論文 96-21-6-15

Traffic Characteristics and Effective Bandwidth Estimation for MPEG Sources

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MPEG 입력 신호원의 트래픽 특성과 효과적 대역폭 추정

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ABSTRACT

One of the important issues for multimedia communications over ATM networks is efficient use of net-work resources since transmitting video at low cost requires high utilization of channel bandwidth. In this paper, we propose an effective bandwidth estimation scheme for MPEG sources in ATM networks. It has been known difficult to allocate effective transmission bandwidth to MPEG sources due to its bursty characteristics. The MPEG traffic stream can be modeled using five traffic parameters: service rate, burst load ratio, interburst load ratio, burst period, and interburst period. Using these parameters, a transmission bandwidth for each source can be estimated, and the estimated bandwidth is applied synchronously to reduce the queue size in a buffer. For a simple network model of a high speed link that multiplexes a number of virtual-circuit connections, simulations on adaptive bandwidth allocation were performed, and the results show that the queueing delay is significantly reduced, when compared to a fixed bandwidth allocation.

요 약

비디오 데이터를 전송할 때 채널 대역 효율을 최대화하는 문제 때문에 ATM 망에서 멀티미디어 통신의 중요한 사항은 효과적인 망입력 데이터들의 활용문제이다. ATM 망에서 MPEG 입력데이타의 효과적 대역폭 추정방식을 제안하였다. MPEG 데이터의 무작위, 대용량 특성으로 인하여 효과적으로 전송 대역폭을 할당하는 문제는 일반적으로 어려운 문제이다.

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論文番號:96061-0213
接受日字:1996年 2月 13日

MPEG 트래픽 스트림을 서비스처리 속도, 버스트 입력율, 버스트간 입력율, 버스트주기, 버스트 간격 주기의 5가지트래픽 파라미터를 사용하여 모델링하였다. 각 데이터 입력원의 전송대역폭을 추정할 수 있었으며, 버퍼의 큐(queue) 용량을 줄이기 위하여 이 추정된 대역폭을 동기식으로 적용하였다. 많은 수의 가상 회로 접속을 다중화한 고속 링크의 망모델에서 적응 대역 할당 방식의 성능을 시뮬레이션 하였다. 그 결과 고정 대역 할당 방식에 비하여 큐잉 지연이 상당히 감소되었다.

I. Introduction

Asynchronous Transfer Mode (ATM) networks provide variable-rate high-speed channels, making variable rate transmission possible even for real-time video signals. From the standpoint of network design, variable bit rate (VBR) video characterization is an important issue for video source modeling and dynamic bandwidth allocation. Many video coding schemes have been proposed for the compression of video signals which generate a high bit rate stream[1-3]. Since the Motion Picture Expert Group (MPEG) video compression scheme has emerged as a standard for multimedia applications[4], there has been a great deal of effort devoted to characterizing MPEG video [5-7]. MPEG video uses a two-layer coding scheme, in which a different coding flow is applied on a picture-by-picture basis to get higher video compression. As a result, video traffic is so bursty compared to packet data that it is known to be difficult to design an efficient bandwidth allocation scheme for VBR video.

Since MPEG-compressed video generates a large volume of synchronous traffic that must be transmitted within a maximum delay, video services generally require high transmission bandwidth and strict average and maximum delay constraints to guarantee quality of service[8]. To achieve high utilization of network resources, networks must provide a mechanism of bandwidth allocation to each virtual circuit connection and an indication of the burstiness of their transmission for long-term congestion control. A number of bandwidth allocation schemes for congestion control that have been proposed have also included simulation results on different classes of traffic in ATM networks[9-10]. One of the key issues in the design of bandwidth allocation mechanism is that of specifying the transmission bandwidth for each virtual circuit connection subject to a guaranteed quality of service requirement[11].

Where video sources generate variable bit rate traffic stream as a result of video compression, it is not easy to find the transmission bandwidth for each connection that satisfies the grade of service. The simplest bandwidth allocation scheme is that of a fixed amount of bandwidth, viz., either the average or the peak bandwidth of the source. If the peak bandwidth is allocated, network utilization will be quite poor due to the wasted bandwidth. If the average rate is allocated, the buffer will overflow during a long burst and further result in a large queueing delay. Hence, estimating an transmission bandwidth that is typically slightly larger than the average bandwidth of a source and allocating this bandwidth to the source can significantly reduce queueing delay, while maintaining high efficiency of network resources, given a finite buffer size.

