



Release Notes for Cisco Unified Customer Voice Portal (Unified CVP) Release 8.0(1)

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Introduction

This document discusses new features, changes, and caveats for Release 8.0(1) of Cisco Unified Customer Voice Portal (Unified CVP) software.

Additional information on new features, and on many of the product changes, are available in the relevant end-user documentation.



Note

For the most up-to-date version of all Cisco documentation, go to the Cisco Web page:
<http://www.cisco.com/web/psa/products/index.html>



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System Requirements

For hardware and third-party software specifications for Release 8.0(1), refer to the *Hardware and System Software Specification (Bill of Materials) for Cisco Unified Customer Voice Portal*, which is accessible from

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

See the *Installation and Upgrade Guide for Cisco Unified Customer Voice Portal, Release 8.0(1)* for additional information. This document is accessible from

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

Related Documentation

Documentation for Cisco Unified Customer Voice Portal, as well as most related documentation is accessible from

<http://www.cisco.com/go/cvp>

The latest Release Notes for Cisco Unified Customer Voice Portal are accessible from

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

Troubleshooting content for Unified CVP 7.x and later is available only on DocWiki:

http://docwiki.cisco.com/wiki/Troubleshooting_Unified_Customer_Voice_Portal

New and Changed Information

Integrated 3G-H324M Gateway Support

The Cisco Integrated 3G-H324M Gateway - or Video Gateway - allows multimedia communications (H.324M) between 3G (third generation) mobile handsets and Cisco AS5xxx Universal Gateways.

Refer to the information on Cisco.com for more in-depth information about the Cisco Integrated 3G-H324M Gateway: <http://www.cisco.com/en/US/docs/video/multicomm/3g324m.html>.

For information on configuring Unified CVP for this feature, please refer to the Unified CVP Configuration and Administration (CAG) Guide

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

VMWare/UCS Support

Unified CVP Release 8.0(1) is now supported on the VMWare platform for specific deployments. See <http://www.cisco.com/go/uc-virtualized> for details on VMWare and UCS support.

Mixed G.729 and G.711 Codec Support

Unified CVP Release 8.0(1) now supports the use of a mixture of G.711 and G.729 Coder-Decoders (CODECs) on different legs of the same call, as long as you have allocated and configured sufficient resources in the solution for transcoding. Your network must be properly sized for VoIP traffic and Codec bandwidth requirements.

CVP supports mixed G.711 and G.729 codecs in Standalone and Comprehensive SIP deployments with Cisco Unified Border Element Enterprise Edition for Cisco (CUBE) and Cisco Unified Communications Manager (Unified CM). Calls that are ingressed through a SIP trunk from the carrier to a CUBE require IOS 15.1(2)T or later T for mixed codec support. You can use any combination of codecs on the legs of a call. For example, a caller can place a call using the G.729 codec, hear an IVR prompt played using the G.711 codec, be transferred to the first Agent using the G.729 codec, and then transferred to the second agent using the G.711 codec.

Unified CVP passes media information in SIP messages to Media Termination Points. Media Termination Points extend supplementary services, such as call hold, call transfer, call park, and conferencing, that are otherwise not available when a call is routed to an endpoint. When using mixed codecs with Unified CVP, you must use a transcoder to translate the codec used on one call leg to the codec used on the next call leg. Unified CVP does not include a transcoder. You must include either Unified CM, CUBE, or a DSP farm in a Unified CVP deployment that uses mixed codecs.

More details on using mixed codecs with Unified CVP are available in the *Solution Network Design Guide for Cisco Unified Voice Portal*.

G.729 Codec Support

G.729 is now supported in Unified CVP Release 8.0(1) in single-codec (G.729 only) and mixed codec (G.711 and G.729) deployments. Note that recorded prompts, Music on Hold, or any other audio media files must be specially encoded for Cisco G.729 to work with Unified Communications Manager.

Prompts and Music-on-Hold media files served from a media server and through a gateway must be modified to be heard by the caller. See [Modifying G.729-encoded Prompts and Music-On-Hold Media for Gateway Compatibility](#), page 15.

Cisco PGW2200 Softswitch Support

Unified CVP Release 8.0(1) now supports the Cisco PGW2200 Softswitch in Comprehensive model.

The Cisco PGW 2200 is a carrier-class softswitch that performs the signaling and call-control tasks (such as digit analysis, routing, and more) within the Public Switched Telephone Network (PSTN) gateway infrastructure. It is typically added to a Unified Contact Center Solution when the customer would like to bring SS7 calls into the contact center. The PGW uses Signaling System 7 (SS7) and Session Initiation Protocol (SIP) to manage call signaling. The PGW utilizes Media Gateway Control Protocol (MGCP) to communicate with an IOS Media Gateway to manage media transmission.

