



# Release Notes for Cisco Unified Customer Voice Portal, Release 7.0(2)

July 2009

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This document updates the March 2009 version with the following changes:

<b>Section</b>	<b>Notes</b>	<b>Date Added</b>
Important Notes	Added, “Disabling Unified Presence Server DNS as a Best Practice”	July 9, 2009
Important Notes	Added “HTTPS Performance Numbers in Configuration and Admin Guide are Incorrect”	July 9, 2009
Important Notes	Added “Call Survivability Feature Now Mandatory for CVP 4.x(x) and 7.x(x)”	July 9, 2009
Streaming of Media in ICM Scripts	Changed wording from “music on hold” to “music while in queue” to differentiate from the Music on Hold feature in Communications Manager.	June 30, 2009
Important Notes	Added “UDP Transport as a SIP Best Practice”	June 30, 2009

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## Introduction

This document discusses new features, changes, and caveats for Maintenance Release 7.0(2) of Cisco Unified Customer Voice Portal (Unified CVP) software.

This document is a supplement to the *Release Notes for Cisco Unified Customer Voice Portal, Release 7.0(1)*, which are available at:

[http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod\\_release\\_notes\\_list.html](http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_release_notes_list.html)

The *Release Notes for Cisco Unified Customer Voice Portal, Release 7.0(2)* **must** be used in conjunction with the just mentioned Release Notes.

Cisco Unified Customer Voice Portal Release 7.0(2) is a maintenance release that contains fixes and a limited set of new functionality. Release 7.0(2) is incremental and cumulative, and can be rolled back.

## New and Changed Information

This section contains the following topics:

- Firefox support
- 7985G video hardphone support
- CUPC Softphone support
- Cisco Security Agent
- VoiceXML Server Gateway Adapters
- McAfee 8.5i support
- Symantec 10.1.5
- Streaming of media in ICM scripts
- Streaming of media for custom ringtones
- SIP SigDigits now in OAMP

### Firefox Support

Firefox version 2.x is now a supported browser for the Video Desktop introduced in CVP 7.0(1).

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## **7985 Video Hardphone Phone Support**

The basic and full video solution introduced in CVP 7.0(1) now fully supports the 7985 video phone.

## **CUPC Softphone Support**

The basic and full video solution introduced in CVP 7.0(1) now fully supports the Cisco Unified Personal Communicator (CUPC) video desktop/softphone.

## **VoiceXML Server Gateway Adapters**

Two new VoiceXML Server Gateway Adapters were added in this release. The new Gateway Adapters should be used only in setups that use third party ASR engines that are currently not supported by any other adapter. Please contact your third party ASR vendor, if planning to use those adapters. Note that those adapters do not have any impact on any other adapters present in the software or the existing applications. The new adapters are:

- Unified CVP 7.0 with Speech
- Unified CVP 7.0 VoiceXML 2.1 with Speech

Note that Cisco Unified Call Studio 7.0(2) is required to create the applications using those new adapters.

## **Cisco Security Agent**

The Cisco Security Agent version 5.2.238 policy 305, both in standalone form and policy export file, has been released to work specifically with Unified CVP 7.0(2).

## **McAfee 8.5i Support**

Unified CVP 7.0(2) has been qualified to work with McAfee 8.5i.

## **Symantec 10.1.5 Support**

Unified CVP 7.0(2) has been qualified to work with Symantec 10.1.5.

## **Streaming of Media in ICM Scripts**

Adds the ability to use streaming media for prompts using micro-apps. In the PlayMedia micro-app you can now specify an RTSP stream for your media. This will launch an RTSP session via the IOS Gateway to the Helix streaming server. This is a way to play a broadcast stream already in progress, instead of playing a .wav file to the caller. This gives the ability to replace music while in queue with a broadcast stream. IOS version 12.4(20)T1 is required on the gateway for this functionality.

## **Streaming of Media for Custom Ringtones**

Allows administrators to use an RTSP URL in the custom ringtones patterns DN list. Now you may play a broadcast stream already in progress as a ringtone when transferring to an agent. IOS version 12.4(20)T is required on the gateway for this functionality.

