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Evaluation of Single Rate Multicast Congestion Control Schemes for MPEG-4 Video Transmission

Christos Bouras, Apostolos Gkamas and Georgios Kioumourtzis

***Abstract*—We present in this paper a simulation-based comparison of one of the best known multicast congestion control schemes - TFMCC - against our proposed Adaptive Smooth Multicast Protocol (ASMP). ASMP consists of a single-rate multicast congestion control, which takes advantage of the RTCP Sender (SR) and Receiver Reports (RR) in order to adjust the sender's transmission rate in respect of the network conditions. The innovation in ASMP lays in the "smooth" transmission rate, which is TCP-friendly and prevent oscillations. We use an integrated simulation environment named Multi-Evalvid-RA for the evaluation of the two congestion control schemes. Multi-Evalvid-RA provides all the necessary tools to perform simulation studies and assess the video quality by using both network related metrics along with video quality measurements. The performance evaluation results show that ASMP is a very efficient solution for rate adaptive multimedia applications and a serious competitor to well know TFMCC.**

***Index Terms*—Multimedia transmission, TCP Friendly, Congestion control, Multicast, Network Simulator (NS-2).**

I. Introduction

Video on Demand (VoD) and real time video applications have gained the interest of industry and the research community over the last few years. An efficient way to disseminate video files to a number of users in terms of bandwidth consumption is through multicasting.

Up to now there are promising approaches in literature, (TFMCC [1], PGMCC [2], RTP with TCP-friendly control [3], LDA+ [4], ERMCC [5]) which are mainly concentrated on the fairness of multimedia applications towards TCP traffic. Most of these proposals have been evaluated through simulations conducted with the ns-2 [6] network simulator software. Those simulations were not based on any multimedia traffic generation model and in the best case trace files were used instead. Therefore, the only quality indicators were purely based on "classic" network metrics (e.g. packet loss ratio, delay jitter etc). However, network metrics cannot characterize the quality of the resulting video transmission and may lead to debatable results because the perceived video quality by the end user is not measured. It is also very difficult to transform or correlate network metrics into QoS metrics of a video transmission.

The above limitations, as part of the simulation environment, undermine the performance evaluation studies in which quality measurements for multimedia data transmission (e.g. Peak Signal to Noise Ratio (PSNR), Mean Opinion Score (MOS), etc) are missing. Different multimedia encodings can result in different perceived video quality, although the transmission is done with exactly the same set of protocols and under the same network conditions. In addition, small variations to network metrics (e.g. packet loss ratio) may have an important effect on objective multimedia metrics (e.g. PSNR). Therefore, it is important to study the performance of any proposed solution by using real video files and associate simulation results with video QoS metrics.

To overcome the above described limitations the designers of Evalvid [7] provided the framework and the tools for video transmission over a real or simulated network. A later work named Evalvid-RA [8] extended the Evalvid by adding rate adaptive video transmission functionality. However, Evalvid-RA was also restricted to unicast transmission and therefore simulations and performance evaluation studies with multicast protocols were excluded. The above limitations have motivated the design and the implementation of Multi-Evalvid-RA, which is a tool-set for rate adaptive video transmission within the framework of ns-2 in the multicast domain. Two proposed rate adaptive multicast protocols are implemented: namely TFMCC and ASMP [9]. In addition, Multi-Evalvid-RA provides the basic API for the implementation of any rate adaptive scheme within or on top of RTP/RTCP protocols [9]. By doing so, any congestion control mechanism can be integrated in Multi-Evalvid-RA by making use of either the feedback mechanism of RTP/RTCP or any other preferable solution.

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The target of this paper is the competitive evaluation of TFMCC and ASMP, based on both objective multimedia metrics along with network metrics. This "joint" performance evaluation process provides the evaluation platform to better understand the benefits and limitations of our proposed protocol. The rest of this paper is organized as follows: In the next section we overview the functionality of TFMCC and ASMP. Section 3 provides a brief discussion on the latest video quality assessment methods. The simulation environment of Multi-Evalvid-RA is described in Section 4. In Section 5 we present the performance evaluation results. We conclude our paper in section 6.

I. Multicast Congestion Control Protocols

In this section we briefly discuss the functionality of TFMCC and ASMP.

A. *TCP-Friendly Multicast Congestion Control (TFMCC)*

TFMCC is a single-rate multicast congestion control protocol that extends TFRC from the unicast to multicast domain. The design goals of TFMCC are to provide a multicast congestion control that is TCP-friendly, highly responsive to network changes and also suitable for multimedia data transmission. One important attribute of TFMCC is the feedback suppression algorithm in which only the receiver with the lowest receiving capacity, which is termed the Current Limiting Receiver (CLR), sends frequent feedback reports. The sender adjusts its transmission rate based on CRL feedback reports. The rest of the receivers in the multicast group, send their feedback at longer time intervals in order to prevent feedback implosion phenomena at the sender side. TFMCC's TCP-friendly bandwidth share is measured by receivers with the use of the following TCP equation:

$$r^{i}{}_{tcp} = \frac{8s}{t_{RTT}\left(\sqrt{\frac{2p}{3}} + \left(12\sqrt{\frac{3p}{8}}\right) l(1+32p^2)\right)} \quad (1)$$

Where, r^{i}_{tcp} is the receiver's i estimation (in bytes/sec), s is the packet size in bytes, p is the packet loss ratio and t_{RTT} is the Round Trip Time (RTT) of the link between the sender and the receiver. The long-term TCP-friendliness is defined in TFMCC's specification to be no more than twice the sending rate of a TCP flow which is traversing the same link as the TFMCC flow. For the time being, TFMCC is under the review of the Internet community as an experimental RFC [1].

