# Acme Packet – Cisco CVP Verizon Trunk Interoperability Appication Note

## Revision History

| Version | Author | Description of Changes | Date Revision Completed |
|---|---|---|---|
| ***1.0*** | Andy Tatum | Initial Draft | 5/4/2012 |
| | | | |




## Abstract

*This document provides an overview of the Cisco CVP/CUCM Interoperability testing completed by Acme Packet Net-Net 3800 and Verizon Business SIP Trunking.*


## Contents

## Contents

# 1 Introduction

Acme Packet Net-Net session border controllers (SBCs) provide critical control functions to enable enterprises to deliver trusted, first-class interactive communications across IP network borders. A broad range of interactive communications services and applications ranging from basic VoIP to service oriented architecture (SOA)-enabled unified communications (UC) and collaboration are supported.

This document aims to provide an overview of the interoperability testing between the Net-Net 3800 SBC and the Cisco CVP environment with Verizon.

## 1.1 Intended Audience

This document is intended for use by Acme Packet HQ and Field Based Engineers. It assumes the reader is familiar with basic operations of the Session Director, and has attended the following training course(s) (or has equivalent experience):

- EDU-CAB-C-CLI Net-Net 4000/3000 Configuration Basics
- EDU-ADV-OE Net-Net Session Director Advanced Configuration

Further, the test plans enclosed assume familiarity with the SD's ACLI command line interface, retrieving and reviewing log files generated by the SD, standard network analysis tools (wireshark/tcpdump), and all protocols involved in the activity.

## 2 Application Overview

The test cases provided in this document are specific to Cisco CVP/CUCM Environments interfacing with Verizon business VoIP SIP trunks. Previous deployments required a Cisco CUBE to interface directly with CVP. This testing is a certification of CVP environments with 3800/4500 Acme SBCs and Verizon business SIP trunks. All test results have been implemented and certified by Tekvizion Labs, and accepted by Cisco Systems.

# 3 Software/Hardware/Tools

## 3.1 Net-Net SBC Hardware and Software Requirements

This section gives a high level view of system requirements and test tools/equipment used to research and test the feature outlined in this Tech Note.

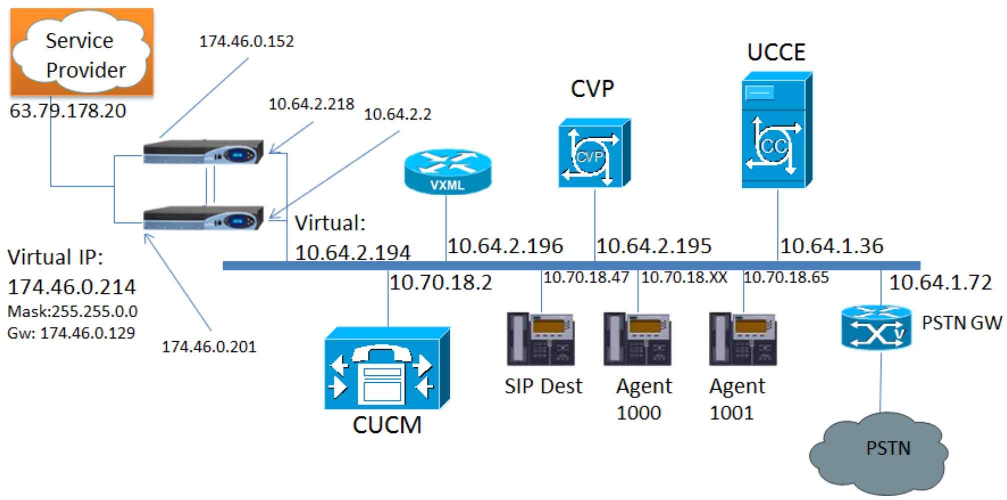
| SBC Platform | Mainboard Rev. | Bootloader | Software Version/Patch |
|---|---|---|---|
| NN3820 | 4.00 | 06/21/2011 | SCX6.2.0 MR-9 |
| NN3820 | 4.00 | 06/21/2011 | SCX6.3.0 GA |
| NN3820 SPL - Script | | | AvayaCiscoUCID64.4.spl |
| | | | SPL Version C2.0.0 |

## 3.2 Test Tool / Third Party Equipment used for Feature research and Testing

The following test tools and/or Third Party Equipment were used during for research and testing of the feature outlined in this Tech Note. **Where applicable; test tool usage instructions, including configuration overview, will be noted.**

| Third Party Platform | Software Version/Patch |
|---|---|
| Cisco VXML | 12.4(13r)T |
| CUCVP | 8.5(1) |
| CUCM | 8.5.1.1 |
| CUICM/CCE | 8.5.2.0 |

### 3.3 Test Bed Diagrams




.

## 4 Verizon SIP Trunk to CVP Test Cases and Use Cases

| | | | |
|---|---|---|---|
| Basic Call Setup CVP Setup | | The following configuration applies to all tests in this section, unless otherwise noted:<br>- G711U preferred codec<br>- Call sent to CVP<br>- Separate VXML browser (non-combo)<br>- SIP over TCP (behind SBC)<br>- Agent with SIP phone<br>- Agent preferred codec G711U<br>- Phone number provisioned in the user portion of request URI<br>- KPML not enabled in CUCM or SBC | |
| Regular inbound call | Establish baseline | 1. Dial toll free number<br>2. Wait for CVP to answer the call<br>3. Hang up | **Pass** |
| CVP hangs up | | 1. Dial toll free number<br>2. Wait for CVP to answer the call<br>3. CVP application hangs up the call | **Pass** |
| Disconnect during ring phase | Verify proper call termination | 1. For this test case only,use the Ring No Answer Script<br>2. Dial toll free number<br>3. Hang up quicky after finish dialing<br>4. Verify that call was properly torn down (PCAP) | **Pass** |
| SIP over UDP | Verify UDP is working properly (UDP used internally, behind SBC. Verizion always uses UDP outside SBC) | 1. For this test case only, configure all devices to use SIP over UDP<br>2. Dial toll free number<br>3. Wait for CVP to answer the call<br>4. Hang up | **Pass** |

| | | | |
|---|---|---|---|
| Long Call | Verify proper and session refresh for long calls | 1. Dial toll free number<br>2. Wait for CVP to answer the call<br>3. Keep the call up for at least 30 min (has to be longer than session refresh timer)<br>4. Hang up<br>5. Verify that session was refreshed accordingly | **Pass** |
| Service Provider Proprietary headers | Verify graceful handling of service provider proprietary headers | 1. For this test case only, configure service provider proprietary headers<br>2. Dial toll free number<br>3. Wait for CVP to answer the call<br>4. Verify DTMF and ASR<br>5. Hang up | **Pass** |
| Non IP-IVR, non-NCR calls with Ring No Answer scripts on. | | 1. Dial toll free number 1866 6747056<br>2. Wait for CVP to answer the call<br>3. Destination does not answer the call<br>4. Verify disposition | **Pass** |
| Non IP-IVR, non-NCR calls with Busy scripts on. | | 1. Dial toll free number 1866 6747056<br>2. Wait for CVP to answer the call<br>3. Destination is busy<br>4. Verify disposition | **Pass** |
| Basic Call Setup to CVP, Negative Testing | | Same as previous section, with the following additions:<br>- Create two dial peers in SBC. One with higher preference should target CVP, the other an inexistent IP address.<br>- Make sure ringback dial peer (91919191) and 92929292 are configured | |
| Non-provisioned number in SBC | Verify proper error message is sent to service provider | 1. For this test case only, do not configure dial peer for incoming number in SBC<br>2. Place call<br>3. Verify error message sent to service provider | Pass |

| | | | |
|---|---|---|---|
| Provisioning error in SBC dial-peer | Verify SBC redirects the call to healthy destination | 1. For this test case only, change the dial peer preference in order to have CVP being the second choice for SBC.<br>2. Dial toll free number<br>3. Wait for CVP to answer the call<br>4. Hang up | **Pass** |
| Unreachable destination in SBC | Verify proper error message is sent to service provider | 1. For this test case only, disable the dial peer that targets CVP (but leave the one that points to an inexistent destination).<br>2. Place toll free call<br>3. Verify error message is sent to service provider (or time out occurs) | Pass |
| Non-provisioned number in CVP | Verify proper error message is sent to service provider | 1. For this test case only, make sure the incoming number is not provisioned in CVP<br>2. Place toll free call<br>3. Verify error message is sent to service provider | **Pass** |
| Non-provisioned number in UCCE | Verify proper error message is sent to service provider | 1. For this test case only, make sure the incoming number is not provisioned in UCCE<br>2. Place toll free call<br>3. Verify error message is sent to service provider | Pass |
| Non-provisioned number in VXML browser | Verify proper error message is sent to service provider | 1. For this test case only, do not provision the VXML browser bootstrap dial peer<br>2. Place toll free call<br>3. CVP will try to recover from the VXML failure, but will eventually give up<br>4. Verify messages | **Pass** |
| Unreachable VXML browser | Verify proper error message is sent to service provider | 1. For this test case only, configure an inexistent VXML browser in CVP<br>2. Place toll free call<br>3. CVP will try to recover from the VXML failure, but will eventualy give up<br>4. Verify messages | **Pass** |

| | | | |
|---|---|---|---|
| VXML browser failover | Verify proper error message is sent to service provider | 1. For this test case only, provision an inexistent VXML browser in addition to a valid one<br>2. Place toll free call<br>3. Make sure CVP attempts the inexistent VXML browser first<br>4. Wait for valid browser to answer the call<br>5. Hang up | **Not Tested** |
| Midcall SBC failure | Verify proper error message is sent to service provider | 1. Place toll free call<br>2. Wait for CVP to answer<br>3. Disconnect SBC<br>4. Verify messages | **Pass** |
| Midcall CVP failure | Verify proper error message is sent to service provider | 1. Place toll free call<br>2. Wait for CVP to answer<br>3. Disconnect CVP server<br>4. Verify disposition (survivability script should be on) | **Pass** |
| Midcall VXML Browser failure | Verify proper error message is sent to service provider | 1. Place toll free call<br>2. Wait for CVP to answer<br>3. Disconnect VXML browser<br>4. Verify disposition (survivability script should be on) | **Pass** |
| Midcall PG failure | Verify proper error message is sent to service provider | 1. Place toll free call<br>2. Wait for CVP to answer<br>3. Disconnect VRU PG<br>4. Verify disposition (survivability script should be on) | **Pass** |
| Queue and Transfer to Agent | | Same as previous section, with the following addition, unless otherwise noted:<br>- Agent in auto-answer<br>- CVP RONA configured<br>- Agent configured with G279 starting on test case 5<br>- SBC configured to end-to-end codec renegotiation | |

