Unified Customer Voice Portal Overview

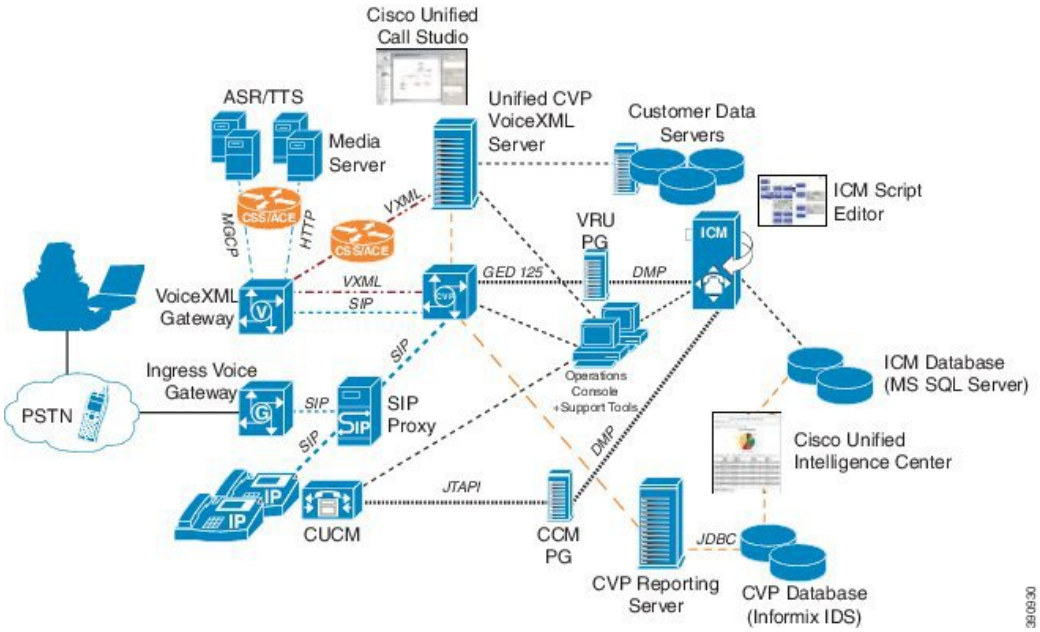
Overview

The Unified Customer Voice Portal (Unified CVP) is a web-based platform that provides carrier-class Interactive Voice Response (IVR) and IP switching services over VoIP networks. The platforms work together so that you can create and deploy IVR applications. The IVR applications include voice and video interaction and traditional numeric input to provide intelligent, personalized self-service over the phone.

Unified CVP is based on VoiceXML (VXML). VXML provides a flexible application development and deployment environment for creating IVR applications. IVR applications control audio input and output, presentation logic, call flow, telephony connections, and error handling. The Unified CVP suite of components enables the VXML to receive and report IVR events and interface with customer database components.

This chapter provides a high-level view of the Unified CVP product as it relates to the VXML and call flow model. The following figure illustrates the Unified CVP and components in a network. The other chapters in this guide provide more details about Unified CVP, including system design considerations such as call flow models and implementation factors.

Figure 1: Unified CVP and Its Components in a Network



Unified CVP Product Components

This section describes the Cisco Unified Customer Voice Portal (CVP) product components.

Note Call Server, VXML Server, and Media Server are combined as one component, which is known as CVP Server. Installing CVP Server installs all three components.

Call Server

The Call Server component provides the following independent services, which all run on the same Windows server:

- **SIP service**: This service communicates with the Unified CVP solution components such as the SIP Proxy Server, Ingress Gateway, Unified CM SIP trunks, and SIP phones. The SIP service implements a Back-to-Back User Agent (B2BUA). This B2BUA accepts SIP invites from ingress voice gateways and typically directs those calls to an available Voice XML gateway port. After completing call setup, the Unified CVP B2BUA acts as an active intermediary for any subsequent call control. While the Unified CVP SIP signaling is routed through this service, this service does not touch the RTP traffic. Integrated into this B2BUA is the ability to interact with the Cisco Unified ICM through the ICM Service. This integration provides the ability for the SIP Service to query the Unified ICM for routing instruction and service control. This integration also allows Unified ICM to begin subsequent call control to do things such as transfers.
- **ICM service**: This service is responsible for all communication between Unified CVP components and Unified ICM. It sends and receives messages on behalf of the SIP Service and the IVR Service.

- **IVR service**: This service creates the Voice XML pages that implement the Unified CVP Micro applications based on RunExternalScript instructions received from Unified ICM. The IVR Service functions as the VRU leg (in Unified ICM Enterprise parlance). Calls must be transferred to it from the SIP Service in order to execute Micro applications. The Voice XML pages created by this module are sent to the Voice XML gateway to be executed.

Call Server can be deployed as part of the Enterprise Windows Domain.

For hardware details, see the latest version of the *Hardware and System Software Specification for Cisco Unified Customer Voice Portal* (formerly called the *Bill of Materials*), available at http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_technical_reference_list.html.

