JOURNAL OF INDUSTRIAL AND
MANAGEMENT OPTIMIZATION
Volume **5**, Number **3**, August **2009**

doi:10.3934/jimo.2009.5.541

pp. **541–552**

TRAFFIC MODELLING AND BANDWIDTH ALLOCATION ALGORITHM FOR VIDEO TELEPHONY SERVICE TRAFFIC

Bong Joo Kim and Gang Uk Hwang

Department of Mathematical Sciences and Telecommunication Engineering Program
Korea Advanced Institute of Science and Technology (KAIST), Daejeon 305-701, Korea

Yeon Hwa Chung

R&D Division, Corporate Customer Supporting BU, Business Group, KT
KT Daeduk 2 Center, Daejeon 305-811, Korea

Abstract. In this paper, we analyze the stochastic characteristics of the measured video telephony service traffic and propose a mathematical model of the video telephony service traffic based on the analyzed stochastic characteristics. From our analysis, a voice traffic can be modelled by a renewal process with deterministic inter-arrival times and fixed packet size. On the other hand, a video traffic is modelled by an on and off source, where on and off periods are according to gamma distributions with respective parameters. Using the mathematical model, we estimate the required bandwidth to satisfy a given Quality of Service (QoS) requirement for the video telephony service traffic. To show the validity of our analysis, numerical studies with the NS-2 simulator are provided.

1. **Introduction.** To provide the real-time video applications, such as video conference and video telephony in the network, it is important to forecast the pertinent amount of bandwidth for a given Quality of Service (QoS) requirement because such applications need usually large amount of bandwidth and can drain network resources [4], [10].

To solve this problem, traffic modelling and bandwidth estimation have been examined extensively in the literature. Saito *et al.* [9] analyzed the characteristics of voice traffic generated by 32 kb/s Adaptive Differential Pulse Code Modulation (ADPCM) and video traffic generated by variable rate coding in an ATM network, and then compared the deterministic and statistical bandwidth allocation methods when video and voice traffic are multiplexed in the ATM network. Chou and Chang [2] estimated the amount of bandwidth in high speed digital networks using the effective bandwidth theory, and experimented the validity with simulated Markov traces and the actual video traces. In [7], Bléfari-Melazzi *et al.* proposed a traffic model, called Renewal Approximation (RA) for ITU-T H.263 video streams which takes into account the different picture types and Group Of Pictures (GOP), and then presented performance evaluation studies with experimental videophone traffic

2000 *Mathematics Subject Classification.* Primary: 90B22; Secondary: 90B15, 90B18.

Key words and phrases. Video telephony service traffic, bandwidth allocation, QoS, fluid flow analysis.

This work was supported by the Korea Science and Engineering Foundation(KOSEF) grant funded by the Korea government(MOST) (No. R01-2007-000-20053-0).

traces. Koutsakis and Paterakis [6] proposed Call Admission Control (CAC) and traffic policing mechanisms for transmitting the encoded videoconference movies in wireless ATM networks. From these previous studies, we find that the stochastic characteristics of the real-time video application intensely affect the required amount of bandwidth and network performance. Therefore, to predict the required bandwidth, it needs to develop an appropriate mathematical model capturing the stochastic characteristics of the video telephony service traffic.

This paper focuses on a mathematical method to estimate the required bandwidth to satisfy a given QoS requirement for the video telephony service traffic consisting of the voice and video traffic. We first analyze the stochastic characteristics of the actual video telephony service traffic which is obtained by measuring at an L3 switch of an access network, and then develop a simple mathematical model that captures the analyzed stochastic characteristics. From our analysis, we see that the voice traffic can be modelled by a renewal process with deterministic inter-arrival times and fixed packet size, and accordingly, the deterministic bandwidth allocation is considered for the voice traffic. On the other hand, we see that the video traffic can be modelled by a fluid flow model with alternating on and off periods. We then apply the fluid flow analysis and develop a simple and efficient bandwidth allocation algorithm to satisfy a given QoS requirement for the video traffic. Although we propose a bandwidth allocation algorithm based on the actual video telephony service traffic which is measured at the specific region, the codecs (ITU-T G.711 and ITU-T H.263) to generate the voice and video traffic in our experiment are widely used in practice [1], [5]. Hence, the application of our bandwidth allocation method is not limited and it can be applied to any video telephony service traffic generated according to ITU-T G.711 and ITU-T H.263.

