Twilio Elastic SIP Trunking Configuration Guide

Cisco CUCM 12.5(SU1) with Cisco vCUBE 14.1

June 2021
## Document History

<table>
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<th>Rev. No.</th>
<th>Description</th>
</tr>
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<tbody>
<tr>
<td>1.0</td>
<td>Twilio Elastic SIP Trunking Configuration Guide</td>
</tr>
<tr>
<td>1.1</td>
<td>Updated based on the feedback from Twilio</td>
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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 Trunk, Virtual Cisco Unified Border Element (vCUBE), and Cisco Unified Communications Manager (CUCM).

2 Document Overview
This configuration guide provides steps for configuring a Twilio Elastic SIP Trunk to a Virtualized Cisco Unified Border Element (vCUBE). The Cisco Unified Communication Manager (CUCM) was also validated and used throughout this testing.

2.1 Twilio Elastic SIP Trunking
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.

2.2 Cisco UBE and Cisco UCM
Cisco Unified Border Element (CUBE) and Cisco Unified Call Manager (CUCM) provide industry-leading reliability, security, scalability, efficiency, and enterprise call and session management and is the core call control application of the collaboration portfolio.

It should be noted that while this application note focuses on the optimal configurations for the Cisco UBE (CUBE) in an enterprise Cisco UCM (CUCM) 14.1 environment, the same SBC configuration model can also be used for other enterprise applications with a few tweaks to the configuration for required features.

In addition, it should be noted that the SBC configuration provided in this guide focuses strictly on the CUCM Server associated parameters. Many SBC applications may have additional configuration requirements that are specific to individual customer requirements. These configuration items are not covered in this guide.
For additional information on CUCM 12.5, please refer to: Cisco UCM 12.5 Information
For additional information on CUBE 14.1, please refer to: Cisco UBE 14.1 information
2.3 tekVizion Labs

tekVizion Labs™ is an independent testing and Verification facility offered by tekVizion PVS, Inc. (“tekVizion”). tekVizion Labs offers several types of testing services including:

- Remote Testing – provides secure, remote access to certain products in tekVizion Labs for pre-Verification and ad hoc testing
- 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.

tekVizion is a systems integrator specifically dedicated to the telecommunications industry. Our core services include consulting/solution design, interoperability/Verification testing, integration, custom software development and solution support services. Our services helps service providers achieve a smooth transition to packet-voice networks, speeding delivery of integrated services. While we have expertise covering a wide range of technologies, we have extensive experience surrounding our practice areas which include: SIP Trunking, Packet Voice, Service Delivery, and Integrated Services.

The tekVizion team brings together experience from the leading service providers and vendors in telecom. Our unique expertise includes legacy switching services and platforms, and unparalleled product knowledge, interoperability and integration experience on a vast array of VoIP and other next-generation products. We rely on this combined experience to do what we do best: help our clients advance the rollout of services that excite customers and result in new revenues for the bottom line. tekVizion leverages this real-world, multi-vendor integration and test experience and proven processes to offer services to vendors, network operators, enhanced service providers, large enterprises and other professional services firms. tekVizion’s headquarters, along with a state-of-the-art test lab and Executive Briefing Center, is located in Plano, Texas.

For more information on tekVizion and its practice areas, please visit tekVizion Labs website at www.tekVizion.com

3 Validation Network Components

The network for the Twilio Elastic SIP Trunk, vCUBE, and CUCM reference configurations is illustrated below:
3.1 Hardware Components (vCUBE and CUCM)
- UCS-C240 VMWare server running ESXi 5.5.0 used for CUCM
- UCS-C240 VMWare server running ESXi 6.7.0 used for vCUBE
- Cisco IP Phone 9971 and 7941G

3.2 Software Requirements (vCUBE and CUCM)
- CUCM v12.5.1.13900-152
- vCUBE v14.1 (SW Version: 17.3.3, Platform CSR1000v)
4 Features

