Comfortable Service: A New Type of Integrated Services Based on Policed Priority Queuing

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Policed Priority Queuing (PPQ), which processes packets in constant time, and statistically guarantee bandwidth, packet loss ratio and queuing delay, is proposed. The behavior of PPQ is analyzed by elementary queuing theory. Comfortable Service (CS), which is based on PPQ and realizes statistical end-to-end QoS guarantee on the Internet, is discussed.

1. Introduction

On the Internet, a flow for which QoS should be guaranteed (QoS flow) and a best-effort flow that does not require QoS guarantee share the same network resources, and they are influenced from each other. Under this circumstance, a mechanism of guaranteeing QoS for the flows requiring QoS is on demand.

Guaranteed service[Sh97] based on Weighted Fair Queuing (WFQ)[Pa92] is proposed as a type of Integrated Services (IntServ). In guaranteed service, it takes $O(\log(n))$ time to insert each input packet into an output queue, or to pick it up from the queue because of sorting, where n is the number of flows that should be treated individually. WFQ strictly guarantees the maximum delay time, however this does not mean all the packets will actually reach the end in the maximum delay time, because the network cannot completely be free from bit errors or packet loss.

In addition, it was considered that QoS flows make the maximum use of bandwidth of links in [Na92]. However on the Internet, best-effort flows do exist too, so the bandwidth that is not consumed by QoS flows can be consumed by best-effort flows. Thus the consumption of all the bandwidth by QoS flows is not taken into account here.

We propose a queuing method, *Policed Priority Queuing (PPQ)*, in which the times to insert each input packet to an output queue and the time to pick it up from the output queue are constant, and the maximum delay time is guaranteed statistically. We then define a new type of IntServ, *Comfortable Service (CS)*, which is based on PPQ and realizes statistical end-to-end QoS guarantee on the Internet.

In Section 2, QoS guarantee in this paper is defined. In Section 3, a router based on PPQ is proposed and explained. In Section 4, a queue at an output interface in a router is analyzed theoretically using M/D/1/K as a queuing model, and the QoS guaranteed by this model is discussed. In Section 5, a simulation verifies the analysis. In Section 6, CS is proposed as a new type of IntServ. In Section 7, our proposals are discussed at various issues.

2. Definition of QoS Guarantee

For the purpose of guaranteeing QoS, which is applicable to real applications, not only bandwidth guarantee, but also specification of a traffic pattern and delay bound guarantee are required.

For example, even if it is just stated that the bandwidth of 10 Mbps is guaranteed, an application cannot judge whether it may send data 1 bit at a time at regular intervals, may send 1280 bytes (10 Kbits) of data at a time with some jitter, or may send 100 Mbits of data at 10 sec intervals. Therefore, a token bucket model is currently used. In addition to bandwidth, traffic pattern and delay bound are specified by the token bucket model.

However, 100% of packet reachability is not yet guaranteed. Transmission error rate or packet loss ratio should be shown for real applications.

In this article, Quality of Services (QoS) guarantee is defined as follows:
When transmitting packets
- with the defined traffic pattern,
- with a specified average bandwidth,

they reach the end
- at the defined probability (at the defined packet loss ratio),
- in a specified maximum delay time.

This QoS guarantee is necessary and sufficient for multimedia applications such as audio and video transmission. Guaranteed service, which does not specify a packet loss ratio, is insufficient for these applications.

3. Structure of Router Based on PPQ for QoS Guarantee

A router based on Policed Priority Queuing (PPQ), which we propose for QoS guarantee, is shown in Fig. 1.

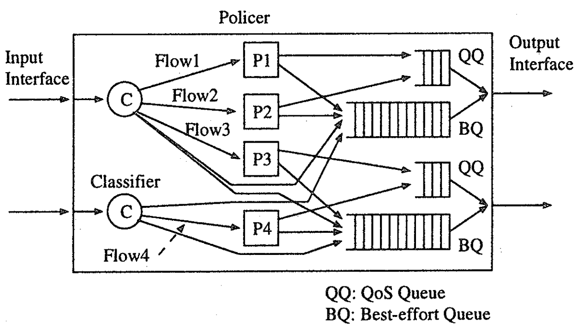

Fig. 1 Structure of a router based on PPQ

The process of packets is as follows:

(1) Packets of multiple flows are input into a router from input interfaces.

(2) By classifiers, each packet is classified into a QoS-guaranteed type (from Flow1 to Flow4 in Fig. 1) or a best-effort type.

