Effective Streaming technology of a layered encoding Video Application supporting QoS mechanism in the Internet

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Abstract: Internet became the most popular network in spite of its weakness in realtime multimedia service. Many experts believe that the Internet has the potential to become the main multimedia distribution network of the near future. Currently, it does not provide any QoS guarantees and, even when it does, guaranteed quality delivery of video may turn out to be too expensive. Unavoidable packet losses and delay jitter caused by congestion in a best effort delivery environment require use of intelligent transport techniques for effective video delivery. According to market needs of better quality of service (QoS) for realtime multimedia services over Internet, they have been standardizing RSVP, IntServ, and DiffServ. This paper combines the benefits of QoS mechanisms such as RSVP/IntServ with scalable video encoding. We propose that more important bit stream is given more priority such that limited network resources are guaranteed for the stream. Various prioritizing approaches are proposed and compared to normal approach by using Network Simulator. The calculated QoS parameters such as packet loss rate are used to calculate degree of degradation in video quality. In this Paper, proposed methods can be implemented adaptively to VoIP protocol, such as H.323, SIP.

1. Introduction

As Internet Protocol(IP) is getting explosively popular, users become to need real-time conversational video service such as video phone and video conference. Since IP uses packet switching in which bandwidth of a channel is shared by many users, this protocol is more appropriate to document service and has much problem to support conversational multimedia service. Previously, conversational video services are mostly implemented over circuit switching networks which guarantee communication quality. Internet protocol is now evolving mainly in order to support real-time multimedia system. It will be equipped with some reservation capability(RSVP), more efficient switching systems(MPLS), and ATM backbone. There is no doubt that conversational multimedia service of high quality will be supported by future Internet. This paper, however, proposes efficient methods to overcome degradation of multimedia quality by the current Internet.

Condition of the current Internet is time varying and does not guarantee quality of service. Some tried to make video codec more robust to loss by using error resilience coding

in MPEG-4 and H.263, and others use scalable capability and appropriate rate control.[9]

In Section 2, The basic characteristics of IP network about QoS is described. Section 3 explains basic scalability in video codec and in VoIP(Voice over IP) to Section 4 which proposes and evaluates some techniques to layered coding video application. It is shown that the proposed techniques are good for QoS of video services. Section 5 concludes this paper and introduces future works.

2. IP Network Characteristics

Current IP networks are supporting for only IP routing services. But, So Mutimedia Services is going to increase that the network needs resouces reservation methods. Because multimedia services are continuos media. Network nodes must be reserve network resources(bandwidths, buffers) between router and router or router and end-point. So current IP Networks are not well suited to the streaming of video content. IP networks exhibit packet losses, delay and jitter (delay variation), as well as variable achievable throughput.[9] Real-time video applications require all packets to arrive in a timely manner. If packets are lost, then the synchronization between encoder and decoder is broken, and errors propagate through the rendered video for some time. If packets are excessively delayed, they become useless to the decoder and are treated as lost. Packet losses and its visual effect on the rendered video, is particularly significant in predictive video coding systems, such as H.263 and MPEG-4. The effect of packet losses can be reduced, but not eliminated, by introducing error protection into the video stream. As such resilience techniques have been widely studied, and at best can only minimize rather then eliminate the effect of packet loss. In present, there are effort for the main multimedia distribution network.[4] Increasing bandwidth is a necessary first step for accommodating these real-time applications, but it is still not enough to avoid jitter during traffic bursts. QoS does not create bandwidth, but manages it so it is used more effectively to meet the wide range or application requirements. The challenge of these IP QoS technologies is to provide differentiated delivery services for individual flows or aggregates without breaking the network in the process. Generally speaking, QoS is the ability of a network element (e.g. an application, a host or a router) to provide

some level of assurance for consistent network data delivery.[8]

Some applications are more stringent about their QoS requirements than others, and for this reason (among others) we have two basic types of QoS available:

- Resource reservation (integrated services): network resources are apportioned according to an application's QoS request, and subject to bandwidth management policy.

- Prioritization (differentiated services): network traffic is classified and apportioned network resources according to bandwidth management policy criteria. To enable QoS, network elements give preferential treatment to classifications identified as having more demanding requirements.

