TCP Compatible Greediness Control Algorithm for Wireless Multimedia Streaming

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***Abstract*— This paper proposes a TCP compatible greediness control mechanism that tunes the greediness of the multimedia streaming process based on client priority, in order to make more efficient use of the wireless network and increase the overall user perceived quality. The majority of streaming solutions use rate adaptation based on congestion avoidance mechanisms that try to obtain as much bandwidth as possible from the limited network resources. However the lack of both knowledge about the characteristics of target devices and cross-layer communication results in fair bandwidth distribution at the transport layer, but creates unfairness at the application layer. This unfairness mostly affects user perceived quality when streaming high quality multimedia. Therefore, there is a need to allow application layer streaming applications tune the aggressiveness of transport layer congestion control mechanisms, in order to create application layer Quality of Experience fairness between competing media streams, by taking their device characteristics into account.**

I. Introduction

Research has proposed different solutions for streaming media over IP-based networks. Many of these multimedia streaming processes use rate adaptation based on congestion avoidance mechanisms which try to obtain as much bandwidth as possible from the transmission resource while minimizing loss, delay and optimize other network parameters. At present, multimedia streaming services optimize video to suit the unique characteristics (e.g. screen size, screen resolution, location, etc.) of the device to which the media is being streamed. However this optimization takes place at the application layer and no provision is made for it at the transport layer. As a result bandwidth is distributed fairly at the transport layer resulting in unequal video quality distribution at the application layer. For example a High Definition Television (HDTV) requires multimedia content to be streamed at a much higher bit rate than a Standard Definition Television (SDTV) to achieve the same user Quality of Experience (QoE) at the application layer, due to the difference in screen size and resolution (see Figure 1). However existing streaming solutions do not account for these differences at the transport layer. This has a negative impact on the overall QoE experienced by the users of various devices. As a result there is a need to optimize the transmission of multimedia content at the transport layer to account for the characteristics of the destination device, in order to create QoE fairness for various multimedia services at the application layer. In this way the overall QoE will be optimized for all users.

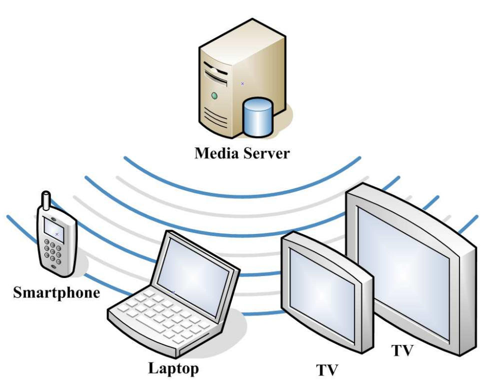

Fig. 1. Typical residential Wireless LAN

This paper proposes a TCP compatible Greediness Control Algorithm (GCA) mechanism that tunes the greediness of the transport layer congestion control mechanism to create application layer QoE fairness in wireless networks. GCA extends the solution proposed by the IETF in [1] by introducing two parameters that allow the streaming application to tune the aggressiveness of the rate estimation. As a result, GCA introduces quality-driven, cross-protocol fairness to the media streaming process. This technique of rate adaptation, combined with a variable bit rate video codec allows for fair prioritization of the multimedia flows based on device characteristics. Results show that this form of prioritization increases the overall user QoE achieved on a number of different devices operating within the home wireless network.

This paper is structured as follows. Section II gives an overview of related works and issues related to multimedia streaming pertinent to the proposed solution. In Section III, the problem statement and solution are outlined. Section IV examines a numerical analysis of the proposed solution. Section V describes the simulation setup, scenarios and testing results. The paper is concluded in Section VI.

II. Related Work

Extensive research has focused on providing a certain level of QoS when streaming multimedia and different approaches have been proposed. These works can be broadly categorized based on the layers of the TCP/IP model that they are deployed at. Apart from the transport layer solutions that use variations of UDP [2] [3] and RTP / RTCP [4], next we present related works most applicable to the proposed scheme.

