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# Adaptive Stream Multicast for Video in Heterogeneous Networks

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**Abstract:** This study first proposes a novel adaptive stream multicast for MPEG-4 FGS video to meet these challenges. Based on the fine-granularity property of MPEG-4 FGS video coding technology, the scheme tries to delivery multimedia multicast service over the heterogeneous networks in a similar way as that to transport water in pipelines, where the valves in pipelines adjust water flux to next pipeline. A new method of computing PSNR is also first advanced to evaluate the MPEG-4 FGS video transmission. Simulated results indicate that the scheme could dispose the heterogeneity of networks and end-systems freely, with permanent stability, flexible scalability and unprejudiced fairness and TCP-friendliness.

**Key words:** MPEG-4 FGS, stream multicast, multicast

## INTRODUCTION

In the future, the existing wired and wireless networks are expected to interconnect each other and converged to a huge and heterogeneous network (Chilamkurti and Soh, 2002). To handle heterogeneous network conditions, layered multicast (Huang *et al*., 1999) was suggested to delivery video service to heterogeneous subscribers. The technique requires video stream to be organized into layers and each layer is sent to a separate multicast group. Under the available bandwidth, subscribers join the appropriate number of layers by join-experiments. Nevertheless, under wireless environments frequent join-experiments will cost amount of resources because of frequently fluctuant bandwidth. The scheme doesn't fit for video stream transmission in wireless networks. What's more, the segmentation of video stream and synchronization of received layered data must be also considered really. This leads to more complexity for terminals. However, most efforts (Roesler *et al*., 2003) are devoted to resolve the questions about fairness, congestion control and TCP-friendliness of layered multicast. All these efforts could not radically remedy the above fatal drawbacks of layered multicast. In this study, an adaptive stream multicast for MPEG-4 FGS video, which takes advantage of the fine-granularity property of MPEG-4 FGS (Chilamkurti and Soh, 2002) video coding technology, is first proposed to address the insurmountable issues of layered multicast. The scheme tries to deliver the MPEG-4 FGS encoded video stream over heterogeneous networks in the same way as that to supply water by pipelines.

## ADAPTIVE STREAM MULTICAST SCHEME

The MPEG-4 video coding with Fine Granularity Scalability (FGS) is a new coding technique expressly designed for video streaming. With FGS coding the video is encoded into a Base Layer (BL) and one Enhancement Layer (EL). Similar to conventional scalable video coding, the base layer must be received completely in order to decode and display a basic quality video.

The discrete cosine transform coefficients of FGS enhancement layer are bit-plane coded and there is no motion compensation within the FGS enhancement layer. So, the enhancement layer can be truncated anywhere at the granularity of bits to adapt to the available network resources and the remaining part can still be decoded to improve upon the basic video quality, while the conventional scalable video coding requires the reception of complete enhancement layers to improve upon the basic video quality. This makes the enhancement layer highly resilient to transmission errors and subsequently well suited to the transmission over error-prone networks such as the best-effort Internet. The base layer is transmitted with high reliability (achieved through appropriate resource allocation and/or channel error correction) and the FGS enhancement layer is transmitted with low reliability (i.e., in a best effort manner and without error control).

With fine granularity property, the FGS-encoded videos can flexibly adapt to changes in the available bandwidth in wired and wireless networks. The FGS video coding has the potential to fundamentally change the video streaming in networks.

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As presented by Cheng and Rito (2003), the enhancement layer of MPEG-4 FGS pre-coded video stream could be truncated anywhere, so its throughput could be adapted by cutting its enhancement layer. That also means that the video stream rate can be adjusted according to the network status of their downstream links by the routers in networks, which works as the valves in pipelines do to regulate the flux of water to a downstream pipeline.

Thus, the ASM-CBCC is devised to deliver the MPEG-4 FGS encoded video stream over heterogeneous networks in the same way as that to supply water by pipelines. With the scheme, sources do nothing but send video stream to a multicast group and each receiver joins the multicast group. The more the receiver gets data, the better it perceives video quality. Simultaneously, routers on multicast distribution tree conduct rate adaptation according to network congestion status. Therefore, the ASM-CBCC is actually a single-channel multicast with router-assisted congestion control.

