Features and Functions

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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.

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


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.

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:


You can also refer to CUBE Deployment with SIP Trunks.
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.

### Video in Queue

Video in Queue (VIQ) is an optional Basic Video feature in Unified CVP. It allows the caller to interact through high-definition video prompt or navigate a video menu using DTMF keys. The following figure displays the topology and call flow for Basic Video.

*Figure 1: Video in Queue*

The Unified CVP Studio VideoConnect element allows the specific video prompt to be played for video endpoints. It also allows the DTMF input during video-prompt playback to be collected and integrated with the Unified Call Studio or Unified CCE scripting environment.

*Note* Video in Queue is not played during a CUCM failover.

See the *Configuration Guide for Cisco Unified Customer Voice Portal* for specific Cisco Unified Border Element or VXML Gateway configuration information for VideoConnect.
See the Element Specifications for Cisco Unified CVP VXML Server and Cisco Unified Call Studio for using the VideoConnect element.

See “Incoming Call Configuration and Media File Management” in the MediaSense User Guide to use media files.

When configuring the Video in Queue for Unified CVP, set the MediaSense **Incoming Call Configuration > Action** to play once.

### Custom SIP Headers

The Custom SIP Header feature enables Unified CVP to pass selected SIP header information to and from Unified ICM for modification within ICM scripts. This feature allows much greater flexibility in providing SIP interoperability with third-party SIP trunks and gateways.

### Passing Information in SIP Headers to Unified ICM

Unified CVP enables the passing of one or more SIP headers to Unified ICM for manipulation within the ICM script. Unified CVP administrator can use the Unified CVP Operations Console Server user interface (Operations Console) to select a specific header, or a header and specific parameters within that header. These SIP headers can be passed to Unified ICM in the SIPHeader field of the New Call and Request Instruction messages sent from the CVP ICM subsystem to Unified ICM.

To access the variable in the ICM script, access the Call.SIPHeader field. Setting this field causes Unified CVP to use that data in outbound SIP calls to IVR or Agents, or REFER or 302 redirect messages.

The amount of space available to send header data to Unified ICM is limited and is truncated to 255 bytes. The SIP protocol RFC provides a function to represent common header field names in an abbreviated form. The compact header format as defined in RFC 3261 (and other RFCs for newly defined headers) is used for the header titles before passing the header to Unified ICM.

Not all headers have a compact format. For example, P-Headers or private headers (for example P-Asserted-Identity) may not have a compact form, and the full header name is passed to ICM.

See the table in the RFC3261 that defines the compact header abbreviations.

### String Formats and Parsing

The following example shows the formatting of a string sent to Unified ICM based on Operations Console SIP configuration screen settings:

```
"User-to-User: 123456789"
"f:Name <sip:from@127.0.0.1:6666>;param1;param2|v:SIP/2.0/UDP viaHost"
```

The delimiter is the bar character.
The data may be parsed with string manipulation syntax in the script such as this example.

Caution

No syntax checking.

There is no syntax checking while adding or modifying headers in the Operations Console. You must be careful that the headers are in correct SIP syntax. The only characters not allowed in Operations Console input are the semicolon and the comma because these are used internally to store the configuration. Typically, if there is a problem with the header syntax, the CVP log shows that the INVITE is not sent due to a SIP stack-parsing exception, and the call is aborted. In other cases, if a mandatory SIP header is modified incorrectly, the call itself may get sent to an unexpected destination or the receiver may not be able to handle the call if the message is not conforming to RFC.

Passing of Headers from the ICM Script

The objective of this feature is to provide a scriptable option to modify SIP headers on the outgoing Unified CVP transfer. You can specify SIP header values in outgoing SIP INVITES only. The specifying can include the addition, modification, or removal of header values.

Note

The SIP header modification feature is a powerful tool which can tweak SIP headers as needed. Exercise caution when applying SIP Profiles and ensure that the profile does not create interop issues, rather than solving them. Unified CVP provides the flexibility to add, modify, or remove outgoing SIP header in the INVITE message only. You can deploy Unified CVP in many scenarios to facilitate interoperability with third-party devices.

