A Mechanism for Multicast Multimedia Data with Adaptive QoS Characteristics

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Abstract. In this paper, we describe a mechanism for adaptive transmission of multimedia data, which is based on real time protocols. The proposed mechanism can be used for multicast multimedia data over heterogeneous networks, like the Internet, and has the capability to adapt the transmission of the multimedia data to network changes. In addition, the adaptive multicast transmission mechanism uses an inter-receiver fairness function in order to treat the group of receivers with fairness in a heterogeneous environment. The proposed mechanism uses a "friendly" to the network users congestion control policy to control the transmission of the data. We evaluate the adaptive multicast transmission mechanism through a number of simulations in order to examine its fairness to the group of receivers and its behavior against TCP and UDP data streams.

1 Introduction

The multicast transmission of real time multimedia data is an important component of many current and future emerging Internet applications, like videoconference, distance learning and video-on-demand. The heterogeneous nature of the Internet makes the multicast transmission of real time multimedia data a challenge. Different receivers of the same multicast multimedia data may have different processing capabilities, different loss tolerance and different bandwidth available in the paths leading to them. Should the sender application let the receiver with the least capacity dictate the adaptation? Is it fair the sender application ignores such a receiver? General speaking the sender application must treat the group of receivers with fairness.

Today, the underlying infrastructure of the Internet does not sufficiently support Quality of Service (QoS) guarantees. The new technologies, which are used for the implementation of networks, like the Asynchronous Transfer Mode (ATM) provide capabilities to support QoS in one network domain but it is not easy to implement QoS among various network domains, in order to provide end-to-end QoS to the user. As a result, in the future users may have the capability to request specific end-to-end QoS even over the Internet, but this is not feasible today. In addition, many researchers stand that the cost for providing end-to-end QoS is too big, and it is better to invest on careful network design and careful network monitoring, in order to identify and upgrade the congested network links ([6]).

The proposed mechanism uses RTP/RTCP (Real time Transmission Protocol / Real time Control Transmission Protocol) ([15]) for the transmission of the multimedia data. The RTP protocol seems to be the de facto standard for the transmission of multimedia data over the Internet and is used by mbone tools (vit, vat, etc) and ITU H.323 applications. In addition RTCP offers capabilities for monitoring the transmission quality of multimedia data. We use RTCP for the implementation of the network monitoring capabilities of the proposed mechanism.

In this paper, 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.

The multicast transmission of multimedia data over the Internet has to accommodate receivers with heterogeneous data reception capabilities. To accommodate heterogeneity, the sender application may transmit one multicast stream and determine the transmission rate that satisfy most the receivers ([3], [1], [16], [9], [13]), may transmit at multiple multicast streams with different transmission rates and allocate receivers at each stream ([8], [5]) or may use layered encoding and transmit each layer to a different multicast stream ([11], [4], [20]). An interesting survey of techniques for multicast multimedia data over the Internet is presented in paper [10]. It is important for adaptive real time applications to have "friendly" behavior to the dominant transport protocols of today's Internet ([7]). Paper [14] presents an end-to-end rate-based congestion control mechanism for the transmission of real time data in the Internet, which follows the macroscopic behavior of TCP.

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 ([13]), network feedback based ([3], [1], [16], [9]) or based on a combination of the above two approaches ([17]). I. Busse et al. in [3] select the appropriate transmission rate based on the percentage of the receivers that are congested and the percentage of the receivers that are loaded. D. Sisalem in [16] proposes the Loss Based Adjustment (LBA) mechanism for selecting the appropriate transmission rate. The LBA mechanism is using the number of receivers (among other parameters) in order to select the appropriate transmission rate. D. Sisalem et al. in [17] extend the LBA mechanism in order to use an equation based adaptation mechanism, except of the network feedback based adaptation mechanism. H. Smith et al. in [18] propose the Target Bandwidth Rate Control (TBRC) mechanism. The TBRC mechanism is using the dependency of the packets during the multicast transmission (among other parameters) in order to maximize the usable bandwidth for the receivers. T. Jiang et al. in [9] introduce the Receiver Fairness (RF) and the Inter-Receiver Fairness (IRF) functions. The sender application is using the RF and IRF functions in order to determine the rate that satisfy most the group of receivers.