B. *Adaptive Smooth Multicast Protocol (ASMP)*

ASMP comprises a single-rate multicast congestion control, which takes advantage of RTCP Sender (SR) and Receiver Reports (RR). The innovation in ASMP is the calculation of smooth transmission rates, which is performed by receivers and is based on RTCP reports. The main objective in ASMP is to adjust the sender's transmission rate in such way that oscillations are reduced, following a smooth fashion. Another important attribute is the long term TCP-friendliness; meaning that the multimedia stream consumes no more bandwidth than a TCP connection which is traversing the same path with the multimedia stream. Moreover, with the use of RTCP feedback reports, ASMP provides better scalability, as the amount of these feedback reports is controlled by the RTCP protocol and they cannot exceed a specified threshold, expressed as percentage of the total available bandwidth. Bandwidth availability is increased for user data without disseminating any additional feedback reports (ACKs or NACKs) than those of RTCP sender and receiver reports. ASMP is implemented on top of RTP/RTCP protocol. This approach has several advantages as the complexity is moved up to the application layer, leaving the operating system and network elements untouched. The only visible drawback is related to high time intervals between two consecutive RTCP feedback reports in a large multicast group. As a result, the sender cannot react quickly when network conditions change very rapidly. However, ASMP increases its responsiveness with the use of network statistical information. This information is based on Congestion Indicators (CI) which provide the warnings of upcoming congestion so that the "smoothness factor" is tuned to a proper value. The smoothness factor regulates the behavior of the congestion control mechanism, making it less or more aggressive, according to the level of congestion in the bottleneck link. In this way, the congestion control in ASMP is "smooth", without suffering from high oscillations, and at the same time is very responsive to network changes. More details about ASMP can be found in [9].

II. Video Quality Assessment Methods

Video quality assessment methods can be classified into two categories in respect of the involvement of human interaction during the evaluation process. In the subjective QoS assessment scheme the perceived video quality is defined through human grading in which the individual viewer determines the quality level.

Objective QoS schemes do not involve human interaction and are classified into three categories. In the first category, we evaluate the transmitted video by comparing the complete decoded video at the end user to the original one sent by the sender. In the second category, we compare only features/metrics between the original and the decoded video and not the whole video sequence. In the third category we do not conduct any comparison between the original and the decoded video at the end user. The evaluation is realized only by assessing the decoded video at the end user side. The Video Quality Expert Group (VQEG) names these methods as the full, the reduced and the no reference methods [11].

Multi-Evalvid-RA implements the full reference method. Apart from network metrics that are related to video transmission quality, we extend the performance evaluation with additional video quality metrics. Two video assessment metrics named PSNR and MOS are measured and evaluated. PSNR is a derivative of SNR. The PSNR computes the maximum possible signal energy to the noise energy, which results in a higher correlation with the subjective quality perception than the conventional SNR. Equation 2 gives the definition of the PSNR of a source image s and destination image d [12].

$$PSNR(s,d) = 20\log\frac{V_{peak}}{MSE(s,d)} \text{ in dB}$$

where (2)

$V_{peak} = 2^k - 1,\ k$ bit color depth

$MSE(s,d) =$ mean square error of s and d

TABLE 1. ITU-R QUALITY AND IMPAIRED SCALE AND PSNR TO MOS MAPPING [7]

PSNR (dB)	MOS	Perceived Quality	Impairment
>37	5	Excellent	Imperceptible
31-37	4	Good	Perceptible, but not annoying
25-30	3	Fair	Slightly annoying
20-24	2	Poor	Annoying
<20	1	Bad	Very annoying

To evaluate the impact of the network on the quality of the received video, we need to compare the received video with the original one sent by the sender. Although, MOS is a subjective evaluation metric for the assessment of video quality, we can obtain a MOS value based on the corresponding PSNR value as shown in Table 1.

III. SIMULATION ENVIRONMENT

The simulation environment consists of three parts and is depicted in Figure 1. During the pre-processing a raw video file, which is usually stored in YUV format, is encoded with the desired video encoder[1] into 30 different encoded MPEG-4 video clips with quantizer scale values in the range 2 to 31 (Table 2). We use the *ffmpeg* [13] free video encoder for the creation of the video clips. An MPEG-4 [14] encoder such us *ffmpeg*, generates three types of frames. Frames are arranged in Group of Pictures (GOP). A GOP consists of exactly one Intra-frame (I-frame), some related Predictive frames (P-frames) and optionally some Bi-directional Frames (B-frames) between these I- and P-frames. The encoded video files are then traced to produce 30 frame-size trace files. We use the `mp4.exe` program of Evalvid to create the trace files. At the end of the pre-processing phase we have 30 `m4v` files with its associated frame size files.