The remainder of this paper is organized as follows. In Section 2, we investigate the stochastic characteristics of the measured video telephony service traffic and propose a mathematical model that captures the analyzed stochastic characteristics. In Section 3, we propose an algorithm to estimate the required bandwidth to satisfy a given QoS requirement for the video telephony service traffic. Section 4 provides numerical studies with the NS-2 simulator to show the validity of our bandwidth allocation algorithm. Finally, we give our conclusions in Section 5.

2. Stochastic characteristics of traffic and analysis.

Video telephony service traffic was measured in Daejeon, South Korea on May, 24, 2005, 15:40∽15:57. The video telephony service traffic packets are captured through a port mirroring at an L3 switch in an access network of a telecommunication company in South Korea, and the captured traffic packets are classified by user's IP addresses. The video telephony service traffic usually consists of a voice traffic based on ITU-T G.711 and a video traffic based on ITU-T H.263. For this video telephony service traffic, video traffic and voice traffic are first separated in the system, and then encoded independently with time index. In addition, they are transmitted throughput independent channels. Therefore, in this study, we consider voice traffic and video traffic separately and independently. The stochastic characteristics of the video telephony service traffic between two tagged IP users are summarized in TABLE 1.

2.1. Characteristics of measured voice traffic.

ITU-T G.711 is used to generate the voice traffic. The G.711 codec [11], also known as Pulse Code Modulation (PCM), is a very commonly used waveform code. In ITU-T G.711, basically a voice packet is generated every 20 ms and the payload size is fixed of size 160 bytes.

TABLE 1. Stochastic characteristics of video telephony service traffic.

voice traffic	inter-arrival time (sec)	mean	1.9998e-002
		variance	3.2793e-006
		scv*	0.0082
	packet size (bytes)		214
video traffic	on period (sec)	mean	6.5350e-003
		variance	5.1882e-006
		scv	0.1215
	off period (sec)	mean	4.7663e-002
		variance	1.5100e-004
		scv	0.0665
	packet size (bytes)		1514 & the rest**

* squared coefficient of variation

** All video packets arriving during on periods except the last packet are of size 1514, but the last packet size is variable in [1, 1514].

Therefore, as shown in TABLE 1, we observe that the measured voice traffic has almost deterministic inter-arrival times and the packets in the voice traffic are of fixed size 214 bytes which include RTP/UDP/IP header and MAC header. Therefore, the voice traffic can be mathematically modelled as a renewal process with deterministic inter-arrival times and fixed packet size.

2.2. **Characteristics of measured video traffic.** ITU-T H.263 video encoding scheme is used to generate the video traffic, and the encoded video traffic packets are transmitted to the IP network after a rate control according to the buffer state in the encoder [12]. From FIGURE 1, we observe that the video traffic has alternating on and off periods, where there are packet arrivals during on periods and no packet arrivals during off periods. To find the appropriate probability density functions (pdf) of the lengths of on and off periods, we checked the histograms and the Q-Q plots of the lengths of on and off periods, respectively. We first compared the histograms of the lengths of on and off periods with distributions of various kinds. In this study, we chose three candidate distributions, the exponential distribution, the normal distribution, and the gamma distribution. For the Q-Q plots, we also tried the exponential distribution, the normal distribution, and the gamma distribution. As shown in FIGURE 2 and FIGURE 3, we see that the gamma distribution is the most suitable distribution for the lengths of on and off periods. Note that there are some mismatched data values for the off periods in the Q-Q plots when we consider the gamma distribution. Refer to FIGURE 3-(b). However, the portion of the mismatched data is about 0.245% of total data. So, it can be negligible. In fact, we later confirm by simulation studies (See FIGURE 5 and 6) that the mismatched data can be negligible when we estimate the required bandwidth for a video telephony service traffic. Therefore, from our analysis, the video traffic can be modelled mathematically by an on and off source, where on and off periods are according to gamma distributions with respective parameters. The respective parameters in gamma distributions are obtained from means and variances of on and off periods given in TABLE 1. The respective parameters in gamma distributions for on and off periods are as follows:

$$\begin{aligned} \text{on period duration} &\backsim \text{Gamma}\Big[8.23, \frac{1}{0.000794}\Big] \\ \text{off period duration} &\backsim \text{Gamma}\Big[15.04, \frac{1}{0.003168}\Big], \end{aligned}$$

where the pdf $f(x)$ of a gamma distribution with parameters α and λ, denoted by Gamma(α, λ), is given by $f(x) = \frac{\lambda^\alpha x^{\alpha-1} e^{-\lambda x}}{\Gamma(\alpha)}$, $x > 0$, where $\Gamma(\alpha) = \int_0^\infty x^{\alpha-1} e^{-x} dx$, $\alpha > 0$. In addition, from our analysis, we find that the packet inter-arrival times during on periods are also according to a gamma distribution with parameters $\alpha = 1.43$ and $\lambda = \frac{1}{0.001}$, and the number of packets arriving during an on period is almost either 5 or 6.