In this paper we present a mechanism for adaptive multicast of multimedia data over networks with the use of one multicast stream based on network feedback. The proposed mechanism is an extension of our work presented in [2]. Paper [2] presents an unicast congestion control mechanism for adaptive multimedia applications. The rest of this paper is organized as follows: In section 2, we give a brief overview of the adaptive multicast transmission mechanism. Section 3 presents the algorithms on which the operation of adaptive multicast transmission mechanism is based. Detailed description of our simulation results is presented in section 4. Finally, section 5 concludes the paper and discusses some of our future work.

2 Overview of Adaptive Multicast Transmission Mechanism

This section gives an overview of the adaptive multicast transmission mechanism operation. We assume that we have a sender application, which transmits multimedia data to a group of n receivers with the use of multicast in one stream. 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.

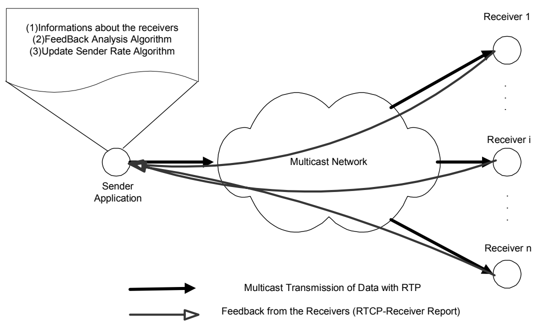

Fig. 1. Architecture of adaptive multicast transmission mechanism

The sender application keeps information about each receiver 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 estimation of receiver i's preferred transmission rate r_i is done with the use of feedback analysis algorithm, which is described in section 3.1. The feedback analysis algorithm is an extension of our work, which is presented in [2].

The sender application uses the IRF and RF_i functions which are presented in [9], in order to determine the transmission rate that satisfy most the group of receivers. RF_i function for the receiver i is defined in [9] 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 RF_i function has values in [0.0, 1.0], and the receiver i is satisfied when the $RF_i \approx 1.0$ and complete satisfied when $RF_i = 1.0$ (when $r_i = r$). The receiver i is not satisfied when the $RF_i << 1.0$. Receiver i can encounter dissatisfaction either of packet loss (when $r_i < r$) or of unutilized bandwidth (when $r_i > r$). IRF function for a group of n receivers is defined in [9] 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 $0 \le a_i \le 1, 0, i = 1, \ldots, n$.

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 $i, i = 1 \ldots n$ in IRF function is $a_i = \frac{1}{n}$ [1]. If the sender application wants to treat unequally the group of receivers, can assign priority to some receivers with the use of

[1] The number n of the receivers can easily be computed by the RTCP protocol

unequal a_i values. The sender application estimates the transmission rate r for the multimedia data with the use of update sender rate algorithm, which is described in section 3.2. Figure 1 shows the architecture of the proposed adaptive multicast transmission mechanism.

3 Algorithms of Adaptive Multicast Transmission Mechanism

This section gives a detailed description of the algorithms on which the operation of adaptive multicast transmission mechanism is based. We present two algorithms: (1) The feedback analysis algorithm which is used for the estimation of receiver i's r_i preferred transmission rate and (2) the update sender rate algorithm which is used for the estimation of sender transmission rate r.

3.1 Feedback Analysis Algorithm - Estimation of Receiver i's r_i Preferred Transmission Rate

Feedback analysis algorithm analyses the feedback information that the receiver i sends to the sender application (with the use of RTCP receiver reports), concerning the transmission quality of the multimedia data. 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. The receiver i's preferred transmission rate r_i represent the transmission rate that this receiver will prefer if it was the only one receiver in the multicast transmission of the multimedia data.

