CDR Field Descriptions

This chapter provides field descriptions for the Call Detail Records (CDRs) in the order in which they appear in the CDR file.

CDR Field Descriptions

The following table describes all fields in the current CDRs in the order in which they appear.

Table 1: CDR Field Descriptions

Field Name	Range of Values	Description
cdrRecordType	0, 1, 2	Defines the type of record. The following valid values apply: • 0—Start call detail record (not used) • 1—End call detail record (CDR) • 2—CMR record Default - For CDRs, this field always remains 1.
globalCallID_callManagerId	Positive Integer	Designates a unique Unified Communications Manager identity. The Global Call ID comprises two fields: globalCallID_callId globalCallID_callManagerId. All records that are associated with a standard call have the same Global Call ID in them. Default - Ensure that this field is populated.

Field Name	Range of Values	Description
globalCallID_callId	Positive Integer	Designates a unique call identity value that is assigned to each call. The system allocates this identifier independently on each call server. Values get chosen sequentially when a call begins. A value gets assigned for each call, successful or unsuccessful. When Unified Communications Manager restarts, it checks the file for the current globalCallID_callId number and assigns the next 1000th number to the next GlobalCallID_callId. The Global Call ID consists of two fields: globalCallID_callId globalCallID_callManagerId. All records that are associated with a standard call have the same Global Call ID in them. **Note** For Unified Communications Manager Release 5.x and later releases, the value in the GlobalCallId CDR field survives over Unified Communications Manager restarts. In Release 4.x and earlier releases, although the GlobalCallId field is time-based, the field gets reused under conditions of heavy traffic. Because of this behavior, problems can occur with customer billing applications and the ability of CAR to correlate CMRs with CDRs and to correlate conference call CDRs. For Release 5.x and later releases, GlobalCallId redesign ensures that the field retains a unique value, at least for a certain number of days. Now, the last used globalCallId_callId value gets written to disk periodically (for every x number of calls). The value gets retrieved after a Unified Communications Manager restart, and the new globalCallId_callId value begins with this number plus x. Default - Ensure that this field is populated.
origLegCallIdentifier	Positive Integer	Identifies the originating leg of a call. Be aware that this value is unique within a cluster. If the leg of a call persists across several subcalls and CDRs (as during a call transfer), this value remains constant. Default - Ensure that this field is populated.
dateTimeOrigination	Integer	Identifies the date and time when the user goes off the hook or the date and time that the H.323 SETUP message is received for an incoming call. The time gets stored as UTC. Default - Ensure that this field is populated.
origNodeId	Positive Integer	Identifies the server, or node within a cluster, to which the originator of the call is registered at the time that the call is made. Default - Ensure that this field is populated.
origSpan	0, Positive Integer	For calls that originate at a gateway, this field indicates the B-channel number of the T1, PRI, or BRI trunk where the call originates, or a zero value for FXS or FXO trunks. For H.323 gateways, the span number remains unknown, and this field contains the call leg ID of the originator. For calls that did not originate at a gateway, the value specifies zero. Default - This field gets populated based on these rules.

Field Name	Range of Values	Description
origIpAddr	Integer	Identifies the v4 IP address of the device that originates the call signaling. For Cisco Unified IP Phones, this field specifies the v4 address of the phone. For PSTN calls, this field specifies the v4 address of the H.323 gateway. For intercluster calls, this field specifies the v4 address of the remote Unified Communications Manager. Default - 0. If the v4 address does not exist for the originating device, this field equals 0. This field gets populated based on these rules.
callingPartyNumber	Text String	Specifies a numeric string of up to 25 characters that indicates the calling party number if the calling party is identified with a directory number. If the calling party uses a blended address in the identity headers, this field contains the directory number portion of the blended address. For calls that originate at a Cisco Unified IP Phone, this field shows the extension number of the line that is used. For incoming H.323 calls, this field specifies the value that is received in the Calling Party Number field in the Setup message. This field reflects any translations that are applied to the Calling Party Number before it arrives at the Unified Communications Manager (such as translations at the gateway). For the server calls, where Unified Communications Manager originates a half call without a calling party, this field may remain empty. CallingPartyNumber could contain a SIP URI. Default - This field gets populated based on these rules.
callingPartyUnicodeLoginUserID	Unicode – UTF_8	Specifies the calling party login user ID. The format of this field specifies UTF_8. Default - Empty string “ ”. If the user ID does not exist, this field stays empty.
origCause_location	0 to 15 For a list of cause code values see Call Termination Cause Codes, on page 25	Specifies the Location field that is indicated in the ISDN release message for clearing causes that are received over ISDN signaling links. See topics that are related to call termination cause codes for a list of the valid values per Q.850. For clearing causes that are created internally by the Unified Communications Manager, this value specifies zero. Default - 0

Field Name	Range of Values	Description
origCause_value	0 to 129 For a list of cause code values see Call Termination Cause Codes, on page 25	Reflects the reason for clearance for the calls that are cleared by the originating party. Unified Communications Managercurrently uses the Q.850 codes and some Unified Communications Managerdefined codes. See topics that are related to call termination cause codes for a listing. For calls that are cleared by the terminating party, this field specifies zero. In addition to the standard values that are described in Q.850, when a call is split by a feature (transfer or conference), the CDR terminates, and this field gets set to 393216. This represents a proprietary value for this field. Default - 0
origPrecedenceLevel	0 to 4	Represents the precedence level of the original leg. For MLPP, each call leg includes a precedence level. • Precedence 0 = FLASH OVERRIDE/ EXECUTIVE OVERRIDE • Precedence 1 = FLASH • Precedence 2 = IMMEDIATE • Precedence 3 = PRIORITY • Precedence 4 = ROUTINE Default - 4
origMediaTransportAddress_IP	0, Integer	Identifies the v4 IP address of the device that originates the media for the call. For Cisco Unified IP Phones, this field specifies the v4 address of the phone. For PSTN calls, this field specifies the v4 address of the H.323 gateway. For intercluster calls, this field specifies the v4 address of the remote phone. Default - 0. If media is not established or the address is not v4, this field equals 0.
origMediaTransportAddress_Port	0, Positive Integer	Identifies the IP port number that is associated with the OrigMediaTransportAddress_IP field. Default - 0. If media is not established, this field stays 0.
origMediaCap_payloadCapability	0, Positive Integer For a full list of codecs, see Codec Types, on page 22	Identifies the codec type that the originator uses to transmit media. Unified Communications Manager currently uses the following payload capability values: 0, 1-16, 18-20, 25, 32, 33, 81-86. See topics related to codec types for a listing of the valid values. Default - 0. If media is not established, this field stays 0.

