have our Script, let's make sure the Agents actually use it. Navigate to **Admin** → **Architect** → **Architect**. A new tab will open. Make sure **Inbound Call** is selected under **Flows**. Click **+ Add**, give it a name and click **Create Flow**.



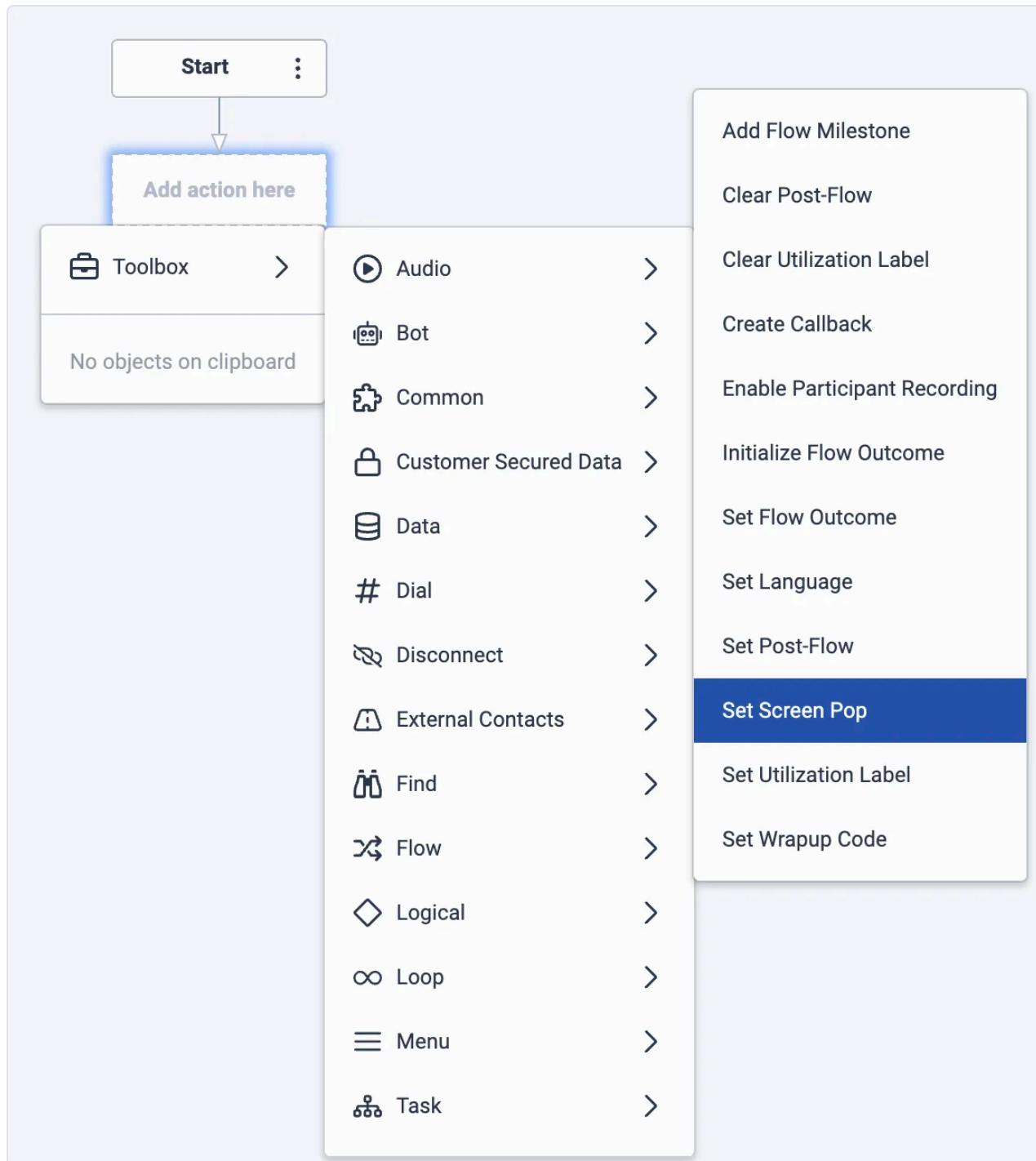
2. First thing we need to do: Add a new **Reusable Task**, and set it as the starting task.



