



## Charter SIP Trunking:

# Cisco Unified Communications Manager 11.0.1 with Cisco Unified Border Element (CUBE 11.5.0) on ISR4321/K9 [IOS-XE 3.17.1 – 15.6(1)S1] using SIP

July 25, 2016



## Table of Contents

Introduction.....	4
Network Topology .....	5
System Components .....	6
<b>Hardware Requirements</b> .....	6
<b>Software Requirements</b> .....	6
Features.....	7
<b>Features Supported</b> .....	7
<b>Features Not Supported</b> .....	7
<b>Caveats</b> .....	7
<b>Configuration</b> .....	8
Configuring Cisco Unified Border Element.....	8
<b>Network Interface</b> .....	8
<b>Global Cisco UBE Settings</b> .....	9
<b>Codecs</b> .....	10
<b>Dial Peer</b> .....	10
<b>Call Flow</b> .....	13
<b>Configuration Example</b> .....	15
Configuring Cisco Unified Communications Manager.....	31
<b>Cisco UCM Version</b> .....	31
<b>Cisco Call Manager Service Parameters</b> .....	31
<b>Offnet Calls via Charter SIP Trunk</b> .....	32
<b>Dial Plan</b> .....	40
<b>Acronyms</b> .....	44
Important Information.....	45



## Table of Figures

Figure 1: Network Topology .....	5
Figure 2: Cisco UBE High Availability .....	5
Figure 3: Outbound Voice Call .....	13
Figure 4: Inbound Voice Call .....	13
Figure 5: Outbound Fax Call .....	14
Figure 6: Inbound Fax Call .....	14
Figure 7: PBX to PBX via Charter Call.....	14
Figure 8: Cisco UCM Version.....	31
Figure 9: Service Parameters.....	31
Figure 10: SIP Trunk Security Profile.....	32
Figure 11: SIP Profile.....	33
Figure 12: SIP Profile (Cont.) .....	34
Figure 13: SIP Profile (Cont.) .....	35
Figure 14: SIP Trunks List .....	36
Figure 15: SIP Trunk to Cisco UBE.....	37
Figure 16: SIP Trunk to Cisco UBE (Cont.).....	38
Figure 17: SIP Trunk to Cisco UBE (Cont.).....	39
Figure 18: Route Patterns List .....	40
Figure 19: Route Pattern for Voice.....	41
Figure 20: Route Pattern for Voice (Cont.).....	42
Figure 21: Route Pattern for Fax .....	43



## Introduction

Service Providers today, such as Charter, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and centralized IP to TDM POP gateways to provide on-net and off-net services.

Charter is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and Charter network, Cisco Unified Border Element (Cisco UBE 11.5.0) ISR 4321/K9 running IOS-XE 3.17.1 – 15.6(1) S1 can be used. The Cisco Unified Border Element provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.0.1 connected to Charter network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco UCM (Cisco Unified Communications Manager). Only configuration settings specifically required for Charter interoperability are presented. Feature configuration and most importantly the dial plan are customer specific and need individual approach.

- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 11.0.1 and Cisco Unified Border Element (Cisco UBE 11.5.0) on ISR 4321/K9 [IOS-XE 3.17.1 - 15.6(1)S1] for connectivity to Charter SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 11.0.1) to PSTN (Charter).
- Testing was performed in accordance to Charter generic SIP Trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), unattended and attended transfers, call forward, conferences and interoperability with Cisco Unity Connection (CUC)
- The Cisco UCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Charter SIP network and Cisco Unified Communications. The configuration described in this document details the important configuration settings to have enabled for interoperability to be successful and care must be taken by the network administrator deploying Cisco UCM to interoperate to Charter SIP Trunking network.

This application note does not cover the use of Calling Search Spaces (CSS) or partitions on Cisco UCM. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:

[http://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/srnd/collab10/collab10/dialplan.html](http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/dialplan.html)

