Performance Evaluation of Congestion Control on Interworking Conventional LANs with B-ISDN

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ABSTRACT

In this paper, we propose congestion control methods following a packet conservation principle which can prevent congestion in the interworking unit(IWU). Those are rooted in the idea of achieving internetworking stability by the rate control mechanism in a transport connection of HSTP, and the congestion status information in IWUs. The suggested algorithm and its effect on congestion in the LAN/B-ISDN IWU are described in detail by first order fluid approximations of stochastic processes.

The performance evaluations are performed with the modeling and simulation. The methodology based on the Discrete Event System Specification (DEVS) formalism is used to develop modular and hierarchical specifications of discrete event models. The specified models are implemented in DEVSIM++, which provides a simulation environment with the DEVS formalism. The results of simulation show both steady-state and transient behavior of the suggested congestion control method.

INTRODUCTION

B-ISDN is deploying in several countries to be used as a test-bed for developing Broadband Telecommunication Services, and also as experimental backbone networks in national environments. There are many high speed networks based on optical fibers, which are widely used in field as local area networks(LANs) and metropolitan area networks(MANs). The deployment plans of ATM networks in many countries show that B-ISDN will be national and even international backbone networks with Gbit/s speed by the end of this century.

Although B-ISDN delivers new high speed LANs which is suitable for interworking with ATM network, it is impossible to replace all the networks currently in use (Sutherland and Burgin 1993). The interworking conventional LANs with B-ISDN is required to provide data communication services to the users. With B-ISDN, a virtual private network service can be provided by use of remote bridges between LANs on different sites that have identical MAC layer protocols. To be more general, routers must be used to allow communications between LANs using different MAC layer protocols as long as identical internetwork protocols are used.

A major problem in the LAN to B-ISDN interworking is the speed difference between two networks connected together (Clark et al. 1989; Jain et al. 1990). Many previous studies have shown that the implementation of the window-based transport protocol can result in wrong behavior in response to network congestion. Jacobson's study for windowbased congestion control shows some examples of wrong behavior, and suggests good algorithms that can be used to make right things happen in the TCP/IP network. Inherently, the window based congestion control method shows the difficulties to detect a congestion occurrence especially to estimate a mean round trip time (Jacobson and Karels 1988). Also, the rate based bandwidth allocation scheme in ATM switch makes the rate-based congestion control for interworking LANs/B-ISDN more attractive.

CONGESTION CONTROL

There are several issues to be solved to interconnect B-ISDN and conventional LANs by the IWU. These are protocol conversion between different data link layer protocols, addressing conversion, state transition matching, transmission speed difference resolution, buffer management, routing, routing table management, and congestion control.

All those issues are important in building a high speed IWU. In this paper, congestion control is addressed as an essential issue for the stable operation of the store-and-forward internetworking. Congestion control is a strategy to prevent or reduce the effect of overload in a network. Several papers have been published to show the strategies of congestion control (Jain 1986). The preallocation scheme is very effective for the continuous rate of data traffic, but it is impossible to allocate a suitable size of resources without loss in case of the variable rate. The scheme of discarding overloaded packets in the routing node is easy to implement, but it does not provide any prevention of congestion. The scheme of restriction of the total number of packets is not easy to implement and does not prevent the congestion in a certain node. Backpressure is a form of hop-by-hop, on/off flow control. It works well for the short-lived overload, but it propagates congestion to adjacent nodes in case of the long-lived overload (Jain 1990). The slow start algorithm provides the prevention mechanism of congestion collapses. Its disadvantages are that it requires to detect lost packets, and it is very hard to estimate the round trip time delay.

HSTP suggests a rate control method to prevent a buffer overflow in the receiving node. Since this rate control method can be applied to prevent buffer overflow in a congested interworking node, we investigate the use of rate control to solve congestion problems in the interworking nodes.

