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Abstract: This paper addresses the problem of congestion control for video transmission in large multicast groups. In order to solve the well-known feedback implosion problem in large multicast groups, we first present a mechanism for filtering RTCP receiver reports sent from receivers to the whole session. The proposed filtering mechanism provides a classification of receivers according to a predefined similarity measure. An end-to-end source and FEC rate control based on this distributed feedback aggregation mechanism coupled with a video layered coding system is then described. The number of layers, their rate and levels of protection are adapted dynamically to aggregated feedbacks. The algorithms have been validated with the NS2 network simulator.

Key-words: multicast, layered,video, congestion control, aggregation, cluster,T CP-compatible, FGS, Fine Grain Scalability

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Contrôle de débit source-canal adaptatif pour la transmission multi-points de video multi-couches basé sur un algorithme de clustering

Résumé : Ce papier traite du problème du contrôle de congestion pour la transmission vidéo multipoint sur réseaux hétérogènes. Afin de règler le problème "connu" d'implosion de la voie de retour, nous présentons tout d'abord un mécanisme permettant le filtrage des rapports de réception RTCP. Le mécanisme proposé fournit une classification des récepteurs selon une mesure de similarité préalablement définie. Un algorithme global de régulation de débit source et canal, basé sur ce mécanisme d'aggrégation de rapport couplé à un sytème de codage vidéo multi-couches est ensuite décrit. Le nombre de couches, leurs débits ainsi que leurs niveaux de protection sont adaptés dynamiquement aux rapports aggrégés. Ces algorithmes ont été validés avec le simulateur de réseau NS2.

Mots-clés : multicast, multi-point, multi-couches, vidéo, contrôle de congestion, aggregation, cluster,TCP-compatible, FGS, granularité fine

1 Introduction

Transmission of multimedia flows over multicast channels is confronted with the receivers heterogeneity problem. In a multicast topology (multicast delivery tree in the $1 \rightarrow N$ case, acyclic graph in the $M \rightarrow N$ case), network conditions such as loss rate and queueing delays are not homogeneous in the general case. Rather, there may be local congestions affecting downstream delivery of the video stream in some branches of the topology. Hence, the different receivers are connected to the source via paths with varying delays, loss and bandwidth characteristics. Due to this potential heterogeneity, dynamic adaptation of multimedia flows over multicast channels, for optimized QoS of multimedia sessions, faces challenging problems. The adaptation of source and transmission parameters to the network state often rely on the usage of feedback mechanisms. However, the use of feedback schemes in large multicast trees faces the potential problem of feedback implosion. The first issue addressed here is therefore the problem of aggregating heterogeneous reports into a consistent view of the communication state. The second issue concerns the design of a source rate control mechanism that would allow a receiver to receive the source signal with quality commensurate with the bandwidth and loss capacity of the path leading to it.

Layered transmission has been proposed to cope with receivers heterogeneity [1, 2, 3]. In this approach, the source is represented using a base layer, and several successive enhancement layers refining the quality of the source reconstruction. Each layer is transmitted over a separate multicast group and receivers decide the number of groups to join (or leave) according to the quality of their reception. At the other side, the sender can decide the optimal number of layers and the encoding rate of each layer according to the feedback sent by all receivers. A variety of multicast schemes making use of layered coding for audio and video communication have been proposed, some of which rely on a multicast feedback scheme [4], [5]. Despite rate adaptation to the network state, applications have to face remaining packet losses. Error control schemes using FEC (*Forward Error Correction*) strongly reduce the impact of packet losses [6, 7, 8]. In these schemes, redundant information are sent along with the original information so that the lost data (or at least part of it) can be recovered from the redundant information. Clearly, sending redundancy increases the probability of recovering packets lost, but it also increases the bandwidth requirements and thus, the loss rate of the multimedia stream. Therefore, it is essential to couple the FEC scheme to the rate control scheme in order to jointly determine the transmission parameters (redundancy level, source coding rate, type of FEC scheme, etc.) as a function of the state of the multicast channel, to achieve the best subjective quality at receivers. For such adaptive mechanisms, it is important to have simple channel models that can be estimated in an on-line manner.

The sender, in order to adapt the transmission parameters to the network state, does not need reports of each receiver in the multicast group. It rather needs a partition of the receivers into homogeneous classes. Each layer of the source can then be adapted to the characteristics of one class or of a group of classes. Each class represents a group of homogeneous receivers according to discriminative variables related to the received signal quality. The clustering mechanism used here follows the above principles. A classification of receiver reports is performed by aggregation agents organized into a hierarchy of local regions. The approach assumes the presence of aggregation agents at strategic positions within the network. The aggregation agents classify receivers according to similar reception behaviors, and filter correspondingly the RTCP receiver reports. By classifying receivers, this mechanism solves the feedback implosion problem and at the same time provides the sender with a compressed representation of the receivers.