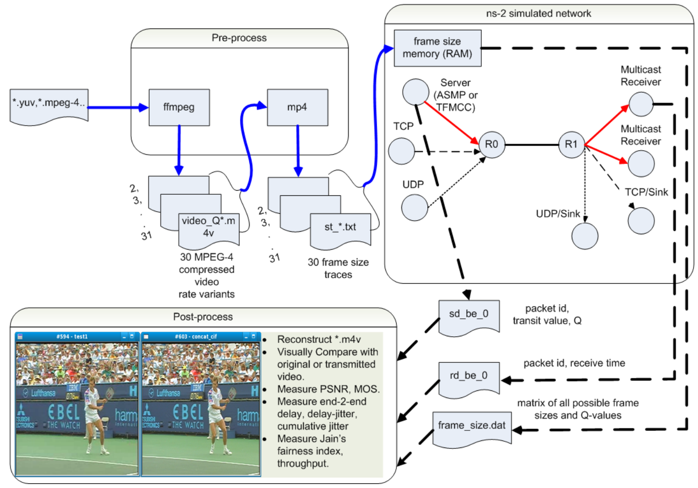

Fig. 1. Overview of the Multi-Evalvid-RA: pre-processing, ns2 simulation environment, post-processing

TABLE 2. ENCODINGS RATES OF MPEG-4 VIDEO

Q scale	Encoding rate (Kb/s)	Q scale	Encoding rate (Kb/s)	Q scale	Encoding rate (Kb/s)
2	2614.4	12	179.0	22	103.2
3	1420.5	13	166.0	23	85.1
4	929.1	14	139.7	24	81.6
5	651.1	15	135.4	25	92.7
6	452.9	16	129.9	26	76.6
7	362.0	17	113.6	27	73.8
8	294.0	18	107.6	28	86.1
9	234.8	19	102.6	29	69.5
10	219.0	20	96.9	30	67.4
11	197.4	21	92.8	31	65.5

The ns-2 creates the simulated network. The video file is transmitted from the server to the group of multicast receivers. The choice of the adaptive multicast transmission protocol is made by canceling out #define ASMP or #define TFMCC option in the `vbr_rateadapt.cc` file. Therefore, the desired congestion control scheme is responsible for the adaptation of the sender's transmission rate. The Variable Bit Rate (VBR) controller of Evalvid-RA considers a new quantizer scale for the next GOP based on the information provided by the congestion control mechanism. During the simulation time we store the traces for both the server (`sd_be_0`) and the receivers (`rd_be_0`) to enable easy calculation of network metrics and video performance evaluation metrics (PSNR, MOS). All the above files are used for the reconstruction of the original transmitted video and for post-processing measurements.

The Evaluation Trace Rate Adaptive (`et_ra.exe`) program of Evalvid-RA reconstructs the received video at the end user. The usage of these tools is described in the `instructions.txt` file, which is part of the Multi-Evalvid-RA package.

[1] Currently H.263 and MPEG-4 are supported by Evalvid.

IV. Simulations and Results

For our simulations we choose a high-motion video sequence (`highway`) with a duration of 73.4 seconds. This high-motion sequence is very challenging as it provides low PSNR values. All video clips are encoded at 25 frames/s with a fixed GoP size of 12 frames.

The frame size of all clips is 352 x 288 pixels, which is known as the Common Intermediate Format (CIF). Table 2 depicts the video encoding rate per quantizer scale.

We conduct simulations under different scenarios to investigate:

- The TCP-friendly behavior of the tested protocols.
- Their performance for video transmission when certain time related constraints are involved.
- Their behavior when sharing network resources with UDP, which dos not employ any congestion control mechanism.

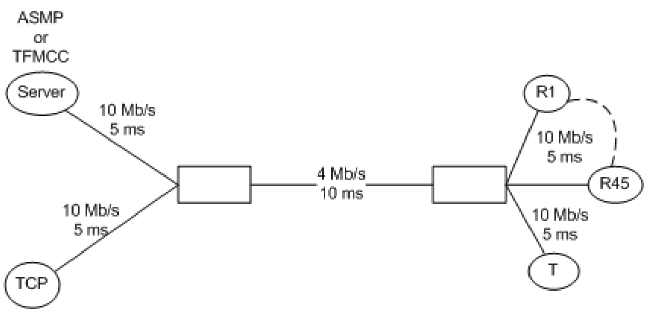

Fig. 2. Dumbbell bottleneck scenario

A. Simulation Topology

As both TFMCC and ASMP aim to have a TCP-friendly behavior we will start with a dumbbell simulation scenario in which one video server shares the same bottleneck link with one TCP Agent (Figure 2). We use a multicast group that consists of 45 receivers (receivers R1 through R45). T represents the TCP receiver. We use in our simulations Drop Tail queue in the routers and set the access link capacity of all agents to 10 Mb/s with an access delay of 5 ms. Thus the total one-way propagation delay is 20 ms. The MTU is set to 1018 bytes for both ASMP and TFMCC. These numbers resulted from 978 bytes payload, 12 bytes for the RTP header (we use the same size for TFMCC header), 8 bytes for the UDP header and 20 bytes for the IP header.