| | | | |
|---|---|---|---|
| Self-service, queue and agent | Baseline | 1. Make agent unavailable<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Make agent available<br>6. Agent answers the call (take note of caller ID presented)<br>7. Caller hangs up | **Pass** |
| No self service, direct to agent | Verify signaling behavior when no VXML is used. | 1. For this test case only, use an application that immediately queues to a skill group<br>2. Make agent available<br>3. Place toll free call<br>4. Agent answers the call<br>5. Caller hangs up | **Pass** |
| Agent hangs up | Verify called party disconnect. | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent hangs up | **Pass** |
| Hold | Verify hold/resume signaling. | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent puts call on hold<br>7. After a few seconds, agent resumes call<br>8. Caller hangs up | Pass |

| | | | |
|---|---|---|---|
| Long hold | Verify interoperability issues when calls are on hold for long periods of time. | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent puts call on hold<br>7. After 30 min, agent resumes call<br>8. Caller hangs up | **Pass** |
| Midcall codec negotiation | Verify end-to-end midcall codec negotiation (Service Provider starts G711, then switches to G729 after reINVITE). SBC DSPs are not engaged. | 1. From this point on, make sure the agent is configured to use G729<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer (G711 all the way to service provider)<br>5. Queue call to skill group<br>6. Agent answers the call<br>7. Make sure no DSPs or transcoders are engaged, and G729 is being used to service provider<br>7. Caller hangs up | Pass |
| Mute (silence) | Verify behavior during long periods of silence. | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent puts call on mute<br>7. After 30 min, agent resumes call<br>8. Caller hangs up | **Pass** |

| | | | |
|---|---|---|---|
| Midcall DSP insertion (SBC) | Verify DSP insertion in SBC. Service Provider does not renegotiate initial G711 codec. | 1. For this test case only, make sure SBC is not configured for end-to-end codec renegotiation. Internal dial peers offer G711 and G729.<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer (G711 all the way to service provider)<br>5. Queue to skill group<br>6. Agent answers the call<br>7. SBC should insert transcoder (G711 to service provider, G729 internally)<br>8. Caller hangs up | **Not Tested** |
| Midcall DSP insertion (CUCM) | Verify DSP insertion in CUCM. Service Provider does not renegotiate initial G711 codec. | 1. For this test case only, make sure SBC is not configured for end-to-end codec renegotiation. Internal dial peers offer only G711. Tanscoder configured in CUCM.<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer (G711 all the way to service provider)<br>5. Queue to skill group<br>6. Agent answers the call<br>7. CUCM should insert transcoder (G711 to service provider, G711 from SBC to Transcoder, G729 to agent)<br>8. Caller hangs up | **Pass** |
| SCCP Phone | Verify any differences with SCCP phones | 1. For this test case only, configure a SCCP phone<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer (G711 all the way to service provider)<br>5. Queue call to skill group<br>6. Agent answers the call<br>7. Make sure no DSPs or transcoders are engaged, and G729 is being used to service provider<br>7. Caller hangs up | **Pass** |

| | | | |
|---|---|---|---|
| Encrypted SIP Phone | Verify any differences in behavior when encryption is used in phoen signaling | 1. For this test case only, configure an encrypted SIP phone<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer (G711 all the way to service provider)<br>5. Queue call to skill group<br>6. Agent answers the call<br>7. Make sure no DSPs or transcoders are engaged, and G729 is being used to service provider<br>7. Caller hangs up | **Pass** |
| No auto answer | Verify signaling changes when agent has to manually answer the call. | 1. For this test case only, do not configure auto-answer<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer<br>5. Queue call to skill group<br>6. Manually answer the call<br>7. Caller hangs up | **Pass** |
| RONA | Verify signaling changes when agent does not answer the call, and call is redirected. | 1. For this test case only, do not configure auto-answer<br>2. Make sure CVP is configured to RONA<br>3. Make agent available<br>4. Place toll free call<br>5. Wait for CVP to answer<br>6. Queue call to skill group<br>7. Do not answer the call<br>8. CVP RONAs back to queue | **Pass** |

| | | | |
|---|---|---|---|
| Long Call | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Keep the call up for at least 30 min (has to be longer than session refresh timer)<br>7. Hang up<br>8. Verify that session was refreshed accordingly | ***Pass*** |
| Hold with sendrecv | Verify whether the SBC can propery change CUCM default hold messaging (sendonly/inactive) to sendrecv | Create a SIP/SDP header transformation that modified reINVITE messages sent TO the service provider. The transformation should change SDP headers a=sendonly or a=inactive should to a=sendrecv<br>1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent puts call on hold<br>7. After a few seconds, agent resumes call<br>8. Caller hangs up | **Pass** |
| Caller with privacy settings (caller ID presentation restriction) | | 1. Make agent unavailable<br>2. Place toll free call using privacy code (*67 before toll free number)<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Make agent available<br>6. Agent answers the call (take note of caller ID presented)<br>7. Caller hangs up | **Pass** |

| | | | |
|---|---|---|---|
| Queue and Transfer to Agent, Negative Testing | | Same as previous section, with the following addition, unless otherwise noted: | |
| No CUCM routes provisioned in CVP | | 1. Do not configure routes for CUCM in CVP<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer<br>5. Queue call to skill group<br>6. Verify disposition | **Pass** |
| Midcall CUCM failure | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Disconnect CUCM Sub<br>7. Verify disposition | **Pass** |
| Midcall phone failure | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Unplug agent's phone<br>7. Verify disposition | **Pass** |
| Ringback service unavailable | | 1. Do not configure the 91919191 dial peer in the VXML browser<br>2. Make agent available<br>3. Place toll free call<br>4. Request to talk to agent<br>5. Agent answers the call<br>6. Caller hangs up<br>7. Verify messages | **Pass** |

| | | | |
|---|---|---|---|
| Midcall CVP failure | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Unplug CVP<br>7. Verify disposition | **Pass** |
| Basic Transfers and Conferences | | Same as previous section, with the following addition, unless otherwise noted.<br>All calls flows in this section start with the same steps:<br>1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer<br>4. Request to talk to agent<br>5. Agent answers the call (G729 end-to-end) | |
| Blind (single step) transfer to queue | Verify CVP-based transfers, internal destination. (Service Provider doesn't "see" the transfer) | 6. Agent requests blind transfer to queue<br>7. CVP pulls the call back from CUCM (BYE), and sends call to VXML browser (call negotiates G711 end-to-end)<br>8. Make another agent available<br>9. New agent answers the call(call negotiations G729 end-to-end)<br>10. Caller hangs up | **Pass** |
| Consultative transfer to queue, completes in queue | Warm transfers (Service Provider doesn't "see" the transfer) | 6. Agent requests consultative transfer to queue<br>7. CUCM puts caller on hold, calls CVP (transcoder engaged)<br>8. Agent completes the transfer (caller in queue)<br>9. Make another agent available<br>10. New agent answers the call (call negotiates G729 end-to-end)<br>11. Caller hangs up | Pass |

| | | | |
|---|---|---|---|
| Consultative transfer to queue, completes when call is answered | Warm transfers (Service Provider doesn't "see" the transfer) | 6. Agent requests consultative transfer to queue<br>7. CUCM puts caller on hold, calls CVP (transcoder engaged)<br>8. Make another agent available<br>9. New agent answers the call (both agents are talking)<br>10. Complete the transfer<br>11. Caller hangs up | Pass |
| Agent-to-agent trasnsfer | | 6. Agent initiates consultative transfer to another agent<br>7. New agent answers the call<br>8. Complete the transfer<br>9. Caller hangs up | **Pass** |
| Consultative transfer to external destination (over SIP trunk), service provider SIP destination | CUCM-controlled transfers. Singaling hairpins in UCCE, transferred leg is seen by Service Provider as an outbound call (iow, Service Provider does not "think" it is a blind transfer). | 6. Agent requests consultative transfer to external destination<br>7. CUCM puts caller on hold, calls CVP (transcoder engaged)<br>8. CUCM calls external destination through SBC<br>9. Destination answers<br>10. Agent completes transfer | Pass |
| Advanced Transfers: take back and transfer (DTMF *8) | | Same as previous section, with the following addition, unless otherwise noted.<br>All calls flows in this section start with the same steps:<br>1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer<br>4. Request to talk to agent<br>5. Agent answers the call (G729 end-to-end) | |

| | | | |
|---|---|---|---|
| Advanced Transfers: REFER | | Same as previous section, with the following addition, unless otherwise noted.<br>- UserToUserInfo varilable not set<br>All calls flows in this section start with the same steps:<br>1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer<br>4. Request to talk to agent<br>5. Agent answers the call (G729 end-to-end) | |
| REFER back to service provider, PSTN destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC<br>8. SBC configured to pass REFER through<br>9. Service provider pulls the call back and completes transfers | **Pass** |
| REFER back to service provider, Service Provider SIP destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC<br>8. SBC configured to pass REFER through<br>9. Service provider pulls the call back and completes transfers | **Pass** |
| REFER back to service provider, malformed destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC (inexistent number)<br>8. SBC configured to pass REFER through<br>9. Service provider pulls the call back and completes transfers | **Pass** |
| REFER back to service provider, no answer, PSTN destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC<br>8. SBC configured to pass REFER through<br>9. Destination does not answer<br>10. Verify disposition | **Pass** |

| | | | |
|---|---|---|---|
| REFER back to service provider, no answer, Service Provider SIP destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC<br>8. SBC configured to pass REFER through<br>9. Destination does not answer<br>10. Verify disposition | **Pass** |
| REFER back to service provider, busy, PSTN destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC<br>8. SBC configured to pass REFER through<br>9. Destination is busy<br>10. Verify disposition | **Pass** |
| REFER back to service provider, busy, Service Provider SIP destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC<br>8. SBC configured to pass REFER through<br>9. Destination is busy<br>10. Verify disposition | **Pass** |
| REFER back to service provider, caller hangs up before completion, PSTN destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC<br>8. SBC configured to pass REFER through<br>9. Caller hangs up while destination is still ringing<br>10. Verify disposition | **Pass** |
| REFER back to service provider, caller hangs up before completion, SIP destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC<br>8. SBC configured to pass REFER through<br>9. Caller hangs up while destination is still ringing<br>10. Verify disposition | **Pass** |
| REFER back to service provider, with GTD | | 6. For this test case only, set the UserToUserInfo variable<br>7. Agent requests blind transfer<br>8. CVP pulls the call back, sends REFER to SBC<br>9. SBC configured to pass REFER through (signaling forward unconditional)<br>9. Verify disposition | **Pass** |

