Scientia Iranica, Vol. 6, No. 1, pp 51-58

SCIENTIA
IRANICA

Research Note

Transmission of Pre-Encoded MPEG Video over ATM Networks and VoD Application

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This paper proposes the use of a video transcoder for improving transmission efficiency of MPEG bit streams. Video services using recorded video in compressed format can benefit a great deal if dynamic control of compressed bit streams during transmission is provided. Since the input of the video transcoder is a standard MPEG bit stream, like the output, its operation at different locations along the transmission path is possible. In Video on Demand applications (VoD), by transcoding coded videos into lower rates, either the user should demand lower video quality or channels with reduced capacity can be supported. The picture quality obtained after transcoding is nearly the same as if the original video was encoded at the same bit rate. For ATM (Asynchronous Transfer Mode) networks, it is shown that unconstrained transmission of compressed bit streams over such networks can lead to severe degradation in picture quality due to cell loss, whereas use of the transcoder as a traffic controller reduces the cell loss and, hence, pictures are degraded gracefully.

INTRODUCTION

Telecommunication infrastructures and services are both converging to the multimedia concept, in which video traffic plays a major role. At present, it is generally accepted by both the telecommunication industry and academic community that MPEG-2 [1] is the main standard for video coding. Furthermore, ATM is already defined as the transfer mode for the broadband integrated services digital networks (B-ISDN) which leads to the assumption that video services based on MPEG-2 will constitute the main traffic load in ATM networks.

The heterogeneity of present communication networks such as circuit switched, ATM, mobile and internet as well as the wide range of video services from very low bit rate under 64 Kbit/s to High Definition Television (HDTV) with the bit rate in excess of 20 Mbit/s, demonstrate that matching the source and network characteristics is a priority. If matching of source to channel cannot be maintained, then the quality of the video can degrade up to an unacceptable level. However, there are numerous scenarios where this matching cannot be guaranteed at its optimum. For example, traffic in the internet can vary enormously during the day, where in some cases continuous transport of video, even at its lowest possible quality becomes impossible. In ATM, the available channel rate to a service can instantaneously change under the influence of other bursty traffic. In the case of Video on Demand (VoD), even though there may not be any restrictions on the channel capacity, users may wish to receive video at the quality of their own choice.

In the past 5 to 7 years, various layered coding techniques have been devised to improve the quality of video services [2,3]. In these coding techniques, an image sequence is encoded into various levels with different degrees of importance where the most significant parts of the bit stream can be protected against poor channel behavior, or the user can use a portion of the bit stream according to his needs. Although these techniques can provide a minimum Quality of Service (QoS), they have several pitfalls. For example, the overall bit rate of a multilayer coder can be much higher than a single layer one [4]. In the case of VoD, the number of available layers can limit the user choice, or may not be accommodated in the future networks, such as video over mobile networks at very low bit rates.

In this paper, the use of a video transcoder is proposed as a solution for the problems described

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above involving compressed video, in particular, pre-encoded video. The video transcoder is based on the authors' previous work [5] adapted for the applications presented in this paper. Two applications are described where video transcoding is utilized to provide transmission efficiency; VoD and ATM networks. In the first application, prior to a set up call, the user demands the QoS (bit rate) of his needs, hence the video server always keeps a single copy of the video which can be distributed at any lower rate. In the second application, depending on the ATM traffic status, the bit rate at the congested node is reduced to ease congestion and hence, to improve the quality of the received pictures.

DYNAMIC TRANSCODING

A generic video transcoder for rate conversion should accept a standard compressed bit stream and produce another standard bit stream at a lower rate. Therefore, syntax compatibility at both input and output bit streams is mandatory. Moreover, pictures decoded from the converted bit stream should have nearly the same quality as if the original raw image sequence were encoded at the converted bit rate. Such a system should also introduce a very low delay in order to be capable of dynamically matching the compressed bit streams to any constrained channel. This is particularly important in video over ATM environment, where fast response of traffic shaper to network demand is highly desirable. Figure 1 depicts the generic environment where the video transcoder is employed. In this figure, the video source is considered to be any system transmitting pre-encoded video bit streams, for instance a video server or a workstation. Furthermore, no specific assumptions are made about the relation between input and output channels and the rate converter (A, B). They may accept constant or variable bit rate video. In addition, the transcoder can be either part of the video source or placed anywhere along the transmission path.

