Performance of Adaptive Multimedia Transmission: The case of "One Multicast Stream" Techniques

Ch. Bouras[1,2] and A. Gkamas[1,2]

[1] Research Academic Computer Technology Institute, Riga Feraiou 61, GR-26221 Patras, Greece
[2] Computer Engineering and Informatics Dept., Univ. of Patras, GR-26500 Rion, Greece
bouras@cti.gr, gkamas@cti.gr

Abstract. A "one multicast stream" mechanism is a mechanism, which can be used for multicast multimedia data with the use of one multicast stream over heterogeneous networks, like the Internet, and has the capability to adapt the transmission of the multimedia data to network changes. In this paper, we describe a "one multicast stream" mechanism for adaptive transmission of multimedia data, which is based on real time protocols. In addition, the proposed mechanism uses an inter-receiver fairness function in order to treat the group of receivers with fairness in a heterogeneous environment. We evaluate the adaptive multicast transmission mechanism through a number of simulations and compare it with a number of similar schemes available to the literature (LBA+, TFMCC, PGM, TBRCA).

1 Introduction

The subject of adaptive multicast of multimedia data over networks with the use of one multicast stream has engaged researchers all over the world. During the adaptive multicast transmission of multimedia data in a single multicast stream, the sender application must select the transmission rate that satisfies most the receivers with the current network conditions. Three approaches can be found in the literature for the implementation of the adaptation protocol in a single stream multicast mechanism: equation based ([6]), network feedback based ([1–3]) or based on a combination of the above two approaches ([9]).

In the proposed mechanism, we concentrate on the implementation of a mechanism for monitoring the network condition and estimating the appropriate transmission rate for multicast multimedia data in one multicast stream, in order to satisfy most the heterogeneous group of receivers. The most prominent feature of the proposed adaptive multicast transmission mechanism is that the proposed mechanism provides the most satisfaction to the group of receivers, with the current network condition, and at the same time is trying to have "friendly" behavior to other network applications. In addition, the network monitoring capabilities, of the proposed mechanism, is based on a combination of parameters in order to determine the network conditions. Moreover, all the required modules for the implementation of the adaptive transmission mechanism are located on the server side only. This means, that any application, which is compatible with the transmission of multimedia data through RTP sessions (for example mbone tools) can access our service and benefit from its adaptive transmission characteristics. More information about the proposed mechanism can be found in [2].

In this paper, we give also a detail comparison of the proposed mechanism with other "one multicast stream" schemes available to the literature. Main target of this comparison is to compare the proposed mechanism performance with the performance of other "one multicast stream" schemes available to the literature against the following criteria: TCP friendliness, Stability, Scalability and Convergence time to stable state. The above parameters set outline well the behavior of a layered encoding congestion control scheme.

2 Overview of Adaptive Multicast Transmission Mechanism

The sender application is using RTP/RTCP protocols for the transmission of the multimedia data. Receivers receive the multimedia data and inform the sender application for the quality of the transmission with the use of RTCP receiver reports. The sender application collects the RTCP receiver reports, analyses them and determines the transmission rate r that satisfy most the group of receivers with the current network conditions. The sender application keeps information about each treceiver i, and each time receives one RTCP receiver report from receiver i, estimates the receiver i's preferred transmission rate r_i (which represent the transmission rate that this receiver will prefer if it was the only one receiver in the multicast transmission of the multimedia data). The sender application uses the IRF and RF_i functions which are presented in [5], in order to determine the transmission rate that satisfy most the group of receivers. RF_i function for the receiver i is defined in [5] as follows:

$$RF_i(r) = \frac{\min(r_i, r)}{\max(r_i, r)} \tag{1}$$

Where r_i is the transmission rate that the receiver i prefers (r_i represents the transmission rate that this receive will prefer, if it was the only one receiver in the multicast transmission of the multimedia data) and r is the transmission rate that the sender application is planning to use. From the equation (1) it is obvious that the receiver i is satisfied when the $RF_i \approx 1.0$ and complete satisfied when $RF_i = 1.0$ (when $r_i = r$). IRF function for a group of n receivers is defined in [5] as follows:

$$IRF(r) = \sum_{i=1}^{n} a_i * RF_i(r) \tag{2}$$

subject to $\sum_{i=1}^{n} a_i = 1$ and $0 \leq a_i \leq 1, 0, i = 1, ..., n$.

