A Mechanism for Greediness Management when Streaming Multimedia to Portable Devices

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Abstract— **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. This is good if we assume that all clients have equal requirements. However, this is rarely the case as devices have unique characteristics (e.g. screen size, screen resolution, location, etc.) that influence streaming related parameters and consequently the experience obtained by the end user. Due to the inherent greediness of streaming protocols and lack of knowledge about the characteristics of the users device, this may result in certain devices receiving higher bandwidth share than they actually need and others not receiving enough, affecting the users perceived quality. Therefore there is a need to allow for client differentiation in order to provide an acceptable service for all clients, taking their device characteristics into account. This paper proposes a Greediness Control Algorithm (GCA), a specially designed application layer solution that tunes the greediness of the multimedia streaming based on client priority, to make more efficient use of the wireless network and increase the overall user perceived quality.**

I. Introduction

In recent years we have seen a number of developments including the introduction of hard-disc based storage on handhelds, the introduction of WiFi in the home environment and the continued rise of the smartphone [1]. This evolution is set to continue with increases in the storage, processing and communications capabilities of these devices. Portable devices will be able to connect to the residential Wireless Local Area Network (WLAN) (IEEE 802.11) infrastructure and receive multimedia content directly from the in home media server. However, WLAN's were designed primarily as best effort data networks and are not well suited for multimedia distribution.

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 that try to obtain as much bandwidth as possible from the transmission resource while minimizing loss, delay and other network parameters. The problem with this is that devices have unique characteristics (e.g. screen size, screen resolution, location, etc.) that influence the experience obtained by the end user, also denoted as Quality of Experience (QoE). For example a laptop with a 13 inch screen requires multimedia content to be streamed at a much higher bit rate than a PDA to achieve the same QoE due to the difference in screen size (see Figure 1). As a result there is a need to prioritize the delivery of multimedia content based on the characteristics of the destination device in order to make more efficient use of the limited wireless network resources.

This paper proposes the Greediness Control Algorithm (GCA), a specially designed application-level prioritization mechanism for multimedia delivery in wireless networks. It uses variable rate adaptation according to both receiver feedback and statically assigned priorities based on device characteristics. GCA extends the solution proposed by the IETF in [2] by introducing α and β parameters that allow the streaming application to tune the aggressiveness of the rate estimation and as a result, introduce true prioritization to the media streaming process. We show that this technique of rate adaptation, combined with a scalable video format allows for fair prioritization of the media flows based on the above mentioned device characteristics. Results show that this form of prioritization increases the overall user QoE achieved via a number of 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 is outlined. Section IV describes the simulation setup and is followed by a discussion. This paper is then concluded in Section V.

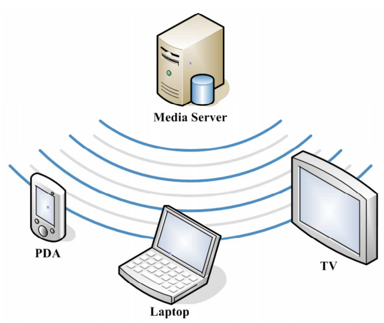

Fig. 1. Typical residential streaming scenario

II. Related Work

Extensive research has focused on providing certain level of QoS when streaming multimedia and different approaches have been proposed. The following sections outline the related works most applicable to the proposed scheme. These works can be broadly categorized based on the layers of the TCP/IP model they are deployed at.

A. Data Link / Physical layer

The IEEE 802.11 [3] [4] [5] 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 [6] 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 are also limited and parameters are not dynamically adjustable.