## Network Topology

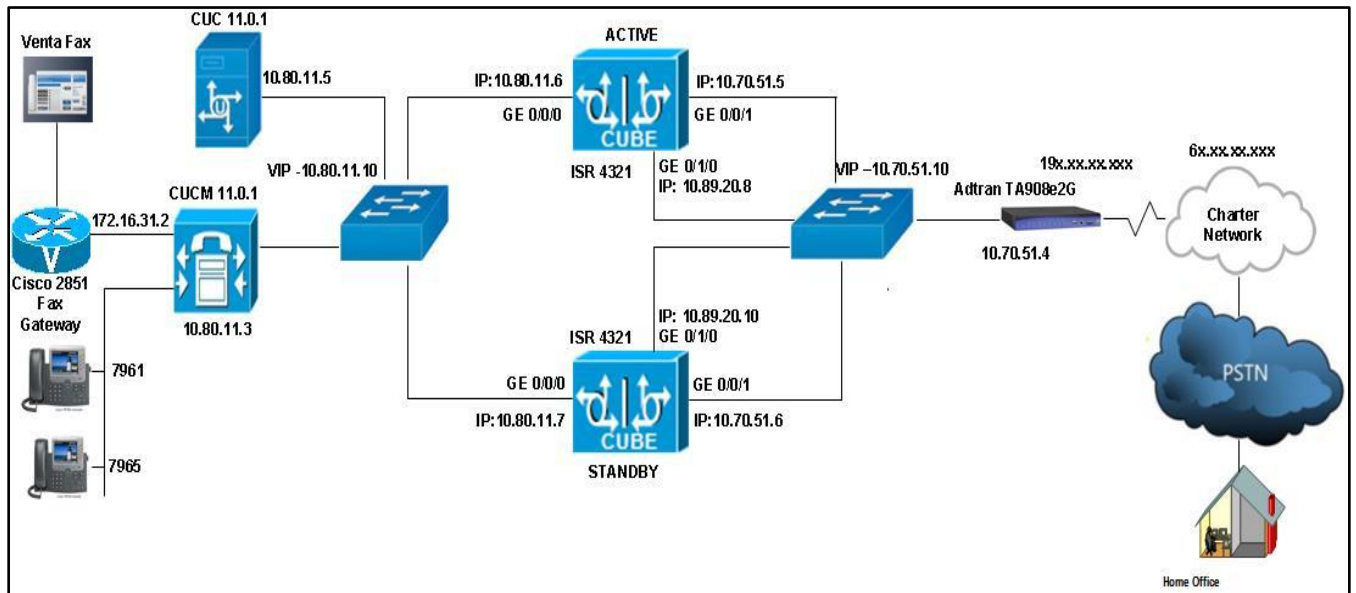


Figure 1: Network Topology

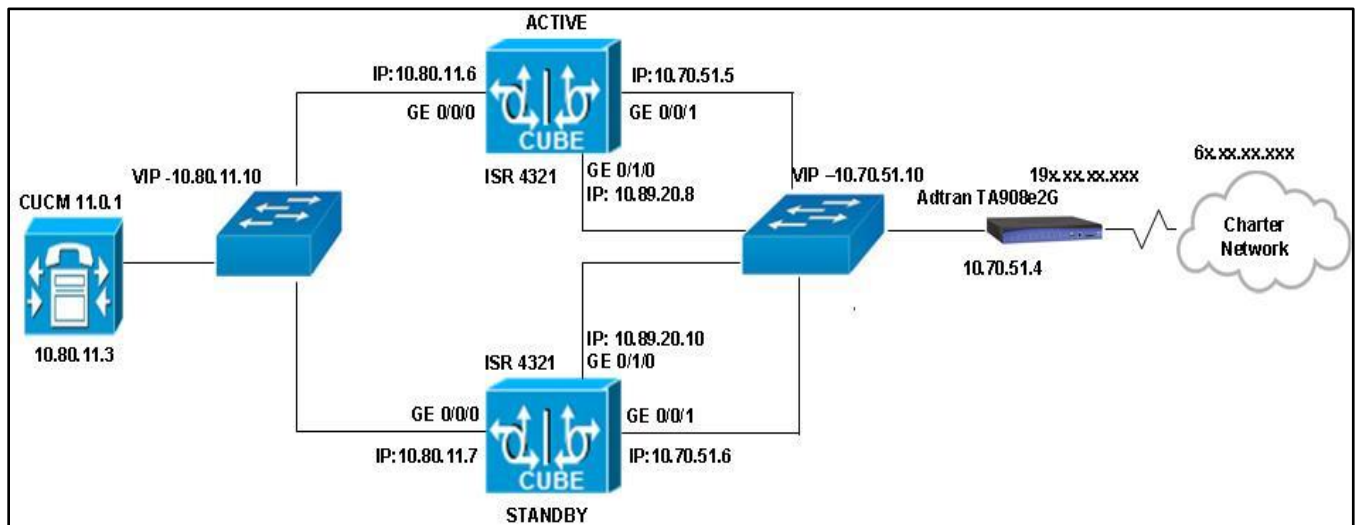


Figure 2: Cisco UBE High Availability



## System Components

### Hardware Requirements

- Cisco UCSC-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 router as CUBE
- Cisco ISR4321/K9 (1RU) processor with 1647061K/6147K bytes of memory with 3 Gigabit Ethernet interfaces
- Processor board ID FLM1925W0X2
- Cisco 2851 Fax Gateway
- IP phones 7961 (SIP) and 7965 (SCCP)
- Adtran Total\_Access\_908e\_2nd\_Gen) – Provided and managed by Charter

### Software Requirements

- Cisco Unified Communications Manager 11.0.1
- Cisco Unity Connection 11.0.1
- IOS-XE 3.17.1 - 15.6(1)S1 for ISR 4321/K9 Cisco Unified Border Element 11.5.0
- Cisco IOS Software, ISR Software (X86\_64\_LINUX\_IOSD-UNIVERSALK9-M), Version 15.6(1)S1, RELEASE SOFTWARE (fc3)
- Cisco IOS XE Software, Version 03.17.01.S
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway
- Adtran Total\_Access\_908e\_2nd\_Gen /R11.4.6.E - Provided and managed by Charter



## Features

### Features Supported

- Incoming and outgoing off-net calls using G711ULaw
- Call hold
- Call transfer (unattended and attended)
- Call conference
- Call forward (all, busy and no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF relay (both directions) (RFC2833)
- Media flow-through on Cisco UBE
- Fax (G.711 pass-through)

### Features Not Supported

- Cisco IP phones used in this test do not support blind transfer
- Fax (T.38) and G729 is not supported by Service Provider
- In HA redundancy mode the primary cube will not take over the primary/active role after a reboot/network outage

### Caveats

- Caller ID is not updated after attended or unattended transfers to off-net phones. This is due to a limitation on Cisco UBE and will be resolved in the next release. The issue does not impact the calls.
- For testing, 911 calls were terminated by Charter



