Cisco Expo 2009

Cisco Unified Customer Voice Portal

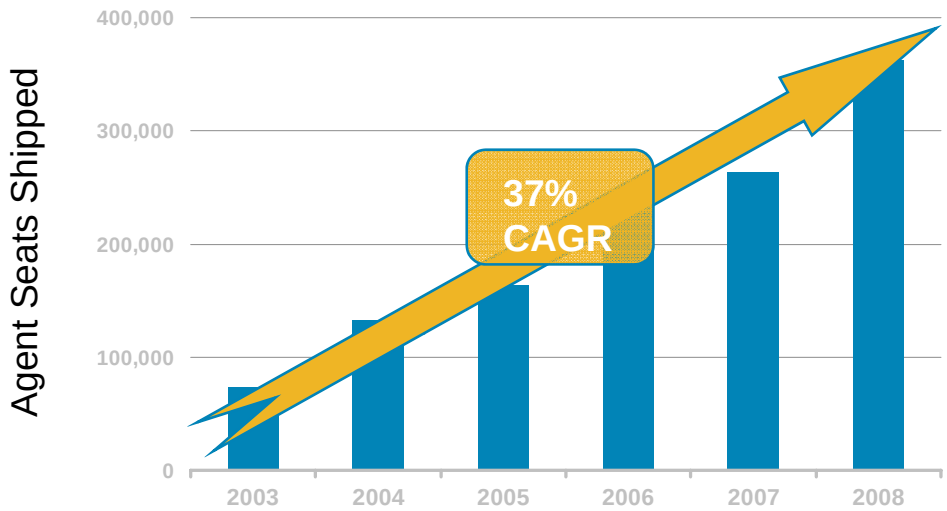
Overview and Design - Mario Gianni

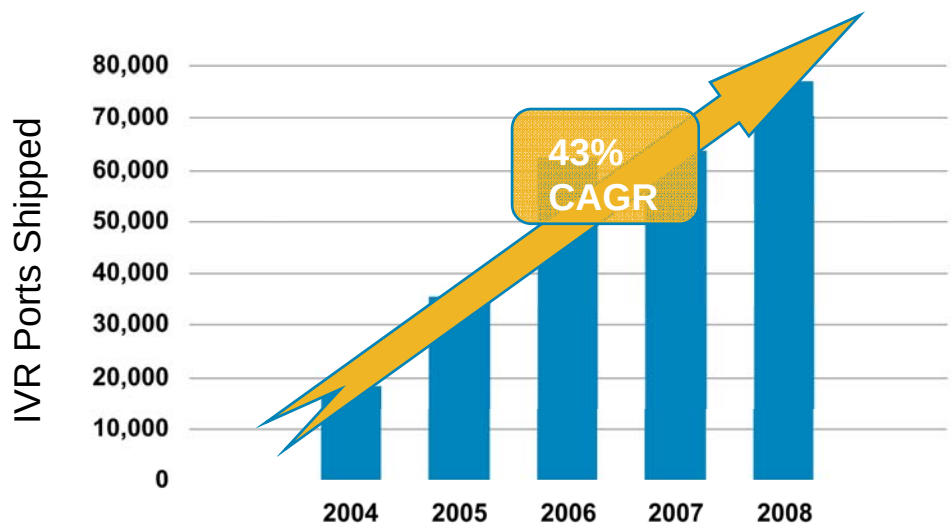
Agenda

- Overview
- Deployment Models
- HW Platform, Operations and Reporting
- Sizing
- CUBE Support
- SIP Design
- CVP 8.0 Planned Features

