TFRC-BASED SELECTIVE RETRANSMISSION FOR MULTIMEDIA APPLICATIONS

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Abstract

Multimedia applications are becoming increasingly popular in IP networks, while in mobile networks the limited bandwidth and the higher error rate arise in spite of its popularity. These are restraining factors for mobile clients using multimedia applications such as voice over IP (VoIP) or video streaming. In some conditions the retransmission of lost and corrupted packets should increase the quality of the multimedia service, but these retransmissions should be enabled only if the network is not in congested state. Otherwise the retransmitted packet will intensify the congestion and it will have negative effect on the audio/video quality. Our proposed mechanism selectively retransmits the corrupted packets based on the TCP-Friendly Rate Control (TFRC) and the actual video bit rate.

1. Introduction

The number of newly appeared multimedia applications is growing intensively in the IP based networks. With the rise of multimedia and network technologies, multimedia has become an indispensable feature on the Internet. Animation, voice and video clips become more and more popular on the Internet. Multimedia networking products like Internet telephony, Internet TV, video conferencing have appeared on the market. These applications are not only used in reliable wired networks but also in wireless environment where the obstacles of the expansion are the higher bit error ratio of the radio link and the limited bandwidth of the mobile links. Third-generation wireless networks are rapidly approaching reality, also providing higher bandwidth levels with the ability to transmit video streams in acceptable quality.

The real-time applications usually encode audio/video in a format that handles loss of full packets. Loss-tolerant real-time multimedia applications such as video conferencing or video streaming prefer UDP to avoid unacceptable delay introduced by packet retransmissions. UDP is considered selfish and ill-behaving because TCP throttles its transmission rate against the network congestion whereas UDP does not have such control mechanisms [1]. Some of the nowadays investigated transport protocols

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(e.g. UDPLite [2], SCTP [3], DCCP [4], etc.) can be more efficient for audio/video streaming applications. The unreliable UDP, UDPLite and DCCP do not retransmit any corrupted packets while SCTP will do it until all the packets arrive correctly to the client. These protocols basically do not adapt themselves to the actual conditions nevertheless it would lead to the increase of effectiveness. When the conditions make it possible to retransmit the lost or damaged packets it is worth to do it, but in some cases the effect of the retransmission is harmful. When the network is in congested state or the RTT (round-trip-time) is so high that the retransmitted packet will not arrive in time, the retransmission will not increase the quality; moreover will increase the load and latency.

The rapid growth in the usage of streaming media has heightened the need for a congestion control protocol suitable for streaming media. Among the proposed streaming-media congestion control protocols, TCP-Friendly Rate Control (TFRC) [5] is one of the promising solutions. TFRC maintains an equal or lesser average sending rate as competing TCP connections, while providing a relatively smooth sending rate to help packets to meet the real-time constraints required by streaming media.

Among the unreliable transport protocols only the Datagram Congestion Control Protocol (DCCP) supports congestion control mechanisms (TCP-Like, TFRC). The congestion control mechanism needs information about the packet loss event; hence the DCCP header includes a sequence number field that identifies the packet. Consequently the streaming server gets information which packet was lost and which was received.

In this paper a new TFRC controlled selective retransmission scheme is proposed for multimedia transmission over noisy wireless channels in order to ensure acceptable video quality at the receiver. To analyze the effectiveness of our approach we used the ns-2 network simulator [6].

The rest of the paper is organized as follows. A review of related work in selective retransmission and TFRC-based video streaming is presented in Section II. In Section III we propose a congestion sensitive retransmission method for multimedia applications. The obtained results are presented in Section IV. Finally, we summarize our paper and outline our future work in the last section.

2. Related Work

The nowadays used transport protocols can be categorized as reliable or unreliable. The reliable protocols like TCP, SCTP retransmits all the lost packets while the unreliable ones like UDP, UDPLite, DCCP do not. For video streaming applications typically the unreliable protocols are used.

Video sequences are compressed in a format such as MPEG to achieve bandwidth efficiency. Video compression exploits redundancy between frames to achieve higher compression. However packet loss can be detrimental to the compressed video with interdependent frames because errors potentially propagate across many frames. The MPEG video frame structure consist of, I picture, intra coded, coded independently of other frames, P or predictive picture, predicted from the previously decoded picture and B or bi-directionally predictive picture, predicted from one previous and future picture. This is the motivation to protect the important frames like I pictures to avoid the propagation of errors.

