

# Twilio Elastic SIP Trunking Configuration Guide

## Cisco CUCM 12.5(SU1) with Cisco vCUBE 14.1

June 2021

## Document History

Rev. No.	Description
1.0	Twilio Elastic SIP Trunking Configuration Guide
1.1	Updated based on the feedback from Twilio

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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 help 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](http://www.tekVizion.com)*

## 3 Validation Network Components

The network for the Twilio Elastic SIP Trunk, vCUBE, and CUCM reference configurations is illustrated below:

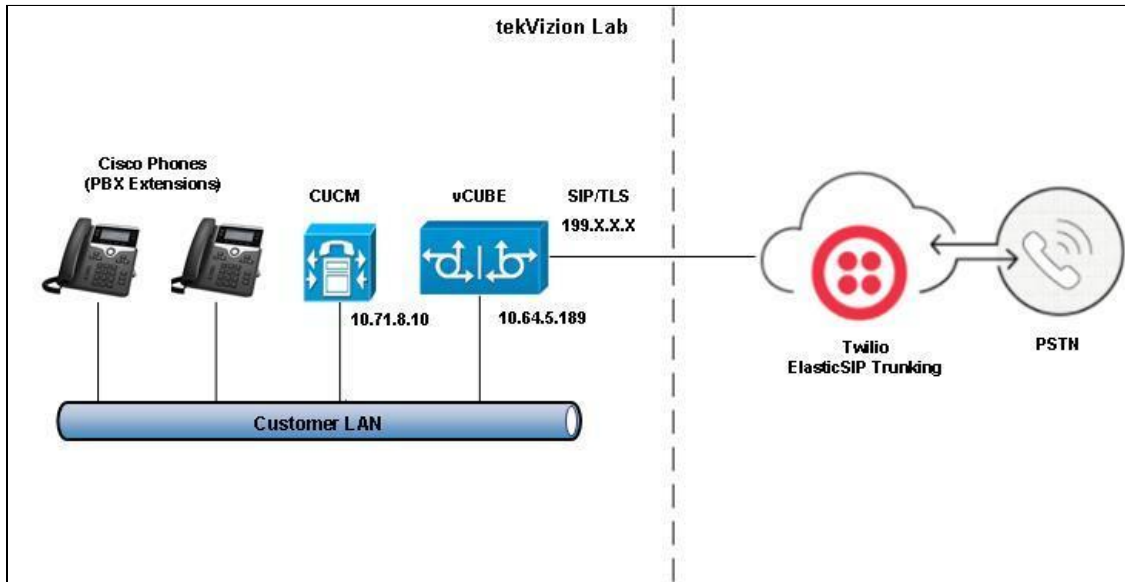


Figure 1 Network Topology

### 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**.

*Table 1: PBX Configuration Steps*

Steps	Description	Reference
Step 1	CUCM Configuration	<a href="#">Section 5.3</a>
Step 2	vCUBE Configuration	<a href="#">Section 5.4</a>

### 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.

*Table 2 – IP Addresses*

Component	Lab Value	Customer Value
<b>vCUBE</b>		
LAN IP Address	10.64.5.189	
LAN Subnet Mask	255.255.0.0	
<b>CUCM</b>		
IP Address	10.71.8.10	
Subnet Mask	255.255.0.0	

### 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**

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | About | Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

**SIP Profile Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Status**

- Status: Ready
- All SIP devices using this profile must be restarted before any changes will take affect.

**SIP Profile Information**

Name*	Standard SIP Profile Twilio
Description	Standard SIP Profile Twilio
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Ager
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, ani
Confidential Access Level Headers*	Disabled

Redirect by Application  
 Disable Early Media on 180  
 Outgoing T.38 INVITE include audio mline  
 Offer valid IP and Send/Receive mode only for T.38 Fax Relay  
 Use Fully Qualified Domain Name in SIP Requests  
 Assured Services SIP conformance  
 Enable External QoS\*\*

Figure 3 CUCM SIP Profile

**Cisco Unified CM Administration**  
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System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**SIP Profile Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

