A NETWORK TRAFFIC SCHEDULER FOR A VOD SERVER ON THE INTERNET*

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Abstract: Most of the Video on Demand (VoD) systems were designed to work in dedicated networks. However, there are some approaches that provide VoD service in nondedicated and best effort networks, but they adapt the media's quality according to the available network bandwidth. Our research activities focus on VoD systems with high quality service on nondedicated networks. Currently, we have designed and developed, to integrate in the VoD server, a network manager that provides: total network control, network state information, and adaptation of the transmission rate in a TCP-Friendly way. The present work describes this network manager, named Network Traffic Scheduler (NTS), which incorporates a congestion control algorithm named "Enhanced Rate Adaptation Protocol" (ERAP). ERAP is an optimization of the well-known protocol denominated "Rate Adaptation Protocol" (RAP). Maintaining the basic behavior of RAP, ERAP increases the efficiency of the NTS by reducing the resources usage (of the server and the network). These components has been extensively evaluated by simulations and real tests in which the resource consumption and the performance were measured. This paper presents the advantages of using ERAP instead of RAP in a VoD server, and its viability to be integrated within the NTS in a VoD server on nondedicated networks.

1 INTRODUCTION

Our research group have designed a distributed Video on Demand (VoD) system architecture for dedicated networks (Qazzaz et al., 2003). This platform works very well in Local Area Network (LAN) environments, but many problems appear that need to be solved whilst trying to provide a service in nondedicated Wide Area Network (WAN) environments.

Therefore, our investigation must be lead to solve other kinds of problems that arise when the VoD system works in a nondedicated network such as the Internet. It is necessary to adapt the main component of the platform, the Video Proxy Server (VPS), to a set of restrictions imposed by the type of communication that is possible to make on this type of networks.

Since at the moment the Internet does not offer a multicast service (in a general form), the first change that must be made is that the VoD service will be made solely on unicast channels (instead of a combination of multicast and unicast).

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Another problem is that on the Internet all the packets are treated in the same way, without discrimination or explicit delivery guarantees, known as the "best-effort service model". The quality of service and the resources are not guarantees in terms of bandwidth, transfer delay, delay variation (jitter), and packet losses. The packet losses can be due to physical reasons and droppings. In the first case, the noise that affects the transmission of the signals or the putting out of service of active communication devices (links, routers, etc.) are included. The dropping of packets is due to the congestion that takes place because the network is overloaded by the demand of network resources and this demand is close or exceeds the network capacity (Tanenbaum, 2002).

When the transmission is made within the network capacity, the packets arrive at their destination (except for a few that are affected by noise in general) and the number of received packets is proportional to the number of sent packets. However, when increasing the traffic, the network cannot handle it, and begins to lose packets. The result is a state of congestion which continues to build up and get worse. When a packet loss exists (for example if any router has dropped it),

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the retransmission timer will expire and the sender will possibly retransmit the packet. More packets in the network make the situation worse, and as the capacity of delivery of the network continues to be the same, the proportion of received packets against the sent packets declines even more. (Kurose and Ross, 2004; Tanenbaum, 2002)

The optical fiber is being used in long distance communication of the backbones so the loss of packets by transmission errors is relatively low. Consequently, in the Internet communication, the most of the expired packets are due to the network congestion, rather the transmission errors. Therefore, all Transport Control Protocol (TCP) algorithms assumes that packets expire is consequence of the network congestion.

In order to adapt to congestion states, the VoD server must decreases the traffic sent by the network, allowing the possibility to identify two types of strategies for this aim: *Dynamic Rate Control* and *Anticipation*.

The Dynamic Rate Control strategy, allows the transmission rate to adapt dynamically according to the network conditions, but to reduce the transmission rate, the media quality must be lowered (Wang and Schulzrinne, 1999). The solutions based on Anticipation are applicable only to prerecorded continuous media, that is to say, the whole multimedia file that can be sent to the client already exists. This strategy, takes advantage of the periods of low use of the server and the network bandwidth, to send media in advance that the client will consume later. In this way, it is possible to tolerate moments in which the network or the server are overloaded.

The algorithm developed in our VPS is named *Credit Based Media Delivery Algorithm* (Cb-Mda), and belongs to the Anticipation category. This type of algorithm, from a general point of view, is a *Logical Channel Scheduler* (LCS), or is also known as *Streams Scheduler*.

