Qwest SIP Trunking: Connecting Cisco Unified Communications Manager 8.0(3) via the Cisco Unified Border Element 1.3 using SIP

April 29, 2011

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Note: Testing was conducted at Qwest VoIP CPE Littleton lab.
Introduction

Service Providers today, such as Qwest Communications, 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 a centralized IP to TDM gateway to provide on-net and off-net services. Qwest Communications SIP Trunking is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either Analog or T1 lines. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element provides demarcation, security, interworking and session management services.

- This application note describes how to configure a Cisco Unified Communications Manager (Cisco UCM) 8.0(3) with a Cisco Unified Border Element 1.3 (Cisco UBE) for connectivity to Qwest Communications SIP trunk service. The deployment model covered in this application note is CPE (Cisco UCM 8.0(3)/Cisco UBE) to PSTN via Qwest SIP Trunking. This document does not address 911 emergency outbound calls. For 911 feature service details contact Qwest Communications, directly.
- Testing was performed in accordance to Qwest SIP Trunk interoperability Test Plan and all features were verified. Key features verified are:
  - Basic Calls
  - Basic Calls with Calling Name and Number as allowed or restricted
  - DTMF Relay
  - Call Conference (Intra-site, PSTN)
  - Call Transfer (Blind, Attended, Early Attended)
  - Hold and Resume
  - Voice Mail
  - T.38 Fax
  - Simultaneous Calls
  - Auto Attendant
  - International Calls
  - G.711 Fax
  - Call Forwarding (Unconditional, Busy, No Reply)
  - Codec negotiation
  - Dial Plans
- The Cisco Unified Border Element configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Qwest Communications SIP network and Cisco Unified Communications. The configuration described in this document details the important commands to have enabled for interoperability to be successful and care must be taken, by the network administrator deploying Cisco UBE, to ensure these commands are set per each dial-peer requiring to interoperate to Qwest Communications SIP network.
- This application note does not cover the use of calling search spaces (CSS) or partitions on Cisco Unified Communications Manager. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:

Note: Testing was conducted at Qwest VoIP CPE Littleton lab.
**Network Topology**

Figure 1. Lab Network Topology

---

**System Components**

**Hardware Components**

- Cisco 2911 with VIC3 – 4FXS/DID (Used as Cisco Unified Border Element and MGCP gateway for FXS)
- Cisco Unified Communications Manager (Cisco MCS 7800 Series server)
- Cisco Unity Connection (Cisco MCS 7800 Series server)
- Cisco IP Phones (SCCP)

---

**Note:** Testing was conducted at Qwest VoIP CPE Littleton lab.
Software Requirements

- Cisco Unified Border Element Release 1.3 with IOS version 15.0(1)M3. This configuration was tested with c2900-universalk9-mz.SPA.150-1.M3.bin
- Cisco Unified Communications Manager Release 8.0(3). This solution was tested with 8.0.3.10000-8
- Cisco Unity Connection Release 8.0(3). This solution was tested with 8.0.3.10000-8

Note: Testing was conducted at Qwest VoIP CPE Littleton lab.
Features

Features Supported

- Voice calls using G.729 and G.711 codecs
- RFC 2833
- RFC 3261
- Calling number presentation / restriction
- Call conferencing
- Call transfer (attended and unattended)
- Call hold and resume
- Call forwarding
- T.38 fax
- G.711 pass-through fax

Features Not Supported

- SIP REFER
- MGCP based T.38 Fallback to pass-through

Note: Testing was conducted at Qwest VoIP CPE Littleton lab.
Caveats

- Although Qwest SIP Trunking supports SIP REFER the Cisco UCM will only accept REFER messages, it will not generate them.
- MGCP based fax calls set up as G.729 will not Fallback to pass-through.
Configuration

Configuring Cisco Unified Border Element

Critical commands are marked in **Bold** with footnotes at the bottom of the page

Version Information:

Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9-M), Version 15.0(1)M3, RELEASE SOFTWARE (fc2)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2010 by Cisco Systems, Inc.
Compiled Sun 18-Jul-10 03:32 by prod_rel_team

ROM: System Bootstrap, Version 15.0(1r)M6, RELEASE SOFTWARE (fc1)

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A summary of U.S. laws governing Cisco cryptographic products may be found at: http://www.cisco.com/wwl/export/crypto/tool/stqrg.html

If you require further assistance please contact us by sending email to export@cisco.com.

