Call Transfer Options

Release Trunk Transfer

Release Trunk Transfer releases the ingress trunk and removes Unified CVP and the gateway from the call control loop. These transfers have the following characteristics:

- They can be invoked by VXML Server (Standalone Call Flow Model) or using Unified ICM.
- Unified ICM Network Transfer using Unified CVP as the routing client does not work because Unified CVP can no longer control the call.
- They are blind, that is, if the transfer fails for any reason then Unified ICM does not recover control of the call. Router Requery is not supported.
- They cause the switch leg to terminate resulting in a Telephony Call Dispatcher (TCD) record being written to the database for the call even though the caller is still potentially talking to an agent. This behavior differs from other types of transfers in which the TCD record is not finalized until the caller hangs up.
- As the ingress trunk is released, you do not have to size gateways to include calls that have been transferred using Release Trunk Transfer. This behavior differs from other types of transfers in which gateway resources continue to be occupied until the caller hangs up.
- Because Unified CVP is no longer monitoring the call, you do not have to size Call Servers to include calls that have been transferred using Release Trunk Transfer. Additionally, Unified CVP Call Director port licenses are not required.

Following are the signaling methods available to trigger a release trunk transfer:

- Takeback-and-Transfer. See Takeback-and-Transfer, on page 2
- Hookflash and Wink. See Hookflash and Wink, on page 2

- Two B Channel Transfer. See Two B Channel Transfer, on page 3

Takeback-and-Transfer

Takeback-and-Transfer (TNT), also known as Transfer Connect, is a transfer method where dual tone multifrequency (DTMF) tones are outpulsed to the PSTN by Unified CVP. TNT outpulses DTMF tones to the PSTN. A typical DTMF sequence is *8xxxx, where xxxx represents a new routing label for the PSTN. Upon detection of a TNT DTMF sequence, the PSTN drops the call leg to the Ingress Gateway port, and then reroutes the caller to a new PSTN location, such as a TDM ACD location. This method is offered by a few PSTN service providers.

Customers can use TNT, if they have an existing ACD site but no IVR and want to use Unified CVP as an IVR. Over time, customers may need to transition agents from the TDM ACD to Unified CCE and use Unified CVP as an IVR, queueing point, and transfer pivot point. Using Unified CVP as more than just an IVR eliminates the need for TNT services.

In Unified CVP deployments with Unified ICM, the DTMF routing label that is outpulsed can be a Unified ICM translation routing label to enable passing of call data to another Unified ICM peripheral, such as a TDM ACD. In this scenario, Unified CVP views the call as completed, and Unified CVP call control is ended. With TNT, if the transfer to the termination point fails, Unified CVP cannot reroute the call. Using some TNT services, you can reroute the callback to Unified CVP. However, Unified CVP treats this call as a new call.

Hookflash and Wink

Hookflash and wink are signaling methods that are associated with a TDM PBX or ACD. With the Hookflash feature, Unified CVP can transfer SIP calls using a hookflash followed by the DTMF destination. This feature allows for deployments in which a PBX is in the front-end of the Unified CVP Ingress Gateway, and in which the PBX provides non-VoIP connectivity to agents. Hookflash applies to analog trunks and wink applies to digital trunks (T1 or E1 channel), although both are similar in function. Both hookflash and wink send an on-hook or off-hook signal to the PBX or ACD, which responds with dial tone (or the PBX winks back on a digital trunk). This signaling causes the Voice Gateway to send a string of routing digits to the PBX or ACD. Upon collection of the routing digits, the PBX or ACD transfers the caller to the new termination point, which can be an ACD queue or service on that same PBX or ACD.

Customers can use hookflash and wink if they have an existing ACD but no IVR, and want to use Unified CVP as an IVR that is installed on the line side of their existing PRX or ACD. Over time, the customer may need to transition agents from the TDM ACD to Cisco Unified CCE and have the Voice Gateways connected to the PSTN instead of the line side of the PBX or ACD. In Unified CVP deployments with Unified ICM, the routing label can be a Unified ICM translation routing label. This label enables passing of call data to the ACD service (and subsequently to the agent in a popup message). With hookflash and wink, if the transfer to the termination point fails, Unified CVP cannot reroute the call. Some PBX or ACD models can reroute the callback to Unified CVP. However, Unified CVP treat this call as a new call.