The main function of the LCS is to plan the different streams in order to make use of the server exit bandwidth. However, the LCS does not have the capacity to manage the communications at the network level. This capacity, necessary to guarantee the QoS, is assigned to a new module that we named *Network Traffic Scheduler* (NTS). The NTS works together with the LCS (Cb-Mda in our case), and a feedback from NTS to LCS is provided to inform of the communication state between the server and clients. When the NTS detects congestion, in the path of communication with a client, it must warn the LCS so that it takes the suitable measures to reduce the transmission rate of the logical channel or stream. For that reason, it is necessary that the NTS and the LCS work coordinately and cooperatively to obtain the best solution to the problem raised. Without information of the real transmissions and the state of the network, the LCS will not be able to work accordingly.

The investigation has been centered in the development of a NTS based on User Datagram Protocol (UDP) packets and with congestion management so that the VPS can be used on the Internet with total guarantees and quality of service. The use of UDP, instead of TCP, has significant advantages with respect to the overload that implies TCP in the transmission. Furthermore, TCP does not give any of the information required by the LCS, and it does not allow that this capacity can be added to it.

The rest of this paper is organized as follows: in section 2 the related works to this article are described. In section 3, the NTS and its interaction with the rest of the components of the VPS are explained. Furthermore, the characteristics that must have their algorithm of congestion control in order to adapt it to the new necessities of the VPS are indicated. The “Rate Adaptation Protocol” (RAP) (Rejaie et al., 1999; Rejaie et al., 1998), a well-known congestion control protocol, whose basic characteristics are suitable in order to adapt to the NTS, is described in section 4. This protocol is optimized to be introduced in the NTS, giving origin to the new protocol "Enhanced Rate Adaptation Protocol" (ERAP), which is presented in section 5. In section 6, the RAP and ERAP behaviours are compared, and the advantages to using ERAP instead of RAP as congestion control algorithm of the NTS are analyzed. Finally, in section 7 the conclusions and future works are described.

2 RELATED WORKS

Congestion control has been studied for many years, nevertheless, the existing protocols are not many if we are restricted to TCP-Friendly protocols for multimedia transmission on the Internet.

Jacob et al. (Jacobs and Eleftheriadis, 1997) presents a congestion control algorithm similar to TCP except that this one does not make retransmissions.

Cen et al. (Cen et al., 1997) proposes the “Streaming Control Protocol” (SCP) for real-time streaming of continuous multimedia data across the Internet. Dorgham Sisalem et al. presents the “Loss-Delay Adjustment Algorithm” (LDA) in (Sisalem and Schulzrinne, 1998) and their variant “LDA+” in (Sisalem and Wolisz, 2000). Sally Floyd et al. exposes the protocol “TCP-Friendly Rate Control” (TFRC) in (Floyd et al., 2000), and later M. Handley et al. describes this protocol in the RFC 3448 (Handley et al., 2003).

Reza Rejaie et al. presents the “Rate Adaptation Protocol” (RAP), and a mechanism of layered quality

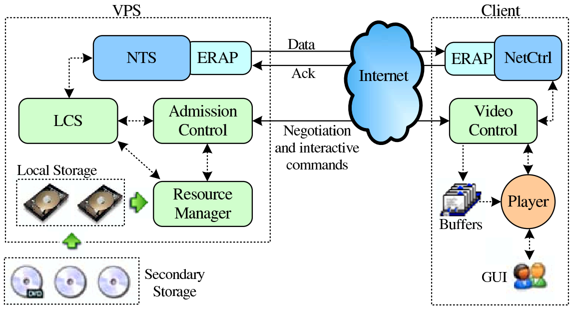


Figure 1: VPS and Client architectures.

adaptation for Internet video streaming in the context of unicast congestion control are described in (Rejaie et al., 2000).

Between the commercial players of video streaming on the Internet we can include: RealPlayer, Windows Media Player, and QuickTime. Although their algorithms of congestion control have not been revealed, there exists publications ((Hessler and Welzl, 2005) and (Chung et al., 2002) among others) that show studies on the performance and its behaviours with respect to if they are or not TCP-Friendly according to the answers that they offer in congestion cases in real networks.

3 NETWORK TRAFFIC SCHEDULER

The NTS is the component of the VPS in charge to manage the communications at the network level guaranteeing a transport with QoS to the LCS. The new architecture of the VPS is presented in figure 1.

The NTS will inform the LCS about the state of communication with each client, in such a way that the LCS can carry out their tasks with real and updated information of the network. If the NTS finds that the communication of a certain connection has improved, then the LCS will be able to increase the transmission rate of that channel. In case the communication gets worse, the LCS will have to reduce the transmission rate immediately.