B. Application Layer

Many streaming solutions use rate adaptation based on congestion control mechanisms that try to obtain as much bandwidth as possible from the transmission resource. 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 [7] is a source based TCP friendly Additive Increase Multiplicative Decrease (AIMD) rate adaptation scheme. Enhanced Loss Delay Adjustment (LDA+) [8] adapts the transmission behavior of UDP based multimedia streams in accordance with the current network congestion state, whereas the Quality Oriented Adaptation Scheme (QOAS) [9] uses estimated end-users perceived quality in the adaptation loop. TCP-Friendly Rate Control (TFRC) [2] 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 [10] (DCCP) has been proposed as an unreliable transport protocol incorporating end-to-end congestion control. The IETF has also standardized a Congestion Manager [11] allowing for congestion control to be carried out more efficiently by sharing congestion information between entities. However all 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. [12] 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. The streaming process uses rate adaptation based on congestion avoidance mechanisms that try to obtain as much bandwidth as possible from the transmission resource. This results in greedy applications (e.g. file transfer) consuming bandwidth that they do not necessarily require. Streaming applications can also be classified as greedy. They assume that all devices have equal bandwidth requirements resulting in efficient use of available bandwidth. For example, consider the situation where three clients with various device characteristics, such as the 20" TV, 12" Laptop and 3" PDA shown in Figure 2. Each device requests a DVD quality video from the media server to be streamed via the WLAN. Client 1 requires the full 6 Mbps video stream to achieve excellent QoE, client 2 requires a 3 Mbps and client 3 requires only 2.0 Mbps to achieve the same level of QoE. Traditional rate adaptation techniques might result in all three devices receiving an equal share (3 Mbps) of the available bandwidth. This would result in client 1 achieving unsatisfactory QoE, client 2 achieving satisfactory QoE, while client 3 receives video at a quality that will not be appreciated due to its smaller screen size and lower resolution display. It will also result in faster battery depletion because the stream has to be adjusted to suit the display. To overcome this greediness 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

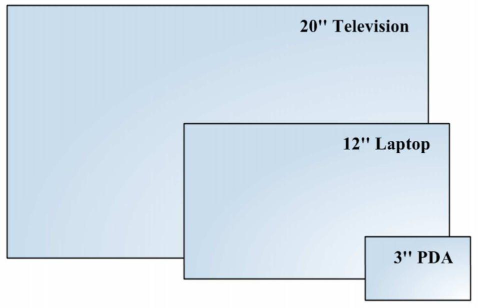

Fig. 2. Typical screen characteristics of multimedia streaming devices

the greediness of the rate control algorithm in order to achieve equal user satisfaction and increase overall QoE.

B. Solution: Greediness Control Algorithm

TFRC is a congestion control mechanism for unicast flows. TFRC is designed to compete fairly with TCP flows operating in the same environment. However TFRC has much smoother variation of throughput compared to TCP which makes it suitable for multimedia streaming applications. TFRC determines its transmission rate based on a simplified version of the TCP Reno throughput (see Equation 1). It determines the sending rate as a function of the Round Trip Time (RTT), loss event rate (p) and packet size (s). 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. This dependency on the receiver makes TFRC a receiver driven protocol. Detailed information about the operation of the TFRC protocol can be found in [2]

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

GCA resembles 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 (see Figure 3).

Using the stochastic TCP model [13] and the methodology used in [14], two parameters that control the greediness and generosity are produced resulting in Equation 2:

$$X = \frac{s}{RTT\sqrt{\frac{2bp(\delta-1)}{\alpha(\delta+1)}} + 3 \times RTO \times p\sqrt{\frac{bp(\delta-1)(\delta+1)}{2\alpha\delta^2}}\,(1+32p^2)} \qquad (2)$$

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.

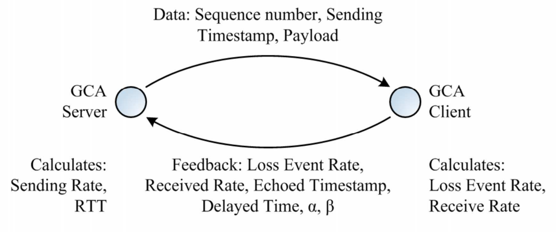

Fig. 3. Illustration of GCA operation

IV. Simulation and Testing

A. Setup

The GCA-based adaptive multimedia streaming scheme 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. The model is implemented using the Network Simulator 2 (NS-2) [15]. IEEE 802.11g parameters were used for the wireless environment simulation, the parameters of which can be found in Table I.

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 resembles the one depicted in Figure 1. Clients are assumed to have devices with characteristics similar to those shown in Figure 2: a big screen digital TV, a laptop and a PDA An FTP background traffic source is also included to simulate the download of a large file in parallel with multimedia streaming, a situation that frequently occurs in reality. Simulation results are presented for the un-prioritized (equal priority is assumed for all streaming processes) and the prioritized cases (differentiated priority is considered for each client). The tables present results of network related measurements in terms of throughput, delay, loss and jitter as well as estimated video quality, measured using PSNR metric. These objective quality metrics compare the situation when the streams are received in ideal conditions - characterised by 0% loss - with the actual received streams.