The Cisco PGW 2200 consists of up to five different elements:

- Cisco Media Gateway Controller (MGC) software, running on Sun Microsystems general computing platforms - required
- Cisco IP Transfer Point - LinkExtenders (ITP-Ls) - required
- LAN switch for IP interconnectivity of Cisco PGW 2200 elements - required
- H.323 Signaling Interface (HSI) adjunct processor- not allowed with contact center

Call agent (or media gateway controller) and softswitch are industry-standard terms used to describe the network element that provides call control functions to telephony and packet networks. The Cisco PGW 2200 in call control mode functions as a call agent or softswitch. A PSTN gateway provides the interface between traditional Signaling System 7 (SS7) networks or non-SS7 networks and Media Gateway Control Protocol (MGCP)-, H.323-, and Session Initiation Protocol (SIP)-based networks, including signaling, call control, and time-division multiplexing (TDM)/IP gateway functions. The Cisco PGW 2200, coupled with Cisco media gateways, functions as a PSTN gateway.

The following PSTN gateway applications are enabled by the Cisco PGW 2200:

- Voice-over-IP (VoIP) transit
- Primary Rate Interface (PRI) Grooming and Time-Division Multiplexing (TDM) Offload
- SIP PSTN gateway
- H.323 PSTN gateway
- Enterprise Services (Hosted Voice and Call Center)

Courtesy Callback

Courtesy Callback is a new feature in Release 8.0(1). Courtesy Callback reduces the time callers have to physically 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.

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) Note: The included sample scripts use this method for determining callback eligibility.
- 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, or system upgrades, etc. may be established as callback criteria)

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 when the system determines that an agent will be available soon (or is available), then the system places a call back 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 much shorter wait.

More details on the Courtesy Callback feature are available in the Configuration and Administration Guide.

Locations-Based Call Admission Control

Locations-Based Call Admission Control (LBCAC) has been enhanced and uses a simplified configuration in Unified CVP Release 8.0(1). In Release 8.0(1) LBCAC bandwidth calculations are more accurate and the feature can now select a local VXML Gateway for VRU treatment at the branch office during warm transfers from an agent.

More details on LBCAC are available in the *Cisco Unified Customer Voice Portal Release 8.x Solution Reference Network Design (SRND)* and the Unified CVP Configuration and Administration Guide.

Operations Console Improvements

Unified CVP Release 8.0(1) introduces many improvements to the Operations Console:

- System Level Configuration - Unified CVP Operations Console Server now provides administration of various system-level configurations. A system-level configuration allows an administrator to manage configurations that can be deployed to many Unified CVP solution devices in a single operation.
- Network Map - The Unified CVP Operations Console Server now generates and distributes a network map to all configured Unified CVP devices as well as other, third-party devices. The network map contains information which allows serviceability tools (such as the Unified CVP Web Services Manager and System CLI tool) to access individual devices from the network map via a defined Serviceability application programming interface (API).
- Bulk Upload of multiple files to multiple endpoints - Operations Console Server administrators can now transfer multiple files to multiple configured devices in a single operation. The bulk administration file transfer features are also system-level configurations and share many of the core feature types.
- IOS template management and deployment - Operations Console administrators can now manage, edit, and deploy IOS configurations (through templates) to supported gateways. This simplifies initial configuration of Unified CVP.
- User Management - Operations Console Server administrators can now configure Unified CVP Web Services users via User Management menu. Web Services users (Serviceability Administration) (see [System CLI/Web Services Manager, page 6](#)) are pushed out via the Web Services system-level configuration to Unified CVP devices. These credentials are used for authentication when using the Unified CVP Web Services Manager or System CLI.
- Integrated Google Custom Search of User Documentation - The Operations Console now has a link from the main page to search Cisco provided documentation via keyword. This feature uses a Google-provided web services interface in order to manage search operations, and only searches Unified CVP documentation and Wiki content posted on Cisco.com so all results are specific to Unified CVP.

Additional improvements and more details on the Operations Console improvements are available in the Operations Console User Guide and the Configuration and Administration Guide.

SIP Enhancements

Multiple SIP enhancements have been introduced in Release 8.0(1) to bring SIP feature support to parity with H.323 feature support:

- SIP Headers - Unified CVP now enables the passing of one or more SIP headers to Unified ICM for use within ICM scripts. You configure which headers are passed to Unified CVP through the Operations Console, on the Call Server SIP tab. You can also modify SIP headers in the Unified ICM script and pass these back to Unified CVP.
- SIP Hookflash - Unified CVP now supports using hookflash with SIP. Using this feature you can transfer SIP calls using a hookflash followed by the DTMF destination. This is useful for deployments in which a PBX “front-ends” the Unified CVP [media](#) gateway, and in which the PBX provides non-VOIP connectivity to agents.