In the experiments reported in this paper we consider two pairs of discriminative variables in the clustering process: the first one constituted of the loss rate and the *goodput* and the second constituted of the loss rate and the throughput of a conformant TCP connection under similar loss and round trip time conditions. We show that approaches in which receivers rate-requests are only based on the *goodput* measure risk to lead to a severe sub-utilization of the network resources. To use a TCP throughput model, receivers have first to estimate their round-trip-time (RTT) to the source. In order to do so we use the algorithm described in [5] jointly with a new application-defined RTCP packet, called *Probe-RTT*.

This distributed feedback aggregation mechanism is coupled with a video FGS layered coding system, to adapt dynamically the number of layers, the rate of each layer and its level of protection. Notice that the aggregation mechanism that has to be supported by the network nodes remains generic and can be used for any type of media. The optimization is performed by the sender and takes into account both the network aggregated state as well as the rate-distortion characteristics of the source. The latter allows to optimize the quality perceived by each receiver in the multicast tree.

The remainder of this paper is organized as follows. Section 2, provides an overview of related resarch on multicast rate and congestion control. Section 3 sets the main lines of the hybrid sender/receiver driven rate control based on a clustering algorithm. The protocol functions to be supported by the receivers, and the receiver clustering mechanism governing the feedback aggregation are described respectively in section 4 and section 5. Section 6 describes the multi-layer source and channel rate control and the multi-layered MPEG-4 Fine Grain Scalable source encoder that has been used in the experiments. Finally, experimental results obtained with the NS2 network simulator with various discriminative clustering variables (goodput, TCP-compatible thoughput), including with the additional usage of forward error correction are discussed in section 7.

2 Related Work

Related work in this area focuses on error, rate and congestion control in multicast for multimedia applications. Layered coding is often proposed as a solution for rate control in video multicast applications over the Internet. Several - sender driven [9], receiver driven [10, 11], or hybrid schemes [12, 4, 13] - approaches have been proposed to address the problem of rate control in a multicast transmission. Receiver-driven approaches consist in multicasting different layers of video using different multicast addresses and let the receivers decide which multicast group(s) to subscribe to. RLM [10] and RLC [11] are two well-known receiver-driven layered multicast congestion control protocols. However, they both suffer from pathological behaviors such as transient periods of congestion, instability, and periodic losses. These problems mainly come from the bandwidth inference mechanism used [14]. For example, RLM uses *join-experiments* that can create additional traffic congestion, during transition periods corresponding to the latency for pruning a branch of the multicast tree. RLC [11] is a TCP-friendly version of RLM based on the generation of periodic bursts that are used for bandwidth inference on synchronization points indicating when a receiver can join a layer. Both the synchronization points and the periodic bursts can lead to periodic congestion and periodic losses [14]. PLM [15] is a more recent layered multicast congestion control protocol based on the generation of packet pairs to infer the available bandwidth. PLM does not suffer from the same pathological behaviors than RLM and RLC but requires a Fair Queuing network.

Battacharya & *al.* [16] present a general framework for the analysis of AIMD (Additive Increase Multiplicative Decrease) Multicast congestion control protocols. This paper shows that because of the so called "Path Loss Multiplicity Problem", unclever use of congestion information sent by receivers to sender, may lead to severe degradation and lack of fairness. This paper formalizes the multicast congestion control mechanism in two components : the Loss Indication Filter (LIF) and the rate Adjustement Algorithm. Our paper presents an implementation that minimises the Loss Multiplicity Problem by using a LIF which is implemented by a clustering mechanism (section 5.2) and a rate Adjustement Algorithm following the algorithm described in sections 4 and 6.

TFMCC [17] is an equation-based multicast congestion control mechanism that extends the TCP-friendly TFRC [18] protocol from the unicast to the multicast domain. TFMCC uses a scalable round-trip time measurements and a feedback suppression mechanism. However, since it is a single rate congestion control scheme, it cannot handle heterogeneous receivers and adapts its sending rate to the current limiting receiver.

FLID-DL [19] is a multi-rate congestion control algorithm for layered multicast sessions. It mitigates the negative impact of long IGMP leave latencies and eliminates the need for probe intervals used in RLC. However, the amount of IGMP and PIM SM sontrol traffic generated by each receiver is prohibitive. WEBRC [20] is a new equation-based rate control algorithm that has been recently proposed. It solves the main drawbacks of FLID-DL using an innovative way to transmit data in waves. However, WEBRC such as FLID-DL are intended for reliable download applications and possibly streaming applications but cannot be used to transmit real-time hierarchical flows such as H.263+.

