Genesys Multicloud CX Release Notes

WebRTC Release Notes

2/22/2026

Contents

Not all releases or changes listed below may pertain to your deployment. Check the table below to see which releases apply to you.

Related documentation:

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RSS:

- For cloud
- For private edition

Important

The Release table lists the initial availability date for each release and the deployment environments for which a release is made available. Except when otherwise stated in the description for a specific release, each release includes all of the features and resolved issues that were introduced on earlier dates, regardless of the deployment environment. The features and resolved issues that apply only to a specific deployment environment are noted as such.

Service	Available	Genesys CX on		Private edition	Highlights	Release
AWS	Azure					
WebRTC Media Service	November 10, 2023	aws	Azure		• New Partitioned cookies flag • Configurable Coturn metrics port • Support of TLS 1.3	100.0.076.0000
WebRTC Media Service	September 29, 2023	aws	Azure		• Coturn server version	100.0.074.0000

Service	Available	Genesys CX on		Private edition	Highlights	Release
					4.6.2. • Metrics in Prometheus format. • Custom environment variables in helm charts. • New command-line option, **ice-addr-filter.** • Resolved issues.	
WebRTC Media Service	June 20, 2023	aws	Azure		• New traces for ICE failures. • Resolved issues.	100.0.072.0000
Webphone	April 19, 2023		Azure		**rtp-inactivity-timeout** parameter part of the the sign-in request.	100.0.038.0000
WebRTC Media Service	March 23, 2023	aws	Azure		Support for Ambassador cloud design pattern on Azure. New	100.0.067.0000

Service	Available	Genesys CX on		Private edition	Highlights	Release
					bandwidth-related metrics. Other improvements and resolved issues.	
WebRTC Media Service	December 20, 2022				Resolved issues.	100.0.057.0000
Webphone	October 12, 2022				Telemetry Service is introduced to collect logs, events, and metrics from software endpoints of Webphone.	100.0.036.0000
WebRTC Media Service	September 16, 2022				Resolved issue and improvements.	100.0.052.0000
WebRTC Media Service	June 21, 2022	aws	Azure		This release includes important fixes.	100.0.050.0000
WebRTC Media Service	June 16, 2022	aws	Azure		WebRTC Media Service now implements observability based on golden signals that enables a	100.0.046.0000

Service	Available	Genesys CX on		Private edition	Highlights	Release
					global view of the overall service health.	
Webphone	May 30, 2022				Support for Genesys Multicloud CX private edition deployments on Azure Kubernetes Service (AKS). This release includes important fixes.	100.0.031.0000
WebRTC Media Service	May 27, 2022				Support for Genesys Multicloud CX private edition deployments on Azure Kubernetes Service (AKS). This release includes important improvements and fixes.	100.0.044.0000
WebRTC Media Service	May 13, 2022	aws	Azure		This release includes important improvements	100.0.038.0000

Service	Available	Genesys CX on		Private edition	Highlights	Release
					and fixes.	
WebRTC Media Service	March 21, 2022	aws			WebRTC Media Service now supports TLS 1.2. It no longer supports TLS 1.0.	100.0.032.0000
WebRTC Media Service	February 24, 2022	aws	Azure		The **Content-length** HTTP header is now handled as a case-insensitive field.	100.0.025.0000
WebRTC Media Service	January 5, 2022	aws			WebRTC Media Service now starts the session cleanup timer when it receives the `/wait` request with an incorrect CSeq number.	100.0.019.0000
WebRTC Media Service	October 21, 2021				Support for deploying all private edition services in a single namespace.	100.0.016.0000