## **SIP SigDigits now in OAMP**

Now that the configuration of SIP SigDigits has been added to OAMP in Unified CVP 7.0(2), customers are urged to use this new facility and to stop performing manual changes to the sip.properties file directly on the Call Server. Any manual changes will be overwritten by the next "save and deploy" OAMP operation. If you had customized the SigDigits value manually prior to installing UNIFIED CVP 7.0(2), you should set this value in OAMP after upgrading to 7.0(2) to insure that your custom value is not inadvertently overwritten at a later time.

## **Installation Notes**

This section provides information about the following installation topics:

- Pre-Installation Issues
- Installation Requirements
- Installation Steps

### **Pre-Installation Issues**

Before you attempt to install, do the following:

- Close and Unlock All Unified CVP Related Files
- Backup VoiceXML Server Customer Audio Files

### **Close and Unlock All Unified CVP Related Files Before Installing**

Before you attempt to install or uninstall Unified CVP Release 7.0(2) Maintenance Release, make sure that all unnecessary applications are shut down and opened files are closed. If a file that the Unified CVP installation program requires is locked, a failed installation or uninstallation could result.

### **Backup VoiceXML Server Custom Audio Files**

VoiceXML Server saves custom audio files in the web deployment directory of either Apache Tomcat or IBM WebSphere Application Server. When installation or

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uninstallation of the Unified CVP 7.0(2) Maintenance Release of Unified CVP includes updating of the VoiceXML Server component, the previous deployment of VoiceXML Server, including any custom audio files in this directory, is completely removed. Therefore, if you wish to continue using the custom audio files from the previous deployment, you must back up the files manually in a temporary directory prior to installing or uninstalling the service release. These files may be restored back to their deployment location. Deployment directory examples:

```
Tomcat deployments - %CVP_HOME%\VXMLServer\Tomcat\webapps\CVP\audio
WebSphere deployments - %WAS_HOME%\profiles\AppSrv01\installedApps\serverCell\CVP VXML
server.ear\CVP.war\audio
```

## Installation Requirements

Unified CVP 7.0(2) can be installed in the following ways:

- an existing installation of Unified CVP 7.0 would already have Unified CVP 7.0(1) installed; Release 7.0(2) would be installed over this
- for a new installation of Unified CVP 7.0 that uses only Microsoft Windows, install Unified CVP 7.0(1) and then immediately install Release 7.0(2) over this---configuration between the installation of Release 7.0(1) and Release 7.0(2) is allowed but optional
- for a new installation of Unified CVP 7.0 that uses AIX, install Unified CVP 7.0(1), then immediately install Release 7.0(2) over this---configuration between the installation of Release 7.0(1) and Release 7.0(2) is allowed but optional

## Windows Installation and Uninstallation Steps

Complete the following steps to install Unified CVP:

1. Open and run the Windows Unified CVP 7.0(2) patch installer executable.
2. Select Install or Uninstall.
3. Follow the instructions displayed on the screens.
4. Restart your server upon successful completion of patch installation or uninstallation.
5. On VXML Server using WebSphere Application Server (WAS), redeploy the CVP.war file
  1. The CVP.war file must be redeployed following both installation and uninstallation.
  2. To do so, you will use the WebSphere web admin (e.g. [http://<server\\_ip\\_address>:9060/ibm/console/login.do](http://<server_ip_address>:9060/ibm/console/login.do))
  3. Goto Applications -> Enterprise Applications
  4. Select "CVP VXML server" and click on "Update"

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5. In the "Preparing for the application installation" page, select "Replace the entire application"
6. Keep all parameters to their default values on Step 1, 2 and 3
7. Click "Finish" on step 4, "Save" the changes to the master configuration
8. Restart WebSphere

### **Informix Services Unable to Stop upon Certain Patch Installations and Uninstallations**

Upon Unified CVP patch installation or uninstallation, one or more Informix-related services may not stop as requested by the Unified CVP patch installer. This symptom has been found to occur when a patch installation or uninstallation is kicked off under heavy Informix load or processes. These services can include:

- ISM Local Execution (nsrexecd)
- ISM Portmapper (portmap)
- ISM Server (nsrd)

An administrator can workaround these issues by stopping Informix processes and services before running a patch installation or uninstallation:

In Windows, the end-user should perform the following steps:

1. Start -> Settings -> Control Panel -> Administrative Tools -> Services
2. All started Informix services should be stopped manually.
3. Verify the Informix services are in a stopped/offline state.
4. Run the Unified CVP patch installer.  
Alternatively, these processes may be stopped/killed within the Windows Task Manager.