VXML Server

The VXML Server executes advanced IVR applications by exchanging VoiceXML pages with the VoiceXML gateway's built-in voice browser. Like almost all other Unified CVP product components, it runs within a Java 2 Enterprise Edition (J2EE) application server environment such as Tomcat. You can add either custom-built or standard J2EE components to interact with back-end hosts and services. The VXML Server applications are written using Cisco Unified Call Studio and are deployed to the VXML Server for execution. The applications are invoked on an as-needed basis by a special Micro application which must be executed from within the Unified CCE routing script.

The VXML Server can also be deployed in a standalone configuration that does not include any Unified ICM components. Applications are invoked as a direct result of calls arriving in the VoiceXML gateway, and a single post application transfer is allowed.

The VXML Server can execute on Windows Server 2012 R2 Standard Edition. For hardware requirements and details, see the *Hardware and System Software Specification for Cisco Unified Customer Voice Portal*(formerly called the *Bill of Materials*), available at http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_technical_reference_list.html

For more information about the VXML Server and its latest features, see the *User Guide for Cisco Unified CVP VXML Server and Cisco Unified Call Studio*.

Media Server

The Media Server component is a simple web server which provides prerecorded audio files, external VoiceXML documents, or external ASR grammars to the VoiceXML gateway. Some of these files can be stored in the local file system on the gateways. However, in practice, most installations use a centralized media server to simplify distribution of prerecorded customer prompt updates. Media Server functionality can also include a caching engine. The gateways themselves, however, can also do prompt caching when configured for caching. Typical Media Servers used are Microsoft IIS and Apache, both of which are HTTP-based.

Note

- The Media Server component in Unified CVP is installed by default, along with Unified CVP Call Server and Unified CVP VXML Server.
- Unified CVP does not support the use of Tomcat as a Media Server.

Media Servers can be deployed as a simplex operation, as a redundant pair, or with supported load balancers in a farm. The VoiceXML Gateway caches.wav files it retrieves from the Media Server. In most deployments, the Media Server encounters low traffic from Unified CVP.

For the most current information on Media Servers, see the latest version of *Hardware and System Software Specification for Cisco Unified Customer Voice Portal* (formerly called the *Bill of Materials*), available at

http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_technical_reference_list.html

Call Studio

Cisco Unified Call Studio is the service creation environment (script editor) for Unified CVP VXML Server applications. It is based on the open source Eclipse framework, which provides an advanced drag-and-drop graphical editing feature. Call Studio also provides options to insert vendor-supplied and custom-developed plug-ins that enable applications to interact with other services in the network. Call Studio basically is an offline tool. The only interaction with the Unified CVP VXML Server is to deliver compiled applications and plugged-in components for execution.

The Call Studio license is associated with the MAC address of the machine on which it is running. You typically designate one or more data center servers for that purpose. Cisco Unified Call Studio cannot run on machines.

For additional hardware details, see the latest version of the *Hardware and System Software Specification for Cisco Unified Customer Voice Portal* (formerly called the *Bill of Materials*), available at http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_technical_reference_list.html

Reporting Server

The Reporting Server is a Windows server that hosts an IBM Informix Dynamic Server (IDS) database management system. The Reporting Server provides consolidated historical reporting for a distributed self-service deployment. The database schema is specified by the Unified CVP product, but the schema is fully published so that customers can develop custom reports based on it. The Reporting Server receives reporting data from the IVR Service, the SIP Service (if used), and the VXML Server. The Reporting Server depends on the Call Server to receive call records.

For standalone VXML Server deployments, one Call Server is needed per Reporting Server. The Reporting Server must be local to the Call Servers and VXML Servers. Deploying the Reporting Server at a remote location across the WAN is not supported. Use multiple Reporting Servers and place them at each site when Call Servers and VXML Servers exist at multiple locations.

The Reporting Server does not perform database administrative and maintenance activities such as backups or purging. However, Unified CVP provides access to such maintenance tasks through the Operations Console Server.

Operations Console

The Unified CVP Operations Console is a Windows server that provides a console for the browser-based administration and configuration for all Unified CVP components. It offers shortcuts to the administration and configuration interfaces of other Unified CVP solution components. The Operations Console is a required component in all Unified CVP deployments. The Operations Console must be run on a separate server from other Unified CVP devices. The Operations Console is, in effect, a dashboard from which an entire Unified CVP deployment can be managed.

The Operations Console must be configured with a map of the deployed solution network. It can then collect and maintain configuration information from each deployed component. You can use standard tools to back up both the network map and the configuration information that are stored locally on the server. The Operations Console provides options to display and modify the network map and stored configuration data and to distribute such modifications to the affected solution components.

The Operations Console can display two views of configuration parameters for managed components. The runtime view shows the status of all configuration parameters as the managed components use them. The configured or offline view shows the status of all configuration parameters that are stored in the Operations Server database and deployed to the device when you run the Save and Deploy option.