(3) An output interface is determined.

  (a) A packet of best-effort flows is queued in the best-effort queue at the output interface determined by packet analysis.

  (b) For packets of a QoS-guaranteed flow, the policer associated with the flow checks if the packets violate the bandwidth requested by the flow in advance. Legal packets are queued in the QoS queue at the output interface, and illegal ones are queued in the best-effort queue at the same output interface.

(4) Packets are transmitted.

The following rules are used when transmitting packets:
- A QoS queue has an absolutely high priority.
- The process is executed in First Come First Serve (FCFS) manner.
- A router transmits packets in a best-effort queue when the corresponding QoS queue becomes empty.
- Packet transmission is not preempted until it is over, when it is once started.

These processes can be executed in constant time except the process at classifiers. By means of hashing or using Content Addressable Memory (CAM), the process at classifiers can also be executed in constant time. Note that since a policer does not configure a queue, the queuing delay does not occur at the policer.

4. Queuing Model and Its Analysis

4.1 Queuing Model M/D/1/K

In PPQ, QoS-guaranteed flows are being processed, almost whenever packets exist in a QoS queue. Thus QoS-guaranteed flows are considered to be independent of best-effort flows. Only the QoS queue is discussed hereafter.

The length of a QoS queue at an output interface is influenced by all the QoS-guaranteed flows directed to the interface. Assuming that each flow is transmitted from the upstream node at exponential distribution intervals (Poisson process), the arrival intervals of mixed flows at the queue in the router also becomes the Poisson process.

We regard the processing time of each packet constant, considering the processing time of the longest packet. We let the MTU be 1280 bytes with taking into account IPv6.

As a result, when let the maximum queue length be K, the behavior of a QoS queue can be treated as M/D/1/K.

4.2 Analysis of Packet Loss Ratio and Delay

Using an M/D/1/K queuing model, we analyze the behavior of a QoS queue and clarify the delay and packet loss ratio.

We let the processing time of one packet be constant in the above discussion. Then, we use packet per second (pps) instead of bit per second (bps) for majoring both router's processing capacity and the rate of arriving packets of QoS-guaranteed flows.

Consider the M/D/1/K queuing model with letting the processing capacity of a certain output interface be μ pps, the packet arrival rate of all the flows arriving at a certain QoS queue be λ pps, and the maximum length of the queue be K. Here $\lambda = \sum_i \lambda_i$, where the arrival rate of flow i is λ_i. The utilization factor of the interface is given by $\rho = \frac{\lambda}{\mu}$.

Generally, it is well known that the probability of queue overflow becomes very low, when ρ is less than 1.

Though the detailed analysis is omitted here, the packet loss ratio L for each ρ and each K is given by Fig. 2. This shows that in the case of the maximum queue length of 26, the packet loss ratio of 10^{-5} is achieved even when 80% of the bandwidth is consumed.

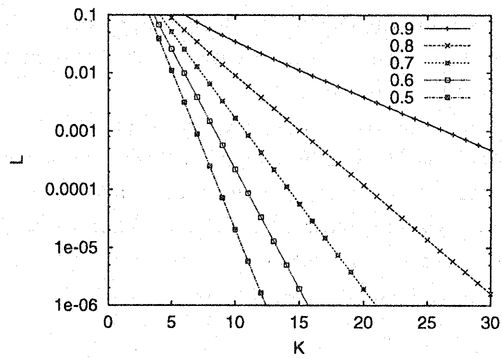

Fig. 2 Relationship between the maximum queue length and the packet loss ratio

4.3 Consideration of Guaranteed QoS

The results obtained Section 4.2 are examined, and how QoS is guaranteed is discussed.

Fig. 2 shows that for example, under the condition that flows are admitted up to 80% of link bandwidth, i.e. $\rho = 0.8$, in order to let packet loss ration be less than 10^{-5}, K should be 26. Therefore, very small packet loss ratio is achieved with short queue length by means of not consuming up the bandwidth for QoS flows.

In other words, under the conditions that the maximum length of the queue is set to 26, and that only the reservations that consume 80% of the bandwidth are admitted, the packet loss ratio of 10^{-5} can be guaranteed for every flow.

Considering use on the Internet, even if only 80% of an output interface is utilized for QoS-guaranteed flows, the rest 20% is consumed by best-effort flows. This means the interface is fully utilized at any time.

The delay a packet experiences is about $\frac{q}{\mu}$, when the queue length is q at the arriving time, so the distribution of the values of delay is approximately calculated from the probability distribution of q.