A MPEG-4 FGS DropTail (MF-DT) priority queue also is schemed out to adapt the throughput of video to downstream links by dropping some enhancement layer packets according to queue length change, which indicates network congestion status. In the queue, different priorities are assigned to the I, P, B, E packets of the MPEG-4 FGS video stream with the priority from high to low, respectively, where, E packets are that of enhancement layer. It also assumes that non-video packets have the same priority as that of I packets. When a queue reaches its limit, the packets with lowest priority will be listed out from the queue's tail. If their priorities are lower than that of an arriving packet, the first packet with lowest priority will be dropped and the packets after it will be moved forward correspondingly. Then the arriving packet is appended to the tail of the queue. Otherwise, the arriving packet will be dropped simply. The MF-DF could be implemented as an extension of present queues and completely compatible with them.

## EVALUATION METRIC OF MPEG-4 FGS VIDEO TRANSMISSION

There are many methods to evaluate the MPEG-4 FGS video transmission by QoS parameters, i.e., packet loss ratio, delay jitter, starvation probabilities and so on, but they don't reflect perceived video quality. The most widespread method is the calculation of peak signal to noise ratio (PSNR) of video sequences. Nevertheless, the method to obtain approximate PSNR for MPEG-4 FGS video doesn't consider the dependencies of video sequences as showed in Fig. 1. So, a novel scheme, which considers inter-frame dependencies and rate-distortion

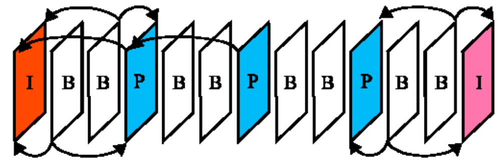


Fig. 1: Typical frame sequence and dependencies for GoP

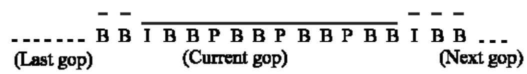


Fig. 2: GoP pattern in video sequences

curves roundly, is designed in this study to approximate the PSNR of MPEG-4 FGS video sequences.

For each reference frame in current GoP as showed in Fig. 3, its direct or indirect prediction frames are assigned to assumable dependency-factor (DF) values according their dependency degree on the current reference frame. In Fig. 2, the I frame in current GoP has thirteen prediction frames, i.e., the last two B frames in last GoP and the residual frames in current GoP. As the two B frames in last GoP are bi-directional prediction frames relying on the third P frame in last GoP and the I frame in current GoP, they are given a DF value 0.5, respectively.

The first two B frames in current GoP gain a DF value 0.75, in respect that they are predicted by the I frame and the first P frame in current GoP and the latter also depends on the former again. Only related to the I frame, the first P frame in current GoP is assigned a DF value 0.5 accordingly. The second two B frames in current GoP with a DF value 0.875, are bi-directionally predicted by the first and second P frame in current GoP, both of which rely on the I frame in current GoP. Here, the cumulative effects of the I frame in current GoP are considered carefully. Similarly, all the DF values for each prediction frame of the I frame in current GoP could be gained, shown in Fig. 3.

In one GOP, different reference frame has different prediction frame number, as shown in Fig. 4, where, all the prediction frames of the first, second and third P frame are given a DF value, respectively. According to Liu *et al.* (2004), the PSNR quality of a MPEG-4 FGS coded video is obtained by adding its base layer PSNR quality and enhancement layer PSNR quality improvement. The base layer PSNR and the enhancement layer improvement PSNR, could be gained by the piecewise linear interpolation of the base layer packet loss ratio PSNR curve (Liu *et al.*, 2004) and the enhancement layer rate-distortion curve, respectively. Therefore, the PSNR quality of a frame could be obtained by adding the two PSNR. However, for the prediction frame of a reference frame, its PSNR quality must be minus the product of its

| | | | | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| B | B | I | B | B | P | B | B | P | B | B | P | B | B |
| 0.5 | 0.5 | 1 | 0.75 | 0.75 | 0.5 | 0.875 | 0.875 | 0.75 | 0.9375 | 0.9375 | 0.875 | 0.5 | 0.5 |