Service	Available	Genesys CX on		Private edition	Highlights	Release
					WebRTC now supports redirection of logs to `stdout`. CPU limits for WebRTC gateway pods can be configured now. Early Adopter Program support for Genesys Multicloud CX private edition deployments on GKE.	
WebRTC Media Service	October 20, 2021	aws	Azure		v100.0.015.0000 Codec configuration support for WebRTC Media Service.	100.0.015.0000
WebRTC Media Service	September 30, 2021	aws	Azure		v 100.0.011.0000 Support for arbitrary UIDs in private edition deployments on OpenShift. Call hold operation handled properly now.	100.0.011.0000
WebRTC Media Service	September 17, 2021	aws	Azure		v 100.0.009.0000. DN@SwitchName format handled properly now.	100.0.009.0000

Service	Available	Genesys CX on		Private edition	Highlights	Release
WebRTC Media Service	August 31, 2021		Azure		v 100.0.008.0000. Fixed WebRTC Media Service processing of ICE/ DTLS completion and ICE errors.	100.0.008.0000
WebRTC Media Service	July 14, 2021	aws			v 100.0.003.0000. Tested for Hunt Group feature in AWS.	100.0.003.0000
WebRTC Media Service	July 9, 2021	aws	Azure		v 9.0.000.89 Early Adopter Program support for Genesys Multicloud CX private edition deployments on OpenShift. Hunt Group feature successfully tested in SIP Cluster deployments with WebRTC Media Service (AWS only). 200 OK status code sent for a pending/wait request when there is an authorization	9.0.000.89

Service	Available	Genesys CX on		Private edition	Highlights	Release
					time out expiration.	

WebRTC Media Service: November 10, 2023

What's New

- A new cookie flag, **Partitioned** is added into webrtc sessions cookies. To handle the behavior, a new boolean option, `send-partitioned-cookie` is introduced (turned on by default). Usage: `-send-partitioned-cookie false` . This is enabled by default to support a breaking change in Google Chrome.
 (WRTCMS-1382)
 Limited to: Private Edition
- A new environment variable, `WEBRTC_COTURN_PROMETHEUS` is introduced for Coturn containers to set Prometheus-related options for Coturn. A new Helm chart option, `coturn.prometheusPort` is available to configure the Prometheus metrics. Metrics are exposed on the **9641** port as an HTTP response under */metrics*. (WRTCMS-1370)
 Limited to: Private Edition
- WebRTC now supports TLS 1.3 for secure connections. (WRTCMS-1371)

WebRTC Media Service: September 29, 2023

- 100.0.074.0000 available October 05, 2023
- Helm charts and containers

What's New

- The WebRTC microservice now uses Coturn server version 4.6.2. The previously used version was 4.5.1. (WRTCMS-1350)
- Now, Coturn can expose metrics in the Prometheus format. To enable this, start Coturn with the `--prometheus` command line argument that can be set using the `WEBRTC_COTURN_CMD_ARGS` environment variable.

  Metrics are exposed on the **9641** port as an HTTP response under */metrics*. (WRTCMS-1349)
- Now, helm charts allow to set custom environment variables for the Gateway and Coturn containers. The `gateway.envVars` and `coturn.envVars` options in the **values-template.yaml** file demonstrate the usage of custom environment variables. (WRTCMS-1325)
- A new monitoring message is available for case if an HTTP connection with the WebRTC agent is experiencing delays and the queue of HTTP-requests is growing:

  Type: SYS

  Method: `HTTPQueue Grows`

  Message: *HTTP queue is 2 or more, connection with agent is unstable*. (WRTCMS-1322)

- The following new statistics are available:
    - `wrtc_rtp_zero_bytes{ type="gateway_rcv" }` - counter, increased if Gateway received 0 bytes from the WEB-leg during the call.
    - `wrtc_rtp_zero_bytes{ type="gateway_snd" }` - counter, increased if Gateway sent 0 bytes to the WEB-leg during the call.
    - `wrtc_rtp_zero_bytes{ type="sip_rcv" }` - counter, increased if Gateway received 0 bytes from the SIP-leg during the call.
    - `wrtc_rtp_zero_bytes{ type="sip_snd" }` - counter, increased if Gateway sent 0 bytes to the SIP-leg during the call. (WRTCMS-1315)
- A new command-line option, **ice-addr-filter**, is now available to ignore ICE candidates collected on the gateway side. IP addresses of candidates that match the filter will be ignored and not sent to the WebRTC client.

