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Virtual Routing Model and PI Rate Control Algorithm in High-band width Wireless LAN

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Abstract

Transmission accuracy of traditional Transmission Control Protocol (TCP) in high-bandwidth wireless network is low, and its efficiency is low. For these problems, the thesis proposes a virtual router model from the perspective of end-to-end, and on this basis of using the classical control theory the thesis proposes an end-to-end PI rate control algorithm--SPI RCP on the transmitter. According to the RTT information collected by the transmitter, the real-time queue length of virtual router is calculated, and by using the PI control theory the transmission rate of the transmitter is calculated to make the queue length of the bottleneck node is stabilized at a target location, at the same time to avoid the wrong adjust of congestion windows caused by wireless link packet loss. The end-to-end design of new algorithm saves router resources, enhances the stability of the queue length, and reduces network congestion packet loss. The simulation results show that compared with the traditional congestion control protocol, the new mechanism can better control the queue length of the router and improve the stability and network throughput of the network load.

Keywords: *congestion control, PI controller, high-bandwidth network, wireless network, virtual router*

1. Introduction

Currently, Transmission Control Protocol (TCP) is the most widely used transport protocol on the Internet; 90% of the amount of data on Internet is transmitted by using the TCP protocol, so TCP plays a crucial role on the overall performance of the network [1]. Now, with the increase of network applications, many network applications require a large amount of data, real-time, and stable data transmission, especially with the popularity of high bandwidth and wireless networks [2]. People want to be able to use laptops and other mobile devices fastly and stablely download data via Wireless Local Area Networks (WLAN). For example, the users download streaming media data from the Internet by using high-bandwidth wireless local area network, and researchers download real experimental data from the experimental data acquisition devices by using high-bandwidth wireless local area network [3]. However, the performance of the traditional TCP in high-bandwidth wireless LAN is not satisfactory. There are two major problems of traditional TCP in high-bandwidth wireless network: 1) In the congestion avoidance stage, Additive-Increase/ Multiplicative Decrease mechanism (AIMD) is used [4] to adjust the congestion window of the sender, while this kind of mechanism in a high-bandwidth network easyly causes congestion window to the transmission and a greater sawtooth wave of network load to [5], which is not conducive to the full bandwidth utilization and stable transmission of data. 2) The traditional TCP treat packet loss as the judgement to determine the network congestion and according the packet loss to reduce the transmitter's congestion window. This kind of adjustment method to the congestion window has a better performance in the conventional wired network with

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lower error rate [6-9]. However, in a high error rate in the wireless network, traditional TCP cannot effectively distinguish conventional radio link error loss and network congestion losses, which results that the transmitter wrongly treat the network congested when the radio link is packet lossed, and thereby wrongly reduces the size of the congestion window, and then reduces the network performance. These factors lead to the users of traditional TCP cannot get higher network performance through high-bandwidth wireless network to download data; meanwhile, the network bandwidth is not fully utilized, which results in a waste of resources.

In the past period of time many researchers have done a lot of research to the problems of the traditional TCP, and worked out a lot of solutions. According to the location of the control algorithm, these solutions can be divided into sender-centric approach [10], the network intermediate nodes-centric approach [11], the receiver-centric approach [12], as well as mobile host-centric approach [13]. The improvement programs based on congestion control algorithm mechanism can be divided into: optimization on the existing congestion control algorithm [14] and redesign congestion control algorithm using new theoretical.

Main innovation of this paper:

- In this paper, for unsatisfactory performance problems of the traditional TCP in the high-bandwidth wireless networks, from end-to-end perspective the author presents a virtual router model, and on this basis proposes an end-to-end PI rate control algorithm -SPI RCP by using the classical control theory.
- Compared with the traditional transmitter congestion control algorithms, the algorithm proposed in this paper no longer adjust the congestion window based on packet loss information, but according to the RTT collected by transmitter calculate the real-time queue length in the virtual router, and uses the PI control theory to calculate the transmission rate of the sender to stabilize the queue length of network bottleneck nodes at the target position, which reduce influence on network performance caused by wireless packet loss and enhance network performance.
- Compared with the existing PI control algorithm, the end-to-end design of the new algorithm saves router resources, enhances the stability of the queue length, and reduces network congestion packet loss. The simulation results show that the new mechanism can well control the real-time queue length on bottleneck nodes to improve network utilization and throughput.