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 use of filters is essential in order to avoid a solely phenomenon to affect the behavior of the feedback analysis algorithm and lead to wrong estimations of the receiver i's preferred transmission rate r_i. More particularly the value of the packet loss rate passes the following filter:

$$LR^i{}_{new} = a * LR^i{}_{old} + (1-a) * LR^i{}_{net} \quad \textbf{(3)}$$

Where: $LR^i{}_{new}$: The new filtered value of packet loss rate for the receiver i. $LR^i{}_{old}$: The previous filtered value of packet loss rate for the receiver i (When the multicast transmission of the data starts $LR^i{}_{old} = 0$). $LR^i{}_{net}$: The value of the packet loss rate from the RTCP receiver report that the receiver i sent. a: This parameter specifies how aggressive the feedback analysis algorithm will be to the values of the packet loss rate, which receives from the RTCP receiver report. For the parameter a stands $0 \le a \le 1$. The value of the delay jitter passes the following filter:

$$J^i{}_{new} = \beta * J^i{}_{old} + (1-\beta) * J^i{}_{net} \quad \textbf{(4)}$$

Where: $J^i{}_{new}$: The new filtered value of delay jitter for the receiver i. $J^i{}_{old}$: The previous filtered value of delay jitter for the receiver i (When the transmission of the data starts $J^i{}_{old} = 0$). $J^i{}_{net}$: The value of the delay jitter from the RTCP receiver report that the receiver i sent. β: This parameter specifies how aggressive the feedback analysis module will be to the values of the delay jitter, which receives from the RTCP receiver report. For the parameter β stands $0 \le \beta \le 1$.

We can designate the operation of the feedback analysis algorithm with the appropriate selection of α and β parameters values. 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. The receiver i encounters dissatisfaction due to packet losses. In this case the receiver i's preferred transmission rate r_i is less than the current transmission rate. (2) Condition load: When the network is in load condition the transmission quality is good. The packet loss rate is in affordable value, which does not cause problems to the presentation of the multimedia data. The current transmission rate satisfies the receiver i. In this case the receiver i's preferred transmission rate r_i is near to the current transmission rate. (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 receiver i encounters dissatisfaction due to unutilised bandwidth. In this case receiver i's the preferred transmission rate r_i is more than the current transmission rate.

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:

$$\begin{aligned} &if\,(LR^i{}_{new} \ge LR_c) \rightarrow congestion \\ &if\,(LR_u < LR^i{}_{new} < LR_c) \rightarrow load \\ &if\,(LR^i{}_{new} \le LR_u) \rightarrow unload \end{aligned} \quad \mathbf{(5)}$$

The analysis of the filtered delay jitter by the feedback analysis algorithm is based on the fact that abrupt increase of delay jitter may denote that the queues of the routers on the transmission path to receiver i had been overloaded and this may cause congestion to the network during the next moments. 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. More particularly the feedback analysis algorithm uses the following procedure for the analysis of filtered delay jitter:

$$if\,(J^i{}_{new} > \gamma * J^i{}_{old}) \rightarrow congestion \quad \mathbf{(6)}$$

Where γ is a parameter, which specifies how aggressive the feedback analysis algorithm will be to the increase of delay jitter. In other words γ specifies quantitatively the expression "abrupt 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:

$$\begin{aligned} & if\,(network = unload) \rightarrow r_{i-new} = r_{i-old} + R_{increase} \\ & if\,(network = load) \rightarrow r_{i-new} = r_{i-old} \\ & if\,(network = congestion) \rightarrow r_{i-new} = r_{i-old} * (1 - LR^{i}{}_{new}) \end{aligned} \tag{7}$$

$$r_{i-old} = r_{i-new}$$

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 $1 - LR^{i}{}_{new}$ (which means that we set the receiver i's preferred transmission rate r_i to be the maximum transmission rate that will not cause packet losses to the receiver i), 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. In addition the preferred transmission rate r_i of the receiver i cannot be greater than a value $r\max$ and cannot smaller than a value $r\min$. The values of $r\max$ and $r\min$ depends on the network and application type.

The operation and the behavior of the feedback analysis algorithm are influenced by the parameters, which are used ($\alpha, \beta, \gamma, LR_c, LR_u, R_{increase}$). The choice of the above parameters depends on the network and the kind of the dominant traffic in it. The appropriate parameters for each network can be defined through a series of experiments and simulations. From our simulations, we found some values that tune the behavior of the feedback analysis algorithm: $\alpha = 0.75$, $\beta = 0.8$, $\gamma = 2$, $LR_c = 0.055$, $LR_u = 0.01$ and $R_{increase} = 50.000bps$. More information about the tuning of the above parameters can be found in [2].