4.1 Features Supported

- OPTIONS
- Basic Outbound Calls
- Basic Inbound Calls
- Calls with RTCP enabled and disabled (Both US and EMEA trunk)
- Mute/Unmute
- Call Cancellation
- Ringing Timeout
- User Busy
- Calling Invalid Extension
- Codec scenarios (G711/G729/OPUS)
- Fax (G711-Passthrough)
- DTMF (RFC2833)
- Toll-free call: 1-800-XXX-XXXX
- Emergency call
- ONND scenarios
- Anonymous call
- Hold/Resume (with/without MOH)
- Session Refresh
- Call Forward (CFA/CFNA/CFB)
- Transfer (Blind/Consultative)
- Conference
- Route Crankback
- Call Admission Control

4.2 Features Not Supported by vCUBE

- CSR vCUBE does not support transcoding. Consequently the following scenarios could not be executed
  - DTMF Inband (CUCM does not generate Inband for vCUBE to passthrough)
4.3 Caveats and Limitations

- The following Twilio data centers were used for the testing.
  - Twilio Public Ashburn, VA and Umatilla, OR edge
    - tekvizion.pstn.ashburn.twilio.com or tekvizion.pstn.twilio.com (for US trunk)
    - tekvizion.pstn.umatilla.twilio.com (for route crankback)
    - tekvizion.pstn.dublin.twilio.com (for EMEA trunk)
- It is required to confirm that the CSR nvram store contains “ios_core.p7b” certification bundle and there is no associated trustpoint configured.
- The entire test was executed only on TLS/SRTP. The TLS connection was only between vCUBE and Twilio.
- By design, Twilio includes a Diversion header for inbound calls. For the PBX call forward scenarios, as CUCM would also add Diversion during call forward, there were 2 Diversion headers in the call forwarded INVITE.
- CUCM was configured to generate Comfort Noise packets and vCUBE passed through the packets through Twilio.
- The RTCP disabled scenario were executed disabling RTCP on the CUCM

5 Configuration

5.1 Configuration Checklist

In this section we present an overview of the steps that are required to configure CUCM and vCUBE for SIP Trunking with Twilio Elastic SIP Trunking.

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</tbody>
</table>

5.2 IP Address Worksheet

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.

<p>| Table 2 – IP Addresses |</p>
<table>
<thead>
<tr>
<th>Component</th>
<th>Lab Value</th>
<th>Customer Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>vCUBE</td>
<td></td>
<td></td>
</tr>
<tr>
<td>LAN IP Address</td>
<td>10.64.5.189</td>
<td></td>
</tr>
<tr>
<td>LAN Subnet Mask</td>
<td>255.255.0.0</td>
<td></td>
</tr>
<tr>
<td>CUCM</td>
<td></td>
<td></td>
</tr>
<tr>
<td>IP Address</td>
<td>10.71.8.10</td>
<td></td>
</tr>
<tr>
<td>Subnet Mask</td>
<td>255.255.0.0</td>
<td></td>
</tr>
</tbody>
</table>

5.3 CUCM Configuration

This section leverages screen shots taken from CUCM used for the interoperability testing to provide a general overview of the PBX configuration.

5.3.1 CUCM Login and Version

Open an instance of a web browser and connect to the CUCM using the following address: https://<CUCMIP>

Log in using an appropriate user ID and password. Verify the system version being tested.

Figure 2: CUCM software version
5.3.2 CUCM SIP Profile Configuration

A new SIP Profile **Standard SIP Profile Twilio** was configured. To add a new SIP Profile, from the Device drop down menu,

1. Navigate to Device Settings  **SIP Profile**.
2. On the screen that appears, click **Add New** and configure the SIP Profile as below.
3. Then click **Save** and then **Apply Config**

---

**Figure 3 CUCM SIP Profile**
### Figure 4 CUCM SIP Profile Contd.

#### SIP Profile Configuration

- **SDP Information**
  - SDP Session-Level Bandwidth Modifier for Early Offer and Re-invite: TIM and AS
  - SDP Transparency Profile: <None>
  - Accept Audio Codec Preferences in Received Offer: Default
  - Require SMTP Inactive Exchange for Mid-Call Media Change: Default
  - Allow RR/RS Bandwidth Modifier (RFC 3556)