A. Physical / MAC Layer

The IEEE 802.11 [5] [6] [7] family is the leading standard for WLAN's. The original standards were designed for best effort services and as a result, lacked support for real-time services. The IEEE 802.11 MAC sub-layer defines two medium access control mechanisms, the basic Distributed Coordination Function (DCF) and the optional Point Coordination Function (PCF). DCF can only support best-effort services, and does not provide any QoS guarantees. Although streaming media has strict bandwidth, delay and jitter requirements, it is tolerant of some loss. In DCF mode, all stations within a BSS compete for the shared resource using CSMA/CA. No mechanism is employed to differentiate between the priorities of stations. As a result stations receive equal priority access to the available resources. PCF was designed for real-time services but it is rarely implemented and suffers from loose specification. IEEE 802.11e [8] MAC enhancements were proposed to address some of the shortcomings of the original specification. It provides the required service differentiation by associating a priority level with each packet. The higher priority packets then receive preferential access to the wireless medium. This preferential access is achieved by varying the contention windows and interframe spacing parameters of the CSMA/CA protocol. However this form of service differentiation only provides better than best effort prioritization as well as only providing service differentiation between media flows that occupy different traffic categories. No provision is made for video streams that require further differentiation due to their physical characteristics. Firmware or even hardware may need to be upgraded to support this form of service differentiation. The granularity of the priorities is also limited and parameters are not dynamically adjustable.

B. Application Layer

Many streaming solutions use rate adaptation based on congestion control mechanisms. However the majority of these techniques have focused on network QoS as opposed to user QoE. Rate adaptation schemes are the least complex and most flexible mechanisms for providing QoS as they use the existing network infrastructure. Adaptation takes place at the application layer, by adjusting the parameters of a multimedia stream to best suit the available network conditions. Most solutions use receiver feedback. Rate Adaptation Protocol (RAP), proposed in [9] is a source based TCP friendly Additive Increase Multiplicative Decrease (AIMD) rate adaptation scheme. Enhanced Loss Delay Adjustment (LDA+) [10] adapts the transmission behavior of UDP based multimedia streams in accordance with the current network congestion state, whereas the Quality Oriented Adaptation Scheme (QOAS) [11] uses estimated end-users perceived quality in the adaptation loop. TCP-Friendly Rate Control (TFRC) [1] is a congestion control algorithm that calculates transmission rate as a function of loss events and round-trip time. More recently Datagram Congestion Control Protocol [12] (DCCP) has been proposed as an unreliable transport protocol incorporating end-to-end congestion control. The IETF has also standardized a Congestion Manager [13] allowing for congestion control to be carried out more efficiently by sharing congestion information between entities. However all of these rate control schemes focus on achieving the highest throughput possible, rather then the highest quality. Higher throughput usually translates into higher quality. However, certain devices may have different throughput requirements due to their physical characteristics. The schemes mentioned above will create greedy devices that result in unfairness between competing streams and inefficient use of available bandwidth. [14] has proposed the use of self limiting sources to control the greediness of multimedia traffic. However this too falls short of true greediness control, as it does not provide protection from greedy background traffic flows.

III. Greediness Control Algorithm

A. Problem Statement

Consider a typical residential IEEE 802.11g WLAN with a number of devices attached. Access to the wireless network is shared equally among these devices, resulting in them competing and receiving a fair share of the available bandwidth. Streaming solutions optimize video at the application layer to suit the characteristics of the device to which the media is being streamed. However these solutions still use rate adaptation techniques based on transport layer congestion avoidance mechanisms that try to obtain as much bandwidth as possible. This results in greedy streaming applications unfairly consuming excessive bandwidth that they do not necessarily require. They assume that all devices have equal bandwidth requirements resulting in inefficient and unfair distribution of available bandwidth. For example, consider the situation where three clients with various device characteristics, such as the 32" HDTV, 20" HDTV and 12" SDTV. Each device