The amount of bandwidth reduction on the input bit stream is dictated by the command $0 < S(t) \leq 1$, which can be either a dynamic feedback signal (e.g. reactive congestion control) or a static signal for a constant reduction of the input bit rate at the chosen setup call. The input R_T^{in} and the output R_T^{out} rates are provided by the following equations:

$$R_T^{\text{in}} = R_C(t) + R_V^{\text{in}}(t) \ , \tag{1}$$

$$R_T^{\text{out}} = R_C(t) + S(t).R_V^{\text{in}}(t) \ , \tag{2}$$

where $R_C(t)$ and $R_V^{\text{in}}(t)$ are the non-reducible and variable parts of the input bit stream, respectively. Since the transcoder only acts on the transform coefficients of the bit stream and motion vectors remain intact, $R_C(t)$ comprises all the header information as well as that of motion vectors. $R_V^{\text{in}}(t)$ are the bits assigned to the transform coefficients of the bit stream [6].

As emphasized earlier, the main issue that must be minimized in any transcoding process is the delay of the rate conversion process. In the case of MPEG-2 coders, delay is due to processing, frame reordering and output buffering including packetization. The processing delay can be reduced to a negligible value, at least in theory, since it only depends on the hardware which can always be made faster. Therefore, delays due to buffering and frame reordering are dominant in magnitude compared to other latencies such as block-based operations and data reordering.

The Video Transcoder

In a recent work, it has been shown that open loop techniques, such as requantization of DCT coefficients, can introduce a considerable amount of distortion into decoded pictures because of error accumulation in the decoder [5]. Therefore, the scheme of Figure 2 is proposed as a video transcoder for networking applications. The input bit stream is converted into a lower rate by using a coarser quantizer for generating the output bit stream and the new quantizer scale is adjusted according to the user command $S(t)$, which provides the amount of required bit rate reduction.

As shown in Figure 2, the transcoding error due to coarser quantization is fed back and added to the next frame to compensate for the drift. This error is

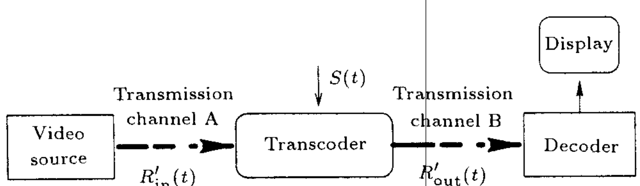

Figure 1. Generic video communication system.

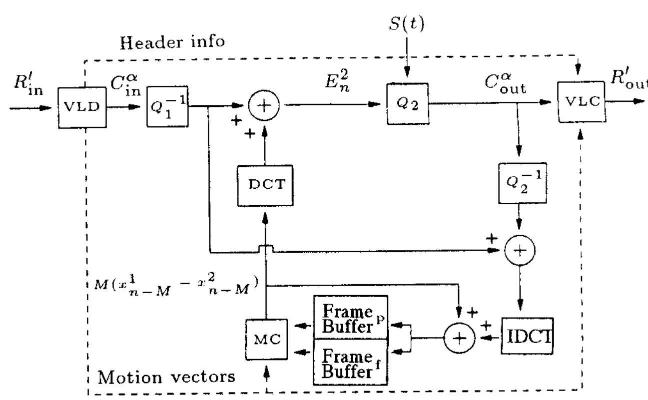

Figure 2. Video transcoder.

