Some experience on video flow regulation with an active network approach

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***Abstract*— We present in this talk firstly some experience we got throughthe design and implementation of an acive network technology based machanism for video flow regulation. This mechanism makes use of several typical active features to perform real-time video flows analysis and responsive codec feedback control. Our tests show visible improvements obtained by our mechanism vs the classical RTCP control scheme. This work is presented in details in this joint paper with Rim Hammi.**

We then plan to discuss extensions of our mechanism including bandwidth broking and policy-oriented flow control. We conclude by putting this work in the evolution of flexible network technologies. We are particularly interested by theWeb Servive approach. Our current research effort is carried on within the RNRT/Amarillo project.

***Index Terms*— Video communication, active networks, Internet, QoS, regulation, responsive control.**

I. Introduction

IN recent years, the development and the apparition of real time application as well as multimedia applications have witnessed an exponential increase. The real time constraints of these applications present a big challenge for their integration. They need different levels of QoS adapted to the data transmission. Real time constraints of applications like video-phone, videoconference, audio and video cause few problems for their integration into networks using the IP protocol. One major point is the lack of an acceptable quality when the video communication goes across a best-effort IP network. The problem could be solved by a resources reservation scheme, such as the time-honored RSVP or the more recent DiffServ. One can also try to adapt the video communication's rate to the channel's current situation, with an adequate and responsive regulation mechanism. This paper is focused on the handling of video communication with the responsive rate control approach. It proposes a framework which uses an emerging technology, namely active networks.

The concept of active networks has been introduced rather recently. This concept has been proposed as an alternative to the client/server paradigm, and the basic idea of active network is to make the network as programmable as possible. This is achieved by introducing into the network switching nodes some user-controllable processing capacities. This last point, i.e. the dynamic placement of user-defined computing functionality, yields the fundamental difference between the active network and today's (traditional) networking technology. Indeed, today's switching nodes are solely devoted to the basic packed forwarding operation, whereas the active network approach gives the possibility to perform some additional processing on some packets when needed. Thus, an active networking capable network infrastructure provides more flexibility for individual flow handling, as well as the possibility of dynamically deploying new services.

An active networking capable network infrastructure provides more flexibility for individual flow handling, as well as the possibility of dynamically deploying new services. Technically speaking, there are two key architectural components that allow dynamic placement of user-codes in an active network. The first one is the capability of transporting runnable codes between nodes via packets, in the same manner as data are exchanged (we assume the exchanges are secured). The second one is the building of execution environments (EE) above a node OS, in order to host and run the application code.

As we can see, the main advantages of the active network reside in the fact that virtually all kind of intelligence (service) can be placed, in a dynamic manner, into a given node to handle a given traffic. One of the main active network functions that we will use in our framework is the ability to execute some operations on routers. We will use it to get access to local information, such as packet loss ratio or bandwidth allocation. Another functionally is the capability of sending and generating of signalization information. We will use this to provide dynamic feedback from network to user.

The handling of video communications is still a focus of investigation with today's networking infrastructure, which is dominated by Internet technology. This is because video traffics are real-time, burst and (selectively) sensitive to loss, and so need to be handled with a

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Dagstuhl Seminar Proceedings 04411
http://drops.dagstuhl.de/opus/volltexte/2005/85

minimum of QoS control. The generic architecture in Internet for multimedia traffic has been standardized in H.323 [13], the key element for traffic handling is the use of RTP/RTCP [26] over UDP links.

The protocol RTP is mainly designed to provide end-to-end transport functions and to insure the integrity and the facility of synchronization of the video stream. This is achieved through some fields in the RTP-header such as the Sequence Number (NS) field and the Timestamp (TS) field. The RTCP, the companion protocol of RTP, is designed to provide QoS feedback to the participants of an RTP session. The feedback is in the form of sender reports and receiver reports. These reports contain information on the quality of data delivery and information of the membership. The QoS feedback is useful to the sender. This later can adjust its transmission rate according to the information sending by the receiver. In fact, one of the main functions of RTCP aims to provide feedback to the sender/receiver (CODEC video for example) with current channel's state. The RTCP is capable of reporting loss ratio, received bandwidth, as well as jitter. The sending period is by default 4 seconds for an unicast session. It is in anyway under 5% of the total bandwidth of a session, so, for a multicast session, the period of individual reports can be even longer than 4 seconds. For the video applications, the 4-sec. period could be too long. Besides, a scalar loss ratio information is surely less accurate than a two-state Gilbert loss model (which captures not only the loss ratio, but also the loss pattern). The RTCP is unfortunately not designed for such user-specific data patterns.

