SIPREC Working Group Y. Pdut

Internet-Draft NICE Systems  
Intended status: Standards Track   
Expires: <EXPIRE DATE> <DATE>

**WEBRTC Gateway Recording Using SIPREC**

# Abstract

“WEBRTC” is a joint effort of IETF and W3C to define an API and protocols for real time communication between browsers.

Real time communication includes voice calling, video calling, screen sharing and peer to peer data channels.

[RFC 7245] states that session recording is a critical requirement in many communications environments such as call centers and financial trading. In some of these environments, all calls must be recorded for regulatory, compliance, and consumer protection reasons. Recording of a session is typically performed by sending a copy of a media stream to a recording device.

According to [I-D.rtcweb-gateways] WEBRTC gateway is an entity which interconnect between WebRTC endpoints and devices that are not WebRTC endpoints.

This document will try to describe several architectures combining WEBRTC, WEBRTC gateways, and SIP based UC environments, and allow recording of sessions using SIPREC.

# Status of This Memo

This Internet-Draft is submitted in full conformance with the provisions of BCP 78 and BCP 79.

Internet-Drafts are working documents of the Internet Engineering Task Force (IETF). Note that other groups may also distribute working documents as Internet-Drafts. The list of current Internet-Drafts is at http://datatracker.ietf.org/drafts/current/.

Internet-Drafts are draft documents valid for a maximum of six months and may be updated, replaced, or obsoleted by other documents at any time. It is inappropriate to use Internet-Drafts as reference material or to cite them other than as "work in progress."

This Internet-Draft will expire on XXXXXXXX.

# Copyright Notice

Copyright (c) 2015 IETF Trust and the persons identified as the document authors. All rights reserved.

This document is subject to BCP 78 and the IETF Trust's Legal Provisions Relating to IETF Documents (http://trustee.ietf.org/license-info) in effect on the date of publication of this document. Please review these documents carefully, as they describe your rights and restrictions with respect to this document. Code Components extracted from this document must include Simplified BSD License text as described in Section 4.e of the Trust Legal Provisions and are provided without warranty as described in the Simplified BSD License.

# Table of Contents

[**Abstract** 1](#_Toc421711504)

[**Status of This Memo** 1](#_Toc421711505)

[**Copyright Notice** 2](#_Toc421711506)

[**Table of Contents** 2](#_Toc421711507)

[**1. Introduction** 2](#_Toc421711508)

[1.1. WEBRTC Gateway Functions 2](#_Toc421711509)

[**2. Architectures** 3](#_Toc421711510)

[2.1. Combined 3](#_Toc421711511)

[2.2. Distributed 4](#_Toc421711512)

[**3. Informative References** 4](#_Toc421711513)

[**4. Normative References** 4](#_Toc421711514)

[**5. Acknowledgments** 4](#_Toc421711515)

[**Contributors** 4](#_Toc421711516)

[**Author's Address** 5](#_Toc421711517)

# Introduction

## WEBRTC Gateway Functions

WEBRTC gateway have several functions:

As a Media Gateway:

* Convert digital media (SRTP/DTLS streams) between disparate telecommunications networks.
* Transcoding between different transmission and coding.
* Convert between different media transports – TDM/VoIP/Radio

1. As a Signaling Gateway:

* Transfer signaling protocol messages between separate nodes.
* Convert from one network operational paradigm to another
* Convert signaling protocol to other.

As a Security Gateway:

* Connect between service provider's access network and a backbone network to provide service to residential and/or enterprise customers.
* Protect the network and other devices from various attacks.

Basically, in Enterprises environments, WebRTC gateway converts WebRTC proprietary signaling and media streams to UC SIP environment.

## WEBRTC Recording Using SIPREC

WEBRTC gateway is being used in order to connect exiting SIP UC networks to WEBRTC devices.

Session recording is a critical requirement in many communications environments such as call centers and financial trading. In some of these environments, all calls must be recorded for regulatory, compliance, and consumer protection reasons [RFC 7245].

SIPREC is the de facto standard for Enterprise SIP voice and video recording.

SIPREC can be leveraged to record WEBRTC session in one of the following ways:

* Add SIPREC functionality to WEBRTC gateway.
* Use existing SIPREC functionality to Edge servers (like in Session Border Controllers).
* Use existing SIPREC functionality in existing SIP servers (like in Call Center Session Manager or Application Server).

## Mapping between SIPREC and WEBRTC

### Signaling

WEBRTC does not define a specific signaling protocol; instead it allows its users to freely select desired signaling protocol. Even due, there are some variants that are more common than others, like:

* SIP over Web Sockets [RFC 7118]
* Javascript Session Establishment Protocol (JSEP) [I-D-rtcweb-jsep]

### Media

According to [TBD] WebRTC clients are REQUIRED to implement the following audio codecs.

