Institution: University of Glasgow

Unit of Assessment:
B11: Computer Science and Informatics

Title of case study:
Development of International Standards which have fuelled the rapid global development of telecommunications technology

1. Summary of the impact

As a key participant in the Internet Engineering Task Force (IETF), Dr Perkins has been instrumental in developing key protocol standards that underpin modern telecommunications. The Real-time Transport Protocol (RTP) acts as a transport layer distributing audio-visual data across the network, whilst the Session Description Protocol (SDP) describes the format and destination of streaming media. These standards are essential components of 3G and 4G mobile phone standards and form the infrastructure for many fixed telephone networks. They are implemented in Apple’s Mac OS X and iOS, Google’s Android, and Microsoft Windows, and feature in billions of devices around the world.

2. Underpinning research

It became clear in the late-1990s that building separate, highly optimised networks to support voice telephony, video and television distribution and data traffic was unnecessary and uneconomic. A single converged data network could support all three classes of traffic. This would enable innovation by providing increased flexibility for application designers, and would lower costs by removing duplicate functionality and simplifying network management. Dr Perkins has been working at Glasgow University since 2003.

Dr Perkins’ work has been in support of that vision, developing robust, adaptive transport protocols for real-time networked multimedia that allow such applications to make effective use of the Internet as such a converged network substrate, and their associated signalling protocols. There are three main threads to this research:

1. Measuring and monitoring network performance to understand its impact on applications. The Internet provides a service that is usable, but not optimised for, real-time multimedia, and it is important to understand the limitations of that service. This research included measuring and modelling variation in packet timing, including dispersion and reordering, in networks used for transmission of high-quality interactive and streaming video. In work carried out at the University of Glasgow, 2007-12, funded by Cisco, Dr Perkins showed the suitability of the Real-time Control Protocol (RTCP) for monitoring and fault isolation in commercial Internet television (IPTV) systems, demonstrating that it can scale to support these deployments [4]. University of Glasgow researchers also conducted a sizeable measurement study of ADSL and cable link performance and its effects on IPTV showing that widely-used Internet loss models are not accurate, often greatly underestimating the effects of loss, and introducing a new two-level Markov loss model for such links [1].

2. Developing techniques to improve reliability and quality of real-time networked multimedia. Work at the University of Glasgow (2007-12; funding by Cisco) studied the effects of residential link impairments on application-layer forward error correction for IPTV [3] and virtualisation of RTCP feedback to ease transition from legacy cable TV to IPTV [2].

3. Congestion control mechanisms for adaptive multimedia. With funding from Microsoft Research (2003, at USC/ISI and Glasgow), and the NSF (2002–06, jointly at the University of Glasgow and USC/ISI) researchers studied the Transmission Friendly Rate Control algorithm, developing one of the first real-world implementations of this algorithm, and demonstrating that while the algorithm can work well on long-distance high-rate paths, stable implementations are infeasible for paths where the network round-trip time is close to the operating system timer limits [5]. This experience led to the ongoing (2012-) development of a circuit breaker algorithm for interactive multimedia traffic [6], intended to allow the widespread safe deployment of high-quality video conferencing.

3. References to the research

1. Martin Ellis, Dimitrios P. Pezaros, Theodore Kypraios, and Colin Perkins, Modelling Packet Loss in RTP-based Streaming Video for Residential Users, Proceedings of the 37th IEEE Conference on Local Computer Networks, Clearwater, FL, USA, October 2012. DOI:10.1109/LCN.2012.6423613 *
2. Alejandra Soni García, Jörg Ott, Martin Ellis, and Colin Perkins, Virtual RTCP: A Case Study of Monitoring and Repair for UDP-based IPTV Systems, Proceedings of the 19th International Packet Video Workshop, Munich, Germany, May 2012. Best student paper. DOI:10.1109/PV.2012.6229743 [REF2] *
3. Martin Ellis, Dimitrios Pezaros, and Colin Perkins, Performance Analysis of AL-FEC for RTP-based Streaming Video Traffic to Residential Users, Proceedings of the 19th International Packet Video Workshop, Munich, Germany, May 2012. DOI:10.1109/PV.2012.6229737
4. Ali C. Begen, Colin Perkins, and Jörg Ott, On the Scalability of RTCP-Based Network Tomography for IPTV Services, 7th IEEE Consumer Communications and Networking Conference, Special Session on IPTV Toward Seamless Infotainment, Las Vegas, NV, USA, January 2010. DOI:10.1109/CCNC.2010.5421780 [REF2] *
5. Ladan Gharai and Colin Perkins, Implementing Congestion Control in the Real World, Proceedings of the IEEE International Conference on Multimedia and Expo, Lausanne, Switzerland, August 2002. DOI:10.1109/ICME.2002.1035802
6. Varun Singh, Stephen McQuistin, Martin Ellis, and Colin Perkins, Circuit Breakers for Multimedia Congestion Control, To appear in Proceedings of the 20th International Packet Video Workshop, San Jose, CA, USA, December 2013. [Available from HEI]