2. Transmitter PI Speed Controller Model and Assumptions

A. transmitter PI speed controller model sender

According to end-to-end network model, when the data transmitter sends data to the receiver, a series of router R1～Rm the data flowed can be combined as a virtual router, and the queue of each router can be combined as a virtual queue, as Figure 1 shows. In the process of data flow achieving the data receiver through the virtual router, in network all the data packets through the virtual router are treated as interference packets. Assuming that all the transmitter of the interference are called Sd, and the receiver of interference data packet is generically referred as Dd, the data transmitter S sends data with the speed of Sv(t) and the interference data transmitter Sd is sent with the speed of vd(t).

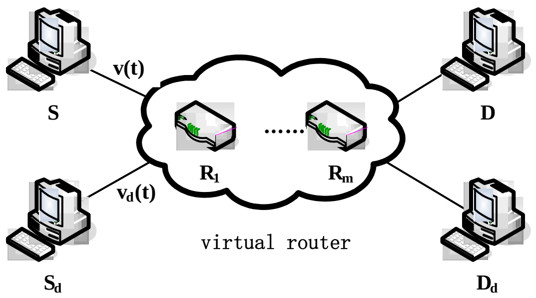

Figure 1. Data Communication Network Model

Shown in Figure 2, the thesis designs a PI speed controller for each data stream in the data transmitter to calculate a transmission rate of the data stream. Meanwhile, a virtual router is designed by the transmitter for each data stream to simulate the router all of the data flowed through. The thesis assumes that the maximum capacity of a virtual queue of virtual router is B Byte, the queue length of the control target is q0 Byte, real-time queue length is q(t) Byte, the queue length of initial router is q(0) and the processing speed is μ bits / second. The average time delay of packets from the transmitter to the virtual router is τf; the average time delay of packets return to transmitter is τb; the average loop delay is τ=τf +τb.

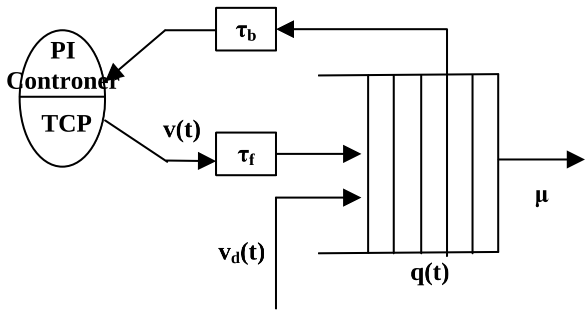

Figure 2. The Queuing Model of Virtual Router for Data Transmitter

According to the queuing model of virtual router in Figure 2, the rate of change of the queue length at the time t can be expressed as follows:

$$\dot{q}(t) = v(t - t_f) + v_d(t) - \mu \tag{1}$$

B. Performance Modeling for Network Calculus

Network calculus is a kind of mathematic tool to manage queue based on the minimum plus algebra. It is mainly used for network modeling and quantitative analysis. Compared with traditional queue management theory, network calculus pays more attention to the worst-case performance instead of the general case. It can be used to solve some basic attributes, such as capacity requirements in node buffer, network transmission delay and node bandwidth needs. It is defined as follows:

Definition 1 (generalized increasing function) For $\forall s \le t$, if $f(s) \le f(t)$, f is a increasing function

Definition 2 (generalized extended function set)

If $F = \{f(t) | , f(t) = 0, \forall t < 0; f(s) \le f(t), t \in [0, +\infty]\}$,

F is a generalized extended function set.

Definition 3 (minimum convolution) For $\forall f, g \in F$ the minimum convolution operation of function f and g is as follows:

$$(f \otimes g)(t) = \inf_{0 \le s \le t} \{ f(t-s) + g(s) \} \cdot \quad (2)$$

Definition 4 (minimum deconvolution) For $\forall f, g \in F$ the minimum deconvolution operation of function *f* and *g* is as follows:

$$(f \varnothing g)(t) = \sup_{s \ge 0} \{ f(t+s) - g(s) \} \cdot \quad (3)$$

Definition 5 (arrival curve) Given a generalized increasing function $\alpha \in F$, if the cumulative function R of the input stream for any time s and t are met:

$$R(t) - R(s) \le \alpha(t-s) \Leftrightarrow R \le R \otimes \alpha \cdot \quad (4)$$

α is called the arrival curve of *R*, in which $0 \le s \le t$.

