ADAPTIVE RETRANSMISSION SCHEME FOR VIDEO STREAMING APPLICATIONS

Árpád Huszák

Budapest University of Technology and Economics

Department of Telecommunications

Budapest, Hungary

huszak@hit.bme.hu

Sándor Imre

Budapest University of Technology and Economics

Department of Telecommunications

Budapest, Hungary

imre@hit.bme.hu

Contact Author: Árpád Huszák

Email: huszak@hit.bme.hu

Abstract

In this paper we present a novel selective retransmission scheme, based on congestion control algorithm. Our method efficient in narrowband networks for multimedia applications which demand higher bandwidth. 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 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 actual video bit rate and the TCP-Friendly Rate Control (TFRC), which is integrated to the preferred DCCP transport protocol.

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INTRODUCTION

The new applications for delivering continuous media is set to become more and more popular, driven by user demand and network technology advances providing quality of service (QoS) and increased bandwidth to the user terminal. 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 mobile networks and new wireless technologies like WiMAX, HSDPA, HSUPA, etc. 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. This feature makes it possible to transmit the coded video in hazardous channels, without retransmitting the corrupted or lost packets. Traditional error control mechanisms generally use retransmission to provide reliability at the expense of latency. Loss tolerant multimedia applications should use retransmissions as well, but the retransmission will be successful only if the retransmitted packet arrives at the receiver before the playback.

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 [RFC 2581] throttles its transmission rate against the network congestion whereas UDP [RFC 768] does not have such control mechanisms. Some of the nowadays investigated transport protocols (e.g. UDPLite [RFC 3828], SCTP [RFC 2960], DCCP [RFC 4340], 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. To efficiently control the retransmissions a selective retransmission scheme is needed.

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) [RFC 3448] 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.

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. A brief overview of the preferred DCCP transport layer protocol is presented in Section III. In Section IV we propose a congestion sensitive retransmission method for multimedia applications. The obtained results are discussed in Section V. Finally, we summarize our paper and outline our future work in the last section.

VIDEO CODING STANDARDS

Video sequences are compressed in a format such as MPEG to achieve bandwidth efficiency. The MPEG standards are based on coding technologies developed for video conferencing and other multimedia applications. During the past ten years, both video and audio coding technology and industrial application background had changed significantly. At present there are four optional video and audio industry coding standard: MPEG-2, MPEG-4, MPEG-4

AVC (for short AVC, also known as JVT, H.264) and AVS. The first three standards are complete by the MPEG experts group, the fourth is China independent formulation. According to the main technical specification, the main coding efficiency of MPEG-4 is 1.4 times of MPEG-2, AVS and AVC are similar and more than twice of MPEG-2.

The structure of the MPEG coded videos is based on the GOP (Group of Pictures). The GOP contains a small number of frames coded so that they can be decoded completely as a unit, without reference to frames outside of the group. There are three types of frame:

Intra coded frames (I) - coded as single frames as in JPEG pictures, without reference to any other frames
Predictive coded frames (P) - coded as the difference from a motion compensated prediction frame, generated from an earlier I or P frame in the GOP.
Bi-directional coded frames (B) - coded as the difference from a bi-directionally interpolated frame, generated from earlier and later I or P frames in the sequence (with motion compensation)

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. This is the motivation to protect the important frames like I pictures to avoid the propagation of errors. In our retransmission scheme the bitrate of video stream plays very important role which highly depends from the GOP structure, the frame rate, coding resolution, etc.

RELATED WORKS IN SELECTIVE RETRANSMISSION

Retransmissions seem to be simple technique to improve the quality with recovering the missing parts of the data stream. It is not necessary to receive all the data packets to enjoy the video movie. If we do not want to retransmit all the lost packets, we should consider which packets should be retransmitted to efficiently improve the video quality. The decision algorithm of the retransmissions should be designed based on different aspects. The decision algorithm can be optimized using the actual network parameters like the retransmission and playout delay, the packet content or the network load.

