ATM Rate Based Traffic Control with Bode Principle

Yuanwei Jing, Hui Zeng, Qingshen Jing, and Ping Yuan

Abstract: Bode principle is applied to carry out traffic control for rate based ATM network, which guarantees the higher buffer utilization, buffer overflow-free, and well utilization of bandwidth. The principle confirms the relationship between the threshold of buffer queue and the network bandwidth, as well as the relationship between the threshold of buffer and source input rate. Theoretic warrant of the buffer threshold is proposed. The reference range of source input rate is provided in theory, which makes the source end respond to the change of network state rapidly and dynamically, and then the effect of time delay to the traffic control is avoided. Simulation results show that the better steady and dynamic performances of networks are obtained by Bode principle.

Keywords: ABR, ATM network, bode principle, traffic control.

1. INTRODUCTION

No matter from the point of networks or the users, a general network is necessary to be set up to transmit both low speed signals and high speed signals, and to be suitable for delay character of voice signals, bit error character of data signals, and bit error and delay character of image signals. Such network is the broadband integrated services digital networks (B-ISDN). In 1988, the asynchronous transfer mode (ATM) was confirmed as B-ISDN information transmission mode.

ATM is an oriented connection technology of high speed exchange and multiplexing, which provides five classes of services: constant bit rate (CBR), variable bit rate (VBR), available bit rate (ABR), unspecified bit rate (UBR) and guaranteed frame rate (GFR). GFR is a new ATM service designed specially to sustain IP trunk sub-network. CBR and VBR are the protected services.ABR uses the remained bandwidth of VBR and CBR. ABR application appoints a peak cell rate (PCR) and minimal cell rate (MCR). The network distributes resource properly to guarantee that all ABR applications can receive MCR capacity. Unused capacity is shared in all ABR source with a fair mode, and ABR guarantees the fair capacity distribution. ABR becomes the most suitable service of traffic control for its fewer cells loss rate and more fair resource share [1-4].

Traffic control is divided into two classes: rate based traffic control and credit based traffic control. ATM traffic management criterion adopts traffic control based on rate. The traffic control is to restrict sending data rate at the level no more than processing rate of receiver, which needs the buffer for data.

If the sending data rate exceeds the applied bandwidth, so-called "leaky bucket" mechanism is adopted for buffer processing, i.e., if user messages exceeds the storage capacity of the leak bucket, parts of them will be lost or delayed. The main algorithms of "leak bucket" mechanism include the leaky bucket and the token bucket algorithms. The leak bucket algorithm regulates that the node sends a packet to network every fixed time, and guarantees that the packet rate flowing to communication sub-network is a constant. For input packets, if queue is not full, then the packets will be parked at the queue tail, otherwise it will be lost.

In practical application of token bucket algorithm, if burst traffic is coming, the corresponding output rate is increased to guarantee that there is no data loss. Only packets with token can be sent. If the rate of input packets exceeds the rate of token, and all tokens are consumed out, the algorithm informs the host computer to wait to input packets. And then the input traffic matches communication capacity. Token bucket algorithm makes output equipment have the ability of processing the burst traffic, and responds to sudden increase of input more quickly.

The basic idea of "leak bucket" is to force packets

Manuscript received December 5, 2005; revised August 19, 2007; accepted December 22, 2007. Recommended by Editor Jae Weon Choi. This work was supported by the National Natural Science Foundation of China under Grant 60274009, the National Program (863) of High Technology Development under Grant 2004AA412030 and the Liaoning Natural Science Foundation under Grant 20052027.

Yuanwei Jing, Hui Zeng, and Ping Yuan are with the College of Information Science and Engineering, Northeastern University, Shenyang 110004, China (e-mails: ywjjing@mail.neu.edu.cn, zh6135@163.com, yuanping@ise.neu.edu.cn).

Qingshen Jing is with the College of Engineering, Peking University, Beijing 100871, China (e-mail: jingqingshen@126.com).

to be sent with a predicted rate. The method is to regulate the average rate of data transmission.

At present, ATM network traffic control is a hotspot problem, and a lot of good results have made with methods of control theory. The researches of ABR traffic control start from binary feedback. The merit is simple to execute. However, they suffer serious problems of stability, exhibit oscillatory dynamics, and requirement of large amount of buffer to avoid cell loss [5]. After that, explicit rate algorithms are considered and investigated broadly. Most of the existing explicit rate schemes lack of two fundamental parts in the feedback control design: the analysis of the closed-loop network dynamics, and the interaction with VBR traffic [6]. The control design problem is formulated as a standard disturbance rejection problem where the available bandwidth acts as a disturbance for the network system [7,8]. Reference [9] designs a PD controller for ABR traffic control of ATM network. It is easy to execute with lower computational burden, and better system stability is obtained. However, only simple network model is discussed in [9], and it is localized for complex network models. Reference [10] designs a variable structure controller for ABR traffic control, which aims at binary traffic control. The sliding mode variable structure algorithm restrains the oscillations of ACR and queue, and smoothes the delay jitter. The reliable implementation mechanism for QoS in ATM networks is provided.

