Rate Control in an MPEG-2 Video Rate Transcoder For Trasport Feedback based Quality-Rate Tradeoff

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Abstract

In this paper we present result from an application integrated congestion control mechanism designed for time-sensitive traffic based on the principle of direct protocol interactivity. In contrast to classical transport protocols we envision a transport mechanism, which is interactive and can provide event notification to the subscriber of its communication service. We then show a friendly adaptive MPEG-2 video transcoding scheme, which directly interacts with the transport protocol and adjusts its production in synch with the impairment events in the transport layer. In this paper we present the application side symbiotic mechanics, and report potential dramatic improvement in time-bounded video delivery. The system can be implemented with relative simple and direct modification of the current TCP transport.

Key Words: *netcentric applications, rate transcoding, video transport protocol, temporal QoS.*

1. Introduction

Congestion is one of the most actively researched areas in networking. However, the mainstream schemes focus on adjusting the delay-bandwidth product of communication and they work fully inside network. Application packets are delayed either in routers or at the network entry-point to cope with occasional congestions. For example, form the point of view of applications a TCP windowing mechanism acts as a network gatekeeper [BrOP94, Jaco88, AlPa99, Tene96]. It eventually performs some form of traffic shaping and introduces time distortion. Such distortion in temporal dimension is considered to be harmless to normal traffic. However, this is not always the case with time sensitive traffic.

Indeed, for time sensitive traffic a congestion control scheme based on delaying traffic in many cases may mean a mere shift in the point of packet discard. The situation of a time sensitive application packet in need of transport can be clarified with an analogy to a patient in need of an ambulance. Instead, of being dropped inside the network, in the classical TCP scheme packets are waited at the entry buffer, when the link is congested. A time sensitive packet (such as an audio or video) is often in effect rendered useless at the source. It resembles a situation where the paramedics draw satisfaction from the fact that the patient is not dying in their ambulance, although the packet dies right at the TCP entry buffer waiting for the transport. TCP window buffer spreads a backlog in time. To make the matter worse, the ambulance however returns at some later point in time and picks up the delayed traffic. Effectively this is non usable from application point of view. For time sensitive communication, it not only spells doom for the current data but for packets those follow. Clearly, one of the critical problems in provisioning an integrated solution is that in the current arrangement the applications are not at all being notified of the congestion or of any other network impairment. Rather applications are put to sleep by the network/operating system process. Many of the time sensitive video system therefore avoid TCP and prefer to use raw UDP. Unfortunately, the random packet loss in UDP under congestion can create equally adverse effect. Experiment has shown that if about 10-20% of the UDP packets are randomly lost at congestion, then most video streams become effectively unusable. Because in reality all real video transport packets contain many important header fields with deep inter packet dependency. It hardly improves the situation if an application if it is lost inside network or at the TCP/UDP sending buffer. Any congestion control scheme based on the principle of time distortion can result in similar problem.

In this paper, we show a new friendly adaptive MPEG-2 video transcoding scheme which is congestion adaptive. The interesting aspect of the scheme is that it directly interacts with the transport protocol and adjusts its production in synch with the impairment events in the transport layer. This feedback allows the system to tradeoff spatial quality of video with temporal quality. Since, this is done in the application level with deep application level knowledge the results are much more tolerable from the final application performance point of view. In this paper we present the application side symbiotic mechanics, and report potential dramatic improvement in time-bounded video delivery. The overall scheme is network wise very simple and yet effective. The effectiveness is derived from the clever synchronization of the multimedia rate control mechanism of MPEG-2.

1.1. Related Works

Congestion control for time-sensitive traffic is a difficult problem. Most of the classical strategies are based on delaying traffic at various network points. The schemes vary from simple packet dropping in network, to admission control (delaying at network egress points), to graceful delaying by prioritization. For last few years it has been felt that applications have to be more integrated in the solution. Particularly promising are the research in the new TCP friendly paradigm [KeWi00, ReHE00, SiWo98, PrCN00]. [SiWo98] presented a TCP rate-based pacing mechanism that particularly takes note of document transfer characteristics. [ReHE00] discussed a general framework where applications can control rates based on their end-to-end measurements (similar end-to-end technique is used in RealPlayer). There are also fully application level proposals. Due to the lack of convenient means to sense network states several works suggested [BrGM99, Wolf97] sending multilevel redundant information for video. Also several other works investigated combining application specific information from several streams into one clearinghouse architectures for aggregated congestion control. For example, recently proposed Congestion Manager [ABCS00, BaRS99] is a system layer component. It provisions aggregated congestion control when multiple streams from the same end-point attempt to send via a separate program called Congestion Manager. Notably, in the process, it also proposed mechanism for application to be aware of the congestion states mediated by the congestion manager. [SiWo98] proposed building TCP friendly application where application relies on real-time transport protocol (RTP) mediated end-to-end measurement. [PrCN00] used multiple probing mechanics for aggregate congestion control.

While there has been several promising work in network or system level to increase TCP friendly-ness, relatively very few work exists that seriously looked into the corresponding issues that arise in an actual time-sensitive application while taking advantage of the suggested ‘friendliness’. Notably, the paradigm fundamentally shifts a major part of the congestion management responsibility to the applications. Time sensitive applications themselves have substantial complexity in adapting. Rate adaptation for any advanced application in general is quite complex. It requires sophisticated layer 4+ techniques. Unlike the network layer only paradigm of congestion management,