B. Un-prioritized Simulation

The first series of simulations evaluated the performance of the proposed GCA that was modeled and implemented

TABLE I
IEEE 802.11g Simulation Parameters

Parameter	Value
Data Rate	54 Mbps
Basic Rate	6 Mbps
RTS Threshold	0
Slot Time	9 μs
SIFS	16 μs
CCATime	4 μs
Preamble Length	16 bits
PCLPHeader Length	24 bits
CWmin	15
CWmax	1023

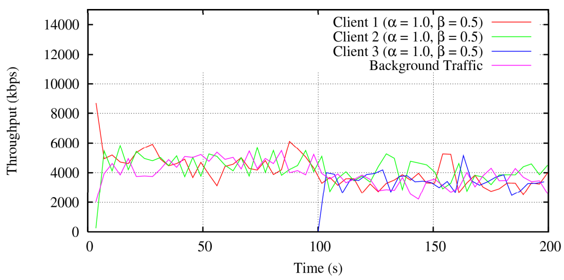

Fig. 4. Un-prioritized throughput analysis

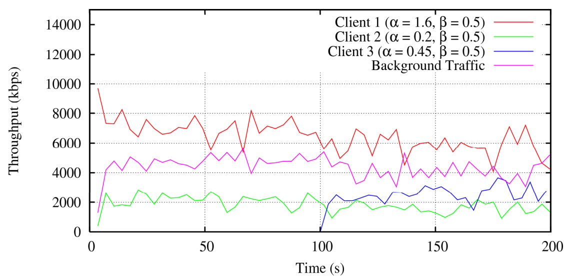

Fig. 6. Prioritized throughput analysis

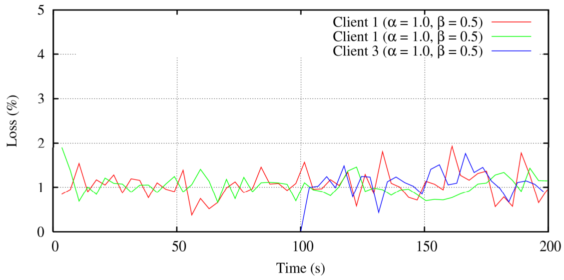

Fig. 5. Un-prioritized loss analysis

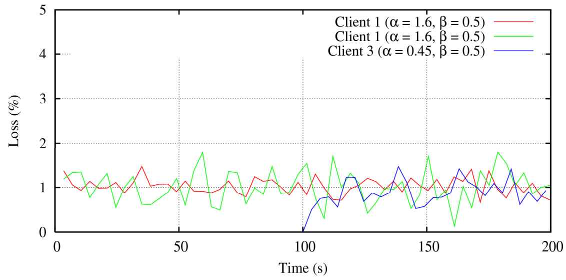

Fig. 7. Prioritized loss analysis

TABLE II
UN-PRIORITIZED SIMULATION RESULTS

$0 < t < 100$	Client 1	Client 2	Client 3
Throughput (kbps)	4863.76	4608.67	-
Delay (ms)	24.05	24.55	-
Loss (%)	0.99	1.06	-
Jitter (ms)	0.53	0.54	-
PSNR (dB)	38.18	37.91	-
$100 < t < 200$	Client 1	Client 2	Client 3
Throughput (kbps)	3435.05	3924.43	3457.78
Delay (ms)	37.37	43.11	25.91
Loss (%)	1.09	1.01	1.12
Jitter (ms)	1.08	0.74	0.69
PSNR (dB)	36.67	37.12	36.98

TABLE III
PRIORITIZED SIMULATION RESULTS

$0 < t < 100$	Client 1	Client 2	Client 3
Throughput (kbps)	7128.15	2118.96	-
Delay (ms)	24.23	24.69	-
Loss (%)	1.04	1.01	-
Jitter (ms)	0.47	0.79	-
PSNR (dB)	80.21	76.86	-
$100 < t < 200$	Client 1	Client 2	Client 3
Throughput (kbps)	5808.3	1570.31	2533.81
Delay (ms)	39.27	40.52	27.21
Loss (%)	0.99	1.00	0.89
Jitter (ms)	0.97	1.21	0.79
PSNR (dB)	74.56	56.12	68.67

using NS-2. The α and β parameters are set to 1.0 and 0.5 respectively, to represent an un-prioritized media flow. Client 1, 2 and the background FTP source begin their transmissions at t = 1 and 2 respectively, followed by client 3 at t = 100. The results of these tests are illustrated in Figure 4 and 5 and summarized in Table II.

