Cisco Unified Customer Voice Portal

Overview and Design - Mario Gianni
Agenda

- Overview
- Deployment Models
- HW Platform, Operations and Reporting
- Sizing
- CUBE Support
- SIP Design
- CVP 8.0 Planned Features
Overview
Cisco’s Market Position Accelerating

Cisco #2 in Worldwide Market Share for Agent Seats and IVR Ports
(Sources: Gartner, Tern Systems)

Source: Cisco analysis
Overview

- Provides ASR/TTS
- VoiceXML application development
- J2EE runtime for serving VoiceXML

Customers

3rd Party Speech Server

CVP VoiceXML Server

Cisco IOS VoiceXML Browser

- Voice Gateway
- Interprets VoiceXML
- DTMF and Speech

Run-time or IP

CVP Call Control

- Telephony switching and on-net queuing

Enterprise Infrastructure

Agents
Detailed Components

- ASR
- TTS
- Media Server
- MRCP
- HTTP
- H.323 or SIP
- SIP Proxy
- CUCM
- TDM
- ACD
- Ops Console
- External Data Sources
- CVP Studio
- Backend Interface
- App GW, SQL, etc.
- GED 125
- DMP
- VRU PG
- ICM
- Informix DB
- CVM Call Server
- CVP VXML Server
- CVP Reporting Server
- ICM Script Editor
Video Integration

- The UCVP 7.0 release adds VIDEO self-service and queuing capabilities to UCVP, fully-integrated with UCCE
- Variety of endpoints supported including 3G Mobile, Video Kiosk, Web, and Soft-clients
- Provided through integration with video components from RADVISION®
- Video-enabled customer service
  - Agents viewable by customer
  - Agents can “push” informational videos to customer
  - Video automatically played during self-service interactions
## CVP Vs. IP-IVR Comparison

<table>
<thead>
<tr>
<th></th>
<th>CVP 7.0</th>
<th>IP-IVR</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Number of Ports per Server</strong></td>
<td>850 SIP 500 H.323</td>
<td>300 (all protocols)</td>
</tr>
<tr>
<td><strong>Gateway Call Control Protocol</strong></td>
<td>SIP/H.323</td>
<td>SIP/H.323/MGCP</td>
</tr>
<tr>
<td><strong>ASR/TTS</strong></td>
<td>MRCP Interface at Voice Gateway</td>
<td>MRCP Interface on IP-IVR Server</td>
</tr>
<tr>
<td><strong>Call Control</strong></td>
<td>H.323/SIP under direction of ICM</td>
<td>JTAPI Under Direction of UC Manager</td>
</tr>
<tr>
<td><strong>Scripting Tool</strong></td>
<td>Call Studio (Eclipse-Based)</td>
<td>CRS Editor</td>
</tr>
<tr>
<td><strong>Integration Options</strong></td>
<td>Java (J2EE) Tomcat Websphere</td>
<td>Custom Java Objects ODBC Databases</td>
</tr>
<tr>
<td><strong>Deployment Models</strong></td>
<td>Centralized Distributed</td>
<td>Centralized Only</td>
</tr>
<tr>
<td><strong>Media Termination</strong></td>
<td>Voice Gateway</td>
<td>IP-IVR Server</td>
</tr>
<tr>
<td><strong>Video Capabilities</strong></td>
<td>Video Queuing/IVR Video Caller/Agent</td>
<td>NA</td>
</tr>
<tr>
<td><strong>Codec Support</strong></td>
<td>Prompt: G.711 Agent: G.711/G.729</td>
<td>G.711 or G.729 (Set at installation)</td>
</tr>
</tbody>
</table>
Deployment Models
Deployment Models

1. **Standalone VXML Server**
   Automated Self Service IVR, No Queuing, Limited Call Control

2. **Call Director**
   IP Enable TDM ACDs, TDM Migration to IP, On-Net Routing & Transfer, Save $

3. **VRU-only**
   IVR and Queuing, Call Control via SS7

4. **Comprehensive**
   Pure IP-based Contact Center, IVR, Call Control, Queuing

5. **Basic Service Video**
   Audio-only IVR, Call Control, Queuing, Video Agent

6. **Full Service Video**
   Video IVR, Call Control, Queuing, Video Agent
Standalone VXML Server

- IVR only solution
- Single site or multi-site
- VXML Server can be centralized
- VoiceXML Gateways at remote site terminate IP or TDM voice
- VXML/HTTP across the WAN

