Institution: University of Glasgow (UofG)		
Unit of Assessment: UoA11 Computer Science and Informatics		
Title of case study: WebRTC standards improve interactive audio/visual communications via the world-wide web		
Period when the underpinning research was undertaken: 2011-2020		
Details of staff conducting the underpinning research from the submitting unit:		
Name(s):	**Role(s) (e.g. job title):**	**Period(s) employed by submitting HEI:**
Dr. Colin Perkins Dr. Stephen McQuistin	Senior Lecturer Research Assistant; Research Associate	2003–present 2018–2020; 2020–present
Period when the claimed impact occurred: 2012 onwards		
Is this case study continued from a case study submitted in 2014? N		

1. Summary of the impact

WebRTC (Web Real-time Communication) enables real-time interactive video conferencing for the first time as a *native* component of web browsers. Dr Perkins' research developed circuit breakers and congestion control feedback to ensure WebRTC is safe to deploy and has good performance. It has been incorporated into the international technical standards for WebRTC, helping browsers and other video conferencing systems recognise and react to network congestion, adapting video quality to match network capacity. Global deployment of WebRTC-based video conferencing has transformed education, telemedicine, business, and family communications — a transformation which has accelerated during the COVID-19 pandemic. Dr Perkins' work helps ensure such use does not overwhelm the network.

2. Underpinning research

WebRTC extends modern web browsers, adding new network protocols and programming interfaces to support secure, high-quality, peer-to-peer, real-time interactive voice, video, and data. This allows new classes of interactive audio/visual web applications, including high-quality video conferencing, to be developed and deployed within any web browser.

This innovation comes with a risk, however. High-quality interactive video needs significant network resources, and rapid deployment of such web applications has potential to congest the network due to the increased traffic demand, degrading the user experience and disrupting other Internet applications.

Research at Glasgow focussed on removing that risk through the development of *circuit breaker* algorithms that monitor progress of an ongoing video call and its impact on other traffic, halting the call if it is causing congestion that will disrupt other users of the network. These algorithms monitor reception quality feedback provided by the WebRTC Real-time Transport Protocol (RTP) and determine if the video is causing persistent and excessive congestion or network disruption based on heuristics derived from knowledge of user perception of video quality, video coding, network dynamics; and behaviour of network transport protocols (e.g., RTP, UDP, TCP/IP).

Using the information obtained, the algorithms determine if it is necessary to reduce the sending rate or terminate the flow, breaking the circuit. This research by Perkins, Stephen McQuistin and Martin Ellis (PhD student), in Glasgow, in collaboration with Aalto University, the University of Aberdeen, and Ericsson Research, was published in the International Packet Video workshop (outputs [3.1], [3.2]), and an IEEE (Institute of Electrical and Electronics Engineers) Infocom workshop [3.3]. Perkins led the incorporation of this research into the WebRTC standards developed by the Internet Engineering Task Force (IETF). The IETF is the leading international standards organisation that develops the technical standards that define the internet (IETF developed the network protocol aspects of WebRTC; the World-Wide Web Consortium (W3C) integrated with the web programming model).

WebRTC also uses detailed *congestion control feedback* to adapt video quality to match available network capacity, rather than simply terminating problematic flows. Devising scalable feedback algorithms that can operate across the range of scenarios where WebRTC is used (from point-to-point phone calls to high-quality multiparty interactive video conferences) is difficult due to timing constraints inherent in the underlying RTP protocol, the limited adaptation rate of video codecs, and limits on the feedback overhead that can be tolerated. Research by Perkins and McQuistin showed how to adopt *Explicit Congestion Notification* from the network for use with the UDP transport underpinning WebRTC [3.4], and in collaboration with researchers from Ericsson and the University of Oxford, Perkins worked to incorporate this into the WebRTC standards.

Finally, Perkins' team developed *scalability improvements* for the network protocols used in WebRTC to make it feasible to deploy on older 3G/4G wireless links, and in developing countries with low-capacity residential Internet infrastructure. These technical measures adapt the feedback and media multiplexing algorithms in RTP when used for WebRTC, reducing cross-reporting to lower overhead, and reducing the number of network ports used to ease NAT traversal and simplify deployment.