Cisco CISCO2911/K9 (revision 1.0) with 479232K/45056K bytes of memory.
Processor board ID FTX1435A0RR
3 Gigabit Ethernet interfaces
1 Serial interface
1 Channelized T1/PRI port
4 Voice FXS interfaces
DRAM configuration is 64 bits wide with parity enabled.
255K bytes of non-volatile configuration memory,
1970223K bytes of USB Flash usbflash0 (Read/Write)
250880K bytes of ATA System CompactFlash 0 (Read/Write)

Configuration register is 0x2102

Note: Testing was conducted at Qwest VoIP CPE Littleton lab.
**Running Configuration:**

Current configuration : 9182 bytes

! Last configuration change at 22:14:12 UTC Thu Apr 14 2011 by qwest

version 15.0
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption

hostname CCM_CUBE

boot-start-marker
boot-end-marker

card type t1 0 0
no logging queue-limit
logging buffered 2000000
no logging rate-limit
enable password 7 15031C09403E

no aaa new-model
no network-clock-participate wic 0

no ipv6 cef
ip source-route
ip cef

! ip dhcp excluded-address 10.1.1.2
ip dhcp excluded-address 10.1.1.3

! ip dhcp pool CiscoCM
  network 10.1.1.0 255.255.255.0
  default-router 10.1.1.1
  option 66 asci 10.1.1.2

! ip domain name yourdomain.com
multilink bundle-name authenticated

! crypto pki token default removal timeout 0

**Note:** Testing was conducted at Qwest VoIP CPE Littleton lab.
crypto pki trustpoint TP-self-signed-1820344965
enrollment selfsigned
subject-name cn=IOS-Self-Signed-Certificate-1820344965
revocation-check none
rsakeypair TP-self-signed-1820344965

crypto pki certificate chain TP-self-signed-1820344965
certificate self-signed 01
3082024F 308201B8 A0030201 02020101 300D0609 2A864886 F70D0101 04050003
31312F30 2D060355 04031326 494F532D 5365C66 2D536967 6E65642D 43657274
69666963 6174652D 31383230 33343439 36353001E 170D3131 30323137 31373336
35335A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03012A49
4F532D53 656C662D 5369676E 65642D43 75726164 65726174 31383230 33343439
36353031E 170D3131 30323137 31373336 35335A17 0D323030 31303130 30303030
305A3031 312F302D 06035504 03012A49
8100A0D8 21F0961B E40B5BA2 D3EC3DB5 1DC9C070 619EF9B3 2E6EFCC1 669A155F
475A5E57 A97A9A3D BCC38882 300D0AE65 899995EC 20C8D67A DFD1EE64 04B01DBD
446B8455 16C324D4 7604C3F0 D7C82295 7E2B26B5 1B8A76FB 5A384714 09EC65B0
5D7071FB EEF12B32 83F30387 835633F1 BA45059D BD99D6A7 5EE5BE15 6528D39A
04CD0203 010001A3 77307530 0F060355 1D310101 FF040530 030101FF 30220603
551D1104 1B301982 1743434D 5F435542 452E796F 7572646F 6D61696E 2E636F6D
301F0603 551D2304 18301680 143F4E3F 7A764FE1 75D76581 3D5C2D45 A19227C5
E1301D06 03551D0E 04160141 3F4EF37A 764FE175 D765815D 5C2D45A1 9227C5E1
300D0609 2A864886 F70D0101 04050003 818008F8 7A72275B 2766112 2A960B11
988BE4FF 98C8C02 721EDADA 142801F2 324C0330 6F7653B0 8689B11F 0A9803C7
07384408 E6B1D2 C4802EAF 6F34950F A8D9F26D C85F096C 7C855579 C5794AE3
FDEA7F9C 7D49F236 DA886764 F1A2FA2 6F650F46 4100CB3A 9C143990 0D413F72
17D7CA57 FE2100A7 5EA72AF7 04AD0386 FF6DAB

voice-card 0
dspfarm
dsp services dspfarm

voice service voip
address-hiding
allow-connections sip to sip
no supplementary-service sip moved-temporarily
fax protocol t38 ls-redundancy 0 hs-redundency 0 fallback none
no fax-relay sg3-to-g3
sip
bind control source-interface Loopback0
bind media source-interface Loopback0