The basic idea of the methods mentioned above is the traffic control based on rate feedback, which belongs to closed-loop control method. However, in practical "leak bucket" mechanism, open-loop idea is applied to avoid congestion from start. Hence, in order to close to network applications, it still has great practical signification for open-loop control research.

In the paper, open-loop idea is combined with closed-loop idea to set up system model, and Bode algorithm is adopted to carry out the traffic control. Bode algorithm has broad application in engineering. So, Bode algorithm with open-loop idea is applied to ATM ABR research.

Three main results are obtained in this paper. First, network does not need the real time control, which lightens the network control burden, and satisfies the performance index. In the references, no concept of non-real time is provided. Second, the relationship of network bandwidth and buffer threshold is given. In some references, the buffer threshold is given directly, and no standard about it is explained and the relation with network bandwidth is not illustrated. Third, the relationship of buffer threshold and input rate breadth is supposed. In most references, input rate is regulated by comparing buffer queue length with threshold. However, source can not obtain the needed input rate of network in time for the existence of network delay, and then effective control can not be carried out.

ATM rate based traffic control with Bode principle is discussed in the paper. Theoretical results different from others are achieved. Simulation results show that Bode principle with dynamic buffer threshold can sustain the higher throughput, less cell loss rate and max-min justice of rate distribution in dynamic environment effectively.

2. MODEL DESCRIPTION

In ABR communications of ATM network, storage and transmission packet exchange service is adopted. Cells and packets are sent to networks from source, then stored in switch nodes. At each node, a FIFO buffer is designed for all output links. The buffer is shared in all VCs coming by the link. Finally, cells and packets are sent to the destination. In order to adequately utilize the assigned bandwidth without buffer overflow, the queue length of output link buffer is controlled in a limited range, by comparing queue length with buffer threshold to reflect congestion condition and decide source sending rate.

An oriented connection grouping exchange network with simple structure [11] is described in Fig. 1.

In Fig. 1, S, D, SN denotes the source, the destination and the switch nodes, respectively, VC_i denotes the virtual circuit, and L_j denotes the links.

The network traffic model is described as [8]

$$\frac{\mathrm{d}y_j(t)}{\mathrm{d}t}=\sum_{i=1}^{m_j}u_{ij}(t-\tau_{ij})-f_j(t), \tag{1}$$

where y_j is the buffer queue length of link L_j, u_{ij} is source rate through the buffer of link L_j, τ_{ij} is the time delay of source rate to the buffer of link L_j, m_j is the number of VC through the buffer of link L_j, f_j is transmission bandwidth of VC_i, and

$$f_j(t)=\begin{cases} w_j(t), & \text{if } y_j(t)>0,\\ \min[\sum_{i=1}^{m_j}u_{ij}(t-\tau_{ij}),\ w_j(t)], & \text{if } y_j(t)=0, \end{cases}$$

where $w_j(t)$ is transmission bandwidth of links.

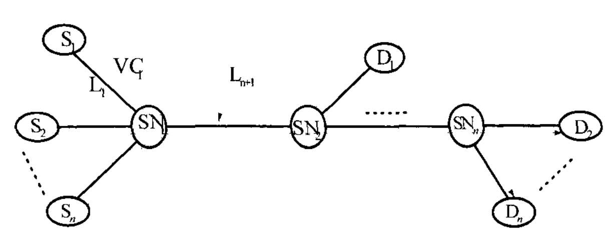


Fig. 1. Oriented connection exchange network.

3. SYSTEM DESIGN

In ATM networks, time delay exists necessarily, and it brings some disadvantage influence to networks. So, time delay must be considered in ATM network control. In general, time delays are divided into four classes: (1) Transmission delay - the required time of transmitting all bits of data packet; (2) Propagation delay - the required time of 1 bit from source to destination; (3) Processing delay - the required time of processing the data packet, before source or switch nodes send data packet and the data packet is sent to application; (4) Queuing delay - the time of waiting in any queue of routing process.

Cell transmission delay is much less than propagation delay, and then the feedback for congestion control is slow. The needed transmission capacity of the application sustained by ATM network changes in a great range, from thousands bits per second to hundreds of millions bits per second, and the communication capacity changes very quickly. So, simple congestion control methods are difficult to be applied successfully [12].

The better control effectiveness can be achieved by combining the open-loop and the closed-loop control methods. The basic idea is to force the packet to be sent with a predicted rate. The method is to adjust the average transmission rate, which belongs to open-loop method. While the closed-loop method is a traffic control based on rate feedback, by comparing buffer queue length with threshold to adjust source input rate. The system model is shown in Fig. 2.