Components:
- VXML Server
- Unified Call Studio
- VoiceXML Gateway
- Reporting Server (optional)
- ASR/TTS (recommended)
- CSS (optional)
Comprehensive

- IVR, call control, queuing
- ICM keeps track of agents
- VXML Session: VXML GW to Call Server or VXML Server

Components:
- Call Server
- VXML Server
- VoiceXML Gateway
- Gatekeeper or SIP Proxy
- ICM and VRU PG
- CUCM or TDM ACD
- Reporting Server (optional)
- ASR/TTS (optional)
- CSS (optional)
Basic Service Video

- Audio-only IVR, call control, queuing, video agent
- Same call flow as comprehensive SIP
- Supports both audio and video calls
- Video supported with SIP only
- Components:
  - Call Server
  - VXML Server
  - VoiceXML Gateway
  - SIP Proxy
  - ICM and VRU PG
  - Communications Manager
  - Reporting Server (optional)
  - ASR/TTS (optional)
- Video Endpoints:
  - Cisco Unified Video Advantage
  - Cisco Unified Personal Communicator
  - TelePresence™
  - 7985G
  - No 3G Gateway caller support
Full Service Video

- Video IVR, call control, queuing, video agent
- Supports both audio and video calls
- CIF/QCIF format support

Components:
- CVP Call Server
- CVP VMS (Darwin Streaming Server)
- Radvision IVP (Linux, iContact, MSP, B2BUA, XML)
- SIP Proxy
- ICM and VRU PG
- CUCM
- CUVC 3545 (MCU/EMP)

Video Endpoints:
- Cisco Unified Video Advantage
- Cisco Unified Personal Communicator
- 7985G
- Radvision/Cisco 3G Video Gateway
Video Media Server

- CVP Video Media Server
- Based on Darwin Media Streaming Server
- Stores and streams video to Radvision MSP/IVP via RTSP
- Video files are stored in 3GP format
- Admin web interface for managing video files, metadata
- Agent web interface for searching/previewing/playing video to callers
- CTIOS/CAD Supported
HW Platform, Operations and Reporting
HW Platform and OS Support

- Call Server, VXML Server, Reporting Server, Video Media Server must use MCS-7845 or higher server
- Windows 2003 supported for all components
  - Windows 2000 no longer supported
- VXML Server
  - WebSphere or Tomcat
  - Supported on AIX as well
- Operations Console
  - MCS-7825 or higher server
- Unified Call Studio
  - Windows XP SP2/Windows Vista. Windows OS compatible HW
Operations Console

- Operations Console is used to provision and administer.
- Operations Console is required to deploy CVP 4.0 and later releases.
- Monitor each CVP Server and display statistics about each one.
- Start, shutdown or gracefully shutdown any of the unified CVP devices.
- CVP components can only be managed by one OAMP Server.
- Cannot be deployed outside the firewall while other servers residing inside.
Studio Script Editor

- Projects and files explorer
- Script element palette
- Application call flow
- Script element properties and configuration
Reporting Server

- No redundancy/replication
- Cradle-to-grave reporting with ICM
- Can join CVP data with data in ICM database
- Ability to filter what data is persisted (security)
- Crystal Report templates provided with software
- CUIS templates available in 8.0
- Backups
  - Scheduled
  - On-demand
- Database used is Informix
- Database size is chosen at install time:
  - Small or all-in one - 2GB database
  - Standard (Medium) - 50 GB database
  - Advance (Large) - 100 GB database
Sizing
# CVP Components Sizing

<table>
<thead>
<tr>
<th>Component</th>
<th>Scalability per Server</th>
<th>Redundancy</th>
<th>HW Platform</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Server (SIP)</td>
<td>850 ports</td>
<td>N+1 to N*2</td>
<td>MCS-7845</td>
</tr>
<tr>
<td>Call Server (H.323)</td>
<td>500 ports</td>
<td>N+1 to N*2</td>
<td>MCS-7845</td>
</tr>
<tr>
<td>VXML Server</td>
<td>750 ports</td>
<td>N+1 to N*2</td>
<td>MCS-7845</td>
</tr>
<tr>
<td>Operations Console</td>
<td>NA</td>
<td>NA</td>
<td>MCS-7825 or higher</td>
</tr>
<tr>
<td>Co-Res (SIP, VXML, Media Server)</td>
<td>750 SIP ports + 750 VXML ports</td>
<td>N+1 to N*2</td>
<td>MCS-7845</td>
</tr>
<tr>
<td>Co-Res (H323, VXML, Media Server)</td>
<td>500 H323 ports + 500 VXML ports</td>
<td>N+1 to N*2</td>
<td>MCS-7845</td>
</tr>
<tr>
<td>VRU PG</td>
<td>9600 ports</td>
<td>N*2</td>
<td>MCS-7845</td>
</tr>
</tbody>
</table>
# VXML Gateway Sizing

