Improving user's satisfaction in adaptive multicast video

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Abstract

Adaptability is the most promising feature to be applied in future robust multimedia applications. In this paper, we propose the Direct Algorithm to improve the degree of satisfaction at heterogeneous destinations in an multi-layered multicast video environment. It relies on a mechanism that dynamically controls the rates of the video layers. This mechanism is based on feedback packets sent by the receivers. The main goal is to efficiently merge these feedback packets at intermediate nodes in order to avoid packet implosion at the source. The proposed scheme is optimized to achieve better global video quality and less bandwidth requirements. We also propose the direct algorithm with a virtual number of layers and discuss the use of multiple criteria when choosing the video layers. The results show that the proposed scheme leads to improved global video quality at heterogeneous receivers with no cost of extra bandwidth.

1 Introduction

Current network architectures are not well suited to provide the quality of service required by most multimedia applications. For multicast communications, the limitations are still subtler. Heterogeneity in intermediate and terminal equipment is the main drawback for the deployment of such applications.

These challenges are intrinsic to video distribution, since many video applications are multicast by nature. To deal with these problems, several strategies have been proposed to improve and guarantee a minimum quality of service. One approach is to use resource reservation in order to provide a certain level of guarantee [1]. An efficient resource reservation, however, requires a previous knowledge of the traffic characteristics, which is not always feasible. Another technique is to conceive adaptive multimedia applications. Introducing adaptability is seen as a key feature for the next generation of video applications. This approach consists in giving to the application a certain level of consciousness on the network condition [2, 3, 4, 5, 6]. In [7], two strategies are proposed to dynamically control the throughput of a traffic source: the variable compression and the hierarchical coding. For the variable coding, the source observe the available bandwidth and computes the minimum signal compression rate that should be applied in order to maximize the bandwidth utilization. In the hierarchical coding, each signal is coded in a base layer and in one or more improvement layers. These layers are correlated according to a causal law, and the quality of the presentation depends on the number of layers that are retrieved. The utilization of multi-layered video, however, introduces several challenges in the deployment of a multicast communication system.

Previous works have proposed some schemes to deal with the real-time aspect of video distribution. In [8], the source transmits one video flow to multiple IP destinations. Based on reports sent by the receivers, the source estimates the average video quality and the congestion level of the network to adjust the video rate. Since this scheme uses only one video layer, global video quality may be degraded due to heterogeneity in the receivers.

In the destination set grouping [9], the source maintains a certain number of video flows

derived from the same raw video. The receivers are classified according to their capabilities and each video flow is addressed to a particular group. Whereas this approach can lead to good fairness, it may be inefficient in terms of bandwidth utilization due to its redundancy of information.

In the receiver-driven layered multicast [10], the source generates a certain number of video layers and transmits each layer to a different IP-multicast group. The receivers subscribe to the number of groups they will. The communication is dynamic in the way receivers can dynamically join or leave groups. Nevertheless, they are limited to the layers the source decides to transmit.

In [11], a number of feedback control packets containing current congestion state are exchanged between the source and the receivers. Based on the information, the source estimates the number of video layers and the respective rates. To avoid implosion of feedback control packets at intermediate nodes, it is implemented an algorithm to merge feedback packets returned by the receivers.

In the literature, however, little attention has been given to the problem of merging feedback control packets in order to maximize the average level of satisfaction at the receivers. We propose in this paper the direct algorithm to efficiently merge feedback packets at intermediate nodes [12]. The main goal is to calculate the throughput of each video layer so as to increase the global video quality at the destinations. The global video quality is estimated in terms of the level of satisfaction at each destination, i.e., the received video quality compared with the maximum quality that could be received. Through the analysis of different network topologies, we prove that the proposed scheme leads to improved global quality of reception at heterogeneous receivers. Indeed, the results show that the proposed mechanism results

in less allocated bandwidth for greater level of user satisfaction. We also address in this paper the generation of feedback packets with a virtual number of layers. In this case, the intermediate nodes consider that the source can transmit more than its maximum capacity. We show that it results in an improvement in the average global video quality. As a subject for future works, we also discuss the use of multiple decision criteria to compute the video layers, for example the level of video degradation at the receivers, the global bandwidth utilization and the number of destinations that receive the maximum video quality.