Field Name	Range of Values	Description
origMediaCap_maxFramesPerPacket	0, Positive Integer	Identifies the number of milliseconds of data per packet that the originating party sends. This field normally gets set to 10, 20, or 30 for G.729 or G.711 codecs, but the field can store any nonzero value. Default - 0. If media is not established, this field stays 0.
origMediaCap_g723BitRate	0	This field is not used in the current release of Unified Communications Manager. Default - This field will remain 0.
origVideoCap_Codec	0, 100 = H.261, 101 = H.263, 103 = H.264	Identifies the codec type that the originator uses to transmit video (H.261, H.263, or H.264.) Default - 0. If media is not established, this field stays 0.
origVideoCap_Bandwidth	0, Positive Integer	Identifies the bandwidth that is measured in units of kbps. Default - 0. If media is not established, this field stays 0.
origVideoCap_Resolution	0, 1 = SQCIF, 2 = QCIF, 3 = CIF, 4 = CIF4, 5 = CIF16 6 = H263 custom resolution 7 = W360P 8 = VGA 9 = W448P 10 = HD720P 11 = HD1080P 12 = CIF2	Indicates the transmitting resolution. In the case of H.264 codec or SIP device, this field refers to the max transmitting resolution the device can transmit for this call. Default - 0. If media is not established, this field stays 0.
origVideoTransportAddress_IP	0, Integer	Identifies the v4 IP address of the device that originates the call. Default - 0. If media is not established or the address is not v4, this field stays 0.
origVideoTransportAddress_Port	0, Positive Integer	Identifies the video RTP port that is associated with the origVideoTransportAddress_IP field. Default - 0. If media is not established, this field stays 0.

Field Name	Range of Values	Description
origRSVPAudioStat	0 to 5	Provides the status of the RSVP audio reservation from originator to terminator. 0 – No reservation. 1 – RSVP Reservation Failure condition at call setup or feature invocation. 2 – RSVP Reservation Success condition at the call setup or feature invocation. 3 – RSVP Reservation No Response (RSVP Agent) condition at the call setup or feature invocation. 4 – RSVP Mid Call Failure Preempted condition (preempted after the call setup). 5 – RSVP Mid Call Failure Lost Bandwidth condition (includes all mid-call failures except MLPP preemption). Default – 0
origRSVPVideoStat	0 to 5	Provides the status of the RSVP video reservation from originator to terminator. 0 – No reservation. 1 – RSVP Reservation Failure condition at call setup or feature invocation. 2 – RSVP Reservation Success condition at call setup or feature invocation. 3 – RSVP Reservation No Response (RSVP Agent) condition at call setup or feature invocation. 4 – RSVP MID Call Failure Preempted condition (preempted after call setup). 5 – RSVP MID Call Failure Lost Bandwidth condition (includes all mid-call failures except MLPP preemption). Default – 0
destLegCallIdentifier	0, Positive Integer	Identifies the terminating leg of a call. This value remains unique within a cluster. If the leg of a call persists across several sub-calls and, consequently, several CDRs (as during a call transfer), this value remains constant. Default - 0. If the destination cannot be reached, this field stays 0.
destNodeId	0, Positive Integer	Identifies the location, or node within a cluster, to which the terminating party of the call is registered at the time that the call is made. Default - 0. If the destination cannot be reached, this field stays 0.

Field Name	Range of Values	Description
destSpan	0, Positive integer	For calls that are received at a gateway, this field indicates the B channel number of the T1, PRI, or BRI trunk where the call is received, or a zero value for FXS or FXO trunks. For H.323 gateways, the span number remains unknown, and this field contains the call leg ID of the destination. For calls not terminating at a gateway, the value specifies zero. Default - 0. If the destination cannot be reached, this field stays 0.
destIpAddr	0, Integer	Identifies the v4 IP address of the device that terminates the call signaling. For Cisco Unified IP Phones, this field specifies the v4 address of the phone. For PSTN calls, this field specifies the v4 address of the H.323 gateway. For intercluster calls, this field specifies the v4 address of the remote Unified Communications Manager. Default - 0. If the destination cannot be reached, this field stays 0. If the v4 address does not exist for this device, the field equals 0.
originalCalledPartyNumber	Text String	Specifies the number to which the original call was presented, prior to any call forwarding. If translation rules are configured, this number reflects the called number after the translations have been applied. If a blended address is used for the called party, this field specifies the directory number portion of the blended address. This field represents a numeric string of up to 48 characters that can be either digits or a SIP URL. Default - Empty string “ ”. If destination cannot be reached, or if the called party number is a directory URI, this field stays empty.
finalCalledPartyNumber	Text String	Specifies the phone number to which the call finally gets presented, until it is answered or rings out. If no forwarding occurs, this number shows the same number as the originalCalledPartyNumber. If the call finally gets presented to a directory URI, the field remains empty. If a blended address is used, this field specifies the directory number portion of the blended address. For calls to a conference bridge, this field contains the actual identifier of the conference bridge, which is an alphanumeric string (for example, b0019901001). This field represents an alphanumeric string that can be either digits or a SIP URL. Default - Empty string “ ”. If destination cannot be reached, this field stays empty.
finalCalledPartyUnicodeLoginUserID	Unicode – UTF_8	Specifies the login user ID. The format of this field specifies UTF_8. Default - Empty string “ ”. If the user ID does not exist, this field stays empty.

Field Name	Range of Values	Description
destCause_location	0 to 15 For a list of cause code values see Call Termination Cause Codes, on page 25	For clearing causes that are received over ISDN signaling links, the ISDN release message indicates this location field. See topics that are related to call termination cause codes for a listing of the valid values per Q.850. For clearing causes that Unified Communications Manager creates internally, this value equals zero. Default - 0. If the destination cannot be reached, this field stays 0.
destCause_value	0 to 129 For a list of cause code values see Call Termination Cause Codes, on page 25	Reflects the reason for the calss that the destination party cleared. See topics that are related to call termination cause codes for a listing of the valid values per Q.850. For calls that the originating party clears, this field stays zero. In addition to the standard values that are described in Q.850, when a call gets split by a feature (transfer or conference), the CDR terminates, and this field gets set to 393216. This represents a proprietary value for this field. Default - 0. If the destination cannot be reached, this field stays 0.
destPrecedenceLevel	0 to 4	Represents the destination legs precedence level. For MLPP, each call leg has a precedence level. • Precedence 0 = FLASH OVERRIDE • Precedence 1 = FLASH • Precedence 2 = IMMEDIATE • Precedence 3 = PRIORITY • Precedence 4 = ROUTINE Default - 4
destMediaTransportAddress_IP	0, Integer	Identifies the v4 IP address of the device that terminates the media for the call. For Cisco Unified IP Phones, this field designates the v4 address of the phone. For PSTN calls, this field designates the v4 address of the H.323 gateway. For intercluster calls, this field shows the v4 address of the remote phone. Default - 0. If the destination cannot be reached or the IP address of the destination is not v4, this field stays 0.
destMediaTransportAddress_Port	0, Positive Integer	Identifies the IP port number that is associated with the DestMediaTransportAddress_IP field. Default - 0. If the destination cannot be reached, this field stays 0.
destMediaCap_payloadCapability	0, Positive Integer For a full list of codecs, see Codec Types, on page 22	Identifies the codec type that the terminating party uses to transmit media. Unified Communications Manager currently uses the following payload capability values: 0, 1-16, 18-20, 25, 32, 33, 81-86. See topics related to codec types for a listing of the valid values. Default - 0. If the destination cannot be reached, this field stays 0.