3.2 Update Sender Rate Algorithm - Estimation of Sender Transmission Rate r

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. When the sender application estimates the new transmission rate r tries to provide to the group of receivers the better satisfaction that the current network conditions can provide.

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 ([19]). We chose an algorithm similar to TCP's rate control algorithm for fairness reasons to the allocation of network resources (like bandwidth), especially during network congestion periods.

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:

$$\begin{gathered} r_{incr} = r_{old} + R_{increase} \\ r_{stay} = r_{old} \\ r_{dcr} = r_{old} * R_{decrease} \\ r_{new} = MaxIFR_{r=r_{incr},r_{stay},r_{dcr}}[IFR(r)] \\ receiver - i_{i=1..n} : r_{i-old} = r_{new} \\ r_{old} = r_{new} \end{gathered} \quad \textbf{(8)}$$

Where r_{new} is the new transmission rate of the sender application, and r_{old} is the previous transmission rate of the sender application. In addition the transmission rate r_{new} cannot be greater than a value $r\max$ and cannot smaller than a value $r\min$. The values of $r\max$ and $r\min$ depends on the network and application type.

The update sender rate algorithm does not take directly into account the current network condition, during the estimation of new transmission rate r_{new} for the sender application. The current network conditions are taken directly into account by the feedback analysis algorithm, during the estimation of receivers' preferred transmission rates r_i. Because the values of the receivers' preferred transmission rates r_i are involved to the calculation of $IFR(r)$ the update sender rate algorithm takes indirectly into account the current network conditions. The simulation that we made (Section 4) shows that the approach of update sender rate algorithm to take in account the satisfaction of the receivers directly and to take in account the current network condition indirectly work well.

With the above described procedure the transmission rate of the sender application is always set to the value that satisfy most the group of receivers with the current network conditions. In our simulations we use the following values for the parameters of the update sender rate algorithm: $r\max = 2.000.000bps$, $r\min = 200.000bps$, $R_{increase} = 50.000bps$ and $R_{decrease} = 0.75$.

4 Simulations

In this section, we present a number of simulations that we made in order to analyze the behavior of the adaptive multicast transmission mechanism. Primary aims of the simulations were the study of adaptive multicast transmission mechanism fairness regarding the group of receivers and mechanism's behavior regarding the dominant traffic model of today's Internet (TCP and UDP traffic). We implemented our mechanism and run simulations in the LBNL network simulator ns-2 ([12]). We run three simulations: (1) Multicast transmission of adaptive multimedia in heterogeneous receivers. (2) Multicast transmission of adaptive multimedia in heterogeneous receivers and UDP traffic at the same time. (3) Multicast transmission of adaptive multimedia in heterogeneous receivers and TCP traffic at the same time. During all the simulations we used the following values for the parameters of our algorithms: $\alpha = 0.75$, $\beta = 0.8$, $\gamma = 2$, $LR_c = 0.055$, $LR_u = 0.01$, $r\max = 2.000.000bps$, $r\min = 200.000bps$, $R_{increase} = 50.000bps$, $R_{decrease} = 0.75$ and $a_i = \frac{1}{n}$, $i, i = 1...n$ where n is the number of the receivers. During our simulations we had 20 receivers.

4.1 Simulation One: Multicast Transmission of Adaptive Multimedia in Heterogeneous Receivers

In this simulation we investigate the behavior of the adaptive multicast transmission mechanism and its capability to treat with fairness a heterogeneous group of receivers. Figure 2 shows the topology of this simulation. The bandwidth of each link is given to the simulation topology and varies from 0.5 Mbps to 2.0 Mbps. All the links in the simulation topology have delay 10 ms and they use the drop-tail[2] (FIFO) policy to their queue. In addition, all the links in the simulation topology are full duplex. During this simulation, we have one sender application (S) that multicast multimedia data to a group of 20 receivers (R1 to R20) with the use of the adaptive multicast transmission mechanism. Receivers R1 to R10 are connected to router n2 and receivers R11 to R20 are connected to router n3. The receivers transmit RTCP receivers reports with the use of the RTCP adaptive feedback mechanism and the sender application runs the update sender rate algorithm every 1 second. We run this simulation for 100 seconds and the sender application starts transmitting data with transmission rate of 1.5 Mbps.