In real-world use, it is unlikely that these QoS protocols will be used independently, and in fact they are designed for use with other QoS technologies to provide top-to-bottom and end-to-end QoS between senders and receivers.

This paper combines the benefits of QoS mechanisms such as RSVP/IntServ with scalable video encoding. We propose that more important bit stream is given more priority such that limited network resources are guaranteed for the stream. Various prioritizing approaches are proposed and compared to normal approach by using Network Simulator.

3. Scalability
3.1 Scalability of Video coding in MPEG-4

The scalability of a video stream refers to the generation of a stream that may only be decoded in part due to limitations of available resources. The scalability may be desired to overcome limitations of available computing power, or to accommodate bandwidth limitations.

All of these methods can be used separately, or together to create a multi-layer scalable bit stream. The resolution, frame rate, and quality of the image can only increase by adding scaling layers. In general, The scalability of video means the ability to achieve video of more than one resolution and/or quality simultaneously. The scalable video coding involves generating a coded representation (bitstream) in a manner which facilitates the derivation of video of more than one resolution and/or quality by the scalable decoding. Bitstream Scalability is the property of a bitstream that allows decoding of appropriate subsets of a bitstream to generate complete pictures of resolution and/or quality commensurate with the proportion of the bitstream decoded.[1][2]

The one basic bitstream is called by the base layer(B), another bitstreams is called by the enhancement layers(E1, E2,...) . There are important information of header, motion vector and DC values for the leaset decoding information in the base layer. The enhancement layer, hence, can certainly be decoded by having bit stream of base layer. If these are decoded together, it is good quality of more than only bit stream of base layer. As the result of the effective scalability, if it is guaranteed for the bit stream of the base

layer in the networks, multimedia services can support the least quality services. Also the error propagation caused by packet loss of the enhancement layer is the fewest, because enhancement layer refers to the base layer. There are three types of scaling: Temporal, Signal-to-Noise Ratio (SNR), and Spatial that are available in Recommendation H.263 and MPEG-4.[1][2][5]

- SNR scalability
  In the case of SNR scalability, the base layer bit stream encodes as low quality(because of high compression), and the enhancement layer bit stream encodes as high quality. For example, the spatial and temporal resolution of the base layer is same as the enhancement layer.

- Spatial scalability
  In the case of spatial scalability, the enhancement bit stream is used to increase the resolution of the image. For example, the resolution of the base layer is QCIF(Quarter CIF) resolution and that of the enhancement layer is CIF resolution. In this case, when the output with QCIF resolution is required, only the base layer is decoded. And when the output with CIF resolution is required, both the base layer and the enhancement layer are decoded. But the complexity of its coding is higher than another coding methods.

- Temporal scalability
  In Object-based Temporal scalability, the frame rate of a selected object is enhanced such that it has a smoother motion than the remaining area. In other words, the frame rate of the selected object is higher than that of the remaining area. For example, if the base layer is coded as 10 frames per sec, the both are coded as 20 frames per sec. The complexity of its coding is most simple in others scalability, so that, now this scalability method is using in Windows Media Player.

There are two types of enhancement layered coding methods. The one is composed of P(Prediction) frame, the other is composed of P frame and B frame. In such as enhancement layer format is PPP(Prediction only), it refers only one frame for estimating motivation in base layer. So that enhancement layer refers to upsampling a frame in basement layer or same layer. PPP mode decreases estimation time, but bit rates is high. This mode is used in FGS. In the other hand PBB mode refers basement layer and enhancement layer at same time. This method is high compression rates, but increases complexity between frames. Also it is effective for preventing error propagation.

Amongs of these scalability types, it is selected spatial scalability and enhancement layer to be composed of P frames in this paper,. The reason of the choice is good for real-time multimedia applications because of small decoding time.[1][2]

- Enhancement layer methods: PPP
  This figure is enhancement layer that be composed of P frame.
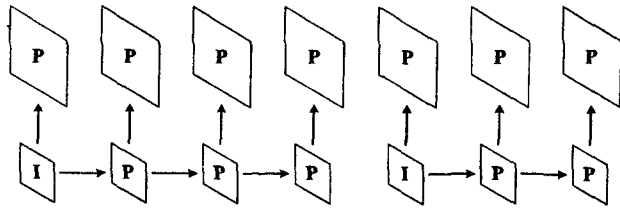
Fig 1. Enhancement layer format(Enhancement layer has only P frames)

● Enhancement layer methods: PBB
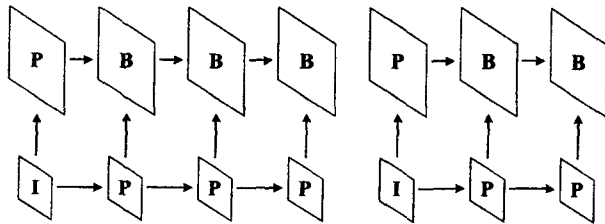This figure is enhancement layer that be composed of PBB frame.