Generally speaking, the control of the video communication has to be defined by application, which follows the so-called end-to-end paradigm. A number of generic issues have been identified ([33], [30]). The central problem is to achieve QoS control by minimizing congestion problem as well as error (loss) problems. In fact, the congestion control for video communication can be achieved through many approach such as the traffic filtering, the transcoding, the layered (hierarchical) coding or the rate-control. The goal of traffic filtering is to adapt a video traffic to a given link capability, this can be done, in the MPEG2 case for example, by dropping B-frames and/or P-frames. Another possible way to achieve traffic filtering is the transcoding, either from one format to another or by modifying DCT quantification coefficients. Under the layered (hierarchical) coding, video stream is provided via one base stream, and several enhancing sub-streams. The congestion control can also be done by rate-control. The generic approach is a dialog between the encoder and the decoder, for the choice of a bandwidth used for video-stream generation.

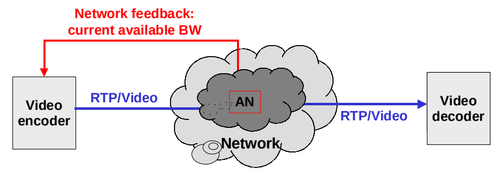

Fig. 1. Network-based rate-control

In contrast to these approaches, we propose in this paper a new concept of reactive control of video communication defined by the network based on the active network technology. This article propose a new mechanism of real time transmission control of video communication through the Internet. The originality of this framework lies in the use of the technology of active network, in order to conceive a new approach of reactive control, made to adapt the video flow to the variations of resources in the Internet. The network, according to our approach, supervises the transmission of video packets and reacts to flow variations by sending to the encoder a recommendation of the available bandwidth in the network for its flow. Figure 1 shows that the network regulate the sender rate by calculating and sending periodically the adequate video throughput, to the encoder, based on the observation of its link load and the state of the flow transmission, without requiring any feedback from the receiver.

In this paper, we propose a framework and present an implementation of reactive control and dynamic regulation of video flow, called *ARM (Active Regulation Mechanism)*. It has been implemented and tested in the active network platform AMARRAGE [2].

The specific goals of our framework are as follows:

- Modularity: Implementation of specific algorithms in the form of modules.
- Extensibility: New user-specific agents (or algorithms) can be dynamically load and modified at run time.
- Flexibility: Creation, configuration and bound of a kernel software modules responsible for performing certain specific functions on specified network flows (video flow in our case).
- Performance: Allows a feedback rate more adequate and specific to the applications and with a lower overhead. The frequency and the choice of information are under the sole control of the applications.

Before presenting our active regulation mechanism, we will first identify the existing methods of correction and controls aiming at adjusting a video communication over the Internet. Following that, we present the various components of the proposed framework. In Section IV, we present our rate-control algorithm. In Section V, we evaluate our implementation. Conclusions are stated in Section VI.

II. Related works

As far as Internet is concerned, the transmission control is different in nature from the one used in RNIS or ATM networks. Best-Effort Internet is subject to congestion that causes loss of packets and jitters variations. In this section, we will mention the various methods of control and traffic regulation conceived to adapt video communication to the constraints of transmission on Internet.

The transmission of a continuous flow, like a video communication, witness very strict times constraints. The most important constraint is the one of the bandwidth that should remain stable and superior to the video data throughput. Unfortunately, public Internet is unable to ensure the available average bandwidth in a constant way. In fact, the network is subject to frequent congestions according to the hour and place of the connection or also according to the service provider. The conditions of a transmission are thus unstable and inpredictable. The bandwidth can suddenly fall to an insufficient level of a video transmission and raise few seconds later always in a sudden way. Therefore, the challenge is to maintain a stable or at least acceptable level of quality. For video streaming, congestion control takes the form of rate control: that is, adapting the sending rate to the available bandwidth in the network. Rate control attempts to minimize the possibility of network congestion by matching the rate of the video stream to the available network bandwidth.

The existing rate-control schemes can be classified into three categories: source-based, receiver-based, and hybrid rate control [33].

Source-Based Rate-Control: Under the source-based rate control, the sender is responsible for adapting the video transmission rate by employing feedback information about the network. The source-based rate control mechanism follow two approaches: probe-based and model-based approach. In the probe-based approach, the sender probes for the available network bandwidth by adjusting the source rate in a way that could maintain the packet loss ratio below a certain threshold. For example, [28] introduce a rate control scheme for real-time traffic in networks, namely RCS. This later is based on the concept of using dummy packets to probe the availability of network resources. RCS is an en-to-end protocol. It needs to be implemented only at the source and destination.