Outgoing SIP header feature do not allow you to remove or add Mandatory SIP headers. Only the modify option is available for basic mandatory headers, such as To, From, Via, CSeq, Call-Id and Max-Forwards. There is no checking for the modifications in the ICM script editor, it is actually enforced by the java SIP stack layer by throwing a DsSipParseException.

Typically, with Unified ICM, if the field is greater than 255 characters then it is truncated. In the SIP subsystem, if there is a problem updating or adding a header with the string provided the Unified ICM script, then you either see an WARN type message in the Unified CVP log, if there is an DsSipParseException, or else sends the INVITE sends unexpected results on the receiver end.

This feature is applicable only for outgoing SIP INVITES (only the initial INVITE, not reinvites). Changes to the INVITE are applied just before it is sent out. There is no restriction on the changes that can be applied. The header length (including header name) after modification should not exceed 255.

Examples of Unified ICM Scripting for Custom SIP Headers

In the script editor, the Set node is used to set the call variable string for SIPHeaderInfo.

In the Unified ICM script delimit the header, operation, and value with a tilde character, and use the bar character to concatenate operations.

Scripting Examples for Outbound Header Manipulations
### Example | Notes
--- | ---
"Call-Info~add~<sip:x@y>;parm1=value1" | Adds a Call-Info header with the proper call info syntax as per RFC3261.
"Via~add~SIP/2.0/UDP viaHost" | Adds a Via header to the message.
"v~add~SIP/2.0/UDP viaHost|f~mod~<sip:123@host>;parm1=value1" | Short Form notation, plus concatenated operations. Adds a Via header and modifies the From header.
"Call-Info~add~parm1=value1" | **Incorrect:** This will fail due to incorrect syntax of Call-Info header per RFC 3261. You will see a WARN message in the CVP log. This is enforced in the stack.
"From~add~<sip:x@y>;parm1=value1" | From header add and modify will do the same thing, since only one From header is allowed in a message per RFC 3261. This is enforced in the stack.
"Call-ID~add~12345@xyz" | Same as From header, only one allowed.
"Call-ID~mod~12345@abc" | Same as From header, only one allowed.
"User-To-User~mod~this is a test|P-Localization-Info~mod~1234567890" | Can be used to concatenate operations in one ICM variable Set Node.
"Call-ID~rem" | Removes the first header called Call-Id in the message.


## Courtesy Callback

Courtesy Callback reduces the time callers have to wait on hold or in a queue. The feature enables your system to offer callers, who meet your criteria, the option to be called back by the system instead of waiting on the phone for an agent. The caller who has been queued by Unified CVP can hang up and subsequently be called back when an agent is close to becoming available (preemptive callback). This feature is provided as a courtesy to the caller so that the caller does not have to wait on the phone for an agent.

Preemptive callback does not change the time a customer must wait to be connected to an agent, but rather enables the caller to hang up and not be required to remain in queue listening to music. Callers who have remained in queue or have undergone the callback treatment will appear the same to agents answering the call.
Scheduling a callback to occur at a specified time is not part of this feature.

The following illustration shows the components needed for the Courtesy Callback feature.

**Figure 2: Courtesy Callback Components**

The Courtesy Callback applications on the VXML Server must not be invoked more than once for the same call.

**Typical Use Scenario**

If the caller decides to be called back by the system, they leave their name and phone number. Their request remains in the system and the EWT executes when the system places a callback to the caller. The caller answers the call and confirms that they are the original caller, and the system connects the caller to the agent after a short wait.
Typical Use Scenario

_Courtesy Callback is supported for IP originated calls as well._

A typical use of the Courtesy Callback feature follows this pattern:

1. The caller arrives at Unified CVP and the call is treated in the normal IVR environment.
2. The Call Studio and Unified ICM Courtesy Callback scripts determine if the caller is eligible for a callback based on the rules of your organization (such as in the prior list of conditions).
3. If a courtesy callback can be offered, the system tells the caller the approximate wait time and offers to call the customer back when an agent is available.
4. If the caller chooses not to use the callback feature, queuing continues as normal.
   Otherwise, the call continues as indicated in the remaining steps.
5. If the caller chooses to receive a callback, the system prompts the caller to record their name and to key in their phone number.
6. The system writes a database record to log the callback information.

_In Note:_ If the database is not accessible, then the caller is not offered a callback and they are placed in queue.