3. Remove the Initial Greeting, and rename the task.



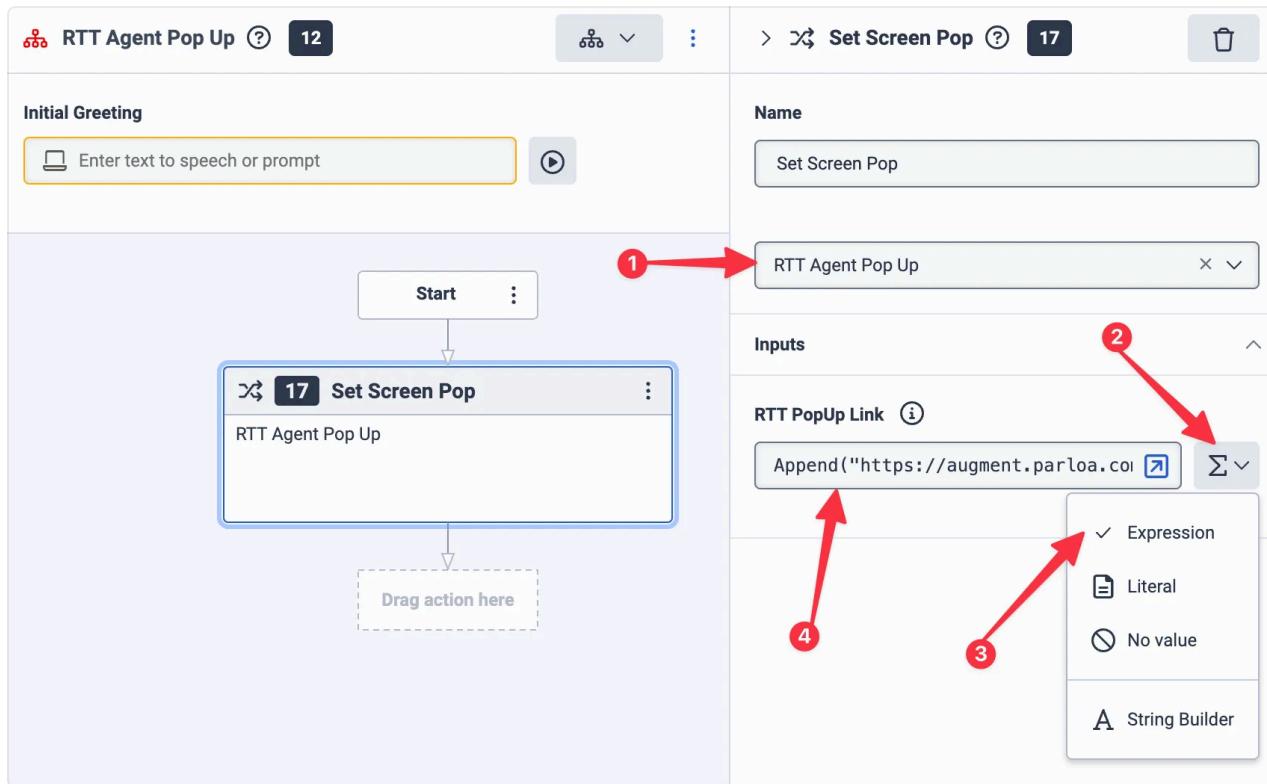
4. Now we'll focus on the flowchart. Add a Set Screen Pop block.