A source adaptive multi-layered multicast - SAMM - algorithm based on feedback packets containing information on the estimated bandwidth available on the path from the source is described in [4]. Feedback mergers are assumed to be deployed in the network nodes to avoid feedback implosion. A mechanism based on "partial suppression" of feedback is proposed in [5]. This approach avoids the deployment of aggregation mechanisms in the network nodes, but on the other hand, the partial feedback suppression will likely induce a flat distribution of the requested rates.

MLDA [12] is a TCP-friendly congestion control scheme in which as in the scheme we propose, senders can adjust their transmission rate according to feedback information generated by receivers. However, MLDA does not provide a way to adapt the FEC rate in the different layers according to packet loss observed at receivers. Since the feedback only includes TCP-friendly rates, MLDA does not need feedback aggregation mechanisms and uses exponentially distributed timers and a *partial suppression* mechanism to prevent feedback implosion.

In [13], a rate based congestion and loss control mechanism for multicast layered video transmission is described. The strategy relies on a mechanism that aggregates feedback information in the networks nodes. However, in contrast with SAMM, the optimization is not performed in the nodes. Source and channel (FEC) rates in the different layers are chosen among a set of requested rates in order to maximize the overall PSNR seen by all the receivers. Receivers are classified according to their available bandwidth, and for each class of rate, two types of information are delivered to the sender: the number of receivers represented by this class and an average loss rate computed over all those receivers. It is supposed here that receivers with similar bandwidths have similar loss rates, which may not be always the case. In this paper, we solve this problem using a distributed clustering mechanism. In our approach, receivers with similar bandwidths, but with different loss rates are not classified within the same class. Therefore, with more accurate clusters, a better adaptation of the error-control process at the source level is possible. Moreover, the global optimization performed is different and leads to better results. Finally, we considered in this paper a new scalable video source.

3 Protocol overview

This section gives an overview of the rate control protocol proposed in this paper. Its design relies on a feedback tree structure, where the receivers are organized into a tree hierarchy, and internal nodes aggregate feedbacks.

At the beginning of the session, the sender announces the range of rates (i.e. a rate interval $[R_{min}, R_{max}]$) estimated from the average rate-distortion characteristics of the source. The value R_{min} corresponds to the bit rate under which the received quality would not be acceptable, and R_{max} to the rate above which there is no significant improvement of the visual quality. This information is transmitted to the receivers at the start of the session. The interval $[R_{min}, R_{max}]$ is then divided into subintervals in order to only allow relevant values for layers rates. This quantization avoids having non quality discriminative layers.

After this initialization, the multicast layered rate control process can start. The latter assumes that the time is divided into feedback rounds. A feedback round comprises four major steps :

- At the beginning of each round the source announces the number of layers and their respective rates, via RTCP sender reports (SR). Each source layer is transmitted to an IP multicast group.
- Each receiver measures network parameters and estimates the bandwidth available on the path leading to it. The estimated bandwidth and the layer rates will trigger

subscriptions or unsubscriptions to/from layers. Estimated bandwidth and loss rates are then conveyed to the sender via RTCP receiver reports (RR).

- Aggregation agents placed at strategic positions within the network classify receivers according to similar reception behaviors, i.e. according to a measure of distance between the feedback parameter values. On the basis of this clustering, these agents proceed with the aggregation of the feedback parameters, providing a representation of homogeneous clusters.
- The source then proceeds with a dynamic adaptation of the number of layers and of their rates in order to maximize the quality perceived by the different clusters.

The next sections describe in details each of the four steps.

4 Protocol functions supported by the receiver

Two bandwidth estimation strategies have been considered : the first approach measures the *goodput* of the path, and the second approach considered estimates the TCP-compatible bandwidth under similar conditions of loss rates and delays. This section describes the functions supported by the receiver in order to measure the corresponding parameters, and the multicast groups join and leave policy that has been retained. The bandwidth values estimated by the receivers are then conveyed to the sender via RTCP RR reports augmented with dedicated fields.

4.1 Goodput-based estimation

A notion of *goodput* has been exploited in the SAMM algorithm described in [3]. Assuming priority-based differentiated services for the different layers, the *goodput* is defined as the cumulated rate of the layers received without any loss. If a layer has suffered from losses, it will not be considered in the *goodput* estimation. The drawback of such a measure is that the estimated bandwidth will be highly dependent on the sending rates, hence it does not allow an accurate estimation of the link capacity. When no loss occurs, in order to best approach the link capacity, SAMM considers values higher than the goodput measured. Nevertheless, a loss rate of 0% is not realistic on the Internet. Experiments have shown that this notion of *goodput* in a best-effort network in presence of cross traffic leads to estimated bandwidths decreasing towards zero during the sessions. Here, the *goodput* is defined instead as the rate received by the end system. A simple mechanism has been designed to try to approach the bottleneck rate of the link . If the loss rate is under a given threshold T_{loss}, the bandwidth value B_t estimated at time t is incremented as

$$B_t = B_{t-1} + \Delta \tag{1}$$

where Δ represents a rate increment and B_{t-1} represents the last estimated value. Let g_t be the observed goodput value at time t. Thus, when the loss rate becomes higher than the threshold T_{loss}, B_t is set as

$$B_t = g_t. \tag{2}$$

In the experiments we have taken $t_{loss} = 3\%$ and the Δ parameter increases similarly to the TCP increase, i.e. of one packet per round-trip time.