The NTS must be equipped with a congestion control algorithm to fulfill their responsibilities. All congestion control algorithm that is used on the Internet must have the property of being TCP-Friendly, that is to say, the use of bandwidth (in a stable state) does not have to be greater than that required by TCP under similar circumstances (Floyd and Fall, 1999). If irresponsible users capture more bandwidth than corresponds to them, the delivery service of the users that cooperate for the good operation of the network could be degraded. Furthermore, the stability and operation of the whole system would be threatened (Gevros et al., 2001).

Different strategies for the congestion control exist, but the NTS must make use of a strategy of type *Black-Box* (this scheme sees the network as a black or closed box) since there is no feedback from the interconnection devices (routers) and the feedback from the receiver is the only one available. The streaming applications generally have better behaviour (and therefore they require it) when the traffic flows in continuous forms and with few throughput variations in the time. This property is known as Smooth Sending Rate. Although it is a characteristic that our LCS does not use, it is good that the congestion control protocol of the NTS includes it for versatility purposes.

Furthermore, it has been decided for a line of work of *rate-based* protocols (instead of *window-based*) because they have the advantage of not transmitting packets in bursts. If a sender has the transmission capacity of b packets/second, is better to send a packet each $1/b$ seconds and not a sequence of b packets every second. A sequence of b packets could be unacceptable for an interconnection device that does not have the sufficient amount of memory to store it temporarily. On the other hand, if it does have the sufficient amount of memory, long queues in routers will increase the end-to-end delay. This can cause retransmissions (when expiring the time of the packets) that produces an increase of the congestion state.

Several rate-based congestion control protocols exist in the literature, between which are LDA, RAP, TFRC, and SCP (see section 2). Many of the existing protocols are proposed by investigators, and in many cases a real implementation does not exist, or this does not adapt to the NTS. Thus, our investigation has been oriented in taking the specifications of the most open of them, and the most suitable to our objectives.

We have chosen the RAP specification because this has the expected properties of black-box, rate-based, and smooth sending rate, and also because it is simple, well documented, and well known as the Network Simulator - NS2 (McCanne and Floyd, 2005; Breslau et al., 2000) includes an implementation of them. Nevertheless, in the official site of RAP no real implementation of this protocol is provided.

The RAP protocol was adapted and optimized to fulfill the requirements of the NTS, giving origin to a new RAP version named *Enhanced Rate Adaptation Protocol* (ERAP).

The NTS and the NetCtrl (see figure 1) use the ERAP protocol for media stream transmission, and the rest of the communications between the VPS and the Client (negotiations and interactive commands) are performed by a TCP connection.

4 THE RAP PROTOCOL

As their authors explain in (Rejaie et al., 1999; Rejaie et al., 1998), the RAP is an end-to-end rate-based congestion control mechanism that utilizes an Additive Increase Multiplicative Decrease (AIMD) algorithm for rate adaptation to achieve TCP-friendliness. The AIMD rate adaptation algorithm can have non TCP-friendly behaviour when a heavy load produces the reduction of TCP's performance. Therefore, a fine-grain rate-adaptation mechanism is added to assist RAP in becoming more stable and reacting to temporary congestion while realizing the AIMD algorithm at a coarser granularity. Basic RAP has a TCP-Friendly behaviour in many situations, and the fine-grain rate-adaptation mechanism expands this behaviour to more circumstances.

The RAP protocol is mainly implemented in the sender. A RAP sender sends data packets with a sequence number, and a RAP receiver sends an acknowledgment (ACK) for each packet. Using this feedback, the RAP sender can detect losses and sample the round-trip-time (RTT). Timeouts and gaps in the sequence space are used to detect packet losses. This protocol only considers the packet losses as a congestion symptom. Unlike TCP, a RAP sender may send many packets before receiving a new ACK.

In principle, an ACK packet includes the sequence number of the delivered data packet, but in order to provide robustness against single ACKs losses, the following redundant information is added to them:

- *lastRecv*: the sequence number of the last received packet
- *lastMiss*: the sequence number of the last missed packet previous to *lastRecv*, or 0 if no packet was missing
- *prevRecv*: the sequence number of the received packet previous to *lastMiss*, or 0 if *lastRecv* was the first packet

For example, if the pattern of packet losses was "1 _ _ 4 _ _ 7", the values are: *lastRecv* = 7, *lastMiss* = 6, and *prevRecv* = 4. A packet with sequence number Seq_i will be considered received if $((lastRecv \geq Seq_i)and(Seq_i > lastMiss))or(Seq_i = prevRecv)$.