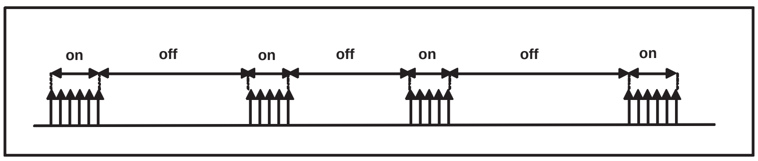


Figure 1. The arrival pattern of a video traffic.

3. Bandwidth allocation analysis.

3.1. Bandwidth allocation for a single voice traffic.

Since the voice traffic can be modelled by a renewal process with deterministic inter-arrival times and fixed packet size, we consider the deterministic bandwidth allocation for the voice traffic. Let BW_{voice} be the required bandwidth for the voice traffic. Then, BW_{voice} is given by

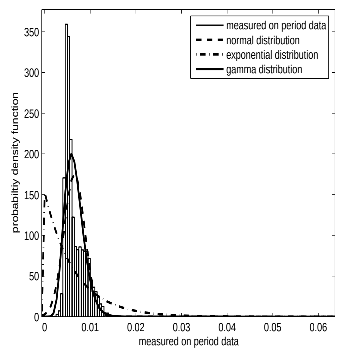
$$BW_{voice} = \frac{X}{E[D]}, \tag{1}$$

where X denotes the packet size in units of bits and $E[D]$ denotes the estimated expectation of the packet inter-arrival time in seconds.

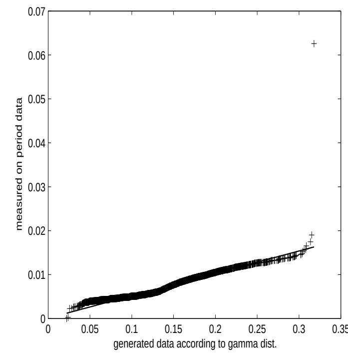
3.2. Bandwidth allocation for a single video traffic.

For the video traffic, on and off periods are according to gamma distributions with non-integer parameters. To make our mathematical model simple, we approximate Gamma(α, λ) distributions for on and off periods by a mixture of Gamma(n, λ) distributions where n are integers. Note that, with this simple approximation, we can construct a Continuous Time Markov Chain (CTMC) to model the video traffic, and the fluid flow analysis can be applied to estimate the pertinent bandwidth for the video traffic. In our approximation for the on period, since the original parameter α is 8.23, we let $n = 8$ with probability 0.8 and $n = 9$ with probability 0.2. Then, the average of n is equal to 8.2 ($\approx \alpha = 8.23$). For the off period, since the original parameter α is 15.04, we let $n = 15$. In summary,

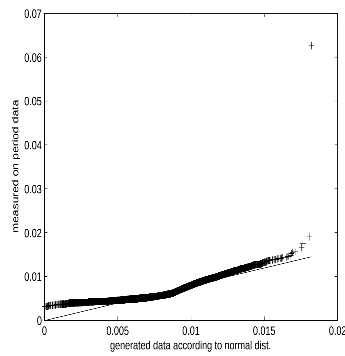
$$\text{on period duration} \backsim \text{Gamma}\Big[8 \text{ with } p \text{ or } 9 \text{ with } q, \frac{1}{0.000794}\Big],$$



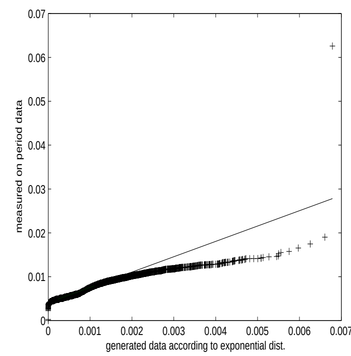
(a) histogram



(b) Q-Q plot with gamma dist.



(c) Q-Q plot with normal dist.



