Twilio Elastic SIP Trunking
Configuration Blueprint

Avaya Aura Communication Manager and Session Manager with:
Avaya Session Border Controller for Enterprise

October 2022
Abstract
This document outlines configuration steps required to integrate Avaya's Aura CM, SM Contact Center with Twilio's Elastic SIP Trunking. Third-party Enterprise -Grade validation testing of these configurations was conducted by the engineers at tekVizion Labs™.

Notice
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Twilio Elastic SIP Trunking (ESIPT)
Twilio Elastic SIP Trunking is a cloud-based solution that provides connectivity for IP-based communications infrastructure to connect to the PSTN for making and receiving telephone calls to the rest of the world via any broadband internet connection. Twilio’s Elastic SIP Trunking service automatically scales, up or down, to meet your traffic needs with unlimited capacity. In just minutes you can deploy globally with Twilio's easy-to-use self-service tools without having to rely on slow providers.

Sign up for a free Twilio trial and learn more about configuring your Twilio Elastic SIP Trunk.

Please note: The IP Addresses, FQDN and configuration names and details given in this document are used for reference purposes only. These same details cannot be used in customer configurations. End users of this document can use the configuration details according to their network requirements.
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5.1 Create an IP-ACL rule

Figure 80 ESIPT Regional Edge URLs

5.2 Associate Phone Numbers on your Trunk

6 TekVizion

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1 Audience

This document is intended for technical staff which have installation and operational responsibilities for the technologies described within this document, including: Twilio Elastic SIP Trunking, Avaya Aura Communication Manager (Avaya Aura CM), Avaya Aura Session Manager (Avaya Aura SM) with Avaya Session Border Controller for Enterprise (Avaya SBCE), to connect to Twilio’s inbound and outbound PSTN Connectivity capabilities.

2 Lab Configuration

The network for the SIP trunk reference configuration is illustrated below and is representative of Avaya Aura CM and Avaya Aura SM with Avaya SBCE configuration with twilio.
2.1 Hardware Components

- UCS-B200 VMWare server running ESXi 6.0 or later used for the following virtual machines
  - Avaya Aura
    - Communication Manager
    - Session Manager
    - Modular Messaging
- Avaya SBCE running on Dell CAD – 208 hardware appliance
- Avaya IP Phone IP Phone(s)– 9630G

2.2 Software Requirements

- Avaya Aura
  - Session Manager: 8.1.3.2
  - Communication Manager: 8.1.0.0
  - System Manager: 8.1.3.2
- Avaya Session Border Controller for Enterprise : 8.1.3.1
3 Features

3.1 Features Supported

- Basic calls using G.711ulaw
- International Call
- Call Transfer
- Call Forwarding
- Call Waiting
- Three-Way Calling
- Call Hold and Resume
- Calling Number Presentation and Restriction
- Busy-out PBX endpoint
- DTMF Inband and RFC2833

3.2 Features Not Supported

- None

3.3 Features Not Tested

- None

3.4 Caveats and Limitations

- Avaya SBC has a limitation to consume the certificate bundle shared by twilio. Because of this the systems administrator will need to split the CA certificates in the bundle and install individually. (Avaya ticket ID 1-19181728682)
  - During testing tekVizion observed importing only the DigiCert certificates from the Twilio bundle was sufficient to enable the TLS certificate verification
4 Avaya Configuration

4.1 Avaya Configuration Checklist

In this section we present an overview of the steps that are required to configure Avaya Aura CM, Avaya Aura SM and Avaya SBCE for SIP Trunking with twilio.

<table>
<thead>
<tr>
<th>Steps</th>
<th>Description</th>
<th>Reference</th>
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<tbody>
<tr>
<td>Step 1</td>
<td>Avaya Aura CM Configuration</td>
<td>Section 4.3</td>
</tr>
<tr>
<td>Step 2</td>
<td>Avaya Aura SM Configuration</td>
<td>Section 4.4</td>
</tr>
<tr>
<td>Step 3</td>
<td>Avaya SBCE Configuration</td>
<td>Section 4.5</td>
</tr>
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</table>

Table 1 – PBX Configuration Steps

4.2 IP Address Worksheet

NOTICE: The specific values listed in the table below and in subsequent sections are used in the lab configuration described in this document and are for illustrative purposes only. The customer must obtain and use the values for your deployment.

<table>
<thead>
<tr>
<th>Component</th>
<th>Lab Value</th>
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<tbody>
<tr>
<td>Avaya SBCE</td>
<td></td>
</tr>
<tr>
<td>LAN IP Address</td>
<td>10.89.33.223</td>
</tr>
<tr>
<td>LAN Subnet Mask</td>
<td>255.255.255.0</td>
</tr>
<tr>
<td>Avaya Aura CM</td>
<td></td>
</tr>
<tr>
<td>IP Address</td>
<td>10.80.33.204</td>
</tr>
<tr>
<td>Subnet Mask</td>
<td>255.255.255.0</td>
</tr>
<tr>
<td>Avaya Aura SM</td>
<td></td>
</tr>
<tr>
<td>IP Address</td>
<td>10.80.33.207</td>
</tr>
<tr>
<td>Subnet Mask</td>
<td>255.255.255.0</td>
</tr>
</tbody>
</table>

Table 2 – IP Addresses
4.3 Avaya Aura CM Configuration

This section provides screen shots taken from Avaya Aura CM and were used for the interoperability testing. These screen shots provide a general overview of the PBX configuration.