- **Parameters used in Phone**
  - Timer Invite Expires (seconds)
  - Timer Register Delta (seconds)
  - Timer Register Expires (seconds)
  - Timer T1 (milliseconds)
  - Retry INVITE
  - Retry Non-INVITE
  - Media Port Ranges
    - Common Port Range for Audio and Video
    - Separate Port Ranges for Audio and Video
  - Start Media Port
  - Stop Media Port
  - DSCP for Audio Calls: Use System Default
  - DSCP for Video Calls: Use System Default

### Figure 5 CUCM SIP Profile Contd.

#### SIP Profile Configuration

- **DSCP for Audio Portion of Video Calls**: Use System Default
- **DSCP for TelePresence Calls**: Use System Default
- **DSCP for Audio Portion of TelePresence Calls**: Use System Default
- **Call Pickup Group Other URI**: x-cisco-service-pickup
- **CM Pickup Group URI**: x-cisco-service-pickup
- **Host Me Service URI**: x-cisco-service-sp
- **User Info**: None
- **DTMF DS Level**: Nominal
- **Call Hold Ring Back**: Off
- **Anonymous Call Block**: Off
- **Caller ID Blocking**: Off
- **Do Not Disturb Control**: User
- **Telekern Level for 7940 and 7960**: Disabled
- **Resource Priority Namespace**: <None>
- **Timer Keep Alive Expires (seconds)**
  - 120
- **Timer Subscribe Expires (seconds)**
  - 120
- **Timer Subscribe Delta (seconds)**
  - 5
- **Maximum Registrations**: 70
- **Off Hook to First Digit Timer (milliseconds)**
  - 15000
- **Call Forward URI**: x-cisco-service-sp
Figure 6 CUCM SIP Profile Contd.

Figure 7 CUCM SIP Profile Contd.
5.3.3 CUCM Device Pool Configuration

5.3.3.1 Codec Preference list

1. Navigate to **System Region Information** Audio Codec Preference List
2. Click **Add New**
3. Provide a Name and Description: **G711_Preferred Codec List** was used
4. Prioritize codecs as shown below

![Figure 9 CUCM Audio Codec Preference List](image)

5.3.3.2 Region

1. Navigate to **System Region**
2. Click **Add New**
3. Provide a Name: **G711 Region** was used in this test (see list of [Twilio Elastic SIP Trunking codecs here](#))
4. Associate the codec preference list **G711_Preferred Codec List** to this Region
5.3.3.3 Device Pool

1. Navigate to **System** Device Pool
2. Click **Add New**
3. Provide a Device Pool Name: **G711_pool** was used
4. Associate the Region: **G711_Region** to this Device Pool
5. Associate the Media resource Group List: **MRGPL**
6. Leave all other parameters at their default settings
7. Click **Save**
Figure 12 CUCM Device Pool Contd.

Figure 13 CUCM Device Pool Contd.
5.3.4 Media Resources

5.3.4.1 Media Resources Group

2. Add New.
3. Provide a Name: MRGP was used.
4. Select Media Resources from the Available Media Resources. (these are assumed to be added earlier and are available for use /registered with this CUCM)

![Media Resource Group Configuration](image)

Figure 14 CUCM Media Resources Group

5.3.4.2 Media Resources Group List

1. Navigate to Media Resources Media Resource Group List
2. Add New.
3. Provide a Name: MRGPL was used.
4. Select the media resource group MRGP from the list of Available Media Resource Groups.
5.3.5 **Twilio SIP Trunk Security Profile**

1. Navigate to: System  Security  Non Secure SIP Trunk Profile
2. Provide a Name: **Non Secure SIP Trunk Profile-Twilio** was used for this test
3. Provide a Description: **Non Secure** was used for this test
4. Select Incoming Transport Type: **TCP+UDP** was used in this test
5. Select Outgoing Transport Type: **UDP** was used in this test
6. Select Incoming Port: 5060
7. Click Save and Apply Config
5.3.6 Twilio SIP Trunk to vCUBE

1. Navigate to Device Trunk
2. Provide a Device Name: Trunk-CUBE-Twilio was used in this test
3. Provide a Description: SIP Trunk to CUBE for Twilio was used
4. Set Device Pool: G711_pool
5. Set Media Resource Group List: MRGPL
6. Set Significant Digits: 4
7. Set Destination Address: Set IP address of vCUBE
8. Set SIP Trunk Security Profile: Non Secure SIP Trunk Profile
9. Set SIP Profile: Standard SIP Profile Twilio
10. Set DTMF Signaling Method: No Preference
Figure 20 CUCM SIP Trunk Configuration Contd.