According to proportional fairness the bottleneck link should be equally shared by the video and TCP flows, which means that multicast receivers must not consume more than one half of the bottleneck bandwidth. Therefore, multicast receivers must not consume more than 2 Mb/s. The rest of the available bandwidth must be shared by the TCP flow and it is expected that the TCP flow will present at least 2 Mb/s throughput.

B. TCP Friendliness - Jain's Fairness Index Measurements

Throughput is measured during the simulation time. We obtain the Jain's Fairness Index by using the `jain.awk` program, which is part of the simulation tool-set. Figures 3 and 4 present the results of ASMP and TFMCC simulations. We present only the results from one multicast receiver in both cases (ASMP and TFMCC) for easier observation.

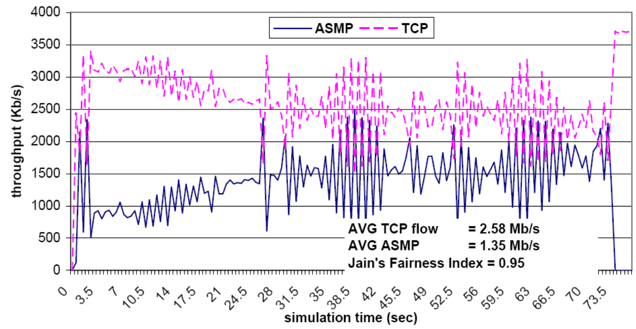

Fig. 3. ASMP vs TCP traffic

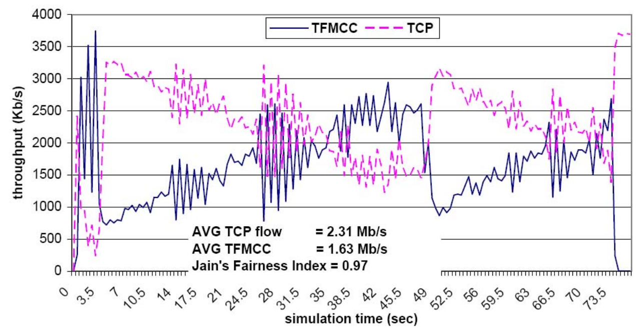

Fig. 4. TFMCC vs TCP traffic

We observe that both ASMP and TFMCC have a mirrored relationship with the TCP traffic. TFMCC has higher performance in terms of the achievable throughput than ASMP at the expense of higher frame loss rate. ASMP has zero losses whereas TFMCC presents a frame loss rate of 2.95% (59 lost frames out of 2000 total transmitted). This frame loss ratio has a negative impact on the video quality as seen by the end user and we will observe the negative impact on the PSNR values later in this paper. The Jain's fairness Index is higher for TFMCC, which indicates that the available bandwidth is better distributed between the multicast and the TCP traffic when TFMCC is involved than with ASMP. On the other hand, ASMP is more TCP-friendly than it ought to be and this is a direct result of its smoothing functions that reduce responsiveness to rapid network changes. A final observation from these results is that both protocols seem to be TCP-friendly as they almost equally share the bandwidth of the bottleneck link with TCP traffic.

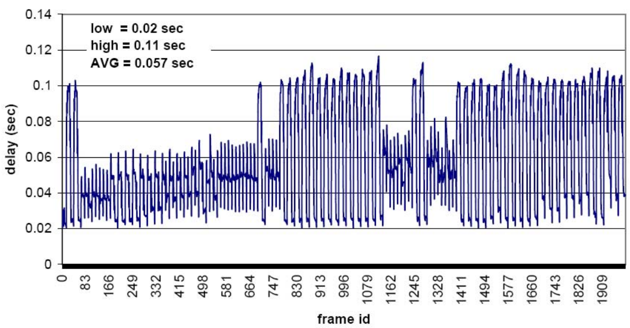

Fig. 5. ASMP end-to-end delay

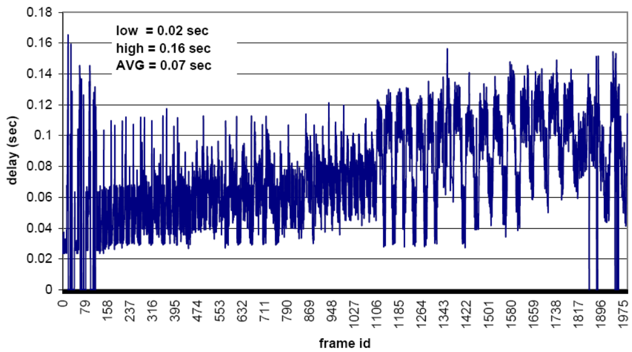

Fig. 6. TFMCC end-to-end delay

C. End-to-end Delay Measurements

We observe in Figures 5 and 6 that TFMCC presents higher end-to-end delay than ASMP. TFMCC in some cases presents end-to-end delay values higher than 150 ms, which is the delay limit of VoIP and other conversational media applications. This is a direct result of the high queue delay in the routers. However, in general terms both protocols have very good performance.

