Buffer and Bandwidth Management for the Expedited Forwarding Traffic Class in Differentiated Services Networks

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Abstract: This paper addresses issues related to buffer and bandwidth management for the Expedited Forwarding (EF) traffic class in IP Differentiated Services (DiffServ) networks. We are concerned with edge nodes connecting a number of end-users with premium traffic contracts to the core network and we examine buffer requirements in those nodes. Our results suggest that even in the case of EF traffic class, an amount buffering is essential to provide certain levels of Bit Error Rate (BER) while achieving good utilisation of the existing bandwidth resources; this is in contrast to the current belief that a buffer size of 1 or 2 packets should be enough for the EF traffic class.

1. Introduction

In Differentiated Services (DiffServ) networks, the Expedited Forwarding (EF) traffic class will be able to provide low loss, delay and jitter guarantees in order to meet the stringent requirements of real time services, such as Voice over IP (VoIP), Videoconferencing and Video on Demand (VoD). The Assured Forwarding (AF) traffic class will provide more "soft" guarantees of qualitative nature in comparison to the "hard" quantitative guarantees of the EF traffic class. In order for such guarantees to be supported, the following two steps are necessary in terms of bandwidth management and admission control:

a. A proportion of link capacity should be reserved for use by EF and AF traffic and these proportions should remain allocated to those traffic classes even during congestion. In order to estimate those proportions, we need to know the bandwidth demands of EF and AF traffic. This can be done through traffic forecasting and through the concept of *effective bandwidth*. The rest of the link capacity will be allocated to Best Effort (BE) traffic. That way it is guaranteed that EF and AF traffic will not starve the lower priority BE traffic. It should be stated that traffic demands for BE traffic are virtually impossible to estimate because there are no traffic contracts and, as such,, it is impossible to use traffic descriptors and calculate related effective bandwidth.

b. An admission control scheme should be employed for EF and AF traffic.

2. Bandwidth Model

2.1 Topology and traffic sources

In our scenario we assume that VoIP and Videoconference sources generate EF traffic. For VoIP sources an ON-OFF model with exponential ON and OFF durations can be used [1] [4]. As in [1], it is assumed that ON and OFF periods have an average of 1.004s and 1.587s respectively. This corresponds to a 38.53% talk-spurt cycle, as recommended by ITU-T specification for conversational speech. The peak transmission rate is 64kbps and the average is 24.8kbps. The model that is used for the Videoconference sources is the one proposed in [2]. The average transmission rate is 3.9Mbps and the peak transmission rate is 10.575Mbps. The characteristics of the transmission rate can be approximated either by a Continuous-State Autoregressive Markov Model or by a superposition of 20 ON-OFF Markov sources where each of them has a peak transmission rate 945kbps and average ON and OFF periods of 0.321s and 1.273s respectively. The term high rate source will be used to refer to a Videoconference source and the term low rate source will be used to refer to a VoIP source. The traffic generated by those traffic sources is forwarded to a shared output link (e.g. the link that connects the edge node, where the end-users are connected, to the core network) through individual input links capable to provide zero bit error rate (BER). The objective is to examine whether the available formu las for the effective bandwidth can give a satisfactory approximation of the output link capacity for a given BER. The cases of 10-1000 links carrying traffic generated by low rate sources, 3-50 input links carrying traffic generated by high rate sources and combinations of these have been examined.

2.2 Effective bandwidth

Given the anticipated number of active sources through traffic forecasting, as mentioned in the previous section, and the desired BER, a certain amount of bandwidth needs to be allocated (in our configuration this represents the capacity of the shared output link). In the case of VoIP sources, the term active sources refers in fact to the number of traffic trunks that must be reserved according to the Erlang B formula for a given call blocking probability. The formula that is used for the estimation of the effective bandwidth is the one suggested in [3]. It is based on the assumption that when the effect of statistical multiplexing is of significance, a Gaussian distribution can rather accurately approximate the distribution of the stationary bit rate of aggregated sources.

The estimation of the effective bandwidth in that case does not take into account any existing buffers (bufferless model) and therefore provides an overestimate of the actual demand. The effective bandwidth for N multiplexed sources is:

$$C \approx m + a's \quad \text{with} \quad a' = \sqrt{-2\ln(e) - \ln(2p)} \qquad (1)$$

where $m = \sum_{i=1}^{N} m_i$ is the mean aggregate bit rate, s is the standard deviation of the aggregate bit rate $(s^2 = \sum_{i=1}^{N} s_i^2)$ and e is the allowed Bit Error Rate (BER). You may note that (1) is also computationally simple and depends only on the BER and the means and variances of the bit rates generated by individual sources. There are also other approaches for the calculation of effective bandwidth [4] [5] but they are only valid for Markovian sources, assume the existence of very large buffers [4] in order to function and if the number of individual sources is not small, they give a greater overestimation of the actual demand than the bufferless model since they don't take into account the effect of statistical multiplexing (additive effective bandwidth).

3. Evaluation through Simulation

3.1 Buffer of one packet length

In theory, even with *no* buffers, the above formula should be able to provide the desired BER. In practise, as evaluated through simulation, it turned out that this could not be achieved even by using a buffer of one packet length – we have found that for desired BER 0.1, the actual obtained BER could not be less than 0.18 i.e. 80% more. In addition, the output link utilisation remained relatively low. Even with the output link capacity set equal to the sum of the peak rates of the individual sources and the buffer size set equal to one packet length, no BER less than 0.05 could be achieved, whereas in theory a zero BER should have been possible.

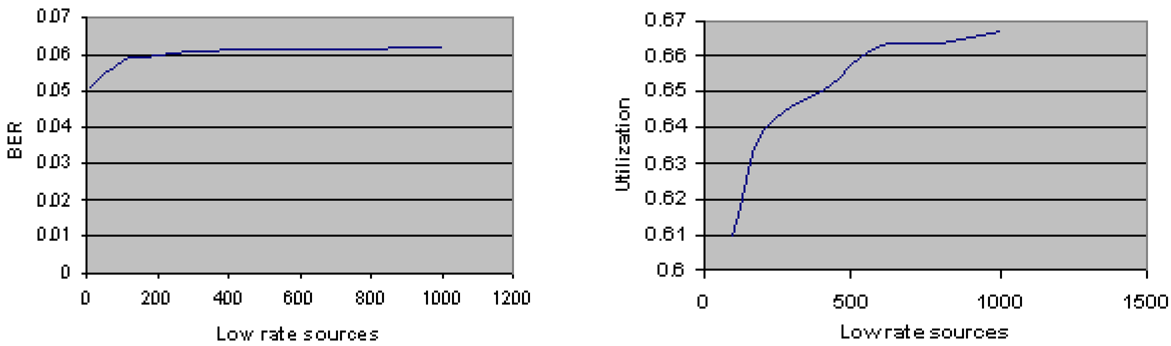

Figure1: BER for output link bandwidth equal to sum of peak rates (left) and utilization of link for output link bandwidth equal to computed effective bandwidth (right).

That is because we have losses due to collisions of packets arriving from different input links simultaneously or within very small time intervals. To demonstrate the nature of this problem one could consider the following example: if there exist two input links carrying traffic generated by two identical constant bit rate sources (one source per link) and the sources happen to be synchronized, in case there is no output buffer, one out of two arriving packets every time will always have to be dropped however large the output link capacity. The effective bandwidth formulas treat traffic as continuous flows (variables). The BER they compute represents the overflow probability in case the sum of the flows (or variables) cannot be fit in the link. In reality, traffic sources are not continuous flows or variables. Traffic arriving from every input link is a stream of individual packets and however large the bandwidth of the output link (one needs to take into account that bandwidth represents strictly serial speed), at each time instance only one packet can be forwarded to the latter. So if there is no buffer (or even 1 packet buffer) and two packets arrive simultaneously or within dt=(packetsize/output link capacity) then definitely one of them will be dropped. This can be a serious problem in cases where the number of input links is large. If the output link connects an edge router to the core network, it could mean that all end-users will suffer a BER >0.05 (provided that the capacity of the output link is set equal to the sum of peak rates) even before their packets have reached the core network.

3.2 Estimation of required buffer size

Since increase of allocated bandwidth is not capable to provide low BER and moreover results to poor utilisation, increasing the buffer size is the other alternative. It should be noted that the low utilization of the output link, even when its capacity is computed using the formula for the bufferless model, combined with the high BER incurred, suggest that if there were no collisions of packets arriving from different input links, the bandwidth computed this way would indeed provide an overestimation of the actual demand. The cases mentioned in previous section were tested for three different BER levels (0.1-0.01-0.001). The output link capacity was set equal to the effective bandwidth (bufferless formula) and the parameter under investigation was the minimum buffer size length so that the des ired BER is achieved. The results indicate that:

a. For input links carrying traffic of low rate sources, the required buffer size increases with the number of the input links (not linearly) for a given BER.

b. For input links carrying traffic of high rate sources, the required buffer size increases with the number of the input links (not linearly) for a given BER.

c. When input links of these two types coexist, the number of links carrying traffic of high rate sources is the one that mainly influences the required buffer size. This is anticipated, since the high rate sources have much larger contribution to the output link capacity.

d. For decreasing BER, the required buffer size increases (for the same number and type of input links).

e. The effective bandwidth estimation gets closer to the actual demand as the number of input sources increases. (This is in contrast to the additive effective bandwidth formulas which need small number of sources and very large buffers in order to provide a relatively good approximation).

f. Since the size of the required buffer is not increasing linearly with the number of sources, the maximum queuing delay decreases with the number of sources. (The computed output link capacity is not increasing linearly as well, but its positive effect on queuing delay compensates for the negative effect caused by the increase of buffer size)

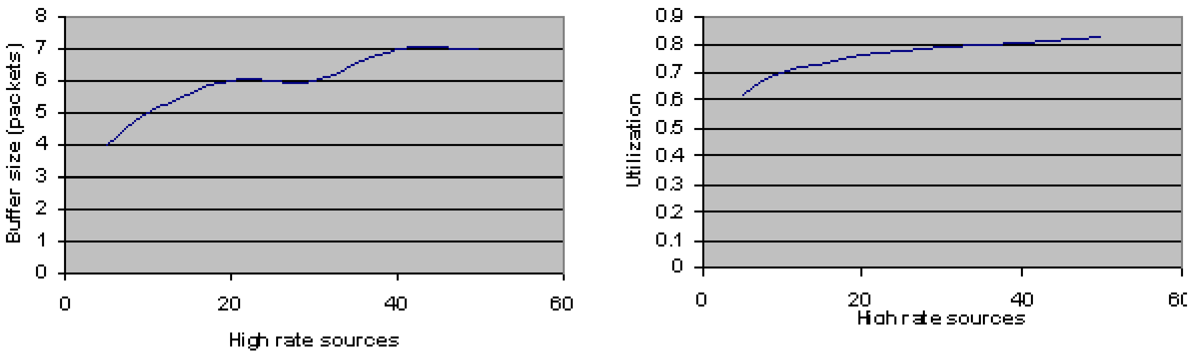

Figure 2: Required buffer size (left) and utilization (right) for BER 0.01.

4. Discussion

In order for low BER to be achieved in nodes with many input links, two measures can be taken: (a) increase the capacity of the output link or (b) increase the buffer size. In the first approach, by allocating bandwidth of the output link equal to the sum of EF traffic carried by input links it turns out that a BER < 0.01can be achieved by using a buffer of four packets length and a BER < 0.1 with a three packets buffer. However in real networks, where lower priority traffic classes will also be employed, if a priority queuing scheme is to be used where lower priority classes won't have access to the bandwidth reserved for EF traffic, that will mean that lower priority classes will have to be deprived of a considerable amount of link bandwidth, which in addition will be exaggeratingly underutilised. If a class-based queuing scheme is to be used, where lower priority classes will be allowed to use the bandwidth reserved for EF traffic, when EF traffic is not using it, then the underutilisation issue is solved. The issue is that this approach cannot be feasible if the sum of peak rates of EF traffic sources exceeds the physical capacity of the output link. Moreover, with respect to admission control, the maximum allowable number of potential subscribers of EF traffic class services needs to be smaller than in the case that the effective bandwidth approach (with larger buffers) is used, since the admission control criterion will be S_{peak} < maximum allowable output link capacity reserved for EF and not Total Effective Bandwidth < maximum allowable output link capacity reserved for EF. Moving from the edge to the core of a network, the effect of the collisions is decreased since the number of input links in a node is smaller and the speed of output links is higher. In real networks, where VoIP calls are expected to arrive as a Poisson process and their maximum allowable simultaneous active population is used for the reservation of bandwidth, BER is expected to be a fraction lower since less or equal-at most- to that number VoIP sources will be active every time instance (provided that the blocking policy of current telephone networks will be employed). In order to examine the influence of the sources' burstiness, the above simulations were run again using the model proposed in [4] for VoIP sources which is less bursty having ON and OFF periods with an average of 0.352s and 0.65s respectively, but no significant differences were observed.

5. Implementation of a more realistic scenario

In a real differentiated services network, where at least both EF and BE traffic classes will be employed, the bufferless formula could be used to allocate link capacity for EF traffic end-users using the required buffer size in order to achieve the desired BER. Since the bandwidth calculated by this formula is an overestimation of the actual demand, the BE traffic class should be allowed to use the bandwidth reserved for the EF traffic class when it is idle in order to improve utilisation. In case there is only a small number of input links carrying EF traffic and the physical capacity of the output link allows it, then allocating bandwidth equal to the sum of peak rates and using small buffers could be considered in order to lower the queuing delay otherwise experienced. A simple scenario was implemented in order to validate the above suggestions. In addition to the EF traffic, BE traffic was added to the output link. The average rate of the BE traffic (generated as a superposition of Pareto ON-OFF sources, which are proven to reproduce the self similar characteristics of TCP controlled traffic) was set equal to the 90% of the output link capacity (representing approximately the proportion of TCP controlled Internet traffic [6] and the number and type of EF traffic sources was chosen accordingly so that their effective bandwidth was equal to the remaining 10% of the output link capacity. A CBQ scheme was employed allowing BE traffic to use excess EF allocated bandwidth. Two configurations were tested and in both cases by using the previously estimated buffer size the desired BER was achieved.

Low rate Sources	High rate Sources	Effective Bandwidth	Desired BER	Estimated buffer size	Achieved BER	Output link Utilization
50	20	~100Mbps	0.01	6 packets	0.0045	0.89
400	5	~40Mbps	0.01	5 packets	0.0029	0.94

Table 1: Validation results for the three configurations.

6. Summary and future work

The simulation results suggest that for network nodes with many input links, even for EF traffic, buffering is needed in order to maintain BER under certain low levels whilst achieving good utilization of the existing bandwidth resources. Moreover, the required buffer size appears not to impose significant burden with respect to delay requirements. The relationship between number of input links, type of traffic carried (aggregated or single source) and desired BER in association with the output buffer size will be further investigated. In addition, more realistic topologies will be tested and potentially some kind of artificial intelligence will be used (neural networks, fuzzy logic) pursuing correct estimation of buffer size even for not tested configurations.

Acknowledgements

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