Figure 3 shows the sender application transmission rate and the values of IRF function. When the sender application starts transmitting data with transmission rate 1.5 Mbps all the receivers, except R1, R2, R3, R11 and R12, encounter

[2] Drop-tail is the most common queue policy to Internet routers.

dissatisfaction due to packet losses because their available bandwidth is less than 1.5 Mbps. The sender application starts reducing the transmission rate in order to treat with fairness all the receivers. The sender application reduces its transmission rate near to 0.6 Mbps (5^{th} second). In this point, the dissatisfaction that the "fast" receivers (for example R1 or R9) encounter due to unutilized bandwidth is more that the dissatisfaction that the "slow" (for example R4 or R12) receivers encounter due to packets losses. The sender application starts increasing the transmission rate in order to treat with fairness all the receivers. At 15^{th} second the transmission rate of the sender application is stabilized near to 1.0 Mbps and the sender application keeps this transmission rate until the end of the simulation. At 15^{th} second the sender application has found the transmission rate that satisfy most the group of receivers with the current network conditions. In addition from 15^{th} second to 100^{th} second the value of *IRF* function is stable because the sender application does not change its transmission rate.

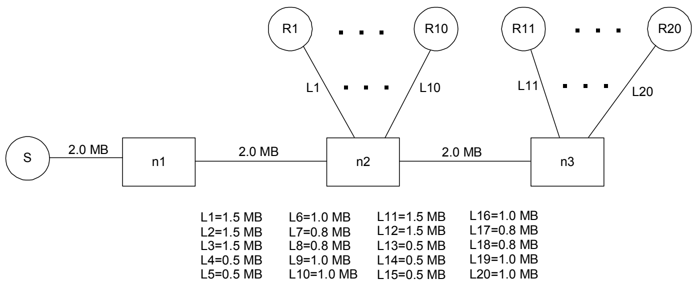

Fig. 2. Topology of simulation one

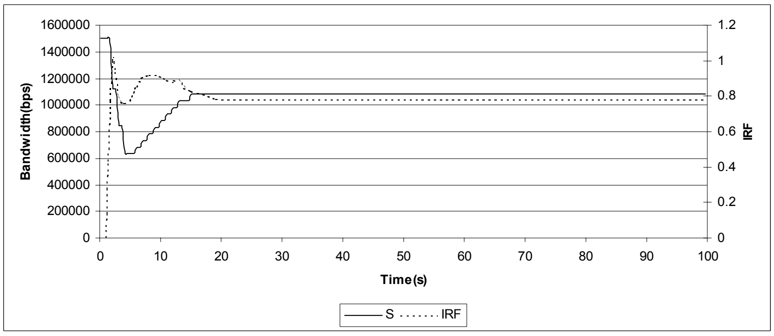

Fig. 3. Sender application bandwidth and IRF function values of simulation one

The adaptive multicast transmission mechanism behaves well: after some time the sender application finds the transmission rate that satisfy most the group of receivers and keeps that transmission rate while the network conditions are not changed. In addition the value of transmission rate (~ 1.0 Mbps) that satisfy most the group of receivers is the expected due to the fact that the most of the receivers prefer transmission rate of 1.0 Mbps.

4.2 Simulation Two: Multicast Transmission of Adaptive Multimedia in Heterogeneous Receivers and UDP Traffic at the Same Time

In this simulation, we transmit at the same time multimedia data with the use of the adaptive multicast transmission mechanism and UDP traffic. During this simulation, we investigate the behavior of the adaptive multicast transmission mechanism during network congestion produced by a greedy UDP traffic. Figure 4 shows the topology of this simulation. The topology of this simulation is the same with the topology of simulation one, except for that we have added two nodes A and B connected to router n1 and router n3 respectively. We have again one sender application (S) that multicast multimedia data to a group of 20 receivers (R1 to R20) with the use of the adaptive multicast transmission mechanism. Receivers R1 to R10 are connected to router n2 and receivers R11 to R20 are connected to router n3. In order to produce UDP traffic, we attach to node A, a CBR (Constant Bit Rate) traffic generator (CBR-Source), which transmits data to a CBR-Receiver attached to node B. The CBR-Source produces UDP traffic with constant transmission rate of 1.5 Mbps. The receivers transmit RTCP receivers reports with the use of the RTCP adaptive feedback mechanism and the sender application runs the update sender rate algorithm every 1 second. We run this simulation for 100 seconds and the sender application starts transmitting data with transmission rate of 1.5 Mbps. The CBR-Source starts the transmission of the data at 30th second, and stops the transmission of the data at 70th second.

