Evaluation of Deficit Round Robin Queue Discipline for Real-time Traffic Management in a RTP/RTCP Environment

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***Abstract*—Multimedia real-time traffic is deemed to be dominant in future communication systems. One of the reference applications to support real-time traffic is the Real-time Transport Protocol (RTP), which can be used to transmit multimedia contents on real-time basis. At the same time, Real-time Transmission Control Protocol (RTCP) is used for receiving feedback and getting information about the network. This paper proposes and evaluates a traffic management implementation in such an RTP/RTCP environment for congestion control. Deficit Round Robin queue discipline is used as the traffic management strategy instead of Random Early Detection and DropTail queue disciplines. A simulation campaign was performed to analyze the effects of implemented traffic strategies in RTP/RTCP environment and compare it with previous solutions. The obtained results highlight a significant difference in terms of jitter delay and packet losses and improvement the bandwidth utilization for real-time flows. Thus, we are able to provide quantitative evidence of the importance of the queue discipline to efficiently manage multimedia content.**

***Index Terms*—Multimedia traffic, real time protocol, real-time transmission control protocol, congestion control, queueing, deficit round robin, random early detection.**

I. Introduction

IT IS expected that next generation communication system will deliver an ever increasing amount of multimedia traffic. Therefore, the capability to transmit this traffic, complying with real-time quality of service (QoS) constraints, is commonly regarded as one of the major upcoming research challenges for the next years.

The present paper investigates transport layer solutions for real-time delivery, especially focusing on the choice of the queue discipline for multimedia flows. In this sense, it is important not only to achieve high efficiency of the queueing policy, but also to be able to correctly manage different kinds of traffic. In fact, multimedia traffic comprises several applications with different characteristics in terms of required QoS. Moreover, it is expected that multimedia traffic will coexist with other best effort data traffic in the same network operations.

Technological solutions to achieve real-time delivery over the internet include in particular the Realtime Transmission Protocol (RTP) and the Real-Time Control Protocol (RTCP) [1]. In [2], RTP/RTCP environment was introduced and implemented within the well-known ns-2 simulator [3].

Along the lines of [2], which we also adopt in this paper, the main functionality of RTP is modeled as involving the identification of payload, the generation of RTP packets, and finally the introduction of RTP packets time stamps and sequence numbers. Real-Time Transmission Control Protocol (RTCP) is used for inquiring the network status and getting feedback. The major advantage of RTCP is that it does not interact with RTP, but it can be used as a network management entity. Thus, RTP and RTCP can operate jointly as direct and feedback loop.

The data exchanged by the nodes through this mechanism enter a buffer queue at each intermediate receiver. One basic cause of delay in the transmission of multimedia traffic is actually the queueing delay at these buffers. Thus, when multimedia or real-time traffic is concerned, it is important to select the correct type of queueing policy in order to provide the users with the required QoS.

To this end, different choices are possible. Previous existing work utilizes very simple queue disciplines, such as a basic DropTail policy [4]. In this paper, we propose to use, within the RTP/RTCP framework, a Deficit Round Robin (DRR) strategy. This is justified by several theoretical benefits, which we aim at validating in practice.

Dynamic Bandwidth allocation for real-time traffic is an important issue and is also supported by [11]. The authors are using queuing delay parameter to tune the congestion control mechanism. But this work does not mainly focus on a particular queue discipline; rather it tries to determine whether to increase the packet train by observing the current queueing delay.

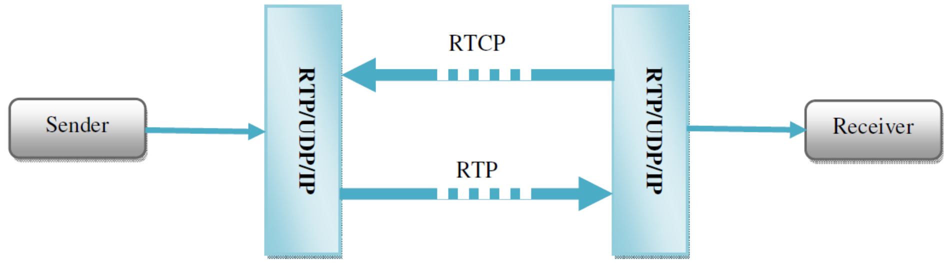

Fig. 1. The RTP/RTCP workflow

As described by [12] the delay jitter parameter is introduced by the congestion in the network as well as inadequate queue discipline usage. Therefore, selecting the proper queue discipline can play a tremendous role in the efficiency of real-time traffic over congested network.