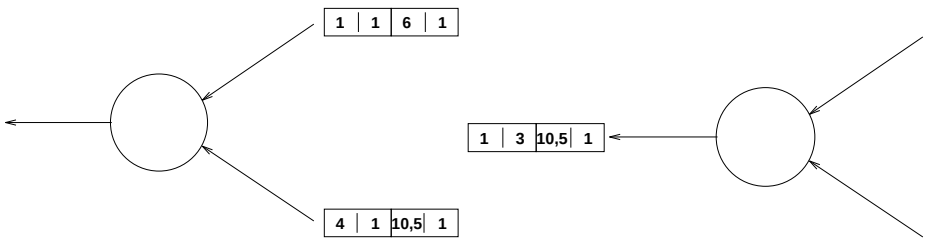
The following of this paper is organized in four sections. In section 2 we propose and analyze an algorithm to improve the global video quality at the receivers. In section 3, we discuss the use of multiple decision criteria for the computation of the video layers. Several analysis and simulations to evaluate the proposed scheme are described in section 4 and section 5 presents some concluding remarks.

2 Improving satisfaction at receivers

We proposed in this paper an efficient algorithm to merge feedback packets at intermediate nodes. This algorithm is also used to calculate the rates of the multiple layers at the source. The computation of the video layers is based on feedback packets sent by the receivers [11]. In this scheme, the source periodically multicasts control packets to the destinations. As these control packets traverse the network, intermediate nodes mark them with the amount of available bandwidth. After receiving the feedback packets, each destination knows the resource constraints within the path from the source. Then, receivers generate returning feedback packets that contain the maximum rate they can receive. As these packets are

sent to the source, intermediate nodes merge upcoming concurrent packets in order to avoid feedback packet implosion.

In the control scheme proposed in [11], the arriving feedback packets are stored in a temporary array with the video rates disposed in a crescent order. If this temporary array has a number of entries greater than the maximum number of layers that the source can generate, then the node performs an algorithm to reduce the number of entries. The algorithm discards the entry j that results in the larger "goodput rate" (G). The goodput rate is the total throughput received by the destinations without loss and is calculated as $G = \sum r_i \times c_i$, where r_i is the rate required by c_i receivers. The number of destinations in the discarded entry is added to the number of destinations in the $(j-1)$th entry. These steps are repeated until the number of entries in the temporary array is reduced to the maximum number of layers that the source can generate.



(a) arriving packet (b) merged packet

Figure 1: Example of the merging procedure.

Although discarding the entry that results in the greater goodput rate, the global "level of satisfaction" at the receivers is not optimized. For instance, suppose the circumstances shown in figure 1(a). Consider also G_j as the goodput rate when discarding entry E_j. In the first step of the merging procedure, we have four entries in the temporary array. Since the

maximum number of video layers that the source can transmit is two, we must then discard two entries[1]. The sequence below shows each step of the reducing algorithm.

Step 1 $G_4 = 1 \times 2 + 6 \times 1 + 10,5 \times 1 = 18,5$

$G_6 = 1 \times 1 + 4 \times 2 + 10,5 \times 1 = 19,5$

$G_{10,5} = 1 \times 1 + 4 \times 1 + 6 \times 2 = 17$

G_6 is the maximum goodput rate, then this entry is eliminated and $c_4 = c_4 + c_6$.

Step 2 $G_4 = 1 \times 3 + 10,5 \times 1 = 13,5$

$G_{10,5} = 1 \times 1 + 4 \times 3 = 13$

In the step 2, G_4 is greater than $G_{10,5}$. Thus, E_4 is discarded and the node forwards the resulting feedback packet, as shown in figure 1(b). In conformity with this packet, the receivers in the sub-tree should receive two video layers, one at 1Mbps and the other at 9,5Mbps. However, these values are not optimal in the case of the example of figure 1. The best result is the one that leads to the greatest level of user's satisfaction. In the example, there are four destinations in the sub-tree. The correspondent capabilities are r_1=1Mbps, r_2=4Mbps, r_3=6Mbps and r_4=10,5Mbps, respectively. If the source sends two layers at L_1=1Mbps and L_2=9,5Mbps, destinations D_1 and D_4 will be entirely satisfied, but D_2 and D_3 will not be served with their maximum capabilities. D_2 will have a degradation of 3Mbps (capacity=4Mbps, received video = 1Mbps) and D_3 a degradation of 5Mbps (capacity=6Mbps, received video = 1Mbps).