The Operations Console allows configuration parameters to be updated or preconfigured even when the target component is not online or running. If the target server (without its services) is online, you can apply the configured settings to that server. These settings become active when that server's services also come online, only then are reflected in the runtime view.

The Operations Console Server is not a redundant component. As such, you cannot duplicate the Operations Console Server within a deployment. It backs up the configuration database regularly, or whenever changes are made.

Additional Components

The following components are not part of the Unified CVP software, but you can use them with the CVP components for a complete solution:

- Cisco Ingress Voice Gateway
- Cisco VoiceXML Gateway
- Cisco Egress Gateway
- Video Endpoints
- Cisco Unified Communications Manager
- Cisco Unified Intelligence Center
- Cisco Unified Contact Center
- SIP Proxy Server
- DNS Server
- Load Balancers
- Cisco Unified Border Element

- Video Media Server
- Automatic Speech Recognition Server/Text-to-Speech Server
- Network Monitor

Cisco Ingress Voice Gateway

The Cisco Ingress Voice Gateway is the point at which an incoming call enters the Unified CVP system. It terminates time division multiplexing (TDM) calls on one side and implements VoIP on the other side. It serves as a pivot point for extension of calls from the TDM environment to VoIP endpoints. Hence, WAN bandwidth is conserved because no hairpinning of the media stream occurs. The Cisco Ingress Voice Gateway also provides for call switching capabilities at the command of other Unified CVP solution components.

Unified CVP Ingress Voice Gateways supports Session Initiation Protocol (SIP). Media Gateway Control Protocol (MGCP) voice gateways are supported if they are registered with Cisco Unified Communications Manager.

For the list of supported gateways, see ISO Gateway Choices. For approved gateway and software combinations, see the *Hardware and System Software Specification for Cisco Unified Customer Voice Portal* at http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_technical_reference_list.html.

VoiceXML Gateway

In this document, the term "VoiceXML Gateway" can either refer to Cisco Virtualized Voice Browser (Cisco VVB) or IOS VoiceXML Gateway that can be deployed on an Ingress Voice Gateway. VoiceXML Gateways are often deployed in farms for Centralized deployment models. VoiceXML Gateway interprets VoiceXML pages either from the Unified CVP Server IVR Service or the VXML Server.

Audio prompts retrieved from a third-party media server can be cached in VoiceXML Gateway to reduce WAN bandwidth and prevent poor voice quality. The VoiceXML document provides a pointer to the location of the audio file to be played or it provides the address of a text-to-speech (TTS) Server to stream the audio to the user. The VoiceXML Gateway interacts with automatic speech recognition (ASR) and TTS Servers through Media Resource Control Protocol (MRCP).

Cisco Virtualized Voice Browser

You can deploy Cisco VVB on a separate virtual machine. This model is suitable for both standalone, and comprehensive deployments. Cisco VVB communicates with ASR/TTS using MRCP.

For more information on Cisco VVB support, see the *Hardware and System Software Specification for Cisco Unified Customer Voice Portal* at http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_technical_reference_list.html.

Cisco IOS VoiceXML Gateway

You can deploy Cisco IOS VoiceXML Gateway on the same router as you deploy Unified CVP Ingress Voice Gateway. This model is suitable for deployments with small branch offices. The Cisco IOS VoiceXML Gateway can also run on a separate router platform. This model is suitable for deployments with large or multiple voice gateways, where only a small percentage of the traffic is for Unified CVP. Using this model, an organization can share public switched telephone network (PSTN) trunks between office users and contact center agents, and route calls based on the dialed number. VoiceXML Gateway can store audio files on flash memory or on a third-party media server.

For the supported Cisco IOS VoiceXML Gateways, see the *Hardware and System Software Specification for Cisco Unified Customer Voice Portal* at http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_technical_reference_list.html.

Unless a Cisco IOS VoiceXML Gateway is combined with an Ingress Voice Gateway, the Cisco IOS VoiceXML Gateway does not require TDM hardware. It interacts with VoIP on one side and HTTP (carrying VXML or .wav files) and MRCP (carrying ASR and TTS traffic) on the other side. As with Ingress Voice Gateways, Cisco IOS VoiceXML Gateways are often deployed in farms for Centralized deployment models, or one in each office in Branch deployments.

Cisco Egress Gateway

The Cisco Egress Voice Gateway is used only when calls need to be extended to TDM networks or equipment, such as a PSTN or a TDM automatic call distributor (ACD). While the Real-time Transport Protocol (RTP) stream runs between the ingress and egress voice gateway ports, the signaling stream logically goes through the Unified CVP Server and Cisco Unified Intelligent Contact Management (Unified ICM). This allows subsequent call control (such as transfers).