Overview

Cisco's Market Position Accelerating



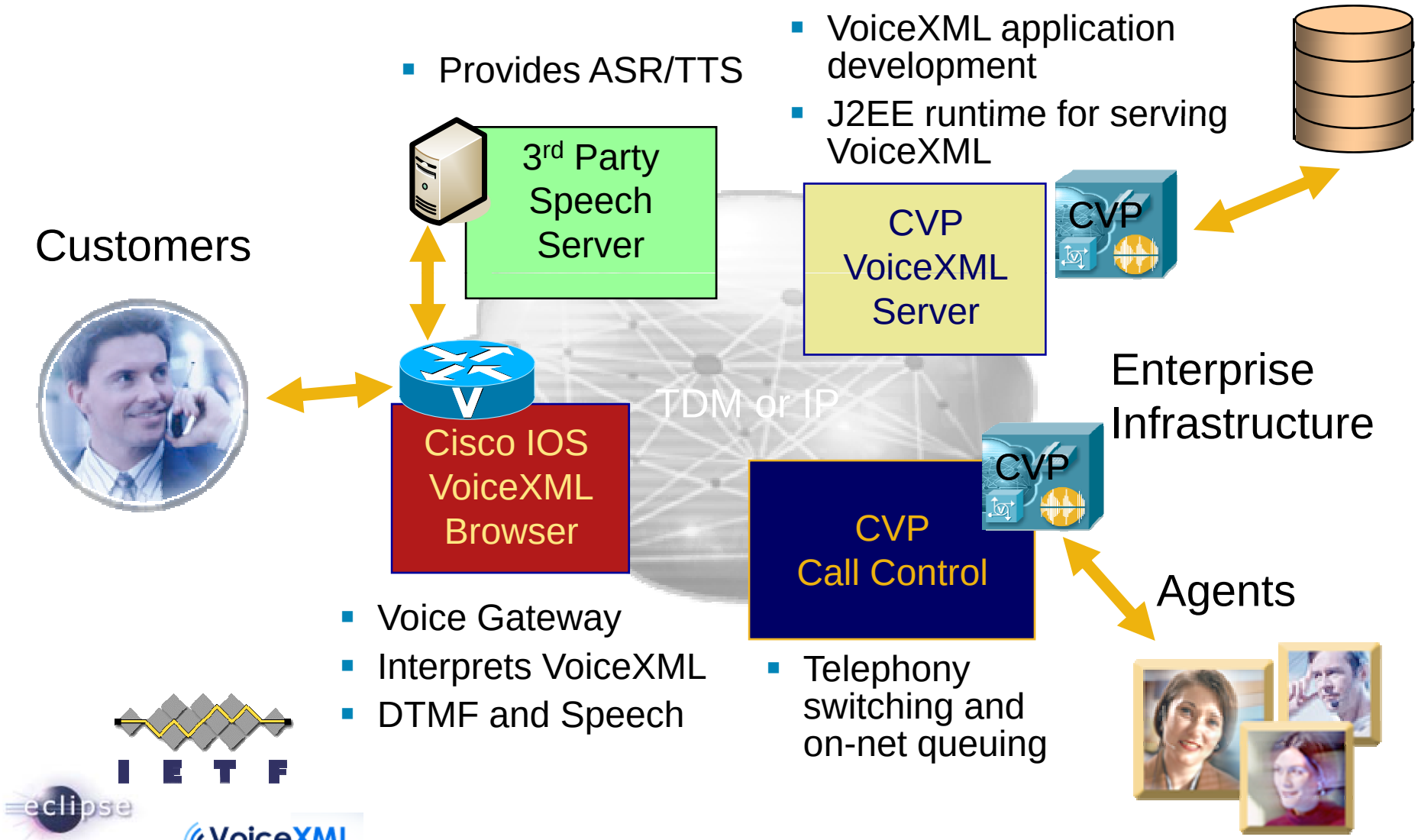


Cisco #2 in Worldwide Market Share for Agent Seats and IVR Ports

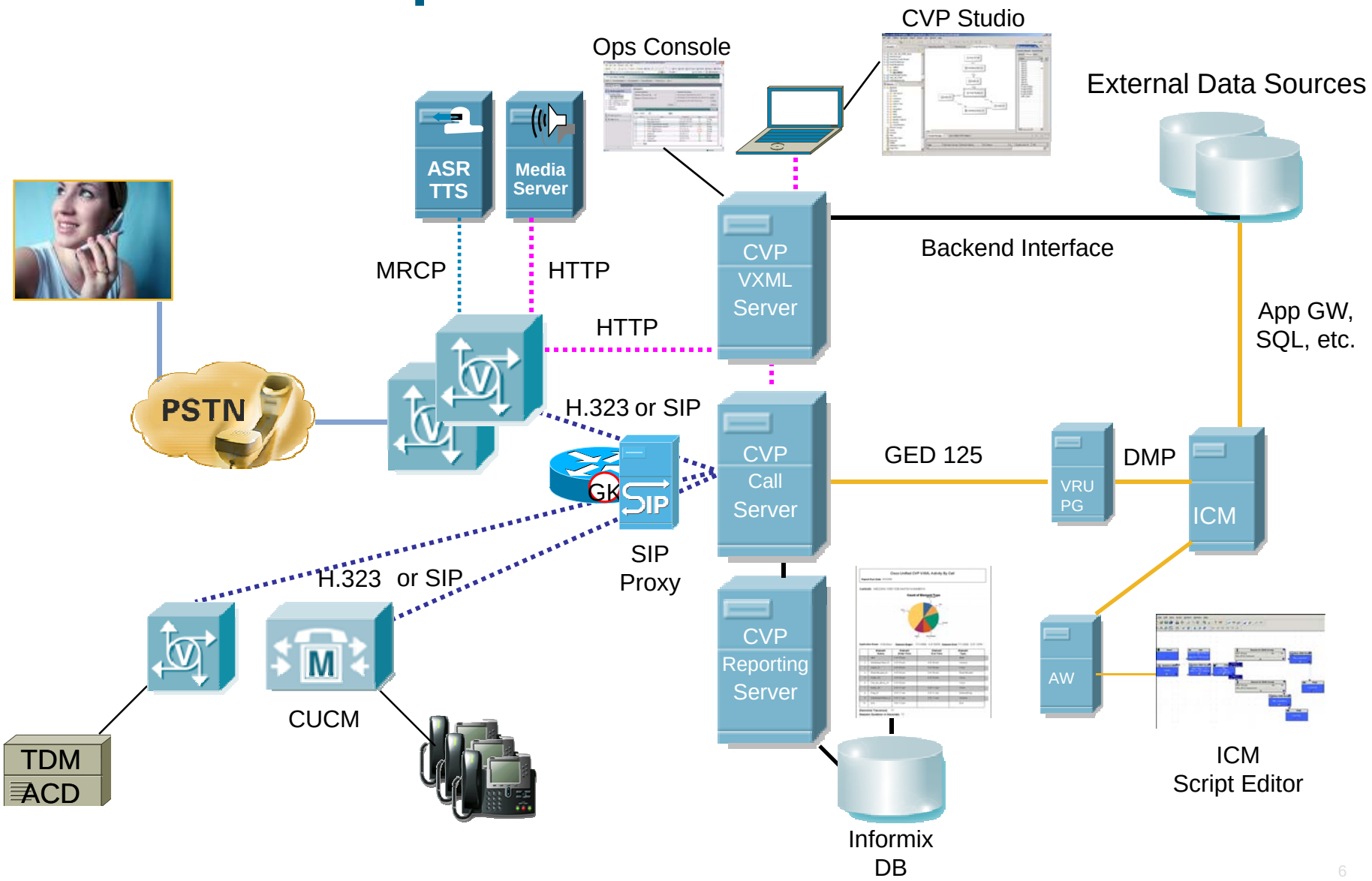
(Sources: Gartner, Tern Systems)