Definition 6 (service curve) Given a generalized increasing function $\beta \in F$, if the cumulative functions for a system S and the stream flowed through S respectively are *R* and R^* on the the input terminal and the output terminal and only if

$$R^*(t) \ge R(s) + \beta(t-s) \Leftrightarrow R^* \ge R \otimes \beta \cdot \quad (5)$$

β is called the service curve of *S* for stream, in which $0 \le s \le t$.

Definition 7 (strict service curve) Given a generalized increase function $\beta \in F$, for a system S and the stream flowed through S the cumulative function *R** of the stream flowed through transmitter at any peak period [*t*, *t+u*] satisfy:

$$R^*(t+u) - R^*(t) \ge \beta(u) \cdot \quad (6)$$

β is called the strict service curve of *S* for stream, in which $u \ge 0$. Strict service curve is a special case of service curve.

With the known data stream arrived at curve α and service curve β, the following several theorems can be draw.

Theorem 1 (backlog upper bound) Supposed the data stream α in arrival curve pass by the system S provided service curve as β, then for any time t, the backlog data in system S satisfies:

$$R(t) - R^*(t) \le v(\alpha, \beta) = \sup_{s \ge 0} \{ \alpha(s) - \beta(s) \} \quad (7)$$

Theorem 2 (delay upper bound) Supposed the data stream α in arrival curve pass by the system S provided service curve as β, then for any time t, the delay of data stream in system S satisfies:

$$d(t) \le h(\alpha, \beta) = \sup_{t \ge 0} (\beta^{-1}(\alpha(t)) - t) \quad (8)$$

Theorem 3 (output upper bound) Supposed the data stream α in arrival curve pass by the system S provided service curve as β, the output upper bound is as follows:

$$\alpha^*(t) = \alpha(t) \varnothing \beta(t) \quad (9)$$

Theorem 4 Supposed the survival curves in system *S*1 and system *S*2 are respectively β_1 and β_2, the total service curve provided by system *S*1 and system *S*2 is β, and then $\beta = \beta_1 \otimes \beta_2$.

Theorem 5 (remaining service curve) Supposed a data stream A consists of two share a single data stream *A*1 and *A*2 (arrival curves respectively are α_1 and α_2). *A*1 and *A*2 flow through the system S in an arbitrary order, and then the system S provides strict service curve to service the data stream A , so:

$$\beta^1 = [\beta - \alpha_2]^+ \cdot \quad (10)$$

Shown as the above formula, $[x]^+ = \max(0, x)$. If $\beta^1 \in F$, β^1 is called the remaining service curve of A_1. It is noteworthy that, β must be the strict service curve, or the formula (9) cannot be established [5].

3. The Design of PI Speed Controller

According to queuing model of the virtual router, a rate control system on transmitter is designed shown in Figure 3.

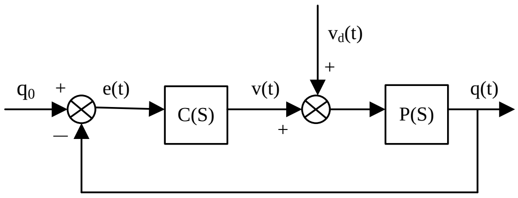

Figure 3. Queuing Model of the Virtual Router Control System

Wherein, q_0 is the queue's control target queue length; $e(t)$ is deviated from the queue length; $q(t)$ queue is the real-time queue length; $C(s)$ is the PI speed controller; $P(s)$ is the control object; $v(t)$ is the transmission rate for the data transmitter; $v_d(t)$ is the transmission speeds of interfering with data.