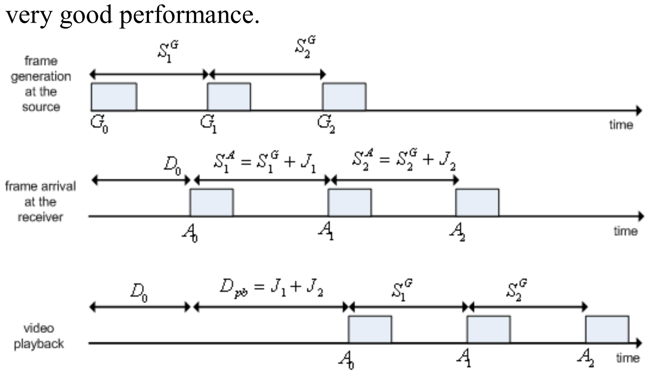

Fig. 7. Media synchronization timing diagram

D. Effects of Cumulative Jitter

The cumulative jitter is defined as the amount of playback delay that must be provided in order to avoid discarding delayed video frames at the client side [16]. Figure 7 shows the frame generation at the sender as well their arrival and playback at the receiver. For two consecutive frames i-1 and i the jitter delay variation is defined in [16] by Equation 3 below:

$$J_i = S_i^A - S_i^G = (A_i - A_{i-1}) - (G_i - G_{i-1}) \tag{3}$$

The cumulative inter-frame jitter or simply cumulative jitter is defined in [16] as "the amount of playback delay D_{pb} that must be provided to avoid discarding any frame in the sequence":

$$CJ_k = \sum_{i=1}^{i} J_i \tag{4}$$

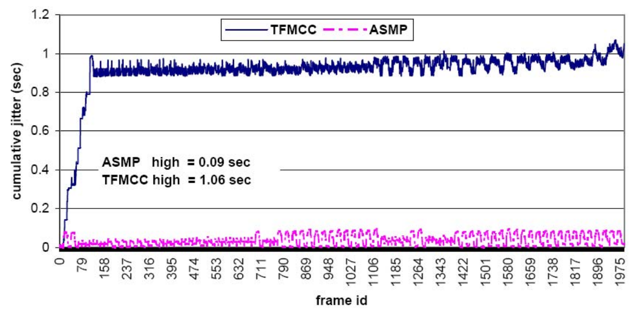

Fig. 8. Cumulative jitter variations

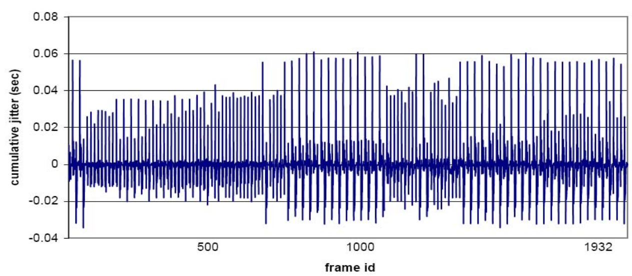

Fig. 9. ASMP inter-frame variations

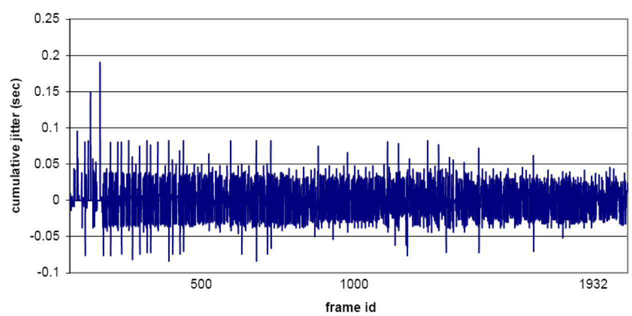

Fig. 10. TFMCC inter-frame variations

This metric is very important for assessing the performance of the underlying congestion control mechanism because it is highly correlated with the level of congestion in the bottleneck link(s). Once the cumulative jitter of a video frame exceeds the playback buffer duration, the video frame is useless and is discarded. A larger playback buffer allows late arriving frames to be recovered at the expense of longer start-up time and extra storage capacity at the end user. We observe in figure 8 that the cumulative jitter of TFMCC is very high when compared with that of ASMP. ASMP presents very high performance as the highest value of cumulative jitter is below 0.1second.

E. Video Objective Performance Metrics

Following our evaluation we measure the PSNR and MOS values. To obtain the PSNR values (Figure 11) we compare the encoded video at the sender side with the one "as seen" by the receivers. PSNR frame-by-frame values are matched to MOS values (Figure 12) to get the objective video evaluation results.