Where r is the transmission rate that the sender application is planning to use and a_i is the weight of the receiver i to the computation of the IRF value. From the equation (2), it is obvious that for greater values of IRF function the group of receivers is more satisfied and for lesser values of IRF function the group of receivers is less satisfied. The sender application in repeated time spaces estimates the transmission rate r for the multicast transmission of the multimedia data. The sender application is using as satisfaction measurement the IRF function defined in equation (2) and is usually treating all receivers as equal, which means that the weight a_i for all the receivers $i, i = 1...n$ in IRF function is $a_i = \frac{1}{n}$ (The number n of the receivers can easily be computed by the RTCP protocol). If the sender application wants to treat unequally the group of receivers, can assign priority to some receivers with the use of unequal a_i values.

3 Algorithms of Adaptive Multicast Transmission Mechanism

The proposed mechanism is using two algorithms: Feedback analysis algorithm and update sender rate algorithm. Feedback analysis algorithm analyses the feedback information that the receiver i sends and every time the sender application receives a RTCP receiver report from the receiver i, runs the feedback analysis algorithm in order to estimate the preferred transmission rate r_i, which will satisfy the receiver i. Feedback analysis algorithm is using the values of packet loss rate and the delay jitter from the RTCP receiver report and passes them through the appropriate filters. The feedback analysis algorithm characterizes the network on the following conditions, based on the filtered values of packet loss rate and delay jitter: (1) Condition congestion: When the network is in congestion condition, the packet loss rate is high and the transmission quality of the data is low. (2) Condition load: When the network is in load condition the transmission quality is good (3) Condition unload: When the network is in unload condition either packet losses does not exist or the packet loss rate is very small.

The changes among the network conditions for the receiver i are based on the filtered values of the packet loss rate and delay jitter concerning this receiver. More particularly, for the packet loss rate we define two values LR_c (congestion packet loss rate) and LR_u (unload packet loss rate), which control the changes among the network conditions based on the following procedure:

$$\begin{aligned} &if(LR^i_{new} \geq LR_c) \rightarrow congestion \\ &if(LR_u < LR^i_{new} < LR_c) \rightarrow load \\ &if(LR^i_{new} \leq LR_u) \rightarrow unload \end{aligned} \tag{3}$$

Where: $LR^i_{new}$the new filtered value of packet loss rate for the receiver i, and J^i_{new} the new filtered value of delay jitter for the receiver i. Feedback analysis algorithm apprehends the abrupt increase of delay jitter as a precursor of network congestion and set the network condition for receiver i to congestion:

$$if(J^i_{new} > \gamma * J^i_{old}) \rightarrow congestion \tag{4}$$

Where γ is a parameter, which specifies how aggressive the feedback analysis algorithm will be to the increase of delay jitter. In order to estimate the new value of the receiver i's preferred transmission rate r_i, we use the following procedure:

$$\begin{aligned}
&if(network = unload) \rightarrow r_{i-new} = r_{i-old} + R_{increase}\\
&if(network = load) \rightarrow r_{i-new} = r_{i-old}\\
&if(network = congestion) \rightarrow r_{i-new} = r_{i-old} * (1 - LR^i_{new})\\
&r_{i-old} = r_{i-new}
\end{aligned} \tag{5}$$