In this study, we design a traffic descriptor which characterizes traffic variation over time, and investigate the properties of the description parameters. Upon combining the parameters, we estimate the transmission bandwidth for each connection. The transmission bandwidth for a source is determined by the current bandwidth multiplied by the average load for a time-span. After determining the transmission bandwidth for all of the connections, an appropriate bandwidth for each connection is estimated and updated to control the queue.

The paper is organized as follows. In Section 2, MPEG video coder is described. In Section 3, a new traffic descriptor is developed by characterizing traffic variation over time. The traffic stream is modeled as a series of bursts and interbursts. The load parameters and the transmission bandwidth are also defined. In Section 4, experimental results are shown and discussed. Finally, our conclusions from this study are described in Section 5.

II. MPEG video coder

MPEG has been a standard for multimedia applications since 1992[12]. The first phase is MPEG-1 at 1.5 Mbps rate. Now a couple of different phases, e.g., MPEG-2 at 6 to 10 Mbps rate and MPEG-4 at low-bit rate, are under standardization by the MPEG working group. In this section, the mechanism of MPEG video coder will be briefly described.

1. Motion Picture Experts Group (MPEG)

MPEG is one of the International Standard Organization (ISO) working groups. It is a part of ISO/IEC/JTC1/SC2/WG11 which has undertaken an effort to develop a standard for video and associated audio on telecommunication channels as well as local area networks. The MPEG activities cover more than video compression, since the compression of the associated audio and the issue of audio-visual synchronization are closely coupled with the video compression. MPEG-video addresses the compression of a video signal and its associated audio to the bit rate of about 1.5 Mbps with an acceptable quality. MPEG standard is a generic standard that means that the standard is independent of a particular applications. Flexibility in coding algorithm is also allowed to encourage the design of enhanced coder.

2. Structure of video data

There are six layers in a hierarchy of MPEG video data. The six layers are video sequence, group of pictures (GOP), picture (or frame), slice, macroblock (MB), and block. A coded video sequence commences with a sequence header which may optionally be followed by a group of pictures header and then by one or more pictures. The video sequence is followed by a sequence end code.

A group of pictures header is intended to assist random access into the sequence. Figure 1 shows an example of GOP. There are three types of pictures that use different coding methods. An Intra-coded (I) picture is coded using information only from itself. A Predictive-coded (P) picture is a picture which is coded using motion compensated prediction from a past I-picture or P-picture. A Bidirectionally predictive-coded (B) picture is a picture which is coded using motion compensated prediction from a past and future I-picture or P-picture.

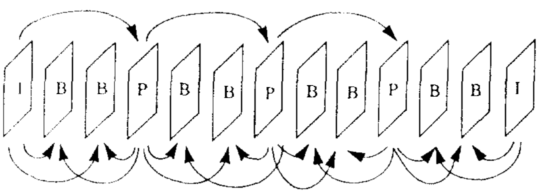

Fig. 1 Example of Group of Pictures (GOP)

In the figure, I is an intrapicture, P is a predictive picture, and B is a bidirectionally-predictive picture. Applications requiring random access, fast-forward playback, or fast-reverse playback may use relatively short group of pictures.

A slice is a series of an arbitrary number of macroblocks. Every slice shall contain at least one macroblock. Slices shall not overlap. A Macroblock (MB) contains a section of the luminance (Y) component and the spatially corresponding chrominance (Cr, Cb) components. A matrix of 16×16 pixels makes a macroblock. In 4:2:0 chroma format, a macroblock consists of 4 Y, 1 Cb, and 1 Cr blocks. A Block is a matrix of 8×8 pixels. The block is a coding unit for discrete cosine transform (DCT). The details for coding flow will be described in the following section.

3. Compression algorithm

An MPEG coder uses two basic techniques: block-based motion compensation for the reduction of temporal redundancy, and transform domain-based compression for the reduction of spatial redundancy. The block diagram of MPEG encoder is shown in Figure 2.

