Method of DiffServ-Based Bandwidth-Sharing among Delay-Sensitive Traffic and Loss-Sensitive Traffic in Backbones

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***Abstract**: Real-time and multimedia applications require an end-to-end QoS guarantee, and various types of applications require various QoS conditions. A DiffServ network should guarantee different QoS conditions for different types of communications. In this paper, the effect of traffic control in a DiffServ core network is experimentally evaluated using bursty traffic generated by an MMPP (Markov-Modulated Poisson Process) model. The situation to be simulated is that there are hundreds of conversational video streams that are delay-sensitive and hundreds of streaming videos that are loss-sensitive. If there are bandwidth-sharing queues such as those follow WFQ (Weighted Fair Queuing) in the core nodes and the two types of video traffic are assigned to two of the queues, the requirements of both types of traffic can be satisfied in a better (more efficient) way by assigning a larger weight to the queue for the conversational video. In our experiment using MMPP-based actual traffic and high-end L3 switches, the optimum ratio of the weights was approximately 1.3 when the traffic rates were the same. The optimum weight shares depend on the nature of the traffic, especially the burstiness.*

***Keywords**:* NGN, Next-generation backbone, QoS measurement, QoS guarantee, DiffServ, Bandwidth sharing, WFQ.

1. Introduction

The traffic of real-time and multimedia communication is increasing on the Internet and will be heavily used on next-generation networks (NGNs). Real-time and multimedia applications require an end-to-end QoS guarantee, and various types of applications require various QoS conditions. For example, voice-phones, video-phones, and multi-player game applications are very delay- and jitter-sensitive, and music streaming is loss sensitive.

Differentiated services (DiffServ) [Nic 98] [Car 98] have enabled better QoS for premium traffic in large-scale IP networks by offering a better communication service to more important (premium) traffic than less important (best-effort) traffic. However, conventional DiffServ models are quantity oriented rather than quality oriented, i.e., more important traffic gets more resources and less important traffic gets less resources, because the QoS requirements were quantity oriented, i.e., the only measure was the bandwidth.

Future real-time and multimedia communication services should be different from quantity-oriented communication services because the QoS requirements are more quality oriented, i.e., measured by multi-dimensional parameters such as latency, jitter, and loss ratio for example. A DiffServ network should and can guarantee different QoS conditions for different types of communications, and probably, the network resources can be used more efficiently by a method of quality-oriented QoS guarantee.

In this paper, the author intends to show a method of quality-oriented QoS guarantee and an example of a multi-service core network with a quality-oriented QoS guarantee by experiments using actual network nodes and computer-generated network traffic instead of a mere computer simulation or theory. We generated and used two types of traffic that simulate conversational and streaming video traffic. In Section 2, application-level QoS classes based on the ITU-T Y.1541 and DiffServ classes that correspond to the above QoS classes are described. In Section 3, the assumed network architecture is explained and the router architecture and usage are described. The experimental methods are described in Section 4, and the results are shown in Section 5. Related work is described in Section 6. The conclusion is given in Section 7.

2. QoS Classes

In this section, the application-level QoS classes, the QoS classes in the core DiffServ network, and the mapping between these two classifications are described.

2.1 Application-level classes

Various multimedia applications will require quite different QoS parameters for each communication flow that is used. Those applications can specify various values for parameters such as bandwidth, latency, jitter, or packet-loss ratio. Although each parameter value may be much different from those in other flows, the QoS requirements may be classified into a small number of classes. In ITU-T recommendation Y.1541 [ITU 06], the QoS requirements are classified into eight classes, and a 3GPP document [3GP 06] classifies them into four classes. The latter four classes correspond to the classes in Y.1541, and these four are considered to be typical. Therefore, this classification, as described below, is used in this paper.

- *Conversational class*
  This is the class for bi-directional real-time traffic. The maximum latency must be small (less than 100 ms),[1] and the maximum jitter must be small too (less than 50 ms). Voice- and video-conversation traffic belong to this class.
- *Interactive class*
  This is the class for non-real-time but delay-sensitive traffic. The maximum latency must be small (less than 100 ms), but the maximum jitter is not specified. Control traffic such as SIP messaging belongs to this class.
- *Streaming class*
  This is the class for unidirectional real-time traffic. The maximum latency must be medium (less than 400 ms), and the maximum jitter must be small (less than 50 ms). Video- and voice-streaming traffic belong to this class.[2]
- *Best-effort class*
  This is the class for non-real-time (and delay-insensitive) traffic. Neither the maximum latency nor jitter is defined (not required to be guaranteed).

The maximum loss ratio is assumed to be 10^{-3} for all the classes above.