7. The caller is disconnected from the TDM side of the call. However, the IP side of the call in Unified CVP and Unified ICM is still active. This keeps the call in the same queue position. No queue music is played, so Voice XML Gateway resources used during this time are less than if the caller had actually been in queue.
8. When an agent in the service or skill category the caller is waiting for is close to being available (as determined by your callback scripts), then the system calls the person back. The recorded name is announced when the callback is made to ensure that correct person accepts the call.
9. The system asks the caller, through an IVR session, to confirm that they are the person who was waiting for the call and that they are ready for the callback.
   If the system cannot reach the callback number provided by the caller (for example, the line is busy, RNA, network problems, etc.) or if the caller does not confirm they are the caller, then the call is not sent to an agent. The agent is always guaranteed that someone is there waiting when they take the call. The system assumes that the caller is already on the line by the time the agent gets the call.
   This feature is called preemptive callback as the system assumes that the caller is already on the line by the time the agent gets the call and that the caller has to wait minimal time in queue before speaking to an agent.
10. The system presents the call context on the agent screen-pop, as normal.

If the caller cannot be reached after a configurable maximum number and frequency of retries, the callback is aborted and the database status is updated appropriately. You can run reports to determine if any manual callbacks are necessary based on your business rules.

Determine Callback Time

The following information provides an overview of how callback time is determined, the determination process and the calculation methods employed.

These are some definitions of key terms used:

- **Wait Time**—The interval of time between when the call enters the queue and when the call leaves the queue.
- **Reconnect Time**—The interval between the point at which the callback is started and the point at which the caller has accepted the callback and is waiting for an agent.
- **Callback in Queue Time**—The interval between when the caller is reconnected, waiting for an agent and when the call leaves the queue.
- **Service Level Agreement (SLA)**—Average of Callback in Queue Time. Average means that roughly 50 percent of calls are within the service level and 50 percent are outside the service level.
- **Average Dequeue Time**—The average number of seconds it takes for a call to leave the queue.
- **Remaining Time**—The number of seconds left to count down to call back the caller.

Callback in Queue Time

The average Callback in Queue Time after a callback is designed to be within an agreed service level. However, Courtesy Callback is also designed so that callers are not called back too early or too late, as both scenarios are undesirable. On the one hand, if callers are called back too early, then they are more likely to have to wait in the queue for a longer period of time, while, if the callback is made too late, there is a greater chance that call center agents could be idle and waiting for calls.

When the dynamics of a call center change, such as when more or fewer agents are available, or when the average handle time changes, it in turn causes the remaining time to change. Therefore, with Courtesy Callback, the Average Dequeue Time is calculated based on various factors such as calls in queue, average handle time, and agents in ready and talking states.

The Average Dequeue Time is updated when a call enters the queue and when it leaves the queue. This information is used for calculations for reducing the Callback in Queue Time and minimizing instances of call center agents waiting for calls.

Process Details and Calculation Methods

The following information details the process used to determine the callback time for calls in the queue. It also shows the method, or formula, used to calculate the Average Dequeue Time as well as the method used to update the remaining time for all Courtesy Callback calls in the queue.

The process for determining callback time is as follows:

1. The Average Dequeue Time (D) is calculated using the formula, $D = (\text{EWT} + F) / N$
   - EWT is the estimated wait time for a new Courtesy Callback call.
   - F is the number of seconds that the first call is already in position in the queue.
The Dequeue Time plays a significant role in the optimal behavior of the Courtesy Callback feature. The average Dequeue Time is calculated based on factors such as call volume, agent availability, and the average handle time for a particular skill group.

The Estimated Wait Time (EWT) is an approximation, and its accuracy is driven by the uniform average handling time and agent availability for a particular skill group. If these factors are not uniform, it may lead to a difference in the estimated wait time announced to the customer and the actual callback time. In addition, the use of microapps may insert calls into the queue that were not included in the EWT calculation. For scripting of calls that includes Courtesy Callback, all calls must be queued on the IVR using VxmlScripting, instead of microapps.

The remaining time for all Courtesy Callback calls in the queue is updated using the formula: 
\[ R(p) = p * D - F - C \]

- \( p = 1, ..., N \)
- \( R(p) \) is the remaining time for the \( p \)th queue position Courtesy Callback call.
- \( C \), the post-callback time, is the sum of the time it takes to retrieve the Courtesy Callback caller back on the phone and the SLA time.