The congestion control scheme being proposed and studied here is base on rate control, and applicable to the connectionless network. The proposed scheme is very simple in nature. It follows a *packet conservation* principle, which can prevent congestion in the IWUs. Those are rooted in the idea of achieving internetworking stability by using the rate control mechanism in a transport connection of HSTP, and using the congestion status information in the interworking nodes. The suggested algorithm has three sets of policies for controlling the traffic placed on the network. These are placed at the IWU of the network, the source node of the network, and the destination node of the network.

a) Source Node (SN) Policy : The transmission rate in a SN is initially increased linearly with time constant α until it receives a congestion experience message from the destination node. Then the transmission rate in the SN is decreased exponentially with time constant β. When the SN receives a message which did not experience congestion, the SN starts to increase the rate again with rate α.

b) Interworking Unit (IWU) Policy : When a packet arrives in an IWU, the IWU checks the buffer status of its own. When a packet leaves a SN, a congestion experience bit is originally 0. If the buffer size in use is greater than some level Q_0, the congestion experience bit is set to 1. Otherwise, the IWU does not change the congestion experience bit.

c) Destination Node (DN) Policy : When a DN receives a packet, it checks the congestion experience bit in the packet. If it is set to 1, the DN recognizes that one of the IWU in the path is congested, and it requests to reduce the transmission rate of the SN. If it is set to 0, it means that all IWUs in the path have enough buffer space to handle the current transmission rate. This information is sent back to the SN.

NUMERICAL ANALYSIS OF PROPOSED SCHEME

The proposed scheme dynamically regulates the transmission rate of the SN to prevent congestion in the intermediate IWUs. Closed-form solutions are obtained for the dynamic showing the both transient and steady state behavior. Source rate is defined as real-valued continuous variables. This allows the adjustments of transmission rates of SNs. The adjustment is requested by the interworking node in congestion state. It is presented by the differential equations of transmission rates. Also, coupled differential equations including transmission rates of SNs and the queue size of a congested IWU can describe the time dependent behavior of the network.

Single Input Case

We define parameters; $Q(t)$ is the actual height of buffer level at the congested IWU at time t, and $\lambda(t)$ is the data transmission rate at the SN at time t. If we define the transfer delay from the SN to congested IWU as τ_f, and the feedback transfer delay from the congested IWU to the SN as τ_r, the delayed signals $Q^*(t)$ and $\lambda^*(t)$ can be defined as

$Q^*(t) = Q(t-\tau_r)$ and $\lambda^*(t) = \lambda(t-\tau_f)$.

We define dynamics of the system as

$$\frac{d}{dt}Q(t) = \hat{Q}(t) = \lambda(t-\tau_f) - \mu = \lambda^*(t) - \mu \quad (1)$$

$$\frac{d}{dt}\lambda(t) = \hat{\lambda}(t) = \begin{cases} \alpha, & \text{if } Q^*(t) = Q(t-\tau_r) \le Q_0 \\ -\lambda(t)/\beta, & \text{if } Q^*(t) = Q(t-\tau_r) > Q_0 \end{cases} \quad (2)$$

Suppose that $Q(t) = 0$ at $t = 0$. Then due to time delay τ_r, $\lambda(t)$ increases at constant rate α from $t = \tau_r$, i.e.,

$$\hat{\lambda}(t) = \begin{cases} \alpha, & \text{for } t \ge \tau_r \\ 0, & \text{for } t < \tau_r \end{cases} . \quad (3)$$

Integrating this with initial condition $\lambda(\tau_r) = 0$ yields

$$\lambda(t) = \begin{cases} \alpha(t-\tau_r), & \text{for } t \ge \tau_r \\ 0, & \text{for } t < \tau_r \end{cases} . \quad (4)$$

To find the buffer level $Q(t)$ for $t \ge \tau_r + \tau_f$ with initial condition $Q(\tau_r + \tau_f) = 0$, integrate $\hat{Q}(t)$ to get

$$Q(t) = \left(\frac{\alpha(t^2-\tau^2)}{2} - (\alpha\tau + \mu)(t-\tau)\right)U(t-\tau), \quad (5)$$

where $U(t)$ denotes unit step function and τ denotes total time delay in the loop, which is denoted by $\tau = \tau_f + \tau_r$. When $Q^*(t) = Q_0$, $\lambda(t)$ begins to decrease exponentially. To find the time t_1 when $\lambda(t)$ begins decrease, solve

$$Q_0 = \left(\frac{\alpha(t_1-\tau_r-\tau)^2}{2} - \mu(t_1-\tau_r-\tau)\right)U(t-\tau_r-\tau) \quad (6)$$

to get

$$t_1 = \tau + \tau_r + \frac{\mu + \sqrt{\mu^2 + 2\alpha Q_0}}{\alpha}. \quad (7)$$