2.2 DiffServ classes and QoS mapping

The above application-level QoS classes are mapped to DiffServ classes of the core network (mapped to per-hop behaviors (PHBs) [Nic 98]). The DiffServ classes in the usage assumed in this paper and the mapping are explained here.

- *EF (Expedited Forwarding) PHB class*
  This class is for virtual-leased-line (VLL) services with an end-to-end bandwidth guarantee. Voice streams in the Conversation class are mapped to this class. Video streams in the Conversation class (such as video in TV phones) may be included in this class. However, the latter streams are not included in this class in this paper because bursty video traffic may use too much resource if Conversation video traffic is assigned a higher priority than that of Streaming video traffic. In addition, the assignment of a higher priority may degrade the quality of voices.
- *AF (Assured Forwarding) PHB class for conversational UDP communication*
  This is a class for video traffic in the Conversation class and for UDP traffic in the Interactive class. This AF class was originally a loosely assured service, i.e., best-effort service with minimum bandwidth guarantees. Conversation video and UDP control traffic are classified into this class because both require a small latency.
- *AF PHB class for streaming*
  This is a class for traffic in the Streaming class. Streaming video traffic is classified into this class.
- *DF (Default Forwarding) PHB class for communication with no QoS specification*
  This is the class for the best-effort service. Best-effort-class traffic is mapped to this class.

3. Traffic Control in DiffServ Networks

The structure of networks, the structure of routers, and the method of controlling the traffic of classes described in the previous section are explained in this section.

3.1 Assumed network architecture

The network structure assumed in this paper is shown in **Figure 1**. The core network is an IP network that consists of a hundred or more edge routers, ten or more core nodes (routers or switches), and 10-Gbps links with 1000 or more traffic flows between the core nodes. The core network is controlled by one or more policy servers. There are two or more edge networks, i.e., LANs or access networks, connected to the core network.

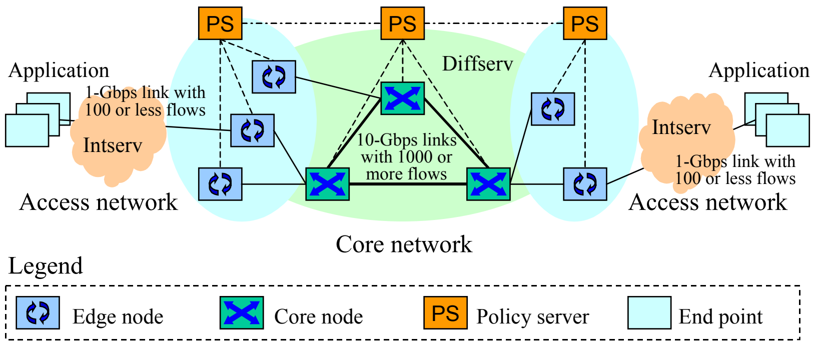


Figure 1. Assumed network architecture

3.2 Assumed core-node architecture and queue usage

The core nodes are assumed to have at least four queues for each outbound network interface (See **Figure 2**). One of them is a higher-priority queue (e.g., a queue following low-latency queuing (LLQ)) and others are lower-priority bandwidth-sharing queues (e.g., queues following (class-based) weighted fair queuing (WFQ)). Commercially available backbone routers such as Cisco's, Juniper's, or Alaxala's have this type of queue sets.

The higher-priority queue is used for the EF class and the bandwidth-sharing queues are used for the AF and DF classes. Very small but nonzero weight should be allocated to the DF class, and the remainder of the weight should be allocated to the AF class. For example, 2% of the total weight can be allocated to the DF class and 98% can be shared by the AF classes. The 98% is the maximum percentage of traf-

[1] The maximum latency is 80 ms in the 3GPP document, but it is 100 ms in the corresponding class of Y.1541.
[2] The maximum jitter may be larger for streaming if a jitter buffer is available.

fic that the AF classes are permitted to use. However, the actual traffic ratios may be less than the allocated percentages. Therefore, if the traffic ratio of the AF classes is less, the DF traffic can use more than 2% of the bandwidth allocated to the bandwidth-sharing queues. The buffer sizes are the same for all the classes used for the measurement.[1]

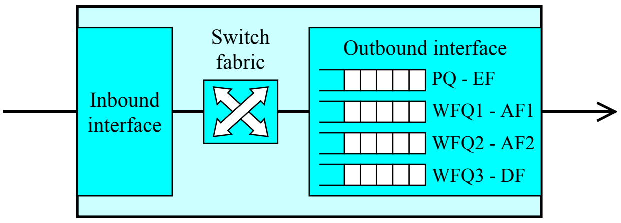


Figure 2. Structure of core-node queues

4. Method of Experiments

A network with two core nodes was constructed, and PC-based traffic generators were developed and used on this network.