### **AIX Installation and Uninstallation Steps**

Complete the following steps to install Unified CVP on AIX:

Note: Once installed, the Unified CVP 7.0(2) maintenance release for AIX cannot be rolled back to 7.0(1)

1. Run cvp702\_aix.bin.
2. The patch includes a new CVP.war file that must be redeployed manually. To do so, you will use the WebSphere web admin (e.g. [http://<server\\_ip\\_address>:9060/ibm/console/login.do](http://<server_ip_address>:9060/ibm/console/login.do))
  1. Goto Applications -> Enterprise Applications
  2. Select "CVP VXML server" and click on "Update"
  3. In the "Preparing for the application installation" page, select "Replace the entire application"
  4. Keep all parameters to their default values on Step 1, 2 and 3

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5. Click "Finish" on step 4, "Save" the changes to the master configuration
6. Restart WebSphere

### **AIX Uninstallation Steps**

To uninstall Unified CVP 7.0(2) from the AIX operating system, perform the following steps:

1. Uninstall the Unified CVP VXML Server web application from WebSphere
  1. Go into the WebSphere web admin (e.g. `http://<server_ip>:9060/ibm/console/login.do`)
  2. Goto Applications -> Enterprise Applications
  3. Select and delete "CVP VXML server"
  4. "Save" the changes to the master configuration
  5. Restart WebSphere
2. Uninstall Unified CVP
  1. Run the `%CVP_HOME/bin/uninstall.sh` script
  2. Confirm your desire to permanently uninstall CVP 7.0(2) from the AIX server

## **Important Notes**

This section provides information about the following topics:

- Call Survivability Feature Now Mandatory for CVP 4.x(x) and 7.x(x)
- HTTPS Performance Numbers in Configuration and Admin Guide are Incorrect
- Disabling Unified Presence DNS as a Best Practice
- UDP Transport as a Best Practice
- Basic video transfers with CUCM 7.0
- Consult transfer with video enabled endpoints and H.323
- Warm transfer with queuing over UDP with CUPS 7.0(1)
- Basic video call with Telepresence endpoint
- MTP and completing transfers while still in queue
- Takeback-and-Transfer (TNT)
- T38 Not Supported

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- CVP VB RAI Logic

### **Call Survivability Feature Now Mandatory for CVP 4.x(x) and 7.x(x)**

All calls that originate in the PSTN and reach Unified CVP via a POTS dial-peer must have call survivability configured on the POTS dial-peer. Any deployments where calls originate to Unified CVP via POTS dial-peer that do not use the survivability feature are not supported unless the feature being used explicitly states that survivability is not required.

For more information on call survivability and the survivability.tcl script, see the section “Call Survivability” in the [Configuration and Administration Guide for Unified Customer Voice Portal](#).

### **HTTPS Performance Numbers in Configuration and Admin Guide are Incorrect**

The published performance numbers for deployments using HTTPS are not correct in the *Configuration and Administration Guide for Unified Customer Voice Portal, Release 7.x(x)*. The correct performance numbers are available in the [Cisco Unified Customer Voice Portal \(CVP\) Solution Reference Network Design \(SRND\)](#). See the sections on sizing.

### **Disabling Unified Presence Server DNS as a Best Practice**

Long delays can be introduced when using DNS in the Unified Presence server. Additionally, configuration is made more complicated when DNS is used. For example, you must configure reverse DNS lookups, and special ACL and presence settings.

It is strongly recommended to disable Unified Presence Server’s DNS to simplify configuration and decrease the possibility of introducing delays. When installing Unified Presence Server, the install prompts you to enter the DNS server (primary and secondary). You may leave the DNS entries blank to disable DNS altogether.

You must configure your routes in Unified Presence to use IP addresses if you disable DNS. Unified Presence Server DNS is not necessary when all routes are configured with IP addresses.