3. References to the research

Outputs (available via links, or on request from HEI):

3.1 Varun Singh, Stephen McQuistin, Martin Ellis, and Colin Perkins, Circuit Breakers for Multimedia Congestion Control, Proceedings of the 20th International Packet Video Workshop, San Jose, CA, USA, December 2013. doi:10.1109/PV.2013.6691439
3.2 Nazila Fough, Fabio Verdicchio, Colin Perkins, and Gorry Fairhurst, Media Usability Circuit Breakers for RTP-Based Interactive Networked Multimedia, Proceedings of the 21st International Packet Video Workshop, Cairns, Australia, June 2015. doi:10.1109/PCS.2015.7170080
3.3 Zaheduzzaman Sarker, Varun Singh, and Colin Perkins, An Evaluation of RTP Circuit Breaker Performance on LTE Networks, Proceedings of the IEEE Infocom Workshop on Communication and Networking Techniques for Contemporary Video, Toronto, Canada, April 2014. doi:10.1109/INFCOMW.2014.6849240
3.4 Stephen McQuistin and Colin Perkins, Is Explicit Congestion Notification usable with UDP?, Proceedings of the ACM Internet Measurement Conference, Tokyo, Japan, October 2015. doi:10.1145/2815675.2815716. This was incorporated into the following IETF standards document: Magnus Westerlund, Ingemar Johansson, Colin Perkins, Piers O'Hanlon, and Ken Carlberg, Explicit Congestion Notification (ECN) for RTP over UDP, IETF, RFC 6679, August 2012 (doi:10.17487/RFC6679)

Grants (PI name, funder name, amount, start and end date):

3.5 Project #58711 with Dr Colin Perkins as PI, funded by Ericsson (GBP50,000), 2012–2014.
3.6 Project #55349 with Dr Colin Perkins as PI, funded by Huawei (EUR216,000), 2011–2017.

4. Details of the impact

Context:

The IETF (https://www.ietf.org/) is a key international organisation developing technical standards for the Internet. It has convened since 1986 as an open community of network operators, vendors, and researchers concerned with smooth operation and evolution of the Internet. IETF standards, known as RFCs, are widely implemented, and describe critical infrastructure components of the Internet. Dr Perkins has been involved in IETF since the mid-1990s, co-authoring >30 RFCs, chairing the Audio/Video Transport (1998–2008), Multiparty Multimedia Session Control Working Group (2000–2007), and Real-time Media Congestion Control Working Groups (2016–). He chairs the Internet Research Task Force (https://irtf.org/), the research arm of IETF, and is a member of the Internet Architecture Board (https://iab.org/)

WebRTC (https://webrtc.org/) is a free, open source, browser extension that brings *native* (built-in) real-time audio-visual conferencing to the web for the first time. Developed by IETF and the World Wide Web Consortium (W3C), WebRTC comprises network protocol extensions to transport interactive multimedia traffic and JavaScript APIs to allow applications to use these features. From the first cross-browser video call in February 2013, WebRTC has been adopted by *all* major web browsers, desktop and mobile, supporting applications including Facebook

Messenger, Google Meet and Hangouts, Discord, Snapchat, Cisco WebEx, Microsoft Teams, and Skype.

Development of circuit breakers and congestion control feedback

Dr Perkins led development of *circuit breakers* and *congestion control feedback* to ensure that WebRTC is safe to deploy, made *scalability improvements* to make WebRTC feasible to deploy on older 3G/4G wireless links and poor-quality residential networks, and was lead author of the WebRTC Media Transport protocol specification in IETF.

Dr Perkins led development of the circuit breaker algorithm [3.1−3.2], worked with Varun Singh (CEO/founder of callstats.io, the leading performance monitoring and management service for WebRTC) and Zahed Sarker at Ericsson Research to validate its behaviour [3.3], and brought the research results into the IETF Audio/Video Transport Core Maintenance working group, leading to publication of the *circuit breaker extensions* as an IETF standard in March 2017 [5.1]. Through the WebRTC working group, Dr Perkins gained consensus that the circuit breaker should be implemented in WebRTC applications. The circuit breaker is referenced as "MUST implement" in the WebRTC media transport specification (DOI:10.17487/RFC8834) [5.8]. Independently, the Real-time Streaming Protocol standard (DOI:10.17487/RFC7826), a core component of the 3GPP Multimedia Broadcast/Multicast Service, cited a draft of the circuit breaker as "MUST implement" [5.1].

Circuit breakers are an essential backstop to protect the network from errant applications, ensuring user experience is maintained, and that growing deployment of interactive real-time media doesn't disrupt other traffic. More sophisticated applications adapt media quality to the available network capacity, rather than using the all-or-nothing approach provided by the circuit breaker. Dr Perkins co-chairs the Real-time Media Congestion Avoidance Techniques working group in the IETF (https://datatracker.ietf.org/wg/rmcat/about/), developing congestion control algorithms for future versions of WebRTC, and was a key member of the design team that developed *congestion control feedback* for WebRTC to allow interoperability of congestion control algorithms from different vendors, and performed simulations to demonstrate its effectiveness and low-overhead [5.5-5.6]. Dr Perkins' work on Explicit Congestion Notification [3.4] is seeing interest for future versions of WebRTC [5.5] and was also published as an IETF standard.