## Configuration

### Configuring Cisco Unified Border Element

#### 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 - for LAN and WAN.

```
interface GigabitEthernet0/0/0
description Charter LAN MS4 1/0/7
ip address 10.80.11.6 255.255.255.0
media-type rj45
negotiation auto
no mop enabled
redundancy rii 1
redundancy group 1 ip 10.80.11.10 exclusive
!
interface GigabitEthernet0/0/1
description Charter WAN MS4 1/0/8
ip address 10.70.51.5 255.255.255.0
negotiation auto
no mop enabled
redundancy rii 2
redundancy group 1 ip 10.70.51.10 exclusive
!
```





## Global Cisco UBE Settings

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

```
voice service voip
ip address trusted list
ipv4 0.0.0.0 0.0.0.0
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol pass-through g711ulaw
sip
bind control source-interface GigabitEthernet0/0/0
bind media source-interface GigabitEthernet0/0/0
session refresh
asserted-id pai
privacy pstn
early-offer forced
no silent-discard untrusted
midcall-signaling passthru
g729 annexb-all!
```

### Explanation

Command	Description
allow-connections sip to sip	Allow IP2IP connections between two SIP call legs
fax protocol	Specifies the fax protocol
asserted-id	Specifies the type of privacy header in the outgoing SIP requests and response messages
early-offer forced	Enables SIP Delayed-Offer to Early-Offer globally
midcall-signaling passthru	Passes SIP messages from one IP leg to another IP leg

© 2016 Cisco Systems, Inc. All rights reserved.

Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on [cisco.com](http://cisco.com)



## Codecs

G711Ulaw is used as the preferred codec for this testing and changed the preferences according to the test plan description

```
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g729r8
```

## Dial Peer

Cisco UBE uses dial-peers to route the call accordingly based on the digits

```
dial-peer voice 200 voip
  description Outbound-from IP PBX to PSTN - WAN facing
  huntstop
  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 profiles 100
  voice-class sip options-keepalive
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-nte
  fax-relay ecm disable
  fax rate disable
  fax nsf 000000
  fax protocol pass-through g711ulaw
  no vad
  !
dial-peer voice 210 voip
  description outgoing call to Charter - LAN facing
```



```
huntstop
session protocol sipv2
session transport udp
incoming called-number .T
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 500 voip
description cube-dp incoming call from PSTN
huntstop
session protocol sipv2
session transport udp
incoming called-number 303835....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 100
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
```



```
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 100 voip
description Inbound-from PSTN to IP PBX - LAN facing
huntstop
destination-pattern 303835....
session protocol sipv2
session target ipv4:10.80.11.3:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
```



## Call Flow

In the sample configuration presented here, Cisco UCM is provisioned with four-digit directory numbers corresponding to the last four DID digits. No digit manipulation is performed on the Cisco UBE.

For incoming PSTN calls, the Cisco UBE presents the full ten-digit DID number to Cisco UCM. The Cisco UCM picks up the last 4 significant Digits configured under SIP Trunk and routes the call based on those 4 digits. Voice calls are routed to IP phones; Fax calls are routed via a 4-digit route pattern over a SIP trunk that terminates on the Fax Gateway and in turn to the VentaFax client connected to the Fax Gateway.

CPE callers make outbound PSTN calls by dialing a “8” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, Cisco fax Gateway sends to Cisco UCM the DID with leading access code “8”. A “8.@” route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the Cisco UBE for Voice call or Fax. For PBX to PBX via Charter, Caller dial 8 prefix followed by the target 1+10Digit DID no for that extension number, 8 was stripped and the 1 +10 digits number was send to Cisco UBE, Cisco UBE sends the full 1+ 10 digits DID under Dial Peer 200 and send to Charter network which will direct back to Cisco UBE and handled same as normal incoming PSTN call. For International calls same pattern 8.@ followed by 011, country code and calling no is used.