Source: Cisco analysis

Overview



Detailed Components



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

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

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

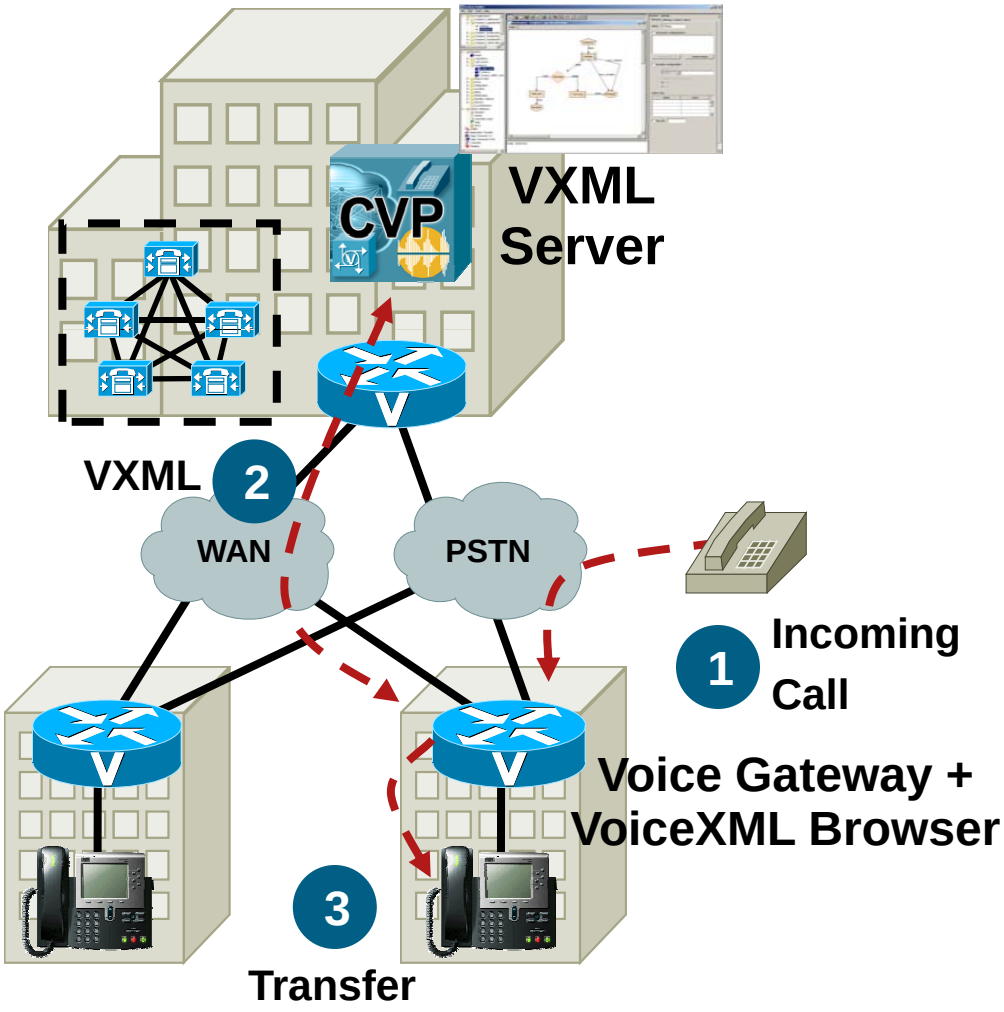
5. **Basic Service Video**

   Audio-only IVR, Call Control, Queuing, Video Agent

6. **Full Service Video**

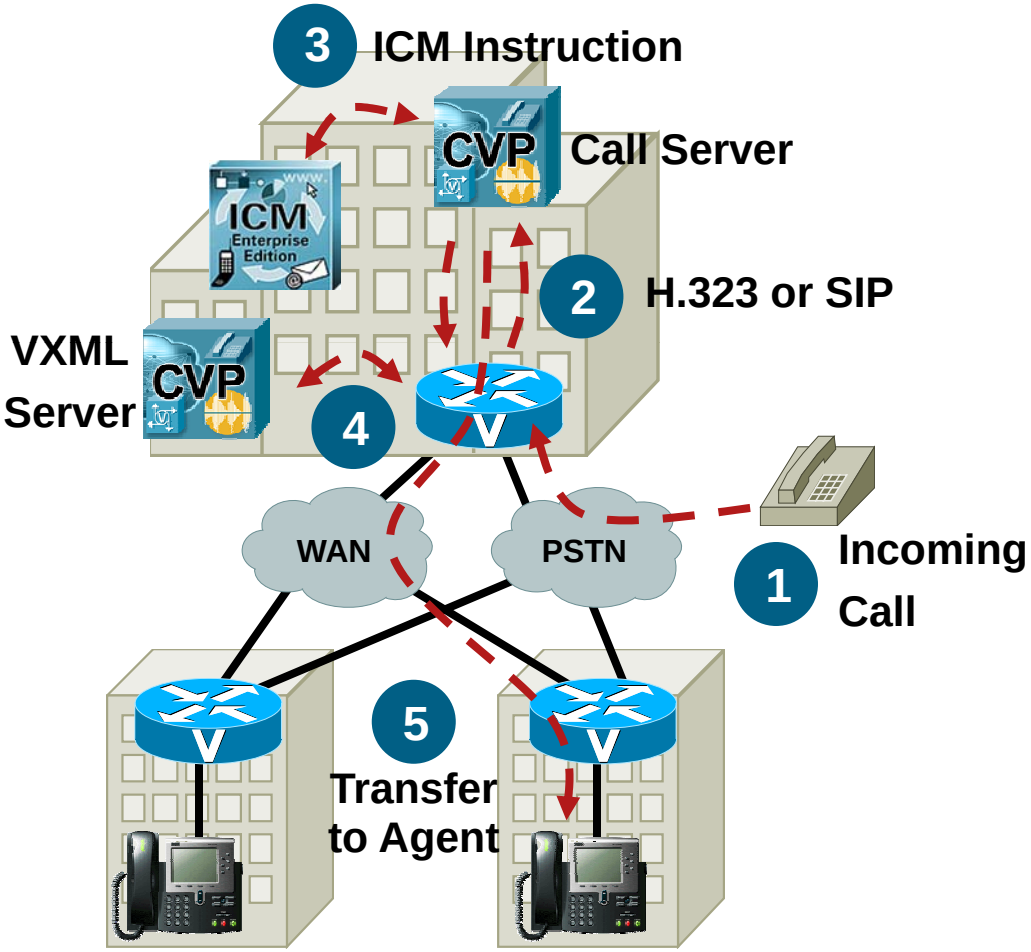
   Video IVR, Call Control, Queuing, Video Agent