Attempts were made to implement a selective retransmission protocol with a decision algorithm (Piecuch, et al, 2000). 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 publications appeared like (Mulabegovic, et al, 2002), which introduces an application layer protocol that sits atop UDP. The protocol offers semi-reliable transport. Instead of trying to guarantee 100% data delivery the protocol simply recovers as many packets as possible within a specified deadline. In addition the protocol incorporates features such as probabilistic redundant NAK transmission and flow control through selective frame dropping. The disadvantage of this protocol is that extra messages are needed for the retransmission requests.

In (Huszák and Imre, 2006) a content based retransmission scheme is introduced. This algorithm is also implemented in the application layer in order to protect the important MPEG frames. In this work DCCP was used instead of UDP. The advantage of DCCP is that no extra messages are needed because the protocol has its own acknowledgement system. Some other approaches describe methods that include categorizing groups of packets in order of importance (Rhee, 1998; Zheng and Atiquzzaman, 2002). 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 (M.Feamster and H.Balakrishnan, 2002) analyzed this approach with SR-RTP (Miyazaki, et al, 2001). 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 (Huszák and Imre, 2007) a source controlled and playout time oriented retransmission scheme for multimedia streaming is introduced. For the retransmission to be successful, retransmitted packet must arrive at the receiver in time for playback. To minimize the probability of wastefully retransmitted packets, a playout buffer is usually set up at the receiver side to prefetch a certain amount of data before playback. The key issue of this

scheme is the correct determination of the playout buffer that must be dependent on the network delay to make the retransmission possible.

TRANSPORT LAYER AND DCCP

The transport layer provides transparent transfer of data between end systems using the services of the network layer (e.g. IP) below to move data between the two communicating systems. This layer is responsible for delivering messages between networked hosts. As part of this, the transport layer protocols are also responsible for fragmentation and reassembly. In addition, some protocols also provide services to manage flow control and end-to-end error recovery. There is a long list of services that can be optionally provided by the transport layer (e.g. Connection-oriented communication, Same Order Delivery, Reliable Data, Flow Control, Congestion avoidance, etc.). None of them are compulsory, because not all applications want all the services available. Some can be wasted overhead, or even counterproductive in some cases.

From the retransmission point of view 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.

Datagram Congestion Control Protocol

The Datagram Congestion Control Protocol is a newly defined transport protocol by the IETF that implements bidirectional, unicast connections of congestion controlled, unreliable datagrams. DCCP was published as RFC 4340, a proposed standard, by the IETF in March, 2006. Although it is really new protocol the Linux kernel (2.6.14 and above for IPv4 and from 2.6.16 also for IPv6) already contains the implementation of DCCP.

Presently the reliable TCP and SCTP are the only alternative protocols to provide congestion control, but the retransmission mechanism is a strong disadvantage for multimedia services due to high end-to-end delay it causes. In delay-sensitive applications, such as streaming media and telephony, timeliness is also preferred. The main goal is to keep the delay and its variance on a minimum level, but on the other hand providing high service quality wit low bit error ratio. Other differences between DCCP and TCP are the different acknowledgement

formats and distinguished kinds of loss. A Data Dropped option declares that a packet was dropped because of corruption, because of receive buffer overflow, and so on. This facilitates research into more appropriate rate-control responses for these non-network congestion losses (although currently such losses will cause a congestion response).

For real-time applications the time constraints are more important than reliability, so media transmissions typically use transport protocols like UDP, where no retransmission occurs, providing minimal packet delay. UDP avoids long delays, but applications that use UDP as transport protocol, must implement congestion control on their own to prevent packet network flooding with a miss-behaved application and ensure fairness in bandwidth share. DCCP combines the best features of the two protocols within media transmission context, supporting congestion control mechanisms. It may be useful to think of DCCP as TCP minus bytestream semantics and reliability, or as UDP plus congestion control, handshakes, and acknowledgements. DCCP is a connection-oriented protocol with special packet types to establish, close and maintain connections. The main packet type for transmitting data is the DCCP-Data. In general use only the 12 or 16 byte length Generic Header is mandatory (Figure 1.).