However, the probability distribution of delay are not needed for QoS guarantee actually, because for achieving the packet loss ratio already mentioned, a receiver has to wait for packet's arriving for the maximum delay time caused by queuing. Then we will discuss just the maximum delay time hereafter.

Since the maximum length of a queue is preset, a packet will be discarded when the queue is full at the arriving time. Thus a packet does never wait for more than the processing time of K packets. As a result, the maximum value of queuing delay is restricted less than $\frac{K}{\mu}$.

Table 1 The maximum value of the queuing delay at a router

Length of queue (Loss ratio)	Processing rate of router (μ)			
	1 Kpps (10 Mbps)	10 Kpps (100Mbps)	100 Kpps (1 Gbps)	1 Mpps (10 Gbps)
5 (10^{-1})	5msec	0.5msec	50 μsec	5 μsec
10 (10^{-2})	10msec	1.0msec	100μsec	10 μsec
16 (10^{-3})	16msec	1.6msec	160μsec	16 μsec
21 (10^{-4})	21msec	2.1msec	210μsec	21 μsec
26 (10^{-5})	26msec	2.6msec	260μsec	26 μsec

The maximum delays are shown in Table 1, when the value of μ is 1 Kpps, 10 Kpss, 100 Kpps, and 1M pps. The packet loss ratio is also shown when letting $\rho = 0.8$. Table 1 shows that the delay is very small and can be ignored against the delay required for actual applications in the case of high-speed networks.☆ For example, when $K = 26$, which achieves the packet loss ratio of 10^{-5}, and $\mu = 1$ Mpps, the queuing delay becomes just 26 μsec by allowing QoS flows up to $\lambda = 800$ Kpps ($\rho = 0.8$). That is, the delay less than 26 μsec is guaranteed with the probability of $1 - 10^{-5}$.

As discussed above, according to PPQ, QoS guarantee can be easily achieved, where the packet loss ratio and the queuing delay are easily estimated.

☆ The delay required for voice communication, which is the most severe for a delay, is less than 200 msec.

5. Evaluation by Simulation

We validate the analysis in the previous section by simulation.

5.1 Network Model

We consider voice transmission as a concrete application, where the average transmission interval is 20 msec, i.e. $\lambda_i = 50$ pps.

Here a sender adds a random delay to avoid unintended synchronization of other senders. The added delay is uniformly distributed within the range of $[0, D_i)$, where the average interval is $D_i = \frac{1}{\lambda_i}$. However, the sending rate does not vary, since the sending time of a certain packet is determined by first adding the average sending interval time to the previous sending time to which a random delay has not been added yet, and then adding a random delay.

For a network model, we consider a model shown in Fig. 3. The rate of the output interface of the router is $\mu = 100$ Kpps (1Gbps), the length of the queue is $K = 26$, and the number of senders is 1600. As a result, $\lambda = 80$Kpps, and $\rho = 0.8$.

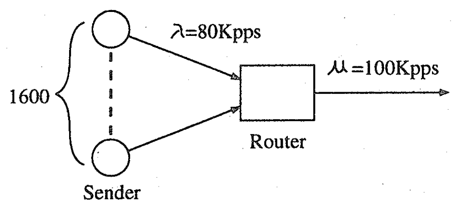

Fig. 3 Simulation network

5.2 Simulation Result

The result simulated during 100 seconds is shown in Fig. 4 and Table 4. Fig. 4 shows that the theoretical values are the approximation of the measured values, and that the measured queue length is shorter than the theoretical queue length in a probability sense. Table 2 also shows that the measured packet loss ratio is smaller.