| | | | |
|---|---|---|---|
| SBC Consumes REFER, all G711 | Service Provider does not see the REFERs (transfer enforced internally) | 6. For this test case only, all phones need to be G711<br>7. Agent (G711) requets blind transfer<br>8. CVP pulls the call back, sends REFER to SBC<br>9. SBC configured to consume REFER<br>10. Destination could be another phone in CUCM (G711)<br>11. Destination answers the phone | **Pass** |
| SBC Consumes REFER, with midcall codec insertion | | 6. For this test case only, initiation agent is G711, but destination phone is G729<br>7. Agent (G711 end-to-end) requests blind transfer<br>8. CVP pulls the call back, sends REFER to SBC<br>9. SBC consumes REFER<br>10. Destination is G729-only phone<br>11. Destination answers call, SBC inserts DSP | **<u>Not Tested</u>** |
| SBC Consumes REFER, midcall codec negotation | | 6. Agent (G729 end-to-end) requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC<br>8. SBC configuted to consume REFER<br>9. Destination is G711-only phone<br>10. Destination answers, G711 negotatiated end-to-end | **Pass** |
| SBC Consumes REFER, midcall codec negotiation, no survivability script | | 6. From this test case on, survivability script turned off<br>7. Agent (G729 end-to-end) requests blind transfer<br>8. CVP pulls the call back, sends REFER to SBC<br>9. SBC configuted to consume REFER<br>10. Destination is G711-only phone<br>11. Destination answers, G711 negotatiated end-to-end | **Pass** |

| | | | |
|---|---|---|---|
| SBC consumes REFER, malformed destination | | 6. Agent (G729 end-to-end) requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC (destination not configured in SBC)<br>8. SBC configuted to consume REFER<br>9. Verify disposition (make sure Requery is configured in UCCE) | **Pass** |
| SBC consumes REFER, destination does not answer | | 6. Agent (G729 end-to-end) requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC (destination not configured in SBC)<br>8. SBC configuted to consume REFER<br>9. Destination rings, and call is not answered<br>10. Verify disposition after several seconds | **Pass** |
| 302 Redirect Consume | | 1. Configure UCCE Script that simply redirect the call (Start -> Refer Label). Destination should be a UCCE phone.<br>2. Place call a toll free call.<br>3. Verify that the SIP INVITE arrived at CVP, and it responded with a 302 Redirect<br>4. SBC should receive Redirect and send call to the provided destination | **Pass** |
| Mobile Agent | | MTPs need to be configured for Mobile Agent. Trunk Groups need to be set for RFC2833 preferred DTMF in order for the MTPs to be dynamically allocated. Ideally, Mobile Agent should use a phone over a SIP trunk. | |
| Self-service, queue and agent in auto-answer | | 1. For this test case only, make agent unavailable<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Make agent available<br>6. Agent answers the call<br>7. Caller hangs up | **Not Tested** |

| | | | |
|---|---|---|---|
| Self-service, queue and agent in manual-answer | | 1. For this test case only, make agent unavailable<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Make agent available<br>6. Agent answers the call<br>7. Caller hangs up | **Not Tested** |
| Consultative transfer to queue, completes in queue | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer<br>4. Request to talk to agent<br>5. Agent answers the call (G729 end-to-end)<br>6. Agent requests consultative transfer to queue<br>7. CUCM puts caller on hold, calls CVP (transcoder engaged)<br>8. Agent completes the transfer (caller in queue)<br>9. Make another agent available<br>10. New agent answers the call (call negotiates G729 end-to-end)<br>11. Caller hangs up | **Not Tested** |
| Consultative transfer to queue, completes when call is answered | - | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer<br>4. Request to talk to agent<br>5. Agent answers the call (G729 end-to-end)<br>6. Agent requests consultative transfer to queue<br>7. CUCM puts caller on hold, calls CVP (transcoder engaged)<br>8. Make another agent available<br>9. New agent answers the call (both agents are talking)<br>10. Complete the transfer<br>11. Caller hangs up | **Not Tested** |

| | | | |
|---|---|---|---|
| Long Call | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Keep the call up for at least 30 min (has to be longer than session refresh timer)<br>7. Hang up<br>8. Verify that session was refreshed accordingly | **Not Tested** |
| Mobile Agent, Negative Testing | | MTPs need to be configured for Mobile Agent. Trunk Groups need to be set for RFC2833 preferred DTMF in order for the MTPs to be dynamically allocated. Ideally, Mobile Agent should use a phone over a SIP trunk. | |
| Midcall SBC failure | | 1. Place toll free call<br>2. Wait for CVP to answer<br>3. Queue call<br>4. Mobile Agent answers<br>5. Disconnect SBC<br>6. Verify disposition | Not Tested |
| Midcall CVP failure | | 1. Place toll free call<br>2. Wait for CVP to answer<br>3. Queue call<br>4. Mobile Agent answers<br>5. Disconnect CVP<br>6. Verify disposition | Not Tested |
| Midcall connection failure | | 1. Place toll free call<br>2. Wait for CVP to answer<br>3. Queue call<br>4. Mobile Agent answers<br>5. Disconnect agent's phone<br>6. Verify disposition | Not Tested |
| Miscellaneous Features | | Survivability Script needs to be turned on | |
| Standalone CVP | | Not comprehensive mode | |

| | | | |
|---|---|---|---|
| Standalone CVP, Negative Testing | | Same as previous section, with the following additions:<br>- Create two dial peers in SBC. One with higher preference should target the VXML browser the other an inexistent IP address. | |
| Standalone Advanced Transfers: REFER | | Same as previous section, with the following addition, unless otherwise noted.<br>- Requires IOS 15.2.1T (for this test only)<br>- Survivability script turned off<br>- SBC Configured with REFER passthrough<br>All calls flows in this section start with the same steps:<br>1. Place toll free call<br>2. Wait for CVP to answer | |
| Verizon IP-IVR (Verizon Business IP Toll Free Specific) | IP-IVR adds different network elements to Verizon's network, which may change how the signaling to CVP works | CVP in comprehensive mode | |
| Regular inbound call | Establish baseline | 1. Dial toll free number<br>2. Wait for CVP to answer the call<br>3. Hang up | **Pass** |
| Incoming DTMF | Verify RFC2833 compliance | 1. Dial toll free number<br>2. Wait for CVP to answer the call<br>3. Use DTMF to navigate through CVP prompts<br>4. Hang up | **Not Tested** |
| Disconnect during ring phase | Verify proper call termination | 1. For this test case only, configure in CVP a "fake" VXML browser (unreachable IP address)<br>2. Dial toll free number<br>3. Hang up quicky after finish dialing<br>4. Verify that call was properly torn down (PCAP) | **Pass** |

| | | | |
|---|---|---|---|
| Long Call | Verify proper and session refresh for long calls | 1. Dial toll free number<br>2. Wait for CVP to answer the call<br>3. Keep the call up for at least 30 min (has to be longer than session refresh timer)<br>4. Hang up<br>5. Verify that session was refreshed accordingly | **Pass** |
| Self-service plus agent | Baseline | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent hangs up | **Pass** |
| Hold | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent puts call on hold<br>7. After a few seconds, agent resumes call<br>8. Caller hangs up | **Pass** |
| Long hold | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent puts call on hold<br>7. After 30 min, agent resumes call<br>8. Caller hangs up | **Pass** |

| | | | |
|---|---|---|---|
| Midcall codec negotiation | Verify end-to-end midcall codec negotiation. CUBE DSPs are not engaged. | 1. For this test case only, make sure the agent is configured to use G729<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer (G711 all the way to service provider)<br>5. Queue call to skill group<br>6. Agent answers the call<br>7. Make sure no DSPs or transcoders are engaged, and G729 is being used to service provider<br>7. Call expected to fail because Verizon Media Server does not support G729 | **Pass** |
| Midcall codec negotiation with Verizon RLT | | 1. For this test case only, make sure the agent is configured to use G729<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer (G711 all the way to service provider)<br>5. Queue call to skill group<br>6. Agent answers the call<br>7. Transcoders should be engaged, with agent using G729 and SP using G711<br>8. Agent presses *7, which triggers RLT<br>9. Verify if codec was renegotiated | **Pass** |
| Mute | | 1. IP-IVR call comes to an agent (should be G711)<br>2. Agent dials *7 (Verizon sends reINVITE offering new codecs, but call does not renegotiate)<br>3. Agent transfers to another phone that is configured for G729 only (the call should now be G729 end-to-end)<br>4. New phone puts the call on mute for at least 30 minutes<br>5. Hang up | **Pass** |

| | | | |
|---|---|---|---|
| Consultative transfer to queue, completes when call is answered | Warm transfers (Service Provider doesn't "see" the transfer) | 6. Agent requests consultative transfer to queue<br>7. CUCM puts caller on hold, calls CVP (transcoder engaged)<br>8. Make another agent available<br>9. New agent answers the call (both agents are talking)<br>10. Complete the transfer<br>11. Caller hangs up | **Pass** |
| DTMF blind transfer, Service Provider releases the call, **PSTN** destination | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent requests blind transfer<br>7. CVP pulls the call back, plays INFO messages to CUBE<br>8. CUBE converts INFO to RFC2833 tones<br>9. Service provider pulls the call back and completes transfers | Not Tested |
| DTMF blind transfer, CVP releases the call **PSTN** destination | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent requests blind transfer to PSTN destination<br>7. CVP pulls the call back, plays INFO messages to CUBE<br>8. UserToUser variable configured, so CVP sends BYE<br>9. CUBE converts INFO to RFC2833 tones (no GTD passthrough) | **Pass** |

| | | | |
|---|---|---|---|
| DTMF blind transfer, CVP releases the call releases the call, SIP destination | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent requests blind transfer to SIP destination<br>7. CVP pulls the call back, plays INFO messages to CUBE<br>8. UserToUser variable configured, so CVP sends BYE<br>9. CUBE converts INFO to RFC2833 tones (no GTD passthrough) | **Pass** |
| DTMF blind transfer, CVP hangs up the call, GTD sent to service provider | Similar to the previous test, but this time verifying what happens when the GTD reaches Verizon Business. In order for CUBE to relay the GTD, use "signaling forward unconditional" in the voice class. | 6. Agent requests blind transfer<br>7. CVP pulls the call back, plays INFO messages to CUBE<br>8. CUBE converts INFO to RFC2833 tones<br>9. Verify disposition | **Pass** |
| DTMF blind transfer, malformed destination | To determine the call disposition when the label CVP sends is not recognized by Verizon Business. The label provided by ICM should be an invalid number (e.g., '12345678'). UserToUserInfo variable not set. | 6. Agent requests blind transfer<br>7. CVP pulls the call back, plays INFO message to CUBE. Destination is not a valid number<br>8. Verify disposition. | **Pass** |
| DTMF blind transfer, incomplete destination | Verify call disposition when label is short | 6. Agent requests blind transfer<br>7. CVP pulls the call back, plays INFO message to CUBE. Destination is a valid number, except that it is missing the last digit.<br>8. Verify disposition (may have to wait several minutes). | **Pass** |

