



# **Nexmo SIP Trunking Configuration Guide**

**CUCM 11.5.1.12900-21  
With  
CUBE 16.05.01b**

**May 2017**

# Contents

1 Introduction.....	3
2 SIP Trunking Network Components.....	3
2.1 Hardware Components.....	3
2.2 Software Requirements.....	4
3 Features.....	4
3.1.1 Features Supported.....	4
3.1.2 Features Not Supported by PBX.....	4
3.1.3 Caveats and Limitations.....	4
4 Configuration.....	5
4.1 IP Address Worksheet.....	5
4.2 Configuring Cisco Unified Communications Manager.....	6
4.2.1 Cisco UCM Version.....	6
4.2.2 Cisco Unified Call Manager Service Parameters.....	6
4.2.3 Offnet Calls via Nexmo SIP Trunk.....	7
4.2.4 Dial Plan.....	14
4.3 Configuring Cisco Unified Border Element.....	14
4.3.1 Network Interface.....	14
4.3.2 Global Cisco UBE Settings.....	14
4.3.3 Codecs.....	15
4.3.4 Dial Peer.....	15
4.3.5 Configuration Example.....	16
4.4 Configure Numbers in Nexmo Account.....	21

# 1 Introduction

This document is intended for Nexmo SIP trunk customer's technical staff and Value Added Retailer (VAR) having installation and operational responsibilities. This configuration guide provides steps for configuring Cisco Unified Communications Manager (Cisco UCM) 11.5.1.12900-21 and Cisco Unified Border Element (Cisco UBE) 16.05.01b to Nexmo SIP Trunking services.

## 2 SIP Trunking Network Components

The network for the SIP trunk reference configuration is illustrated below and is representative of a Cisco UCM and Cisco UBE configuration to Nexmo SIP trunking.