Standalone VXML Server



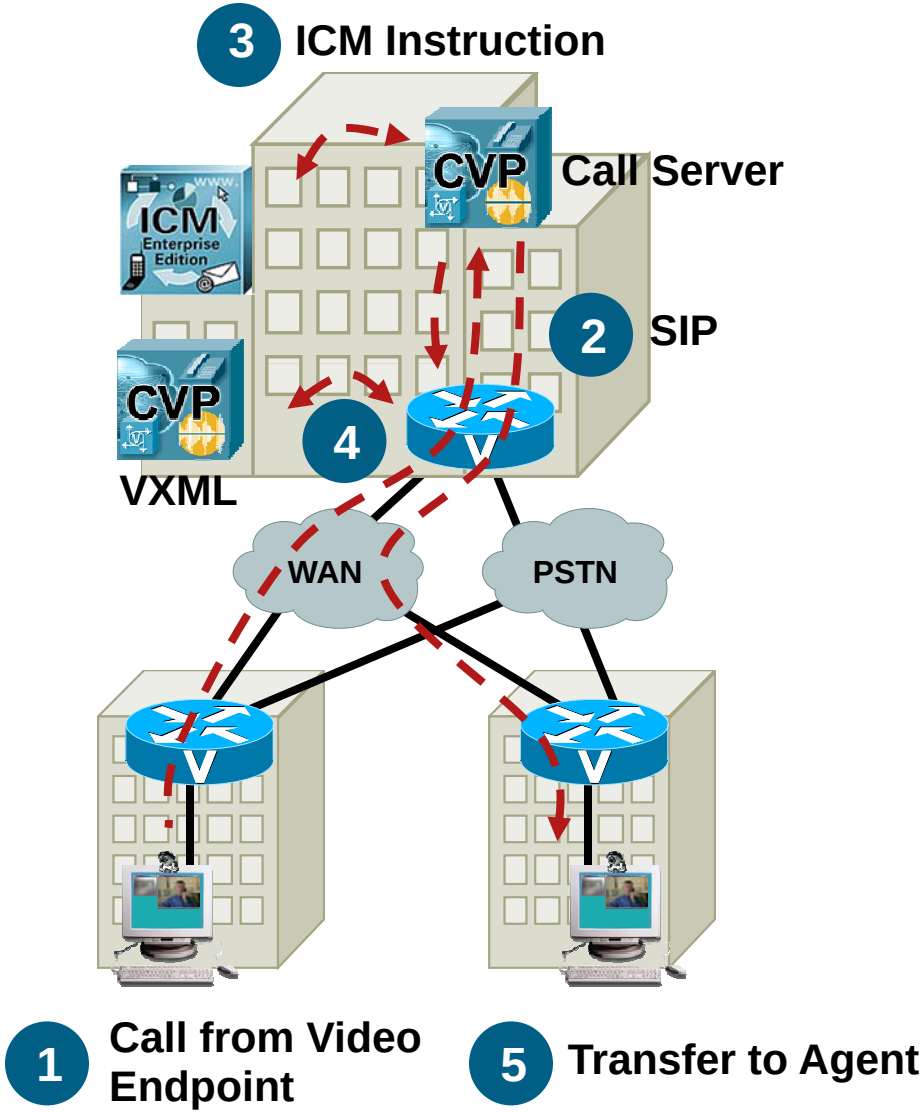
- IVR only solution
- Single site or multi-site
- VXML Server can be centralized
- VoiceXML Gateways at remote site terminate IP or TDM voice
- VXML/HTTP across the WAN
- Components:
  - VXML Server
  - Unified Call Studio
  - VoiceXML Gateway
  - Reporting Server (optional)
  - ASR/TTS (recommended)
  - CSS (optional)

Comprehensive



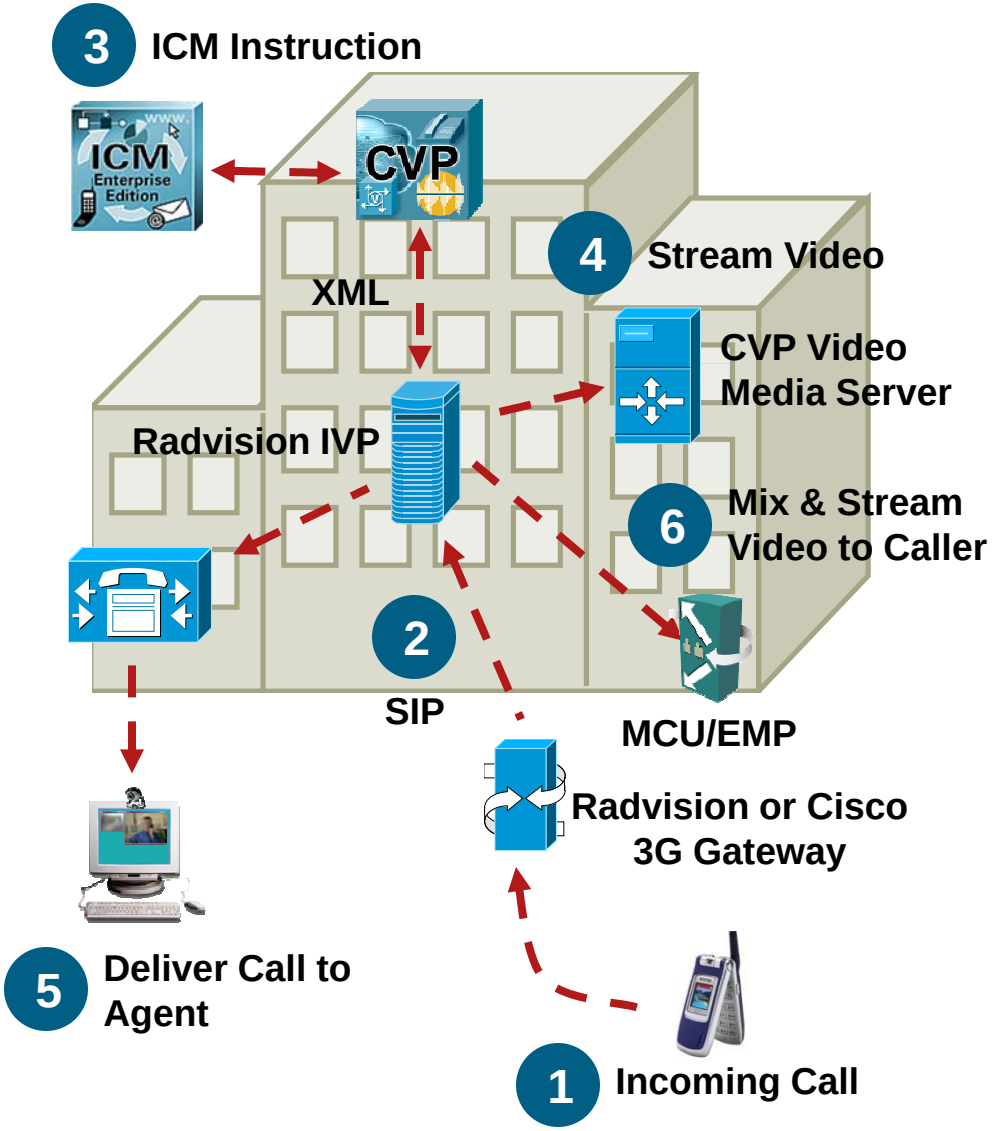
- IVR, call control, queuing
- ICM keeps track of agents
- VXML Session: VXML GW to Call Server or VXML Server
- Components:
  - Call Server
  - VXML Server
  - VoiceXML Gateway
  - Gatekeeper or SIP Proxy
  - ICM and VRU PG
  - CUCM or TDM ACD
  - Reporting Server (optional)
  - ASR/TTS (optional)
  - 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