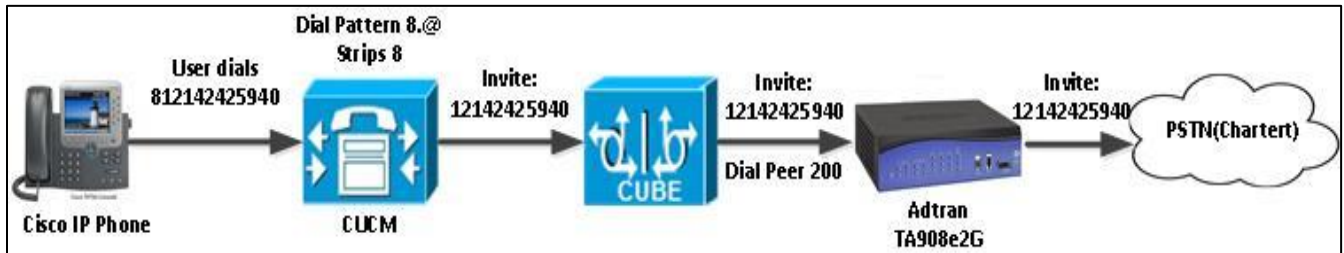


Figure 3: Outbound Voice Call

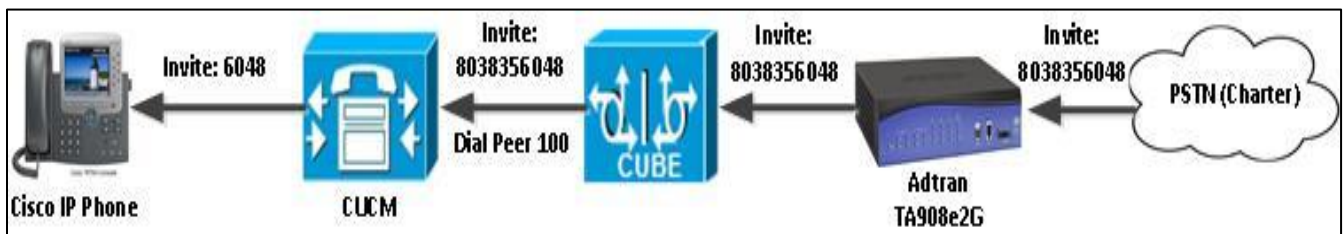


Figure 4: Inbound Voice Call

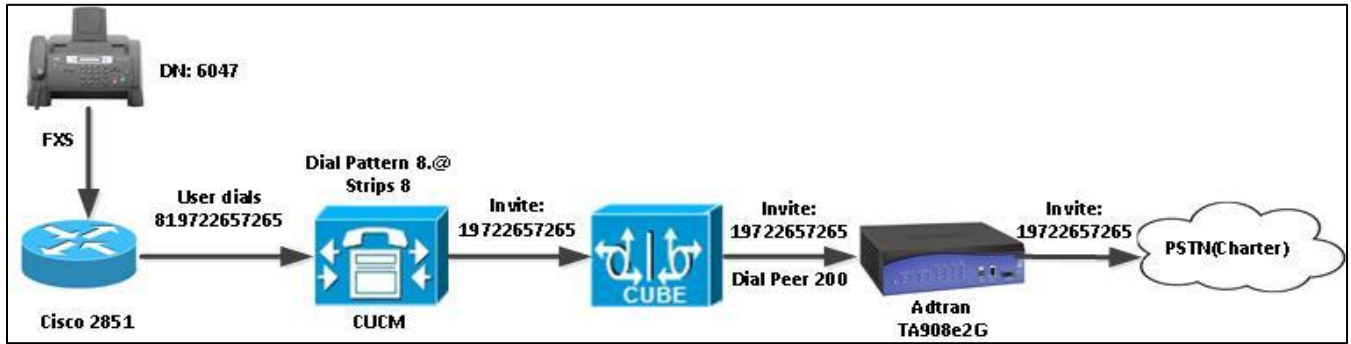


Figure 5: Outbound Fax Call

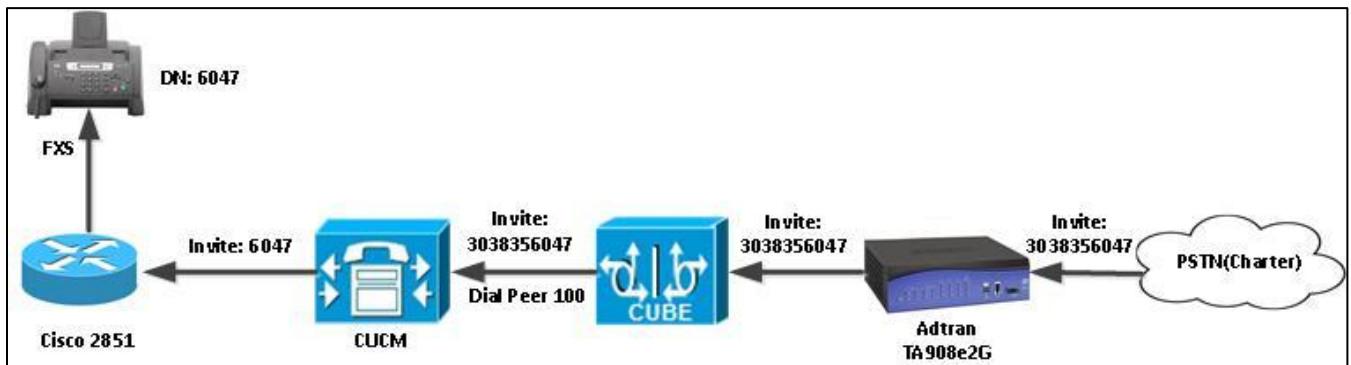


Figure 6: Inbound Fax Call

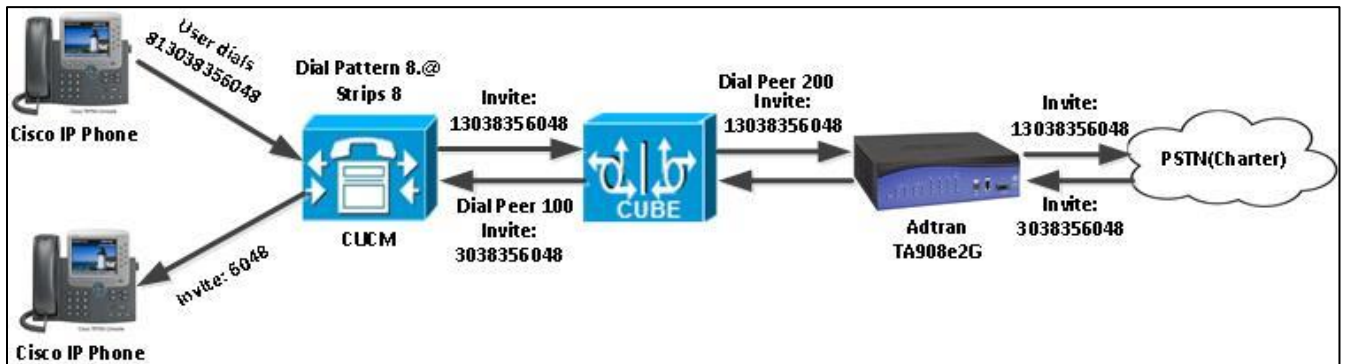


Figure 7: PBX to PBX via Charter Call



## Configuration Example

The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously

### *Active Cisco UBE*

```
version 15.6
service config
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keep alive disable-kernel-core
!
hostname CharterCube1
!
boot-start-marker
boot system bootflash:isr4300-universalk9.03.17.01.S.156-1.S1-std.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
vrf definition mgmt-intf
!
no logging rate-limit
no aaa new-model
```



```
no ip domain lookup
!
subscriber templating
multilink bundle-name authenticated
!
voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
fax protocol pass-through g711ulaw
sip
rel1xx supported "rel100"
session refresh
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
g729 annexb-all
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!
voice class sip-profiles 100
request INVITE sip-header Diversion modify "<sip:(.*)@(.)>" "<sip:303835\1@\2"
!
license udi pid ISR4321/K9 sn FDO19220MW3
license boot level appxk9
license boot level uck9
```





```
!  
spanning-tree extend system-id  
!  
redundancy  
mode none  
application redundancy  
group 1  
name b2bhaCharter  
priority 100 failover threshold 75  
timers delay 30 reload 60  
control GigabitEthernet0/1/0 protocol 1  
data GigabitEthernet0/1/0  
track 1 shutdown  
track 2 shutdown  
!  
vlan internal allocation policy ascending  
!  
track 1 interface GigabitEthernet0/0/0 line-protocol  
track 2 interface GigabitEthernet0/0/1 line-protocol  
!  
interface GigabitEthernet0/0/0  
description Charter LAN MS4 1/0/7  
ip address 10.80.11.6 255.255.255.0  
media-type rj45  
negotiation auto  
no mop enabled  
redundancy rii 1  
redundancy group 1 ip 10.80.11.10 exclusive  
!
```



```
