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WebRTC Gateways
draft-ietf-rtcweb-gateways-00

Abstract

This document specifies conformance requirements for a class of WebRTC-compatible endpoints called "WebRTC gateways", which interconnect between WebRTC endpoints and devices that are not WebRTC endpoints.

Requirements Language

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119 [RFC2119].

Status of This Memo

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1. Introduction

The WebRTC model described in [I-D.ietf-rtcweb-overview] is focused on direct browser to browser communication as its primary use case. Nevertheless, it is clearly interesting to have WebRTC endpoints connect to other types of devices, including but not limited to SIP phones, legacy phones, CLUE-based teleconferencing systems, XMPP-based conferencing systems, and entirely proprietary devices or systems.

WebRTC gateways are a specific type of WebRTC-compatible endpoints which enable the exchange of media streams between WebRTC endpoints on one side, and the other types of devices mentioned above on the other side.

This document describes the requirements that need to be placed on such gateways, both the requirements on WebRTC endpoints that can be relaxed and the additional requirements that need to be applied.

A WebRTC gateway is a WebRTC-compatible endpoint, and will thus not be conformant with all requirements for a WebRTC endpoint (it does not do everything a WebRTC endpoint does), but is able to interoperate with WebRTC endpoints.

1.1. Implications of the gateway environment

A gateway will be limited in the functionality it can offer by the system or class of devices it is gatewaying to. For instance, a gateway into the telephone system will not be able to relay data or video, no matter how much it is required. Therefore, a number of functions that are mandatory to support in WebRTC endpoints are not mandatory on gateways; the requirement on the gateway is that it is able to negotiate those features away correctly.

1.2. Signalling model

The WebRTC model is that signalling is outside the scope of the specification. This document does not change that.

Nevertheless, any practical gateway needs to deal with signalling. For that, this document assumes that the overall system consists of an application running in the WebRTC browser, possibly one or more signalling relays that mediate signalling and thereby enable communication between the application and the gateway, and the actual gateway that is responsible for handling the media flows.

The application, the signalling relays (if any) and the gateway together need to be able to:

- o adhere to the offer/answer semantics

- o deal with the description of configuration coming from the browser; this is specified in SDP format in the WebRTC browser API

- o generate the information that is needed by the browser to set up the session, and express that information in the form of SDP.

The shorthand notation "The gateway MUST/SHOULD/MAY support <SDP function xxx>" used below means that an application running in the Web browser, the signalling relays, and the gateway together MUST/SHOULD/MAY support this functionality; it is not a requirement that this happens at the media gateway itself.

2. WebRTC non-browser requirements that can be relaxed

WebRTC gateways are intended to communicate with WebRTC endpoints. WebRTC gateways are no User Agents. They are therefore expected to conform to the requirements for WebRTC non-browsers in [I-D.ietf-rtcweb-overview], with the exceptions defined in this section.

A WebRTC gateway which is expected to be deployed where it can be reached with a static IP address (as seen from the client) does not need to support full ICE; it therefore MAY implement ICE-Lite only.

ICE-Lite implementations do not send consent checks, so a gateway MAY choose not to send consent checks too, but MUST respond to consent checks it receives.

A gateway is expected to not need to hide its location, so it does not need to support functionality for operating only via a TURN server; instead it MAY choose to produce Host ICE candidates only.

If a gateway serves as a media relay into another RTP domain, it MAY choose to support only features available in that network. This means that it MAY not (need to) support Bundle and any of the RTP/RTCP extensions related to it, RTCP-Mux, or Trickle Ice. However, the gateway MUST support DTLS-SRTP, since this is required for interworking with WebRTC endpoints.

If a gateway serves as a media relay into a network or to devices not implementing the WebRTC Datachannel, it MAY choose to not support the Datachannel.

3. Additional WebRTC gateway requirements

(nothing yet)

4. IANA Considerations

This document makes no request of IANA.

Note to RFC Editor: this section may be removed on publication as an RFC.

5. Security Considerations

A WebRTC gateway may operate in two security modes: Security-context termination and security-context relaying.

Relaying is only possible where signed and encrypted content can be passed through unchanged, and where keys can be exchanged directly between the endpoints.

When the gateway terminates the security context, it means that the WebRTC user has to place trust in the gateway to perform all verification of identity and protection of content in the realm on the other side of the gateway; there is no way the end-user can detect a man-in-the-middle attack, an identity spoofing attack or a recording done at the gateway. For many scenarios, this is not going to be seen as a problem, but needs to be considered when one decides to use a gatewayed service.

6. Acknowledgements

Several comments from Christer Holmberg and Andrew Hutton were included.

7. Change history

Changes from draft-alvestrand-rtcweb-gateways-00

o Aligned terminology with draft-rtcweb-overview-12

o Rewrote text on signaling to improve clarity

o Editorial nits

Changes from draft-alvestrand-rtcweb-gateways-01 o Aligned terminology with draft-rtcweb-overview-13 ("non-browser")

o Nits

Changes from draft-alvestrand-rtcweb-gateways-02

o Re-submitted as WG draft

o Addressed a comment from Andrew Hutton that deployment in open internet is an option, not a fact.

8. Normative References

[I-D.ietf-rtcweb-overview]
Alvestrand, H., "Overview: Real Time Protocols for Browser-based Applications", draft-ietf-rtcweb-overview-13 (work in progress), November 2014.

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

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