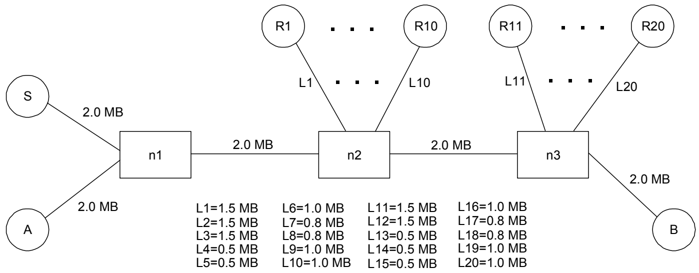

Fig. 4. Topology of Simulations two

In this simulation the sender application except of treat with fairness the group of receivers, it must share the bandwidth of the congested links between the router n1, n2 and between router n2, n3 with the CBR-Source, when the CBR-Source transmits data. Figure 5 shows the sender application transmission rate, the CBR-Receiver bandwidth and the values of *IRF* function. The sender application finds the transmission rate that satisfies most the group of receivers (15th second) after some instability in the transmission rate. When the transmission of UDP traffic starts (at 30th second), congestion occurs to links between the router n1, n2 and between router n2, n3. The receivers prefer smaller transmission rates due to congestion condition, and the sender application reduces its transmission rate near to 0.5Mbps and keeps this transmission rate for the next 40 seconds, during which the transmission of UDP traffic takes place. When the transmission of UDP traffic stops (70th second), the sender application gradually reserves again the available bandwidth. The value of *IRF* function is stable when the transmission rate of the sender application is stable, and floats between 0.77 and 0.97 when the transmission of UDP traffic takes place. The *IRF* function has higher values, when the transmission of the UDP traffic takes place, because all the receivers encounter packet losses due to congested links between the router n1, n2 and between router n2, n3 and all the receivers are satisfied with the small transmission rate that the sender application selects.

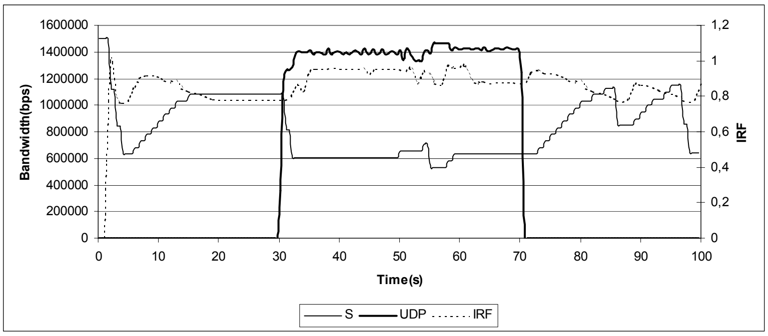

Fig. 5. Sender application bandwidth, UDP-Receiver bandwidth and IRF function values of simulation two

It is obvious from Figure 5 that the proposed mechanism has "friendly" behavior to UDP traffic and good behavior during network congestion condition. When the transmission of UDP traffic starts the sender application reduces its transmission rate and when the transmission of UDP traffic stops the sender application reserves again the available bandwidth.

4.3 Simulation Three: Multicast Transmission of Adaptive Multimedia in Heterogeneous Receivers and TCP Traffic at the Same Time

In this simulation, we transmit at the same time multimedia data with the use of the adaptive multicast transmission mechanism and TCP traffic. During this simulation, we investigate the behavior of adaptive multicast transmission mechanism against TCP traffic. Figure 6 shows the topology of this simulation. The topology of this simulation is the same with the topology of simulation two except for the capacity of some links has changed. We have again one sender application (S) that multicast multimedia data to a group of 20 receivers (R1 to R20) with the use of the adaptive multicast transmission mechanism. Receivers R1 to R10 are connected to router n2 and receivers R11 to R20 are connected to router n3. In order to produce TCP traffic, we connect to node A and B, a FTP server and a FTP client respectively. The FTP server transmits a file to FTP client using "4.3BSD Tahoe TCP" protocol [19]. The receivers transmit RTCP receivers reports with the use of the RTCP adaptive feedback mechanism and the sender application runs the update sender rate algorithm every 1 second. We run this simulation for 100 seconds and the sender application starts transmitting data with transmission rate of 1.5 Mbps. The transmission of the file from the FTP server to FTP client, starts at 30th second and stops at 70th second.