For more information on the delay issue on how to disable the Unified Presence DNS server, see the Cisco Unified Presence Server *Release Notes* at: [http://www.cisco.com/en/US/docs/voice\\_ip\\_comm/cups/6\\_0\\_5/english/rel\\_notes/cup602E\\_Srn.html#wp248243](http://www.cisco.com/en/US/docs/voice_ip_comm/cups/6_0_5/english/rel_notes/cup602E_Srn.html#wp248243)

### **UDP Transport as a SIP Best Practice**

Use UDP transport when you are using SIP with Unified CVP to prevent timing issues and longer failover response times. Timing issues can sometimes occur when using TCP

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as the transport protocol. Additionally, there can be slower failover response times when using TCP. These issues are not present when using UDP as the transport protocol.

Note that TCP transport is the default selection on the Unified Communication SIP Trunk. Setting this to UDP for the trunk that interconnects with Unified CVP may resolve timing issues seen with TCP.

You can set the transport to UDP on the SIP Trunk security profile used for the trunk on route-pattern matching by going to the following menu in *Unified Communication Manager Administration*:

1. Select **System > Security Profile > SIP Trunk Security Profile**
2. Click **Find**, and select the appropriate SIP Trunk.
3. Set the "Security Device Mode" to Non-Secure, and change "Outgoing Transport Type" to UDP
4. Click **Save**.

### **Basic video transfers with CUCM 7.0**

Single step or Consultative transfers is successful in basic mode but has no video if MTP is checked on the SIP trunk used for warm transfer queuing.

The following call flows fail in that there is no video once connected (7960 and 7985 endpoints)

- Single Step Transfer with queuing at Agent2- - no video after transfer
- Consult Transfer with queuing at Agent2- - no video at consult or after transfer

Though Unified CVP, versions 4.0(1), 4.0(2), 4.1(1) and 7.0(1), no longer requires MTP checked, MTP allocation is forced by the recommended workaround to IOS defect CSCsm12716. The CSCsm12716 issue is specific to completing a transfer while an agent is in queue. Customers need to test with MTP unchecked on warm transfer trunk to make the above call flows work.

### **Consult transfer with video enabled endpoints and H.323**

Consultative transfers with video enabled endpoints do not work with the H.323 call control protocol. The workaround is to disable video capabilities at the endpoints (e.g. disable video on the phones).

### **Warm transfer with queuing over UDP with CUPS 7.0(1)**

If you are experiencing call disconnects when completing consultative transfers with queuing over UDP when using CUPS version 7.0(1), use the following workaround:

- use the TCP transport on the CUPS static routes pointing to CUCM.
- use the TCP transport on the SIP trunk security profile on CUCM. It has to be the SIP trunk used for the VRU leg of the transfer. This is the trunk used with MTP regarding CSCsm12716 mentioned above.

Note: On earlier versions of CUPS where the "record route header" feature can be and is disabled, this issue won't occur even when using UDP.

Note: Another workaround is to point CUCM SIP Trunk direct to Unified CVP instead of CUPS, and UDP may be retained. This is the trunk used with MTP regarding CSCsm12716 mentioned above.

Note: TCP may automatically be used on a CUPS call if the MTU size of the SIP message exceeds 1300 bytes by default. This generally happens when signaling forward unconditional is used on the ingress gateway. To preserve UDP transport usage you may change CUP service parameter setting for MTU upconversion to 1800 instead of 1300.

### **Basic video call with Telepresence endpoint**

In a certain scenario, basic video calls from a Telepresence endpoint will require MTP to be enabled on the SIP trunk pointing to Unified CVP. The scenario is as follows:

1. Telepresence caller dials into Unified CVP
2. receives audio-only IVR
3. caller is then is transferred to an Agent on an audio-only IP Phone

MTP must be enabled on the SIP trunk or else one-way audio is encountered. (Caller could not hear audio without MTP enabled)

### **MTP and completing transfers while still in queue**

If the agent completes the transfer to caller while still in queue, it causes the mid call media change for the voice browser on the VXML gateway, which the gateway does not currently support (CSCsm12716). Therefore, MTP allocation becomes necessary.