In Fig. 2, $G_c(s)$ is the series leading correction item which compensates the system delay. The traffic is viewed as a deterministic and variable fluid flow, and the network queuing process is continuous in time. The transfer function of original system is

$$\frac{y(s)}{u(s)}=\frac{e^{-\tau s}}{s}, \tag{2}$$

where τ is the round-trip time delay of network. Because

$$e^{-\tau\omega j}=\cos\tau\omega-j\sin\tau\omega, \tag{3}$$

we have

$$\left.\frac{y(s)}{u(s)}\right|_{s=j\omega}=\frac{\cos\tau\omega-j\sin\tau\omega}{j\omega}. \tag{4}$$

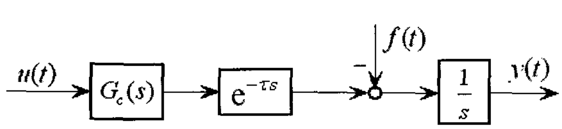


Fig. 2. ABR traffic control system.

Frequency analysis method is a typical method of applying frequency character to study control systems, which can describe the system frequency characters directly. The analysis method is simple and the physical conception is clear. To improve the system stability and instantaneous performance, and to restrain noise, the physical essence and the solution approach can be seen definitely from system frequency characters. The frequency characters are described by curves commonly, including gain and phase character curves.

In Bode principle, the phase margin γ and the gain margin h are used to describe system stability. Phase margin means that if system phase lag increases γ again for frequency ω_c, then the system is critical stable. The mathematical description is

$$\gamma=\pi+\angle G(j\omega_c)H(j\omega_c). \tag{5}$$

According to the definition of phase margin, there is

$$\gamma=\pi-\frac{\pi}{2}-\arctan\tan\tau\omega_c=\frac{\pi}{2}-\arctan\tan\tau\omega_c, \tag{6}$$

where ω_c the is cut-off frequency which can be obtained from equation $|G(j\omega_c)H(j\omega_c)|=1$, or

$$20\log\left|\frac{\sqrt{\cos^2\tau\omega_c+\sin^2\tau\omega_c}}{\omega_c}\right|=1.$$

Therefore,

$$\omega_c=1. \tag{7}$$

When system phase margin is between $\pi/6\sim\pi/3$, the better performance is got according to engineering experience. Note that arctantan $\theta\in(-\frac{\pi}{2},\frac{\pi}{2})$ for any angle θ. If the system phase margin is equal to or larger than the expected phase margin γ^*, then, we suppose

$$\gamma^*\le\frac{\pi}{2}-\arctan\tan\tau\omega_c\le\frac{\pi}{3}. \tag{8}$$

From (8), we have

$$k\pi+\frac{\pi}{6}\le\tau\omega_c\le k\pi+\frac{\pi}{2}-\gamma^*,\ k=0,1,2,\cdots. \tag{9}$$

Noting (7) yields

$$k\pi+\frac{\pi}{6}\le\tau\le k\pi+\frac{\pi}{2}-\gamma^*,\ k=0,1,2,\cdots. \tag{10}$$

If inequality (10) holds, the phase margin requirement is satisfied.

Other index indicating system stability is the gain margin h. It means that, if the coefficient of transfer function of open-loop system increases to h times of the original one, then the system is critical stable. For h, we have

$$20\log h = 20\log\frac{1}{\left|G(j\omega_g)H(j\omega_g)\right|} = -20\log\left|G(j\omega_g)H(j\omega_g)\right|, \tag{11}$$

where ω_g satisfies the following equation

$$-\frac{\pi}{2} - \arctan\tan\tau\omega_g = -\pi, \tag{12}$$

i.e.,

$$\text{arctantan } \tau\omega_g = \frac{\pi}{2}. \tag{13}$$

Therefore

$$\tau\omega_g = l\pi + \frac{\pi}{2},\ l = 0,1,2,\cdots. \tag{14}$$

Then,

$$\omega_g = \frac{(l+\frac{1}{2})\pi}{\tau},\ l = 0,1,2,\cdots. \tag{15}$$

From (11), the gain margin satisfies

$$20\log h = -20\log\left|\omega_g\right| = 20\log\frac{\tau}{(l+\frac{1}{2})\pi}. \tag{16}$$

From (10), we can get

$$\frac{(k+\frac{1}{6})\pi}{(l+\frac{1}{2})\pi} \le \frac{\tau}{(l+\frac{1}{2})\pi} \le \frac{(k+\frac{1}{2}-\frac{\gamma''}{\pi})\pi}{(l+\frac{1}{2})\pi}. \tag{17}$$

Based on the discussion above, we can see that, if (10) holds, the gain margin requirement is satisfied. So, time delay lies in a limited range, the gain and phase margins are satisfied, which is not considered to dispense with emendation and compensation.

For ATM network, non-real time network traffic control lightens the burden of network and satisfies the performance index at the same time.

Theorem 1: For ATM network with time delay, if (10) holds, then the expected gain and phase margins are satisfied, and it is not considered to dispense with emendation and compensation.

For the case that the time delay range dose not satisfy performance index, the series leading correction is processed. The transfer function of series leading correction is $G_c(s) = (1+aTs)/(1+Ts)$, where a, and T are parameters in revised transfer function which can be achieved by expected performance index.

