CenturyLink SIP Trunking: Connecting Cisco Unified Communications Manager 8.6.1 via the Cisco Unified Border Element 8.6 using SIP

March 22, 2013

Table of Contents
Introduction ........................................................................................................................................... 2
Network Topology .................................................................................................................................. 3
System Components .............................................................................................................................. 4
    Hardware Components ...................................................................................................................... 4
    Software Requirements ................................................................................................................... 4
Features .................................................................................................................................................. 4
    Features Supported .......................................................................................................................... 4
    Features Not Supported .................................................................................................................. 4
Caveats .................................................................................................................................................. 4
CUBE Configuration ............................................................................................................................ 5
    Version Information: ...................................................................................................................... 5
    Running Configuration: .................................................................................................................. 6
Cisco Unified Communications Manager Configuration ................................................................. 14
    CUCM Version ............................................................................................................................... 14
    Route Plan Report ......................................................................................................................... 15
    Route Patterns ............................................................................................................................... 16
        Route Pattern 0 .......................................................................................................................... 17
        Route Pattern 00 ....................................................................................................................... 18
        Route Pattern 9*67.@ ............................................................................................................... 19
        Route Pattern 9.00X+ ............................................................................................................. 20
        Route Pattern 9.5551212 ....................................................................................................... 21
        Route Pattern 9.@ ..................................................................................................................... 22
        Route Pattern 9.X11 .................................................................................................................. 23
        Route Pattern 9.XXX5551212 ................................................................................................. 24
MGCP Gateway ...................................................................................................................................... 25
    MGCP FXS Port ............................................................................................................................. 26
SIP Trunk ................................................................................................................................................ 27
SIP Profile .............................................................................................................................................. 29
Acronyms .............................................................................................................................................. 31
Important Information .......................................................................................................................... 32

© 2010 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
Page 1 of 33
EDCS# 1045727 Rev 2

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

Service Providers today, such as CenturyLink 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. CenturyLink 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) 8.6.1 with a Cisco Unified Border Element 8.6 (Cisco UBE) for connectivity to CenturyLink Communications SIP trunk service. The deployment model covered in this application note is CPE (Cisco UCM 8.6.1/Cisco UBE) to PSTN via CenturyLink SIP Trunking. This document does not address 911 emergency outbound calls. For 911 feature service details contact CenturyLink Communications, directly.
- Testing was performed in accordance to CenturyLink 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 – RFC 2833
  - Call Conference (Intra-site, PSTN)
  - Call Transfer (Blind, Attended, Early Attended)
  - Hold and Resume
  - Voice Mail
  - T.38 Fax
  - Simultaneous Calls
  - International Calls
  - 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 CenturyLink 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 CenturyLink 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 CenturyLink VoIP CPE Littleton lab.
Network Topology

Figure 1. Lab Network Topology

© 2010 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
Page 3 of 33
EDCS# 1045727 Rev 2

Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
System Components

Hardware Components
- Cisco 2911 with VIC – 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)

Software Requirements
- Cisco Unified Border Element Release 8.6 with IOS version 15.1(3)T1. This configuration was tested with flash:c2900-universalk9-mz.SPA.151-3.T1.bin
- Cisco Unified Communications Manager Release 8.6.1. This solution was tested with 8.6.1.20000-1
- Cisco Unity Connection Release 8.6.1. This solution was tested with 8.0.3.10000-8

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

Features Not Supported
- SIP REFER
- MGCP based T.38 Fallback to pass-through

Caveats
- Although CenturyLink 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.
- Could not configure the CUCM MGCP gateway to remove T38, therefore no G711 fax passthru testing was performed.

Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
CUBE Configuration

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

**Version Information:**

Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9-M), Version 15.1(3)T1, RELEASE SOFTWARE (fc2)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2011 by Cisco Systems, Inc.
Compiled Sun 27-Mar-11 07:36 by prod_rel_team

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

CCM_CUBE uptime is 1 week, 3 days, 17 hours, 47 minutes
System returned to ROM by reload at 20:21:45 UTC Thu Sep 8 2011
System image file is "flash:c2900-universalk9-mz.SPA.151-3.T1.bin"
Last reload type: Normal Reload
Last reload reason: Reload Command

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:

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.
250880K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:

License UDI:

<table>
<thead>
<tr>
<th>Device#</th>
<th>PID</th>
<th>SN</th>
</tr>
</thead>
</table>

© 2010 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
Page 5 of 33
EDCS# 1045727 Rev 2

**Note:** Testing was conducted at CenturyLink VoIP CPE Littleton lab.
Technology Package License Information for Module:`c2900`