Basically, the algorithm is conformed to by two timers, the *ipgTimer* and the *rttTimer*, that along with the reception of ACKs are the triggers of the three events that direct the algorithm.

Furthermore, the following important variables are included: *IPG* (inter packet gap, used to control the transmission rate), *SRTT* (smoothed - or estimated - round trip time, defines the periodicity of the transmission rate increment), and *Timeout* (defines the time to live or expiration of the packet); and a list named *transmissionHistory*. This list stores the sent packets but not the acknowledged packets, which change to the *ignored* state when the loss of a packet is detected. Thus, the algorithm only reacts once, at the most, when a burst of packet losses occurs, related to the same congestion case. The *SRTT* and *Timeout* variables are updated based on the last sample RTT using the Jacobson/Karels algorithm (Jacobson and Karels, 1988).

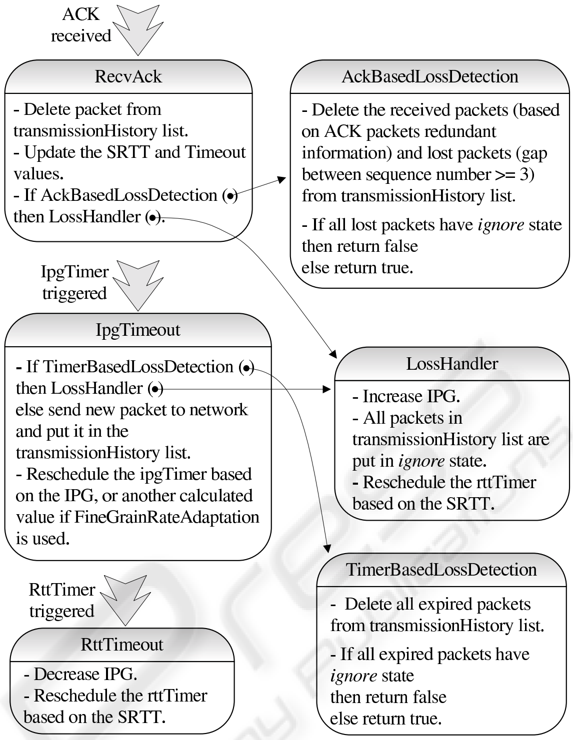


Figure 2: The RAP algorithm.

The diagram of the figure 2 describes the operation of the RAP, where the arrows represent the events that direct it, and the boxes represent their main procedures. To start the algorithm, initially, the *IpgTimeout* and the *RttTimeout* procedures are invoked.

The algorithm uses an adaptation scheme (AIMD) so that if does not detect congestion, the transmission rate is periodically and additively increased, and if it detects congestion, the transmission rate is immediately and multiplicatively decreased.

When no packet losses are detected, the transmission rate, S_i, is increased by a certain value α (i.e. $S_{i+1} = S_i + \alpha$) updating the value of the IPG (inter packet gap) based on equation (1).

$$S_i = \frac{PacketSize}{IPG_i} \qquad IPG_i = \frac{IPG_i * C}{IPG_i + C} \qquad (1)$$

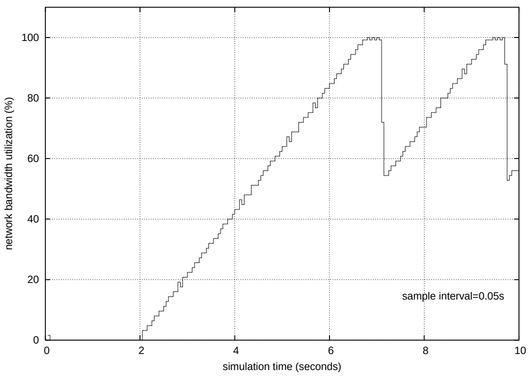


Figure 3: Characteristic RAP behaviour.

In this equation, C has the dimension of time and determines the value of α. RAP assigns to C the value of *SRTT* to emulate the TCP window adjustment mechanism in the steady state. At any rate, α represents the increment of one packet in each adjusting point. When packet losses are detected, the transmission rate is multiplicatively decreased by a certain value β (i.e. $S_{i+1} = \beta * S_i$), updating the value of the IPG based on equation (2).

$$IPG_{i+1} = IPG_i/\beta \qquad \beta = 0.5 \tag{2}$$

RAP gives a value of $\beta = 0.5$ in order to follow the behaviour adopted by TCP.

