# Host-Based Service Differentiation with Congestion Feedback

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*Abstract*—**This paper investigates the possibility to differentiate services by using endpoint traffic controls, while sharing a single best-effort network. The starting point is a scheme based on a combination of probe-based admission control in the end-systems for streaming traffic and error correction to isolate different flows. There are inherent limitations in pure host-based control mechanisms and we therefore investigate the improvement that can be achieved when explicit congestion notification is used in routers. In particular we investigate the sensitivity to different active queue management mechanisms and parameter settings. Simulations show that a high ECN marking rate helps to differentiate between flows, but it is important to make sure that the chosen AQM works well for the chosen parameters. This study also shows that a main benefit of ECN is to enable the extension of the scheme to wireless environments.**

***Keywords- admission control; ECN; error correction; quality of service; congestion control***

## I. INTRODUCTION

It is a challenging task for both technical and economical reasons to provide quality of service for conversational services in the Internet. Due to the different uses of IP networks, many specific QoS problems can be solved with tailor-made solutions, for instance in virtual private networks. However, a common solution in the hosts would be very attractive in order to provide maximum flexibility and to open up for new uses of the internet protocol stack. Compared with proposals that are mainly based on implementing new mechanisms in the routers, a new host protocol would give more control to the application developers. Just as the wide spread of the TCP/IP protocol stack has encouraged the development of new types of services, a uniformly available quality of service mechanism in the hosts could help in the development of streaming and real-time services on the Internet.

The paper takes its starting point in previously presented work where a scheme was devised to differentiate between streaming and elastic sessions in a single-class network [9, 16, 17]. The main principle is to use probe-based admission control for streaming sessions and make them share a best-effort network fairly with TCP traffic. The problem of serving streaming sessions with different requirements on delay, sending rate and loss rate is addressed by adding forward error correction (FEC) (e.g. Reed-Solomon codes) and by using the notion of TCP fairness to define admission thresholds for the streaming sessions. The admission threshold is set such that a streaming session is admitted if a TCP session over the same path would get a higher average throughput than the peak rate of the streaming session. This requires estimation of the loss rate and the round-trip time (RTT) of the path in order to calculate the equivalent TCP throughput for the path. One of the main problems pointed out in our previous work is that TCP fairness is not an appropriate goal for streaming sessions, since the admission will depend unduly on parameters such as the packet size and the RTT. We therefore consider other goals where flows can be prioritized based on criteria, such as the willingness to pay. This follows work by other authors, where it is shown how network providers can implement pricing schemes based on congestion notifications [3, 8]. Starting from our previously proposed host-based service differentiation it would only require changes in the admission policies of the hosts to take into account also explicit congestion notifications (ECN) and to work in such a congestion pricing framework. These extensions to the policies are presented in this paper.

The deployment of quality of service schemes in the Internet is difficult since it is not a single network where a common policy can be enforced. Due to the differing goals and strategies of ISPs, it is not feasible to deploy schemes that would require upgrades of the whole Internet at once, or that would require all routers on a path to use the same policies. Therefore, we study the behavior of probe-based admission control on paths where different parts have different properties, such as wired and wireless parts or routers with and without ECN. Our results show that it is possible to get good quality of service in this case as well, when the policies in the hosts are properly chosen. However, the admission control is quite sensitive to the AQM parameters.

The main contribution of this paper is the study of the addition of explicit congestion signaling to the host-based differentiated services scheme that we proposed in previous work. There are two major gains: first, the estimation of the state of the network is easier due to the increased signaling and, second, it is to a larger extent possible to differentiate between losses due to noise and congestion.

In Section II we describe related work and its relation to this study. Section III describes the protocol and contains simulation results regarding the sensitivity to different parameters in the AQM and admission policies. In Section IV

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the use of ECN for differentiation and for heterogeneous networks is investigated. In Section V the conclusions of this study are presented.

## II. RELATED WORK

The principle of probe-base admission control is to send a number of packets, a probe, to estimate the condition of a path before starting a new session over it. Most of the research on probe-based admission control has focused on integrating it into quality of service frameworks such as Diffserv [18]. This means that the admission controlled real-time traffic is assumed to have its own service class, which is separated from the other traffic by means of scheduling in the routers. Within the service class, the principle is to use a lower priority for the probes to avoid disturbing the ongoing user traffic.

Breslau et al. studied several fundamental properties of probe-based admission control and provide guidelines to what can and cannot be achieved [2]. Their conclusion is that probe-based admission control can perform almost as well as traditional MBAC with measurements and decisions performed by the routers. However, they also show that there are several limitations with probe-based admission control schemes. One is the difficulty to isolate different probe-based admission control controlled flows from one another, which makes it hard to serve users with non-similar requirements on rate and loss. This problem is addressed by using FEC in our proposed scheme.

Another issue pointed out by Breslau et al. is the need for long probes to estimate the loss rate. Although this issue might be exaggerated—a probe last only a small fraction of a streaming flow—this can be mitigated by the use of early congestion notification (ECN), which has mainly been studied in combination with TCP [20]. ECN uses three bits in the IP header to convey congestion information, the first bit indicates that the end host is capable of responding to ECN marks, the second bit is set by a router if it is congested and the third bit is an echo bit which is set by the receiver to notify the sender that a congestion mark has been received. Another possibility that is currently studied in the IETF transport area working group is to use a multilevel congestion notification for admission control purposes. This is not considered here since we prefer to remain with the most common ECN semantics.

Our assumption for this work is that some of the routers along a path use some form of active queue management (AQM) to determine which packets to loose or mark. The most well-known AQM method is random early detection (RED) [5]. The router keeps track of the average queue length and when it is between an upper and a lower threshold a packet is dropped with a probability that increases linearly with increasing queue length. Actually there are many more parameters that can be modified and several studies have been made on how to choose the parameters statically or adaptively [21]. To reduce the feedback latency another proposal is to use a proportional-integral controller, which uses the momentary queue size rather than an average value; it includes an integral factor to regulate the steady-state queue length [10]. Random early marking (REM) uses both the queue length and the aggregate input rate as a congestion measure, and the marking rate is an exponentially increasing function of the congestion measure [1]. A different approach is to use a virtual queue with lower service rate than the actual queue, and mark packets when the virtual queue is full. This gives an earlier feedback signal than relying on the level in the real queue [13]. This has been identified as a reason for better stability of virtual queuing schemes than real-queues [14], but the results from different studies are partly ambiguous [6]. A problem for AQM mechanisms is to find appropriate parameter settings, and adaptive algorithms for parameter settings are therefore used in many of the AQM mechanisms [13, 21].

Probe-based admission control in combination with ECN has been studied by T. Kelly in [12]. The main use of ECN in that paper was to reduce the setup time, which is possible due to the increased feedback compared with tail dropping queues. The studied marking scheme was the virtual queue proposed by Gibbens and F. Kelly [8]. Also Key and Massoulié investigated the possibility to reduce the probing time with ECN [14]. The authors distinguish between large and small scale systems and find that there is a need for different probing strategies. The efficiency of different AQM schemes in combination with probing has also been investigated by Ganesh et al [6]. The purpose is to find schemes that work with short probe lengths, and the authors find that AVQ and REM perform very similarly and are both robust and approaches the performance of a centralized admission control, whereas tail dropping performs significantly worse. In this paper we also make a comparison of AQM schemes, but for a given probing scheme that is designed to work together with TCP, which changes the prerequisites.

F. Kelly studied another framework for probe-based admission control, where congestion marking is added in the routers. This could be used for congestion pricing to provide a proper incentive for the users to reduce the congestion [11]. Mathematical models for investigation of the behavior have been proposed by Gibbens and F. Kelly [7]. The ideas of using congestion marking for pricing has also been further developed in the M3I project, where several models for introducing pricing into the Internet are proposed [3]. This indicates how different business models can be used by operators and how mediators can help in providing services with predictable price and quality to the customers.

This paper extends the previous literature on probe-based admission control with ECN by considering error-prone links and scenarios where not all routers support ECN. These issues are addressed by increasing the robustness by FEC and appropriate admission policies.

## III. AQM AND ECN FOR ADMISSION CONTROL

In this section we evaluate the advantage of using explicit congestion feedback for host-based admission control. We take a practical approach and investigate issues related to existing AQM algorithms and parameter settings.

### A. *Protocol Description*

The protocol used for real-time sessions is based on probe-based admission control, i.e. a number of packets are sent before the actual start of the flow to estimate the characteristics of the path. Here the parameters used to characterize the path are the loss rate and the congestion marking rate, as opposed to loss rate and RTT in previous work [9, 16, 17]. The congestion-marking rate (i.e., share of received packets that are marked) is compared to an admission threshold which is intended to control the traffic in the network by rejecting sessions at times of congestion. The loss rate (i.e., share of sent packets that are lost) is used to determine whether the quality will be good enough for the application; we assume that there is a maximum tolerable loss level for each application. The congestion marking rate is estimated as the average marking rate over the probe duration (analogous for the loss rate) and it is compared to the marking threshold: if the marking rate is too high the flow is blocked. Blocked flows will backoff for an exponentially distributed random time. The expected backoff time increases as $2^n$, where $n$ is the number of failed probing attempts, as described in [9]. The maximum number of probing attempts is three. If the marking rate is below the admission threshold but the loss rate is above the loss requirement of the application, the session can add FEC to bring the loss rate down to the required level. The resulting loss rate with FEC is approximated using an assumption of independent losses; it is rounded upwards since this assumption underestimates the actual loss rate after FEC. Due to the low granularity of the FEC parameters this approximation will suffice in most cases.

The amount of redundancy that can be added depends on the estimated marking rate during the probing: When the total sending rate is increased by added redundancy, the marking threshold is reduced by the corresponding percentage. For instance, if the original admission threshold for a flow would be five percent marking, and it would need to add 10 percent of redundancy to get an acceptable loss probability, then the admission threshold would actually be a 5/1.1 percent marking rate. The amount of redundancy is the minimum of what estimated necessary to provide a low enough loss rate. In addition, we put an upper bound on the amount of FEC that can be added to avoid excessive use of FEC, this is normally 20 percent of the data rate. If the loss requirement cannot be fulfilled under these constraints on the FEC parameters the flow is rejected. For paths without ECN-enabled routers the only limitation is the acceptable loss rate and the upper bound on the added redundancy. The algorithm for computing the FEC parameter setting and the admission control is described in the pseudo-code.

In order to assure reasonable chances of achieving low enough loss rate a margin is added to the loss requirement, which means that $T_l$ is reduced [18]. This is made using an assumption that the estimate from the probe measurement of the packet loss probability along the path would be normally distributed. The margin is computed as the confidence interval that limits the probability of exceeding the loss rate to 25 percent. Note that the empirical probability may differ from this since the distribution is only approximately normally distributed. There is no added margin for the marking threshold; the margins are only to limit the number of admitted sessions that experience too high loss rates.

```
Procedure after probing

  exp = 1                                   % BW expansion
  $P_{Rloss} = 0$                           % Loss after FEC
  $P_{loss}$ = #lost / #sent                % Loss estimation
  $P_{mark}$ = #marked / #received          % Marking estimation
  if $P_{mark} > T_m$                       % $T_m$=admission thr.
      backoff
  else if $P_{loss} < T_l$                  % $T_l$=application req.
      accept
  else
      $P_{Rloss} = f(P_{loss}, exp, block)$       % Loss after FEC rec.
      while $P_{Rloss} > T_l$
          # redundancy packet++
          exp= 1+ #redundancy packets / FEC block length
          $P_{Rloss} = f(P_{loss},exp,block)$     % Loss after FEC rec.
      if $P_{mark} < T_m$ / exp
          accept
      else
          backoff
```

### B. *Simulation Parameters*

All simulations in the paper are made using NS-2 [19]. To get reliable results each scenario has been run for 2000 simulated seconds 10 times and to avoid transient effects the first 100 seconds of each simulation run has been discarded following visual inspection of the results.The accuracy of the presented results is not explicitly stated for all results for sake of brevity, but it is pointed out where the accuracy is insufficient to draw conclusions. The TCP background traffic consists of file transfers with a lognormal distribution of the file sizes with mean 160 kB and variance 400 kB. The sessions arrive as a Poisson process with intensity depending on the scenario, as described in the corresponding sections. Unless other parameters are stated the TCP version is TCP NewReno, the maximum queue size is 50 packets, the TCP packet size is 576 bytes and the UDP packet size is 188 bytes. The bandwidth of all links is 10 Mb/s and the end-to-end delay, excluding queuing, is 50 ms. The UDP sessions have exponentially distributed lengths with average 50 seconds are constant bit rate and use a probing time of 3 seconds, corresponding to 399 packets. This choice is based on results from previous studies [16]. The FEC block size is 20 packets, corresponding to 150 milliseconds delay.

The simulations in Section III use a dumbbell topology with a single bottleneck link. The real-time sessions have a transmission rate of 200 kb/s, and three different classes with loss requirements of maximum 0.5%, 1.0% and 1.5% packet loss respectively.

### C. *Gain from ECN*

With ECN it is possible to get an early warning about congestion before packets are lost. In the case of probe based admission control it also opens the possibility of getting more feedback information compared to when only losses are used as congestion signals. To evaluate the gain of ECN the following three alternatives have been compared:

- Drop-tail router without any ECN marking
- RED without ECN marking
- RED with ECN[1]

By comparing these three alternatives it is possible to evaluate both the effect of the different queue management, i.e. RED instead of drop-tail, and the effect of ECN marking. The admission threshold for the marking rate in the third case is chosen to be identical to the loss requirement of the applications. A dumbbell topology was used and the offered UDP load was varied from 45 percent to 135 percent of the available capacity. In addition there is TCP file transfers corresponding to an average load of 1 Mb/s. Hence the dominating part of the traffic is UDP, which makes the result better reflect the interaction between admission control and ECN. Figure 1 shows how these different queuing strategies can control the loss rate as the load increases. It is clear that the drop-tail router has the highest packet loss rate in the network and that ECN further reduces the loss rate compared with RED with packet drops, in particular at high load.

In Figure 2 the average blocking probability for the flows are indicated. It is interesting to note that the drop-tail router actually has a substantially higher blocking probability than the RED queue, but the resulting loss rate is still higher with the drop-tail queue. With RED the losses are less correlated than for a drop-tail queue, therefore the probing works better with RED. The blocking rate is increased further when ECN is used, which is the price of maintaining a very low loss rate. This results in a much lower probability of admitted sessions experiencing higher loss rate than their requirement. For example, approximately one percent of the sessions experience too high loss at 90% load in the case with RED and ECN, compared to over forty percent of the sessions when ECN signaling is not used.

We also remark that none of the TCP sessions are starved, i.e. the same amount of TCP traffic is transferred in all scenarios. However, the finishing time of the TCP sessions may vary. Based on this we may conclude that a plain drop-tail is not a good solution, and that ECN can help in maintaining low loss rates in the network.

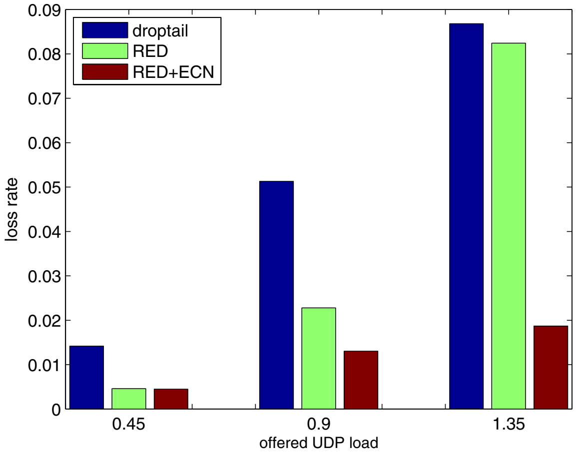

**Figure 1.** The loss rate for different queue types at different levels of offered UDP traffic. Note that the loss rate after FEC recovery is lower so that almost all of the sessions meet their loss requirements.

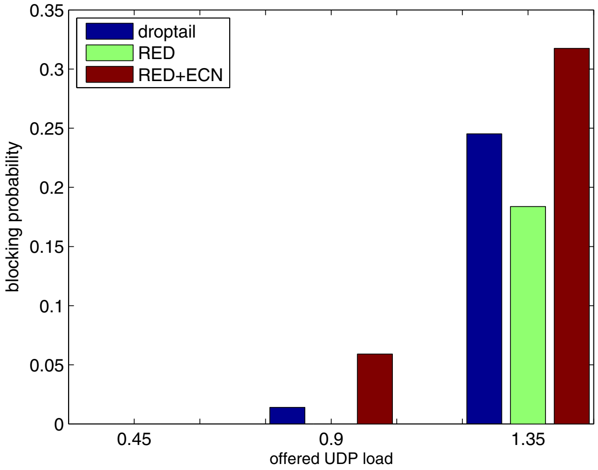

**Figure 2.** Blocking probability for different levels of offered load, RED without ECN is most efficient in utilizing the network.

### D. *Sensitivity to different AQM mechanisms*

Since there are several proposed AQM algorithms, we investigate which ones are useful in combination with probe-based admission control. The purpose is to investigate how sensitive the probe-based admission control is to the AQM scheme, and to choose suitable representative AQM schemes for the following simulation study. These simulations use a mix of UDP sessions with three different admission thresholds: 2, 4 and 6 percent marking rate. In addition there are TCP file transfers corresponding to 1 Mb/s and 10 always-on TCP flows.

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[1] The RED parameters are the same as the static case in the next section

**Table 1.** The marking rates and loss rates for different AQM schemes with ECN marking, all results are in percent. The utilization is here defined as good-put as a percentage of the link capacity, i.e. it does not include probe packets, redundancy or retransmitted packets, therefore the numbers may seem a bit low. The average offered UDP load is in percent of the bottleneck link capacity.

| Offered UDP load | 60 | | | 120 | | | 180 | | |
|---|---|---|---|---|---|---|---|---|---|
| | Loss | Mark | Util. | Loss | Mark | Util. | Loss | Mark | Util. |
| RED (adapt.) | 0.9 | 0.18 | 90.0 | 32.2 | 0.0 | 80.9 | 33.8 | 0.0 | 80.9 |
| RED (static) | 0.45 | 1.60 | 89.2 | 1.20 | 3.62 | 88.1 | 1.48 | 4.60 | 84.1 |
| PI | 6.1 | 0.42 | 89.7 | 31.7 | 0.01 | 79.3 | 33.4 | 0.01 | 79.9 |
| REM | 0.0 | 6.48 | 46.1 | 0.03 | 9.48 | 33.3 | 0.17 | 21.81 | 18.1 |
| AVQ | 0.0 | 0.81 | 94.8 | 0.02 | 4.86 | 91.0 | 0.10 | 6.94 | 82.1 |

A problem is that some of the AQM schemes stop marking packets when they get overloaded and instead drop packets. This is based on an assumption that the congestion control protocols (usually TCP) will react also to losses. However, our protocol is designed to work in environments where losses could be also caused by bit errors; therefore it is not appropriate to react to the losses as if they are caused by congestion when explicit congestion notification is available. If the congestion measure (e.g. the average queue length) is pushed above the threshold where the AQM algorithm stops marking only the loss requirement is limiting the admission, which leads to high loss rates in the network when FEC is used. In Table 1 we can see that adaptive RED and PI have this problem, at least with the parameters we have used. (For adaptive RED the parameters are set as described in [19, 21]). With careful engineering of the parameters it could be possible to avoid this, as can be seen from the results for RED with static parameters. The parameters are set so that the marking rate increases from 0 at the lower threshold of 10 packets up to 0.5 at the maximum threshold of 50 packets, which is also equal to the queue size. Since it is only the probing sessions and not the already accepted ones that react to the marks, the aggregate reaction to congestion notifications is not as dynamic as when all sessions use TCP congestion control and a faster increase in the marking rate can therefore help.

Since the simulations show that AVQ works well in combination with probe-based admission control we will use AVQ or static RED in the following simulations. The virtual queue scheme reacts to congestion before the queue starts to fill up, which can also be an advantage compared to RED that only reacts when the actual queue is starting to build up. The service rate of the virtual queue is 90 percent of the real queue in all simulations. At moderate overload this is sufficient to keep the network running without any losses. REM provides a low loss rate and a significantly higher average marking rate than the other alternatives, but the utilization of the network is very low, therefore it is not a good alternative. The reason is the rapid increase in marking rate when the congestion measure increases, which does not work well with either the admission control policy or the TCP congestion control. With a different admission policy in the host, it could be possible to make an admission decision based on a short probe, as described in [6]. However, it would probably not work well in combination with TCP.

From these simulations we can conclude that the probe-based admission control is quite sensitive to different AQM schemes. The reason for this is that most AQM schemes have been designed under the assumption of TCP congestion control in the end-systems (illustrates the possible complications of cross-layer design). Therefore, it is necessary to either jointly design the algorithms and parameters in the end-systems and the routers, or to choose an AQM scheme that works well in combination with the probe-based admission control. In the rest of the paper we will use either AVQ or RED where the parameters have been set statically to ensure stability.

### *E. Admission Policy*

Next we investigate various admission strategies, under the assumption that there are applications with differing requirements on the acceptable loss rates. One can think of different approaches for setting the ECN parameters in order to achieve goals like admitting sessions based on either priorization of different users or applications or some fairness criterion. Since we are considering the possibility of using a single class within the network where the admission controlled traffic also has to co-exist with TCP traffic, the ECN marking policies are adapted to work with TCP. Of course, the marking rate depends on the load, but the AQM scheme will determine how rapidly the ECN marking rate will increase as a function of the load, and hence a hypothetical equilibrium marking rate if the offered traffic would be constant. For the admission control it is an advantage to have high marking probabilities which makes it easier to estimate the marking rate with good accuracy during a short probe period. However, a too high marking rate is negative for TCP since the congestion control reacts drastically as soon as it encounters an ECN mark; therefore the marking rate should not increase to rapidly since that would lead to a low TCP throughput. It is difficult to give any general guidelines as to what level does not starve TCP, since this depends strongly on parameters such as the round trip time of the TCP flows, but we keep the admission thresholds at a few percent marking probability to avoid starving long-distance TCP sessions. We first look at how to set the parameters when RED with ECN-marking is used for the active queue management.

We have considered the following policies, with the motivations for each one:

1. Set the marking threshold equal to the loss requirement of each application. This corresponds to TCP congestion control reaction to ECN marks, i.e. they should be treated equally to losses.
2. Use an identical marking threshold, regardless of the loss requirement of each application. This would be fairer since there is no reason to differentiate the admission of sessions with different loss requirements as long as they get their required quality levels. These levels can then be provided using some margin to assure a low probability of exceeding the requirement and/or by applying FEC.

For both of the cases above we can consider the possibility to set the marking threshold higher to allow more accurate estimates of the marking rate. In the first case this would mean that we set the threshold as a multiple of the loss requirement. In the second case the threshold is arbitrary to begin with, and therefore it could simply be raised to a higher rate.

For these simulations an offered load of 1 Mb/s TCP traffic and 13.5 Mb/s UDP traffic is used and the results are presented in Table 2. The results show that it is possible to maintain a lower loss rate when the marking threshold is higher. The standard deviations of the loss rates are less than 0.05 percent; for the blocking rate, the standard deviations of the results are around 2 percent so we can not make any conclusions about the average blocking rates. However, when the threshold is a multiple of the loss requirements (column one and two) the blocking probability differs substantially between the classes, as explained in the next section. The TCP flows are not starved in any of the cases, i.e. all the files are completely transferred.

**Table 2.** The percentage of lost packets and blocked sessions for different parameter settings in the hosts. The first column is for a marking threshold equal to the loss requirement, the second for threshold equal to three times the loss requirement and in the last two columns all sessions have identical admission thresholds.

| Policy | $T_m = T_l$ | $T_m = 3T_l$ | $T_m = 1\%$ | $T_m = 3\%$ |
|---|---|---|---|---|
| Loss | 0.21 | 0.13 | 0.19 | 0.12 |
| Blocking | 39.3 | 39.9 | 39.6 | 39.9 |

## IV. HETEROGENEOUS PATHS

In this section we investigate how ECN can help in enabling service differentiation and support in more complex scenarios where the paths contain routers or links with different properties. The purpose is to test how robust the protocol is to variations in the path characteristics.

### A. *ECN for differentiation*

We are investigating the possibility to use ECN marking to differentiate flows with respect to blocking, as suggested in [8]. This requires specific admission thresholds for each flow or class of flows; therefore it is important to investigate how these should be set. We compare three policies that are based on differing marking thresholds, where the highest marking threshold has the highest priority and the lowest threshold the lowest priority. All sessions are assumed to have a requirement of a maximum loss rate of 0.5 percent, and the admission thresholds for the sessions are in the first case (×1 in Figure 3) 0.5%, 1.0% and 1.5%, in the second case (×2 in Figure 3) 1.0%, 2.0%, 3.0% and in the last case (×3 in Figure 3) 1.5%, 3.0% and 4.5%.

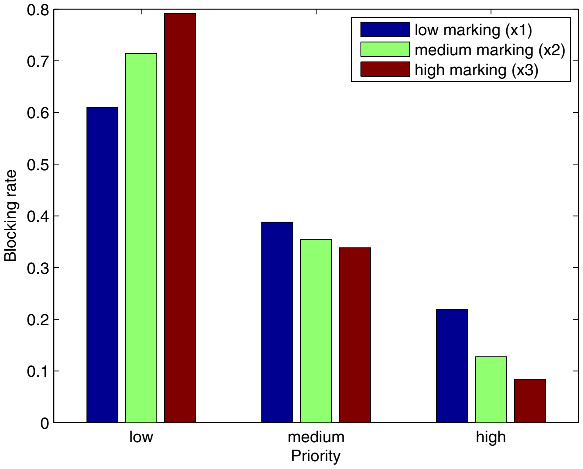

**Figure 3.** The priorities of the different flows determine which admission threshold they should use. The resulting differentiation is more pronounced for the higher marking rates.

The results in Figure 3 are from a scenario where the offered UDP load is 13.5 Mb/s and the offered TCP traffic is 1 Mb/s and the link capacity 10 Mb/s. For all the cases the loss rates are under 0.2%, and approximately the same for the different marking policies. However, the differentiation is more pronounced for higher marking rates. In the previous section we could see that a higher marking rate tends to reduce the loss rate, but a more pronounced differentiation means that the prioritized sessions can be admitted at times of higher congestion, which has an opposite effect of increasing the loss rate, this can explain why the loss rates are not lower for higher marking rates. Since the loss rates are low, the amount of added redundancy is less than 0.2% of the data rate in all cases, and more than 99.5% of the admitted sessions experience a lower average loss rate than their requirement. Hence, there is no statistically relevant difference between the different marking regimes in terms of providing sufficiently low loss rates. The standard deviation in the blocking rate, estimated from the different simulation runs in the plots is less than three percentage points for all results.

The average marking rates are 2.4%, 3.4% and 4.4%, respectively for the different policies. The absolute differences between the marking thresholds of different priorities are of course larger when the average marking rate is higher, which makes the differentiation more accurate. The conclusion from this experiment is that higher marking thresholds are beneficial for differentiation of multiple priorities.

### *B. Wireless paths*

We now study a scenario where losses are not exclusively caused by congestion, but where also transmission errors contribute to the losses. This would typically correspond to a wireless network or other networks where the physical channel conditions are challenging. In this case losses are not a good indication of congestion, therefore congestion marking can improve the possibilities of controlling the network efficiently and also to differentiate session loss based on congestion rather than wireless error-induced loss. Another aspect is that FEC might help to provide sufficient quality to admitted sessions. With the guidance of both loss estimates and congestion notifications it is possible to set the amount of redundancy appropriately.

To evaluate the ability to differentiate based on congestion rather than transmission errors we compare sessions with the same bottleneck, but where some of the sessions also have an error-prone link on the path, whereas other sessions only suffer from congestion-induced losses. We compare a scenario where the router uses a RED queue without ECN, with a scenario where the router uses RED with ECN, both with the static parameters described in Section III.D. The traffic in this scenario consists of TCP file transfers corresponding to an average load of 1.35 Mb/s and offered UDP load of 9 Mb/s.

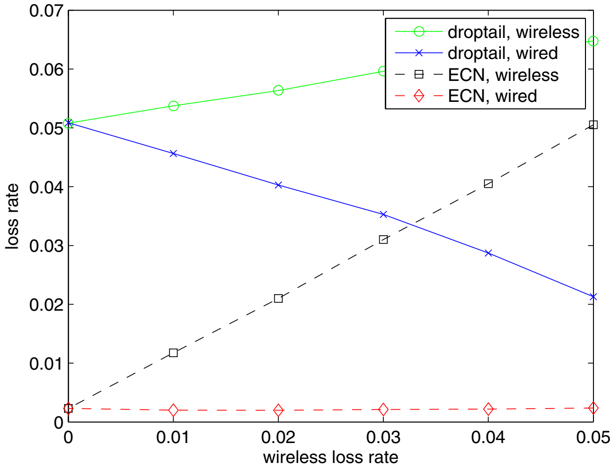

**Figure 4.** For the wired sessions the loss probability is only caused by congestion, for the wireless sessions the total loss probability includes both congestion losses and losses on the wireless link. With ECN there are hardly any losses caused by congestion.

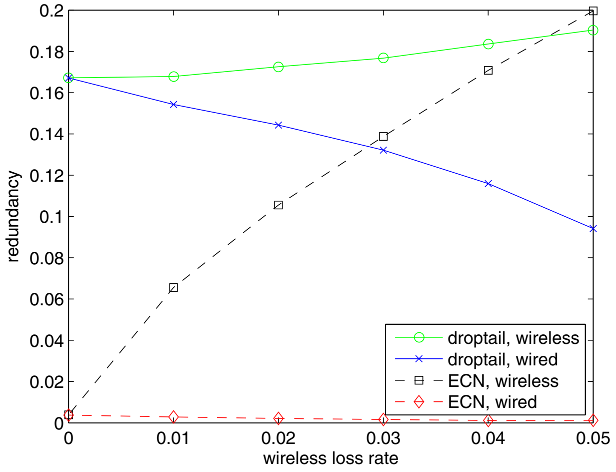

**Figure 5.** The amount of added redundancy as a fraction of the user data is plotted as a function of the noise-induced losses, which means that the total loss rate is as in Figure 4.

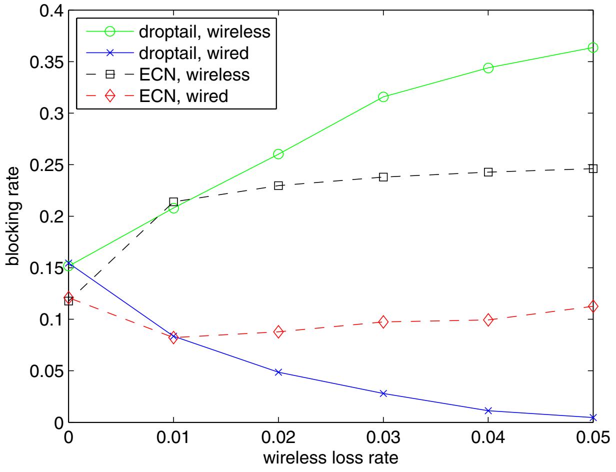

**Figure 6.** The blocking rate as a function of the wireless loss rate shows that when ECN is used the blocking increases only due to the extra redundancy. Without ECN the blocking increases also due to the higher loss rate.

Figure 4 shows how the total loss rate depends on the wireless loss rate. It can be seen that without ECN the load is so high that the loss rate is high even without any wireless losses. When the error-induced losses increase, the loss rate on the wired path drops while the loss rate on the wireless path increases further. Without ECN the only bound on the sending rate is that the redundancy should not exceed 20 percent. As long as that suffices to fulfill the loss requirement of the application the sessions are admitted, which results in congestion. This can be avoided by using policies with an admission threshold also for the loss rate, for example a TCP fairness criterion [16]. However, if that loss threshold would be too low, it would be virtually impossible for sessions to be

**Table 4.** Simulation results with one ECN router and one drop-tail router for the first four columns. The last four columns are the classes that only pass through the ECN router. For both paths the results are the average of 100 kb/s and 200 kb/s traffic types.

| Type/ Path | 1/1 | 2/1 | 3/1 | 4/1 | 1/2 | 2/2 | 3/2 | 4/2 |
|---|---|---|---|---|---|---|---|---|
| Red. | 12.09 | 12.43 | 7.06 | 7.20 | 0.10 | 0.44 | 0.03 | 0.16 |
| Block | 80.09 | 61.05 | 80.40 | 59.32 | 73.47 | 48.49 | 73.82 | 45.00 |
| Fail | 40.18 | 41.34 | 1.25 | 1.46 | 6.96 | 6.04 | 0.42 | 0.28 |

admitted when the wireless losses increase. In Figure 5 we can see that in the non-ECN scenario most of the sessions add almost the maximum amount of redundancy even when there are no error-induced losses. When the router signals congestion using ECN the loss rate of the path follows the wireless loss rate quite well, and the redundancy is only added to recover the wireless losses.

Figure 6 shows that the resulting blocking probability for the different paths behaves differently depending on whether ECN is used or not: Without ECN there is an increasing discrimination against the wireless path with an increasing loss rate, whereas when ECN is deployed, the difference in blocking probability is smaller and does not increase notably with increasing loss rate. The sessions that pass the wireless link need to add some redundancy to recover losses, hence the total traffic is increased and the blocking rate increased. But as long as the added FEC is sufficient to recover the lost packets the flows have a good chance of being admitted.

These simulations show the benefits of having explicit congestion information: the admission decisions are based on the degree of congestion rather than on losses caused by transmission errors so that sessions can be admitted and use FEC to obtain good enough quality.

### *C. Heterogeneous Policies*

The last scenario that we investigate is when not all the routers on the path use the same congestion-marking policy, resulting in various degrees of congestion in different parts of the network. This could correspond to routers belonging to different ISPs that have different policies, for example some could employ congestion pricing as suggested in [3]. To investigate this we simulate a simple scenario with two routers using different policies. In the simulation experiment a user with an ISP implementing ECN marking is considered. A second router on the path uses drop-tail queuing. The second router could be considered as a potentially lossy part of the path, since the admission control will only be based on the loss rate if a flow is not blocked by congestion on the first router. Only if the loss rate on the second router is so high that it cannot be compensated by FEC will the session be blocked due to the congestion in the second router. The UDP traffic consists of the set of eight classes described in Table 3. The offered traffic is on average 8 UDP sessions of each type, with half of the sessions passing only through the first router and the other half passing both the ECN enabled and the best effort router. This corresponds to an offered UDP-load of 19.2 Mb/s on the first router. On the first router there are also 10 constantly sending TCP sessions, and shorter TCP file transfers corresponding to a load of 0.5 Mb/s. On the second router there are 30 constantly sending TCP sessions and 0.5 Mb/s of shorter TCP file transfers.

**Table 3.** Specifications for the different real-time applications, all sources use constant bit rate.

| Type | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 |
|---|---|---|---|---|---|---|---|---|
| Rate (kb/s) | 100 | 100 | 100 | 100 | 200 | 200 | 200 | 200 |
| Loss thr. | 0.5 | 0.5 | 1.5 | 1.5 | 0.5 | 0.5 | 1.5 | 1.5 |
| Mark thr | 1 | 4 | 1 | 4 | 1 | 4 | 1 | 4 |

The simulation gives the following results:

- The loss rate for the admitted flows with a single bottleneck is 0.9 percent and for the path with two bottlenecks it is 3.2 percent.
- The marking rate is of course the same for both paths, since it is only the first router that marks the packets, and the average rate is 4 percent.

In Table 4 we have summarized further results from the simulations, where path 1 passes both routers and path 2 only passes the ECN enabled router. It turns out to be negligible differences between sessions with 100 kb/s and 200 kb/s sending rates; hence these classes have been merged in the table. This indicates that the load added by the probe itself has a negligible impact on the loss rate at the given degree of multiplexing, i.e. a 10 Mb/s link. Table 4 shows how much redundancy each session adds on average, to compensate for the loss rate experienced during the probe. As expected, the sessions with stricter requirements on the loss rate add more redundancy, and the sessions that only pass through the ECN router hardly add any redundancy at all. The table also shows the resulting blocking rates for the different classes. It is clear that the sessions with high marking thresholds are less likely to be blocked than the ones with low marking thresholds. However, the blocking rate does not vary substantially between sessions with differing loss requirement. This indicates that the higher rate due to the added redundancy does not make much difference to the admission threshold, and hence the admission probability.

The last row in Table 4 shows the percentage of admitted sessions that suffer from a higher loss rate than the requirement for each class. It is mainly the sessions with requirement on 0.5 percent loss rate that suffer from too high loss rate, and in particular the ones that pass through two bottlenecks where it is more than 25 percent of the accepted sessions. This indicates that the approximations of the loss rate after FEC recovery and the normal distribution of probe losses are not accurate enough in this case, which is partly explained by the high TCP load on the drop-tail router, which tends to cause high correlation in the loss process.

## V. CONCLUSION

Pure host-system based protocols may be a feasible way to provide service differentiation in the short term, but it has limitations that can be mitigated by support from the network. A simple form of network support is explicit congestion notification, which can be used in combination with different queue management algorithms and congestion control protocols. Since ECN has also been considered as a component in general quality of service architectures it is interesting to investigate how it can improve the performance of our previously proposed host-based protocol.

The results of this study show that:

- The AQM scheme has to be chosen with care, if it has been designed with TCP in mind it may need modifications to work well also with probe-based admission control.

- More feedback information, from a high marking rate with corresponding high admission thresholds can help the performance by providing better possibilities for service differentiation and low loss rates.

- An extension to heterogeneous and wireless environment is enabled by the addition of ECN to the scheme.

As a conclusion, ECN improves the scheme both by providing more feedback information than packet losses alone, but the main benefit is that is enables the hosts to differentiate between losses caused by errors and losses caused by congestion which is becoming more important as wireless access networks are getting more common.

## ACKNOWLEDGMENT

This work was carried out during the tenure of an ERCIM fellowship for the first author.

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