A statistical method to evaluate adaptive video applications based on subband coding over best effort networks, particularly with ABR services

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Abstract: With the proliferation of Networked Multimedia Systems, different resource management within the best effort service paradigm, capable of handling such systems efficiently are being tested. These environments requires the design of adaptive compression algorithms, which must adapt to the changing network conditions. Usually this adaptive process tries to manage the multiresolution concept, based on subband coding and the Rate-Distortion Theory.

In this paper we present a statistical method to evaluate the performance of adaptability of such algorithms and an overview of its main mechanisms, which are responsible for this adaptation, specially in the case of video applications.

1 Introduction

The design of adaptive compression algorithms to support multimedia applications that can adapt to changing network conditions within the best effort paradigm, is currently being the subject of intense study. Most of these applications, because are aimed at human beings do not require a guarantee QoS and their performance can be degraded using perceptual criteria. But at least, these applications require a minimum output rate at connection set-up, which can be dimensioned to achieve a minimum perceptual quality or a threshold, which avoids to be just noticeable the degradation produced.

This resource management, can be supported by, RSVP (Resource Reservation Protocol) in Internet or by ABR (Available Bit Rate) services in ATM (Asynchronous Transfer Mode). It should be noticed that in reference [ADM96] is described the ATM and RSVP mapping of services, pointing out the best effort paradigm as a common service for both. These trends in best effort services are justified because end users are willing to pay less for just an acceptable quality, as well as their service providers make profit of the spare bandwidth.

In this situation, the compression algorithms must keep working properly although there were changing network conditions. At large, these algorithms are carried out by multiresolution, based on subband decomposition and using the Rate-Distortion Theory.

In this paper we focuses the study of evaluation for adaptive video compression algorithms based on subband coding over best effort networks. The main objective is to determine a statistical method to evaluate the performance of adaptability of such algorithms, according to the loss probability for the subbands that carry the relevant part of the information. Here after, we centre the study in ABR services over ATM networks.

The rest of the paper is structured as follow. In section 2 are described the features provided by an ABR connection for a video coder. In section 3 are explained the basic adaptive process of video coders. In section 4 is described the adaptation to the service provided by the network. In section 5 is shown the probabilistic study for video coders if based on subband coding. Finally, section 6 presents the conclusions and ideas for future work.

2 ABR connections

Video-based applications that are rate adaptive, can obtain substantial benefits by using ABR connections. These benefits can be summarised in the following three aspects. First, these applications typically require a minimum encoding rate for a video stream by defining a Minimum Cell Rate at connection set up. Second, when explicit rate feedback is used and the ABR connections supporting these applications are multiplexed on a dedicated queue at the switches, the cell transfer delay is more predictable because the congestion control mechanism keeps the queues almost empty. And third, the periodic feedback mechanism keeps each source informed of the available bandwidth it has at their disposal, which can be used by the algorithm to adapt quickly to new network conditions.

2.1 Estimation of Available Bandwidth

This feedback mechanism is conveyed by a special type of cells called Resource Management (RM) cells, which are inserted by the source in the normal data cell flow sent toward the destination terminal. Once the RM cells get to the destination terminal then they are sent back to the source, collecting on their way congestion state information supplied by the switches.

The transmission rate of an ABR source (in our case an adaptive video coder) is computed by taking into account both the information conveyed by the RM cells and a set of source behaviour rules [RJK96]. The rate at which an ABR source can transmit at any given time is called *Allowed Cell Rate* (ACR). The video coder must track the values taken by the ACR to estimate the bandwidth that a connection has available.

But the ACR changes are the order of hundreds of μs and cannot be used directly by the coder, which works at the frame time scale. Therefore it only requires to estimate the connection available bandwidth at its time scale by filtering the ACR values. One technique that is easy to implement is to perform an exponential weighted averaging of the ACR samples, that is $MACR = MACR + \alpha(ACR - MACR)$, where MACR (Mean ACR) is the estimated available bandwidth and α determines the cut-off frequency (f_c)of the low pass filter. This cut-off frequency is normalized by the sampling frequency. The normalized cut-off frequency of MACR expression is given by:

$$\cos w_c = \frac{\alpha^2 + 2\alpha - 2}{2(\alpha - 1)}$$

where $w_c = 2\pi f_c$. One of the problems to determine the value of α is that the sampling frequency is not constant. It depends on the inter-arrival times of the RM cells, which itself is a function of the available bandwidth that changes with time. A common trade-off value is $\alpha = \frac{1}{16}$.

The video coder uses the value of MACR as a forecast for the available bandwidth that the connection will have in its time scale, at the frame rate time scale or 25 frames per second.

3 Adaptive video compression algorithm

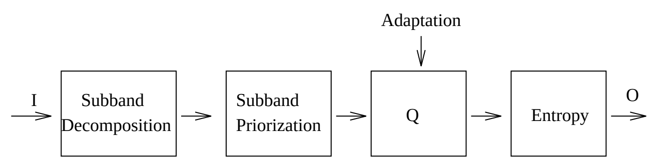

Figure 1: *Functional block diagram of a video coder based on subband coding*

A video compression algorithm is based on three steps: a decomposition process, a quantization process and finally a entropy coding process, as can be seen in figure 1. But if adaptive performance is required, each process requires a suitable design.

3.1 Subband coding and subband priorities

Adaptability means multiresolution, and multiresolution can be implemented using a subband decomposition or subband coding. Our study refers to video coders based on subband coding.