Table 1 shows the level of global degradation for all possibilities of video rate. It should be emphasized that the smallest global degradation corresponds to the largest global video quality at the destinations. We can observe that the best global video quality corresponds to the base layer at 1Mbps and the enhancement layer at 6Mbps. The merging algorithm

[1]Since the first entry corresponds to the receivers with less capabilities, it is never discarded.

presented above, however, does not lead to the best values of video rates. This is because the algorithm performs the packet reduction in successive steps.

Table 1: Global average video degradation.

	$L_1 = 1; L_2 = 3$		$L_1 = 1; L_2 = 5$		$L_1 = 1; L_2 = 9,5$	
Destination	Received	Degradation	Received	Degradation	Received	Degradation
$D_1 = 1$	1	0	1	0	1	0
$D_2 = 4$	4	0	1	3	1	3
$D_3 = 6$	4	2	6	0	1	5
$D_4 = 10,5$	4	6,5	6	4,5	10,5	0
Global Degradation		8,5		7,5		8

Consider that a node must reduce a temporary packet that contains a fragment as shown in figure 2. In the picture, f_i is the required video rate and f_i^* corresponds to the number of destinations that want to receive this rate.

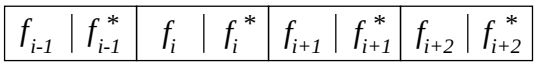


Figure 2: Temporary array.

Consider that the best values for the layers are f_{i-1} and f_{i+1}. This happens if

$$f_i^*(f_i - f_{i-1}) > f_{i+1}^*(f_{i+1} - f_i) < f_{i+2}^*(f_{i+2} - f_{i+1}). \quad (1)$$

The successive method does not lead to this result if

$$f_i^*(f_i - f_{i-1}) < f_{i+1}^*(f_{i+1} - f_i) + f_{i+2}^*(f_{i+1} - f_i) \tag{2}$$

and

$$f_{i+2}^*(f_{i+2} - f_{i+1}) < f_{i+1}^*(f_{i+1} - f_i) + f_{i+1}^*(f_i - f_{i-1}). \tag{3}$$

Combining equations (2), (3) and (1), we have that if

$$f_{i+1}^*(f_{i+1} - f_i) < f_i^*(f_i - f_{i-1}) < f_{i+1}^*(f_{i+1} - f_i) + f_{i+2}^*(f_{i+1} - f_i) \tag{4}$$

and

$$f_{i+1}^*(f_{i+1} - f_i) < f_{i+2}^*(f_{i+2} - f_{i+1}) < f_{i+1}^*(f_{i+1} - f_i) + f_{i+1}^*(f_i - f_{i-1}) \tag{5}$$

are satisfied, then the global video quality is not optimized if we use the successive method.

To avoid the situations as in the above example, this paper proposes the Direct Algorithm to merge feedback packets. This algorithm computes the final merged feedback packet in only one step. In a general case where the source can generate L video layers ($C_1 ... C_L$) and the packet that must be reduced has N entries, the direct algorithm performs as follows. We make $C_1 = f_1$ and we calculate $C_2 ... C_L$ so as to minimize the *loss sum* (L_S), calculated as

$$L_S = \sum_{i=2}^{L} f_i^* \times (f_i - r_k), \tag{6}$$

where k is the maximum number of layers that the rate f_i can support[2], and $r_k = \sum_{j=1}^{k} C_j$.

[2]For example, if $f_i = 10$ and we have three layers at 5, 4 and 3, then $k = 2$, because the receiver is able to receive only the first two layers.

The values of $C_2 ... C_L$ are calculated in order to coincide with a combination of f_j. In the direct algorithm, all of the exceeding entries are discarded in the same loop. Furthermore, determining $min\{L_S\}$ we guarantee that the global average video quality is locally optimized at the intermediate nodes. However, depending on the number of intermediate nodes that perform the merging algorithm, entries that should be chosen as actual video layers may be inappropriately discarded.

Consider the situation depicted in figure 3. In this example, the source can transmit up to two video layers. In node N_2, the entry that corresponds to the destination that wants to receive 5Mbps is discarded, and this information is lost. If this information had not been eliminated, the algorithm would find the rates that lead to the best global video quality, i.e., 1Mbps and 5Mbps. In this case, the global degradation would be 8Mbps. When performing the packet merging at N_1, however, the obtained throughputs are 1Mbps and 10Mbps, which results in global degradation of 10Mbps.