| | | | |
|---|---|---|---|
| DTMF consultative transfer, CVP releases the call releases the call, SIP destination | | 1. Place inbound call<br>2. Call is queued, etc<br>3. Agent answers<br>4. Agent uses his PHONE (NOT the dektop) and dials *8866XXXXXXX. Agent stays on the line<br>5. Destination answers. Verify destination is talking to agent.<br>6. Agent enters *#<br>7. Verify that Verizon tears down call to agent. Destination should continue talking to caller.<br>8. Caller hangs up. | **Pass** |
| DTMF consultative transfer, CVP does NOT release the call releases the call, SIP destination | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent requests blind transfer to SIP destination<br>7. CVP pulls the call back, plays INFO messages to CUBE<br>8. CUBE converts INFO to RFC2833 tones (no GTD passthrough)<br>9. Verify disposition (make take several minutes) | **Pass** |
| REFER back to service provider, **PSTN** destination | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer<br>4. Request to talk to agent<br>5. Agent answers the call<br>6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to CUBE<br>8. CUBE configured to pass REFER through<br>9. Service provider pulls the call back and completes transfers | **Pass** |

| | | | |
|---|---|---|---|
| REFER back to service provider, Service Provider SIP destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to CUBE<br>8. CUBE configured to pass REFER through<br>9. Service provider pulls the call back and completes transfers | **Pass** |
| REFER back to service provider, malformed destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to CUBE (inexistent number)<br>8. CUBE configured to pass REFER through<br>9. Service provider pulls the call back and completes transfers | **Pass** |
| REFER back to service provider, no answer, **PSTN** destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to CUBE<br>8. CUBE configured to pass REFER through<br>9. Destination does not answer<br>10. Verify disposition | **Pass** |
| REFER back to service provider, no answer, Service Provider SIP destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to CUBE<br>8. CUBE configured to pass REFER through<br>9. Destination does not answer<br>10. Verify disposition | **Pass** |
| REFER back to service provider, busy, **PSTN** destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to CUBE<br>8. CUBE configured to pass REFER through<br>9. Destination is busy<br>10. Verify disposition | **Pass** |

| | | | |
|---|---|---|---|
| REFER back to service provider, busy, Service Provider SIP destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to CUBE<br>8. CUBE configured to pass REFER through<br>9. Destination is busy<br>10. Verify disposition | **Pass** |
| REFER back to service provider, caller hangs up before completion, **PSTN** destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to CUBE<br>8. CUBE configured to pass REFER through<br>9. Caller hangs up while destination is still ringing<br>10. Verify disposition | **Pass** |
| REFER back to service provider, caller hangs up before completion, SIP destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to CUBE<br>8. CUBE configured to pass REFER through<br>9. Caller hangs up while destination is still ringing<br>10. Verify disposition | **Pass** |
| REFER back to service provider, with GTD forwarded to SP, SIP destination | | 6. For this test case only, set the UserToUserInfo variable<br>7. Agent requests blind transfer<br>8. CVP pulls the call back, sends REFER to CUBE<br>9. CUBE configured to pass REFER through (signaling forward unconditional)<br>9. Verify disposition | **Pass** |
| REFER back to service provider, with GTD blocked by CUBE, SIP destination | | 6. For this test case only, set the UserToUserInfo variable<br>7. Agent requests blind transfer<br>8. CVP pulls the call back, sends REFER to CUBE<br>9. CUBE configured to pass REFER through (signaling forward none)<br>9. Verify disposition | **Pass** |

| | | | |
|---|---|---|---|
| CUBE Consumes REFER, all G711 | Goal: have CUBE send UPDATE message to Service Provider | 6. For this test case only, all phones need to be G711<br>7. Agent (G711) requets blind transfer<br>8. CVP pulls the call back, sends REFER to CUBE<br>9. CUBE configured to consume REFER<br>10. Destination could be another phone in CUCM (G711)<br>11. Destination answers the phone | **pass** |
| Encrypted SIP Phone | | 6. Agent requests consultative transfer to IP toll free number<br>7. Destination answers<br>8. Agent completes the transfer<br>9. Caller hangs up | **Pass** |
| Consulting call over IPCC trunk, agent hangs up | | 6. Agent requests consultative transfer to IP toll free number<br>7. Destination answers<br>8. Agent hangs up | **Pass** |
| Ring No Answer | | 1. Use the "no answer" script<br>2. Dial toll free number<br>3. Verify disposition | **Pass** |
| Busy | | 1. Use the busy script<br>2. Dial toll free number<br>3. Verify disposition | **Pass** |
| Inbound call to IP-IVR with privacy configured | | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call, verify calling number as "anonymous"<br>6.Caller hangs up | **Pass** |

| | | | |
|---|---|---|---|
| DTMF consultative transfer, CVP releases the call releases the call, SIP destination | | 1. Place inbound call<br>2. Call is queued, etc<br>3. Agent answers<br>4. Agent uses his PHONE (NOT the dektop) and dials *8972XXXXXXX. Agent stays on the line<br>5. Destination answers. Verify destination is talking to agent.<br>6. Agent enters *#<br>7. Verify that Verizon tears down call to agent. Destination should continue talking to caller.<br>8. Caller hangs up. | **Pass** |
| NCR | NCR is Verizon's ability to redirect calls in case thy are not answered. | CVP in comprehensive mode. Make sure ICM has a "busy" and a "no answer" script. No IP-IVR used in the first 3 cases. | |
| Regular inbound call | Establish baseline | Use toll free 866-674-7057<br>1. Dial toll free number<br>2. Wait for CVP to answer the call<br>3. Hang up | **Pass** |
| NCR with no answer | | 1. Use the "no answer" script<br>2. Dial toll free number<br>3. Verify that Verizon redirected the call | **Pass** |
| NCR with Busy | | 1. Use the busy script<br>2. Dial toll free number<br>3. Verify that Verizon redirected the call | **Pass** |
| Regular inbound call | Establish baseline | From this test on, use IP-IVR<br>1. Dial toll free number<br>2. Wait for CVP to answer the call<br>3. Hang up | **Pass** |
| NCR with no answer | | 1. Use the "no answer" script<br>2. Dial toll free number<br>3. Verify that Verizon redirected the call | **Pass** |
| NCR with Busy | | 1. Use the busy script<br>2. Dial toll free number<br>3. Verify that Verizon redirected the call | **Pass** |

| | | | |
|---|---|---|---|
| **SBC Failover** | For this entire section, observice if the FIRST call fail. If so, make a note and repeat the test a few times. | SIP over TCP as first focus. If we encounter issues, switch to UDP. | |
| New calls | Establish baseline | 1. Disconnect "primary/active" SBC<br>2. Dial toll free number<br>3. Wait for CVP to answer the call<br>4. Use DTMF to navigate through CVP prompts<br>5. Hang up | **Pass** |
| Call in progress, CVP | Verify impact of SBC failure in calls in progress | 1. Make call in to CVP<br>2. Wait for CVP to answer call<br>3. Disconnect active SBC<br>4. Use DTMF to navigate through CVP prompts<br>5. Hang up, and verify proper tear down (PCAP, no zombie calls on CVP, etc) | **Pass** |
| Disconnect during ring phase | Verify impact of SBC failure in the middle of SIP INVITE process | 1. For this test case only, configure in CVP a "fake" VXML browser (unreachable IP address)<br>2. Dial toll free number<br>3. Disconnect active SBC<br>4. Hang up<br>5. Verify that call was properly torn down (PCAP), and there are no zombie calls in CVP (may take a few seconds) | **Pass** |
| Call in progress, Long Call | Verify impact of SBC failure in session refresh timers | 1. Dial toll free number<br>2. Wait for CVP to answer the call<br>3. Disconnect active SBC<br>4. Keep the call up for at least 30 min (has to be longer than session refresh timer)<br>5. Hang up<br>6. Verify that session was refreshed accordingly | **Pass** |

| | | | |
|---|---|---|---|
| Call in progress, Queue | | 1. Make agent unavailable<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Disconnect active SBC<br>5. Request to talk to agent (queue to skill group in ICM)<br>6. Make agent available<br>7. Agent answers the call<br>8. Caller hangs up | **Pass** |
| Call in progress, Agent | | 1. Make agent unavailable<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Make agent available<br>6. Agent answers the call<br>7. Disconnect active SBC<br>8. Caller hangs up | **Pass** |
| Agent hangs up | | 1. Make agent unavailable<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Make agent available<br>6. Agent answers the call<br>7. Disconnect active SBC<br>8. Agent hangs up | **Pass** |

| | | | |
|---|---|---|---|
| Hold | Verify hold/resume signaling. | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent puts call on hold<br>7. Disconnect active SBC<br>8. After a few seconds, agent resumes call<br>9. Caller hangs up | **Pass** |
| Mute (silence) | Verify behavior during long periods of silence. | 1. Make agent available<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Request to talk to agent (queue to skill group in ICM)<br>5. Agent answers the call<br>6. Agent puts call on mute<br>7. Disconnect active SBC<br>8. After 30 min, agent resumes call<br>9. Caller hangs up | **Pass** |
| SBC Transcoded call | Verify if calls transcoded by SBC can failover gracefully | 1. For this test case only, make sure SBC is not configured for end-to-end codec renegotiation. Internal dial peers offer G711 and G729.<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer (G711 all the way to service provider)<br>5. Queue to skill group<br>6. Agent answers the call<br>7. SBC should insert transcoder (G711 to service provider, G729 internally)<br>8. Disconnect active SBC<br>9. Verify disposition<br>10. Caller hangs up (look for clean disconnect) | **Not Tested** |

| | | | |
|---|---|---|---|
| CUCM Transcoded Call | Verify if SBC failover impacts CUCM when DSPs are in the media path | 1. For this test case only, make sure SBC is not configured for end-to-end codec renegotiation. Internal dial peers offer only G711. Tanscoder configured in CUCM.<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer (G711 all the way to service provider)<br>5. Queue to skill group<br>6. Agent answers the call<br>7. CUCM should insert transcoder (G711 to service provider, G711 from SBC to Transcoder, G729 to agent)<br>8. Disconnect active SBC<br>9. Verify disposition<br>10. Caller hangs up (look for clean disconnect) | **Pass** |
| UDP | Verify any discrepancies when UDP is used | For this test case only, configure SBC to use UDP in the inbound dial peer<br>1. Make agent unavailable<br>2. Place toll free call<br>3. Wait for CVP to answer (self service application)<br>4. Disconnect active SBC<br>5. Request to talk to agent (queue to skill group in ICM)<br>6. Make agent available<br>7. Agent answers the call<br>8. Caller hangs up | **Pass** |

| | | | |
|---|---|---|---|
| SCCP Phone | Verify any differences with SCCP phones | 1. For this test case only, configure a SCCP phone<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer (G711 all the way to service provider)<br>5. Queue call to skill group<br>6. Agent answers the call<br>7. Make sure no DSPs or transcoders are engaged, and G729 is being used to service provider<br>8. Disconnect active SBC, verify call stays up<br>9. Place call on hold and resume, verify call stays up<br>10. Caller hangs up | **Pass** |
| Encrypted SIP Phone | Verify any differences in behavior when encryption is used in phoen signaling | 1. For this test case only, configure an encrypted SIP phone<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer (G711 all the way to service provider)<br>5. Queue call to skill group<br>6. Agent answers the call<br>7. Make sure no DSPs or transcoders are engaged, and G729 is being used to service provider<br>8. Disconnect active SBC, verify call stays up<br>9. Place call on hold and resume, verify call stays up<br>10. Caller hangs up | **Pass** |