4.1 Network structure

The structure of the experimental network is shown in **Figure 3** and the photos are shown in Photos 1 and 2. In the core network, two simulated core switches using a high-end L3 switch called the Hitachi GS4000 was used.[2] The switches were connected by a gigabit link. Although a 1-Gbps link is used, this network should simulate a larger-scale network with a thousand or more flows are aggregated. The GS4000 can be configured to have four queues, i.e., a queue that follows LLQ and three queues that follow WFQ, in each outbound interface.

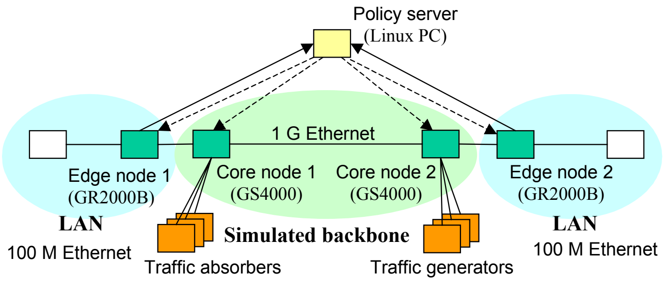


Figure 3. Structure of experimental network

Although edge routers were not used in the experiments described in this paper, there were two edge routers in the experimental network. PCs and application servers were also connected through LAN switches and the edge routers. The

Photo 1. The experimental network nodes and servers

Photo 2. The traffic generators

applications send QoS requests (resource-reservation requests) using a protocol called SNSLP [Kan 08] (Simplified NSLP), which is similar to NSLP (NSIS Signaling Layer Protocol) [Man 06] and RSVP (Resource ReSerVation Protocol) [Bra 97], and the requests were forwarded to the policy server. The policy server configures the edge routers and core nodes and controls the weights of the weighted fair queues of the GS4000.

4.2 Problems and method of traffic generation

The author intended to generate a mixture of various types of traffic including one-way and two-way conversations, and including voice, video, data, and control traffic. However, PC-based traffic generators instead of actual Internet or NGN traffic were used because real-world traffic in a large-scale core network could not be handled. The traffic generators that simulated aggregated flows with hundreds of microflows and traffic absorbers (i.e., PCs for measurement) were connected to the core node directly through Gigabit Ethernet links.

The three problems below that concern the traffic generators must have been solved.

- *Actual packet generation according to self-similar stochastic model*
  The author intended to use traffic of a self-similar stochastic process to make measurements in an environment similar to real-world networks possible. Conventionally, experiments on self-similar traffic were usually performed on simulators. In contrast, in this experiment, actual packets must be generated.
- *Gigabit-order packet-generation performance*
  The Gigabit Ethernet link between the two core nodes needed to be filled with ten or less traffic generators because it is difficult to schedule and to control a large number of generators.
- *Generation of several types of UDP traffic*
  The experiment required several types of traffic including real-time voice, real-time video, and streaming video.

[1] RED (Random Early Detection) worked when the total packet size was greater than 80% of the queue size (the drop ratio was 1 at the end (100%) of queues), but all the measured traffic was mostly insensitive to RED because that was UDP traffic.
[2] VLANs are used to simulate two switches. Because no tagged packet can move between VLANs with different VLAN IDs, two or more switches can be simulated by using one VLAN switch.

They have different stochastic features.

To solve the above problems, a traffic-generator program based on an MMPP (Markov-Modulated Poisson Process) model and an absorber program were developed. The MMPP is a model to analyze [Hef 86] [Mus 03] [Hey 03] and to simulate [Abd 05] aggregated Internet traffic. The packet size distributions of voice and video traffic are also simulated. The burstiness of the traffic can be controlled by changing the parameter values of the MMPP. The distributions of packet length were decided considering the nature of the applications. The rate of generated traffic was adjusted by changing the period of packet generation in the generator. It is not yet very clear how the generated traffic can be close to real Internet traffic because very few measurement results on *UDP* traffic on the Internet or NGN are available. However, traffic generated by the MMPP must be closer than conventionally-used Poisson-models because it can simulate self-similarity, long-dependence, and burstiness. The detailed method of the traffic generation and measurement are described in the supplement.

Three types of simulated aggregated traffic, i.e., conversational voice, conversational video, and streaming video, were generated. This video traffic simulated 100 or more traffic flows (more than 500 Mbps). Each type of traffic was generated by two generator processes, which ran on two CPUs of a dual-CPU PC, with slightly different MMPP parameters. These types of traffic passed through the core link.

4.3 Traffic on core link

The purpose of this experiment is to show the effectiveness of the core policy deployed by the policy server. However, in this experiment, instead of using the policy server, the weights of the queues were changed manually through the command line interface (CLI) during the experiment because the traffic generators did not send QoS requests (i.e., SNSLP packets), and the policy server, thus, could not deploy a policy to core nodes.