Attempts were made to implement a selective retransmission protocol with a decision algorithm [7][8][9]. This algorithm decides whether or not to request a retransmission for a message that was detected as lost. The decision to retransmit is determined by the Euclidean distance calculated by using the loss and latency ratio. This protocol does not examine the reason of the packet loss and does not use congestion avoidance mechanisms.

Other approaches describe a method that includes categorizing groups of packets in order of importance [10][11]. It takes advantage of the motion prediction loop employed in most motion compensation based codecs. By correcting errors in a reference frame caused by earlier packet loss it prevents error propagation. Feamster and Balakrishnan [12] analyzed this approach with SR-RTP [13]. This RTP extension provides semantics for requesting the retransmission of independently-processible portions of the bitstream and a means for reassembling fragmented portions of independently processible units.

Related works base their retransmission algorithm either on the packet importance or on the delay introduced by the retransmission. As their results show the quality of the video stream can be improved with the selective retransmission method. These proposals are effective in networks with high bandwidth where no congestion occurs. For all that it makes no sense to retransmit a lost packet if it will be lost again or it will cause the loss of other packets. In the next section we will introduce our congestion controlled selective retransmission method.

3. Proposed Retransmission Scheme

We propose a selective retransmission scheme which disable or enable the retransmission of lost packets according to the current state of the network. When the network is in congested state or near to this state the retransmissions should be disabled. When the buffers of the network routers are overloaded the additional load will make the things worst. A retransmitted packet will be dropped at the routers or it will cause the loss of other packets. The proposed method uses the TFRC congestion avoidance algorithm to decide whether the lost packet should be retransmitted. The generally used UDP does not use congestion control; hence our TFRC-based algorithm uses DCCP instead of UDP. DCCP transport protocol was designed for multimedia applications with integrated congestion control mechanisms. These mechanisms needs information about the lost packets therefore the DCCP packet header includes sequence number field. This makes it possible to identify the lost packets and manage the retransmission queue.

TFRC is an equation based congestion control algorithm and it can only estimate the free bandwidth in the network. The algorithm proposes a sending rate in function of the network parameters like RTT (round-trip-time), packet loss ratio.

$$T = \frac{s}{R\sqrt{\frac{2p}{3}} + 4R(3\sqrt{\frac{3p}{8}} \cdot p \cdot (1 + 32p^2))} \tag{1}$$

The equation uses weighted average of the parameters, to avoid the radical decrease or increase of the sending rate. Hereby the effect of a single packet loss is negligible. Single packet losses usually occur due to wireless channel failure without need to reduce the sending rate.

The bitrate of a MPEG source may vary but we suppose that it is usually below the calculated TFRC sending rate when the network is not overloaded. When it is overloaded the estimated packet error ratio and the RTT (which may fluctuate during congestion) will be higher causing the decrease of TFRC sending rate. The video bitrate and the TFRC sending rate is independent so in the case of network congestion the determined TFRC sending rate should be lower then the multimedia stream rate. Our proposal is to disable the retransmission when the TFRC rate is under the video bitrate.

$$X_{MPEG}(t) > \frac{s}{R\sqrt{\frac{2p}{3}} + 4R(3\sqrt{\frac{3p}{8}} \cdot p \cdot (1+32p^2))} \tag{2}$$

To save on bandwidth, such video streams are often compressed, which leads to highly bursty, variable bit-rate (VBR) output streams. Transmitting a VBR stream over packet switched networks is difficult without packet loss due to congestion, or without wasting substantial bandwidth with a peak rate reservation.

Our mechanism proves to be effective when the TFRC sending rate varies near the video bitrate or the video bitrate is high enough to reach the TFRC rate. In other situations the TFRC-based selective retransmission method is applicable too, but of course it will enable the retransmission for the whole duration of the video transmission when the TFRC rate is much higher then the video bitrate and disable when it is lower.

