Title	QoS
Author(s)	,
Citation	
Issue Date	2007-03
Type	Thesis or Dissertation
Text version	author
URL	http://hdl.handle.net/10119/3577
Rights	
Description	Supervisor: , ,

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A QoS control of the Videostream have a burst

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February 2007

1.Introduction

In recent years, high speed transmission lines such as ADSL or FTTH spread to the home, and transmission of MPEG-2 streams on Internet receives attention. Since MPEG-2 images are highly compressed, these are distracted in continuity when data loss occurs. In the case of the application that does not demand real time constraint, like VoD (Video on Demand), there are many devices to keep quality, providing buffers at the receiver edge. However, we can not use buffers when we communicate animation with real time constraint such as Video phone, because the packet delay causes fatal quality deterioration. Losses of packets becomes fatal. Because the Internet provides only Best Effort Service, there are delays, delay variations, and losses of packets. Consequently, it can not keep QoS. Therefore, we should control quality deterioration factor. Studies on guaranteeing QoS are flourish.

On the other hand, we completely keep quality in synchronous network. The next generation synchronous network, GMPLS (Generalized Multi- Protocol Label Switching), is expected as a network backbone.The trend to integrate network infrastructure to IP network technology, asynchronous network, methods how utilize the characteristics of synchronous network, packet scheduling mechanisms, are not established.

The author proposes a new network service for MPEG-2 stream transmission. He pays his attention to GOP structure of MPEG-2 that characterizes MPEG-2 traffic.The evaluation of the proposed transmission service is performedin computer simulation experiments, and the comparison betweenthe proposed service and priority scheduling with the admission control are carried out.

The author defines a term "a burst" as a packet sequence that consists of one frame data.

2.Realtime of videostream

Because videostream is very high bitrate data, it is demanded a great many resources. And because it is compression, bitrate changes in real time communication. In addition, there were much a service class and scheduling given priority, and it has been suggested to make good use of real time security of an animation stream and a not crowded resource. However, a large quantity of resources are allotted to a top priority class by conventional technique and cannot answer it for the demand that I want to use plural top priority classes for too much.

3.Switching

Packet switching that is ansynchronous is a very efficient to do multiplexing. However, because it is ansynchronous, a delay variation of a packet occurs and cannot guarantee quality. On the other hand, I can completely guarantee quality because line exchange occupies a resource of the line and can use it. However, I become a high cost very much because multiplexing is hard to do it.

4.Traffic of MPEG2

MPEG2 is used both a time model compression and a space type compression. Encode and decode are performed by a unit such as GOP. I picture is generated by DCT(Descread Cosine Transform). P picture is generated by for difference with a I picture. B picture is bidirectional predicted from I or P picture, and it is made. In this way because it is made, when an error occurred, the error range is decided from a kind of picture. The error range of I picture is the widest. MPEG2 has variable bitrate. The admission control calculated from max bandwide have a weak point can't transport many flows.

5.QoS control mechanism

I suggested QoS control mechenisms which consider MPEG2. Admission control and packet scheduling synchronize to GOP. I suggested a timeslot. Timeslot is synchronized to GOP. I challenged to transport a burst for 33.3 ms.

I defined three Service Classes, EF(Expedited Forwarding) Class, PF(Predicted Forwarding) Class and BE(Best Effort)Class. The Class has the highest priority is EF. Next one is PF, and BE is the lowest. EF guarantees maximum of the bandwide. PF guarantees average of the bandwide. When a control server assigned it to each timeslot in response to call receptionist application of each class and assign it to a timeslot of the lowest band. Because a timeslot is transmission of a message time of an I picture, I try that I follow the

transmission of a message of an important I picture thereby. Packet scheduling algorithm uses a multi-level round robin. In addition, a thing more than bandwidths of a backbone is expected so that it is allocated on average in a PF class. Naturally a delay occurs if it surpass a bandwidth. Because the packet which delayed does not need to become a picture, I decide to throw away a packet by the importance. I performed a simulation experiment of priority scheduling with a call admission control mechanism to do comparison with this suggestion method. An evaluation item is following four items.

1. The total number that can transmit each service class
2. Delay of frame
3. Delay variation of I picture
4. The total number of a frame omission

6.Simulation

I performed a simulation experiment with a computer to evaluate this suggestion method. As a result, as for the suggestion technique, in the EF class, a lot of transmission of a message was enabled than priority scheduling with a admission control mechanism. On the other hand, real time becomes light so that suggestion technique delays transfer with a timeslot about a delay of a frame.

7.Conclusion

I suggested a QoS control mechanism for GMPLS which considered a traffic characteristic of MPEG2 and evaluated an effect of this. Number of a class to guarantee a band and a trade-off of real time occurred hereby. For a problem given in this suggestion method, the first is the problem that real time lowers. The second is the problem that a delay occurs on the Egress. In the last, it is necessary to support more GOP.