* best indicators of research quality

4. Details of the impact

The last 15 years have seen enormous changes in telecommunications, with global impact. The rise of voice-over-IP (VoIP) has caused the wholesale replacement of large parts of the traditional telephone network and the rapid deployment of 3G and 4G mobile telephony systems that embrace VoIP building on the 3rd Generation Partnership Project IP Multimedia Subsystem. We are seeing similar changes in other industries: video conferencing and telepresence is now commonplace; IP-based TV set-top boxes are replacing traditional cable TV systems; digital distribution is disrupting the cinema industry; and IP-based distribution is widely used for surveillance cameras and security systems. Perkins' work has been to build, and improve, protocol standards that allow vendors to make interoperable systems, to help build and move towards a single protocol framework for internet-based real-time multimedia, rather than balkanised non-interoperable proprietary systems.

Two network protocol standards underpin the overwhelming majority of these deployments. The Real-time Transport Protocol (RTP) provides the transport layer that delivers audiovisual data across the network, while the Session Description Protocol (SDP) forms the basis of the signalling mechanisms that describe the format of the media data and the destination of the media flow. The Internet Engineering Task Force (IETF), the main international technical standards body governing the Internet, developed both RTP and SDP.

The RTP and SDP protocol standards are essential components of 3G and 4G mobile phone standards [9], implemented in hundreds of millions of devices around the world. They form the infrastructure for many fixed telephone networks, as part of equipment that is rapidly replacing the traditional circuit switched telephone network. They are implemented in Apple's Mac OS X and iOS, in Google's Android operating system, and in Microsoft Windows as core telephony and video conferencing frameworks [10]. They are implemented in TV set-top boxes distributed by many ISPs. And they form the basis of many commercial security camera systems, and other surveillance applications.

As co-chair of the IETF's Audio/Video Transport Working Group from 1998-2008, Dr Perkins managed the evolution of RTP from a proposed standard protocol, primarily of interest to researchers and with minimal deployment, to a full Internet standard [7] with numerous commercial implementations that is now deployed in billions of devices worldwide. He has contributed to numerous standards relating to robust multimedia transport, media quality feedback and monitoring, and security of voice telephony, video conferencing, and IPTV systems, and has written the definitive book on the protocol [8]. Research at Glasgow extended early standards on robust VoIP transmission [RFC2198] with more recent standards for reliable text conversation for hearing-impaired users [RFC4103], for forward-shifted redundancy [RFC6354], and in-progress work on temporal and spatial redundancy for video streaming. University of Glasgow work on monitoring fed into two international standards [11, 13] that provide guidelines for the development of monitoring features of RTP, and further standards [12, RFC4585] optimising those monitoring features. Working with industry to transfer these ideas, the University of Glasgow are credited in the development of a further nine monitoring standards [RFCs 6776, 6798, 6843, 6958, 6990, 7002-7005], and research on scalability of the RTCP monitoring framework for IPTV systems ([4], above) showed that such monitoring can scale to multimillion-user populations, supporting adoption by widely deployed commercial IPTV set-top box implementations.

Dr Perkins served as co-chair of the IETF's Multiparty Multimedia Session Control working group from 2000-07. One of his main contributions in this time was to act as editor for a significant revision and clarification to the SDP signalling standard *(2,861 citations in Google Scholar since the 2006 version [14]; updated version in progress, aiming for publication late early 2014).* This has a successful aside, based on the need for signalling to support research in IPTV, robust media transport, and congestion control.

Since 2011, University of Glasgow researchers contributed to the Web Real-time Conferencing working group of the IETF and the associated working group in the World-Wide Web Consortium (W3C). These groups are bringing standard native video conferencing features to web browsers, to allow features without plug-ins such as Adobe Flash Player or Google Talk. Based on his research experience, Dr Perkins became co-author of the draft specification for Media Transport and the use of RTP in web browsers [14], ensuring they will support robust media transport and effective quality monitoring. Based on prior multimedia congestion control research, and on-going research on circuit breaker algorithms for multimedia transport protocols ([6], above), Dr Perkins is now developing a draft [15] that will form a normative part of the Web Real-time Conferencing standards to ensure safe media delivery that doesn't congest the network. These specifications are currently incomplete drafts, but they are already giving guidance to and a framework for development for many companies. In addition, they are already implemented and deployed at Internet scale in the Google Chrome and Mozilla Firefox web browsers [16]. These two browsers alone represented ca 60% of the browser base in June 2013, and are used by millions of people.