Source port			Dest port
Data offset	CCVal	CsCov	Checksum
Res	Type	X	Sequence Number (low bits)

Figure 1. DCCP generic header

The size of the header depends on the length of the Sequence Number field. The *X* bit indicates whether 24 or 48 bit long sequence numbers are used. Unlike TCP sequence numbers, which are byte-based, DCCP sequence numbers increment by one per packet. In our selective retransmission algorithm we will utilize this feature. DCCP similarly to UDPLite is designed to provide a partial checksum that only covers as much of the user data that the sending application specifies in the DCCP Generic Header. Errors in the rest of the packet are ignored because they are assumed to be acceptable for the destination application. To avoid complexity, the protocol requires that the sensitive data in a packet start at the beginning. The *CsCov* field specifies how many bytes are sensitive to errors. This feature will be also utilized in our algorithm.

DCCP connections are congestion controlled, but unlike in TCP, DCCP applications have a choice of congestion control mechanism. DCCP uses Congestion Control Identifiers (CCID) to determine the congestion control mechanism. Currently two identifiers are being defined:

CCID2 that implements a TCP-like Congestion Control (Floyd, 2004) and CCID3 that implements a TCP-Friendly Rate Control (TFRC) (Floyd, 2004), but DCCP is easily extensible to further forms of unicast congestion control.

TCP-Friendly Rate Control (TFRC)

The TFRC congestion control algorithm plays a very important role in our selective retransmission scheme. TFRC is designed to be reasonably fair when competing for bandwidth with TCP flows. The penalty of having smoother throughput than TCP while competing fairly for bandwidth is that TFRC responds slower than TCP to changes in available bandwidth. It should only be used when the application has a requirement for smooth throughput, like in case of video streaming applications. We use this mechanism in our selective retransmission scheme to estimate the available bandwidth in order to avoid congestion.

TFRC congestion control consists in an equation-based rate control mechanism, designed to keep a relatively steady sending rate while still being responsive to congestion. A TFRC sender adjusts its rate as a function of the measured rate of loss events, where a loss event consists of one or more packets dropped within a single round-trip time. In order to compute a TCP-friendly sending rate, TFRC uses the following equation:

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

where the sending rate T, in bytes/sec, is modeled as a function of packet size s, round trip time R, steady-state loss event rate p. 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. TFRC computes p as a weighted average of the most recent n loss intervals (typically, $n = 8$), where a loss interval corresponds to the number of packets received between two consecutive loss events. The weights have been chosen so as to obtain a good trade-off between responsiveness and rate stability. The value of p is calculated by the receiver and fed back to the sender by means of feedback messages. The sender also uses the feedback messages to keep an estimate R of the round-trip time.

$$R = qR + (1 - q)R_{sample} \tag{2}$$

TFRC is not sensitive to the precise value for the filter constant q, but the default value is 0.9.

The TFRC protocol can not distinguish congestion loss and wireless loss; therefore the sending rate will be lower than the really achievable rate. Using alternative congestion control methods which effectively estimates the congestion loss, the performance of the congestion control scheme can be improved. New methods appeared like ARC (Akan and Akyildiz, 2004) and WLED integrated with ARC (WLED-ARC) (Singh, et al, 2006) to handle the wireless losses in the sending rate calculations.

THE 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. To decide whether to retransmit a lost packet congestion control protocol is used. These protocols help us to estimate the available bandwidth without causing congestion. In an overloaded network a retransmitted packet will be dropped again 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 uses acknowledgements and the packet header includes sequence number field. This makes it possible to identify the lost packets and manage the retransmission queue.

The bitrate of the video 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.

