QoS Control and Interworking of Overbooked Elastic and Brittle ATM Traffic

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ABSTRACT

Early deployments of broadband ATM networks are likely to use *overbooked* Permanent Virtual Connections in which more connections are set up than Connection Admission Control (CAC) would ordinarily permit. Since, hopefully, not all users will simultaneously use their connections, the QoS levels required for each connection will be met. A critical problem concerns the behaviour when more than the expected maximum number of users are active - does the network gracefully degrade? Internet type applications under the control of TCP posses a degree of *elasticity* that enable them to adjust, whereas others (often real time applications) are more *brittle*. Under such circumstances appropriate ATM level controls are required that maintain the differing QoS requirements. The interaction of these controls with higher layer protocols, notably TCP, is crucial.

Various control mechanisms are described and compared. It is concluded that TCP performs well in conjunction with the ATM Forum's Guaranteed Frame Rate and VBR.3 traffic classes. Both cases benefit from Partial Packet Discard and a mechanism that assures minimum bandwidth requirements. Network partitioning using PVCs gives additional control. When used together these controls can reduce undesirable interactions between elastic and brittle services thus enabling easier end-to-end service provision.

INTRODUCTION

A number of services are planned for use over public Wide Area Broadband networks. Two examples are *high speed data services* (e.g. Internet access and Teleworking) and *digital video* (e.g. Video Conferencing and Interactive Television, iTV).

There is considerable debate between the IP and ATM communities as to the preferred protocol for these. This paper primarily considers the ATM environment but with the assumption that much of the data will be IP, with TCP as the layer 4 transport protocol, although there will additionally be some applications using native ATM.

Ideally access to the ATM network would be controlled by ATM's Switch Virtual Circuits, (SVCs) as a bandwidth access control mechanism. That is, when users require a particular service then they would request an ATM connection with the required bandwidth and quality of service requirements, and be charged accordingly. For instance, iTV may require a constant bit rate connection operating at a few Mbps, whereas Internet traffic may be bursty. If the network is capable of supporting a particular connection then the ATM's connection admission control (CAC) would accept it - otherwise it would be rejected.

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Unfortunately, many early deployments of such networks are not using SVCs, but instead use Permanent Virtual Circuits (PVCs). In order to make this economic it is necessary to *overbook* the number of connected customers. That is, establish connections to more customers than the ATM CAC would ordinarily permit.

This presents important quality of service issues since there is the possibility of complete network collapse under overload conditions. SVCs present a form of graceful degradation in which already connected users maintain good QoS, with new connections denied service. With PVCs, customers' applications will launch traffic over existing connections, and if the network is already heavily loaded this may result in a sufficiently large number of cells being lost from all connections such that they all become unusable. However, an alternative solution of over dimensioning of the network, incurs high cost to Network Operators but would work.

Application Elasticity

Whether or not the overbooked network degrades gracefully under overload depends on the degree of *elasticity* within the applications. The majority of applications used currently on the Internet are elastic in the sense that when bandwidth becomes scarce then everything just slows down. Do all users care if it takes 15 seconds to load a Web page that normally takes 10 seconds? Charging issues for such applications are discussed in [1].

Conversely, many Telecomms services are quite *inelastic* or *brittle*. As an example consider speech. Although adaptive compression algorithms have been developed, they are not widely deployed. Most fixed public networks still use constant bit rate 64kbps for voice, although more costly networks such as Mobile and International network use compression, possibly adaptively.

A further example is iTV. Typical video servers contain pre-compressed streams of MPEG [2] data. This may require, depending on the coding and quality parameters, at least 2Mbps and if such bandwidth is not available then the service becomes unusable.

SERVICES OVERVIEW

Brittle Service: Interactive Television (iTV)

An essential feature of video compression is that it can be regarded as a variable-rate (VBR) mechanism. Usually, there is more information to be passed to the decoder as interframe changes increase due to temporal activity (motion). However the nature of video compression and coding algorithms in use is to shape the output to the network as a constant bitrate stream. The effect of this is that during occasions of little motion, the codec is able to substitute data representing better spatial (intrafield) resolution. In short, there is a trade-off between constant quality at variable bitrate or variable quality at constant bitrate (CBR). The latter case is true of most iTV deployments today, and given the existing investment in VLSI to support CBR, there is little momentum to standardise on an MPEG/VBR specification.