We used traffic generators to generate simulated aggregated voice and video traffic. The major focus of this experiment was to observe the difference in QoS parameters generated by the difference in weight shares between the conversational and streaming videos. Although voice traffic was used in this experiment in addition to videos, there are no parameters in the core node to control the EF traffic, except the size of the buffer that is usually mostly empty. Therefore, the author did not focus on the voice traffic but instead focused on the two types of video traffic in this paper. The conversational video traffic is sensitive to latency and jitter. The streaming video is less sensitive than the conversational video to latency but probably more sensitive to loss ratio. Therefore, the conversational video should be allocated a larger weight than the streaming video if their rates are the same.

For example, we can use 1.5 for the weight ratio. If the sum of weights for the conversational and streaming video traffic are 1 and the estimated rates of the traffic are 0.4 and 0.4 Gbps (i.e., the same), we can decide that the weight for the conversational video is 0.6 and that for the streaming video is 0.4. Then, the ratio becomes 1.5 (0.6 / 0.4). If the traffic rates are not the same, we can change the weights proportionally. For example, if the traffic rates are 0.6 and 0.2 Gbps for the conversational video and the streaming video, respectively, we can decide that the weights are 0.82 (= $0.6 * 1.5 / (0.6 * 1.5 + 0.2)$) and 0.18.

4.4 Experimental procedure

In this experiment, six traffic-generation processes were successively (manually) invoked within about three seconds. After they were invoked, they transitioned among three phases, i.e., calibration phase 1, experiment phase, and calibration phase 2 (See Section 5.3). In the two calibration phases, the values measured in the experiment phase are adjusted according to an assumption that the latencies in the two calibration phases are half of the round-trip delay measured by a `ping` command. The reason such a method was used instead of using the NTP (Network Time Protocol) was because measuring the latency and jitter with errors less than 100 µs was necessary, but synchronizing the clocks of senders and receivers with errors less than 1 ms using the NTP was difficult.

5. Experimental Results

The effect of WFQ weight control, which is the main focus of this experiment, is shown in Section 5.2. However, before showing that, we compared QoS parameters under MMPP traffic flows and those under Poisson traffic flows.

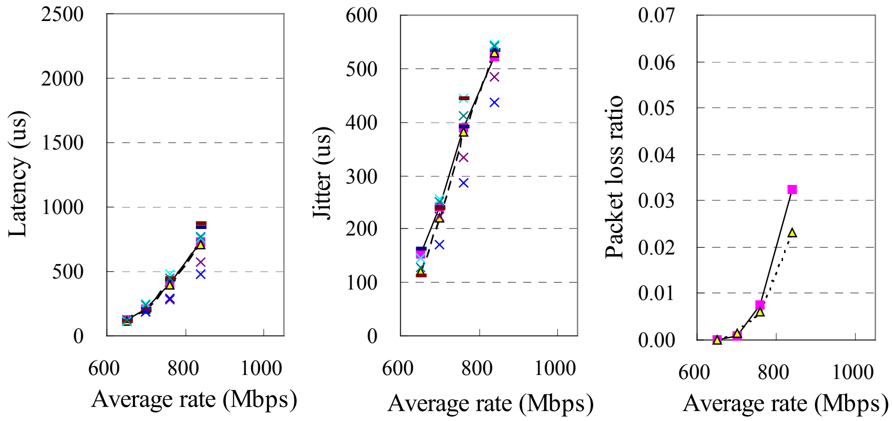
5.1 Comparison of MMPP and Poisson traffic

The relationships between the average traffic rate and the latency, jitter, and loss ratio were measured for both MMPP and Poisson traffic. The results of the MMPP traffic and Poisson traffic are shown in **Figures 4** and **5**, respectively.[1] The conversation video (solid line) and streaming video (broken line) are shown. The weights for these types of traffic were the same.[2]

When the average traffic rate increased, the QoS smoothly

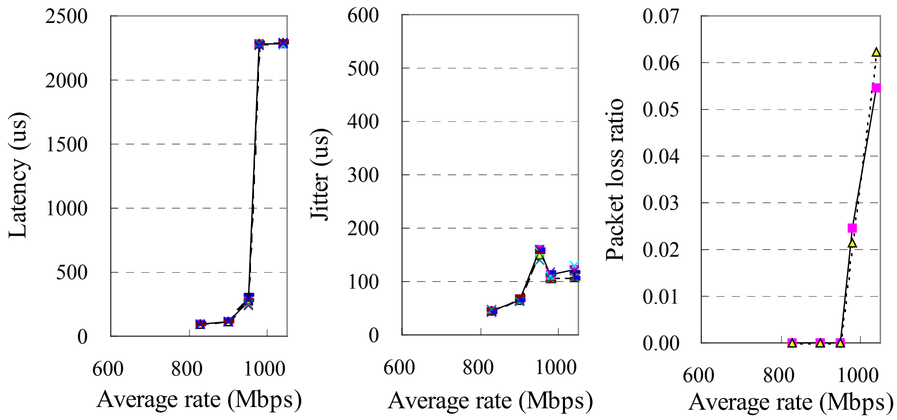
[1] The following parameters were used: $\lambda_1 = \lambda_2 = \lambda_3 = \lambda_4 = \lambda_5 = \lambda_6 = 1.5$ (See Section 7.1).

