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Genesys Engage cloud Release Note

WebRTC Media Service

5/4/2025

WebRTC Media Service

For information on later releases of WebRTC, click here: WebRTC

Note: Not all changes listed below may pertain to your deployment.

April 22, 2021 (9.0.000.88)

What's New

- WebRTC metrics reporting to Prometheus monitoring system is expanded with a new metric named `wrtc_max_clients_per_instance` to display the maximum number of agents that the WebRTC instance can handle.
- WebRTC Media Service now supports proactive authentication renewal. This feature allows WebRTC endpoints to refresh WebRTC authentication before its expiration in the WebRTC Media Service. The proactive authentication renewal feature prevents service interruption due to the authentication expiration of the WebRTC endpoints.

- To reduce the number of stale HTTP connections, WebRTC Media Service now responds with the `200 OK` message to the previous `/wait` request if a new `/wait` request is received.
- ElasticSearch messaging is now extended with the following new fields:
    - `direction`—to distinguish between incoming/outgoing messages
    - `logfile`—to display the name of the current workflow logfile
    - `color`—to mark the color of deployment that produced the ElasticSearch message
- WebRTC JavaScript API now supports proactive authentication renewal.
- WebRTC JavaScript API client now sends JSAPI-version in the `/sign_in` request.

Resolved Issues

- WebRTC Media Service now allows multiple browser tabs to be opened for the same session. The first signed-in browser tab becomes the master tab within the session. This master tab handles all the incoming and outgoing calls and interactions within WebRTC Media Service. Other browser tabs (clients) within the same session are kept in the session queue. The maximum queue size is 5 clients. Any subsequent sign-in beyond this limit is rejected.

  When the master tab is signed out, the next tab in the queue becomes the master. When all the browser clients sign out, the WebRTC Media Service session is closed. WebRTC Media Service handles these multiple browser tabs by providing a unique `tab id` for each tab. This `tab id` is generated by the WebRTC JSAPI library and passed to WebRTC Media Service via an HTTP request parameter. For example: `'webrtc-uri/sign_in?tab=id12345'`. Previously, all opened browser tabs sent long-polling/wait requests with the same `session id` to WebRTC Media Service, which caused session conflicts. (WRTCMS-557)

- WebRTC Media Service now correctly handles the `rtcp-mux` attribute in the WebRTC client re-INVITE SDP and uses only one RTP component for the Interactive Connectivity Establishment (ICE) session. Previously, this attribute was handled incorrectly when an initial offer was sent from the WebRTC Media Service.

  The incorrect handling of the `rtcp-mux` attribute sometimes caused WebRTC Media Service to select two components for the ICE session (RTP+RTCP) instead of one (RTP). As a result, ICE checked for the RTCP component failed and the media path was not established (to provide audio). (WRTCMS-689)

January 12, 2021 (9.0.000.80)

What's New

- TURN Server is upgraded to version 4.5.1.3.
- WebRTC metrics reporting to Prometheus monitoring system are expanded with a new metric named `wrtc_client_errors {type="dropped_on_signout"}`. This metric is incremented when an active call is dropped due to an agent sign-out. Also, the default value of the `http-session-timeout` option is changed from 20 seconds to 60 seconds.
- WebRTC JavaScript API now uses flags in body parameters to provide additional information to the WebRTC Media Service. This functionality is used to renew session authentication. This feature is

available in WebRTC JavaScript 9.0.000.80.

Resolved Issues

- WebRTC Media Service no longer starts an authentication expiration timer when the browser client sends a subsequent sign-in request due to page reload or connection errors. Previously in this scenario, WebRTC Media Service incorrectly started an authentication expiration timer for 10 seconds. As a result, a `403 Authentication Expired` response was sent to the client and the client was stuck in an unauthenticated state. (WRTCMS-550)

October 14, 2020 (9.0.000.62)

What's New

This release includes only resolved issues.