The model-based approach is based on the throughput model of TCP. Specially the TCP-friendly model [19], [22], [20]. Under the model-based rate control, the video connection could avoid congestion in a similar way to that of TCP and it can complete fairly with TCP flow. For example, [15] present an architecture and algorithms for support of Internet Video employing image processing and networking techniques. Their architecture uses the TCP Congestion Control (TCP-CC) algorithm as a congestion indicator. The adaptable media object uses a technique called Dynamic Rate Shaping (DRS). On the other hand, [5] propose a unicast streaming flow and congestion control scheme called SCP (Streaming Control Protocol). Like TCP, SCP employs sender-initiated congestion detection through positive acknowledgement, and uses a congestion-window based policy to back-off exponentially. In [19] and [29] rate adjustment is based on TCP throughput model [7], [21].

In [27] and [23], two rate adaptation protocols, namely LDA and RAP, are presented. Both of them perform control for real time communications by means of mechanisms similar to those of TCP [16]. Sisalem and Schulzrinne [27] present the scheme called the Loss-Delay based adjustment Algorithm (LDA) for adapting the transmission rate of multimedia applications to the congestion level of the network. The LDA algorithm relies on the en-to-end Real Time Transport Protocol (RTP [26]) for feedback information. Whereas, Rejai and al [23] present an end-to-end TCP-friendly Rate Adaptation Protocol (RAP), which employs an additive-increase, multiplicative decrease (AIMD) algorithm. The RAP protocol is mainly implemented at the source. A RAP source sends data packets with sequence numbers, and a RAP sink acknowledges each packet, providing end-to-end feedback. Each acknowledgment (ACK) packet contains the sequence number of the corresponding delivered data packet. Using the feedback, the RAP source can detect losses and sample the round-trip-time (RTT).

Receiver-Based Rate-Control: In this approach, the receivers regulate the receiving rate by adding/dropping channels while the sender does not participate in the rate control. The receiver-based rate control is typically used in multicasting scalable video.

The receiver-based rate-control mechanisms follow also two approach the probe-based and the model-based approach. This later is also based on the TCP-fiendly model.

Hybrid Rate-Control: Under the hybrid rate-control, the receivers adjust the rate of video streams by adding/dropping channels. Unlike the receiver-based rate-control model, the sender in hybrid rate-control approach adjusts also the transmission rate of each channel based on feedback from the receivers [9].

In contrast to previous rate-control mechanisms, the

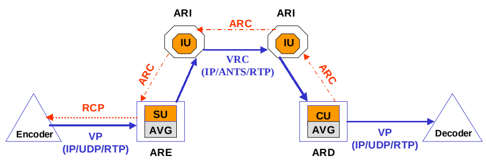

Fig. 2. The Active Regulation Mechanism architecture

proposed approach in this paper allows the network to adapt directly the source throughput thanks to the active network technology. Under this scheme, that we called *Network-Based Rate-Control*, the network is responsible for adapting the video transmission rate by employing active feedback information about the network and the state of transmission of the video flow. Our framework follow two approach: probe-based and model-based approach. They need to be implemented at the active routers. Under the probe-based approach, the active network probes for the available network bandwidth and the state of the video packets transmission, which is presented in the Section III. The model-based approach is based on the compensation algorithm presented in the Section IV. The proposed algorithm, witch is implemented in the active router, calculates the current available bandwidth for the video flow using the active feedback information collected by the active network. Under the *Network-Based Rate-Control*, the network regulate the sender rate by calculating and sending periodically the adequate video throughput, to the encoder, based on the observation of its link load and the state of the flow transmission, without requiring any feedback from the receiver (see figure 1).

III. The Active Regulation Mechanism ARM

A. Architecture

As we said in the introduction, the main idea of our mechanism is to allow routers to adapt the transmission of video flow to the available resources in the Internet.

Adopting the technology of active network, we suggest a new architecture of signalization and control of video real time transmission over Internet. The main idea is to introduce in the network capacities of treatment, control and dynamic regulation of throughput able to solve the real time transmission problems.

We have considered the following aspects for the definition of the different services of active control :

- The need to calculate the critical parameters of transmission including the loss rate, delay, jitters, bandwidth, video throughput, etc.
- Dialogue and exchange of transmission and reception report between the routers.
- Direct dialogue between the network and the sender.
- Knowledge of the state of the channel transporting the video flow.
- Rapid actions of prevention and correction at the level of the network routers.

The *Active Regulation Mechanism* is installed on the router as a kernel plugin. Figure 7 gives the most important components of the Active Router kernel implemented in Unix system.

The architecture of ARM is presented in figure 2. The entities orchestrating the control of the RTP session and which adapt the transmission are : the ARE active router (the Active Router closet to the Encoder), the ARD active router (the Active Router closet to the Decoder), the intermediate active routers ARI (Active Router Intermediate) and a user-agents (active measurement module). These later are dynamically installed at the active routers ARE, ARD and ARI. The role of each agent depends on its location :

- The main role of the agent called CU (Client User-agent) at ARD and the agents called IU (Intermediate User-agent) at ARI is to take information about the local link state, including particularly bandwidth, loss ratio and loss pattern, as well as the delay jitters.
- The main role of the agent called SU (Server User-agent) at ARE is to decide the rate-control to be taken, according to the current link state.