Figure 21 CUCM SIP Trunk Configuration Contd.
5.3.7 Route Pattern

1. Navigate to Call Routing > Route/Hunt > Route Pattern
2. Select Add New to create a new Route Pattern
3. Set Route Pattern: 9.@ (This is to enable outbound dialing from CUCM to vCUBE using the access code as “9”)
4. Set Gateway/Route List: Trunk-CUBE-Twilio was used in this test
5. Set Discard Digits: PreDot (This option is to remove the prefix “9” from called party number while sending the call out to vCUBE)
Figure 24 CUCM Route Pattern Configuration

Figure 25 CUCM Route Pattern Configuration Contd.
5.4 vCUBE Configuration

vCUBE is configured through CLI as CLI mode offers more flexibility and convenience compared to GUI mode. NOTICE: the IP Address values used in this section are for reference only and are specific to the tekVizion test environment. These MUST be considered ONLY as reference. Each IP Address is described as a footnote.

5.4.1 Global vCUBE settings

The global configure settings enables CUBE application on the router, enables voice services with VoIP, and configures trusted IP address lists, enable SIP configuration mode and many more:

```
enable
configure terminal
voice service voip
ip address trusted list
  ipv4 54.171.127.192
  255.255.255.192
  ipv4 54.244.51.0
  255.255.255.0
  ipv4 54.172.60.0
  255.255.255.0
  ipv4 172.16.29.0
  255.255.255.0
rtcp keepalive
address-hiding
mode border-element
media disable-detailed-stats
allow-connections sip to sip
fax protocol pass-through g711ulaw
trace
sip
  session refresh
  srtp-auth sha1-80
  early-offer forced
  midcall-signaling passthru
privacy-policy passthru
```

---

1 Enables privileged EXEC mode
2 Enters global configuration mode
3 Enters voice service configuration mode specifying VoIP as the voice encapsulation type
4 Enables trust with Signaling IPs for Europe Ireland Gateways
5 Enables trust with Signaling IPs for North America Oregon Gateways
6 Enables trust with Signaling IPs for North America Virginia Gateways
7 Enables trust with Cisco Phone (PBX extension) IPs
8 Enables CUBE application
9 Allows connections between SIP endpoints in a VoIP network
10 Enables VoIP trace feature which can be used to help troubleshoot issues
11 Enables global SIP configuration mode
12 Converts a delayed-offer to early offer
5.4.2 vCUBE - TLS SIP trunk to Twilio

The following configuration changes are specific to trunk configuration for Twilio.

5.4.2.1 Codecs

Two set of codecs were configured as part of this validation testing and each one is associated with an outbound dial peer for Twilio. The first one is for the US trunk (Twilio Ashburn and Umatilla datacenters) and the second one is for the Europe trunk (Dublin datacenter):

voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g711alaw
  codec preference 3 g729r8

voice class codec 2
  codec preference 1 g711alaw
  codec preference 2 g729r8

5.4.2.2 SIP Profile

The SIP profiles are configured to modify SIP headers. The SIP profile is associated with outbound dial peers for Twilio. The following SIP profiles were used for the test:

voice class sip-profiles 200
  request REINVITE sip-header From modify "(<.*:.*)(@.*>)"
    "1@tekvizion.pstn.twilio.com>"13
  request CANCEL sip-header From modify "(<.*:.*)(@.*>)"
    "1@tekvizion.pstn.twilio.com>"14
  request INVITE sip-header To modify "(<.*:.*)(@.*>)"
    "1@tekvizion.pstn.twilio.com;user=phone>"15
  request REINVITE sip-header To modify "(<.*:.*)(@.*>)"
    "1@tekvizion.pstn.twilio.com>"16
  request INVITE sip-header From modify "(<.*:.*)(@.*>)"
    "1@tekvizion.pstn.twilio.com;user=phone>"17
  request INVITE sip-header P-Asserted-Identity modify "(<.*:.*)(@.*>)"
    "1@tekvizion.pstn.twilio.com>"18