Investigations were made to regulate video quality to adjust video rate to the desired sending rate which is determined by TCP-Friendly Rate Control algorithm [14]. Although it is recommended that the TFRC system regulates sending rate more than once in RTT, it is unrealistic to control video quality so frequently, in some cases, at the rate higher than video frame rate. With adaptive coding mechanisms difference between the TFRC rate and the video rate is minimal but not zero. In these solutions the retransmission determination function varies its output frequently.

$$A(t) = \operatorname{sgn}(X_{MPEG}(t) - X_{TFRC}(t)) \tag{3}$$

The retransmission should be enabled according to variable $A(t) \in \{-1, 1\}$ When it is -1 the retransmission is enabled and if it is 1 the retransmission is disabled.

The TFRC-based retransmission method does not need any additional traffic load to manage its functionality. All the needed information is provided by the DCCP protocol and the integrated congestion control algorithms. The transmitted packets should be stored in buffer on the server side to later retransmit the lost ones if possible.

We propose to use a LIFO (last-in first-out) queue with a queue management algorithm to increase the probability of reception. The duration of congestion should be long therefore the buffer should be filled with old packets that are not requested more by the client. If a packet loss occurs immediately after the congestion period the retransmission of old and useless packets should take so long time that the latest packet stored in the buffer will became old too. The LIFO buffer is more effective in this situation then FIFO (first-in firs-out). It is true that LIFO queue will cause the increase of probability of drop of older packets from the buffer due to the queue management, but when the network is not congested the buffer level is low and the probability of drop is negligible.

To avoid the situation when the buffer is full of old packets the queue management process deletes the older packets from the queue. To decide whether a packet is old or not the current playout buffer level is needed. This parameter determines the time remained till the final process that should be compared with the time needed for transmission, retransmission request, retransmission and management processes. The estimated time for these transmissions is three times the actual one-way delay of the link in the moment of first transmission. The extra delay (t_e) is introduced by additional management processes.

$$\frac{S_{PlayoutBuffer}\left[byte\right]\cdot 8}{X_{MPEG}\left[bps\right]} > \frac{3RTT\left[s\right]}{2} + t_e\left[s\right] \tag{4}$$

If the given inequality is true for a packet it should be dropped. The retransmitted packets should be lost too, but in real time multimedia transmission the delay of two or more retransmissions of the same packet is usually not acceptable. In our method we do not use multiple retransmissions therefore the retransmitted packets are deleted from the queue.

Late retransmissions in real time application are undesirable because the receiver side process already skipped the lost packets. The unrequired retransmissions waste network bandwidth and CPU cycles, contribute to congestion and may delay new data. The time available for recovery may be increased with no perceptible deterioration in quality to the user, by introducing limited buffering at the receiver. This is called playout buffering and the buffering delay is called playout or control delay. The determination of the playout buffer length is out of scope of this paper.

4. Simulation Results

In order to test the performance of the TFRC-based selective retransmission scheme, described in the previous section in, we analyzed some scenarios with ns-2. The simplex test network is illustrated on the following figure.

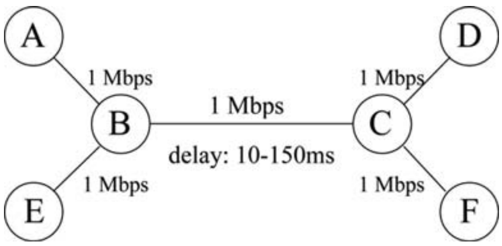

Fig. 1. Network topology

The analyzed video stream is transmitted from node A to node D in DCCP/IP packets, while the background traffic is generated by node E and received by node F. This background traffic uses UDP with variable bitrate. The TFRC-based selective retransmission method is implemented in node A. The bandwidth of the links is 1 Mbps that is high enough for all test scenarios. Node B uses a DropTail (FIFO) queue with length of 10 which should be overloaded in case of congestion.

We used a constant bitrate video stream in the simulations but of course our method is applicable for variable bitrate streams too. In the case of variable bitrate streams the current coding rate must be known that makes the selective retransmission scheme more complex. In the simulations the video bitrate was always lower than bandwidth of the links therefore congestion was caused only due to the background traffic.

In the first scenario the background traffic is off and the links are reliable therefore no loss occurs due to channel unreliability. The calculated sending rate by TFRC and the actual sending rate of the MPEG stream are illustrated on the next figure.

