Spectrum Enterprise SIP Trunking Service
Cisco CUCM/CUBE
IP PBX Configuration Guide

About Spectrum Enterprise:

Spectrum Enterprise is a division of Charter Communications following a merger with Time Warner Cable and acquisition of Bright House Networks. Spectrum Enterprise is a national provider of scalable, fiber technology solutions. The Spectrum Enterprise portfolio includes networking and managed services solutions, including Internet access, Ethernet and Managed Network Services, Voice, TV and Cloud solutions. Our industry-leading team of experts works closely with clients to achieve greater business success.

About this document:

Spectrum Enterprise assures IP PBX compatibility by conducting interoperability testing to ensure any potential compatibility issues have been resolved prior to installation. Please review the IP PBX configuration instructions in this guide prior to your installation date.

Be advised that this document may contain references to Charter or Charter Business. All references to Charter should be read as Spectrum Enterprise.

Thank you,

Spectrum Enterprise

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Cisco CUCM/CUBE IP PBX Configuration Guide

Charter COMMUNICATIONS

1 Introduction

The document describes how to configure the Cisco Unified Communication Manager (CUCM)/ Cisco Unified Border Element (CUBE) IP PBX to interoperate within the Charter network. It does not provide any information how to provision, configure or use the features of the IP PBX. Please refer to the documentation provided with the IP PBX or contact the vendor.

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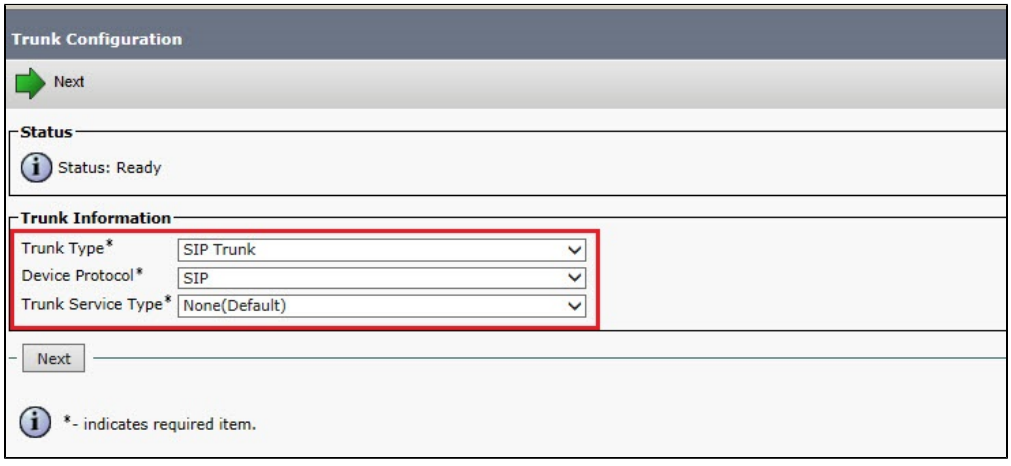
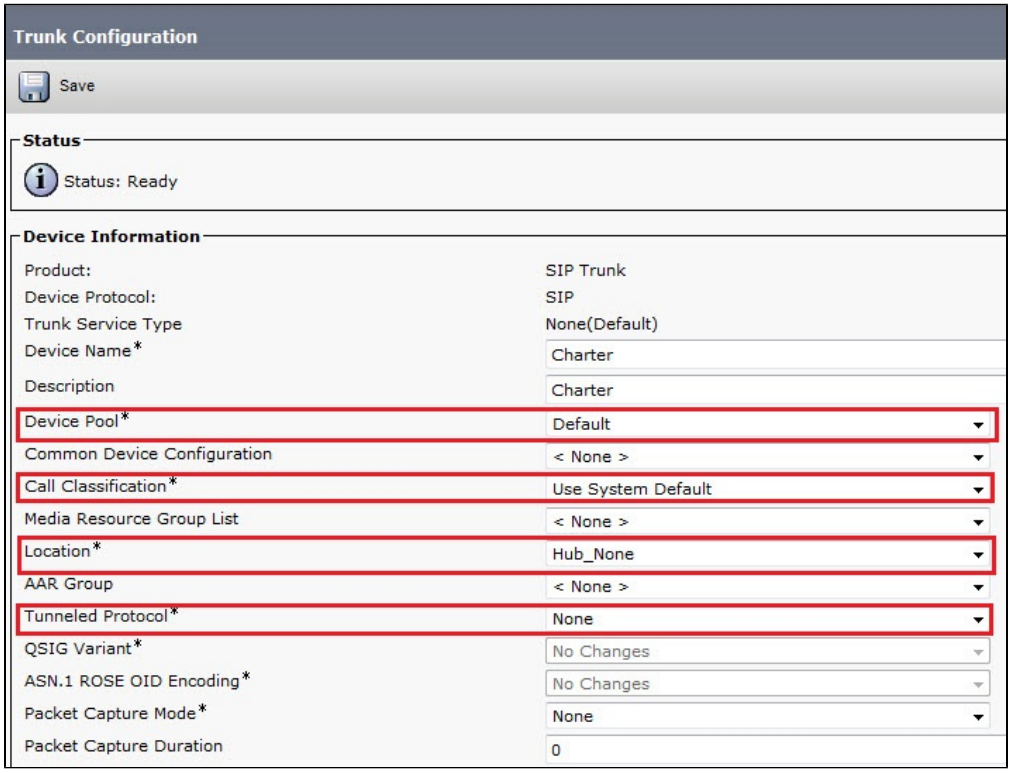
2 Trunk Profile Configuration