Suppose that the cut-off frequency is ω_c^*, and the max leading angle frequency is ω_m. Obviously, there should be

$$-L'(\omega_c^*) = L_c(\omega_m) = 10\log a, \tag{18}$$

i.e.,

$$-20\log(\frac{1}{\omega_c^*}) = 10\log a. \tag{19}$$

It is easy to see from (19) that $a = (\omega_c^*)^2$. To guarantee system response speed and make use of phase leading character of networks adequately, let $\omega_m = \omega_c^*$. Then, we have

$$T = \frac{1}{\omega_m\sqrt{a}} = \frac{1}{a}. \tag{20}$$

Therefore, the revised system transfer function is

$$G(s) = \frac{(1+aTs)e^{-\tau s}}{(1+Ts)s} = \frac{(1+s)e^{-\tau s}}{(1+Ts)s}. \tag{21}$$

4. PERFORMANCE INDEX ANALYSIS

4.1. Stability

The source input traffic of the network system is supposed to be step-changed, i.e., $u(t) = \alpha I(t)$ where α is a constant larger than zero, and $I(t)$ is a unit step signal. The transfer function of input and output is

$$\frac{y(s)}{u(s)} = \frac{(1+aTs)e^{-\tau s}}{s(1+Ts)}. \tag{22}$$

Under the time domain transformation, the queue length corresponding to source input is

$$\begin{aligned} y_u(t) &= \alpha[(1-a)Te^{-\frac{1}{T}(t-\tau)} + (t-\tau) - (1-a)T]I(t-\tau) \\ &= \alpha[(a-1)T(1-e^{-\frac{1}{T}(t-\tau)}) + (t-\tau)]I(t-\tau). \end{aligned} \tag{23}$$

And, if the worst disturbance input is $f(t) = \beta I(t-T_1)$, where β is a constant larger than zero, and $t \ge T_1$. The system transfer function under the disturbance is

$$\frac{y(s)}{f(s)} = -\frac{1}{s}. \tag{24}$$

By transforming (24) to time domain, we get

$$y_f(t) = -\beta(t - T_1). \tag{25}$$

From (25), we can see that $y_f(t) < 0$. Therefore, the buffer queue length is

$$\begin{aligned} y(t) &= y_u(t) + y_f(t) \le y_u(t) \\ &= \alpha[(a-1)T(1 - \mathrm{e}^{-\frac{1}{T}(t-\tau)}) + (t-\tau)]I(t-\tau) \\ &\le \alpha[(a-1)T + (t-\tau)]. \end{aligned} \tag{26}$$

With $t \ge \tau$, the buffer threshold is supposed to be r^0. To guarantee that the buffer is not overflow and the system is stable, we must have

$$y(t) \le \alpha[(a-1)T + (t-\tau)] \le r^0. \tag{27}$$

That is,

$$\alpha \le \frac{r^0}{(a-1)T + (t-\tau)}. \tag{28}$$

The inequality relationship between the average gain of input rate and the buffer threshold is given in (28). If only source sending rate is less than the value, buffer is no overflow. In general references, only system stability is proved, but the supposing principle of buffer threshold is not given, or controller output is used to be source sending rate. For the round-trip time delay, however, the source end can not receive the rate value in time. The given inequality relation (28) can pass such problems.

Theorem 2: Suppose the source input rate is step-changed, i.e., $u(t) = \alpha I(t)$, and the system is in the worst disturbance, i.e., $f(t) = \beta I(t - T_1)$. If the inequality (28) is satisfied, then the buffer queue length is less than threshold r^0, and the system stability and quick response are guaranteed.

4.2. Utilization of buffer and links

The buffer queue length is

$$\begin{aligned} y(t) &= y_u(t) + y_f(t) \\ &= \alpha[(a-1)T(1 - \mathrm{e}^{-\frac{1}{T}(t-\tau)}) + (t-\tau)]I(t-\tau) \\ &\quad -\beta(t - T_1). \end{aligned} \tag{29}$$

If the buffer queue length is closer to the threshold, the buffer has the higher utilization ratio. Then, to get the higher utilization ratio, suppose $y(t) = r^0$ (in order to deal with outburst data flow, only a part of buffer capacity is generally taken as the buffer threshold. So, it is naturally to let $y(t) = r^0$), that is,

$$\begin{aligned} r^0 &= \alpha[(a-1)T(1 - \mathrm{e}^{-\frac{1}{T}(t-\tau)}) + (t-\tau)]I(t-\tau) \\ &\quad -\beta(t - T_1). \end{aligned} \tag{30}$$

Rewriting (30) and by (28), we have

$$\begin{aligned} &\beta(t - T_1) \\ &= \alpha[(a-1)T(1 - \mathrm{e}^{-\frac{1}{T}(t-\tau)}) + (t-\tau)]I(t-\tau) - r^0 \\ &\le \frac{r^0}{(a-1)T + (t-\tau)} \\ &\quad \times [(a-1)T(1 - \mathrm{e}^{-\frac{1}{T}(t-\tau)}) + (t-\tau)] - r^0 \\ &= r^0[1 + \frac{(a-1)T(-\mathrm{e}^{-\frac{1}{T}(t-\tau)})}{(a-1)T + (t-\tau)}] - r^0 \\ &= -\frac{(a-1)T\mathrm{e}^{-\frac{1}{T}(t-\tau)} r^0}{(a-1)T + (t-\tau)}, \end{aligned} \tag{31}$$

where $t \ge T_1$, so

$$\beta \le -\frac{(a-1)T\mathrm{e}^{-\frac{1}{T}(t-\tau)} r^0}{[(a-1)T + (t-\tau)](t - T_1)}. \tag{32}$$

The relation between available bandwidth and buffer threshold is given by the above inequality, then buffer is used adequately, and buffer threshold is confirmed by network bandwidth resource, the higher communication link bandwidth is guaranteed.