- Enhanced UI - You can now pass generic data to and from Unified ICM through Unified CVP using User-to-User Information (UUI). For example, it is sometimes desirable to capture data from an external system (such as caller-entered digits from a third-party IVR) and pass that data to Unified ICM on a new call.
- KPML Support - Unified CVP now supports Key Press Markup Language (KPML) for SIP. KPML is a feature that enables monitoring of DTMF signals “out-of-band” (OOB), as opposed to the (RFC2833) in-band DTMF monitoring method that Unified CVP typically uses.
- Server Groups/Dynamic Routing - Unified CVP now supports dynamic routing with SIP. Dynamic Routing allows the originating endpoint to have knowledge of the status of the destination address before attempting to send the SIP INVITE. When the destination is unreachable over the network, or is out of service at the application layer, then the originating SIP user agent gains knowledge of this status through a new heartbeating mechanism. Unified CVP 8.0(1) adds a SIP heartbeating mechanism with endpoints.
- Trunk Group Utilization - This new feature allows gateways to push memory, trunk, and CPU status to Unified CVP. This information can be used for real time routing and Unified ICM reporting and scripting.
- Post Call Survey - Unified CVP now supports Post Call Survey for SIP calls. A Post Call Survey is a survey that takes place after normal call treatment and is typically used to determine whether a customer was satisfied with their call center experience. This feature enables you to configure a call flow that, after the agent disconnects from the caller, the call is optionally sent to a DNIS configured for a Post Call Survey.

Details on all the enhancements are available in the Operations Console User Guide and the Configuration and Administration Guide.

System CLI/Web Services Manager

Unified CVP Supports a new serviceability Command Line Interface (CLI) called Unified System CLI. Unified System CLI enables you to collect diagnostic (health and status) information on Unified CVP servers and to collect device-specific information from each supported node connected to the Unified CVP server from which you are using the Unified System CLI.

Unified System CLI accesses a new web services layer in Unified CVP. This layer is referred to as the Web Services Manager (WSM). It provides REST-like API's and supports secure authentication and data encryption on each of the interfaces. WSM interacts with various subsystems and infrastructure handlers, consolidates the responses, and publishes an xml result. The Web Services Manager supports http requests and sends a predefined XML response. Unified System CLI uses WSM to collect and present the data available to WSM from the various Unified CVP components.

More details on System CLI and Web Services Manager are available in the Configuration and Administration Guide and the Operations Console User Guide.

Call Studio Enhancements

Unified Call Studio adds the following features in Release 8.0(1):

User Interface Updates:

- Additional elements have been added to the Elements View. These elements are used with Courtesy Callback.
- New Global Search Menu Tab - Search for a specific element in an entire application.

Additional Enhancements:

- A search/replace feature has been added in 8.0(1) that automatically updates all substitution tags, decision editor and decision XML referencing the renamed element.
- A checkbox has been added to the 'Call Studio License Warning' dialog that states “Don't remind me again unless the license is re-activated”. If the user checks this checkbox, the 'Call Studio License Warning' dialog does not appear again.
- Licensing Changes — The licensing mechanism for Call Studio now uses a new FlexLM-based license.

Additional details are available in the VXML and Call Studio documentation set.

Unified CVP VXML Server Enhancements

The following features are now available for Unified CVP VXML Server:

- Load-balancer interaction has been optimized with two new features:
 - Enhanced HTTP probes for load balancers: this feature enables you to control the HTTP response code sent when licensed ports are exceeded. Instead of returning a 200 OK message and putting call requests into “on hold” states, Unified CVP can return 500 Server Errors so that load balancers stop sending calls to the Unified CVP server in question.
 - Requests to a load balancer can now be reduced to only the first HTTP request of a session. This minimizes the impact of load balancer failures on VXML Server sessions.
- A new debug mode is now available to help track down “bad_url” errors.
- VXML Server now generates trace logs in the same format as those generated by other Unified CVP components (such as the Call Server and Reporting Server). These trace logs are in addition to the activity/error logs produced by the application and global loggers.
- The Unified CVP Call Server now automatically pass the call identifier (call GUID), ANI and DNIS of a call to the Unified CVP VXML Server
- Licensing Updates:
 - By default, Unified CVP VXML Server supports 30 ports for 30 days for evaluation.
 - Licensing has changed to use the same licensing mechanism as Unified CVP Call Servers.
 - For Upgrades, the Operations Console contains a tool (**System > License Conversion**) to convert the old license into a file that allows you to obtain a new-style license from cisco.com.
 - The license location has changed. The license is now stored in `%CVP_HOME%\conf\license\cvp.license`
 - Licenses are now additive. You do not need to remove an existing license than install a new license to add ports. To add additional ports you apply an additional license file that increases the number of ports and updates the saved license to indicate the total number of ports.