**SDP Information**

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites\*: TIAS and AS

SDP Transparency Profile: < None >

Accept Audio Codec Preferences in Received Offer\*: Default

Require SDP Inactive Exchange for Mid-Call Media Change

Allow RR/RS bandwidth modifier (RFC 3556)

**Parameters used in Phone**

Timer Invite Expires (seconds)\*: 180

Timer Register Delta (seconds)\*: 5

Timer Register Expires (seconds)\*: 3600

Timer T1 (msec)\*: 500

Timer T2 (msec)\*: 4000

Retry INVITE\*: 6

Retry Non-INVITE\*: 10

Media Port Ranges:  Common Port Range for Audio and Video  
 Separate Port Ranges for Audio and Video

Start Media Port\*: 16384

Stop Media Port\*: 32766

DSCP for Audio Calls: Use System Default

DSCP for Video Calls: Use System Default

Figure 4 CUCM SIP Profile Contd.

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System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**SIP Profile Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

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 URI\*: x-cisco-serviceuri-pickup

Call Pickup Group Other URI\*: x-cisco-serviceuri-opickup

Call Pickup Group URI\*: x-cisco-serviceuri-gpickup

Meet Me Service URI\*: x-cisco-serviceuri-meetme

User Info\*: None

DTMF DB Level\*: Nominal

Call Hold Ring Back\*: Off

Anonymous Call Block\*: Off

Caller ID Blocking\*: Off

Do Not Disturb Control\*: User

Telnet 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 Redirections\*: 70

Off Hook To First Digit Timer (milliseconds)\*: 15000

Call Forward URI\*: x-cisco-serviceuri-cfwdall

Figure 5 CUCM SIP Profile Contd.

**Cisco Unified CM Administration**  
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System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**SIP Profile Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

Speed Dial (Abbreviated Dial) URI\*

Conference Join Enabled  
 RFC 2543 Hold  
 Semi Attended Transfer  
 Enable VAD  
 Stutter Message Waiting  
 MLPP User Authorization

**Normalization Script**

Normalization Script   
 Enable Trace

	Parameter Name	Parameter Value
1	<input type="text"/>	<input type="text"/>

**External Presentation Information**

Anonymous External Presentation  
 External Presentation Number   
 External Presentation Name

**Trunk Specific Configuration**

Reroute Incoming Request to new Trunk based on\*   
 Resource Priority Namespace List

Figure 6 CUCM SIP Profile Contd.

**Cisco Unified CM Administration**  
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**SIP Profile Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

SIP Rel1XX Options\*   
 Video Call Traffic Class\*   
 Calling Line Identification Presentation\*   
 Session Refresh Method\*   
 Early Offer support for voice and video calls\*

Enable ANAT  
 Deliver Conference Bridge Identifier  
 Enable External Presentation Name and Number  
 Reject Anonymous Incoming Calls  
 Reject Anonymous Outgoing Calls  
 Send ILS Learned Destination Route String  
 Connect Inbound Call before Playing Queuing Announcement

**SIP OPTIONS Ping**

Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)\*   
 Ping Interval for Out-of-service Trunks (seconds)\*   
 Ping Retry Timer (milliseconds)\*   
 Ping Retry Count\*

Figure 7 CUCM SIP Profile Contd.