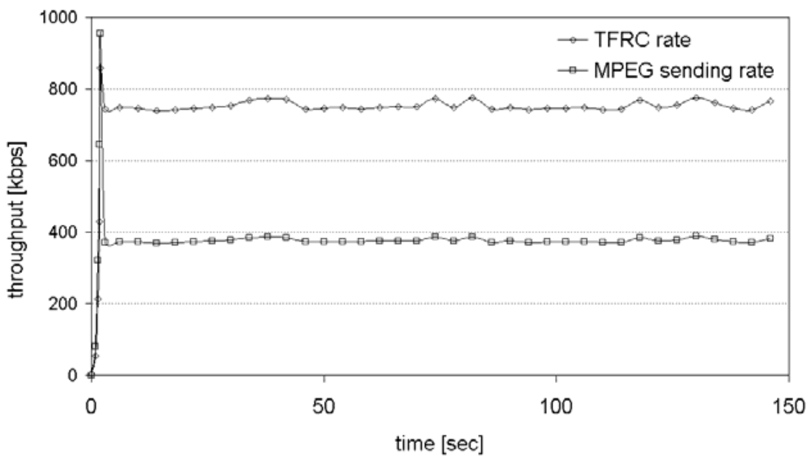

Fig. 2. Scenario 1.: Background traffic off without packet loss

The difference between the calculated TFRC rate and the video stream rate is high; hence according to our method the retransmission was enabled for the whole duration of the simulation.

The TFRC sending rate is significantly influenced by the packet loss ratio. The reason of the packet loss is not differentiated by the source therefore the loss due to congestion and channel unreliability has the same effect on the loss ratio parameter used in the TFRC equation. The only difference is in the RTT variation so it should be taken into consideration what is already done by the TFRC algorithm. It uses weighted average of loss ratios where a single loss has no significant effect on this estimation. The weighted average function smooths the variation of loss ratio so our algorithm should not care on the RTT in addition.

In the next scenario we analyzed the TFRC-based retransmission scheme with different packet loss ratios on the A-B link to find the limits of the method. The background traffic is still off but the TFRC algorithm radically reduces its suggested sending rate.

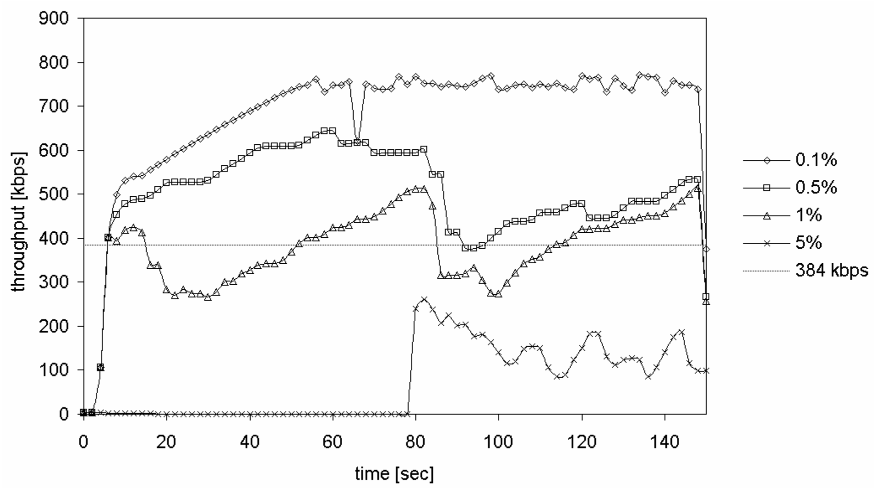

Fig. 3. Scenario 2.: TFRC sending rate in function of packet loss

The results show that the retransmission is enabled all the time when the packet loss is lower then 0.5%, the video stream rate is 386 kbps and the measured RTT is about 140 ms. The selective retransmission scheme will probably deliver all the packets in this case. Only those packets will be missing that are lost again during the retransmission. On extremely bad channels where the packet loss is 5% the connection should not be built up for a long time because the DCCP-Request and DCCP-Response packets were lost too. In the case of 1% packet loss ratio only a 270 kbps stream should be transmitted without disabling the retransmission. The retransmitted packets significantly increase the MPEG video quality especially when I-frame data has been delivered correctly to client. We made our examinations for MPEG-2 video streams but of course obtained improvement of quality is true for other audio and video stream formats.

As we mentioned before the TFRC-based selective retransmission is efficient when the TFRC sending rate varies near the video bitrate or the video bitrate is high enough to reach the TFRC rate. In the second scenario when the packet loss probability is 1% two periods are determined when the retransmission is disabled. In spite of these periods the video quality is improved. Figure 4 shows the evolution of video quality due to retransmissions in enabled periods.