[2] The parameters for this experiment were as follows. The Poisson distribution parameters were $\boldsymbol{\lambda}_1 = \boldsymbol{\lambda}_2 = \boldsymbol{\lambda}_3 = \boldsymbol{\lambda}_4 = \boldsymbol{\lambda}_5 = \boldsymbol{\lambda}_6 =$ (0.25, 0.45, 0.65, 0.85, 1.05, 1.25, 1.45, 1.65, 1.85, 2.05) (n = 10). The birth-and-death process parameters were p = 0.1 and q = 0.1. The following values were used as the default, and the rate of traffic was adjusted by changing the values proportionally. The conversational voice parameters were $\tau_1 = 49$ and $\tau_2 = 51$. The conversational video parameters were $\tau_3 = 64$ and $\tau_4 = 65$. The streaming video parameters were $\tau_5 = 63$ and $\tau_6 = 66$. The relationship between these parameter values and states of a real-world network was not known. However, these values were used because they generated bursty traffic that matched the purpose of the experiment.



(a) Latency (µs) (b) Jitter (µs) (c) Packet loss ratio

Figure 4. Relationship between average rate of traffic and QoS (1. Bursty case (MMPP))



(a) Latency (µs) (b) Jitter (µs) (c) Packet loss ratio

Figure 5. Relationship between average rate of traffic and QoS (2. Poisson distribution case)

decreased in the MMPP case. In contrast, the QoS was good until the traffic rate increased to 95% (950 Mbps) of the link capacity, but the QoS (except jitter) suddenly decreased when the traffic exceeded 95%. This result seemed to reflect the nature of Ethernets.

5.2 Results of experiments on the relationship of WFQ weight ratio and QoS

The main results of the experiments are shown in this section. Only the MMPP traffic was used. The average network load was approximately 70% in these experiments because, looking at Figure 4 (c), 60% (600 Mbps) seems to be too low (probably no QoS policy is required), and 80% (800 Mbps) seems to be too high (probably no QoS policy can save the situation).

First, the latency, jitter, and loss ratio were measured while the ratio of the weights of queues for the conversational video and the streaming video were switched between 1.00 (49% and 49%) and 1.72 (62% and 36%), and other parameters were fixed. The rates of the conversational video and the streaming video were mostly the same. The results are displayed in **Figure 6**.[1] Four measured data were obtained for each case for latency and jitter. Both the measured and averaged values are shown in the figure. The averaged values are connected by lines. Although the measurement errors are not sufficiently small to draw quantitative propositions, the following qualitative propositions can be drawn from the results.

The conversational video is sensitive to latency, so using a larger weight for the conversational video than that of the streaming video should be better. The results suggest that the latency, jitter, and loss ratio of the conversational video were better when the ratio of the weights was 1.72 rather than 1.00. However, the loss ratio increased by a factor of 4, and the latency and jitter increased 40 to 50%. This severe increase in the loss ratio occurred because the maximum length of the queue, i.e., the buffer size, was not increased even when the average queue length increased significantly. Although the latency and jitter of the streaming video became larger when the ratio was 1.72, they were still much smaller than the maximum value specified for the Streaming class (i.e., 100 and 50 ms) because the buffer size was smaller. However, if the traffic goes through many backbone links, the latency may accumulate, so the latency caused by one link must be sufficiently smaller than the specified value.

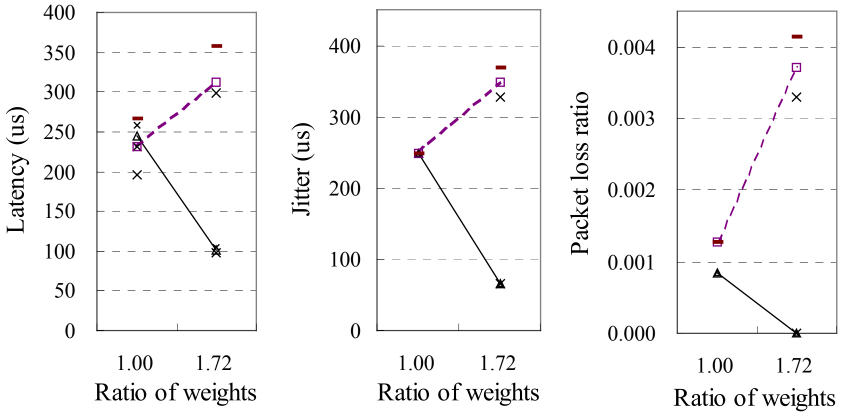
Second, increasing the buffer size in the experimental conditions was difficult, so we tried another method to satisfy the QoS parameters specified for the Streaming class. We used the same MMPP parameters for both the conversational and streaming videos in the above experiment. However, we could use weaker parameters that cause less burstiness for the streaming video. This would cause an effect similar to that of

