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Web Real-Time Communications Deployment Guide

Hardware Sizing Information

12/15/2025

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Hardware Sizing Information

Hardware Sizing and Performance Information

Network Sizing Guidelines

The Genesys WebRTC implementation uses the G.711 and VP8 codecs:

- The G.711 codec is used for audio and requires 64kbps of bandwidth in each direction (incoming and outgoing).
- The VP8 codec is used for video encoding. The bitrate requirement depends on the quality of the streams, starting with a minimum of 100kbps and going up to 2000kbps or more for a single-party HD call. More detail is provided in the following table, which lists bandwidth requirements in kilobits per second.

Video Resolution	Gateway		Browser	
	Incoming	Outgoing	Incoming	Outgoing
SD		256	256	128
HQ		512	512	256
ED		1024	1024	512
HD		2048	2048	1024

Performance Testing Scenarios

Genesys performed load testing for the following hardware and software platforms to create the sizing guidelines for Genesys WebRTC Service 8.5.2.

Important

VGA video resolution was used for this testing.

Performance Testing Configuration

	Linux Virtual Machine	Microsoft Windows Virtual Machine
OS	Red Hat Linux Enterprise Server v6.3x86_64 Kernel 2.6.32-279	Windows Server 2008 R2 Enterprise x64
Processor Type	Intel® Xeon® CPU X5675; 2 vCPUs; hyper-threading disabled	Intel® Xeon® CPU X5675; 2 vCPUs; hyper-threading disabled
Speed	3.06 GHZ	3.06 GHZ

	Linux Virtual Machine	Microsoft Windows Virtual Machine
Memory Size (RAM)	5 GB	5 GB
Hard Disk Space	35 GB	35 GB

Important

Genesys recommends Red Hat Enterprise Linux as the preferred platform for Genesys WebRTC Service.

Performance Test Results

Performance testing was conducted using VGA video resolution.

Important

Due to a Windows memory leak and due to the fact that the Windows version of the Genesys WebRTC Service runs as a 32-bit process (running in compatibility mode for 64-bit Windows), most of the performance testing was done with Red Hat Enterprise Linux. Windows testing is still ongoing, so there are no conclusive test results at this time, but Genesys expects that your results with Windows should be slightly lower than the Linux results and recommends that you limit traffic to 110 simultaneous calls.

Linux Performance Test Results

Description	Max Concurrent Calls	CAPS
Browser-to-browser audio+video without Xcoding	120	1.65
Browser-to-browser audio without Xcoding	190	2.47
Browser-to-SIP endpoint audio+video with Xcoding	190	2.41
Browser-to-SIP endpoint audio with Xcoding	250	3.06
Browser-to-browser audio+video SRTP without Xcoding	120	1.65
Browser-to-SIP endpoint audio+video SRTP without Xcoding	140	1.85
Multiple instances of browser-to-browser audio+video without	240	3.28

Hardware Sizing Information

Description	Max Concurrent Calls	CAPS
Xcoding (Tested with 2 instances. Since larger configurations have not been tested, Genesys recommends scaling horizontally by adding new hosts.)		