A NEW REAL-TIME MULTIMEDIA CONTROL PROTOCOL FOR DISTANCE LEARNING

Lirong He, Ian Rogers, Lisha He
School of Computer Science, University of Manchester,
Manchester, UK
ian.rogers@manchester.ac.uk

Abstract

Real-time multimedia education is becoming an increasingly prevalent part of the Internet with the spread of broadband networks. Via the Internet, users can easily work with other participants by sharing examples, such as in a computer lab. It is convenient for students that they can communicate with teachers at any time and in numerous places. Real-time multimedia systems provide users with a video and audio interface enabling them to see and hear other people. Real-time multimedia facilities for e-learning also can be supplemented by face-to-face teaching in tutorial groups at regular meetings.

Real-time distance learning already makes use of the Internet and network links, but such links can be poor at handling real-time data. Network bottlenecks, variable time delays and network congestion can lead to the multimedia data being severely degraded and ultimately unintelligible. There is a requirement to manage the use of the network traffic so as to maximise the trade-off between multimedia quality and appropriate use of bandwidth.

Techniques exist for the transmission of real-time data over the Internet Protocol (IP), namely the Real-time Transport Protocol (RTP) and the Real-Time Control Protocol (RTCP). We present a new technique for the transmission of real-time data over the transport network layer called the Real-Time Multimedia Transport Control Protocol (RTMTCP). The paper shows the new protocol and control mechanism to be an effective way of improving quality and reducing bandwidth for real-time multimedia distance learning in a way not previously possible.

Building upon the real-time multimedia network protocols, this paper presents a new control method designed for distance learning. This method was developed for application to distance learning specifically over a link between the UK and China. The control method uses feedback and priority weightings to determine whether packets should be sent over the network and if so using which codec. The control method maximizes the quality of the transmitted data in un-congested networks, reduces its bandwidth usage in congested networks and attempts to reduce the chance of making the network congested.

The new RTMTCP method is prototyped and evaluated against the existing RTP/RTCP based schemes. Models of the UK and China network link, the control mechanism, the feedback mechanisms, the means for automatic priority detection and the real-time distance learning system were created. Our evaluation shows RTMTCP to be an improvement over RTCP for real-time distance learning.

Keywords
Real-time Multimedia Distance Learning, Quality of Service (QoS), RTP, RTCP, RTMTCP.

1. INTRODUCTION

During the mid 20th Century, with the use of multimedia facilities (radio and television) in distance education, the number of open universities that were established all over the world grew [1]. For example, the Open University of the United Kingdom was established in 1969 and Canada's Athabasca University was founded in the early 1970s [2]. With the growing popularity of multimedia technologies in the early 90's, CD-ROMs with a variety of educational software were developed. The attraction of multimedia presentation brought another dimension to educational material. By the early 1990s, distance education started making use of the Internet. With increasing access to the Internet on campuses and in homes, many universities began to use the Internet to support courses offered on campus and to offer entire courses online. The Open University and Athabasca University offer a large portion of their courses online. Virtual universities, universities who offer courses on the Internet, use a variety of information technology tools and are found in many countries. Examples include the Western Governor's University in the United States and the Simon Fraser University of British Columbia in Canada. The emergence of Internet-based virtual universities marks the beginning of a new era of distance education development. In the late 90's, again due to the bloom of E-commerce, virtual university and e-learning portals were built for different type of students, from those continuing education, to mission based job training, and to life long learning [3].

A new form of distance learning using the Internet is that involving real-time teaching in a virtual classroom. Many virtual classroom and distance-learning courses are making use of multimedia communications. The transport layer protocol known as the Real-Time Transport Protocol (RTP) [4-6] is an IP-based datagram protocol providing support for the transport of real-time data such as video and audio streams. RTP marks the packets from each application with a specific payload type identifier and adds time stamp and sequence numbers, so that audio and video data can be synchronised at the receiving host. An adjunct to the RTP is a control protocol known as the Real-Time Control Protocol (RTCP) [4-6]. RTCP produces sender and receiver reports containing sender identification, quality of service, lost packets and other factors to be interchanged between the RTP sender and receiver.