13 To update re-INVITE From header to contain FQDN instead of IP before sending out to Twilio
14 To update CANCEL From header to contain FQDN instead of IP before sending out to Twilio
15 To update INVITE To header to contain FQDN instead of IP before sending out to Twilio
16 To update re-INVITE To header to contain FQDN instead of IP before sending out to Twilio
17 To update INVITE From header to contain FQDN instead of IP before sending out to Twilio
18 To update INVITE PAI header to contain FQDN instead of IP before sending out to Twilio and also to include user=phone after FQDN (user=phone is not needed for all the scenarios)
request ANY sip-header Diversion modify "sip: \(+1\.*\)@(\.*)>
"sip:\1@tekvizion.pstn.twilio.com;user=phone>"  
request ANY sip-header Diversion modify "sip:(00..*)@(.*)>
"sip:+1814926\1@tekvizion.pstn.twilio.com;user=phone>"  

5.4.2.3 SIP-UA

SIP user-agent configuration:

sip-ua
  no remote-party-id
  transport tcp tls v1.2
  connection-reuse
  crypto signaling default trustpoint TP-self-signed-2894276916  

5.4.2.4 Crypto Trustpoint

This is a default self-signed certificate generated by vCUBE. Testing revealed there is no need to configure any new certificate enrollments. The basic certificate bundle ios_core.p7b that exists in nvram is sufficient to trust the certificates sent by the Twilio datacenters (Ashburn, Umatilla and Dublin) considered for this testing:

crypto pki trustpoint TP-self-signed-2894276916
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-2894276916
  revocation-check none
  rsakeypair TP-self-signed-2894276916

5.4.2.5 Cisco CA bundle

This is to update the Cisco CA bundle with the latest certificates. It is important to ensure that the corresponding Twilio Data center certificates are available as part of this bundle. The following command is used to view the certificates in the bundle:

"show crypto pki trustpool"

#crypto pki trustpool import url https://www.cisco.com/security/pki/trs/ios_core.p7b

---

19 To update ANY Diversion header which has number starting with "+1" to contain FQDN instead of IP before sending out to Twilio and also to include user=phone after FQDN (user=phone is not needed for all the scenarios). This is for the Diversion header that is sent by Twilio itself.

20 To update ANY Diversion header which has number starting with "00" to contain FQDN instead of IP before sending out to Twilio and also to include user=phone after FQDN (user=phone is not needed for all the scenarios). This is for the Diversion header included by CUCM for the call forward scenarios.

21 Use listener port for sending requests over UDP

22 Configures the SIP gateway to use its trustpoint when it establishes or accepts TLS connection. The trustpoint label refers to the vCUBE’s certificate.
5.4.2.6 Translation Profile

The translation profile is to apply translation rule for the calling and called number types. This is associated with the outbound dial peers to Twilio and CUCM:

voice translation-profile 1
  translate calling 1
  translate called 1
voice translation-profile 2
  translate calling 2
  translate called 2
voice translation-profile 3
  translate calling 3
  translate called 3
voice translation-profile 5
  translate calling 5
  translate called 5

5.4.2.7 Translation Rule

The translation rules are used to manipulate the numbers before sending them to Twilio or CUCM. These are invoked by Translation profiles:

voice translation-rule 1
  rule 1 /\^1\([\-]*\)/ /\1/
voice translation-rule 2
  rule 1 /\^\([0-9]{1,3}\)[\-\ ]{1,3}\)/ /\1\1/
voice translation-rule 3
  rule 1 /\^44\([\-\ ]{1,3}\)/ /\1/
voice translation-rule 5
  rule 1 /\^\([2-9]{1,3}\)[\-\ ]{1,3}\)/ /\1\1/
address. Inbound dial peers are for the incoming legs to vCUBE and outbound dial peers are for the outgoing legs from vCUBE.