ATM cell loss and **errors** of video packets containing important MPEG [13] stream synchronisation data, of which the Program Clock Reference (PCR) is most important, can greatly affect service performance. The PCR is applied to the MPEG stream at approximately 0.1 second intervals, to synchronise the phase-locked loop (PLL) in the video decoder. The PLL then provides a timebase for the timestamps used in video reproduction. The ATM Forum Video on Demand [2] specification recommends the AAL5 criteria for efficient

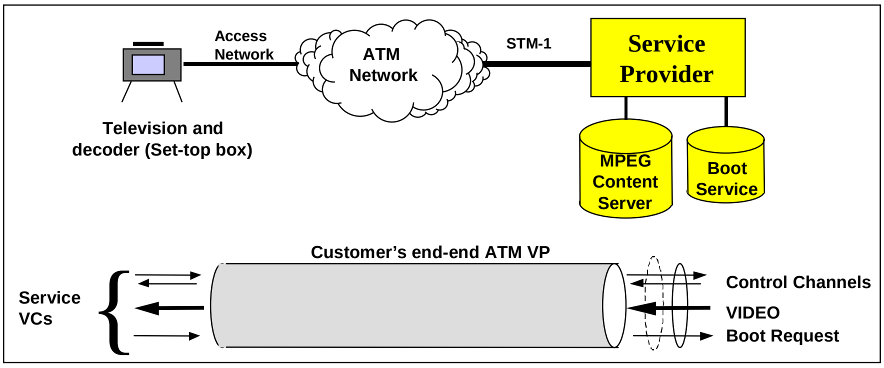

Figure 1: Network Configuration for Interactive TV

transport, whilst minimising jitter in the reconstruction of MPEG data. Consequently, the MPEG2 packet containing the important PCR data is inserted into an AAL5 Protocol Data Unit (PDU), not exceeding two MPEG transport packets. Thus, whilst the decoder is unaware of the PCR location, the maximum jitter cannot exceed the length of one MPEG2 packet (188bytes). This then relies on the ATM transport mechanism not losing the cell containing the PCR data and for systems with a cell loss ratio in the order of 1×10^{-7}, the risk of losing an MPEG picture group (about half a second of video) is of the order of hourly.

The video stream is not particularly sensitive to absolute delay, although it is sensitive to **cell delay variation**. A typical receiver's buffer can absorb a few 10's of milliseconds of delay, but greater variation results in cell loss or discard.

Practical implementations of iTV over ATM require the use of several VCs as shown in Figure 1. Their requirements on the network can be defined by the following functionality:

- **Boot service**: when the decoder (set-top box) is switched on, it sends a request to the service provider for a boot image (decoders have minimal functionality and are usually unable to store the image on a permanent basis). The primary benefit of this is that the service provider has control over set-top box middleware releases for service look and feel;
- **Control channels** carry messages *upstream* from the set-top box passing control that request TV programs, fast forward, data for interactive shopping, etc. and *downstream* from the service provider to load software and perform network management operations;
- **Video channel**: MPEG2 transport stream at the designated constant bitrate.

In general the most brittle of these VCs is the video stream. Any cell loss due to overbooking (or indeed other reasons) could have a number of effects on video quality, ranging from losses within individual frames to the absence of one or more picture groups.

The other VCs are more tolerant. Boot requests are usually repeated until acknowledged by the service provider and the upstream control channel has a latency budget determined by the response time the service provider wishes to honour following a user request. Losses in this VC are easily remedied by re-pressing the remote control. Similarly, it is likely that a control channel to the set-top box will not have any substantial realtime criticality.

The evolution of network provisioning for an iTV service indicates the maintenance of a rigid CBR QoS contract for downstream video. Other VCs can be tailored to fit overbooking and other QoS types, such as UBR for the boot channel and VBR for the control mechanism.

Elastic Traffic: Internet

The Internet still offers only a best effort service. QoS control rests with the layer 4 transport layer, applications and users who adjust to the Internet's performance. In particular TCP is widely used in order to perform end to end flow congestion control and error recovery.