According to Figure 3, the formula (11) can be used to calculate the transmission rate of transmitter data:

$$\begin{aligned} v(t) &= K_p e(t) + K_I \int_0^t e(t)dt \\ &= K_p(q_0 - q(t-\tau_b)) + K_I \int_0^t (q_0 - q(t-\tau_b))dt \end{aligned} \qquad (11)$$

According to the formula (10) and formula (11) the control systems can be calculated the output response affected by input signals and interference signals. Superposition principle is applied to obtain the total output response simultaneously affected by the input signal and interference signals. The total output response $Q(s)$ output response of the total Q (s) is as follows:

$$Q(s) = \frac{V_d(s)s^2 + (K_p q_0 - 2\mu)s + K_I q_0}{s^3 + K_P e^{-s\tau} s^2 + K_I e^{-s\tau} s} \qquad (12)$$

Let $e^{-\tau s} \approx 1 - \tau s$, when input $R(s) = \frac{q_0}{s}$, according to the equation (12) characteristic equation of the closed-loop control system can be obtained as follows:

$$s^2 + \frac{K_P - K_I \tau}{1 - K_P \tau} s + \frac{K_I}{1 - K_P \tau} = 0 \qquad (13)$$

To ensure the stability of the closed loop control system, the two roots (i.e. the poles of the second closed-loop control system) of the formula (13) are on the left half plane of S plane.

The standard formula of closed loop transfer function in second-order control system is as follows [12]:

$$G_C(s) = \frac{\omega_n^2}{s^2 + 2\zeta\omega_n s + \omega_n^2} \qquad (14)$$

Wherein, ζ is the damping ratio of control system; ω_n is the undamped natural oscillation angular frequency of the control system.

The characteristic equation in equation (14) is as follows:

$$s^2 + 2\zeta\omega_n s + \omega_n = 0 \qquad (15)$$

Compared equation (13) and equation (15), a group of formulas of K_p 、K_I and ω_n、ζ can be obtained:

$$K_P = \frac{\omega_n^2 \tau + 2\zeta\omega_n}{\omega_n^2 \tau^2 + 2\zeta\omega_n \tau + 1} \qquad (16)$$

$$K_I = \frac{{\omega_n}^2}{{\omega_n}^2\tau^2 + 2\zeta\omega_n\tau + 1} \quad (17)$$

Therefore, when the appropriate parameters ζ and ω_n are choosing, controller parameters K_p、 K_I and the ideal control system transient response can be got.

4. Realization of Congestion Control Algorithm

Similar to the traditional TCP algorithms, the process of new designed algorithm adjusting the congestion window is also divided into two phases: the slow start phase and congestion avoidance phase. However, the specific implementation of each stage different from the conventional TCP.

Slow start phase: once the data stream is connected, the congestion window is initialized to 2, and it is adjusted by the way of exponential growth mode. After several transmission cycle, according to the received information of ACK, parameters (such as RTT, μ, *etc.*,) required by PI control is initialized, and then the congestion control algorithms transferres to congestion avoidance phase.

Congestion avoidance phase: According to the target queue length of virtual router and the real-time queue length obtained by calculation, the transmitter's transmission rate can be calculated by PI controller and the congestion window can adjusted. In order to improve the fairness with existing protocols, when the congestion window obtained by calculation is less than 2, the slow start phase will be backed.

Several Key Mechanisms on PI Congestion Control Algorithms:

1. In order to make TCP protocol current used in the network based on the windows compatible, the calculated transmission rate $V(t)$ is transferred into the transmitting window $W(t)$, that is, $W(t)＝v(t)*\tau$. The send window's size is limited as the integer in [2，W_{max}].

2. When RTT is recorded, the RTT of packet is recorded, in which it is successfully transmitted only once, and the RTT of packet cannot be recorded, if it is the retransmission data. When the packet is retransmitted, the original data packet may not be lost, and maybe it is just delay to reach the receiver. If the RTT records the results, there may appeare some mistakes.

3. The calculation methods of the average loop delay τ: the exponentially weighted moving average method is used to calculate the average loop delay time, assuming τ_{new} is the latest RTT sampling, $\tau=0.8*\tau+0.2*\tau_{new}$.