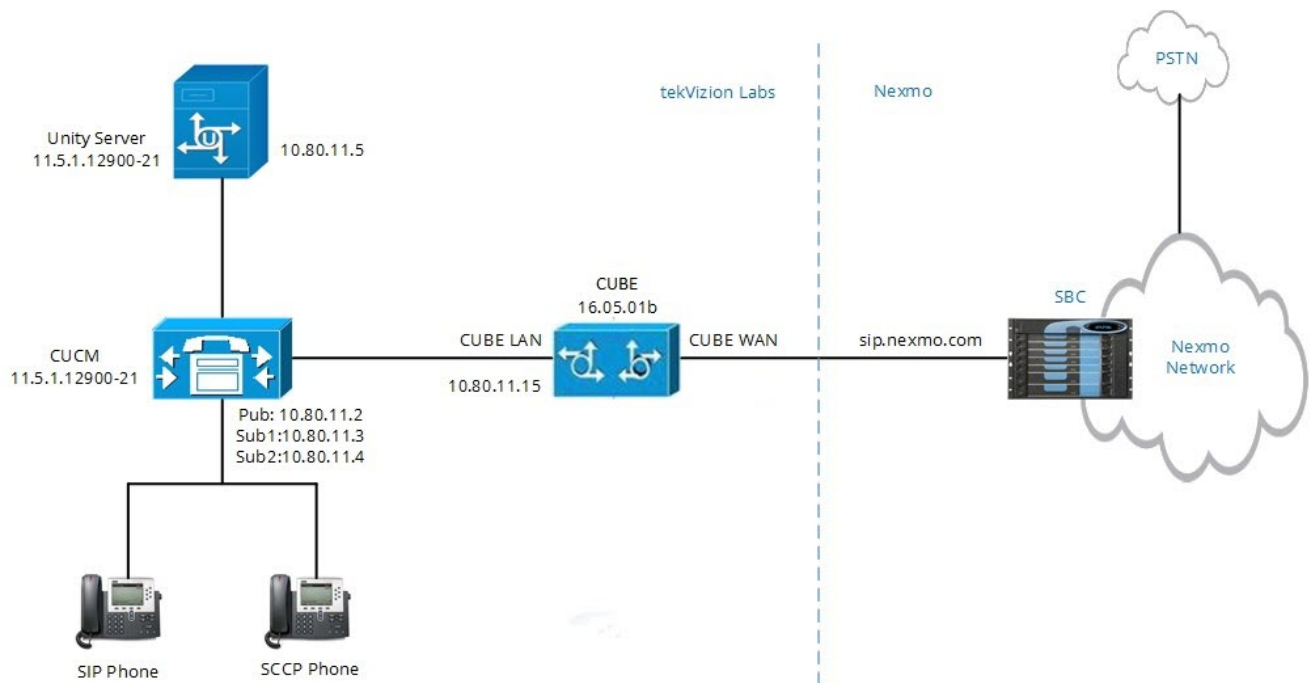


Figure 1: Topology Diagram

### 2.1 Hardware Components

- Cisco UCS-C240-M3S VMWare host running ESXi 5.5.0 Standard
- Cisco ISR4321/K9 router as CUBE
- Cisco ISR4321/K9 (1RU) processor with 1684579K/6147K bytes of memory with 3 Gigabit Ethernet interfaces
- Processor board ID FLM1925W0X0
- IP Phones 7942(SCCP), 7841(SIP)

## 2.2 Software Requirements

- Cisco Unified Communications Manager 11.5.1.12900-21
- Cisco Unity Connection 11.5.1.12900-21
- IOS 16.05.01b for ISR4321/K9 Cisco Unified Border Element
- Cisco IOS Software [Everest], ISR Software (X86\_64\_LINUX\_IOSD-UNIVERSALK9-M), Version 16.5.1b, RELEASE SOFTWARE (fc1)
- Cisco IOS XE Software, Version 16.05.01b

## 3 Features

### 3.1.1 Features Supported

- Incoming and outgoing off-net calls using G711ULAW & G711ALAW voice codecs
- Calling Line (number) Identification Presentation
- Calling Line (number) Identification Restriction
- Call hold and resume
- Call transfer (unattended and attended )
- Call Conference
- Call forward (all, no answer)
- DTMF relay both directions (RFC2833)
- Media flow-through on Cisco UBE

### 3.1.2 Features Not Supported by PBX

- None

### 3.1.3 Caveats and Limitations

- Caller ID is not updated after attended or semi-attended transfers to off-net phones. This is due to a limitation on Cisco UBE. The issue does not impact the calls.

## 4 Configuration

### 4.1 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 1 – IP Addresses*

Component	Lab Value	Customer Value
<b>Cisco UBE</b>		
LAN IP Address	10.80.11.15	
LAN Subnet Mask	255.255.255.0	
WAN IP Address	192.65.79.XXX	
WAN Subnet Mask	255.255.255.128	
<b>Cisco UCM IP PBX</b>		
System IP Address	10.80.11.2	