<table>
<thead>
<tr>
<th>Voice Gateway Platform</th>
<th>Dedicated VXML GW</th>
<th>Voice Gateway and VoiceXML</th>
<th>Minimum DRAM Recommended</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>All calls VXML-Controlled; no PSTN interfaces present</td>
<td>All calls are PSTN calls and all calls are VXML-Controlled</td>
<td></td>
</tr>
<tr>
<td></td>
<td>DTMF</td>
<td>ASR/TTS</td>
<td>DTMF</td>
</tr>
<tr>
<td>Cisco 1861</td>
<td>5</td>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td>Cisco 2801</td>
<td>7</td>
<td>6</td>
<td>6</td>
</tr>
<tr>
<td>Cisco 2811</td>
<td>30</td>
<td>24</td>
<td>25</td>
</tr>
<tr>
<td>Cisco 2821</td>
<td>48</td>
<td>36</td>
<td>36</td>
</tr>
<tr>
<td>Cisco 2851</td>
<td>60</td>
<td>56</td>
<td>56</td>
</tr>
<tr>
<td>Cisco 3825</td>
<td>180</td>
<td>140</td>
<td>210</td>
</tr>
<tr>
<td>Cisco 3845</td>
<td>200</td>
<td>155</td>
<td>230</td>
</tr>
<tr>
<td>AS5350XM*</td>
<td>240</td>
<td>192</td>
<td>240</td>
</tr>
<tr>
<td>AS5400XM*</td>
<td>240</td>
<td>192</td>
<td>240</td>
</tr>
</tbody>
</table>

The numbers assume the only activities running on the GW are VXML with basic routing and IP connectivity. These figures apply to NTE 75% CPU, VAD off, Cisco IOS 12.4.15T5. The numbers represent performance with Call Studio generated scripts running on CVP VoiceXML Application Servers. VoiceXML 2.0 and MRCPv2 were tested. The 1861 requires 12.4.20T as a minimum release.

* NTE 80% CPU.
# VXML Gateway Sizing

VXML Sessions with Javascript-Intensive Applications: 12.4.15T

<table>
<thead>
<tr>
<th>Voice Gateway Platform</th>
<th>Dedicated VXML GW</th>
<th>Voice Gateway and VoiceXML</th>
<th>Minimum DRAM Recommended</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>All calls VXML-Controlled; no PSTN interfaces present</td>
<td>All calls are PSTN calls and all calls are VXML-Controlled</td>
<td></td>
</tr>
<tr>
<td></td>
<td>DTMF</td>
<td>ASR/TTS</td>
<td>DTMF</td>
</tr>
<tr>
<td>Cisco 1861</td>
<td>2</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>Cisco 2801</td>
<td>3</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>Cisco 2811</td>
<td>10</td>
<td>5</td>
<td>10</td>
</tr>
<tr>
<td>Cisco 2821</td>
<td>20</td>
<td>15</td>
<td>15</td>
</tr>
<tr>
<td>Cisco 2851</td>
<td>30</td>
<td>25</td>
<td>25</td>
</tr>
<tr>
<td>Cisco 3825</td>
<td>70</td>
<td>55</td>
<td>85</td>
</tr>
<tr>
<td>Cisco 3845</td>
<td>80</td>
<td>60</td>
<td>95</td>
</tr>
<tr>
<td>AS5350XM*</td>
<td>105</td>
<td>85</td>
<td>110</td>
</tr>
<tr>
<td>AS5400XM*</td>
<td>105</td>
<td>85</td>
<td>110</td>
</tr>
</tbody>
</table>

* NTE 80% CPU

© 2008 Cisco Systems, Inc. All rights reserved. Cisco Confidential
# VXML Gateway Sizing