Dr Perkins was editor of the IETF WebRTC media transport standard [5.7] that incorporates these results and extensions, and has been a long-term editor of the Session Description Protocol [5.8] signalling standard. These are two of the mandatory-to-implement core specifications that define WebRTC.

Development of internet-based communication for low-capacity links

Dr Perkins collaborated with engineers from Ericsson, Huawei, and Vidyo to develop extensions to the RTP standards that improve its scalability and reduce overheads, allowing WebRTC to work effectively on low-capacity links, including dial-up connections in developing regions, and on poor quality 3G/4G wireless links. These have enhanced the reach of WebRTC, allowing it to compete with traditional telephony in developing regions, and providing a more robust user experience in wireless environments [5.2-5.4].

Circuit breaks and congestion control feedback have facilitated the growth of WebRTC

WebRTC is integral to a new ecosystem of applications and services such as Skype for Business, Zoom, and Google Duo, facilitating communication in areas such as education, telemedicine and business. A Google news search for WebRTC finds 71,400 articles relating to the technology, its applications, and companies building on the platform. For example, Attend Anywhere has led in the transformation of healthcare delivery across the UK and Australia, initially using Vidyo, and then WebRTC from 2014 to provide high-quality secure video consultations across the healthcare sector [5.9].

The global restriction in travel in response to the coronavirus pandemic combined with the cross-platform browser capability of WebRTC has facilitated real-time audio-visual communication for business, healthcare and education across the Internet [5.10f]. In the first 4 months of the COVID-19 pandemic, real-time voice and video has grown more than 200% in traffic and daily conferencing minutes together with 20-fold increases in users of conferencing platforms [5.10a]. In April 2020, Google announced that over 2 million new users were connecting on Google Meet, collectively spending more than 3,800 years of secure meetings each single day [5.10b]. This has continued to be supported as people work and learn from home, with 100,000 schools across 20 countries connecting using Zoom [5.10c] . The UofG work on circuit breakers, congestion control feedback, and scalability for WebRTC have helped the network adapt to these changes [5.10d], allowing businesses to connect with customers, patients with their doctors [5.9], students with their teachers, and friends and families across the globe to stay in touch during the global pandemic. The resultant global market is anticipated to reach USD50 billion by 2026 [5.10e].

5. Sources to corroborate the impact

Links to IETF standards documents are included below. The IETF publishes detailed proceedings, presentations, working documents, and video recordings of meetings in an open access manner on their website [5.11], so Dr Perkins' participation in leading the development is a matter of public record. The following IETF members may be contacted for testimony describing Dr Perkins' contribution to IETF:

- CTO of Collaboration, Cisco Systems, Inc., and former chair of the IETF WebRTC Working Group
- Area Director of Transport Area, Ericsson Research
- CEO/founder of callstats.io

5.1 C. S. Perkins and V. Singh, Multimedia Congestion Control: Circuit Breakers for Unicast RTP Sessions, Internet Engineering Task Force, RFC 8083, 2017 (DOI:10.17487/RFC8083).

A draft version of this standard is cited as "MUST implement" in *the Real-Time Streaming Protocol, v2.0*, standard (DOI:10.17487/RFC7826) and the final version is similarly cited in the standard for *Media Transport and Use of RTP in WebRTC* [5.8]. This standard is cited as a Best Current Practice in the *Network Transport Circuit Breakers* (DOI:10.17487/RFC8084) and *UDP Usage Guidelines* (DOI:10.17487/RFC8085). The IETF standards for transport of VC-2 High Quality Video (DOI:10.17487/RFC8450), Mixed Excitation Linear Prediction Enhanced speech codec (DOI:10.17487/RFC8130), Tactical Secure Voice Cryptographic Interoperability Specification speech codec (DOI:10.17487/RFC8817), the Timed Text Mark-up Language (DOI:10.17487/RFC8759), and ISO/IEC 21122 (JPEG XS) video mandate, or recommend in one case recommend, use of the circuit breaker.

5.2 J. Lennox, M. Westerlund, Q. Wu, and C. Perkins, Sending Multiple RTP Streams in a Single RTP Session, Internet Engineering Task Force, RFC 8108, 2017 (DOI:10.17487/RFC8108).