Finally, we implemented this policy within the ns-2 simulator and we assess its performance in a test topology by means of a simulative campaign, evaluating several metrics of interest. In this way, we are able to verify that the proposed solution is properly able to fulfill QoS requirements of multimedia traffic.

The rest of this paper is organized as follows. In Section II, the RTP/RTCP environment is described. In Section III we illustrate our proposed solution, based on the employment of DRR, and presents a test topology to evaluate it. In Section IV we discuss simulation results. Finally, Section V concludes the paper and reviews the overall significance of the proposed approach.

II. Description of the RTP/RTCP Environment

RTP was developed by the Audio-Video Transport Working Group. It uses regimented packet format for multimedia contents that is, audio and video [1]. It was designed for multicast applications. RTP provides the following services: Identification of payload, time stamping, sequence numbering, and delivery notification. It provides end-to-end network transport functions, but is not responsible for guaranteed QoS. Therefore, RTP and RTCP are most frequently used in a joint manner, because RTP is used to transport multimedia data and RTCP is used for monitoring the network QoS [7]. A scheme depicting this interaction is represented in Fig. 1.

For multimedia sources, adaptive transmission rate algorithm is introduced in [8] using a TCP friendly rationale. The algorithm considers the maximum transmission rate, minimum transmission rate and the granularity. In the adaptive algorithm, the sender changes its transmission rate according to the adaptive algorithm schema. The authors of [8] have simulated TCP-friendly and constrained TCP-friendly flow control and their results proved that the constrained TCP-friendly version reaches a higher degree of fairness than the plain TCP-friendly one. This combination of RTP for real-time flows and RTCP to monitor the QoS of the network was also used in [9], where new feedback control mechanism for video transmissions are presented. By means of simulation, this contribution compares the packet losses of UDP flow against the UDP_RTP flow. Results proved that UDP_RTP can improve the video transmission.

RTCP was also used in [13] to get the network information and tune the system accordingly for real-time traffic. But the emphasis of this paper is on multimedia traffic management in ad hoc networks. Authors are also using RTCP for getting end-to-end feedback information about the packet loss performance as well as delay jitter. RTP/RTCP protocol suite is well known for getting feedback from the receiver and it is also utilized by [14] to exemplify the real-time traffic flow for unicast and multicast environment. In the present analysis, we are also focusing on multicast scenario, particularly for real-time traffic. The authors of [14] present a vast survey of multimedia synchronization and the main technique is exemplified by the existing RTP/RTCP suite. Congestion Control is a challenging issue particularly for real-time traffic, this argument is also supported by [15], where however it is proposed to use RED queuing discipline for congestion control or for dropping the packet at the time of congestion. However, this may violate the need for real-time traffic to receive fair bandwidth allocation and especially the requirement for lossless delivery of information.

Therefore, we also take a RTP/RTCP environment as the starting point of our evaluation. However, as will be argued in the next sections, we stress the importance of an efficient queueing discipline at the buffers. For this reason, in the following we review and discuss this point. Further, we comparatively evaluate different choices in this respect.

III. Outline of the Proposed Solution

The simplest queue discipline, called DropTail, follows a very basic policy, i.e., it treats all the packets equally in a single queue, and each packet is served in the same order as received. It involves very low computational complexity and easily predictable behavior [4]. Among the drawbacks of this queue discipline are increased delay, jitter and packet losses for real-time applications. Further, queuing delay increases the congestion.

To perform an improved congestion control, Random Early Detection (RED) can be used. This queue discipline uses one queue and one dropping probability, which is used to determine whether to process the packets further or discard them. However, one major drawback of this queue discipline is loss of packets due to dropping. Therefore, by dropping packets it decreases the QoS perceived by the users. Real-time flows are more sensitive; therefore, one should avoid such mechanism, which harms the quality of sensitive traffic. RED algorithm of Active Queuing Management is used to get high throughput, but average queue length is a very sensitive parameter for the level of congestion control [5]. Finally, the policy is unable to setup a proper value for the queue length, a task which is left to the network operators, according to the type of the networks and the specific needs of the system.

Deficit Round Robin (DRR) is a fair queue discipline and is much more efficient than the previous fair queuing algorithms [6]. It is an extension of WRR Weighted Round Robin (WRR). In spite of its advantages, DRR is still an easily implementable queue discipline and has $O(1)$ running time per packet. Moreover, it can handle various sizes of packets, which makes it suitable for heterogeneous multimedia applications.