Follow the step-by-step procedure.

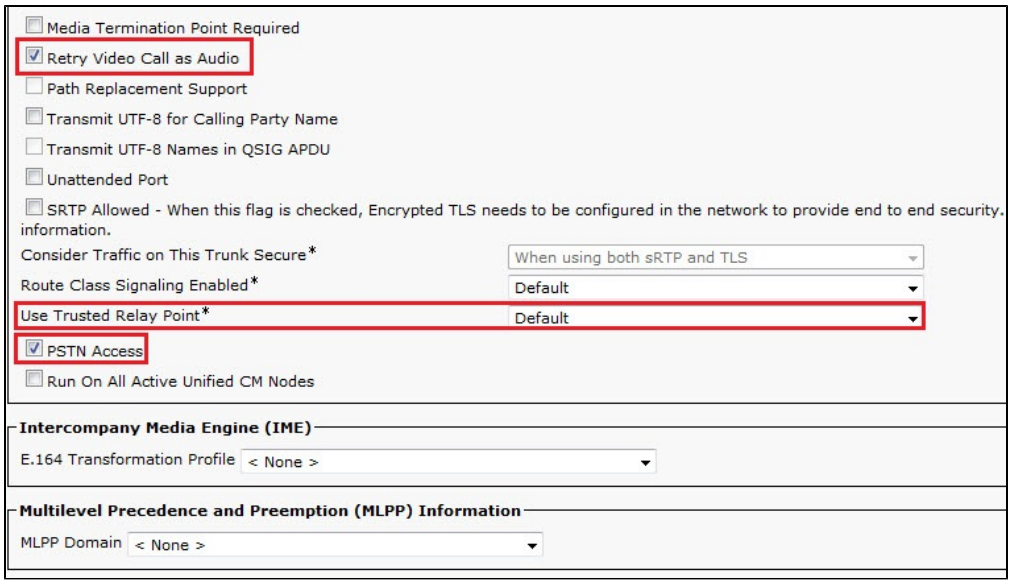
Steps	Action
1	Login to CUCM Administration
2	Navigate to **Device-> Trunk**
3	Click **Add New**
4	Follow the corresponding configuration figures for configuration of the SIP profile. Trunk Type: **SIP Trunk** Device Protocol: **SIP** Trunk Service Type: **None(Default)** Click on **Next**
5	Go to the next table.