Field Name	Range of Values	Description
destMediaCap_maxFramesPerPacket	0, Positive Integer	Identifies the number of milliseconds of data per packet that the terminating party of the call sends. This field normally gets set to 10, 20, or 30 for G.729 or G.711 codecs but can store any nonzero value. This field can specify zero if the media is never established. Default - 0. If the destination cannot be reached, this field stays 0.
destMediaCap_g723BitRate	0	This field is not used in the current release of Unified Communications Manager. Default - This field stays 0.
destVideoCap_Codec	0, 100 = H.261, 101 = H.263, 103 = H.264	Identifies the codec type that the terminating party uses to transmit video (H.261, H.263, or H.264). Default - 0. If the destination cannot be reached, this field stays 0.
destVideoCap_Bandwidth	0, Positive Integer	Identifies the bandwidth, and is measured in units of kbps. Default - 0. If the destination cannot be reached, this field stays 0.
destVideoCap_Resolution	0, 1 = SQCIF, 2 = QCIF, 3 = CIF, 4 = CIF4, 5 = CIF16 6 = H263 custom resolution 7 = W360P 8 = VGA 9 = W448P 10 = HD720P 11 = HD1080P 12 = CIF2	Indicates the transmitting resolution. In the case of H.264 codec or SIP device, this field refers to the max transmitting resolution the device can transmit for this call. Default - 0. If media is not established, this field stays 0.
destVideoTransportAddress _IP	0, Integer	Identifies the v4 IP address of the device that receives the call. Default - 0. If the destination cannot be reached or the IP address of the destination is not v4, this field stays 0.
destVideoTransportAddress_Port	0, Positive Integer	Identifies the video RTP port that is associated with the destVideoTransportAddress_IP field. Default - 0. If the destination cannot be reached, this field stays 0.

Field Name	Range of Values	Description
destRSVPAudioStat	0 - 5	Designates the status of the RSVP audio reservation from terminator to originator. 0 – No reservation. 1 – RSVP Reservation Failure condition at the call setup or feature invocation. 2 – RSVP Reservation Success condition at call setup or feature invocation. 3 – RSVP Reservation No Response (RSVP Agent) condition at call setup or feature invocation. 4 – RSVP Mid Call Failure Preempted condition (preempted after call setup). 5 – RSVP Mid Call Failure Lost Bandwidth condition (includes all mid call failures except MLPP preemption). Default – 0
destRSVPVideoStat	0 - 5	Designates the status of the RSVP video reservation from terminator to originator. 0 – No reservation. 1 – RSVP Reservation Failure condition at call setup or feature invocation. 2 – RSVP Reservation Success condition at call setup or feature invocation. 3 – RSVP Reservation No Response (RSVP Agent) condition at call setup or feature invocation. 4 – RSVP Mid Call Failure Preempted condition (preempted after call setup). 5 – RSVP Mid Call Failure Lost Bandwidth condition (includes all mid call failures except MLPP preemption). Default – 0
dateTimeConnect	0, Integer	Identifies the date and time that the call connects. The time gets stored as UTC. If the call is never answered, this value shows zero. Default - 0. If the call is never connected, this field stays 0.
dateTimeDisconnect	Integer	Identifies the date and time when the call is cleared. This field gets set even if the call never connects. The time gets stored as UTC. Default - Ensure that this field is populated.

Field Name	Range of Values	Description
lastRedirectDn	Text String	Specifies a numeric string of up to 25 characters. The numeric string can contain digits or a SIP URL. For forwarded calls, this field specifies the phone number of the next to last hop before the call reaches its final destination. If only one hop occurs, this number matches the OriginalCalledPartyNumber. If a blended address is used for call addressing, this field contains only the directory number portion of the blended address. For calls that are not forwarded, this field matches the OriginalCalledPartyNumber and the FinalCalledPartyNumber. For calls to a conference bridge, this field contains the actual identifier of the conference bridge, which is an alphanumeric string (for example, b0019901001). Default - Empty string " ". If the call is never redirected, or if the next to last hop address is a directory URI, this field remains empty.
pkid	Text String	Identifies a text string that the database uses internally to uniquely identify each row. This text string provides no meaning to the call itself. Default - A unique ID should always populate this field.
originalCalledPartyNumberPartition	Text String	Identifies unique partition name that is associated with the OriginalCalledPartyNumber field because Unified Communications Manager supports multiple Cisco Unified IP Phones with the same extension number in different partitions. For calls that egress through an H.323 gateway, this field uniquely specifies the partition name that is associated with the route pattern that points to the gateway. Default - Empty string " ". If the original called party does not have a partition, this field remains empty.
callingPartyNumberPartition	Text String	Identifies unique partition name that is associated with the CallingPartyNumber field because Unified Communications Manager supports multiple Cisco Unified IP Phones with the same extension number in different partitions. For calls that ingress through an H.323 gateway, this field remains blank. Default - Empty string " ". If the original called party does not have a partition, this field remains empty.
finalCalledPartyNumberPartition	Text String	Identifies unique partition name that is associated with the FinalCalledPartyNumber field because Unified Communications Manager supports multiple Cisco Unified IP Phones with the same extension number in different partitions. For calls that egress through an H.323 gateway, this field uniquely specifies the partition name that is associated with the route pattern that points to the gateway. Default - Empty string " ". If the final called party does not have a partition, this field remains empty.