| | | | |
|---|---|---|---|
| No auto answer | | 1. For this test case only, do not configure auto-answer<br>2. Make agent available<br>3. Place toll free call<br>4. Wait for CVP to answer<br>5. Queue call to skill group<br>6. Disconnect active SBC while the call is ringing at the agent<br>7. Manually answer the call<br>8. Caller hangs up | **Pass** |
| Blind transfer, Service Provider releases the call, SIP destination (*8 enabled) | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, plays INFO messages to SBC<br>8. SBC converts INFO to RFC2833 tones<br>9. As soon as CVP starts playing the tones, disconnect SBC<br>10. Verify disposition | **Pass** |
| REFER back to service provider, PSTN destination | | 6. Agent requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC<br>8. SBC configured to pass REFER through<br>9. While the call is still ringing at the destination, disconnect active SBC<br>10. Verify disponistion (graceful handoff of NOTIFY?) | **Pass** |
| SBC consumes REFER, destination does not answer | | SBC configured to consume REFER<br>6. Agent (G729 end-to-end) requests blind transfer<br>7. CVP pulls the call back, sends REFER to SBC<br>8. SBC configuted to consume REFER<br>9. Destination rings, and call is not answered<br>10. While call is ringing, disconnect active SBC<br>11. Verify disposition | **Pass** |

| | | | |
|---|---|---|---|
| Mobile Agent, idle | | 1. Log Mobile Agent in<br>2. Make sure connection to phone (over SIP trunk) is established (MTP should be engaged)<br>3. Disconnect active SBC<br>4. Verify failover | **Not Tested** |
| Mobile Agent, on call | | 1. Log Mobile Agent in<br>2. Answer ACD call<br>3. Disconnect active SBC<br>4. Verify failover<br>5. Hold/resume<br>6. Verify call is still there<br>7. Blind transfer call | **Not Tested** |

## 5 Summary and Conclusion

### 5.1 Summary

| No. of Test Cases | Pass | Fail | N/S, N/T |
|---|---|---|---|
| 151 | 136 | 0 | 15 |
| | | | |

### 5.2 Caveats

For Cisco-GUID header support 6.3 GA or later is required and the AvayaCiscoUCID64.4.spl . .4 in the SPL script fixes formatting in the header for Cisco-Guid, not Cisco-GUID, as expected by cisco SIP equipment. Earlier versions are supported, but do not conform to cisco formatting. The only relevance of 6.3 is support for SPL which is not supported in 6.2 or earlier.

Some test required more specific local-policies to route calls. These local-policies were removed so the specifics did not convolute the actions in the configuration.

Lastly this CVP version has a TCP issue when doing stateful failover on the SBC in HA. The SBC during failover sends a TCP RST, after failing over. This causes the CVP to reset the TCP connection and re-transmit the last unanswered SIP message. The issue with CVP delays the re-establishment of the TCP connection for 34 seconds. Cisco is researching this issue. We tested failover with CUCM using TCP and did not see the same issue. There is no issue with UDP. The issue seems to be related to the TCP stack on CVP only.

## 6 Author's Address

Andy Tatum
Acme Packet, Inc.
100 Crosby Drive
Bedford MA 01730
Email: atatum@acmepacket.com

## 7 Disclaimer

The content in this document is for informational purposes only and is subject to change by Acme Packet without notice. While reasonable efforts have been made in the preparation of this publication to assure its accuracy, Acme Packet assumes no liability resulting from technical or editorial errors or omissions, or for any damages resulting from the use of this information. Unless specifically included in a written agreement with Acme Packet, Acme Packet has no obligation to develop or deliver any future release or upgrade or any feature, enhancement or function.

## 8 Full Copyright Statement

# 9 Appendix – A Net-Net SBC Configuration

## 9.1 Net-Net SBC Sample Configuration

```
Acmesystem1# show running-configuration
codec-policy
        name                           PCMU-Only
        allow-codecs                   PCMU
        order-codecs
        last-modified-by               admin@107.2.151.96
        last-modified-date             2012-04-11 18:12:25
host-routes
        dest-network                   10.0.0.0
        netmask                        255.0.0.0
        gateway                        10.64.1.1
        description
        last-modified-by               admin@71.237.114.64
        last-modified-date             2012-03-26 17:50:30
local-policy
        from-address
                                       *
        to-address
                                       *
        source-realm
                                       inside
        description
        activate-time                  N/A
        deactivate-time                N/A
        state                          enabled
        policy-priority                none
        last-modified-by               admin@10.64.201.127
        last-modified-date             2012-03-26 14:52:06
        policy-attribute
                next-hop                       63.79.178.20
                realm                          outside
                action                         none
                terminate-recursion            disabled
                carrier
                start-time                     0000
                end-time                       2400
                days-of-week                   U-S
                cost                           0
                app-protocol
                state                          enabled
                methods
                media-profiles
                lookup                         single
                next-key
                eloc-str-lkup                  disabled
                eloc-str-match
local-policy
        from-address
                                       *
```

**Comment [AT1]:** There were some test cases that involved codec offer manipulation. These codec policies if needed can be applied at the realm-config level.

```
        to-address
                                       *
        source-realm
                                       outside
        description
        activate-time                  N/A
        deactivate-time                N/A
        state                          enabled
        policy-priority                none
        last-modified-by               admin@10.64.1.153
        last-modified-date             2012-04-12 11:25:16
        policy-attribute
                next-hop                       10.64.2.195
                realm                          inside
                action                         replace-uri
                terminate-recursion            enabled
                carrier
                start-time                     0000
                end-time                       2400
                days-of-week                   U-S
                cost                           0
                app-protocol
                state                          enabled
                methods
                media-profiles
                lookup                         single
                next-key
                eloc-str-lkup                  disabled
                eloc-str-match
local-policy
        from-address
                                       *
        to-address
                                       10.70.18.2
        source-realm
                                       outside
        description
        activate-time                  N/A
        deactivate-time                N/A
        state                          enabled
        policy-priority                none
        last-modified-by               admin@216.41.24.2
        last-modified-date             2012-03-30 10:13:11
        policy-attribute
                next-hop                       10.70.18.2
                realm                          inside
                action                         replace-uri
                terminate-recursion            enabled
                carrier
                start-time                     0000
                end-time                       2400
                days-of-week                   U-S
                cost                           0
                app-protocol
                state                          enabled
                methods
                media-profiles
```