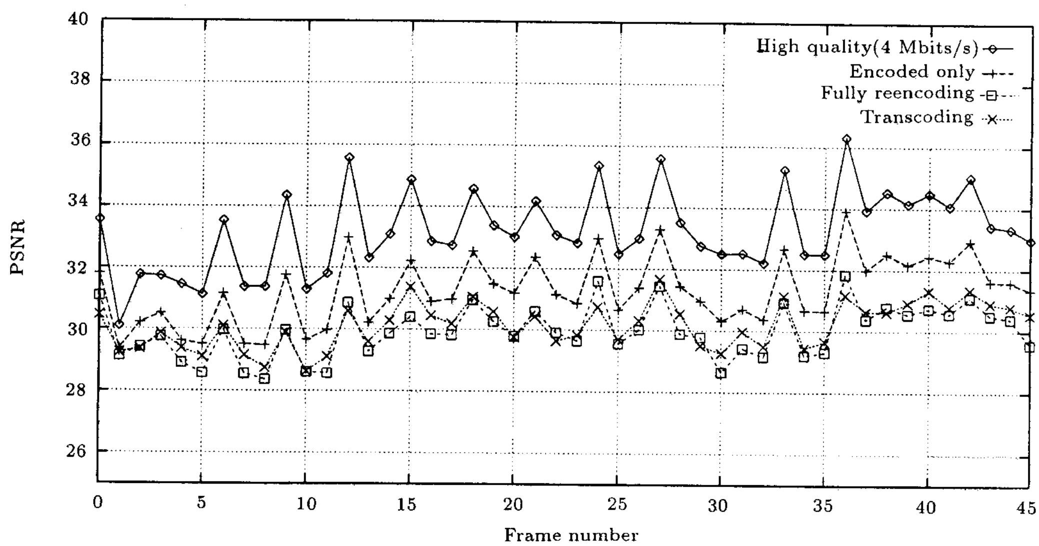

Figure 3. Conversion from 4 Mbit/s into 3 Mbit/s.

accumulated in the pixel domain due to motion compensation (MC), which is primarily defined as a pixel domain operation. Since B pictures use predictions from both the previous and next I or P pictures, two frame buffers are required to accumulate the error and prevent picture drift. However, taking into account that B pictures are not used as predictions, some additional distortions can be tolerated in these types of picture and the cost of the system can be reduced by using only one error buffer in the feedback loop. The drift through P frames, which are the important ones, can still be prevented while using only one buffer in the feedback loop. The I frames reset the error accumulation and do not require corrections. A more complete description of this video transcoder and performance measures can be found in [7].

VIDEO ON DEMAND - MULTICASTING

In this section, the application of the video transcoder in video multicasting is studied. It is assumed that one single high quality bit stream is to be distributed by a video server into three classes of user, each assigned to a different channel bandwidth. These correspond to CBR transmission over channels with capacities of 4 Mbit/s, 3 Mbit/s and 2 Mbit/s. For simulation purpose, three Groups of Pictures (GoP) of the same test image sequence (MOBILE) with N = 15 and M = 3 are encoded. N is the GOP length and M is the distances between the anchor pictures [1]. The bit stream encoded at the highest quality (4 Mbit/s) was transcoded into 3 Mbit/s and 2 Mbit/s in order to meet the bandwidth constraints of the second and third class of users, respectively.

In order to compare the picture quality of transcoded sequences, two more tests are carried out each with 3 Mbit/s and 2 Mbit/s. In one test, the original image sequence was encoded by a standard MPEG-2 encoder at 3 Mbit/s and 2 Mbit/s. This is referred to as "encoded only" in the figures. In the next test, the 4 Mbit/s bit stream was decoded and re-encoded again at 3 Mbit/s and 2 Mbit/s for comparison, which is referred to as "re-encoded".

Figures 3 and 4 illustrate the peak signal to noise ratio (PSNR) for the two cases: conversion of the 4 Mbit/s bit stream into 3 Mbit/s and 2 Mbit/s. In both figures, the PSNR of the high quality bit stream (encoded at 4 Mbit/s) is shown for comparison purposes.

As the figures demonstrate, the picture quality of the transcoded bit streams in both cases is very close to that obtained from the bit streams directly encoded from the original video "encoded only" at the same reduced rates of 3 Mbit/s and 2 Mbit/s. The difference is less than 1 dB in almost all frames which is not noticeable in terms of perceptual quality. This is interesting since, as shown, it is not necessary to keep several copies of the same video stored in the video server to meet different user demands. By keeping one single high quality bit stream and using the proposed video transcoder for conversion at lower rates as required, it is possible to provide the user with the same quality of service as if the original video were directly encoded at the same rate.

Moreover, Figures 3 and 4 show that PSNR curves of transcoded bit streams and those fully re-encoded are almost identical. If the computational complexity of both systems is taken into account, then the transcoder is far less complex than a cascade of decoder-encoder. Since the transcoder uses the incoming motion vectors, there is no need for new motion estimation which gives major gains in computational complexity. Motion estimation is undoubtedly the most demanding cost function of the encoder [8].