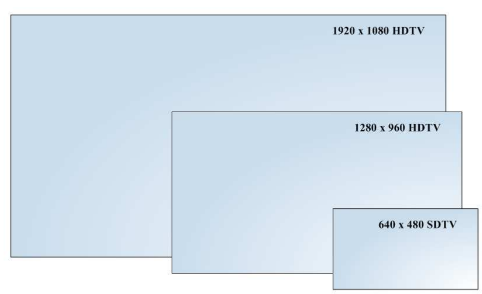

Fig. 2. Scaled screen resolution of multimedia streaming devices

requests a unique H.264 video stream (see Table I from the media server to be streamed via the WLAN. If conventional rate control schemes such as the ones outlined in section II-B were deployed in this scenario it would result in all clients receiving an equal share of available bandwidth. Assuming there is only 18 Mbps of available bandwidth this may result in clients 2 and 3 receiving their required bandwidth while client 1 receives only 70% of what it actually requires. Although this allocation of bandwidth might appear fair from a transport layer perspective, from the application layers QoE point of view, this allocation is grossly unfair. This problems stems from the fact that these rate control techniques do not consider requirements of the media they are carrying or the device to which the media is being streamed. A fairer solution for this scenario would be for each of the clients to share the burden of the congested network equally. To overcome this greedy behaviour it is necessary to tune the parameters of the rate control algorithms to take into account the actual requirements of the device to which the media is being streamed. This can be achieved by introducing parameters that allow the control of the greediness of the rate control algorithm in order to achieve equal user satisfaction and increase overall QoE.

TABLE I
DEVICE CHARACTERISTIC VIDEO REQUIREMENTS

	Client 1	Client 2	Client 3
Device type	32' HDTV	20' HDTV	12' SDTV
Format	H.264	H.264	H.264
Resolution (pixels)	1920x1080	1280x720	640x480
Average Bit Rate (Mbps)	9	6	3
Max Bit Rate (Mbps)	20	14	8

B. Solution: Greediness Control Algorithm

GCA is a congestion control mechanism for unicast flows, which is designed to compete fairly with TCP flows operating in the same environment. GCA is based on the TFRC protocol. It inherits many of TFRC's characteristics which make it suitable for multimedia streaming applications. GCA determines its transmission rate based on a simplified version of the TCP Reno throughput equation (see Equation 1). It determines the sending rate as a function of the Round Trip Time (RTT), loss event rate (p), packet size (s) and the number of packets acknowledged by a single TCP acknowledgment (b). These parameters are calculated on the receiver side of the connection where they are periodically sent back in the form of feedback to the sender.

$$X = \frac{s}{RTT\sqrt{\frac{2bp}{3}} + 3 \times RTO \times p\sqrt{\frac{3bp}{8}}\,(1+32p^2)} \tag{1}$$

GCA resembles the TFRC protocol mechanism as it involves a sender transmitting data packets to the receiver, which periodically returns feedback to the sender. The headers of these packets contain essential information that allow the calculation of RTT, loss event rate and receive rate. Accurate calculation of the loss event rate and RTT are essential for correct operation of the protocol. The loss event rate relates the lost packets to the total number of packets sent. This is calculated by taking a weighted average of a number of consecutive loss intervals. Using the stochastic TCP model presented in [15] and the methodology used in [16], two parameters that control the greediness and generosity are produced, resulting in Equation 2:

$$X = \frac{s}{RTT(\sqrt{\frac{2p(\delta-1)}{\alpha(\delta+1)}} + 12 \times p\sqrt{\frac{p(\delta-1)(\delta+1)}{2\alpha\delta^2}}\,(1+32p^2))} \tag{2}$$

Using this equation and by varying α and δ where $\delta = 1/\beta$, it is possible to configure GCA flows so that they are either more or less aggressive, thus prioritizing the carried traffic.

IV. NUMERICAL ANALYSIS

Figures 3 through 5 show the numerical analysis of the above equation. A number of cases were evaluated by varying different parameters to evaluate the scheme's performance. When the values of α and β are set to 1.0 and 0.5 respectively