  The value is the CIDR block or particular IP address as shown in the below examples:

  `-ice-addr-filter 172.17.0.1.1`

  `-ice-addr-filter 172.17.0.1/24`

  `-ice-addr-filter 172.17.0.1/16` (WRTCMS-1314)

Resolved Issues

- All log messages are now escaped properly. Previously, some SIP and other log messages for **stdout** logging mode were not escaped properly for correct exposure in the JSON format. (WRTCMS-1347)
  Limited to: Private Edition
- WebRTC Gateway no longer reports dynamic call-stats if RTP is not fully established. Previously, dynamic call-stats of RTP quality could contain incorrect reporting in case of a long time to answer on the agent side. (WRTCMS-1346)
- If the `not-reinvite-web-on-empty` option is set to *true* and the WebRTC Gateway receives a re-INVITE from the SIP-side during a DTLS negotiation, the re-INVITE is now processed correctly. (WRTCMS-1337)
  Limited to: Private Edition
- Now, exposed Prometheus statistics for lost, error, and jitter stats are processed correctly. (WRTCMS-1329)
- Now, the JSAPI client will send an empty list of candidates if the ICE timer has expired and no ICE candidates are collected. Previously, if the JSAPI client cannot collect any candidate from the TURN server, it continuously restarted the ICE timer to gather ICE candidates. (WRTCMS-1323)

WebRTC Media Service: June 20, 2023

- 100.0.072.0000 available June 23, 2023
- Helm charts and containers

What's New

- The following new traces are added for ICE failures:
    - SYS, ICEFailed, ICE State moved to FAILED, remote candidates are present
    - SYS, ICEFailed, ICE State moved to FAILED, no remote candidates

- SYS, ICEFailed, ICE Failed during established connection

  Also, a new metric, `wrtc_system_errors{type="failedice"}`, is added for ICE failures. (WRTCMS-1214)

- Gateway jitter is now also reported as a histogram for the **wrtc_rtp_gateway_jitter_ms_bucket** metric. (WRTCMS-1281)

Resolved Issues

- WebRTC Gateway now correctly processes the `Content-length` header irrespective of the casing of the text it contains. Previously, WebRTC Gateway incorrectly handled responses from GWS Configuration Service if the `Content-length` header contained text in lower case. (WRTCMS-1278)
- If log messages are written to stdout, the *\r* and *\n* characters are escaped and multiline messages now occupy only a single line. Previously, log messages from different threads overlapped with each other because of the *\r* and *\n* characters in SIP or other messages, causing a single line message to span multiple lines. (WRTCMS-1273)
- WebRTC Gateway metrics have been reworked and a new tenant dimension has been added to the tenant-related metrics. (WRTCMS-1218)
- Now, in case of a SIP registration failure. WebRTC Gateway does not drop an active call but preserves the RTP stream to allow agents to continue talking with customer. (WRTCMS-1178)
- Previously, WebRTC Gateway experienced 10ms gaps in jitter calculation. Now, jitter calculation is more reliable. Also, call stats are reported in monitoring messages for calls every 10 seconds. (WRTCMS-1280)
- Now, `http-request GET/options` to the monitoring port shows all current options in plain text format. For example, `curl http://localhost:10052/options">http://localhost:10052/options">http://localhost:10052/options">http://localhost:10052/options`. Also, at startup, WebRTC Gateway pushes a monitoring message with type SYS and method `Options`, and a message that contains the complete set of options in JSON format. (WRTCMS-1297)
- In case of connectivity issues with GAuth, Environment or Config services of GWS, WebRTC Gateway now pushes monitoring messages with type SYS and methods `Auth Service Failed`, `Env Service Failed`, `Cfg Service Failed`, and messages that contain the info about the failure, which can help understand the reason for the connectivity issue without having to download logs. (WRTCMS-1298)
- The JSON format for the dashboard located in the helm chart now shows correctly updated metrics. (WRTCMS-1302)
- WebRTC Gateway no longer crashes if the DN or Agent Name contains **%s**. Previously, WebRTC Gateway could crash in such cases. (WRTCMS-1310)