[1] The following parameters were used. The Poisson distribution parameters are $\lambda_1 = \lambda_2 = \lambda_3 = \lambda_4 = \lambda_5 = \lambda_6 = (0.25, 0.45, 0.65, 0.85, 1.05, 1.25, 1.45, 1.65, 1.85, 2.05)$ (n = 10). The birth-and-death process parameters are p = 0.1 and q = 0.1. The conversational voice parameters are $\tau_1 = 49$ and $\tau_2 = 50$. The conversational video parameters are $\tau_3 = 63$ and $\tau_4 = 65$. The streaming video parameters are $\tau_5 = 62$ and $\tau_6 = 64$. Basically, the values were measured twice and the average values were plotted.

the buffer-size increase because the average queue size will become smaller. The measured results obtained using this condition are shown in **Figure 7**.[1] Six measured data were obtained for each case for latency and jitter. Both the measured values and averaged values (connected by lines) are shown in the figure. The following qualitative propositions can be drawn from the results.

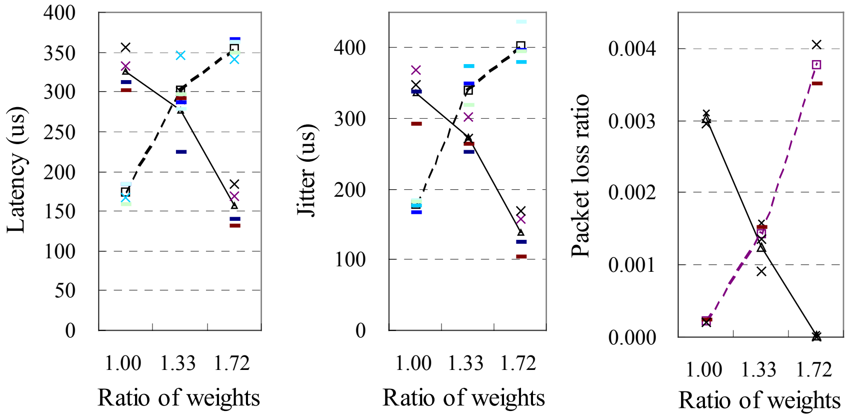
When the ratio of the weights was 1.00, the streaming video performed better than the conversational video in terms of the latency, jitter, and loss ratio. The loss ratio was greater than the specified value for the conversational video. However, the result was reversed when the ratio of the weights was 1.72. The loss ratio was greater than the specified value for the streaming video. When the ratio is 1.33 (56% and 42%), although the distribution of the measured values is large, all the QoS parameters were mostly balanced. The loss ratios of both conversational and streaming videos are about 1.5×10^{-3}. They were still greater than the specified value, i.e., 10^{-3}, because the buffer size was still shorter than required or the traffic was still more bursty than expected, but they were close to the specified value. This means the requirements of both the conversational and the streaming videos were mostly satisfied when the ratio was 1.33.

We also performed experiments using 24 subjects, and compared the subjective quality of voice traffic when the ratio of the weights is 1.00 and 1.72.[2] We obtained a similar result as above, which means that the QoE (quality of experience) of the conversational voice traffic using AF (not EF) is better when the ratio is 1.72.



(a) Latency (µs) (b) Jitter (µs) (c) Packet loss ratio

Figure 6. Relationship between WFQ weight ratio and QoS (1. When burstiness is the same)



(a) Latency (µs) (b) Jitter (µs) (c) Packet loss ratio

Figure 7. Relationship between WFQ weight ratio and QoS (2. When the streaming video is less bursty)

6. Related Work

Many researchers including Striegel and Manimaran [Str 02] studied delay- and/or loss-based service differentiation in DiffServ. In particular, Christin et al. [Chr 02] studied quantitative assured forwarding services. Christines' method took delay and loss into account. This type of service differentiation enabled quantitative (quantity-oriented) differentiation, but quality-oriented differentiation was not focused on.