The position of the routers was chosen in order to control the transmission over the best-effort (IP) network as well as supervising the reception of the video data. In fact, the ARD active router, close to the receiver, witnesses the quality of video reception received by the decoder. It would be able to signal a transmission problem, losses for example. The ARE active router on the side of the encoder could on itself directly commands to the sender its behavior by means of information and statistics calculated by the user-agents. Besides, intermediary active routers bring local information useful for the control of the RTP session. They provide among other things like the link state, the state of the buffers and the waiting files (information from the MIB).

1) The video real time capsule VRC: For the purpose of our framework, we have integrate in the ANTS-AMARRAGE platform an Active Video Gateway (AVG) service that ensures in real time the conversion of the video packet into capsules and vice versa. Video datagram travel until the first active router (ARE). This latter transforms, by means of the active gateway, the video signal into a useful form by active network. It converts video packets (IP/UDP/RTP/Video) in active packets (IP/ANTS/RTP/Video) called VRC (Video Real

Corresponding IP header			+ Header specific to ARM						+ Payload
Source address	Destination address	TTL	V	Type	Previous address	NS	NS of RTP	TS of RTP	RTP / Video

Fig. 3. The Video Real time Capsule format

Corresponding IP header			+ Header specific to ARM								
Source address	Destination address	TTL	V	Type	Previous address	CSN (Num. cap)	Jitters	LR (Loss Ratio)	BVB	LI ARI Information (Info MIB, etc)	

Fig. 4. The Active Reporting Capsule format

time Capsule). On the level of the last active router (ARD), these capsules VRC will be converted into UDP datagrams. The function of this gateway is to allow real time video flow to leave the active networks towards its final destination: the terminal containing the decoder.

Video Real time Capsules are active capsules created by the active networks to ensure the transmission of the RTP packets and its video data (H.263+ in our case) through active nodes. The payload of this capsules is formed by the RTP packet. The header contains besides the fields of ANTS [31], [32], a certain number of supplementary fields. They allow the active routers ARE, ARI and ARD to control the session. The header of a video capsule is schematized in figure 3.

Apart from the usual ANTS fields, we added in the active header the following fields:

- **Sequence Number:** it represents the sequence number of the capsule.
- **RTP Sequence Number:** it is the NS field of RTP. This field was copied in the header of the capsule to facilitate the treatment done by the active nodes.
- **RTP Timestamp:** it also corresponds to the field Timestamp of RTP header. This field allows to identify the packets belonging to the same image.

2) The active report capsule ARC: The transfer of the measured information between the active router needs the creation of an active capsule called ARC (Active Report Capsule). This capsule is made to ensure the control of the real time traffic. In fact, an ARC is sent from the ARD to the ARE. When living the ARD, it carried the channel state viewed by receiver. It can also carry other information, such as the maximum/minimum rate at which the user accepts to receive the communication. On its journey up to ARE, local information on each ARI are added to the ARC.

If we consider that active routers are placed on the network (including network-network boundaries), it is reasonable to assume that the active report capsules follow the same path as the (downstream) video stream (figure 2). With our assumption on active report capsule routing, the ARE now has a precise view of the channel used by video communication : it know not only the final receive state, but also the state at each boundary (location of active routers).

The Active Report Capsule format is presented in figure 4. The header contains four supplementary fields which represent the information, measured by the user-agents, of the video flow:

- **CSN (Capsule Sequence Number)**: sequence number of capsule. This field is incremented by 1 for every transmitted ARC capsule. The initial value is 1.
- **Jitters**: this field gives the Jitters variation in milliseconds starting from the last transmitted report.
- **LR (Loss Ratio)**: this field shows the rate of the losses of video packet (VRC capsule) witnessed in the active router ARD starting from the last transmitted report.
- **BVB (Bandwidth of Video Basic flow)**: this field informs ARE router of the throughput of the video flow calculated in the ARD router starting from the last active report capsule. This field represents the bit-rate of the basic flow (H.263+ or MPEG4 flows).
- **LI (Local Information)**: this field retrieves all information necessary to the level of the intermediary active routers. On its journey up to ARE, ARC adds in this field local information on each ARI and calculated by the IU agents (MIB information, etc).

When receiving the ARC capsules, the ARE router starts the analysis of the information sent by the ARD and the ARI. The transmission parameters of the video flow will be then used by the algorithm of regulation, implemented on ARE, in order to calculate the adequate video bit-rate. The throughput calculated by the algorithm will be sent to the encoder by the mean of the RCP (ReCommendation Packet) packet presented in the next Section.