The buffer management scheme used in previous RTP/RTCP framework, such as [2], is simply to employ Droptail at the sender and receivers ends, whereas RED (Random Early Detection) queue discipline is used at the router ends. The RED queuing policy uses priority levels to drop packets; however, since the RED queue is applied at the link between the two routers, therefore it treats all the flows identically. Actually, RTP flows should be given some priority over non real-time flows. Therefore, considering the jitter and packet losses as QoS metrics, jitter was introduced due to the employment of Droptail and packet loss was increased due to the packet dropping by the RED queue. Alternatively, traffic management strategy can be applied to this scenario for real-time flows, so that real-time can be given higher priority over non real-time traffic. This will result in enhancing the QoS in terms of increased share of bandwidth and decreased jitter and packet losses for real-time flows. In the proposed traffic management strategy, Deficit Round Robin (DRR) is used as a queue discipline for the source and destinations of real-time flows as well as for the link between the routers.

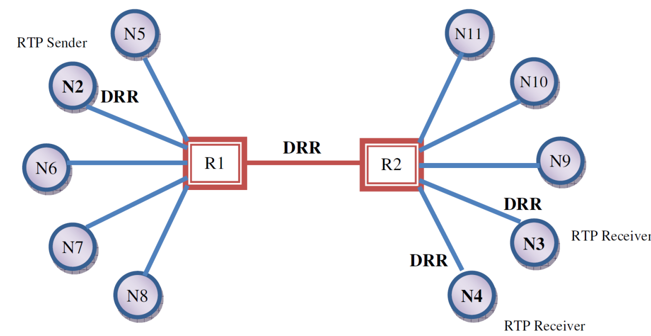

Fig. 2. Evaluation network topology

The network topology is described in Fig. 2. In the above figure, N2 is the node 2, which the sender of RTP flows, R1 and R2 are the edge routers and N3 and N4 are the receiver nodes. Instead of simple DropTail strategy, we have implemented a DRR queue discipline; in this manner, the jitter delay and the packet loss rate are significantly decreased, whereas the bandwidth utilization for real-time flows is improved. Results have been proven through simulation, described in the following section.

IV. Performance Evaluation

We utilized the ns-2 environment [3] to perform a simulation campaign. Focusing on a topology as previously described, we implemented the DRR policy for its use at the queueing buffers. The RTP sender and receiver use the DRR queue, similarly, DRR is also used instead of RED queuing system, but for all the other nodes Droptail is used. The queue length is set to 50 packets for each of the queue, whereas for the DRR queue between the routers has the capacity of 100 packets. Further, RTP-RTCP multicast environment is arranged in such a way that node 2, node 3 and node 4 join the multicast group, where node 2 is the transmitter of RTP traffic whereas node-3, node-4 are the RTP traffic receiver.

We simulated 100 seconds of transmissions and we evaluated the resulting system performance in terms of three different metrics: bandwidth allocation, jitter, and smooth loss. We compare our results with those reported in [2], where no specific traffic management strategy was adopted; thus, the curves referring to this approach are labeled “without traffic management.” We will show instead that in our proposed solution, buffer management plays very important role in terms of bandwidth allocation and other traffic management aspects; for this reason, we refer to our solution as “with traffic management” in the graphs. The following subsections detail the analysis for each of the investigated metrics.

