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

March 22, 2013

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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

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

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

License Info:
License UDI:

-----------------------------------------------
Device# PID SN
-----------------------------------------------

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

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

* Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
Note: 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(.*)$/ /1/
! 
voice translation-rule 3
rule 1 l,*(\.|\/) /1/
! 
voice translation-rule 9
rule 2 /^9\1$/ /1/
! 
voice translation-rule 10
rule 1 l,*(\.*\.)$/ /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 CenturyLink password 7 0518165E331D5A
username cisco privilege 15 secret 5 $1$kuyq$nQxV4Kd3/zsk6/opnU/5/
redundancy
! 
controller T1 0/0/0
cablelength long 0db
channel-group 0 timeslots 1-24
! 
interface Loopback0

3 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

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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 ^72028391[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
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
dtmf-relay rtp-nte
no vad
!

Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
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-nge
  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-nge
  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-nge
  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-nge
  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-nge
  no vad
!
dial-peer voice 204 voip

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Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
translation-profile outgoing DIGITSTRIP-1
preference 1
destination-pattern ^10+
session protocol sipv2
session target ipv4:10.182.0.24
session transport udp
voice-class sip pass-thru content sdp
dtmf-relay rtp-nte
no vad
!
dial-peer voice 205 voip
translation-profile outgoing DIGITSTRIP-1
preference 1
destination-pattern ^15551212
session protocol sipv2
session target ipv4:10.182.0.24
session transport udp
voice-class sip pass-thru content sdp
dtmf-relay rtp-nte
no vad
!
dial-peer voice 304 voip
translation-profile outgoing DIGITSTRIP-1
preference 2
destination-pattern ^10+
session protocol sipv2
session target ipv4:10.182.3.24
session transport udp
voice-class sip pass-thru content sdp
dtmf-relay rtp-nte
no vad
!
dial-peer voice 305 voip
translation-profile outgoing DIGITSTRIP-1
preference 2
destination-pattern ^15551212
session protocol sipv2
session target ipv4:10.182.3.24
session transport udp
voice-class sip pass-thru content sdp
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

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

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

Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
Cisco Unified Communications Manager Configuration

CUCM Version

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

<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>2</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c2911_CUBE</td>
</tr>
<tr>
<td>127</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP002992AAB</td>
</tr>
<tr>
<td>137</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0012CDA2D7D</td>
</tr>
<tr>
<td>137</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0012CDA2D7D</td>
</tr>
<tr>
<td>132</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0026CBA8287B</td>
</tr>
<tr>
<td>135</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP000121FFECCE</td>
</tr>
<tr>
<td>140</td>
<td>Generated_Everyone</td>
<td>Conference</td>
<td></td>
</tr>
<tr>
<td>141</td>
<td>Generated_Everyone</td>
<td>Voice Mail Port</td>
<td></td>
</tr>
<tr>
<td>142</td>
<td>Generated_Everyone</td>
<td>Directory Number</td>
<td>SEP0011CUBEB</td>
</tr>
<tr>
<td>143</td>
<td>Generated_Everyone</td>
<td>Voice Mail Port</td>
<td></td>
</tr>
<tr>
<td>222</td>
<td>Generated_Everyone</td>
<td>Hunt Pilot</td>
<td></td>
</tr>
<tr>
<td>9557.0</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c2911_CUBE</td>
</tr>
<tr>
<td>9.0000</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c2911_CUBE</td>
</tr>
<tr>
<td>9.5551212</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c2911_CUBE</td>
</tr>
<tr>
<td>9.9</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c2911_CUBE</td>
</tr>
<tr>
<td>9.111</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c2911_CUBE</td>
</tr>
<tr>
<td>9.0000551212</td>
<td>Generated_Everyone</td>
<td>Route Pattern</td>
<td>c2911_CUBE</td>
</tr>
<tr>
<td>XXXXXX</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.
Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
### Route Pattern 0