Theorem 3: Suppose the network system is under the worst disturbance, i.e., $f(t) = \beta I(t - T_1)$. If (32) holds, then, for any $t \in [0, +\infty)$, the system has higher buffer utilization and communication link bandwidth utilization.

4.3. Buffer utilization situation

Suppose that the buffer utilization ratio is A. To get the buffer utilization ratio under a specified performance index, the value range of A is discussed. Consider the following equation.

$$\begin{aligned} Ar^0 &= \alpha[(a-1)T(1 - \mathrm{e}^{-\frac{1}{T}(t-\tau)}) + (t-\tau)]I(t-\tau) \\ &\quad -\beta(t - T_1). \end{aligned} \tag{33}$$

Substituting (32) into (33) yields

$$\begin{aligned} Ar^0 &\ge \alpha[(a-1)T(1 - \mathrm{e}^{-\frac{1}{T}(t-\tau)}) + (t-\tau)]I(t-\tau) \\ &\quad + \frac{(a-1)T\mathrm{e}^{-\frac{1}{T}(t-\tau)} r^0}{(a-1)T + (t-\tau)}. \end{aligned} \tag{34}$$

From inequality (28), we see that, even though α gets its maximum value, i.e., $\alpha = \frac{r^0}{(a-1)T+(t-\tau)}$, the equality (33) still holds. So, we get

$$A \geq \frac{(a-1)T}{(a-1)T+(t-\tau)}. \quad (35)$$

Buffer utilization situation can be got in theory after determining each parameter. The higher buffer utilization ratio is achieved for $t \geq \tau$.

4.4. Relationship between frequency and time indexes

Bode principle is applied broadly in engineering. The frequency domain index is used in Bode principle, which has some relation with the time domain index. In practical application, the time domain index can reflect performance indexes directly. So, the relation description between the frequency domain and time domain in Bode principle is still worth to be studied.

$$M_r = \frac{1}{\sin\gamma} \quad (1 \leq M_r \leq 1.8), \quad (36)$$

$$\sigma = 0.16 + 0.4(M_r - 1), \quad (37)$$

$$t_s = \frac{K\pi}{\omega_c}, \quad (38)$$

$$K = 2 + 1.5(M_r - 1) + 2.5(M_r - 1)^2, \quad (39)$$

where M_r is the oscillation performance, σ is the overshoot, t_s is the transient time.

5. SIMULATION RESULTS

Consider the network system shown in Fig. 1. The bandwidth of each link is supposed to be 100Mbit/s. The buffer queue length is controlled to guarantee that the link congestion dose not occur, and the adequate and fair bandwidth utilization is obtained.

Let $\gamma^* \geq 5\pi/18$, $\omega_c^* = 2.27$rad/s, $\tau = 2$, $t = 2.5$, $T_1 = 2.45$. Then $a = (\omega_c^*)^2 = 5.14$, and $T = 1/a = 0.195$. The worst case of the change of ABR bandwidth is considered in simulations. The actual ABR communication bandwidth of link L_{n+1} is given as

$$f(t) = 100I(t) + 1000I(t-3) - 1000I(t-5) + 1000I(t-7) - 1050I(t-10).$$

It is shown in Fig. 3.

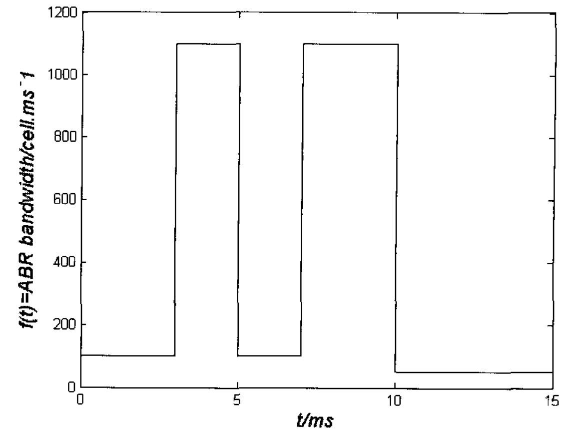


Fig. 3. Step change ABR communication bandwidth.

From (10), we know that if τ is in

$$k\pi + \pi/6 \leq \tau \leq k\pi + \pi/2 - 5\pi/18,$$

i.e.,

$$3.14k + 0.523 \leq \tau \leq 3.14k + 2.27.$$

In such case, the performance index is satisfied without network control.