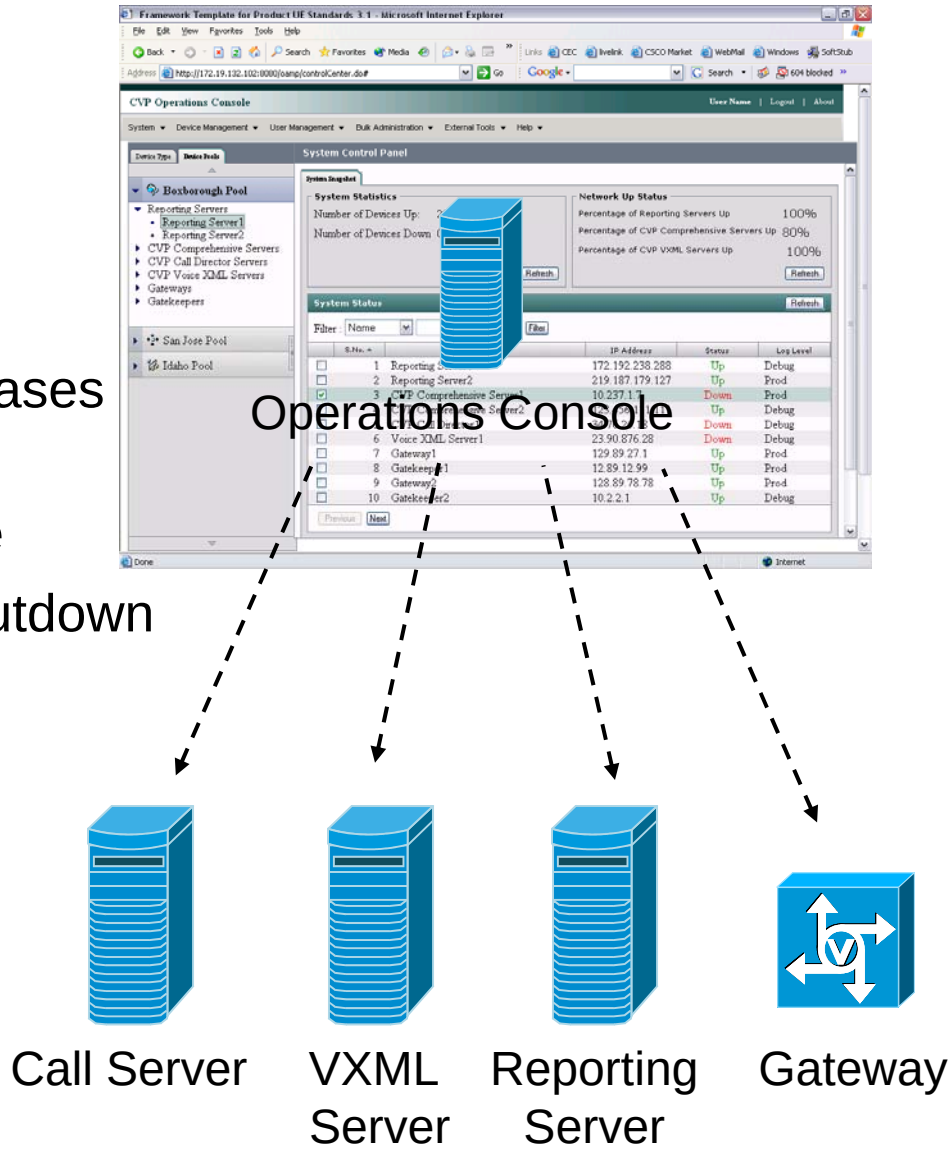
HW Platform, Operations and Reporting

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



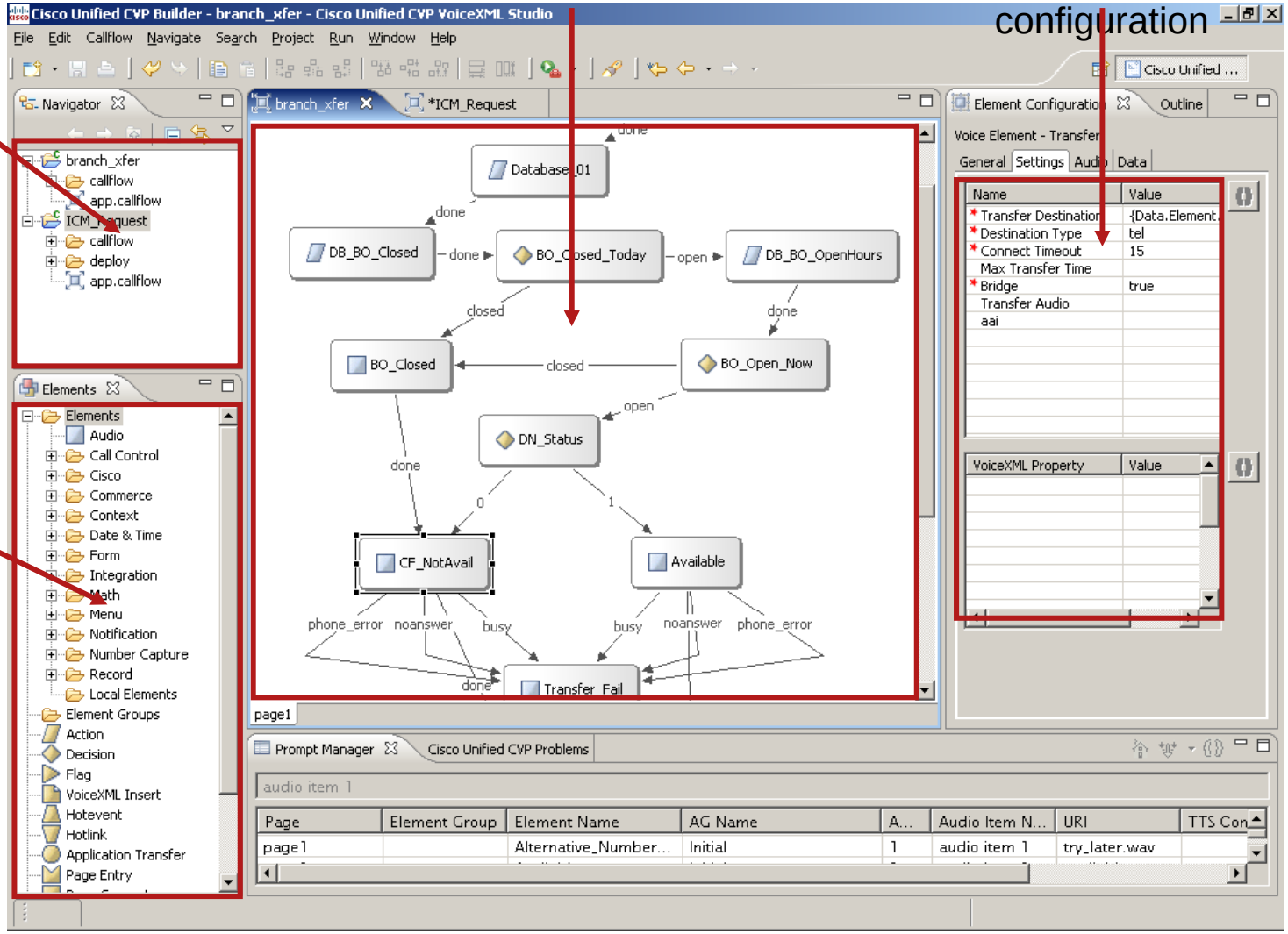
Studio Script Editor

Projects and files explorer

Application call flow

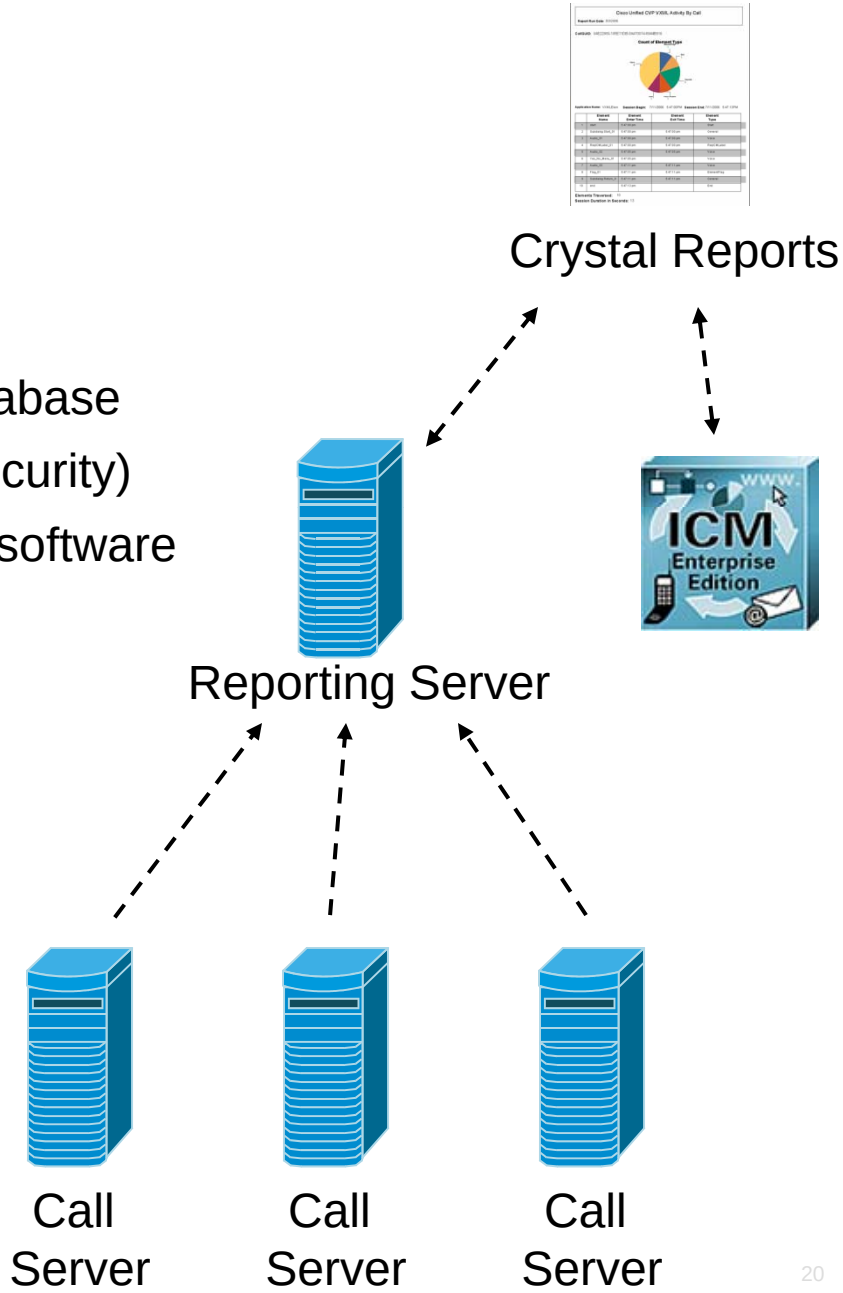
Script element properties and configuration

Script element palette



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 lab/demo use only
  - Standard (Medium) - 50 GB database
  - Advance (Large) - 100 GB database



Sizing

CVP Components Sizing

Component	Scalability per Server	Redundancy	HW Platform
Call Server (SIP)	850 ports	N+1 to N*2	MCS-7845
Call Server (H.323)	500 ports	N+1 to N*2	MCS-7845
VXML Server	750 ports	N+1 to N*2	MCS-7845
Operations Console	NA	NA	MCS-7825 or higher
Co-Res (SIP, VXML, Media Server)	750 SIP ports + 750 VXML ports	N+1 to N*2	MCS-7845
Co-Res (H323, VXML, Media Server)	500 H323 ports + 500 VXML ports	N+1 to N*2	MCS-7845
VRU PG	9600 ports	N*2	MCS-7845