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

**Pattern Definition**
- **Route Partition**: Generated_Everyone
- **Description**: 
- **Numbering Plan**: Not Selected
- **Route Filter**: None
- **PRLP Precedence**: None
- **Apply Call Blocking Percentage**: Default
- **Resource Priority Namespace Network Domain**: None
- **Route Class**: Default
- **Gateway/Route List**: 129.11_CPE (Edit)
- **Route Option**: Route this pattern
- **Call Classification**: OnNet
- **Allow Device Override**: Provide Outside Dial Tone
- **Provide Device Override**: Allow Overlap Sending
- **Urgent Priority**: Required Authorization Code
- **Authorization Level**: 0
- **Required Client Heter Code**: None

**Calling Party Transformations**
- **Use Calling Party's External Phone Number**: Mask
- **Prefix Digits (Outgoing Calls)**: None
- **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**
- **Forward Digits**: None
- **Called Party Transform Mask**: None
- **Prefix Digits (Outgoing Calls)**: 1
- **Called Party Number Type**: Cisco Callmanager
- **Called Party Numbering Plan**: Cisco Callmanager

**ISDN Network-Specific Facilities Information Element**
- **Network Service Protocol**: Not Selected
- **Network Service**: Not Selected
- **Service Parameter Name**: Not Exist
- **Service Parameter Value**: Not Exist

---

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**Note:** Testing was conducted at CenturyLink VoIP CPE Littleton lab.
**Route Pattern 00**

<table>
<thead>
<tr>
<th>Status</th>
<th>Status: Ready</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Pattern Definition</strong></td>
<td>Route Pattern</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td>Generates 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>PRLP Precedence</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Configure Call Blocking Percentage</strong></td>
<td>None</td>
</tr>
<tr>
<td><strong>Resource Priority Namespace Network Domain</strong></td>
<td>ResourcePriorityNamespaceNetworkDomain</td>
</tr>
<tr>
<td><strong>Route Class</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Gateway/Routing List</strong></td>
<td>Route1</td>
</tr>
<tr>
<td><strong>Route Option</strong></td>
<td>Route this pattern</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>Call Classification</strong></th>
<th>Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Allow Device Override</strong></td>
<td>None</td>
</tr>
<tr>
<td><strong>Provide Outside Dial Tone</strong></td>
<td>None</td>
</tr>
<tr>
<td><strong>Allow Overlap Sending</strong></td>
<td>None</td>
</tr>
<tr>
<td><strong>Emergency Override</strong></td>
<td>None</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>Called Party Transformations</strong></th>
<th>Prefix Digits (Outgoing Calls)</th>
<th>1</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Called Party Transform Mask</strong></td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td><strong>Called Party Number Type</strong></td>
<td>Cisco CallManager</td>
<td></td>
</tr>
<tr>
<td><strong>Called Party Numbering Plan</strong></td>
<td>Cisco CallManager</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>Connected Party Transformations</strong></th>
<th>Connected Line ID Presentation</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Connected Name Presentation</strong></td>
<td>Default</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th><strong>ISDN Network Specific Facilities Information Element</strong></th>
<th>Network Service Protocol</th>
<th>-- Not Selected --</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Network Service</strong></td>
<td>Service Parameter Name</td>
<td>Service Parameter Value</td>
</tr>
<tr>
<td>Not Selected</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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**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+

**Route Pattern Configuration**

<table>
<thead>
<tr>
<th>Status</th>
<th>Ready</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pattern Definition</td>
<td>Route Pattern: 9.00X+</td>
</tr>
<tr>
<td>Description</td>
<td>Generated, Everyone</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>CNIP</td>
</tr>
<tr>
<td>Route Addr</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>HLR/SCP Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Apply Call Blocking Percentage</td>
<td>No Error</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>c9711_CUBE</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Default</td>
</tr>
</tbody>
</table>

**Calling Party Transformations**

- Use Calling Party’s External Phone Number 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**

- Redirect Digit: Pro-At
- Called Party Transform Mask
- 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 Protocol</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
<td>-- Not Selected --</td>
<td></td>
</tr>
</tbody>
</table>