3) The recommendation packet RCP: We have seen in the previous section that the first active router (ARE) analyzes the information sent by ARD and ARI. With this full and up-to-date vision of the channel, the ARE active router can provide an accurate feedback to the encoder. The feedback is carried in a packet, called RCP (ReCommendation Packet), in order to adapt the rate of the video communication (figure 2). These raw values are transmitted into UDP.

ARE has necessary information allowing it to detect transmission problems, losses packets for example, as well as anticipating a near loss due to congestion. The mechanism allows the use of any adequate algorithm for rate-determination (compensation algorithm, for example, presented in Section IV). The active router ARE calculates by means of this algorithm the value of the bandwidth available for video traffic. It is the RCP packets that ensure the transfer of the adequate rate of the video communication to the encoder. They bring in the requirements of the network as well as those of the decoder by means of an additional channel open between the active router ARE and the video source. A port number will be exchanged between the encoder and the first active network (ARE) before the establishment of the RTP session. In the case of a videoconference, every active transmitter will receive bandwidth recommendations from the first active router which is close to it in the broadcasting tree [11]. Thus, the encoder is no longer isolated, it is constantly informed of the state of the video communication over the Internet.

B. Measurement of the QoS parameter

The transmission control of real time video communication needs instant observation and supervision of the flow and the link state. The task of this function is to calculate the transmission criteria of video data and evaluate the position of packets in the global traffic. The expected system is achieved by means of the user-agents that we have respectively implemented at the level of the actives routers ARE, ARD and ARI. Their roles are to supervise the communication and to calculate the necessary statistics for the control of the video flow.

The function of these measurement modules are different. As we have mentioned in the previous Section, the role of each measurement function depends on its location in the active router. In fact, the user-agent at the ARD router, called CU, measures the criteria of transmission of video data including losses, delay, jitters and the throughput of video flow. The location of this module of measurement in the last active router (ARD) gives the state of the video data received in the decoder. The so-called SU module analyzes on its way the flow sent by the source. It provides ARE router a complementary information to control the communication.

CU module implemented in the ARD active router has an important role. It calculate the parameters necessary for the active real time controls and signalization that we have developed. The location of this module at the level of the last on board active router ARD, allows it to determine the state of transmission of the video data in the network. These calculated parameters reflect the state of the flow received by the decoder. We estimate that the distance achieved by the flow between ARD and the terminal containing the receiver is done without damages or with minimum damages only.

The CU user-agent calculates on the level of ARD the transmission parameters of the video data through the network. It measures the rate of losses, the video throughput, the delay and the jitters. These parameters are calculated and sent every second to ARE active router. The choice of the frequency of measurement is discusses in the next Section (Section III-B.1). These measurement parameters replace those usually provided by the reception report RR of RTCP protocol. The measures achieved by this module provide more rich and more precise statistics than those of RTCP report.

The parameters of QoS, measured by CU function, will be sent by the ARD towards ARE by means of the ARC (Active Report Capsule).

1) Proposed method: The procedure of collecting real time transmission parameters is done in two steps :

1) **Observation:** When they reach the ARD router, VRC capsules will be demultiplexed towards the active video gateway (AVG). CU module, installed in this latter, proceeds to inspecting the fields of these capsules before being converted into UDP packets. In fact, CU memorizes the sequence number of every capsule as well as its timestamp. Notice that packets belonging to the same image have the same timestamp.

   These information are memorized for every supervision cycle and are then used to calculate the throughput, the arrival delay, the rate and model of losses. It consists of keeping the necessary information for the establishment of the measures of the flow state. This is achieved by examined the fields of the header of every capsule's entrance and by calling the class and under-class methods of the ANTS platform [31].

   In this way we are able to memorize the arrival

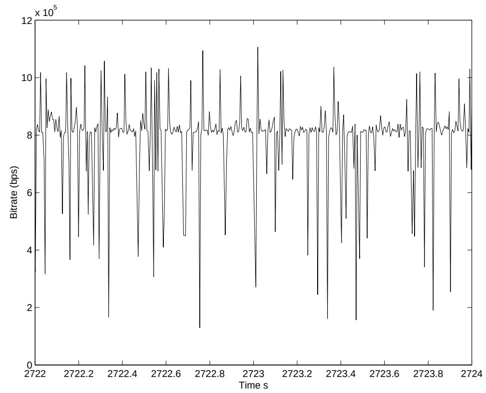

Fig. 5. The instantaneous rhythm of the video flow

time, the sequence number and the size of each capsule together with the number of transmitted images and their sizes.

The first two references are useful for the measurement of the bandwidth. These information will be used by CU module to identify the video bit rate. It is also possible to calculate the instantaneous bit rate of H.263+ basic flow. Figure 5 gives the instantaneous rhythm of the video flow received in the ARD active router. We notice that the throughput calculated by CU module is done periodically every second. We will find more explanations in the next section.