Courtesy Callback can support a default wait time of 30 minutes with an exception of maximum 90 minutes.

Example Scripts and Audio Files

The courtesy callback features is implemented using Unified ICM scripts. Modifiable example scripts are provided on the Unified CVP install media in the \( \text{'CVP}\text{\textbackslash Downloads and Samples'} \) folder. These scripts determine whether or not to offer the caller a callback, depending on the criteria previously described. The files provided are:

-CourtesyCallback.ICMS, the ICM script
-CourtesyCallbackStudioScripts.zip, a collection of Call Studio scripts

Sample audio files that accompany the sample studio scripts are installed to the
\(<\text{CVP\_HOME\textbackslash OPSConsoleServer\textbackslash CCDownloads\textbackslash CCBAudioFiles.\text{zip}}\text{\ and also as part of the Media Files installation option.}\)

If CCBAudioFiles.zip is used, the contents must be unzipped onto the media server. CCBAudioFiles.zip has Courtesy Callback-specific application media files under en-us\textbackslash app and media files for Say It Smart under en-us\textbackslash sys. If you already have media files for Say It Smart on your media server, then only the media files under en-us\textbackslash app are needed.
The default prompts work for most of the default Call Studio scripts. However, the application designer must review and provision the Say It Smart plugin prompts for specific cases that are not covered within the default set of prompts.

The sample scripts are configured to use the default location of http://<server>:<port>/en-us/app. The default location of the sample audio files must be changed in the sample scripts to accommodate your needs (that is, substitute the media server IP address and port in <server> and <port>).

The following example scripts are provided:

- **BillingQueue**
  This script is responsible for playing queue music to callers that either choose to not have a callback occur or must reenter the queue for a short period after receiving a callback.
  You may customize this script to suit your business needs.
- **CallbackEngine**
  This script keeps the VoIP leg of a callback alive between when a caller elects to have a callback and when a caller receives the callback.
  Do **not** customize this script.
- **Callback Entry**
  This script handles the initial IVR when a caller enters the system and when the caller is provided with the opportunity to receive a callback.
  You may customize this script to suit your business needs.
- **CallbackQueue**
  This script handles the keepalive function of a call while a caller is in queue and listening to the music played within the BillingQueue script.
  Do **not** customize this script.
- **CallbackWait**
  This script handles the IVR portion of a call when a customer is called back.
  You may customize this script to suit your business needs.

**Callback Criteria**

Examples of callback criteria you can establish include:

- Number of minutes a customer is expected to be waiting in queue exceeds some maximum number of minutes (based on your average call handling time per customer)

- **Assigned status of a customer** (gold customers may be offered the opportunity to be called back instead of remaining on the line)
• The service a customer has requested (sales calls for example, or system upgrades may be established as callback criteria)

**Courtesy Callback Design Considerations**

The following design considerations apply to the Courtesy Callback (CCB) feature:

• During Courtesy Callback, callback is made using the same Ingress Gateway through which the call arrived.

**Note**

In Courtesy Callback, outbound calls cannot be made using any other Egress Gateway.

• Calls that allow Callback must be queued using a Unified CVP VXML Server.

• The Unified CVP Reporting Server is a prerequisite for Courtesy Callback.

• Answering machine detection is not available for this feature. During the callback, the caller is prompted with a brief IVR session message and acknowledge with DTMF that they are ready to take the call.

• Calls that are transferred to agents using DTMF *8, TBCT, or hookflash cannot use the Courtesy Callback feature.

• Courtesy Callback feature does not support Agent call transfers to CCB Queue, over a computer telephony integration (CTI) route point.

• Callbacks are a best-effort function. After a limited number of attempts to reach a caller during a callback, the callback is terminated and marked as failed.

• Customers must configure the allowed or blocked numbers that Callback is allowed to place calls through the Unified CVP Operations Console.

• Media inactivity detection feature on the VXML Gateway can affect waiting callback calls. For more information, see the *Configuration Guide for Cisco Unified Customer Voice Portal*.

• Courtesy Callback requires an accurate EWT calculation for its optimal behavior.