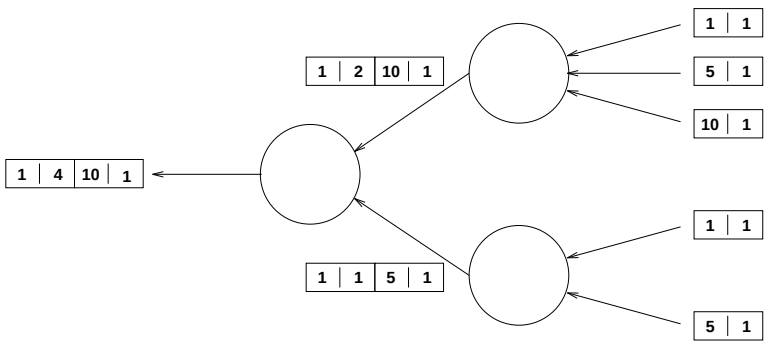


Figure 3: Example of information loss when merging packets.

To minimize this problem, we also propose the merging algorithm with virtual layers. In this algorithm, the intermediate nodes consider that the source can transmit more than the actual number of layers. This strategy may avoid information loss as in the example of

figure 3. The intermediate nodes are informed that the source can transmit V layers, with $V > C$ (C is the actual number of layers transmitted by the source.)

In the same example above, consider that $V = 3$. The merging algorithm is not performed at node N_2. Consequently, the entry that corresponds to the receivers that want to receive 5Mbps is not discarded. At node N_1, this information is still existing. When performing the merging algorithm, node N_1 discards the entry that stores 10Mbps, which leads to the best result.

3 Multiple decision criteria

We have proposed the direct algorithm to efficiently distribute layered video among heterogeneous receivers. The proposed scheme computes the video layers in such a way to minimize the global level of degradation at the receivers.

In this section we argue that the computation of the video layers should also take into account some other premises. Consider the example shown in figure 4, where one destination wants to receive 1Mbps, two destinations want the video at 4Mbps, and one receiver that wants to receive 10Mbps. In this example, the source is limited to transmit no more than 2 video layers.

The rate of the base layer is set to 1Mbps. For the enhancement layer, the source has to choose between 3Mbps and 9Mbps. Notwithstanding, in both cases the global level of video degradation is 6Mbps, resulting in a deadlock decision since no other rules have been established to deal with this sort of problem.

If the source decides to transmit the enhancement layer at 3Mbps, a greater number

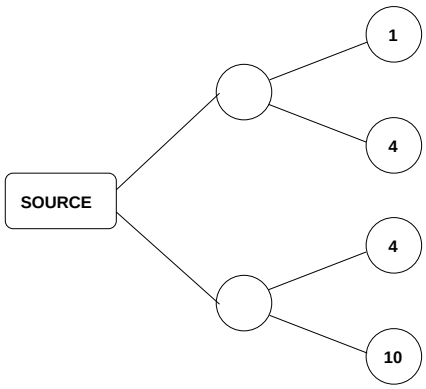


Figure 4: Example of deadlock decision.

of less powerful receivers is benefited. If the enhancement layer is set to 9Mbps, the source considers that one destination with more capacity should be better satisfied than two limited destinations. As a matter of fact, this decision must rely on the type of the application. For video on the demand, for example, it may be more interesting that the destination with more capacity receives the better video quality. On the other hand, in a environment of videoconference the best choice might be to satisfy a number of receivers as great as possible.

Besides the number of destinations, it is also important to take into account the quantity of the required bandwidth. In the same example of figure 4, if the source transmits the layers at 1Mbps and 3Mbps, the global bandwidth utilization is 4Mbps. In the other case (layers at 1Mbps and 9Mbps), the global bandwidth utilization is 10Mbps, even leading to the same value of quality degradation.