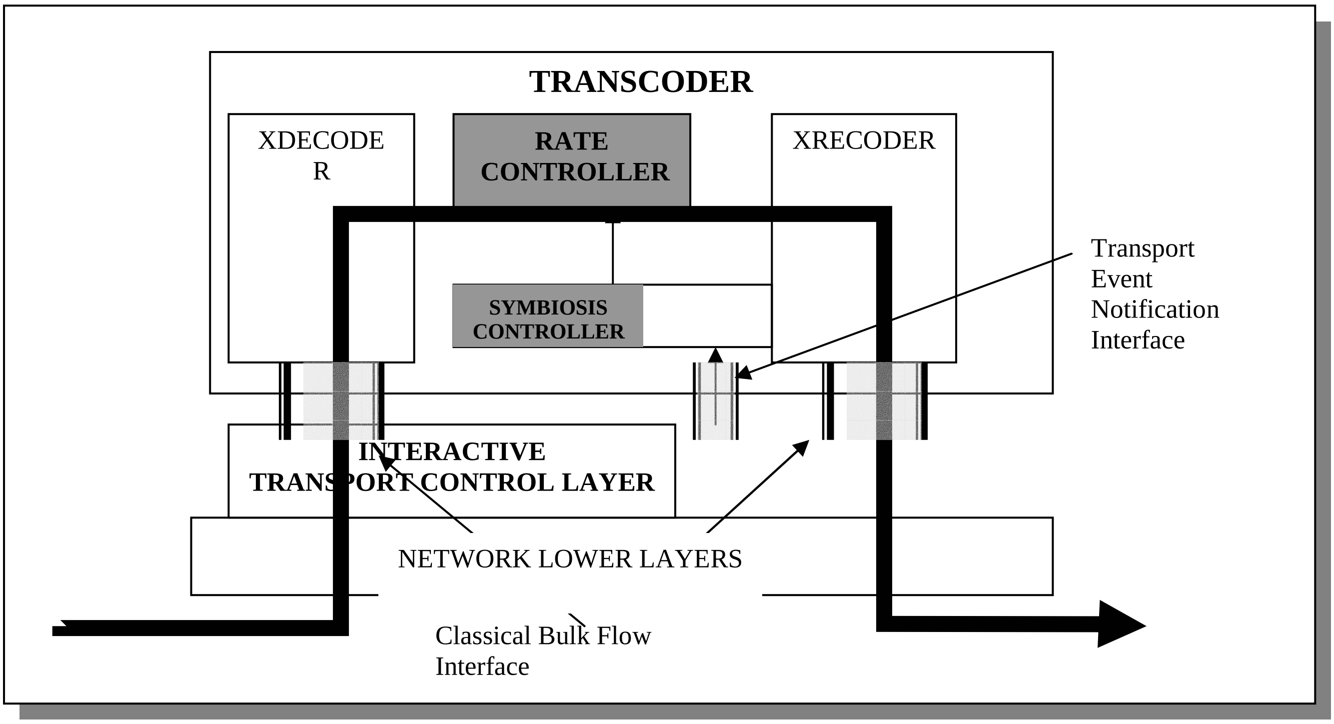

Fig-1(a) Interactive transport system and smart transcoder

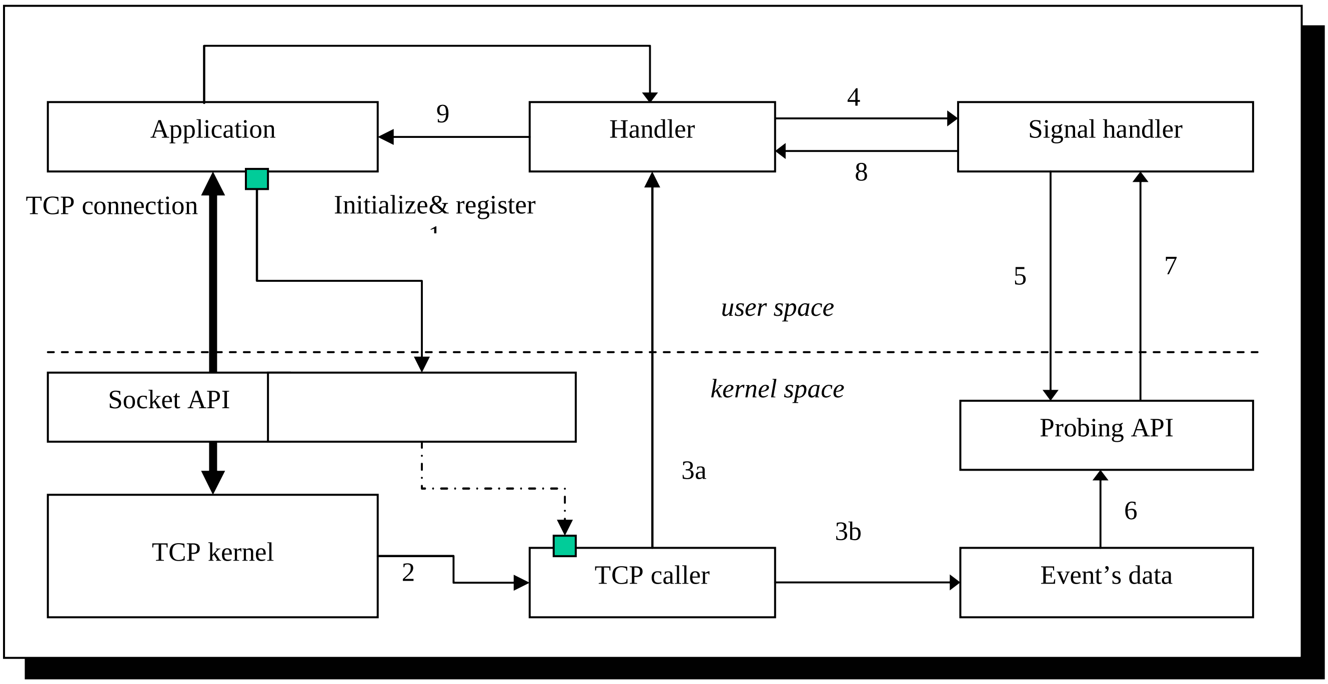

Fig-1(b) the TCP interactive extension. The added registration API allows demanding applications to subscribe to events by 1. When a timeout event is detected the kernel initiates 2. 3A via TCP caller then invokes the handler. Optionally, applications can probe additional event data via signal handler/ and additional API via calls 4,5,6,7,8 and 9. If event data is subscribed then 3B occurs concurrently with 3a.

a key research problem in this paradigm is the design of such rate adaptation techniques.

In this paper, we focus precisely here and present the symbiosis mechanics of a MPEG-2 ISO-13818-2 [ISO96] video streaming system [KYGP01, KhYa01] that we have implemented. We demonstrate a full application domain congestive rate adaptation and the interface mechanics to an interactive transport layer. We then share some interesting performance results.

The general principle we follow is simple and intuitive. It seems an effective delay conformant solution for time sensitive traffic may be built if the original data volume can be reduced by its originator-- the application. To our knowledge this is one the first performance report of MPEG-2 video rate transcoding system, which exploits direct feedback.

The particular scheme we propose here has several novel aspects compared to other recent works. First it depends on an active and direct notification mechanism by the underlying transport protocol, rather than indirect end-to-end feedback. If there is any congestion, we propose an interactive transport protocol, which can directly notify the application.

Secondly, we have designed a transcoder system rather than an encoder. This transcoder actively participates in a lazy symbiotic *exponential-back-off and additive-increase* like scheme [PeDa00]. (This is also one of the first to our knowledge). The advantage of this design is that it isolates the video server operation from the congestion management. It has been designed to sit either at the transport entry-point and perform conventional end-to-end paradigm based video transport like a conventional encoder. Or, it can also sit inside a network using technology such as active net or multi-protocol label switching for targeted and localized congestion management. The transcoding mechanism observes the local transport layer characteristics and can accordingly adjust the outgoing MPEG-2 stream bit-rate. Such a configuration can facilitate video communication between network segments with widely different bandwidth. However, a transcoding system is computationally more challenging than a conventional encoder. It is required to match the frame rate, and there should be much faster than typical encoding. The advantage is that it subsumes the functionality of encoder based system.