The biggest problem facing the use of RTP in distance learning occurs in achieving a good quality of speech and image transmission. This problem results in the loss of lesson time if connections falter and the slowing of progress in lessons. Audibility problems often occur when large groups participate in real-time distance learning. This problem occurs because important speech packets are lost. The teacher has to make pupils listen to each other, and repeat their answers or other comments. Although virtual classrooms already make use of the Internet and dedicated network links, such links can be poor at handling real-time data. This has caused extra stress on teachers [7]. Network bottlenecks, variable time delays and network congestion can lead to the multimedia data being severely degraded and ultimately unintelligible. There is a requirement to manage the use of the network traffic so as to maximise the trade-off between multimedia quality and appropriate use of bandwidth. This leads to the creation of the Real-Time Multimedia Transport Control Protocol (RTMTCP) [10].

To solve these problems for distance learning, we have designed a new technique for the transmission of real-time data, over the transport network layer, called the Real-Time Multimedia Transport Control Protocol (RTMTCP). This paper is the first, to our knowledge, to have incorporated the concept of joint flow/congestion control methods, priority weighting and receiver feedback to achieve high quality communication for real-time multimedia distance learning.

The rest of paper is organised as follows. Section 2 presents the requirement of real-time distance learning over networks. The design, the analysis and evaluation of the RTMTCP protocol are presented in sections 3 and 4. Section 5 summarises and concludes this paper.

2. REQUIREMENTS IN REAL-TIME MULTIMEDIA DISTANCE LEARNING

2.1 Speech and image requirements

The bandwidth requirements of image data packets for face-to-face communications are typically greater than audio data packets [8]. For distance learning, audio data is normally the most crucial element [10]. Therefore, the good transmission of audio/speech data is a key requirement for the RTMTCP protocol.

2.2 Time delay requirement

Current Internet based distance education programmes suffer from communication problems caused by packet delay. When excessive packet delay exists, the conversation can become impossible. In the literature it is considered that a total delay of below 150ms is the ideal for two-way real-time communication, and a total delay of up to 250ms is still acceptable for long distance real-time communication [9]. Therefore speech packets delayed by more than $(250 - P_d)$ ms will not arrive in a period suitable for playback and are considered lost. P_d is the processing delay caused by compressing and decompressing the audio data. For high bit-rate speech, that is not heavily compressed, the processing delay is negligible. For low bit rate highly compressed speech the processing delay, dependent on the selection of the codec, becomes more important. It is a requirement for the RTMTCP protocol that it is able to dynamically switch audio codecs so as to maximise the intelligibility of the audio data for varying network conditions.

2.3 Packet priority assignment requirement

Real-time multimedia data is typically comprised not only of speech and image data, but in a virtual classroom, whiteboard and other data may also be sent. Each type of data stream places different QoS requirements on the network. Audio data is to be considered high priority, and this is determinable by examining the packet type. However, there are occasions when speech and image data should be considered a higher priority than normal. Teachers and students use sound and visual gestures to signify certain things. For example, a raised hand can imply a question; a raised voice can be used to add emphasis to a sentence. It is therefore the case that, in a distance learning environment, the automatic recognition of audio and visual gestures will enable a measure of a packet's priority. It is a requirement for the RTMTCP protocol that packet types have a field for expressing the relative importance of the packet.

3. THE DESIGN OF THE NEW PROTOCOL (RTMTCP)

The RTMTCP protocol gives a better compromise between QoS and bandwidth adaptation than the existing RTCP protocol. Fig. 1 gives an overview of the RTMTCP protocol, and shows its use in a complete system. The RTMTCP protocol uses feedback including speech packet loss and packet loss probability to define the new network state for varying network conditions.