4.3.1 Avaya Aura CM Login

- Avaya Aura CM configuration is done via SAT simulator through PuTTY.
- Log in using an appropriate User ID and Password.

Figure 2: Avaya Aura CM login
4.3.2 IP Node Name

- Use the Change node-names ip command to verify that node names are defined for Avaya Aura CM (procr) and Session Manager (Lab133-SM81). The node names are needed for configuring the Signaling Group.

![PutTY Command Output](image)

*Figure 3 IP Node Name*
4.3.3 IP Codec Set

- Use `change ip-codec-set 1` to define a list of codecs for calls between Avaya Aura CM and SM.

![Figure 4 IP Codec Set](image)
4.3.4 IP Network Region

- Use change ip-network-region 1 to define the network region
- Authoritative Domain: Domain name lab.tekvizion.com
- Codec Set: Enter codec set 1 created in Section 4.3.1
- Intra-region IP-IP Direct Audio: yes
- Intra-region IP-IP Direct Audio: yes

![Figure 5 IP Network Region]
4.3.5 Signaling Group

- Command add signaling group 2 was used to create Signaling Group. Use change signaling group 2 to modify existing signaling group.
- Set Group Type: sip
- Set Transport Method: tcp
- Set Peer Detection Enable: y
- Set Near-end Node Name: procr
- Set Near-end Listen Port: 5060
- Set Far-end Node Name: Lab133-SM81
- Set Far-end Listen Port: 5060
- Set Far-end Network Region: 1
- Set Far-end Domain: lab.tekvizion.com
- Set DTMF over IP: rtp-payload
- Set Direct IP-IP Audio Connections: n
- Leave other fields to default value

Figure 6 Signaling Group
4.3.6 Trunk Groups

- Trunk group 1 is used for trunk to Avaya SM. Command `add trunk group 1` was used to create Trunk Group. Use `change trunk group 1` to modify existing trunk group.
- Set **Group Type**: sip
- Set **Group Name**: PSTN
- Set **TAC**: #001
- Set **Direction**: two-way
- Set **Service Type**: public-ntwrk
- Set **Member Assignment Method**: auto
- Set **Signaling Group**: 2 (created in section 4.3.3)
- Set **Number of Members**: 10

![Figure 7 Trunk Group](image)
- Set Preferred Minimum Session Refresh Internal (sec): 900

Figure 8 Trunk Group Continuation
- Set Numbering Format: private

Figure 9 Trunk Group Continuation
- Set Telephone Event payload Type: 101
- Set Identity for calling Party Display: From
- Leave all other fields to default values

Figure 10 Trunk Group Continuation
4.3.8 Route Pattern

- Use change-route-pattern x command to specify the routing preference. Route pattern 1 is used for SIP trunk to Avaya SM.
- Set Pattern Name: PSTN
- Set Grp No: 1 (created in Section 4.3.4)
- Set FRL: 0
- Set Numbering Format: unk-unk
- Leave all other fields to default values

![Figure 11 Route Pattern](image_url)
4.3.9 Outbound Call Routing

For outbound call to twilio, Automatic Route Selection (ARS) is used. Use command change ars analysis x to configure the routing table.

- Set **Dialled String**: 214242
- Set **Min**: 10
- Set **Max**: 12
- Set **Route Pattern**: 1 (created in section 4.3.5)
- Set **Call Type**: natl (for national and intl for International dialing)

![Figure 12 Outbound Call Routing](image_url)
4.3.10 Inbound Call Routing

twilio sends 10 digit DID numbers to Communication Manager via Session Manager for Incoming calls. The command change inc-call-handling-trmt trunk-group 1 is used to terminate the calls to proper destinations.

- Set Number Len: 10 is used for example.
- Set Number Digits: Input twilio assigned DID numbers.
  - Set Del: 10 is used for example.
- Set Insert: The proper target Extension Number is given for each assigned DID account.

![Figure 13 Outbound Caller ID](image)

4.4 Avaya Aura Session Manager Configuration

4.4.1 Avaya Aura SM login

- Avaya Aura Session Manager Configuration is accomplished through the Avaya Aura System Manager
- Access Avaya Aura System Manager Web login screen via https://<IP Address/FQDN>
- Enter the login credentials
- Click Log On
Figure 14 Avaya Aura SM login
4.4.2 Domain

- Navigate to Elements > Routing

![Figure 15 Routing](image)

- Navigate to Routing > Domains
- Click New

![Figure 16 Add Domain](image)
• Set Name: Enter the domain name of Avaya Aura PBX, lab.tekvizion.com
• Set Type: sip
• Click Commit (not shown here)

![Figure 17 Domain](image1.png)