Where: r_{i-new}: The new value of the receiver i's preferred transmission rate r_i. r_{i-old}: The old value of the receiver i's preferred transmission rate r_i. $R_{increase}$: The factor with which the sender application increases the transmission rate in the case of available bandwidth. When the network condition of receiver i is unload, we increase the preferred transmission rate r_i by adding a factor $R_{increase}$, in order to decrease the dissatisfaction of receiver i due to unutilized bandwidth. When the network condition of receiver i is congested, the preferred transmission rate r_i is reduced by multiplying with the factor $1 - LR^i_{new}$, in order to decrease the dissatisfaction of receiver i due to packet losses. When the network condition of receiver i is load we do not change the receiver i's preferred transmission rate r_i, because the receiver i is satisfied with the current transmission rate. The sender application in repeated time spaces estimates the transmission rate r for multicast the multimedia data with the use of update sender rate algorithm. The estimation of the sender application transmission rate r is aiming to increase the satisfaction of the group of receivers based on the satisfaction measurement that the function $IRF$of equation (2) provides.

The update sender rate algorithm is using an Additive Increase Multiplicative Decrease (AIMD) mechanism in order to estimate the new transmission rate r. This algorithm is similar to the algorithm that the TCP rate control uses.

When the sender application is estimating the new transmission rate r, it has three opportunities: (1) To increase the transmission rate by adding a factor, $R_{increase}$ (r_{incr}). (2) To keep the previous transmission rate (r_{stay}). (3) To decrease the transmission rate by multiplying with a factor less that 1, $R_{decrease}$ (r_{dcr}).

The update sender rate algorithm is selecting as new transmission rate r, the transmission rate r from r_{incr}, r_{stay}, r_{dcr} which provides the most satisfaction to the group of receivers, which means the transmission rate r from r_{incr}, r_{stay}, r_{dcr} that has the greater IRF value. In addition the update sender rate algorithm is updating the old value of the preferred transmission rates of all the receivers in order the feedback analysis algorithm to be aware of the current transmission rate. Here is the summary of the update sender rate algorithm operation:

$$
\begin{aligned}
&r_{incr} = r_{old} + R_{increase} \\
&r_{stay} = r_{old} \\
&r_{dcr} = r_{old} * R_{decrease} \\
&r_{new} = MaxIFR_{r=r_{incr},r_{stay},r_{dcr}}[IFR(r)] \\
&receiver - i_{i=1..n} : r_{i-old} = r_{new} \\
&r_{old} = r_{new}
\end{aligned} \tag{6}
$$

Where r_{new} is the new transmission rate of the sender application, and r_{old} is the previous transmission rate of the sender application.

4 Description of other "one multicast stream" schemes

In this section we describe the major "one multicast stream" mechanisms available to the literature:

- LBA+ ([9]): LBA+ stands for "Enhanced Loss - Delay Based Adaptation Algorithm" and it is an end-to-end mechanism for the transmission of multimedia data with the use of "one multicast stream" technique. LBA+ is based on the network congestion condition and is using receivers' feedback in order to estimate the network congestion condition. LBA+ is using the RTP/RTCP protocol for the collection of network statistics (packet loss rate and delay jitter) and it uses this information in order to estimate TCP friendly transmission rates. In addition to that LBA+, is using also an analytical model of TCP in order to estimate TCP friendly transmission rates.
- TFMCC ([11]): TFMCC stands for "TCP - Friendly Multicast Congestion Control" and it is a mechanism for one multicast stream multimedia data transmission which is based on the use of an analytical model of TCP in order to obtain TCP friendliness. TFMCC main characteristics are the scalable RTT time measurements, the advance mechanism that prevents the feedback implosion problem and the friendliness against the other data stream that are using the network.
- PGM ([8]): PGM stands for "Single Rate Multicast Congestion Control Scheme" and it is a mechanism for one multicast stream multimedia data transmission which is based on the use of an analytical model of TCP in order to obtain TCP friendliness. Main characteristic of PGM is the innovative algorithm for the selection of the receiver which acts as representative of the receivers group. This algorithm assumes that the representative receiver does not change across the time but is just moving to the different network locations.
- TBRCA ([10]): TBRCA stands for "Target Bandwidth Rate Control Algorithm" and is used for the transmission of multimedia data with the use of one multicast stream. TBRCA operation is based on the feedback that it receives from the receivers. Basic target of TBRCA is to maximize the overall amount of multimedia information that the receivers' group receives and

at the same time TBRCA tries to serve also receivers with low bandwidth network connection.