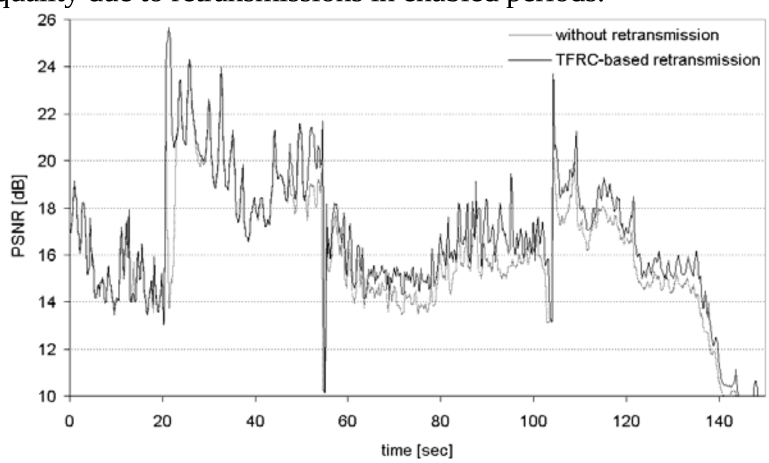

Fig. 4. MPEG video quality improvement

The average Peak Signal to Noise Ratio (PSNR) of the stream without retransmission is 15.86 dB. With the selective retransmission method it is 16.6 dB. Peak Signal to Noise Ratio is a coarse and controversial indicator of picture quality that is derived from the root mean squared error (RMSE). It compares the frames with the same frame number. Due to packet losses the frame numbers should be shifted therefore not the originally same frames are compared. Sometimes the PSNR value is almost the same but the visual difference is significant.

In these scenarios the RTT was roughly constant (about 140 ms) and the TFRC sending rate variation was due to packet loss occurrences. In the following tests the RTT will vary according to the level of congestion. In the test network the level of congestion is equal with the buffer level of node B. To analyze the TFRC-based selective retransmission method in congested network the background traffic is set on. In this scenario the total bandwidth demand of the background traffic and the video stream is higher then the available link capacity in short periods. The available free capacity of the B-C link is shown in the next figure.

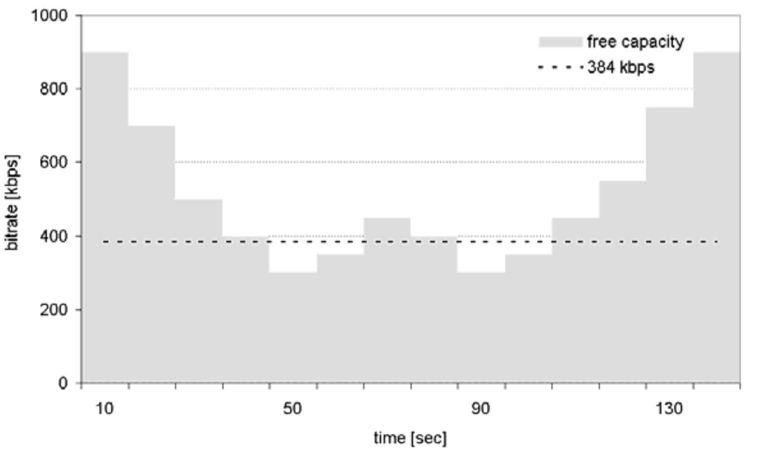

Fig. 5. Available bandwidth on B-C link (Scenario 3)

The packet drop probability of the A-B link is 0.1% but the large number of packet drops is due to the overflow of the buffer of node B. In 150 seconds about 4800 packets were transmitted from which 5 was corrupted due to channel corruption and about 50 due to congestion. Due to the large number of packet drops and the increase of RTT, the TFRC varies the sending rate to find the highest sending rate. The video rate is 384 kbps in this scenario therefore this is the actual sending rate of the source although the TFRC specifies higher. Nevertheless the actual sending rate will be the TFRC rate when it is higher then the video rate. The periods when the retransmission is enabled according to our scheme is illustrated in Figure 6.