Secure VXML Sessions with HTTPS: 12.4.15T

<table>
<thead>
<tr>
<th>Voice Gateway Platform</th>
<th>Dedicated VXML GW</th>
<th>Voice Gateway and VoiceXML</th>
<th>Minimum DRAM Recommended</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>All calls VXML-Controlled; no PSTN interfaces present</td>
<td>All calls are PSTN calls and all calls are VXML-Controlled</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>DTMF</td>
<td>ASR/TTS</td>
<td>DTMF</td>
</tr>
<tr>
<td>Cisco 1861</td>
<td>3</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>Cisco 2801</td>
<td>4</td>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td>Cisco 2811</td>
<td>15</td>
<td>10</td>
<td>15</td>
</tr>
<tr>
<td>Cisco 2821</td>
<td>30</td>
<td>20</td>
<td>20</td>
</tr>
<tr>
<td>Cisco 2851</td>
<td>40</td>
<td>35</td>
<td>30</td>
</tr>
<tr>
<td>Cisco 3825</td>
<td>115</td>
<td>90</td>
<td>125</td>
</tr>
<tr>
<td>Cisco 3845</td>
<td>125</td>
<td>100</td>
<td>135</td>
</tr>
<tr>
<td>AS5350XM*</td>
<td>155</td>
<td>120</td>
<td>138</td>
</tr>
<tr>
<td>AS5400XM*</td>
<td>155</td>
<td>120</td>
<td>138</td>
</tr>
</tbody>
</table>

* NTE 80% CPU

© 2008 Cisco Systems, Inc. All rights reserved. Cisco Confidential
Bandwidth Usage

Signaling and WAV Files

- H.323 uses about 7,000 bytes per call in each direction
- SIP uses about 17,000 bytes per call in each direction
  - H.323 bps = CPS * 56,000
  - SIP bps = CPS * 136,000
- WAV files
  - Depends on size and number of files
  - Depends on how often prompts are refreshed in GW cache
## Bandwidth Usage

<table>
<thead>
<tr>
<th>CPS</th>
<th>BHCA</th>
<th>Agents</th>
<th>Queue Time</th>
<th>VXML Ports</th>
<th>H.323</th>
<th>SIP</th>
<th>VXML</th>
</tr>
</thead>
<tbody>
<tr>
<td>.03</td>
<td>100</td>
<td>6</td>
<td>45 sec</td>
<td>9</td>
<td>2 kbps</td>
<td>4 kbps</td>
<td>40 kbps</td>
</tr>
<tr>
<td>.14</td>
<td>500</td>
<td>21</td>
<td>27 sec</td>
<td>18</td>
<td>8 kbps</td>
<td>19 kbps</td>
<td>106 kbps</td>
</tr>
<tr>
<td>.28</td>
<td>1000</td>
<td>38</td>
<td>24 sec</td>
<td>27</td>
<td>16 kbps</td>
<td>38 kbps</td>
<td>168 kbps</td>
</tr>
<tr>
<td>1.39</td>
<td>5000</td>
<td>172</td>
<td>17 sec</td>
<td>84</td>
<td>78 kbps</td>
<td>189 kbps</td>
<td>600 kbps</td>
</tr>
</tbody>
</table>

* Equations used to find these values can be found in the SRND
CUBE Support
CVP and CUBE

- CUBE acts as a demarcation point
- Co-resident VXML Gateway and CUBE SBC is supported
- SIP-SIP, SIP-H.323 translation
- PSTN / 3rd Party SIP Trunks
- CUCM originated SIP calls
- CUBE is supported in flow-through mode only
Centralized Model

- Bring in PSTN SIP Trunk to the Enterprise
- SIP / H.323 translation for versions of CM or CVP that do not support SIP
- All components located in Data Center
- VoiceXML runs on a separate platform from CUBE
Distributed Model

- Bring PSTN SIP Trunk in to the Enterprise
- SIP / H.323 translation for versions of CM or CVP that do not support SIP
- CUBE and VXML Gateway distributed in branch offices, terminating PSTN SIP Trunk
- VXML runs on the same platform from CUBE
SIP Design
SIP Service

- SIP is the preferred protocol for unified CVP
- SIP provides improved scalability and performance to unified CVP
- With SIP, you are able to interoperate with both Cisco and non-Cisco SIP edge devices
- SIP and H.323 can co-exist at the same time
- SIP Proxy, DNS SRV or Static Routes replace Gatekeeper function
- SIP packets tagged with proper QoS markings Vs. ACL for H.323
- CVP does not support KPML or SIP-Notify for DTMF, RFC 2833 only
  - MTP will be allocated if IP phones and SIP trunk on CUCM are not configured for, or do not support RFC2833
SIP Routing