In this paper, in the next section, we first provide the system overview. In section 3 we then present the symbiotic rate control mechanism-- the key application component that provides the key network aware solution. The model has been developed by closely following the MPEG-2 Test Model 5 (TM5). MPEG-2 TM-5 signifies a real video coder with substantial complexity of itself. While the detail can be found in [Mpeg00], in this paper we describe the salient part of the rate control architecture that is critical to this symbiosis. Finally, in section 4 we share performance of the scheme. The results to be presented has been obtained using a real implementation of the symbiotic Transcoder [KYGP01, KhYa01], and letting it run on a simulated version of the proposed TCP interactive. With one of the first real and direct MPEG-2 symbiosis (to our knowledge), we report a potential dramatic improvement such a symbiotic scheme can offer.

2. Rate Adaptive Transcoder

2.1. System Configuration:

We have developed a three-part system model-- *server, transcoder* and *the client*. The middle component *transcoder* [KPOY01, KHFH96] can be placed in a suitable network junction point, which intercepts the stream. This is slightly different from encoder-decoder *server-client* system model. This approach has several advantages as opposed to implementing the rate adaptation at the end-point (encoder). It subsumes the functionalities of server-client model. In addition, it allows rate adaptation on video stream that is already encoded and thus enables serving stored video at a dynamically selected rate. This decoupling also has the benefit that the transcoder can be made to auto sense local asymmetry in link capacities and can be dynamically deployed inside network for streaming. For example it can sit at a node splicing a fiber and a wireless network, and thus can downscale an incoming high-bandwidth video multicast stream for an outgoing low-capacity wireless links. Additionally, there also exists the possibility of bring down transcoding operation inside network by emerging technologies such as active networking [TSSW97, GuTe98]. Fig-1 explains the system components.

2.2. Transport Control:

The transcoder sits on top of the interactive transport control layer-- TCP Interactive. Unlike conventional TCP, this interactive transport layer, when there is an internal timer-out event, passes on the current window resize event to the subscriber layer. The interface is almost identical to the TCP classic, except, upon opening the socket, the application binds an interrupt handler routine to the designated socket event. When, the event occurs the TCP triggers the handler. The binding is optional. If application chooses not to bind any handler the system defaults to the silent mode identical to TCP classic.

The internal architecture of the TCP interactive is shown in Fig-1(b). The added registration API helps applications to subscribe to TCP events, in this case the timer out. We have added a simple extension to TCP kernel. The main unit is called TCP Caller unit. It is activated if an application subscribes. It keeps track of the TCP timeout event. More inquisitive applications can also probe into selected TCP states. When a timeout event is detected the kernel initiates 2. 3A via TCP caller then invokes the handler. Optionally, applications can probe additional event data via signal handler/ and additional API (4,5,6,7,8,9). If event data is subscribed then 3B occurs concurrently with 3a.

2.3. Transcoder Architecture:

The transcoder unit has a decoder, and a re-encoder[1]. The re-encoder has a feedback rate control mechanism, which is capable of working in two modes: *normal* mode and *frugal* mode. In frugal mode the rate can be controlled at frame level. The actual control signal to the rate controller is generated by an application unit called *symbiosis controller*. The symbiosis controller accepts input signal from the transport layer to realize the symbiosis. Below we describe the MPEG-2 transcoder rate and symbiosis control mechanisms that we have developed for this experiment.

3. Rate Control Mechanism

The rate control mechanism is illustrated in Fig-2. The complexity of the system arises from several reasons. Due to the *variable length coding* (VLC), it is not possible to predict the exact amount of bits that will be produced from a macro-block for a given choice of coding parameters. Secondly, the perceptual content and activity in a particular picture area also dictates the inherent amount of bits that may be required to encode it. Also the bit requirements per macro-block depends on the picture type (I, B or P) as well other subjective factors. The proposed mechanism is also a double-loop feedback control mechanism where the output bit-rate is continually sensed to determine overall piecewise constant rate, with appropriate accounting for variations in frame/picture type like TM-5. A second internal feedback loop further tracks the efficacy of key conversion factors/constants for additional stability.

The output bit-rate is controlled by the quantization-step. After motion estimation and compensation, the prediction errors for each 8x8 blocks are computed. These 64 pixel differences are then transformed into 64 DCT coefficients. Each of the DCT coefficients is however, quantized using a separate step, because the human visual system responds differently to distortion in various DCT coefficients. However, to control the overall output bit rate MPEG-2 in its linear quantization mode uses a scale factor called *mquant* to determine the actual quantization steps, which are applied on these DCT coefficients. The quantized output for intra-and non-intra frames are respectively given by:

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[1] It is in general full logic MPEG-2 transcoding is a daunting computational task particularly because of the encoding. However, a number of recent techniques (including ours) have been identified for accelerated transcoding. We note that primary encoding at encoder and the secondary encoding employed in the transcoder are not the same. In the re encoding many information are available adhoc. Several computations (such as motion vector estimation) can be bypassed-- significantly under cutting the transcoding cost [KPOY01, KHY01].

$$y = \frac{f(x, quant\_step) + .75 \times mquant}{2 \times mquant}$$

$$y = \frac{16 \times f(x, quant\_step)}{mquant}$$

Here x is the DCT coefficient, $y=f(x, quant\_step)$ is determined from ISO/IEC 13818-2 tables [ISO96]. As *mquant* increases, the effective quantization steps become larger, more information is lost, encoding requires lower bits, and also the quality of the picture degrades, and vice verse. To account for few of these factors, in the topmost level the value of *mquant* for each macroblock is calculated as a product of two primary factors (a) the *buffer fullness* and (b) the *macroblock activity*. The *mquant* for the jth frame is computed as a product of two parameters: $mquant_j = Q_j \times N\_act_j$. The final value of $mquant_j$ is coded either in the slice or in the macroblock header [ISO96]. The part that is relevant[2] for this experiment is the Q_j. It is a modulation parameter, that determines how the allocation of frame-bits itself is varied.

3.1. Feedback Quantization Mechanism:

The system has two modes of operation: *normal* mode and *frugal* mode. In normal mode, the objective of feedback system is to maintain the output bit rate at piece-wise per GOP (group-of-picture). In frugal mode, it moves into a variable-rate encoding mode with proper proportioning for frame types, and the macro-block activity however, without any carryover. The saving earned during the frugal mode, however, is stored and can be (optionally) carried over to the point where normal mode is resumed to attain overall target rate. The control mechanism maintains three virtual buffers for separately tracking the bits consumed by the I, B, and P frames. To encode a frame of type x, for each macroblock, first a quantity called buffer fullness d_j^x of its corresponding buffer is determined. This is then used to determine the modulation factor Q_j.

$$\mathrm{Q_j} = \left[\frac{31 \times e_j^x}{r} \right] \text{ where,} \qquad \ldots(1)$$

$$\mathrm{r} = \left\lfloor \frac{2 \times c(t)}{frame\_rate} + 0.5 \right\rfloor$$

Here, r is called *reaction parameter* and is estimated from the current overall bit rate goal c(t). The quantity e_j^x is the *effective buffer fullness* and is computed from *virtual buffer fullness* d_j^x. The notation refers to the jth macroblock inside x type frame. These quantities are determined as following:

$$e_j^x = d_j^x - d_0^x \cdot S(t) \text{, and} \qquad \ldots(2)$$

$$d_j^x = d_0^x + B_{j-1} - \frac{(j-1) \cdot T^x(t)}{mb\_count}$$

In normal mode the *effective buffer fullness* is given by the *virtual buffer fullness*, but during frugal mode it is decoupled from initial buffer fullness, and is only estimated based on the frugal state target bit rate. A value of 1 to the state function *S(t)* moves the system to the frugal state, and zero to normal state. In the frugal mode, the bit generation temporarily reduces. However, the virtual buffer fullness quantity is continually updated. This enables the carryover of the savings made during frugal mode operation when the system returns to normal mode.