Cisco Integrated Services Router Generation 3 Gateway

Cisco's Generation 3 Integrated Services Router(ISR G3) 4xxx series gateway is certified with Cisco Unified Customer Voice Portal (CVP) version 10.0(1) or later. You can deploy the ISR G3 gateways with Unified CVP in a Cisco Unified Contact Center Enterprise (CCE) solution or in a standalone Unified CVP self-service deployment. For more information about 4xxx series gateway variants, see *Hardware and System Software Specification for Cisco Unified Customer Voice Portal* available at, http://www.cisco.com/c/en/us/support/customer-collaboration/unified-customer-voice-portal/products-technical-reference-list.html.

Note

- ISR G3 gateways do not have a built-in Voice-XML (VXML) browser. Therefore, deploying the ISR G3 ingress gateway with Unified CVP requires the use of a separate ISR G2 gateway to provide the VXML browser.
- Support of the ISR G3 gateways with Unified CCE Outbound Option with Call Progress Analysis requires Unified CCE version 10.5(2) or later.

Video Endpoints

Unified CVP Basic Video Service supports the following features:

- Cisco Unified Video Advantage
- Cisco TelePresence
- Video in Queue (VIQ)

Cisco Unified Communications Manager

Cisco Unified Communications Manager (Unified CM) is the main call processing component of a Cisco Unified Communications system. It manages and switches VoIP calls among IP phones. The Unified CM

combines with Cisco Unified Intelligent Contact Management (Unified ICM) to form Cisco Unified Contact Center Enterprise (Unified CCE). Unified CVP interacts primarily with Unified CM as a means for sending PSTN-originated calls to Unified CCE agents.

The following common scenarios require calls to Unified CVP to originate from Unified CM endpoints:

- An office worker (not an agent) on an IP phone dials an internal help desk number.
- An agent initiates a consultative transfer that gets routed to a Unified CVP queue point.
- A Cisco SCCP Dialer port transfers a live call to a Unified CVP port for an IVR campaign.

A single Unified CM can originate and receive calls from SIP. PSTN calls that arrived on MGCP Voice Gateways registered with Unified CM can be routed or transferred to Unified CVP only through SIP (and not through Cisco Unified Border Element).

Unified CM is an optional component in the Unified CVP solution. Its use in the solution depends on the type of call center being deployed. TDM-based call centers using ACDs, for example, typically do not use Unified CM (except when they are migrated to Cisco Unified CCE), nor do strictly self-service applications that use the Unified CVP Standalone self-service deployment model. Unified CM is used as part of the Cisco Unified CCE solution, in which call center agents are part of an IP solution that uses Cisco IP Phones, or when migrated from TDM ACDs.

Only specific versions of Unified CM are compatible with Unified CVP solutions. Unified CVP is supported with SIP only, see the *Hardware and System Software Specification for Cisco Unified Customer Voice Portal* at http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_technical_reference_list.html.

Cisco Unified Intelligence Center

Cisco Unified Intelligence Center (Unified Intelligence Center) is a web-based reporting application that provides real-time and historical reporting in an easy-to-use, wizard-based application for Unified CCE and Unified CVP. It allows Contact Center supervisors and business users to report on the details of every contact across all channels in the Contact Center from a single interface. Unified Intelligence Center allows you to extend the boundaries of traditional contact center reporting to an information portal where data can be easily integrated and shared throughout the organization.

Administrators can use Unified Intelligence Center to control access to features, reports, and data by granting privileges only to authorized individual users or groups of users. For example, you can assign each supervisor to a group of agents, skills, and call types that are the most relevant to them so that each report provides focused, actionable insights into data that is appropriate to their role.

Several features in this product allow you to extend the Unified Intelligence Center platform beyond traditional Contact Center reporting and into an enterprise-wide information portal. You can use data from nontraditional sources to improve business efficiency and effectiveness.

Cisco Unified Contact Center

Use either Cisco Unified Contact Center Enterprise (Unified CCE) or Cisco Unified Intelligent Contact Management (Unified ICM) when advanced call control, such as IP switching and transfers to agents, is required in Unified CVP. Unified ICM provides call center agent-management capabilities and call scripting capabilities. Variable storage capability and database access through the Unified CCE or Unified ICM application gateways are also powerful tools. A Unified CVP application can take advantage of these capabilities because Unified CVP applications can be called from within a Unified CCE or Unified ICM script in a non-standalone Unified CVP deployment model.

The Unified CVP Call Server maintains a GED-125 Service Control Interface connection to Unified CCE or Unified ICM. GED-125 is a third-party control protocol in which a single socket connection is used to control many telephone calls. For Unified CCE or Unified ICM, Call Server is a voice response unit (VRU) connected to Unified CCE or Unified ICM, as all other GED-125 VRUs are connected. Unified CVP is a VRU peripheral to Unified CCE or Unified ICM.

Note You can use the hosted versions of Unified CCE and Unified ICM for advanced call control.

SIP Proxy Server

SIP Proxy Server routes SIP messages among SIP endpoints. SIP Proxy Server is required for Unified CVP high-availability architecture for call switching. SIP Proxy Server is designed to support multiple SIP endpoints of various types and to implement load balancing and failover among these endpoints. Deployment of a SIP Proxy in the solution enables a more centralized configuration of the dial plan routing configuration.