7. Conclusion

A DiffServ network should guarantee different QoS conditions for different types of communications, especially, real-time and multimedia communications using voice and video, for example. This paper showed that if there are bandwidth-sharing queues that follow WFQ, in core nodes and conversation and streaming video streams are assigned to two of the queues, the requirements of both types of traffic can be better satisfied by assigning a larger weight to the queues for the conversational video. In our experiment, the optimum ratio of the weights was approximately 1.3 when the traffic rates were the same. The optimum weight share depends on the nature of the traffic (especially the burstiness).

[1] The following parameters were used. The Poisson distribution parameters (except the streaming video) are $\lambda_1 = \lambda_2 = \lambda_3 = \lambda_4 = (0.25, 0.45, 0.65, 0.85, 1.05, 1.25, 1.45, 1.65, 1.85, 2.05)$ (n = 10). The birth-and-death process parameters are p = 0.1 and q = 0.1. The conversational voice parameters are $\tau_1 = 49$ and $\tau_2 = 50$. The conversational video parameters are $\tau_3 = 63$ and $\tau_4 = 65$. The streaming video parameters are $\lambda_5 = \lambda_6 = (0.55, 0.65, 0.75, 0.85, 0.95, 1.05, 1.15, 1.25, 1.35, 1.45)$ (n = 10), $\tau_5 = 62$, and $\tau_6 = 64$. Basically, the values were measured twice and the average values were plotted.

[2] We used a voice application called *voiscape* [Kan 05] instead of using a video application for the evaluation because we could not prepare a video application for the experiments.

The experiments in this paper used very specific conditions, i.e., the traffic model (an MMPP with specific parameters), core node architecture, and fixed buffer size. Therefore, as future work, the above proposition should be tested in various environments.

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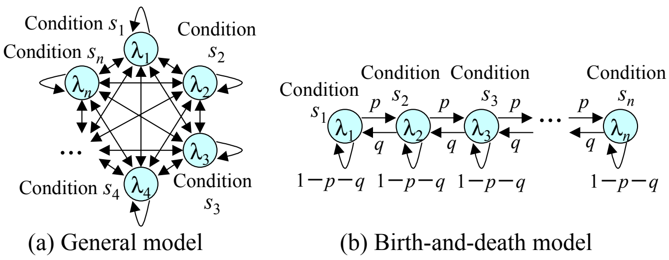
8. Supplement: MMPP-Based Traffic Generation

8.1 MMPP Model

Many researchers have studied methods using the MMPP (Markov-Modulated Poisson Process) for generating network traffic with long-term self-dependence [Hef 86] [Mus 03] [Hey 03]. In the MMPP model, the generator process transitions among the states of a Markov chain, and, in each state, traffic that has a Poisson distribution, whose parameters depend on the state, is generated. A generalized MMPP model is shown in **Figure 8** (a). The generator process transitions among a predefined number of states and generates packets according to the Poisson distribution with parameter λ_i when the process is in state s_i. The probability of packet generation in state s_i is $P(\lambda_i)$ (where $P(\lambda)$ means a Poisson distribution). Any state can be followed by any state, i.e., an arbitrary state transition may occur, in this model. If we differentiate values of λ_i (i = 1, 2, …), the traffic becomes bursty.



(a) General model (b) Birth-and-death model

Figure 8. MMPP Models

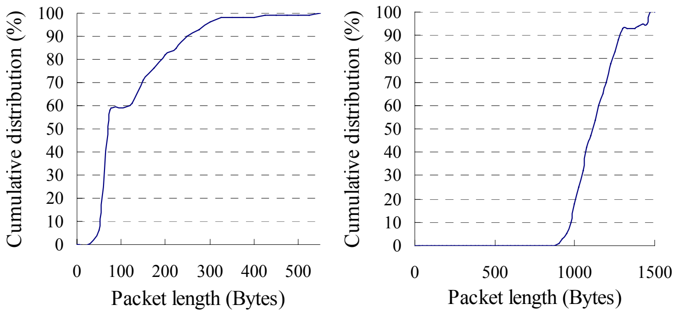
However, this generalized model has an extremely large number of parameters, so estimating them is difficult. In addition, the meaning of this model is not clear. There are more restricted models including a model shown in Figure 8 (b), i.e., the birth-and-death model. In this model, the birth probability is p and the death probability is q. In this model, a virtual source of traffic is generated or killed one by one. State s_0 is the state with no traffic sources, and the probability of packet generation, $P(\lambda_0)$, should be zero. While the state transitions among s_1, s_2, …, and s_n, the probability $P(\lambda_i)$ should increase. In the experiments in this paper, the birth-and-death model was used.

8.2 Distribution of packet length

In our experiments, both simulated voice-packet distribution (**Figure 9** (a)) and simulated video-packet distribution (Figure 9 (b)) are used. These distributions were decided in reference to Reine et al. [Rei 03] and some other papers. Explanations are added here.