The actual buffer level $Q(t)$ reaches Q_0 at $t_0 = t_1 - \tau_r$, i.e., τ_r earlier than $Q^*(t)$. Integrating, for $t > t_1$, $\hat{\lambda}(t) = -\frac{\lambda(t)}{\beta}$ yields

$$\lambda(t) = c_1 e^{-t/\beta} U(t-t_1), \quad (8)$$

where the integration constant c_1 can be determined, by using Eqs. (4) and (8), from continuity condition of $\lambda(t)$ signal at $t = t_1$, i.e., $\lambda(t_1) = \alpha(t_1 - \tau_r) = c_1 e^{-t_1/\beta}$. From this we get $c_1 = \alpha(t_1 - \tau_r)e^{-t_1/\beta}$ and, therefore,

$$\lambda(t) = (\alpha\tau + \mu + \sqrt{\mu^2 + 2\alpha Q_0})e^{-(t-t_1)/\beta}U(t - t_1). \tag{9}$$

Now integration of $\hat{Q}(t) = \lambda^*(t) - \mu$, leads to

$$Q(t) = -\alpha\beta t_0 e^{-(t-t_1-\tau_f)/\beta} - \mu t + c_2. \tag{10}$$

Using Eqs. (5) and (10) and applying continuity condition of $Q(t)$ signal at $t = t_1 + \tau_f$, we find c2 and, therefore,

$$Q(t) = \alpha\beta t_0(1 - e^{-(t-t_1-\tau_f)/\beta}) - \mu(t - \tau) + \frac{\alpha(t_1+\tau_f-\tau)^2}{2}. \tag{11}$$

Consider $Q(t)$ for a moment. As can be seen from the expression for $Q(t)$, there is no guarantee that $Q(t)$ begins to decrease at $t = t_1 + \tau_f$. It is obvious that, as $t \to \infty$, $Q(t)$ decreases monotonically due to $-\mu t$ term, which yields constant negative gradient. To find the maximum value of $Q(t)$, solve following equation for t_M

$$\frac{dQ(t)}{dt} = \alpha t_0 e^{-(t_M-t_1-\tau_f)/\beta}) - \mu = 0 \tag{12}$$

to get

$$t_M = t_1 + \tau_f - \ln\frac{\mu}{\alpha t_0}. \tag{13}$$

Note that, in view of Eq. (7), the argument of ln function is always less than 1 and, therefore, the last term in Eq. (13) takes a positive values. Substituting t_M into Eq. (11), we get

$$Q(t) = \alpha\beta t_0(1 - (\frac{\mu}{\alpha t_0})^{1/\beta}) - \mu(t_1 - \tau_r - \ln\frac{\mu}{\alpha t_0}) + \frac{\alpha(t_1+\tau_f-\tau)^2}{2} \tag{14}$$

When $Q^*(t) = Q_0$, $\lambda(t)$ starts to increase linearly again. Let this moment be denoted by t_2. Then t_2 is the solution of

$$Q_0 = \alpha\beta t_0(1 - e^{-(t_2-t_1-\tau)/\beta}) - \mu(t_2 - \tau - \tau_r) + \frac{\alpha(t_1+\tau_f-\tau)^2}{2}. \tag{15}$$

<u>Multiple Input Case</u>

Suppose there are n inputs in the buffer, time delays in each input flow control device, and also time delays in transfer of measured information of actual buffer level. We define parameter $Q(t)$ as the actual height of buffer level at time t, $Q_i^*(t)$ is the delayed information of $Q(t)$ transferred to i'th input $\lambda_i(t)$, $\lambda_i(t)$ is the i'th data transmission rate at the source at time t, and $\lambda_i^*(t)$ is the delayed input flow into the buffer of the interworking node. Then, the delayed signals $Q_i^*(t)$ and $\lambda_i^*(t)$ for $i=1,\ldots n$ can be defined as

$$Q_i^*(t) = Q(t - \tau_r) \text{ and } \lambda_i^*(t) = \lambda_i(t - \tau_f). \tag{16}$$

We define dynamics of the system by using unit step function $U_i(t)$ as follows.