## 4.2 Configuring Cisco Unified Communications Manager

### 4.2.1 Cisco UCM Version

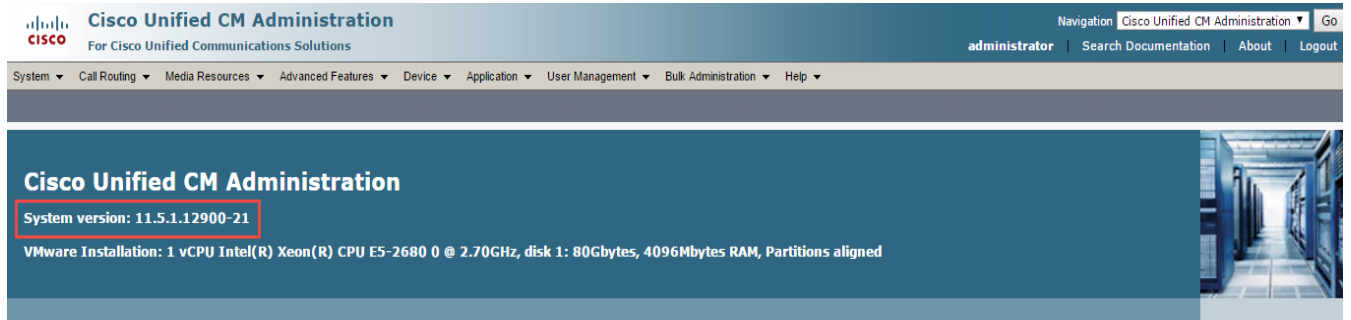


Figure 2: Cisco UCM Version

### 4.2.2 Cisco Unified Call Manager Service Parameters

Navigation: System → Service Parameters

1. Select **Server**: clus21pub--CUCM Voice/Video (Active)
2. Select **Service**: Cisco CallManager (Active)
3. All other fields are set to default values

**Select Server and Service**

Server\*

Service\*

All parameters apply only to the current server except parameters that are in the cluster-wide group(s).

---

**Cisco CallManager (Active) Parameters on server clus21pub--CUCM Voice/Video (Active)**

Parameter Name	Parameter Value	Suggested Value
<b>Call Throttling</b>		
<a href="#">Code Yellow Entry Latency</a> *	<input type="text" value="20"/>	20
<a href="#">Code Yellow Exit Latency Calculation</a> *	<input type="text" value="40"/>	40
<a href="#">Code Yellow Duration</a> *	<input type="text" value="5"/>	5
<a href="#">Max Events Allowed</a> *	<input type="text" value="2000"/>	2000
<a href="#">System Throttle Sample Size</a> *	<input type="text" value="10"/>	10

Figure 3: Service Parameters

### 4.2.3 Offnet Calls via Nexmo SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and Nexmo Network and calls are routed via Cisco UBE. From Cisco UBE, we have pointed the trunk to sip.nexmo.com and opened the firewall for the list of IPs in the portal provided by Nexmo.

#### 4.2.3.1 SIP Trunk Security Profile

Navigation: System → Security → SIP Trunk Security Profile

1. Set **Name**: *Non Secure SIP Trunk Profile* is used as an example
2. Set **Outgoing Transport Type**: UDP in this example
3. SIP trunks to Nexmo should use UDP as a transport protocol for SIP. This is configured using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.

SIP Trunk Security Profile Information	
Name*	Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile for Nexmo
Device Security Mode	Non Secure ▼
Incoming Transport Type*	TCP+UDP ▼
Outgoing Transport Type	UDP ▼
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
X.509 Subject Name	
Incoming Port*	5060
<input type="checkbox"/> Enable Application level authorization	
<input type="checkbox"/> Accept presence subscription	
<input type="checkbox"/> Accept out-of-dialog refer**	
<input type="checkbox"/> Accept unsolicited notification	
<input type="checkbox"/> Accept replaces header	
<input type="checkbox"/> Transmit security status	
<input type="checkbox"/> Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	Use Default Filter ▼

Figure 4: SIP Trunk Security Profile

### 4.2.3.2 SIP Profile Configuration

Navigation: Device → Device Settings → SIP Profile

1. Set **Name**: *Standard SIP Profile* is used as an example

SIP Profile Information	
Name*	Standard SIP Profile
Description	SIP Profile for Nexmo
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-Agen
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, anc
Confidential Access Level Headers*	Disabled
<input type="checkbox"/> Redirect by Application	
<input type="checkbox"/> Disable Early Media on 180	
<input type="checkbox"/> Outgoing T.38 INVITE include audio mline	
<input type="checkbox"/> Offer valid IP and Send/Receive mode only for T.38 Fax Relay	
<input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests	
<input type="checkbox"/> Assured Services SIP conformance	
<input type="checkbox"/> Enable External QoS**	
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
<input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	
<input type="checkbox"/> 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	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> 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
DSCP for Audio Portion of Video Calls	Use System Default



Figure 5: SIP Profile Configuration

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
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial
<input checked="" type="checkbox"/> Conference Join Enabled	
<input type="checkbox"/> RFC 2543 Hold	
<input checked="" type="checkbox"/> Semi Attended Transfer	
<input type="checkbox"/> Enable VAD	
<input type="checkbox"/> Stutter Message Waiting	
<input type="checkbox"/> MLPP User Authorization	