Consider the following recommendations to optimize the EWT, when using Precision Queues for Courtesy Callback:

* Queue the calls to a single Precision Queue

* Do not include a Consider If expression when you configure a step.

* Do not include a wait time between steps or use only one step in the Precision Queue.

**Note**

Make sure that you use simple Precision Queue definitions (for example, with one step and one-to-one agent mapping). The complexity of Precision Queues makes calculating accurate EWT difficult.
Post Call Survey

A contact center typically uses a post call survey to determine whether a customer was satisfied with their call center experience. For example, did the customer find the answer they were looking for using the self service or did they have a pleasant experience with the contact center agent.

The Post Call Survey (PCS) feature enables you to configure a call flow so that after the agent hangs up, the caller is transferred to a DNIS that prompts the caller with a post call survey.

There are two responses a caller can have to a post call survey request:

1. The caller is prompted during IVR treatment as to whether they would like to participate in a post call survey. If they choose to do so, they are automatically transferred to the survey call after the agent ends the conversation.

2. The caller is prompted to participate, but declines the post call survey. A Unified ICM script writer can use an ECC variable to turn off the ability for Post Call Survey on a per-call basis. By setting the ECC variable to n, the call will not be transferred to the PCS DNIS.

For reporting purposes, the post call survey call has the same Call-ID and call context as the original inbound call.

Typical Uses

The caller is typically asked whether they would like to participate in a survey during the call. In some cases, the system configuration based on dialed numbers determines if the post call survey gets invoked at the end of conversation with agents. When the customer completes the conversation with an agent, the customer is automatically redirected to a survey. The post call survey gets triggered by the hang-up event from the last agent.

A customer can use the keypad on a touch tone phone and voice with ASR/TTS to respond to questions asked during the survey. From the Unified CCE point of view, the post call survey call is just like another regular call. During the post call survey, the call context information is retrieved from the original customer call.

Design Considerations

Observe the following conditions when designing the Post Call Survey feature:

- A Post Call Survey is triggered by the hang-up event from the last agent. When the agent hangs up, the call routing script launches a survey script.

- The mapping of a dialed number pattern to a Post Call Survey number enables the Post Call Survey feature for the call.

- The value of the expanded call variable user.microapp.isPostCallSurvey controls whether the call is transferred to the Post Call Survey number.

  - If user.microapp.isPostCallSurvey is set to y (the implied default), the call is transferred to the mapped post call survey number.

  - If user.microapp.isPostCallSurvey is set to n, the call ends.
• To route all calls in the dialed number pattern to the survey, your script does not have to set the user.microapp.isPostCallSurvey variable. The variable is set to y by default.

• REFER call flows are not supported with Post Call Survey. The two features conflict: REFER call flows remove Unified CVP from the call and Post Call Survey needs Unified CVP because the agent has already disconnected.

• For Unified CCE reporting purposes, when a survey is initiated, the call context of the customer call that was just transferred to the agent is replicated into the call context of the Post Call Survey call.

Call Admission Control

Call admission control is the function for determining if there is enough bandwidth available on the network to carry an RTP stream. Unified CM can use its own locations function or RSVP to track bandwidth between the Ingress Gateway and destination IP phone locations.

For more information about call admission control, see the chapter on Distributed Deployment.

In networks, Resource Reservation Protocol (RSVP) is a protocol 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 via the Unified CVP Call Server in SIP. As an alternative, the recommended solution for Call Admission Control is to employ 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:

Queue-at-the-Edge Branch Office Deployment Model

The following figure illustrates a typical branch office deployment.

Figure 3: Typical Branch Office Deployment.

You can deploy Unified CVP in a single cluster Unified CM deployment to provide queue-at-the-edge functionality. In this deployment model, branch-located Ingress Gateways are typically used to allow callers access using local phone numbers rather than centralized or non-geographic numbers. This consideration is especially 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 (ACDs) into the CVP switching solution. Apart from the gateways all other CVP components are centrally located and WAN links provide data connectivity from each branch location to the central data center. (Although the media server is centrally located, commonly used VRU media is cached at the local branch.)

In the Unified CVP branch office deployment model using queue-at-the-edge, the only equipment at the branch office is an Ingress Gateway (optionally acting as a VoiceXML Gateway as well), IP phones for Unified CCE agents, IPT (user) phones, and agent desktops.