5.3 J. Lennox, M. Westerlund, Q. Wu, and C. S. Perkins, Sending Multiple Media Streams in a Single RTP Session: Grouping RTCP Reception Statistics and Other Feedback, Internet Engineering Task Force, RFC 8861, Approved March 2016, Published January 2021 (https://mailarchive.ietf.org/arch/msg/ietf-announce/jbYr09VKagqXceJe9XdFE3C0Sew/) (DOI:10.17487/RFC8861).

5.4 M. Westerlund, C. S. Perkins, and J. Lennox, Sending Multiple Types of Media in a Single RTP Session, Internet Engineering Task Force, RFC 8860, Approved Nov. 2020, Published Jan. 2021 (https://mailarchive.ietf.org/arch/msg/avt/UuyNF3ZUTcOfRdQ_P96BB0fNOhU) (DOI:10.17487/RFC8860).

5.5 Z. Sarker, C. S. Perkins, V. Singh, and M. A. Ramalho, RTP Control Protocol (RTCP) Feedback for Congestion Control, Internet Engineering Task Force, RFC 8888, Approved Nov. 20201 (https://mailarchive.ietf.org/arch/msg/avt/UuyNF3ZUTcOfRdQ_P96BB0fNOhU), Published January 2021 (DOI:10.17487/RFC8888).

5.6 C. S. Perkins, RTP Control Protocol (RTCP) Feedback for Congestion Control in Interactive Multimedia Conferences, Internet Engineering Task Force, Work in progress (draft-ietf-rmcat-rtp-cc-feedback).

5.7 C. S. Perkins, M. Westerlund, and J. Ott, Media Transport and Use of RTP in WebRTC, Internet Engineering Task Force, RFC 8834, Approved July 2015, Published January 2021

(http://mailarchive.ietf.org/arch/msg/ietf-announce/4edyPGe3Q8wDVneRiVHEXLMF3GM/) (DOI:10.17487/RFC8834). This is the core specification for delivery of audio-visual media data in WebRTC, and is mandatory to implement by all implementations of WebRTC.

5.8 A. Begen, P. Kyzivat, C. S. Perkins, and M. Handley, SDP: Session Description Protocol, Internet Engineering Task Force, RFC 8866, Approved August 2019, Published January 2021 (https://mailarchive.ietf.org/arch/msg/ietf-announce/qRxLtmbDkaQvDNvtgyER7t2Rpkk/) (DOI:10.17487/RFC8866). This is an update to DOI:10.17487/RFC4566, which has ~3,900 citations in Google Scholar.

5.9 Corroborating testimony available from CTO, Attend Anywhere. www.attendanywhere.com/journey.html

5.10 Blog links relating to internet usage in the pandemic:
  a. https://blog.zoom.us/a-message-to-our-users/
  b. https://cloud.google.com/blog/products/g-suite/how-google-meet-supports-two-million-new-users-each-day
  c. https://blog.zoom.us/navigating-a-new-chapter-for-zoom/
  d. http://www.circleid.com/posts/20200707-evolving-the-internet-through-covid-19-and-beyond/
  e. https://www.gminsights.com/industry-analysis/video-conferencing-market?utm_source=PrNewswire.com&utm_medium=referral&utm_campaign=Paid_PrNewswire
  f. https://www.ietf.org/blog/webrtc-pandemic/

5.11 IETF working groups and meeting minutes:
  a. IETF RTCWEB working group: https://datatracker.ietf.org/wg/rtcweb/about/ (the standard collectively known as WebRTC comprises two parts: the network protocols developed by the IETF RTCWEB working group, and the API developed by the W3C WebRTC working group developed the API, for the standard collectively known as WebRTC)
  b. IETF Audio/Video Transport Core Maintenance (AVTCORE) working group: https://datatracker.ietf.org/wg/avtcore/about/ (the AVTCORE working group develops the core media transport protocol standards; RTCWEB standards reference these for details of real-time media transport protocols).
  c. IETF Real-time Media Congestion Avoidance Techniques (RMCAT) working group: https://datatracker.ietf.org/wg/rmcat/about/ (the RMCAT working group develops media rate adaptation and congestion control algorithms; the RTCWEB standards reference these to specify congestion control).
  d. Materials for older meetings are linked from the Proceedings page at https://www.ietf.org/how/meetings/past/ – much of Dr Perkins' work on WebRTC media transport, circuit breakers, and congestion feedback is in this category, and includes presentations and discussion in the RTCWEB working group at IETF meetings 81, 84-87, 89-90, and 94, the AVTCORE working group at IETF meetings 84-87, 89-94, 96, and 99-106, the RMCAT working group at IETF meetings 97, 99, 101-102, and 104-106.