**Comment [AT2]:** This is required for CVP and outside the default configuration

**Comment [AT3]:** This is required for CUCM and outside the default configuration

```
                lookup                          single
                next-key
                eloc-str-lkup                   disabled
                eloc-str-match
local-policy
        from-address
                                        10.70.18.2
        to-address
                                        *
        source-realm
                                        inside
        description
        activate-time                   N/A
        deactivate-time                 N/A
        state                           disabled
        policy-priority                 none
        last-modified-by                admin@107.2.151.96
        last-modified-date              2012-04-19 17:46:21
        policy-attribute
                next-hop                        63.79.178.21
                realm                           outside
                action                          none
                terminate-recursion             disabled
                carrier
                start-time                      0000
                end-time                        2400
                days-of-week                    U-S
                cost                            0
                app-protocol
                state                           enabled
                methods
                media-profiles
                lookup                          single
                next-key
                eloc-str-lkup                   disabled
                eloc-str-match
media-manager
        state                           enabled
        latching                        enabled
        flow-time-limit                 86400
        initial-guard-timer             300
        subsq-guard-timer               300
        tcp-flow-time-limit             86400
        tcp-initial-guard-timer         300
        tcp-subsq-guard-timer           300
        tcp-number-of-ports-per-flow    2
        hnt-rtcp                        disabled
        algd-log-level                  NOTICE
        mbcd-log-level                  NOTICE
        red-flow-port                   1985
        red-mgcp-port                   1986
        red-max-trans                   10000
        red-sync-start-time             5000
        red-sync-comp-time              1000
        media-policing                  enabled
        max-signaling-bandwidth         10000000
        max-untrusted-signaling         100
```

```
        min-untrusted-signaling        30
        app-signaling-bandwidth        0
        tolerance-window               30
        rtcp-rate-limit                0
        trap-on-demote-to-deny         enabled
        syslog-on-demote-to-deny       disabled
        syslog-on-demote-to-untrusted  disabled
        anonymous-sdp                  disabled
        arp-msg-bandwidth              32000
        fragment-msg-bandwidth         0
        rfc2833-timestamp              disabled
        default-2833-duration          100
        rfc2833-end-pkts-only-for-non-sig enabled
        translate-non-rfc2833-event    disabled
        media-supervision-traps        disabled
        dnsalg-server-failover         disabled
        last-modified-by               admin@71.237.114.64
        last-modified-date             2012-03-20 18:22:54
network-interface
        name                           M00
        sub-port-id                    0
        description                    Inside interface to Cisco
        hostname
        ip-address                     10.64.2.194
        pri-utility-addr               10.64.2.218
        sec-utility-addr               10.64.2.2
        netmask                        255.0.0.0
        gateway                        10.64.1.1
        sec-gateway
        gw-heartbeat
                state                          disabled
                heartbeat                      0
                retry-count                    0
                retry-timeout                  1
                health-score                   0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                    11
        hip-ip-list
        ftp-address
        icmp-address
        snmp-address
        telnet-address
        ssh-address
        signaling-mtu                  0
        last-modified-by               admin@107.2.151.96
        last-modified-date             2012-05-03 10:11:46
network-interface
        name                           M10
        sub-port-id                    0
        description                    Outside interface to Verizon
        hostname
        ip-address                     174.46.0.214
        pri-utility-addr               174.46.0.152
        sec-utility-addr               174.46.0.201
```

```
        netmask                         255.255.255.128
        gateway                         174.46.0.129
        sec-gateway
        gw-heartbeat
                state                           disabled
                heartbeat                       0
                retry-count                     0
                retry-timeout                   1
                health-score                    0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                     11
        hip-ip-list
        ftp-address
        icmp-address
        snmp-address
        telnet-address
        ssh-address
        signaling-mtu                   0
        last-modified-by                admin@107.2.151.96
        last-modified-date              2012-05-03 10:12:27
network-interface
        name                            wancom1
        sub-port-id                     0
        description
        hostname
        ip-address
        pri-utility-addr                169.254.1.1
        sec-utility-addr                169.254.1.2
        netmask                         255.255.255.252
        gateway
        sec-gateway
        gw-heartbeat
                state                           disabled
                heartbeat                       0
                retry-count                     0
                retry-timeout                   1
                health-score                    0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                     11
        hip-ip-list
        ftp-address
        icmp-address
        snmp-address
        telnet-address
        ssh-address
        signaling-mtu                   0
        last-modified-by                admin@71.237.114.64
        last-modified-date              2012-03-20 13:32:47
network-interface
        name                            wancom2
        sub-port-id                     0
```

```
        description
        hostname
        ip-address
        pri-utility-addr                169.254.2.1
        sec-utility-addr                169.254.2.2
        netmask                         255.255.255.252
        gateway
        sec-gateway
        gw-heartbeat
                state                           disabled
                heartbeat                       0
                retry-count                     0
                retry-timeout                   1
                health-score                    0
        dns-ip-primary
        dns-ip-backup1
        dns-ip-backup2
        dns-domain
        dns-timeout                     11
        hip-ip-list
        ftp-address
        icmp-address
        snmp-address
        telnet-address
        ssh-address
        signaling-mtu                   0
        last-modified-by                admin@71.237.114.64
        last-modified-date              2012-03-20 13:34:33
phy-interface
        name                            M00
        operation-type                  Media
        port                            0
        slot                            0
        virtual-mac                     00:08:25:04:0c:fe
        admin-state                     enabled
        auto-negotiation                enabled
        duplex-mode                     FULL
        speed                           100
        overload-protection             disabled
        last-modified-by                admin@71.237.114.64
        last-modified-date              2012-03-20 14:51:49
phy-interface
        name                            M10
        operation-type                  Media
        port                            0
        slot                            1
        virtual-mac                     00:08:25:04:0c:ff
        admin-state                     enabled
        auto-negotiation                enabled
        duplex-mode                     FULL
        speed                           100
        overload-protection             disabled
        last-modified-by                admin@71.237.114.64
        last-modified-date              2012-03-20 14:51:19
phy-interface
        name                            wancom1
        operation-type                  Control
```

```
        port                            1
        slot                            0
        virtual-mac
        wancom-health-score             8
        overload-protection             disabled
        last-modified-by                admin@71.237.114.64
        last-modified-date              2012-03-20 13:26:42
phy-interface
        name                            wancom2
        operation-type                  Control
        port                            2
        slot                            0
        virtual-mac
        wancom-health-score             9
        overload-protection             disabled
        last-modified-by                admin@71.237.114.64
        last-modified-date              2012-03-20 13:28:06
realm-config
        identifier                      inside
        description                     realm to Cisco cvp/cucm
        addr-prefix                     10.0.0.0
        network-interfaces
                                        M00:0
        mm-in-realm                     disabled
        mm-in-network                   enabled
        mm-same-ip                      enabled
        mm-in-system                    enabled
        bw-cac-non-mm                   disabled
        msm-release                     disabled
        qos-enable                      disabled
        generate-UDP-checksum           disabled
        max-bandwidth                   0
        fallback-bandwidth              0
        max-priority-bandwidth          0
        max-latency                     0
        max-jitter                      0
        max-packet-loss                 0
        observ-window-size              0
        parent-realm
        dns-realm
        media-policy
        media-sec-policy
        srtp-msm-passthrough            disabled
        in-translationid
        out-translationid
        in-manipulationid
        out-manipulationid
        manipulation-string
        manipulation-pattern
        class-profile
        average-rate-limit              0
        access-control-trust-level      none
        invalid-signal-threshold        0
        maximum-signal-threshold        0
        untrusted-signal-threshold      0
        nat-trust-threshold             0
        deny-period                     30
```

```
        cac-failure-threshold          0
        untrust-cac-failure-threshold  0
        ext-policy-svr
        diam-e2-address-realm
        symmetric-latching             disabled
        pai-strip                      disabled
        trunk-context
        early-media-allow
        enforcement-profile
        additional-prefixes
        restricted-latching            none
        restriction-mask               32
        spl-options
        accounting-enable              enabled
        user-cac-mode                  none
        user-cac-bandwidth             0
        user-cac-sessions              0
        icmp-detect-multiplier         0
        icmp-advertisement-interval    0
        icmp-target-ip
        monthly-minutes                0
        net-management-control         disabled
        delay-media-update             disabled
        refer-call-transfer            disabled
        refer-notify-provisional       none
        dyn-refer-term                 disabled
        codec-policy
        codec-manip-in-realm           disabled
        constraint-name
        call-recording-server-id
        xnq-state                      xnq-unknown
        hairpin-id                     0
        stun-enable                    disabled
        stun-server-ip                 0.0.0.0
        stun-server-port               3478
        stun-changed-ip                0.0.0.0
        stun-changed-port              3479
        match-media-profiles
        qos-constraint
        sip-profile
        sip-isup-profile
        block-rtcp                     disabled
        hide-egress-media-update       disabled
        last-modified-by               admin@107.2.151.96
        last-modified-date             2012-05-03 10:14:09
realm-config
        identifier                     outside
        description                    Realm to Verizon
        addr-prefix                    63.79.178.0
        network-interfaces
                                       M10:0
        mm-in-realm                    disabled
        mm-in-network                  enabled
        mm-same-ip                     enabled
        mm-in-system                   enabled
        bw-cac-non-mm                  disabled
        msm-release                    disabled
```

```
qos-enable                      disabled
generate-UDP-checksum           disabled
max-bandwidth                   0
fallback-bandwidth              0
max-priority-bandwidth          0
max-latency                     0
max-jitter                      0
max-packet-loss                 0
observ-window-size              0
parent-realm
dns-realm
media-policy
media-sec-policy
srtp-msm-passthrough            disabled
in-translationid
out-translationid
in-manipulationid
out-manipulationid
manipulation-string
manipulation-pattern
class-profile
average-rate-limit              0
access-control-trust-level      none
invalid-signal-threshold        0
maximum-signal-threshold        0
untrusted-signal-threshold      0
nat-trust-threshold             0
deny-period                     30
cac-failure-threshold           0
untrust-cac-failure-threshold   0
ext-policy-svr
diam-e2-address-realm
symmetric-latching              disabled
pai-strip                       disabled
trunk-context
early-media-allow
enforcement-profile
additional-prefixes
restricted-latching             none
restriction-mask                32
spl-options
accounting-enable               enabled
user-cac-mode                   none
user-cac-bandwidth              0
user-cac-sessions               0
icmp-detect-multiplier          0
icmp-advertisement-interval     0
icmp-target-ip
monthly-minutes                 0
net-management-control          disabled
delay-media-update              disabled
refer-call-transfer             disabled
refer-notify-provisional        none
dyn-refer-term                  disabled
codec-policy
codec-manip-in-realm            disabled
constraint-name
```

```
        call-recording-server-id
        xnq-state                      xnq-unknown
        hairpin-id                     0
        stun-enable                    disabled
        stun-server-ip                 0.0.0.0
        stun-server-port               3478
        stun-changed-ip                0.0.0.0
        stun-changed-port              3479
        match-media-profiles
        qos-constraint
        sip-profile
        sip-isup-profile
        block-rtcp                     disabled
        hide-egress-media-update       disabled
        last-modified-by               admin@107.2.151.96
        last-modified-date             2012-05-03 10:14:46
redundancy-config
        state                          enabled
        log-level                      INFO
        health-threshold               75
        emergency-threshold            50
        port                           9090
        advertisement-time             500
        percent-drift                  210
        initial-time                   1250
        becoming-standby-time          180000
        becoming-active-time           100
        cfg-port                       1987
        cfg-max-trans                  10000
        cfg-sync-start-time            5000
        cfg-sync-comp-time             1000
        gateway-heartbeat-interval     0
        gateway-heartbeat-retry        0
        gateway-heartbeat-timeout      1
        gateway-heartbeat-health       1
        media-if-peercheck-time        0
        peer
                name                           acmesystem2
                state                          enabled
                type                           Secondary
                destination
                        address                        169.254.1.2:9090
                        network-interface              wancom1:0
                destination
                        address                        169.254.2.2:9090
                        network-interface              wancom2:0
        peer
                name                           acmesystem1
                state                          enabled
                type                           Primary
                destination
                        address                        169.254.1.1:9090
                        network-interface              wancom1:0
                destination
                        address                        169.254.2.1:9090
                        network-interface              wancom2:0
        last-modified-by               admin@71.237.114.64
```

```
        last-modified-date             2012-03-20 13:49:26
session-agent
        hostname                       10.64.2.195
        ip-address                     10.64.2.195
        port                           5060
        state                          enabled
        app-protocol                   SIP
        app-type
        transport-method               StaticTCP
        realm-id                       inside
        egress-realm-id
        description                    CVP
        carriers
        allow-next-hop-lp              enabled
        constraints                    disabled
        max-sessions                   0
        max-inbound-sessions           0
        max-outbound-sessions          0
        max-burst-rate                 0
        max-inbound-burst-rate         0
        max-outbound-burst-rate        0
        max-sustain-rate               0
        max-inbound-sustain-rate       0
        max-outbound-sustain-rate      0
        min-seizures                   5
        min-asr                        0
        time-to-resume                 0
        ttr-no-response                0
        in-service-period              0
        burst-rate-window              0
        sustain-rate-window            0
        req-uri-carrier-mode           None
        proxy-mode
        redirect-action
        loose-routing                  disabled
        send-media-session             enabled
        response-map
        ping-method                    OPTIONS
        ping-interval                  90
        ping-send-mode                 keep-alive
        ping-all-addresses             disabled
        ping-in-service-response-codes
        out-service-response-codes
        load-balance-dns-query         hunt
        media-profiles
        spl-options
        in-translationid
        out-translationid
        trust-me                       disabled
        request-uri-headers
        stop-recurse                   300-399,404-599
        local-response-map
        ping-to-user-part
        ping-from-user-part
        li-trust-me                    disabled
        in-manipulationid
        out-manipulationid             CVP_Manip
```

