RESTful HTTP WebRTC Testing

HTTPFLEX TEST APPLICATION

Easy-to-use, high-performance and high-capacity load testing solution to enable OMA RESTful WebRTC-compliant device, network and service testing throughout development and deployment.

SPEC SHEET

KEY FEATURES

- 1.28 million WebRTC endpoints
- 1.28 million TLS and DTLS sessions
- 1.28 million SRTP and SRTCP streams
- HTTP and HTTPs support
- SRTP and SRTCP multiplexing tests
- Real-time voice and video quality of service (QoS) for all streams
- OPUS and VP8 WebRTC codecs
- STUN, TURN and ICE NAT server tests
- Both ICE-Lite and ICE-Full support
- Policy and charging function (PCRF) emulation
- IMS/VoLTE and VoIP core network emulation
- Mix of VoIP, VoLTE and WebRTC service tests

OVERVIEW

Web real-time communications (WebRTC) technology enables real-time communication via browsers. Service providers have deployed WebRTC and are doing trials with the technology to extend their IMS services such as voice over IP (VoIP), voice over LTE (VoLTE), video over LTE (ViLTE) and rich communication services (RCS) to Internet (web) users, hence enabling real-time communication between IMS and web users such as voice, video and chat. The interworking between IMS and web users is made possible by the WebRTC gateway, which translates signaling and transcodes media between these two technology domains.

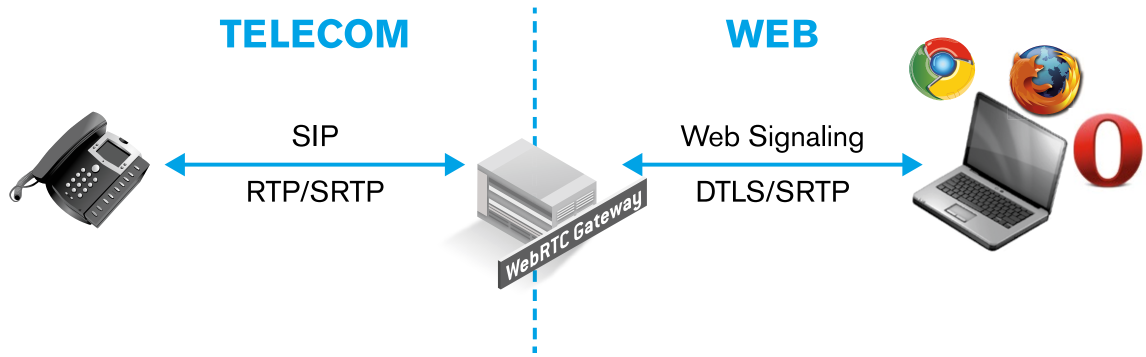
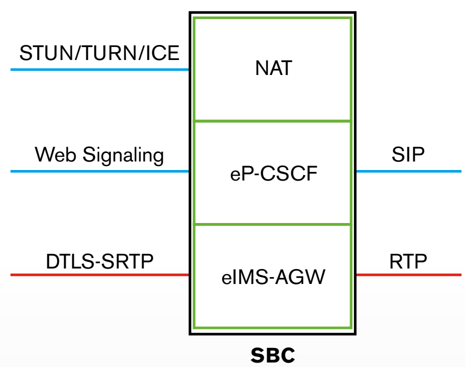


Figure 1. Interworking via WebRTC gateway

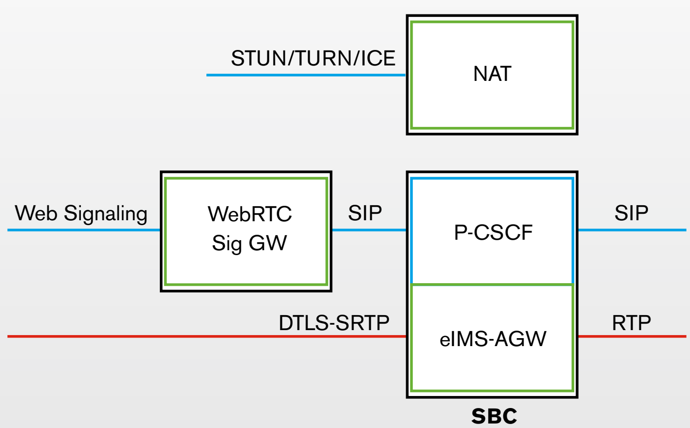
WebRTC GATEWAY

A WebRTC gateway is typically implemented and deployed in one of the two following variations:

1. Session border controller (SBC) supporting WebRTC signaling (eP-CSCF), media (e-IMS-AGW) and NAT (STUN/TURN/ICE) functions.



2. SBC supporting only WebRTC media (eIMS-AGW) function, and a separate device supporting WebRTC signaling (WebRTC signaling gateway) and STUN/TURN/ICE (NAT) functions.



RESTful HTTP

WebRTC specifications do not define standard web signaling protocol, leaving web application architects and developers to decide. The signaling protocol is a requirement to transmit and exchange user device capabilities such as codec and ICE credentials from one endpoint to another in order to set up the endpoint for a communication session. Despite the flexibility and freedom provided to the application designers to choose the signaling protocol, having no standard signaling protocol introduces challenges in delivering interoperable WebRTC services.

To overcome the interoperatbility challenge, Open Mobile Alliance (OMA) has defined a standardized RESTful HTTP network API that allows a a web-user application (e.g., Javascript running in a WebRTC-enabled browser) to signal video, voice and data session over IP with another communication endpoint in the network. It includes common data types, naming conventions, fault definitions and namespaces.