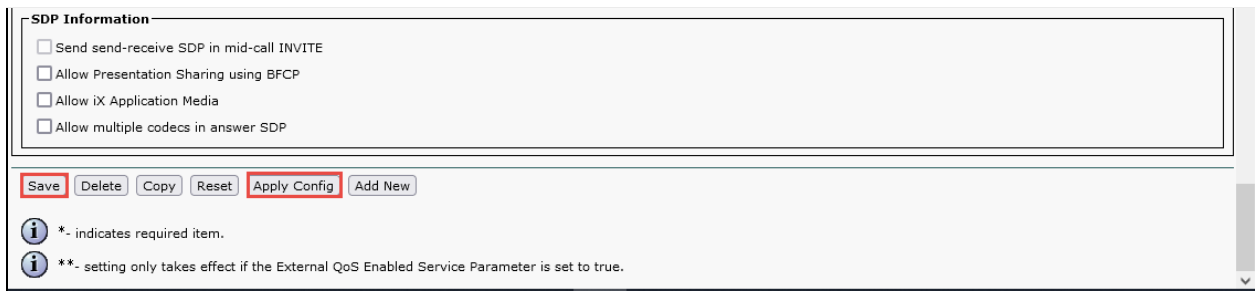


Figure 8 CUCM SIP Profile Contd.,

### 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

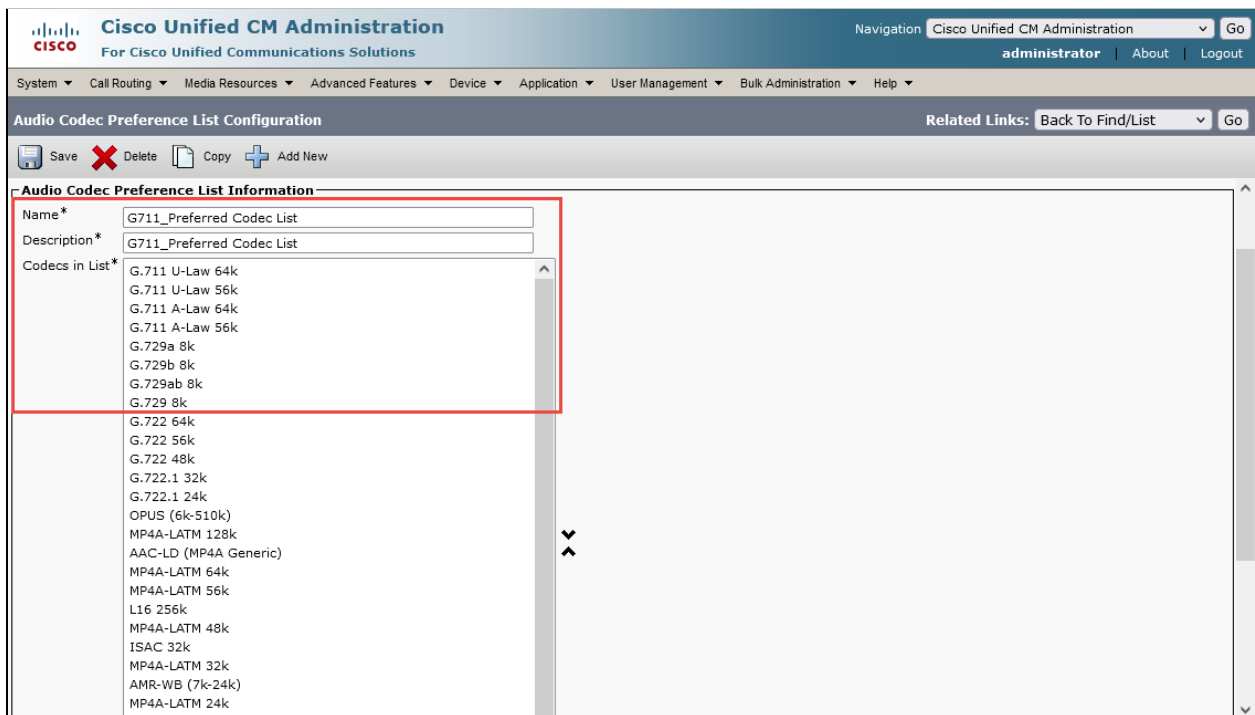


Figure 9 CUCM Audio Codec Preference List

#### 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

**Region Configuration**

Save Delete Reset Apply Config Add New

**Region Information**

Name\* G711\_Region

**Region Relationships**

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	G711_Preferred Codec List	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)
G711_Region	G711_Preferred Codec List	64 kbps (G.722, G.711)	Use System Default (384 kbps)	Use System Default (2000000000 kbps)

Figure 10 CUCM 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**

**Device Pool Configuration**

Save Delete Copy Reset Apply Config Add New

**Device Pool Settings**

Device Pool Name\* G711\_pool

Cisco Unified Communications Manager Group\* Default

Calling Search Space for Auto-registration < None >

Adjunct CSS < None >

Reverted Call Focus Priority Default

Intercompany Media Services Enrolled Group < None >

MRA Service Domain < None >

**Roaming Sensitive Settings**

Date/Time Group\* CMLocal

Region\* G711\_Region

Media Resource Group List MRGPL

Location < None >

Network Locale < None >

SRST Reference\* Disable

Connection Monitor Duration\*\*\*

Single Button Barge\* Default

Join Across Lines\* Default

Physical Location < None >

Device Mobility Group < None >

Figure 11 CUCM Device Pool



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System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Device Pool Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

