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
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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:

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!

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the type of privacy header in the outgoing SIP requests and response messages</td>
</tr>
<tr>
<td>early-offer forced</td>
<td>Enables SIP Delayed-Offer to Early-Offer globally</td>
</tr>
<tr>
<td>midcall-signaling passthru</td>
<td>Passes SIP messages from one IP leg to another IP leg</td>
</tr>
</tbody>
</table>
**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](image)

![Figure 4: Inbound Voice Call](image)
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**

```plaintext
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
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 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
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.S156-1.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 tgc-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 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

![Cisco UCM Version Screenshot](image)

**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

![Service Parameters Screenshot](image)

**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](image)

**Explanation**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Transport Type</td>
<td>TCP + UDP</td>
<td>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.</td>
</tr>
<tr>
<td>Outgoing Transport Type</td>
<td>UDP</td>
<td></td>
</tr>
</tbody>
</table>
**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

<table>
<thead>
<tr>
<th>SIP Profile Information</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Name</strong></td>
</tr>
<tr>
<td>Charter SIP Profile</td>
</tr>
<tr>
<td><strong>Description</strong></td>
</tr>
<tr>
<td>Default SIP Profile</td>
</tr>
<tr>
<td><strong>Default MTP Telephony Event Payload Type</strong></td>
</tr>
<tr>
<td>101</td>
</tr>
<tr>
<td><strong>Early Offer for G.Clear Calls</strong></td>
</tr>
<tr>
<td>Disabled</td>
</tr>
<tr>
<td><strong>User-Agent and Server header information</strong></td>
</tr>
<tr>
<td>Send Unified CM Version Information as User-Agent</td>
</tr>
<tr>
<td><strong>Version in User Agent and Server Header</strong></td>
</tr>
<tr>
<td>Major And Minor</td>
</tr>
<tr>
<td><strong>Dial String Interpretation</strong></td>
</tr>
<tr>
<td>Phone number consists of characters 0-9, *, #, and +</td>
</tr>
<tr>
<td><strong>Confidential Access Level Headers</strong></td>
</tr>
<tr>
<td>Disabled</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>SDP Information</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>SDP Session-level Bandwidth Modifier for Early Offer and Re-invites</strong></td>
</tr>
<tr>
<td>TIAS and AS</td>
</tr>
<tr>
<td><strong>SDP Transparency Profile</strong></td>
</tr>
<tr>
<td>Pass all unknown SDP attributes</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameters used in Phone</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Timer Invite Expires (seconds)</strong></td>
</tr>
<tr>
<td>180</td>
</tr>
<tr>
<td><strong>Timer Register Delta (seconds)</strong></td>
</tr>
<tr>
<td>5</td>
</tr>
<tr>
<td><strong>Timer Register Expires (seconds)</strong></td>
</tr>
<tr>
<td>3800</td>
</tr>
<tr>
<td><strong>Timer T1 (msec)</strong></td>
</tr>
<tr>
<td>550</td>
</tr>
<tr>
<td><strong>Timer T2 (msec)</strong></td>
</tr>
<tr>
<td>4000</td>
</tr>
<tr>
<td><strong>Retry INVITE</strong></td>
</tr>
<tr>
<td>6</td>
</tr>
<tr>
<td><strong>Retry Non-IN VITE</strong></td>
</tr>
<tr>
<td>10</td>
</tr>
<tr>
<td><strong>Media Port Ranges</strong></td>
</tr>
<tr>
<td>Common Port Range for Audio and Video</td>
</tr>
<tr>
<td>Separate Port Ranges for Audio and Video</td>
</tr>
<tr>
<td><strong>Start Media Port</strong></td>
</tr>
<tr>
<td>16384</td>
</tr>
<tr>
<td><strong>Stop Media Port</strong></td>
</tr>
<tr>
<td>32766</td>
</tr>
</tbody>
</table>

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**

<table>
<thead>
<tr>
<th>Call Pickup URI*</th>
<th>x-cisco-service-uri-pickup</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Pickup Group Other URI*</td>
<td>x-cisco-service-uri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group URI*</td>
<td>x-cisco-service-uri-pickup</td>
</tr>
<tr>
<td>Meet Me Service URI*</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info*</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level*</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back*</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block*</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking*</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control*</td>
<td>Off</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)*</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)*</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)*</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirections*</td>
<td>70</td>
</tr>
</tbody>
</table>

| Off Hook To First Digit Timer (milliseconds)* | 15000 |
| Call Forward URI* | x-cisco-serviceuri-cfwdall |
| Speed Dial (Abbreviated Dial) URI* | x-cisco-serviceuri-abbrev |

- **Conference Join Enabled**
- **RFC 2543 Hold**
- **Semi Attended Transfer**
- **Enable VAD**
- **Stutter Attended Transfer**
- **MLPP User Authorization**

#### Normalization Script

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

#### Incoming Requests FROM URI Settings

- **Caller ID DN**
- **Caller Name**

---

Figure 12: SIP Profile (Cont.)