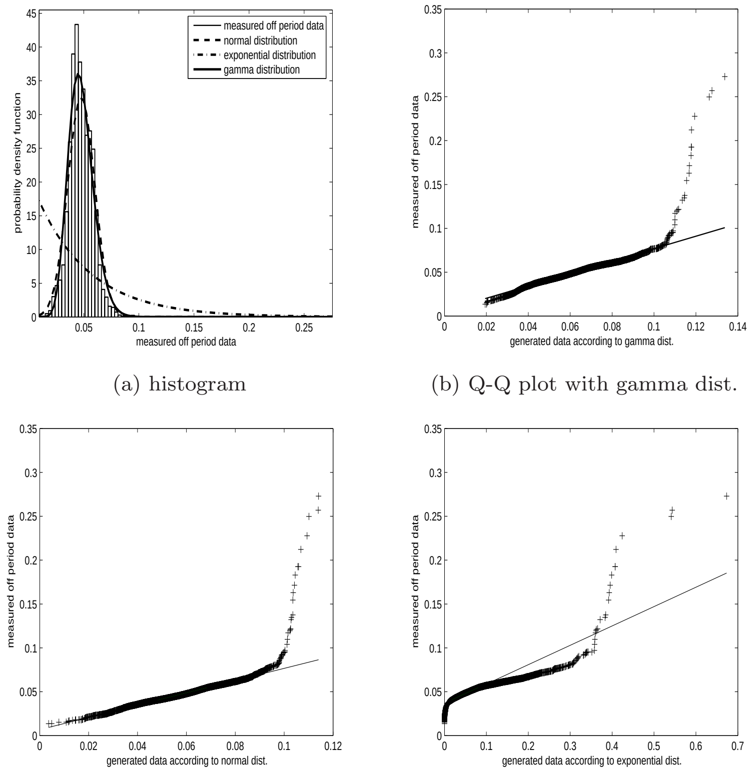
(d) Q-Q plot with exponential dist.

FIGURE 2. Histogram and Q-Q plots for on period

$$\text{where } p=\text{P}\{\text{n}=8\}=0.8,\ q=\text{P}\{\text{n}=9\}=0.2,$$

$$\text{off period duration} \ \backsim \ \text{Gamma}\Big[15, \frac{1}{0.003168}\Big].$$

Then, we construct a CTMC with state space $\{1, 2, \cdots, 9, 10, \cdots, 24\}$ to model alternating on and off periods. To model the on period, we use the subset $\{1, 2, \cdots, 9\}$ as the state space of on periods. The state transitions of the CTMC during the on period are as follows. For state $i(\leqslant 7)$, the CTMC always has state transitions to state $(i+1)$, for state $i = 8$, the CTMC has state transitions to state 10 with probability 0.8 and to state 9 with probability 0.2, and for state $i = 9$, the CTMC always has state transitions to state 10. Note that, when the CTMC visits state 10 from its lower states 8 or 9, it implies that the on period ends. To model the off period, we use the remaining states $\{10, 11, \cdots, 24\}$ as the state space of off periods. The state transitions of the CTMC during the off period are as follows. For state $i(\leqslant 23)$, the CTMC always has state transitions to state $(i+1)$ and for state $i = 24$, the CTMC always has state transitions to state 1, which implies the end of the off period. Since the lengths of on and off periods are according to gamma distributions, we assume that the sojourn times in states are according to exponential distributions. Let λ_{on} be the parameter of the exponential distribution for state i $(1 \leqslant i \leqslant 9)$. Then, the expectation of the length of the on period for the CTMC is $\frac{8.2}{\lambda_{on}}$. From TABLE 1,



(a) histogram (b) Q-Q plot with gamma dist.

(c) Q-Q plot with normal dist. (d) Q-Q plot with exponential dist.

FIGURE 3. Histogram and Q-Q plots for off period

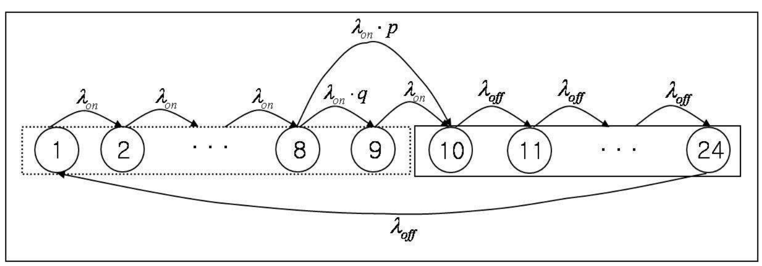


FIGURE 4. State transition diagram for CTMC.

by letting $\frac{8.2}{\lambda_{on}} = 6.535 \times 10^{-3}$, we get $\lambda_{on} = 1.255 \times 10^3$. Next, let λ_{off} be the parameter of the exponential distribution for state i ($10 \leqslant i \leqslant 24$). Then, similarly as above, we have $\lambda_{off} = 3.147 \times 10^2$. The state transition diagram for CTMC is given in FIGURE 4.