4.4.3 Locations

• Navigate to Routing > Locations
• Select New

![Figure 18 Locations](image2.png)
• Set Name: Lab133_81

![Location Details](image)

**Figure 19** Locations continuation

• Under *Location Pattern*, select Add to add IP Address Patterns for different networks that communicates within the location
• Set *IP Address Pattern*: 10.80.33.x
• Leave all other fields to default values
• Click Commit

![Location Pattern](image)

**Figure 20** Locations continuation
4.4.4 Adaptations

- twilio uses E164 numbering format for SIP Trunking Service. Adaptation was created at the Session Manager to manipulate the digits sent to twilio network via Avaya Session Border Controller for Enterprise (Avaya SBCE).
- Navigate to Routing > Adaptations. Click New
- Set Adaptation Name: Adaptation_For_Twilio
- Set Module Name: DigitConversionAdapter
- Set Module Parameter Type: Name-Value Parameter is selected from the drop down, Click Add
- Set Name/Value: fromto/true
- Set Name/Value: osrcd/10.89.33.207 (Avaya Aura SM IP is entered)
- Set Name/Value: odstd/10.89.33.223 (Avaya SBCE LAN IP is entered)
- Under Digit Conversion for Incoming Calls to SM, click Add

<table>
<thead>
<tr>
<th>Matching Pattern</th>
<th>Min/Max</th>
<th>Delete Digits</th>
<th>Address to Modify</th>
</tr>
</thead>
<tbody>
<tr>
<td>+15675</td>
<td>12/36</td>
<td>2 – Deletes +1 from +15675 patterns</td>
<td>Destination – Modifies digits in TO header and sends it to Avaya CM</td>
</tr>
</tbody>
</table>

*Figure 21 Digit Conversion to Avaya CM*

- Under Digit Conversion for Outgoing Calls from SM, click Add

<table>
<thead>
<tr>
<th>Matching Pattern</th>
<th>Min/Max</th>
<th>Delete Digits</th>
<th>Insert Digits</th>
<th>Address to Modify</th>
</tr>
</thead>
<tbody>
<tr>
<td>214242</td>
<td>-10/36</td>
<td>0</td>
<td>+1 – Insert +1 in front of 214242 patterns</td>
<td>Destination – Modifies the digits in TO header and sends it to twilio</td>
</tr>
</tbody>
</table>

*Figure 22 Digit Conversion to twilio*

- Leave all other fields at default values
- Repeat the same for all your outbound dial DID individually.
Click Commit

Figure 23 Adaptation for twilio
4.4.5 SIP Entities and Entity Links

SIP Entity for Avaya Aura Session Manager

- Navigate to: Elements >Routing > SIP Entities.

Figure 24 SIP Entity for Avaya SM
- Click New
- Set Name: Enter name of the host, Lab133_SM81
- Set FQDN or IP Address: Enter the SIP address of the Session Manager
- Set Type: Session Manager is selected from the drop down
- Set Location: Select the location (created in Section 4.4.3)
- Under Listen Port:
  - Set TCP/TLS Failover Port: 5060/5061
- Click Add to assign Domain lab.tekvizion.com for the following Ports and Protocol

![Figure 25 SIP Entity for Avaya SM continuation](image)
- Port 5060 and Protocol TCP/UDP
- Port 5061 and Protocol TLS
- Click Commit

Figure 26 SIP Entity for Avaya SM continuation
SIP Entity and Entity Links for Avaya Aura Communication Manager

- Set Name: Lab133_CM81
- Set FQDN or IP Address: Enter the IP address of Avaya Aura Communication Manager
- Set Type: CM
- Click Commit

![SIP Entity Details](image)

* Figure 27 SIP Entity and Entity Links for Avaya CM

- Under Entity Links, Click New

![Entity Links](image)

* Figure 28 SIP Entity and Entity Links for Avaya CM continuation
- Set Name: Lab133-SM81_Lab133CM_SIP_TCP_5060_TCP
- Set SIP Entity 1: Select the SIP entity Lab133-SM81
- Set SIP Entity 2: Lab133-CM81
- Set Protocol: TCP
- Set Ports: 5060
- Set Connection Policy: trusted
- Leave all other fields to default values
- Click Commit

![Entity Links](image)

**Figure 29 SIP Entity and Entity Link for Avaya CM continuation**

**SIP Entity and Entity Links for Avaya SBCE**

- Set Name: SIP ENTITY_ESBC_TWILIO
- Set FQDN or IP Address: Enter the IP address of Avaya SBCE interface facing Avaya Aura SM
- Set Adaptation: Select the Adaptation for Avaya SBCE configured in Section 4.4.4
- Set Location: Select the location created in Section 4.4.3
- Click Commit

![SIP Entity Details](image)

**Figure 30 SIP Entity and Entity Link for Avaya SBCE**
Under Entity Links, Click New

- Set Name: TWILIO
- Set SIP Entity 1: Select the SIP Entity Lab133-SM81
- Set SIP Entity 2: SIP ENTITY_ESBC_TWILIO
- Set Protocol: UDP
- Set Ports: Set both Ports to 5060
- Set Connection Policy: trusted
- Leave all other fields to default values
- Click Commit