VXML Gateway Sizing

Voice Gateway Platform	Dedicated VXML GW		Voice Gateway and VoiceXML		Minimum DRAM Recommended
	All calls VXML-Controlled; no PSTN interfaces present		All calls are PSTN calls and all calls are VXML-Controlled		
	DTMF	ASR/TTS	DTMF	ASR/TTS	
Cisco 1861	5	4	4	2	256MB
Cisco 2801	7	6	6	4	256MB
Cisco 2811	30	24	25	20	256MB
Cisco 2821	48	36	36	30	256MB
Cisco 2851	60	56	56	48	512MB
Cisco 3825	180	140	210	130	512MB
Cisco 3845	200	155	230	145	512MB
AS5350XM*	240	192	240	160	512MB (default)
AS5400XM*	240	192	240	160	512MB (default)

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

Voice Gateway Platform	Dedicated VXML GW		Voice Gateway and VoiceXML		Minimum DRAM Recommended
	All calls VXML-Controlled; no PSTN interfaces present		All calls are PSTN calls and all calls are VXML-Controlled		
	DTMF	ASR/TTS	DTMF	ASR/TTS	
Cisco 1861	2	2	2	2	256MB
Cisco 2801	3	2	2	2	256MB
Cisco 2811	10	5	10	5	256MB
Cisco 2821	20	15	15	15	256MB
Cisco 2851	30	25	25	20	512MB
Cisco 3825	70	55	85	50	512MB
Cisco 3845	80	60	95	60	512MB
AS5350XM*	105	85	110	70	512MB (default)
AS5400XM*	105	85	110	70	512MB (default)

* NTE 80% CPU

VXML Gateway Sizing

Secure VXML Sessions with HTTPS: 12.4.15T

Voice Gateway Platform	Dedicated VXML GW		Voice Gateway and VoiceXML		Minimum DRAM Recommended
	All calls VXML-Controlled; no PSTN interfaces present		All calls are PSTN calls and all calls are VXML-Controlled		
	DTMF	ASR/TTS	DTMF	ASR/TTS	
Cisco 1861	3	2	2	2	256MB
Cisco 2801	4	4	4	2	256MB
Cisco 2811	15	10	15	10	256MB
Cisco 2821	30	20	20	15	256MB
Cisco 2851	40	35	30	25	512MB
Cisco 3825	115	90	125	75	512MB
Cisco 3845	125	100	135	85	512MB
AS5350XM*	155	120	138	95	512MB (default)
AS5400XM*	155	120	138	95	512MB (default)

* NTE 80% CPU

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

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

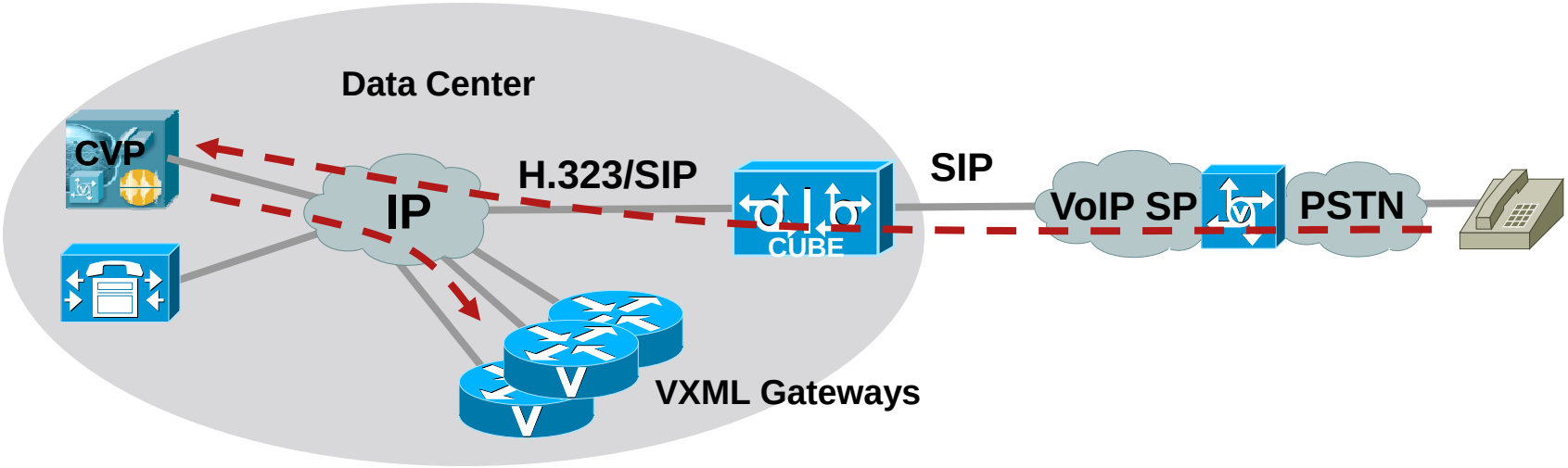
* Equations used to find these values can be found in the SRND

CUBE Support

CVP and CUBE

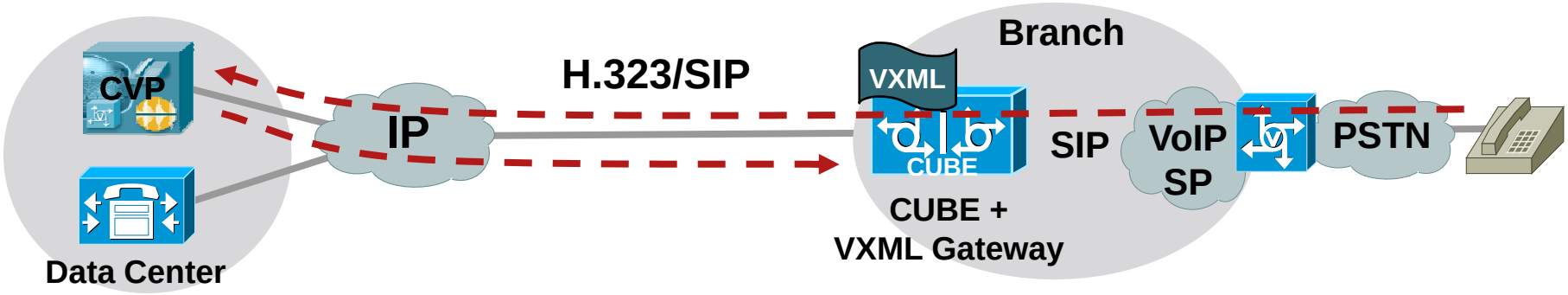
- CUBE acts as a demarcation point
- Co-resident VXML Gateway and CUBE SBC is supported
- SIP-SIP, SIP-H.323 translation
- PSTN / 3rd Party SIP Trunks
- CUCM originated SIP calls
- CUBE is supported in flow-through mode only