<b>Normalization Script</b>							
Normalization Script	< None > ▼						
<input type="checkbox"/> Enable Trace							
	<table border="1"> <thead> <tr> <th></th> <th>Parameter Name</th> <th>Parameter Value</th> </tr> </thead> <tbody> <tr> <td>1</td> <td></td> <td></td> </tr> </tbody> </table>		Parameter Name	Parameter Value	1		
	Parameter Name	Parameter Value					
1							
<b>Incoming Requests FROM URI Settings</b>							
Caller ID DN							
Caller Name							

Figure 6: SIP Profile Configuration – Cont.

**Trunk Specific Configuration**

Reroute Incoming Request to new Trunk based on\*

Resource Priority Namespace List

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

Allow Passthrough of Configured Line Device Caller Information

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\*

---

**SDP Information**

Send send-receive SDP in mid-call INVITE

Allow Presentation Sharing using BFCP

Allow iX Application Media

Allow multiple codecs in answer SDP

Figure 7: SIP Profile Configuration (cont.)

### 4.2.3.3 SIP Trunk Configuration

Create SIP trunk to Cisco UBE

**Navigation:** Device → Trunk

Name ^	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile
<a href="#">Trunk to CUBE 10.80.11.15</a>	Nexmo CUBE		<a href="#">Default</a>	<a href="#">8.@</a>				SIP Trunk	Unknown - OPTIONS Ping not enabled		<a href="#">Non_Secure SIP Trunk Profile</a>

Figure 8: SIP Trunk List

**Device Information**

Product: SIP Trunk  
 Device Protocol: SIP  
 Trunk Service Type: None(Default)

Device Name\*:

Description:

Device Pool\*:  ▼

Common Device Configuration:  ▼

Call Classification\*:  ▼

Media Resource Group List:  ▼

Location\*:  ▼

AAR Group:  ▼

Tunneled Protocol\*:  ▼

QSIG Variant\*:  ▼

ASN.1 ROSE OID Encoding\*:  ▼

Packet Capture Mode\*:  ▼

Packet Capture Duration:

Media Termination Point Required

Retry Video Call as Audio

Path Replacement Support

Transmit UTF-8 for Calling Party Name

Transmit UTF-8 Names in QSIG APDU

Figure 9: SIP Trunk to Cisco UBE

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\*

Route Class Signaling Enabled\*

Use Trusted Relay Point\*

PSTN Access

Run On All Active Unified CM Nodes

---

**Intercompany Media Engine (IME)**

E.164 Transformation Profile

---

**MLPP and Confidential Access Level Information**

MLPP Domain

Confidential Access Mode

Confidential Access Level

---

**Call Routing Information**

Remote-Party-Id

Asserted-Identity

Asserted-Type\*

SIP Privacy\*

**Inbound Calls**

Significant Digits\*

Connected Line ID Presentation\*

Connected Name Presentation\*

Calling Search Space

AAR Calling Search Space

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.

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

Figure 10: SIP Trunk to Cisco UBE – Cont.

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

**Caller Information**

Caller ID DN

Caller Name

Maintain Original Caller ID DN and Caller Name in Identity Headers

**SIP Information**

**Destination**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	10.80.11.15		5060

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 [View Details](#)

DTMF Signaling Method\* No Preference

**Normalization Script**

Normalization Script < None >

Enable Trace

	Parameter Name	Parameter Value
1		

**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

Figure 11: SIP Trunk to Cisco UBE – Cont.

## 4.2.4 Dial Plan

**Navigation:** Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

Cisco IP phone dial “8”+11 digit number to access PSTN via Cisco UBE. “8” is removed before sending to Cisco UBE.