---

**Figure 31** SIP Entity and Entity Link for Avaya SBCE continuation

**Figure 32** SIP Entity and Entity Link for Avaya SBCE continuation
4.4.7 Routing Policies

Routing policy to Avaya Aura CM

- Navigate to: Routing > Routing Policies. Click New
- Set Name: SM_to_CM
- Click Select under SIP Entity as Destination and the SIP Entities window is displayed

![Routing Policy Details](image)

- Check the radio button beside Lab133-CM81 as destination SIP Entity (configured in Section 4.4.5)
- Click Select and return back to Routing Policy Details page

![SIP Entities](image)

**Figure 33 Routing Policy for Avaya CM**

**Figure 34 Routing Policy for Avaya CM continuation**
Leave all other fields at default values

- Click Commit

Routing Policy Details

General

* Name: to_Avaya_TWILIO_ESBC

Disabled: [ ]

* Retries: 0

Notes: [ ]

SIP Entity as Destination

<table>
<thead>
<tr>
<th>Select</th>
<th>FQDN or IP Address</th>
<th>Type</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td>La8112-CMB1</td>
<td>10.99.33.204</td>
<td>CM</td>
<td></td>
</tr>
</tbody>
</table>

Figure 35 Routing Policy for Avaya CM continuation

Routing policy to Avaya SBCE

- Set Name: to_Avaya_TWILIO_ESBC
- Click Select under SIP Entity as Destination and SIP Entities window is displayed.

Routing Policy Details

General

* Name: to_AvAYA_TWILIO_ESBC

Disabled: [ ]

* Retries: 0

Notes: [ ]

SIP Entity as Destination

Select

Figure 36 Routing Policy for Avaya SBCE
- Check the radio button beside SIP ENTITY_ESBC_TWILIO as destination SIP Entity (configured in Section 4.4.5)
- Click Select and return back to Routing Policy Details page

Figure 37 Routing Policy for Avaya SBCE continuation

- Leave all other fields to default values
- Click Commit

Figure 38 Routing Policy for Avaya SBCE continuation
4.4.8 Dial Patterns

Dial Pattern for Avaya Aura CM

- Navigate to: Routing > Dial Patterns. Click New
- Set Pattern: 5675 (first 4 digit of Twilio DID assigned to the PBX phone)
- Set Min: 4
- Set Max: 36
- Under Originating Locations and Routing Policies, Click Add, at the new window
  - Originating Location: Select Lab133-81 (created in Section 4.4.3)
- Routing Policies: Select SM_to_CM under Routing Policies
- Click Select to return to Dial Pattern Details page
- Leave all other fields to default values.
- Click Commit

Figure 39 Dial Pattern to Avaya CM
Dial Pattern to twilio via Avaya SBCE

- Navigate to: Routing > Dial Patterns. Click New
- Set Pattern: 214242
- Set Min: 6
- Set Max: 12
- Under Originating Locations and Routing Policies, Click Add, at the new window
  - Originating Location: Select Lab133-81 (created in Section 4.4.3)
  - Routing Policies: Select to Avaya TWILIO_ESBC under Routing Policies
- Click Select to return to Dial Pattern Details page
- Leave all other fields to default values.
- Click Commit

![Dial Pattern Details](image)

**Figure 40 Dial Pattern to twilio via Avaya SBCE**

4.5   Avaya SBCE Configuration

4.5.1   Avaya SBCE login

- Log into Avaya Session Border Controller for Enterprise (SBCE) web interface by typing “https://X.X.X.X/sbc”.
- Enter the Username and Password
- Click Log In
Unauthorized access to this machine is prohibited. This system is for the use authorized users only. Usage of this system may be monitored and recorded by system personnel.

Anyone using this system expressly consents to such monitoring and is advised that if such monitoring reveals possible evidence of criminal activity, system personnel may provide the evidence from such monitoring to law enforcement officials.

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Figure 41 Avaya SBCE Login

- Under Device, select ASBCETwilio from drop down to expand the configuration for Avaya SBCE.

Figure 42 Selection of Avaya SBCE Device
4.5.2 Server Interworking

Server Interworking for Avaya SM

- Navigate to: Configuration Profiles > Server Interworking
- Select the predefined Interworking Profile avaya-ru, click Clone
- Set Clone Name: AASM8.1
- Click Finish

Figure 43 Server Interworking profile for Avaya SM
### General

**Hold Support**
- None
- RFC2543 - c=0.0.0.0
- RFC3264 - a=sendonly
- Microsoft Teams

<table>
<thead>
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<th>Handling</th>
<th>Option 1</th>
<th>Option 2</th>
<th>Option 3</th>
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<tr>
<td>180</td>
<td>None</td>
<td>SDP</td>
<td>No SDP</td>
</tr>
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<td>181</td>
<td>None</td>
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<td>182</td>
<td>None</td>
<td>SDP</td>
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<tr>
<td>183</td>
<td>None</td>
<td>SDP</td>
<td>No SDP</td>
</tr>
</tbody>
</table>