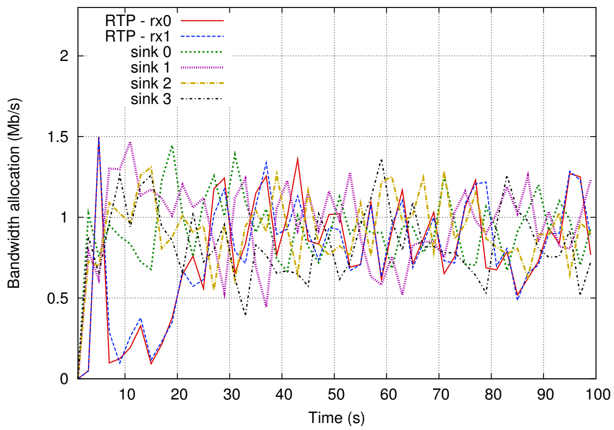

Fig. 3. Bandwidth allocated to the flows without traffic management

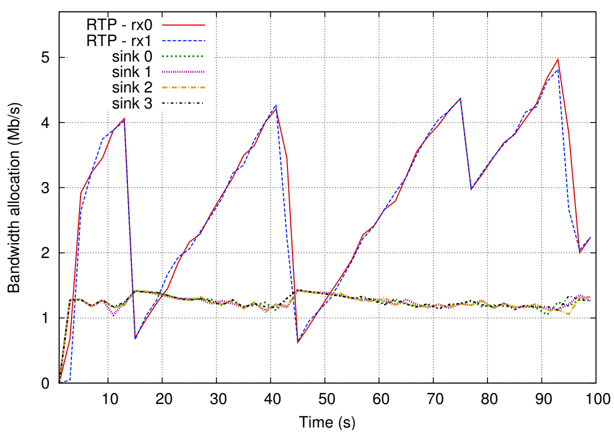

Fig. 4. Bandwidth consumptions of the flows with traffic management

Fig. 5. Jitter analysis

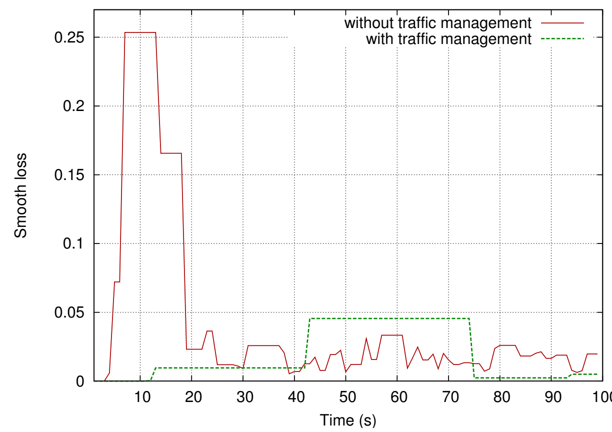

Fig. 6. Smooth loss analysis

A. Bandwidth Allocation

Figs. 3 and 4 compare the bandwidth allocation of the flows, for the two cases without and with traffic management, respectively. In Fig. 3, it is observed that all the flows get almost equal (and not very high) amount of bandwidth. No preference is given to real-time flows. Finally, the bandwidth allocations of all flows fluctuate considerably, and in a very variable manner from flow to flow.

Conversely, when we apply our proposed traffic management strategy, there is a great difference of bandwidth consumption between the real-time flows and the others. Fig. 4 shows that the two RTP flows get a higher amount of bandwidth than what reported in Fig. 3, and also they receive a better allocation than the other flows, which is a sign that real-time traffic is correctly provided with better QoS than best effort traffic. It is also worth observing that the oscillations are significantly reduced. Still, they are not avoided; observe, for example, that the bandwidth assigned to real-time flows drops in many points, due to congestion. Yet, the plots of Fig. 4 highlight a more regular behavior, that is, similar kinds of flow enjoy similar QoS at the same time.

B. Jitter

Thanks to its better traffic management capability, our proposed strategy is also able to effectively decrease the jitter values as compared to the previous solution with Droptail and RED. Fig. 5 shows a comparison of the jitter values for multimedia nodes between the solution of [2] and new proposed solution with traffic management for RTP/RTCP environments. We distinguish jitter values for our proposed strategy "with traffic management" and the implementation of [2] which is referred to as "without traffic management."

The plot emphasizes that jitter is significantly decreased when our traffic management technique is applied on

RTP/RTCP environment. It is also important to notice that the jitter values decrease and become smoother as time goes by, but still the curve of the proposed queue discipline stays considerably below the one without traffic management features.

C. Smooth Loss

Real-time traffic is very sensitive and requires more priority as compared to non real-time flows. Thus, by applying some traffic management strategies, real-time traffic can be given precedence over non real-time. Thus, when real-time traffic is prioritized, packet losses are likely to be decreased for real-time flows. This is investigated in Fig. 6, which represents the smooth losses [10] comparison between the previous RTP/RTCP solutions with our solution containing the traffic management component. The smoothing factor is set to $\alpha = 0.9$. Again we report two plots, "with" and "without" our traffic management approach.

The considerably better behavior in terms of packet losses is expected, depending on the application in use, to reflect in an enhanced QoS for real-time flows.

V. Conclusions

We considered an RTP/RTCP environment where RTP is used to transmit multimedia data and RTCP is used in conjunction with RTP to get the network statistics and maintain the overall end-to-end network QoS in a feedback-based manner. In related research work, RED queue discipline, or even simpler DropTail, are used to manage multimedia traffic. However, there are other queue disciplines, which can improve the transmission of RTP traffic. When applied to this scenario, Deficit Round Robin is able to significantly improve the performance in terms of bandwidth allocation, jitter, and packet losses for real-time flows.

The advantages of DRR have been proven by means of simulation results comparing existing solutions with a proposed approach based on DRR, in terms of all these performance metrics. The correct choice and setup of queueing policy at the network nodes has been proven to be key for meeting QoS constraint of real-time multimedia traffic.

Possible extensions of the present work include the analysis of similar traffic management strategies in different topologies and/or with different combinations of existing queue disciplines. Moreover, it is also possible to envision the application of similar approaches to RTP/RTCP environments realized over wireless networks.

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