**Inbound Dial Peer for CUCM**
This dial peer is for the incoming call leg from CUCM:

dial-peer voice 1 voip
description Incoming from CUCM
session protocol sipv2
session transport udp
incoming called-number [0-9]T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet1
voice-class sip bind media source-interface GigabitEthernet1
dtmf-relay rtp-nte

**Inbound Dial Peer for Twilio**
This dial peer is for the incoming call leg from Twilio:

dial-peer voice 3 voip
description Incoming from Twilio US
max-conn 1
session transport tcp tls
incoming called-number +1..........
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet2
voice-class sip bind media source-interface GigabitEthernet2
dtmf-relay rtp-nte
srtp

dial-peer voice 5 voip
description Incoming from Twilio UK
session transport tcp tls
incoming called-number +44..........
voice-class codec 2
voice-class sip bind control source-interface GigabitEthernet2
voice-class sip bind media source-interface GigabitEthernet2
dtmf-relay rtp-nte
srtp
no vad

**Outbound Dial Peer to Twilio**
This dial peer is for the outgoing call leg from vCUBE towards Twilio:

dial-peer voice 2 voip
description Outgoing to Twilio Ashburn Datacenter
translation-profile outgoing 2
preference 1
shutdown
destination-pattern [0-9]T
rtp payload-type comfort-noise 13
session protocol sipv2
session target dns:tekvizion.pstn.twilio.com
session transport tcp tls
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 200
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet2
voice-class sip bind media source-interface GigabitEthernet2
dtmf-relay rtp-nte sip-kpml sip-notify
srtp
dial-peer voice 20 voip
description Outgoing to Twilio Dublin Datacenter
translation-profile outgoing 5
preference 2
shutdown
destination-pattern [0-9]T
rtp payload-type comfort-noise 13
session protocol sipv2
session target dns:tekvizion.pstn.dublin.twilio.com
session transport tcp tls
voice-class codec 2
voice-class sip asserted-id pai
voice-class sip profiles 200
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet2
voice-class sip bind media source-interface GigabitEthernet2
dtmf-relay rtp-nte sip-kpml sip-notify
srtp

Outbound Dial Peer to CUCM
This dial peer is for the outgoing call leg from vCUBE towards CUCM:

dial-peer voice 4 voip
description Outgoing US number to CUCM
translation-profile outgoing 1
destination-pattern +1.......... 
session protocol sipv2
session target ipv4:10.71.8.10:5060
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet1
voice-class sip bind media source-interface GigabitEthernet1
dtmf-relay rt-n-te
no vad

dial-peer voice 6 voip
description Outgoing UK number to CUCM
translation-profile outgoing 3
destination-pattern +44...........
session protocol sipv2
session target ipv4:10.71.8.10:5060
session transport udp
voice-class codec 2
voice-class sip bind control source-interface GigabitEthernet1
voice-class sip bind media source-interface GigabitEthernet1
dtmf-relay rt-n-te
no vad
6  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).

6.1  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 CUBE SBC(s).
6.2 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.

Under the General Settings you can enable different features as desired.
Features

To learn more about SIP Trunking features, please see our user documentation.

Call Recording

- **Enabled**: Calls will be recorded.

### Call Recording

- **Record from ringing**

### Recording Trim

- **Disabled**: Silence will not be trimmed from recording

Secure Trunking

- **Enabled**: 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)

- **Enabled**: Twilio will consume an incoming SIP REFER from your communications infrastructure and create an INVITE message to the address in the Refer-To header

- **Enable PSTN Transfer**: Allow Call Transfers to the PSTN via your Trunk.

Symmetric RTP

- **Enabled**: 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 SDP

Additional Features

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**

  ![Show Localized URIs](https://www.tekvizion.com)

- **.psn.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>

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](#)

**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 CUBE SBC. 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:

**Add Origination URL**

**Origination SIP URI**

sip:199.182.124.230:edge=ashburn

**Priority**

10

Numeric range from 0 to 65535.

**Weight**

1.0

Numeric range from 1 to 65535.