midcall-signaling passthrough

voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g7298

*Required for MOH and offset transfer
voice class sip-profiles 1
request INVITE sip-header Diversion modify "[^ ]+(127@)" "<sip:6143568\1"
request INVITE sip-header Diversion modify "[^ ]+(1[3,4][1-9]@)" "<sip:7202839\1"

voice translation-rule 1
rule 1 /[^\s]+/ \1

voice translation-rule 3
rule 1 /[^\s]+/ \1

voice translation-rule 9
rule 2 /[^\s]+/ \1

voice translation-rule 10
rule 1 /[^\s]+/ \1

voice translation-profile DIGITSTRIP-1
 translate called 1

voice translation-profile DIGITSTRIP-9
 translate called 9

voice translation-profile Last10
 translate called 10

voice translation-profile Last3
 translate called 3

application
 license udi pid CISCO2911/K9 sn FTX1435A0RR
 hw-module pvdm 0/0
 username qwest password 7 0518165E331D5A
 username cisco privilege 15 secret 5 $1$kuyq$SnQxV4Kd3/zsik6/opnU/5/
 redundancy
 controller T1 0/0/0
 cablelength long 0db
 channel-group 0 timeslots 1-24

*Required for offnet call forwarding / single number reach if CUCM dial plan is less then 10 digits. The example is for a 3 digit dial plan

Note: Testing was conducted at Qwest VoIP CPE Littleton lab.
! interface Loopback0
   ip address 10.182.0.218 255.255.255.252
!
interface GigabitEthernet0/0
   no ip address
   duplex auto
   speed auto
!
interface GigabitEthernet0/0.100
   description LAN Switch
   encapsulation dot1Q 100
   ip address 10.1.1.1 255.255.255.0
   ip nat inside
   ip virtual-reassembly
!
interface GigabitEthernet0/1
   no ip address
   shutdown
   duplex auto
   speed auto
!
interface GigabitEthernet0/2
   no ip address
   shutdown
   duplex auto
   speed auto
!
interface Serial0/0/0:0
   description WAN
   ip address 65.113.26.242 255.255.255.252
   ip nat outside
   ip virtual-reassembly
   encapsulation ppp
!
   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 nat inside source list 199 interface Serial0/0/0:0 overload
   ip route 0.0.0.0 0.0.0.0 65.113.26.241
!
   access-list 199 permit ip 10.1.1.0 0.0.0.255 any
!
!
control-plane
!
!

Note: Testing was conducted at Qwest VoIP CPE Littleton lab.
voice-port 0/1/0
description Fax
timeouts ringing infinity
caller-id enable
!
voice-port 0/1/1
!
voice-port 0/1/2
!
voice-port 0/1/3
!
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager config server 10.1.1.2
!
mgcp
mgcp call-agent 10.1.1.2 service-type mgcp version 0.1
mgcp dmf-relay voip codec all mode nte-ca
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
mgcp package-capability fm-package
mgcp default-package fxr-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax rate 14400
mgcp fax t38 ls_redundancy 1
mgcp fax t38 hs_redundancy 1
no mgcp fax-relay sg3-to-g3
mgcp bind control source-interface GigabitEthernet0/0.100
mgcp bind media source-interface GigabitEthernet0/0.100
!
mgcp profile default
!
sccp local GigabitEthernet0/0.100
sccp ccm 10.1.1.2 identifier 1 version 7.0
sccp
!
sccp ccm group 1
  bind interface GigabitEthernet0/0.100
  associate ccm 1 priority 1
  associate profile 1 register mtpf866f25ff9e0
!
dspfarm profile 1 transcode
  codec g711ulaw
  codec g729r8
  codec g729abr8
  maximum sessions 6
  associate application SCCP
!
dial-peer voice 101 voip
  preference 1
  destination-pattern ^72028391[3-4][1-9]
  session protocol sipv2
  session target ipv4:10.1.1.2