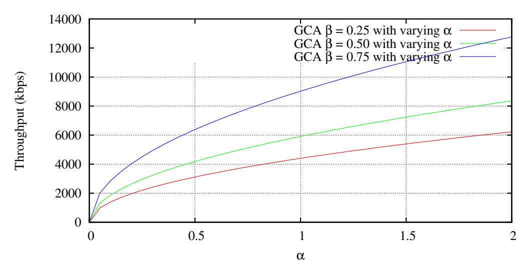

Fig. 3. The effect alpha has on throughput

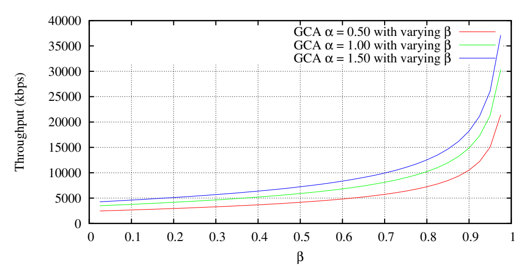

Fig. 4. The effect beta has on throughput

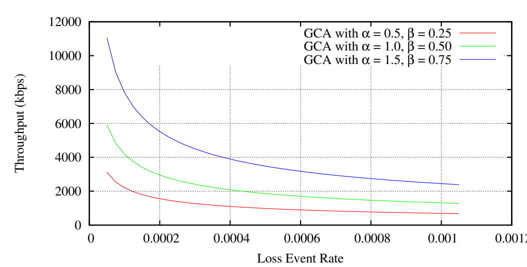

Fig. 5. Typical values of alpha and beta for varying Loss Event Rates

the prioritized rate estimation equation simplifies to the non-prioritized case. Figure 3 evaluates the effect a variation of the alpha parameter has on throughput for various values of β. The α parameter was varied between 0 and 2. It can be seen from the graph that by manipulating the value of α it is possible to obtain $\sqrt{\alpha}$ throughput for a fixed value of β. A similar increase in throughput is obtained by varying β for fixed values of α. The increase in throughput when β is varied between 0 and 1 follows an exponential increasing curve.

The proposed GCA rate estimation equation was also evaluated by varying the loss event rate for various combinations of α and β. These results show how it is possible to control the greediness of media flows by varying the α and β parameters. Figure 5 shows the results of this analysis. The curve representing GCA with $\alpha = 1.0$ and $\beta = 0.5$ shows how a normal media flow reacts to variations in the Loss Event Rate. When $\alpha = 0.5$ and $\beta = 0.25$, a conservative stream is created that obtains about 50% less throughput. It is possible to obtain almost 100% more bandwidth by setting $\alpha = 1.5$ and $\beta = 0.75$ and thus creating a very greedy stream. This analysis illustrates how it is possible to introduce service differentiation to congestion control mechanism by varying the α and β parameters.

V. Simulation-based Testing

A. Setup

The GCA streaming solution outlined in section III-B was implemented by a simulation model and a number of tests were carried out to evaluate the scheme's performance in alleviating this discrepancy between transport and application layer QoE fairness. The model is implemented using the Network Simulator 2 (NS-2) [17]. IEEE 802.11g parameters were used for the wireless environment simulation.

Simulations involved varying the number of clients receiving multimedia data with 1024 byte packet size from a central server / source connected to the WLAN. The simulation topology consisted of a media server connected to an access point with 4 wireless clients attached. Clients are assumed to be devices with characteristics and requirements detailed in Table I. A background traffic source (BG) is also included to suitably load the network by simulating the transfer of a large file in parallel with multimedia streaming. Simulation results demonstrate the problem using a conventional TFRC based streaming solution and also the solution using proposed GCA outlined in Section III-B. Tables II and III present a summary of results of network related measurements in terms of throughput, delay, loss and jitter as well as objectively assessed video quality using the PSNR [18].

B. Conventional TFRC based Streaming Solution Simulation

The first series of simulations evaluated the performance of the existing streaming solution that employs TFRC based rate control. This simulation is designed to illustrate the effect that transport layer fairness provided by conventional rate control mechanisms has on application layer video equality for the scenario outlined in Section III-A. Clients 1, 2, 3 and a background traffic source begin their transmissions at one second increments beginning at $t = 0s$. The throughput analysis of these simulations are illustrated in Figures 6 and comprehensive summary of results can be found in Table II.

Results show that a conventional streaming solution employing TFRC is reasonably fair when competing with other flows. Figure 6 illustrates throughput analysis of this competition. An analysis of transport layer statistics for this simulation concludes that clients are being treated equally. Clients 1 and 2 are receiving approximately 4 Mbps each and client 3 is receiving its optimal 3 Mbps. The background traffic source is receiving a slightly lower throughput of just 2.5 Mbps. However when the application layer objective video quality metrics are analyzed the fairness anomaly becomes apparent.