Consequently, it is shown that the packet loss

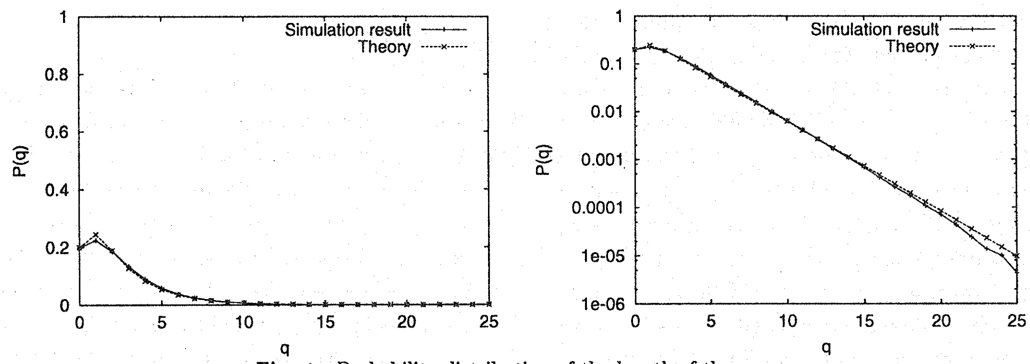

Fig. 4 Probability distribution of the length of the queue

Table 2 The number of arrived/discarded packets and packet loss ratio

	# of arrived packets	# of discarded packets	Packet loss ratio
Measured values	7999194	50	6.25×10^{-6}
Theoretical value	-	-	8.95×10^{-6}

ratio for each flow is kept smaller than the theoretical value during communication, by means of constructing a router based on PPQ.

The reason why the measured value does not completely correspond to the theoretical value is considered as follows; although we assumed the Poisson process as arrival intervals, the measured arrival intervals are not so bursty than the Poisson process. That is, the theoretical values are overestimation. We discuss whether this overestimation cause a problem in Section 7.3.

6. "Comfortable" Service

We define Comfortable Service (CS) as a new service for QoS guarantee. Here comfortable means not completely guaranteed but statistically guaranteed, and comfortable for actual applications.

First, we aim at the packet loss ratio of less than 10^{-5} at a single router. Then, assuming that there is a case that at most 100 routers reside on the path, the end-to-end packet loss ratio becomes less than about 10^{-3}.

Under the condition that the routers on the path of a flow employs PPQ, when a sender sends packets

- at exponential distribution intervals or at less bursty intervals,
- with an average bandwidth of λ_i, which is reserved in advance,

end-to-end QoS guarantee

- with a probability of more than $1 - 10^{-3}$ (at the packet loss ratio of 10^{-3}),
- and with the queuing delay which is the sum of the transmission delays and the queuing delays K/μ of all the routers.

is realized.

Each router can freely set up the values of K and ρ as long as it achieves the packet loss ratio of 10^{-5}. Here the packet loss ratio caused by bit errors on links should be further lower than 10^{-5}.

7. Issues

7.1 Is Packet Loss Ratio of 10^{-3} of Use?

The packet loss ratio of 10^{-3} may seem to be no use for communication requiring a severe loss ratio such as high-quality video transmission, however this is not correct.

The Shanon's theorem shows that channel capacity of a communication channel is determined by bandwidth and error probability of the channel, and that an error probability can be made as small as wished by means of block coding, as long as a transmitting rate is lower than the channel capacity. Instead, the theorem also implies that it takes more time for coding and decoding.

Actually, error correction coding can drasticly reduce an error probability. Since many packets are handled together in strong error correction coding, delay for spooling these packets occurs. However, the delay is just 1/30 seconds even when 100 packets of 1280 bytes are handled together for error correction in a 30 Mbps communication. This can be ignored for humans. In addition, if this is a communication for digital video transmission of 30 frame per second, the delay of 1/30 seconds is originally inevitable for JPEG coding. Error correction coding does not cause a new delay in this case.

Therefore, an error probability can be reduced by sender's elaborate process. It is important to give the defined low error probability before communication, but is not important to provide various error probabilities for various applications.

7.2 Limitation of PPQ

PPQ is valid only on fast links, and not on slow links, since the delay is in inverse proportion to the rate of an output interface. On slow links of less than 10 Mbps, there is a case that WFQ can decrease the maximum delay more.

7.3 Is M/D/1/K Model Appropriate?

Estimation by the M/D/1/K model may be exaggerated, since the actual arrival intervals of a flow tends to be less bursty and more periodic than exponential distribution. In this case, more bandwidth can be assigned to QoS-guaranteed services by means of letting ρ be larger. However on the Internet, it is required to assign bandwidth to best-effort services if any. It is not required to utilize almost all the bandwidth for QoS guarantee.

Consequently, the M/D/1/K model is appropriate and sufficient for QoS guarantee on the Internet in the real world.

8. Conclusion

We proposed Policed Priority Queuing (PPQ), in which packets are processed in constant time, and the packet loss ratio and the maximum delay time is guaranteed statistically. These theoretical values were examined by network simulation, and it was proven that the proposed queuing method is available for real use. We also defined Comfortable Service (CS) provided by PPQ for end-to-end QoS guarantee on the Internet. PPQ and CS are scalable to the number of flows, able to be implemented on routers easily, and suitable to the Internet in the real world.

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