<table>
<thead>
<tr>
<th>Technology</th>
<th>Technology-package</th>
<th>Current</th>
<th>Type</th>
<th>Technology-package</th>
<th>Next reboot</th>
</tr>
</thead>
<tbody>
<tr>
<td>ipbase</td>
<td>ipbasek9</td>
<td>Permanent</td>
<td>ipbasek9</td>
<td></td>
<td></td>
</tr>
<tr>
<td>security</td>
<td>None</td>
<td>None</td>
<td>None</td>
<td>None</td>
<td></td>
</tr>
<tr>
<td>uc</td>
<td>uck9</td>
<td>Permanent</td>
<td>uck9</td>
<td></td>
<td></td>
</tr>
<tr>
<td>data</td>
<td>None</td>
<td>None</td>
<td>None</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Configuration register is 0x2102

Running Configuration:

Current configuration : 9182 bytes

!
! Last configuration change at 22:14:12 UTC Thu Apr 14 2011
!
version 15.1
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 domain name yourdomain.com
multilink bundle-name authenticated
!
crypto pki token default removal timeout 0
!

© 2010 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
Page 6 of 33
EDCS# 1045727 Rev 2

Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
Testing was conducted at CenturyLink VoIP CPE Littleton lab.
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 /^1(.*$/ \A1/

voice translation-rule 3
rule 1 /.*,\(\.*\)/ \A1/

voice translation-rule 9
rule 2 /^9(.*$/ \A1/

voice translation-rule 10
rule 1 /.*,\(.........\)/ \A1/

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 CenturyLink password 7 0518165E331D5A
username cisco privilege 15 secret 5 $1$kuyq$nQxV4Kd3/zsik6/opnU/5/

redundancy

controller T1 0/0/0
  cablelength long 0db
  channel-group 0 timeslots 1-24

interface Loopback0

\^ Required for offnet call forwarding / single number reach if Cisco UCM dial plan is < 10 digits. The example is for a 3 digit dial plan

Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
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 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 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
!
control-plane
!
!
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

Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
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 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
!
!
dial-peer voice 101 voip
preference 1
destination-pattern ^7202839[3-4][1-9]
session protocol sipv2
session target ipv4:10.1.1.2
session transport udp
dtmf-relay rtp-nte
no vad
!
!
!
!
dial-peer voice 201 voip
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
dtmf-relay rtp-nte
no vad
!
dial-peer voice 301 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 202 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 sip profiles 1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 203 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 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 sip profiles 1
dtmf-relay rtp-nte
no vad
!
dial-peer voice 303 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 sip profiles 1
dtmf-relay rtp-nte
no vad
!

dial-peer voice 204 voip
Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
retry invite 36
!
!
!
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