Field Name	Range of Values	Description
lastRedirectDnPartition	Text String	Identifies unique partition name that is associated with the LastRedirectDn field because Unified Communications Manager supports multiple Cisco Unified IP Phones with the same extension number in different partitions. For calls that egress through an H.323 gateway, this field specifies the partition name that is associated with the route pattern that points to the gateway. Default - Empty string " ". If the last redirecting Party does not have a partition or the call was never redirected, this field stays empty.
duration	0, Positive integer	Identifies the difference between the Connect Time and Disconnect Time. This field specifies the time that the call remains connected, in seconds. This field remains zero if the call never connects or if it connects for less than 1 second. Default - 0
origDeviceName	Text String	Specifies the text string that identifies the name of the originating device. Default - Ensure that this field is populated.
destDeviceName	Text String	Specifies the text string that identifies the name of the destination device. Default - Empty string" ". If the original device does not have a name, this field stays empty.
origCallTerminationOnBehalfOf	0, Positive Integer For a complete list of OnBehalfOf fields, see OnBehalfof Codes, on page 33	Specifies code that identifies why the originator was terminated. For example, if the originator of the call hangs up the phone, the OnBehalfOf code shows "12" for Device. If the call terminates because of a transfer, the OnBehalfOf code shows "10" for Transfer. See topics related to CDR field descriptions for a list of the codes. This release added new OnBehalfOf codes. Default - 0
destCallTerminationOnBehalfOf	0, Positive Integer For a complete list of OnBehalfOf fields, see OnBehalfof Codes, on page 33	Specifies code that identifies why the destination was terminated. For example, if the destination of the call hangs up the phone, the OnBehalfOf code shows "12" for Device. If the call terminates because of a transfer, the OnBehalfOf code shows "10" for Transfer. See topics related to CDR field descriptions for a list of the codes. This release added new OnBehalfOf codes. Default - 0
origCalledPartyRedirectOnBehalfOf	0, Positive Integer For a complete list of OnBehalfOf fields, see OnBehalfof Codes, on page 33	Specifies code that identifies the reason for redirection of the original called party. For example, if the original called party was redirected because of a conference, the OnBehalfOf code specifies "4." See topics related to CDR field descriptions for a list of the codes. This release added new OnBehalfOf codes. Default - 0

Field Name	Range of Values	Description
lastRedirectRedirectOnBehalfOf	0, Integer For a complete list of OnBehalfOf fields, see OnBehalfof Codes, on page 33	Specifies code that identifies the reason for redirection of the last redirected party. For example, if the last redirected party was redirected on behalf of a conference, the OnBehalfOf code specifies "4." See topics related to CDR field descriptions for a list of the codes. This release added new OnBehalfOf codes. Default - 0
origCalledPartyRedirectReason	0, Integer For a complete list of OnBehalfOf fields, see Redirect Reason Codes, on page 31	Identifies the reason for a redirect of the original called party. See topics related to redirect reason codes for a complete list of the codes. Default - 0
lastRedirectRedirectReason	0, Integer For a complete list of OnBehalfOf fields, see Redirect Reason Codes, on page 31	Identifies the last redirect reason for redirection. See topics related to redirect reason codes for a complete list of the codes. Default - 0
destConversationID	0, Integer	Specifies a unique identifier that is used to identify the parties of a conference call. For conference chaining scenarios, the origConversationID and destConversationID fields identify which conferences are chained together. Default - 0
globalCallId_ClusterId	Text String	Specifies a unique ID that identifies a cluster of Unified Communications Managers. The field is generated at installation and is not used by Unified Communications Manager. The fields globalCallId_ClusterId + globalCallId_CMId + globalCallId_CallId make up this unique key. Default - This field should always be populated.
joinOnBehalfOf	0, Integer For a complete list of OnBehalfOf fields, see OnBehalfof Codes, on page 33	Specifies code that identifies the reason for a join. For example, if the join takes place on behalf of a transfer, the OnBehalfOf code specifies "10." See topics related to CDR field descriptions for a list of the codes. Default - 0
comment	Text String	Allows features to add text to the CDRs. This text can describe details about the call. For example, the following field flags malicious calls: Tag—CallFlag Value—MALICIOUS Default - Empty string " ".

Field Name	Range of Values	Description
authCodeDescription	Text String	Provides a description of the FAC. Default - Empty string " " or null.
authorizationLevel	0, Integer	Displays the level of the FAC. Default - 0
clientMatterCode	Text String	Displays the client matter code. Before the system extends a call, the user enters a client matter code that can be used for assigning account or billing codes to calls. Default - Empty string " " or null.
origDTMFMethod	0, Positive Integer	Displays the DTMF method that the originator uses. 0 - No DTMF - Use ANY matched DTMF. 1 - OOB - Use OOB if endpoints behind SIPTrunk support it. 2 - 2833 - Use RFC2833 if endpoints behind SIPTrunk support it. 3 - OOB and 2833 - Use both KPML and RFC2833 if endpoints behind SIPTrunk can support both. 4 - Unknown Default - 0 (No preference)
destDTMFMethod	0, Positive Integer	Displays the DTMF method that the destination uses. 0 - No DTMF - Use ANY matched DTMF.1 - OOB - Use OOB if endpoints behind SIPTrunk support it.2 - 2833 - Use RFC2833 if endpoints behind SIPTrunk support it.3 - OOB and 2833 - Use both KPML and RFC2833 if endpoints behind SIPTrunk can support both.4 - Unknown. Default - 0 (No preference)
callSecuredStatus	0, Positive Integer	Displays the highest security status that is reached during a call. For example, if the call is originally unsecured, and later the call changes to secured, the CDR contains 1 for "Secured" even though different portions of the call have different status values. 0 - Non-secured 1 - Authenticated (not encrypted) 2 - Secured (encrypted) Default - 0 (Non-secured)
origConversationID	Integer	Identifies the conference ID that is associated with the originating leg of the call. In most cases, this field equals 0. For conference chaining scenarios, the origConversationID and destConversationID fields identify which conferences are chained together. Default - 0
origMediaCap_Bandwidth	0, Positive Integer	Displays the media bandwidth that is used at the origination of the call. Default - 0

Field Name	Range of Values	Description
destMediaCap_Bandwidth	0, Positive Integer	Displays the media bandwidth that is used at the destination of the call. Default - 0
authorizationCodeValue	Text String	Displays the Forced Authorization Code (FAC) that is associated with the call. Default - Empty string “ ” or null.
outpulsedCallingPartyNumber	Text String	Comprises an alphanumeric string of up to 50 characters. The calling party number gets outpulsed from the device. This field gets populated only when normalization or localization takes place at the device. Default - Empty string “ ” or null.
outpulsedCalledPartyNumber	Text String	Comprises an alphanumeric string of up to 50 characters. The called party number gets outpulsed from the device. This field gets populated only when normalization or localization takes place at the device. Default - Empty string “ ”or null.
origIpv4v6Addr	Text string	Comprises an alphanumeric string of up to 64 characters. This field identifies the IP address of the device that originates the call signalling. The field can be either IPv4 or IPv6 format depending on the type of IP address that gets used for the call. For Cisco Unified IP Phones, this field is the address of the Cisco Unified IP Phone. For PSTN calls, this field is the address of the gateway. For intercluster calls, this field is the address of the remote Unified Communications Manager. The IP address is either in dotted decimal format or in colon separated hexadecimal format. Default - The IP address of the originating device as reported by the device or used for the call after media negotiation.
destIpv4v6Addr	Text string	Comprises an alphanumeric string of up to 64 characters. This field identifies the IP address of the device that terminates the call signalling. The field can be either in IPv4 or IPv6 format depending upon the type of IP address that gets used for the call. For Cisco Unified IP Phones, this field is the address of the Cisco Unified IP Phone. For PSTN calls, this field is the address of the gateway. For intercluster calls, this field is the address of the remote Unified Communications Manager. The IP address is either in dotted decimal format or in colon separated hexadecimal format. Default - Empty String “ ” or null. If the destination does not get reached, this field stays empty.