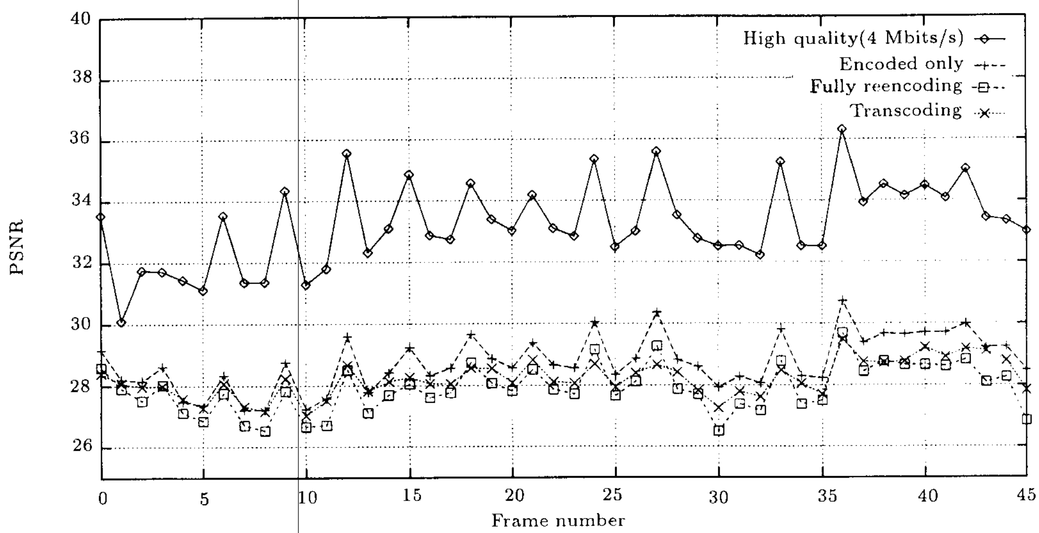

Figure 4. Conversion from 4 Mbit/s into 2 Mbit/s.

CONGESTION CONTROL

The traffic generated by VBR MPEG encoders possesses particular characteristics which have a major influence on transmission through ATM networks. Due to the different types of picture, I, P and B, the bit rate profile of MPEG streams follows the same pattern as the GoP structure which results in a strong periodic pattern along the sequence with a great deal of implications on ATM performance [9,10]. The periodic nature of the bit rate plays the most important role in multiplexing MPEG sources since the cross correlation between independent sources can be very high. If the GoP patterns of different bit streams happen to be coincident, then cell loss in the network can reach extremely high values [11]. Moreover, these losses will occur at peaks in bit rate produced by I and P frames, thus the errors introduced will propagate throughout the sequence.

In most of the simulation studies described in the literature, the multiplexing buffer of an ATM switch is modelled as a finite capacity queuing system with buffer size B (in cells) and service rate C. Such a model is illustrated in Figure 5. The queuing discipline is First-In-First-Out (FIFO) and the sources are pooled according to the arrival instant, i.e., the first cell to arrive is the first to enter in the buffer. The service rate of the multiplexer is adjusted in order to obtain the desired network load. Following the approach used in [12], the video traffic trace can be arranged as a circular list with a uniformly distributed random start for each source.

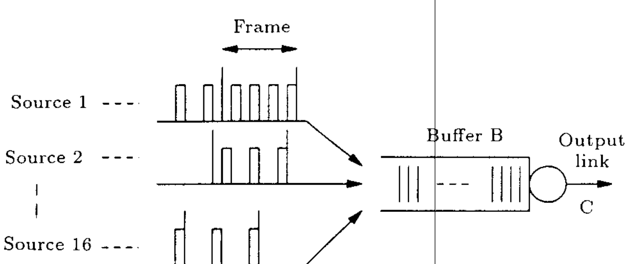

Figure 5. ATM multiplexer.

Traffic Control with Transcoders

Closed-loop control, implemented through Explicit Congestion Notification (ECN), has been used as an efficient mechanism to allow good network utilization while maintaining adequate QoS [13,14]. Two schemes are currently considered as adequate: Explicit Forward Congestion Indication (EFCI) and Explicit Backward Congestion Notification (EBCN) [15]. Both of them employ a particular type of Operation And Maintenance (OAM) cell, called ATM Resource Management (RM) cell, to convey network information about the congestion [16].