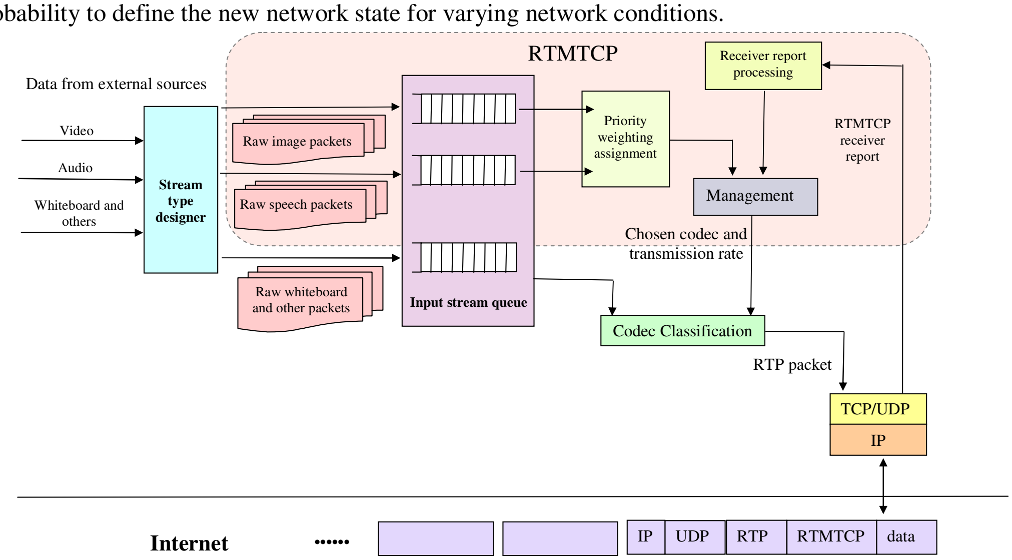

Fig. 1. RTP/RTMTCP protocol overview

3.1 Determination of the network states

The probability of speech packet loss (λ_{sn}) is defined to be: $\lambda_{sn} = 1 - \frac{S_n}{H_{sn}}, (n = 1, 2, ...)$, where S_n is the number of speech packets received during the n^{th} control period with a total transmission delay of less than $(250 - P_d)$, and H_{sn} is the total number of speech packets from the sender during the n^{th} control period. A new threshold value for speech packet loss (λ_{sc}) is also used by the RTMTCP control system. The speech packet loss threshold (λ_{sc}) is used to determine when the speech quality is less than fair (calibrated using Mean Opinion Score tests – MOS [10]) for a given codec, frame size, packet loss concealment method and voice sample.

As with RTCP, the network will be defined to have three different states, as shown in Table 1.

State	Selection criteria
Congested	if $(1 \ge \lambda_n \ge \lambda_c) \vee ((\lambda_n > \lambda_u) \wedge (\lambda_{sn} \ge \lambda_{sc}))$
Loaded	if $(\lambda_u < \lambda_n < \lambda_c) \wedge (\lambda_{sn} < \lambda_{sc})$
Unloaded	if $\lambda_n \le \lambda_u$

Table 1. RTMTCP network state selection

3.2 Flow/congestion control method

The transmission rate control algorithm is as follows:

1. The sender assigns priority weightings to each speech or image packet using automatic priority assignment. An example of a priority weighting assigned using edge detection hand model is shown in Fig. 2.

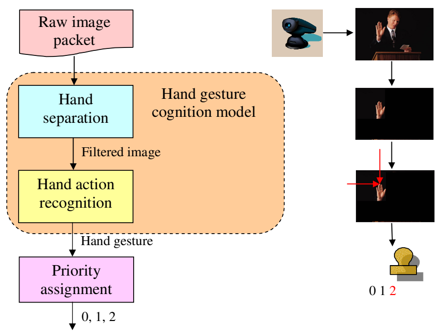

Fig. 2. Hand gesture recognition

2. If the priority weighting of a given packet is high (i.e. 2), the packet will be sent using the low bit-rate with good QoS codec.
3. Otherwise if the priority weighting of a given packet is low (i.e. 0), the packet will not be sent.
4. Otherwise, the priority weighting is 1, the codec will vary dependent on the network state (described in Table 1) as described below:
   a. If the network state is congested, the sender decreases its transmission rate (A) by a suitable amount (ΔA) unless it is at its minimum value.
   b. Otherwise if the network state is unloaded, the sender increases its transmission rate (A) by a suitable amount (ΔA), unless it is already at its maximum value.
   c. Otherwise if the network state is loaded, then the sender does not change the transmission rate.

4. EVALUATION

A network model was used to compare the performance of RTP/RTCP and RTP/RTMTCP as implemented in Java. The parameters of the model were typical of the link between the UK and China, and its use for distance learning. For evaluation, the results of bandwidth use and maximum possible number of acceptable calls for RTP/RTCP or RTP/RTMTCP, are shown in Fig. 3.