Reporting Enhancements

Many enhancements have been incorporated into Unified CVP reporting:

- Up to doubling of available space via compression of backups
- Integration with CUIS
- Increased Data Retention Times

- CLI/WSM integration

Additional enhancements and details are available in the Reporting Guide for Unified CVP.

Licensing Enhancements

Licensing has been streamlined and updated. Additional details are available in the Installation and Upgrade Guide.

AIX Support

AIX is not supported in Release 8.0(1). When upgrading to Release 8.0(1) from a previous Unified CVP release, AIX cannot be upgraded during this process.

Full Video Support

Full Video is no longer a supported feature in Unified CVP. Basic Video is still supported.

Documentation Changes

Planning your Unified CVP solution is an important part of the process in setting up Unified CVP, so make sure you first read the *Cisco Unified Customer Voice Portal Release 8.x Solution Reference Network Design (SRND)* guide *before* configuring your Unified CVP solution. With Unified CVP 8.x, the Planning Guide for Cisco Unified Customer Voice Portal has been incorporated into the SRND document.

Installation Notes

- [Important: Upgrades and Customized Log Messages, page 8](#)
- [Unified CVP Call Server Licensing, page 9](#)
- [Unified CVP Reporting Server Licensing, page 9](#)
- [Unified Call Studio Licensing, page 9](#)
- [Unified VXML Server Licensing, page 10](#)
- [Installation on WebSphere, page 10](#)

Important: Upgrades and Customized Log Messages

You can customize Unified CVP log definition files to control the log message contents and properties. If you have customized CVPLogMessages.xml and/or CVPLogMessageRes.properties and you are upgrading to Unified CVP 8.0(1), then you must consider the following information.

During the upgrade to Unified CVP Release 8.0(1), the existing message definition files CVPLogMessages.xml and CVPLogMessagesRes.properties are renamed to CVPLogMessages.xml.bak and CVPLogMessagesRes.properties.bak respectively. Unified CVP 8.0(1) versions of these files are installed (CVPLogMessageRes.properties is now named LogMessageRes.properties). If you customized

these files previously, then you must reapply customizations of these files manually, based on the backed-up files. The definitions of the following message IDs has changed in release 8.0(1): 1980, 7000, 9011, 9020, 9021, 9022, 11100, 11102, 11103, 11104, 11105, 11106, 11107. When reapplying customizations, you must consider the changed definitions for these IDs.

Unified CVP Call Server Licensing

A Unified CVP Call Server license is required for all Unified CVP Call Server machines on which Call Server services are required.

As of Release 8.0(1), the Unified CVP Call Server provisions a 30-port license for all Unified CVP Call Server installations that have no license file. The evaluation license expires in 30 days.

You must obtain a new software license for upgrades.



Note

Media files are not retained on Unified CVP Call Server or VXML Server upgrade if in the past the server has been used as a media server.

The *Solution Network Reference Design Guide* provides detailed information about how Unified CVP components and ports are licensed.

Unified CVP Reporting Server Licensing

As of Release 8.0(1), if the Unified CVP Reporting Server installation has no license file, the Reporting Server operates with limited functionality by allowing only 10,000 database writes per day for 30 days in the following manner:

- Start counting the number of database writes on startup.
- Every time Unified CVP writes messages to the database, the Reporting Server checks the local date.
- When Unified CVP passes 10,000 database writes (the sum of both inserts and updates) for a day, it sends an alarm and no longer writes messages to the database.
- When the local date changes, Unified CVP resumes writing to the database and tracking the number of writes.

A Unified CVP Reporting Server license is required for a call center to operate with reporting. When the Unified CVP Reporting Server starts, it begins counting messages. Each time a message is written to the database, the Unified CVP Reporting Server checks the local date. When database writes reach 10,000 (the sum of both inserts and updates) in a single day, an alarm is sent and messages are no longer written to the database. When the local date changes, database writes resume, as does the count.

The evaluation license expires in 30 days.

You must obtain a new software license for upgrades.

Unified Call Studio Licensing

Unified Call Studio can be used for 30 days after installation without an active license. This may be useful for simple testing or evaluation purposes. After 30 days, an active license must be applied to continue using Call Studio.

Licensing has changed for Call Studio in release 8.0(1), you must obtain a new license to use Call Studio after 30 days. Complete details on licensing are available in the *Installation Guide for Unified CVP, Release 8.0(1)*.