Resolved Issues

- WebRTC Media Service no longer prints RTP/RTCP packets to the log files. Previously, WebRTC Media Service terminated unexpectedly when printing RTP/RTCP packets to the log file that contained the `%r` symbol. (WRTCMS-522)

September 25, 2020 (9.0.000.61)

What's New

- When the number of concurrent authentication attempts exceeds the limit set by the new `auth-conn-limit` option (whose default value is 280), WebRTC Media Service rejects the next authentication attempt with an error code of `429 Too Many Requests`.
- WebRTC Media Service now gets the session ID from cookies instead of from the request URI's session identification parameter. This feature is supported in WebRTC Java Script API v9.0.000.61.

August 31, 2020 (9.0.000.57)

What's New

This release includes only resolved issues.

Resolved Issues

- WebRTC Media Service moved the environment functionality to a separate thread. Also, WebRTC Media Service now provides an anti-throttling mechanism that resolves the main thread deceleration. (WRTCMS-476)
- Starting with release 9.0.000.47, WebRTC JavaScript client supports Edge Chromium 2020 browser. (WRTCMS-483)

June 26, 2020 (9.0.000.56)

What's New

- WebRTC Media Service now adds the `color` label to WebRTC metrics that report to Prometheus monitoring system. The value of the `color` label is configured in the `api-path` option that is used during deployment for active and inactive deployment colors. That is, if WebRTC Media Service belongs to green deployment and if `api-path=green`, then all metrics will contain the label `color=green`.

May 06, 2020 (9.0.000.52)

What's New

- WebRTC Media Service now quickly detects if the selected transport for a SIP Tenant is out of service and searches for a new active transport using active in-service detection for SIP Tenants. Using active in-service detection, WebRTC Media Service sends a request to the DNS server to resolve the SIP Server/SIP Proxy FQDN. It then selects one SRV/A record from the list of the resolved destinations and sends a `SIP OPTIONS` request to the selected transport. If a `SIP 200 Ok` response is received within the configured time that is defined in the option `oos-force`, the selected transport is marked as `in-service` and used for subsequent SIP transactions. The next check for this transport, which is configured in `oos-check`, will be performed after the defined time period expires.

April 15, 2020 (9.0.000.51)

What's New

This release includes only resolved issues.

Resolved Issues

- WebRTC Media Service now correctly retrieves an inbound call that was put on hold by a browser-based WebRTC agent after the second call was released. Previously, the third-party call control (3PCC) `retrieve` was not processed correctly. (WRTCMS-431)

April 03, 2020 (9.0.000.49)

What's New

- The Coturn server of WebRTC Media Service is upgraded to version 4.5.1.1 to allow the WebRTC Media Service to work with the Federal Information Processing Standards (FIPS) compliant version of OpenSSL library.

Resolved Issues

- The Third-party call control (3PCC) retrieve operation in WebRTC Media Service now works properly when it is initiated by a browser-based WebRTC agent that uses the Opus codec. Previously, the agent did not receive the media. (WRTCMS-420)

- When WebRTC Media Service processes remote Interactive Connectivity Establishment (ICE) candidates, it now checks only the `relay` candidates. This improves the media connection establishment time. Previously, WebRTC Media Service checked all types of candidates including the `host`. (WRTCMS-416)

- To improve the media connection establishment time, WebRTC Media Service does not `re-Invite` the WebRTC participant from the SIP server side if:
    - There is a SIP `re-Invite` without Session Description Protocol (SDP).
    - The SIP `re-Invite` SDP media direction and the active/inactive state was not changed, and it does not have any new media types.

      For the above cases, WebRTC Gateway creates new offer/answer SDP based on the previously used local SDP. (WRTCMS-415)

- WebRTC Media Service now properly processes the second SIP `INVITE` without SDP by replying with a SIP `200 Ok` response with an SDP offer. Previously, when WebRTC Media Service received two subsequent SIP `INVITE` requests without SDP, it sent a SIP `200 Ok` response for the second `INVITE` without an SDP offer. It violates the Request for Comments (RFC) 3261 standard and can lead to media connectivity establishment failure. This can affect the following scenarios:

    - Route call to WebRTC agent with recording
    - 3PCC call from WebRTC agent with recording
    - Consult call from WebRTC agent with recording
      (WRTCMS-413)

- WebRTC Media Service no longer allows G722 codec usage for media connections. (WRTCMS-409)

January 31, 2020 (9.0.000.47)

What's New

- The WebRTC Media Service now provides call quality statistics using the HTTP call message `PUT-STATS`. This message is automatically sent by WebRTC Gateway to the WebRTC JavaScript client at the end of each call and contains call quality statistics in JSON format.