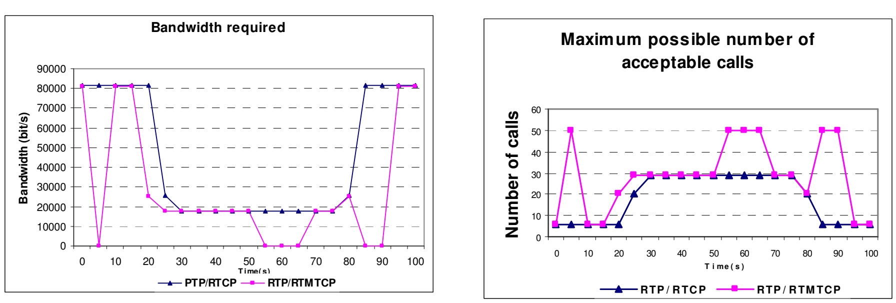

Fig. 3. Evaluation results

Our results show that our worst case bandwidth and number of acceptable call performance for RTP/RTMTCP matches RTP/RTCP. However, on average the number of acceptable calls is greater for RTP/RTMTCP and with a lower bandwidth requirement. The overall system is able to allocate parameters to improve the quality of real-time multimedia communication by dynamically optimizing the packet streams. In its best case, the RTP/RTMTCP system does not transmit any data. In a loaded network, RTMRTCP is better able to adapt its codec choice dependent on speech packet loss probability. By these means RTMTCP frees up network bandwidth, avoids reaching a congested state and increases the maximum possible number of calls. This allows a greater amount of distance learning communication. We summarize the differences between the RTP/RTMTCP and RTP/RTCP protocol for real-time distance learning in Table 2.

Protocols	Speech and image requirements	Speech	Image	Time delay requirement	Bandwidth	Packet priority assignment requirement	Easy to use
RTP/RTCP	No	Poor	Poor	No	Moderate to high	No	Yes
RTP/RTMTCP	Yes	Good	Good	Yes	Low	Yes	Yes

Table 2. The differences between the RTP/RTCP and RTP/RTMTCP protocols

The protocols of RTP/RTMTCP that provides audio, video and data transfer give many opportunities for versatile real-time interaction during video conference lessons. The teacher and the students can hear and see each other; they can exchange written or picture material using document camera, audio, graphics or telefax. The e-learning lesson based on RTP/RTMTCP enables the performing of learning tasks and natural interaction, communication and cooperation to aid the integration of the remote classes.

5. SUMMARY & CONCLUSION

This paper has presented a new RTMTCP protocol for achieving high quality real-time multimedia communication. Four new and important concepts, that are speech packet loss probability, priority weighting, new network states and new flow/congestion control mechanism, have been investigated for improving QoS. The RTMTCP protocol places weak real-time multimedia distance learning requirements on the RTCP protocol. From our results, the new protocol with control mechanism is an effective way of improving quality, reducing bandwidth and increasing the number of participants in real-time multimedia distance learning. The RTMTCP method is able to have an increased number of network connections when the network state is congested, improved quality when the network is un-congested and to avoid reaching a congested state.

References

[1] Wang, L., Wang, X., "A Universal Web-based Teaching and Learning Supplementary System", proceedings of International Conference on DEDL, Beijing, China, 1999.

[2] Tam, S.W., "A Vision for the 21st century: Distance Teaching vs Distance Learning and Development", proceedings of International Conference on DEDL, Beijing, China, 1999.

[3] Ubell, R., "Engineers turn to e-learning". *IEEE Spectrum*, 2000, 37(10): p. 59-63.

[4] El-Marakby, R., Hutchison, D., "Towards Managed Real-Time Communications in the Internet Environment", proceedings of the Fourth IEEE Workshop on High-Performance Communication Systems, Greece, 1997.

[5] Sun, L.F., Wade, G., Lines, B., Ifeachor, E., "Impact of Packet Loss Location on Perceived Speech Quality", proceedings of the 2nd IP-Telephony workshop, USA, 2001.

[6] Chin, K.V., Hui, S.C., Foo, S., "Packet Voice Recovery Techniques for Real-Time Internet Voice Communication", proceedings of the IEEE APCC/ICCS' 98, Singapore, 1998.

[7] Rönkä, A., "Classroom-focused distance education in practice", *http://www.helsinki.fi/kasv/nokol/projektit/kilpis/engkilp.html.*

[8] Minoli, D., Minoli, E., "Delivering Voice over IP Networks", John Wiley & Sons, 1998.

[9] Hassan, M., Nayandoro, A., "Internet Telephony: Services, Technical Challenges, and Products", *IEEE Communication Magazine*, 2000.

[10] He, L, "Real-time facilities for distance learning over IP networks", *PhD thesis*, University of Manchester, UK, 2004.