**Local Route Group Settings**  
Standard Local Route Group: < None >

**Device Mobility Related Information\*\*\*\***  
 Device Mobility Calling Search Space: < None >  
 AAR Calling Search Space: < None >  
 AAR Group: < None >  
 Calling Party Transformation CSS: < None >  
 Called Party Transformation CSS: < None >

**Geolocation Configuration**  
 Geolocation: < None >  
 Geolocation Filter: < None >

**Call Routing Information**

**Incoming Calling Party Settings**  
 If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default		< None >
International Number	Default		< None >

Figure 12 CUCM Device Pool Contd.

**Cisco Unified CM Administration**  
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**Device Pool Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

Unknown Number: Default | < None >  
 Subscriber Number: Default | < None >

**Incoming Called Party Settings**  
 If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >
Subscriber Number	Default	0	< None >

**Phone Settings**  
**Caller ID For Calls From This Phone**  
 Calling Party Transformation CSS: < None >

**Connected Party Settings**  
 Connected Party Transformation CSS: < None >

**Redirecting Party Settings**  
 Redirecting Party Transformation CSS: < None >

Save | Delete | Copy | Reset | Apply Config | Add New

Figure 13 CUCM Device Pool Contd.

### 5.3.4 Media Resources

#### 5.3.4.1 Media Resources Group

1. Navigate to Media Resources -> Media Resource 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)

The screenshot displays the 'Media Resource Group Configuration' page in Cisco Unified CM Administration. The page title is 'Media Resource Group Configuration' and it shows the configuration for a group named 'MRGP'. The 'Name' field is set to 'MRGP' and the 'Description' is 'All resources included'. Under the 'Devices for this Group' section, the 'Selected Media Resources' list contains: ANN\_2 (ANN), ANN\_3 (ANN), ANN\_4 (ANN), CFB\_2 (CFB), and CFB\_3 (CFB). The 'Available Media Resources' list is currently empty. The 'Save' button at the bottom left is highlighted with a red box.