You can configure Unified CCE Skill Groups, dial plans and routing priorities so that callers who Ingress at one branch are connected by preference to agents who are located at the same branch. In these cases, the RTP traffic flows directly from Ingress Gateway to IP phone, and does not need to traverse the WAN (although signaling and data may traverse the WAN).
The goal of this model is to first route the calls locally to an agent available in the branch office, if possible, and keep the media streams local. If the local agent is not available, only the call gets routed to the agent on another branch office over the WAN link; the originating call and the initial VRU treatment are done locally.

Another advantage of this deployment configuration is that in the event of WAN link failure, the call can still be routed locally using the CVP survivability application running on the pots dial-peer for TDM originated calls.

Enhanced Location Call Admission Control

ELCAC Concepts

The following definitions are important to the ELCAC feature:

- **Phantom Location**—A default location with unlimited bandwidth used when calculating calls that are hairpinned over a SIP trunk or when the SIP call is queued at the local branch, to enable correct bandwidth calculations. The Phantom location should be assigned to the gateway or trunk for CVP.

- **siteID**—The siteID is a string of numbers that Unified CVP appends to the label it receives from Unified ICM. Depending on the siteID the dial plan can be configured to route the call to a specific destination, such as the branch VXML Gateway or Egress Gateway, or UCM node. The siteID can be appended at the front of the label, at the end, or not at all. This configuration is separate from the Unified CM location configuration, and is specific to Unified CVP. The siteID is used to indicate the real location of the call and allow the bandwidth to be deducted from the correct location. siteID is unique across multiple Unified CM clusters. Multiple siteIDs can still route to the same branch office (if needed) by mapping the unique siteIDs to same branch gateways in proxy routes.

- **Shadow Location**—This new location is used for intercluster trunks between two Cisco Unified Communications Manager clusters. This location is not used as intercluster ELCAC is not supported in Unified CVP.

Locations are created in UCM. Unified CVP gets these locations when you synchronize the location information from the UCM on operations console. You can associate a siteID for these locations on operations console and then associate your gateways to these locations. Based on this configuration, CVP creates two hash objects. One hash would map location to a siteID and the second hash would store mapping of GW IP address to location name and siteID. These hash objects are used to route the call to appropriate GW to provide edge queuing (using siteID), and pass around the location information on the call legs for UCM to do proper CAC calculations.

For a Unified CVP branch office deployments the following considerations apply:

- Control the number of calls that goes over the WAN link to branch offices based on the available bandwidth of the WAN link.

- For the queue-at-the-edge functionality, the call originating from a specific branch office should be routed to a local VXML Gateway on priority.

When you are using the Unified CVP intracluster Enhanced Location CAC model deployment, you must control the number of calls that go over the WAN link to branch offices. The decision to admit calls is based on the CAC computations, which represent the bandwidth used by the call. These computations are valid whether the calls are IP calls between two phones within Cisco Unified Communications Manager, calls over SIP trunks, or calls originated from TDM-IP Gateway.
For queue-at-the-edge functionality, the call originating from a specific branch office must be routed to a local Unified CVP VXML Gateway based on priority. That is, always choose a local branch agent if possible.

Unified CVP supports topology modeling with Enhanced Location Call Admission Control (ELCAC) for intracluster. It does not support intercluster Enhanced Location CAC. Location Bandwidth Manager is enabled for intracluster CAC, but disabled for intercluster CAC. For more information on ELCAC topology modeling, see 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.

Comparison of Enhanced Location Call Admission Control Feature

The Enhanced Location Call Admission Control (ELCAC) feature addresses two important issues with the prior CAC feature:

1. Bandwidth miscalculations in CAC with IP originated callers, as well as with any post transfers from agents.
2. Inability to deterministically select a local VXML GW for VRU treatment at the branch office during warm transfers from an agent due to no correlation between the two calls at consult.