Note: Testing was conducted at Qwest VoIP CPE Littleton lab.
session transport udp
dtmf-relay rtp-nte
no vad
!
dial-peer voice 201 voip
  translation-profile outgoing Last10
  preference 1
  destination-pattern ^1[1-9]........
  session protocol sipv2
  session target ipv4:10.182.0.24
  session transport udp
  voice-class sip profiles 1
dtmf-relay rtp-nte
  no vad
!
dial-peer voice 102 voip
  preference 1
  destination-pattern ^6143568127
  session protocol sipv2
  session target ipv4:10.1.1.2
  session transport udp
  voice-class codec 1
dtmf-relay rtp-nte
  no vad
!
dial-peer voice 202 voip
  translation-profile outgoing Last10
  preference 2
  destination-pattern ^1[1-9]........
  session protocol sipv2
  session target ipv4:10.182.3.24
  session transport udp
  voice-class sip profiles 1
dtmf-relay rtp-nte
  no vad
!
dial-peer voice 203 voip
  translation-profile outgoing Last3
  preference 1
  destination-pattern ^1[4,7,9]11
  session protocol sipv2
  session target ipv4:10.182.0.24
  session transport udp
  voice-class codec 1
  voice-class sip profiles 1
dtmf-relay rtp-nte
  no vad
!
dial-peer voice 301 voip
  translation-profile outgoing DIGITSTRIP-1
  preference 1
  destination-pattern ^1011.+
  session protocol sipv2
  session target ipv4:10.182.0.24
  session transport udp
  voice-class codec 1

Note:  Testing was conducted at Qwest VoIP CPE Littleton lab.
voice-class sip profiles 1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 302 voip
translation-profile outgoing DIGITSTRIP-1
preference 2
destination-pattern ^1011.+
session protocol sipv2
session target ipv4:10.182.3.24
session transport udp
voice-class codec 1
voice-class sip profiles 1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 204 voip
translation-profile outgoing Last3
preference 2
destination-pattern ^1[4,7,9]11
session protocol sipv2
session target ipv4:10.182.3.24
session transport udp
voice-class codec 1
voice-class sip profiles 1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 10 pots
preference 1
service mgcpapp
port 0/1/0
!
!
gateway
timer receive-rtp 1200
!

sip-ua
   retry invite 3
!
!
!
!
gatekeeper
shutdown
!
!
line con 0
session-timeout 60
exec-timeout 30 0
session-disconnect-warning 59
login local
line aux 0
line vty 0 4
access-class 23 in
exec-timeout 30 0
privilege level 15
login local
transport input telnet ssh
line vty 5 15
access-class 23 in
privilege level 15
login local
transport input telnet ssh
!
scheduler allocate 20000 1000

Note: Testing was conducted at Qwest VoIP CPE Littleton lab.
Configuring the Cisco Unified Communications Manager

CUCM Version

Note: Testing was conducted at Qwest VoIP CPE Littleton lab.
**Route Plan Report**

**Note:** Testing was conducted at Qwest VoIP CPE Littleton lab.
## Route Patterns

![Cisco Unified CM Administration](image)

**Note:** Testing was conducted at Qwest VoIP CPE Littleton lab.
Route Pattern 9*67.@

<table>
<thead>
<tr>
<th>Pattern Definition</th>
<th>Description</th>
<th>Numbering Plan</th>
<th>Route Filter</th>
<th>HLSPP Precedence</th>
<th>Resource Priority Namespace Network Domain</th>
<th>Route Class</th>
<th>Gateway/Route List</th>
<th>Route Option</th>
<th>Call Classification</th>
<th>Allow Device Override</th>
<th>Provide Outside Dial Tone</th>
<th>Allow Overlap Sending</th>
<th>Urgent Priority</th>
<th>Require Forced Authorization Code</th>
<th>Authorization Level</th>
<th>Require Client Caller Code</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>9*67.@</td>
<td>Generated_Everyone</td>
<td>None</td>
<td>Default</td>
<td>None</td>
<td>Default</td>
<td>c9311_CUBE</td>
<td>Route this pattern</td>
<td>Offset</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
</tr>
<tr>
<td>Description</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Block this pattern</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>No</td>
<td>Yes</td>
</tr>
</tbody>
</table>

**Calling Party Transformations**
- Use Calling Party's External Phone Number Mask
- Calling Party Transform Mask
- Prefix Digits (Outgoing Calls)

- Calling Party Presentation: Restricted
- Connected Line ID Presentation: Default
- Connected Name Presentation: Default

**Called Party Transformations**
- Called Party Presentation: Restricted
- 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