Note When PBXs and gateways have constrained support, hookflash transfer becomes difficult. If possible, avoid using the PBX for Unified ICM switching. Also, terminate all incoming calls on Unified CVP ingress gateways to allow Unified CVP to route calls to the PBX rather than on the ingress gateways.

Following guidelines and notes apply for hookflash transfers:

- Cisco 1700 Series Gateways are not tested with hookflash transfers.

- Cisco 2800 and 3800 Series Gateways can support Analog FXO or Digital FXO (T1/CAS). This function is considered line-side hookflash to the PBX. However, E&M is not supported at this time. You can adjust the hookflash duration with the **timing hookflash-out** command in **voice-port**. This feature is useful if you have a PBX that has a nonconfigurable hookflash duration, and it gives you the ability to adjust the hookflash duration on the gateway side.
- Cisco 5x00 Series Gateways are tested with T1/CAS and the **e&m-fgb dtmf dnis** command. E&M is considered "trunk-side hookflash" to the PBX, and not all switches support trunk-side hookflash. Additionally, the hookflash duration on the Cisco 5x00 Series Gateways is 200 ms, and you must configure the PBX for the same duration. This option varies with switch type and a proof-of-concept with the switch used.
- In Deployment Model No. 1, Standalone Self-Service, a TCL script is required to produce the hookflash. A TCL script is provided with Unified CVP.
- For Digital FXO (T1 CAS) Trunks, configure Dialed Number Identification Service (DNIS) on the gateway, based on the T1/E1 channel on which the call arrives. The PBX is programmed to route DNIS calls over T1 trunks. Configure the DNIS of the gateway because the call arrives to the gateway on that trunk.
- For Digital FXO (T1 CAS) Trunks, configure Dialed Number Identification Service (DNIS) on the gateway, based on the T1/E1 channel on which the call arrives. The PBX is programmed to route DNIS calls over T1 trunks. Configure the DNIS of the gateway because the call arrives to the gateway on that trunk.

  Note The disadvantage to this approach is that the gateway trunk allocation must be predetermined. You must know the percentage of calls that arrive to a particular DNIS so that the trunk groups on the gateway can be allocated accordingly.

  An alternate method, also known as converse on step, can be used on some PBXs where DTMF tones indicating DNIS and ANI are sent to the IVR. This method requires a single main Unified ICM routing script to input DNIS digits using a Get Data (GD) Microapplication and to invoke the correct subscript based on the collected DNIS digits. This method requires close coordination between Cisco, the PBX vendor, and the customer.
- For FGB E&M trunks in Cisco 5x100 Series Gateways, ANI and DNIS can be sent by using "*" as the delimiter, for example, *ANI*DNIS*. For configuration details, see *ANI/DNIS Delimiter for CAS Calls on CT1*, available at http://www.cisco.com/c/en/us/td/docs/ios/redirect/eol.html.

Note
- Hookflash is supported on 2X and 3X gateways only.
- Hookflash applies to TDM-originated calls only. After Unified CVP invokes hookflash, Unified CVP is no longer in control of the call.

Two B Channel Transfer

Two B Channel Transfer (TBCT) is an Integrated Services Digital Network (ISDN)-based release trunk signaling function that is offered by some public switched telephone network (PSTN) service providers. When a TBCT is invoked, the Ingress Gateway places the initial inbound call on hold briefly while a second call leg (ISDN B Channel) is used to call the termination point. When the termination point answers the call, the gateway sends ISDN signaling to the PSTN switch to request to complete the transfer, bridge the call through the PSTN switch, and remove the call from the Ingress Gateway. As with a TNT transfer, the termination point might be a TDM PBX or ACD connected to the PSTN.

This process may be necessary for a customer with an existing ACD site but no IVR, who wants to use Unified CVP initially as just an IVR. Over time, the customer might want to transition agents from the TDM ACD to Cisco Unified CCE and use Unified CVP as an IVR, queueing point, and transfer pivot point (which eliminates the need for TBCT services and using Unified CVP to perform reroute on transfer failure).