Find Route Patterns where	Pattern	begins with	Find	Clear Filter	+	-
<input type="checkbox"/>	Pattern ^	Description	Partition	Route Filter	Associated Device	
<input type="checkbox"/>	8.@	To Nexmo CUBE 10_80_11_15			<a href="#">Trunk to CUBE 10.80.11.15</a>	

Figure 12 Route Pattern List

## 4.3 Configuring Cisco Unified Border Element

### 4.3.1 Network Interface

Configure Ethernet IP address and sub interface. The IP address and VLAN encapsulation used are for illustration only, the actual IP address can vary. For SIP trunks two IP addresses must be configured—LAN and WAN.

```
interface GigabitEthernet0/0/0
ip address 192.65.79.XXX 255.255.255.128
negotiation auto
interface GigabitEthernet0/0/1
ip address 10.80.11.15 255.255.255.0
negotiation auto
```

### 4.3.2 Global Cisco UBE Settings

In order to enable Cisco UBE IP2IP gateway functionality, enter the following:

```
voice service voip
ip address trusted list
ipv4 173.193.199.24
ipv4 174.37.245.34
ipv4 5.10.112.121
ipv4 5.10.112.122
ipv4 119.81.44.6
ipv4 119.81.44.7
```

```
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
sip
  session refresh
  asserted-id pai
  early-offer forced
  midcall-signaling passthru
  g729 annexb-all
```

### 4.3.3 Codecs

G711ulaw and G711alaw voice codecs are used for this testing. Codec preferences used to change according to the test plan description

```
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g711alaw
```

### 4.3.4 Dial Peer

Cisco UBE uses dial-peer to route the call based on the digit to route the call accordingly.

```
dial-peer voice 1 voip
  description incoming dial-peer from CUCM to CUBE
  session protocol sipv2
  session transport udp
  incoming called-number .T
  voice-class codec 1
  dtmf-relay rtp-nte
  no vad
dial-peer voice 2 voip
  description outgoing dial-peer from CUBE to CUCM
  destination-pattern 120.....
  session protocol sipv2
  session target ipv4:10.80.11.2
  session transport udp
  voice-class codec 1
  voice-class sip options-keepalive
```

```
dtmf-relay rtp-nte
no vad
dial-peer voice 3 voip
description incoming dial-peer from NEXMO to CUBE
session protocol sipv2
session transport udp
incoming called-number 120.....
voice-class codec 1
dtmf-relay rtp-nte
no vad
dial-peer voice 4 voip
description outgoing dial-peer from CUBE to NEXMO
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
dtmf-relay rtp-nte
no vad
```

#### 4.3.5 Configuration Example

User Access Verification

Username: cisco

Password:

nexmo#

nexmo#sh run

Building configuration...

Current configuration : 5992 bytes

version 16.5

service timestamps debug datetime msec

service timestamps log datetime msec

service password-encryption

platform qfp utilization monitor load 80

no platform punt-keepalive disable-kernel-core

hostname nexmo

boot-start-marker



```
boot system flash isr4300-universalk9.16.05.01b.SPA.bin
boot-end-marker
vrf definition Mgmt-intf

address-family ipv4
  exit-address-family
  address-family ipv6
  exit-address-family
enable secret 5
no aaa new-model
ip name-server 8.8.8.8
subscriber templating
multilink bundle-name authenticated
crypto pki trustpoint TP-self-signed-1017057749
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-1017057749
  revocation-check none
  rsakeypair TP-self-signed-1017057749
crypto pki certificate chain TP-self-signed-1017057749
  certificate self-signed 01
    30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
    31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
    69666963 6174652D 31303137 30353737 3439301E 170D3137 30353130 31353233
    34315A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
    4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 30313730
    35373734 39308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
    0A028201 0100BF99 0B3B8C33 835DC696 011A6384 ACF8B705 E34D0B17 9BF7A355
    BAB68AED 970A3529 C4780464 92AD7408 96C38292 F286685A 0C3A285C 614EC7A7
    E0D3F7B3 D38037E0 C828DBB8 F08F5474 8A453D68 D3FAAB83 004BA2F3 55201661
    1E4F6DBD 9C0771B4 E8EF4B08 C70CDAD1 8C5F8B00 3C07FEC2 375FE2E3 73BD4F47
    FD1B4F88 D6D19FAB C23069E0 F91E6099 FB7B00D4 0D7D5419 F5570F93 EFBB5C79
    EE86DC0B 72043F04 C7F2B07E 0E681425 705762BF 8B7A0360 25C1077A 2A2BC17A
    68F75A15 7E2439F7 770D90F1 0E8C00F3 65AA0D65 6B891C32 BA19C16E 3B902974
    4A296DB1 8E3E7AD3 694A03AF FA3B5051 D1762F4E 26CBCF74 57DEA2B8 35FDAA31
    44E65C43 76B30203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
    301F0603 551D2304 18301680 144171AB 9DC3C6B5 F0CA2C01 78ADDAA8 FB66024B
    70301D06 03551D0E 04160414 4171AB9D C3C6B5F0 CA2C0178 ADDAA8FB 66024B70
    300D0609 2A864886 F70D0101 05050003 82010100 003606AE 1AFB9104 447F53BB
    71338C17 F4848B40 9F4A9AA7 9CB791AE 44B73856 241CB923 FD0B0109 2F51F91B
```