**Note:** Testing was conducted at Qwest VoIP CPE Littleton lab.
Route Pattern 9.@

<table>
<thead>
<tr>
<th>Pattern Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
</tr>
<tr>
<td>Description</td>
</tr>
<tr>
<td>Numbering Plan</td>
</tr>
<tr>
<td>Route Filter</td>
</tr>
<tr>
<td>NLSP Precedence</td>
</tr>
<tr>
<td>Resource Priority</td>
</tr>
<tr>
<td>Route Class</td>
</tr>
<tr>
<td>Gateway/Route List</td>
</tr>
<tr>
<td>Route Option</td>
</tr>
</tbody>
</table>

**Note:**
Testing was conducted at Qwest VoIP CPE Littleton lab.
MGCP Gateway

**Gateway Configuration**

- **Product**: Cisco 2911
- **Protocol**: MGCP
- **Domain Name**: CCN_CUBE.yourdomain.com
- **Description**: 3911-LSE
- **Cisco Unified Communications Manager Group**: Default

**Configured Slots, VICs and Endpoints**

- **Module in Slot 0**: VWIC-MBDC
  - **Submit 0**: <None>
  - **Submit 1**: VIC-MFXG/OID
  - **Submit 2**: <None>
  - **Submit 3**: <None>

**Product Specific Configuration Layout**

- **Global ISON Switch Type**: 4505
- **Switchback Timing**
- **Switchback uptime-delay (min)**: 10
- **Switchback schedule (hh:mm)**: 12:00
- **Type Of DTMF Relay**: NTE-C4
- **Network Port Through**: Enable
- **Cisco Fax Relay**: Disable
- **T30 Fax Relay**: Enable
- **RTT Package Capability**: Enable
- **MT Package Capability**: Disable
- **RES Package Capability**: Disable
- **PRE Package Capability**: Enable
- **SST Package Capability**: Enable
- **RTP Unreachable OnOff**: Enable
- **RTP Unreachable timeout (ms)**: 1000
- **RTCP Report Interval (sec)**: 0
- **Simple SDR**: Enable

---

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**Note**: Testing was conducted at Qwest VoIP CPE Littleton lab.
Note: Testing was conducted at Qwest VoIP CPE Littleton lab.
### SIP Trunk

#### Status

- Status: Ready

#### Device Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Product</td>
<td>SIP Trunk</td>
</tr>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>None (Default)</td>
</tr>
<tr>
<td>Device Name</td>
<td>c9511_CUBE</td>
</tr>
<tr>
<td>Description</td>
<td>SIP Trunk</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>SIP Trunk</td>
</tr>
<tr>
<td>Call Classification</td>
<td>None &gt;</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>None &gt;</td>
</tr>
<tr>
<td>Location*</td>
<td>None &gt;</td>
</tr>
<tr>
<td>AAA Group</td>
<td>None &gt;</td>
</tr>
<tr>
<td>Packet Capture Mode*</td>
<td>None &gt;</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>None &gt;</td>
</tr>
</tbody>
</table>

- Media Termination Point Required: No
- Retry Video Call as Audio: No
- Transmit UDN for Calling Party Name: No
- Unattended Port: No
- RTP Allowed: No
- RTP Class Signaling Enabled*: Default
- Use Trusted Relay Point*: Default
- PSTN Access: No

#### Intercompany Media Engine (IME)

- E.164 Transformation Profile: None

#### Multilevel Precedence and Preemption (MPLP) Information

- MPLP Domain: None

#### Call Routing Information

- Remote-Party-Id: Default
- Asserted-Identity: Default
- Asserted-Type*: Default
- SIP Privacy*: Default

#### Inbound Calls

- Significant Digits*: 0
- Connected Line ID Presentation*: Default
- Connected Name Presentation*: Default
- Calling Search Space: Generated_CSS_1
- AAR Calling Search Space: None
- Prefix CN: None

- Redirecting Diversion Header Delivery - Inbound: Yes

#### Incoming Calling Party Settings

- Prefix: None
- Strip Digits: None
- Calling Search Space: Default
- Use Device Pool CSS: Yes

#### Connected Party Settings

- Connected Party Transformation CSS: None
- Use Device Pool Connected Party Transformation CSS: Yes

---

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**Note:** Testing was conducted at Qwest VoIP CPE Littleton lab.
**Outbound Calls**

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Called Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Party Selection</td>
<td>Originator</td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Caller ID DN</td>
<td></td>
</tr>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
</tbody>
</table>

Note: Testing was conducted at Qwest VoIP CPE Littleton lab.