3.2. Buffer Fullness Estimation and Carryover:

Virtual buffer fullness is determined from three quantities: (i) the number of bits generated so far by encoding previous j-1 macroblocks inside this frame (B_{j-1}), (ii) the initial fullness of buffer before beginning the encoding of this frame (d_j^0), and (iii) the target bits allocated to this frame (T^x). The initial values for the buffer fullness are computed at the beginning of encoding a frame. For the encoding of first frame of a GOP these are given by $d_0^I = 10 \times \frac{r}{31}, \;\; d_0^P = k_P \cdot d_0^I, \;\; and \;\; d_0^B = k_B \cdot d_0^I$. Here k_B and k_P are universal constants and depend on the quantization matrices. For standard MPEG-2 quantization matrix their values are k_P =1.0 and k_B =1.4. For subsequent frames the final fullness of the previous frame is passed on as the initial fullness of the next frame buffers. During the encoding of a frame for each macro-block the actual amount of bits produced is measured immediately after it's encoding. Thus, Once DCT is done, all subsequent coding of the current macroblock including VLC have to be completed before the next macroblock can be quantized.

3.3. Target Rate Proportioning:

To calculate the target bit for each frame, at the beginning of each GOP, first a rough allocation for the entire GOP is estimated. This is estimated from the target stream bit rate, frame rate and the total number of frames in the

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[2] The motivation behind the *activity factor* is that human visual perception is less sensitive to distortions in noisier textured areas and more sensitive to distortion in image areas with uniform texture. We used a slightly modified region based activity assignment algorithm for estimation of N_act_j. This is like TM-5 but in addition, it allows spatial distribution of the bits to be controlled for a given allocation of frame bits.

GOP. Each GOP initially has one I and n_B and n_P B, and P frames respectively.

$$R_{GOP} = \left\lfloor \frac{(1+n_{P-remaining}+n_{B-remaining})\times c(t)}{frame\_rate} + 0.5 \right\rfloor \qquad \ldots(3)$$

To account for the variations in the frame types complexities, a TM-5 like adjustment is made. This is performed with the quantities called *global complexity measures* [X_I: X_P: X_B]. These are computed by averaging the actual quantization values used during the encoding of all the macroblocks including the skipped ones) and the actual number of bits generated S_X, where $X_X = S_X \cdot Q_X$. These averages are maintained for each frame type (x=I, P, and B) and updated at the end of each frame encoding. Finally, the actual target bit rate for each frame type is computed using the following usual TM-5 models (where k's are various defined constants):

$$T^I(t) = \left\lfloor \frac{R(t)}{1+\frac{n_P \cdot X_P}{k_P \cdot X_I}+\frac{n_B \cdot X_B}{k_B \cdot X_I}} + 0.5 \right\rfloor \qquad \ldots(4a)$$

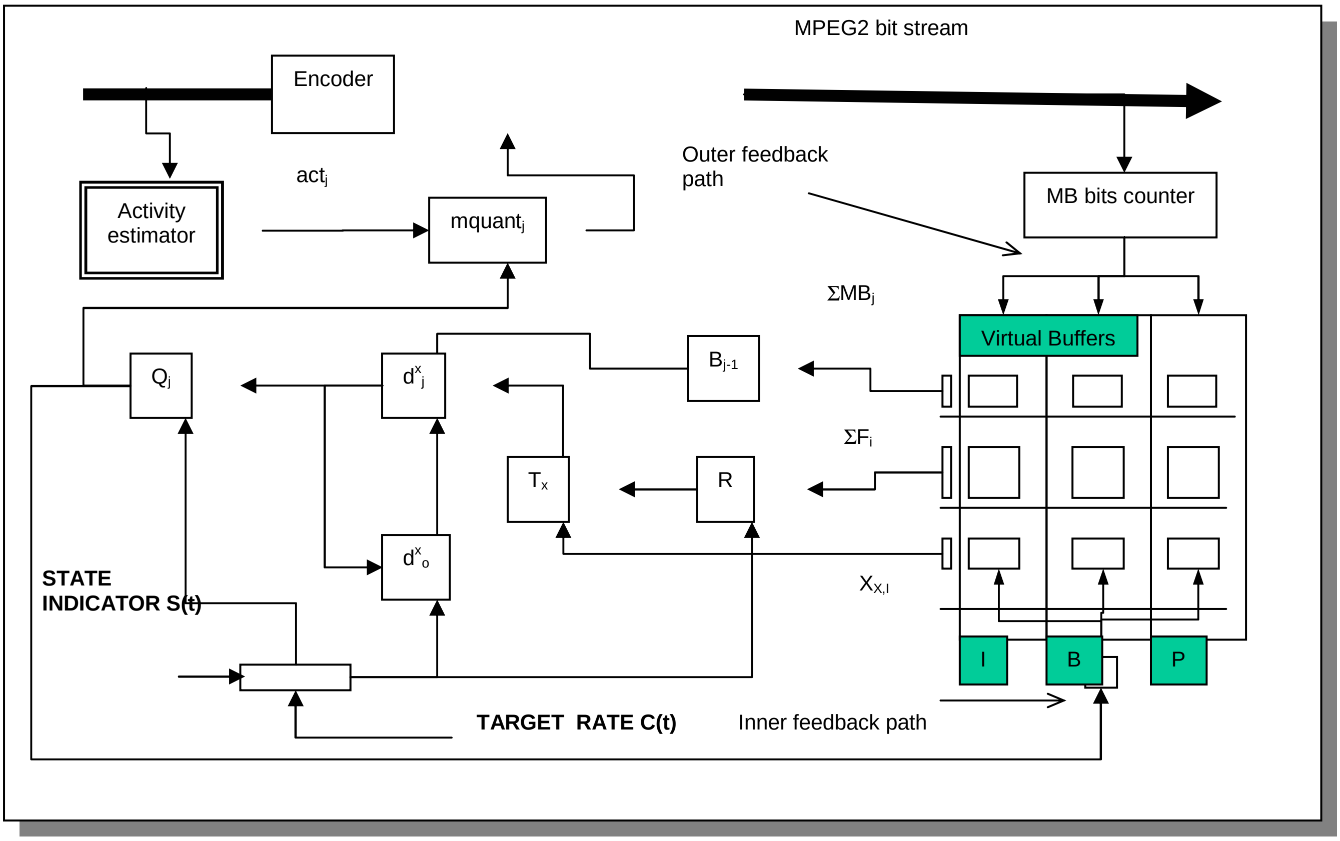


Fig-2 MPEG-2 feedback based symbiotic rate control system

$$T^P(t) = \left\lfloor \frac{R(t)}{n_P + \frac{n_B \cdot k_P \cdot X_B}{k_B \cdot X_P}} + 0.5 \right\rfloor \qquad \ldots(4b)$$

$$T^B(t) = \left\lfloor \frac{R(t)}{n_B + \frac{n_P \cdot k_B \cdot X_P}{k_P \cdot X_B}} + 0.5 \right\rfloor \qquad \ldots(4c)$$