* PCMA/PCMU - 1 channel with a rate of 8000 Hz and a ptime of **20 -** see section 4.5.14 of [RFC3551]
* Telephone Event - [RFC4734]
* Opus [draft-ietf-codec-opus]

# Architectures

Typically there are two potential architectures:

* WebRTC GW as SRC – the GW fork the media directly to SRS. The GW supports SIPREC as a signaling protocol to SRS
* SIP Server as SRC – GW converts signaling to SIP towards SIP server. SIP server acts as SRC.

## WebRTC Gateway as SRC

In this architecture, the WEBRTC gateway, SIP server, and SIPREC SRC are combined into one solution.

The solution includes the following entities:

* WEBRTC End Point – browser or other type of device supporting WEBRTC protocols and API.
* WEBRTC Gateway – convert signaling and media from WEBRTC protocols to UC protocols.
* SIP Server – entry point for the UC network. Can be Session border controller, Session Manager, or other type of SIP server.
* SIPREC SRC - a SIP User Agent (UA) that acts as the source of the recorded media, sending it to the SRS [RFC 7245 - 2].
* SIPREC SRS - a SIP User Agent (UA) that is a specialized media server or collector that acts as the sink of the recorded media [RFC 7245 - 2].
* UC Cloud – Unified enterprise network of devices, servers and other nodes supporting SIP.

The media flow from WebRTC client to WebRTC gateway using WebRTC standard CODECs and WebRTC standard transport layer. The WebRTC gateway convert stream to UC stream – RTP or SRTP and standard CODEC. The media also stream in the SIPREC SRC and forked to SIPREC SRS.

Security

<TBD>

NAT Traversal

<TBD>

## SIP Server as SRC

In this solution, the WEBRTC Gateway converts the incoming WEBRTC signaling stream to SIP protocol. This allows the enterprise to leverage exiting SIP server (such as Session Manager or Session Border Controller). The SIPREC SRC is part of the SIP Server and fork the media the same method like for other SIP streams

The solution includes the following entities:

* WEBRTC End Point – browser or other type of device supporting WEBRTC protocols and API.
* WEBRTC Gateway – convert signaling and media from WEBRTC protocols to UC protocols.
* SIP Server – entry point for the UC network. Can be Session border controller, Session Manager, or other type of SIP server.
* SIPREC SRC - a SIP User Agent (UA) that acts as the source of the recorded media, sending it to the SRS [RFC 7245 - 2]. In this architecture, the SRC is part of the SIP server, and forks the media from SIP SERVER to SIP SRC
* SIPREC SRS - a SIP User Agent (UA) that is a specialized media server or collector that acts as the sink of the recorded media [RFC 7245 - 2].

The media is directed from WEBRTC End Point, using WEBRTC media protocols and CODECS, to WEBRTC Gateway. The WEBRTC Gateway converts the media to SIP network specific media protocols and CODECs and forward the media to SIP SERVER. The SIP Server forward the traffic to UC network. SIP Server also fork received media from both sides (WEBRTC network and UC network) to SIPREC SRC which in turn forward it to SIPREC SRS for recording.

If there is a direct media between WEBRTC and UC SIP network, with termination in UC End Points, the SIP server will not be able to fork media to SIPREC SRS.

### Security

<TBD>

### Nat traversal

<TBD>

## Metadata

<REFER TO SIPREC METADATA>

<What about WEBRTC specific metadata>?

# Informative References

[RFC 7811] I. Baz Castillo, “The WebSocket Protocol as a Transport for the Session Initiation Protocol (SIP)”, [RFC 7811](https://tools.ietf.org/html/rfc7118)

[RFC 7245] A. Hutton, “An Architecture for Media Recording Using the Session Initiation Protocol”, [RFC 7245](https://tools.ietf.org/html/rfc7245)

[RFC 7478] C. Holmberg, “Web Real-Time Communication Use Cases and Requirements”, [RFC 7478](https://datatracker.ietf.org/doc/rfc7478/), March 205

[I-D.rtcweb-overview] H. Alvestrand, “Overview: Real Time Protocols for Browser-based Applications”,   
[draft-ietf-rtcweb-overview-13](https://tools.ietf.org/html/draft-ietf-rtcweb-overview-13)

[I-D.rtcweb-gateways] H. Alvestrand, "WebRTC Gateways",   
[draft-alvestrand-rtcweb-gateways-02](https://tools.ietf.org/html/draft-alvestrand-rtcweb-gateways-02) (work in progress), March 2015.

[I-D-rtcweb-jsep] J. Uberti, “Javascript Session Establishment Protocol”, draft-ietf-rtcweb-jsep-09

[RFC3551]

[RFC4734]

[draft-ietf-codec-opus]

# Normative References

[RFC2119] Bradner, S., "Key words for use in RFCs to Indicate Requirement Levels", BCP 14, RFC 2119, March 1997.

# Acknowledgments

# Contributors

The following individuals contributed to this document (listed in no specific order):

# Author's Address

Yaron Pdut (editor)

NICE Systems

Phone: +972-58-5585822

EMail: yaronp@gmail.com