WebRTC Test Solution

QualityAssurer Series—QA-805

Easy-to-use, high-performance and high-capacity load testing solution to enable WebRTC device, network and service testing during the various stages of development and deployment.

SPEC SHEET

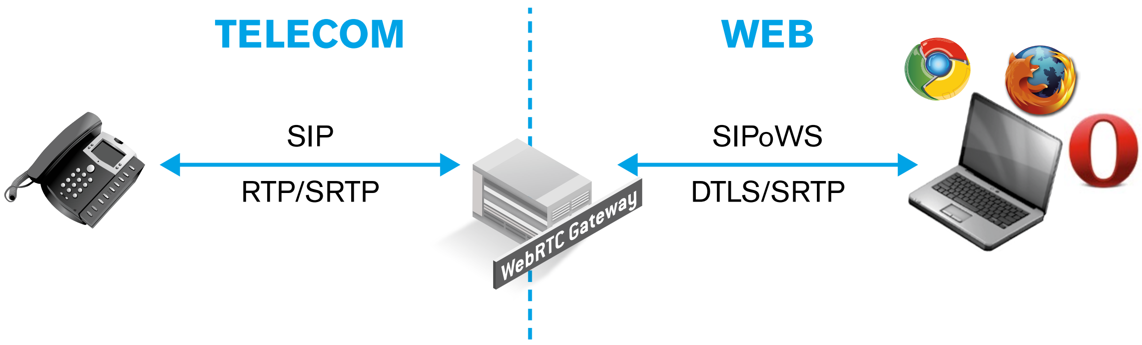
KEY FEATURES

- 10 million WebRTC endpoints
- 5 million TLS and DTLS sessions
- 1.28 million SRTP and SRTCP streams
- TLS over WebSocket and SRTP/SRTCP multiplexing tests
- Real-time QoS metrics for all streams
- Both voice and video QoS metrics
- OPUS and VP8 WebRTC codecs
- NAT/firewall emulation (ICE/STUN)
- ICE-Lite and ICE-Full testing support
- HTTP WebSocket setup and teardown tests
- Distributed-denial-of-service (DDoS) attacks and theft-of-service (ToS) tests
- Media pinhole opening and closing tests
- Policy-and-charging-function (PCRF) emulation
- IMS/VoLTE and VoIP core network emulation
- Mix of VoIP, VoLTE and WebRTC service tests

OVERVIEW

Access to the Internet via web browsers has become a fundamental requirement for both personal and business use. Web real-time communication (WebRTC) is a technology that enables real-time communication via browsers. More than one billion devices, including tablets, smartphones and PCs, already support WebRTC already, and that number is expected to increase to over six billion by 2019.

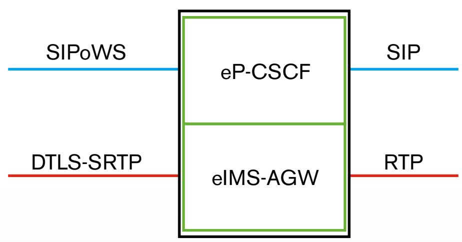
Mobile network operators (MNOs) can capitalize on this opportunity by extending their IP multimedia subsystem (IMS) services to Internet users, thereby enabling end-to-end communication between the IMS and web clients. The interworking between the IMS and web clients is made possible by the WebRTC gateway, which converts the signaling and media between the two technology domains.



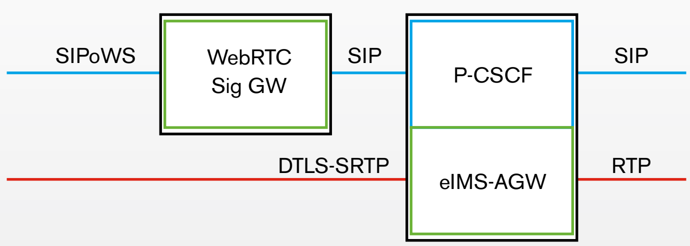
WebRTC GATEWAY IMPLEMENTATION

A WebRTC gateway solution is typically implemented by network equipment manufacturers (NEMs) in one of the two following variations:

1. Session border controller (SBC) supporting both WebRTC signaling (eP-CSCF) and media (eIMS-AGW) functions.



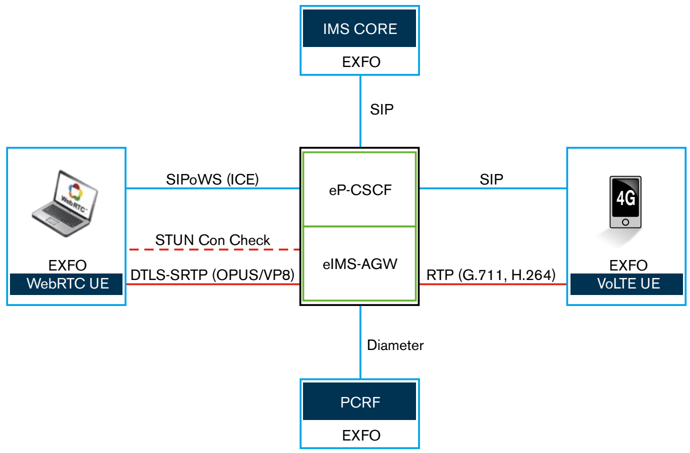
2. SBC supporting only the WebRTC media (eIMS-AGW) function in conjunction with a WebRTC signaling (eP-CSCF) gateway supporting only the signaling function.



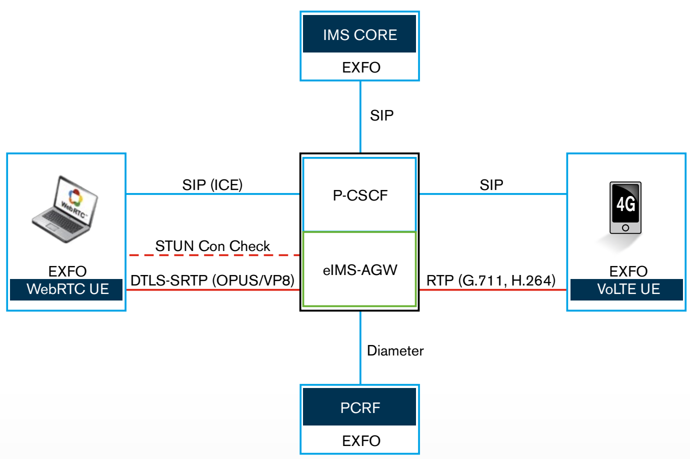
EXFO'S WEBRTC TEST CONFIGURATION

EXFO's QA-805—a high-performance, high-capacity Voice-over-Internet protocol (VoIP), SBC and IMS testing product—supports the comprehensive and flexible set of WebRTC testing applications and features, thereby enabling testing of WebRTC devices and networks for the following test configurations:

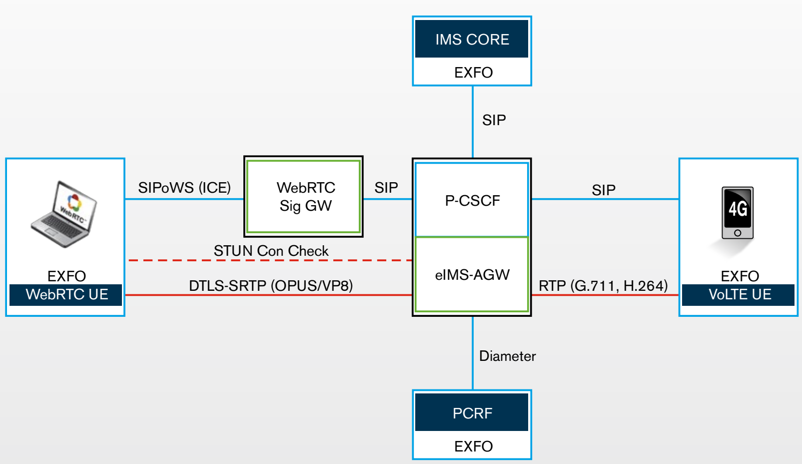
1) SBC supporting both WebRTC signaling (eP-CSCF) and media (eIMS-AGW) functions



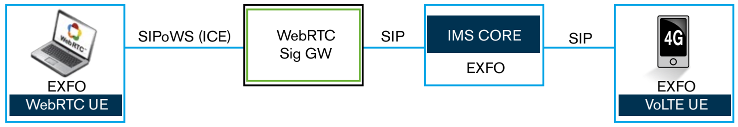
2) SBC supporting only the WebRTC media function (eIMS-AGW)



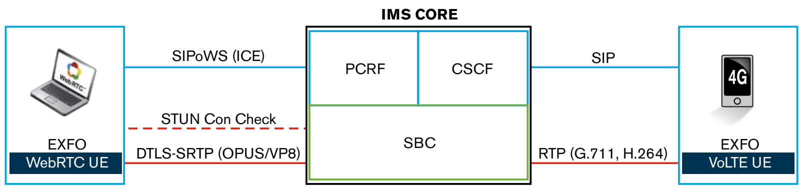
3) SBC supporting only the WebRTC media (eIMS-AGW) function together with a stand-alone WebRTC signaling gateway