$$\hat{Q}(t) = \sum_{i=1}^{n} \lambda_i(t - \tau_{fi})U_i(t - \tau_{fi}) - \mu \tag{17}$$

$$\hat{\lambda}_i(t) = \begin{cases} \alpha_i, & \text{if } Q_i^*(t) = Q(t - \tau_{ri}) \le Q_0 \\ -\lambda_i(t)/\beta_i, & \text{if } Q_i^*(t) = Q(t - \tau_{ri}) > Q_0 \end{cases} \tag{18}$$

Suppose that $Q(t) = 0$ at $t = 0$. Then, due to time delay τ_{ri}, $\lambda_i(t)$ begins to increase at constant rate α_i from $t = \tau_{ri}$, i.e.,

$$\hat{\lambda}_i(t) = \begin{cases} \alpha_i, & \text{for } t \ge \tau_{ri} \\ 0, & \text{for } t < \tau_{ri} \end{cases}. \tag{19}$$

Integration yields

$$\lambda_i(t) = \begin{cases} \alpha_i(t - \tau_{ri}), & \text{for } t \ge \tau_{ri} \\ 0, & \text{for } t < \tau_{ri} \end{cases}. \tag{20}$$

To find the buffer level $Q(t)$, integrate Eq. (17) for $t \ge \tau_m$ with initial condition $Q(\tau_m) = 0$, where τ_m denotes the minimum time delay among the total loop delays and can be defined as $\tau_m = \min\{\tau_1, \tau_2, \ldots, \tau_n\}$, and τ_i denotes total time delay in i^{th} loop, i.e., $\tau_i = \tau_{fi} + \tau_{ri}$. Then, we get

$$Q(t) = \sum_{i=1}^{n}(\frac{\alpha_i(t-\tau_i)^2}{2} - \alpha_i\tau_i(t - \tau_i))U_i(t - \tau_i) - \mu(t - \tau_m)U(t - \tau_m). \tag{21}$$

To find the time when $\lambda_i(t)$ begins to decrease exponentially, we first calculate the time t_0 when $Q(t)$ equals its reference value Q_0. Since Eq. (21) contains unit step function initiated from different time, this equation can be solved under the assumption that all signals $\lambda_i(t)$ contribute to the buffer level, which is reasonable because otherwise some signals $\lambda_i(t)$ for some is never come to the buffer. From physical point of view, the validity of this assumption depends on system parameters, i.e., total time delay τ_i of each signal loop, increasing rate α_i, and reference level of buffer Q_0. If there is a loop for which α_i is very large with comparatively small loop delay τ_i, then the corresponding signal $\lambda_i(t)$ would fill the buffer up to Q_0 even before some λ_i with large loop delays begin to be activated. If τ_i and α_i are sufficiently small or similar in magnitude for all i and Q_0 is sufficiently large, then above assumption holds. If above assumption does not hold, then with actual parameters it must be checked numerically by iteration that which signals actually contribute to the increase of buffer level $Q(t)$ up to Q_0. Assuming our assumption is valid, all delayed unit step functions may be dropped to yield an equation which may readily be solved for t_0, i.e.,

$$Q_0 = \sum_{i=1}^{n}(\frac{\alpha_i(t_0-\tau_i)^2}{2} - \alpha_i\tau_i(t_0-\tau_i)) - \mu(t_0-\tau_m). \qquad (22)$$

From this equation t_0 can be found as

$$t_0 = \frac{2\sum_{i=1}^{n}\alpha_i\tau_i+\mu+\sqrt{(2\sum_{i=1}^{n}\alpha_i\tau_i+\mu)^2+2\alpha_t(Q_0+\mu\tau_m-3\sum_{i=1}^{n}\alpha_i\tau_i/2)}}{\alpha_t} \qquad (23)$$

where $\alpha_t = \sum_{i=1}^{n}\alpha_i$. Note that the validity of our assumption is ensured if $t_0 \geq \tau_M$, where $\tau_M = \max\{\tau_1, \tau_2, \ldots, \tau_n\}$.
The time t_{1_i} when λ_i begins to decrease exponentially can be found as $t_{1_i} = t_0 + \tau_{r_i}$ for $i = 1, \ldots, n$.