**Comment [AT4]:** Required for CVP

```
        manipulation-string
        manipulation-pattern
        p-asserted-id
        trunk-group
        max-register-sustain-rate      0
        early-media-allow
        invalidate-registrations       disabled
        rfc2833-mode                   none
        rfc2833-payload                101
        codec-policy
        enforcement-profile
        refer-call-transfer            enabled
        refer-notify-provisional       none
        reuse-connections              NONE
        tcp-keepalive                  none
        tcp-reconn-interval            0
        max-register-burst-rate        0
        register-burst-window          0
        sip-profile
        sip-isup-profile
        kpml-interworking              inherit
        last-modified-by               admin@107.2.151.96
        last-modified-date             2012-05-09 12:50:04
session-agent
        hostname                       63.79.178.21
        ip-address                     63.79.178.21
        port                           5060
        state                          enabled
        app-protocol                   SIP
        app-type
        transport-method               UDP
        realm-id                       outside
        egress-realm-id
        description                    Verizon trunk1
        carriers
        allow-next-hop-lp              enabled
        constraints                    disabled
        max-sessions                   0
        max-inbound-sessions           0
        max-outbound-sessions          0
        max-burst-rate                 0
        max-inbound-burst-rate         0
        max-outbound-burst-rate        0
        max-sustain-rate               0
        max-inbound-sustain-rate       0
        max-outbound-sustain-rate      0
        min-seizures                   5
        min-asr                        0
        time-to-resume                 0
        ttr-no-response                0
        in-service-period              0
        burst-rate-window              0
        sustain-rate-window            0
        req-uri-carrier-mode           None
        proxy-mode
        redirect-action
        loose-routing                  disabled
```

```
        send-media-session              enabled
        response-map
        ping-method                     OPTIONS
        ping-interval                   90
        ping-send-mode                  keep-alive
        ping-all-addresses              disabled
        ping-in-service-response-codes
        out-service-response-codes
        load-balance-dns-query          hunt
        media-profiles
        spl-options
        in-translationid
        out-translationid
        trust-me                        disabled
        request-uri-headers
        stop-recurse
        local-response-map
        ping-to-user-part
        ping-from-user-part
        li-trust-me                     disabled
        in-manipulationid
        out-manipulationid              Verizon_Manip
        manipulation-string
        manipulation-pattern
        p-asserted-id
        trunk-group
        max-register-sustain-rate       0
        early-media-allow
        invalidate-registrations        disabled
        rfc2833-mode                    none
        rfc2833-payload                 101
        codec-policy
        enforcement-profile
        refer-call-transfer             disabled
        refer-notify-provisional        none
        reuse-connections               NONE
        tcp-keepalive                   none
        tcp-reconn-interval             0
        max-register-burst-rate         0
        register-burst-window           0
        sip-profile
        sip-isup-profile
        kpml-interworking               inherit
        last-modified-by                admin@107.2.151.96
        last-modified-date              2012-05-09 12:50:14
session-agent
        hostname                        10.70.18.2
        ip-address                      10.70.18.2
        port                            5060
        state                           enabled
        app-protocol                    SIP
        app-type
        transport-method                StaticTCP
        realm-id                        inside
        egress-realm-id
        description                     CUCM
        carriers
```

```
allow-next-hop-lp              enabled
constraints                    disabled
max-sessions                   0
max-inbound-sessions           0
max-outbound-sessions          0
max-burst-rate                 0
max-inbound-burst-rate         0
max-outbound-burst-rate        0
max-sustain-rate               0
max-inbound-sustain-rate       0
max-outbound-sustain-rate      0
min-seizures                   5
min-asr                        0
time-to-resume                 0
ttr-no-response                0
in-service-period              0
burst-rate-window              0
sustain-rate-window            0
req-uri-carrier-mode           None
proxy-mode
redirect-action
loose-routing                  enabled
send-media-session             enabled
response-map
ping-method                    OPTIONS
ping-interval                  90
ping-send-mode                 keep-alive
ping-all-addresses             disabled
ping-in-service-response-codes
out-service-response-codes
load-balance-dns-query         hunt
media-profiles
spl-options
in-translationid
out-translationid
trust-me                       disabled
request-uri-headers
stop-recurse
local-response-map
ping-to-user-part
ping-from-user-part
li-trust-me                    disabled
in-manipulationid
out-manipulationid             CUCM-Out
manipulation-string
manipulation-pattern
p-asserted-id
trunk-group
max-register-sustain-rate      0
early-media-allow
invalidate-registrations       disabled
rfc2833-mode                   none
rfc2833-payload                0
codec-policy
enforcement-profile
refer-call-transfer            disabled
refer-notify-provisional       none
```

```
        reuse-connections               NONE
        tcp-keepalive                   none
        tcp-reconn-interval             0
        max-register-burst-rate         0
        register-burst-window           0
        sip-profile
        sip-isup-profile
        kpml-interworking               inherit
        last-modified-by                admin@107.2.151.96
        last-modified-date              2012-05-09 12:50:23
session-agent
        hostname                        63.79.178.20
        ip-address                      63.79.178.20
        port                            5060
        state                           enabled
        app-protocol                    SIP
        app-type
        transport-method                UDP
        realm-id                        outside
        egress-realm-id
        description                     Verizon Tunk2
        carriers
        allow-next-hop-lp               enabled
        constraints                     disabled
        max-sessions                    0
        max-inbound-sessions            0
        max-outbound-sessions           0
        max-burst-rate                  0
        max-inbound-burst-rate          0
        max-outbound-burst-rate         0
        max-sustain-rate                0
        max-inbound-sustain-rate        0
        max-outbound-sustain-rate       0
        min-seizures                    5
        min-asr                         0
        time-to-resume                  0
        ttr-no-response                 0
        in-service-period               0
        burst-rate-window               0
        sustain-rate-window             0
        req-uri-carrier-mode            None
        proxy-mode
        redirect-action
        loose-routing                   enabled
        send-media-session              enabled
        response-map
        ping-method                     OPTIONS
        ping-interval                   90
        ping-send-mode                  keep-alive
        ping-all-addresses              disabled
        ping-in-service-response-codes
        out-service-response-codes
        load-balance-dns-query          hunt
        media-profiles
        spl-options
        in-translationid
        out-translationid
```

```
        trust-me                       disabled
        request-uri-headers
        stop-recurse
        local-response-map
        ping-to-user-part
        ping-from-user-part
        li-trust-me                    disabled
        in-manipulationid
        out-manipulationid             Verizon_Manip
        manipulation-string
        manipulation-pattern
        p-asserted-id
        trunk-group
        max-register-sustain-rate      0
        early-media-allow
        invalidate-registrations       disabled
        rfc2833-mode                   none
        rfc2833-payload                0
        codec-policy
        enforcement-profile
        refer-call-transfer            disabled
        refer-notify-provisional       none
        reuse-connections              NONE
        tcp-keepalive                  none
        tcp-reconn-interval            0
        max-register-burst-rate        0
        register-burst-window          0
        sip-profile
        sip-isup-profile
        kpml-interworking              inherit
        last-modified-by               admin@107.2.151.96
        last-modified-date             2012-05-03 10:20:07
sip-config
        state                          enabled
        operation-mode                 dialog
        dialog-transparency            enabled
        home-realm-id
        egress-realm-id
        nat-mode                       None
        registrar-domain
        registrar-host
        registrar-port                 0
        register-service-route         always
        init-timer                     500
        max-timer                      4000
        trans-expire                   32
        invite-expire                  180
        inactive-dynamic-conn          32
        enforcement-profile
        pac-method
        pac-interval                   10
        pac-strategy                   PropDist
        pac-load-weight                1
        pac-session-weight             1
        pac-route-weight               1
        pac-callid-lifetime            600
        pac-user-lifetime              3600
```

```
        red-sip-port                    1988
        red-max-trans                   10000
        red-sync-start-time             5000
        red-sync-comp-time              1000
        add-reason-header               disabled
        sip-message-len                 4096
        enum-sag-match                  disabled
        extra-method-stats              disabled
        rph-feature                     disabled
        nsep-user-sessions-rate         0
        nsep-sa-sessions-rate           0
        registration-cache-limit        0
        register-use-to-for-lp          disabled
        refer-src-routing               disabled
        add-ucid-header                 disabled
        proxy-sub-events
        allow-pani-for-trusted-only     disabled
        pass-gruu-contact               disabled
        sag-lookup-on-redirect          disabled
        set-disconnect-time-on-bye      disabled
        last-modified-by                admin@198.178.8.81
        last-modified-date              2012-04-12 12:58:45
sip-interface
        state                           enabled
        realm-id                        inside
        description                     singnalling for inside realm
        sip-port
                address                         10.64.2.194
                port                            5060
                transport-protocol              TCP
                tls-profile
                multi-home-addrs
                allow-anonymous                 agents-only
                ims-aka-profile
        sip-port
                address                         10.64.2.194
                port                            5060
                transport-protocol              UDP
                tls-profile
                multi-home-addrs
                allow-anonymous                 agents-only
                ims-aka-profile
        carriers
        trans-expire                    0
        invite-expire                   0
        max-redirect-contacts           0
        proxy-mode
        redirect-action
        contact-mode                    strict-route
        nat-traversal                   none
        nat-interval                    30
        tcp-nat-interval                90
        registration-caching            disabled
        min-reg-expire                  300
        registration-interval           3600
        route-to-registrar              disabled
        secured-network                 disabled
```

**Comment [AT5]:** Strict routing is a request from Cisco, and is different from the default SBC configurations

```
        teluri-scheme                  disabled
        uri-fqdn-domain
        spl-options                    GUID-Node-ID=0x000825a05f30
        trust-mode                     all
        max-nat-interval               3600
        nat-int-increment              10
        nat-test-increment             30
        sip-dynamic-hnt                disabled
        stop-recurse                   401,407
        port-map-start                 0
        port-map-end                   0
        in-manipulationid
        out-manipulationid
        manipulation-string
        manipulation-pattern
        sip-ims-feature                disabled
        subscribe-reg-event            disabled
        operator-identifier
        anonymous-priority             none
        max-incoming-conns             0
        per-src-ip-max-incoming-conns  0
        inactive-conn-timeout          0
        untrusted-conn-timeout         0
        network-id
        ext-policy-server
        default-location-string
        charging-vector-mode           pass
        charging-function-address-mode pass
        ccf-address
        ecf-address
        term-tgrp-mode                 none
        implicit-service-route         disabled
        rfc2833-payload                101
        rfc2833-mode                   dual
        constraint-name
        response-map
        local-response-map
        ims-aka-feature                disabled
        enforcement-profile
        route-unauthorized-calls
        tcp-keepalive                  none
        add-sdp-invite                 disabled
        add-sdp-profiles               GUID-Node-ID=0x000825a05f30
        sip-profile
        sip-isup-profile
        tcp-conn-dereg                 0
        register-keep-alive            none
        kpml-interworking              disabled
        tunnel-name
        last-modified-by               admin@107.2.151.96
        last-modified-date             2012-05-03 10:16:27
sip-interface
        state                          enabled
        realm-id                       outside
        description                    signalling for outside realm
        sip-port
                address                        174.46.0.214
```

**Comment [AT6]:** SPL is only available in 6.3 and later. When this SPL option is created it automatically creates “add-sdp-profile”. This option can be also added globally at the spl-config