Webphone: April 19, 2023 Azure

What's New

- The Webphone client now sends the **rtp-inactivity-timeout** parameter in the sign-in request to WebRTC Gateway. (WRTCMS-1204)

WebRTC Media Service: March 23, 2023 aws Azure

- 100.0.067.0000 available April 04, 2023
- Helm charts and containers

What's New

- WebRTC now retries sending a SIP REGISTER request in case of an error or timeout when SIP Registration fails due to a network disruption or the SIP Proxy instance is unavailable. Previously, when a SIP REGISTER transaction failed, WebRTC Gateway immediately terminated the corresponding session and that may have resulted in unexpected call terminations. This functionality is turned off by default. (WRTCMS-1186)
- WebRTC Gateway now supports RTP inactivity timeout to detect dropped or no-audio calls. This is turned off by default. (WRTCMS-1185)
- The following new bandwidth-related metrics are added for the WebRTC Gateway:
  - `wrtc_current_audio_bandwidth_web_inbound_kbps`
  - `wrtc_current_audio_bandwidth_web_outbound_kbps`
  - `wrtc_current_audio_bandwidth_sip_inbound_kbps`
  - `wrtc_current_audio_bandwidth_sip_outbound_kbps`
    (WRTCMS-1145)
- Helm-charts are updated to support the Ambassador design pattern in Azure. (WRTCMS-1077)
  Limited to: Genesys CX on Azure

Resolved Issues

- The KEDA autoscaler is now created with `ignoreNullValues` set to **false** to enable KEDA to show a warning instead of transferring 0 to HPA. Also, the name of the metric in `KEDA ScaledObject` is changed; `color` is added into the `metricName`. (WRTCMS-1237)
  Limited to: Genesys CX on Azure
- WebRTC now uses log streaming to Grafana Cloud Loki for Docker based services in Engage Cloud on AWS. (WRTCMS-981)
  Limited to: Genesys CX on AWS
- Now, session cookies are cleared on receiving a `sign-out` API request. (WRTCMS-915)
- WebRTC clients can now automatically re-login at a new active WebRTC deployment after cutover. (WRTCMS-861)

WebRTC Media Service: December 20, 2022

- 100.0.057.0000 available December 20, 2022
- Helm charts and containers

Resolved Issues

- Now, the session cookie is cleared on the sign-out API request. (WRTCMS-915)

For private edition

- Now, WebRTC Gateway will not re-invite WebRTC client for a SIP re-invite session without an SDP offer if the WebRTC Gateway already has an active SDP offer for the SIP side. To enable this functionality, **not-reinvite-web-on-empty** must be set to `true` by using of command-line arguments.
  Example: **-not-reinvite-web-on-empty true**

For more information, see https://all.docs.genesys.com/WebRTC/Current/WebRTCPEGuide/Configure">https://all.docs.genesys.com/WebRTC/Current/WebRTCPEGuide/Configure">Configure WebRTC and refer to **gateway.arguments** Helm chart value in this page. (WRTCMS-1194)

Webphone: October 12, 2022

- 100.0.036.0000 available October 12, 2022
- Helm charts and containers

What's New

- WebRTC introduces Telemetry Service to collect logs, events, and metrics from software endpoints of the Webphone component. Also, the Webphone component collects all the logs from the browser and send them to the Telemetry Service. (WRTCMS-902)

Resolved Issues

- The Moment.js package vulnerability is fixed. (WRTCMS-1130)