Field Name	Range of Values	Description
origVideoCap_Codec_Channel2	0, 100 = H.261, 101 = H.263, 103 = H.264,	Identifies the codec type that the originator uses to transmit video (H.261, H.263, or H.264) for the second video channel. Default - 0. If media does not get established, this field displays 0. Also, if H.239 and BFCP are not supported for this call, this field displays 0.
origVideoCap_Bandwidth_Channel2	0, Positive integer	Identifies the bandwidth measured in units of kbps for the second video channel. Default - 0. If media does not get established, this field displays 0. Also, if H.239 and BFCP are not supported for this call, this field displays 0.
origVideoCap_Resolution_Channel2	0, 1 = SQCIF, 2 = QCIF, 3 = CIF, 4 = CIF4, 5 = CIF16 6 = H263 custom resolution 7 = W360P 8 = VGA 9 = W448P 10 = HD720P 11 = HD1080P 12 = CIF2	Indicates the transmitting resolution for the second video channel. In the case of H.264 codec or SIP device, this field refers to the maximum transmitting resolution the device can transmit for this call. Default - 0. If media is not established, this field stays 0. Also, if H.239 and BFCP are not supported for this call, this field displays 0.
origVideoTransportAddress_IP_Channel2	0, Integer	Identifies the v4 IP address of the device that originates the call for the second video channel. Default - 0. If media does not get established, this field displays 0. Also, if H.239 and BFCP are not supported for this call, this field displays 0.
origVideoTransportAddress_Port_Channel2	0, Positive integer	Identifies the video RTP port associated with the origH239VideoTransportAddress_IP field for the second video channel. Default - 0. If media does not get established, this field displays 0. Also, if H.239 and BFCP are not supported for this call, this field displays 0.
origVideoChannel_Role_Channel2	0 = Presentation role, 1 = Live role, Positive integer	Identifies the H.239 video channel role of the device that originates. Default - 0. If media does not get established, this field displays 0. Also, if H.239 is not supported, this field displays 0.
destVideoCap_Codec_Channel2	0, 100 = H.261, 101 = H.263, 103 = H.264	Identifies the codec type that the terminating party uses to transmit video (H.261, H.263, or H.264) for the second video channel. Default - 0. If the destination cannot be reached, this field stays 0. Also, if H.239 and BFCP are not supported for this call, this field displays 0.

Field Name	Range of Values	Description
destVideoCap_Bandwidth_Channel2	0, Positive integer	Identifies the bandwidth measured in units of kbps for the second video channel. Default - 0. If media does not get established, this field displays 0. Also, if H.239 and BFCP are not supported for this call, this field displays 0.
destVideoCap_Resolution_Channel2	0, 1 = SQCIF, 2 = QCIF, 3 = CIF, 4 = CIF4, 5 = CIF16 6 = H263 custom resolution 7 = W360P 8 = VGA 9 = W448P 10 = HD720P 11 = HD1080P 12 = CIF2	Indicates the transmitting resolution for the second video channel. In the case of H.264 codec or SIP device, this field refers to the maximum transmitting resolution the device can transmit for this call. Default - 0. If media is not established, this field stays 0. Also, if H.239 and BFCP are not supported for this call, this field displays 0.
destVideoTransportAddress_IP_Channel2	0, Integer	Identifies the v4 IP address of the device that receives the call. Default - 0. If media does not get established, this field displays 0. Also, if H.239 and BFCP are not supported for this call, this field displays 0.
destVideoTransportAddress_Port_Channel2	0, Positive integer	Identifies the video RTP port associated with the destH239VideoTransportAddress_IP field. Default - 0. If media does not get established, this field displays 0. Also, if H.239 and BFCP are not supported for this call, this field displays 0.
destVideoChannel_Role_Channel2	0 = Presentation role, 1 = Live role, Positive integer	Identifies the H.239 video channel role of the device that receives the call. Default - 0. If media does not get established, this field displays 0. Also, if H.239 is not supported, this field displays 0.
incomingProtocolID	0 = Unknown, 1 = SIP, 2 = H323, 3 = CTI/JTAPI, 4 = Q931, Integer	Identifies the protocol (SIP, H.323, CTI/JTAPI, or Q.931) used between Cisco Unified CM and the upstream voice product in the call path.
incomingProtocolCallRef	Varchar(32)	Identifies the globally unique call reference identification for the protocol. The value is received from the upstream voice product. The value is alpha–numeric and truncated to 32 characters.

Field Name	Range of Values	Description
outgoingProtocolID	0 = Unknown, 1 = SIP, 2 = H323, 3 = CTI/JTAPI, 4 = Q931, Integer	Identifies the protocol (SIP, H.323, CTI/JTAPI, or Q.931) used between Cisco Unified CM and the downstream voice product in the call path.
outgoingProtocolCallRef	Varchar(32)	Identifies the globally unique call reference identification for the protocol. The value is passed to the next downstream voiced product. The value is alpha–numeric and truncated to 32 characters.
currentRoutingReason	Positive Integer For field values see Routing Reason Values for External Call Control, on page 21	Displays the reason why the call was intercepted for the active call. This field is used with the external call control feature. See topics related to routing reason values for external call control for a list of reasons. Default value is 0.
origRoutingReason	Positive Integer For a complete list of OnBehalfOf fields, see OnBehalfof Codes, on page 33	Displays the reason why the call was intercepted for the first time. This field is used with the external call control feature, See topics related to routing reason values for external call control for a list of reasons. Default value is 0.
lastRedirectingRoutingReason	Positive Integer For a complete list of OnBehalfOf fields, see OnBehalfof Codes, on page 33	Displays why the call was intercepted for the last time. This field is used with the external call control feature. See topics related to routing reason values for external call control for a list of reasons. Default - Empty string.
huntPilotPartition	Text String	Indicates the partition for the hunt pilot DN. Default - Empty string.
huntPilotDN	Text String	Indicates the hunt pilot DN through which the call is routed. Default - Empty string.
calledPartyPatternUsage	Positive Integer	Indicates the pattern of the called party. Default value specifies 5 (PATTERN_ROUTE). • If the huntPilotDN is populated, use the huntPilotDN field value as the hunt pilot. • If the huntPilotDN is not available, check the pattern usage (7 =PATTERN_HUNT_PILOT) in the CDR table to identify the call type. If this call is a hunt list call, use the finalCalledPartyNumber as the huntPilotDN.