B5CD1660 D54BEF67 354213D4 2A442000 B0662481 36D063B3 9BD7D567 46A85C9A  
9AC3E4CD 4B373ECB C8F91089 AF698DCD 37002793 AE1B645A 5F5C1EA2 CBEF72D5  
0763A01E D25FC6C1 A06AF364 47AC82E4 134C463B 176D32CD 16A0AD15 383FB164  
D62134E5 218478F0 5B389D19 75A2C399 C1CC40B5 6AC3DAB2 8AA5D21D 25728B12  
6696650C 5220DB5F A22A304C 8F37EA5C A1C2C37B 7C58F5D2 4B214B5E A1C99E67  
A741E30D 798A7C2F 92F15D55 D8E74340 3A3AF3EB 048EE669 85B8F7FD 5B607C98  
AB1BB24D 0C8B76C4 FAC45B66 52CF5BC0 9CCDFE0B

voice service voip

ip address trusted list

ipv4 173.193.199.24

ipv4 174.37.245.34

ipv4 5.10.112.121

ipv4 5.10.112.122

ipv4 119.81.44.6

ipv4 119.81.44.7

address-hiding

mode border-element license capacity 20

allow-connections sip to sip

sip

session refresh

asserted-id pai

early-offer forced

midcall-signaling passthru

g729 annexb-all

voice class codec 1

codec preference 1 g711ulaw

codec preference 2 g711alaw

license udi pid ISR4321/K9 sn FDO19220MQ9

license boot suite AdvUCSuiteK9

license boot level uck9

diagnostic bootup level minimal

spanning-tree extend system-id

username cisco privilege 15 password 7

redundancy

mode none

interface GigabitEthernet0/0/0

ip address 192.65.79.160 255.255.255.128

negotiation auto

interface GigabitEthernet0/0/1

```
ip address 10.80.11.15 255.255.255.0
negotiation auto
interface GigabitEthernet0/1/0
no ip address
negotiation auto
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto
threat-visibility
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 172.16.24.0 255.255.248.0 10.80.11.1
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
control-plane
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
mgcp profile default
dial-peer voice 4 voip
description outgoing dial-peer from CUBE to NEXMO
destination-pattern .T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
dtmf-relay rtp-nte
no vad
dial-peer voice 1 voip
description incoming dial-peer from CUCM to CUBE
session protocol sipv2
```

```
session transport udp
incoming called-number .T
voice-class codec 1
dtmf-relay rtp-nte
no vad
dial-peer voice 2 voip
description outgoing dial-peer from CUBE to CUCM
destination-pattern 120.....
session protocol sipv2
session target ipv4:10.80.11.2
session transport udp
voice-class codec 1
voice-class sip options-keepalive
dtmf-relay rtp-nte
no vad
dial-peer voice 3 voip
description incoming dial-peer from NEXMO to CUBE
session protocol sipv2
session transport udp
incoming called-number 120.....
voice-class codec 1
dtmf-relay rtp-nte
no vad
sip-ua
credentials number 12014647035 username 911236e3 password 7 realm sip.nexmo.com
authentication username 911236e3 password 7
sip-server dns:sip.nexmo.com:5060
line con 0
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
login local
no network-clock synchronization automatic
```

## 4.4 Configure Numbers in Nexmo Account

1. Login to the Nexmo account using the credentials provided at the time of registration. A **Key** and **Secret** will be displayed on the dashboard and this can be used as the username and password for Registration SIP Trunks.