The EFCI scheme is an end-to-end flow control whereby information about the buffer occupancy or available bandwidth is sent to the destination and then back to the source. Under this scheme, the RM cells are sent forward through the network and depending on the "intelligence" of each switch, the information carried by these cells may refer to available resources in the switch, congestion levels and timing. When these cells arrive at the destination, they are sent back to the source instructing it to react according to the information received. Under the EBCN scheme, the switching nodes directly notify the source by sending the RM cell back to the source carrying information about the current status of the switch. In this case, the RM cell does not travel along the transmission path and, therefore, does not convey information about the most stringent bottleneck in the transmission path, but only about the switch where it came from.

Application of either EFCI or EBCN for video services, which have tight delay constraints, requires quick response of the video traffic generator in adjusting the transmission rate, since the effectiveness of reactive congestion control depends on the speed of the source response to the switch demand [17,18]. Adjusting the transmitted bit rate results in graceful degradation of decoded pictures which is far better than distortions imposed by lost cells of unconstrained transmission.

Here, pre-encoded VBR video is assumed to be stored in the video server and the delivered traffic is controlled by video transcoders. The simulation model used is depicted in Figure 6, where the network switch monitors the multiplexing buffer occupancy and sends a feedback signal accordingly.

The switching buffer occupancy (B_f) is sampled at the rate of T_f times per second, which corresponds to the transmission rate of the RM cells. Using T_f = 2.2 ms with SIF pictures at 25 Hz, the transcoder can receive one feedback signal in each slice period, simplifying the rate adjusting procedure. Note that in SIF pictures at 25 Hz, slice period is 1/18 of 40 ms frame duration and MB period is 1/22 of that of the slice. The sampling period can be made as small as one MB time interval, which is the finest possible adjustment for the quantizer step size. However, at higher sampling rates, the RM cell rate may overload the feedback channel of the transmission network and more overhead is inserted in the transcoded bit stream. Alternatively, the rate of RM cells can be determined by the number of incoming source cells as defined for the available bit rate (ABR) service [15], e.g., one RM cell for every 32 data cells.

In this model, the buffer is implemented with two pre-defined thresholds, T_L, T_H, such that at the sampling instance: a negative feedback signal (NEG) is sent to the transcoder if the buffer occupancy is above the maximum threshold (T_H), a null signal (NULL) is sent if it is between T_H and the minimum threshold T_L, and a positive signal (POS) when it is below T_L. The transcoder increments its current quantizer scale when a NEG signal is received, takes no action under the NULL signal and the POS makes it to decrement the current quantizer scale. The RM cells that carry this feedback signal may reach the transcoder after a certain delay due to line propagation and reverse queuing delay.

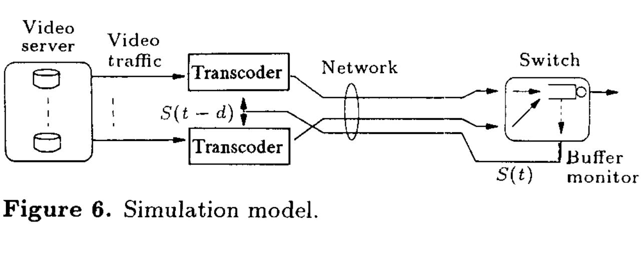

Figure 6. Simulation model.

Simulation Results

Two types of experiment were carried out in order to evaluate: i) the cell loss performance and, ii) the picture quality. While the former is useful to evaluate the effects of feedback delay on the response of the feedback system, the latter shows how much a video sequence can be degraded under network congestion in the absence of transcoders. In this simulation, a practical buffer size of B = 100 ATM cells was used with fixed thresholds at $T_H = 2B/3$ and $T_L = B/3$. Two image sequences of 30 seconds were used to generate the VBR traffic encoded with an average rate of 3.9 Mbit/s, corresponding to the input bit rate of the transcoders and the highest bit rate delivered to the network. It is assumed that the VBR video bit stream is smoothed over one picture.