---

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

- **Description:** Generated, Everyone
- **Route Pattern:** Route Pattern 9.5551212
- **Numbering Plan:** HAP
- **Route Filter:** None
- **HPLP Precedence:** Default
- **Resource Priority Namespace Network Domain:** None
- **Route Class:** Default
- **Gateway/Route List:** NAT, CUBE
- **Route Option:** Route this pattern
- **Call Classification:** CallWait
- **Calling Party Transformations:**
  - 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
- **Common Party Transformations:**
  - Discard Digits
  - Called Party Transform Mask
  - Prefix Digits (Outgoing Calls): 1
  - Called Party Number Type: Cisco CallManager
  - Called Party Numbering Plan: Cisco CallManager
- **ISDN Network Specific Facilities Information Element:**
  - Network Service Protocol
  - Network Service Parameter Name
  - Network Service Parameter Value
  - Carrier Identification Code

**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

Note: Testing was conducted at CenturyLink VoIP CPE Littleton lab.
**Route Pattern 9.X11**

### Route Pattern Configuration

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

**Pattern Definition**

- **Route Pattern**: 9.X11
- **Description**: Generated, Everyone
- **Numbering Plan**: NAPF
- **Route Filter**: None
- **NUPP Precedence**: Default
- **Apply Call Blocking Percentage**: No
- **Resources Priority Namespace Network Domain**: Default
- **Route Class**: Default
- **Gateway/Route List**: CND\_CUBE (Edit)
- **Route Option**: Route this pattern, Block this pattern

**Call Classification**

- **Call Classification**: Offnet

**Calling Party Transformations**

- **Use Calling Party's External Phone Number Mask**: Yes
- **Calling Party Transform Mask**: Not Set
- **Prefix Digits (Outgoing Calls)**: 1
- **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

**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 Set</td>
<td>Not Set</td>
</tr>
</tbody>
</table>

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

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

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

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

<table>
<thead>
<tr>
<th>MGCP FXS Port</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Product</strong></td>
</tr>
<tr>
<td><strong>Gateway</strong></td>
</tr>
<tr>
<td><strong>Device Protocol</strong></td>
</tr>
</tbody>
</table>

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

### Status

- **Status:** Ready

### Device Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Product</strong></td>
<td>SIP Trunk</td>
</tr>
<tr>
<td><strong>Device Protocol</strong></td>
<td>SIP</td>
</tr>
<tr>
<td><strong>Trunk Service Type</strong></td>
<td>None (Default)</td>
</tr>
<tr>
<td><strong>Device Name</strong></td>
<td>cisc1_cuse</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td>SIP Trunk</td>
</tr>
<tr>
<td><strong>Device Pool</strong></td>
<td>SIP Trunk</td>
</tr>
<tr>
<td><strong>Common Device Configuration</strong></td>
<td>None (Default)</td>
</tr>
<tr>
<td><strong>Call Classification</strong></td>
<td>Use System Default</td>
</tr>
<tr>
<td><strong>Media Resource Group List</strong></td>
<td>None (Default)</td>
</tr>
<tr>
<td><strong>Location</strong></td>
<td>Hub Zone</td>
</tr>
<tr>
<td><strong>AAA Group</strong></td>
<td>None (Default)</td>
</tr>
<tr>
<td><strong>Packet Capture Mode</strong></td>
<td>None (Default)</td>
</tr>
<tr>
<td><strong>Packet Capture Length</strong></td>
<td>0</td>
</tr>
</tbody>
</table>

- **Media Termination Point Required**
- **RTP Video Call as Audio**
- **Transmit RTP for Calling Party Name**
- **Unattended Port**
- **SIP Multicast**
- **SFIP Allowed** - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.
- **Route Class Signaling Enabled**
- **Use Trusted Sip Port**
- **PSTN Access**

### Intercompany Media Engine (IME)

- **E.164 Transformation Profile**

### Multilevel Precedence and Preemption (MUPP) Information

### Call Routing Information

- **Remote-Party-Id**
- **Asserted-Identity**
- **Asserted-Trust**
- **SIP Privacy**

### Inbound Calls

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Significant Digits</strong></td>
<td>5</td>
</tr>
<tr>
<td><strong>Connected Line ID Presentation</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Connected Name Presentation</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Calling Search Space</strong></td>
<td>Generated_CSS_1_6</td>
</tr>
<tr>
<td><strong>AAI Calling Search Space</strong></td>
<td>None (Default)</td>
</tr>
<tr>
<td><strong>Prefix CN</strong></td>
<td></td>
</tr>
</tbody>
</table>