4. Calculation on real-time queue length: $q(t)=q(t\text{-}T)+(\tau_{new}–\tau_{old})*\mu$. Wherein $q(t)$ represents the current sampling real-time queue length; q(t-1) represents the last sampling real-time queue length; τ_{old} is the sampling RTT at t-T time; T is the calculation period of transmission rate. The initial is supposed as q(0)=0.

5. Once the network packet loss occurred, it will be retransmitted immediately, but the retransmission does not reduce the value of congestion window.

5. Performance Evaluation

In the Qualnet simulation software [13], the new protocol is implemented based on Reno protocol code. Through the simulation, the author makes a performance comparison between the new control protocol SPI RCP captain and the controller PI RED of classic queue router length.

In the process of simulation, classical dumbbell topology is used, shown in Figure 4. Data transmitter is linked via a wired link and router R; the link bandwidth is of 100Mbps; the delay from S1 ~ S5 to router are10ms, 15ms, 20ms, 25ms, 30ms. Data receiving terminal connected with the AP via a wireless link; the wireless protocol uses 802.11b, and the bandwidth is 11Mbps. The router R connected with the AP via a 1Gbps wireless link and the AP via a wired link, and the delay is 50ms. Suppose the cache of the router R and receiver is greatly, the cache set of AP is 60 packets; the packet size is 1000B; AP is the network bottleneck nodes.

Data transmitter $S_1 \sim S_4$ respectively send FTP long data stream to the data receiving terminal $D_1 \sim D_4$ via the router R and wireless AP. To simplify the model and analyze the dynamic stability of the system when the data stream flow through the network, data stream flowed through the classic analog step-style UDP network is used, in which state streams' start time is 50s and the end time is 100s. The data is sent by using CBR; the transmission rate is 2Mbps; the transmission rate of the other times data stream are zero; each simulation time is 300s.

In the simulation process, when the conventional PI RED queue length control protocol is used, the transmitter uses TCP Reno, and PI controller's parameters are set according to the setting principles [6]. The queue length of the set target is 20 packets. When the newly proposed SPI RCP is used, the AP uses the FIFO algorithm and sets the queue length of the virtual router control target as 20 packets; ζ is 0.6; ω_n is $0.4/\tau$.

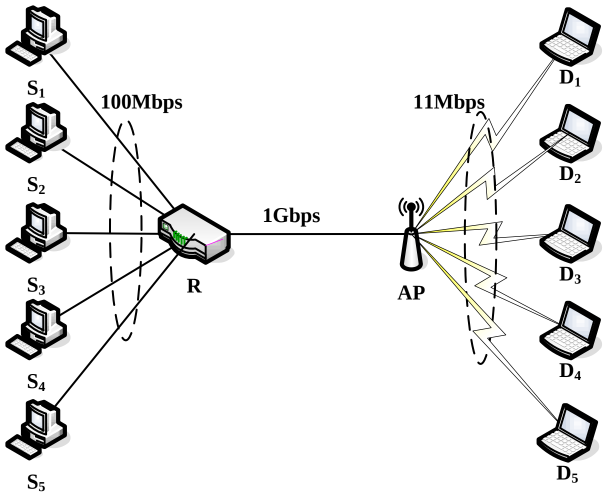

Figure 4. Dumbbell Topology

Experiment 1: The wireless link packet loss is set as zero and the performance of network is analyzed only under the circumstance of congestion packet loss. In the simulation process, the cached real-time queue length and the real-time packet loss rate of bottleneck node AP are recorded. And then the average value of every 0.1s is adopted as a sample value, and the results are shown in Figure 5 and Figure 6. Through these two parameters, it can be analyzed that the stability of bottleneck nodes' queue length caused by the new protocol and the classic routers queue length control protocol, and the performance of the router congestion packet loss rate is also analyzed.