Once each frame is encoded the bits used is measured and the encoded frame is subtracted from the initial GOP size ($R_{new} = R - S_X$) to estimate the remaining available bits. Also, the number of frames n_B or n_P gradually decreases. The target size for subsequent frames in the GOP, which are either type P or B, are estimated from the remaining bits R, and the remaining number of frames. Finally $Q_j = \lfloor 31 \times d_j^x \cdot r^{-1} \rfloor$ is computed by dividing the buffer-fullness by the TM-5 *reaction parameter*. When the system is in normal mode the rate control mechanism does not need to sense the target bit rate at every frame. However, when it moves into frugal mode it senses the current target-rate per frame.

3.4. Symbiotic Rate Determination:

The transcoder only focuses on the rate adaptation. However, the actual values of the rate dynamics are controlled by a separate mechanism called *symbiosis controller*. The control parameter of the rate controller *target bit-rate* c(t) is determined by a two variable min/max mechanism.

The idea is to closely mimic the rate provided by the underlying transport layer, however, it is done in a way that maximizes applications requirements. In this experiment we have designed a symbiosis, which responds to a timeout event. Let the target bit rate during normal mode generation is given by C_{max}. When, a time-out event occurs in the channel (designated by an event variable $\xi = 1$), we let the subscriber rate retract to a smaller but yet non zero quantity. We define this point by parameter called *rate retraction ratio* ρ.

The idea is that based on the specific video instance and a tolerance level on its quality the system should still be able to generate video however, with lesser visual quality based on precise quality/ delay tradeoff boundaries of the video. Based on the tolerance we define a ratio called *rate retraction ratio:*

$$\rho = \frac{C_{\min}}{C_{\max}}.$$

For symbiosis with the underlying TCP, we define a running generation threshold function as following:

$$\begin{aligned} c_T(t) &= \frac{1}{2} c(t-1) \; when \, \xi = 1 \\ &= c_T(t-1) \; otherwise \end{aligned} \qquad \ldots(5)$$

It retracts to half its current size when fault occurs. The running control function c(t) is then given by:

$$\begin{aligned} c(t) &= \rho \cdot c_{\max} \; when \, \xi = 1 \\ &= 2 \cdot c(t-1) \; when \, c(t) \geq \frac{1}{2} c_T(t-1) \\ &= \min[C_{\max}, x(t-1)+1] \; when \, c(t-1) \geq c_T(t-1) \end{aligned} \qquad \ldots(6)$$

The control function performs *binary-exponential-backoff* and *additive increase* within the limits given by generation parameters ρ and normal mode target bitrate C_{max}. The system enters the frugal state $S(t) = 1$, when then loss event occurs (i.e. $\xi = 1$), and stays in the frugal state until the control (target bit-rate) recovers to the normal target bit-rate.

4. Experiment Results

For our experiment we have built a simulator for the proposed interactive TCP and subjected it with real MPEG-2 transcoder generated traffic. The simulator allows the underlying data link layer to have various impairment conditions. Below we share some representative results. This experiment describes the performance for the case of a MPEG-2 ISO/IEC 13818-2 broadcast DTV (704x480) resolution video encoded with base frame rate of 4 Mbps. In this section we also show the frame wise detail event trace of what happens to the first 250 of the frames. We simulated both the classical transport channel (labeled as TCP) as well as interactive transport channel (iTCP). We let the video generator (transcoder) feed into the video stream.