Let $\boldsymbol{Q} = (q_{ij})_{24\times 24}$ be the transition rate matrix of the CTMC described above. From our analysis, $\boldsymbol{Q}$ is given by

$$\boldsymbol{Q} = \left[\begin{array}{ccccc|ccccc}
-\lambda_{on} & \lambda_{on} & 0 & \cdots & 0 & 0 & 0 & 0 & \cdots & 0 \\
0 & -\lambda_{on} & \lambda_{on} & \cdots & 0 & 0 & 0 & 0 & \cdots & 0 \\
0 & 0 & -\lambda_{on} & \cdots & 0 & 0 & 0 & 0 & \cdots & 0 \\
\vdots & \vdots & & \ddots & \vdots & \vdots & & \ddots & & \vdots \\
0 & & \cdots & -\lambda_{on} & \lambda_{on}\cdot q & \lambda_{on}\cdot p & 0 & 0 & \cdots & 0 \\
0 & & \cdots & 0 & -\lambda_{on} & \lambda_{on} & 0 & 0 & \cdots & 0 \\
\hline
0 & 0 & 0 & \cdots & 0 & -\lambda_{off} & \lambda_{off} & 0 & \cdots & 0 \\
0 & 0 & 0 & \cdots & 0 & 0 & -\lambda_{off} & \lambda_{off} & \cdots & 0 \\
\vdots & \vdots & \vdots & \ddots & \vdots & \vdots & & \ddots & & \vdots \\
0 & 0 & 0 & \cdots & 0 & 0 & & \cdots & -\lambda_{off} & \lambda_{off} \\
\lambda_{off} & 0 & 0 & \cdots & 0 & 0 & & \cdots & 0 & -\lambda_{off}
\end{array}\right].$$

Let r be the arrival rate during on periods. For simplicity, we assume that the arrival rate during on periods is constant, i.e,

$$r = \frac{\text{E[amount of the packets arriving during on period]}}{\text{E[length of on period]}}.$$

Since the estimated expectation of the amount of the packets arriving during an on period is 7.82487 (Kbytes), r is equal to 9.57903 (Mbps)$\left(= \frac{7.82487\times 10^3\times 8}{6.535\times 10^{-3}}\right)$. Then, the video traffic can be modelled by a fluid flow model with alternating on and off periods. With the fluid flow model made so far, we use the fluid flow analysis to compute the pertinent bandwidth for the video traffic corresponding to a given QoS requirement as follows.

Let $\boldsymbol{R}$ be the arrival rate matrix of dimension 24. Note that $\boldsymbol{R}$ is a diagonal matrix such that the i-th diagonal element is $r(= 9.57903)$ for $1 \leqslant i \leqslant 9$ and 0 for $10 \leqslant i \leqslant 24$. That is, $\boldsymbol{R}$ is given by

$$\boldsymbol{R} = \left[\begin{array}{cccc|cccc}
r & & & & & & & \\
& r & & & & & & \\
& & \ddots & & & & & \\
& & & r & & & & \\
\hline
& & & & 0 & & & \\
& & & & & 0 & & \\
& & & & & & \ddots & \\
& & & & & & & 0
\end{array}\right].$$

Let C be the service rate and $\boldsymbol{D}$ be the drift matrix defined by $\boldsymbol{D} \triangleq [\boldsymbol{R} - C\boldsymbol{I}]$, where $\boldsymbol{I}$ is the identity matrix of dimension 24. That is, $\boldsymbol{D}$ is given by

$$\boldsymbol{D} = \left[\begin{array}{cccc|cccc}
r-C & & & & & & & \\
& r-C & & & & & & \\
& & \ddots & & & & & \\
& & & r-C & & & & \\
\hline
& & & & 0 & & & \\
& & & & & 0 & & \\
& & & & & & \ddots & \\
& & & & & & & 0
\end{array}\right].$$

In our fluid flow analysis, we need to find the (generalized) eigenvalue and eigenvector pair $(z_i, \boldsymbol{\psi_i})$ of the following equation:

$$z_i \boldsymbol{\psi_i} \boldsymbol{D} = \boldsymbol{\psi_i} \boldsymbol{Q}. \tag{2}$$