A subband decomposition is a process where the information is decomposed in subbands. Each subband has different resolution level of the original video, in a way that if we add all subbands in a reverse decomposition process, we obtain the original video. Several methods to implement a subband coding can be found in the literature [SN96].

Obviously, depending on the video information assigned to each subband, not all subband have same importance from the human visual system point of view, because human has different responses to these subbands. Then, this perceptual priority determines the order in which subband are going to be transmitted.

3.2 Rate-Distortion Theory

Once the decomposition process has been carried out, we must adapt the number of bits per subband to the available resources provided by the network, implementing a bit allocation algorithm. This establishes the quantization step using the Rate-Distortion Theory[SN96]. That is to minimize the total distortion introduced after the quantification process of all subbands, using Lagrange factors, once given the available bandwidth forecast or rate (in bits per pixel).

4 Adaptation to the Service Provided by the Network

Once the information generated by each subband is available, the coder creates an information unit per subband; each information unit must contain information to reconstruct each individual subband. The information unit is called the *Service Specific Convergence Sublayer* Packet Data Unit (SSCS-PDU) in ATM jargon. This PDU is then transferred to the *Common Part Convergency Sublayer* (CPCS), where it is encapsulated in a CPCS- PDU and then segmented by the Segmentation and Reassemble (SAR) sublayer.

The order in which the different SSCS-PDUs get transferred to the CPCS will define the order in which the different subbands will be transmitted. This order is determined by their perceptual priority as said below. Also notice that when a new set of subbands arrive to the ATM layer ready for transmission, any stale information must be flushed to keep synchronization. This queue flushing mechanism will only deteriorate minimally the quality because it will only affect the low priority subbands.

5 Probabilistic study of subband transmission

Over different network configuration and congestion situation, if the video compression algorithm based on subband decomposition must be adaptive, this means that video coders must maintain continuously *subjective quality constant* by minimizing the loss probability for the subbands that carry the relevant part of the information.

On the one hand, the probability of transmission of each subband can be considered an random variable, related to the network congestion situation and the intrinsic video sequence characteristics. This random variable has a Bernuilli distribution with $p(T)$ and $1 - p(T)$ as transmission or not probability[Jai92]. Obviously this decision it is taken by the adaptive algorithm according this information.

On the other hand, the subbands to be transmitted, are the ones on which we are interested in. In the transmission, two things can occur, or a correct arrival with a $p(A)$ probability, or an incorrect arrival with $p(E)$ probability, where $p(T) = p(AT) + p(ET)$.

Nevertheless, the more relevant parameter for the adaptability process is the conditioned probability for each subband, of a right arrival when that subband had been transmitted $p(A/T) = p(AT)/p(T)$.

This conditioned probability shows different aspects from the video compression algorithm:

- if the algorithm is adaptive enough and it is based on a good forecast mechanism, actually if a subband is determined to be transmitted by the algorithm, it should arrive correctly

- by pryorizing the transmission of the subbands according to a perceptual criteria and by observing the ABR source behaviour rules, the more relevant subbands should keep a high $p(A/T)$

- the changing network conditions should not modify $p(A/T)$, which only depends on the performance of the adaptive algorithm

Because we are working with a closed number of trials, a set of samples within the sample space (determined by all possible combinations for a transmission) should give us enough information to decide how adaptive a video compression algorithm is. To do that we need a more reliable information based on the confidence interval for each $\hat{p}(A/T)$.

The confidence interval determines the interval within the approximated $\hat{p}(A/T)$ value is valid (right) with a $1-\alpha$ per cent of probability. The confidence interval for $\hat{p}(A/T)$ is determined by the number of available samples and the variance of the $p(A/T)$ samples. It is given by next expression[Run96] for a estimated $\hat{p}$:

$$\left[\hat{p} - N_{Z_{\alpha/2}}\frac{\sqrt{\hat{p}(1-\hat{p})}}{N_t}, \hat{p} + N_{Z_{\alpha/2}}\frac{\sqrt{\hat{p}(1-\hat{p})}}{N_t}\right] \quad (1)$$

where N_t is the number of available samples or the number of times each subband has been transmitted. Usually $1-\alpha$ is 90%, which means $N_{Z_{\alpha/2}} = 1.96$. Finally just to use 1 equation, some requirements must be kept, that is $\hat{p}\ N_t > 5$ condition[Run96].

5.1 Sample Space

Several network configurations may be used to carry out this study. As much bigger the number of trials is, more reliable the $\hat{p}(A/T)$ value will be. Obviously more reliable $\hat{p}(A/T)$ values will be given by estimation with a little confidence interval. Then not only network configuration should be used in the experiment, but different patterns of traffic in the connection, different number of ABR sources, different distances, different video sequences and so.

6 Conclusions

This paper shows that the successful transmission of video over connections that only offer a *best-effort* service may be measured by means of a probabilistic study.

A proposed video coder must maintain the *subjective quality constant* by minimizing the loss probability for the subbands that carry the relevant part of the information, according to any criteria, in this case for video given by a human visual system. This must be achieved by prioryzing the transmission of the subbands according to the perceptual criteria and by observing the ABR source behaviour rules.

Notice that the ABR service guarantees that sources which obey the feedback signal supplied by the network will achieve low losses. This is an interesting point of ABR services from the adaptive application point of view.

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