AVAYA CPaaS

Integrating Avaya CPaaS through SIP

Note: This integration is only available for the **U.S.** and **Canada**. Numbers outside of this region will be supported in a future release.

BRIEFING

Avaya CPaaS can be integrated with other communication platforms that are SIP compatible where Avaya CPaaS becomes the engine for another program. This document describes configuring SIP Trunking without requiring XML.
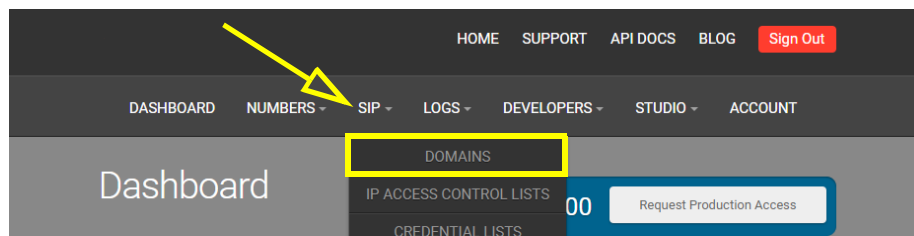
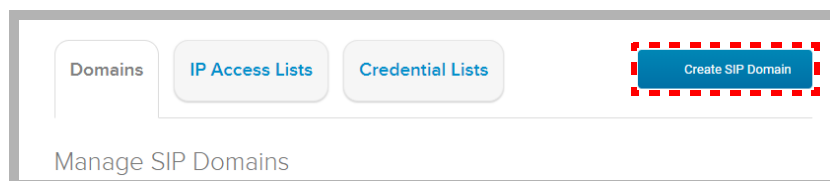Avaya CPaaS must be configured to provide the necessary details to the other applications.

LAUNCH

Open a web browser, go to **cloud.zang.io**, and login using your corporate credentials.

Create a Domain

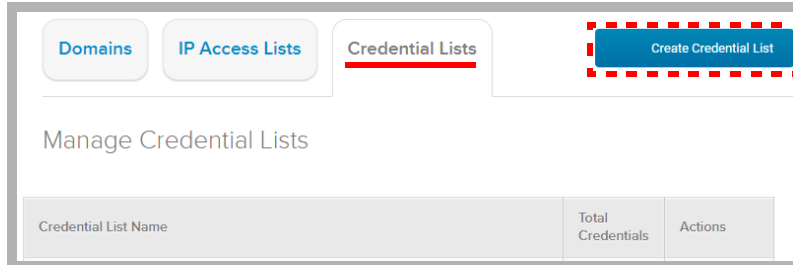- From the Dashboard, go to **SIP > Domains**.

- Click **Create SIP Domain**.

- Give the domain a name. This name will have **.sip.zang.io** automatically appended to it. Enter a human friendly name for convenience, then click **Create Domain**.

Create User Credentials

- From the Dashboard, go to **SIP > Domains**.
- Go to the **Credential Lists** tab, and click **Create Credential List**.

- Enter a name for the list and click **Create List**.

- Once the list has been created, click **Add Credential**. Enter a person's name, give them a username and a password to access this connection, then click **Create Credential**.
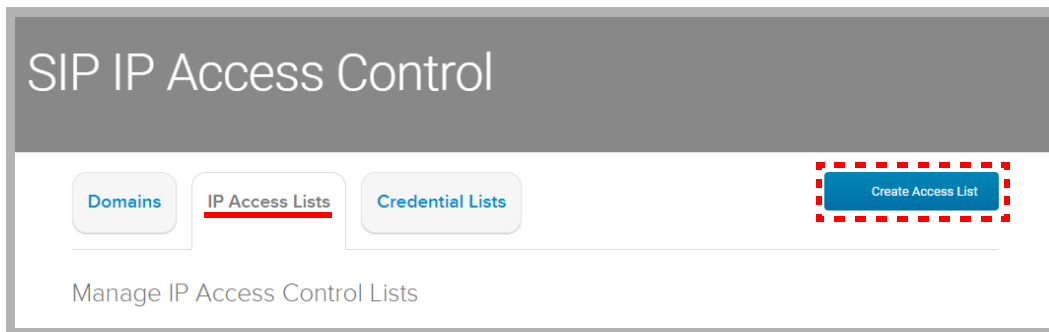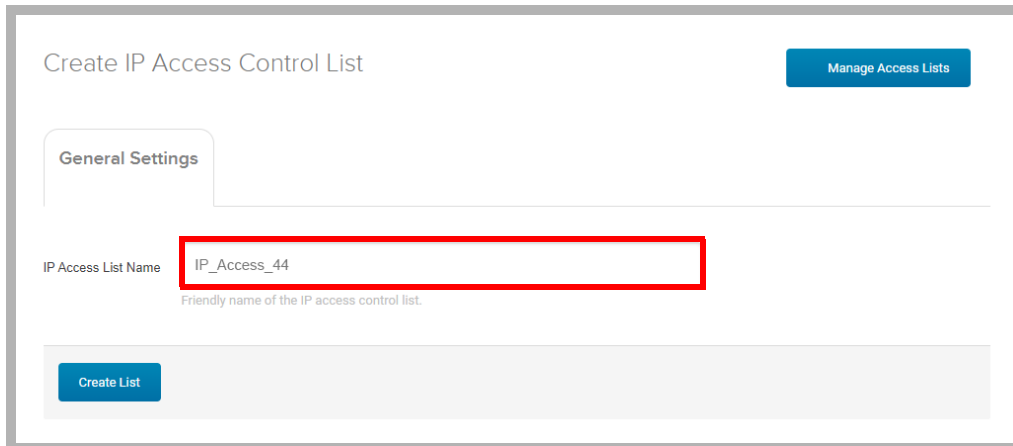
Create IP Access Control List (ACL)