The immediate beneficiaries of this work are the Internet standards community, and implementers of the various protocol standards. Glasgow research has helped ensure the maintenance of a consistent and coherent protocol framework, providing a clear direction for the growth of the protocol standards at a time when the industry was expanding massively. This has brought research insights directly into the standards process, and perhaps as importantly, has provided a neutral voice – untainted by allegations of favouritism or commercial interest – at the heart of the standards process. All the standards developed are available online and free of charge, and most are open to implement without fear of IPR restrictions. The open and participatory nature of the process, and the global deployment of the resulting standards, shows the benefit of this approach.

5. Sources to corroborate the impact

RFCs (Internet standards) mentioned above can be retrieved from http://tools.ietf.org/html/rfcXXXX replacing 'XXXX' with the 4-digit RFC number. Sample RFCs and other sources corroborating the impact include:

7. The specification of "RTP: A Transport Protocol for Real-Time Applications" was produced by the IETF Audio/Video Transport working group and published as RFC 3550 in July 2003. The RFC confirms my involvement as working group chair, and the minutes of the working group (http://tools.ietf.org/wg/avt/minutes) corroborate the impact of my work steering the development of the protocol.
8. Colin Perkins, "RTP: Audio and Video for the Internet", Addison-Wesley, June 2003, ISBN 0-672-32249-8. 232 citations in Google Scholar. This is the definitive book on RTP, with 249 Google Scholar citations and almost 6,000 copies sold.
9. The 3rd Generation Partnership Project (http://www.3gpp.org/) IP Multimedia Subsystem (IMS) forms the basis for multimedia services in mobile telephony. The main components of the IMS are three IETF standards: SIP, SDP, and RTP. An overview of the 3GPP IMS, documenting the role of RTP and SDP in the system, can be found in G. Camarillo and M. A. García-Martín, "The 3G IP Multimedia Subsystem (IMS): Merging the Internet and the Cellular Worlds", 3rd Edition, ISBN: 978-0470516621, Wiley (September 2008).
10. Developer documentation for MacOS X, iOS, Android, and Windows will corroborate the use of RTP and SDP in these products. In the case of Apple iOS and MacOS X, RTP provides the media transport functionality of the FaceTime video conferencing product announced by Steve Jobs in the keynote address at the Apple World Wide Developers Conference in June 2010 (the slides from that keynote state that Facetime is based on open standards including RTP). Microsoft implemented RTP in NetMeeting, Windows Live Messenger, and Microsoft Lync (formerly known as Microsoft Office Communicator); the Microsoft produces extend RTP slightly, as in http://msdn.microsoft.com/en-us/library/cc431492(v=office.12).aspx
11. Jörg Ott and Colin Perkins, Guidelines for Extending the RTP Control Protocol (RTCP), Internet Engineering Task Force, September 2010, RFC 5968. 20 citations on Google Scholar.
12. Colin Perkins and Thomas Schierl, Rapid Synchronisation of RTP Flows, Internet Engineering Task Force, November 2010, RFC 6051. 42 citations on Google Scholar.
13. Qin Wu, Geoff Hunt, and Phil Arden, Guidelines for Use of the RTP Monitoring Framework, Internet Engineering Task Force, November 2012, RFC 6792. I gave extensive advice to the authors of this, building on [11] and [4], to show how RTCP can be used in a scalable and extensible manner. This is corroborated through my acknowledgement in the specification, and from the minutes of the IETF XRBLOCK working group (charter and meeting minutes available at http://tools.ietf.org/wg/xrblock/).
14. Mark Handley, Van Jacobson, and Colin Perkins, "SDP: Session Description Protocol", Internet Engineering Task Force, RFC 4566, July 2006. An update to this specification (http://tools.ietf.org/html/draft-ietf-mmusic-rfc4566bis-09; last revised September 2013) is a current work item of the IETF Multiparty Multimedia Session Control working group (charter and meeting minutes available at http://tools.ietf.org/wg/mmusic/).
15. Colin Perkins, Magnus Westerlund, and Jörg Ott, "Web Real-Time Communication (WebRTC): Media Transport and Use of RTP", Internet Engineering Task Force, work in progress, September 2013. http://tools.ietf.org/html/draft-ietf-rtcweb-rtp-usage-09. This is a work item of the IETF Real-Time Communication in Web Browsers working group (charter and meeting minutes available at http://tools.ietf.org/wg/rtcweb/).
16. Colin Perkins and Varun Singh, "Multimedia Congestion Control: Circuit Breakers for Unicast RTP Sessions", Internet Engineering Task Force, work in progress, July 2013. http://tools.ietf.org/html/draft-ietf-avtcore-rtp-circuit-breakers-03. This is a work item of the IETF Audio/Video Transport Core Maintenance working group (charter and meeting minutes available at http://tools.ietf.org/wg/avtcore/)
17. The WebRTC project website (http://www.webrtc.org/) outlines the implementation status of the standards in the Google Chrome and Firefox browsers, along with providing links to the protocol standards, implementation source code, interoperability testing reports, etc. The WebRTC code is available in the latest stable versions of the Google Chrome and Firefox browsers, and requires no special download.