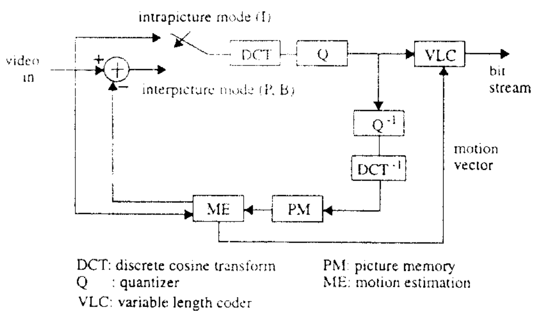

Fig. 2 Block diagram of MPEG video coder

Because of the conflicting requirements of random access and highly efficient compression, three main picture types are defined as explained in the previous section. Intrapictures (I pictures) are coded without reference to other pictures. They provide access points to the coded sequence where decoding can begin, but are coded with only moderate compression. Predictive pictures (P pictures) are coded more efficiently using motion compensated prediction from a past intra-or predictive picture, and are generally used as a reference for further prediction. Bidirectionally-predictive pictures (B pictures) provide the highest degree of compression, but require both past and future reference pictures for motion compensation. Bidirectionally-predicted pictures are never used as references for prediction.

The basic unit of coding within a picture is the macroblock (16×16 pixels). For a given macroblock, a coding mode is chosen as a function of the picture type. Depending on the coding mode, a motion compensated prediction of the contents of the block based on past or future reference pictures is formed. This prediction is subtracted from the actual data in the current macroblock to form an error signal. This error signal is separated into 8×8 blocks and a discrete cosine transform (DCT) is performed on each block. The resulting 8×8 block of DCT coefficients is quantized and the two-dimensional block is scanned in a zig-zag order to convert it into a one-dimensional string of quantized DCT coefficients. Figure 3 shows the DCT transform and its zig-zag scan.

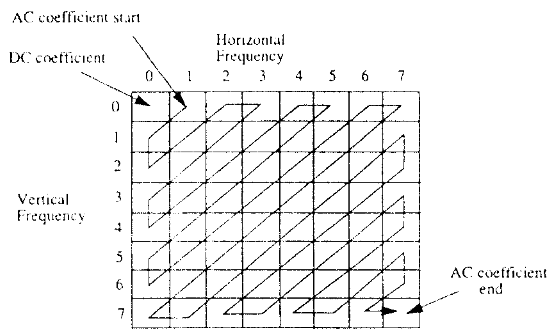

Fig. 3 Zig-zag scan of the DCT coefficients

The coefficients are scanned in a zig-zag order because the low frequency coefficients are scanned first by doing that. In general, low frequency components of image signals have more energy than high frequency components. Run-length coding is used for the quantized coefficient data. Finally the Huffman coding is used for variable length coding. A consequence of using different picture types and variable length coding is that the overall data rate is variable. In the following section, we will look at the statistical characteristics of MPEG video traffic.

4. Traffic characteristics

MPEG video coder uses various coding schemes for each type of pictures and can achieve the compression ratio up to 200. As a result, the generated traffic is highly fluctuating over time domain and called the bursty traffic. The burstiness measure most widely used is the peak-to-average ratio[13]. The peak rate of

MPEG video is up to five times higher than the average rate due to its coding scheme depending on the picture types. The statistical data of MPEG traffic for various video scenes are presented in Table 1. Due to the large fluctuations, the standard deviation is usually comparable to the mean of the traffic.

Table 1. Statistics of MPEG traffic for five video scenes

scenes	mean (cell/frame)	standard deviation (cell/frame)	peak (cell/frame)	peak-to-average ratio
star trek	111	99	474	4.27
football	202	180	1004	4.97
bike	90	86	304	3.38
table tennis	173	182	582	3.36
us	60	52	263	4.38

Samples of real MPEG video traffic at frame level are shown in Figure 4. Inspecting the traffic data at frame level, it has been observed that the rates of I frame and P frame are much higher than the rates of B frames, regardless of the format of the GOP[14]. It is due to the reason that the bidirectional prediction and a large quantization step is used for B frames. It is also observed that the bit rate of I frame is higher than that of P frame except the cases of scene changes