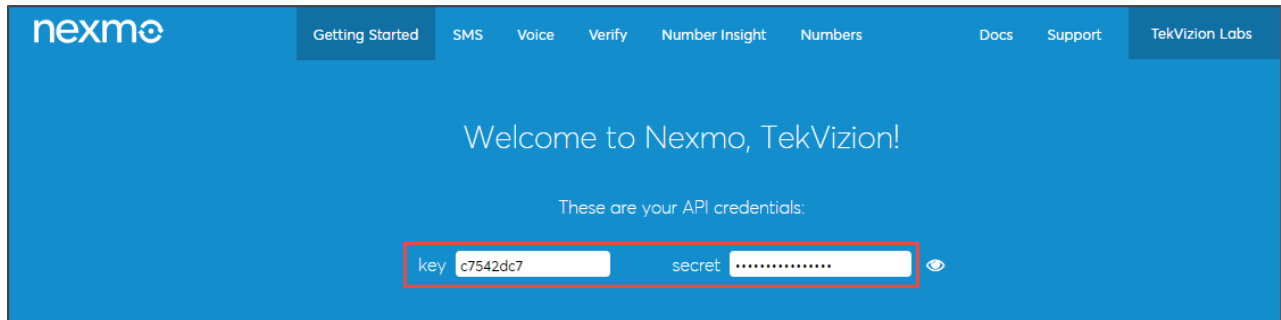


Figure 13: Nexmo Dashboard

2. In order to provide the URL to which the call has to be routed from Nexmo, navigate to the **Numbers** tab
3. Click **Edit** against each number as shown below

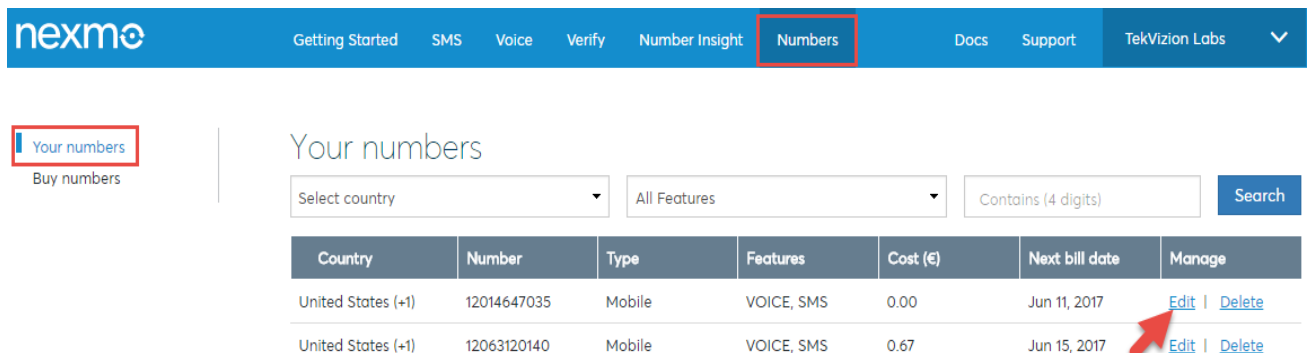


Figure 14: Your Numbers

1. A pop-up will be displayed
2. Select the “**Forward to**” and provide the URL to which the calls route
3. Click **Update** to save the changes

Settings for 12014647035 ×

SMS

Webhook URL

Voice

Forward to	URL
Forward to SIP ▼	12014647035@nexmo.tekvizionlabs.⋮

Status webhook URL

Figure 15: Your Numbers – Cont.