interface GigabitEthernet0/0/1
description Charter WAN MS4 1/0/8
ip address 10.70.51.5 255.255.255.0
negotiation auto
no mop enabled
redundancy rii 2
redundancy group 1 ip 10.70.51.10 exclusive
!
interface GigabitEthernet0/1/0
description CUBE HA MS5 3/0/36
ip address 10.89.20.8 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto
no mop enabled
!
interface Vlan1
no ip address
shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0/0/0
ip route 10.64.0.0 255.255.0.0 10.80.11.1
ip route 10.70.0.0 255.255.0.0 10.70.51.1
ip route 10.80.11.0 255.255.255.0 10.80.11.1
ip route 172.16.0.0 255.255.0.0 10.80.11.1
```

© 2016 Cisco Systems, Inc. All rights reserved.

Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on [cisco.com](http://cisco.com)



```
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 200 voip
description Outbound-from IP PBX to PSTN - WAN facing
huntstop
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 profiles 100
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 210 voip
description outgoing call to Charter - LAN facing
```

© 2016 Cisco Systems, Inc. All rights reserved.

Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on [cisco.com](http://cisco.com)



```
huntstop
session protocol sipv2
session transport udp
incoming called-number .T
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 500 voip
description cube-dp incoming call from PSTN
huntstop
session protocol sipv2
session transport udp
incoming called-number 303835....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 100
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
```



```
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 100 voip
description Inbound-from PSTN to IP PBX - LAN facing
huntstop
destination-pattern 303835....
session protocol sipv2
session target ipv4:10.80.11.3:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
!
sip-ua
keepalive target ipv4:10.70.51.4:5060
timers keepalive active 180
sip-server ipv4:10.70.51.4:5060
!
!
```

```
line con 0
```



```
stopbits 1
line aux 0
stopbits 1
line vty 0 5
exec-timeout 0 0
password
login
!
!
end
```



