2009 International Conference on Machine Learning and Computing
IPCSIT vol.3 (2011) © (2011) IACSIT Press, Singapore

An Equation Based Rate Control for Adaptive Video Streaming over Cellular Networks

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Abstract. Rate control is an important issue for video streaming in cellular networks. This paper describes an equation based rate control for video streaming over cellular networks. In our method, the sending rate is calculated as a function of round trip time (RTT), loss event rate (p), packet size (s) and new control parameters (alpha ,beta) that are able to provide flexible and smooth transmission rate and slowly responsible congestion control and also adaptability to unpredictable wireless channel conditions. We have simulated this method in UMTS and according to analytic and simulation results, this method in addition to network stability increase throughput with low fluctuation.

Keywords: adaptive video streaming, cellular network, rate control, TFRC, UMTS.

1. Introduction

As communications technology is being developed, user's demand for multimedia services raises. On the other hand the current income of the cellular operators from voice calls is saturated. One of the most promising services is the transmission of rich multimedia content, which is why most of the cell phones manufactured nowadays, have multimedia capabilities. But bandwidth is a valuable and limited resource for UMTS and every wireless network, in general. So currently, the poor video quality on the low and fluctuated bandwidth networks is making the use of streaming applications very difficult and it is a problem that prevents the operators from charging for video services [1], [2], [3].It is essential for every wireless network to have an efficient bandwidth allocation algorithm. Congestion control is aimed at solving this problem by adapting the streaming rate to the network conditions. A widely popular rate control scheme over wired networks is equation-based rate control also known as TCP friendly rate control (TFRC). It is an equation-based congestion control algorithm explicitly designed for best-effort unicast multimedia traffic. It is designed to be reasonably fair when sharing bandwidth with TCP flows. TFRC also has a much lower throughput variation over time compared with TCP making it an ideal for streaming media. TFRC is not a full transport layer protocol, rather a congestion control mechanism that can be used by an existing transport protocol [4], [5]. There are basically three main advantages for rate control using TFRC: first, it does not cause network instability, which means that congestion collapse is avoided. Second, it is fair to TCP flows, which are the dominant source of traffic on the Internet. Third, the TFRC's rate fluctuation is lower than TCP, making it more appropriate for streaming applications which require constant video quality. Some applications require a fixed interval of time between packets and vary their segment size instead of their packet rate in response to congestion. According to References [4], [5] TFRC is not designed for those applications. In this work we introduce an improvement method for adaptive video streaming over cellular networks that is based on new equation and provide flexible, smooth transmission rate and slowly responsible congestion control. It has also adaptability to unpredictable wireless channel conditions. We have

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simulated this method in UMTS and according to analytic and simulation results, this method in addition to network stability increases throughput with low fluctuation.

This paper is structured as follows. Section 2 is a description of the related works. In Section 3 the TFRC mechanism with new equation for the UMTS is evaluated. Section 4 is dedicated to the analytic and simulation results, Section 5 presents conclusion. Some ideas for future work are also outlined.