5 Performance evaluation of "one multicast stream" schemes

In this section we compare the performance of the proposed mechanism with other "one multicast stream" mechanisms available to the literature regarding the following parameters: TCP friendliness, stability, scalability and convergence time to stable state. The above parameters set outline well the behavior of a "one multicast stream" congestion control scheme.

Figure 1 shows the bandwidth distribution to a bottleneck link shared by the proposed mechanism and a TCP connection. It is obvious from figure 1 that the behavior of the proposed mechanism against TCP traffic is not friendly with the firm definition of the term (a TCP friendly flow is a flow that consumes no more bandwidth than a TCP connection, which is traversing the same path with that flow ([7])). The sender application would have ideal behavior if it reduces its transmission rate and keeps it steady while the transmission of TCP traffic takes place. Nevertheless, the TCP traffic has transmission rate of more than 0.5 Mbps many times and maximum transmission rate of 0.8Mbps during the simulation, which is good performance for TCP transmission. In addition, the sender application many times realizes bandwidth and provides it to TCP source and in one case (32$^{\text{nd}}$ second) the sender application realizes 0.3 Mbps of its bandwidth.

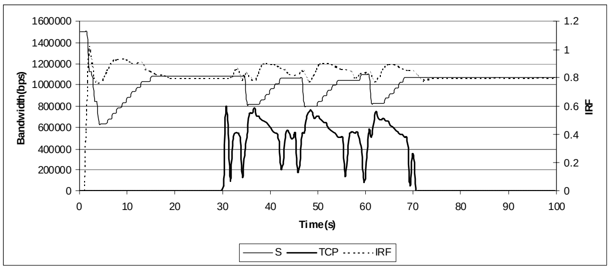

Fig. 1. Proposed application bandwidth, TCP bandwidth and IRF function values

Figure 2 shows the bandwidth allocation during the transmission of TCP traffic together with traffic produced by LBA+. The scenario of the experiment includes the transmission of one TCP stream and one LBA+ stream over the same network path over the Internet. As someone can see in figure 1 LBA+ has friendly behavior (not with the firm definition of the term) against TCP traffic and some times the LBA+ traffic receives more bandwidth than TCP traffic and some times the opposite. In addition, LBA+ has a relative stable operation

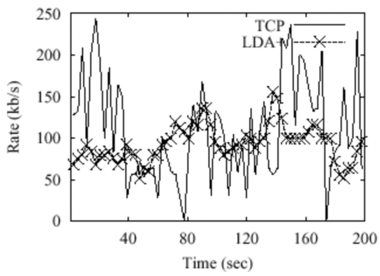

Fig. 2. LBA+ performance against TCP traffic

and its transmission rate does have heavy fluctuations. Regarding the scalability issue, LBA+ is using the feedback mechanism provided by RTCP. Concerning the time to stable state the LBA+ does not have good performance and need significant time to obtain stable state. Comparing the proposed mechanism with LBA+ we can draw the following conclusions: LBA+ has better performance than the proposed mechanism regarding TCP friendliness. The proposed mechanism has better behavior than LBA+ regarding the transmission stability and in addition the proposed mechanism take into account fairness issues regarding receivers.

Figure 3 shows how the TFMCC shares a bottleneck link with TCP traffic. The simulation scenario includes the transmission of one TFMCC sessions together with 15 TCP connections over a bottleneck link. As figure 3 shows, TFMCC follows the behavior of TCP traffic. In addition TFMCC has not stable operation and does has heavy fluctuations to its transmission rate. Regarding scalability issue, TFMCC is using a suppression mechanism to the receivers' feedback with very good results. Comparing the TFMCC behavior with the proposed mechanism behavior we can draw the following conclusions: TFMCC has better performance regarding TCP friendliness and the proposed mechanism has better performance regarding the stability of transmission rate. In addition TFMCC has better performance than the proposed mechanism regarding the scalability issue.