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

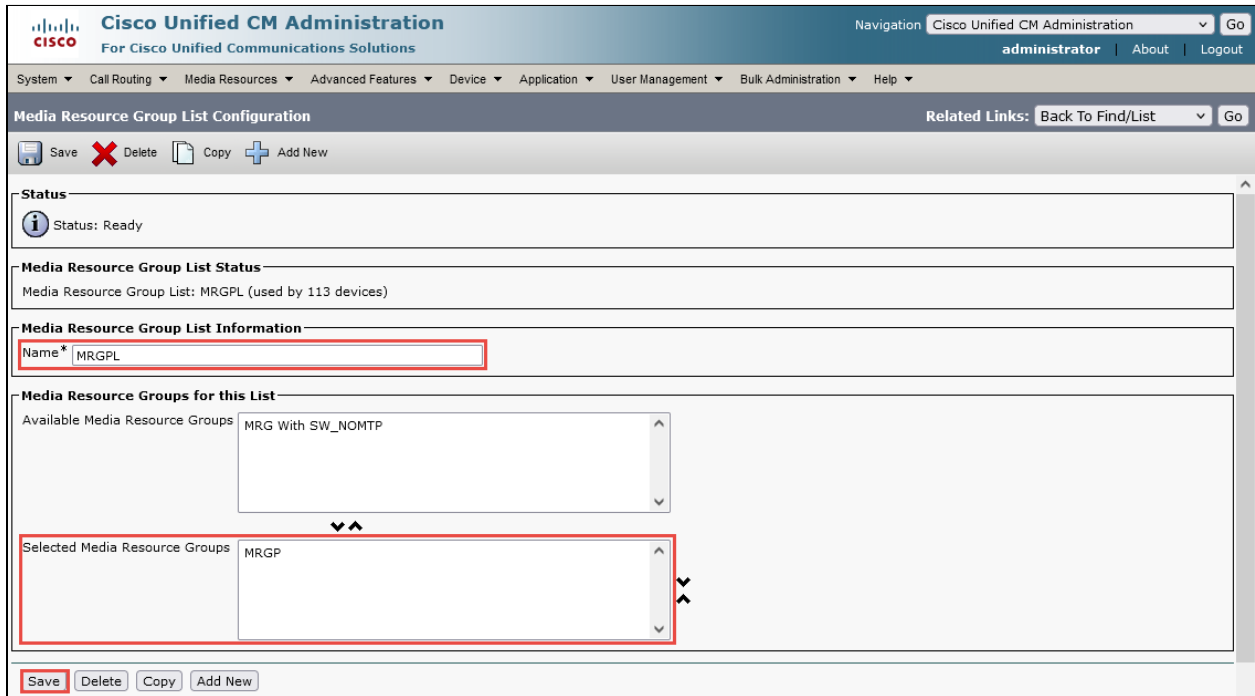


Figure 15 CUCM Media Resources Group List

### 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

**Cisco Unified CM Administration**  
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**SIP Trunk Security Profile Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

**Status**  
Status: Ready

**SIP Trunk Security Profile Information**

Name\* | Non Secure SIP Trunk Profile Twilio  
 Description | Non Secure SIP Trunk Profile Twilio  
 Device Security Mode | Non Secure  
 Incoming Transport Type\* | TCP+UDP  
 Outgoing Transport Type | UDP

Enable Digest Authentication  
 Nonce Validity Time (mins)\* | 600  
 Secure Certificate Subject or Subject Alternate Name

Incoming Port\* | 5060

Enable Application level authorization  
 Accept presence subscription

Figure 16 CUCM SIP Trunk Security Profile

Accept out-of-dialog refer\*\*  
 Accept unsolicited notification  
 Accept replaces header  
 Transmit security status  
 Allow charging header  
 SIP V.150 Outbound SDP Offer Filtering\* | Use Default Filter

Save | Delete | Copy | Reset | Apply Config | Add New

Figure 17 CUCM SIP Trunk Security Profile Contd.

### 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

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

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**Trunk Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

**Device Information**

Product: SIP Trunk  
 Device Protocol: SIP  
 Trunk Service Type: None(Default)  
 Device Name\*: Trunk-CUBE-Twilio  
 Description: SIP Trunk to CUBE for Twilio  
 Device Pool\*: G711\_pool  
 Common Device Configuration: < None >  
 Call Classification\*: Use System Default  
 Media Resource Group List: MRGPL  
 Location\*: Hub\_None  
 AAR Group: < None >  
 Tunneled Protocol\*: None  
 QSIG Variant\*: No Changes  
 ASN.1 ROSE OID Encoding\*: No Changes  
 Packet Capture Mode\*: None  
 Packet Capture Duration: 0

Media Termination Point Required  
 Retry Video Call as Audio  
 Path Replacement Support  
 Transmit UTF-8 for Calling Party Name

Figure 18 CUCM SIP Trunk Configuration

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | About | Logout

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**Trunk Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

Transmit UTF-8 Names in QSIG APDU  
 Unattended Port  
 SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.  
 Consider Traffic on This Trunk Secure\*: When using both sRTP and TLS  
 Route Class Signaling Enabled\*: Default  
 Use Trusted Relay Point\*: Default  
 PSTN Access  
 Run On All Active Unified CM Nodes

**Intercompany Media Engine (IME)**

E.164 Transformation Profile: < None >

**MLPP and Confidential Access Level Information**

MLPP Domain: < None >  
 Confidential Access Mode: < None >  
 Confidential Access Level: < None >

**Call Routing Information**

Remote-Party-Id  
 Asserted-Identity  
 Asserted-Type\*: Default

Figure 19 CUCM SIP Trunk Configuration Contd.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Trunk Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

SIP Privacy\* | Default

Trust Received Identity\* | Trust All (Default)

**Inbound Calls**

Significant Digits\* | 4

Connected Line ID Presentation\* | Default

Connected Name Presentation\* | Default

Calling Search Space | < None >

AAR Calling Search Space | < None >

Prefix DN |

Redirecting Diversion Header Delivery - Inbound

**Incoming Calling Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**Incoming Called Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