Field Name	Range of Values	Description
incomingICID	Text String	Specifies alphanumeric string up to 50 characters. This field is populated with the IMS Identifier(ICID) from the P-Charging Vector at the incoming call leg of the call. This field will be empty when the call leg has no IMS or SIP trunk with P-Charging-Vector enabled. Default = Empty String " "
incomingOrigIOI	Text String	Specifies alphanumeric string up to 50 characters. This field is populated with the originating Interoperator Identifier(IOI) from the P-Charging Vector at the incoming call leg of the call. This field will be empty when the call leg has no IMS or SIP trunk with P-Charging-Vector enabled. Default = Empty String " "
incomingTermIOI	Text String	Specifies alphanumeric string up to 50 characters. This field is populated with the terminating Interoperator Identifier(IOI) from the P-Charging Vector at the incoming call leg of the call. This field will be empty when the call leg has no IMS or SIP trunk with P-Charging-Vector enabled. Default = Empty String " "
outgoingICID	Text String	Specifies alphanumeric string up to 50 characters. This field is populated with the IMS Identifier(ICID) from the P-Charging Vector at the outgoing call leg of the call. This field will be empty when the call leg has no IMS or SIP trunk with P-Charging-Vector enabled. Default = Empty String " "
outgoingOrigIOI	Text String	Specifies alphanumeric string up to 50 characters. This field is populated with the originating Interoperator Identifier(IOI) from the P-Charging Vector at the outgoing call leg of the call. This field will be empty when the call leg has no IMS or SIP trunk with P-Charging-Vector enabled. Default = Empty String " "
outgoingTermIOI	Text String	Specifies alphanumeric string up to 50 characters. This field is populated with the terminating Interoperator Identifier(IOI) from the P-Charging Vector at the outgoing call leg of the call. This field will be empty when the call leg has no IMS or SIP trunk with P-Charging-Vector enabled. Default = Empty String " "

Field Name	Range of Values	Description
outpulsedOriginalCalledPartyNumber	Text String	Specifies alphanumeric string up to 50 characters. The Original called party number outpulsed from the device. Refer to section on *Redirecting Number Transformation* for details. Default = Empty String " "
outpulsedLastRedirectingNumber	Text String	Specifies alphanumeric string up to 50 characters. The Last Redirecting number outpulsed from the device. Refer to section on *Redirecting Number Transformation* for details. Default = Empty String " "
wasCallQueued	Positive Integer	Specifies whether the call has been put into a queue or not. A value of 0 means that the call is not put into any queue; 1 means the call has been put into a queue.
totalWaitTimeInQueue	Positive Integer	Specifies how long a caller has been put into a queue. The value is specified in second. The value is 0 if the call is never put into any queue.
callingPartyNumber_uri	Text String	Specifies an alphanumeric string of up to 254 characters that identifies the calling party if the calling party uses a directory URI for call addressing. If the calling party uses a blended address in the identity headers, this field contains the directory URI portion of the blended address. Default - Empty string “ ”. If the calling party does not use a directory URI, the field stays empty.
originalCalledPartyNumber_uri	Text String	Specifies a string of up to 254 alphanumeric characters that specifies the directory URI to which the original call was addressed, prior to any call forwarding, provided the call was addressed to a directory URI. If a blended address is used for the called party, this field specifies the directory URI portion of the blended address. Default - Empty string “ ”. If destination cannot be reached, or if the called party is a directory number, this field stays empty.
finalCalledPartyNumber_uri	Text String	Specifies an alphanumeric string of up to 254 characters that indicate the directory URI address to which the call finally gets presented, if the final address is a directory URI. If no forwarding occurs, this field shows the same directory URI as the originalCalledPartyNumber_uri field. If a blended address is used for the called number, this field specifies the directory URI portion of the blended address. For calls to a conference bridge, this field contains the actual identifier of the conference bridge, which is an alphanumeric string (for example, b0019901001). Default - Empty string “ ”. If destination cannot be reached, or if a directory number is used for called addressing, this field stays empty.

Field Name	Range of Values	Description
lastRedirectDn_uri	Text String	Specifies an alphanumeric string of up to 254 characters. For forwarded calls that use a directory URI for addressing, this field specifies the directory URI of the next to last hop before the call reaches its final destination. If only one hop occurs, this number matches the originalCalledPartyNumber_uri. If a blended address is used, this field contains only the directory URI portion of the blended address. For calls that are not forwarded, this field matches the originalCalledPartyNumber_uri and the finalCalledPartyNumber_uri. For calls to a conference bridge, this field contains the actual identifier of the conference bridge, which is an alphanumeric string (for example, b0019901001). Default - Empty string “ ”. If the call is never redirected, or if the address is a directory number, this field remains empty.

Routing Reason Values for External Call Control

Unified Communications Manager supports the external call control feature, which enables an adjunct route server to make call-routing decisions for Unified Communications Manager by using the Cisco Unified Routing Rules Interface. When you configure external call control, Unified Communications Manager issues a route request that contains the calling party and called party information to the adjunct route server. The adjunct route server receives the request, applies appropriate business logic, and returns a route response that instructs Unified Communications Manager on how the call should get routed, along with any additional call treatment that should get applied.

The adjunct route server can instruct Unified Communications Manager to allow, divert, or deny the call, modify calling and called party information, play announcements to callers, reset call history so adjunct voicemail and IVR servers can properly interpret calling/called party information, and log reason codes that indicate why calls were diverted or denied.

The following table includes the reasons that can display for the currentRoutingReason, origRoutingReason, or lastRedirectingRoutingReason fields.

Table 2: Routing Reason Values for External Call Control

Field Value	Reason	Description
0	PDPDecision_NONE	This value indicates that the route server did not return a routing directive to the Unified Communications Manager.
1	PDPDecision_Allow_Fulfilled	This value indicates that Unified Communications Manager allowed a call.

Field Value	Reason	Description
2	PDPDecision_Allow_Unfulfilled	This value indicates that Unified Communications Manager disallowed a call.
3	PDPDecision_Divert_Fulfilled	This value indicates that Unified Communications Manager diverted the call.
4	PDPDecision_Divert_Unfulfilled	This value indicates that Unified Communications Manager was not able to divert the call.
5	PDPDecision_Forward_Fulfilled	This value indicates that Unified Communications Manager forwarded the call.
6	PDPDecision_Forward_Unfulfilled	This value indicates that Unified Communications Manager was unable to forward the call.
7	PDPDecision_Reject_Fulfilled	This value indicates that Unified Communications Manager rejected the call.
8	PDPDecision_Reject_Unfulfilled	This value indicates that Unified Communications Manager was not able to reject the call.

Cisco Call Detail Records Codes

This chapter section information about the codec types and codes that are used in the Call Detail Record fields.

Codec Types

The following table contains the compression and payload types that may appear in the codec fields.