Figure 13: SIP Profile (Cont.)

Explanation

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default MTP Telephony Event Payload Type</td>
<td>101</td>
<td>RFC2833 DTMF payload type</td>
</tr>
<tr>
<td>SIP Rel1XX Options</td>
<td>Send PRACK for 1xx Messages</td>
<td>Enable Provisional Acknowledgements (Reliable 100 messages)</td>
</tr>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)</td>
<td>60</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)</td>
<td>120</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
</tbody>
</table>
**SIP Trunk Configuration**
Create SIP trunks to Cisco UBE

**Navigation:** Device > Trunk

---

**Figure 14:** SIP Trunks List

<table>
<thead>
<tr>
<th>Device</th>
<th>Description</th>
<th>Route Pattern</th>
<th>Partition</th>
<th>Route Group</th>
<th>Priority</th>
<th>Trunk Type</th>
<th>SIP Trunk Status</th>
<th>SIP Trunk Destination</th>
<th>SIP Trunk Security Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>CUCM to NV Gateway</td>
<td>SIP Trunk to NV Gateway</td>
<td>6141 Pool</td>
<td>6141 Pool</td>
<td>SIP Trunk</td>
<td>Full Service</td>
<td>Time In Full Service: 1 day 0 hour 52 minutes</td>
<td>Charter Non Secure SIP Trunk Profile</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Charter</td>
<td>SIP Trunk to Charter CUBE</td>
<td>SIP Trunk</td>
<td>SIP Trunk</td>
<td>Full Service</td>
<td>Time In Full Service: 0 day 3 hours 27 minutes</td>
<td>Charter Non Secure SIP Trunk Profile</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unity Connection</td>
<td>To IV</td>
<td>Default</td>
<td>3032</td>
<td>SIP Trunk</td>
<td>Unknown - OPTIONS Ring not enabled</td>
<td>Unity Connection Trunk Security Profile</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
**Figure 15: SIP Trunk to Cisco UBE**
### Intercompany Media Engine (IME)

<table>
<thead>
<tr>
<th>E.164 Transformation Profile</th>
</tr>
</thead>
</table>

### MLPP and Confidential Access Level Information

<table>
<thead>
<tr>
<th>MLPP Domain</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>Confidential Access Mode</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>Confidential Access Level</th>
</tr>
</thead>
</table>

### Call Routing Information

- **Remote-Party-IID**
- **Asserted-Identity**
- **Asserted-Type**
  - Default
- **SIP Privacy**
  - Default

### Inbound Calls

<table>
<thead>
<tr>
<th>Significant Digits</th>
<th>4</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**Prefix ON**
- 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.

<table>
<thead>
<tr>
<th>Clear Prefix Settings</th>
<th>Default Prefix Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number Type</td>
<td>Prefix</td>
</tr>
<tr>
<td>Incoming Number</td>
<td>Default</td>
</tr>
</tbody>
</table>

### 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.

<table>
<thead>
<tr>
<th>Clear Prefix Settings</th>
<th>Default Prefix Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number Type</td>
<td>Prefix</td>
</tr>
<tr>
<td>Incoming Number</td>
<td>Default</td>
</tr>
</tbody>
</table>

Figure 16: SIP Trunk to Cisco UBE (Cont.)
Figure 17: SIP Trunk to Cisco UBE (Cont.)