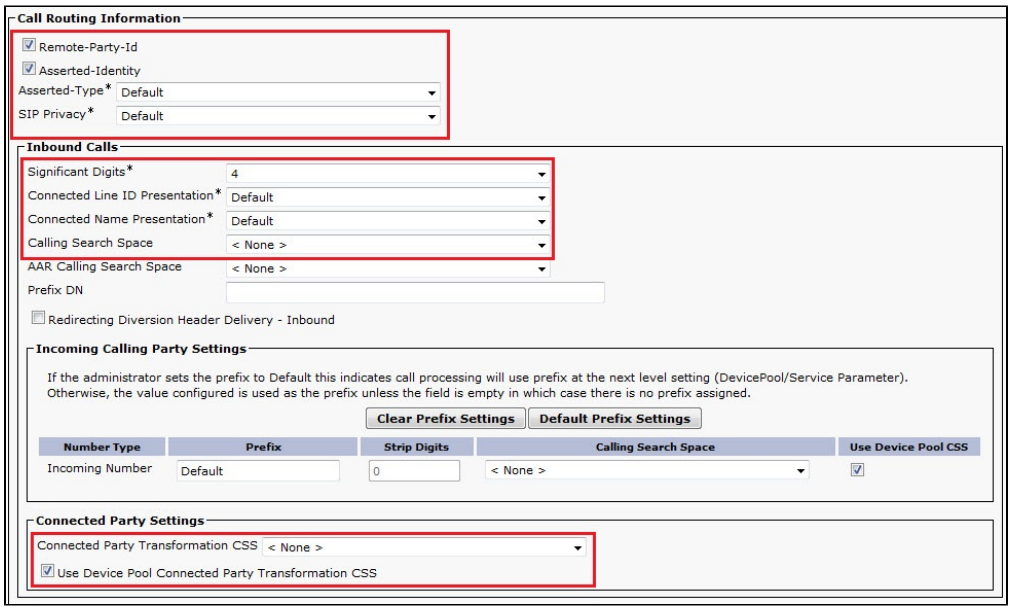
Steps	Action
1	Device Name: **“Charter” is used here for example**
2	Description: **Any relevant description can be used**
3	Device Pool: **Default**
4	Common Device Configuration: **None**
5	Call Classification: **Use System Default**
6	Media Resource Group List: **None**
7	Location: **Hub_None**
8	AAR Group: **None**
9	Tunneled Protocol: **None**
10	Confirm QSIG Variant: **No Changes**
11	Confirm ASN.1 ROSE OID Encoding: **No Changes**
12	Set Packet Capture Mode: **None**
13	Set Packet Capture Duration: **0**
14	Go to the next table.



Steps	Action
1	Confirm Media Termination Point Required: **unchecked**
2	Confirm Retry Video Call as Audio: **checked**
3	Confirm Path Replacement Support: **unchecked**
4	Confirm Transmit UTF-8 for Calling Party Name: **unchecked**
5	Confirm Transmit UTF-8 Names is QSIG APDU: **unchecked**
6	Confirm Unattended Port: **unchecked**
7	Confirm STP Allowed: **unchecked**
8	Confirm Consider Traffic on This Trunk Secure: **disabled**
9	Route Class Signaling Enabled **: Default**
10	Use Trusted Relay Point: **Default**
11	Confirm PSTN Access**: checked**
12	Confirm Run On All Active Unified CM Nodes: unchecked
13	E.164 Transformation profile: **None**
14	MLPP Domain: **None**
15	Go to the next table.



Steps	Action
1	Confirm Remote-Party-Id: **checked**
2	Confirm Asserted-Identity: **checked**
3	Asserted-Type: **Default**
4	SIP Privacy: **Default**
5	Significant Digits: **4**
6	Connected Line ID Presentation: **Default**
7	Connected Name Presentation: **Default**
8	Calling Search Space: **None**
9	AAR Calling Search Space: **None**
10	Confirm Prefix DN: **blank (no data)**
11	Redirecting Diversion Header Delivery-Inbound: **unchecked**
12	Under Incoming Calling Party Settings: Confirm Number Type: **Incoming Number** Prefix: **Default** Confirm Strip Digits: **is disabled** Calling Search Space: **None** Confirm Use Device Pool CSS: **is checked**
13	Connected Party Transformation CSS: **None**
14	Confirm Use Device Pool Connected party Transformation CSS: **is checked**
15	Go to the next table.



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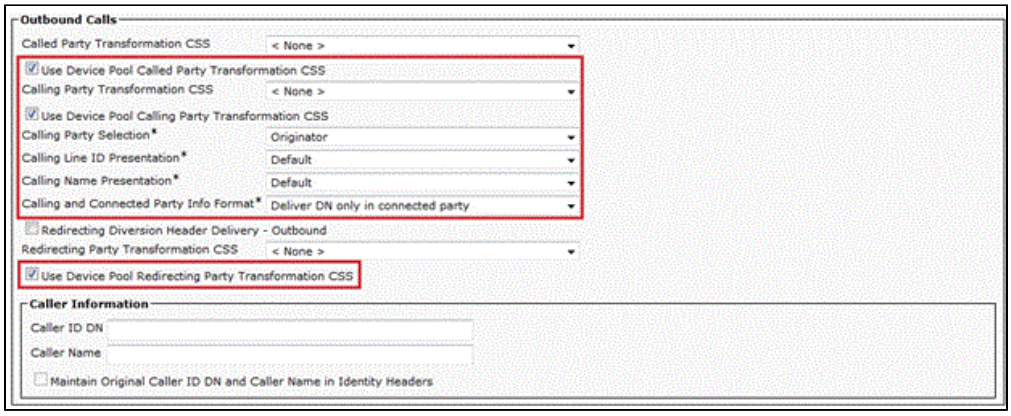
Steps	Action
1	Set Called Party Transformation CSS: **None**
2	Confirm Use Device Pool Called Party Transformation CSS: **checked**
3	Set Calling Party Transformation CSS: **None**
4	Confirm **Use Device Pool Calling Party Transformation CSS**: is checked
5	Set Calling Party Selection: **Originator**
6	Set Calling Line ID Presentation: **Default**
7	Set Calling Name Presentation: **Default**
8	Set Calling and Connected Party Info Format: **Deliver DN only in connected party**
9	Confirm Redirecting Diversion Header Delivery-Outbound: **unchecked**
10	Set Redirecting Party Transformation CSS: **None**
11	Confirm Use Device Pool Redirecting Party Transformation CSS: **checked**
12	Go to the next table.



