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Web Real-Time Communications Deployment Guide

Interoperability Considerations

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Contents

Interoperability Considerations

The Genesys WebRTC Gateway can be used with several types of component, including Genesys servers and many of the leading industry-standard web browsers. This topic provides information on which of these are supported.

Interoperable Components

- Genesys SIP Server
- Genesys Universal Routing Server (URS)
- Genesys Orchestration Server (ORS)
- Genesys Voice Platform (GVP) Media Control Platform (MCP)
- Google Chrome Browser (desktop and mobile)
- Mozilla Firefox Browser (desktop and mobile)
- Opera Web Browser (desktop and mobile)
- Genesys SIP Endpoint SDK
- Genesys Workspace Desktop Edition (WDE) SIP Endpoint
- Genesys Workspace Web Edition (WWE)
- Third-party SIP soft-phones (Bria and LinPhone)
- Genesys Epi Phone

Supported Genesys Product Versions

The following table indicates which versions of the listed Genesys components are supported for use with the WebRTC Gateway.

Component	8.0	8.1	8.5
Management Framework	Yes	Yes. Preferred Configuration Server version is 8.1.3 or above.	—
Genesys Administrator	No	Yes	—
SIP Server	No	Yes. Preferred SIP Server version is 8.1.1 or above. **Note**: *not verified with SIP cluster*.	—
GVP	No	Yes. Preferred GVP version is 8.1.7.	Yes. Works with GVP 8.5.0. However, GVP 8.5.1 or above is the

Component	8.0	8.1	8.5
			preferred version.
URS	No	Yes. Preferred URS version is 8.1.4 or above.	—
ORS	—	Yes. Preferred ORS version is 8.1.4 or above.	—
Workspace Desktop Edition SIP Endpoint	—	—	Yes. Preferred version is 8.5.1 or above.
Workspace Web Edition	—	—	Yes. Preferred version is 8.5.1 or above.

Supported Endpoints

> **Important**
>
> It is recommended that the latest stable version of a supported browser is used. However, in some cases, it may be required to use an older version due to some bug or incompatible change in the latest version. See the WebRTC Product Alerts and Release Notes for known issues with specific browser versions.

Web Endpoint	Version
Chrome	54 and above
Firefox	49 and above
Opera	41 and above

SIP Endpoint	Version
Bria	4.1 and above
Genesys SIP Endpoint SDK	8.5.2 and above

Supported Call Scenarios

Peer-to-Peer

Peer-to-peer sessions can be either symmetric or asymmetric:

- **Symmetric Session**—Each peer offers the same media types—such as audio to audio or video to video.
- **Asymmetric Session**—Different media types—such as audio to video or video to audio—are offered by

the peers.

Browser-to-Browser	• Symmetric session • Asymmetric session
Browser-to-SIP (or vice versa)	• Symmetric session • Asymmetric session
Browser-to-MCP	• Video calls • Audio calls (both sides must be audio-only) • Announcement • VoiceXML dialog • ASR input support • DTMF support (Chrome only—Firefox does not currently support sending DTMF tones)

With Genesys Routing

Direct Agent Routing	• Call to Route Point • No treatment play • Agent is always available
Agent Routing with Treatment	• Call to Route Point • Agent is not Ready, MCP plays Video (audio-only is also supported) • Agent becomes Ready • Treatment stops • Call routed to Agent
Routed to MCP for Transfer	• Call to Route Point • Routed to MCP • MCP plays Video (audio-only is also supported) • VoiceXML transfer • Bridge

	• Media redirect

IVR

- MCP is involved
- VoiceXML dialog
  - Conditional logic possible thru Attached Data
  - For example, profile information can be passed as data and the VXML app can play the appropriate message or transfer to the appropriate agent based on that information
  - ASR input supported
  - DTMF input supported—**for Chrome only**
- VoiceXML transfer
  - Bridge
  - Media redirect

Other

Call Recording	• Recording audio only • Call from browser to Route Point • Routed to MCP • MCP plays music and records
Hold/Retrieve	• Call routed to Agent (SIP) • Agent puts the caller on Hold • Agent later Retrieves the call • Works fine with SIP Endpoint SDK and Bria • Music and Video on Hold scenario works as well (with MCP) **Note:** Firefox does not support renegotiation. For more information, see Genesys WebRTC Developer's Guide.
Call upgrade/downgrade scenario	• Update the call in progress by starting or stopping the sending of local audio or video • Limited testing done and a demo added with new media session every time a call upgrade or downgrade happens, as Firefox does not

	support renegotiation

Third-Party Call Control (3PCC)

For third-party call control (3pcc), set the configuration options as shown in the following table.