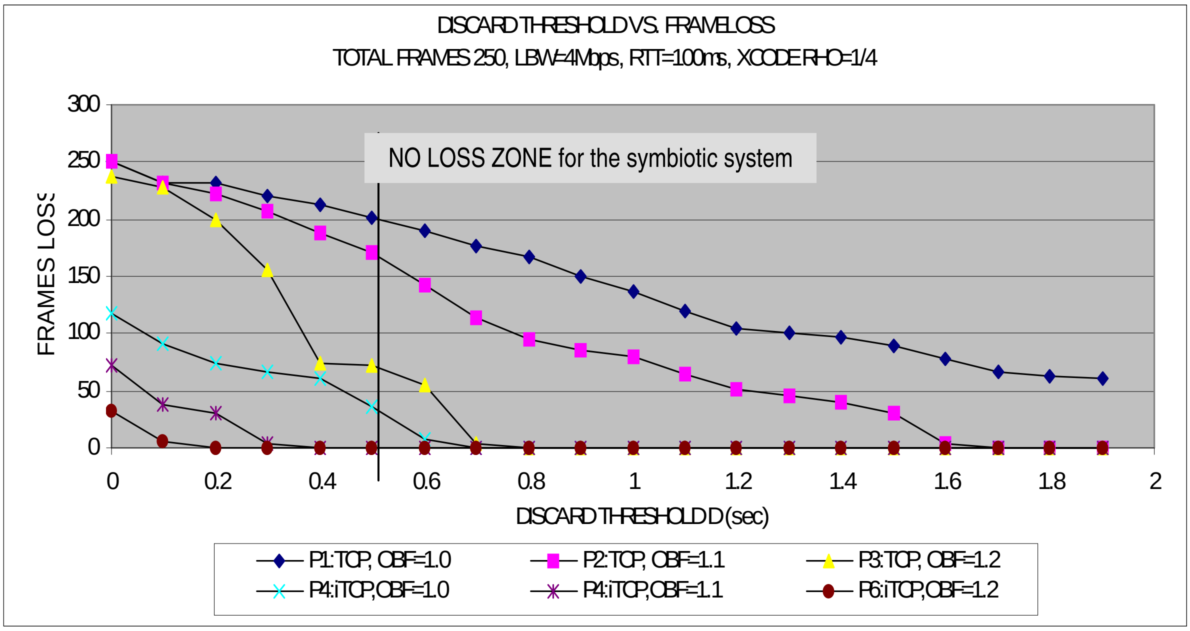


Fig-3

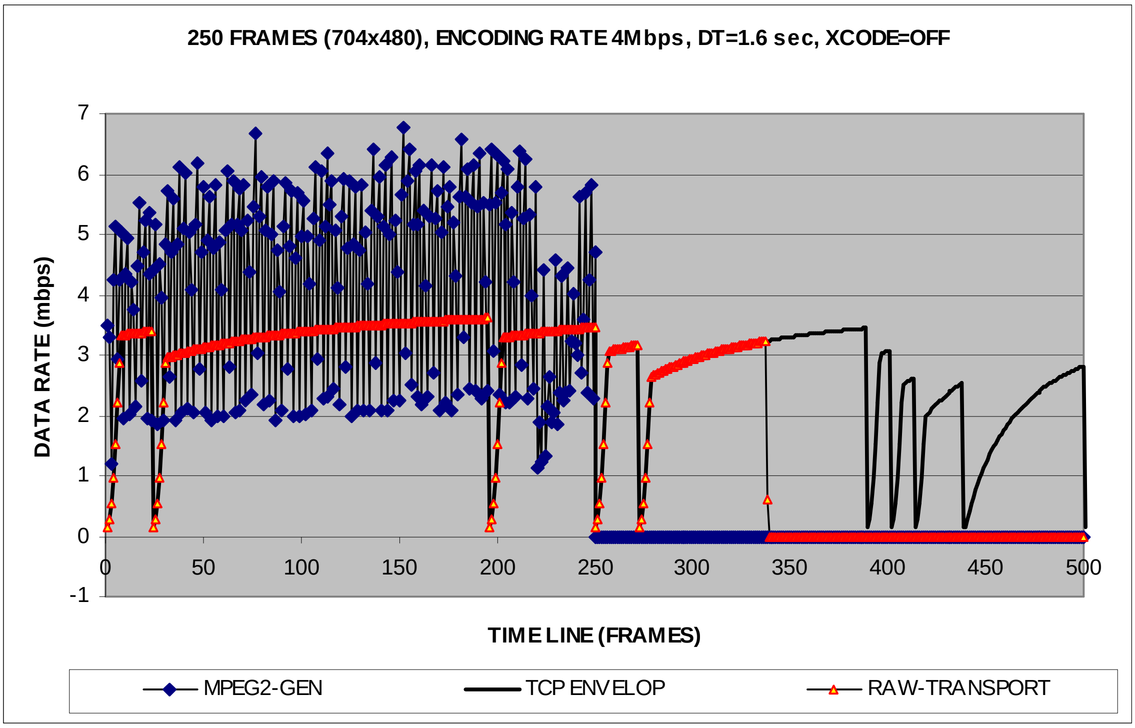
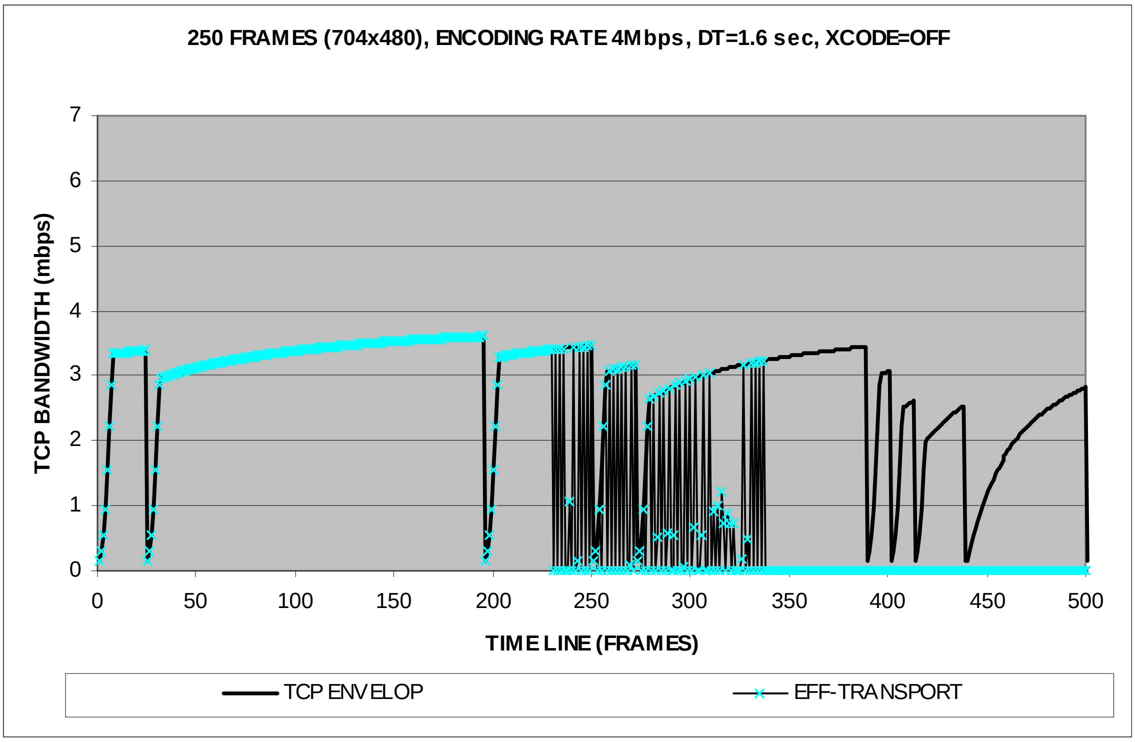

250 FRAMES (704x480), ENCODING RATE 4Mbps, DT=1.6 sec, XCODE=OFF
DATA RATE (mbps)
TIME LINE (FRAMES)
7
6
5
4
3
2
1
0
-1
0
50
100
150
200
250
300
350
400
450
500
MPEG2-GEN
TCP ENVELOP
RAW-TRANSPORT

Fig-4(a)


250 FRAMES (704x480), ENCODING RATE 4Mbps, DT=1.6 sec, XCODE=OFF
TCP BANDWIDTH (mbps)
TIME LINE (FRAMES)
7
6
5
4
3
2
1
0
0
50
100
150
200
250
300
350
400
450
500
TCP ENVELOP
EFF-TRANSPORT