WebRTC Media Service: September 16, 2022

- 100.0.052.0000 available September 16, 2022
- Helm charts and containers

Resolved Issues

- WebRTC Gateway now correctly encodes the HTTP requests to GAuth Service. Previously, the WebRTC Gateway incorrectly encoded the HTTP requests. This led to incorrect processing on new GAuth deployments. (WRTCMS-1057)
- WebRTC Gateway now correctly exposes the availability of GAuth Service. Previously, the WebRTC Gateway was not able to detect the 503 error from the GAuth Service. This led to incorrect exposure of the GAuth metric availability. (WRTCMS-1055)

WebRTC Media Service: June 21, 2022

- 100.0.050.0000 available September 16, 2022
- Helm charts and containers

Resolved Issues

- WebRTC JS library now uses a generic API path for all "`/sign_in`" requests and sends the requests to the appropriate endpoint. (WRTCMS-1062)

For private edition

- WebRTC Media Service now avoids the creation of a stale session while authenticating a `sign_in` request from a client. Previously, when WebRTC Media Service handled the scenario in the following way, it led to a situation where the WebRTC Gateway created a stale session without a HTTP client, affecting the active agent's sessions.
  1. The WebRTC client sends a `sign_in` request to the WebRTC Gateway.
  2. The WebRTC Gateway starts the authentication procedure on GAuth service.
  3. The client is disconnected from the WebRTC Gateway and the authentication process is completed. (WRTCMS-1066)

WebRTC Media Service: June 16, 2022

- 100.0.046.0000 available September 16, 2022
- Helm charts and containers

What's New

- WebRTC Media Service now implements observability based on golden signals that enables a global view of the overall service health. The four golden signals include metrics for traffic, latency, errors, and saturation. For more information, see https://sre.google/sre-book/monitoring-distributed-systems/#xref_monitoring_golden-signals">https://sre.google/sre-book/monitoring-distributed-systems/#xref_monitoring_golden-signals">Google SRE book. (WRTCMS-976)

Webphone: May 30, 2022

- 100.0.031.0000 available May 30, 2022
- Helm charts and containers

Resolved Issues

- Webphone now correctly handles the pre-released version (v1.22.8-gke.200) of Kubernetes. Previously, the ingress object was not deployed properly for the pre-released version. (WRTCMS-1035)
- Webphone now decreases the number of active agent sessions in the active session metric now decrements when an agent logs out of the session. Previously, the active session metric kept increasing even when the agent logged out of the session. (WRTCMS-956)

For private edition

- Webphone supports deployments on Azure Kubernetes Service (AKS) in Genesys Multicloud CX private edition. (CPE-3881)

WebRTC Media Service: May 27, 2022

- 100.0.044.0000 available May 27, 2022
- Helm charts and containers

What's New

- Network policies can now be enabled and configured independently for WebRTC Gateway and Coturn service. (WRTCMS-985)
- Starting with version 100.0.025+0001, WebRTC Media Service now includes the **enableServiceLinks** Helm chart option in the deployment section of the **values.yaml** file. This option controls whether service information is added to the environment variables or not. Valid values are true and false. The default value is false. (WRTCMS-942)

Resolved Issues

- WebRTC Media Service now deploys the Coturn service even if the **deployment.coturnService** section includes no values. (WRTCMS-1034)
- WebRTC Media Service now correctly handles the pre-released version (v1.22.8-gke.200) of Kubernetes. Previously, the ingress object was not deployed properly for the pre-released version. (WRTCMS-1010)

For private edition

- WebRTC Media Service supports deployments on Azure Kubernetes Service (AKS) in Genesys Multicloud CX private edition. (CPE-3880)
  More info:

WebRTC Media Service: May 13, 2022

What's New

- WebRTC Gateway now logs a message into Elasticsearch/stdout even if there are no relay candidates received from the client side, where the **wrtc_system_error {type="norelay", domain="%DOMAIN NAME%"}** metric is incremented. (WRTCMS-966)
- The Elasticsearch\Grafana log messages now include the following fields for troubleshooting purposes:
  - **loginName** - Agent name for the given WebRTC session.
  - **callUUID** - Genesys Call UUID that is tracked through voice and WebRTC services. (WRTCMS-965)

Resolved Issues

- WebRTC Gateway now clears the existing DTLS context when a new SIP re-invite is received. Previously, when the WebRTC Gateway did not complete processing of the existing DTLS context and a new SIP re-invite was received, the existing DTLS context was used for re-establishing media connection which might have resulted in no media for the call. (WRTCMS-1014)
- WebRTC Media Service now correctly handles all the audio calls in SIP re-invite state.