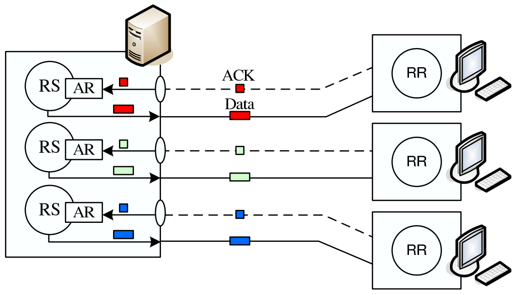
The expected behaviour would be of a progressive increase of the transmission rate in absence of congestion and a abrupt decrement of this upon congestion (tooth of mountain range) as can be observed in the figure 3.

5 THE ERAP PROTOCOL

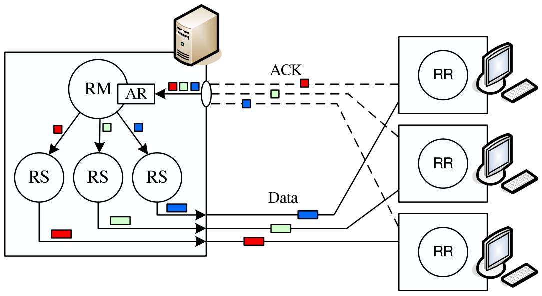
The Enhanced Rate Adaptation Protocol (ERAP) is an adaptation and optimization of the RAP protocol to be introduced in the NTS. The NTS must work with a lot of connections in order to utilize the maximum possible of the available bandwidth and to serve the greater amount of possible requests.

With the purpose of diminishing the use of resources and increasing the efficiency of the NTS, the ERAP protocol was designed with the following particular characteristics:

1. **Encapsulation:** A ERAP packet is encapsulated within a UDP packet to be transmitted by the Internet.
2. **Reliability**: as our system does not analyze streams of video, making a separation in frames, it is impossible to determine if certain packets contain data of low or high relevance. Therefore, all lost packets must be retransmitted. Is not possible to retransmit a lost packet with its initial sequence number, because of the fact that a packet with an old sequence number (smaller or equal to the last sequence number less than 3) immediately will be considered as lost and retransmitted.
   In order to solve this problem, the packets that need to be retransmitted, are transmitted as if they were new packets (i.e. with a sequence number equal to the last, plus one). However, a new sequence number is added to each packet, named *application_sequence_number*, that the receiver uses to maintain the real sequence of packets, and to identify the duplicated packets.
3. **Centralized ACKs' reception:** The decentralized ACKs' reception does not work well when it is used in a server with many active sessions. According to the model proposed by RAP, each Sender must handle his own packets' reception. In this manner, in a real implementation that uses sockets (to transmit the packets), each Sender would have a thread in charge of packets' reception. Therefore, by each active video or established session a thread would be used.
   However, ERAP centralizes the reception of ACKs from multiple destinations, and in this way, only a single thread is necessary instead of so many threads as connections as the server has. Thus, when a packet arrives at the server, it is received by a central module named RapManager and later it is given to the corresponding ErapSender.
   The RAP and ERAP architectures are showed in the figure 4 (a) and the figure 4 (b), respectively, where *RS* is a RapSender agent, *RM* is a RapManager agent, *RR* is a RapReceiver agent, and *AR* is a Acks Reception Thread.
4. **Header Size Reduction:** so that the protocol is versatile and it does not overload the network, a minimum set of information has been defined to the data and ACK packets. RAP has only one header used for data packets and ACKs packets, that contain the following integer data: *seqno*, *size*, *rap_flags*, *lastRecv*, *lastMiss*, and *prevRecv*. *Seqno* is the sequence number, *size* is the packet size, *rap_flags* identify the type (ACK or DATA) of packet, and the rest of the fields are only used when the packet is of ACK type.
   On the other hand, ERAP determines two types of headers, the *data header* and the *ACK header*, avoiding the unnecessary data transport in the headers. The *data header* has only two integer data: *seqno*, and *appSeq*. The first field is the sequence number, and the second field is the application sequence number added to manage the retransmissions. As the packets are received by different



(a) RAP architecture with decentralized ACKs' reception



(b) ERAP architecture with centralized ACKs' reception

Figure 4: RAP and ERAP architectures.

ports, ERAP automatically identifies the type of them (ACK or DATA) and, therefore, the *rap_flags* field is not used. The rest of the fields (*lastRecv*, *lastMiss*, and *prevRecv*) are specifics of the *ACK header*. The *size* field is not necessary because each ERAP packet is encapsulated in only one UDP packet.
The *ACK header* has the following fields of integer type: *port*, *seq*, *lastRecv*, *lastMiss*, and *prevRecv*. The port field is added so that the centralized ACKs' reception module (RapManager) can distinguish to what connection the ACK belongs. If the Receiver wants to send ACK packets to the Sender with a source IP (Internet Protocol) address different to the IP address by which receives the data packets, then the ACK header will contain an extra field named *IP* that will have the IP address of the Receiver by which it receives the data packets, allowing it to identify the session correctly.