Table 3: Codec Types

Value	Description
1	NonStandard
2	G711Alaw 64k
3	G711Alaw 56k
4	G711mu-law 64k
5	G711mu-law 56k

Value	Description
6	G722 64k
7	G722 56k
8	G722 48k
9	G7231
10	G728
11	G729
12	G729AnnexA
13	Is11172AudioCap
14	Is13818AudioCap
15	G.729AnnexB
16	G.729 Annex AwAnnexB
18	GSM Full Rate
19	GSM Half Rate
20	GSM Enhanced Full Rate
25	Wideband 256K
32	Data 64k
33	Data 56k
40	G7221 32K
41	G7221 24K
42	AAC
43	MP4ALATM_128
44	MP4ALATM_64
45	MP4ALATM_56
46	MP4ALATM_48
47	MP4ALATM_32
48	MP4ALATM_24
49	MP4ALATM_NA
80-	GSM

Value	Description
81	ActiveVoice
82	G726 32K
83	G726 24K
84	G726 16K
86	iLBC
89	iSAC
90	OPUS
100	H261
101	H263
102	Vieo
103	H264
104	H264_SVC
105	T120
106	H224
107	T38Fax
109	H265
110	H264_UC
111	XV150_MR_711U
112	NSE_VBD_711U
113	XV150_MR_729A
114	NSE_VBD_729A
115	H264_FEC
120	Clear_Chan
222	Universal_Xcoder
257	RFC2833_DynPayload
258	PassThrough
259	Dynamic_Payload_PassThru
260	DTMF_OOB

Value	Description
261	Inband_DTMF_RFC2833
299	NoAudio
302	v150_LC_SSE

Call Termination Cause Codes

The following tables contain call termination cause codes that may appear in the Cause fields in CDRs.

Note Cause Code is defined in call control as Natural number. It is a 32 bit unsigned (long) positive integer with values ranging from 0 to +4,294,967,295.

Table 4: Call Termination Cause Codes

Code	Description
0	No error
1	Unallocated (unassigned) number
2	No route to specified transit network (national use)
3	No route to destination
4	Send special information tone
5	Misdialed trunk prefix (national use)
6	Channel unacceptable
7	Call awarded and being delivered in an established channel
8	Preemption
9	Preemption—circuit reserved for reuse
16	Normal call clearing
17	User busy
18	No user responding
19	No answer from user (If "No Answer Ring duration" value is greater than the T301 Timer value and after T301 Timer expiry, Call Forwarding No Answer(CFNA) Feature would be invoked).

Code	Description
20	Subscriber absent
21	Call rejected
22	Number changed
25	Natural Exchange Routing Error
26	Non-selected user clearing
27	Destination out of order
28	Invalid number format (address incomplete)
29	Facility rejected
30	Response to STATUS ENQUIRY
31	Normal, unspecified
34	No circuit/channel available
38	Network out of order
39	Permanent frame mode connection out of service
40	Permanent frame mode connection operational
41	Temporary failure
42	Switching equipment congestion
43	Access information discarded
44	Requested circuit/channel not available
46	Precedence call blocked
47	Resource unavailable, unspecified
49	Quality of Service not available
50	Requested facility not subscribed
53	Service operation violated
54	Incoming calls barred
55	Incoming calls barred within Closed User Group (CUG)
57	Bearer capability not authorized
58	Bearer capability not presently available

Code	Description
62	Inconsistency in designated outgoing access information and subscriber class
63	Service or option not available, unspecified
65	Bearer capability not implemented
66	Channel type not implemented
69	Requested facility not implemented
70	Only restricted digital information bearer capability is available (national use)
79	Service or option that is not implemented, unspecified
81	Invalid call reference value
82	Identified channel does not exist
83	A suspended call exists, but this call identity does not
84	Call identity in use
85	No call suspended
86	Call having the requested call identity has been cleared
87	User not member of CUG (Closed User Group)
88	Incompatible destination
90	Destination number missing and DC not subscribed
91	Invalid transit network selection (national use)
95	Invalid message, unspecified
96	Mandatory information element is missing
97	Message type nonexistent or not implemented
98	Message is not compatible with the call state, or the message type is nonexistent or not implemented
99	An information element or parameter does not exist or is not implemented
100	Invalid information element contents
101	The message is not compatible with the call state
102	Call terminated when timer expired; a recovery routine that is executed to recover from the error

Code	Description
103	Parameter nonexistent or not implemented - passed on (national use)
110	Message with unrecognized parameter discarded
111	Protocol error, unspecified
122	Precedence Level Exceeded
123	Device not Preemptable
125	Out of bandwidth (Cisco specific)
126	Call split (Cisco specific)
127	Interworking, unspecified
129	Precedence out of bandwidth
130	Natural Isolated Code
131	Call Control Discovery PSTN Failover (Cisco specific)
132	IME QOS Fallback (Cisco specific)
133	PSTN Fallback locate Call Error (Cisco specific)
134	PSTN Fallback wait for DTMF Timeout (Cisco specific)
135	IME Failed Connection Timed out (Cisco specific)
136	IME Failed not enrolled (Cisco specific)
137	IME Failed socket error (Cisco specific)
138	IME Failed domain blocked (Cisco specific)
139	IME Failed prefix blocked (Cisco specific)
140	IME Failed expired ticket (Cisco specific)
141	IME Failed remote no matching route (Cisco specific)
142	IME Failed remote unregistered (Cisco specific)
143	IME Failed remote IME disabled (Cisco specific)
144	IME Failed remote invalid IME trunk URI (Cisco specific)
145	IME Failed remote URI not E164 (Cisco specific)

Code	Description
146	IME Failed remote called number not available (Cisco specific)
147	IME Failed Invalid Ticket (Cisco specific)
148	IME Failed unknown (Cisco specific)
155	DCC Allowed Percentage Exceeded

Table 5: Cisco-Specific Call Termination Cause Codes

Decimal Value Code	Hex Value Code	Description
262144	0x40000	Conference Full (was 124)
393216	0x60000	Call split (was 126) This code applies when a call terminates during a transfer operation because it was split off and terminated (was not part of the final transferred call). This code can help you to determine which calls terminated as part of a feature operation.
458752	0x70000	Conference drop any party/Conference drop last party (was 128)
16777257	0x1000029	CCM_SIP_400_BAD_REQUEST
33554453	0x2000015	CCM_SIP_401_UNAUTHORIZED
50331669	0x3000015	CCM_SIP_402_PAYMENT_REQUIRED
67108885	0x4000015	CCM_SIP_403_FORBIDDEN
83886081	0x5000001	CCM_SIP_404_NOT_FOUND
100663359	0x600003F	CCM_SIP_405_METHOD_NOT_ALLOWED
117440591	0x700004F	CCM_SIP_406_NOT_ACCEPTABLE
134217749	0x8000015	CCM_SIP_407_PROXY_AUTHENTICATION_REQUIRED
150995046	0x9000066	CCM_SIP_408_REQUEST_TIMEOUT
184549398	0xB000016	CCM_SIP__410_GONE
201326719	0xC00007F	CCM_SIP_411_LENGTH_REQUIRED
234881151	0xE00007F	CCM_SIP_413_REQUEST_ENTITY_TOO_LONG
251658367	0xF00007F	CCM_SIP_414_REQUEST_URI_TOO_LONG
268435535	0x1000004F	CCM_SIP_415_UNSUPPORTED_MEDIA_TYPE
285212799	0x1100007F	CCM_SIP_416_UNSUPPORTED_URI_SCHEME