Fig 2. Enhancement layer format (Enhancement layer has P frames and B frames)

3.2 Scalability of VoIP

In H.323 and SIP, there are many possible methods for layering of the video and organization of the corresponding RTP sessions. The reason that the layers may need to be separated is that they are used for either decoder power scaling, or for bandwidth usage scaling. An important feature of the layered codec is that at any time an endpoint may discard any or all enhancement layers, without affecting the quality of the base video, in order to provide decoder power scaling. This figure is described in H.323. If we use multimedia phone, our multimedia traffics are a voice, videos. Amongs of these traffics, the bitrates of video traffic is too much, therefore, it is made scalable of video traffic for managing effective networks. Just for my information, scalable coding is same as calling by layered coding.[5]
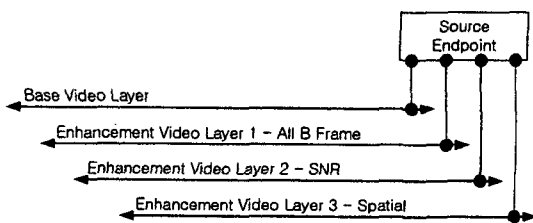

Fig 3. Model with layered video

Several important characteristics of the nature of scalable coding usage should be considered when using network QoS for delivery of scalable coded video streams. An enhancement layer cannot be decoded properly without receiving the layers on which it is dependent. Enhancement video layers may be discarded without affecting the decoding of the layer on which they are dependent.

If available, network QoS may be used to help guarantee that a video stream will be delivered by the network. Since layered video may be delivered using multiple streams, delivered on separate network connections, different QoS can be used on each video layer. QoS used on layered video streams should be specified when the logical channel is opened.

4. Experiment

It is important that a dependent video layer has the information on which they are dependent at the time the dependent layer is to be decoded. This leads to general rules regarding use of QoS. The first, Dependent layers(base layer) that are delivered using network QoS should have the layer they are dependent on, also delivered using QoS. The second, the base layer should be delivered using network QoS, if any other video layers in the conference are to be delivered using QoS. The third, the nearer the video layer is to the base layer, the stronger the delivery guarantees should be. This source is news pictures.
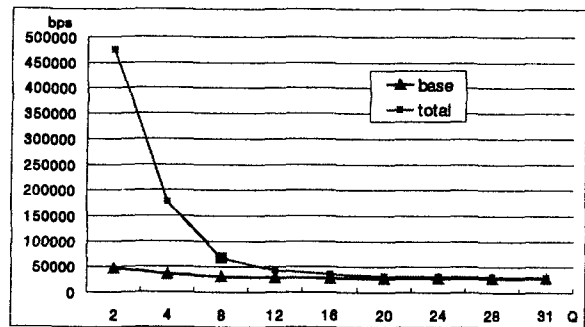

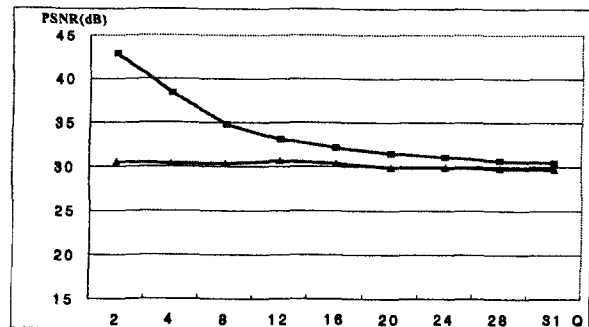Fig. 4 bitrates per Quantization

Fig. 5 PSNR per Quantization

For the experimental, it is used the bit stream which have characteristics such as the figures. These figures show the bit rates and the PSNR(Picture signal noise rates) of an experimental bit strem. It is variable the bit rates to be due to Q(quantization) values. Only base layer bit rates is less than 50kbps, but both of the base layer and the enhancement layer are 475kbps. Fig 5 shows compare to PSNR between the only base layer bit stream and the total bit stream.