5. **Optimized events management:** When in a session, the transmission of packets is suspended temporarily, and later this is re-initiated, the protocol would have to continue the transmission supposing that the network state has not changed. In this way, the network state information, learnt during the previous course (of use of connection) would be wasted. At this moment of inactivity (i.e. when there are no more packets to send) it does not make sense to continue shooting events of the protocol for that session. Then, ERAP deactivates the timers and maintains the state of the session so that when the transmission is resumed, the timers will be reactivated and the transmission rate will be continued from the previous point.
On the other hand, RAP stops only the *ipgTimer* and the *rttTimmer* continues being triggered, that will invoke the *RttTimeout* procedure repeatedly, with the consequent CPU cost. This procedure will only decrease the *IPG* if during the past *SRTT* time, a certain percentage (modifiable) of the *SRTT/IPG* packets were sent. If RAP sent less than that, the rate is not increased.

6. **Generation of information required by the LCS:** as it were explained in section 1, it is necessary that the NTS and the LCS work coordinately and cooperatively in order to optimize the transmission. The information that the NTS must give to the LCS and that ERAP must generate is: total server exit bandwidth (Mbps), available bandwidth (Mbps) for each connection at each moment, and the number of enqueued Mbits in the queue of the NTS that waits to be transmitted (considering all the packets of all the connections). All these measures only include the length of the Application Data Unit (ADU) without counting the overhead of the protocols, that is to say, the bits corresponding to the headers of the own ERAP or any used protocol of the Internet protocol stack (link, network, transport). RapManager is the central module in charge of collecting information from all ERAP connections and the network to generate this data.

6 RAP VS. ERAP

The NTS-ERAP and the RAP have been extensively evaluated by simulations and real tests in which the consumption of resources and the performance are measured. The simulator used is the NS2 version 2.29 released on October 19, 2005. This version of the NS2 includes the implementation of the RAP protocol that we used to compare with ERAP. The results presented in this section show the advantages of using ERAP instead of RAP in a VoD server.

Figure 5 shows the topology used for the simulations, where *R1* and *R2* are routers, *C1*...*Cn* are clients, and *S1* is the VPS.

The link *R1*-*R2* is the bottleneck and *R1* is the bottleneck point. The routers are simulated as elements with FIFO scheduling and drop-tail queuing. When the NTS-ERAP is simulated, *S1* is composed of a RapManager and *n* RapSender, whereas, when the RAP is simulated, *S1* only has *n* RapSender.

Table 1: Simulation parameters.

Packet Size	100, 512, 1024 bytes
Data Header Size	RAP: 24 bytes ERAP: 8 bytes
ACK Size	RAP: 24 bytes ERAP: 20 bytes
Links Bandwidth	R1-R2: 1 MB/s Others: 2 MB/s
Links Delay	10 ms
Queue Size R1-R2	10 packets

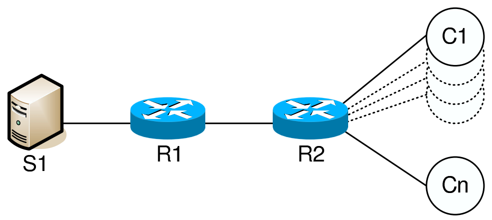


Figure 5: Simulated topology.

The simulation parameters used are shown in table 1.

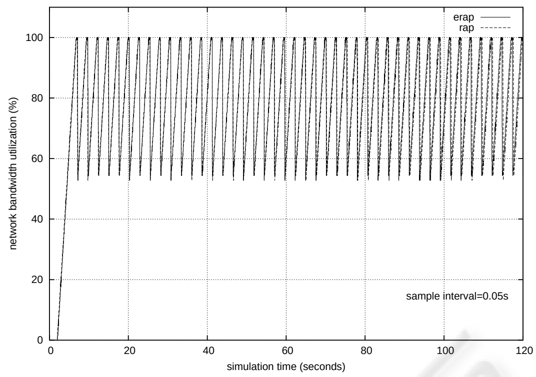
Because of the fact that ERAP does not try to modify the basic behaviour presented by the RAP protocol, one of the objectives of the simulations was to show that both protocols act in a similar way reducing or increasing the transmission rate, upon detection of congestion or in absence of it, respectively.