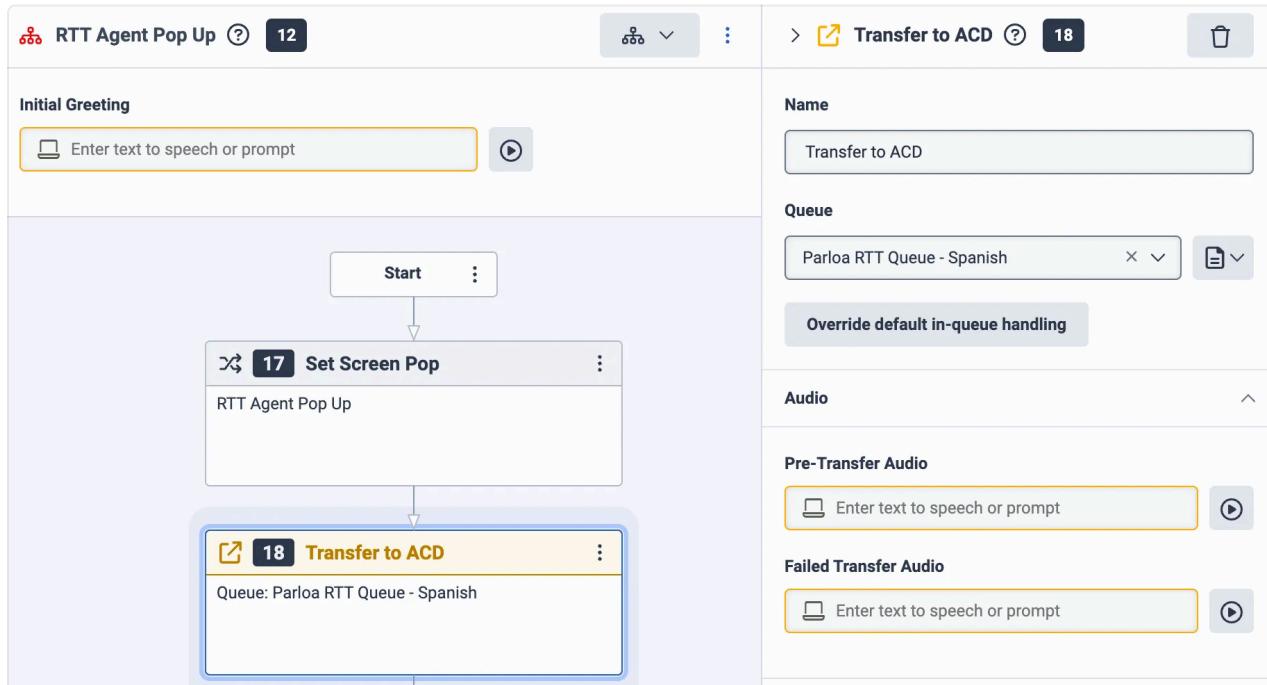
5. Select the Script we created in the previous section. In the **Inputs**, change to **Expression** and paste the following:

```
Append("https://augment.parloa.com/api/v0/conversations/", Call.UUIDData,
```

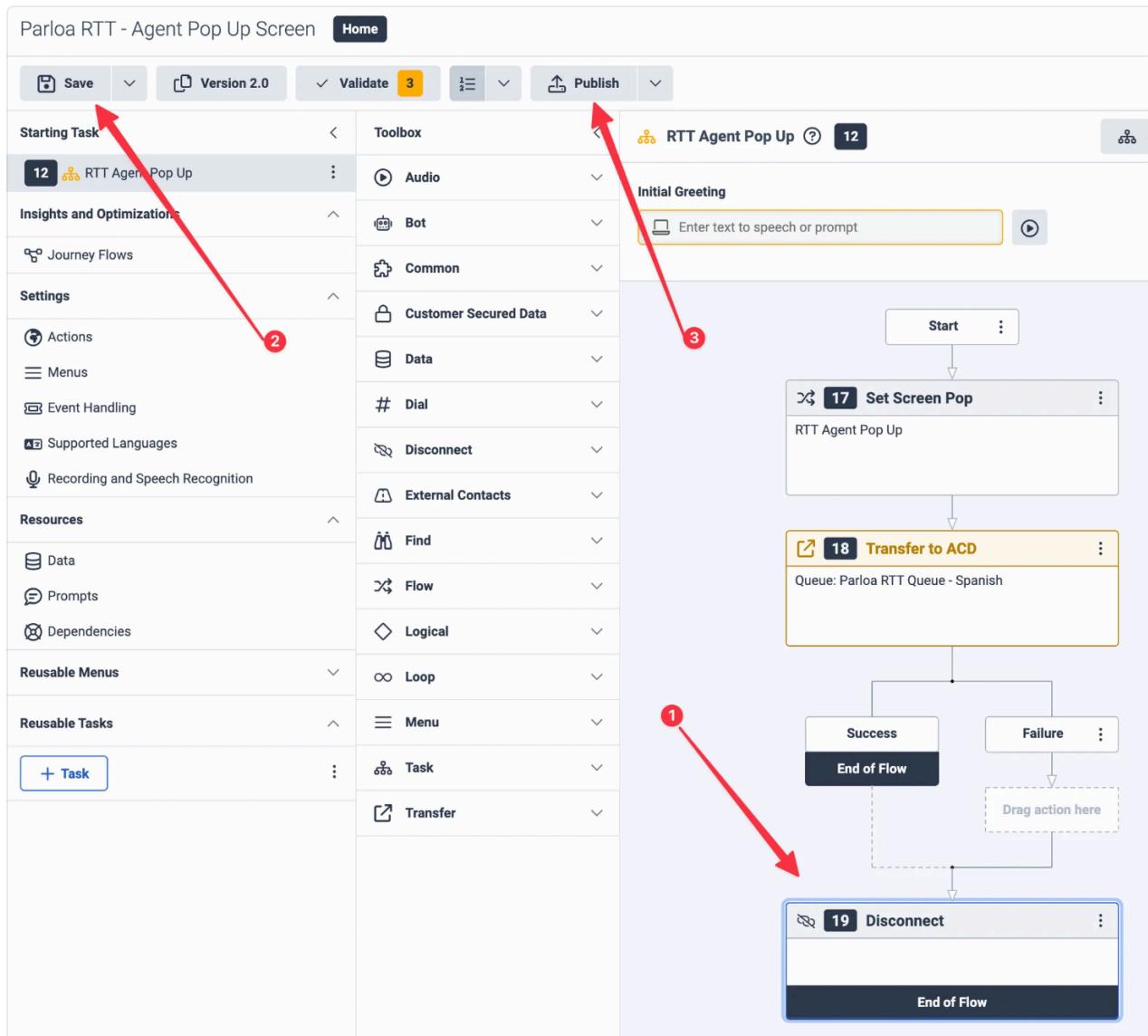
This will allow the the Script to open the correct conversation in the Augmented UI that corresponds to the call the agent received in Genesys.