You can configure a SIP Proxy with multiple static routes to do load balancing and failover with outbound calls. The static routes can point to an IP address or a DNS.

Domain Name System (DNS) Service Record (SRV) is also supported. However, it is not qualified for use on the Cisco Unified SIP Proxy (CUSP) Server. It is qualified for the devices that must reach the CUSP Server, such as Unified CVP, Ingress Voice Gateway, and Unified Communications Manager.

You can deploy Unified CVP without a SIP Proxy Server, depending on the design and complexity of the solution. In such cases, some of the functions that a SIP Proxy Server provides are provided by the Unified CVP Server SIP service.

Following are the benefits of using a SIP Proxy Server:

- You can use priority and weight routing with the routes for load balancing and failover.
- If a SIP Proxy Server exists in your SIP network, then Unified CVP acts as an additional SIP endpoint. The Unified CVP fits incrementally into the existing SIP network.

If you do not use a SIP Proxy Server, then the Ingress Voice Gateways and Unified CMs must point directly to Unified CVP. In such a deployment, perform the following tasks:

- Perform load balancing using DNS SRV lookups from gateway to DNS Server; balance SIP calls using this procedure.
- Perform load balancing of calls outbound from Unified CVP (outbound call leg) using DNS SRV lookups.
- Perform failover of SIP rejections (code 503 only), if you configure SRV records with ordered priorities.

DNS Server

You can install Domain Name System (DNS) Server anywhere in the network. Its purpose is to resolve hostnames to IP addresses. Unified CVP can make both Type A record lookups and SRV Type record lookups. If a DNS Server is slow to respond, is unavailable, or is across the WAN, the performance of Unified CVP is affected.

The DNS Server is used during SIP interactions in the following situations:

- When a call arrives at an Ingress Voice Gateway, the dial peer can use DNS to alternate calls between the two SIP Proxy Servers. The SIP Proxy Servers can also use DNS to distribute incoming calls among multiple SIP Services. If SIP Proxy Servers are not being used, then the Ingress Voice Gateway can use DNS directly to distribute inbound calls among multiple SIP Services.
- Unified CCE directs the SIP service to transfer calls to the VRU leg and can use DNS to alternate such requests between two SIP Proxy Servers. If SIP Proxy Servers are not being used, the SIP Service can use DNS directly to distribute VRU legs among multiple VoiceXML Gateways.
- When transferring a call to an agent using a SIP Proxy Server, the SIP Proxy cannot use DNS SRV for outbound calls; it must be configured with multiple static routes to do load balancing and failover. The static routes can point to an IP address or a regular DNS A host record. If SIP Proxy Servers are not being used, then the SIP Service can use DNS to locate the IP address of the target agent.

The use of the DNS Server for SIP routing is optional in Unified CVP. You do not need to have a dedicated DNS Server, as the existing DNS server handles the additional load of Unified CVP. For every call destined for Unified CVP that comes into the network, there are approximately three to four DNS lookups. You can determine the number of DNS queries per second by determining the number of calls per second for the solution, and multiplying that number by 4.

DNS lookups are needed for DNS SRV queries, not necessarily for A record queries, which can also be configured locally in the system file. You can use Unified CVP Server Groups alternately to avoid DNS SRV lookups.

Load Balancers

Application Control Engine

As a load-balancing device, ACE determines which server in a set of load-balanced servers receives the client request for service. Load balancing helps fulfill the client request in the shortest amount of time without overloading either the server or the server farm as a whole.

See *Cisco ACE 4700 Series Appliance Server Load-Balancing Configuration Guide* (http://www.cisco.com/en/US/docs/app_ntwk_services/data_center_app_services/ace_appliances/vA3_1_0/configuration/slb/guide/slbgd.html) to learn more about load-balancing with ACE.

To migrate from CSS to ACE, use the CSS2ACE Converter tool. See http://wwwin.cisco.com/dss/adbu/dcas/adoptions/cssmigration/ for more information.

To configure Unified CVP for ACE. See the *Configuration Guide for Cisco Unified Customer Voice Portal* available at http://www.cisco.com/en/US/products/sw/custcosw/ps1006/tsd_products_support_series_home.html.

You must have an ACE license to use ACE under load conditions. The minimum licensing requirements for ACE are:

- 1-Gbps throughput license (ACE-AP-01-LIC)
- A non-default SSL feature license, if you intend to use ACE for SSL
- Application Acceleration License (ACE-AP-OPT-LIC-K9), which allows more than 50 concurrent connections on ACE

See the *ACE product documentation* and *ACE release notes* for more licensing information.

Note There are two features for the VXML Server that assist with load balancing:

- Limiting load balancer involvement
- Enhanced HTTP probes for load balancers

See the configuration options ip_redirect and license_depletion_probe_error in the *User Guide for Cisco Unified CVP VXML Server and Cisco Unified Call Studio*, available at http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_user_guide_list.html.