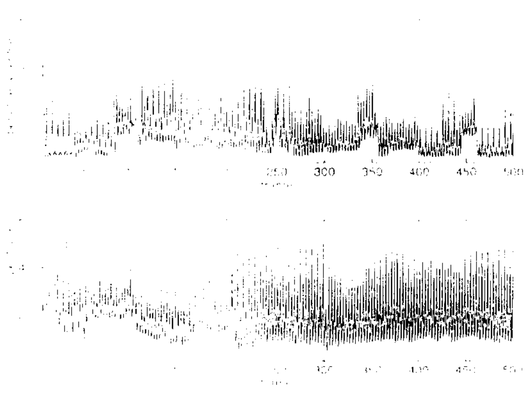

Fig. 4 Samples of real MPEG traffic at frame level
(a) star trek (b) football

and dynamic motion changes. It is due to the fact that P frame is coded like I frame when the energy of prediction error is higher than the energy of original frame.

Since a bursty traffic causes congestion and queueing delay on networks unless the peak bandwidth is not allocated, it is one of the issues to estimate the 'effective' bandwidth for a variable bit rate source in ATM networks[15]. The effective bandwidth is a certain value of transmission bandwidth for a virtual-circuit connection between its peak rate and the average rate which satisfies the requirements of quality of service (QoS). In the next section, we will describe our traffic model for VBR sources in networks and how to estimate the effective bandwidth of the sources.

Ⅲ. Effective bandwidth of MPEG sources

We have investigated the burstiness of MPEG sources and proposed a new traffic descriptor to understand the characteristics of the sources and to model them. Since MPEG has been standardized for multimedia applications, a number of studies have been conducted on the description of MPEG video traffic [16-18]. The purposes of our traffic descriptors are as follows: (1) to predict traffic variation over time and (2) to reduce the effect of bursty traffic on network congestion and queueing delay. Since the traffic variation over time must be expressed in terms of its effect on network congestion, given a bandwidth for the source, our traffic descriptor uses an approach to characterize the overload during a bursty period and find an transmission bandwidth for each connection.

1. Traffic model

MPEG video source generates dynamically varying traffic that alternates between active and silent periods. The duration of the active and the silent periods depend on the video coding scheme, motion changes, and scene changes. If a service rate is allocated to a video source, the fluctuating traffic stream can be modeled as a sequence of bursts and interbursts[19]. Figure 5 illustrates the definition of bursts and interbursts for a given service rate. In the figure, given the service rate m, a burst period is defined as the time interval in which the cell rate of traffic is greater than the service rate, and an interburst period is defined as the time interval between two consecutive bursts.

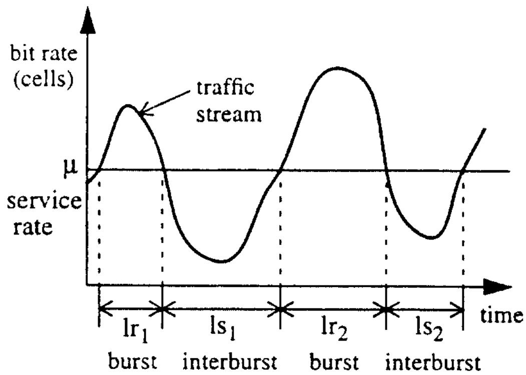

Fig. 5 Definition of bursts and interbursts.

For the same traffic pattern, the periods of burst and interburst depend on the service rate allocated to the source. The time unit in the definition of bursts and interbursts is a time interval in which the cell rate of traffic is measured. For example, the unit is a slice or frame for VBR video. The burst and interburst, therefore, are measured at the slice level in this study. The slice is a horizontal strip of macroblocks in a picture.

2. Burst load and Interburst load

In the previous section, the burst and interburst were defined for a given service rate. For a fixed service rate, the number of cells that arrive during a burst interval is called the burst load. When the cell arrival rate is greater than the service rate, which means that more cells than can be served arrive, the network is overloaded during a burst period and the overloaded cells accumulate in a buffer. On the other hand, the network is underloaded during an interburst period and the cells in the buffer are drained. Figure 6 illustrates the concept of the burst load and the interburst load.