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

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 (Wakamiya, et al, 2001). 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.

$$A(t) = \text{sgn}\,(X_{MPEG}(t) - X_{TFRC}(t)) \qquad (4)$$

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.

$$\frac{S_{PlayoutBuffer}\left[byte\right]\cdot 8}{X_{MPEG}\left[bps\right]} > \frac{3RTT\left[s\right]}{2} + t_e\left[s\right] \tag{5}$$

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.

Our proposed retransmission method is implemented in the application layer using the information provided by the transport layer. The realization of the selective retransmission scheme is possible without modifying the DCCP protocol. We should use socket calls to get the necessary information (sending rate, packet sequence number, weighed packet loss ratio, weighed round-trip-time) for the client/server application.

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 network simulator. In most of the simulations a simple test network was used, illustrated on the following figure.

a) Topology1 (CBR/UDP, FTP/TCP)   **b)** Topology2 (WWW/TCP)
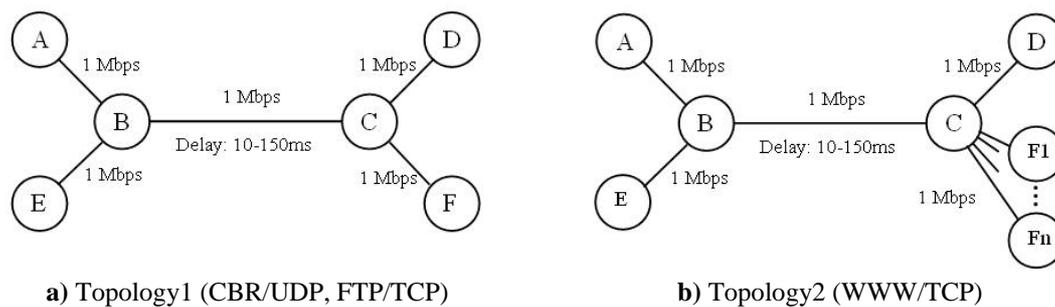
Figure 2. Network topologies

In all of the scenarios 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 ($F1…Fn$). In case of Topology1 this background traffic uses UDP with adjustable but fix bitrate or FTP over TCP, while in case of Topology2 the traffic between node E and $F1…Fn$ is WWW traffic. The TFRC-based selective retransmission method is implemented in node A. The bandwidth of the links is 1Mbps 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. The "wireless link" was the bottleneck link, in order to introduce wireless packet losses using a simple random drop model with the given loss probability. With these scenarios we are able to analyze the effects of a heavy loaded network and frequent wireless losses on the proposed selective retransmission algorithm. To analyze the quality of the MPEG-2 (384kbps) and H.264 (500kbps and 160kbps) video streams, the PSNR (Peak-Signal-to-Noise-Ratio) objective quality parameter was used. The most traditional ways of evaluating quality of digital video processing system are calculation of the Signal-to-Noise Ratio (SNR) and Peak Signal-to-Noise Ratio (PSNR) between the original video signal and signal passed through this system. PSNR is the most widely used objective video quality metric.

In the first scenario the DCCP using TFRC congestion control was analyzed. The background traffic is off and the links are reliable therefore no loss occurs due to channel unreliability. In this case the TFRC sending rate was about 750kbps in average, that mean that the DCCP

protocol with TFRC could utilize the network capacity in cc. 75%.The difference between the calculated TFRC rate and the MPEG-2 video stream rate (384kbps) was 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 smoothes the variation of loss ratio, therefore our algorithm should not care on the RTT in addition.