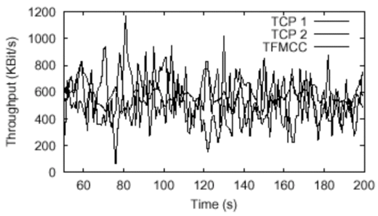

Fig. 3. TFMCC performance against TCP traffic

Figure 4 shows the bandwidth allocation during the transmission of TCP traffic and traffic produced by PGM over a bottleneck link. As figure 4 shows PGM has friendly behavior against TCP and both PGM and TCP get similar bandwidth shares. In addition, PGM has a stable operation. Regarding scalability capabilities, PGM is using a mechanism for the selection of a representative receiver (which is used as representative of receivers' group) and PGM controls the transmission of the multimedia data based on the feedback produced by the representative receiver. One drawback of PGM is the fact that needs support by the network devices in order to avoid the feedback implosion problem. Concerning the time to stable state the PGM has good performance and obtain stable state very quickly. Comparing the PGM behavior with the proposed mechanism behavior we can draw the following conclusions: PGM has better performance regarding TCP friendliness and the proposed mechanism has better performance regarding their stability of transmission rate. Moreover the proposed mechanism does not assume any support by the network devices.

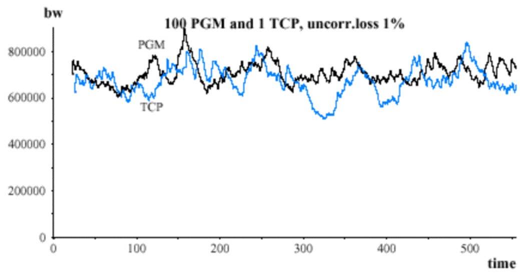

Fig. 4. PGM performance against TCP traffic

Figure 5 shows the bandwidth allocation during the transmission of TCP traffic and traffic produced by TBRCA. The scenario of the experiment includes the transmission of TCP traffic and TBRCA traffic over a 155 Mbps ATM virtual circuit (VC) with background traffic that consumes 154 Mbps. As someone can see in figure 5, initially only TBRCA traffic is transmitted to the VC and TBRCA consumes all the available bandwidth. When the transmission of TCP traffic starts, TCP traffic starts consuming bandwidth and after some time TCP traffic consumes more bandwidth than TBRCA. Moreover TBRCA has a relative stable operation. Regarding scalability issues, TBRCA requires support from the network devices in order to avoid feedback implosion problem. Comparing the TBRCA behavior with the proposed mechanism behavior we can draw the following conclusions: TBRCA is a TCP friendly mechanism but in some cases TBRCA traffic receives significantly smaller bandwidth shares than TCP traffic. The proposed mechanism has better performance regarding the stability of transmission rate. Moreover the proposed mechanism does not assume any support by the network devices.

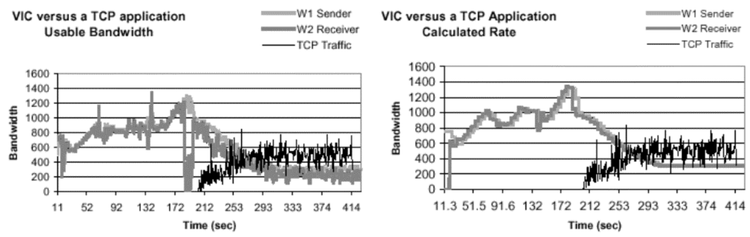

Fig. 5. TBRCA performance against TCP traffic

Table 1 summarizes the comparison of the proposed mechanism against the others "one multicast stream" schemes. As this table shows, the proposed mechanism has not good performance against TCP traffic (with the strict definition of the term TCP friendly: both TCP traffic and proposed mechanism do not get the same bandwidth share) and in general terms has good performance, comparing with the other "one multicast stream" schemes. On the other hand, the proposed mechanism does starve the TCP traffic and the TCP traffic has good performance with the existence of traffic produced by the proposed mechanism. The main advantage of the proposed mechanism comparing with the other schemes is the fact that the proposed mechanism in the only one mechanism that takes into account the issue of fairness among the receivers.