By the definition of the drift matrix $\boldsymbol{D}$, the equation (2) becomes

$$z_i \boldsymbol{\psi_i} [\boldsymbol{R} - C\boldsymbol{I}] = \boldsymbol{\psi_i} \boldsymbol{Q}. \tag{3}$$

If C is considered to be a function of eigenvalues z_i, denoted by $h(z_i)$, then equation (3) is represented as follows:

$$\boldsymbol{\psi_i} \boldsymbol{A}(z_i) = h(z_i) \boldsymbol{\psi_i},$$

where $\boldsymbol{A}(z) \triangleq [\boldsymbol{R} - \frac{1}{z}\boldsymbol{Q}]$.

Let $h_1(z)$ be the maximal real eigenvalue of $\boldsymbol{A}(z)$ which is a real eigenvalue such that $Re[h_k(z)] < h_1(z)$, where $h_k(z)$ is another eigenvalue of $\boldsymbol{A}(z)$. Then, it can be shown that C is equal to the maximal real eigenvalue of $\boldsymbol{A}(z_1)$ from the following proposition [3].

Proposition 1. *For $C \in \left(\bar{\lambda}, \hat{\lambda}\right)$, the dominant eigenvalue z_1 is the unique solution in $(-\infty, 0)$ satisfying*

$$h_1(z_1) = C,$$

where $\bar{\lambda}$ is mean arrival rate and $\hat{\lambda}$ is the peak arrival rate. Here, the dominant eigenvalue z_1 means the largest eigenvalue among negative eigenvalues z_i of the equation (2).

The detailed proof of Proposition 1 is given in [3]. Moreover, we can show that for sufficiently large buffer size x, we have [3],[8]

$$P\{\text{buffer content} > x\} \approx e^{z_1 x},$$

from which we get

$$z_1 = \lim_{x \to \infty} \frac{\log\left[P\{\text{buffer content} > x\}\right]}{x}.$$

This implies that, for a buffer size B and a given packet overflow probability $\epsilon = P\{\text{buffer content} > B\}$, z_1 can be obtained by

$$z_1 \approx \frac{log(\epsilon)}{B}. \tag{4}$$

Then, using z_1 from equation (4), the required bandwidth BW_{video} of the video traffic to satisfy the given packet overflow probability ϵ is obtained by computing the maximal real eigenvalue of $\boldsymbol{A}(z_1)$. In summary,

$$BW_{video} = \text{maximal real eigenvalue of } \boldsymbol{A}(z_1).$$

3.3. **Bandwidth allocation for multiple video telephony service traffic.** Now, we provide how to get the required bandwidth for multiple independent video telephony service traffic. Here, we assume that all video telephony traffic are homogeneous. First, for N multiple voice traffic, since each voice traffic is a renewal process with bandwidth $\frac{X}{E[D]}$ in (1), the required bandwidth for N multiple voice traffic is $\frac{N \cdot X}{E[D]}$. Next, to compute the required bandwidth for N multiple video traffic, we consider the eigenvalue problem for N multiple video traffic in the same manner as in the single video traffic.

Let $\boldsymbol{Q}_k$ and $\boldsymbol{R}_k$ be the transition rate matrix and the arrival rate matrix for k-th individual video traffic, respectively. By the similar derivation for the single video traffic, we can find that for k-th video traffic, there exists a maximal real eigenvalue $h_{k,1}(z)$ such that

$$h_{k,1}(z) > Re[h_{k,2}(z)] \geqslant Re[h_{k,3}(z)] \geqslant \cdots \qquad (z < 0),$$

where $h_{k,i}(z)$ be the i-th eigenvalue of $\boldsymbol{A}_k(z) \triangleq \left[\boldsymbol{R}_k - \frac{1}{z}\boldsymbol{Q}_k\right]$, $1 \leqslant k \leqslant N$. Then, it can be shown that C is equal to the sum of the maximal real eigenvalue of $\boldsymbol{A}_k(z)\,(1 \leqslant k \leqslant N)$ from the following proposition [3].

Proposition 2. *For $C \in \left(\bar{\lambda}, \hat{\lambda}\right)$, the dominant eigenvalue z_1 is the unique solution in $(-\infty, 0)$ satisfying*

$$\sum_{k=1}^{N} h_{k,1}(z_1) = C,$$

where $\bar{\lambda}$ is mean arrival rate of the aggregated traffic, i.e., $\bar{\lambda} \triangleq \sum_{k=1}^{N} \bar{\lambda}_k$ and $\hat{\lambda}$ is the peak arrival rate of the aggregated traffic, i.e. $\hat{\lambda} \triangleq \sum_{k=1}^{N} \hat{\lambda}_k$.