- *Voice*: The sizes of more than half of the packets are 75 bytes or less. The length of many voice packets is 40 bytes. That means many voice packets do not have a UDP payload; they only contain a header. This is probably because voice applications, such as Skype, generate packets with no payload when silent.[1]
- *Video*: A video frame usually contains 1500 bytes or more data. Therefore, if a LAN is used, many packets are 1500 bytes in length or close to that.

The packet size is quantized; it is a multiple of 25. There is probably no significant effect of this quantization in our experiments.



(a) Simulated voice traffic (b) Simulated video traffic

Figure 9. Cumulative distributions of packet length of simulated voice and video traffic

8.3 Detailed description of traffic handling

The traffic generator program generates one or more packets for each period τ, which is typically 50 µs. The period is measured by the clock (`gettimeofday()`) of the PC using busy waiting. In each period, *m* packets (*m* is fixed to be three in our experiments) are generated successively. The number of packets generated is $\lambda_i m$ (where λ_i is the parameter of MMPP).

Three PCs (HP Compaq DC5100SF/CT-P3.0 Pentium 4 (3.0 GHz) Dual CPU) were used for the traffic generators and three of the same types of PCs were used for traffic absorbers. To avoid latency and jitter, only one generator process ran on each CPU. Therefore, six processes were used. The parameters of process P, λ_P (= (λ_{P1}, λ_{P2}, …, λ_{Pn})), p_P, q_P, and τ_P, could be independently specified for each process. Approximately 60 Mbps of simulated voice traffic or 300 Mbps of simulated video traffic could be generated by each CPU. If the period shortened more, the latency became larger when packets were successively generated.

A traffic generator inserts the sending time into the UDP payload of the packet. A traffic absorber computes the latency and jitter from the sending and receiving times. However, the internal clocks of the traffic generator and absorber are not synchronized. Therefore, the latency values must be adjusted. To enable the adjustment, each experiment was performed following the steps below.

- *Calibration phase 1*: Before the experiment phase, only probe packets were generated for approximately 10 seconds. The number of probe packets was limited, so the packets did not cause jitter and additional latency. The latency in this phase is called the silent-time latency.
- *Experiment phase*: The traffic generators generate MMPP-model-based traffic, and latency and jitter were measured in the traffic absorbers. Averages of both latency and jitter were computed using 1000 successive packets. 1000 was used for the average number of packets because a larger-scale average increased jitter error.
- *Calibration phase 2*: After the experiment phase, only probe packets were generated for approximately 10 seconds. The latency in this phase must have been the same as that in the calibration phase 1 (i.e., the silent-time latency).

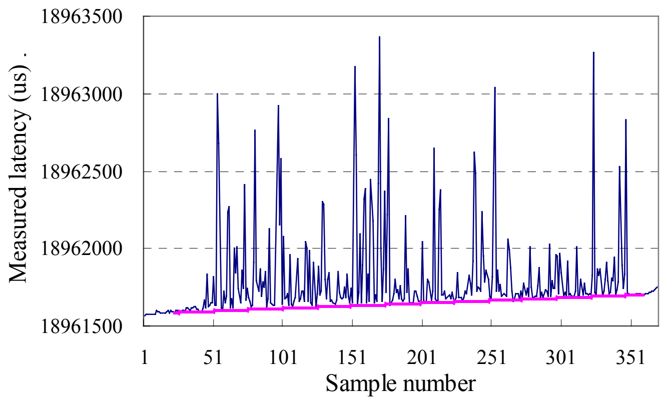


Figure 10. Latency measurement result

The measured latency values (without adjustment) can be plotted as shown in **Figure 10**. They can be adjusted in the following method. A baseline that connects the latency values in two calibration phases is shown in Figure 10. This line corresponds to the silent-time latency. The reason that the slope of this line is not zero is that rates of the traffic-generator clock and the traffic-absorber clock are different (i.e., there is clock skew). We can subtract the time that this line represents from the measured value and add the estimated silent-time latency (64 µs) to obtain an estimated latency.[2]

The packet loss ratio was not measured in the traffic absorber. Instead, the packet loss ratio was estimated by using the CLI of GS4000. By using the `show qos queueing` command, the rate of passed traffic and that of discarded traffic could be measured for each output queue.

[1] The minimum length of packets in our distribution was 50 bytes, instead of 40 bytes, because a time stamp and some more additional information must be included in the packets in the experiments.

[2] Moon et al. [Moo 99], Zhang et al. [Zha 02], and Anagnostakis et al. [Ana 03] describe methods for obtaining the exact latency when skew exists. However, we could obtain exact latency values by a simpler method using time calibration in this experiment.