Unified VXML Server Licensing

As of Release 8.0(1), Unified VXML Server supports 30 concurrent sessions for 30 days by default before a license is applied. To utilize more simultaneous sessions, Unified VXML Server must be activated with a valid license.

**Note**

Media files are not retained on Unified CVP Call Server or VXML Server upgrade if in the past the server has been used as a media server.

Installation on WebSphere

There are manual steps that must be performed to finalize Unified CVP installation on WebSphere. For example, the default port on WebSphere for web applications is 9080; it must be set to 7000 to work with Unified VXML Server and other Unified CVP components. See *Installation and Upgrade Guide for Cisco Unified Customer Voice Portal, Release 8.0(1)* for additional information.

After the installation completes, install the WAR file via WebSphere's standard web application deployment process. The WAR file is named CVP.war; the file resides in CVP_HOME/VXMLServer/war. After the CVP.war is deployed, modify the module class loader order for CVP.war using the WebSphere Administrative Console. To do this, in the WebSphere Administrative Console, navigate to **Applications > Enterprise Applications > CVP_war > Manage Modules > Cisco Unified CVP VXML Server** and from the **Class loader order** drop-down, select **Classes loaded with application class loader first**. Click **Apply**, and then **OK**. Finally, save the changes to the master configuration. Refer to WebSphere documentation for additional details.

Important Notes

The following section contains important notes and restrictions that applies to Release 8.0(1):

- [Integrated 3G-H324M Gateway Support, page 2](#) Integrated 3G-H324M Gateway may Require Adjustment to UDP Retransmission Count Field Value, page 11
- [New CLI \(Command Line Interface\) Required To Disable the New Toll Fraud Feature, page 11](#)
- [PGW Support - Configuration Interoperability Between Unified CVP and PGW, page 12](#)
- [PGW Does Not Support Local REFER, page 13](#)
- [PGW Does Not Support Locations-Based Call Admission Control \(LBCAC\), page 13](#)
- [OAMP Data Delete \(purge\) Configuration May Be Different Than Current Setting on the Reporting Server, page 13](#)
- [Courtesy Callback Does Not Recognize DTMF if a Second Party is Conferenced In, page 13](#)
- [Validation in Call Studio does not Validate for items missing in Record Element, page 14](#)
- [During Courtesy Callback, Survivability Script is not Heard if Call Server Connection to VXML Server is Terminated, page 14](#)
- [Call Server Indicates Device has Unapplied Changes - Unable to Deploy Changes to Call Server, page 14](#)
- [RTCP Interval/Timer Can Cause Survivability to Activate Too Soon, page 14](#)
- [Gateway Template Variable Replacement Fails for Multiple Devices, page 15](#)
- [CiscoUnifiedCVPInstall.log lists Installation on <Unsupported OS>, page 15](#)
- [Modifying G.729-encoded Prompts and Music-On-Hold Media for Gateway Compatibility, page 15](#)
- [Changing the IP Address of a Call Server, page 16](#)
- [Leading zero gets removed from incoming DNIS, page 16](#)

Integrated 3G-H324M Gateway may Require Adjustment to UDP Retransmission Count Field Value

If you are configuring a Cisco 3G-H324M Gateway, you may need to increase the default value of 2 for this configuration because 3G requires additional call setup time.

New CLI (Command Line Interface) Required To Disable the New Toll Fraud Feature

There is a new CLI in IOS required to disable the new Toll Fraud feature which was added in PI13:

```
voice service voip
no ip address trusted authenticate
```

The CVP IOS templates, as deployed from OAMP, are also updated with this CLI.



Note

If the CLI is not used, then all calls from Unified CVP to the VXML gateway may be rejected with a 'SIP code 400 Forbidden'.

There is also a way to use the feature with Unified CVP without disabling it. This can be seen in the IOS documentation at the following links:

<https://Supportforums.cisco.com/docs/DOC-12228>

<http://www.ciscosecure.net/application/pdf/paws/112083/tollfraud-ios.pdf>

PGW Support - Configuration Interoperability Between Unified CVP and PGW

The following section contains important notes related to configuring PGW to operate with Unified CVP:

Dual Tone Multifrequency (DTMF) - The following DTMF commands are required on the Media Gateway Control Protocol (MGCP) in order for DTMF to be processed correctly.

