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Web Real-Time Communications Deployment Guide

New in this Release

12/19/2025

New in this Release

Check out the new features that have been added in the latest releases of the Genesys WebRTC Service.

New in Release 8.5.2

Release 8.5.208.16 of the WebRTC Gateway

- A boolean option, [rsm] web-disable-sdes, has been added with a default value of `true`, which disables SDES-SRTP in initial SDP offers to the web client. With a value of `true`, the RTP profile in an initial offer SDP from the gateway to a web client will use "UDP/TLS/RTP/SAVPF" instead of "RTP/SAVPF", adhering to the WebRTC standard. Even with a value of `true`, incoming offers with SDES-SRTP will still be accepted, though the use of SDES-SRTP is obsolete and discouraged. (MWA-575)
- The Genesys WebRTC Gateway now has G.722 audio codec enabled by default. Note that G.722 codec is given higher priority than G.711 codecs by the browsers as well as the WebRTC gateway, therefore G.722 may be chosen over G.711 during call establishment. (MWA-98)

Release 8.5.206.25 of the WebRTC Gateway

- The following configuration option has been added, which helps work around some renegotiation issues with Firefox. When an older version of Firefox without bundle support is used (version 37 and lower), set this option to `false`. (MWA-547)
web-offer-bundle-only
Section: `rsm`
Default Value: `true`
Valid Values: `true`, `false`

Release 8.5.201.95 of the WebRTC Gateway and Release 8.5.210.03 of the WebRTC JavaScript API

- The Genesys WebRTC Service now supports adding video to an audio-only call
- The Genesys WebRTC Gateway now supports remote CTI control by providing the SIP extensions event package known as the BroadSoft SIP extensions

Release 8.5.201.30 of the WebRTC Gateway

- Support for third-party call control (3pcc) basic functions and two-step procedures

Release 8.5.200.95 of the WebRTC Gateway and Release 8.5.200.07 of the WebRTC JavaScript API

This first release of the Genesys WebRTC Service includes:

- Communications:
 - Supports two-way audio-only calls
 - Supports two-way audio and video calls
 - Supports transcoding on supported media types in both directions
 - Supports transitions between audio-only and video/audio sessions (within browser limitations)
 - Supports incoming calls from either the web or the SIP side
 - Supports sending context data from a web client to the SIP Server as attached data when a call is established
 - Supports sending mid-call user data either to SIP Server as mapped data, or to the remote peer
 - Supports call transfer with Genesys SIP Server
 - Supports sending DTMF tones as telephone-events
- Web browser support:
 - Google Chrome
 - Google Chrome for Android
 - Mozilla Firefox
 - Mozilla Firefox for Android
 - Opera (Desktop and Mobile)
- Supports anonymous access from the web (such as in click-to-dial scenarios)
- Flexible deployment on:
 - A dedicated server
 - A shared server with other Genesys components
 - Multiple load-balanced instances for scalability
- Security:
 - HTTPS, SIP TLS, and secured connection to Configuration Server
 - Web-side encryption of RTP traffic according to IETF recommendations (SDS-SRTP and DTLS-SRTP)
 - Can be configured to use SIP-side SRTP when outside the trusted area
 - Supports Client Side Port Definition (CSPD), allowing for customization of the:
 - SIP-side RTP port
 - Web-side RTP port
 - Transport address and port for a client-side connection
 - Use of multiple NICs, to separate public and private interfaces when both are used (DMZ deployment)
 - Firewall traversal with support of ICE
- Codecs:
 - G.711 A-law and μ -Law audio format

- G.729 audio on SIP/RTP side
- H.264 profiles up to and including Profile 3.1 on SIP/RTP side
- VP8 video
- Real-Time media transcoding whenever required
- RTCP AVPF (partially supported)
- telephone-events
- Support of HTTP over IPv4 and IPv6 interfaces in the same instance
- Integration with Genesys Management Framework
- SNMP for alarms, traps and MIBs
- Platforms:
 - Windows Server 2008 32- and 64-bit
 - RedHat Linux 6.0 64-bit
 - VMWare ESXi 5
- The WebRTC Gateway is built with OpenSSL library version 1.0.1g. This version of OpenSSL is not affected by the [TLS heartbeat read overrun issue](#).
- The WebRTC Gateway supports receiving INFO data in www-form-urlencoded format from the browser in the middle of a call, and forwarding it to the SIP Server using the SIP INFO method.
- The WebRTC Gateway accepts + as a valid first character of a DN.
- The WebRTC Gateway includes support for Cross-Origin Resource Sharing (CORS).
- JavaScript libraries for integration with web applications, communications, and attached data transfer
- The WebRTC JSAPI provides a configuration parameter to specify the time (in milliseconds) to wait for an answer from the peer side after making an offer. If that timeout expires and the offer is still not answered, then the JSAPI sends the onPeerNoanswer event to the client application. The minimum valid value for this timeout is 18000 (18 seconds) and the default value is 60000 (60 seconds).
- The WebRTC JSAPI provides a mechanism for mid-session data transfer for the following scenarios:
 - Between two peers
 - From a peer to the SIP Server as mapped user data