## Standby Cisco UBE

**CharterCube2#sh running-config**

```
version 15.6
service config
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname CharterCube2
!
boot-start-marker
boot system flash bootflash:isr4300-universalk9.03.17.01.S.156-1.S1-std.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no logging rate-limit
no aaa new-model
no ip domain lookup
!
subscriber templating
!
multilink bundle-name authenticated
```



```
voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
fax protocol pass-through g711ulaw
sip
rel1xx supported "rel100"
session refresh
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
g729 annexb-all
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!
!
voice class sip-profiles 100
request INVITE sip-header Diversion modify "< sip:(.*)@(.*)>" "< sip:303835\1 @\2"
!
license udi pid ISR4321/K9 sn FDO19220MQ9
!
spanning-tree extend system-id
!
```





```
redundancy
mode none
application redundancy
group 1
  name b2bhaCharter
  priority 100 failover threshold 75
  timers delay 30 reload 60
  control GigabitEthernet0/1/0 protocol 1
  data GigabitEthernet0/1/0
  track 1 shutdown
  track 2 shutdown
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
  description Charter CUBE1 LAN MS4 1/0/11
  ip address 10.80.11.7 255.255.255.0
  media-type rj45
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.11.10 exclusive
!
interface GigabitEthernet0/0/1
  description Charter CUBE1 WAN MS4 1/0/12
  ip address 10.70.51.6 255.255.255.0
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 10.70.51.10 exclusive
```



```
!  
interface GigabitEthernet0/1/0  
  description CUBE HA MS5 3/0/38  
  ip address 10.89.20.10 255.255.255.0  
  negotiation auto  
!  
interface GigabitEthernet0  
  vrf forwarding Mgmt-intf  
  no ip address  
  negotiation auto  
!  
interface Vlan1  
  no ip address  
  shutdown  
!  
ip forward-protocol nd  
no ip http server  
no ip http secure-server  
ip tftp source-interface GigabitEthernet0/0/0  
ip route 0.0.0.0 0.0.0.0 10.70.51.1  
ip route 10.64.0.0 255.255.0.0 10.80.11.1  
ip route 10.80.11.0 255.255.255.0 10.80.11.1  
ip route 172.16.0.0 255.255.0.0 10.80.11.1  
!  
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 200 voip
description Outbound-from IP PBX to PSTN - WAN facing
huntstop
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 profiles 100
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 210 voip
description outgoing call to Charter - LAN facing
huntstop
session protocol sipv2
session target sip-server
session transport udp
incoming called-number .T
voice-class codec 1
```



```
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte

fax-relay ecm disable

fax rate disable

fax nsf 000000

fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 500 voip
description cube-dp incoming call from PSTN
huntstop

session protocol sipv2
session target sip-server
session transport udp
incoming called-number 303835....

voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 100
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte

fax-relay ecm disable

fax rate disable

fax nsf 000000

fax protocol pass-through g711ulaw
no vad
```



```
!  
dial-peer voice 100 voip  
description Inbound-from PSTN to IP PBX - LAN facing  
huntstop  
destination-pattern 303835....  
session protocol sipv2  
session target ipv4:10.80.11.3:5060  
session transport udp  
voice-class codec 1  
voice-class sip asserted-id pai  
voice-class sip options-keepalive  
voice-class sip bind control source-interface GigabitEthernet0/0/0  
voice-class sip bind media source-interface GigabitEthernet0/0/0  
dtmf-relay rtp-nte  
fax-relay ecm disable  
fax rate disable  
fax nsf 000000  
fax protocol pass-through g711ulaw  
no vad  
!  
!  
sip-ua  
keepalive target ipv4:10.70.51.4:5060  
timers keepalive active 180  
sip-server ipv4:10.70.51.4:5060  
!  
!  
line con 0  
stopbits 1  
line aux 0
```



```
stopbits 1
line vty 0 4
exec-timeout 0 0
password
login
!
!
End
```



## Configuring Cisco Unified Communications Manager Cisco UCM Version

The screenshot shows the Cisco Unified CM Administration web interface. At the top, there is a navigation bar with the Cisco logo and the text "Cisco Unified CM Administration For Cisco Unified Communications Solutions". A search bar contains "Cisco Unified CM Administration" and a "Go" button. Below the navigation bar, there are several menu items: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Help. A warning message states: "Smart Call Home is not configured. To configure Smart Call Home or disable the reminder, please go to Cisco Unified Serviceability > Call Home or Click here." Below the warning, the main heading is "Cisco Unified CM Administration". A red box highlights the "System version: 11.0.1.21900-11". Below this, it says "VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 80Gbytes, 4096Mbytes RAM, Partitions aligned". On the right side, there is a photograph of server racks in a data center.

Figure 8: Cisco UCM Version

### Cisco Call Manager Service Parameters

Navigation: System > Service Parameters

1. Select Server\* = Clus21Sub1--CUCM Voice/Video (Active)
2. Select Service\* = Cisco Call Manager (Active)
3. All other fields are set to default values

The screenshot shows the "Select Server and Service" configuration page. The "Server\*" dropdown is set to "clus21sub1--CUCM Voice/Video (Active)" and the "Service\*" dropdown is set to "Cisco CallManager (Active)". Below this, a note states: "All parameters apply only to the current server except parameters that are in the cluster-wide group(s)." The main section is titled "Cisco CallManager (Active) Parameters on server clus21sub1--CUCM Voice/Video (Active)". It contains a table with three columns: "Parameter Name", "Parameter Value", and "Suggested Value".

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

Figure 9: Service Parameters



## Offnet Calls via Charter SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and Charter Network and calls are routed via Cisco UBE

### SIP Trunk Security Profile

**Navigation:** System > Security > SIP Trunk Security Profile

1. Name\*= Charter Non Secure SIP Trunk Profile
2. Description = non Secure SIP Trunk Profile authenticated by null String

SIP Trunk Security Profile Information	
Name*	Charter Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null Strin
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 10: SIP Trunk Security Profile

#### Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to Charter SBC 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 Profile Configuration

SIP Profile will be later associated with the SIP trunk

**Navigation:** Device > Device Settings > SIP Profile

1. Name\*= Charter SIP Profile
2. Description = Default SIP Profile

SIP Profile Information	
Name*	Charter SIP Profile
Description	Default SIP Profile
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"/> Use Fully Qualified Domain Name in SIP Requests	
<input type="checkbox"/> Assured Services SIP conformance	

SDP Information	
SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
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

Figure 11: SIP Profile



DSCP for Audio Calls	Use System Default						
DSCP for Video Calls	Use System Default						
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						
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 12: SIP Profile (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

