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WebRTC Private Edition Guide

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Learn about WebRTC and how it works in Genesys Multicloud CX private edition.

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Web Real-Time Communication (WebRTC) is a real-time communication over the internet that enables an agent to connect with the Genesys contact center environment to perform their business operations.

WebRTC is a shared (multitenant) service that acts as the signaling and media gateway. The signaling gateway is used to interwork WebRTC with Session Initiation Protocol (SIP), and the media gateway is used to terminate the Interactive Connectivity Establishment (ICE) and Secure Real-time Transport Protocol (SRTP).

WebRTC bridges the calls that are initiated/received by the browser. The SIP Server handles these calls as a SIP call to provide core Genesys features such as routing and IVR. These features are handled by Genesys for browser endpoints with the help of MCP in the call flow. Third-party component CoTURN is used to implement TURN and STUN servers.

Supported Kubernetes platforms

WebRTC is supported on the following cloud platforms:

- Azure Kubernetes Service (AKS)
- Google Kubernetes Engine (GKE)

See the WebRTC Release Notes for information about when support was introduced.