  Previously, when WebRTC Media Service handled the scenario in the following way (below), it led to a situation where the audio call got stuck in the SIP re-invite state, declining all subsequent SIP re-invite attempts with the `486 Busy Here` error:

  1. The WebRTC Call is established.
  2. The agent performs a consult call.
  3. The agent puts the consult call on hold and **SIP NOTIFY** is sent to the WebRTC Gateway.
  4. The WebRTC Gateway processes the SIP NOTIFY and sends an **OK** message so that the SIP Server sends an **EVENT-TALK** message to the client.
  5. While the hold operation is in progress, the agent releases the consult call and SIP Server sends a SIP re-invite request to the WebRTC Gateway to recover the main call.
  6. The WebRTC Gateway processes the SIP re-invite and sends **OFFER SDP** to the client.
  7. Simultaneously, the client finishes processing the **EVENT-TALK** message and sends **OFFER SDP** to the WebRTC Gateway. (WRTCMS-1005)

WebRTC Media Service: March 21, 2022

What's New

- WebRTC Media Service now supports TLS 1.2. It no longer supports TLS 1.0. (WRTCMS-959)

WebRTC Media Service: February 24, 2022

- 100.0.025.0000 available February 24, 2022
- Helm charts and containers

Resolved Issues

- The **Content-length** HTTP header is now handled as a case-insensitive field. Previously, when the **Content-length** HTTP header was handled as a case-sensitive field, the body of the HTTP request was parsed incorrectly if the **Content-length** header field had lowercase characters. (WRTCMS-923)

WebRTC Media Service: January 05, 2022

Resolved Issues

- WebRTC Media Service now starts the session cleanup timer when it receives the `/wait` request with an incorrect CSeq number. Previously while receiving the `/wait` request with an incorrect CSeq number, WebRTC Media Service sent a `400 Bad Request` to the client but did not restart the cleanup timer. After this, if the browser client sent a new `/sign_in` request with a new tab ID without signing out from the previous session, the session stopped responding in the WebRTC Media Service side. (WRTCMS-883)

WebRTC Media Service: October 21, 2021

- 100.0.016.0000 available October 21, 2021
- Helm charts and containers

For private edition

- WebRTC now supports configurable namespaces. For more information, see . (WRTCMS-836)
- CPU limits for WebRTC gateway pods can be configured now. (WRTCMS-828)
- WebRTC now supports redirection of logs to `stdout` . (WRTCMS-823)
- As of December 23, 2021, WebRTC supports deployments on Google Kubernetes Engine (GKE) in Genesys Multicloud CX private edition, as part of the Early Adopter Program. (CPE-1963)

WebRTC Media Service: October 20, 2021

What's New

- WebRTC Media Service allows configuring the list of codecs used for browser-based WebRTC endpoints.
Configure the new option, `webrtc.codecs` on the Cloud Cluster application/Agent group/Person level to enable WWE 9 to provide the list of codecs that can be used for the WebRTC endpoint. The option `webrtc.codecs` defines the comma-separated list that specifies codecs that can be used for the WebRTC endpoint. The default values for this option are `opus, pcmu, pcma`. The changes take effect after the next platform configuration refresh interval.
The browser-based WebRTC endpoint provides the list of codecs to WebRTC Media Service in the `/sign_in` request body. These codecs are used by WebRTC Media Service for OFFER/ANSWER during SDP negotiation.

> **Important**
> For information on `webrtc.codecs` support in WWE, see the page.