Table 1. Comparison of the proposed mechanism with the other "one multicast stream" schemes

	Prop. mech.	LBA+	TFMCC	PGM	TBRCA
TCP friendliness	average	good	very good	very good	good
Stable transmission rate	yes	average	no	average	average
Convergence time	good	average	good	good	average
Stable operation	very good	no	no	good	average
Scalability	average (RTCP)	average (RTCP)	very good (suppression mechanism)	good (representative receiver)	average (network supported)
Limitations	no	no	no	requires support from the network devices	requires support from the network devices
Fairness among receivers	yes	no	no	no	no

6 Conclusion - Future Work

In this paper, we are concentrating to the design of a mechanism for monitoring the network condition and estimating the appropriate rate for the transmission of the multimedia data in order to treat with fairness the receivers. In addition, we compare the proposed mechanism performance with the performance of other "one multicast stream" schemes available to the literature. Main conclusion of this comparison is that the proposed mechanism has good performance and its main drawbacks are the friendliness against TCP traffic and its main advantage is the fact that addresses the issue of fairness among the receivers.

Our future work includes the improvement of the proposed mechanism's behavior against TCP traffic. In addition we will investigate the behavior of the proposed mechanism during the multicast transmission in very large group of receivers. The multicast transmission in very large group of receivers encounters the feedback implosion problem ([1]). Furthermore, we will investigate the scalability of proposed mechanism and how the proposed mechanism will deal with the feedback implosion problem. Moreover, we plan to extend the proposed mechanism with the use of multicast in multiple streams in order to treat with more fairness a heterogeneous group of receivers.

References

1. Bolot, J. C., Turletti, T., Wakeman, I. Scalable feedback control for multicast video distribution in the internet. In Proceedings of SIGCOMM 1994, pages 139–146, London, England, August 1994. ACM SIGCOMM.
2. C. Bouras, A. Gkamas, "Multimedia Transmission with Adaptive QoS based on Real Time Protocols", International Journal of Communications Systems, Wiley InterScience, 2003; 16, pp. 225 - 248.
3. Busse, I., Deffner, B., Schulzrinne, H. Dynamic QoS control of multimedia applications based on RTP, Computer Communications, Jan. 1996.
4. Floyd, S. Fall, K. Promoting the Use of End-to-End Congestion Control in the Internet, Submitted to IEEE/ACM Transactions on Networking, 1998.
5. Jiang, T., Zegura, E. W., Ammar, M. Inter-receiver fair multicast communication over the internet. In Proceedings of the 9th International Workshopon Network and Operating Systems Support for Digital Audio and Video (NOSSDAV) , pages 103–114, June 1999.
6. Pandhye, J., Kurose, J., Towsley, D., Koodli, R. A model based TCP-friendly rate control protocol, Proc. International Workshop on Network and Operating System Support for Digital Audio and Video (NOSSDAV), Basking Ridge, NJ, June 1999.
7. Pandhye J., Firoiu V., Towsley D., Kurose J., "Modeling TCP throughput: a simple model and its empirical validation", ACMSIGCOMM, September 1998.
8. Rizzo L., "pgmcc: A TCP - friendly Single - Rate Multicast Congestion Control Scheme", Proceedings of SIGCOMM'2000, Stockholm, August 2000.
9. Sisalem D., Wolisz A., "LDA + TCP - Friendly Adaptation: A Measurement and Comparison Study," in the 10th International Workshop on Network and Operating Systems Support for Digital Audio and Video (NOSSDAV'2000), June 25 - 28, 2000, Chapel Hill, NC, USA.
10. Smith, H., Mutka, M., Rover, D. A Feedback based Rate Control Algorithm for Multicast Transmitted Video Conferencing, Accepted for publication in the Journal of High Speed Networks.
11. Widmer J., Handley M., "Extending Equation - based Congestion Control to Multicast Applications", Proc. ACM SIGCOMM (San Diego, CA), Aug. 2001.