ICM Managed Transfer

Unified CVP performs ICM Managed Transfer function, which provides gateway-based switching for Unified ICM and Unified CCE installations.

In Unified CVP deployments with Unified ICM, Unified ICM provides all call control. VoiceXML call control from the VXML Server is not supported when Unified ICM is deployed with Unified CVP.

Unified ICM Managed transfer is used to transfer the call to any of the following new termination points:

- A Cisco Unified Communications Manager phone
- An egress port on the same gateway as the ingress port
- A distant Egress Gateway that has a TDM connection to a TDM ACD or PBX (making use of toll bypass features)
- A Unified CVP VoiceXML gateway for queuing or self-service activities

To terminate a call, the Voice Gateway selects an outgoing POTS or VoIP dial peer based on the destination specified by Unified ICM. When a Unified ICM VoIP transfer occurs, the Ingress Voice Gateway port is not released. If the termination point is an Egress Voice Gateway, then a second Voice Gateway port is utilized. Unified CVP continues to monitor the call, and Unified ICM also retains control of the call and can instruct Unified CVP to transfer the call to a new destination.

This type of transfer is used when Unified CVP is used as a call treatment platform and queue point for Unified CCE agents. Unified CVP can also be used to provide call treatment to front-end calls to TDM ACD locations supported by Unified ICM. This type of transfer allows calls to be transferred between peripherals supported by Unified ICM, with full call context and without any return of the voice path.

Calls that are transferred in this way have the following characteristics:

- Unified ICM Network Transfer using Unified CVP as the routing client functions properly because Unified CVP continues to control the call.
- These transfers are supervised, meaning that if the transfer fails for any reason, the Unified ICM routing script does recover control through Router Requery method.
- The switch leg does not terminate until the caller hangs up. The TCD record that is written for the switch leg of the call encompasses the entire life of the call, from initial ingress to hang up.
- Gateways sizing is done to include calls that have been transferred using ICM Managed Transfers because the ingress trunk is not released.

- Call Servers sizing is done to include the calls that have been transferred using ICM Managed Transfers because Unified CVP continues to monitor the call. Additionally, Unified CVP Call Director port licenses are required, except for calls that are connected to Cisco Unified Communications Manager agents.

Network Transfer

Unified CVP allows Network Transfer to transfer calls to another destination after they have been answered by an agent.

When a call is transferred from Unified CVP to an agent, and that agent wants to transfer the call to another agent, the agent can make that transfer using either the agent IP phone or the agent desktop. Transfers from the IP phone are made using CTI route points that point to a Unified ICM script. Transfers from the agent desktop are made using the Dialed Number Plan.

There are two flags in Unified ICM to control the Network Transfer:

- **NetworkTransferEnabled**—This flag is part of the Unified ICM script. When enabled, it instructs the Unified ICM to save the information about the initial routing client (the routing client that sent the NewCall route request).
- **NetworkTransferPreferred**—This flag is enabled on the Unified CVP Peripheral Gateway configuration. When enabled, any route request from this routing client sends the route response to the initial routing client instead of the routing client that sent the route request.

The following points explain how you can do a network transfer:

- You can use Network Transfer to perform a blind transfer only from agent 1 to agent 2 through Unified CVP. In this case, Unified ICM instructs Unified CVP to route the call back from agent 1, and then route it either to a VoiceXML Gateway (for IVR treatment) or to another destination (for example, to agent 2).
- You cannot use Network Transfer to perform a warm transfer or conference with Unified CVP because the call leg to agent 1 must be active while agent 1 performs a consultation or conference. Unified CVP cannot route the call back from agent 1 during the warm transfer or conference.

If a caller dials the same number regardless of a blind transfer, warm transfer, or conference, then follow these best practices:

- Do not enable the **NetworkTransferEnable** flag in the Unified ICM script.
- Dial the CTI Route Point of the same Unified CCE Peripheral Gateway for any transfer or conference request to preserve the call context during the transfer. Dialing the Route Pattern or CTI Route Point of another Peripheral Gateway does not preserve the call context.
- Use **SendToVru** as the first node in the Unified ICM routing script.