`TServer` section of your SIP Server application object	• `sip-hold-rfc3264`—Enable SIP Hold per RFC 3264. Must be set to `true`.
`TServer` section of your SIP Switch WebRTC Extension DN objects	• `dual-dialog-enabled`—Enable Dual Dialog Mode. Must be set to `false`. • `refer-enabled`—Enable REFER support. Must be set to `false`. • `sip-cti-control`—Enable SIP CTI Control. Must have an empty value when Broadsoft extensions are not used for WebRTC. However, when Broadsoft extensions are used, set this parameter to `talk,hold`.

Reporting Considerations

Genesys WebRTC interactions are treated internally as SIP interactions with SIP Server. Because of this, WebRTC calls will normally be reported as SIP calls when Genesys Info Mart or Genesys Interactive Insights are configured in the deployment.

In order to allow customers to report on these interactions independently, the WebRTC Gateway includes the `reporting-service-type` configuration option in the `[rsmp]` section. The default value for this parameter is `WebRTC`. Once this parameter has been set, the WebRTC Gateway adds a custom SIP header called `X-genesys-service_type`—which contains this value—to each SIP call it makes.

On the SIP Server side, the following two parameters must be set in the `[TServer]` section:

- `userdata-map-all-calls=true`
- `userdata-map-trans-prefix=X-genesys-`

When these parameters have been set, if a SIP call comes to SIP Server from the WebRTC Gateway, it passes the `service_type` obtained from the request to ICON as user data. The standard Genesys Info Mart configuration includes integration of the `service_type` Key-Value pair as Genesys Info Mart's `interaction_descriptor.service_type` field.

SIP Endpoint Configuration

Component	Specific/Preferred Configuration
Bria	• Disable H264 • Enable VP8 **Note:** Bria has issues with H.264 (support for packetization mode, setting H.264 profile level, video quality, and so on), so VP8 is preferred for use with Bria.
Genesys SIP Endpoint SDK	• Supports H.264 (from version 8.5.1), and VP8. Set VP8 priority to 1 (highest). • `auto_accept_video = 1`
Genesys Workspace SIP Endpoint	Supports audio and video

SIP Endpoint Known Issues

- There are compatibility issues with various types of SIP Endpoint, as discussed in the section on SIP Endpoint configuration

Media Codec Considerations

For deployment models that support agents on SIP endpoints, additional measures must be taken to ensure that a WebRTC-enabled browser can interoperate with the SIP endpoint's media codecs. If the SIP endpoint does not support audio or video codecs required by WebRTC browsers, the deployment will require real-time media transcoding. The WebRTC Service has an automatic media transcoder to ensure the media path can be bridged between the browser and a SIP endpoint. It is important to note that media transcoding, especially video transcoding, can be a process-intensive task, so it is recommended that the SIP endpoint be configured with codecs that are compatible with the codecs that have been implemented on the WebRTC side.

Please refer to, Codec Support for a list of codecs that are supported by the Genesys WebRTC Service.

For calls established between two WebRTC browsers, the codecs will always be compatible since the WebRTC specification requires compatible codecs.

Genesys Media Server may also be involved as part of the conversation with a WebRTC browser, in order to serve media such as music or video on hold. Genesys Media Server may serve media files that use any of its supported media formats. If necessary, the media files will be automatically transcoded to a codec that is compatible with the WebRTC browser. As a way to optimize processing, Media Server supports automatic caching of transcoded media, which means that a media file only gets transcoded once. Subsequent calls to the same Media Server instance can reuse the cache without needing to transcode again.

Third-Party Software

The following components are mandatory for running the Genesys WebRTC Service:

Platform	Component Name
Windows	Microsoft Visual Studio 2005 SP1 ATL Security Update
Windows	Microsoft Visual Studio 2008 SP1 ATL Security Update
Linux	GLib 2.14+