Third-Party Load Balancers

Note The Unified CVP solution components recommend the use of load balancers to provide load distribution and high availability for HTTP, HTTPS, and MRCP traffic. Load Balancers provides the capability to load balance the rendering of the VXML pages between VoiceXML Gateways and VXML Server and also fetching of the media files for IVR scripts execution from media servers.

The Unified CVP now supports any third-party load balancer possessing the following features:

- Both SSL offloading and SSL pass-through has to be supported.
- Load balancer high availability.
- Session stickiness should not be mandatory.
- Persistence is cookie-insert.
- Distribution algo is round-robin.

The interoperability notes and the known caveats for most commonly used third-party load balancers, such as the Big-IP F5 and the Citrix NetScalar 1000v, can be referred from the following locations:

- For BIG-IP F5 refer to:

  http://www.cisco.com/c/dam/en/us/solutions/collateral/enterprise/interoperability-portal/bigip.pdf

- For Citrix NetScalar 1000v refer to:

  http://www.cisco.com/c/en/us/solutions/enterprise/interoperability-portal/voice_portal.html

Cisco Unified Border Element

Cisco Unified Border Element is a Cisco IOS Session Border Controller (SBC) that interconnects independent Voice over IP (VoIP) and video over IP enterprise networks for data, voice, and video transport. SBCs are an industry class of devices that are critical components for scaling networks from VoIP islands within a single customer network to an end-to-end IP community, both inside the enterprise and to communicate beyond the enterprise across service provider networks or to other enterprises and small and medium businesses (SMBs).

Cisco Unified Border Element is an integrated Cisco IOS software application that runs on Cisco Integrated Services Routers (ISRs), Cisco universal gateways, and ASR routers. For details on supported versions, see the *Hardware and System Software Specification for Cisco Unified Customer Voice Portal* at http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_technical_reference_list.html.

Design Considerations

Please observe the following restrictions when deploying CUBE with SIP Trunks:

- Before 15.2(1)T IOS release, a dial-peer was required to pass the Refer-to header URI through CUBE. Starting from 15.2(1)T release onwards refer-to-passing command can be used without the need for a dial-peer.
- CUBE must be configured in media pass through mode in the Unified CVP deployment. Media flow around mode cannot be used because it is not supported or validated. Only media pass through mode, the default mode on the dial-peer, is supported for CUBE.
- CUBE does not currently support passing the Refer-To header URI designation from CVP when a REFER call flow is initiated. It rewrites the destination address based on the dial peer configuration. Therefore the dial plan must be configured on CVP and CUBE. The note below explains the behavior.
- REFER passthrough cannot be used in conjunction with Survivability. The script does not let REFER messages be relayed to a SIP service provider regardless of other CUBE configuration.
- REFER consume cannot be used in conjunction with Survivability and Router Requery. Survivability always accepts the REFER, even if the transfer does not complete. Unified CCE deems the transfer successful and does not attempt to requery.
- Survivability cannot be used when service provider Alternate Destination Routing (ADR) is used because the script does not let error messages (ring-no-answer or busy) reach the service provider. Manipulation in the script does not let error messages (ring-no-answer or busy) reach the service provider. Manipulation in the Remote-Party-ID header is required instead.
- If GTD is present on the incoming call or if Unified CCE sets a value for the UUI variable, Unified CVP will send a BYE immediately after outpulsing digits in a DTMF transfer. If a delay is required between the digits then comma should be used at the end of the label.
- If GTD is not present on the incoming call, Unified CCE does not set a value for the UUI variable and the service provider does not disconnect a call after receiving digits in a DTMF transfer. Unified CVP will send a BYE request after the SIP.ExternalTransferWait timer expires. Previous versions of Unified CVP did not disconnect the call.
- Survivability is required when Courtesy Callback is used.

Video Media Server

Video Media Server is required for uploading, storing, and playing back of video prompts. Cisco MediaSense is a Video Media Server that provides network-based multimedia capture, streaming, and recording. Cisco MediaSense records conversations on the network rather than on a device. This process simplifies the architecture, lowers costs, provides optimum scalability, and facilitates use by analytics applications from Cisco technology partners.

Automatic Speech Recognition Server and Text-to-Speech Server

Speech recognition servers can provide Automatic Speech Recognition (ASR), Text-to-Speech (TTS), and DTMF recognition services for a VoiceXML Gateway. Communication between the speech recognition servers and the VoiceXML Gateway uses Media Resource Control Protocol (MRCP).

The following table describes support for the MRCP protocol on IOS VXML Gateway and Cisco VVB:

Application	Supports MRCP v1	Supports MRCP v2
Micro-Apps with speech server	IOS VXML and VVB	VVB
Studio applications with speech server	IOS VXML and VVB	OS VXML and VVB

For capacity and redundancy reasons, an Application Control Engine (ACE) is used to mediate between a VoiceXML Gateway and a farm of ASR or TTS servers. If ACE is not used, then a VoiceXML Gateway can support a maximum of two ASR or TTS Servers.