**Refer Handling**
- None

**URI Group**
- None

**Send Hold**
- None

**Delayed Offer**
- None

**3xx Handling**
- None

**Diversion Header Support**
- None

**Delayed SDP Handling**
- None

**Re-Invite Handling**
- None

**Prack Handling**
- None

**Allow 16X SDP**
- None

**T.38 Support**
- None

**URI Scheme**
- SIP
- TEL
- ANY

**Via Header Format**
- RFC3261
- RFC2643

**SIPS Required**
- None

**MediaSec Handling**
- None

---

**Figure 44** Server Interworking profile for Avaya SM Continuation

---

**Data-driven Customer engagement - at scale**

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### Interworking Profiles: AASM8.1

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#### General

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<td>Min-SE</td>
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<tr>
<td>Max Timer</td>
<td>---</td>
</tr>
<tr>
<td>Trans Expire</td>
<td>2 seconds</td>
</tr>
<tr>
<td>Invite Expire</td>
<td>---</td>
</tr>
<tr>
<td>Retry After</td>
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</tbody>
</table>

#### Timers

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<td>Record Routes</td>
<td>Both Sides</td>
</tr>
<tr>
<td>Include End Point IP for Context Lookup</td>
<td>Yes</td>
</tr>
<tr>
<td>Extensions</td>
<td>Avaya</td>
</tr>
<tr>
<td>Diversion Manipulation</td>
<td>No</td>
</tr>
<tr>
<td>Has Remote SBC</td>
<td>Yes</td>
</tr>
<tr>
<td>Route Response on Via Port</td>
<td>No</td>
</tr>
<tr>
<td>Relay INVITE Replace for SIPREC</td>
<td>No</td>
</tr>
<tr>
<td>MOBX Re-INVITE Handling</td>
<td>No</td>
</tr>
<tr>
<td>NATting for 301/302 Redirection</td>
<td>Yes</td>
</tr>
</tbody>
</table>

**Figure 45 Server Interworking profile for Avaya SM Continuation**
Server Interworking for twilio

**Figure 46 Server Interworking profile for Twilio**

<table>
<thead>
<tr>
<th>General</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Hold Support</td>
<td>None</td>
</tr>
<tr>
<td></td>
<td>RFC2543 - c=0.0.0.0</td>
</tr>
<tr>
<td></td>
<td>RFC3264 - a=sendonly</td>
</tr>
<tr>
<td></td>
<td>Microsoft Teams</td>
</tr>
<tr>
<td>180 Handling</td>
<td>None</td>
</tr>
<tr>
<td></td>
<td>SDP</td>
</tr>
<tr>
<td></td>
<td>No SDP</td>
</tr>
<tr>
<td>181 Handling</td>
<td>None</td>
</tr>
<tr>
<td></td>
<td>SDP</td>
</tr>
<tr>
<td></td>
<td>No SDP</td>
</tr>
<tr>
<td>182 Handling</td>
<td>None</td>
</tr>
<tr>
<td></td>
<td>SDP</td>
</tr>
<tr>
<td></td>
<td>No SDP</td>
</tr>
<tr>
<td>183 Handling</td>
<td>None</td>
</tr>
<tr>
<td></td>
<td>SDP</td>
</tr>
<tr>
<td></td>
<td>No SDP</td>
</tr>
<tr>
<td>Refer Handling</td>
<td></td>
</tr>
<tr>
<td>Send Hold</td>
<td></td>
</tr>
<tr>
<td>Delayed Offer</td>
<td></td>
</tr>
<tr>
<td>3xx Handling</td>
<td></td>
</tr>
<tr>
<td>Diversion Header Support</td>
<td></td>
</tr>
<tr>
<td>Delayed SDP Handling</td>
<td></td>
</tr>
<tr>
<td>Re-invite Handling</td>
<td></td>
</tr>
<tr>
<td>Prack Handling</td>
<td></td>
</tr>
<tr>
<td>Allow 18X SDP</td>
<td></td>
</tr>
<tr>
<td>T.38 Support</td>
<td></td>
</tr>
<tr>
<td>URI Scheme</td>
<td>SIP</td>
</tr>
<tr>
<td></td>
<td>TEL</td>
</tr>
<tr>
<td></td>
<td>ANY</td>
</tr>
<tr>
<td>Via Header Format</td>
<td>RFC3261</td>
</tr>
<tr>
<td></td>
<td>RFC2861</td>
</tr>
<tr>
<td>SIP/S Required</td>
<td></td>
</tr>
<tr>
<td>Media/sec Handling</td>
<td></td>
</tr>
</tbody>
</table>