2) **Measurement:** The information memorized by CU module intend to calculate the critical parameters of video transmission:

- **The delay variation** It is the arrival time between the first capsule of one image and the first capsule of the next image. In fact, in the CIF case, images are generally partitioned into 3 or 4 packets. Each packet have the same timestamp and the arrival time between two successive capsules belonging to the same image is small. Indeed, we have chosen to calculate the arrival time between two successive images.

  The delay variation shows a congestion in one of the intermediary routers. In fact, the delay variation shows losses generally. The establishment of the measure of this parameters is used to anticipate the oncoming overflow of the queuing. The analysis of the behavior of this parameters helps ARE to determine the adequate video rate. It helps to stop losses or even avoid them. In fact, thanks to this parameter, the ARE router can command the necessary throughput for the video transmission.
- **The loss rate** The supervision of the sequence number of every capsule arriving to the ARD router is meant to detect the losses. The measurement module CU compares the values of this field with those expected and adds the difference by taking into consideration of course the fact that packets can arrive in disorder. Thus, CU can calculate the rate of losses for every sample of measurement.
- **The basic video throughput** The measures of the loss and the allocated bandwidth aim at the layout and the regulation of the video throughput. The measures are mainly based on the measurement of the throughput of the basic flow.

We have chosen a 1 second period, an empirical optimal value suggested by experiences. Indeed, the duration of observation would be between 500 ms and 2 to 3 seconds. It shouldn't neither exceed 2 to 3 seconds, at the risk of being less performing than RTCP signalization, nor being above 500 ms at the risk of causing spasmodic regulations for the encoder and unpleasant visual effects.

Moreover, 500 ms corresponds only to fifteen or so images in better cases, about 15 packets well rhythmed which would be undoubtedly insufficient as a number of samples.

On the other hand, we should be aware of the fact that the precision and the granularity of measures are conditioned by the computer performances that contain the platform and its exploitation system. As a matter of fact, the execution time of an application depends on many factors [8]. The frequency and the architecture of the processor, the memory quantity, the speed of different buses as well as the network cards influence the execution time and the good functioning of an active application. However, many other factors could also package the CPU needs. The exploitation system (OS) as well as the execution environment (EE) can also influence the application performances.

Moreover, the addition of a function of observation and measurement and the implementation of an algorithm of flow regulation written in Java at the level of the active nodes need certainly some additional resources (CPU, memory, etc). It is then important to control the use of these new functions integrated in the active network platform.

Also, we noticed that the resources consumed during an observation time of 2 seconds are slightly inferior to those required for 1 second, while the precision of the measures of the characteristics of the real time video transmission is more important for a 1 second frequency

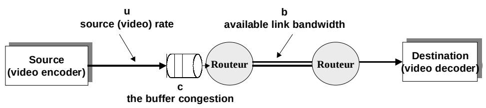

Fig. 6. Elementary network configuration with one source, one buffer, one link

than 2 seconds.

2) Functionalities: The principle of our modules of measurement is to supervise the quality of transmission of the video flow. They measure the parameters of QoS. They replace those provided usually by the decoder. The information we get through CU, IU and SU modules are more rich. The video throughput and the losses model are new entities that the receiver report RR of the real time transmission control protocol cannot provide. In addition, these measures give a continuous report on the state of the flow and its evolution in the network.

SU, IU and CU measurement functions provide respectively ARE, ARI and ARD active routers necessary tools for the RTP session controls. The QoS parameters taken by CU are sent periodically by ARD to ARE. This latter analysis the received information and those taken by SU user-agent to evaluate the situation of the video flow inside the global traffic. Using these measures achieved by the two modules as well as the local information (MIB information for example) given by IU (Intermediary User-agent), implemented at the intermediary active router, the active network is able to react in real time upon the resources variations on the link. The extensive use of this facility of IU module is under experimentation.

IV. RATE CONTROL ALGORITHM

In this section, we present a rate-control algorithm called compensation algorithm. This algorithm calculates the adequate rate of video traffic according to the available bandwidth as well as the delay caused by the network. By using this algorithm, the network tries to avoid future losses. The routers supervise the increase of the transmission delay, that generally means a close loss, and they react by recommending to the encoder to decrease its throughput if they anticipate a congestion. The value of the throughput recommended by the network is calculated by a so-called algorithm of compensation.

The algorithm of regulation presented in this section is based on the analysis of a configuration reduced to a source (video encoder) a receiver (video decoder), a buffer and a link (figure 6). This model, in spite of its simplicity, allows to reveal three characteristics and constraints that we will necessarily encounter in a more intricate network technologies:

- The unstable dynamic of the buffer (problems of memory overflow causing congestion), which is imperative to stabilize by an appropriate retroaction ;
- The fact that the delay, as perceived by the video application, is a non linear function of buffer congestion and the historical of the source transmission, and can thus be perfectly handled by controlling these two variables ;
- The interest of anticipating the variations of the bandwidth of the link.