In this simulation, the sender application except of treat with fairness the group of receivers it must share the bandwidth of the congested links between the router n1, n2 and between router n2, n3 with the TCP traffic when the FTP transmission of the file take place. Figure 7 shows the sender application transmission rate, the TCP source bandwidth and the values of *IRF* function. The sender application finds the transmission rate that satisfies most the group of receivers (15th second) after some instability in the transmission rate. When the transmission of TCP source starts (at 30th second), congestion occurs to links between the router n1, n2 and between router n2, n3. The receivers prefer smaller transmission rates due to congestion condition, and the sender application releases bandwidth in order the TCP traffic to use it. In contrast with the previous simulation, the sender application does not keep steady its transmission rate during the 30th and the 70th seconds, when the transmission of TCP traffic takes place. When the transmission of the TCP traffic takes place, the sender application realizes some bandwidth (about 0.3 Mbps) for a while and reserves it again. When the transmission of TCP traffic stops (70th second) the sender application gradually reserves again the available bandwidth. The value of *IRF* function is stable when the transmission rate of the sender application is stable and floats between 0.79 and 0.90 when the transmission of TCP traffic takes place, because the transmission of TCP traffic produce instability to the adaptation mechanism and the sender application changes continually its transmission rate.

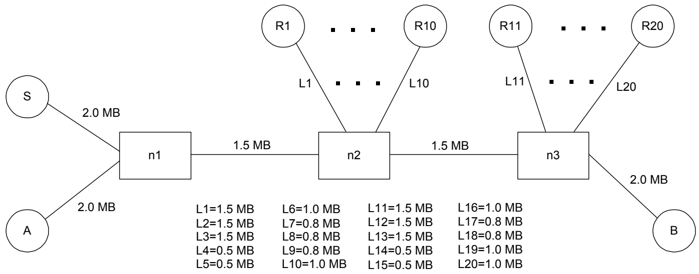

Fig. 6. Topology of Simulations three

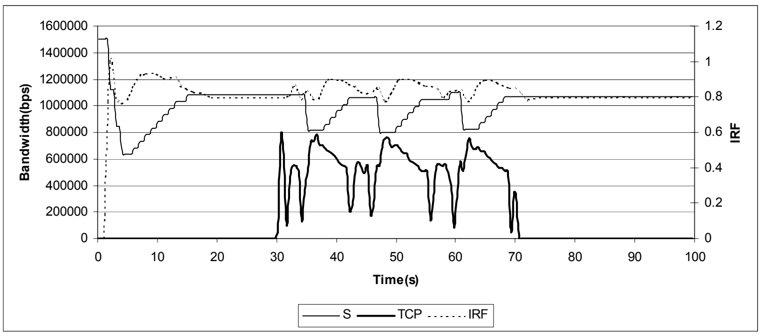

Fig. 7. Sender application bandwidth, TCP source bandwidth and IRF function values of simulation three

It is obvious from Figure 7 that the behavior our mechanism to TCP traffic is not so "friendly" as the behavior to UDP traffic. 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^{nd} second) the sender application realizes 0.3 Mbps of its bandwidth.

5 Conclusion - Future Work

In this paper, we present a mechanism for multicast transmission of adaptive multimedia data in a heterogeneous group of receivers. 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 investigate the behavior of the proposed mechanism against the dominant transport protocols of today's Internet (TCP and UDP). The proposed mechanism uses RTP/RTCP protocols for the transmission of multimedia data. Through a number of simulations, we draw the following conclusions: (1) The proposed mechanism treats with fairness the group of receivers. (2) The proposed mechanism has “friendly” behavior both to UDP and TCP traffic streams.

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.

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