2. Related Works

Rate control is an important issue in both wired and wireless streaming applications. In References [1], [2] the authors present streaming video over UMTS transport channels through the use of the modified the TFRC mechanism that is mainly used in wired networks. In Reference [3] the authors presents adaptive streaming algorithm (ASA) that improves significantly the quality of service in varying network conditions and monitors its performance using queuing methodologies. The algorithm utilizes the available buffers on the way of the streaming data in a unique way and controls buffers' occupancy levels by controlling the transmission and the encoding rates of the streaming server to achieve high QoS for the streaming. In Reference [6] the authors presents, TCP Reno, treats the occurrence of packet loss as a manifestation of network congestion. This assumption may not apply to networks with wireless channels, in which packet loss is often induced by noise, link error, or reasons other than network congestion. Equivalently, TCP Vegas uses queuing delay as a measure of congestion [7]. Thus, the authors [6] propose an enhancement of the TCP Reno and TCP Vegas for the wireless networks, namely TCP Veno. In Reference [8], two algorithms are presented that formulate resource allocation in wireless networks. These procedures constitute a preliminary step towards a systematic approach to jointly design TCP congestion control algorithms, not only to improve performance, but also more importantly, to make interaction more transparent. In Reference [9] the performance characteristics of TCP New Reno, TCP SACK, TCP Veno and TCP Westwood under the wireless network conditions are studied and the authors propose a new TCP scheme, called TCP New Jersey, which is capable of distinguishing wireless packet losses from congestion. In other scheme [10] when a packet is lost, TFRC goes beyond layering abstraction and enquires the link layer about the recent signal strength. The packet loss is recognized to be due to wireless channel error if recent signal strength is low and due to congestion otherwise. Explicit Loss Notification (ELN) can also be applied to notify TCP/TFRC sender when a packet loss is caused by wireless channel errors rather than congestion [11], In this case, TFRC can take into account only the packet loss caused by congestion when adjusting the streaming rate. End-to-end statistics can be used to help detect congestion when a packet is lost .For example, by examining trends in the one-way delay variation, Parsa and Garcia-Luna-Aceves [12] interpret loss as a sign of congestion if one-way delays are increasing and a sign of wireless channel error otherwise. One-way delay can be associated with congestion in the sense that it monotonically increases if congestion occurs as a result of increased queuing delay, and remains constant otherwise. Similarly, Barman and Matta [13] proposed a loss differentiation scheme based on the assumption that the variance of round trip time is high when congestion occurs, and is low otherwise. Cen et. al. presents an end-to-end based approach to facilitate streaming over wireless [14]. They combine packet inter-arrival times and relative one way delay to differentiate between packet loss caused by congestion, and that due to wireless channel errors. There are two key observations behind their approach; first, relative one way delay increases monotonically if there is congestion; second, inter-arrival time is expected to increase if there is packet loss caused by wireless channel errors. Therefore, these two statistics can help differentiate between congestion and wireless errors. However, the high wireless error misclassification rate may result in under-utilizing the wireless bandwidth, as shown in [19]. Yang et. al. [15] also proposes a similar approach to improve video streaming performance in presence of wireless error, under the assumption that wireless link is the bottleneck. Yang et.al. [16] proposes a cross-layer scheme that uses link layer information to determine whether a packet loss is caused by channel error or congestion, assuming that only the last link is wireless. A similar assumption is made by Akan and Akyildiz in [17] to derive a wireless TFRC-like equation based protocol to facilitate video streaming. The disadvantage of end-to-end statistics based approaches is that congestion detection schemes based on statistics are not sufficiently accurate, and they either require cross layer information or modifications to the transport protocol stack. It is also possible to enable the routers with ECN markings capability to do rate control using ECN as the measure of congestion [18]. As packet loss no longer corresponds to congestion, ECN based rate control does not adjust sending rate upon observing a packet loss.

3. TFRC Mechanism with New Equation for UMTS

The equation used in TFRC reflects the throughput behaviour of a TCP flow, in this way any equation that models realistically a TCP flow could be used.

In our method we use a novel rate estimation formula presented in Equation (2). It is based on a modified version of the TCP Reno throughput equation (which is designed to compete fairly with TCP) and the sending rate (X) is calculated as a function of round trip time (RTT), loss event rate (p), packet size (s) and new control parameters (α, β) [19], that are able to provide flexible and smooth transmission rate and slowly responsible congestion control and also adaptability to unpredictable wireless channel conditions. Using this equation and by varying α and β, it is also possible to configure flows.

$$X = \frac{ks}{RTT\sqrt{\frac{2p(\delta-1)}{\alpha(\delta+1)}}+3RTO\sqrt{\left(\frac{p(\delta-1)(\delta+1)}{8\alpha\delta^2}\right)}\,p(1+32p^2)} \tag{1}$$