RAP and ERAP have presented almost the same performance as far as the use of network bandwidth, the small difference that is observed is because RAP uses greater ACK packets (4 bytes) than ERAP.

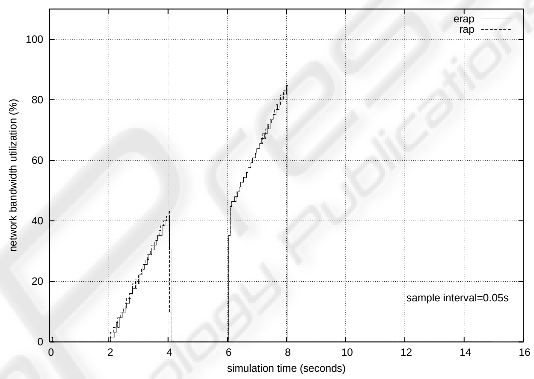
Figure 6 (a) shows the percentage of network bandwidth usage for a total simulation time of 120 seconds, with a data packet size of 100 bytes, and with a continuous data sending by only one active connection.

The transmission rate is increased until the maximum transmission capacity (100%) of the R1-R2 link is used, so that the router R1 begins to drop packets, and, upon detection of packet losses, the transmission rate is decreased immediately. These two phases, the increase and decrease of the transmission rate, are continuously repeated.

As described in item 5 of section 5, ERAP and RAP solve the inactivity problem in different ways. Nevertheless, for our specific environment, in which the packets will be given in a form of burst by the LCS, behaviour differences practically do not exist. This can be observed in the figure 6 (b), corresponding to a simulation with an inactivity period between time 4 and 6, and a data packet size of 100 bytes. Also, the deactivation of both timers at inactivity mo-



(a) Continuous trasmission for 120 seconds



(b) Inactivity times behaviour

Figure 6: RAP and ERAP network utilization.

ments that ERAP carries out has two advantages, the saving of CPU and memory resources. The queue of event scheduler will be much smaller. Instead of having $X + N$ timers, N being the total number of connections, and X the number of connections that have been selected by the LCS to be served at certain moment or slot of time, will only have $X * 2$ timers. In this way, when reducing the events queue, the used memory is reduced and the useless triggering of timers is avoided saving CPU cycles.

ERAP makes better use of the resources when the ACKs' reception is centralized (item 3 of section 5). In this approach, a single thread is necessary in order to receive the ACK packets corresponding to all connections. In this way, a great amount of memory and CPU resources are saved. Each empty thread (i.e. without data and code) occupies approximately 10 MB of memory. If a decentralized ACK packet reception is used, N active sessions (therefore, with N threads) will require $N * 10$ MB of memory, and the system will quickly begin to start swapping.

With respect to the CPU usage, when having many threads, the scheduling and context changes will be

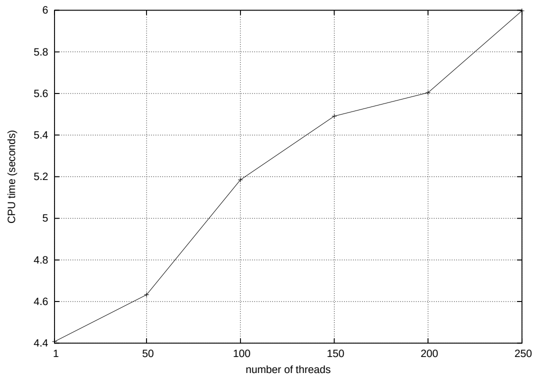


Figure 7: CPU usage according to the number of packets receiving threads.