Fig. 3: DF values for prediction frames of I reference frame

(a)

| | | | | | | | | | | |
|---|---|---|---|---|---|---|---|---|---|---|
| B | B | P | B | B | P | B | B | P | B | B |
| 0.5 | 0.5 | 1 | 0.75 | 0.75 | 0.5 | 0.875 | 0.875 | 0.75 | 0.5 | 0.5 |

(b)

| | | | | | | | |
|---|---|---|---|---|---|---|---|
| B | B | P | B | B | P | B | B |
| 0.5 | 0.5 | 1 | 0.75 | 0.75 | 0.5 | 0.5 | 0.5 |

(c)

| | | | | |
|---|---|---|---|---|
| B | B | P | B | B |
| 0.5 | 0.5 | 1 | 0.5 | 0.5 |

Fig. 4: DF values for prediction frames of (a) the 1st P frame (b) the 2nd P frame (c) the 3rd P frame in one GoP

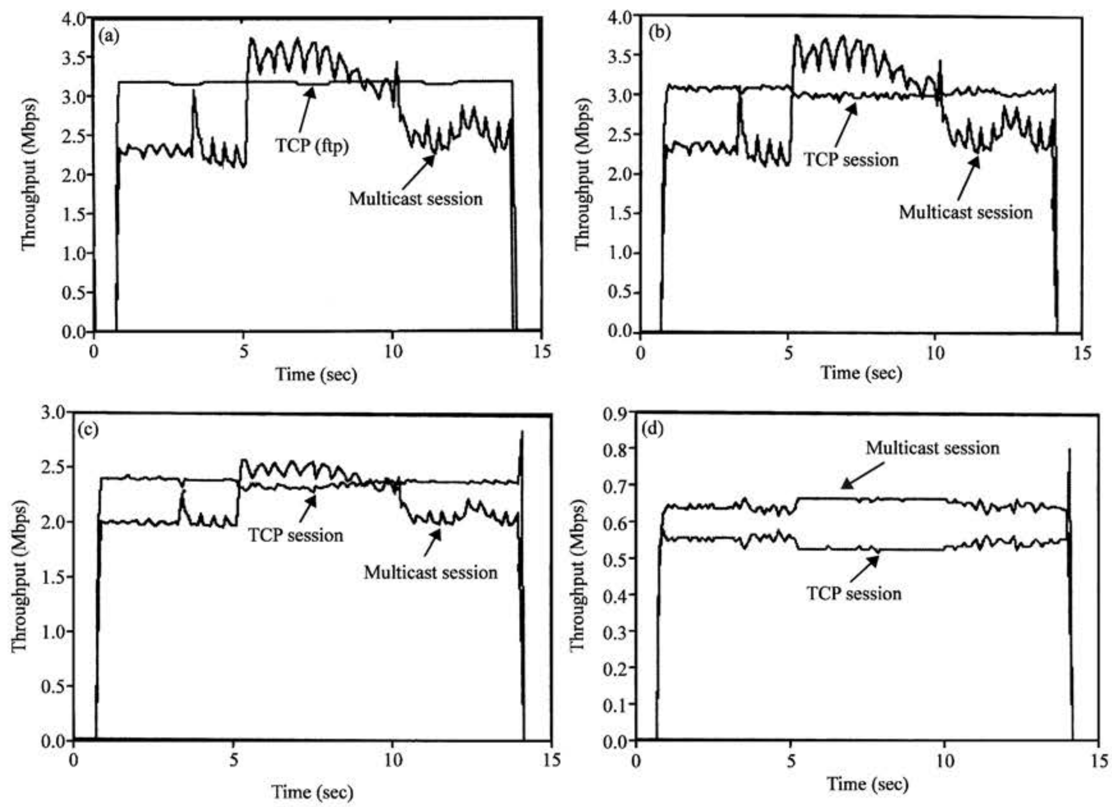


Fig. 5: (a) Original throughput of TCP and multicast under 100 M bottleneck bandwidth; (b) Throughput of TCP and multicast under 100 M bottleneck bandwidth; (c) Throughput of TCP and multicast under 50 M bottleneck bandwidth and (d) Throughput of TCP and multicast under 10 M bottleneck bandwidth

DF value and the different PSNR of the reference frame between before and after transmission in networks.