In the case τ =2, it dose not satisfy the condition. In other words, it is necessary to control the network system. The transfer function is

$$G_c(s) = \frac{1+aTs}{1+Ts} = \frac{1+s}{1+Ts} = \frac{1+s}{1+0.195s}.$$

Then, the revised system transfer function is

$$G(s) = \frac{1+aTs}{1+Ts} \cdot \frac{e^{-\tau s}}{s} = \frac{1+s}{1+0.195s} \cdot \frac{e^{-\tau s}}{s}.$$

Since $\gamma'' = \gamma(\omega_c'') + \varphi_m$, where

$$\gamma(\omega_c^*) = \frac{\pi}{2} - \arctan\tan\tau\omega_c^*$$
$$= \pi/2 - 89\pi/200 = 11\pi/200,$$

$$\varphi_m = \arcsin\frac{a-1}{a+1} = \arcsin\frac{4.14}{6.14} = 47\pi/200,$$

we have

$$\gamma^* = \gamma(\omega_c^*) + \varphi_m = 11\pi/200 + 47\pi/200$$
$$= 58\pi/200 \geq 5\pi/18.$$

The expected phase margin is $\gamma^* \geq 5\pi/18$. Hence, the system requirement is satisfied.

Fig. 4 shows the gain and phase characters. The gain in delay loop is 1, and it dose not influence the gain margin. Since the transfer function contains delay, the phase curve is not equal to the phase margin. From Fig. 4, however, the revised phase increases more than $2\pi/9$, and the phase margin has linear relation with the phase curve. Then, the phase lag is considered to be revised effectively.

In general, the low frequency segment of open-loop frequency character represents the system stability; the intermediate frequency segment represents the

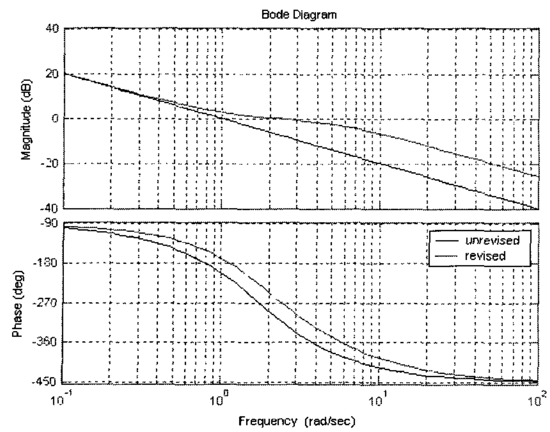


Fig. 4. Gain and phase character curve of unrevised and revised.

system dynamics; and the high frequency segment represents the system complexity and the noise restraining performance. Therefore, the design of control systems with frequency method is to add a suitable emendation equipment, which makes the open-loop frequency character have an expected shape. In other words, the gain in low frequency segment is large enough to guarantee the requirement of stability error; the intermediate frequency segment holds adequate wide bandwidth to guarantee the suitable phase margin; and the gain of high frequency segment should be decreased as fast as possible to reduce the noise influence.

From Fig. 4, we can see that the revised gain margin is increased. The intermediate frequency segment holds adequate wide bandwidth, therefore, the phase margin of the system is guaranteed. The gain of high frequency segment is lower, hence, the noise influence in networks is eliminated preferably.

In practical application, the time domain index can reflect some performance index directly, such as the transient time, etc. When $\gamma^* \geq 5\pi/18$, according to the relationship between frequency and time domains in Bode principle, we can get

$$M_r = \frac{1}{\sin\gamma} = \frac{1}{\sin 5\pi/18} = \frac{1}{0.766} = 1.3,$$

and

$$\sigma = 0.16 + 0.4(M_r - 1) = 0.16 + 0.4(1.3 - 1) = 0.28,$$

where $1 \leq M_r \leq 1.8$. Then, the maximal overshoot is 28%. From

$$K = 2 + 1.5(M_r - 1) + 2.5(M_r - 1)^2 = 2.675,$$

$$t_s = \frac{K\pi}{\omega_c} = \frac{2.675 \times 3.14}{2.27} = 3.7,$$

we see that the transient time is 3.7s. The network has better performance seen from time domain index. If the index is not satisfied, then the object requirement can be obtained by adjusting the parameters.

Next, we discuss the system stability, the buffer utilization and the communication link utilization situation. From Fig. 3, we know that the available bandwidth is $\beta = 1100$. According to (32), we have,

$$1100 \leq \frac{0.8 \times 0.077}{(0.8 + 0.5) \times 0.05} r^0 = 0.947 r^0.$$

Then, we get $r^0 \geq 1161$. The buffer threshold of link L_{n+1} is supposed to be $r^0 = 1170$. From (28), we have

$$\alpha \leq \frac{r^0}{[(a-1)T + (t-\tau)]} = \frac{1170}{1.3} = 900.$$

We may suppose $\alpha = 860$. According to (35), the buffer utilization ratio is $A \geq 90.5\%$. In Fig. 6, the buffer utilization ratio is $A \geq 94.1\% > 90.5\%$. Then, not only the higher buffer utilization is obtained but also the reliability of theory is shown.