The Internet Engineering Task Force, IETF, is developing new protocols and architectures that give additional QoS control for delay sensitive, real-time traffic such as voice and video. Important developments include RTP [3], the Integrated Services [4] environment that includes RSVP, and Differentiated Services [5].

In this paper it shall be assumed that the majority of Internet traffic is still controlled by TCP, and hence the performance interaction between TCP's flow control and the underlying network, specifically ATM, is of particular interest.

TCP Performance Analysis

One key performance metric for TCP is sustained throughput. That is, at which rate would a file of theoretically infinite size be transferred? The situation in which broadband access is provided by ADSL is considered in [6] where equations provided by [7] are extended in order estimate maximum TCP throughput. The key approximation is that:

$$Throughput \approx Window\_size / RTT \qquad (1)$$

for which *RTT* is the Round Trip Time and,

$$Window\_Size = \min(TCP\_Buffer\_Size, cwnd) \qquad (2)$$

where *TCP_Buffer* is the minimum of the sender and receiver buffer sizes for the TCP configuration (typically 8 or 64KBytes), and *cwnd* is the mean congestion window which from [7] is:

$$cwnd \approx C.MSS / \sqrt{segment\_loss\_ratio} \qquad (3)$$

where *MSS* is the (maximum) segment size, and *C* is a constant that depends on the TCP implementation, for which *C=0.93* is appropriate for a typically TCP Reno implementations with delayed achnowledgements [7].

Additionally it is necessary to estimate the RTT. With no queuing in the system this is the round trip propagation delay plus the length (in time) of a TCP segment, giving:

$$RTT\_1 = propagation\_delay + MSS/TCP\_rate + smaller\_terms, \qquad (4)$$

Where:

- *propagation_delay* is the time taken for a single bit to complete a round trip, with no queuing elsewhere in the network;
- *TCP_Rate* is the maximum rate at which TCP can be transmitted, and is constrained by lower layers such as the physical line rate, and protocol header overheads;
- *smaller_terms* would include the length of an acknowledgement, plus segment processing time within the end system.

One contribution to the smaller terms that can be considered is that due to delayed acknowledgements. A rough approximation is that this will add an extra *MSS/TCP_rate* to the delay, since this is the average time taken for a second segment to arrive, which then triggers the delayed acknowledgement to be sent. Thus we shall take the RTT to be (at least):

$$RTT\_1 = propagation\_delay + 2 * MSS/TCP\_rate \tag{5}$$

If there is queuing within the system the simulation evidence shows that the RTT will equal:

$$RTT = Window\_Size/TCP\_Rate \tag{6}$$

This is equivalent to supposing that the queuing is at the point at which the bandwidth is constrained to be *TCP_Rate*, and that this queue absorbs all other delays (e.g. propagation delays), provided that they are less the amount in (5).

Combining these equations using *estimated_throughput=min(window_size)/max(RTT)* gives,

$$Throughput \approx \min\left(TCP\_Rate, \frac{\min(Buffer\_size, cwnd) - MSS}{propagation\_delay + MSS/TCP\_rate} \right). \tag{7}$$

The above has been compared with a simulation model, and it is found to be very close for small segment loss ratios (i.e.<0.01%) and a slight overestimate of up to 20% at larger loss ratios (i.e. between 0.01% and 1%). It is quite inaccurate at higher segment loss ratios since it does not take account of slow start, for which the approach of [8] is more appropriate. Such results are broadly consistent with measured throughput in [7].

Thus, in order for TCP to achieve a given level of *throughput* it is required that:

$$TCP\_Rate > throughput \tag{8}$$

in order for the physical rate to be higher than the required throughput, and:

$$Buffer\_size > TCP\_Rate * propagation\_delay + 2 * MSS \tag{9}$$

the traditional product of delay and bandwidth, with a small correction term, and,

$$segment\_loss\_ratio \le \left(\frac{0.93 MSS}{TCP\_rate * propagation\_delay + 2 * MSS} \right)^2 \tag{10}$$

in order to keep the congestion window sufficiently open.