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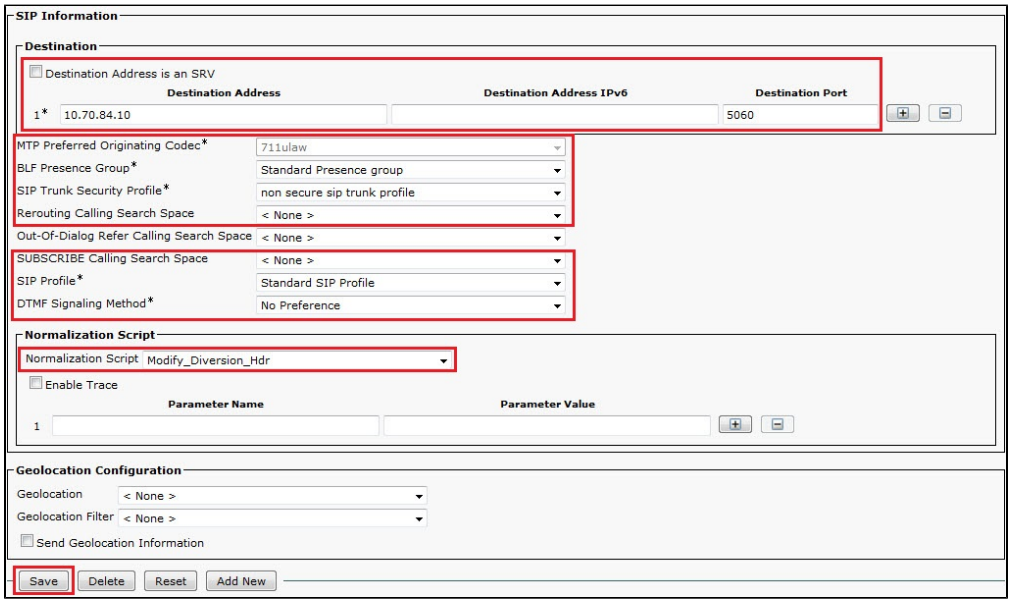
Steps	Action

1	Under SIP Information-Destination: Confirm Destination Address is an SRV: **unchecked** Destination Address: Use the actual LAN IP Address of Adtran (or E-SBC) in the network. Confirm Destination Address IPv6: **blank(no data)** Set Destination Port: **5060**
2	Comfirm MTP Preferred Originating Codec: **711ulaw**
3	BLF Presence Group: **Standard Presence group**
4	SIP Trunk Security Profile: **non secure sip trunk profile**
5	SIP Profile: **Standard SIP Profile**
6	DTMP Signaling Method: **RFC 2833**
7	Normalization Script: **Modify_Diversion_Hdr** (Script must be configured before setting option)
8	Click **Save** button to save SIP profile
9	Procedure completed



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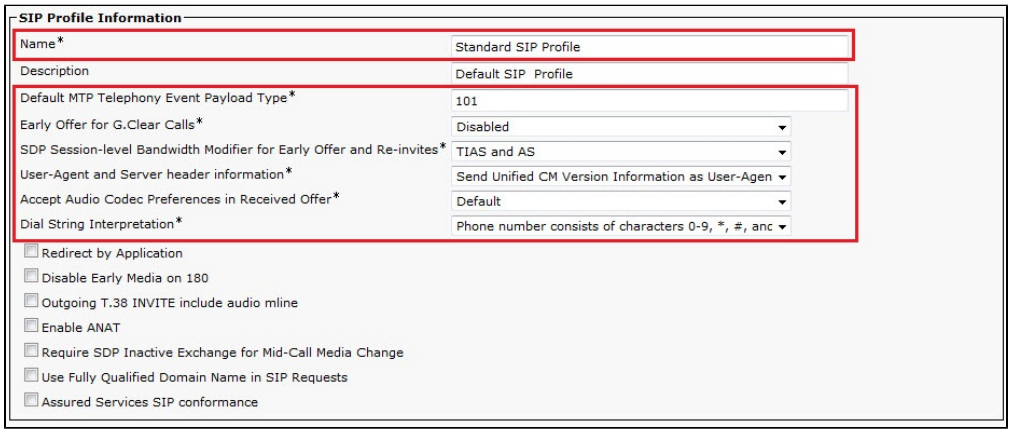
3 SIP Profile Configuration

Follow the step-by-step procedure.