In equation (1), k is a constant factor between 0.7 and 1.3 [1] and RTO (retransmission timeout) is equal to 4RTT [4], [5], so we have $\delta = \frac{1}{\beta}$ and:

$$X = \frac{ks}{RTT\left(\sqrt{\frac{2p(\delta-1)}{\alpha(\delta+1)}}+12p\sqrt{\left(\frac{p(\delta-1)(\delta+1)}{8\alpha\delta^2}\right)}\,(1+32p^2)\right)} \tag{2}$$

By replacing $\alpha = 1$ and $\beta = 0.5$ we have equation (3) that is well known TFRC simplified throughput equation.

$$X = \frac{ks}{RTT\left(\sqrt{\frac{2p}{3}}+12p\sqrt{\left(\frac{3p}{8}\right)}\times(1+32p^2)\right)} \tag{3}$$

According to Reference [20] the relationship between α and β to be TCP-friendly is:

$$\alpha = \frac{4(1-\beta^2)}{3} \tag{4}$$

This relationship offers a wide selection of possible values for α and β to achieve desired transient behaviours, such as responsiveness and reduced rate fluctuations. We use the typical scenario for streaming video over UMTS is shown in Figure1.It is include a UMTS radio cell covered by a Node B connected to a radio network controller (RNC) and a user equipment (UE) connected to a bi-directional channel (DCH) to transmit packet data and is reserved only for a single user. The common channels are the forward access channel (FACH) in the downlink and the random access channel (RACH) in the uplink.

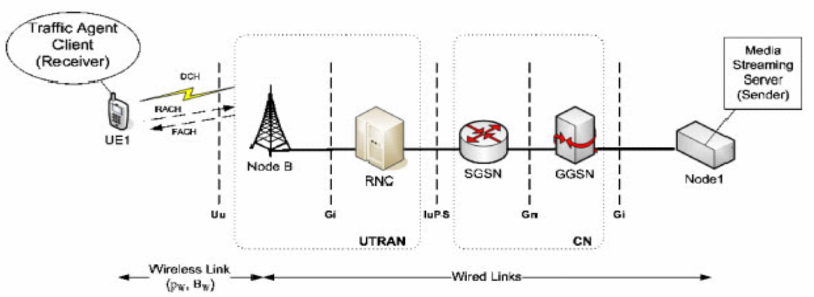

Fig. 1 Typical Scenario for Streaming Video over UMTS

It is assumed that the wireless link has available bandwidth Bw, and packet loss rate Pw, caused by wireless channel error. So the maximum throughput that could be achieved in the wireless link is $Bw(1-Pw)$. There could also be packet loss caused by congestion at wired nodes (GGSN, SGSN, RNC, Node B). This means that the streaming throughput is $X(1-P)$ and P is end-to-end packet loss rate observed by the receiver. Under the above assumptions, the wireless channel as underutilized if $X(1-P) \leq Bw(1-Pw)$. We use the TFRC model described in Equation (2) to analyze the problem. According to Reference [1], the end-to-end packet loss rate P is computed as follows:

$$P = P_{GGSN} + (1-P_{GGSN})\,P_{SGSN} + (1-P_{GGSN})\,(1-P_{SGSN}) \tag{5}$$

$= P_{RNC} + (1\text{-}P_{GGSN})\ (1\text{-}P_{SGSN})\ (1\text{-}P_{RNC})\ P_{Node\ B} + (1\text{-}\ P_{GGSN})\ (1\text{-}\ P_{SGSN})\ (1\text{-}P_{RNC})\ (1\text{-}\ P_{Node\ B})\ P_w$

Since we are comparing our work with previous work on this subject in Reference [1], we will use the same assumptions in our simulation as follows:

- There is no congestion due to streaming traffic to the nodes GGSN, SGSN and RNC.
- There is no congestion at Node B due to the streaming application.
- There is no queuing delay caused at Node B, so the RTT has the minimum value.
- Pw is random and varies from 0 to 0.16 and the backward route is error-free and congestion-free.
- Downlink bit rate Bw is 384 (kbps) and Uplink bit rate is 128 (kbps).
- Downlink TTI is 10(ms) and Uplink TTI is 20(ms).
- Node1 to GGSN bandwidth is 10(Mbps) and average delay is 15(ms).
- GGSN to SGSN bandwidth is 622(Mbps) and average delay is 10(ms).
- SGSN to RNC bandwidth is 622(Mbps) and average delay is 1 (ms).
- RNC to Node B bandwidth is 622(Mbps) and average delay is 15(ms).
- RTCP report rate Every 1 s and Packet size S is 800(bytes).
- $0 < \beta < 1$, $(\delta = \frac{1}{\beta})$ and according to TCP Friendly behaviour $0 < \alpha < 1.33$

The communication between the sender and the receiver is based on RTP/RTCP sessions. The sender (server), use the RTP protocol to transmit the video stream and the mobile user (receiver) in recurrent time space sends RTCP reports to the sender. These reports contain information about the current conditions of the wireless link during the transmission of video between the server and the mobile user. The server using the feedback information and estimates the appropriate rate of the streaming video to avoid network congestion. From Equations (2) and (6) the server is responsible for adjusting the sending rate with the calculated value and estimates the smoothed transmission rate, using the m most recent values**.**

$$X^{Smoothed} = \frac{\sum_{i=1}^{m} wi\,.\,X_{m+1-i}^{Smoothed}}{\sum_{i=1}^{m} wi} \tag{6}$$

So it is essential to keep a history of the previous calculated values for the transmission rate. In our simulations we use m=8 and the values for the weights $\omega_i \in \{1,1,1,1,0.8,0.6,0.4,0.2\}$. Thus, we have chosen to keep track of eight values according to Reference [1]. The server extracts the feedback information from the RTCP report and passes it through an appropriate filter. The use of filter is essential for the operation of the mechanism in order to avoid wrong estimations of the network conditions and for prevent a single spurious packet loss having an excessive effect on the packet loss estimation, by using the equation (7) the server smoothes the values of packet loss rate.

$$P^{Smoothed} = \frac{\sum_{i=1}^{m} wi\,.\,P_{m+1-i}^{smoothed}}{\sum_{i=1}^{m} wi} \tag{7}$$

RTT is roundtrip time between the server and the client. We assume the delay time of server to client is denoted by Dsc and the delay time of client to server is denoted by Dcs, so we can say $RTT = Dsc + Dcs$.

When the client receives a RTP packet from the server, it uses the timestamp of the RTP packet and the local time that the packet is received in order to estimate the Dsc. We don't have the same delay into both directions for UMTS it means $Dsc \neq Dcs$, so we cannot say the RTT between the sender and the receiver is twice the Dsc, ($RTT \neq 2Dsc$). In order to smooth estimation of the RTT due to the potential unsynchronized clocks between the server and the client and due to the potential asymmetry of the path between the sender and the receiver we have to estimate the accurate RTT. So we use a parameter θ and we have:

$$RTT_{accurate} = (1 + \theta)Dsc \tag{8}$$

For computing θ, the sender estimates the RTT_{sample} every time it receives a receiver report from the client by using the following equation:

$$RTT_{sample} = (rxTime - txTime) - delayTime \tag{9}$$

In equation (9), rxTime is time the feedback packet was received by the sender, txTime is transmission time of last received data packet at the receiver and delayTime is time at receiver between reception of last packet and transmission of feedback packet. So we can estimate an appropriate value of θ using the following equation:

$$\theta = \frac{RTT_{sample} - D_{SC}}{D_{SC}} \tag{10}$$

4. Analytic and Simulation Results

We present simulation and experimental results in this section. To validate the above-presented analysis, we carry out experiments using MATLAB and NS-2 simulator [21].