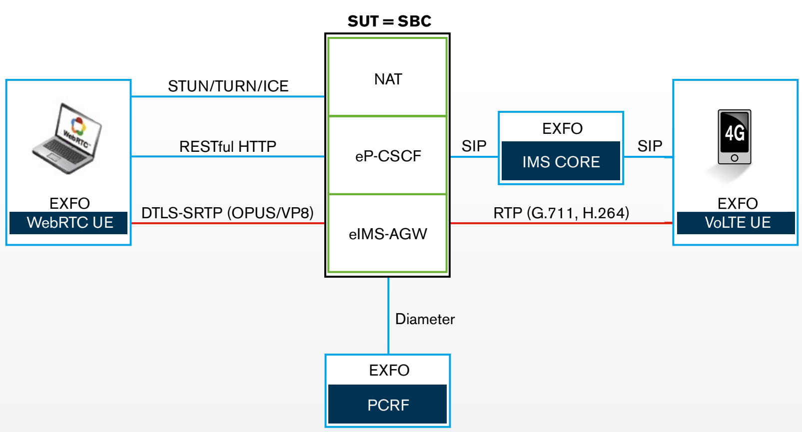
httpFlex Application

The httpFlex application emulates RESTful HTTP WebRTC enpoints in compliance with OMA RESTful Network API for WebRTC signaling v1.0 specifications. Its graphical user interface is flexible and easy to use, enabling customers to use out-of-box pre-canned call flows or to define their own custom call flows. It can be used to simulate real-world normal and abnormal calls during various lifecycle phases from lab testing to live network testing. The httpFlex application runs over the QualityAssurer QA-805 test platform.

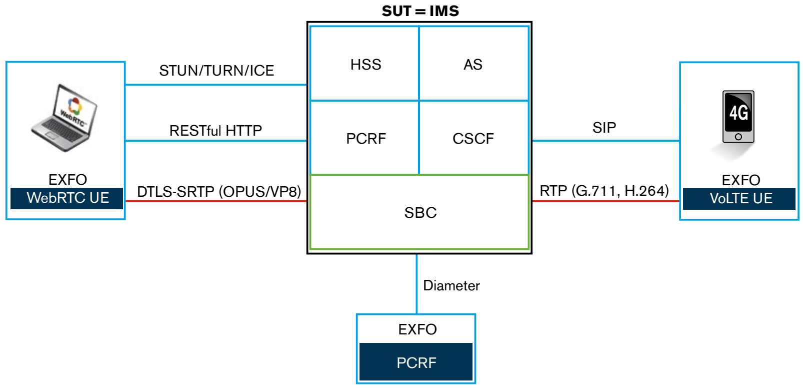
Test Configuration

The QualityAssurer QA-805 test platform—the industry's leading high-performance and high-capacity platform—supports the comprehensive set of applications and features to test the WebRTC gateway as well as the network end to end. The following picture depicts both test configurations:

1) SBC supporting WebRTC signaling (eP-CSCF), media (e-IMS-AGW) and NAT (STUN/TURN/ICE) functions. In this test configuration, the httpFlex application is used together with EXFO's volteFlex, proxyFlex, and hssFlex applications for SBC wraparound testing; httpFlex emulates WebRTC endpoints, volteFlex emulates VoIP/VoLTE/ViLTE endpoints, proxyFlex emulates CSCFs in the IMS core and hssFlex emulates the HSS and PCRF.



2) End-to-end IMS network testing with a mix of different types of endpoints, such as WebRTC, VoLTE and ViLTE endpoint. In this test configuration, httpFlex is used together with the volteFlex and hssFlex applications to test the IMS network; httpFlex emulates WebRTC endpoints, volteFlex emulates VoIP/VoLTE/ViLTE endpoints while hssFlex emulates PCRF.



KEY USE CASES

Simulates millions of WebRTC enpoints 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 scale linearly to accomadate growing loads without any performance/capacity degradation
Tests how many RESTful HTTP sessions the WebRTC gateway is able to set up per second, hold and maintain successfully over a period of time with different notification refresh timeout values
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 order 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) and traversal-using-relays-around-NAT (TURN) 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 expected QoS with different quality-parameter settings, such as ToS/differentiated services code point (DSCP), VLAN and MPLS
Performs high-availability tests to determine resilience of the WebRTC gateway 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, VoIP, VoLTE, ViLTE and RCS/message session relay protocol (MSRP) services

SPECIFICATIONS

Platform	QA-805
Modules and interfaces	W²CM-10GbE (8 X 1 GigE and 2 X 10 GigE) W²CM-10GbE-Lite (8 X 1 GigE and 2 X 10 GigE) W²CM-4GbE (4 X 1 GigE) W²CM-Sig (8 X 1 GigE and 2 X 10 GigE, signaling only)
WebRTC protocols	RESTful HTTP, ICE-Full/Lite, STUN, TURN, DTLS-SRTP, SRTP and SRTCP multiplexing
Transport and IP protocols	TCP, TLS, IPv4, IPv6
WebRTC codecs	OPUS, G.711, VP8 and H.264
RESTful HTTP WebRTC endpoint capacity	1.28 million per QA-805 platform
Number of DTLS-SRTP sessions	1.28 million concurrent 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
Interworking	IP (IPv4 to IPv6) Transport (TCP to TLS to TCP/UDP/SCTP) Signaling (RESTful HTTP to SIP) Media (RTP/RTCP to SDES-SRTP/SRTCP to DTLS-SRTP/SRTCP) Note: For interworking testing volteFlex application is required
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
Negative testing	Create invalid WebRTC messages, create invalid and error call flow, mix valid and invalid call, STUN/TURN inactivity and connectivity failure, etc.
Automation	TCL command line interface

ORDERING INFORMATION

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