Integrating, for $t > t_{1_i}$, $\hat{\lambda}_i(t) = -\lambda_i(t)/\beta_i$, yields $\lambda_i(t) = c_{1_i}e^{-t/\beta_i}U(t-t_{1_i})$. Where the integration constant c_{1_i} can be determined from continuity condition of $\lambda_i(t)$ signal, i.e.,

$$\lambda_i(t) = \alpha_i(t_{1_i} - \tau_{r_i}) = c_{1_i}e^{-t_{1_i}/\beta_i}. \qquad (24)$$

From this we get $c_{1_i} = \alpha_i(t_{1_i} - \tau_{r_i})e^{t_{1_i}/\beta_i}$ and, therefore,

$$\lambda_i(t) = \alpha_i(t_{1_i} - \tau_{r_i})e^{-(t-t_{1_i})/\beta_i}U_i(t-t_{1_i}) \qquad (25)$$

$$= \alpha_i t_0 e^{-(t-t_{1_i})/\beta_i}U_i(t-t_{1_i}) \qquad (26)$$

The first λ_i^* signal reaches the buffer at $t = t_*$ where $t_* = \min\{t_{1_1} - \tau_{f_1}, t_{1_2} - \tau_{f_2}, \ldots, t_{1_n} - \tau_{f_n}\}$.

Now integrating for $t \geq t_*$

$$\hat{Q}(t) = \sum_{i=1}^{n}\lambda_i^*(t)U_i(t-t_{1_i}-\tau_{f_i}) - \mu, \qquad (27)$$

leads to

$$Q(t) = Q(t_*) + \sum_{i=1}^{n}\alpha_i\beta_i t_0(1-e^{-(t-t_{1_i}-\tau_{f_i})/\beta_i})U_i(t-t_{1_i}-\tau_{f_i}) - \mu(t-t_*) \qquad (28)$$

where $Q(t_*)$ is determined from Eq.(33) as

$$Q(t_*) = \sum_{i=1}^{n}(\frac{\alpha_i(t_*-\tau_i)^2}{2} - \alpha_i\tau_i(t_*-\tau_i)) - \mu(t_*-\tau_m). \qquad (29)$$

Eq. (28) shows that Q(t) increase further from $Q(t_*)$ for a while until exponential terms are saturated and then it decrease due to persistent effect of $-\mu t$. To determine the maximum value of Q(t), note first in Eq. (28) that Q(t) is defined as summation of $-\mu t$ and exponential functions activated in different time, and also that all terms in summation increase monotonically. In other words, the instant when Q(t) takes its maximum value is heavily influenced by relative sizes between the rate of exponential increase (β_i) of each signal and the rate of decrease of buffer level (μ). In addition, it also depends on when a particular exponential function starts. In summary, the instant t_M, when Q(t) attains its maximum, depends on the various system parameters and must be determined by numerical method.

Being aside from exact calculation of Q_{max}, we turn our attention to estimation of an upper bound of Q(t), denoted by Q_{sub}, is given by $Q_{sub} = Q(t_*) + t_0\sum_{i=1}^{n}\alpha_i\beta_i$.
The value of Q_{sub} may be used, with some safety factor, as a design parameter of buffer size.

In what follow, we will try to find approximate estimation of t_M as follows: Assume the maximum of Q(t) occurs after all signals $\lambda_i^*(t)$ for $i = 1, \ldots, n$ reach the buffer or, equivalently, assume that $t_M \geq \max\{t_{1_1} - \tau_{f_1}, t_{1_2} - \tau_{f_2}, \ldots, t_{1_n} - \tau_{f_n}\}$.

From differentiation of Eq. (28) by dropping unit step functions, the following equation results.

$$\sum_{i=1}^{n}\alpha_i t_0 e^{-(t-t_{1_i}-\tau_{f_i})/\beta_i}U_i(t-t_{1_i}-\tau_{f_i}) - \mu = 0. \qquad (30)$$

Although the closed form solution t_M is still not possible, an approximate solution t_M is available in closed form. Define $t_{**} = \max\{t_{1_1} - \tau_{f_1}, t_{1_2} - \tau_{f_2}, \ldots, t_{1_n} - \tau_{f_n}\}$, and $\beta_{min} = \min\{\beta_1, \beta_2, \ldots, \beta_n\}$.
Since all the exponential terms decrease to zero eventually, we may solve $\sum_{i=1}^{n}\alpha_i t_0 e^{-(t_M - t_{**})/\beta_{min}} - \mu = 0$ to get $t_M = t_{**} - \ln\frac{\mu}{\alpha_t t_0}$. The exact solution t_M may be quite different from t_M because we estimated the exponential decrease overly. Substituting t_M into Eq. (28) will yield an approximate value of upper bound of Q(t).