Steps	Action

1	Set Name: **Appropriate name.** Standard SIP Profile is used for example.
2	Description: **Default SIP Profile**
3	Default MTP Telephony Event Payload Type: **101**
4	Early Offer for G.Clear Calls: **Disabled**
5	SDP Session-level Bandwidth Modifier for Early Offer and Re-invites: **TIAS and AS**
6	User-Agent and Server header information: **Send Unified CM Version Information as User-Agent Header**
7	Accept Audio Codec Preferences in Received Offer: **Default**
8	Dial String Interpretation: **Phone number consists of characters 0-9,*,#,and +** (others treated as URI addresses)
9	Confirm Redirect by Application: **unchecked**
10	Confirm Disable Early Media on 180: **unchecked**
11	Confirm Outgoing T.38 INVITE include audio mline: **unchecked**
12	Confirm Enable ANAT: **unchecked**
13	Confirm Require SDP Inactive Exchange for Mid-Call Media Change: **unchecked**
14	Confirm Use Fully Qualified Domain Name in SIP Requests: **unchecked**
15	Confirm Assured Services SIP conformance: **unchecked**
16	Go to next the table.



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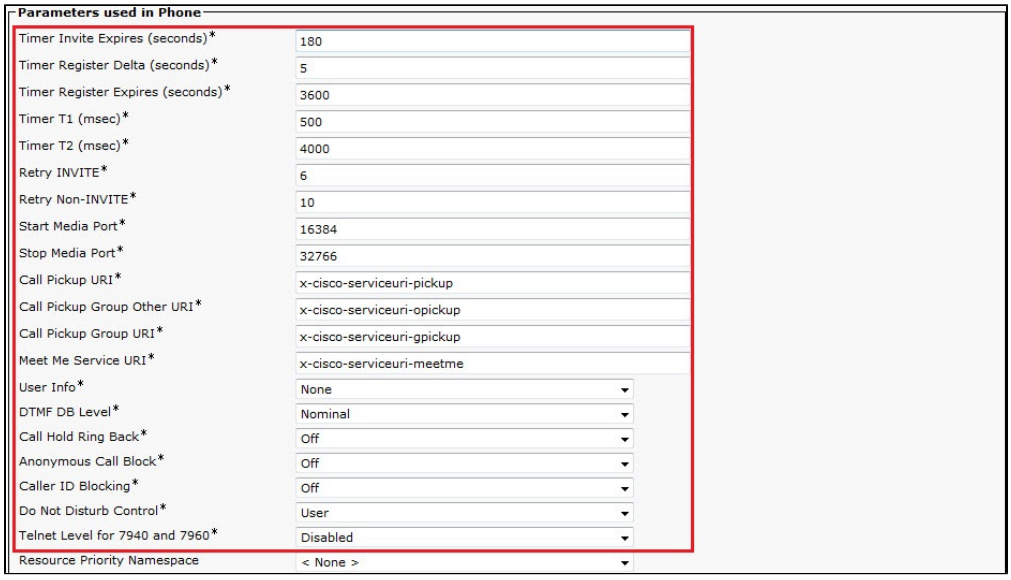
Steps	Action
1	Timer Invite Expires (seconds): **180**
2	Timer Register Delta (seconds): **5**
3	Timer Register Expires (seconds): **3600**
4	Timer T1 (msec): **500**

5	Timer T2 (msec): **4000**
6	Retry INVITE: **6**
7	Retry Non-INVITE: **10**
8	Start Media Port: **16384**
9	Stop Media Port: **32766**
10	Call Pickup URI: **x-cisco-serviceuri-pickup**
11	Call Pickup Group Other URI: **x-cisco-serviceuri-opickup**
12	Call Pickup Group URI: **x-cisco-serviceuri-gpickup**
13	Meet Me Service URI: **x-cisco-serviceuri-meetme**
14	User Info: **None**
15	DTMF DB Level: **Nominal**
16	Call Hold Ring Back: **Off**
17	Anonymous Call Block: **Off**
18	Caller ID Blocking: **Off**
19	Do Not Disturb Control: **User**
20	Telnet Level for 7940 and 7960: **Disabled**
21	Resource Priority Namespace: **None**
22	Go to the next table