Finish
Interworking Profiles: Twilio

<table>
<thead>
<tr>
<th>General</th>
<th>Timers</th>
<th>Privacy</th>
<th>URI Manipulation</th>
<th>Header Manipulation</th>
<th>Advanced</th>
</tr>
</thead>
<tbody>
<tr>
<td>Record Routes</td>
<td>Both Sides</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Include End Point IP for Context Lookup</td>
<td>Yes</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Extensions</td>
<td>Avaya</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
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<td>No</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Has Remote SBC</td>
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<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Route Response on Via Port</td>
<td>No</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Relay INVITE Replace for SIPREC</td>
<td>No</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>MOBX Re-INVITE Handling</td>
<td>No</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>NATing for 301/302 Redirection</td>
<td>Yes</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Figure 47 Server Interworking profile for Twilio Continuation
4.5.3 SIP Servers

SIP Server for Avaya SM

- Navigate to Services > SIP Servers
- Set **Profile Name**: Avaya_SM
- Set **Server Type**: Select Call Server from the drop down
- Set **IP Address/FQDN**: Enter the Avaya Aura Session Manager SIP IP Address
- Set **Port**: 5060
- Set **Transport**: UDP

![Image of SIP Server configuration](image)

**Figure 48 SIP Server for Avaya SM**

**Figure 49 SIP Server for Avaya SM Continuation**
SIP Server for Twilio

SIP Server for Twilio

Figure 50 SIP Server for Twilio
4.5.4 Topology Hiding

Topology hiding profile for Avaya SM

- Topology Hiding profiles are added for Avaya SM to overwrite and hide certain headers
- Navigate to: Configuration Profiles > Topology Hiding
- Select the newly created profile Avaya_SM and Click Edit
- Set Header: Request-Line, To, From are selected
- Set Replace Action: Overwrite
- Set Overwrite Value: lab.tekvizion.com
- Click Finish (not shown here)

Figure 51 Topology Hiding Profile for Avaya SM
Topology hiding profile for Twilio

Figure 52 Topology Hiding Profile for Twilio
4.5.5 Routing

Routing for Avaya SM

- Navigate to: Configuration Profiles > Routing
- Set Profile Name: Avaya_SM_routing
- Set Priority/Weight: 1
- Set SIP Server profile: select Avaya SM (configured in above SIP Servers section) from the dropdown (not shown here)
- The Server IP, Port and Transport Protocol populates automatically

Routing for Twilio

NOTE: Twilio’s Ashburn, VA USA edge was used for this testing. Please refer to the following for a full list of Twilio Edge URLs here.

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4.5.6 End Point Policy Groups

End Point Policy Group for Avaya SM

- A new End Point Policy Group is created for Avaya Aura Session Manager.
- Navigate to: Domain Policies > End Point Policy Groups
- Select default-low under Policy Groups
- Click Clone
- Set Clone Name: Avaya SM
- Click Finish

![Figure 55 End Point Policy Group for Avaya SM](image-url)
● Select the newly created Group Avaya SM, Click Edit
● Set Signaling Rule: Avaya SM
● Click Finish

![Edit Policy Set](image-url)

Figure 56 End Point Policy Group for Avaya SM Continuation
End Point Policy Group for twilio

- Repeat the same steps to create End Policy Group for twilio

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4.5.8 Network Management

- Navigate to: Network & Flows > Network Management > Interfaces.
- Interfaces which are enabled for Avaya LAN and Twilio are shown below

![Network Management Interfaces](image1)

- Navigate to: Network & Flows > Network Management > Networks.
- IP addresses which are configured for Avaya LAN and twilio interface are shown below

![Network Management Networks](image2)

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4.5.9 Media Interface

- Navigate to: Network & Flows > Media Interface.
- Set Name: Med_LAN is given here
- Set IP Address: Select LAN (A1, VLAN0) from the drop down and the IP address populates automatically. The IP address for Interface facing Avaya Aura SM is 10.89.33.223
- Set Port Range: 35000-40000

**Figure 60 Media Interface facing Avaya SM**

- Repeat the same steps to create a Media Interface facing twilio.

**Figure 61 Media Interface facing twilio**
4.5.10 Signaling Interface

- Navigate to: Network & Flows > Signaling Interface.
- Configure Signaling Interface towards Avaya SM LAN and twilio as shown below.

![Signaling Interface](image)

*Figure 62 Signaling Interface facing Avaya SM LAN and twilio*
4.5.11 Endpoint Flows

- Set **Flow Name**: Avaya SM
- Configure flow for Avaya SM LAN as shown below

![Edit Flow: Avaya SM](image)

**Figure 64 Endpoint Flows for Avaya SM LAN**
- Set *Flow Name*: Twilio
- Configure flow for twilio as shown below

![Edit Flow: Twilio](image)

*Figure 65 Endpoint Flows for twilio*
4.5.12 TLS Configuration

The following are necessary steps to modify the configuration from protocol UDP to TLS between Avaya SBCE and twilio:

- Navigate to: TLS management > Certificates. Click Install
- Set Type: Select CA Certificate
- Set Name: globalrootCA
- Set Allow weak Certificate/Key: Checked
- Set Certificate File: Click Choose File to select twilio Root CA (received from twilio)
- Click Upload

Note: - Avaya SBCE has a limitation to consume the CA Bundle certificate, if you received CA bundle from your customer then need to split the certificates and upload individually.