PERFORMANCE EVALUATION

Discrete event simulation has been used to investigate the behavior of systems under design. Because of the abstractness of numerical analysis, we tried to develop an analytical model in the previous chapter and derived some approximated results. Due to the increasing demands on computing complexity with analytic model, it is failed to derive all aspects of the dynamical behavior of congestion control methods under design. Also, we assumed source rate and queue size as real-valued continuous variables to derive the closed-form solutions. Since it may create some errors in the derived results, we develop simulation models to provide more reliable and accurate results than analytic ones. Also, it is aimed to see the omitted other aspects of the congestion control methods under design. Since the temporal issues of systems are significant, discrete event modeling and simulation are considered as the best solution for develop models.

We develop simulation models for the performance evaluation of a congestion control method in the DEVSIM++ modeling and simulation environment. DEVSIM++ realizes the DEVS formalism (Zeigler 1984) for modeling. It is associated abstract simulation concepts for simulation. DEVSIM++ provides modeler facilities for the specification of atomic models and coupled models within DEVS framework. Detailed descriptions for the DEVSIM++ modeling and simulation environment can be found in (KIM 1991).

The DEVS formalism specifies discrete event models in a hierarchical and modular form in the set-theoretic manner. Within the formalism, one must specify the basic models, from which larger ones are built. Then, one specifies how these models are connected together in hierarchical fashion. A basic model, called an atomic model, has specifications for the dynamics of the mode with seven tuples; input event set, sequential state set, output event set, internal transition function, external transition function, output function, and time advance function. The second form of the model, called a coupled model, tells how to couple several component models together to form a new model. This latter model can be employed as a component in a larger coupled model, thus giving rise to the construction of complex models in a hierarchical fashion.

A congestion control model is described in a modular and hierarchical manner using the DEVS methodology as shown in Fig. 1. Four atomic DEVS models are developed as the components of the simulated system. Then two DEVS coupled models, an experimental frame module (EF) and a congestion control model module (MODEL), are developed to couple these four atomic models. The coupling of the EF module and the MODEL module results in a coupled model 'CONGSIM' which constructs whole simulation environment

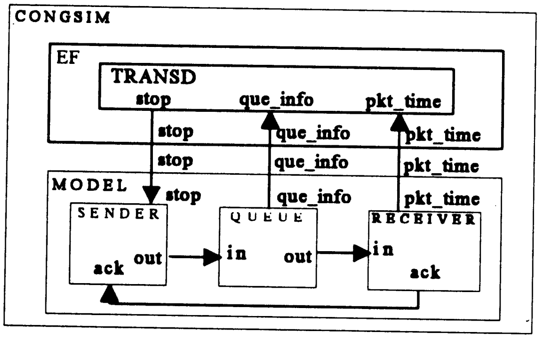

Fig. 1. DEVS model of Congestion Control System

Due to the increasing demands on computing capability for the modeling of complex systems, we build an abstract model of congestion control system following DEVS modeling formalism and simulation methodology. The atomic model 'SENDER' is a sending node in a network which follows packet generation policies as defined in chapter 2. The rates of packet generation varies based on alpha, beta, and the congestion control information in the arrived packet from the 'RECEIVER'. The atomic model 'QUEUE' is an interworking unit in the network. It places arriving packets in a FIFO queue and sends the packets from the queue to 'RECEIVER'. If the queue level reaches certain point, it sets the congestion experience flag in the sending packet. When the atomic model 'RECEIVER' receives a packet, it checks the congestion experience flag, and sends this information to 'SENDER' for the rate control. The atomic model 'TRANSD' gathers simulation results and sends 'stop' signal to 'SENDER' to end the simulation. More detailed descriptions of models are omitted due to the limited space on this paper.

Fig. 2. shows the rate variation in a source node which shows the transient state results. It includes single source node case, two source node case, and four source node case. Each 'SENDER' independently sends packets to different 'RECEIVER' in this model. The results displays that the sending rate increases linearly first. Then, it decreases exponentially when congestion is occurred.