Fig. 5 shows the unrevised buffer queue length. If the time delay exists in the needed emendation range, and no emendation is made, then the buffer queue length will exceed the threshold. From Fig. 5, we can see that, the buffer threshold is 1200 cells, and the queue length is more than 2500 cells, then serious network congestion happened.

Fig. 6 shows the revised buffer queue length. From Fig. 6, we see that, no buffer overflow occurs, and the higher buffer utilization is achieved. Therefore, the leading correction and compensation makes network get better stability and dynamic performance.

Fig. 7 shows the comparison of the buffer queue length under different control methods. PID and P methods are used to compare with Bode principle. In Fig. 7, the blue curve is result of PID method of closed-loop model, whose transfer function is

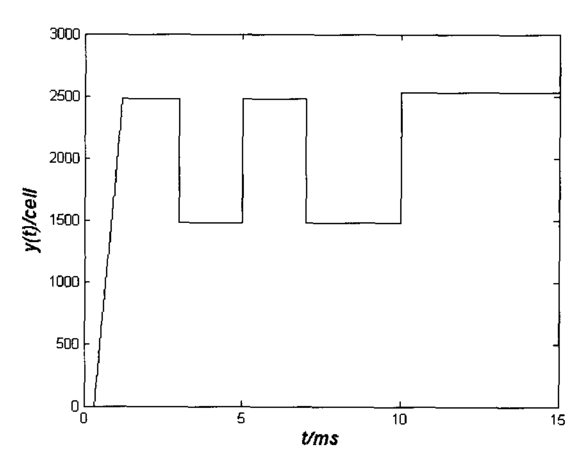


Fig. 5. Buffer queue length (unrevised).

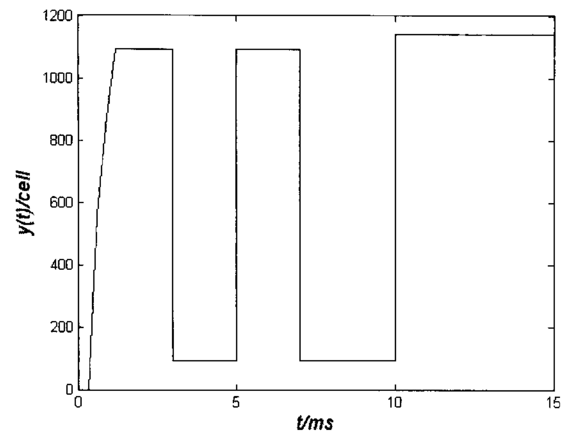


Fig. 6. Buffer queue length (revised).

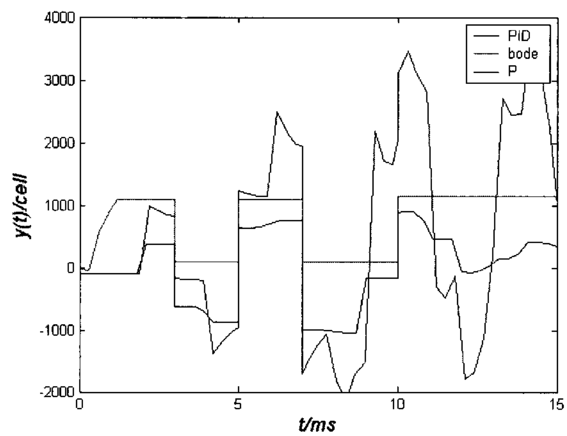


Fig. 7. Buffer queue length under different control methods.

$$G_{PID}(s) = (4.8s^2 + 3s + 0.6)/(4s^2 + 4s).$$

The parameters are adjusted by Ziegler-Nichols method. The response speed is quite quick. The excessive adjustment is, however, serious and the buffer is overflowed. The black curve is the result of P control method. The system dose not overflow, but the response speed is slow and the buffer utilization ratio is low. While Bode principle adopted in the paper guarantees overflow-free and higher buffer utilization ratio. Then, the better control effectiveness is gotten.

From the above analysis, we know that the source input rate has relation with buffer threshold, and the relation is got under the condition of assuring system stability. Hence, the system stability and the minimal cell loss are guaranteed. Otherwise, the maximal link utilization of network is got from theoretical analysis and simulation results.

Round-trip time is variable, which is considered in the design, and the necessary control range of time-delay is given. So, stability of virtual circuit with long and variable round-trip time can be guaranteed. From algorithm design and performance analysis, we can see that not only the system is stable, but also the better dynamics and adaptability to the environment change are achieved. The network parameters, such as the network bandwidth, the time delay, and the performance index, are variable. If the bandwidth varies, then other parameters can make corresponding varies to guarantee the system stability and dynamic requirement.

ABR service defines the fair capacity share, and the fair bandwidth allocation among ABR flow can be realized theoretically [7]. Otherwise, the proposed method is simple and flexible in algorithm analysis and design.