Cell Loss Performance

Figure 7 shows cell loss ratio (CLR) of the unconstrained versus constrained transmission using transcoders at network feedback delays of 0, 5, 10, 40 milliseconds. For delays of about 40 milliseconds and above, it was found that the feedback system has no influence on the CLR suffered by the transmitted stream. This is because the buffer size is not large enough to accommodate the arrival of unexpected cells due to latency in the feedback path and the high burstiness of the video traffic at the frame level. Larger buffer sizes would stand laıger delays in the feedback path although a larger delay is introduced in forward transmission. Therefore, network routes via satellites, for instance, which may cause delays in excess of 250 milliseconds, require larger buffers in order to make the feedback control system more efficient. Note that, because of the small buffer used, congestion occurs at the frame level which is mainly due to the coding algorithm. To a certain extent, the traffic burstiness at frame level could be reduced by using large smoothing buffers at the source, e.g., smoothing the traffic over

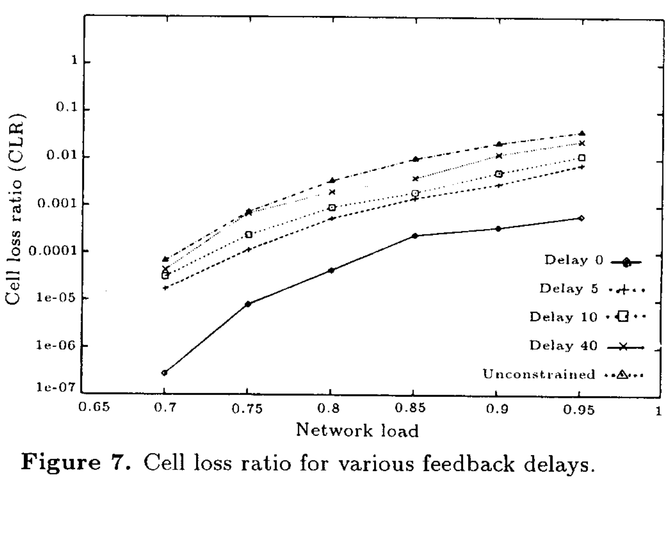

Figure 7. Cell loss ratio for various feedback delays.

one GoP. However, a much greater delay would be introduced as well.

Picture Quality

The picture quality obtained after decoding an unconstrained bit stream and a transcoded one is demonstrated in Figure 8. The unconstrained transmission at a network load of about 0.75 produces a CLR = 10^{-3} whereas, if the bit rate is reduced such that the load drops to 0.55, then the CLR will be practically zero (in the experiments, no cells were lost at a network load of 0.55). The network load is defined as the ratio of the aggregate traffic offered to the multiplexer of Figure 5 to its service rate. The decoder implements a simple concealment method where the lost MBs in each picture are simply copied from the last decoded reference picture.

Therefore, if the bit rate of the video sources is reduced by 20%, no errors due to cell loss will occur at the expense of a lower network utilization. As Figure 8 shows, if the bit rate is transcoded before entering into the network then, on average, the picture quality drops about 3 dB and no cells are lost, whereas for unconstrained transmission, the quality has several negative peak drops of the order of 15-20 dB due to cell loss. Since large fluctuations in picture quality are much more annoying than a poorer constant picture quality, the use of the transcoder to reduce the bit rate is a better solution than allowing cell loss in the network [19].

It should be noted that the relation between cell loss and perceived image quality is not always straightforward. Picture quality, when cell loss occurs, depends on the information carried by the lost cells and not necessarily on the cell loss rate. The problem is even more complicated in the case of MPEG coded pictures. Cell loss in B pictures may have no noticeable effect on picture quality, and losses in P pictures are still less disturbing than in I ones. However, a drop of 15-20 dB, as shown in Figure 8 for unconstrained transmission, is significant enough to be noticed in all types of pictures.