Figure 20 CUCM SIP Trunk Configuration Contd.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Trunk Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**Connected Party Settings**

Connected Party Transformation CSS | < None >

Use Device Pool Connected Party Transformation CSS

**Outbound Calls**

Called Party Transformation CSS | < None >

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS | < None >

Use Device Pool Calling Party Transformation CSS

Calling Party Selection\* | Originator

Calling Line ID Presentation\* | Default

Calling Name Presentation\* | Default

Calling and Connected Party Info Format\* | Deliver DN only in connected party

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS | < None >

Use Device Pool Redirecting Party Transformation CSS

Figure 21 CUCM SIP Trunk Configuration Contd.

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Trunk Configuration** Related Links: Back To Find/List Go

Save Delete Reset Add New

**Presentation Information**

Anonymous Presentation  
Presentation Number:   
Presentation Name:   
 Send Presentation Name and Number only in the FROM header and not in the other identity headers

**SIP Information**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Sta
1*	10.64.5.189		5060	N/A	

MTP Preferred Originating Codec\*: 711ulaw  
BLF Presence Group\*: Standard Presence group  
SIP Trunk Security Profile\*: Non Secure SIP Trunk Profile  
Rerouting Calling Search Space: < None >  
Out-Of-Dialog Refer Calling Search Space: < None >  
SUBSCRIBE Calling Search Space: < None >  
SIP Profile\*: Standard SIP Profile Twilio [View Details](#)  
DTMF Signaling Method\*: No Preference

Figure 22 CUCM SIP Trunk Configuration Contd.

**Normalization Script**

Normalization Script: < None >  
 Enable Trace

	Parameter Name	Parameter Value
1	<input type="text"/>	<input type="text"/>

**Recording Information**

None  
 This trunk connects to a recording-enabled gateway  
 This trunk connects to other clusters with recording-enabled gateways

**Geolocation Configuration**

Geolocation: < None >  
Geolocation Filter: < None >  
 Send Geolocation Information

Save Delete Reset Add New

Figure 23 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)

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Route Pattern Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Add New

**Pattern Definition**

Route Pattern\*   
 Route Partition   
 Description   
 Numbering Plan\*   
 Route Filter   
 MLPP Precedence\*   
 Apply Call Blocking Percentage  
 Resource Priority Namespace Network Domain   
 Route Class\*   
 Gateway/Route List\*  (Edit)  
 Route Option  Route this pattern  Block this pattern   
 Call Classification\*   
 External Call Control Profile   
 Allow Device Override  Provide Outside Dial Tone  Allow Overlap Sending  Urgent Priority  
 Require Forced Authorization Code  
 Authorization Level\*   
 Require Client Matter Code

**Calling Party Transformations**

Figure 24 CUCM Route Pattern Configuration

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

**Route Pattern Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Add New

Use Calling Party's External Phone Number Mask  
 Calling Party Transform Mask   
 Prefix Digits (Outgoing Calls)   
 Calling Line ID Presentation\*   
 Calling Name Presentation\*   
 Calling Party Number Type\*   
 Calling Party Numbering Plan\*

**Connected Party Transformations**

Connected Line ID Presentation\*   
 Connected Name Presentation\*

**Called Party Transformations**

Discard Digits   
 Called Party Transform Mask   
 Prefix Digits (Outgoing Calls)   
 Called Party Number Type\*   
 Called Party Numbering Plan\*

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol   
 Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value=" &lt; Not Exist &gt;"/>	<input type="text"/>

Save | Delete | Copy | Add New

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:

```
enable1
configure terminal2
voice service voip3
ip address trusted list
  ipv4 54.171.127.1924 255.255.255.192
  ipv4 54.244.51.05 255.255.255.0
  ipv4 54.172.60.06 255.255.254.0
  ipv4 172.16.29.07 255.255.255.0
rtcp keepalive
address-hiding
mode border-element8
media disable-detailed-stats
allow-connections sip to sip9
fax protocol pass-through g711ulaw
trace10
sip11
  session refresh
  srtp-auth sha1-80
  early-offer forced12
  midcall-signaling passthru
  privacy-policy passthru
```