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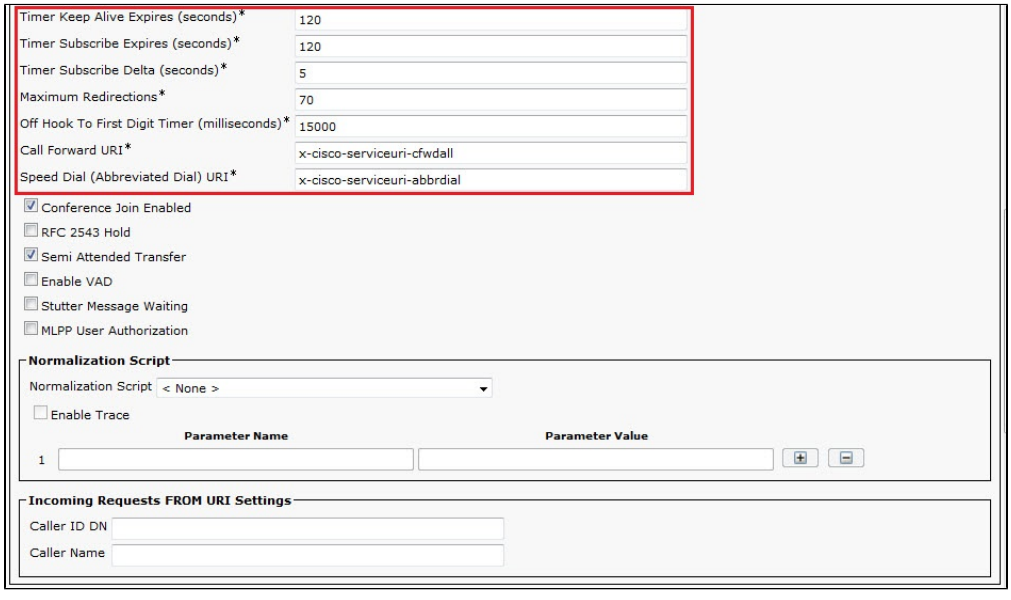
Steps	Action
1	Set Timer Keep Alive Expires (seconds): **120**
2	Set Timer Subscribe Expires (seconds): **120**
3	Set Timer Subscribe Delta (seconds): **5**
4	Set Maximum Redirections: **70**
5	Set Off Hook To First Digit Timer (milliseconds): **15000**
6	Set Call Forward URI: **x-cisco-serviceuri-cfwdall**
7	Set Speed Dial (Abbreviated Dial) URI: **x-cisco-serviceuri-abbrdial**
8	Confirm Conference Join Enabled: **checked**
9	Confirm RFC 2543 Hold: **unchecked**
10	Confirm Semi Attended Transfer: **checked**
11	Confirm Enable VAD: **unchecked**
12	Confirm Stutter Message Waiting: **unchecked**
13	Confirm MLPP User Authorization: **unchecked**
14	Set Normalization Script: **None**
15	Confirm Enable Trace: **unchecked**
16	Confirm Caller ID DN: **Blank**
17	Confirm Caller Name: **Blank**
18	Go to the next table



