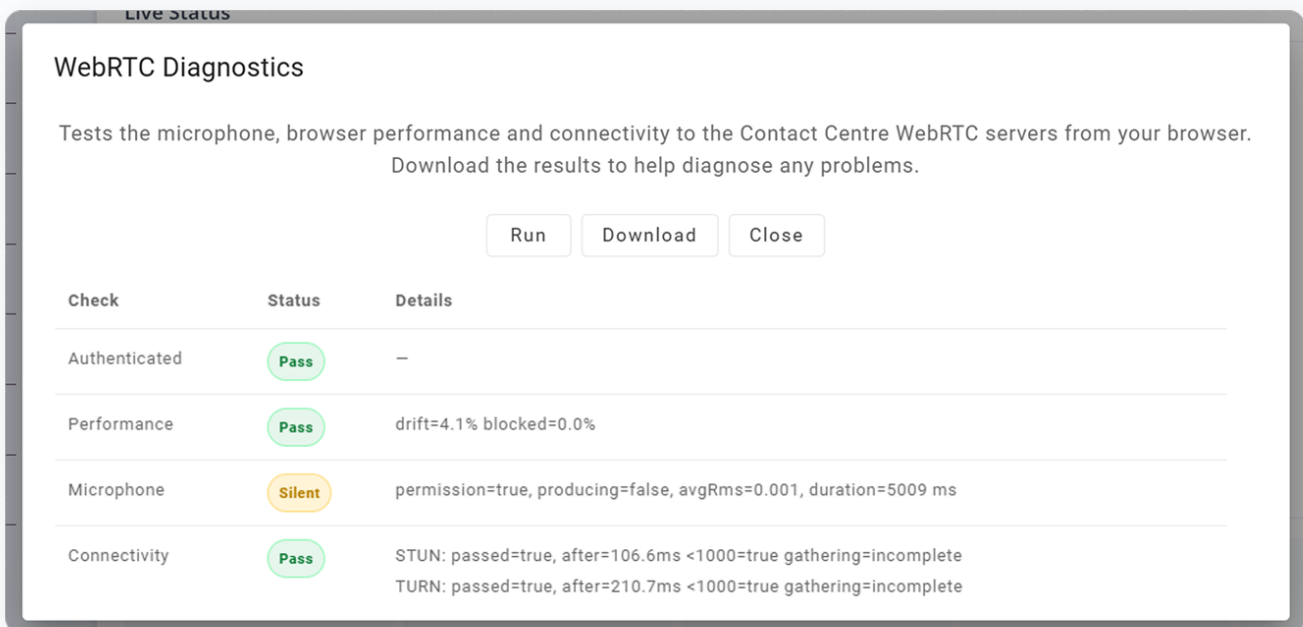












Understanding WebRTC diagnostic tests

The WebRTC diagnostics tool is designed to help agents quickly identify and resolve common issues that may affect their ability to receive or make WebRTC calls. It checks their WebRTC authentication, browser performance, microphone access, and network connectivity to the Contact Centre’s WebRTC servers.

What the Test Checks



Test	What It Does	What to Look For
Authenticate	Confirms your browser is properly authenticated to the WebRTC registration service and has received valid ICE server details.	 Pass - Your session is ready for calls.  Fail - Reload the page and sign in again.
Performance	Measures your browser’s CPU performance and responsiveness.	 Pass - Browser performance is good.  Fail - Browser or system is overloaded; close unused apps or tabs.

Microphone	Checks microphone permission and listens briefly for sound.	<p> Pass - Audio detected.</p> <p> Silent - Permission granted but no sound detected; check mic input or speak during test.</p> <p> Fail - Permission denied; allow mic access and retry.</p>
Connectivity	Tests network reachability of your STUN and TURN WebRTC servers.	<p> Pass - All servers responded correctly.</p> <p> Warn - One path succeeded but another timed out.</p> <p> Fail - No valid ICE candidates; network or firewall issue.</p>

Understanding the Connectivity Test

The connectivity check gathers ICE candidates, which represent possible routes for WebRTC media. You'll see separate results for STUN and TURN servers:

Type	What It Means	Typical Issues
STUN (Server-Reflexive)	Verifies basic UDP connectivity and NAT traversal.	UDP 3478 blocked or STUN server unreachable.
TURN (Relay)	Verifies media can be relayed through the TURN server (used when direct paths fail).	Expired TURN credentials, blocked UDP/TCP 3478 or 5349, strict corporate firewalls.

Healthy networks usually show both srflx (STUN) and relay (TURN) candidates. Partial results may still allow calls but can indicate restricted connectivity.

Useful Resources

- [Diagnosing Call Quality Issues with WebRTC Diagnostics](#)