---

<sup>1</sup> Enables privileged EXEC mode

<sup>2</sup> Enters global configuration mode

<sup>3</sup> Enters voice service configuration mode specifying VoIP as the voice encapsulation type

<sup>4</sup> Enables trust with Signaling IPs for Europe Ireland Gateways

<sup>5</sup> Enables trust with Signaling IPs for North America Oregon Gateways

<sup>6</sup> Enables trust with Signaling IPs for North America Virginia Gateways

<sup>7</sup> Enables trust with Cisco Phone (PBX extension) IPs

<sup>8</sup> Enables CUBE application

<sup>9</sup> Allows connections between SIP endpoints in a VoIP network

<sup>10</sup> Enables VoIP trace feature which can be used to help troubleshoot issues

<sup>11</sup> Enables global SIP configuration mode

<sup>12</sup> 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>"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;user=phone>"18
```

<sup>13</sup> To update re-INVITE From header to contain FQDN instead of IP before sending out to Twilio

<sup>14</sup> To update CANCEL From header to contain FQDN instead of IP before sending out to Twilio

<sup>15</sup> To update INVITE To header to contain FQDN instead of IP before sending out to Twilio

<sup>16</sup> To update re-INVITE To header to contain FQDN instead of IP before sending out to Twilio

<sup>17</sup> To update INVITE From header to contain FQDN instead of IP before sending out to Twilio

<sup>18</sup> 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>"19
request ANY sip-header Diversion modify "sip:(00..*)@(.*)>"
"sip:+1814926\1@tekvizion.pstn.twilio.com;user=phone>"20
```

### 5.4.2.3 SIP-UA

SIP user-agent configuration:

```
sip-ua
no remote-party-id
transport tcp tls v1.2
connection-reuse21
crypto signaling default trustpoint TP-self-signed-289427691622
```

### 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
```

<sup>19</sup> 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

<sup>20</sup> 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

<sup>21</sup> Use listener port for sending requests over UDP

<sup>22</sup> Configures the SIP gateway to use its trustpoint when it establishes or accepts TLS connection. The trustpoint label refers to the vCUBE’s certificate.

Reading file from [https://www.cisco.com/security/pki/trs/ios\\_core.p7b](https://www.cisco.com/security/pki/trs/ios_core.p7b)  
 Loading [https://www.cisco.com/security/pki/trs/ios\\_core.p7b](https://www.cisco.com/security/pki/trs/ios_core.p7b)  
 % PEM files import succeeded.

#### 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 123
  rule 1 /^+1\(.*)/ ^1/
voice translation-rule 224
  rule 1 /^\([1-9].....\) / +1\1/
voice translation-rule 325
  rule 1 /^+44\(.*)/ ^1/
voice translation-rule 526
  rule 1 /^\([2-9].....\) / +44\1/
```

#### 5.4.2.8 Dial Peers

Dial Peers are static route table, mapping phone numbers to interfaces or IP addresses. A dial peer is associated or matched to each call leg according to the destination