Cisco does not sell or support any ASR or TTS software or servers. However, Cisco tests Unified CVP with Nuance products. A certification process is currently being developed to allow additional vendors to qualify their products against Unified CVP VoiceXML, and the World Wide Web Consortium (W3C) provides a rich feature set to support the ASR grammars. It is easy to implement and support inline grammars, by which the set of acceptable customer responses is passed to the VoiceXML Gateway. Another form is external grammars, where Unified ICM passes a pointer to an external grammar source. The VXML Server adds this pointer to the VoiceXML document that it sends to the VoiceXML Gateway, which then loads the grammar and uses it to check ASR input from the caller. In this case, the customer creates the grammar file. A third type of grammar is the built-in grammar. For a complete explanation of grammar formats, see the W3C website at http://www.w3.org/TR/speech-grammar/.

The text for TTS is passed directly from the VXML Server to the VoiceXML gateway. This action is referred to as inline TTS in this document.

The actual speech recognition and speech synthesis are performed by a separate server that communicates with the VoiceXML Gateway through MRCP. Currently, Nuance is the only tested ASR and TTS engine. The ASR and TTS engine also supports (with limitations) voice recognition and synthesis for multiple languages.

For information on Nuance, see http://www.nuance.com.

Nuance is a third-party product, which the customer or partner must purchase directly from the vendor. The customer also receives technical support directly from the vendor. However, that does not mean that the vendor's latest software version can be used. Unified CVP is tested with specific versions of each vendor's product. Cisco Technical Assistance Center (TAC) does not support Unified CVP customers who use different ASR and TTS versions apart from those which have been tested. For details on the supported ASR and TTS products, see the *Hardware and System Software Specification for Cisco Unified Customer Voice Portal* at http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_technical_reference_list.html.

Network Monitor

You can use Simple Network Management Protocol (SNMP) station to monitor the solution deployment status. See the *Operations Console User Guide for Cisco Unified Customer Voice Portal*, available at http://www.cisco.com/c/en/us/support/customer-collaboration/unified-customer-voice-portal/products-user-guide-list.html.

Call Flows

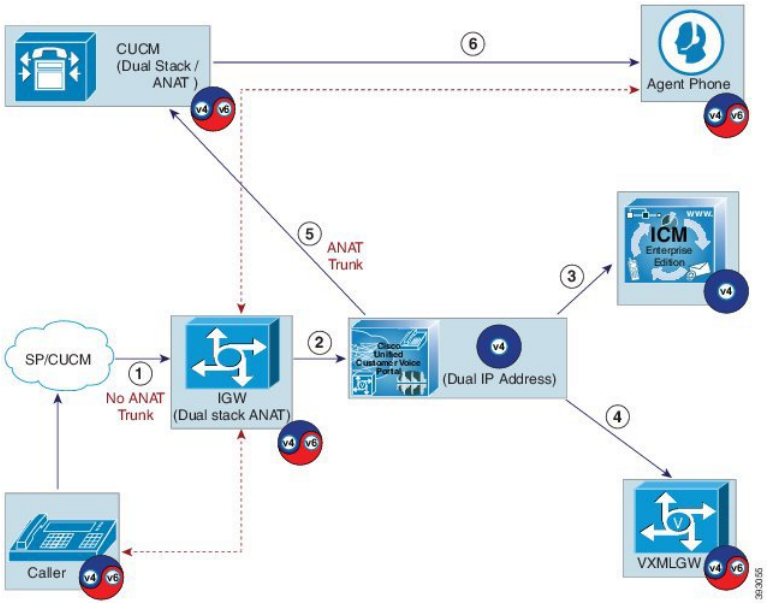
IPv6 Architecture for Unified CVP

Beginning with Unified CVP Release 11.0(1), you can use either IPv6-only or a mix of IPv4 and IPv6 endpoints. Servers that communicate with those endpoints can now accept IPv6 connections, in addition to IPv4 connections. Communication between servers continues to use IPv4 connections.

Note Ensure that you use a separate Ingress Gateway and VXML Gateway for both the Standalone and Comprehensive call flow models.

The following figure displays the Comprehensive call flow model for IPV6. The solid lines indicate the signal paths and dashed lines indicate the media path.

Figure 2: Comprehensive call flow model for IPv6



A typical Comprehensive call flow for IPV6 is as follows:

1. A customer call arrives from the service provider SIP trunk or internal CUCM trunk to Ingress Gateway (either IPv4 or IPv6 trunk).
2. The Ingress Gateway sends the call to Unified CVP.

Note The Ingress Gateway to CVP connection is on IPV4 with Alternative Network Address Types (ANAT) enabled.

3. Unified CVP sends the incoming call request to ICM in IPv4 and gets a VRU label.

4. Unified CVP sends the call to the VoiceXML Gateway. The caller hears the IVR.
5. Unified CVP sends the call to an agent with IPv4 signal and IPv4 and IPv6 media.

Note The Unified CVP to Unified Communications Manager trunk is dual stack with ANAT enabled.