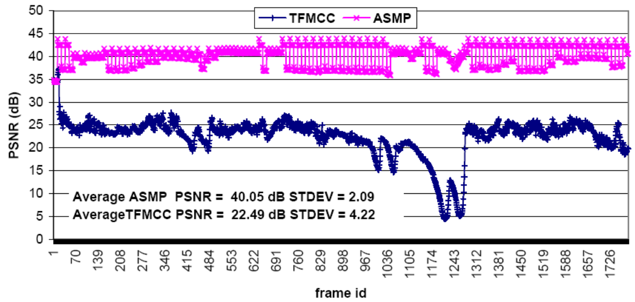

Fig. 11. Resulting PSNR values (frame by frame) of TFMCC and ASMP

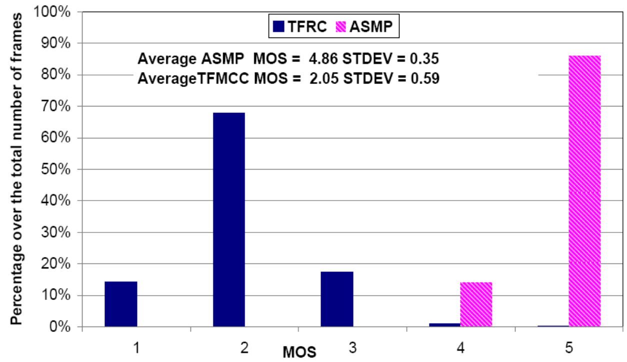

Fig. 12. Resulting MOS values of TFMCC and ASMP

We observe from the MOS values that the biggest part of the video sequence in ASMP is graded as "excellent" with the exception of some frames that are graded as "good". TFMCC's performance is graded from "bad" to "poor" for the biggest part of the video sequence with a small part (approximately 15%) that is graded as "fair". This is a result of the frame loss rate (2.95%) that causes PSNR degradation due to congestion in the bottleneck link

F. Simulations with End-to-End Delay Constraints

In this simulation we use the same dumbbell topology without the TCP background traffic. Our objective is to assess the video quality at the end user when absolute play-out buffer constraints are involved. Under this scenario video frames that are received with an end-to-end delay less than or equal to the absolute play-out buffer p_d will be rendered at the correct sequence position. Video frames received with end-to-end delay bigger than the p_d cause a previous frame to be displayed when that frame had to be displayed. The late arriving frame is not dropped, however, since it is the most updated frame of the received video. To better explain the consequences of the absolute play-out buffer time let us consider the example in Figure 13. Frame 2 is displayed twice due to increasing network delay. Frame 3 is never displayed. When it was received, Frame 4 had also been received, which was more up-to-date according to the planned schedule.

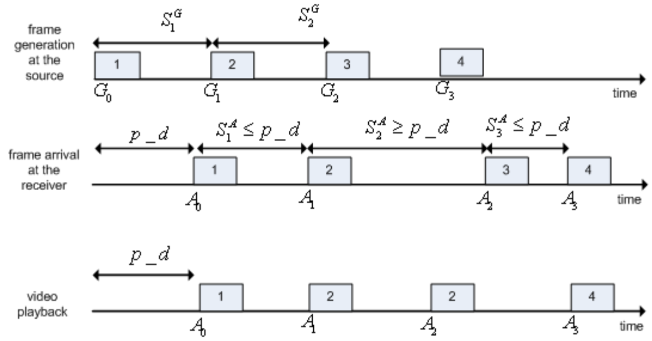

Fig. 13. Effects of absolute play-out time constraint

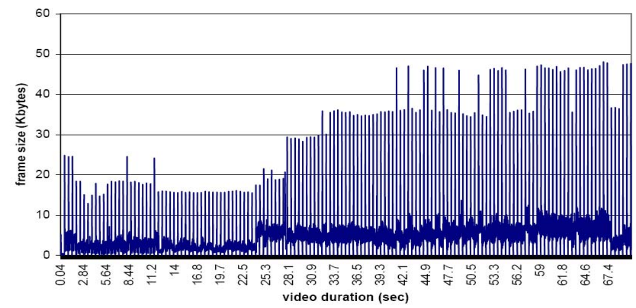

Fig. 14. Concat_cif.yuv frame sizes (Kbytes)

In this simulation we set the absolute play-out delay at 150 ms, which reflects the recommended one-way delay for conversational media. We create a raw video that consists of different video sequences that differ in complexity with the following order: *News* (frame 1-300) has medium complexity, *Akiyo* (300-600) has very low complexity as it contains more static information, *Stefan* (600-700) is very complex due to continuous moving pictures, and lastly *Paris* (700-1753) which has high to medium complexity. This sequence, named concat_cif.yuv, is encoded in MPEG4 format with temporal resolution of 25 frames per second, and GOP pattern IBPIBPIBP with a size of 25 frames per GoP. The encoded frame sizes are depicted in Figure 14. The peaks in the graph correspond to I-frames, which are larger than B and P-frames. Therefore, by combining various video sequences of different complexity we can better simulate video transmission as the temporal resolution is changed over time.