We analyzed the TFRC behavior with different packet loss ratios to find the limits of the method. The background traffic is still off but the TFRC algorithm radically reduces its suggested sending rate.
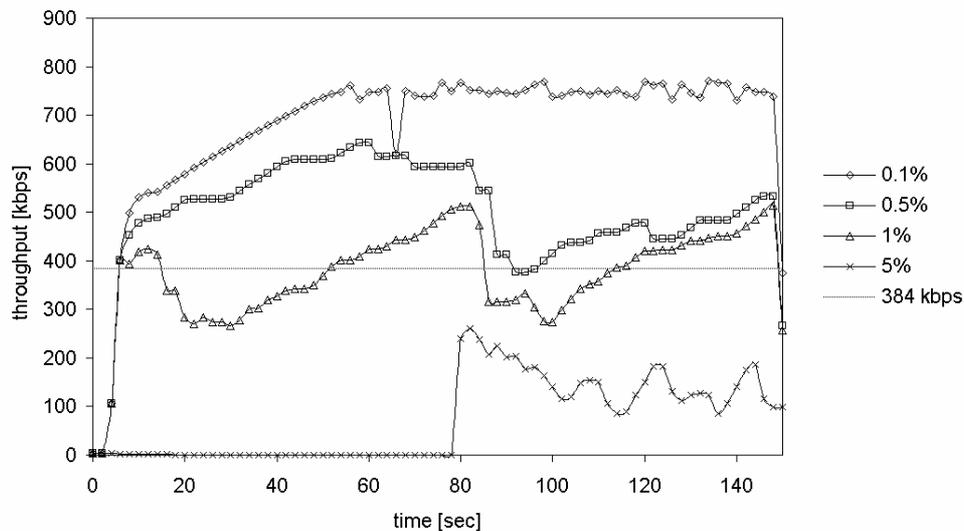
Figure 3. The variation of the TFRC sending rate during the 150sec long simulation period with different packet loss ratios

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 386kbps and the measured RTT is about 140ms. 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 270kbps 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.

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.
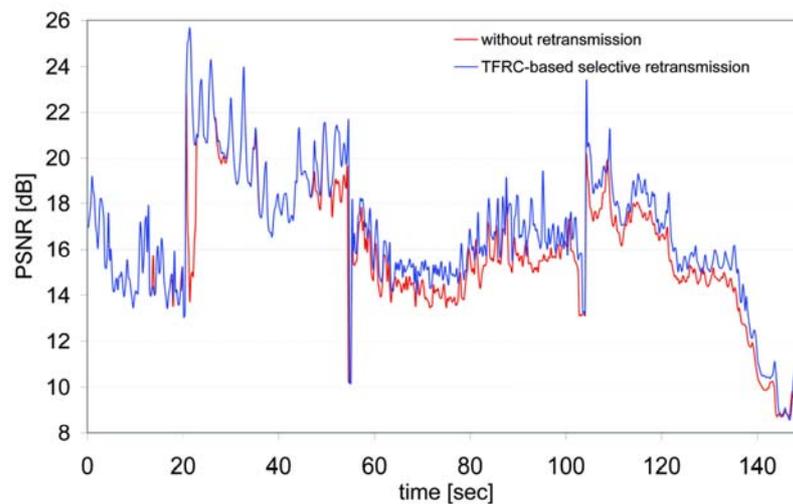
Figure 4. The MPEG video quality improvement, using the subjective PSNR quality measuring technique

The average Peak Signal to Noise Ratio (PSNR) of the stream without retransmission is 15.86dB. With the selective retransmission method it is 16.6dB. 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. We used CBR/UDP to generate the

background traffic. In order to analyze the reaction of the TFRC on changing channel conditions, we changed the bitrate of the background traffic every 10sec. 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 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 MPEG-2 video rate is 384kbps 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 5. The available free capacity of the *B-C* link is also visible in the next figure.
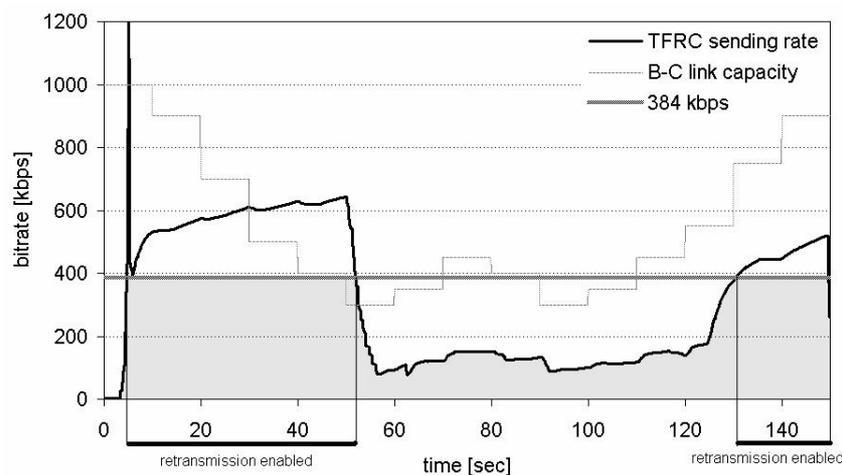
Figure 5. .TFRC rate and the retransmission periods