The detailed proof of Proposition 2 is given in [3]. Since we assume that all video telephony traffic are homogeneous, from Proposition 2, we know the required bandwidth for N multiple video traffic is $N \cdot BW_{video}$. Since each video telephony service traffic consists of a voice and a video traffic, for N multiple video telephony service traffic, the total required bandwidth $BW^{(N)}$ is given by

$$BW^{(N)} = N \cdot \left(BW_{voice} + BW_{video}\right).$$

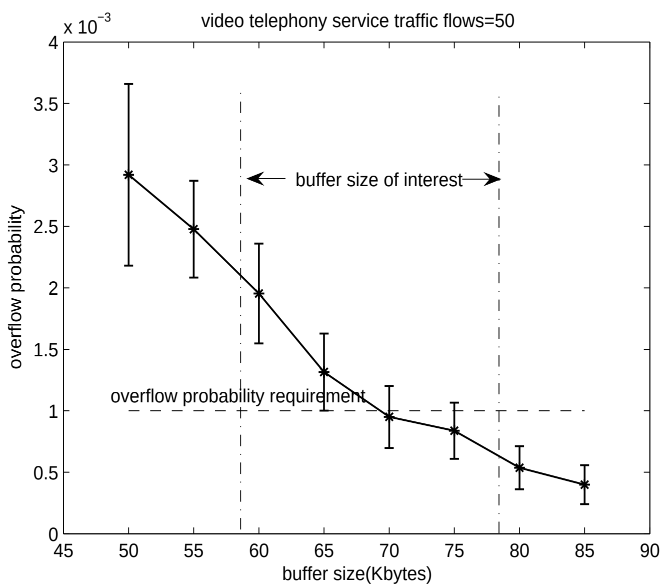


Figure 5. Video telephony service traffic flows=50

4. **Numerical results.** We perform simulation studies by using the NS-2 simulator to check the validity of our results. We simulate with the estimated bandwidth based on our analysis, and then investigate whether the estimated bandwidth is proper

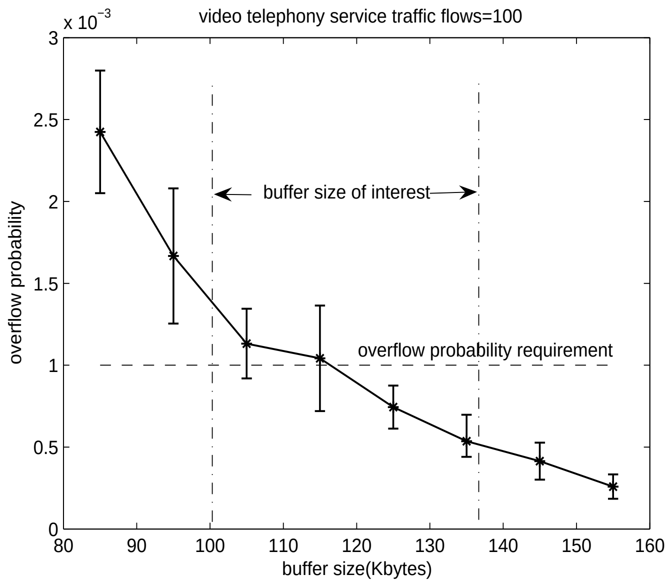


Figure 6. Video telephony service traffic flows=100

or not. In simulations, we use the measured data for the packet inter-arrival times to generate packets, and the packet overflow probability requirement is assumed to be 10^{-3}. We perform simulations with multiple traffic rather than with a single traffic to verify the effectiveness of our bandwidth allocation algorithm since there are multiple video telephony service traffic in practice. In addition, we see that in practical network design, the maximum queueing delay of 4 ms to 8 ms per hop could be acceptable to guarantee the end-to-end delay requirement of the real-time video applications. So, we consider a buffer size which corresponds to the maximum queueing delay from 4 ms to 8 ms and investigate the performance results for the buffer size of interest. The estimated values of the required bandwidth (which are used in simulations) when video telephony service traffic flows are 50 and 100, are given in TABLE 2 and 3, respectively.