4 Analysis

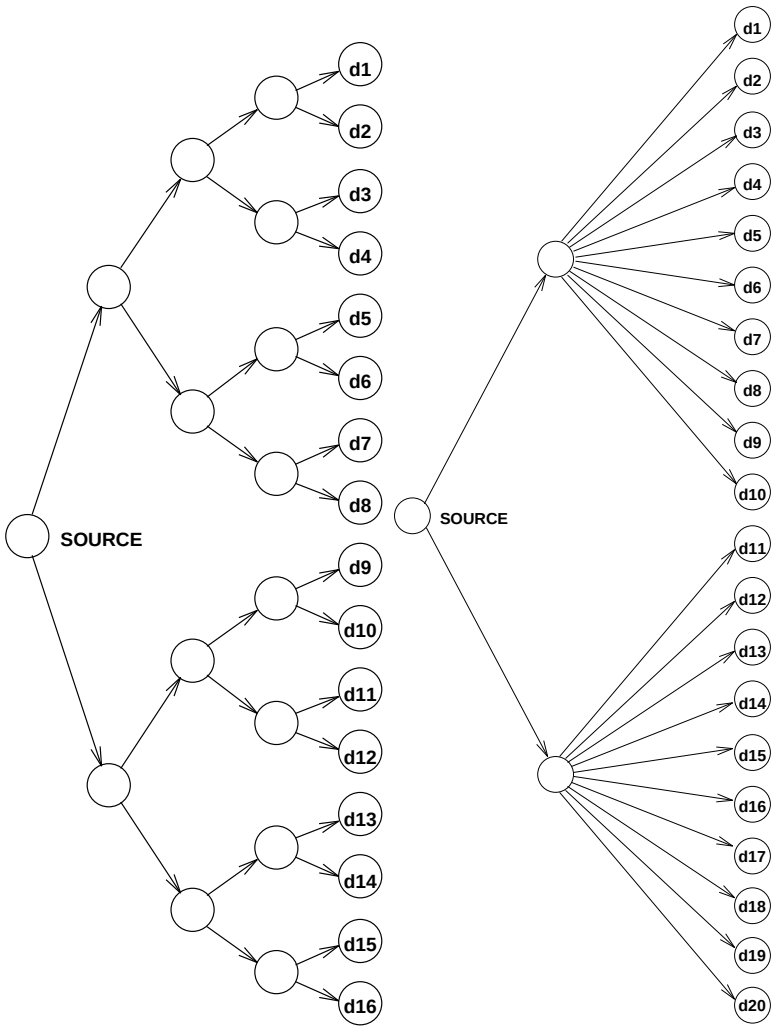
In this section, we analyze the proposed algorithm and evaluate the level of video degradation and the network bandwidth utilization at the receivers. We have performed several simula-

tions upon different scenarios. Each scenario corresponds to a multicast video session with one sender and several destinations. We consider in our simulations a source with different capabilities, i.e., with different values for the maximum number of layers. We also evaluate the tradeoffs between the number of layers and the level of satisfaction and to analyze the introduction of virtual layers at the intermediate nodes.

The first topology is shown in figure 5(a). This topology represents a network with the receivers sparely distributed among the branches. We have also performed simulations for a multicast session with concentrated receivers (figure 5(b)). The receivers' capabilities were randomly generated with average rates between 1 and 15.

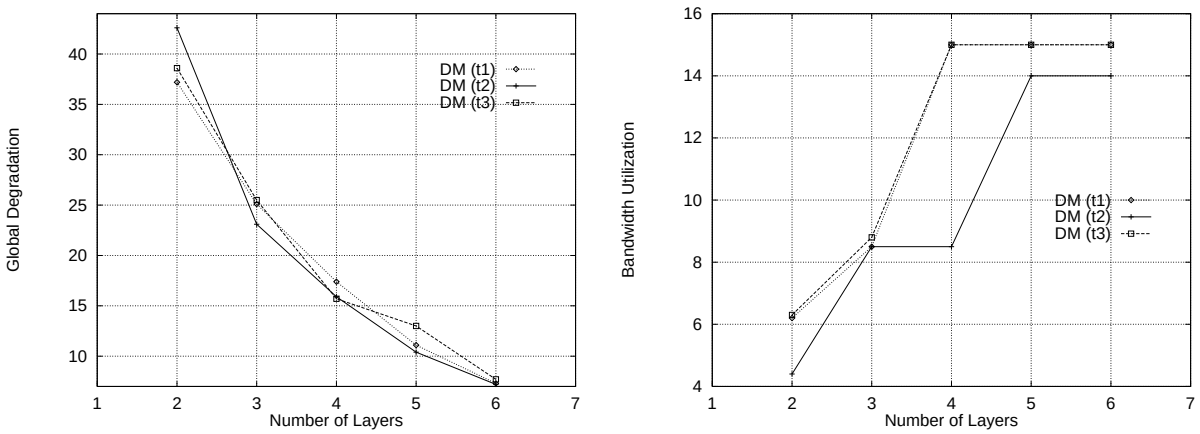
Figure 6(a) shows the global video quality degradation for the direct method. The curves correspond to the results for three different merging loops at $t = t_0$, $t = t_1$ and $t = t_2$. The correspondent levels of bandwidth utilization are shown in figure 6(b). We observe from these two graphics that, for example, at $t = t_2$, the bandwidth utilization is the same if the source transmits 3 or 4 layers. When transmitting 3 layers, however, the global degradation is approximately 60% greater than if the source transmits 4 layers. The source should balance extra complexity (when increasing the number of layers) and quality to decide how many layers should be sent.