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.
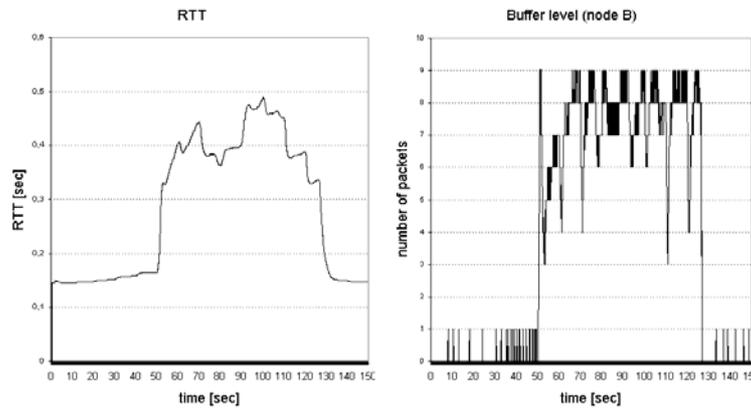
Figure 6. RTT and buffer level of node *B*

In this scenario the network was congested in 50% of the time. 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.

In the previous simulations the 384kbps MPEG-2 video stream was analyzed using Topology1 (see Figure 2.), while in the following investigations the H.264 video was examined in Topology2. In these simulations WWW background traffic was set up. The WWW flows are simulated with Pareto model, with average page sizes of 10kbyte and with average waiting time of 5 seconds. The buffer size of the nodes was set to 20 in the following simulations.

The increase of the number of WWW users or the packet loss probability will cause the decrease of the available bandwidth for the video stream. The TFRC algorithm can not distinguish between wireless and congestion loss, therefore any kind of loss will shrink the utilizable bandwidth. We have analyzed this effect varying the number of WWW users and the packet loss ratio on link *B-C*. The next figure shows the result of a simulation when the packet loss probability was set to a very low value (0.01%), while the load of the background traffic was changing according to the number of WWW users.
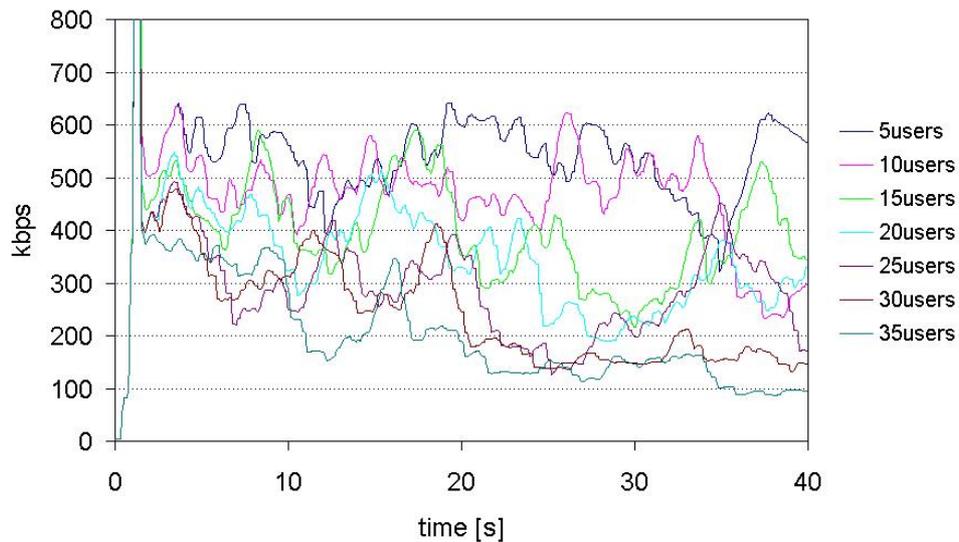
Figure 7. Available bandwidth (TFRC calculated) for the video stream