- **Redirecting Diversion Header Delivery - Inbound**

### Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates calling 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>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Incoming Number</strong></td>
<td>Default (None)</td>
</tr>
<tr>
<td><strong>Prefix</strong></td>
<td>None (Default)</td>
</tr>
<tr>
<td><strong>SLIP Digits</strong></td>
<td>None (Default)</td>
</tr>
<tr>
<td><strong>Calling Search Space</strong></td>
<td>Generated_CSS_1_6</td>
</tr>
<tr>
<td><strong>Use Device Pool CSS</strong></td>
<td>Default</td>
</tr>
</tbody>
</table>

### Connected Party Settings

- **Connected Party Transformation CSS**
- **Use Device Pool Connected Party Transformation CSS**
### Outbound Calls

- **Called Party Transformation CSS**: <None>
- **Use Device Pool Called Party Transformation CSS**: <None>
- **Calling Party Transformation CSS**: <None>
- **Use Device Pool Calling Party Transformation CSS**: <None>
- **Calling Party Selection**: Originator
- **Calling Line ID Presentation**: Default
- **Calling Name Presentation**: Default
- **Caller ID ON**: 
- **Caller Name**: 

**Redirecting Diversion Header Delivery - Outbound**: Required for offnet call forwarding / single number reach

### SIP Information

- **Destination Address**: 10.182.0.218
- **Destination Address IPv6**: 
- **Destination Address is an SRV**: 
- **Destination Port**: SCCP
- **MTP Preferred Originating Codec**: GECOMP/GECOMP
- **Presence Group**: Standard Presence group
- **SIP Trunk Security Profile**: Non Secure SIP Trunk Profile
- **Routing Calling Search Space**: <None>
- **Out-Of-Calling Searchspace Calling Search Space**: <None>
- **SUBSCRIBE Calling Search Space**: <None>
- **SIP Profile**: SIP Trunk IP Profile
- **DTMF Signaling Method**: RFC 2833

### Geolocation Configuration

- **Geolocation**: <None>
- **Geolocation Filter**: <None>
- **Send Geolocation Information**: 

---

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

### SIP Profile Information

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>SIP Trunk IP Profile</th>
<th>Default SIP Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Default MTP Telemetry Event Payload Type</td>
<td>Default SIP Profile</td>
<td>101</td>
</tr>
<tr>
<td></td>
<td>Resource Priority Namespace List</td>
<td>&lt; None &gt;</td>
<td>Disabled</td>
</tr>
<tr>
<td></td>
<td>EarlyOfferForQClearCalls</td>
<td>&lt; None &gt;</td>
<td>Disabled</td>
</tr>
<tr>
<td></td>
<td>SIP Session-level Bandwidth Modifier for Early Offer and Re-invite</td>
<td>&lt; None &gt;</td>
<td>Disabled</td>
</tr>
<tr>
<td></td>
<td>User-Agent and Server header information</td>
<td>&lt; None &gt;</td>
<td>Disabled</td>
</tr>
</tbody>
</table>

- Redirect by Application
- Disable Early Media on 100
- Outgoing T.26 INVITE include audio alone
- Enable ANAT
- Require SIP Inactive Exchange for Mid-Call Media Change
- Use Fully Qualified Domain Name in EIP Requests
- Allow Presentation Sharing using ECP

### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)</td>
<td>160</td>
</tr>
<tr>
<td>Timer Registrar Expires (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>16384</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>13766</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>k-cisco-servicean-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>k-cisco-servicean-pickup</td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>k-cisco-servicean-pickup</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>k-cisco-servicean-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>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 Subscribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe 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>1500</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>k-cisco-servicean-divert</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>k-cisco-servicean-abbrevial</td>
</tr>
</tbody>
</table>

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**Note:** Testing was conducted at CenturyLink VoIP CPE Littleton lab.
Required for far end ringback
Send codec’s in initial INVITE, Cisco UBE set to pass through to CenturyLink

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.
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