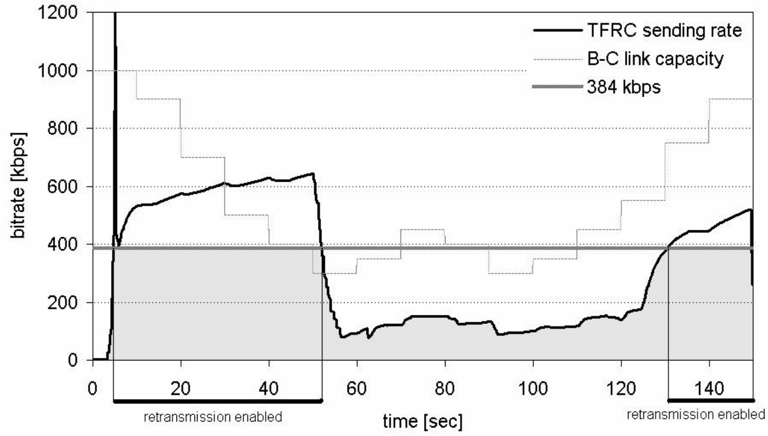

Fig. 6. TFRC rate in Scenario 3

The TFRC reduces the offered sending rate immediately below the video rate when congestion occurs. From this moment the retransmission is disabled. The TFRC will increase the offered rate after the buffer of node B is getting empty and the measured RTT is decreasing. It takes time to pour out the packets therefore the TFRC rate increase is restrained.

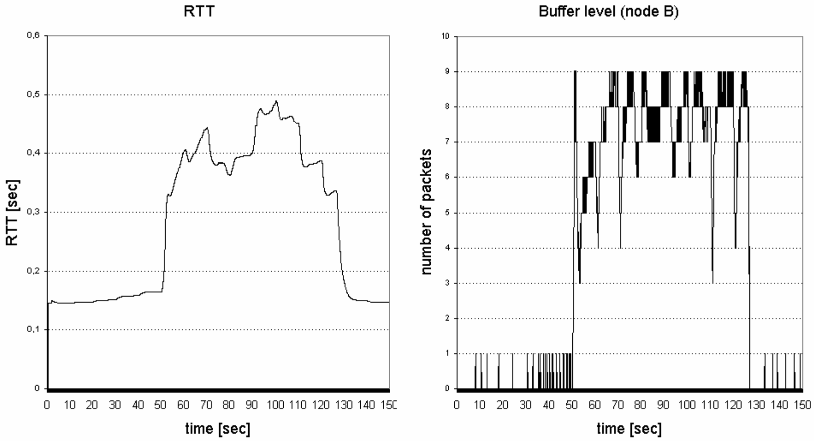

Fig. 7. RTT and buffer level of node B in Scenario 3

In real time applications retransmission is not recommended when the RTT is high. Our TFRC-based selective retransmission method indirectly takes the RTT into consideration. The Figure 7 shows that in the 50-130sec period the RTT values are the extremely high. As Figure 6 illustrates our method disables the retransmissions in the same time period. In congestion periods the probability of packet delivery is lower and the introduced delay is much higher so the retransmitted packets should not arrive to the client in time.

In Scenario3 the network was congested in 50% of the time. In this scenario the video quality improvement is significant in the first 50 seconds while in the congested period there is no difference between the two cases (without retransmission and TFRC-based retransmission). In the last period of the simulation when the retransmission is enabled again only one packet was retransmitted that was lost due to channel unreliability. The PSNR analysis shows the significant difference in the first period.

From the 130. second all the packets are delivered correctly therefore there is no difference from the original stream. The measured average PSNR using the TFRC-based retransmission scheme is 36dB while without it 19.9dB.

5. Conclusion

In order for video streaming to succeed on the Internet, systems must account for the anomalies of packet loss and changes in bandwidth and delay that make the delivery of real-time video on the Internet challenging. In this paper, we have proposed a new selective retransmission scheme for multimedia transmission over noisy wireless channel using TFRC-based decision mechanism.

We have analyzed the effects of packet loss on the quality of MPEG video and proposed a model to improve the quality of service. We have shown that, by recovery of the data in the bitstream considering the current state of the network, significant performance gains can be achieved without much additional penalty in terms of latency.

The evaluations were done on MPEG streams, but the selective retransmission algorithm is capable for other data type transmissions where high latency is not acceptable and the loss of few packets is tolerable.

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