For Genesys Softphone in WebRTC mode, you can use the existing options `codecs.enabled.audio` and `codecs.enabled.video`. (WRTCMS-758)

WebRTC Media Service: September 30, 2021

- 100.0.011.0000 available September 22, 2021
- Helm charts and containers

Resolved Issues

- The WebRTC Media Service JavaScript library now correctly processes call hold operations. Previously, when an agent tried to put a consultation call on hold, the library silently ignored the request from SIP Server. As a result, the call hold and its subsequent operations were delayed for 30 seconds due to the hold operation timeout. (WRTCMS-798)

For private edition

- WebRTC now supports the use of arbitrary, or random, user IDs (UIDs) in OpenShift.
  - The Dockerfile has been modified to specify container and file ownership as `user=500` (genesys) and `group=0` (root).
  - The securityContext settings exposed in the default **values.yaml** file specify the user and group IDs for the genesys user (`500:500:500`). You must override these Helm chart values if you want OpenShift to use arbitrary UIDs. For more information, see .
  - WebRTC is deployed using ServiceAccounts that use the **restricted** Security Context Constraint (SCC). In an earlier implementation, Genesys required you to deploy all private edition services using a ServiceAccount associated with the custom **genesys-restricted** SCC, to control permissions for the **genesys** user (500). Genesys now expects OpenShift to use arbitrary UIDs in your deployment, and the **genesys-restricted** SCC has been deprecated. If you previously deployed WebRTC using the **genesys-restricted** SCC, Genesys recommends that you redeploy WebRTC so that you use arbitrary UIDs.

  (WRTCMS-764)
  More info:

WebRTC Media Service: September 17, 2021

Resolved Issues

- WebRTC Media Service now handles the `DN@SwitchName` format properly in sign-in request. Previously, WebRTC Media Service incorrectly handled the `DN@SwitcName` format for JSON-based signaling protocol, resulting in an invalid SIP DN being registered. (WRTCMS-792)

WebRTC Media Service: August 31, 2021

Resolved Issues

- WebRTC Media Service no longer processes SIP re-invite until the current ICE/DTLS establishment process completes. Previously, because SIP re-invite did not require that the WebRTC connection be reestablished, WebRTC Media Service did not wait for the completion of the current ICE/DTLS establishment process. This resulted in the agent missing some part of the call (for example, the call did not include the whisper). (WRTCMS-773)
- The WebRTC Media Service Java Script API library now handles Interactive Connectivity Establishment (ICE) errors and reconnects the media if possible. Previously, the library did not handle the ICE errors caused by temporarily network disconnection during active calls. This resulted in the media connection being lost. (WRTCMS-750)

WebRTC Media Service: July 14, 2021

- 100.0.003.0000 available July 14, 2021
- Helm charts and containers

What's New

- **AWS only.** The Hunt Group feature (parallel call distribution strategy) is successfully tested in SIP Cluster deployments with WebRTC Media Service. Supported in WebRTC Media Service v9.0.and v100.0.x. Calls are initiated/received by the browser-based Workspace Web Edition (WWE) 9.0.x. (WRTCMS-725)

WebRTC Media Service: July 09, 2021

- 9.0.000.89 available June 30, 2021
- Helm charts and containers

What's New

- **AWS only.** The Hunt Group feature (parallel call distribution strategy) is successfully tested in SIP Cluster deployments with WebRTC Media Service. Supported in WebRTC Media Service v9.0.and v100.0.x. Calls are initiated/received by the browser-based Workspace Web Edition (WWE) 9.0.x. (WRTCMS-725)

Resolved Issues

- WebRTC Media Service now sends a 200 OK status code for a pending/wait request when there is an authorization time out expiration (

For private edition

- Starting with this release, WebRTC is available for select customers in Genesys Multicloud CX private edition, as part of the Early Adopter Program. Deployments on OpenShift Container Platform (OpenShift) are supported. (WRTCMS-661)
  More info:

Prior Releases

For information about prior releases of WebRTC, click here: WebRTC Release Notes