Decimal Value Code	Hex Value Code	Description
83886207	0x1500007F	CCM_SIP_420_BAD_EXTENSION
369098879	0x1600007F	CCM_SIP_421_EXTENSION_REQUIRED
402653311	0x1800007F	CCM_SIP_423_INTERVAL_TOO_BRIEF
419430421	0x19000015	CCM_SIP_424_BAD_LOCATION_INFO
503316501	0x1E000015	CCM_SIP_429_PROVIDE_REFER_IDENTITY
1073741842	0x40000012	CCM_SIP_480_TEMPORARILY_UNAVAILABLE
1090519081	0x41000029	CCM_SIP_481_CALL_LEG_DOES_NOT_EXIST
1107296281	0x42000019	CCM_SIP_482_LOOP_DETECTED = 0x42000000 + EXCHANGE_ROUTING_ERROR
1124073497	0x43000019	CCM_SIP_483_TOO_MANY_HOOPS
1140850716	0x4400001C	CCM_SIP_484_ADDRESS_INCOMPLETE
1157627905	0x45000001	CCM_SIP_485_AMBIGUOUS
1174405137	0x46000011	CCM_SIP_486_BUSY_HERE
1191182367	0x4700001F	CCM_SIP_487_REQUEST_TERMINATED
1207959583	0x4800001F	CCM_SIP_488_NOT_ACCEPTABLE_HERE
1258291217	0x4B000011	CCM_SIP_491_REQUEST_PENDING
1291845649	0x4D000011	CCM_SIP_493_UNDECIPHERABLE
1409286185	0x54000029	CCM_SIP_500_SERVER_INTERNAL_ERROR
1442840614	0x56000026	CCM_SIP_502_BAD_GATEWAY
1459617833	0x57000029	CCM_SIP_503_SERVICE_UNAVAILABLE
2801795135	0xA700003F	CCM_SIP_503_SERVICE_UNAVAILABLE_SER_OPTION_NOAV
1476395110	0x58000066	CCM_SIP__504_SERVER_TIME_OUT
1493172351	0x5900007F	CCM_SIP_505_SIP_VERSION_NOT_SUPPORTED
1509949567	0x5A00007F	CCM_SIP_513_MESSAGE_TOO_LARGE
2701131793	0xA1000011	CCM_SIP_600_BUSY_EVERYWHERE
2717909013	0xA2000015	CCM_SIP_603_DECLINE
2734686209	0xA3000001	CCM_SIP_604_DOES_NOT_EXIST_ANYWHERE
2751463455	0xA400001F	CCM_SIP_606_NOT_ACCEPTABLE

Redirect Reason Codes

The following table contains the available Redirect Reason Codes that may appear in a record.

Q.931 Standard Redirect Reason Codes	
Value	**Description**
0	Unknown
1	Call Forward Busy
2	Call Forward No Answer
4	Call Transfer
5	Call Pickup
7	Call Park
8	Call Park Pickup
9	CPE Out of Order
10	Call Forward
11	Call Park Reversion
15	Call Forward all
Nonstandard Redirect Reason Codes	
18	Call Deflection
34	Blind Transfer
50	Call Immediate Divert
66	Call Forward Alternate Party
82	Call Forward On Failure
98	Conference
114	Barge
129	Aar
130	Refer
146	Replaces
162	Redirection (3xx)
177	SIP-forward busy greeting
178	Call Forward Unregistered

Q.931 Standard Redirect Reason Codes	
Value	**Description**
207	Follow Me (SIP-forward all greeting)
209	Out of Service (SIP-forward busy greeting)
239	Time of Day (SIP-forward all greeting)
242	Do Not Disturb (SIP-forward no answer greeting)
257	Unavailable (SIP-forward busy greeting)
274	Away (SIP-forward no answer greeting)
303	Mobility HandIn
319	Mobility HandOut
335	Mobility Follow Me
351	Mobility Redial
354	Recording
370	Monitoring
399	Mobility IVR
401	Mobility DVOR
402	Mobility EFA
403	Mobility Session Handoff
415	Mobility Cell Pickup
418	Click to Conference
434	Forward No Retrieve
450	Forward No Retrieve Send Back to Parker
464	Call Control Discovery (indicates that the call is redirected to a PSTN failover number)
480	Intercompany Media Engine (IME)
496	IME Connection Timed Out
512	IME Not Enrolled
528	IME Socket Error
544	IME Domain Blacklisted

Q.931 Standard Redirect Reason Codes	
Value	**Description**
560	IME Prefix Blacklisted
576	IME Expired Ticket
592	IME Remote No Matching Route
608	IME Remote Unregistered
624	IME Remote IME Disabled
640	IME Remote Invalid IME Trunk URI
656	IME Remote URI not E164
672	IME Remote Called Number Not Available
688	IME Invalid Ticket
704	IME Unknown
720	IME PSTN Fallback
738	Presence Enabled Routing
752	Agent Greeting
783	NuRD
786	Native Call Queuing, queue a call
802	Native Call Queuing, de-queue a call
818	Native Call Queuing, redirect to the second destination when no agent is logged in
834	Native Call Queuing, redirect to the second destination when the queue is full
850	Native Call Queuing, redirect to the second destination when the maximum wait time in queue is reached

OnBehalfof Codes

The following table contains the available OnBehalfof Codes that may appear in a CDR record.

Table 6: OnBehalfof Codes

Value	Description
0	Unknown

Value	Description
1	CctiLine
2	Unicast Shared Resource Provider
3	Call Park
4	Conference
5	Call Forward
6	Meet-Me Conference
7	Meet-Me Conference Intercepts
8	Message Waiting
9	Multicast Shared Resource Provider
10	Transfer
11	SSAPI Manager
12	Device
13	Call Control
14	Immediate Divert
15	Barge
16	Pickup
17	Refer
18	Replaces
19	Redirection
20	Callback
21	Path Replacement
22	FacCmc Manager
23	Malicious Call
24	Mobility
25	Aar
26	Directed Call Park
27	Recording
28	Monitoring

Value	Description
29	CCDRequestingService
30	Intercompany Media Engine
31	FallBack Manager
32	Presence Enabled Routing
33	AgentGreeting
34	NativeCallQueuing
35	MobileCallType