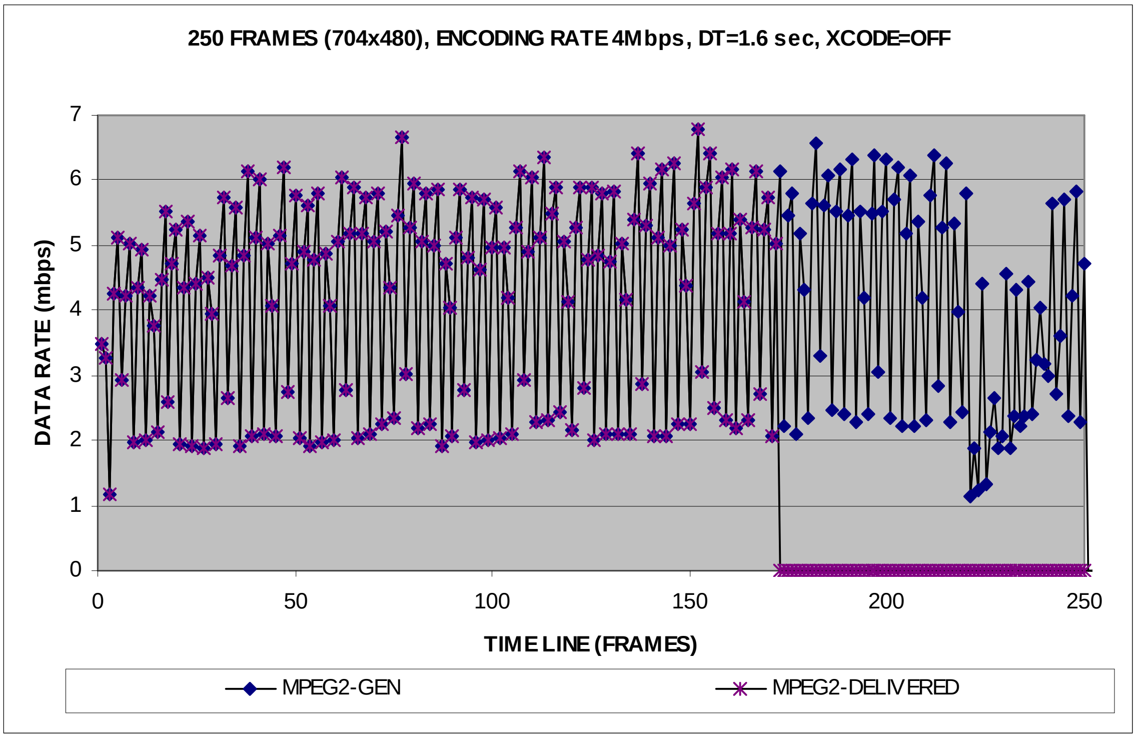

250 FRAMES (704x480), ENCODING RATE 4Mbps, DT=1.6 sec, XCODE=OFF
DATA RATE (mbps)
0
1
2
3
4
5
6
7
0
50
100
150
200
250
TIME LINE (FRAMES)
MPEG2-GEN
MPEG2-DELIVERED

Fig-4(c)

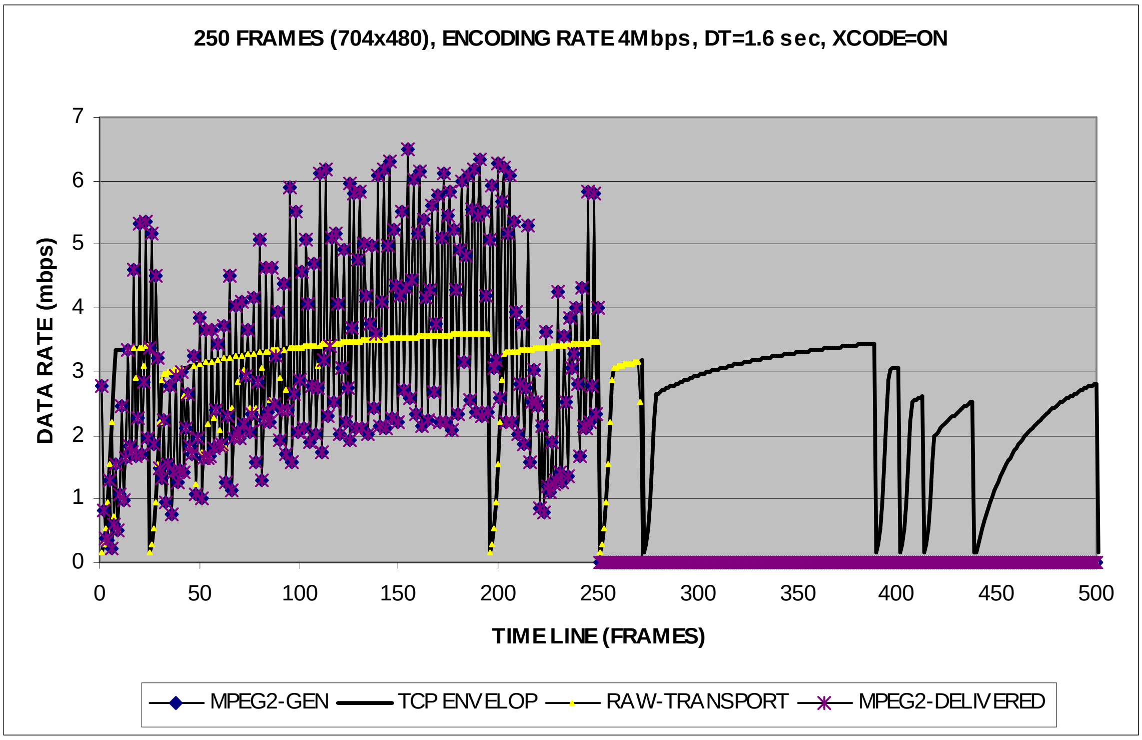

250 FRAMES (704x480), ENCODING RATE 4Mbps, DT=1.6 sec, XCODE=ON
DATA RATE (mbps)
0
1
2
3
4
5
6
7
0
50
100
150
200
250
300
350
400
450
500
TIME LINE (FRAMES)
MPEG2-GEN
TCP ENVELOP
RAW-TRANSPORT
MPEG2-DELIVERED

Fig-4(d)

For fairness, for the iTCP system (Fig-1(b)) we added additional delay from the call back mechanics and the handler process fork delay by the kernel. We simulated channel time-out event using a uniform random distribution. We further assumed, that this event is independent of the video stream size (due to congestion deep inside the network).

In the classical mode, we let the transcoder operate in error unaware mode and generate the video using TM-5 [Mpeg00] rate control at 4 Mbps. Transport control protocol buffered the generated data while the transport layer exercised binary back-off and additive recovery at time-out events. In the interactive mode the transcoder according to the symbiosis controller varied the video rate for interactive TCP. The video data was received into an analyzer. The transcoder and the analyzer both recorded the delivery time of each frame date as they were transported according to their coding sequence. A frame is considered 'failed' if its delivery time exceeds a given *discard threshold* d.

The objective of our first experiment is to observe how the rate of frame discard varies with various choices of the threshold.

We were also curious to see how the discard rate varies with the allocated transport bandwidth. Correspondingly, we varied the channel (link layer) bandwidths from 100-120% over the coding bandwidth. Fig-3 plots the dramatic difference between the performances of the two channels. It plots the number of failed frame (y-axis) with various cut-off delays (x-axis).

In this symbiosis we used $\rho=0.25$. The top three curves (P1, P2, P3) show the frame loss for classical channel for three path bandwidths (1.0, 1.1 and 1.2 times encoding target rate). As can be seen, even if the acceptable delay is set as high as 1.2 seconds, more than 50 (+20%) frames are lost for both P1 and P2. Curves P4, P5 and P6 respectively now show the improvement in performance from the iTCP integrated solution for the same three cases. Even at much smaller 0.7 seconds cutoff delay, complete recovery has been possible. No frame was lost.

