Distributed Deployment

Distributed Gateways

Unified CVP can use different types of gateways depending on the deployment model. This section discusses each type of voice gateway and their effects in a distributed deployment.

Ingress or Egress Voice Gateway at the Branch

In this deployment model, Ingress Voice Gateways located at a branch office are typically used to provide callers with access using local phone numbers instead of using centralized or non-geographic numbers. This capability is important in international deployments spanning multiple countries. Egress Gateways are located at branches either for localized PSTN breakout or for integration of decentralized TDM platforms into the Unified CVP switching solution. All other Unified CVP components are centrally located, and WAN links provide data connectivity from each branch location to the central data center.

Ingress or VoiceXML Gateway at the Branch

Consider other voice services that run at the branch that can affect Ingress or VoiceXML Gateways. For example, the branch is a remote Cisco Unified Communications Manager (Unified CM) site supporting both ACD (Agent Desktop provides call control capabilities ready/not ready, wrap up.) agent and non-agent phones. In this model, the PSTN gateway is used for ingress of Unified CVP calls as well as Ingress and Egress of normal user calls. In circumstances when the VoiceXML and Voice Gateway functions reside at the same branch location but on separate devices, special attention has to be given to the dial plan to ensure that the VRU leg is sent to the local VoiceXML resource. This is because the Unified CVP Call Server **settransferlabel** label applies only to coresident VoiceXML and Voice Gateway configurations.

When the Ingress Voice Gateway and the VoiceXML Gateway at a branch do not reside on the same Gateway, there are two ways to ensure that the calls are handled within the branch and not sent through the WAN to a different VoiceXML Gateway:

- Configure Unified ICM with multiple customers, one Unified ICM configuration.

  The Unified ICM configuration differentiates between calls based on the Dialed Number. The Dialed Number is associated with a customer representing the branch site. When a NetworkVRU is needed, the NetworkVRU associated with the customer in Unified ICM is selected and the caller is sent to that NetworkVRU. This method allows you to have multiple NetworkVRUs, each with a unique label. The disadvantage of this method is that each NetworkVRU requires its own VRU scripts in Unified ICM.

- Configure Unified CVP using the SigDigits feature.

  The SigDigits feature allows you to use the dial plan on the SIP Proxy to route calls to the correct site. When the call arrives at an Ingress Voice Gateway, the gateway prepends digits before sending the call to Unified CVP. Those prepended digits are unique to that site for a dial plan.

  When the call arrives at Unified CVP, Unified CVP strips the prepended digits and stores them in memory, resulting in the original DID on which the call arrived. Unified CVP then notifies Unified ICM of the call arrival using the original DID and matches a Dialed Number in Unified ICM.

  When Unified ICM returns a label to Unified CVP to transfer the call to a VoiceXML gateway for IVR treatment or to transfer the call to an agent phone, Unified CVP prepends the digits that it stored in memory before initiating the transfer. The dial plan in the SIP Proxy must be configured with the prepended digits to ensure that the calls with a certain prepended digit string are sent to specific VoiceXML Gateways or Egress Gateways.

  When the VoiceXML Gateway receives the call, the CVP bootstrap service is configured to strip the digits again, so that when the IVR leg of the call is set up, the original DN is used on the incoming VoiceXML request.

  

  Note The digits can be prepended to translation route DNs, and that the egress or receiving component (such as Unified CM) may need to strip digits to see the original DN.

  The term SigDigits is used to describe this feature because the command in Unified CVP to turn on the feature and specify how many significant digits should be stripped is called Prepend Digits for SIP in the operations console.

  This method is preferred because it involves the least amount of Unified ICM configuration overhead: a single NetworkVRU and single set of VRU scripts and Unified ICM routing scripts. This allows all of the Unified CVP Servers and VoiceXML Gateways to function as a single network-wide virtual IVR from the perspective of Unified ICM.

  The SigDigits feature can also be used to solve multicluster call admission control problems. (See Call Admission Control Considerations , on page 5, for more information.)

Colocated VXML Servers and VoiceXML Gateways

Either all gateways and servers are centralized or each site has its own set of colocated Unified CVP VXML Servers and VoiceXML Gateways.

Colocation has the following advantages:

- A WAN outage does not impact self-service applications.
- No WAN bandwidth is required for VoiceXML.

Colocation has the following disadvantages:

- Extra Unified CVP VXML Servers are required when using replicated branch offices.
- Additional overhead is required when deploying applications to multiple Unified CVP VXML Servers.