6 Required for SIP Trunk diversity / fail over to alternate SIP trunk on time outs

Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
### Route Plan Report

#### Find and List Route Plan Report

- **Status:** 24 records found

#### Find and List Route Plan Report (1 - 24 of 24)

<table>
<thead>
<tr>
<th>Pattern/Directory Number</th>
<th>Pattern</th>
<th>Type</th>
<th>Route Detail</th>
</tr>
</thead>
<tbody>
<tr>
<td>#</td>
<td>Generated_Everyone</td>
<td>Message Waiting</td>
<td></td>
</tr>
<tr>
<td>1</td>
<td>Generated_Everyone</td>
<td>Message Waiting</td>
<td></td>
</tr>
<tr>
<td>10</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c9111_CUBE</td>
</tr>
<tr>
<td>127</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c9111_CUBE</td>
</tr>
<tr>
<td>137</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0022803A6A6</td>
</tr>
<tr>
<td>137</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0022632A2D70</td>
</tr>
<tr>
<td>137</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0022632A2D70</td>
</tr>
<tr>
<td>137</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0022632A2D70</td>
</tr>
<tr>
<td>137</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0022632A2D70</td>
</tr>
<tr>
<td>137</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0022632A2D70</td>
</tr>
<tr>
<td>137</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0022632A2D70</td>
</tr>
<tr>
<td>137</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0022632A2D70</td>
</tr>
<tr>
<td>137</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0022632A2D70</td>
</tr>
<tr>
<td>137</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0022632A2D70</td>
</tr>
<tr>
<td>141</td>
<td>Generated_Everyone</td>
<td>Voice Mail Port</td>
<td>CUCM1-V72</td>
</tr>
<tr>
<td>143</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>AAVN/S0/913/3@CUCM1-V72</td>
</tr>
<tr>
<td>145</td>
<td>Generated_Everyone</td>
<td>Voice Mail Port</td>
<td>CUCM1-V72</td>
</tr>
<tr>
<td>145</td>
<td>Generated_Everyone</td>
<td>Hunt Pilot</td>
<td>CUCM1-V72</td>
</tr>
<tr>
<td>9797-2</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c9111_CUBE</td>
</tr>
<tr>
<td>9.0004</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c9111_CUBE</td>
</tr>
<tr>
<td>9.555112</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c9111_CUBE</td>
</tr>
<tr>
<td>9.555112</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c9111_CUBE</td>
</tr>
<tr>
<td>9.555112</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c9111_CUBE</td>
</tr>
<tr>
<td>9.555112</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c9111_CUBE</td>
</tr>
<tr>
<td>X0000</td>
<td>Generated_Route_Point</td>
<td>Directory Number</td>
<td>Assistant_RP</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Pattern</th>
<th>Description</th>
<th>Partition</th>
<th>Route Filter</th>
<th>Associated Device</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>x</td>
<td>Generated Everyone</td>
<td>c9911_CUBE</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>0</td>
<td>Generated Everyone</td>
<td>c9911_CUBE</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>6*</td>
<td>Generated Everyone</td>
<td>c9911_CUBE</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>6<em>67</em>6</td>
<td>Generated Everyone</td>
<td>c9911_CUBE</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5<em>5X</em>6</td>
<td>Generated Everyone</td>
<td>c9911_CUBE</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5*5X0132</td>
<td>Generated Everyone</td>
<td>c9911_CUBE</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>5*6</td>
<td>Generated Everyone</td>
<td>c9911_CUBE</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>6X*11</td>
<td>Generated Everyone</td>
<td>c9911_CUBE</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>9.X000012345</td>
<td>Generated Everyone</td>
<td>c9911_CUBE</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
### Route Pattern 00

<table>
<thead>
<tr>
<th><strong>Route Pattern</strong></th>
<th>00</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Description</strong></td>
<td>Generated_Everyone</td>
</tr>
<tr>
<td><strong>Numbering Plan</strong></td>
<td>Not Selected</td>
</tr>
<tr>
<td><strong>Route Filter</strong></td>
<td>None</td>
</tr>
<tr>
<td><strong>FLPP Precedence</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Apply Call Blocking Percentage</strong></td>
<td>None</td>
</tr>
<tr>
<td><strong>Resource Priority Namespace Network Domain</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Route Class</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Gateway/Route List</strong></td>
<td>2011_CURR (Edit)</td>
</tr>
<tr>
<td><strong>Route Option</strong></td>
<td>Route this pattern</td>
</tr>
<tr>
<td><strong>Call Classification</strong></td>
<td>OnNet</td>
</tr>
<tr>
<td><strong>Allow Device Override</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Provide Outside Dialed Tone</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Allow Overlap Sending</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Urgent Priority</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Require Forced Authorization Code</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Authentication Level</strong></td>
<td>5</td>
</tr>
</tbody>
</table>

**Connected Party Transformations**
- **Connected Line ID Presentation**: Default
- **Connected Name Presentation**: Default

**Called Party Transformations**
- **Excess Digits**: None
- **Called Party Transform Mask**: 0
- **Prefix Digits (Outgoing Calls)**: 1
- **Called Party Number Type**: Cisco CallManager
- **Called Party Numbering Plan**: Cisco CallManager

**ISDN Network Specific Facilities Information Element**

<table>
<thead>
<tr>
<th>Network Service</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
<td>-- Not Selected --</td>
<td>-- Not Selected --</td>
</tr>
</tbody>
</table>

© 2010 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
Page 18 of 33
EDCS# 1045727 Rev 2

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

This route pattern is used for caller ID blocking.

Strips route pattern adds prefix digit 1 to for CUBE dial peers.

Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
### Route Pattern 9.00X+

**Description:**
- Testing was conducted at CenturyLink VoIP CPE Littleton lab.

**Pattern Definition**
- **Route Pattern:** 9.00X+
- **Description:** Generated_Everyone
- **Numbering Plan:** NCFP
- **Route Matrix:** < None >
- **RIPR Precedence:** Default
- **Apply Call Blocking Percentage:**
- **Resource Priority Namespace Network Domain:** < None >
- **Route Class:** Default
- **Gateway/Route List:** c2011_CUBE
- **Route Option:**
  - Route this pattern
  - Block this pattern
  - No Error
- **Cell Classification:** Default
- **Allow Device Override:**
- **Provide Outside Dial Tone:**
- **Allow Overload Sending:**
- **Uncert Priority:**
- **Require Forced Authorization Code:**
- **Authorization Level:**
- **Require Client Hatter Code:**