- Now that we have our script ready, we need to transfer to a queue. Add a **Transfer to ACD** block, and choose your preferred queue. In our example, our agents will speak Spanish so we are going to transfer to a Spanish speaking queue.



- Lastly, add a **Disconnect** block in the end of the flowchart. After this, click **Save** and **Publish**.



Call Routing to the Agent Queue

Now that we have our Inbound Call Flow and Script for the agents, let's configure another route to trigger this new Inbound Flow.

Navigate to **Admin → Routing → Call Routing**, and click **+ Add** to create a new route. Give it a name, select the **Inbound Call Flow** we just created for the Agent Inbound Flow, and select the virtual number that you configured in the Augmented UI.

Call Route

Basic Information

Name *

Division *

Home

Only people with Routing > Call Route > View permission in this division can use this call route.

When is this call route open?

Always

Based on a schedule group

Select...

The schedule group defines open, closed, and holiday times to match against. A different call flow will be used for each schedule.

This call route closes in emergency cases

What call flow should be used?

Route to *

Inbound Numbers

Select...

+49 9990 0000



Translation

Translation Engines

Glossaries

Telephony

→ Inbound Flow

← Outbound Flow

Call Management

Agent Queues by Language Pairing

Queue Identifier

sip:+4999900000@parloa-rtt-inbound.b

Augmented UI

And you're done! If you followed these steps properly, you should be able to reach Parloa RTT platform, then your agent will receive Parloa's outbound call and should automatically open up the corresponding call in the Agent Desktop Screen

The screenshot shows the RTT Documentation software interface. At the top, there is a navigation bar with links to Activity, Directory, Documents, Clients, Performance, Apps, and Admin. On the left side, there is a vertical sidebar with various icons for navigation. The main area is divided into two sections: 'Conversations' and 'Dashboard'.
Conversations Section:
- A card for a call with 'Berlin, Germany' (Queue: Parloa RTT Queue - Spanish) with a duration of 0:30.
- Buttons for 'Answer' and 'Decline'.
Dashboard Section:
- A title 'Dashboard' with a back arrow.
- A message: 'Call active - started 00:32min ago'.
- A section for 'Berlin, Germany' with a blue phone icon and a red 'Leave Call' button.
- Information: Agent: es-ES, Caller: de-DE, ID: 7f8b4, Suggestions: GenAI.
- A note: 'Call started: 05.11.2024 - 15:07'
- A message: 'Berlin, Germany joined 15:07:43'.
- A note: 'Agent joined 15:07:44'.
- A text input field: 'Write a message in undefined...' with a 'Send' button.