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

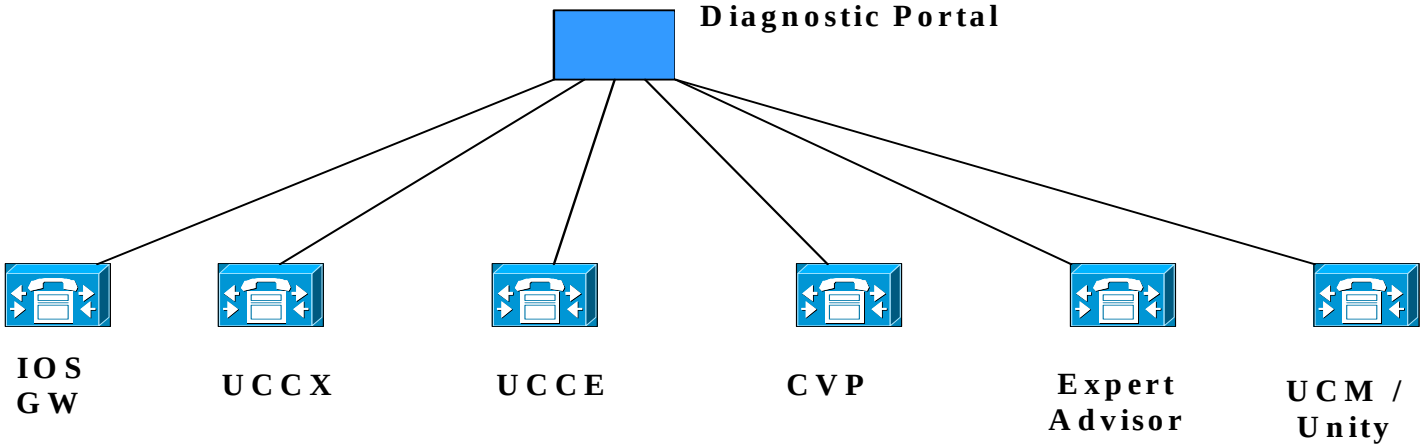
- Part of Unified Communications 8.0
- Enhancements:
  - Improved Scalability
  - UC Diagnostic Portal Support
  - Serviceability Improvements
  - Courtesy Callback
  - Improved SIP Capabilities
  - Licensing Improvements
  - Video Enhancements
  - Reporting Enhancements

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

Product		Version
UC Components	CVP	CVP 8.0 - Anacapa
	ICM	7.5, 8.0
	Support Tools	2.4
	UCCM	6.1.3, 7.1, 8.0
	CUOM	8.0
	CTI OS Desktop	7.5, 8.0
	Cisco Agent Desktop	7.5, 8.0
	CUIS	8.0
	Diag Portal	8.0
Gateway	37xx	12.4(15)T, 12.4(18c)
	28xx, 29xx	12.4(25m), 12.4(pi12)
	38xx, 39xx	
	5350XM	
	5400XM	
Gatekeeper	37xx	12.4(15)T, 12.4(18c)
	28xx	12.4(25m), 12.4(pi12)
ASR/TTS	Legacy Scansoft	
	Nuance SWMS	3.1.17
	Nuance OSR	3.0.16
	Nuance RealSpeak	4.0.12
	Legacy Nuance	
	Nuance MRCP	1.0.0 Service Pack 12
	Nuance ASR	8.5 SP070930 (SP#8)
	Nuance Vocalizer	5.0
	Nuance Quantum	
	Nuance NSS	5.0.4
	Nuance Recognizer	9.0.5
	Nuance RealSpeak	4.5.0, 5.0

Product		Version
Content Switch	CSS	WebNS 8.20
	ACE	3.2.1
Security	CSA	6.0.1
OS		Win2K3 SP2, CCBU Win2K3 R2 OEM
Virus Scan	McAfee VirusScan Enterprise	8.7i
	Symantec AntiVirus Corp Edition	10.0+
	Trend Micro ServerProtect for NT	5.7
Vmware	ESX	4.0
Web Browser	Internet Explorer	IE 6.0 SP1, IE 7.0, IE 8.0
	FireFox	FF2, FF3
Database	IBM Informix Dynamic Server	10.00.xC9
Web Server	Websphere (WAS)	v6.1 32 bits
	Tomcat	v5.5.27
Java	JRE	1.5 and 1.6 (except Studio)
Remote Admin	Symantec pcANYWHERE	12.5
	RealVNC	4.1
	Windows Terminal Services	Server admin only
Proxy Server	CUPS	7.x, 8.0(1)
	CUSP	1.1.4
Radvision Video	iContact	2.0.0.0.27
	Interactive Video Platform	2.0.2.0.5
	MSP	3.2.0.0.86
	Scopia 3G Gateway	3.1.1.0
Video Endpoints	CUVA	2.1.2
	Telepresence	CTS1000, CTS3000
	Hard Phone	7962, 7965, 7975, RT
	Soft Phone	CUPC 7.0.2

Resources

- Unified Communications SRNDs
  - www.cisco.com/go/designzone
- Hardware and Software Specification for CVP
  - http://www.cisco.com/en/US/docs/voice_ip_comm/cust_contact/contact_center/customer_voice_portal/cvp7_0/reference/guide/cvp702bom.pdf
- CVP Product Page
  - www.cisco.com/go/cvp

Registrujte se za Cisco Networkers
25-28. januar 2010. Barselona
28-31. mart 2010. Bahrein