- `mgcp dtmf-relay voip codec all mode nte-gw`
- `mgcp rtp payload-type nte 101`

Music on Hold (MoH) - The following commands are required on the MGCP in order for MoH to work:

- `mgcp behavior mode-attrb-in-spd disable`
- `no voice-fastpath enable`
- `mgcp behavior mdcx-sdp ack-with-sdp`

Retry Timer for Gateway Payload - The following commands are required on the PGW to optimize the caller experience during a PGW failover situation:

- `numan-add:cause:custgrpId="xxxx", causevalue=221,setname="CSCOADRST1", where xxxx is the customer Group ID for your deployment. CSCOADRST1 is an example of a set name only.`
- `prov-ed:profile:name="sipi-profile",*.InviteAttempts="2"`
- `prov-ed:profile:name="sipo-profile",*.InviteAttempts="2"`

Sending GTD information to Unified CVP from the PGW is not supported - For a dialpeer on a gateway that handles calls between the PGW and the gateway, add the following to the dialpeer configuration on the gateway:

- `signaling forward none`

For example:

```
dialpeer voice 65001 voip
    signaling forward none
    incoming called-number 65T
    voice-class codec 1
```

PGW Does Not Support Sending GTD Information to Unified CVP

This feature is not supported in the PGW/CVP deployment model because some call flows with GTD enabled do not work. GTD in SIP messages is used by Unified CVP to pass along certain PSTN data to ICM. This limitation requires that SIP messages generated on and received by the PGW not contain GTD.

PGW Does Not Support Local REFER

PGW does not support SIP-to-SIP Internetworking; therefore, do not configure REFER labels in Unified CVP if these REFER labels will cause the PGW to send a SIP INVITE to another SIP UA.

PGW Does Not Support Locations-Based Call Admission Control (LBCAC)

Locations-Based CAC is not supported with the PGW. In order for LBCAC to function correctly, Unified CVP and Cisco Communications Manager (CUCM) must understand the location of the caller's media termination point in order to choose the optimal Voice XML (VXML) gateway for the call, as well as to calculate bandwidth consumption. Location information is currently not sent to Unified CVP in the format that is expected by Unified CVP and CUCM.

OAMP Data Delete (purge) Configuration May Be Different Than Current Setting on the Reporting Server

When importing Operations Console Server system-configuration, the Data Delete (purge configuration) settings may be different than the purge configuration that is currently set on the Reporting Server itself.

This setting mismatch occurs when you export to save the current Operations Console Server system-configuration, change and deploy new Data Delete settings (purge configuration) for a configured Reporting Server. Then, import the previously saved Operations Console Server system-configuration and restart the local Operations Console and ORM service.

When you re-sign in, the Data Delete settings for the given Reporting Server will be the old data delete configuration settings, which is different than what is present on the Reporting Server itself.

To avoid this data delete setting mismatch, redeploy the Data Delete settings to the desired settings against the given Reporting Server following an import of system-configuration.

All administrators must be aware of data delete settings to avoid inadvertently resetting them to a lower retention period.

This issue occurs because the Operation Console Server displays purge configuration settings that are stored in the local Operations Console Server database. More correctly, the Operations Console Server should display purge configuration settings that are currently set on the Reporting Server database. Otherwise, the previously noted synchronization issue can occur. This issue is being tracked with the identifier CSCsz19347,

Courtesy Callback Does Not Recognize DTMF if a Second Party is Conferenced In

When a caller is called back by Unified CVP using the new Courtesy Callback feature, the caller must press "1" to accept the call. If the caller conferences in a second party before pressing "1" to accept the call, then key press made by the caller (DTMF) is not recognized by the system. The system disconnects the call after the response timeout is reached and then attempts to make another callback to the caller. This issue is being tracked with the identifier CSCtc22787.

Validation in Call Studio does not Validate for items missing in Record Element

The validation feature in Call Studio does not verify if entries have been made for required items in the Record Element; specifically the PATH or FTP Site items. If these items are not provided, Call Studio will still return a “Validation Successful” message when validation is done on the VXML application. However, the application fails with an exception after being deployed with the items omitted.

The record element is used in the new Courtesy Callback feature, and this issue may arise when using Courtesy Callback if the PATH or FTP Site items have not been filled out in the record element. This issue is being tracked with the identifier CSCtf65567.

During Courtesy Callback, Survivability Script is not Heard if Call Server Connection to VXML Server is Terminated

If the connection between the Call Server and the VXML Server is severed during a courtesy callback between the time a caller answers the callback and before they press “1” to accept the call, then the call disconnects but the survivability script error message is not played to the caller. The port remains active on the gateway after the call is disconnected but eventually closes. This defect is being tracked by the identifier CSCtf31396.

Call Server Indicates Device has Unapplied Changes - Unable to Deploy Changes to Call Server

In rare cases in systems managed by multiple Operations Console servers a situation can occur where a Call Server indicates that a device has unapplied changes. However, attempts to save and deploy the changes to the Call Server have no effect. This defect is being tracked by the identifier CSCtf61991.