---

**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 13: SIP Profile (Cont.)

Explanation

Parameter	Value	Description
Default MTP Telephony Event Payload Type	101	RFC2833 DTMF payload type
SIP Rel1XX Options	Send PRACK for 1xx Messages	Enable Provisional Acknowledgements (Reliable 100 messages)
Ping Interval for In-service and Partially In-service Trunks (seconds)	60	OPTIONS message parameters- interval time
Ping Interval for Out-of-service Trunks (seconds)	120	OPTIONS message parameters- interval time



## SIP Trunk Configuration

Create SIP trunks to Cisco UBE

**Navigation:** Device > Trunk

Find and List Trunks													
Trunks (1 - 8 of 8) <span style="float: right;">Rows per Page 50 ▼</span>													
Find Trunks where Device Name ▼ begins with ▼ <input type="text"/> Find Clear Filter <input type="button" value="⊕"/> <input type="button" value="⊖"/>													
Select item or enter search text ▼													
<input type="checkbox"/>	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	
<input type="checkbox"/>	<a href="#">CUCM to FAXgateway</a>	SIP Trunk to FAX Gateway		<a href="#">G711 Pool</a>	6047				SIP Trunk	Full Service	Time In Full Service: 1 day 0 hour 52 minutes	<a href="#">Charter Non Secure SIP Trunk Profile</a>	
<input type="checkbox"/>	<a href="#">Charter</a>	SIP Trunk to Charter CUBE		<a href="#">G711 Pool</a>	8.@				SIP Trunk	Full Service	Time In Full Service: 0 day 3 hours 27 minutes	<a href="#">Charter Non Secure SIP Trunk Profile</a>	
<input type="checkbox"/>	<a href="#">Unity_Connection</a>	To VM		<a href="#">Default</a>	2302				SIP Trunk	Unknown - OPTIONS Ping not enabled		<a href="#">Unity_Connection_Trunk_Security_Profile</a>	

Figure 14: SIP Trunks List



<b>SIP Trunk Status</b>	
<b>Service Status:</b> Full Service	
<b>Duration:</b> Time In Full Service: 0 day 0 hour 3 minutes	
<b>Device Information</b>	
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	Charter
Description	SIP Trunk to Charter CUBE
Device Pool*	G711 Pool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_MTP
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
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> 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
<input checked="" type="checkbox"/> PSTN Access	
<input checked="" type="checkbox"/> Run On All Active Unified CM Nodes	

Figure 15: SIP Trunk to Cisco UBE



**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"/>

---

**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.

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 16: SIP Trunk to Cisco UBE (Cont.)



<b>Connected Party Settings</b>		
Connected Party Transformation CSS	< None >	
<input checked="" type="checkbox"/> Use Device Pool Connected Party Transformation CSS		
<b>Outbound Calls</b>		
Called Party Transformation CSS	< None >	
<input checked="" type="checkbox"/> Use Device Pool Called Party Transformation CSS		
Calling Party Transformation CSS	< None >	
<input checked="" type="checkbox"/> 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	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Outbound		
Redirecting Party Transformation CSS	< None >	
<input checked="" type="checkbox"/> Use Device Pool Redirecting Party Transformation CSS		
<b>Caller Information</b>		
Caller ID DN	<input type="text"/>	
Caller Name	<input type="text"/>	
<input type="checkbox"/> Maintain Original Caller ID DN and Caller Name in Identity Headers		
<b>SIP Information</b>		
<b>Destination</b>		
<input type="checkbox"/> Destination Address is an SRV		
Destination Address	Destination Address IPv6	Destination Port
1* 10.80.11.10		5060
MTP Preferred Originating Codec*	711ulaw	
BLF Presence Group*	Standard Presence group	
SIP Trunk Security Profile*	Charter 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*	Charter SIP Profile <a href="#">View Details</a>	
DTMF Signaling Method*	No Preference	
<b>Normalization Script</b>		
Normalization Script	< None >	
<input type="checkbox"/> Enable Trace		
	Parameter Name	Parameter Value
1	<input type="text"/>	<input type="text"/> <input type="button" value="+"/> <input type="button" value="-"/>
<b>Recording Information</b>		
<input checked="" type="radio"/> None		
<input type="radio"/> This trunk connects to a recording-enabled gateway		
<input type="radio"/> This trunk connects to other clusters with recording-enabled gateways		
<b>Geolocation Configuration</b>		
Geolocation	< None >	
Geolocation Filter	< None >	
<input type="checkbox"/> Send Geolocation Information		

Figure 17: SIP Trunk to Cisco UBE (Cont.)

© 2016 Cisco Systems, Inc. All rights reserved.

Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on [cisco.com](http://cisco.com)



## Explanation

Parameter	Value	Description
Device Name	Charter	Name for the trunk
Device Pool	G711pool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_MTP	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.11.10	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	Charter Non Secure SIP Trunk Profile	SIP Trunk Security Profile configured earlier
SIP Profile	Charter SIP Profile	SIP Profile configured earlier

## Dial Plan

### Route Pattern Configuration

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

Route patterns are configured as below:

- Cisco IP phone dial "8". 1+10 digits number to access PSTN via Cisco UBE
  - "8" is removed before sending to Cisco UBE
- For FAX call, Access Code "8"+ 1+10 digits number is used at Cisco Fax gateway
  - "8" is removed at Cisco UCM
  - The rest of the number is sent to Cisco UBE to Charter network
- Incoming fax call to 6047 will be sent to Cisco Fax gateway