  The WebRTC JavaScript library then passes the call quality statistics to WWE 9. WWE 9 then includes the call quality statistics in the `AttributeUserData` of the `RequestDistributeUserEvent` message that is distributed to SIP Server at the end of the interaction. The corresponding key-name is retrieved from the value of `interaction-workspace/webrtc.quality.statistics.key-name` configured at the Application/Person level.

> **Important**
> This feature is applicable only for browser-based WebRTC clients with WWE 9.

Resolved Issues

- WebRTC Gateway now includes `SameSite=None` and `Secure` attributes to all `Set-Cookie` headers generated by the Gateway. (WRTCMS-300)

November 07, 2019 (9.0.000.43 UPDATE)

What's New

Legacy SIP Environment

- WebRTC Media Service now works with SIP Server operating in Standalone mode. Previously, WebRTC Media Service was integrated only with SIP Cluster.

Microservice Monitoring

- WebRTC metrics reporting to Prometheus monitoring system is expanded with a new metric named `wrtc_system_error` to capture internal problems that occur in the WebRTC Media Service. Based on this new metric, the Prometheus monitoring system will raise an alert when an internal problem occurs.

October 02, 2019 (9.0.000.41 UPDATE)

What's New

Elasticsearch

- WebRTC Media Service now supports Elasticsearch by sending different data into Elasticsearch. This feature allows you to store and quickly analyze large volumes of data in real-time.

Resolved Issues

- Stability improvements have been made in WebRTC Media Service to prevent WebRTC gateway from terminating unexpectedly under certain conditions. (WRTCMS-253)

July 03, 2019 (9.0.000.37 UPDATE)

What's New

WebRTC Agent timeout

- WebRTC Agent will now be signed out of the session if the connection between the Endpoint and WebRTC gateway is broken for more than the configured time. For improved security reasons, this time period is set to **5** seconds by default. However, it is configurable up to 15 minutes.

April 11, 2019 (9.0.000.37)

What's New

SIP addresses

- WebRTC Media Service now retrieves the SIP address from Genesys Web Services (GWS) version 9 automatically and users are not required to configure the SIP address while provisioning Agent Desktop. The Agent Desktop supported version is 9.0.000.21 and above.

December 21, 2018 (9.0.000.27)

What's New

- WebRTC Media Service now supports OAuth 2.0 authentication and authorization method to validate the user credentials passed from Agent Desktop. The Genesys Softphone compatible version to support OAuth 2.0 is 9.0.004.05 and above and the Agent Desktop version is 9.0.000.17 and above.

June 29, 2018 (9.0.000.15)

What's New

Initial release

This is the initial release of WebRTC Media Service for the Genesys Engage cloud (formerly known as PureEngage Cloud [PEC]) solution. Agents can handle both inbound and outbound voice calls through WebRTC-capable devices like Genesys Softphone by communicating with the Genesys Engage cloud software through the WebRTC Media Service. The WebRTC Media Service supports Genesys Softphone version 9.0.003.04+.

The key features of the WebRTC Media Service are:

- Supports G.711 and Opus codecs.
- Provides real-time media transcoding whenever required.
- Supports audio calls only.
- Signalling and media encryption capabilities of WebRTC Media Service ensures appropriate security for voice communications over the public network.

Known Issues

- When WebRTC is used by WWE with zero-footprint (browser-based endpoint), you can only use one (current) tab with WWE in the browser. Opening a second tab with WWE results in double-registration and permanent re-registration of the agent. This limitation is applicable to the following supported browsers:
    - Edge Chromium 2020
    - Google Chrome
    - Firefox