The formula which gives the necessary throughput values is :

$$d_{t+1} = k(1 - \frac{d_t}{d_M})d_n \tag{1}$$

where d_t is the value of video flow throughput to rectify (calculated by the active probe given in Section III), d_M is the maximum throughput of the link, d_n is the nominal throughput video flow and finally k is a constant which assume that

$$k < \frac{d_M}{d_n}$$

If we assume that d_n is a constant, we will search the conditions of convergence from d_t to $\bar{d}$ where $\bar{d}$ is the equilibrium point of the formula 1:

$$\bar{d} = k(1 - \frac{\bar{d}}{d_M})d_n$$

Suppose $\Delta d_{t+1} = d_{t+1} - \bar{d}$. Then we have :

$$\Delta d_{t+1} = k(1-\frac{d_t}{d_M})d_n - \bar{d} = k(1-\frac{d_t}{d_M})d_n - k(1-\frac{\bar{d}}{d_M})d_n$$

$$\Longrightarrow \Delta d_{t+1} = -\frac{kd_n}{d_M}\Delta d_t$$

$$\Longrightarrow \Delta d_t = (-\frac{kd_n}{d_M})^t \Delta d_0$$

Therefore, we reach the following conclusion : d converges towards $\bar{d}$ if $\frac{kd_n}{d_M} < 1$, that is to say if :

$$k < \frac{d_M}{d_n}$$

Algorithm

Let d_{t+1}, d_t, d_n, d_M, J, LR, T_J, T_L, $\mathbf{D}_{BF}$, $\mathbf{D}_{EF}$ and dx denote the recommended rate, the rate of basic video flow to rectify, the nominal rate, the maximum throughput of the link, the current jitters, the current packet-loss ratio, the threshold for jitters, the threshold for packet-loss, *the recommended basic flow rate*, *the recommended rate of the enhancing sub-streams* and a constant, respectively. The sending rate is adjusted according to the following table:

TABLE I
A RATE CONTROL ALGORITHM

	$\mathbf{L}R$	$\mathbf{D}_{BF}$	$\mathbf{D}_{EF}$
$\mathbf{J} \geq \mathbf{T}_J$	$< T_L$	$d_{t+1} * 4/5$	$d_{t+1} * 1/5$
	$> T_L$	$min(d_{t+1}, d_t/2)$	0
$\mathbf{J} \leq \mathbf{T}_J$	0	$min(d_n, d_t + dx) * 4/5$	$min(d_n, d_t + dx) * 1/5$
	> 0	$d_{t+1} * 4/5$	$d_{t+1} * 1/5$

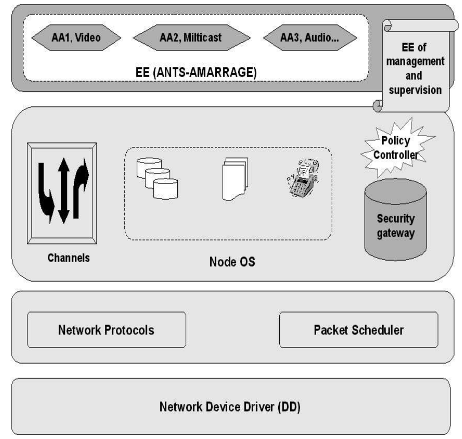

Fig. 7. The ANTS Active Router

V. EXPERIMENTATION

A. Test network setup

Our experience with the active networks is carried on the AMARRAGE platform, which is an active execution environment (EE) derived from the ANTS package [31], [32]. The latter is one of the pioneer, and the most popular, active network implementation. The ANTS-AMARRAGE platform is jointly conceived and developed by the AMARRAGE [2] project consortium which is a French RNRT project [25].

The active network node is divided among the Node Operating System (NodeOS), the Execution Environment (EE) and the Active Applications (AA) (see figure 7):

- The NodeOs is an extended Linux kernel. It performs: i) the demultiplexing of the active network packets (called capsules), ii) the encapsulated by using the Active Network Encapsulation Protocol (ANEP) [1], iii) and to multiple EEs located on the same network node. Suppressing UDP removes an indirection level in the active packets processing. Active services thus operate right above IP.
- Each node supports a Java-based EE, which is permanently available to AA developers. The virtual machine is fully compliant with IPv4 and IPv6 protocols. It provides a programming interface used to access to the Internet Protocol network services.
- Our experience with the active networks, is done by using an active execution environment (EE) derived from the ANTS package [31], [32]. The initial architecture is extended to support the specifics additional needs of significant active applications such as video, videoconference, etc. Those additional functions are secure deployment, configuration, and control of AN software.