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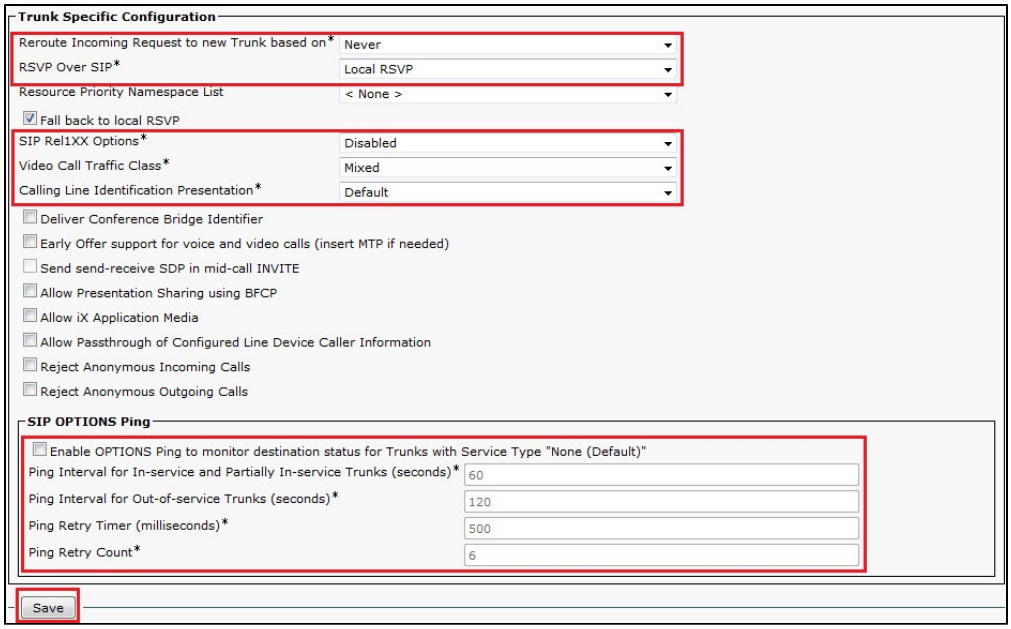
Steps	Action
1	Reroute Incoming Request to new Trunk based on: **Never**
2	RSVP Over SIP: **Local RSVP**
3	Resource Priority Namespace List: **None**
4	Confirm Fall back to local RSVP: **is checked**
5	SIP Rel1XX Options: **Disabled**
6	Video Call Traffic Class: **Mixed**
7	Calling Line Identification Presentation: **Default**
8	Confirm Deliver Conference Bridge Identifier: **is unchecked**
9	Confirm Early Offer support for voice and video calls (insert MTP if needed): **is unchecked**
10	Confirm Send send-receive SDP in mid-call INVITE: **is unchecked**
11	Confirm Allow Presentation Sharing using BFCP: **is unchecked**
12	Confirm Allow iX Application Media: **is unchecked**
13	Confirm Allow Passthrough of Configured Line Device Caller Information: **is unchecked**
14	Confirm Reject Anonymous Incoming Calls: **is unchecked**
15	Confirm Reject Anonymous Outgoing Calls: **is unchecked**
16	Confirm Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)": **is unchecked**
17	Ping Interval for In-service and Partially In-service Trunks (seconds): **60**
18	Ping Interval for Out-of-service Trunks (seconds): **120**
19	Ping Retry Timer (milliseconds): **500**
20	Ping Retry Count: **6**
21	Click **Save** button to apply changes
22	Procedure completed



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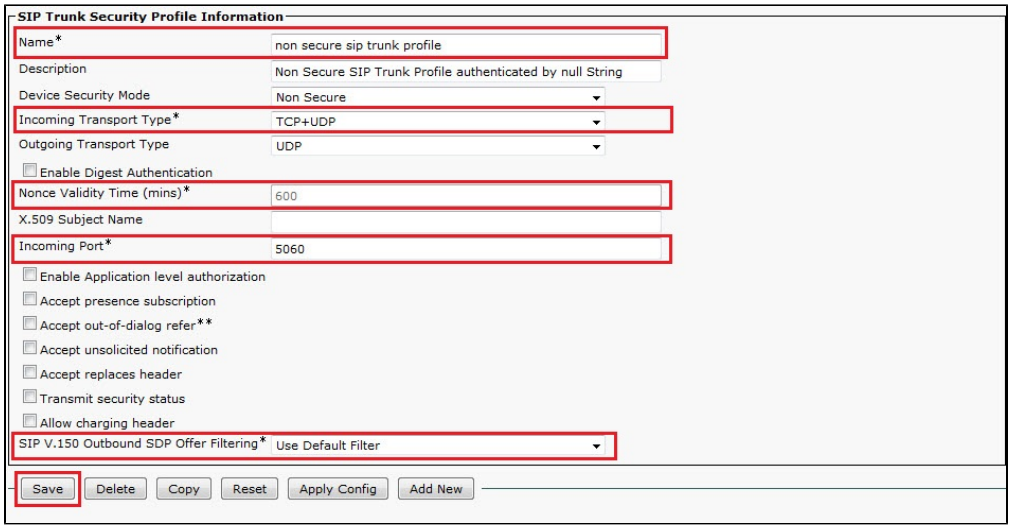
4 SIP Trunk Security Profile Configuration

For SIP trunk security profile configuration, follow the step-by-step procedure.