In figure 7, the direct method is evaluated for different numbers of virtual layers. In this example, the source is supposed to transmit up to 3 video layers for figure 7(a) and up to 4 video layers for figure 7(b). In both cases, the curves saturate for a number of virtual layers greater than 6, which corresponds to the maximum average level of satisfaction at the receivers. In this case, the cost of additional complexity is not compensated by an increase in the number of layers.



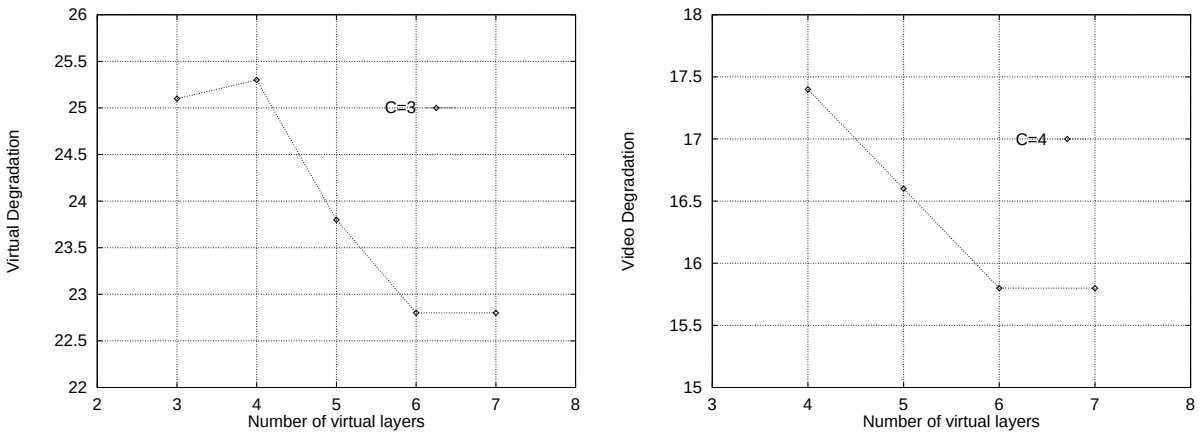
(a) (b)

Figure 5: Simulation topologies: (a) uniformly distributed and (b) concentrated.



(a) (b)

Figure 6: Topology 1: (a) global degradation and (b) bandwidth utilization.



(a) (b)

Figure 7: Source transmitting (a) 3 layers and (b) 4 layers.

To compare the direct and the successive method, we have simulated the two topologies for the number of video layers varying from 2 to 6. Table 2 shows some results obtained for several values of C. For the direct method, the number of virtual layers is set to $L = C + 3$. Observe that the direct algorithm results in better global video quality without requiring additional bandwidth.

Table 2: Global average video degradation.

Topology 1

	Direct Method		Successive Method	
C	Degradation	Utilization	Degradation	Utilization
2	361	71	413	85
3	228	8	233	8
4	158	15	174	15
5	111	15	111	15

Topology 2

	Direct Method		Successive Method	
C	Degradation	Utilization	Degradation	Utilization
2	539	75	539	75
3	339	115	421	115
4	223	115	223	115
5	163	143	188	143

5 Conclusions

In this paper, we have addressed the problem of managing control information for multicast distribution of multi-layered video. We proposed the direct algorithm to merge feedback packets at intermediate nodes. Contrary to other proposals found in the literature, this algorithm performs the merging procedure in only one loop. This leads to improved global level of satisfaction at heterogeneous destinations. The results show that the proposed method achieves lower degradation without additional allocated bandwidth.

We have also proposed the direct method with virtual number of layers, i.e., the intermediate nodes consider that the source can transmit a number of layers greater than its maximum capacity. The simulations show an improved global video quality since merging procedures at intermediate nodes that might discard key entries are no longer performed. In some cases, the algorithm achieves the maximum level of satisfaction, i.e., as if there were no packet merging.

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