<table>
<thead>
<tr>
<th>Connected Party Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Party Transformation CSS</td>
</tr>
<tr>
<td>Use Device Pool Connected Party Transformation CSS</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Outbound Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td>Called Party Transformation CSS</td>
</tr>
<tr>
<td>Use Device Pool Called Party Transformation CSS</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
</tr>
<tr>
<td>Calling Party Selection</td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
</tr>
<tr>
<td>Calling and Connected Party Info Format</td>
</tr>
<tr>
<td>Redirecting Diversion Header Delivery - Outbound</td>
</tr>
<tr>
<td>Redirecting Party Transformation CSS</td>
</tr>
<tr>
<td>Use Device Pool Redirecting Party Transformation CSS</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Caller Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller ID DN</td>
</tr>
<tr>
<td>Caller Name</td>
</tr>
<tr>
<td>Maintain Original Caller ID DN and Caller Name in Identity Headers</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>SIP Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Destination Address is an SRV</td>
</tr>
<tr>
<td>Destination Address</td>
</tr>
<tr>
<td>1*</td>
</tr>
</tbody>
</table>

| SIP Trunk Security Profile | Charter Non Secure SIP Trunk Profile |

| Redirecting Calling Search Space | < None > |
| Out-Of-Dialog Refer Calling Search Space | < None > |
| SUBSCRIBE Calling Search Space | < None > |

| SIP Profile | Charter SIP Profile |

| DTMF Signaling Method | No Preference |

<table>
<thead>
<tr>
<th>Normalization Script</th>
</tr>
</thead>
<tbody>
<tr>
<td>Normalization Script</td>
</tr>
<tr>
<td>Enable Trace</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Recording Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>None</td>
</tr>
<tr>
<td>This trunk connects to a recording-enabled gateway</td>
</tr>
<tr>
<td>This trunk connects to other clusters with recording-enabled gateways</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Geolocation Configuration</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geolocation</td>
</tr>
<tr>
<td>Geolocation Filter</td>
</tr>
<tr>
<td>Send Geolocation Information</td>
</tr>
</tbody>
</table>

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Explanation

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

**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 Table](image-url)

Figure 18: Route Patterns List
### Pattern Definition

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>PSTN calling</td>
</tr>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td>PSTN calling</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NAIP</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>MLPP Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Apply Call Blocking Percentage</td>
<td></td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Route Class</td>
<td>Default</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>Charter</td>
</tr>
</tbody>
</table>

### Calling Party Transformations

- **Use Calling Party’s External Phone Number Mask**
- **Calling Party Transform Mask**: PreDot
- **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

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Discard Digits</td>
<td>PreDot</td>
</tr>
<tr>
<td>Called Party Transform Mask</td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
<tr>
<td>Called Party Number Type</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Called Party Numbering Plan</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

### ISDN Network-Specific Facilities Information Element

<table>
<thead>
<tr>
<th>Network Service Protocol</th>
<th>-- Not Selected --</th>
</tr>
</thead>
<tbody>
<tr>
<td>Carrier Identification Code</td>
<td></td>
</tr>
<tr>
<td>Network Service</td>
<td>Service Parameter Name</td>
</tr>
<tr>
<td>-- Not Selected --</td>
<td>&lt; Not Exist &gt;</td>
</tr>
</tbody>
</table>

---

Figure 19: Route Pattern for Voice
Figure 20: Route Pattern for Voice (Cont.)
<table>
<thead>
<tr>
<th>Pattern Definition</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>6047</td>
</tr>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td>Route pattern to FX Gateway</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>&lt; Not Selected -- &gt;</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>MUPP Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Apply Call Blocking Percentage</td>
<td></td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Route Class</td>
<td>Default</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>CUCM_to_FXgateway</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
</tr>
<tr>
<td>Call Classification</td>
<td></td>
</tr>
<tr>
<td>External Call Control Profile</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**Calling Party Transformations**

| 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     | 303335XXXX |
| 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                | < Not Selected -- > |
| Service Parameter Name         | < Not Exist > |
| Service Parameter Value        |       |

**Figure 21: Route Pattern for Fax**
Explanation

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

Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>CPE</td>
<td>Customer Premise Equipment</td>
</tr>
<tr>
<td>Cisco UBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>MTP</td>
<td>Media Termination Point</td>
</tr>
<tr>
<td>POP</td>
<td>Point of Presence</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>ESBC</td>
<td>Enterprise Session Border Controller</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
</tbody>
</table>
Important Information

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