Gateways at Branch with Centralized VXML Server

Advantages of centralized VoiceXML:

- Administration and reporting are centralized.
- Unified CVP VXML Server capacity can be shared among branch offices.

Disadvantages of centralized VoiceXML:

- Branch survivability is limited.
- WAN bandwidth must be sized for additional VoiceXML over HTTP traffic.

Cisco Unified Communications Manager

In a Unified CVP environment, Unified CM can be an Ingress or Egress Gateway. It is more common for Unified CM to be an Egress Gateway because calls typically are from the PSTN, queued by Unified CVP, and then switched to Unified CM for handling by an agent. If the call is not from the PSTN, but from an IP phone, the Unified CM is an Ingress Voice Gateway from the perspective of Unified CVP.

Unified CM as an Egress Gateway

To deploy Unified CM with Unified CVP, you must use Unified CM call admission control for calls between the Ingress Voice Gateway and the agent IP phone. Therefore, Unified CM recognizes the call coming from the centralized Unified CVP Call Server instead of from the Remote Ingress Voice Gateway.

Unified CM as an Ingress Voice Gateway

When an IP phone initiates a call to Unified CVP, the Unified CM acts as the Ingress Voice Gateway to Unified CVP. A SIP trunk is used to send calls to Unified CVP. For more information on Unified CVP call flows, see Calls Originated by Cisco Unified Communications Manager.

Multicast Music-on-Hold

Multicasting is used for Music-on-Hold (MOH) with supplementary services on Unified CM as an alternative to the unicast MOH. There are two ways to deploy MOH using this feature:

- With Unified CM multicasting the packets on the local LAN
- With the branch gateway multicasting on their local LAN

Use the latter method when survivable remote site telephony (SRST) is configured on the gateway. This method enables the deployment to use MOH locally and avoid MOH streaming over the WAN link.

Note Refer to the following location for information about configuring MOH on the Call Manager Enterprise (CME):

http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmemoh.html#wpmkr1022205

Multicast MOH Usage Guidelines

The following guidelines apply when using Multicast MOH:

- Do not use this setting globally, or on a dial peer on the Ingress or Egress Gateway:

  ```
  modem passthrough nse codec g711ulaw
  ```

  This setting might cause Unified CM to stop the MOH after a timeout period of 10 to12 seconds.
- Do not set media inactivity on the Ingress Voice Gateway because multicast MOH does not send RTP or RTCP, and the call might get disconnected due to media-inactivity configuration. The setting media-inactivity criteria does not support multicast traffic.
- SIP-based multicast MOH is not supported on a 5400 platform because CCM-manager-based MOH subsystems are not supported on 5400 platform. This limitation also affects the ability of a TDM caller to hear multicast packets broadcasted from the Unified CM MOH server.

Call Survivability in Distributed Deployments

Distributed deployments require design guidelines for other voice services that are being run at the branch. For example, the branch is a remote Unified CM site supporting both ACD agent and nonagent phones. This deployment also implies that the PSTN Gateway is used not only for ingress of Unified CVP calls but for ingress or egress of the regular non-ACD phone calls.

Branch reliability in WANs may be an issue in a centralized Unified CVP model because they are typically less reliable than LAN links. The call survivability function must be considered for both the Unified CVP and non-CVP calls. For Unified CM endpoint phones, survivability is accomplished by using a Cisco IOS feature known as Survivable Remote Site Telephony (SRST). For further details on SRST, see the latest version of the *Cisco Unified Communications SRND Based on Cisco Unified Communications Manager*, available at:

http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-implementation-design-guides-list.html

For Unified CVP calls, survivability is handled by a combination of services from a TCL script (survivability.tcl) and SRST functions. The survivability TCL script monitors the SIP connection for all calls that ingress through the remote gateway. If a signaling failure occurs, the TCL script takes control of the call and redirects it to a configurable destination. The destination choices for the TCL script are configured as parameters in the Cisco IOS Gateway configuration.

Note When the called number is in "E164" format, the survivability script removes the "+" sign from the called number before forwarding it to Unified CVP. This is because Unified CVP or ICM does not support the "+" sign in the beginning of DNIS.

Alternative destinations for this transfer include another IP destination (including the SRST call agent at the remote site), *8 TNT, or hookflash. With transfers to the SRST call agent at the remote site, the most common target is an SRST alias or a basic ACD hunt group. For further information about these SRST functions, see the *Cisco Unified Communications Solution Reference Network Design (SRND) based on Cisco Unified Communications Manager*.