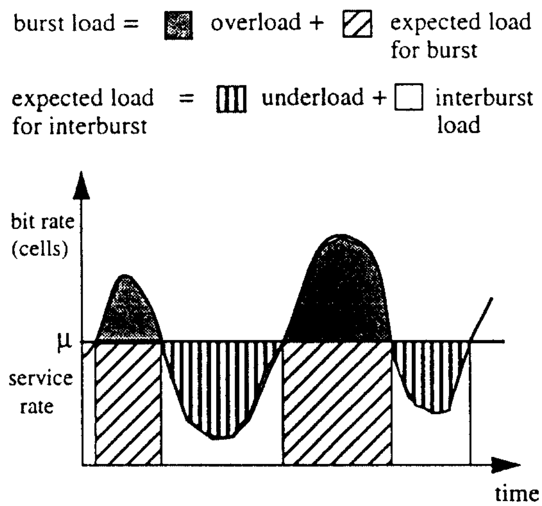

Fig. 6 Burst load and interburst load.

The expected load during a burst interval is given by the service rate multiplied by the corresponding burst period. The burst load is equivalent to the overload plus the expected load during the burst period where the overload is the number of cells accumulated in a buffer during the burst period. The interburst load is shown in the figure, and the expected load during an interburst is equal to the interburst load plus the underload where the underload is the number of cells drained from the buffer during the interburst period. The absolute value of burst load or interburst load varies since it depends on the cell rate of the source. To evaluate the impact of the traffic variation on the network, an appropriate traffic descriptor is required.

3. Traffic description parameters

In this section, a new traffic descriptor is defined using the concept of the burst load and the interburst load. To evaluate how much overload (underload) a burst (an interburst) brings to the network, the total number of cells during a burst (an interburst) period is measured, and the ratio of the total load during a burst (an interburst) to the expected load during the same time period can be parameterized to give the degree of the overload (underload). Since there will be a number of bursts and interbursts that alternate over time, the average values of the burst load ratio and the interburst load ratio can be significant parameters to specify the overall traffic variation. Hence, the definition for burst load ratio (r) and interburst load ratio (s) are defined by

$$r = mean\,[B_k/E_k] \qquad (1)$$

where B_k is the burst load of the k-th burst and E_k is the expected load during the burst; and

$$s = mean\,[I_k/E_k] \qquad (2)$$

where I_k is the interburst load of the k-th interburst and E_k is the expected load during the interburst.

The burst load ratio expresses the degree of overload during a time period, while the interburst load ratio expresses the degree of underload during the same period. The overload parameter that is the burst load ratio can also be used to specify a burstiness measure[20]. The average burst period and the average interburst period are represented by lr and ls respectively.

So far, given a service rate allocated to a source, we have defined the traffic parameters called the burst load ratio, interburst load ratio, burst period, and interburst period. A traffic descriptor that is composed the five parameters (m, r, s, lr, ls) gives a complete description of traffic variation over time. In the next section, we will show that our traffic descriptor is effective in describing the impact of bursty traffic on video networks with respect to network congestion and queueing delay.

4. Properties of traffic parameters

The properties of our traffic descriptor have been investigated using real MPEG video streams. A 50 second 'football' scene at the slice level has been generated from a software MPEG encoder modified from the Stanford PVRG (portable video research group) version. The resolution of a picture is 352×240, where a slice is defined as a horizontal strip of 16 rows that is a sequence of macroblocks, though the slice can be defined as any set of macroblocks in a frame. Hence, a picture consists of 15 slices.

The burst load ratio has the property that is inversely proportional to the service rate, while the interburst load ratio does the opposite. It is explicit from the definition that the burst load ratio decreases as the reference service rate increases, and the opposite holds for the interburst load ratio. Figure 7-(a) and (b) show the sample traffic stream of the 'football' scene and its distribution. Since the traffic stream

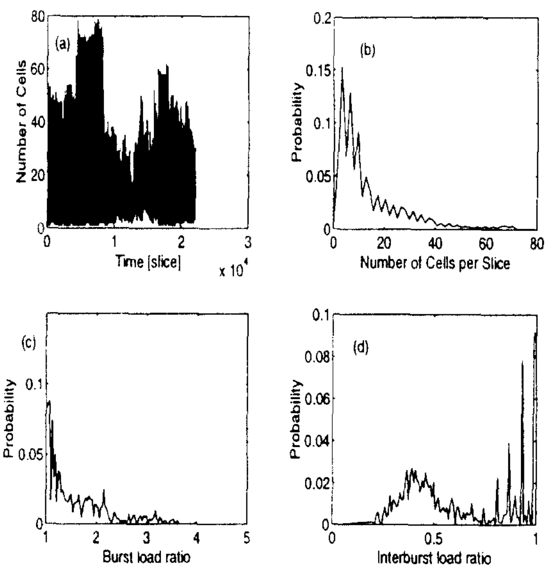

Fig. 7 Traffic parameters of 'football' scene.
(a) traffic at slice level
(b) distribution of slice traffic
(c) distribution of burst load ratio
(d) distribution of interburst load ratio