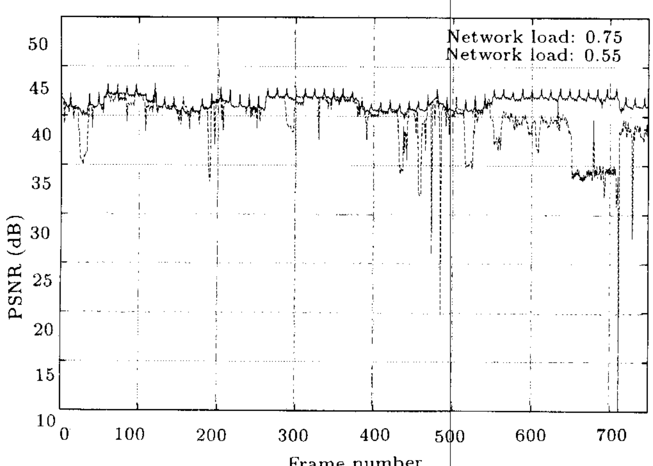

Figure 8. Unconstrained transmission (CLR = 10^{-3} and CLR = 0).

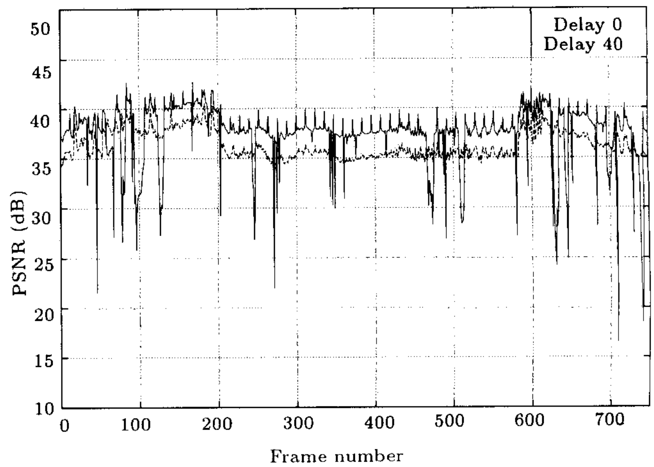

Figure 9. PSNR of a 30 sec. sequence, feedback delay 0 ms and 40 ms.

In Figure 9, the effect of feedback delay on the transcoder performance is demonstrated for another sequence at a network load of 85%. Here, the effect of the 40 ms delay is compared against that of the immediate response. As shown in the figure, when the transcoder reacts immediately to the network demand, there are no lost cells and the picture quality remains roughly constant along the sequence. On the other hand, with a feedback delay of 40 ms, cell loss is unavoidable leading to abrupt changes in picture quality because when the transcoder reacts to the network feedback, it is already too late to avoid cell loss. These results suggest that in networks with large delays, some switches may have local transcoders in order to exert effective control over the video traffic.

Dynamic Transcoding Switch

In the model of Figure 6, it is assumed that: i) all video sources converge on the same output link of the ATM switch and, ii) each bit stream is allocated its own transcoder for the duration of the connection. Although this is valid in a number of realistic scenarios, a rather different situation may arise when the video server is capable of accepting hundreds or thousands of simultaneous connections routed through different switches [20]. In this case, assigning one transcoder to each connection does not seem a practical solution, especially if one takes into account that congestion is not likely to occur more than a few times in the lifetime of each connection. This would result in a very poor transcoding utilization factor U_T which can be defined as:

$$U_T = \frac{T_a}{T_b + T_a} , \tag{3}$$

where T_a is the time during which the transcoder is active and T_b the time during which transcoding is not required and, hence, the source bit rate is the same as the output bit rate, i.e., the transcoder is acting like a by-pass.

A possible approach to increase the transcoding utilization factor and, hence, reduce the number of transcoders required for a video server is to use a special architecture named here as dynamic transcoding switch. A generic diagram of such an architecture is depicted in Figure 10, where the H-dec blocks are header decoders which decode and store the overhead information of the video sequence in local buffers. (It is necessary to keep the information updated from the beginning of the transmission.) In MPEG-2 bit streams, this comprises the sequence header and optionally the Group of Pictures (GoP) header. This is because transcoding may not start at the beginning of the bit stream but somewhere in the middle, hence, the information contained in the sequence header has to be given to the transcoder. For instance, the quantization matrices are defined in the sequence header which, in turn, may be repeated several times in the same bit stream. On the other hand, a GoP carries information about whether the first B pictures of the GoP are dependent on previous reference pictures or can be decoded properly just after the first I picture (in this case only backward prediction is used). For random decoding, if interpolative coding were used in these pictures, decoding would not be possible because one of the references may not be available. In this case, open loop transcoding can be chosen for these pictures without significant loss of performance.