Figure 15 depicts the PSNR values without any play-out time constraint. ASMP presents very high performance with PSNR values above 30 dB, while TFMCC does not scale well with the changing complexity of the video file over time. We observe that with the *Akiyo* video sequence (frames 300-600), the PSNR values of TFMCC are above 30 dB. With the *Stefan* sequence (frames 600-700) PSNR values drop by approximately 15 dB. ASMP scales better with video sequences of high temporal resolution. We observe that the Stefan sequence causes a reduction of PSNR values by approximately 10 dB. However, ASMP clearly outperforms TFMCC in this simulation scenario.

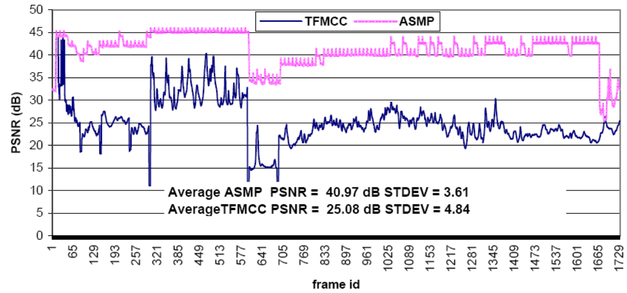

Fig. 15. Resulting PSNR values (frame by frame) without play-out time constraint

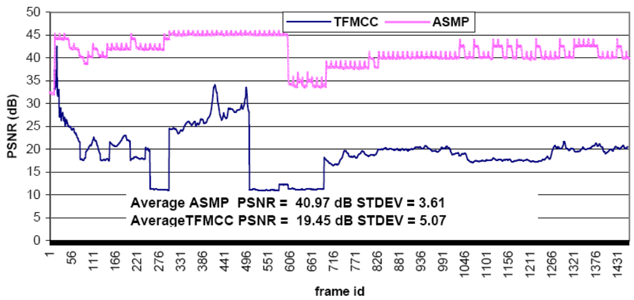

Fig. 16. Resulting PSNR values (frame by frame) with play-out time constraint (150 ms)

Next in figure 16 we observe that the play-out constraint of 150 ms affects only TFMCC's performance. PSNR values drop by approximately 5 dB in all video sequences. This is a direct result of high end-to-end delay values. TFMCC presents high one-way delay while ASMP's one-way delay is lower than 150 ms (Figure 17); at equilibrium is below 50ms. Therefore, ASMP's performance is not affected by the 150 ms play-out constraint.

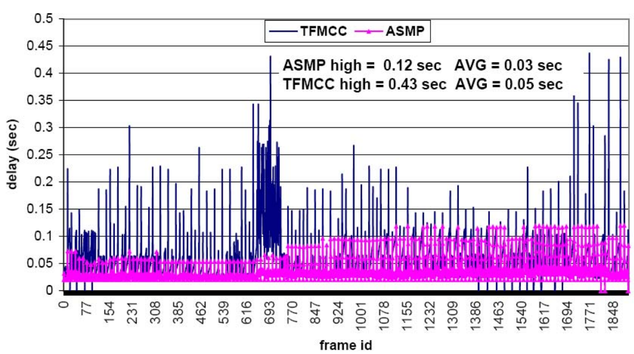

Fig. 17. TFMCC and ASMP end-to-end delay

G. Responsiveness to Dynamics of Competing UDP Traffic

In this simulation we investigate the ability of ASMP and TFMCC to react and adjust the sender's transmission rate when competing for network recourses with UDP traffic that does not employ any congestion control mechanism. We use the same dumbbell simulation scenario in Figure 2 and replace TCP background traffic with one UDP flow. The bottleneck link has 4 Mb/s capacity and 10 ms delay. To better test the responsiveness of the two protocols we vary the available bandwidth as a square wave by injecting UDP traffic throughout the simulation time. UDP traffic is transmitted by a Constant Bit Rate (CBR) source at 2 Mb/s.