## PERFORMANCE ANALYSIS AND EXPERIMENTS

The ASM-CBCC is implemented with network simulator (Huber, 2004) and evaluated in present simulated experiments. In simulated scenarios, a multicast session and a TCP connection are established to test the TCP-friendliness of the scheme. Here, the first 400 frame video trace of the MPEG-4 FGS encoded film The Firm (Huber, 2004) and FTP are the traffic of the multicast session and the TCP connection, respectively. Their throughputs under different shared link bandwidth, as showed in Fig. 5, indicate that the ASM-CBCC is in general fair toward the competing TCP connection and multicast session.

We also test the inter-session fairness of the scheme and its capability of coping with heterogeneous networks. According to the PSNR computing method, the perceived PSNR quality of MPEG-4 FGS video sequences in this scenario is approximated. The results as shown in Fig. 6 indicates that the scheme has good inter-session

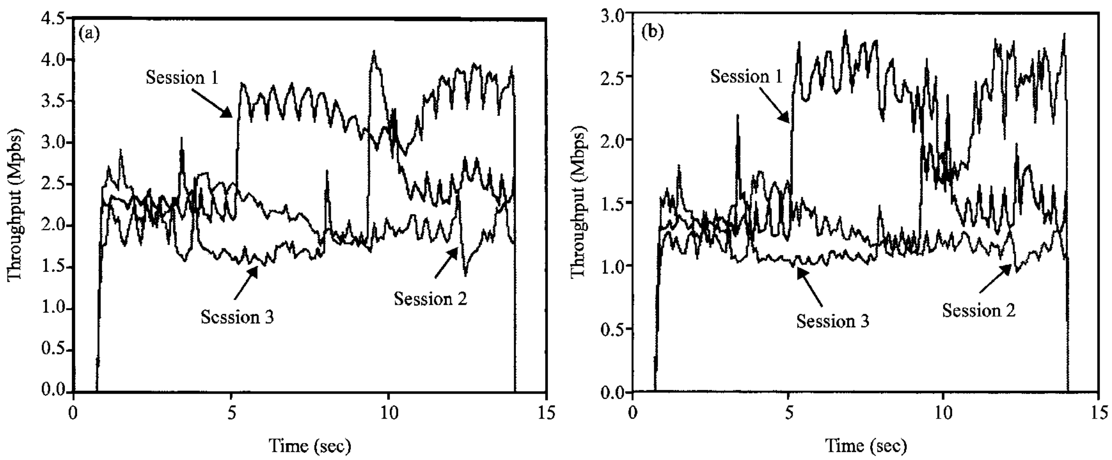


Fig. 6: (a) Original throughput of the three multicast under 100 M bottleneck bandwidth and (b) Throughput of the three multicast under 100 M bottleneck bandwidth

fairness and robust capability to deal with the heterogeneity of networks.

The same simulated experiments with large-scale subscribers reveal the nicer stability of the scheme. And subscribers also could obtain the same video quality as that with small-scale receivers.

## CONCLUSION

Simulated experiments show the permanent stability of the ASM-CBCC with flexible scalability and unprejudiced fairness and TCP-friendliness. Compared with layered multicast, the simple, robust and effective scheme is suitable for deploying in wireless circumstances with heterogeneous subscribers. At the same time, a novel method to calculate PSNR value for MPEG-4 FGS video sequences in simulated networks is also proposed to evaluation of MPEG-4 FGS video transmission.

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