shows a characteristic that is fluctuating quite a bit, the allocation of the average bandwidth to the source resulted in a long queueing delay. The distributions of the burst load ratios and the interburst load ratios for the whole stream, given the average service rate are shown in Figure 7-(c) and (d). It is seen that the burst load ratio increases to 5. This implies that five times more cells than expected have arrived during the burst period.

Most values of the interburst load ratio are around 0.5 which implies that about half of the expected cells have arrived during an interburst. Overall, the interburst period is longer than the burst period. We note that the distribution of burst load ratio resembles the Exponential distribution, while that of the distribution of the interburst load ratio looks like the Gaussian distribution with the mean of 0.5 except for some peak values near 1.

5. Estimation of effective bandwidth

The notion of transmission bandwidth is that of minimum transmission bandwidth allocation to each video source subject to the quality of service requirements. The transmission bandwidth is a value between the average rate and the peak rate of the source. With the same average rate, the transmission bandwidth is larger for a bursty traffic because the cells in a bursty source have a tendency of batch arrival. Hence, a larger bandwidth should be allocated to a more bursty source to maintain the same quality of service.

Using the traffic descriptor parameters designed in the previous section, an average load ratio a is defined by the period-weighted average of the burst load and the interburst load is given by

$$a=(r \cdot l_r + s \cdot l_s)/(l_r + l_s) \qquad (3)$$

The average load ratio signifies an index of average cell arrivals in terms of multiples of the given service rate. By multiplying the service rate, it becomes the average number of cells that arrive during a time-span. The effective bandwidth for the time frame is then given by

$$\mu' = \alpha \cdot \mu \qquad (4)$$

where a is the average load ratio and the m is the current bandwidth. This equation means that the effective bandwidth is updated to the current bandwidth multiplied by the factor of the average load ratio. In conclusion, the significance of the effective bandwidth for a time frame is the average cell arrivals during the previous time frame.

The traffic analysis at cell interarrival level gives a microscopic view of traffic variations. On the other hand, since our traffic model gives a macroscopic view of the traffic variations, it is more feasible for real-time implementation than a cell-level model. In the following section, we will discuss simulation results on bandwidth adaptation using the concept of the effective bandwidth for real MPEG traffic.

Ⅳ. Experimental results

Some preliminary simulation experiments have been conducted in order to test the performance of the proposed bandwidth estimation scheme. The results show that our scheme is promising in the design of ATM networks for achieving both the efficient use of network resources and small queueing delay. In the simulation, the MPEG software encoder was used to generate VBR traffic streams that are multiplexed onto a single link. The bit rates of the traffic are measured at the slice level to apply our bandwidth allocation scheme for each source.

We conducted several simulations to study the performance of our scheme. First, the performance of a fixed bandwidth allocation with the average and the peak bandwidth of the source was investigated. The variation of the queue size under the assumption of an infinite buffer size is shown in Figure 8-(a). The peak value of the queue size was 4801 cells, and the average queue size was 1243 cells which resulted in a large queueing delay. At the other extreme, the peak bandwidth was allocated. The resulting channel utilization was 0.12, an obvious waste of bandwidth. As a result, both the average rate and the peak rate allocations are not effective for MPEG video. The transmission bandwidth for the source was estimated using the traffic descriptor parameters, and the results of the queue size variation via our bandwidth adaptation scheme is shown in Figure 8-(b). The adaptation period was 220 ms, which implies that the estimated transmission bandwidth is adapted every 100 slices. The simulation results showed that the peak queue size decreased to 552 cells, and the average queue size was 54 cells, which is approximately 4% of the average bandwidth allocation. The channel utilization is 0.92 which is an effective utilization of the network.