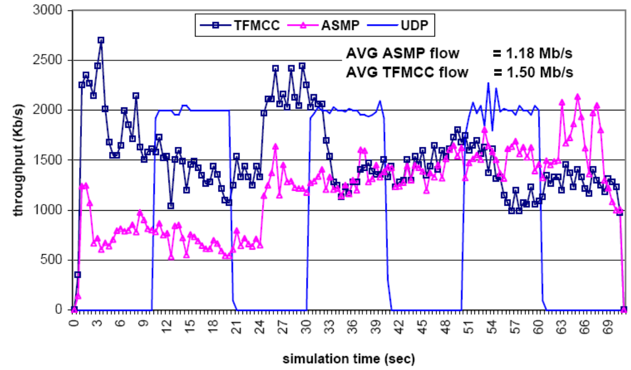

Fig. 18. Throughput of TFMCC and ASMP flows

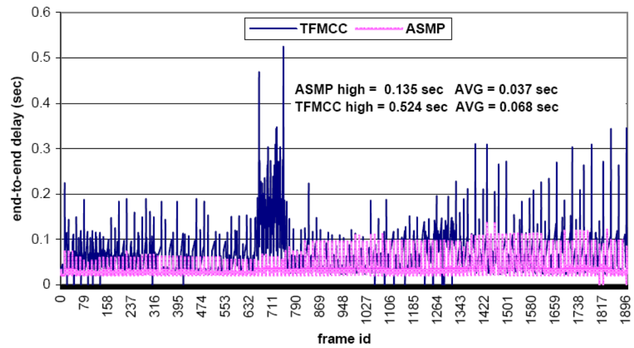

Fig. 19. End-to-end delay of TFMCC and ASMP flows

TFMCC presents higher throughput than ASMP and reacts faster to networks changes due to UDP traffic. TFMCC seems to be a more efficient congestion control mechanism when bandwidth utilization is of a major concern. However, TFMCC overshoots as in the slow start phase the sending rate is increased exponentially unless a packet loss event is observed. We can observe the increase of the transmission rate of TFMCC in the first simulation seconds (Figure 18) in which the transmission rate is above 2.5 Mb/s. At this stage the bottleneck link is already congested. As the VBR controller of Evalvid-RA picks a new quantizer scale every next GoP there is a time gap before it adapts to new network conditions. That means that even in the event of a congested link the VBR controller will continue to transmit the current GoP with an out-of date rate. The direct result is high end-to-end delay (Figure 19) and high frame loss rates (0.017% that is 30 lost frames out of 1753 total transmitted). ASMP is a conservative congestion control mechanism at the expense of lower bandwidth utilization. It does not employ any slow start mechanism and increases its transmission rate gradually by 1 packet per RTT. It takes longer for ASMP to reach its highest transmission rate than TFMCC but it ensures a smooth increase of the transmission rate in order to avoid packet drops. The frame loss rate in ASMP is very low (0.0017%) which results to higher PSNR values (Figure 20).

MOS values of the received video are presented in Figure 21. We observe that ASMP clearly outperforms TFMCC as approximately 85% of the received video is graded as "excellent".

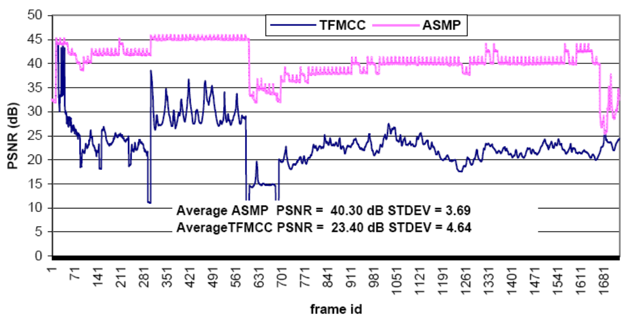

Fig. 20. Resulting PSNR values (frame by frame) of TFMCC and ASMP

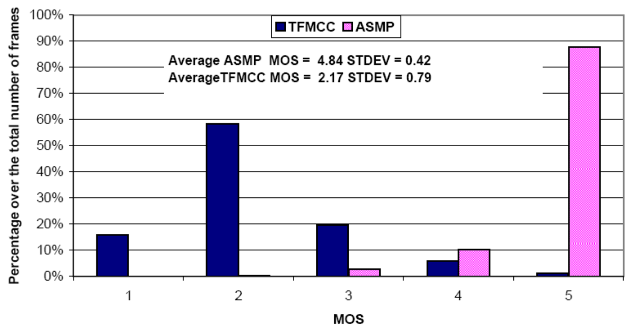

Fig. 21. Resulting MOS values of TFMCC and ASMP

Although, the above simulation scenario is very simple it does provide a better understanding on the correlation between high bandwidth utilization, frame losses and video QoS constraints.

V. Conclusions / Future Work

In this paper we compare the performance of ASMP against known related work we conducted comparative simulations with TFMCC. ASMP outperformed TFMCC is all simulation scenarios. The TCP-friendliness is differently defined in ASMP from that in TFMCC in which the sending rate is generally within a factor of two of the sending rate of a TCP flow that traverses the same link with that TFMCC flow. ASMP is a "moderate" protocol in which the sending rate consumes no more bandwidth than a TCP connection which is traversing the same path with ASMP. TFMCC proved to be a more efficient congestion control mechanism than ASMP when bandwidth utilization was of major concern. However, the high transmission rates of TFMCC tended to rapidly occupy the biggest portion of the available bandwidth in the bottleneck link, which created congestion and frequent frame losses. The direct result of those frame losses was PSNR degradation.

We still need to investigate ASMP's scalability with a larger number of receivers (e.g. thousands of receivers). Moreover, we will carry out a more in-depth investigation to the effect of "smoothening the transmission rate" on other competing traffic types and loss error schemes. It is also our intention to use our solution as part of a multi-rate transmission scheme. In addition, we will implement ASMP in real world experiments and investigate its behavior with real multimedia traffic. Finally the source code, simulation examples and documentation of Multi-Evalvid-RA is available in [17].

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