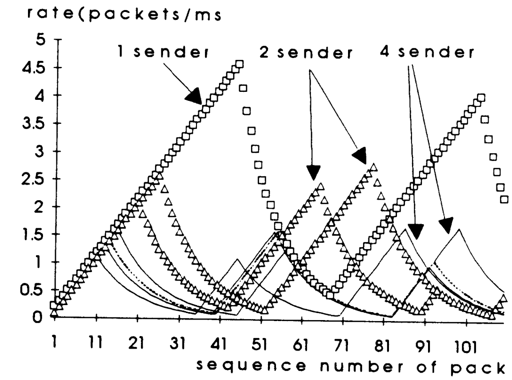

Fig. 2. The Rate Variation of Source Nodes

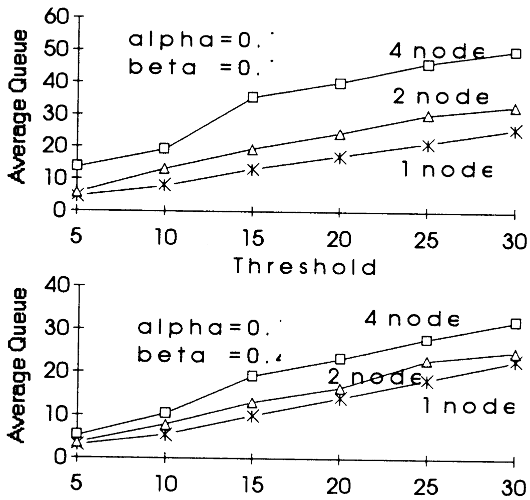

Fig. 3. Average Queue size in the IWU

Fig. 3. shows the average queue size in a interworking unit which is modeled as a 'QUEUE'. It describes the average queue size in the single source node

case, two source node case, and four source node case. The x-axis plots a set of thresholds which is used to set the congestion experience flag of leaving packets in the IWU. Two cases, when beta is 0.1 and 0.4, are plotted to compare the effect of the beta value. Since the sending rate in the source node decreases exponentially with parameter beta, the average queue size decreases when the beta value increases.

Fig. 4. shows the average transmission rate in a source node which is modeled as a 'SENDER'. It describes the average value of the packet's sending rate which varies based on the alpha value and the number of source nodes. The x-axis plots a set of alpha values and the y-axis plots average rates in packets per milli seconds. Since the sending rate in the source node increases with parameter alpha, the average rate increases when the parameter increases.

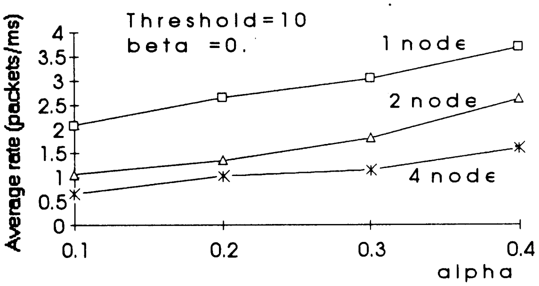

Fig. 4. Average Transmission Rate in a Source Node

CONCLUSION

A rate control method has been considered as a congestion control scheme in the interworking node. We investigate the use of rate control by assuming interworking nodes detect congestion, keep the status information, and feedback the status information to the sending node for the rate control. With this assumption, we propose congestion control method which can prevent congestion in the interworking nodes. Sets of policies for controlling the traffic are defined for the interworking nodes of the network, the source and the destination nodes of the network.

We build deterministic analytic models on rate-based congestion control to show the time dependent behavior in the congested network. Closed-form solutions are obtained for the dynamic showing the both transient and steady state behavior. Suggested analytic model is almost equivalent to previous study (Bolot and Shankar 1990). Only difference is that we use the level of queue in a congested interworking node to control the transmission rate in a source node. And, we expand the model to multiple sources and destination nodes. Our study result shows that the congestion control scheme dynamically regulates the transmission rate of the source node to prevent congestion in the intermediate interworking nodes. Since the coupled differential equations are solved by assuming source rates and queue size as real-valued continuous variables, the derived result may have a little variance from the operation result in the real environment. Thus, we developed discrete event simulation models and derived several performance results. The results shows the behavior of the congestion control system and the effect of parameters.

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