You can mitigate this MTP usage with a "dual SIP Trunk" configuration. The first CVP/CUPS SIP Trunk with MTP \*unchecked\* and the second SIP Trunk with MTP checked to the same CVP/CUPS destination. The VRU label route pattern should be applied on this MTP trunk. This will allow you to only use MTP allocation on the warm transfers with queueing. All the inbound calls will avoid MTP allocation.

When a warm transfer with no queuing (ie no vru label + corr id) is performed, there will

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be no second call from the trunk to Unified CVP. There will only be the SIP reinvite to the caller on the transfer completion. The MTP outbound call is only made when EAPIM VRU label is routed out the trunk to Unified CVP, ie the outbound call.

For the above to work, it is necessary to create an alternate SIP Trunk Security Profile, apart from the default one "non secure sip security profile". Use all the same settings as the default one, but change the Listen Port to something else, like 5065. Now on the second trunk that has the MTP, apply this security profile, and reset both SIP Trunks.

The reason for this is because you cannot create 2 duplicate sip trunks in CUCM with the same host/ip and listen port.

Again, you are never using this trunk for inbound calls, so you never need to make a static route from CUPS/CVP to port 5065 for example. Just use 5060 that is associated on the other non-MTP trunk. Also, if you are going to have other DNs for the IP originated callers, not the warm transfer with queuing calls, then use the non-MTP trunk on that route pattern.

IOS-based Hardware MTP should be used over CUCM Software MTP for performance considerations.

### **Takeback-and-Transfer (TNT)**

The Takeback-and-Transfer (TNT) feature is not supported with the SigDigit feature.

### **T38 Not Supported**

CVP does not support T.38. When implementing FAX services within the CVP environment you must match FAX calls with separate dial-peers on the ingress GW and send the call to the relevant call control entity - either CUCM or FAX server directly.

### **CVP VB RAI Logic**

The RAI logic in CVP depends on four different parameters:

- RAIThreshold
- maxTotalCalls
- maxIVRPorts
- takebackPercentage

All of these parameters are configurable via the VBAdmin.

The RAI is NOT a factor of just the total calls but the IVR ports and takeback percentage, transferred calls do play a role in it. The formula used for sending RAI follows.

The RAI indicator instructs the Gatekeeper to stop routing further incoming calls to the H.323 service if the number of active incoming calls exceeds:

- $(RAIMaxThreshold/100)*MaxTotalCalls$
- or–
- $(RAIMaxThreshold/100)*maxIVRports - (currentTransferredCalls * (takebackPercentage/100))$

The RAI indicator instructs the Gatekeeper to start routing incoming calls to the H.323 service if the number of active incoming calls drops below:

- $(RAIMinThreshold/100)*MaxTotalCalls$
- or–
- $(RAIMinThreshold/100)*maxIVRports - (currentTransferredCalls * (takebackPercentage/100))$

## **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. 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

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To access the Bug Toolkit, go to

[http://www.cisco.com/cgi-bin/Support/Bugtool/launch\\_bugtool.pl](http://www.cisco.com/cgi-bin/Support/Bugtool/launch_bugtool.pl)

Then complete the following steps:

1. Log on with your Cisco.com user ID and password.
2. Click the **Launch Bug Toolkit** hyperlink.
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 *<product name>* in the Product Name field.
4. Click **Next**. The Cisco Unified Customer Voice Portal search window displays.
5. Choose the filters to query for caveats. You can choose any or all of the available options:
  - a. Select the Cisco Unified Call Studio 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. 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. 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.

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- Bug Status Group—Check the Fixed check box for resolved caveats.
  - Release Note Enclosure—The default specifies Valid Release Note Enclosure.
- e. Click **Next**. 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.

## Resolved Caveats in This Release

This section lists caveats specifically resolved by Unified CVP, Maintenance Release 7.0(2). Click the highlighted identifier to access the bug in the Cisco system.