Comparison with OrigIP Trunk Feature on Unified CM

Before Unified CM implemented the phantom trunk and siteID feature for bandwidth calculation, there was the existing feature used by Unified CVP that enabled the correct trunk to be selected depending on the original ID of the caller. This feature enabled Unified CM to select the correct trunk for the TDM gateway, instead of only using the single Unified CVP trunk, and it only applies to incoming calls on the trunk. With this feature, distinct SIP profiles and trunk settings could be used for each branch gateway without being limited to the settings of the single Unified CVP trunk. This feature has no impact on bandwidth calculations.

Router Requery with ELCAC

When a call is rejected by the UCM due to not enough bandwidth, a SIP message 488 Not Acceptable Here is returned to Unified CVP, where it triggers a router requery over the GED-125 interface to the VRU peripheral, and the UCCE Router may return another agent label if requery is configured properly.

Design Considerations

The following considerations apply when using ELCAC:

- The SIP trunk configured between Unified CVP and Unified CM should be associated with Phantom location. A new location called shadow location is added in Unified CM 9.0 for inter-cluster ELCAC, but it is not supported in Unified CVP.
- In multi-cluster CUCM deployments, consider over subscribing bandwidth on WAN links based on the anticipated peak call volume or choose a centralized branch office deployment model, as intercluster ELCAC is not supported on Unified CVP.
- In single-cluster CUCM deployments, ELCAC is supported only for Hub and Spoke topology with Unified CVP.
• A trunk configured with MTP required will not work with the ELCAC siteID feature. The reason is when MTP is inserted, the media is terminated between the end point and MTP resource, not between the two end points.

• If a MTP/Transcoder/TRP media resource is inserted by the Unified CM media layer, the incoming location information is not used.

• If the intercluster call is not looped back to the same cluster, the former behavior of Location CAC logic will apply.

• Each site is uniquely identified by one siteID. Multiple gateways at the same site would need to align to the same siteID, but if two clusters happen to use the same location name, then two siteIDs can map to the same physical branch.

• A second Unified CM cluster may have the same location as the first cluster, but be required to use a unique siteID on Unified CVP. You can define a route in the proxy server to send those cluster calls to the common VXML Gateway at the same location, but used by both the clusters.

• Each cluster would manage the bandwidth for devices in its cluster. If two clusters happen to use the same physical location, then they would each separately manage the bandwidth for the phones that they manage.

High Availability and Failover

The following considerations apply when using LBCAC:

• During the CAC failure, Unified CVP returns a failure code to Unified CCE that triggers router requery.

• If a branch does not have a VXML Gateway, then use the VoiceXML Gateway at the Central data center.

Additional ELCAC Information

The previous version of Unified CVP provided a method of configuring CAC. This method is superseded by the ELCAC method presented here. Both configuration methods are provided in the Configuration Guide for Cisco Unified Customer Voice Portal, available at:


Network-Based Recording

The network-based recording (NBR) feature supports software-based forking for Real-time Transport Protocol (RTP) streams. Media forking provides the ability to create midcall multiple streams (or branches) of audio and video associated with a single call and then send the streams of data to different destinations. To enable network-based recording using CUBE, refer to the configuration guide. You can configure specific commands or use a call agent. CUBE acts as a recording client and MediaSense recorder acts a recording server.

Note
Network-based recording works with the call survivability feature.
The following figure displays the topology and call flow for network-based recording.

A typical call flow for network-based recording is as follows:

1. A customer calls using an IP phone or by using PSTN.
2. The Ingress Gateway sends the call to Unified CVP.
3. Unified CVP sends the incoming call request to ICM and gets a VRU label.
4. Unified CVP sends the call to the VoiceXML Gateway. The caller hears the IVR. However, the call is not recorded.
5. After the agent is available, Unified CVP connects the caller to the agent.
6. Network-based recording starts for this conversation.

**Limitations**

- For agent to agent call transfer, network-based recording does not work but phone-based recording does. If you want to use network-based recording, you can use an ISR gateway between Unified CVP and Cisco Unified CM.

- The NBR feature is currently supported on ISR G2 gateways, in an inbound contact center deployment only.

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**Note**

The NBR feature is supported only on selected IOS image trains. For more information about the supported IOS image trains, see the *Unified CCE Solution Compatibility Matrix*, available at [http://docwiki.cisco.com/wiki/Compatibility_Matrix_for_Unified_CCE](http://docwiki.cisco.com/wiki/Compatibility_Matrix_for_Unified_CCE).