6. Unified CM Communications Manager connects the caller to an available agent. The agent can be either on IPv4 or IPv6 phone configuration.

Note Cisco VVB does not support IPv6.

For information about IPv6 configuration for CVP, see the *Configuration Guide for Cisco Unified Customer Voice Portal* at http://www.cisco.com/c/en/us/support/customer-collaboration/unified-customer-voice-portal/products-installation-and-configuration-guides-list.html.

IPv6 Design Considerations

Following are the design considerations for IPv6:

- For Call Survivability function: The incoming trunk to Gateway must be configured as IPv4 only.
- For Courtesy Callback and Refer feature: If the same incoming trunk is used for outbound dialing, then the session target in Ingress Gateway dialpeer must be same (either IPv4 or IPv6) as the incoming trunk.

Video Endpoints Design Considerations

Following are the design considerations for Video Endpoints:

- Configure the incoming trunk to Gateway in IPv4 mode only.
- Disable ANAT in the Ingress Gateway.
- Configure the Agent endpoint either in IPv4 or dual IP mode only.

Typical SIP Unified CVP Call Flow

The description follows a typical SIP Unified CVP Call Flow (comprehensive call flow model). However, it is an illustration and is not an actual solution. This can only be considered as an introduction to the overall flow of information in a Unified CVP solution.

The call flow consists of an incoming call requiring initial self-service, followed by a queue treatment, and finally delivery to a Unified ICM agent. The following diagram presents a general SIP-based solution.

The following is a typical SIP Unified CVP call flow:

1. The call arrives at an Ingress Voice Gateway and sends an invite message to the SIP Proxy Server that forwards the message to the SIP Service.
2. The Proxy Server determines the IP address of the Unified CVP Server for the dialed number and forwards the invite to the selected Unified CVP Server SIP service.

3. The SIP service consults Unified ICM through the Unified CVP Server ICM Service, which causes Unified ICM to run a routing script.
4. The routing script typically initiates a transfer of the call to a VoiceXML Gateway port through the SIP service.
5. The VoiceXML Gateway sends a message to the IVR service, which requests scripted instructions from Unified ICM.
6. Unified ICM exchanges VRU instructions with the VoiceXML gateway through the IVR service. The instructions can include requests to invoke more sophisticated applications on the Unified CVP VXML server. These requests result in multiple exchanges between the Unified CVP VXML Server and the VoiceXML Gateway to provide self-service.
7. To transfer to a live agent, the Unified ICM routing script queues the caller for an available agent. While waiting for an available agent, Unified ICM provides additional instructions to the VoiceXML Gateway to provide queueing treatment to the caller.
8. When an agent becomes available, Unified ICM sends a message to the Unified CVP Server SIP Service, which forwards a message through the SIP Proxy Server to the Ingress Gateway and to the Unified CM to transfer the call away from the VoiceXML Gateway port and deliver it to the Unified CM agent IP phone.

Design Process

When designing a Unified CVP deployment consider the following high-level steps:

1. Choose a call flow model for your functional deployment.
2. Determine where the Unified CVP components are going to be deployed (in the data center or at a branch).
3. Choose the amount of availability and resiliency that is justifiable or required.
4. Size the deployment to provide the justifiable or required grade of service for the initial deployment and near-term growth.

Note SIP is the only supported call control signaling protocol.

Call Flow Models

The first step in the design process is to determine the functionality you need. Unified CVP offers a number of call flow models to support differing needs. The deployment model you choose depends on the call flow preferences, geographic distribution requirements, and hardware configurations that best satisfy your company's needs.

- Unified CVP VXML Server (standalone) — Provides a standalone VRU with no integration to Unified ICM for queuing control or subsequent call control. Used to deploy self-service VXML applications.
- Call Director — Provides IP switching services only.

  This model is useful if you want to:

    - Only use Unified CVP to provide Unified ICM with VoIP call switching.
    - Prompt and collect data using third-party VRUs and ACDs.
    - Avoid using an Unified CVP VXML Server.

- Comprehensive — Provides IVR services, queue treatment, and IP switching services. This model is useful if you want to,
    - Use Unified CVP to provide Unified ICM with VoIP call switching capabilities.
    - Use Unified CVP to provide Unified ICM with VRU services, including integrated self-service applications, queuing, and initial prompt and collect.
    - Use the video IVR, video queuing, and video agent capabilities.
    - Use an optional Unified CVP VXML Server.
    - Prompt or collect data using optional ASR and TTS services.

- VRU Only — Provides IVR services, queuing treatment, and switching for PSTN endpoints. This model relies on the PSTN to transfer calls between call termination endpoints.

  This model is useful if you want to:
    - Use Unified CVP to provide Unified ICM with VRU services including integrated self-service applications and initial prompt and collect.
    - Avoid using an Unified CVP for switching calls.
    - Use an optional Unified CVP VXML Server.
    - Prompt or collect data using optional ASR and TTS services.

For more details and design considerations for each of these functional deployment models, see Functional Deployment for details.