The simulation results for video telephony service traffic are shown in FIGURE 5 and FIGURE 6. In the figures, we show the experimental overflow probabilities and 95 percent confidence intervals of empirical values. As shown in FIGURE 5, we find that, when we use our estimated bandwidth $BW^{(N)}$, the experimental overflow probabilities are matched with the required overflow probability for the buffer size greater than 65 Kbytes. Similarly, as shown in FIGURE 6, the experimental overflow probabilities are matched with the required overflow probability for the buffer size greater than 115 Kbytes. Although there is a little discrepancy between the simulation results and the required overflow probability, we think that this can be acceptable from the engineering point of view. In addition, from FIGURE 5 and FIGURE 6 we observe that our method works more effectively as the number of video telephony service traffic is increasing.

5. **Conclusions.** In this paper, we analyze the stochastic characteristics of the measured video telephony service traffic which consists of a voice traffic based

TABLE 2. The estimated bandwidth $BW^{(N)}$ when the number of video telephony service traffic flows is 50, i.e., $N = 50$

buffer size (Kbytes)	60	65	70	75
bandwidth (Mbps)	86.1378	85.802	85.5169	85.2179

TABLE 3. The estimated bandwidth $BW^{(N)}$ when the number of video telephony service traffic flows is 100, i.e., $N = 100$

buffer size (Kbytes)	105	115	125	135
bandwidth (Mbps)	148.6211	148.21	147.86	147.5779

on ITU-T G.711 and a video traffic based on ITU-T H.263, and then develop a measurement-based mathematical model for the actual video telephony service traffic. Using our mathematical model, we estimate the required bandwidth of the video telephony service traffic for a given overflow probability requirement. According to our analysis, we can easily compute the required bandwidth for the video telephony service traffic regardless of the number of aggregated traffic. Our simulation studies with the NS-2 simulator show that our mathematical model is appropriate to analyze the video telephony service traffic and to estimate the required bandwidth for a given overflow probability requirement.

Acknowledgments. We would like to thank the referees very much for their valuable comments and suggestions.

REFERENCES

[1] M. Chhawchharia and A. Guchhait, *High capacity VoIP services in 802.11 networks*, in "Proc. the 2^{nd} International Symposium on Wireless Pervasive Computing," (2007), 321–326.
[2] Le–Sheng Chou and Cheng-Shang Chang, *Experiments of the theory of effective bandwidth for Markov sources and video traces*, in "Proc. Annual Joint Conference of the IEEE Computer and Communications Societies Infocom' 96," (1996), 497–504.
[3] A. I. Elwalid and D. Mitra, *Effective bandwidth of general markovian traffic sources and admission control of high speed networks*, IEEE/ACM Trans. on Networking, **1** (1993), 329–343.
[4] P. Hatalsky, H. Smith and M. Carlton, *Call admission control for video phone applications*, in "Proc. the 11^{th} IEEE International Conference on Networks," (2003), 307–312.
[5] W. Kai-Hong Ho, Wai-Kong Cheuk and D. Pak-Kong Lun, *Content-based scalable H.263 video coding for road traffic monitoring*, IEEE Trans. on Multimedia, **7** (2005), 615–623.
[6] P. Koutsakis and M. Paterakis, *Call-admission-control and traffic-policing mechanisms for the trasmission of videoconferece traffic from MPEG-4 and H.263 video coders in wireless ATM networks*, IEEE Trans. on Veh. Technol., **53** (2004), 1525–1530.
[7] N. Bléfari-Melazzi, G. Russo and P. Talone, *Traffic modelling of video sources codified at very low bit rate with the H.263 test model*, in "Proc. IEEE International Conference on Communications," (1997), 528–533.
[8] D. Mitra, *Stochastic theory of a fluid model of producers and consumers coupled by a buffer*, Adv. Appl. Prob., **20** (1988), 646–676.
[9] H. Saito, M. Kawarasaki and H. Yamada, *An analysis of statistical multiplexing in an ATM transport network*, IEEE Journal on Selected Areas in Communications, **9** (1991), 359–367.
[10] H. Sakate, H. Yamaguchi and K. Yasumoto, *Resource management for quality of service guarantees in multi-party multimedia application*, in "Proc. the 6th IEEE International Conference on Network Protocols," (1998), 189–196.
[11] ITU-T Recommendation G.711, "Pulse Code Modulation (PCM) of Voice Frequencies".

[12] ITU-T Recommendation H.263, “Video Coding for Low Bit Rate Communication”.

Received August 2008; 1st revision November 2008; final revision April 2009.

E-mail address: KimBongJoo@kaist.ac.kr
E-mail address: guhwang@kaist.edu
E-mail address: yhj@kt.com