**Comment [AT7]:** The GUID-Node-ID is added in this field automatically when it’s added in the spl-options on the sip-interface

```
            port                              5060
            transport-protocol                UDP
            tls-profile
            multi-home-addrs
            allow-anonymous                   agents-only
            ims-aka-profile
    sip-port
            address                           174.46.0.214
            port                              5060
            transport-protocol                TCP
            tls-profile
            multi-home-addrs
            allow-anonymous                   agents-only
            ims-aka-profile
    carriers
    trans-expire                      0
    invite-expire                     0
    max-redirect-contacts             0
    proxy-mode
    redirect-action
    contact-mode                      none
    nat-traversal                     none
    nat-interval                      30
    tcp-nat-interval                  90
    registration-caching              disabled
    min-reg-expire                    300
    registration-interval             3600
    route-to-registrar                disabled
    secured-network                   disabled
    teluri-scheme                     disabled
    uri-fqdn-domain
    options                           100rel-interworking
    spl-options
    trust-mode                        all
    max-nat-interval                  3600
    nat-int-increment                 10
    nat-test-increment                30
    sip-dynamic-hnt                   disabled
    stop-recurse                      401,407
    port-map-start                    0
    port-map-end                      0
    in-manipulationid
    out-manipulationid
    manipulation-string
    manipulation-pattern
    sip-ims-feature                   disabled
    subscribe-reg-event               disabled
    operator-identifier
    anonymous-priority                none
    max-incoming-conns                0
    per-src-ip-max-incoming-conns     0
    inactive-conn-timeout             0
    untrusted-conn-timeout            0
    network-id
    ext-policy-server
    default-location-string
    charging-vector-mode              pass
```

**Comment [AT8]:** This option feature is added because PRACK is not supported by default by CVP. The SBC will internetwork Rel100 on the Verizon interface with no Rel 100 on the Cisco SIP interface

```
        charging-function-address-mode pass
        ccf-address
        ecf-address
        term-tgrp-mode                 none
        implicit-service-route         disabled
        rfc2833-payload                101
        rfc2833-mode                   preferred
        constraint-name
        response-map
        local-response-map
        ims-aka-feature                disabled
        enforcement-profile
        route-unauthorized-calls
        tcp-keepalive                  none
        add-sdp-invite                 disabled
        add-sdp-profiles
        sip-profile
        sip-isup-profile
        tcp-conn-dereg                 0
        register-keep-alive            none
        kpml-interworking              disabled
        tunnel-name
        last-modified-by               admin@107.2.151.96
        last-modified-date             2012-05-03 10:17:03
sip-manipulation
        name                           Verizon_Manip
        description
        split-headers
        join-headers
        header-rule
                name                           refer_to_prepend
                header-name                    Refer-To
                action                         manipulate
                comparison-type                case-sensitive
                msg-type                       request
                methods                        REFER
                match-value
                new-value
                element-rule
                        name                           remove_add_plus_1
                        parameter-name
                        type                           uri-user
                        action                         replace
                        match-val-type                 any
                        comparison-type                pattern-rule
                        match-value
                        new-value                      "\+1"+$ORIGINAL
                element-rule
                        name                           changeuriuser
                        parameter-name
                        type                           uri-host
                        action                         none
                        match-val-type                 any
                        comparison-type                case-sensitive
                        match-value
                        new-value                      63.79.178.21
        header-rule
```

```
            name                           remove_content_from_refer
            header-name                    Content-Type
            action                         none
            comparison-type                case-sensitive
            msg-type                       any
            methods                        REFER
            match-value
            new-value
    header-rule
            name                           modACKsendonly
            header-name                    Content-type
            action                         manipulate
            comparison-type                case-sensitive
            msg-type                       any
            methods                        ACK
            match-value
            new-value
            element-rule
                    name                           modACK
                    parameter-name                 application/sdp
                    type                           mime
                    action                         find-replace-all
                    match-val-type                 any
                    comparison-type                case-sensitive
                    match-value                    sendonly
                    new-value                      sendrecv
            element-rule
                    name                           modACK_inactive
                    parameter-name                 application/sdp
                    type                           mime
                    action                         find-replace-all
                    match-val-type                 any
                    comparison-type                case-sensitive
                    match-value                    inactive
                    new-value                      sendrecv
    header-rule
            name                           addplusReferto
            header-name                    Refer-to
            action                         none
            comparison-type                case-sensitive
            msg-type                       request
            methods                        REFER
            match-value
            new-value
            element-rule
                    name                           addplus
                    parameter-name
                    type                           uri-user
                    action                         replace
                    match-val-type                 any
                    comparison-type                case-sensitive
                    match-value
                    new-value                      "\+1"+$ORIGINAL
            element-rule
                    name                           changeuriuser
                    parameter-name
                    type                           uri-host
```

```
                        action                          none
                        match-val-type                  any
                        comparison-type                 case-sensitive
                        match-value
                        new-value                       63.79.178.21
        last-modified-by                admin@107.2.151.96
        last-modified-date              2012-04-25 15:40:13
sip-manipulation
        name                            CUCM-Out
        description
        split-headers
        join-headers
        header-rule
                name                            ModTo
                header-name                     To
                action                          manipulate
                comparison-type                 case-sensitive
                msg-type                        request
                methods                         INVITE
                match-value
                new-value
                element-rule
                        name                            ModTo
                        parameter-name
                        type                            uri-host
                        action                          find-replace-all
                        match-val-type                  any
                        comparison-type                 case-sensitive
                        match-value
                        new-value                       $REMOTE_IP
        header-rule
                name                            ModURI
                header-name                     request-uri
                action                          manipulate
                comparison-type                 case-sensitive
                msg-type                        request
                methods                         INVITE
                match-value
                new-value
                element-rule
                        name                            ModURIUser
                        parameter-name
                        type                            uri-host
                        action                          find-replace-all
                        match-val-type                  any
                        comparison-type                 case-sensitive
                        match-value
                        new-value                       $REMOTE_IP
        last-modified-by                admin@107.2.151.96
        last-modified-date              2012-04-17 18:04:53
sip-manipulation
        name                            CVP_Manip
        description
        split-headers
        join-headers
        header-rule
                name                            PAI_Store
```

```
            header-name                    P-Asserted-Identity
            action                         store
            comparison-type                case-sensitive
            msg-type                       any
            methods                        INVITE
            match-value
            new-value
        header-rule
            name                           Add_RPID
            header-name                    Remote-Party-ID
            action                         add
            comparison-type                boolean
            msg-type                       any
            methods                        INVITE
            match-value                    $PAI_Store
            new-value
$PAI_Store.$0+;party=calling;screen=no;privacy=off
            element-rule
                name                           Fix_URI_Display
                parameter-name
                type                           uri-display
                action                         replace
                match-val-type                 any
                comparison-type                pattern-rule
                match-value                    (\w+)
                new-value                      $1+\-\-CVP+" "
        header-rule
            name                           Store_From
            header-name                    From
            action                         store
            comparison-type                case-sensitive
            msg-type                       any
            methods                        INVITE
            match-value
            new-value
        header-rule
            name                           Add_RPID_No_PAI
            header-name                    Remote-Party-ID
            action                         add
            comparison-type                boolean
            msg-type                       any
            methods                        INVITE
            match-value                    !$PAI_Store
            new-value
$Store_From.$0+;party=calling;screen=no;privacy=off
            element-rule
                name                           Fix_URI_Display
                parameter-name
                type                           uri-display
                action                         replace
                match-val-type                 any
                comparison-type                pattern-rule
                match-value                    \"?(\w+)\"?
                new-value                      $1+\-\-CVP+" "
            element-rule
                name                           remove_tag_param
                parameter-name                 tag
```

```
                type                            header-param
                action                          delete-element
                match-val-type                  any
                comparison-type                 case-sensitive
                match-value
                new-value
        element-rule
                name                            Chk_Add_URI_Dsply
                parameter-name
                type                            uri-display
                action                          add
                match-val-type                  any
                comparison-type                 boolean
                match-value                     .*
                new-value                       \-\-CVP+" "
    header-rule
        name                            modRPIDurihost
        header-name                     Remote-Party-ID
        action                          manipulate
        comparison-type                 case-sensitive
        msg-type                        any
        methods                         INVITE
        match-value
        new-value
        element-rule
                name                            modRPIDhost
                parameter-name
                type                            uri-host
                action                          find-replace-all
                match-val-type                  any
                comparison-type                 case-sensitive
                match-value
                new-value                       $LOCAL_IP
    header-rule
        name                            modTo
        header-name                     To
        action                          manipulate
        comparison-type                 case-sensitive
        msg-type                        request
        methods                         INVITE
        match-value
        new-value
        element-rule
                name                            modTo
                parameter-name
                type                            uri-host
                action                          find-replace-all
                match-val-type                  any
                comparison-type                 case-sensitive
                match-value
                new-value                       $REMOTE_IP
    last-modified-by                admin@107.2.151.96
    last-modified-date              2012-05-08 10:41:23
spl-config
    spl-options
    plugins
        name                            AvayaCiscoUCID64.4.spl
```

**Comment [AT9]:** This is only supported in 6.3 GA and later. The SPL must be uploaded to /code/spl/ on the SBC. It is a requirement that the script be signed. Only signed spl scripts should have the extension of .spl . "show spl" will verify the SBC has accepted the spl: i.e: acmesystem2# sho spl
SPL Version: C2.0.0

[sipd] File: AvayaCiscoUCID64.4.spl version: 4 signature: signed and valid "

```
        last-modified-by                admin@107.2.151.96
        last-modified-date              2012-04-24 18:09:19
steering-pool
        ip-address                      10.64.2.194
        start-port                      16384
        end-port                        32767
        realm-id                        inside
        network-interface
        last-modified-by                admin@71.237.114.64
        last-modified-date              2012-03-20 18:24:45
steering-pool
        ip-address                      174.46.0.214
        start-port                      16384
        end-port                        32767
        realm-id                        outside
        network-interface
        last-modified-by                admin@71.237.114.64
        last-modified-date              2012-03-20 18:25:16
system-config
        hostname
        description
        location
        mib-system-contact
        mib-system-name
        mib-system-location
        snmp-enabled                    enabled
        enable-snmp-auth-traps          disabled
        enable-snmp-syslog-notify       disabled
        enable-snmp-monitor-traps       disabled
        enable-env-monitor-traps        disabled
        snmp-syslog-his-table-length    1
        snmp-syslog-level               WARNING
        system-log-level                WARNING
        process-log-level               NOTICE
        process-log-ip-address          0.0.0.0
        process-log-port                0
        collect
                sample-interval                 5
                push-interval                   15
                boot-state                      disabled
                start-time                      now
                end-time                        never
                red-collect-state               disabled
                red-max-trans                   1000
                red-sync-start-time             5000
                red-sync-comp-time              1000
                push-success-trap-state         disabled
        call-trace                      disabled
        internal-trace                  disabled
        log-filter                      all
        default-gateway                 0.0.0.0
        restart                         enabled
        exceptions
        telnet-timeout                  0
        console-timeout                 0
        remote-control                  enabled
        cli-audit-trail                 enabled
```

```
link-redundancy-state           disabled
source-routing                  disabled
cli-more                        disabled
terminal-height                 24
debug-timeout                   0
trap-event-lifetime             0
default-v6-gateway              ::
ipv6-signaling-mtu              1500
ipv4-signaling-mtu              1500
cleanup-time-of-day             00:00
snmp-engine-id-suffix
snmp-agent-mode                 v1v2
last-modified-by                admin@216.41.24.2
last-modified-date              2010-12-16 13:04:37
```