**SEPI Information**

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Destination Address</td>
<td>10.182.0.218</td>
</tr>
<tr>
<td>Destination Address IPv6</td>
<td></td>
</tr>
<tr>
<td>Destination Port</td>
<td>5060</td>
</tr>
<tr>
<td>MIP Preferred Originating Codec</td>
<td>G29.0/G729e</td>
</tr>
<tr>
<td>Presence Group</td>
<td>Standard Presence group</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>Remote Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Out-Of-Calling Refer Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>SIP Trunk IP Profile</td>
</tr>
<tr>
<td>DTMF Signing Method</td>
<td>SPC 2833</td>
</tr>
</tbody>
</table>

**Geolocation Configuration**

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Geolocation Filter</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Send Geolocation Information</td>
<td></td>
</tr>
</tbody>
</table>
### SIP Profile

#### SIP Profile Information

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Default SIP Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP Trunk SP Profile</td>
<td>Default SIP Profile</td>
<td></td>
</tr>
</tbody>
</table>

- **Default RTP Telephony Event Payload Types**: 101
- **Resource Priority Namespace List**: `<Note>`
- **Early Offer for O/Clear Cell**
  - Enabled: **Disabled**

- **Redirect by Application**
- **Disable Early Media on 180**
- **Outgoing T.38 INVITE include audio wire**
- **Enable AMI**
- **Require SDP Inactive Exchange for Mid-Call Media Change**

#### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)*</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)*</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)*</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msco)*</td>
<td>900</td>
</tr>
<tr>
<td>Timer T2 (msco)*</td>
<td>400</td>
</tr>
<tr>
<td>Retry INVITE*</td>
<td>2</td>
</tr>
<tr>
<td>Retry Non-INVITE*</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port*</td>
<td>14334</td>
</tr>
<tr>
<td>Stop Media Port*</td>
<td>12746</td>
</tr>
<tr>
<td>Call Pickup URL*</td>
<td>x-cisco-service-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URL*</td>
<td>x-cisco-service-pickup</td>
</tr>
<tr>
<td>Call Pickup Group URL*</td>
<td>x-cisco-service-pickup</td>
</tr>
<tr>
<td>Next Me Service URL*</td>
<td>x-cisco-service-meetme</td>
</tr>
<tr>
<td>User Info*</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level*</td>
<td>Normal</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>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960*</td>
<td>Disabled</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>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)*</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URL*</td>
<td>x-cisco-service-confall</td>
</tr>
<tr>
<td>Abbreviated Dial URL*</td>
<td>x-cisco-service-abbrev</td>
</tr>
</tbody>
</table>

- **Conference Join Enabled**
- **RFC 2833 hold**
- **Semi Attended Transfer**
- **Enable VAD**
- **Skinner Message Waiting**

#### Trunk Specific Configuration

<table>
<thead>
<tr>
<th>Option</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Incoming Request to new Trunk based on</td>
<td>Never</td>
</tr>
<tr>
<td>RSVP Over SDP*</td>
<td>Local RSVP</td>
</tr>
<tr>
<td>Final back to local RSVP</td>
<td></td>
</tr>
<tr>
<td>SIP RELAY Options*</td>
<td>Send PRACK if 1xx contains SDP</td>
</tr>
</tbody>
</table>

**Note:** Testing was conducted at Qwest VoIP CPE Littleton lab.
MGCP Gateway config (test setup was on the shared 2911)

ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager config server 10.1.1.2
!
mgcp
mgcp call-agent 10.1.1.2 service-type mgcp version 0.1
mgcp dtmf-relay voice codec all mode nte-ca
mgcp package-capability rtp-package
mgcp package-capability sst-package
mgcp package-capability pre-package
mgcp package-capability fm-package
mgcp default-package fxr-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax rate 14400
mgcp fax t38 ls_redundancy 1
mgcp fax t38 hs_redundancy 1
no mgcp fax-relay sg3-to-g3
mgcp bind control source-interface GigabitEthernet0/0.100
mgcp bind media source-interface GigabitEthernet0/0.100
!
mgcp profile default

Note: Testing was conducted at Qwest VoIP CPE Littleton lab.
## Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>MGCP</td>
<td>Media Gateway Control Protocol</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>Cisco UBE</td>
<td>Cisco Unified Border Element</td>
</tr>
</tbody>
</table>

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