Voice mail and recording servers do not send Real-Time Control Protocol (RTCP) packets in reverse direction toward the caller (TDM Voice Gateway), which can falsely trigger the media inactivity timer of the survivability script. It is important to apply the survivability.tcl script carefully to the dial peers because a call might drop if it goes to the voice mail or to a recording element. One method is to use a separate dial peer for voice mail or recording calls, and do not associate the Unified CVP survivability script for those dial peers. Another method is to disable the media inactivity on the survivability script associated with the voice mail or recording dial peers.

For further information on configuration and application of these transfer methods, see the latest version of *Configuration Guide for Cisco Unified Customer Voice Portal*, available at:

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

You can also refer to CUBE Deployment with SIP Trunks.

Note To take advantage of alternate routing on signaling failures, you must use the survivability service on all gateways pointing to Unified CVP. Always use this service, unless you have a specific implementation that prevents using it.

Router requery is not supported when using SIP REFER with Unified CVP Comprehensive Call Flow when the survivability service is handling the REFER message from Unified CVP. Router requery with REFER can be supported in other call flows when Cisco IOS is handling the REFER without the survivability service or if Unified CM is handling the REFER. For third-party SIP trunks, the support of router requery with REFER is dependent on their implementation and support for SIP REFER.

Call Admission Control Considerations

Call admission control can be considered as a solution and not just a Unified CVP component. These considerations are most evident in the distributed branch office model where there are other voice services, such as Unified CM, sharing the same gateways with Unified CVP and the amount of bandwidth between the sites is limited. Be sure that, call admission control methods are in place on the network so that the same call admission control method is used for all the calls traversing the WAN from that site. If two call admission control methods can admit four calls each and the WAN link can handle only four calls, then it is possible for both call admission control entities to admit four calls onto the WAN simultaneously. This control method impairs the voice quality. If a single call admission method cannot be implemented, then each call admission control method must have bandwidth allocated to it. This situation is not desirable because it leads to inefficient bandwidth overprovisioning.

Two call admission control methods can be used in a Unified CVP environment: Unified CM Locations and Unified CM RSVP Agent. In a single-site deployment, call admission control is not necessary.

Unified CM performs call admission by assigning devices to certain locations and keeping track of the number of calls that are active between these locations. Unified CM knows the number of calls that are active and the codec use for each call, so that it can calculate the bandwidth used and limit the number of calls allowed.

A thorough conceptual understanding of call admission control features is important. These features are explained in the *Cisco Unified Communications SRND Based on Cisco Unified Communications Manager*, available at:

http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-implementation-design-guides-list.html

Unified CM Call Admission Control

If Unified CM sends or receives calls from Unified CVP and there are Unified CVP gateways and IP phone agents collocated at remote sites, it is important to understand the call flows in order to design and configure call admission control correctly.

SIP Call Flows

With SIP-based call flows, Cisco Unified CM Release 6.0 (and earlier releases) can look at only the source IP address of the incoming SIP INVITE from Unified CVP. This limitation causes a problem with call admission control because Unified CM cannot identify the gateway that the Unified CVP call originated.

You can use the SIP trunk feature to look beyond the source IP address and to inspect information in the SIP header when determining the device that originated a call. This enhancement allows the SIP trunk to be dynamically selected by the original source IP address instead of the remote port on Unified CVP. The SIP profiles and settings can be used on the source trunks that are different from the Unified CVP trunk.

The Call-Info header in the SIP INVITE specifies the originating device in the following format:

<sip: *IPAddress:port*>;purpose=x-c isco-origIP

The *IPAddress:port* value indicates the originating device and its SIP signaling port.

This source IP SIP trunk selection feature does not impact the bandwidth monitoring for call admission control. In Unified CM, bandwidth monitoring is performed with SIP using locations configuration on Unified CVP and Unified CM. The following header is used by the location server in Unified CM to manipulate bandwidth information for call admission control:

Call-Info: [urn:x-cisco-remotecc:callinfo];x-cisco-loc-id="PKID";x-cisco-loc-name="Loc-NAME"

Resource Reservation Protocol

Resource Reservation Protocol (RSVP) is used for call admission control, and it is used by the routers in the network to reserve bandwidth for calls. RSVP is not qualified for call control signaling through the Unified CVP Call Server in SIP. The recommended solution for CAC is to use the Locations configuration on Unified CVP and in Unified CM.

For more information on RSVP, see the latest version of the *Cisco Unified Communications SRND Based on Cisco Unified Communications Manager*, available at:

http://www.cisco.com/c/en/us/support/unified-communications/unified-communications-manager-callmanager/products-implementation-design-guides-list.html