- SIP Routing is the method in which a DN is resolved to an IP
  - Also used for load-balancing and redundancy
  - Gatekeeper is the only method with H.323 for CVP
- Several options available:
  - SIP Proxy
  - DNS SRV
  - Static Routes
  - Combination of above
- In general, centralized dial plan method is preferred over distributed/static method
  - In some cases a static route may be preferred (lab or small deployments)
SIP High Availability

- SIP HA can be achieved in multiple ways
  - Redundant SIP Proxies
  - DNS SRV records
  - Local SRV records
  - Static Routes
  - Combination of above

- UDP is preferred transport for SIP with CVP
  - TCP Timers cause significant delays with failover
  - While primary Proxy is down caller hears ringing for 10 seconds with UDP, 50 seconds with TCP
CVP 8.0 Planned Features
CVP 8.0 Major Features

- Part of Unified Communications 8.0
- Enhancements:
  - Improved Scalability
  - UC Diagnostic Portal Support
  - Serviceability Improvements
  - Courtesy Callback
  - Improved SIP Capabilities
  - Licensing Improvements
  - Video Enhancements
  - Reporting Enhancements
Improved Scalability

- Goal: reduce equipment amount of servers necessary to deploy CVP
- At least 1200 sessions per server (goal is 1500)
- Support for VMWare
- PG & CVP on one server (via VMWare)
- UCS Support
UC Diagnostic Portal Support

- Centralized portal for all UC applications
- Provides consistent and unified view of common serviceability mechanisms
UC Diagnostic Portal Features with CVP

- System Call Tracking
- Trace/debug level setting/getting
- Log file collection
- Trace file collection
- Application Version information
- Application Platform information
- Application Licensing information
- Application Configuration information
Serviceability Improvements

- Daylight Savings Tool
  - Easily allows changes if local laws modify start/end of DST
- Clarity & Consistency of events & alarms
  - Customer facing events are treated as customer documentation, and written from customer perspective.
  - Events inform customer System Administrator for various conditions on the system that may require System Administrator action - not intended to be engineering debug trace.
  - Events will use common industry terminology (e.g. Newton’s Telecom Dictionary) – do not use internal terminology.
- Existing, local Serviceability tools continue to be supported
Courtesy Callback

- Provides option for caller to leave name and dial-back number once call is in queue
- System automatically calls back prior to connecting to agent (pre-emptive call back)
- No Desktop modifications required
- Works with all ACDS that are supported by ICM
Improved SIP Capabilities

- Dynamic Routing Support
- Location Based Routing and Call Admission Control
- Passing/Receiving SIP Headers
- Improved UUI processing
- SIP support for KPML
- Hook flash transfer support
- SIP Sigdigits enhancement
- Post Call Survey Support
Licensing Enhancements

- Integrated Licensing (moving to FlexLM and IP Node Locking)
- Support for additive licenses
  - e.g. purchase of additional ports
- Automated tool to convert licenses to new format
- PUT support
Video Enhancement

- Support for Cisco IOS 3G gateway as an alternative to the Radvision 3G gateway in our existing video offering
- Cisco IOS 3G gateway supported for Basic and Comprehensive Video deployments
# CVP 8.0 Solution Matrix