Figure 66 Upload twilio Root CA
Client Profile for twilio

- Navigate to: TLS management > Client Profiles. Click Add
- Set Profile Name: TWILIO is given for interface facing twilio
- Set Certificate: select server certificate asbce8.pem for Avaya SBCE interface facing twilio
- Set Peer Certificate Authorities: Select globalrootCA.crt which is uploaded in previous step
- Set Verification Depth: 5
- Click Next
- Set Version: Select all 3 TLS versions
- Click Finish
Server Profile for twilio

- Navigate to: TLS management > Server Profiles. Click Add
- Set Profile Name: TWILIO is given for interface facing twilio
- Set Certificate: Select server certificate asbce8.pem for Avaya SBCE interface facing twilio
- Set Peer Verification: None
- Click Next

Figure 69 Server Profile facing twilio
- Set Version: Check all 3 TLS versions
- Click Finish

![Edit Profile](image)

<table>
<thead>
<tr>
<th>Renegotiation Parameters</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Renegotiation Time</td>
<td>0 seconds</td>
</tr>
<tr>
<td>Renegotiation Byte Count</td>
<td>0</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Handshake Options</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Version</td>
<td>TLS 1.2, TLS 1.1, TLS 1.0</td>
</tr>
<tr>
<td>Ciphers</td>
<td>Default, FIPS, Custom</td>
</tr>
<tr>
<td>Value</td>
<td>HIGH:!DH:!ADH:!MD5:!eNULL:!eNULL:@STRENGTH</td>
</tr>
</tbody>
</table>

**Figure 70 Server Profile facing twilio** Continuation
Edit SIP Server

- Navigate to: Services > SIP Servers
- Under General tab, Click Edit
- Set Transport: Select TLS from Dropdown
- Set Port: 5061
- Set TLS Client Profile: Select Client Profile TWILIO
- Click Finish

Figure 71 SIP Server Profile – twilio
Configure SRTP

- Navigate to: Domain Policies > Media Rules
- Select Media Rule default-high-enc, Click Clone
- Set Clone Name: Twilio-mediarule
- Click Finish

![Figure 72 Media Rule – twilio](image-url)
- Select newly created Media Rule Twilio-mediarule, Click Edit
- Set Preferred Format #1: SRTP_AES_CM_128_HMAC_SHA1_32
  #2: SRTP_AES_CM_128_HMAC_SHA1_80
  #3: SRTP_AES_192_CM_HMAC_SHA1_32
- Click Finish

Figure 73 Media Rule – twilio Continuation
Edit End Point Policy Groups

- Navigate to: Domain Policies > End Point Policy Groups
- Select Twilio under Policy Groups
- Click Edit

Policy Groups: Twilio

Figure 74 Edit End Point policy Group – twilio
- Set **Media Rule**: Select Twilio-mediareule
- Click **Finish**

![Edit Policy Set](image)

*Figure 75 Edit End Point policy Group –twilio Continuation*

**Edit Signaling Interface**

- Navigate to: Network & Flows > Signaling Interface
- Select interface SI_WAN
- Click **Edit**

![Signaling Interface](image)

*Figure 76 Edit Signaling Interface – twilio*
Set TLS Port: 5061

- Set TLS Profile: Select TWILIO
- Set TCP/UDP Port: Delete the values as only TLS is used.
- Click Finish

Figure 77 Edit Signaling Interface – twilio continuation
Edit Server Flows

- Navigate to: Network & Flows > End Point Flows > Server Flows
- Select Server Flow Twilio, Click Edit

End Point Flows

![End Point Flows Diagram]

Figure 78 Edit Server Flow – twilio
- Set Transport: TLS
- Set End Point Policy Group: Select Twilio
- Click Finish

![Edit Server Flow - twilio continuation](image)

5 Twilio Elastic SIP Trunking Configuration

From your Twilio Console, navigate to the Elastic SIP Trunking area (or click on the icon on the left vertical navigation bar).

If Elastic SIP Trunking is not visible via the navigation bar, select “Explore Products +”, locate Elastic SIP Trunking from the center of the screen and click the thumb pin icon. Doing this will add Elastic SIP Trunking to the navigation bar.
5.1 Create an IP-ACL rule
Click on Authentication in the left navigation, and then click on IP Access Control Lists.

Create a new IP-ACL, for example the ACL list name used for this testing was “Tekvizion”, and add the public IP Addresses assigned to the Avaya SBCE(s).

5.2 Create a new Trunk
For each geographical region desired (e.g., North America, Europe), create a new Elastic SIP Trunk.

Now click on Trunks again on the left vertical navigation bar, and create a new Trunk.

Name your new SIP Trunk, then configure it in the following steps.
Under the General Settings you can enable different features as desired. See [ESIPT documentation](https://www.twilio.com/docs/sip) for more information.

**General Settings**

- **Friendly name**
  - `tekvizion`
  - A human-readable descriptive text, up to 64 characters long.

**Trunk SIP**

- `TRUNK:k8e5f4s0890f000c8afaci90f9e0720`

**Features**

To learn more about SIP Trunking features, please see our user documentation if you have the proper permissions to view.