Bode principle adopted in the paper satisfies the standard of high performance ABR traffic control algorithm. In theory, not only network stability and dynamic performance are proved, but also the higher buffer utilization and lower cell loss are ensured. In practical application, the algorithm is more suitable for engineering technology personnel, which is simple and flexible in implementation.

6. CONCLUSIONS

The main function of traffic control is to protect the network and its users to realize the expected network performance, such as cell loss ratio or transmission time-delay and etc. The other function is to optimize network resource utilization to obtain higher network utilization ratio. In the paper, Bode principle is adopted to deal with the problem of ATM network traffic control. Not only effective utilization of buffer capacity and link bandwidth are guaranteed, but also network parameters are considered, which is favorable for network management and maintenance. Therefore, the study has broad application prospect and research importance.

REFERENCES

[1] S. Kamolphiwong, A. E. Karbowiak, and H. Mehrpour, "Flow control in ATM networks: A survey," *Computer Communication*, vol. 21, no. 11, pp. 951-968, November 1998.

[2] L. P. An, N. Ansari, and A. Arulambalam, "TCP/IP traffic over ATM networks with FMMRA ABR flow and congestion control," *Computer Networks and ISDN Systems*, vol. 29, no. 17-18, pp. 2091-2102, 1998.

[3] S. Willliam, *Computer Network and Protocol Technology*, China Electric Power Press, Beijing, 2005.

[4] ATM Forum, *Traffic Management Specification Version 4.0*, April 1996.

[5] F. Bonomi, D. Mitra, and J. B. Seery, "Adaptive algorithms for feedback-based flow control in high-speed wide-area ATM networks," *IEEE Journal on Selected Areas in Communications*, vol. 13, no. 7, pp. 1267-1283, July 1995.

[6] R. Jain, "Congestion control and traffic management in ATM networks: Recent advances and a survey," *Computer Networks and ISDN Systems*, vol. 28, no. 13, pp. 1723-1738, 1996.

[7] W. Shen, L. S. Hu, and H. H. Shao, "The design of traffic control system in networks-Kalman control algorithm," *Journal of China Institute of Communications*, vol. 15, no. 4, pp. 575-578, April 2003.

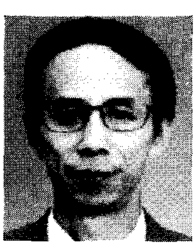
[8] S. Mascolo, "Congestion control in high-speed communication networks using the Smith principle," *Automatica*, vol. 35, no. 12, pp. 1921-1935, 1999.

[9] I. Lengliz and F. Kamoun, "A rate-based flow control method for ABR service in ATM networks," *Computer Networks*, vol. 34, no. 1, pp. 129-138, 2000.

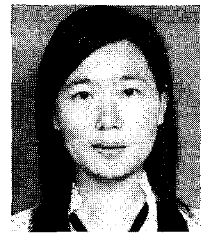
[10] F. Y. Ren, C. Lin, and F. B. Wang, "Variable structure controller for ABR flow control," *Journal of Software*, vol. 3, no. 14, pp. 562-568, 2003.

[11] C. Song, R. Nagarajan, and Y. T. Wang, "First-order rate-based flow control with dynamic queue threshold for high-speed wide-area ATM networks," *Computer Networks and ISDN Systems*, vol. 29, no. 17-18, pp. 2201-2212, 1998.

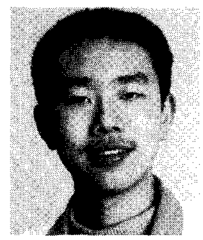
[12] X. R. Xie, *Computer Networks*, Dalian University of Technology Press, Dalian, 2004.



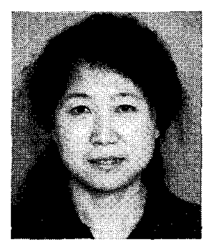
Yuanwei Jing is a Professor in the College of Information Science and Engineering at Northeastern University, China. He received the M.S. and Ph.D. degrees in Automatic Control from Northeastern University in 1984 and 1988. He developed post-doctoral research in the faculty of Electrical Engineering at Saint Cyril and Saint Methodius University in 1996. He was a Senior Visit Scholar in the Faculty of Computer Science Telecommunications at the University of Missouri-Kansas City in 1998-1999. His research interests include structure analysis and control of complex control system, control of telecommunication and computer networks.



Hui Zeng is a Ph.D. candidate in the College of Information Science and Engineering at Northeastern University, China. She received the M.S degree in Control Theory and Control Engineering from North-eastern University in 2005. Her major interest is ATM network traffic control and management.



Qingshen Jing is a Research Student in the College of Engineering at Peking University, China. He joined the research group of Prof. Long Wang at Peking University. His major interests include theoretical and applied mechanics, computations, and simula-tions.



Ping Yuan is an Associate Professor in the College of Information Science and Engineering at North-eastern University, China. She received the Ph.D. degree in Computer Application from Northeastern University in 2005. Her research areas include satellite communications, information systems, and network control.