Note Extra ports are used during the consultation, blind transfer, or conference calls. They are released after the originating consultation is terminated.

SIP Refer Transfer

In some scenarios, Unified CVP transfers a call to a SIP destination and does not have Unified ICM and Unified CVP retain any ability for further call control. Unified CVP can perform a SIP Refer transfer, which allows Unified CVP to remove itself from the call, and free licensed Unified CVP ports. The Ingress Voice Gateway port remains in use until the caller or the terminating equipment releases the call. SIP Refer transfers are used in both Comprehensive and Call Director deployments.

Invoke a SIP Refer transfer by any of the following methods:

- Unified ICM sends Unified CVP a routing label with a format of rfXXXX (For example, rf5551000).
- An application-controlled alternative is to set an ECC variable (user.sip.refertransfer) to the value **y** in the Unified ICM script, and then sends that variable to Unified CVP.

Note Direct Refer transfer using label works only if**Send To VRU** node is used before the Refer.

You can invoke the SIP Refer transfer after Unified CVP queue treatment has been provided to a caller. SIP Refer transfers can be made to Cisco Unified Communications Manager or other SIP endpoints, such as a SIP-enabled ACD.

Router requery on a failed SIP Refer transfer is supported using SIP with the Unified CVP, but only on calls where the survivability service is not handling the SIP Refer request.

Intelligent Network Release Trunk Transfers

Customers who use Deployment Model No. 4 (VRU Only with NIC Controlled Routing) rely on call switching methods that do not involve Unified CVP. In that scenario, all switching instructions are exchanged between a Unified ICM Network Interface Controller (NIC) and the PSTN. Examples of these NIC interfaces include Signaling System 7 and Call Routing Service Protocol (CRSP). The NIC is also used as an interface into the Peripheral Gateway in deployments that involve the device. Peripheral Gateway deployments perform Intelligent Network Release Trunk Transfers.

VoiceXML Transfer

VoiceXML call control is supported only in Standalone deployments in which call control is provided by the VXML Server. Deployment Model No. 3b, which also incorporates VXML Server, does not support VoiceXML call control. In Unified ICM integrated deployments, ICM controls all calls.

The VXML Server can invoke the following types of transfers:

Table 1: Types of VoiceXML Transfers

Release Trunk Transfer	VoiceXML Blind Transfer	VoiceXML Bridged Transfer
Result in the incoming call being released from the Ingress Voice Gateway.	Result in the call being bridged to an Egress Voice Gateway or a VoIP endpoint. However, the VXML Server releases all subsequent call control.	Result in the call being bridged to an Egress Voice Gateway or a VoIP endpoint. However, the VXML Server retains call control so that it can return a caller to an IVR application or transfer the caller to another termination point.
Are invoked using the **subdialog_return** element. VXML Server can invoke a TNT transfer, Two B Channel transfer, HookFlash/Wink transfers, and SIP Refer Transfers. For TDM Release Trunk Transfers (TNT, TBCT and Hookflash/Wink), the VoiceXML Gateway must be combined with the Ingress Gateway for the Release Trunk Transfer to work.	Are invoked using the Transfer element in Cisco Unified Call Studio. These transfers transfer the call to any dial peer that is configured in the gateway.	Bridged transfers do not terminate the script. The VXML Server waits until either the ingress or the destination call ends. The script ends only if the ingress call leg hangs up. If the destination call leg hangs up first, the script recovers control and continues with additional self-service activity. Note that the VXML Server port license remains in use for the duration of a bridged transfer, even though the script is not actually performing any processing.

VoiceXML blind transfers differ from VoiceXML bridged transfers in the following ways:

- VoiceXML blind transfers do not support call progress supervision; bridged transfers support it. This means that if a blind transfer fails, VXML Server script does not recover control and cannot attempt a different destination or take remedial action.
- VoiceXML blind transfers cause the VXML Server script to end. Always connect the “done exit” branch from a blind transfer node to a subdialog_return and a hang up node.

Note Cisco VVB supports Blind Transfer only under VoiceXML Transfer.