Let us suppose for experimentation. The first, bit stream with having characteristic such as above figures is delivered

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from the source to destination. The second, it is guaranteed for base layer bit stream as QoS protocol such as RSVP.
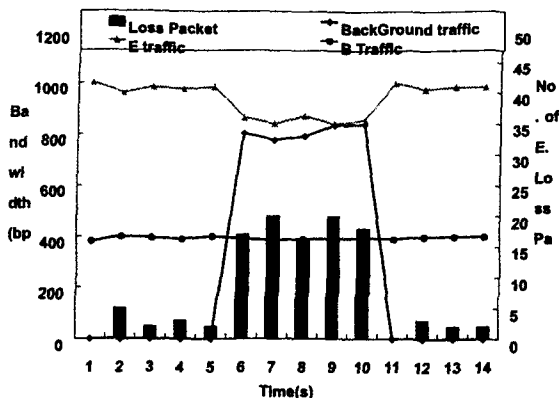
4.1 Experimental result

Fig. 6 Distribution of Packet −loss

This paper combines the benefits of QoS mechanisms such as RSVP/IntServ with scalable video encoding. We propose that more important bit stream is given more priority such that limited network resources are guaranteed for the stream. Various prioritizing approaches are proposed and compared to normal approach by using Network Simulator.[6][7] The calculated QoS parameters such as packet loss rate are used to calculate degree of degradation in video quality. In limited bandwidth, it transfers base layer traffic , enhancement layer traffic and TCP back traffic to the destination. The traffic of best priority is base layer traffic using QoS mechanism. Enhancement layer has no priority and also TCP traffic. therefore as above, when overflow the bandwidth, the base layer traffic is guaranteed and enhancement layer experiences packet losses.[3][10]
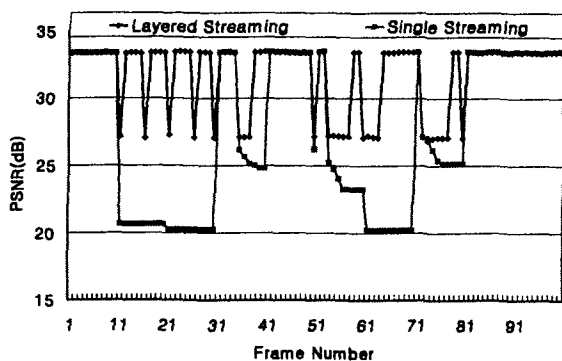

Fig. 7 PSNR variation about Packet -loss

As result of Fig 6, the pattern of packet losses in multi layers is decoded by applying to original source. So that when having packet losses, it is shown that PSNR is decreased. Specially, when having packet losses in single streaming, PSNR is more rapid decreased than in layered streaming. And as it uses scalable coding, error propagation is lower than no scalable coding. There is reason that the base layer traffic is more important information than

enhancement layer traffic, and the base layer traffic is guaranteed that it is maintenance regular video quality in endpoint application. Methods which protect important bit stream in networks exist already such as QoS Protocol (RSVP/IntServ, DiffServ) . So in this experimentation, the base layer needs protection against packet losses using RSVP/IntServ.

5. Conclusion

We propose that more important bit stream is given more priority such that limited network resources are guaranteed for the stream using QoS protocol mechanism. Various prioritizing approaches are proposed and compared to normal approach by using Network Simulator. In current IP networks, all packets consider as same information. However, if networks are guaranteed for these packets which have more important information, networks is supported better services to user. This result is adaptable to real-time applications. Real-time applications have QoS mechanism of being able of negotiation functions with network side node for the QoS, using such as RSVP, DiffServ. ISPs are supporting to service without increasing network resources. In this paper, we propose the streaming technology that combines with taking advantages of MPEG-4 and network(QoS protocol) technologies. Currently, Internet service is evolving in order to support better real-time multimedia service. Evolution includes enhancement in backbone and switching. New approaches such as RSVP and MPLS take advantage of basic concepts of resource reservation and virtual connection.

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