**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**
- **Account Code:** STD
- **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
- **Center Identification Code:**

**Note:** Testing was conducted at CenturyLink VoIP CPE Littleton lab.
Route Pattern 9.5551212

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

Strips route pattern adds prefix digit 1 to for CUBE dial peers
Route Pattern 9.X11

Strips route pattern adds prefix digit 1 to for CUBE dial peers

Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
Route Pattern 9.XXX551212

<table>
<thead>
<tr>
<th>Pattern Definition</th>
<th>9.XXX551212</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NAPR</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>PLMN Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>C911_CUBE</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
</tr>
<tr>
<td></td>
<td>Block this pattern</td>
</tr>
</tbody>
</table>

**Calling Party Transformations**

- **Strip Calling Party's External Phone Number Mask**
- **Prefix Digits (Outgoing Calls)**
- **Calling Line ID Presentation**
- **Calling Name Presentation**
- **Calling Party Number Type**
- **Calling Party Numbering Plan**

**Connected Party Transformations**

- **Connected Line ID Presentation**
- **Connected Name Presentation**

**Strips route pattern adds prefix digit 1 to for CUBE dial peers**

---

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

**Note:** Testing was conducted at CenturyLink VoIP CPE Littleton lab.
MGCP FXS Port

<table>
<thead>
<tr>
<th>Device Information</th>
<th>MGCP FXS Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>Gateway</td>
<td>CUCM_CUBE.yourdomain.com</td>
</tr>
<tr>
<td>Device Protocol</td>
<td>Analog Access</td>
</tr>
<tr>
<td>Registration</td>
<td>Registered with Cisco Unified Communications Manager LTLCICCH</td>
</tr>
<tr>
<td>IP Address</td>
<td>10.1.1.1</td>
</tr>
<tr>
<td>End-Point Name</td>
<td>AALN/50/SU1@CUCM_CUBE.yourdomain.com</td>
</tr>
<tr>
<td>Description</td>
<td>AALN/50/SU1@CUCM_CUBE.yourdomain.com</td>
</tr>
<tr>
<td>Device Pool</td>
<td>00722</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td></td>
</tr>
<tr>
<td>Mobile Resource Group List</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>Generated_CSS_1_E</td>
</tr>
<tr>
<td>AAA Calling Search Space</td>
<td>None</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_Name</td>
</tr>
<tr>
<td>AAA Group</td>
<td>Hub_Name</td>
</tr>
<tr>
<td>Location*</td>
<td>None</td>
</tr>
<tr>
<td>AAA Group*</td>
<td>None</td>
</tr>
<tr>
<td>Network Locales</td>
<td>None</td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Default</td>
</tr>
<tr>
<td>GeoLocation</td>
<td>None</td>
</tr>
<tr>
<td>X</td>
<td>Y</td>
</tr>
<tr>
<td>Multi-Iden Precedence and Preemption (MILPP) Information</td>
<td></td>
</tr>
<tr>
<td>MILPP Domain</td>
<td>None</td>
</tr>
<tr>
<td>MILPP Indication Not available on this device</td>
<td></td>
</tr>
<tr>
<td>MILPP Preemption Not available on this device</td>
<td></td>
</tr>
</tbody>
</table>

| Port Information (Loop Start) | |
| Port Direction | Bothways |
| Attendant Dn* | 143 |
| Unattended Port | |

| Product Specific Configuration Layout | |
| HostBash Timer (SD-1550ms)* | 50 |
| Inter-digit Duration Timer (SD-500ms)* | 100 |
| Input Gain (-6.14 db)* | 0 |
| Output Attenuation (-6.14 db)* | 3 |
| Echo Cancellation Enable* | Enable |
| Echo Cancellation Coverage (ms)* | 64 |
| Rings Number* | Default |
| Impedance* | Default GW config |

Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
SIP Trunk

<table>
<thead>
<tr>
<th>Status</th>
<th>Status: Ready</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>Device Information</th>
<th></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>cisco.CME</td>
</tr>
<tr>
<td>Description</td>
<td>SIP Trunk</td>
</tr>
<tr>
<td>Device Pool*</td>
<td>SIP Trunk</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Call Classification*</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>H.450 Resource Group List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location*</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Packet Capture Mode*</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
</tbody>
</table>

- Media Termination Point Required
- E-164 Transformation Profile
- < None >

- Intercompany Media Engine (IME)
- MLPP Domain
- < None >

- Multilevel Precedence and Preemption (MLPP) Information
- MLPP Domain
- < None >