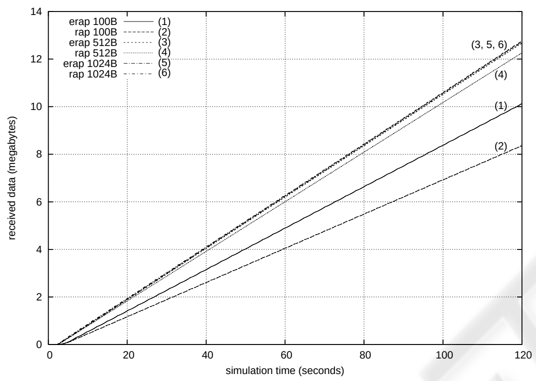
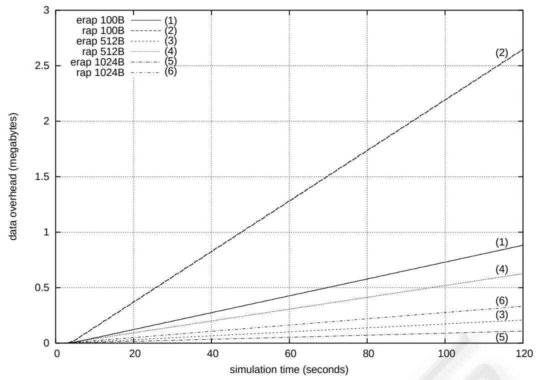
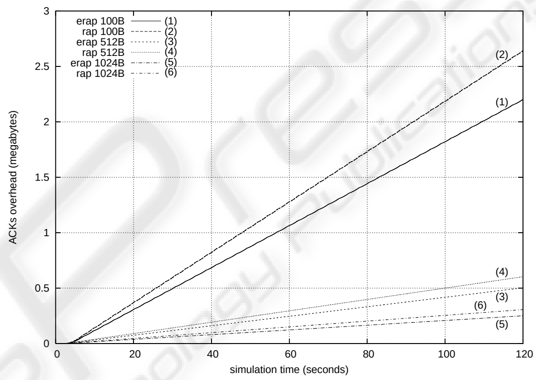


Figure 8: Amount of application data received by the client in the time.



(a) Data overhead in the time



(b) ACKs overhead in the time

Figure 9: RAP and ERAP protocols overheads.

more expensive. In order to verify this affirmation, tests were made in which a sender application sends an UDP traffic load, equitably distributed between N destination ports of another computer. And, in this other computer, N threads (one by port) were waiting to receive the corresponding packets.

The graph in figure 7 shows the sum of the times consumed in user and system mode for each set of receiving threads (1, 50, 100, 150, 200, and 250 threads), with a load of 10^6 packets uniformly distributed between participants threads (i.e. 1 thread: 10^6packets/thread, 200 threads: 5000 packets/thread). A clear increase of the consumption of CPU time is observed when the number of threads are increased. Another disadvantage, is that the number of threads that each process can have is limited by the operating system, therefore the number of sessions that the server can support will be restricted.

The header size reduction (item 4 of section 5), using different headers to data and ACK packets, improves the performance of the protocol. Simulations were made of 120 seconds in length with continuous transmission through only one active connection and sizes of data packet of 100, 512, and 1024 bytes.

In figure 8 it is observed that when the ERAP protocol is used, the amount of application data received by the client is increased, because of that, ERAP has a smaller data header.

The overhead decrease in the network, that presents the ERAP protocol as opposed to RAP, can be observed in figure 9, where the bandwidth consumption caused by data and ACK headers at each moment of the simulation time, are presented in the figure 9 (a) and 9 (b), respectively.

7 CONCLUSIONS AND FUTURE WORKS

This paper is part of an innovative investigation line focus on VoD systems that guarantee a high quality of service in the Internet domain, or another nondedicated and best effort network. We present a Network

Traffic Scheduler (NTS) to be included in a Video Proxy Server (VPS). The VPS must support a high workload, therefore, the NTS was designed to maximize the performance and minimize the resource consumption.

To achieve the better system perform of the VPS, the NTS must cooperate with it, and also must cooperate with the Internet's operation. Therefore, the NTS must include a congestion control protocol that is TCP-Friendly. Based on the specification of the RAP congestion control protocol, the ERAP has been developed to be included in the NTS module. The results presented in this paper show that ERAP improves the RAP protocol, not from the point of view of the behaviour in the adjustment of the transmission rate (that is practically the same), but from the point of view of the resource usage as much of the server as of the network. This improvement also allow to conclude that the ERAP protocol avoid the limitant factors arising on RAP protocol, giving to the video server more flexibility and opportunity to attend the client's petitions. The optimized events management, the header size reduction, and the centralized ACKs' reception, were the strategies to obtain such resource usage diminution.

The NTS has been developed and evaluated extensively by means of simulations. The future works are centered mainly in the integration of the NTS module with the rest of the VPS architecture. The benefits of this new platform, in contrast to the pre-existing platform that does not include the NTS, will be evaluated.

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