Identifier	Component	Headline
<a href="#">CSCsl52876</a>	ss_sip	Telepresence enters 5 on GD but we hear 55 on PD
<a href="#">CSCsm81385</a>	ss_ivr	Hanging up on a Video result in wrong FinalObjectID being returned
<a href="#">CSCsl56232</a>	snmp	SNMP returns no values for certain stats using v1 enterprises.9
<a href="#">CSCsl74893</a>	vxml_server	High CPU usage contribution from ApplicationLoggerBase.runLogger()
<a href="#">CSCso18465</a>	ios	survivability.tcl causes voice channels to hang on gateways
<a href="#">CSCso95910</a>	ss_sip	CVP didn't send ACK to VXML GW
<a href="#">CSCso87847</a>	ss_ged	CAP microapp resets CED to null.
<a href="#">CSCsr04130</a>	ss_sip	ringtone media pattern matcher has issue with exact matches

<a href="#">CSCsr12054</a>	ss_ivr	Switch leg is not released after ICM sends Release
<a href="#">CSCsm95855</a>	ss_sip	ANI not updated on outbound INVITE
<a href="#">CSCso22895</a>	install	Installer for CVP should always update CVP to the most current DST
<a href="#">CSCsl99549</a>	vxml_server	Improve CS so that can recover from customer error at start of call
<a href="#">CSCsm51425</a>	vxml_server	Web Services element should allow substitution in auth credentials
<a href="#">CSCsm95027</a>	vbrowse	Option needed to separate ANI into CLI portion and trunk info portion
<a href="#">CSCsm72930</a>	tools	ST is not fetching log if CVP server is on timezone that is ahead of GMT

## Open Caveats in This Release

This section contains a list of defects that are currently pending in Unified CVP 7.0(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](http://www.cisco.com/cgi-bin/Support/Bugtool/launch_bugtool.pl)

### Generic

Identifier	Component	Headline
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<a href="#">CSCso78599</a>	ss_sip	No survivability on call after SIP refer done out of CVP
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<a href="#">CSCsq27851</a>	ss_ivr	Barge-in through Play Media, barges through all following play media
<a href="#">CSCso22596</a>	vxml_server	VXML self-service is not working with HTTPS on
<a href="#">CSCsm53958</a>	ss_ivr	CVP report video-answer for an audio only agent extension
<a href="#">CSCsl62453</a>	radvision	Garbage title text seen for Japanese, Simplified Chinese
<a href="#">CSCsm67071</a>	radvision	systemError seen in IVP logs after full video load tests
<a href="#">CSCsm67098</a>	radvision	add-media-response Rejected seen from IVP while under full video load
<a href="#">CSCsl20948</a>	radvision	No error message when recording to directory with no free space

## Video transfer

The following issues should be considered when using Basic or Full Mode Video with transfer call flows:

Identifier	Component	Headline
<a href="#">CSCsr28202</a>	ss_ivr	Full: No ringtone video observed by caller after single step transfer
<a href="#">CSCsq55948</a>	doc	Basic Mode: No video after call completion when MTP checked on WT Trunk
<a href="#">CSCsq38752</a>	doc	SendtoVRU needed to see Ringtone Video during Consult Transfer
<a href="#">CSCsq61134</a>	CUCM	Conference: HW MTP Needed to complete basic conf using Route Point
<a href="#">CSCsq76162</a>	radvision	Consult Transfer - 2nd Agents VCR controls do not function on phone.

<a href="#">CSCsq76204</a>	radvision	Caller Presses Hold, Agent sees frozen video
<a href="#">CSCsq95117</a>	Telepresence	Telepresence: Cannot answer when call ID restricted on trunk to CVP
<a href="#">CSCsq94960</a>	doc	MTP required on SIP trunk for Telepresence calling audio agent -IP Phone
<a href="#">CSCsr20746</a>	doc	CVP/H.323 does not support video enabled phones
<a href="#">CSCso95949</a>	radvision	Not able to barge through VPM after SST (but able after consult xfer)
<a href="#">CSCsm95822</a>	radvision	video fails on warm consult xfer from audio-only agent to video-agent

## 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>

Subscribe to the *What's New in Cisco Product Documentation* as a Really Simple Syndication (RSS) feed and set content to be delivered directly to your desktop using a reader application. The RSS feeds are a free service and Cisco currently supports RSS version 2.0.

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