Figure 2 plots this for a *TCP_Rate* of 2Mbps for two MSS values. If the TCP buffer size is 64Kbytes, then equation (9) implies that the propagation delay can be at most 256ms (ignoring the small 2*MSS term). Provided this is satisfied, then the corresponding segment loss ratio must be lower than the plotted line for the required throughput. This confirms that TCP's sustainable throughput is constrained by:

- **TCP implementation** particularly TCP sender and buffer sizes, and selected MSS,
- the **round trip time** (*RTT*), which in the case of ADSL can be significantly affected by the Forward Error Correction (FEC) and Interleaving Algorithms,
- **loss and errors** caused not only by buffer overflow but also errors at the physical layer (some of which may be corrected by the ADSLs FEC).

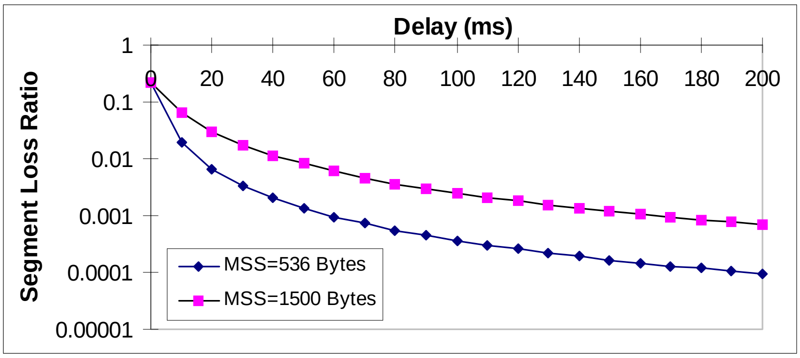

Figure 2: Loss vs. Delay for 2Mbps throughput for a single TCP connection

For ADSL the degree of **Asymmetry** can also be a constraint. An MSS of 536 bytes typically requires 13 ATM cells and an ACK requires 2 ATM cells. With delayed acknowledgement one ACK is sent for 2 segments. Thus it is required that the downstream bandwidth be no more than 13 times the upstream bandwidth.

ATM LAYER

Let us now consider interworking between overbooked elastic and brittle traffic within an ATM environment, as could occur when pre-compressed MPEG video is mixed with TCP controlled Internet traffic. The tight performance constraints for the MPEG imply that it should be given priority within ATM buffers, both in terms of loss (since in theory TCP can retransmit), and delay (TCP is sensitive to maximum delay, but not to delay variation).

Key requirements of the ATM layer are:

R1. If CAC is available for the MPEG traffic then it should be used to determine whether the connection's QoS requirements can be met. If accepted, then subsequent connections should not affect the assured QoS of previously accepted connections. Such CAC may either be within the switch itself, or distributed at a network level.

R2. If CAC is available for the elastic traffic then it should be used to guarantee minimum bandwidth requirements. But when the network is lightly loaded then it should allow connections to expand into excess capacity (especially if the user is charged for that). If CAC is not available then the minimum bandwidth guarantees may also be overbooked.

R3. The queuing and service disciplines should ensure fairness (in some sense) amongst elastic connections. In particular, connections that have higher minimum guaranteed rates should receive a greater share of the available bandwidth than those with lower levels.

R4. If MPEG is also overbooked then the minimum cell rates should be guaranteed (if possible) for the other traffic; i.e. one service class should not interfere with traffic within another service class.

Some ATM controls are now described that can be used in order to solve the above problems.

ATM Traffic Class Selection

Different services have different QoS requirements, suggesting that it is appropriate to place them into different ATM traffic classes. However, the specific behaviour of traffic classes depend upon both **standard facilities** (e.g. policies that are specified in standards such as whether to police against Peak or Sustained rates) and **proprietary facilities** (e.g. implementation details associated with the traffic class such as *Buffer sizes*, and their partitioning between high and low priority traffic (CLP=0 or 1), and *Buffer scheduling* algorithms). Some network elements do not even recognise traffic classes, and merely forward cells to their appropriate destinations. Most do implement class-based priority mechanisms. There are a number of suitable traffic classes for the elastic TCP/IP type traffic [12].

- **VBR.1-nrt & VBR.2-nrt**: The *non-tagging* options that discards cells that exceed the depth of the SCR's leaky-bucket.
- **VBR.3-nrt**: The *tagging* option that marks such violating cells as CLP=1. At subsequent nodes whose buffers are almost full, such low priority cells are discarded. This is called **Selective Cell Discard, SCD,** and with an overbooked network it would permit applications to expand to run at PCR (since CLP=1 cells arrive at lowly occupied buffers).
- **UBR**: ATM's best effort service traffic class.
- **ABR**: Designed as ATM's *elastic* traffic class, that adjusts cell rates to available bandwidth. This can guarantee a minimum rate, MCR. ABR is currently out of favour due to end-to-end network implementation complexities and scaling for public network use.

Another alternative is **GFR**, Guaranteed Frame Rate, to be discussed in the next section.

ATM Frame Awareness

The performance achieved with TCP/IP can be improved considerably by adding frame based awareness. Consider a typical protocol stack as in Figure 3. If one cell is lost then the whole IP frame is useless; there is no point in keeping the remaining cells. Since IP data is usually transported over AAL5 then the ATM layer can apply controls to AAL5 frames including Early Packet Discard (**EPD**), Partial Packet Discard (**PPD**), and Guaranteed Frame Rate (GFR), [9].

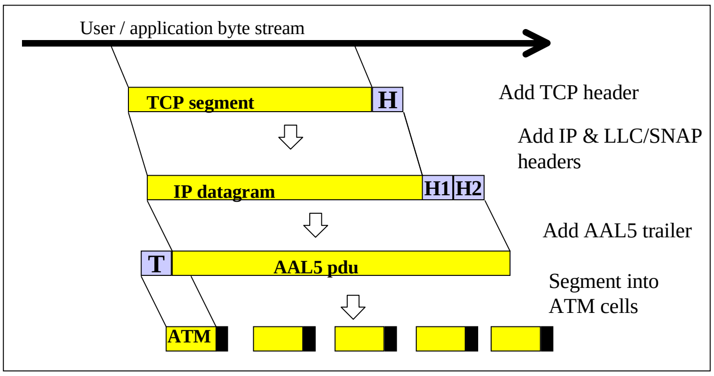

Figure 3: Layer Fragmentation and Encapsulation

GFR is similar to VBR.3 except that it uses frame based tagging and policing. GFR decides at the start of a new AAL5 frame whether it will be able to accept a frame of the maximum permitted length (a worst case assumption). If so, the whole frame is accepted as CLP=0, otherwise it is all tagged as CLP=1.

Queueing Disciplines

A **simple priority queue** is shown in Figure 4, in which the brittle MPEG traffic can be given priority. This will satisfy requirements *R1* and *R2*, but not *R3* and *R4*.

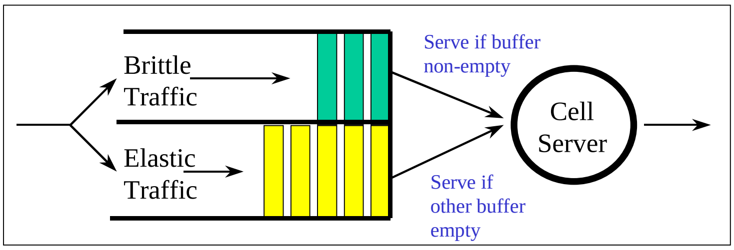

Figure 4: Simple Priority Queue

For example consider the case in which there are only two connections for the elastic traffic, with MCRs of 1Mbps and 10Mbps. Firstly, suppose that 11Mbps per second is available for elastic traffic (or just over so as to ensure that queue lengths do not grow indefinitely). Then if the sources are greedy (as would happen with TCP driving them) then such an arbitrary scheme will allocate 5.5Mbps to each and violate requirement 3.

Secondly, if the MPEG traffic is overbooked, then it could squeeze out all of the elastic traffic, violating requirement *R4*.

Quality Control Paths, QCPs

The QCP provides additional bandwidth control at the ATM layer [10]. It works in a similar manner to the simple priority queue except that lower priority traffic classes are given a minimum reading rate, thus constraining the impact of overbooked higher-priority classes.

A simple CAC algorithm allocates a reading rate to each traffic class. For CBR traffic the rate would be set just less than ΣPCR; ABR or UBR the reading rate should be ΣMCR.

The buffer's scheduling algorithm then decides when to read from each QCP. If that particular queue is empty then a cell is selected from the highest priority non-empty QCP. This will satisfy requirement R4, but unless per-connection controls (e.g. policing or tagging) are also used, it will fail requirement R3.

Network Partitioning

It will be shown that the above facilities can be combined to control of aggregate traffic. An additional issue is that there may be a number of service providers, each of which may be delivering a mixture of elastic and brittle services.

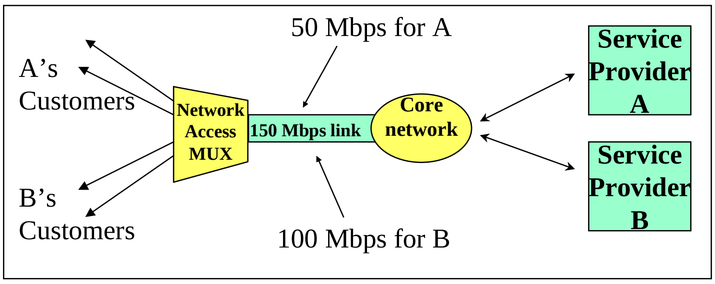

Figure 5 : Network Partitioning

A possible control for such a situation is network or service level control (e.g. an access server) instead of ATM switch layer control. For instance, when SVCs are not available some form of CAC could be applied at the service layer. In its most simple form this corresponds to counting the number of active sessions or connections.

Consider the iTV service. If the service provider knows the parts of the network over which the connections are routed, and carefully controls the number of active sessions, then QoS is guaranteed. For instance, in the case of DSL, an access multiplexer (DSLAM) may connect to more customers than the link between it and the core network can simultaneously support (150Mbps in Figure 5). The service provider can then control overbooking by knowing which customers are connected to which DSLAM. In Figure 5, Service Provider 'A' would ensure that it is sending no more than 50Mbps to the Access Multiplexer, even though it may have 100 customers on it, each of whom may be capable of requesting 2Mbps. Thus 'A' must then ensure that no more than 25 customers on its Access Mux are given service simultaneously.

There does remain an issue of ensuring that service providers do in fact adhere to their contracts. (e.g. ensuring that A does not serve 26 customers, in the hope that B is not serving its full quota), whilst simultaneously permitting the elastic traffic to use any spare bandwidth.

VP Policing

A possible control is to partition each service providers' traffic into separate VPs. The service providers themselves can then decide how to use the bandwidth within their VPs. For example, at network ingress, elastic traffic could be GFR tagged, with brittle traffic traversing the network at high priority. The network provider need then only police the VPs.

An interesting issue concerns whether elastic traffic should be allowed to *expand* in order to consume spare bandwidth within other service providers' VPs. This may require more priorities than those offered by ATM's simple current CLP=0 or 1 scheme, and suggests a use for the IETF's diffserv [14] multiple drop precedences. This would permit, for example, an individual user's traffic that is sent out of profile from the source to be marked differently from an aggregate of users' traffic that is found to be out of profile within the network.

MODELED SYSTEM

In order to compare a number of the control schemes the model illustrated in Figure 6 was constructed with the following key features:

- **TCP layer**: TCP RENO facilities are included such as Slow Start, Fast Retransmit, Fast Recovery and Congestion Avoidance. Since the performance measure of interest is sustainable throughput, connection establishment and termination facilities are not included. Multiple TCP connections can be modeled, and the largest practically available was 25 connections with 32Kbyte buffers. A delay of 20ms was included in each direction, to give an RTT of at least 40ms (maximum theoretical throughput of 6.4Mbps).
- **ATM layer**: Traffic is spaced in order to comply with the PCR. Dual leaky buckets are available for either tagging (VBR.3), discard (VBR.1), or GFR. Simple priority or QCP queues are included, with 2 partitions (one each for the brittle and elastic traffic). SCD and PPD can be used if required. TCP traffic was shaped to PCR=5Mbps (11792c/s); if tagged the SCR or MCR was 500Kbps (1179c/s).
- **Intermediate layers**: The simplest possible protocol stack is modeled as in Figure 3.
- **Physical Layer**: The server speed is 150Mbps (roughly an STM-1), or 354Kcells/sec.
- **iTV Traffic**: Modeled as a Poisson source of cells for the MPEG channel only. Each video channel is 2Mbps of ATM payload (5208 c/s) or 2.2Mbps of physical bandwidth.

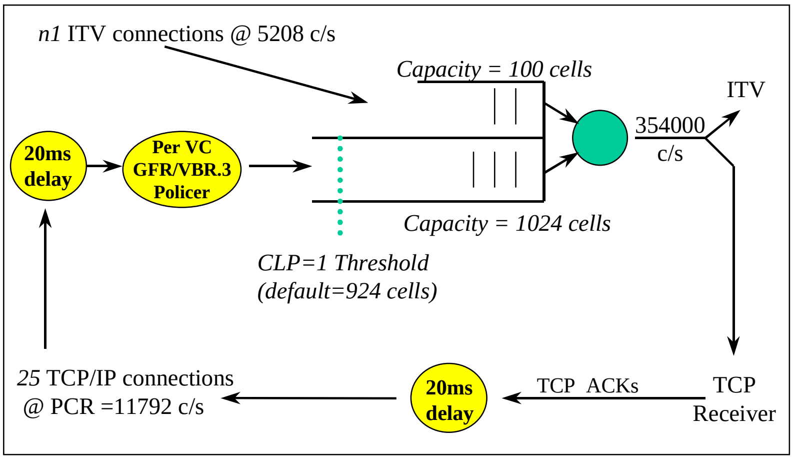

Figure 6: Modeled System

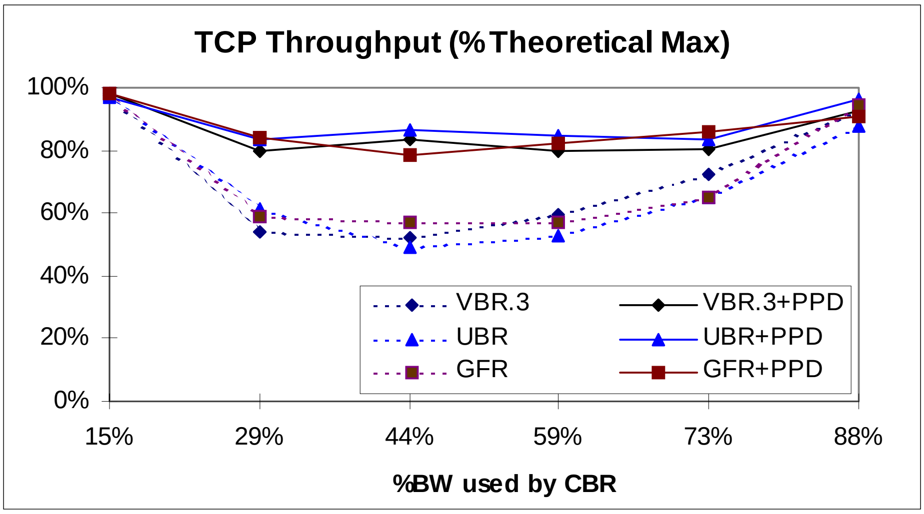

Figure 7: TCP Throughput for varying ATM layer Controls

RESULTS

Six scenarios are modeled - the following three traffic classes, each with and without PPD:

- **UBR**: Essentially uncontrolled - only ATM buffer overflow and delay affect TCP.
- **VBR.3**: With PCR=11792c/s, SCR=1179c/s, MBS=100 cells. The SCD threshold is set at 924 (so that a CLP=1 cell is only accepted if there are less than 924 cells in the VBR buffer, which has total capacity of 1024 cells).
- **GFR**: With PCR=11792c/s, MCR=1179c/s, MBS=100 cells (a new frame is accepted if the bucket would accept MBS/2=50 cells at PCR - this is not strictly in line with the GFR specification that requests that MBS=2xAAL5 frame size). SCD as for VBR.3.

The number of iTV channels was varied so as to use from 15% to 88% of the available bandwidth. The TCP traffic can take the remaining bandwidth. At each point the theoretical maximum TCP goodput is calculated (by calculating inefficiency due to headers) and the achieved throughput is plotted as a percentage of this. From this it is concluded:

1. PPD greatly improves performance. Additional benefits of EPD are expected to be small. This may be due to the small segment size.
2. UBR, GFR and VBR.3 all give similar performance. However, evidence in [11] suggests that UBR is not fair with its allocation of excess bandwidth; each connection gets an equal amount, whereas those with higher MCR's or SCR's should get more.
3. When greatly overloaded performance is similar in each case.
4. With UBR the buffer used by IP is capable of filling. However, since GFR and VBR.3 use tagging and SCD, the buffer never filled. Consequently any high priority (i.e. CLP=0) elastic traffic is virtually assured to get through.

QCP Benefits

The priority queue is now replaced by a QCP mechanism in which low priority elastic traffic is assured at least 12.5Mbps (enough for 25 connections with an MCR of 0.5Mbps each). When the high priority brittle traffic consumes less than 137.5Mbps there is negligible difference in performance from Figure 7. However, if it exceeds this then the QCP will randomly discard the extra brittle traffic whilst assuring the elastic traffic's minimum bandwidth requirement. In such a situation both GFR and VBR.3 (both with PPD) were found to operate at 100% efficiency, taking 0.5Mbps each.

Proposed CAC Algorithm

It is proposed that the CAC algorithm should allocate bandwidth for each class of:

- **For elastic traffic**: Guarantee the minimum = *ΣMCR / (Overbooking factor x 0.99)*
- **For brittle traffic**: Guarantee the maximum =*ΣPCR / (Overbooking factor x 0.99)*

The reading rates for each QCP should be set to these values, with connections accepted provided that the total bandwidth allocated to the QCPs is less than the physical link speed. The *overbooking factor* is set per class. For example, an overbooking of 2 would enable twice as many connections to be established. With SVCs this would be set to a value of *1*.

Outstanding issues are the required buffer sizes and discard thresholds. Given the complexity of analysing Internet traffic, and in particular its tendency to show long range dependency, these are non-trivial problem. However, the values used in Figure 6 worked well in this case.

Internet Issues

The discussion here has focused upon ATM layer controls. It is expected (speculated?) that networks of the future will use both ATM and IP, with ATM confined to wide area networks. If the Internet is to be a universal network transport mechanism for all services, it must also find solutions to the above control problems (e.g. [3], [4], [5]). Otherwise real-time services will be directly presented to the ATM layer, with the Internet constrained to carry elastic traffic only. Perhaps key will be the interaction between the two.

It is interesting to compare the concepts presented here with the IETF's work on Differentiated services [5], which also proposes the use of per-class queuing in the network interior (called diffserv domains). But in contrast to ATM's single CLP bit it proposes three levels of priority (drop precedence) for its near-equivalent of elastic traffic, AF classes [14].

CONCLUSIONS

This paper has discussed the issues of controlling overbooking within ATM networks via the appropriate handling of elastic Internet and brittle Video traffic. It is found that:

- MPEG video traffic is sensitive to both loss and delay variation (but not absolute delay) and should be given priority over delay in-sensitive traffic.
- TCP goodput is sensitive to round trip delay, loss, and implementation variations.
- Interactive Television's Control and Boot channel traffic can be mixed with elastic traffic, provided that the ATM layer uses controls (e.g. PPD, EPD, SCD) to ensure that the probability of buffer overflow is low. Otherwise three priority levels are required.
- TCP benefits from lower layers that discard cells *intelligently*. In particular PPD greatly improves performance. Additional benefits of EPD are expected to be small (but this may be due to the small MTU).

- Overall TCP *goodput* under three modeled schemes are very similar. However relying on UBR to interact with TCP is crude, and uses network resources (i.e. buffers) inefficiently. GFR and VBR.3 are equally good, but other research [11] suggests that GFR is fairer.
- Given the good performance of GFR and VBR.3, it is unlikely that the extra complexity of implementing ABR or per-VC queuing would be worth it.
- The QCP algorithm that guarantees some minimum bandwidth to lower priority traffic can prevent over-booked higher priority traffic from denying them service completely.
- A simple CAC algorithm is proposed for the QCP case, although ascertaining optimum buffer lengths and cell discard thresholds remains an open issue.
- Using VPs to partition network can reduce interaction between service providers' traffic.

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