In this section an analysis of the throughput equation (Equation2), presented in Section 3, is performed to determine how each of the parameters affect the overall throughput of the system. The results of this analysis are illustrated in Figures 2 through Figure 5. This analysis uses the assumptions presented in previous section. The throughput represented by $\alpha = 1$ and $\beta = 0.5$ in each plot is known as the reference point. The first set of analysis results, illustrated in Figure 2, evaluates the effect varying α in the interval 0.0−1.33 has on throughput for different values of $\beta \epsilon \{0.2, 0.3, 0.4, 0.5, 0.6, 0.7, 0.8\}$. Analysis of the curve represented by $\beta = 0.5$, indicates that the throughput changes considerably over the range α. A similar variation is observed for other values of β. Next, the effect of varying the β between 0.0−1.0 for discrete values of $\alpha \epsilon \{0.2, 0.4, 0.6, 0.8, 1, 1.2, 1.33\}$ is evaluated. The result of this analysis is illustrated in Figure 3. First inspection of the results indicates that the throughput grows exponentially when β is varied between 0.0 − 1.0. The reference point throughput only experiences a relatively small variation for β between 0.0 − 0.8. Infinite throughput is experienced as β converges on 1.0. This makes $\beta > 0.8$ unsuitable for bit-rate tuning of multimedia streams due to its instability. The linearity between 0.0 − 0.8 makes β suitable as a tuning parameter. Figures 4 and 5 analyses the effect of varying loss event rate p have on certain values of α and β. As expected the higher the loss event the lower the throughput obtained. These graphs also illustrate ability of this method to maintain proportional fairness between various flows for varying loss event rates.

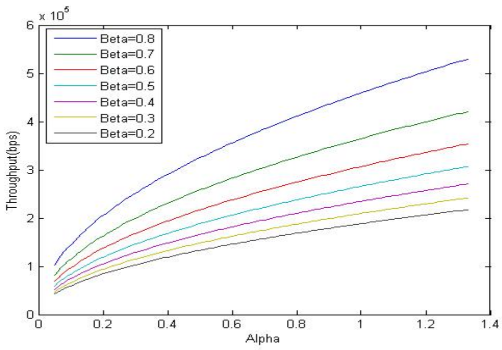

Fig. 2 The Effect of Varying the α Parameter on Throughput for Various Values of β.

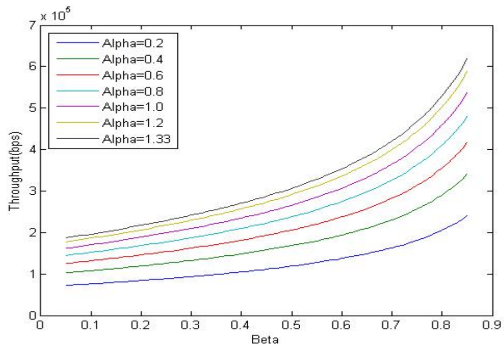

Fig. 3 The Effect of Varying the β Parameter on Throughput for Various Values of α.

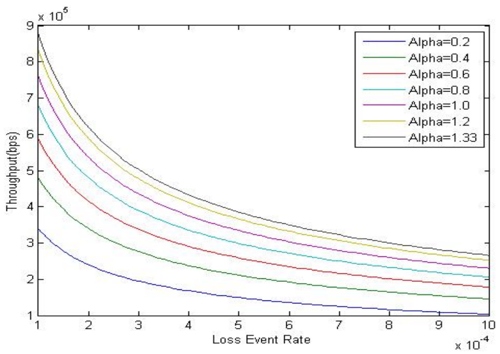

Fig. 4 The Effect of Varying the Loss Event Rate on Throughput for Various Values of α.