4) Stand-alone WebRTC signaling gateway



5) End-to-end network testing with a mix of different types of endpoints, such as WebRTC, Voice over LTE (VoLTE) and business VoIP endpoint



KEY USE CASES
Simulates millions of WebRTC endpoints and other network devices for WebRTC gateway testing, thus enabling selection of the right gateway for WebRTC deployment.
Tests that the WebRTC gateway is able to linearly scale to accommodate growing loads without any performance/capacity degradation.
Tests how many WebSocket and TLS-over-WebSocket connections the WebRTC gateway is able to set up per second, and hold and maintain successfully over a period of time.
Verifies the impact on WebRTC gateway performance as a result of resource-intensive ICE procedures that include prioritization of IP candidates and pair exchange, and orders them into a checklist.
Validates how many concurrent signaling and media sessions WebRTC gateway (data transport layer security-secure real-time transport protocol, DTLS SRTP/SRTCP, and DTLS SRTP/SRTCP multiplexing) can support concurrently.
Verifies resource-intensive session-traversal-utilities-for-NAT (STUN) authentication using short-term ICE attributes (ice-ufrag and ice-pwd) credentials and DTLS fingerprint authentication in order to verify that the certificate presented in the DTLS handshake does not deteriorate the WebRTC gateway performance.
Benchmarks the maximum number of concurrent DTLS-SRTP voice and video media sessions that the WebRTC gateway is able to set up and maintain simultaneously, both with and without transcoding.
Generates and analyzes line-rate (1G and 10G) voice and video streams with a mix of codecs, such as OPUS, VP8, H.264 and G.711.
Checks that the WebRTC gateway is able to deliver the desired quality of service (QoS) with different quality-parameter settings, such as ToS/differentiated services code point (DSCP), VLAN and MPLS.
Performs high-availability tests to determine how resiliently the WebRTC gateway operates under overload conditions, security attacks, card and port failover scenarios.
Exercises the entire operator network infrastructure by testing end-to-end service delivery with a mix of WebRTC, business VoIP, VoLTE and rich-communications-suite (RCS)/message-session-relay-protocol (MSRP) services.

SPECIFICATIONS

Platform	QA-805
Modules and Interfaces	W²CM-10GigE (8 X 1 GigE and 2 X 10 GigE) W²CM-10GigE-Lite (8 X 1 GigE and 2 X 10 GigE) W²CM-4 GigE (4 X 1 GigE) W²CM-Sig (8 X 1 GigE, signaling only)
WebRTC Protocols	SIP over WebSocket, TLS over WebSocket, ICE-Full/Lite, STUN, DTLS-SRTP, SRTP/SRTCP multiplexing
Transport and IP Protocols	UDP, TCP, TLS, SCTP, IPv4, IPv6
WebRTC Codecs	OPUS, G.711, VP8 and H.264
WebRTC Endpoint Capacity	10 million per QA-805 Platform
Number of DTLS-SRTP Sessions	1.28 million concurrent SDES-SRTP and SDES-SRTCP and/or DTLS-SRTP and DTLS-SRTCP streams per QA-805 Platform
Quality Measurements	Voice (ITU-T G.107 E-Model) Video (RFC 4445; VQT MDI) Jitter, loss, delay, etc.
Network Configuration	Unique MAC addresses, VLAN tag, MPLS label, ToS and DSCP settings
Security Testing	DOS, DDOS, Theft of Service, Rouge Media, Pinhole Closure, WebSocket flood, Path Verification, etc.
Interworking	IP (IPv4 to IPv6) Transport (TCP over TLS to TCP/UDP/SCTP) Signaling (SIP over WebSocket to SIP) Media (RTP/RTCP to SDES-SRTP/SRTCP to DTLS-SRTP/SRTCP)
Statistics and Logging	Signaling trace monitor, call records, user-defined KPI, summary and call-flow statistics, table, histogram and chart format, and report generation in HTML and .CSV
Call Profiling	Mix of real-world network-traffic service testing, including VoIP, business VoIP, VoLTE, RCS/MSRP, video, DTMF and more, from a single profile
Negative Testing	Create invalid WebRTC messages, create invalid and error call flow, mix valid and invalid call, WebSocket inactivity, STUN inactivity, connectivity failure, etc.
Automation	TCL command-line interface

ORDERING INFORMATION

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