---

<sup>23</sup> To remove “+1” before sending the number to CUCM

<sup>24</sup> To add “+1” before sending the number to Twilio

<sup>25</sup> To remove “+44” before sending the number to CUCM

<sup>26</sup> To add “+44” before sending the number to Twilio

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

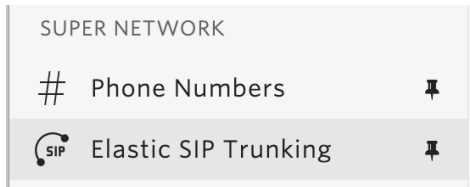
```

```
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 rtp-nte
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
```

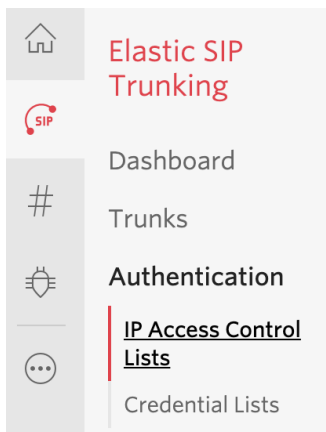
## 6 Twilio Elastic SIP Trunking Configuration

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).



## Tekvizion

### Properties

FRIENDLY NAME

IP-ACL SID AL520093ee3d260df1e3cb0a41a071deb1f

ASSOCIATED SIP TRUNKS [tekvizion](#)

ASSOCIATED SIP DOMAINS —

### IP Address Ranges

IP Access Control Lists may have up to 100 IP addresses.

+	IP ADDRESS RANGE	FRIENDLY NAME
	199.182.124.230 / 32 199.182.124.230 - 199.182.124.230	

## 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.

Create A New SIP Trunk ×

Name your new SIP Trunk, then configure it in the following steps.

FRIENDLY NAME

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 .pstn.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.

▼ Show Localized URIs

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:

**i Attention:** We have updated the syntax for localized SIP hostnames to use our new Edge Locations.  
[View legacy Termination SIP URIs](#)

<b>North America Virginia</b>	tekvizion.pstn.ashburn.twilio.com
<b>North America Oregon</b>	tekvizion.pstn.umatilla.twilio.com
<b>Europe Dublin</b>	tekvizion.pstn.dublin.twilio.com
<b>Europe Frankfurt</b>	tekvizion.pstn.frankfurt.twilio.com
<b>South America Sao Paulo</b>	tekvizion.pstn.sao-paulo.twilio.com
<b>Asia Pacific Singapore</b>	tekvizion.pstn.singapore.twilio.com
<b>Asia Pacific Tokyo</b>	tekvizion.pstn.tokyo.twilio.com
<b>Asia Pacific Sydney</b>	tekvizion.pstn.sydney.twilio.com

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**

**Priority**

i

Numeric range from 0 to 65535.

**Weight**

i

Numeric range from 1 to 65535.

**Enabled**

enabled

Cancel

Add

Continue to add the other Origination URIs, so you have the following configuration:

## Origination URIs

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

► **Show more about provisioning for high service availability**

+	ORIGINATION URI	PRIORITY	WEIGHT	ENABLED	
	sip:199.182.124.230;edge=ashburn	10	10	✓	✗
	sip:199.182.124.231;edge=umatilla	10	10	✓	✗

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.

### Numbers

Add a number ▾

Filter Parameter: Number ▾      Number: +1415GETTWLO      Filter      Configure Emergency Calling ▾      Remove from trunk

<input type="checkbox"/>	Number	Friendly Name	Emergency Status	Emergency Address	Country	
<input type="checkbox"/>	+18149260011	(814) 926-0011	Enabled	375 Beale St. Ste. 300, San Francisco, CA, 94105	United States	<a href="#">View details</a>
<input type="checkbox"/>	+15675220022	(567) 522-0022	Disabled	-	United States	<a href="#">View details</a>
<input type="checkbox"/>	+447403922739	447403922739	Not Available	-	United Kingdom	<a href="#">View details</a>
<input type="checkbox"/>	+15407810033	(540) 781-0033	Disabled	-	United States	<a href="#">View details</a>

## 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$IFUFZjSRnpLmC7kdnFZeYoxjm8Wzk952nE7Vv0IzpkU
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>"
  request CANCEL sip-header From modify "(<.*:.*)(@.*>)"
"\1@tekvizion.pstn.twilio.com>"
  request INVITE sip-header To modify "(<.*:.*)(@.*>)" "\1@tekvizion.pstn.twilio.com>"
  request REINVITE sip-header To modify "(<.*:.*)(@.*>)" "\1@tekvizion.pstn.twilio.com>"
  request INVITE sip-header From modify "(<.*:.*)(@.*>)"
"\1@tekvizion.pstn.twilio.com;user=phone>"
  request INVITE sip-header P-Asserted-Identity modify "(<.*:.*)(@.*>)"
"\1@tekvizion.pstn.twilio.com;user=phone>"
  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>"
!
voice translation-rule 1
  rule 1 /^+1\(.*)/ \1/
!
voice translation-rule 2
  rule 1 /^\[1-9].....\)/ /+1\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
 rsakeypair 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.X27 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
!
control-plane
!
dial-peer voice 1 voip
 description Incoming from CUCM
 session protocol sipv2
 session transport udp

```

---

<sup>27</sup> 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
```