For a group of N video inputs, there are only N-M transcoders available, thus, each transcoder is shared by more than one bit stream using time multiplexing. Providing that not all bit streams are being transmitted through congested nodes, the number of transcoders needed is much less than N. Those sources that do not need a transcoder are just by-passed, through the transcoding switch, directly to the network. If during transmission any particular source needs to be reduced in its bit rate, then it is switched through one of the available transcoders which also receives updated information from the respective H-dec decoder. This allows the transcoder to have all the necessary header information to start transcoding immediately. The philosophy behind such a scheme is similar to statistical multiplexing in ATM switches where the bandwidth allocated to a number of sources is less than the sum of their peak rates. This works well on average, providing that the peak rates do not occur at the same time. Similarly, in transcoding, it is assumed that not all connections are congested at the same time hence, on average, not all bit streams need to be simultaneously transcoded.

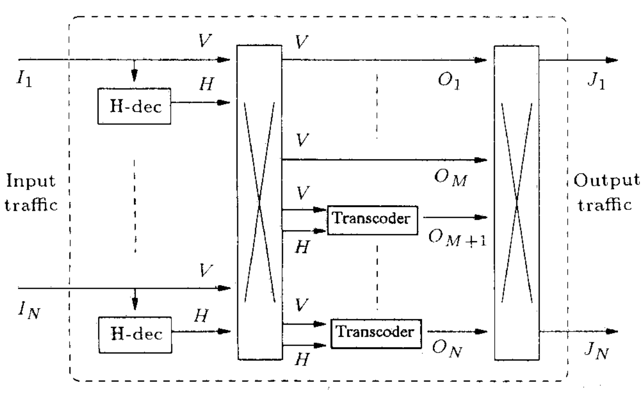

Figure 10. Architecture of a dynamic transcoding switch.

Two strategies can be adopted to switch on a transcoder: i) transcoding starts at the next I picture or, ii) transcoding starts immediately at the next picture regardless of its type. In the former, closed loop transcoding can be performed whereas in the latter open loop transcoding has to be used at least until the next I picture arrives. In this case, since open loop transcoding is used for less than one GoP, depending on the GoP size, the drift may not be significant. Moreover, the transcoder has to wait either for the next GoP or the next picture; thus, the worst delay case is one frame or one GoP, respectively. Although this would not be very efficient in systems which can only afford low feedback delay, such as the one described in the previous section, it is still very useful in other applications that can afford larger delays (see for instance [21]). In the very worst case, by starting transcoding at the next picture, cell loss is allowed to happen only during a maximum period of one picture which is not likely to be an I picture. Therefore, although switching delay limits the performance for some applications, the flexibility, reduced cost and complexity of a dynamic transcoding switch make it useful in video storage systems connected to ATM or heterogeneous networks.

CONCLUSION

Video transcoding was proposed for controlling the bit rate of compressed bit streams. This system decouples the video encoders from the characteristics of the transmission networks and provides efficient transmission of compressed bit streams. It is a low complex system capable of reducing the bit rate of standard bit streams without drift, introducing low delay in both transmission and response to network demand.

It has been shown, in VoD services, that only one high quality bit stream of each video sequence needs to be stored in the server even when transmission at lower rates is required. By transcoding the high quality bit stream accordingly, the transcoded picture quality is, in practical terms, similar to the original video frames if they were encoded at the same bit rate. The simulation results also illustrate that a re-encoding system, which is much more complex than the transcoder, produces nearly the same picture quality as the transcoder.

Furthermore, the application of the transcoder was simulated in reactive congestion control of VBR MPEG-2 coded streams under the EBCN scheme. It was shown how picture quality can be improved using the transcoder as congestion controller, taking into account the feedback delay. The performance of the reactive control system drops to values close to unconstrained transmission of the same bit stream when feedback delay is larger than the frame period, unless larger buffers are used. In case of either unconstrained transmission or large feedback delays, severe degradation is introduced in decoded pictures, whereas by using the transcoder as a bit rate controller, just a small amount of graceful degradation is introduced.

ACKNOWLEDGEMENTS

Pedro A.A. Assunção would like to acknowledge the financial support of Instituto Politécnico de Leiria and JNICT-Programa PRAXIS XXI, Portugal.

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