The efficiency of the proposed selective retransmission scheme depends on success of retransmitted packets. Of course if we can retransmit as more packets as possible, we will achieve better video quality. We have analyzed two H.264 videos with bitrates of 500kbps and 160kbps, which are the thresholds of the retransmissions decision. The retransmission ratios are calculated as the quotient of number of retransmitted and lost packets. The results are shown in the next figures.
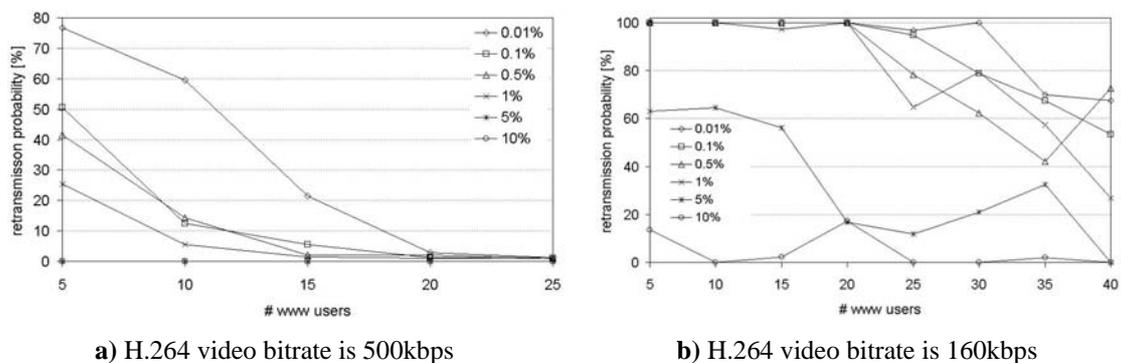
a) H.264 video bitrate is 500kbps   **b)** H.264 video bitrate is 160kbps

Figure 8. Retransmission probability

The network conditions were the same, the only difference was the video bitrate. As we can see in the figures in the case of the low bitrate video more packets was retransmitted. The calculated TFRC sending rate was the same in both cases, but when the TFRC bitrate decrease under 500kbps the retransmission were disable for the 500kbps video, while the retransmission were still enable for the 160kbps one.

The user should not see anything from the operation of the selective retransmission method, only the improvement of the video quality should be observed. That means that our proposed scheme is efficient if the video quality is higher. To measure the video quality the objective PSNR method was used to measure the differences between the original video stream and the transmitted one.

a) PSNR with selective retransmission   b) PSNR without selective retransmission
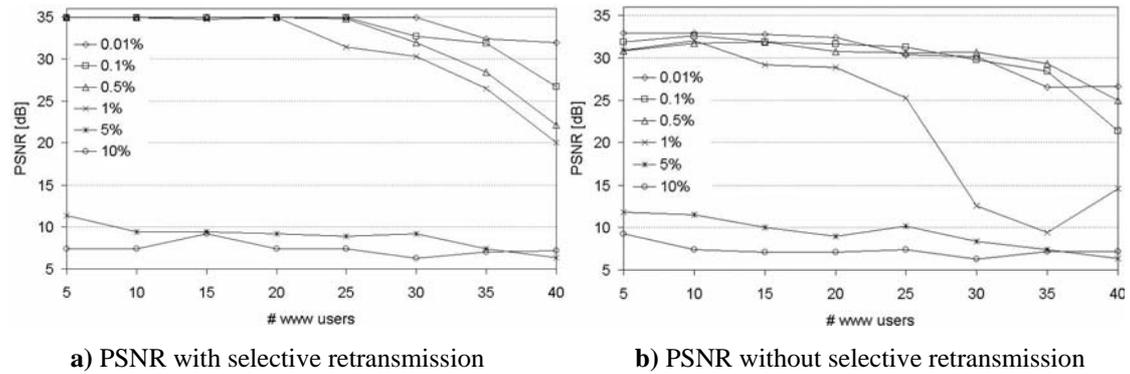
Figure 9. H.264 (160kbps) video quality measurements