RTCP Interval/Timer Can Cause Survivability to Activate Too Soon

The IOS Templates included in Unified CVP Release 8.0(1) do not contain gateway configuration for RTCP timers. However, RTCP timer configuration is required for Unified CVP Survivability to operate correctly. RTCP reporting is a mechanism that allows the gateway to monitor a call and verify that both endpoints are connected. In the configuration example below, the interval denotes 6000 milliseconds and the timer is a modifier so that the amount of “dead air” that the system allows is $(6000 * 2)$ 12000 milliseconds or 12 seconds.

ip rtcp report interval 6000

timer receive-rtcp 2

In the context of Unified CVP, “dead air” occurs during times when the system is waiting for the caller to enter digits. For example, when using Courtesy Callback, the system asks the caller to enter their phone number. If the caller does not enter their complete phone number during the time allowed by the RTCP timer and interval configuration (12 seconds in the above example), then the Unified CVP Survivability script can kick in.

You must set the timer/interval combination low enough so that proper failover/survivability occurs when required, but high enough to give callers enough time to properly enter in information. You must configure the RTCP settings specific to your call deployment and scripts. If your deployment utilizes scripts that require multiple-digit input from callers, then this setting should allow for a longer interval/timer combination.

Gateway Template Variable Replacement Fails for Multiple Devices

When deploying a template with a Gateway template variable (IP address or location ID) to multiple Gateway devices at the same time, the Gateway template variable is replaced with the literal value of the first selected Gateway in the list for all devices. The second and later gateways all use the IP address from the first gateway, instead of using the correct variable replacement of their own IP address.

To prevent this, deploy templates that contain these template variables to one Gateway at a time rather than multiple Gateways in a single operation. This issue is being tracked by the identifier CSCtg42420.

CiscoUnifiedCVPInstall.log lists *Installation on <Unsupported OS>*

When installing Unified CVP on a Windows 2003 system installed with the IBM OEM Windows 2003 installer, a message appears in the logs stating Installation on <Unsupported OS>. This is a cosmetic issue only and can be ignored. The IBM OEM Windows 2003 install is a supported platform. This issue is being tracked by the identifier CSCtg44775.

Modifying G.729-encoded Prompts and Music-On-Hold Media for Gateway Compatibility

Unified CVP 8.0(1) supports G.729 in single-codec deployments. However, prompts and music on hold media files that are stored on a media server and played through a gateway must be modified to work with the gateway. The following procedure should only be used to modify G.729 encoded media files that have to be sent through a gateway from a Unified CVP media server. Music on Hold that is played from the Unified Communications Manager does not need to be modified

-
- Step 1** Log into Unified Communication Manager's MOH File management section. (Media Resources -> MOH Audio file management)
 - Step 2** Upload the .wav file that you want to convert to G.729. (Unified CM will convert to various formats)
 - Step 3** SSH to Unified Communication Manager's Admin CLI
 - Step 4** Run the following command to see the MOH file list: **admin:file list activelog mohprep**

```
admin:file list activelog /mohprep
CiscoMOHSourceReport.xml
SampleAudioSource.alaw.wav
SampleAudioSource.g729.wav
SampleAudioSource.ulaw.wav
SampleAudioSource.wb.wav
SampleAudioSource.xml
```

```
dir count = 0, file count = 6
```

- Step 5** Run the following command to SFTP the file to the share: **admin:file get activelog mohprep/<filename>.wav**

```
admin:file get activelog mohprep/EyesOnMePiano.wb.wav
Please wait while the system is gathering files info
...done.
Sub-directories were not traversed.
Number of files affected: 1
Total size in Bytes: 1343506
Total size in Kbytes: 1312.0176
Would you like to proceed [y/n]? y
SFTP server IP: <SERVER IP>
SFTP server port [22]:
User ID: <USERNAME>
Password: *****
Download directory: \
```

Make sure to use \ when downloading if your SSHd server is windows based.

- Step 6** Obtain a copy of a HEX editing tool. For example, one can be found at <http://www.hhdsoftware.com/Family/hex-editor.html> (This tool is not endorsed or supported by Cisco).
- Step 7** Edit the .wav file with the HEX editor, and set the header 16-bit value at offset 14 hex to 5339 (DB 14 lo-hi byte)

```
52 49 46 46 48 14 00 00 57 41 56 45 66 6D 74 20
16 00 00 00 xx xx 01 00 40 1F 00 00 E8 03 00 00
```

- Step 8** Change **xx xx** to **DB 14** and save the .wav file.