The performance of our bandwidth allocation scheme was also investigated with respect to the adaptation period as shown in Figure 9. In the figure, the channel utilization, the maximum queue size, and the average queue size are presented with respect to the adaptation period. The adaptation period varies from 0 sec (peak-rate allocation) to 3 sec (This is similar to the case of average bandwidth allocation). High channel utilization is obtained in overall, but the average queue size and the maximum queue size increase rapidly as the adaptation period is getting longer. This implies that the queueing delay is affected quite a bit at the cost of more processing time (frequent adaptation), but the channel utilization is not so critical to the frequency of adaptation.

A key result of the queue size are presented in Table 2 for a performance comparison of the fixed rate allocation and the adaptive bandwidth allocation. For both cases, the same amounts of bandwidth were allocated to the source. In the second column of the table, the average and the maximum queue sizes of fixed bandwidth allocation are presented for a single source. In the third column of the table, the result of applying our bandwidth allocation scheme are given. From the result, it is clear that our adaptive bandwidth allocation scheme is effective in reducing the

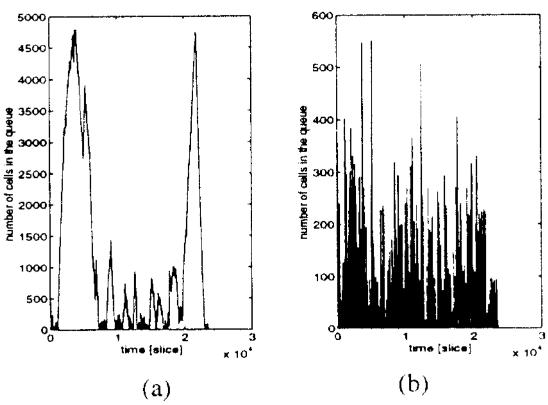

Fig. 8 Queue size variation.
(a) fixed bandwidth allocation
(b) dynamic bandwidth allocation (adaptation period = 220 ms)

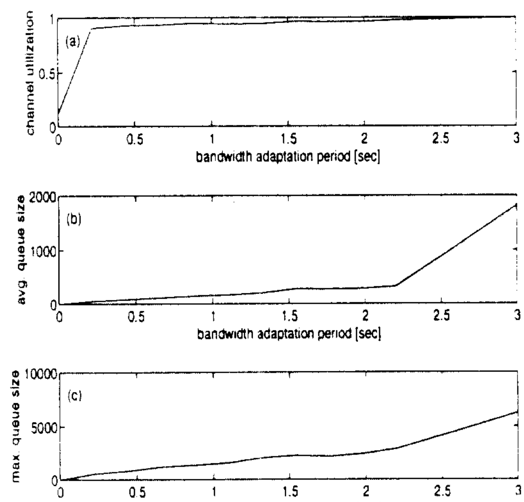

Fig. 9 Performace of bandwidth adaptation schemes vs. the adaptation period
(a) channel utilization
(b) average queue size
(c) maximum queue size

queueing delays for both single and multiple video sources.

Table 2. Comparison between fixed bandwidth allocation and adaptive bandwidth allocation

	fixed allocation	adaptive allocation
average queue size	1234	54
maximum queue size	4801	552

V. Conclusions

In this paper, we have proposed a scheme to estimate the transmission bandwidth for MPEG sources based on the traffic description approach to achieve high network utilization. MPEG video has bursty characteristics in nature, since the coding scheme is highly dependent on the types of pictures. The bit rate of MPEG video is also dependent on the picture types. The bursty MPEG video is modeled using five traffic parameters viz., service rate, burst load ratio, interburst load ratio, burst period, and interburst period. The traffic parameters are easy to measure, and have nice properties that describe the effect of traffic variation on network congestion and queueing delay. Since our traffic model gives more macroscopic viewpoint than cell level models, it is feasible for real-time implementation with less processing time.

A synchronous bandwidth adaptation scheme has been simulated using the traffic parameters, and shown to be effective in reducing the queueing delay with efficient use of network resources. As a result, both high channel utilization and low queueing delay can be achieved by applying our bandwidth adaptation scheme to real MPEG traffic. The simulation results also show that the maximum queue sizes are reduced significantly compared to that of fixed bandwidth allocation. This is a desirable characteristic for video traffic, since the maximum queueing delay is more critical than the average in delay-sensitive traffic like video.

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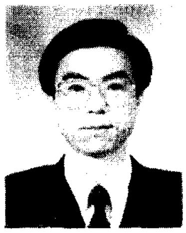

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