- Call Routing Information
- Remote-Party-Id
- Asserted-Identity
- Asserted-Extension:
- Default
- SIP Privacy*:
- Default

- Inbound Calls
- Significant Digits
- Connected Line ID Presentation
- Connected Name Presentation:
- Default
- Calling Search Space:
- AAR Calling Search Space:
- AAR Calling Search Space:
- Prefix ON
- Redirecting Diversion Header Delivery - Inbound

- Incoming Calling Party Settings
- Use Device Pool/Connected Party Transformation CSS

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

<table>
<thead>
<tr>
<th>Parameter</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>Use Device Pool Called Party Transformation CSS</td>
<td></td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td></td>
</tr>
<tr>
<td>Calling Party Selection</td>
<td>Originate</td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Caller ID CN</td>
<td></td>
</tr>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
</tbody>
</table>

**Note:** Testing was conducted at CenturyLink VoIP CPE Littleton lab.

### SIP Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Destination Address</td>
<td>16.182.0.218</td>
</tr>
<tr>
<td>Destination Address IPv6</td>
<td></td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td></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>DTMF Signaling Method</td>
<td>SIP Trunk IP Profile</td>
</tr>
<tr>
<td>Presence Group</td>
<td>Standard Presence group</td>
</tr>
<tr>
<td>Route Trunk Security Profile</td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>RTP Preferred Originating Codec</td>
<td>G729/G729e</td>
</tr>
<tr>
<td>Destination Port</td>
<td>Cisco</td>
</tr>
</tbody>
</table>

**Required for offnet call forwarding / single number reach**

### Geolocation Configuration

<table>
<thead>
<tr>
<th>Parameter</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 Information

<table>
<thead>
<tr>
<th>Name</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default MTP Telesphony Event Payload Type</td>
<td>Default SIP Profile</td>
</tr>
<tr>
<td>Resource Priority Namespace List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Early Offer for G.729 Calls</td>
<td>Disabled</td>
</tr>
<tr>
<td>SDP Session-level Bandwidth Modifier for Early Offer and Re-invite</td>
<td>TIA8 and ASaconect</td>
</tr>
<tr>
<td>User-Agent and Server header information</td>
<td>Send unified Call Version Information as User-Agent</td>
</tr>
</tbody>
</table>

### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invites Expires (seconds)</td>
<td>160</td>
</tr>
<tr>
<td>Timer Registrar Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Registrar Expires (seconds)</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (mssec)</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (mssec)</td>
<td>4000</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>16554</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>33766</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>k-cisco-serviceاني-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>k-cisco-serviceاني-oplookup</td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>k-cisco-serviceاني-oplookup</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>k-cisco-serviceاني-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 Wing 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>Talent Level for 7940 and 7960</td>
<td>Enabled</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscription Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscription Data (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirects</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>k-cisco-serviceاني-forward</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>k-cisco-serviceاني-abbrevdial</td>
</tr>
</tbody>
</table>

© 2010 Cisco Systems, Inc. All rights reserved.

Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Page 29 of 33

EDCS# 1045727 Rev 2

**Note:** Testing was conducted at CenturyLink VoIP CPE Littleton lab.
**Trunk Specific Configuration**

<table>
<thead>
<tr>
<th>Option</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Re-route Incoming Request to new Trunk based on</td>
<td>Never</td>
</tr>
<tr>
<td>RSVP Over SIP*</td>
<td>Local RSVP</td>
</tr>
<tr>
<td>Fall back to local RSVP</td>
<td></td>
</tr>
<tr>
<td>SIP Rel 10X Options*</td>
<td>Send PRACK if 1xx Contains SDP</td>
</tr>
<tr>
<td>Deliver Conference Bridge Identifier</td>
<td></td>
</tr>
<tr>
<td>Early Offer support for voice and video calls (insert HTTP if needed)</td>
<td></td>
</tr>
<tr>
<td>Send send/receive SDP in mid-call INVITE</td>
<td></td>
</tr>
</tbody>
</table>

**SIP OPTIONS Ping**

<table>
<thead>
<tr>
<th>Option</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable OPTIONS Ping to monitor destination status for Trunks with Service Type &quot;None&quot; (Default)</td>
<td></td>
</tr>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)*</td>
<td>60</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)*</td>
<td>120</td>
</tr>
<tr>
<td>Ping Retry Timer (milliseconds)*</td>
<td>500</td>
</tr>
<tr>
<td>Ping Retry Count*</td>
<td>5</td>
</tr>
</tbody>
</table>

**Note:** Testing was conducted at CenturyLink 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>

**Note:** Testing was conducted at CenturyLink VoIP CPE Littleton lab.
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.
Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.