<table>
<thead>
<tr>
<th>UC Components</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Product</strong></td>
<td><strong>Version</strong></td>
</tr>
<tr>
<td>CVP</td>
<td>CVP 8.0 - Anacapa</td>
</tr>
<tr>
<td>ICM</td>
<td>7.5, 8.0</td>
</tr>
<tr>
<td>Support Tools</td>
<td>2.4</td>
</tr>
<tr>
<td>UCCM</td>
<td>6.1.3, 7.1, 8.0</td>
</tr>
<tr>
<td>CUOM</td>
<td>8.0</td>
</tr>
<tr>
<td>CTI OS Desktop</td>
<td>7.5, 8.0</td>
</tr>
<tr>
<td>Cisco Agent Desktop</td>
<td>7.5, 8.0</td>
</tr>
<tr>
<td>CUIS</td>
<td>8.0</td>
</tr>
<tr>
<td>Diag Portal</td>
<td>8.0</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Gateway</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Product</strong></td>
<td><strong>Version</strong></td>
</tr>
<tr>
<td>37xx</td>
<td>12.4(15)T, 12.4(18c)</td>
</tr>
<tr>
<td>28xx, 29xx</td>
<td></td>
</tr>
<tr>
<td>38xx, 39xx</td>
<td>12.4(25m), 12.4(pi12)</td>
</tr>
<tr>
<td>5350XM</td>
<td></td>
</tr>
<tr>
<td>5400XM</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Gatekeeper</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Product</strong></td>
<td><strong>Version</strong></td>
</tr>
<tr>
<td>37xx</td>
<td>12.4(15)T, 12.4(18c)</td>
</tr>
<tr>
<td>28xx</td>
<td>12.4(25m), 12.4(pi12)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>ASR/TTS</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Product</strong></td>
<td><strong>Version</strong></td>
</tr>
<tr>
<td>Legacy Scansoft</td>
<td>3.1.17</td>
</tr>
<tr>
<td>Nuance SWMS</td>
<td></td>
</tr>
<tr>
<td>Nuance OSR</td>
<td>3.0.16</td>
</tr>
<tr>
<td>Nuance RealSpeak</td>
<td>4.0.12</td>
</tr>
<tr>
<td>Legacy Nuance</td>
<td></td>
</tr>
<tr>
<td>Nuance MRCP</td>
<td>1.0.0 Service Pack 12</td>
</tr>
<tr>
<td>Nuance ASR</td>
<td>8.5 SP070930 (SP#8)</td>
</tr>
<tr>
<td>Nuance Vocalizer</td>
<td>5.0</td>
</tr>
<tr>
<td>Nuance Quantum</td>
<td></td>
</tr>
<tr>
<td>Nuance NSS</td>
<td>5.0.4</td>
</tr>
<tr>
<td>Nuance Recognizer</td>
<td>9.0.5</td>
</tr>
<tr>
<td>Nuance RealSpeak</td>
<td>4.5.0, 5.0</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Product</th>
<th><strong>Version</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>Content Switch CSS</td>
<td>WebNS 8.20</td>
</tr>
<tr>
<td>Security CSA</td>
<td>6.0.1</td>
</tr>
<tr>
<td>OS</td>
<td>Win2K3 SP2, CCBU Win2K3 R2 OEM</td>
</tr>
<tr>
<td>Virus Scan</td>
<td>McAfee VirusScan Enterprise 8.7i</td>
</tr>
<tr>
<td>Symantec AntiVirus Corp Edition</td>
<td>10.0+</td>
</tr>
<tr>
<td>Trend Micro ServerProtect for NT</td>
<td>5.7</td>
</tr>
<tr>
<td>Vmware ESX</td>
<td>4.0</td>
</tr>
<tr>
<td>Web Browser Internet Explorer</td>
<td>IE 6.0 SP1, IE 7.0, IE 8.0</td>
</tr>
<tr>
<td></td>
<td>FireFox</td>
</tr>
<tr>
<td>Database IBM Informix Dynamic Server</td>
<td>10.00.xC9</td>
</tr>
<tr>
<td>Web Server Websphere (WAS)</td>
<td>v6.1 32 bits</td>
</tr>
<tr>
<td></td>
<td>Tomcat</td>
</tr>
<tr>
<td>Java JRE</td>
<td>1.5 and 1.6 (except Studio)</td>
</tr>
<tr>
<td>Remote Admin Symantec pcANYWHERE</td>
<td>12.5</td>
</tr>
<tr>
<td></td>
<td>RealVNC</td>
</tr>
<tr>
<td></td>
<td>Windows Terminal Services</td>
</tr>
<tr>
<td>Proxy Server CUPS</td>
<td>7.x, 8.0(1)</td>
</tr>
<tr>
<td></td>
<td>CUSP</td>
</tr>
<tr>
<td>Radvision Video iContact</td>
<td>2.0.0.0.27</td>
</tr>
<tr>
<td></td>
<td>Interactive Video Platform</td>
</tr>
<tr>
<td></td>
<td>MSP</td>
</tr>
<tr>
<td></td>
<td>Scopia 3G Gateway</td>
</tr>
<tr>
<td>Video Endpoints CUVA</td>
<td>2.1.2</td>
</tr>
<tr>
<td></td>
<td>Teleepresence</td>
</tr>
<tr>
<td></td>
<td>Hard Phone</td>
</tr>
<tr>
<td></td>
<td>Soft Phone</td>
</tr>
</tbody>
</table>
Resources

- Unified Communications SRNDs
  - www.cisco.com/go/designzone

- Hardware and Software Specification for CVP

- CVP Product Page
  - www.cisco.com/go/cvp
Registrujte se za Cisco Networkers
28-31. mart 2010. Bahrein