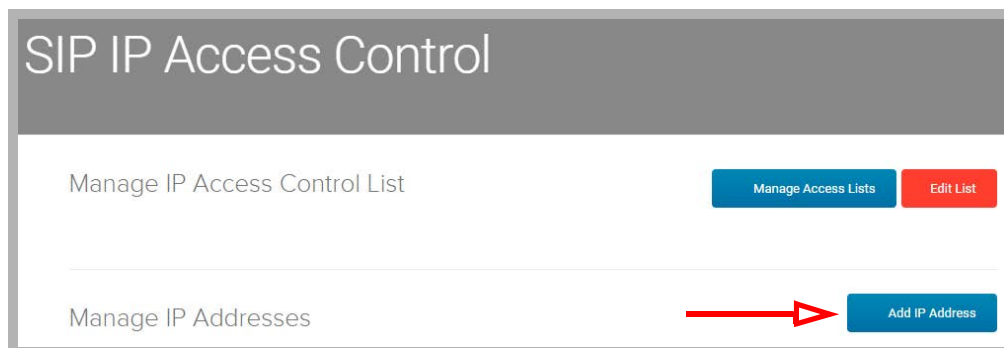- From the Dashboard, go to **SIP > Domains**.
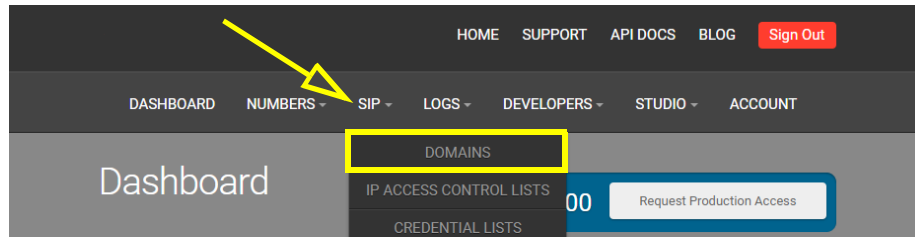
- Go to the **IP Access Lists** tab and select **Create Access List**.

- Enter a name for the list and click **Create List**.

- Click **Add IP Address**.

- Provide a user friendly name and enter an IP address. This is the publicly accessible IP address of your PBX for outbound traffic. When ready, click **Create IP Address**.

Create IP Address

Manage Access List

General Settings

Friendly Name	Your_Address_55
	Friendly name for this IP address.
IP Address	111.222.222.000
	Full and correct IP address (without ranges) that you wish to whitelist.