a) PSNR with selective retransmission   b) PSNR without selective retransmission
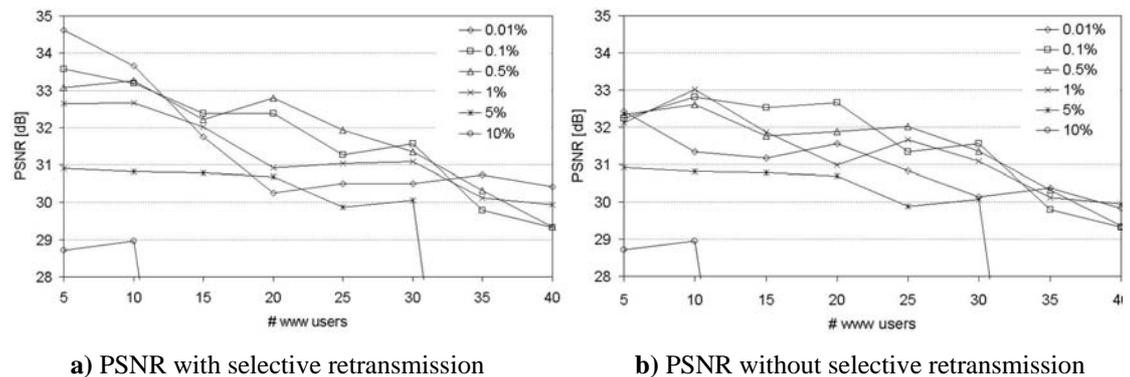
Figure 10. H.264 (500kbps) video quality measurements

The figures show that the measured PSNR quality is equal or higher in case of selective retransmission scheme. The effectiveness of our method highly depends on the bitrate of the video, as well on the network conditions. As Figure 8. and Figure 9. show all of the lost packets of the 160kbps video stream were retransmitted when the number of WWW users was between 0 and 20. In this case the calculated TFRC sending rate was higher then 160kbps. In same network conditions not all the lost packets of the 500kbps video stream was retransmitted (see Figure 8. and Figure 10.), therefore the PSNR values were different.

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.

The introduced retransmission scheme can easily improve the video quality using the DCCP transport protocol. This protocol provides all the needed information for the decision of retransmission. 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-2 and H.264 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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Key Terms and Definitions:

Video streaming: Multimedia that is constantly received by, and normally displayed to, the end-user while it is being delivered by the provider.

Transport protocols: Transport layer is the second highest layer in the TCP/IP reference models, where it responds to service requests from the application layer and issues service requests to the Internet layer.

Congestion control: Controlling traffic entry into the network to avoid congestive collapse by attempting to avoid oversubscription of link by taking resource reducing steps, such as reducing the rate of sending packets.

Video quality: Characteristic of a video passed through a video transmission/processing system, a formal or informal measure of perceived video degradation (typically, compared to the original video).

Quality of Service: It can provide different priority to different users or data flows, or guarantee a certain level of performance to a data flow in accordance with requests from the application program or the internet service provider policy.

Video coding: Reducing the quantity of data, but on the other hand retaining as much of the original's quality as possible. Compressed video can effectively reduce the bandwidth required to transmit digital video.