The results of this simulation show that the un-prioritized GCA is reasonably fair when competing with other flows. Figure 4 shows the effect that this fairness has on the throughput of the other multimedia and background FTP flows. For the first 100 seconds, Clients 1, 2 and the FTP source receive approximately an equal share of available bandwidth, each receiving approximately 4.7 Mbps. This is below the 6 Mbps required to transmit the DVD-encoded stream at its maximum quality. After the addition of Client 3 at t = 100 s, a decrease in the throughput and an increase in the throughput fluctuations are experienced by all clients. Results also indicate (see Figure 5) that loss remains constant at an average of about 1% due to good adaptive streaming solution. The delay also remains constant throughout the simulation with only a slight increase with the introduction of Client 3.

The video quality measurements (see Table II) show clearly what effect this fairness has on the end user as each of the clients multimedia quality is graded on average with PSNR = 37 dB.

In this simulation scenario the streaming server transmitted a PAL video stream (720 x 576 pixels) to each of the connected clients. The un-prioritized GCA-based rate control algorithm adjusts the bit rate of all the transmissions to suit the current network conditions. resulting in an adapted version of the same 720 x 576 pixel stream being delivered to both a 20 inch TV and a PDA with a 352 x 288 pixels screen resolution. As the PDA scales down the resolution of the received stream to suit its screen size, all the advantage of high bit rate streaming is lost in the case of the PDA. If this scaling was done on the server side it would result in a lower throughput for the PDA and eventually higher bandwidth share for the TV which is currently under provisioned.

C. Prioritized Simulation

The second series of simulations evaluated the performance of the proposed different priority-based Greediness Control Algorithm (GCA). As in the un-prioritized simulation, Clients 1, 2 and the background FTP source begin their transmissions at t = 1 s, t = 2 s, and t = 3 s respectively followed by Client 3 at t = 100 s. The results of these tests are illustrated in Figure 6 and 7 and summarized in Table III. Different client priorities were assigned to account for the device characteristics outlined in section III-A and to resemble the situation presented in Figure 2. There priorities were assigned for exemplification purposes and do not directly map to specific device characteristics.

The results of this simulation show that it is possible to achieve service differentiation by controlling the greediness of the media flows. Figure 6 presents the effect that this form of prioritization has on the throughput of the various multimedia streams. Each of the clients receives a prioritized media stream that is tailored to its physical characteristics. Before t = 100 s, Client 1 receives a 7.1 Mbps stream which corresponds to the highest priority flow, while Client 2 receives just 2.1 Mbps as it has lower priority. The background FTP file transfer receives an adequate 4.5 Mbps. At t = 100 s, Client 3 is added and transfers multimedia over the wireless network using GCA. It is assigned a priority between that of Clients 1 and 2. This addition determines further adaptation from existing traffic and results in all streaming processes receiving relatively less bandwidth than before in order to accommodate the new stream. The loss analysis illustrated in Figure 7 shows that loss remains constant at roughly 1% throughout the entire simulation. This confirms that GCA is very effective at avoiding congestion and maintaining stability within the wireless network. The delay measurements show no negative effect on packet delivery times either. However the proposed scheme resulted in an important increase in the objective video quality metrics values. An average increase in PSNR of $\approx$ 31 dB was experienced. This is because the multimedia streaming is now tailored to the destination devices and best suits their characteristics.

These results also show significant increase in the overall user QoE. All devices are receiving streams that are tailored to their characteristics resulting in more efficient use of the available resources. The 20 inch TV is receiving a PAL (720 x 576 pixel) equivalent stream, the laptop receives a VGA (640x280 pixel) equivalent clip and the PDA is receiving a video with CIF (352 x 288 pixel) resolution. It can be concluded that GCA-based adaptation brings bandwidth efficiency and maintains prioritized fairness.

V. Conclusion

This paper proposes a Greediness Control Algorithm (GCA) for adaptive wireless multimedia streaming. The paper motivates the need to control the greediness of multimedia streaming and presents a solution based on client device prioritization. The solution introduces two parameters that allow control of greediness and generosity in adaptive multimedia streaming. These values allow tuning of the aggressiveness of the congestion control mechanism thus allowing for prioritization.

NS2 simulation results, show that the proposed GCA achieves rate and consequently quality differentiation and most importantly obtains an increase in the overall user QoE. Future development will focus on dynamically changing priorities and further refinement of the proposed algorithm. Subjective assessment of end-user perceived quality is also envisaged.

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