Changing the IP Address of a Call Server

When it is necessary to change the IP Address of a Call Server, a specific set of steps must be followed. The steps are documented in the Customer Voice Portal DocWiki, available online at: http://docwiki.cisco.com/wiki/General_Troubleshooting:_Server_IP_Address_Has_Been_Changed

Leading zero gets removed from incoming DNIS

In Unified CVP leading zero(s) gets removed from incoming DNIS when survivability is in use, and the call gets disconnected. A temporary solution is to change the destination-pattern on the VOIP dial peer to identify the incoming DNIS without leading zero(s) and to apply a translation rule on the outgoing DNIS to insert the leading zero(s) so that the upstream SIP Proxy route and the ICM dialed number remains the same.

Bug Toolkit

In addition to the resolved caveats listed above, you can also find the latest resolved caveat information through Bug Toolkit, which is an online tool that is available for customers to query defects according to their own needs.



Tip

You need an account with Cisco.com (Cisco Connection Online) to use the Bug Toolkit to find open and resolved caveats of any severity for any release.

To access the Bug Toolkit, log onto

<http://tools.cisco.com/Support/BugToolKit/action.do?hdnAction=searchBugs>

To access Bug Toolkit, you need the following items:

- Internet connection
- Web browser
- Cisco.com user ID and password

Procedure



Tip

To access the Bug Toolkit, go to

http://www.cisco.com/cgi-bin/Support/Bugtool/launch_bugtool.pl

- Step 1** Log on with your Cisco.com user ID and password.
- Step 2** Click the **Launch Bug Toolkit** hyperlink.
- Step 3** If you are looking for information about a specific caveat, enter the ID number in the "Enter known bug ID:" field.
- To view all caveats for Cisco Unified Customer Voice Portal, go to the "Search for bugs in other Cisco software and hardware products" section, and enter **Cisco Unified Customer Voice Portal** in the Product Name field.
- Step 4** Click **Next**. The Cisco Unified Customer Voice Portal search window displays.
- Step 5** Choose the filters to query for caveats. You can choose any or all of the available options:
- a. Select the Cisco Unified Customer Voice Portal Version:
 - Choose the major version for the major releases.
A major release contains significant new features, enhancements, architectural changes, and/or defect fixes.
 - Choose the revision for more specific information.
A revision release primarily contains defect fixes to address specific problems, but it may also include new features and/or enhancements.
 - b. Choose the Features or Components to query; make your selection from the "Available" list and click **Add** to place your selection in the "Limit search to" list.
To query for all caveats for a specified release, choose "All Features" in the left window pane.



Note

The default value specifies "All Features" and includes all of the items in the left window pane.

- c. Enter keywords to search for a caveat title and description, if desired.



Note To make queries less specific, use the All wildcard for the major version/revision, features/components, and keyword options.

- d. Choose the Set Advanced Options, including the following items:
- Bug Severity level—The default specifies 1-3.
 - Bug Status Group—Check the Fixed check box for resolved caveats.
 - Release Note Enclosure—The default specifies Valid Release Note Enclosure.
- e. Click **Next**.

Step 6 Bug Toolkit returns the list of caveats on the basis of your query. You can modify your results by submitting another query and using different criteria.

Open Caveats in This Release

This section contains a list of defects that are currently pending in Cisco Unified Customer Voice Portal Release 8.0(1). Defects are listed by component and then by identifier.



Tip

If you have an account with Cisco.com, you can use the Bug Toolkit to find caveats of any severity for any release. Bug Toolkit may also provide a more current listing than is reflected in this document. To access the Bug Toolkit, log onto http://www.cisco.com/cgi-bin/Support/Bugtool/launch_bugtool.pl

Table 1 Open Caveats for Cisco Unified Customer Voice Portal Release 8.0(1)

Identifier	Component	Headline
CSCtf31396	appsvr	Survivability not heard when call server goes down during callback
CSCsz19347	oamp	OAMP reflects the wrong purge setting after the database is restored
CSCtf61991	oamp	Unable to clear unapplied changes box from control center
CSCtg42420	oamp	Gateway template variable replacement fails for multiple devices
CSCtc22787	vxml_server	Callback does not recognize DTMF if 2nd party conferenced in
CSCtf65567	vxml_studio	Wrong configuration in element should be caught during validation
CSCtg44775	install	IBM OEM Win 2003 is identified as an <Unsupported OS>
CSCti03406	video-h324	Video Call failed
CSCth90215	video-h324	When audio transcoder is needed, video call got dropped
CSCth87505	video-h324	Video Conference Call fails, caller only sees one agent or screen freezes
CSCti00581	video-h324	When a video call is placed from a Sony 3G phone, no audio is heard

Obtaining Documentation and Submitting a Service Request

For information on obtaining documentation, submitting a service request, and gathering additional information, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

<http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html>

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