Create IP Address

Grant Credential Access

Once the Credentials List has been created, you must grant access to the list.
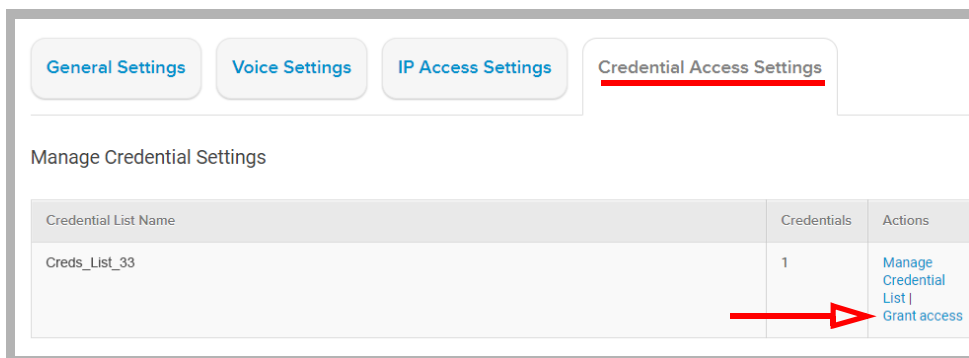
- From the Dashboard, go to **SIP > Domains**.
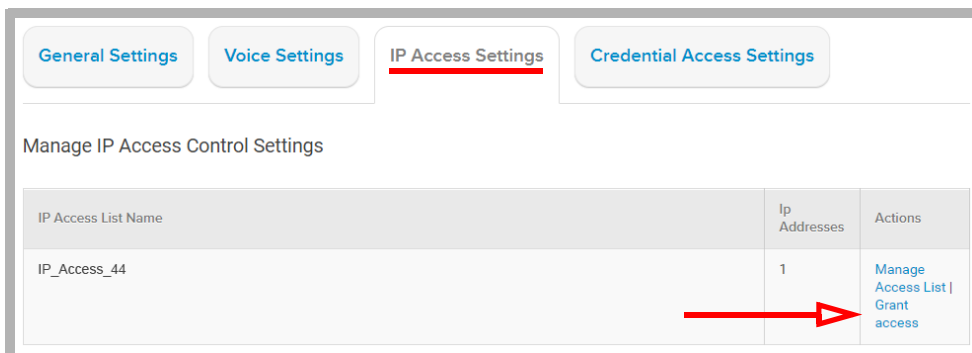- Click the name of the domain you just created.
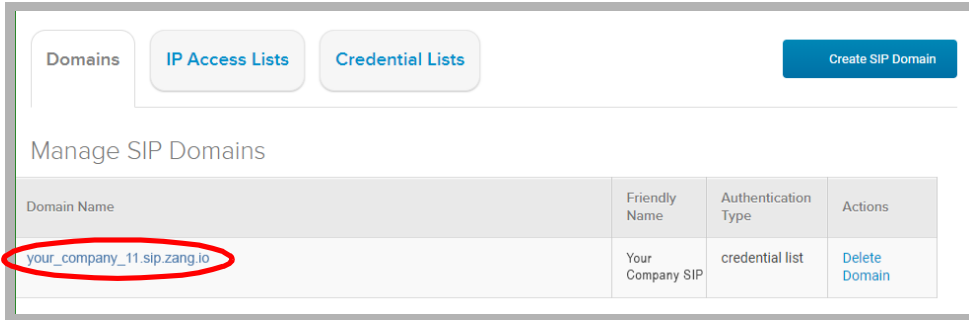
- Open the **Credential Access Settings** tab.

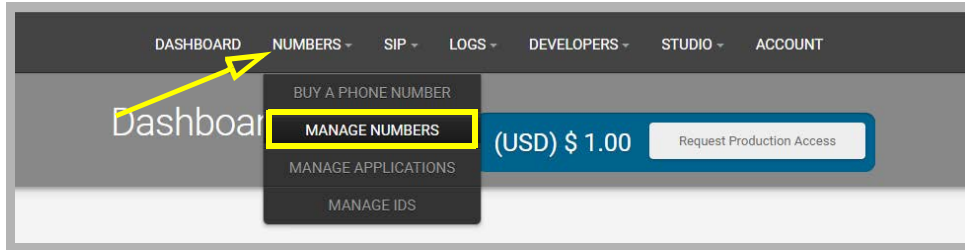- In the Actions column, click **Grant Access**.

- Open the **IP Access Settings** tab. In the Actions column, click **Grant Access**.
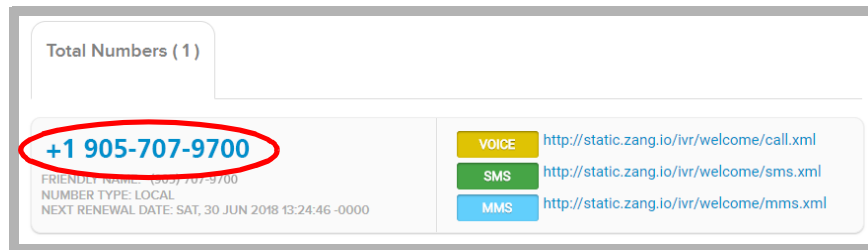
Incoming Traffic

Now that the domain has been created and access configured, you need to direct incoming calls to your SIP endpoint.

- From the Dashboard, go to **Numbers > Manage Numbers**.

- Select the number you want to use.

Hint: If you do not already have one, you will be prompted to purchase a number.

- Under **Voice Request URL**, enter **sip://** followed by the public IP address for your SIP calls. For example: sip://111.222.111.000. When ready, click **Save**.

> **Note**: If you want to use a non-standard port, or use TCP instead of UDP, use the following format:
>
> sip://111.222.111.000**:5062;Transport=TCP**

> **Caution**: This procedure will not work with Applications (e.g. **Numbers > Manage Applications**). It must be performed for each number separately.

> **Note**: If you want to use your Avaya CPaaS number with XML in the future, you will need to contact customer support for assistance.

OUTGOING TRAFFIC

The preceding steps for creating the credentials and the IP ACL are required for outgoing traffic. There are additional configurations required on the PBX / SIP endpoint. Where these are found depends upon the hardware that you are using. Consult with your product documentation for full details.

In the appropriate places on your hardware, configure the following:

- **Domain Name /Realm**: Use the domain created above: **your_company_11.sip.zang.io** .
- **Proxy Address**: Use **trunk.zang.io** . Or if you want to use SRV records: **us-east-trunk-zang.io** .
- **Proxy Port**: **5065** .

For your network requirements (i.e. firewall) here are some details.

- **Source IPs for SIP Signaling**: 104.198.251.179   35.232.140.209
- **Source IP and Port Range for media**: Any. CPaaS is not involved in the media path as media flows directly from our peers to your SIP Endpoint / PBX.