Route Patterns (1 - 7 of 7)						Rows per Page 50
Pattern	Description	Partition	Route Filter	Associated Device	Copy	
<a href="#">2302</a>	Route pattern to unity			<a href="#">Unity_Connection</a>		
<a href="#">6047</a>	Route pattern to FAX Gateway			<a href="#">CUCM_to_FAXgateway</a>		
<a href="#">8.@</a>	PSTN calling			<a href="#">Charter</a>		

Figure 18: Route Patterns List





- Pattern Definition -		
Route Pattern*	8.@	
Route Partition	< None >	
Description	PSTN calling	
Numbering Plan*	NANP	
Route Filter	< None >	
MLPP Precedence*	Default	
<input type="checkbox"/> Apply Call Blocking Percentage		
Resource Priority Namespace Network Domain	< None >	
Route Class*	Default	
Gateway/Route List*	Charter	<a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="No Error"/>	
Call Classification*	OffNet	
External Call Control Profile	< None >	
<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending
<input type="checkbox"/> Urgent Priority		
<input type="checkbox"/> Require Forced Authorization Code		
Authorization Level*	<input type="text" value="0"/>	
<input type="checkbox"/> Require Client Matter Code		
- Calling Party Transformations -		
<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask		
Calling Party Transform Mask	<input type="text"/>	
Prefix Digits (Outgoing Calls)	<input type="text"/>	
Calling Line ID Presentation*	Default	
Calling Name Presentation*	Default	
Calling Party Number Type*	Cisco CallManager	
Calling Party Numbering Plan*	Cisco CallManager	
- Connected Party Transformations -		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
- Called Party Transformations -		
Discard Digits	PreDot	
Called Party Transform Mask	<input type="text"/>	
Prefix Digits (Outgoing Calls)	<input type="text"/>	
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
- ISDN Network-Specific Facilities Information Element -		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code	<input type="text"/>	
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 19: Route Pattern for Voice



- Pattern Definition -		
Route Pattern*	2302	
Route Partition	< None >	
Description	Route pattern to unity	
Numbering Plan	-- Not Selected --	
Route Filter	< None >	
MLPP Precedence*	Default	
<input type="checkbox"/> Apply Call Blocking Percentage		
Resource Priority Namespace Network Domain	< None >	
Route Class*	Default	
Gateway/Route List*	Unity_Connection	<a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern	No Error
Call Classification*	OffNet	
External Call Control Profile	< None >	
<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending
<input type="checkbox"/> Urgent Priority		
<input type="checkbox"/> Require Forced Authorization Code		
Authorization Level*	0	
<input type="checkbox"/> Require Client Matter Code		
- Calling Party Transformations -		
<input type="checkbox"/> Use Calling Party's External Phone Number Mask		
Calling Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Calling Line ID Presentation*	Default	
Calling Name Presentation*	Default	
Calling Party Number Type*	Cisco CallManager	
Calling Party Numbering Plan*	Cisco CallManager	
- Connected Party Transformations -		
Connected Line ID Presentation*	Default	
Connected Name Presentation*	Default	
- Called Party Transformations -		
Discard Digits	< None >	
Called Party Transform Mask		
Prefix Digits (Outgoing Calls)		
Called Party Number Type*	Cisco CallManager	
Called Party Numbering Plan*	Cisco CallManager	
- ISDN Network-Specific Facilities Information Element -		
Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 20: Route Pattern for Voice (Cont.)



<b>Pattern Definition</b>	
Route Pattern*	6047
Route Partition	< None >
Description	Route pattern to FAX Gateway
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
<input type="checkbox"/> Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	CUCM_to_FAXgateway <a href="#">(Edit)</a>
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="No Error"/>
Call Classification*	OffNet
External Call Control Profile	< None >
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	<input type="text" value="0"/>
<input type="checkbox"/> Require Client Matter Code	
<b>Calling Party Transformations</b>	
<input type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	<input type="text"/>
Prefix Digits (Outgoing Calls)	<input type="text"/>
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager
<b>Connected Party Transformations</b>	
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
<b>Connected Party Transformations</b>	
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
<b>Called Party Transformations</b>	
Discard Digits	< None >
Called Party Transform Mask	303835XXXX
Prefix Digits (Outgoing Calls)	<input type="text"/>
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager
<b>ISDN Network-Specific Facilities Information Element</b>	
Network Service Protocol	-- Not Selected --
Carrier Identification Code	<input type="text"/>
Network Service	-- Not Selected --
Service Parameter Name	< Not Exist >
Service Parameter Value	<input type="text"/>

Figure 21: Route Pattern for Fax



## Explanation

Setting	Value	Description
Route Pattern	8.@ for Voice & International Calls, 6047 for Fax Call and 2302 for Unity Connection	Specify appropriate Route Pattern
Gateway/Route List	Charter for Route Pattern 8. @, 6047 for SIP Trunk To Fax Gateway and 2032 for Unity Connection.	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 8.@	North American Numbering Plan
Call Classification	OffNet for Route Pattern 8.@, 6047 and 2032	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 8.@	Specifies how to modify digit before they are sent to Charter network

## Acronyms

Acronym	Definition
CPE	Customer Premise Equipment
Cisco UBE	Cisco Unified Border Element
Cisco UCM	Cisco Unified Communications Manager
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
ESBC	Enterprise Session Border Controller
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol



## Important Information

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS. IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.