- **Call Recording**
  - On by default. Calls will be recorded.

- **Recording Trim**
  - On by default. Silence will not be trimmed from recording.

- **Secure Trunking**
  - On by default. TLS must be used to encrypt SIP messages on port 5061, and SRTP must be used to encrypt the media packets. Any non-encrypted calls will be rejected.

- **Call Transfer (SIP REFER)**
  - On by default. Twilio will consume an incoming SIP REFER from your communications infrastructure and create an INVITE message to the address in the Refer-To header.

- **Caller ID for Transfer Target**
  - Set caller ID as Transferred

- **Enable PSTN Transfer**
  - On by default. Allow Call Transfers to the PSTN via your Trunk.

- **Symmetric RTP**
  - On by default. Twilio will detect where the remote RTP stream is coming from and start sending RTP to that destination instead of the one negotiated in the SIP.

In the Termination section, select a Termination SIP URI.

**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**

- `tekvizion`

- `psta.twilio.com`

- **Show Localized URIs**
Click on "Show localized URI’s" and copy and paste this information as you will use this on your SBC to configure your Trunk.

If you wish to manually connect to a specific geographic region, you may do so by pointing your communications infrastructure to any of the following localized Termination SIP URIs:

**Attention:** We have updated the syntax for localized SIP hostnames to use our new Edge Locations. View legacy Termination SIP URIs

<table>
<thead>
<tr>
<th>Region</th>
<th>URI</th>
</tr>
</thead>
<tbody>
<tr>
<td>North America Virginia</td>
<td>tekvizion.pstn.ashburn.twilio.com</td>
</tr>
<tr>
<td>North America Oregon</td>
<td>tekvizion.pstn.umatilla.twilio.com</td>
</tr>
<tr>
<td>Europe Dublin</td>
<td>tekvizion.pstn.dublin.twilio.com</td>
</tr>
<tr>
<td>Europe Frankfurt</td>
<td>tekvizion.pstn.frankfurt.twilio.com</td>
</tr>
<tr>
<td>South America Sao Paulo</td>
<td>tekvizion.pstn.sao-paulo.twilio.com</td>
</tr>
<tr>
<td>Asia Pacific Singapore</td>
<td>tekvizion.pstn.singapore.twilio.com</td>
</tr>
<tr>
<td>Asia Pacific Tokyo</td>
<td>tekvizion.pstn.tokyo.twilio.com</td>
</tr>
<tr>
<td>Asia Pacific Sydney</td>
<td>tekvizion.pstn.sydney.twilio.com</td>
</tr>
</tbody>
</table>

*Figure 80 ESIPT Regional Edge URLs*

Next, Assign the IP ACL ("Tekvizion") that was created in the previous step:

**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**

| Tekvizion | X |  | + |

**Credential Lists**

Click to select a Credential List

In the Origination section, we’ll need to add Origination URI’s to route traffic towards the Avaya SBCE. The recommended practice is to configure a redundant mesh per geographic region (in this context a region is one of North America,
Europe, etc). In this case, we configure two Origination URIs, each egressing from a different Twilio Edge.

Click on ‘Add New Origination URI’, we’ll depict the configuration for North America:

**Origination**

Incoming traffic to your communications infrastructure from the PSTN.

**Origination URIs**

Configure the IP address (or FQDN) of the network element entry point into your communications infrastructure (e.g. IP-PSX, SBC).

> Show more about provisioning for high service availability

<table>
<thead>
<tr>
<th>Origination URI</th>
<th>Priority</th>
<th>Weight</th>
<th>Enabled</th>
</tr>
</thead>
<tbody>
<tr>
<td>sip.192.65.79.180,edge,umatilla</td>
<td>10</td>
<td>10</td>
<td>✓</td>
</tr>
<tr>
<td>sip.192.65.79.179,edge,ashburn</td>
<td>10</td>
<td>10</td>
<td>×</td>
</tr>
</tbody>
</table>

**Add Origination URL**

<table>
<thead>
<tr>
<th>Origination URI</th>
<th>Priority</th>
<th>Weight</th>
</tr>
</thead>
<tbody>
<tr>
<td>sip.192.65.79.180,edge,umatilla</td>
<td>10</td>
<td>10</td>
</tr>
</tbody>
</table>

**Priority**

Numeric range from 0 to 65535.

**Weight**

Numeric range from 1 to 65535.

**Enabled**

Status: enabled

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5.2 Associate Phone Numbers on your Trunk

In the Numbers section of your Trunk, add the Phone Numbers that you want to associate with each Trunk. Remember to associate the Numbers from a given country in the right Trunk. For example, associate US & Canada Numbers with the North American Trunk and European Numbers with the European Trunk etc.
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- **Verification Testing** – Verification of interoperability performed on-site at tekVizion Labs between two products or in a multi-vendor configuration
- **Product Assessment** – independent assessment and verification of product functionality, interface usability, assessment of differentiating features as well as suggestions for added functionality, stress and performance testing, etc.

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