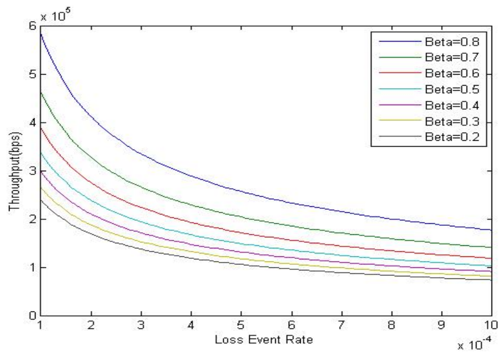

Fig. 5 The Effect of Varying the Loss Event Rate on Throughput for Various Values of β.

We have simulated for 200 seconds and the server initially transmits the video with 256 kbps. We initially use $\alpha = 1$ and $\beta = 0.5$ to adjust the transmission rate of the video. When the overall sending rate is increased, we observe increased packet losses due to congestion, so we measured this packet loss rate according to equations (5) and (7). Then we estimate the new transmission rate of the video according to equation (2) by varying parameters. The server estimates the accurate RTT and calculate smoothed transmission rate by using the most recent values of the calculated sending rate according to equation (6). When the server transmits the video with the greater bit rate and observes an increase in the packet loss rate, it decreases the sending rate of the video to avoid network instability. The throughput of the video in the wireless link is showed in Figure 6. The y-axis presents the throughput in kbps while the x-axis represents the duration of the simulation in second. As it is shown, the server initially uses the video with bit rate 256 kbps and 20 s after the beginning of the simulation the packet loss is occurred in the wireless link and it is due to congestion. So our mechanism calculates the smoothed transmission rate to be under the value of 256 kbps to avoid congestion problems and maintain a TCP-Friendly behaviour. In approximately half of the simulation time, we observe the maximum packet loss rate which results to the minimum estimated transmission rate. Additionally, 100 s after the beginning of the simulation, when the network becomes stable, the smoothed transmission rate of the video will be increase to maximize the throughput. In Figure 6 we can also compare the throughput of the video in the wireless link for this work and Reference [1]. It shows increasing in throughput with low fluctuation. In General, This method in addition to network stability increase throughput with low fluctuation and achieve desired transient behaviours.

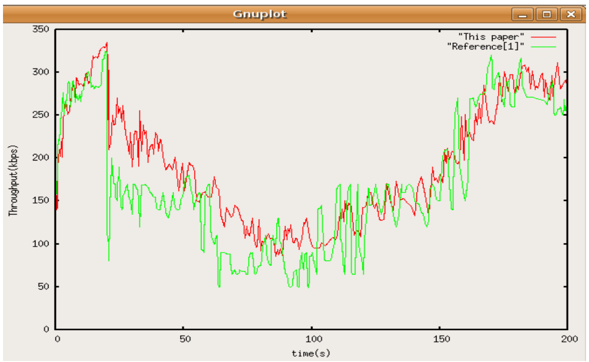

Fig. 6 Comparison of the Video Throughput in Wireless Link

5. Conclusion

Rate control is an important issue in video streaming applications. In this work, we focus on a mechanism for equation based congestion control in video transmission over UMTS network. All previous works on this subject were based on the simplified TCP Reno equation in which sending rate is determined as a function of the packet size, the round trip time and the packet loss rate. In order to change transmission rate, packet size or distance between to sequential transmission must be modified. By changing packet size, TFRC's characteristic will change. In this research another throughput equation is used and transmission rate is function of some other parameters such as α and β in addition to pervious mentioned parameters.

In our method by using α and β transmission rate will be more smoothed in congested conditions and is related to loss rate. We demonstrated that higher throughput could be achieved and it provides smoother throughput variations. Although out method increased throughput and made network more stable using rate control , but it does not guarantee good quality in delivered video. As a future work by using large receiver side buffering and automatic configuration of α and β we can improve quality of service. Another suggestion for future work is fairness in bandwidth allocation according to user requirements.

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