Steps	Action
1	Name: **An appropriate name**
2	**Description**: A description of your choosing
3	**Device Security Mode**: Non Secure
4	**Incoming Transport Type**: TCP+UDP
5	**Outgoing Transport Type**: UDP
6	Confirm Enable Digest Authentication: **is unchecked**
7	Nonce Validity Time (mins): **600**
8	X.509 Subject Name: **A name of your choosing**
9	Incoming Port: **5060**
10	Confirm Enable Application level authorization: **is unchecked**
11	Confirm Accept presence subscription: **is unchecked**
12	Confirm Accept out-of-dialog refer: **is unchecked**
13	Confirm Accept unsolicited notification: **is unchecked**
14	Confirm Accept replaces header: **is checked**

15	Confirm Transmit security status: **is unchecked**
16	Confirm Allow charging header: **is unchecked**
17	SIP V.150 Outbound SDP Offer Filtering: **Use Default Filter**
18	Click **Save** button to apply changes.
19	Procedure completed



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5 CUBE Configuration

Copy and paste the following:

```
version 15.3
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
!
boot-start-marker
boot system flash c2900-universalk9-mz.SPA.153-3.M1.bin
boot-end-marker
!
aqm-register-fnf
!
logging buffered 51200 warnings
enable secret 5 $1$iOXs$1q3Nr3rorSRJDaeqi7b0L.
!
no aaa new-model
clock timezone CST -6 0
clock summer-time CDT recurring
!
!
ip cef
no ipv6 cef
multilink bundle-name authenticated
!
!
!
!
!
!
!
!
voice-card 0
!
!
!
```

Voice Service Trusted IP Address List for CUCM

```
 voice service voip
 ip address trusted list
  ipv4 10.70.84.4
  ipv4 10.70.84.5
  ipv4 10.70.84.6
  ipv4 10.70.84.3
  ipv4 10.64.3.45
 rtcp keepalive
 address-hiding
 mode border-element
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip
 no supplementary-service sip moved-temporarily
 no supplementary-service sip refer
 fax protocol pass-through g711ulaw
 modem passthrough nse codec g711ulaw
 sip
  asserted-id pai
  privacy-policy passthru
  privacy-policy send-always
!
```

List Codecs in Order of Preference

```
 voice class codec 1
 codec preference 1 g711ulaw
!
!
!
!
!
!
license udi pid CISCO2921/K9 sn FTX1744AJ82
hw-module ism 0
!
hw-module pvdm 0/0
!
!
!
username cisco privilege 15 password 0 cisco
!
redundancy
!
!
!
!
!
!
interface Embedded-Service-Engine0/0
 no ip address
 shutdown
!
interface GigabitEthernet0/0
 description LAN side of CUBE to CUCM
```

LAN IP Address of the CUBE

```
ip address 10.70.84.10 255.255.255.0
 duplex full
 speed 100
!
interface ISM0/0
 no ip address
 shutdown
 service-module fail-open
!
interface GigabitEthernet0/1
 description WAN side of CUBE to Adtran #1
```

WAN IP Address of the CUBE

```
ip address 10.64.3.32 255.255.0.0
 duplex auto
 speed auto
!
!
ip default-gateway 10.64.1.1
ip forward-protocol nd
!
ip http server
ip http access-class 23
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip route 0.0.0.0 0.0.0.0 10.64.1.1
!
!
!
!
control-plane
!
!
voice-port 0/0/0
!
voice-port 0/0/1
!
voice-port 0/0/2
!
voice-port 0/0/3
 !
 !
 !
 !
!
!
no mgcp timer receive-rtcp
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
!
!
```

Incoming Dial Peer

```
dial-peer voice 100 voip
description Incoming dialpeer and outgoing NANP (1+10 digits) to Charter
huntstop
destination-pattern ^1[2-9]..[2-9]......$
translate-outgoing calling 1
session protocol sipv2
session target sip-server
incoming called-number .T
voice-class codec 1
dtmf-relay rtp-nte
fax protocol pass-through g711ulaw
no vad
!
```

Outgoing Dial Peer

```
dial-peer voice 300 voip
description Outgoing 303835XXXX calls to CUCM
huntstop
preference 1
destination-pattern 303835....
session protocol sipv2
session target ipv4:10.70.84.5
voice-class codec 1
dtmf-relay rtp-nte
no vad
!
```

Outgoing International Dial Peer

```
 dial-peer voice 102 voip
 description Outgoing International to Charter
 huntstop
 destination-pattern ^011T
 session protocol sipv2
 session target sip-server
 voice-class codec 1
!
```

User Agent IP Address for e-SBC

```
 sip-ua
 no remote-party-id
 retry invite 3
 timers trying 100
 mwi-server ipv4:10.64.1.42 expires 3600 port 5060 transport tcp
unsolicited
 sip-server ipv4:10.64.3.45
!
.
.
Omitted
.
.
end
```