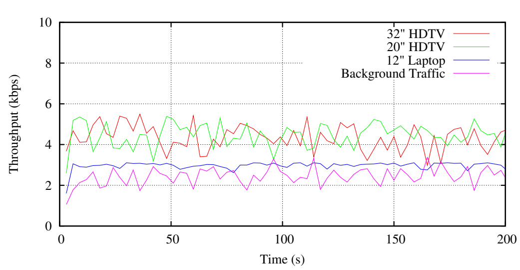

Fig. 6. TFRC based streaming solution simulation throughput analysis

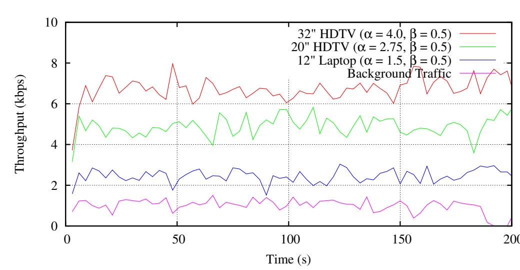

Fig. 7. GCA based streaming solution throughput analysis

TABLE II
TFRC based streaming solution simulation results summary

	Client 1	Client 2	Client 3	BG Traffic
Throughput (kbps)	4,152.23	4,223.65	2,934.89	2,458.86
Delay (ms)	23.47	22.88	21.69	22.21
Loss (%)	1.53	1.65	1.78	1.46
Jitter (ms)	0.62	0.67	0.56	-
PSNR (dB)	35.73	45.24	93.25	-

TABLE III
GCA based streaming solution results summary

	Client 1	Client 2	Client 3	BG Traffic
Throughput (kbps)	6,939.12	5,124.98	2,458.13	1,112.28
Delay (ms)	22.66	22.83	21.78	22.76
Loss (%)	1.57	1.68	1.89	1.62
Jitter (ms)	0.47	0.57	0.64	-
PSNR (dB)	83.67	78.24	76.56	-

These results show that client 1 and 2 are receiving extremely poor PSNR of $\approx 40dB$ while client 3 achieves high quality with PSNR of 90 dB. Although the transport layer network related metrics show equality between clients, the application layer quality metrics show huge inequalities which directly impact the overall user QoE. The SDTV is receiving near perfect video quality while the HDTV's receive unacceptable quality. These results imply that in order to observe the best quality using this streaming solution, it is better to have small screen devices.

C. Greediness Control Algorithm Simulations

The performance for the proposed GCA streaming solution was evaluated under the same simulation conditions as the TFRC based simulation. Clients 1, 2, 3 and the background traffic source begin their transmissions at $t = 1.0s$, $t = 2.0s$, $t = 3.0s$ and $t = 4.0s$ respectively. The results of these tests are illustrated in Figure 7 and summarized in table III. Clients were assigned priorities to account for the device characteristics outlined in Table I. The values of α and β were assigned for exemplification purposes and at this point do not directly map to specific device characteristics.

Figures 7 illustrates the effect that this simulation has on throughput. The most noticeable difference between these results is the level of service differentiation achieved between clients. Each of the clients receive transport layer equality in terms of delay and jitter, each experiencing approximately 20 ms delay and 0.5 ms jitter. However, the service differentiation introduced by the α and β parameters, as expected, has resulted in a throughput inequality. This inequality has resulted in clients 1 and 2 achieving a 69% and 24% increase in throughput, while client 3 has experienced a 21% drop in throughput when compared with the TFRC based streaming solution. The background traffic source has also a significant 56% drop in throughput. However from an application layer perspective these changes in throughput have resulted in fair distribution of video quality between clients. PSNR metrics show very high levels of quality for all devices of approximately 80 dB. Although no application layer statistics are obtained for the background traffic source it is expected that due to the best effort nature of the service there is very little impact on its quality. These results show important increases in the overall user QoE by tailoring the transport layer rate control mechanism to suit the application layer optimization of the media being carried. It can be concluded that GCA-based adaptation brings bandwidth efficiency, maintains TCP compatibility and determines an important overall increase in end-user perceived quality.

VI. Conclusion

This paper proposes the Greediness Control Algorithm (GCA) for wireless multimedia streaming. The paper motivates the need to control the greediness of multimedia streaming process in order to eliminate a fairness anomaly that occurs whereby bandwidth is distributed fairly to competing streams at the transport layer but results in unfair quality distribution when looked at from an application layer perspective. GCA solves this problem by controlling the greediness and generosity of the competing multimedia streaming processes based on their device characteristics.

Simulation results show that the proposed GCA achieves its objective of controlling the greediness of the rate control and as a result introduces application layer QoE fairness while maintaining transport layer stability. GCA delivers QoE equality to users of the multimedia devices. Future development will focus on further refinements of the proposed GCA together with a solution for mapping α and β parameters to actual device requirements. Further testing will also be carried out with a more diverse range of background traffic types. Subjective assessment of end-user perceived quality is also envisaged.

Acknowledgment

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