The video traffic is a CIF full rate (25 frames/s) H.263+ sequence. The CODEC we use is a special version developed by France Telecom R&D for the purpose of our active regulation mechanism. For all of our experiments we use the test video Foreman, with a higher degree of motion.

B. Video quality measurements

In figure 8, we evaluate the performance of the proposed framework. Results are illustrated at CIF resolution using 6 seconds of video at 25 fps for the sequence Foreman and packet loss rates of 4%. The peak signal-to-noise ratio (PSNR) is used as an objective image quality measure, and is defined by $10\frac{\log 255^2}{\sigma_e^2}$, where σ_e^2 is the mean-square error between the processed and original luminance images.

First, we investigate the loss prediction performance of our ARM mechanism. In this test, the active control and rate regulation are performed using the original frames of the Foreman sequence. The left of figure 8 shows the original 16th, 40th and 76th frames. The middle of this images shows the correction done with the RTCP protocol. Finally, the right images show the proposed framework. It can be seen that the ARM mechanism anticipates the loss of packets appearing with the existing correction mechanism (RTCP), (figure 8 (b)). The following table (table II) shows that the ARM has succeed to maintain a limited variation of quality of image before, inside and after the congestion period. In contrast to this approach, RTCP can not avoid the loss which decrease the quality of perceptive image (16.87dB).

C. Fast reaction to congestion

Our experiments show one major advantage of our active regulation mechanism, which is the fact that the active router can adapt the video traffic to the currently available bandwidth in a responsive manner, with a

TABLE II
COMPARISON OF THE PSNR VALUE OF OUR PROPOSED FRAMEWORK WITH THE RTCP

Congestion	ARM	RTCP
Before	22.6330	21.0545
Inside	19.8261	16.8706
After	21.4576	20.7903

whole vision of the link. With our controlling framework, we are able to react faster than RTCP (figure 8 (b) and (c)) with richer information obtained from dynamic measures.

The main improvements of this framework related to the classical RTCP scheme, are the follows:

- It gives a complete vision of the whole channel, in a user-specific manner. The user can actually choose the information by put the adequate code into active report capsule (ARC) and by installing the adequate measurement agent at the adequate locations (ARD,ARI).
- It allows a feedback rate more adequate and specific to the application, and with a lower overhead (we report only information really pertinent to the rate-control). In fact, the frequency and the choice of information are under the sole control of the application. In our experiments, the ARC traffic is about 0.06 % of the total bandwidth of a session.
- It allows the dynamic installation (and modification) of user-specific rate-control algorithm (user-agent at ARE).

Thus, we can virtually get any kind of information at a desired frequency, and we can perform the most efficient regulation with an adequate algorithm [24].

We want to point out that our active-network based mechanism offers a schema to some other active-network solutions, such as the proposition describing in [17] which is based on frame omitting and/or filtering or in [3], [18] which is based on real time multimedia Transcoding.

VI. CONCLUSION

This article has shown a new mechanism of real time transmission control of video communication through the Internet. The originality of this framework lies in the use of the technology of active network, in order to conceive a new approach of reactive control, made to adapt the video flow to the variations of resources in the Internet. It suggests the importance of this technology as well as the given possibilities to introduce innovating services and concepts.

The basic idea of our mechanism is to introduce, in the network, capacities of treatment and control able to solve the transmission problems of real time video data, and a better adopted control service taking into consideration the state of the network and the requirement of the receiver. The network, according to our approach, supervises the transmission of video packets and reacts to flow variations by sending to the video encoder a recommendation of the available bandwidth in the network for its flow. Against all the other adjustment techniques set these days, it is the network that directly adopts the video source throughput.

We validated our solution experimentally through an implementation on the active platform AMARRAGE [2]. The algorithm of regulation used to adopt the real time video transmission to available resources in the network allowed a receiver which link is congestioned, to receive an adequate quality. In fact, the instruction of the active network to decrease the throughput of this receiver has given it a service continuity with a considerable quality drop more or less acceptable. A compromise has therefore been chosen between a correct reduction of the quality of perceptive image or an unacceptable momentarily degradation (figure 9).

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(a)

(b)

(c)

Fig. 8. Comparison of the ARM with the RTCP Best-Effort. (a)The 16th frame, in the left the original frame, the middle is the RTCP environment (21.05dB) and in the right the ARM (22.63dB). (b)The 40th frame. the proposed algorithm without loss (22.69dB): prediction of loss appear in the middle (RTCP 16.87dB). (c)The 76th frame, in the middle the RTCP correction (23.43dB) and in the right the ARM correction (30.55dB)

Fig. 9. Effect of ARM prevention (right) versus degradation under RTCP paradigm (left)

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