Fig-4(a)-(d) explain the internal mechanics of this entire arrangement. The x-axis plots the time line in terms of video frame (coding) sequence. At 30 fps rate each frame is approximately spaced 33 ms apart. Fig-4(a) plots. Fig-4(a) shows the data generation rate, the TCP window envelope, and the network transfer rate (that occurred within this envelop). As evident, in the unaware scheme due to the back-offs the buffer congestion propagated in time. The congestion grew worse over time. The last of the frames was transported almost near the time-line point 350. How much of the data that was transported was actually useful to the applications? To illustrate the situation, in Fig-4(b) we have plotted the effective transfer rate for a discard threshold 1.6 sec. As evident, although almost 30% of the data that network carried was effectively useless from the application point of view. Fig-4(c) plots the same data at application level. It shows in terms of frame line, which has been generated and which was effectively delivered. Fig-4(d) now re-plots all four quantities for the TCP interactive experiment. It shows the data generated by the transcoder, the TCP envelop (same as before), the network transport rate, and the effective frame delivery. As evident the generated MPEG-2 data now resembles closely the TCP envelope. The MPEG-2 generated matches identically with the MPEG-2 delivered. Consequently, there was no loss in the effective transfer rate.

The application level trade-off that occurred in this experiment is now illustrated in the 2-dimensional plots for these two situations. In Fig-5(a) each of the frames are plotted as a point in the video quality/ frame delay plane. As can be seen from the region of the two QoS distributions, in classical TCP, although frames have been generated with SNR quality ranging between 55-38 dB, but many of these frames are lost in transport, and was never delivered. In contrast, the proposed TCP interactive can deliver all the frames with .6-.7 delay guarantee at 55-20 dB quality[3]. Fundamentally, what **TCP interactive** has offered is a qualitatively (as opposed to the quantitative

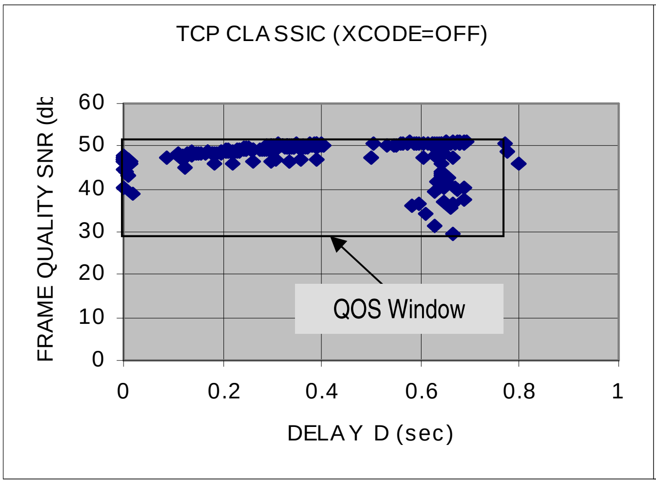
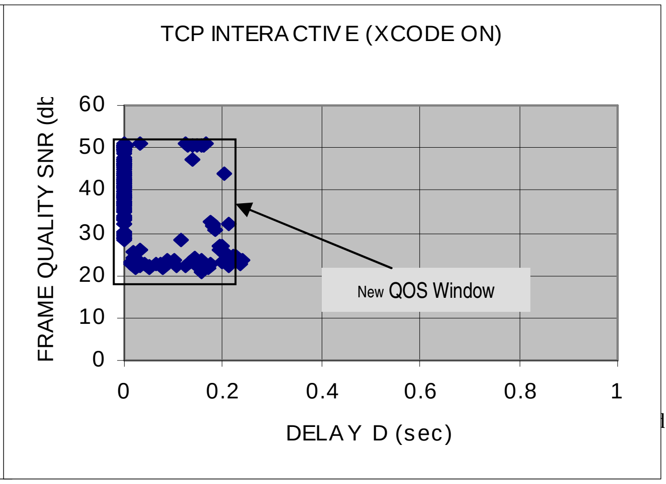


Fig-5(a)



leo

Fig-5(b)

improvements offered by any unaware solution) new empowering mechanism, where the catastrophic frame delay can be traded off for acceptable reduction in SNR quality, resulting in revolutionary advance compared to the almost primitive and time insensitive ways that congestion is generally handled today.

5. Conclusions and Current Work

In this paper, we have presented a case of rate symbiosis mechanism in line with current advances in TCP friendly systems. We have presented the case through a simple 'interactive' generalization of the classical transport control protocol, and a novel implementation of a symbiotic MPEG-2 transcoder. The proposed *principle of protocol interactivity* can enable fundamentally new solutions to many of today's hard to tackle problems. In this paper we have demonstrated the case of quality conformant congestion control for time-sensitive traffic.

The approach exposed the overall advantage of network 'friendly' applications. However, is also departs from the mainstream TCP friendly systems those has been suggested recently in two senses.

First, it does not consider adding new major component in network software structure. One of the principal strength of the proposed scheme is its relative simplicity at network layers –yet its effectiveness. It only expects some form of interactivity directly from the concerned network protocols as a general interface feature. Thus there is no expectation of additional services (such as combined congestion control from multiple applications). Nor it excludes those possibilities, as such features may have added benefits for multi-stream congestion control.

Secondly, it is not dependent on indirect probing tools or separate new network utilities. (nor it excludes their use when available). Interestingly some of the information measured by these tools at the upper layer might be already available (or are being estimated/tracked) at lower layers anyway. At least this is the case with TCP congestion. The suggested direct protocol interactivity thus seems to be the logical path that can avoid potential duplication of efforts.

Nevertheless, the approach will increase complexity in network layer. This is an unavoidable cost if we are to build network aware systems. Other schemes (probe, tool or component mediated schemes) also have the same underlying assumptions. Novel system component mediated schemes too eventually require the existing protocols to be interactive with themselves and thus will require almost similar modification in the base protocols.

The augmentation of the notification feature increases the normal mode delay of TCP even it is slight. However, the actual cost will depend on the intensity of coupling. Designer of application symbiosis unit must be aware of the potential cost of tight coupling between handler and caller. In a more detailed design it is possible to add control over the coupling. However, as shown by the results-- with a prudent design the impact on the network level transfer rate (based on low layer measurement), if any, can be widely surpassed by the gain made at application layer. However, an interesting aspect of this scheme is that a wrong design will only affect the application and will have no effect on the fairness and other network internal issues. Also, notably, the entire scheme is less invasive than many other recent approaches in congestion management (such as ECN or RED, which require router intervention and/or IP layer intervention across network). However, the proposed interactivity is not an alternate to these, rather is a complimentary scheme.

Applications however are not forced to use notification. Current time tested transport protocol TCP will remain adequate for many simpler and less demanding applications. However, the advent of complex network wide applications requiring sophisticated inter-process communication will increasingly require interactivity. The proposed scheme however, is not a general solution for congestion control. It is applicable where the opportunity exists for adjusting the data rate and quality tradeoff. This is generally true for multimedia data.

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