**Enabled**

Enabled

Continue to add the other Origination URIs, so you have the following configuration:
In this example, Origination traffic is first routed via Twilio’s Ashburn edge, if that fails then we’ll route from Twilio’s Umatilla edge.

6.3 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.
7 Appendix

7.1 vCUBE Running Configuration

Current configuration : 9865 bytes

! Last configuration change at 16:15:47 UTC Wed Jun 9 2021 by cisco

version 17.3
service config
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
service call-home
platform qfp utilization monitor load 80
platform punt-keepalive disable-kernel-core
platform console virtual

hostname twilio

boot-start-marker
boot-end-marker

enable secret 9
$9$A4u6SYu7H3ZidE$IFUFZjSRnpLmC7kdncZfZeYoxjm8Wzk952nE7Vv0izpKU
enable password 7 060506324F41

no aaa new-model

ip name-server 8.8.8.8

login on-success log

subscriber templating

multilink bundle-name authenticated

voice service voip
ip address trusted list
ipv4 177.71.206.192 255.255.255.192
ipv4 54.171.127.192 255.255.255.192
ipv4 54.65.63.192 255.255.255.192
ipv4 54.169.127.128 255.255.255.192
ipv4 54.252.254.64 255.255.255.192
ipv4 54.172.60.0 255.255.254.0
ipv4 172.16.29.0 255.255.255.0
rtcp keepalive
address-hiding
mode border-element
media disable-detailed-stats
allow-connections sip to sip
fax protocol pass-through g711ulaw
trace
sip
session refresh
srtp-auth sha1-80
early-offer forced
midcall-signaling passthru
privacy-policy passthru

voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g711alaw
codec preference 3 g729r8

voice class codec 2
codec preference 1 g711alaw
codec preference 2 g729r8

voice class sip-profiles 200
request REINVITE sip-header From modify "(<.*::*)(@.*>)"
"1@tekvizion.pstn.twilio.com;user=phone>"
request CANCEL sip-header From modify "(<.*::*)(@.*>)"
"1@tekvizion.pstn.twilio.com;user=phone>"
request INVITE sip-header To modify "(<.*::*)(@.*>)" "1@tekvizion.pstn.twilio.com;user=phone>"
request REINVITE sip-header To modify "(<.*::*)(@.*>)" "1@tekvizion.pstn.twilio.com;user=phone>"
request INVITE sip-header From modify "(<.*::*)(@.*>)"
"1@tekvizion.pstn.twilio.com>"
request INVITE sip-header P-Asserted-Identity modify "(<.*::*)(@.*>)"
"1@tekvizion.pstn.twilio.com>"
request ANY sip-header Diversion modify "sip:(\+1...........)@(.*)>"
"sip:1@tekvizion.pstn.twilio.com>"
request ANY sip-header Diversion modify "sip:(00..*)@(.*)>"
"sip:+18149261@tekvizion.pstn.twilio.com>"

voice translation-rule 1
rule 1 /^\+1\(\.*\)/ \1/

voice translation-rule 2
rule 1 /^\[(1-9]........\)/ /+1/
voice translation-rule 3
  rule 1 /\^\+44\(.*\)/ /\1/

voice translation-rule 5
  rule 1 /\^\([2-9]\........\)/ /\+44\1/

voice translation-profile 1
  translate calling 1
  translate called 1

voice translation-profile 2
  translate calling 2
  translate called 2

voice translation-profile 3
  translate calling 3
  translate called 3

voice translation-profile 5
  translate calling 5
  translate called 5

voice translation-profile BLOCK
  translate calling 4

crypto pki trustpoint SLA-TrustPoint
  enrollment pkcs12
  revocation-check crl

crypto pki trustpoint TP-self-signed-2894276916
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-2894276916
  revocation-check none
  rsa keypair TP-self-signed-2894276916

crypto pki certificate pool
  cabundle nvram:ios_core.p7b

license udi pid CSR1000V sn 990PJD089R7
  diagnostic bootup level minimal
  memory free low-watermark processor 71497

  spanning-tree extend system-id

username cisco password 7 030752180500

redundancy

interface GigabitEthernet1
ip dhcp client client-id ascii 990PJD089R7
ip address 10.64.5.189 255.255.0.0
negotiation auto
no mop enabled
no mop sysid

interface GigabitEthernet2
ip address 199.X.X.X 255.255.255.192
negotiation auto
no mop enabled
no mop sysid

interface GigabitEthernet3
no ip address
shutdown
negotiation auto
no mop enabled
no mop sysid

ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet1

ip route 0.0.0.0 0.0.0.0 199.182.124.193
ip route 10.71.0.0 255.255.0.0 10.64.1.1
ip route 54.0.0.0 255.0.0.0 199.182.124.193
ip route 172.16.0.0 255.255.0.0 10.64.1.1
ip route 172.17.0.0 255.255.0.0 10.64.1.1
ip ssh version 2

current-plane

dial-peer voice 1 voip
description Incoming from CUCM
session protocol sipv2
session transport udp

---

27 Since the actual public IP used for the test cannot be exposed during documentation, it is hidden.
incoming called-number [0-9]T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet1
voice-class sip bind media source-interface GigabitEthernet1
dtmf-relay rtp-nte
!
dial-peer voice 2 voip
description Outgoing to Twilio Ashburn Datacenter
translation-profile outgoing 2
preference 1
destination-pattern [0-9]T
rtp payload-type comfort-noise 13
session protocol sipv2
session target dns:tekvizion.pstn.twilio.com
session transport tcp tls
voice-class codec 1
voice-class sip asserted-id
voice-class sip profiles 200
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet2
voice-class sip bind media source-interface GigabitEthernet2
dtmf-relay rtp-nte cisco-rtp
srtp
!
dial-peer voice 3 voip
description Incoming from Twilio US
max-conn 1
session transport tcp tls
incoming called-number +1.......... 
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet2
voice-class sip bind media source-interface GigabitEthernet2
dtmf-relay rtp-nte
srtp
!
dial-peer voice 4 voip
description Outgoing US number to CUCM
translation-profile outgoing 1
destination-pattern +1.......... 
session protocol sipv2
session target ipv4:10.71.8.10:5060
session transport udp
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet1
voice-class sip bind media source-interface GigabitEthernet1
dtmf-relay rtp-nte
!
dial-peer voice 5 voip
description Incoming from Twilio UK
session transport tcp tls
incoming called-number +44........
voice-class codec 2
voice-class sip bind control source-interface GigabitEthernet2
voice-class sip bind media source-interface GigabitEthernet2
dtmf-relay rtp-nte
srtp
no vad
!
dial-peer voice 6 voip
description Outgoing UK number to CUCM
translation-profile outgoing 3
destination-pattern +44........
session protocol sipv2
session target ipv4:10.71.8.10:5060
session transport udp
voice-class codec 2
voice-class sip bind control source-interface GigabitEthernet1
voice-class sip bind media source-interface GigabitEthernet1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 20 voip
description Outgoing to Twilio Dublin Datacenter
translation-profile outgoing 5
preference 2
shutdown
destination-pattern [0-9]T
rtp payload-type comfort-noise 13
session protocol sipv2
session target dns:tekvizion.pstn.dublin.twilio.com
session transport tcp tls
voice-class codec 2
voice-class sip asserted-id pai
voice-class sip profiles 200
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet2
voice-class sip bind media source-interface GigabitEthernet2
dtmf-relay rtp-nte sip-kpml sip-notify
srtp
!
sip-ua
no remote-party-id
transport tcp tls v1.2
connection-reuse
crypto signaling default trustpoint TP-self-signed-2894276916
!
line con 0
password 7 131112193D5D1E7B7B2A
login
stopbits 1
line vty 0 4
exec-timeout 120 0
login local
transport input ssh
!
call-home
! If contact email address in call-home is configured as sch-smart-licensing@cisco.com
! the email address configured in Cisco Smart License Portal will be used as contact
email address to send SCH notifications.
contact-email-addr sch-smart-licensing@cisco.com
profile "CiscoTAC-1"
active
destination transport-method http
!
end