4.2 TCP-compatible bandwidth estimation

The second strategy considered for estimating the bandwidth available on the path relies on the analytical model of TCP throughput [21], known also as the TCP-compatible rate control equation. Notice however that the application of the model in a multicast environment is not straightforward.

4.2.1 TCP throughput model

The average throughput of a TCP connection under given delay and loss conditions is given by [21]:

$$T = \frac{s}{RTT\sqrt{\frac{2p}{3}} + T_o min(1, 3\sqrt{\frac{3p}{8}})p(1 + 32p^2)}, \tag{3}$$

where p, RTT, s and T_o represent respectively the congestion event rate [18] [17], the round-trip time, the packet size and the retransmit timeout value of the TCP algorithm.

4.2.2 Parameters estimation

In order to be able to use the above analytical model, each receiver must estimate the RTT on its path. This is done using a new application-defined RTCP packet that we called (Probe-RTT). To prevent feedback implosion, only leaf aggregators are allowed to send Probe-RTT packets to the source. In case receivers are not located in the same LAN than their leaf aggregator, they should add the RTT to their aggregator; this can be easily estimated locally and without generating undesirable extra traffic. The source periodically multicast RTCP reports including the RTT computed (in ms) for the latest Probe-RTT packets received along with the corresponding SSRCs. Then, each receiver can update its RTT estimation using the result sent for its leaf agregator. The estimation of the congestion event rate p is done as in [18] and the packet size s is set to 1000 bytes.

4.2.3 Singular receivers

In highly heterogeneous environments, under constraints of bounded numbers of clusters, the rate received by some end systems may strongly differ from their requests, hence from the TCP-compatible throughput value. The resulting excessively low values of congestion event rates lead in turn to overestimated bandwidth values, hence to unstability. In order to overcome this difficulty, the TCP-compatible throughput B_t at time t is estimated as

$$B_t = min(T, max(S_{rate} + T_{rate}, B_{t-1})) \tag{4}$$

where S_{rate} is the rate subscribed to and T_{rate} is a threshold chosen so that the increase between two requests is limited (i.e. $T_{rate} = K * s/RTT$ with K a constant). B_{t-1} is the last estimated value of the TCP-compatible throughput. When the estimated throughput value T is not reliable, the history used in the estimation of loss rates is re-initialized using the method described in [18]. We will see in the experimentation results that the above algorithm is still reactive and responsive to changes in network conditions.

4.2.4 Slowstart

The slowstart mechanism adopted here differs from the approaches described in [18] and [17]. At the begining of the session or when a new receiver joins the multicast transmission tree, the requested rate is set to R_{min}. Then, after having a first estimation of RTT, To and p, T can be computed and the resulting requested rate B_t^{slow} is given by

$$B_t^{slow} = max(T, g_t + K \times s/RTT) \tag{5}$$

where g_t is the observed goodput value at time t and K is the same constant as the one used in section 4.2.3. The estimation given by (5) is used until we observed the first loss. After the first loss, the loss history is re-initialized taking g_t as the available bandwidth and proceeding with Eq. (4).

4.3 Join/Leave policy

Each receiver estimates its available bandwidth B_t and joins or leaves layers accordingly. However, the leaving mechanism has to take into account the delay between the instant a feedback is sent and the instant the sender adapts the layer rates accordingly. Undesirable oscillations of subscription may occur if receivers decide to unsubscribe a layer as soon as the TCP-friendly throughput estimated is lower than the current rate subscribed to. It is essential to let enough time for the source to adapt its sending rates, and then, only decide to drop a layer if the request has not been satisfied. It's why in order to still be reactive, we have chosen a delay of $K * RTT$ before leaving a layer except in the case where the loss rate becomes higher than a chosen acceptable bound T_loss. These coupled mechanisms permit to avoid a waste of bandwidth due to IGMP (Internet Group Management Protocol) traffic.

4.4 Signalling protocol

The aggregated feedback informations (i.e. estimated bandwidth, loss rate) are periodically conveyed towards the sender in RTCP receiver reports(RR), using the RTCP report extension mechanism. The RR are augmented with the following fields: