WEIGHT UPDATING METHODS FOR A DYNAMIC WEIGHTED FAIR QUEUING (WFQ) SCHEDULER

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Abstract: This work aims to analyze different weight updating methods for a dynamic Weighted Fair Queuing (WFQ) scheduler providing Quality of Service (QoS) guarantees for the applications of the IST PHOENIX project and for new value-added services in general. Two weight updating methods are investigated in terms of granted delays to concerned service classes and buffer utilization of the related queues at a given IP interface. In particular, a novel weight updating based on the Knightly's theory is proposed. Simulation results have demonstrated that a dynamic WFQ based on either of the weight updating methods can well support a proportional relative model of QoS in a Diff-Serv architecture in an IP-based Next Generation Network. The designed system is extremely simple and effective, and with low computational overhead by employing an innovative technique to evaluate the trend of the entering traffic aggregates, in order to trigger a scheduler's weights updating only when needed.

1 INTRODUCTION

The PHOENIX IST project main goal is to effectively exploit the available bandwidth on wireless links (WLAN, UMTS, 4G…) that is dynamic by nature, providing optimised solutions for multimedia transmission over IP-based wireless networks. To reach this goal, a scheme was developed (PHOENIX) offering the possibility to let the application world (source coding, ciphering) and the transmission world (channel coding, modulation) talk together over an IP protocol stack, so that they can jointly develop an end-to-end optimised wireless communication (termed as JSCC/D compliant system).

However, adequate Quality of Service (QoS) guarantees must be provided along the communication path through-out the network, both for the user data and the control/signalling information to be exchanged.

Advances in computing and network technologies have made it possible to support QoS guarantees in packet switching network. There are two approaches: the Integrated Service (IntServ) (Wroclawski, 1997) and Differentiated Service (DiffServ) (Nichols, 1998)(Blake, 1998). IntServ tried to ensure end-to-end and per flow QoS of data traffics. However, it is quite difficult to implement on a large scale, such as in the Internet. For this reason, DiffServ, which only provides a limited number of service classes, was then proposed. It aggregates the flows with the same QoS requirements, and serves the packets according to predefined traffic contracts. Instead of achieving per-flow QoS, DiffServ provides guarantees at an aggregate level.

Relative and absolute service differentiation are two categories of the DiffServ model. Relative DiffServ ensures the ratios of the quality level between classes, while absolute DiffServ absolute service guarantees. Obviously, the first is more

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straightforward to deploy as well as effective with proper resource provisioning in the network. There is a variant in this category, called proportional service differentiation (Li, 2000).

Weighted Fair Queuing (WFQ) (Keshav, 1997) is a scheduling discipline nowadays widely applied to QoS-enabled routers. In WFQ, single flows or traffic classes are served on the basis of the weight assigned to the related queue. The weight can be dynamically determined (Panza, 2005)(Li, 2000)(Kun, 2000) according to the mutual ratio between QoS factors assigned to each queue in order to achieve QoS guarantees, such as delay and loss, in a proportional manner.

In this paper, two different methods for the weight updating of a dynamic version of the WFQ scheduler are analyzed and compared with respect to the goal of effectively supporting a proportional relative QoS model in a DiffServ architecture. As a new proposal, the latter method is based on the sophisticated Knightly's theory (Knightly, 1998).

Moreover, the novel idea of applying a statistical test allows to save computational power by triggering a weight updating when needed only.

The remainder of this paper is organized as follows. Next section briefly explains the main achievements in the field, concerning dynamic versions of WFQ, as well as Knightly's theory, the basis for the novel weight updating method. Then, a description of the work is carried out, the most relevant simulation results and analyses are presented.

Finally, last section summarizes the main achievements and conclusions.

2 WEIGHTED FAIR QUEUEING DISCIPLINE AND KNIGHTLY'S THEORY

2.1 Dynamic WFQ

A Dynamic WFQ (Panza, 2005)(Li, 2000)(Kun, 2000) is able to dynamically and consistently adapt the queue weights according to the time-variant amount of the incoming traffic and the pre-assigned target QoS.

The fundamental issue is to correlate the burstiness of the traffic with the weight value in order to achieve given delay and loss guarantees.

The measurement of the burstiness of the aggregate entering each queue could be realized by evaluating the resulting buffer dimension (Panza, 2005)(Li, 2000), which is ultimately related to the worst-case delay experienced by packets in the queue.

For what concerns the QoS, we aim to support a proportional relative model. Each queue is assigned a static factor so that the guarantees provided to the set of queues should be in line with the mutual ratio of the said factors. For example, if the queue Qi and Qj have the factors Pi and Pj respectively, with Pj=2Pi, the QoS provided to Qj should be two times better then the one granted to Qi. Hence, no absolute QoS assurances are supported in this case.

As proposed in (Panza, 2005)(Li, 2000), where also preliminary results are reported, if in a given interval Tn, Bn represents the average buffer filling of Qn, which is associated with the factors Pn, the queue weights could be determined by the resolution of a linear system whose equations are of the form:

(Bi/Bj)(Pi/Pj) = (Wi/Wj)

where, Wi and Wj are the weights to be assigned to the queues Qi and Qj, respectively.

2.2 Knightly's Theory

Knightly's theory (Knightly, 1998) is based on a parametric characterization of the traffic, referring to an analytical representation, in terms of a temporal function, of the cumulative arrivals process for the concerned traffic. This can be described by a specific traffic constraint function, i.e. a function of temporal variables that represents an upper-bound envelope. It is a model for the arrival traffic derived from the classic leaky bucket characterization of the arriving cumulative traffic, a monotonically increasing piece-wise function *b(t)* of different slopes. For each slope added to the model, two further parameters are to be introduced (i.e. intersection σ and angular coefficient ρ). For example, a dual leaky bucket model has the following analytical expression

$$b(t)=\min\{\sigma 1+\rho 1\, t, \sigma 2+\rho 2\, t\} \quad (1)$$

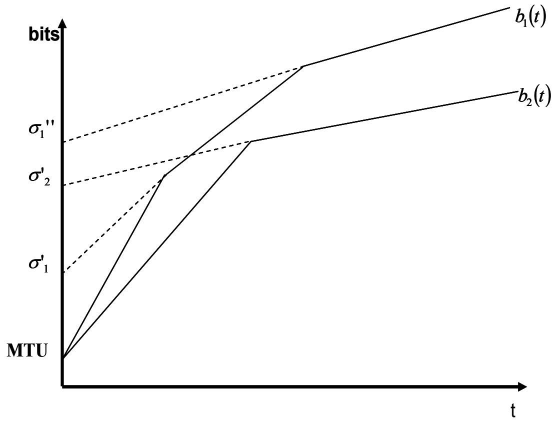


Figure 1: Examples of traffic constraint function of dual and triple leaky bucket type (MTU = Maximum Transfer Unit).

In the case of audio and video traffic flows, as demonstrated by experimental analysis (Knightly, 1997), this function can be simply constituted by two or three slopes (see the figure above for example), respectively. If we denote with $A_j(t)$ the number of bits generated by the source stream j in the time interval [0, t], the function $b_j(t)$ represents the arrival envelopes, and it verifies

$$A_{j(t)} \leq b_j(t),\ \forall\ t>0 \quad (2)$$

Where $A_j(t)$ is a discrete random variable that assumes values in the range [0, $b_j(t)$], whose statistical distribution is unknown.

The proposed model statistically characterizes the traffic by means of the mean rate of the transmission rate variance $RV_j(t)$ in the interval [0, t]. Formally:

$$RV_j(t)= \sigma^2(A_t(t)/t) \qquad \varphi_j=lim_{t\to\infty}(A_j(t)/t) \quad (3)$$

In the hypothesis of a stationary process for the stream j it has been demonstrated that the variance is upper limited, according to the relation:

$$RV_j(t) \leq ((\varphi_j b_j(t))/t)-\varphi_j^2 \quad (4)$$

An admission control algorithm can be applied on a given link to the total traffic envelope obtained as the sum of the envelopes related to the individual traffic streams. Depending on the characterization of the incoming traffic statistic, it is possible to calculate (approximately, (Knightly, 1998)), the probability that the permanence time D of a packet in the considered network node be higher than a given threshold d. This represents the violation probability of the worst-case delay d on the considered link, and is approximated by the following expression:

$$P(D>d) \approx \max_{0\leq t\leq\beta} 1/\sqrt{(2\pi)}\exp[-(C(t+d)-\mu(t))^2/(2\sigma^2(t))] \quad (5)$$

where:

$$\mu(t)=t\cdot\varphi \quad (6)$$

$$\sigma^2(t)=t^2\cdot RV(t) \quad (7)$$

$$\beta=min\{t>0|b_j(t)\leq Ct\} \quad (8)$$

β indicates the time instant in which the service linear function associated with the link capacity C(t) crosses the envelope (modeled with a piece-wise function) of the arrival traffic. This is the same interval in which, if the traffic followed the worst-case, it would see its own packets getting into the buffer and wait for having access to the interface, therefore experiencing delay.

This violation probability is upper limited by the probability of the total backlog, that is the total amount of packets in the time interval d (the predefined limit on the delay), which exceed the maximum number of packets that can be served in the same period.

In (6) an approximation for a large traffic aggregate composed of statistically independent source flows was applied. By the central limit's theorem, it is possible to assume a Gaussian distribution of the difference between the backlog and the available service at the concerned link interface.

3 WORK DESCRIPTION

The work aims to investigate two weight updating methods for a dynamic WFQ scheduler in order to achieve proportional relative QoS guarantees. The former, see subsect. 2.1, based on the resolution of a linear system, while the latter on Knightly's theory, as described in the next subsection.

Comparisons have been carried out in the same working conditions, parameter settings and network scenarios, as well as target QoS assurances.

3.1 A Dynamic WFQ Based on the Knightly's Theory

As stated in the previous paragraphs, the WFQ is the concerned scheduling algorithm, enhanced with an integrated method of measurement and characterization of the traffic aggregates, which in our context corresponds to the different classes of service in a DiffServ-enabled 4G network, in order to tune at run-time the weights of the different queues.

In particular, for each traffic aggregate distinguished by the DSCP value (Nichols, 1998), the parameters of the characterizing traffic constraint functions are measured or indirectly calculated. Those parameters are: the mean rate, the burstiness, the peak rate and the peak rate duration.

Actually, for an initial simulation analysis, the updating mechanism is triggered by the result of an effective statistical test (see subsect. 3.2) applied to the average arrival rate only. The said test is executed for all the incoming active aggregates at the considered IP router interface. It is enough that just one parameter (i.e. the average arrival rate in a queue) presents a variation detected by the test (with the assigned level of reliability), to force an updating of the parameters of all the aggregates and a tuning of the scheduler's weights. If there's no significant change in a measurement period for the functional parameters, the weights remain the same.

The newly proposed updating algorithm based on Knightly's theory has been designed with particular attention to the number of operations to be executed. The goal is to reduce as much as possible the computational overhead, in order to have a solution suitable even for broadband networks, where the high value of the router service rate could entail constraints on the complexity of the deployed mechanisms.

The weight value for each queue is obtained from a system of equations similar to (5) (subsect. 2.2), where the minimum service capacity to be allocated to each queue is the parameter to calculate, that in turns corresponds to the queue weight. Of course, the system also fulfils the condition that the sum of the bandwidth allocated to the queues must be equal to the service rate of the router interface (in (5) indicated as C_{link}).

In practice, we have to deal with an N^{th} order equation, where the parameter *d* to be determined, referring to the equation (5), represents the maximum delay for the N^{th} class (the class granted with the best service in terms of delay). To be noted that the maximum delay of each class can be expressed in terms of the maximum delay of a given class (in this case of the N^{th} one) by means of the assigned QoS factors.

This non-linear equation can be easily solved at run time with an approximated algorithm based on the well-known Newton iterative method that allows to easily obtain the searched solution with a configurable maximum error. More in detail, with an extremely limited number of iterations (actually, never more than 4), it is possible to determine the *d* value with an error lower than a microsecond.

Newton's algorithm needs an initial value for the parameter to be determined. It is represented by the value employed in the previous bandwidth allocation (currently still valid).

Furthermore, we remind that the computational overhead is dramatically reduced by the application of the statistical test that leads to an updating frequency of about 30% of the measurement windows (with a reliability level of 90%).

3.2 An Effective Statistical Test

The t-Student test (Spiegel, 2000) application to a set of empirical observations of rate values is conditioned to the truth of the following assumptions:

I. the samples are independent and with identical distribution (*iid*)
II. the samples belong to a normal statistical distribution.

Actually, by the Central Limit theorem (Spiegel, 2000) the test can be applied to whatever distribution if the observation set is large enough (in practice > 30).

The iid hypothesis is verified when the number of observations is small compared to the population cardinality (Spiegel, 2000).

With a proper measurement window length both the assumption are valid (Cao, 2001)(Cao, 2002).

With a given reliability level, the test is positive when the hypothesis "the traffic (i.e. the mean rate) of a service class has not changed" is to be rejected.

3.3 Traffic Description and Parameter Settings

We considered traffic aggregates composed of both TCP and UDP flows. The latter, H.263 coded video streams at different bit rates, ranging from 64 to 256 Kbit/s as mean value, generated by real traces (H.263/MPEG4) of video streaming and conferencing applications (see the table below for more details). By their nature of typical compressed video flows, the related bit rate is highly variable with a burstiness factor (peak to mean rate ratio) of even 10.

Table 1: Characteristics of a considered H.263 coded video flow.

Encoder Input	176x144 pel (QCIF)
N° of pixels for Chrominance	88
Frame Rate	25 fps
Quantization Parameter	5
Pattern	IBPBPBPBPBP
Integer pel search window	15 pels

Specifically, every traffic aggregate included 3 video streams at 64 Kbit/s and 12 video streams at 256 Kbit/s, for a total average rate (together with the TCP flows) of about 22 Mbit/s.

For a second set of simulations, we also generated H.264 JSCC/D compliant video flows with a bit rate ranging from 60 to 320 Kbit/s. These video flows concerns application scenarios such as video-call, video on demand, video conference and live news. More specifically, the related aggregate included 16 video streams at 60 Kbit/s, 6 video streams at 200 Kbit/s, 4 video streams at 220 Kbit/s and 6 video streams at 320 Kbit/s for a total average rate (together with the TCP flows) of about 22 Mbit/s.

The transmission interface was 100 Mbit/s (e.g. from a FastEthernet network card), where a dynamic WFQ scheduler managed 4 queues, each one fed with a different aggregate and assigned with a QoS factor equals to 1, 2, 3 and 4, respectively (i.e. the traffic in the 4th queue granted with the best service in terms of delay and loss).

For these simulation scenarios, we selected: a time window measurement process of 1.8 s (Panza, 2005), a sampling period of 40 ms and a first-order low pass filter having the zero (termed as k parameter) equal to 0.95 in order to average the gauged values. The measured magnitude is the buffer filling for the weight updating method based on the linear system resolution, whilst it is the average arrival rate for the newly conceived one, based on Knightly's theory.

In the next section, we report and analyze the simulation results, produced by means of OPNET Modeler tool by OPNET Technology Inc..

4 SIMULATION RESULTS

4.1 Updating Method Based on the Linear System

The t-Student test is applied in order to decrease the weight update frequency. Of course, the actual frequency depends on the traffic dynamic; but if the last estimation of the traffic is still valid for all the aggregates in the new time window, the queue weights are not updated, thus saving computational power.

The analysis has been carried out considering as the measured and driving magnitude either the buffer filling or the traffic mean rate, at the decision times.

The minimum required buffer dimension in order to avoid or bound the loss for that queue (when the factors assigned to the other queue remain the same) has been gauged both for the 99th and the 100th percentile of the delay. Of course, the latter value corresponds to absence of loss. Even the average delay in each queue and at the interface as a whole are reported.

Table 2: Minimum buffer sizes for the 99th and 100th percentiles of the delay in the 4 queues.

Queue	99th perc. Min. buffer size (bytes)	100th perc. Min. buffer size (bytes)
1	89673	117963
2	75983	82864
3	59945	74254
4	55677	68501

Table 3: Average delay in each queue and at the interface as a whole.

Queue	Average Delay (ms)	Average Delay at the interface (ms)
1	1.808	0.998
2	0.934	
3	0.701	
4	0.549	

It is worthwhile to point out that the statistical test might not be applied, because the iid hypothesis of the collected samples is not verified a priori (the buffer dimension depends somehow on the story of the system and this could introduce not negligible correlation to be analyzed). Anyway, for the sake of comparison between the two investigated weight updating methods, we collected results with the test application.

The 99th percentiles of the delay for the 4 queues are 11.2, 6.7, 5.4 and 4.4 ms, respectively.

4.2 Updating Method Based on Knightly's Theory

For the sake of comparison, the same sources and link were used for this set of simulations as in the previous analysis (see subsect. 3.3). The driving magnitude is the mean rate and the statistical test is applied.

The graphs below depict the packet-by-packet averaged delay and the CDF (Cumulative Distribution Function) of the delay in the 4 queues.

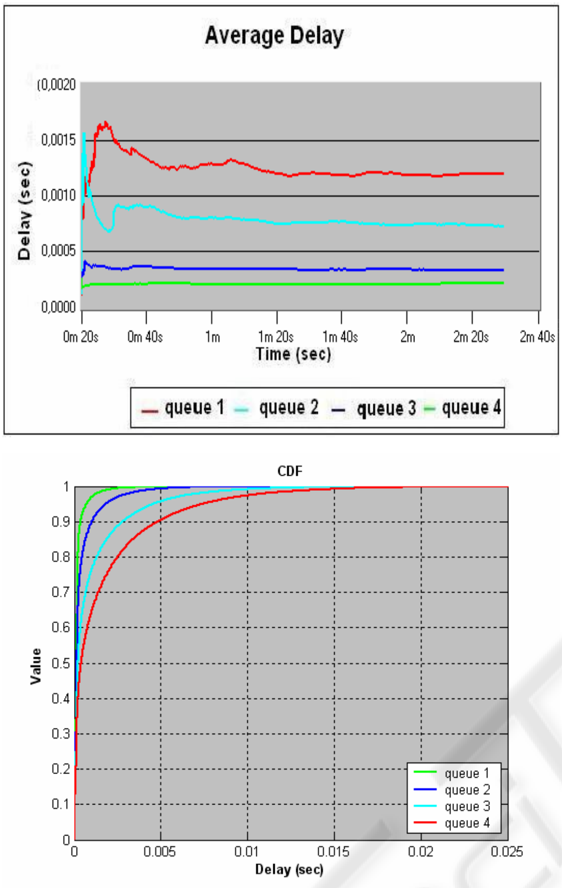


Figure 2: Average delay and CDF of the packet-by-packet delay.

As expected, the delay differentiation is still in line with the assigned quality of service factors to each queue: the higher the factor the lower the delay. This is true even for the packet-by-packet delay when collected in a bucket mode (i.e. every point in the graph corresponds to the mean delay value about a given number of consecutive packets).

The tables below report the relevant figures for the buffer size and the delay for the different queues.

Table 4: Minimum buffer sizes for the 99th and 100th percentiles of the delay in the different 4 queues.

Queue	99th perc. Min. buffer size (bytes)	100th perc. Min. buffer size (bytes)
1	91945	111755
2	64382	108775
3	45734	65190
4	18030	55257

Table 5: Average delay in each queue and at the interface as a whole.

Queue	Average Delay (ms)	Average Delay at the interface (ms)
1	1.772	0.884
2	1.087	
3	0.445	
4	0.221	

The 99th percentiles of the delay are 14.12, 9.58, 4.29 and 1.82 ms, respectively for the 4 queues.

4.3 Results with JSCC/D Compliant Traffic Flows

An analysis for both the scheduling systems was carried out also when one out of the four queues (specifically, the first one) was feed with a traffic aggregate composed by JSCC/D compliant video flows. For an effective comparison, exactly the same parameters were used and results collected. The following tables report the average delay for the four queues and for the interface as a whole.

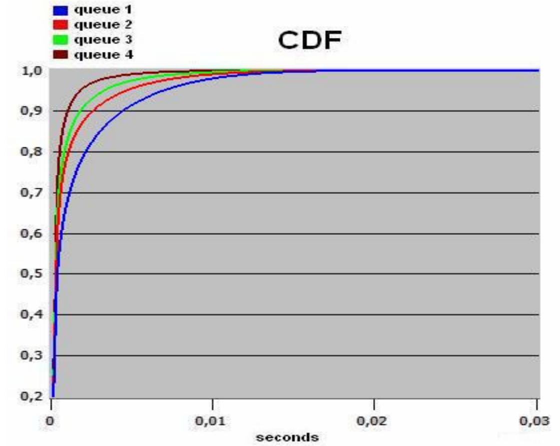
Table 6: Average delay in each queue and at the interface as a whole with the former weight updating algorithm.

Queue	Average Delay (ms)	Average Delay at the interface (ms)
1	1.60	0.98
2	1.02	
3	0.77	
4	0.53	

Table 7: Average delay in each queue and at the interface as a whole with the Knightly's theory based weight updating algorithm.

Queue	Average Delay (ms)	Average Delay at the interface (ms)
1	1.78	0.9
2	1.04	
3	0.53	
4	0.265	

The graphs below depict the CDF of the delay in the 4 queues obtained with the former scheduler and the latter based on Knightly's theory, respectively.



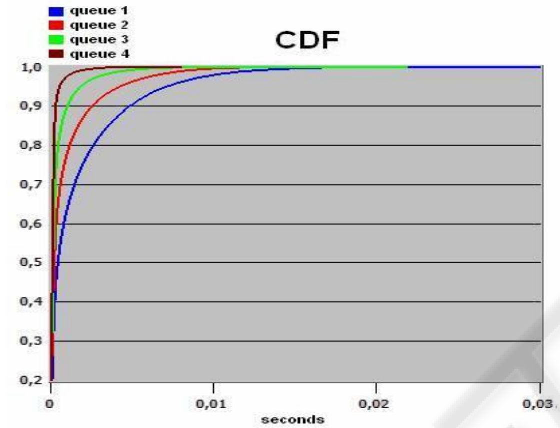


Figure 3: CDF of the delay when the weight updating algorithm is either based on the linear system or Knightly's theory.

The 99th percentiles of the delay are 12.3, 10.0, 7.9 and 5.0 ms, respectively for the 4 queues when the weight updating algorithm is based on the linear system, and 12.57, 8.47, 4.76 and 1.92 ms, when the weight updating is based on Knightly's theory.

4.4 Analysis

The collected results about the buffer dimension and the delay confirm the consistency of the designed systems. A dynamic version of the WFQ scheduler based on either the investigated weight updating methods can provide QoS guarantees according to a proportional relative model. The higher the assigned quality of service factor to a given queue, the lower the delay and the minimum required buffer dimension in order to avoid or bound the loss for that queue (when the factors assigned to the other queue remain the same).

More specifically, we can observe that the traffic aggregates in the 3rd and 4th queues (those assigned with the greatest QoS factors) are granted a lower delay when the updating is based on Knightly's theory, but the contrary happens to the 1st and 2nd queues. Such results are also confirmed by the measurements about the buffer dimension. Therefore, with the novel and more sophisticated method a higher delay differentiation is provided among classes of service.

Looking at the ratio between the figures of the different queues for both the delays and the buffer dimension, we can state that the weight updating based on Knightly's theory more effectively differentiates the QoS provided to the different traffic aggregates, which is more in line with the assigned QoS factors.

The newly proposed weight updating mechanism can provide a better differentiation in terms of granted delays to the various configured classes of service. This means higher performance in terms of supporting a proportional relative model for QoS.

However, also the first mechanism achieves good results, with a lower computational overhead.

5 CONCLUSIONS

In this work, we have investigated and analyzed two different weight updating methods for a dynamic version of the WFQ scheduler, i.e. where the weights are tuned at run-time according to both the actual traffic that enters each queue and the mutual ratio between the QoS factors assigned to the concerned classes of service at the considered IP router interface.

A simulation analysis has been carried out in network scenarios and with configuration parameters suitable for DiffServ-enabled 4G networks.

Novel proposals for an updating method based on the Knightly's theory and the idea of applying a statistical test able to save computational power, have been described and deployed.

By analyzing the collected results about the delays (99th percentile and average value) granted to the various service classes and the buffer dimension (to lose only the packets that experience a delay greater than the 99th percentile or not at all), we can conclude that the weight updating method based on the Knightly's theory is able to provide a better differentiation in terms of granted delays to the various configured classes of service, that is, higher performance in terms of supporting a proportional relative model for QoS. This has been verified with traffic aggregates composed of either traditional (non adaptive) or JSCC/D compliant video streams.

However, also the method built on a simple system of linear equations achieves good results, with a lower computational overhead.

The better selection between them, really depends on the specific network scenario, the required QoS guarantees for the classes of service to be supported and the available computational resources.

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