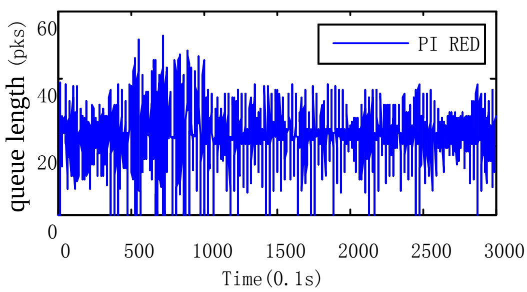

Figure 5(a). Real-time Queue Length

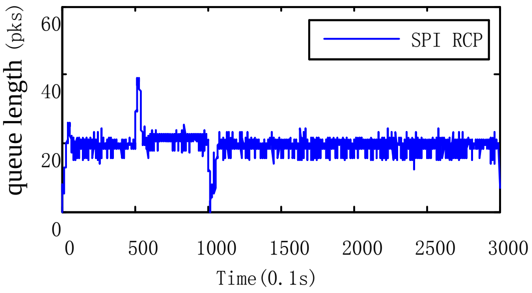

Figure 5(b). Real-time Queue Length

From the two sub-graphs in Figure 5 it is shown that there is a significant difference in the control effects of real-time queue length caused by the two protocols for bottleneck node AP. When the classical PI RED queue length control algorithm is used, the queue length of AP can be stabilized at a relatively fixed range with 20 packets in the center, while there is still a large fluctuation of queue length, especially there are UDP data adding the internet between 50s and 100s. In contrast, the new proposed controller can make a better stability around the target queue length (20 packets). Although there are some changes when the UDP data stream input and output the internet, they will back to the target position soon. Therefore, the new proposed algorithm enables the real-time queue length of the bottleneck nodes stabilized at the target location, so that more cache space is leaved for the bottleneck nodes and the robustness of the system dealing with burst data stream is improved.

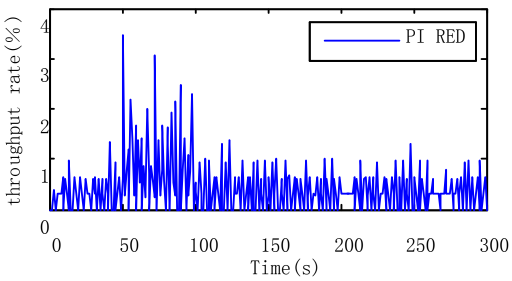

Figure 6(a). Real-time Packet Loss Rates

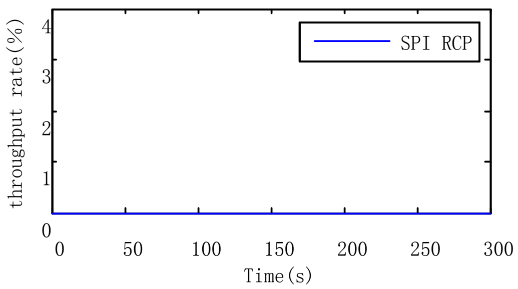

Figure 6(b). Real-time Packet Loss Rates

The two sub-figures in Figure 6 show real-time packet loss rate when the system uses the two kinds of protocols. When the PI RED algorithm is used, the packet loss rate without the UDP data stream is remained in a small fluctuation range, in which the average value is by 0.4%; When there exists UDP data streams within 50s ~ 100s, the packet loss rate is fluctuated a lot, in which the average value is up to 0.86%. When the new protocol is used, in the simulation process there is no packet loss, and then the desired result is achieved. After analyzed the reasons, it can be found that the PI RED control algorithm via implicit loss inform the transmitter reduce the transmission rate, so when the queue length is increased, the algorithm will lose some packets to achieve the purpose of stabilizing the queue length. The new designed algorithm actively probe the changes router's queue length via transmitter uses the changes of RTT, and the control theory is used to adjust the transmission rate. It can immediately reduce the transmission rate after the queue length exceeding the target queue length, rather than adjust transmission rate after congestion occurs. So the packet loss rate of the new algorithm can be controlled in a very low rate. Compared with the real-time queue length of SPI in Figure 5, it can be shown that, under the control of the new algorithm, there is no full condition of the queue, so there is no packet loss.

A large number of simulations show that in the current experimental configuration, when the transmission rate of the UDP data stream is less than 2.8Mbps, the new control algorithm allows the queue length stabilized at the target location, and no packet loss occurred. But when the transmission rate of the UDP data continues increases, the new control algorithms will not be able to make the queue length stabilized at the target location, and there will be packet loss. Because the UDP data account a large proportion of system load and at this time both new algorithm and traditional algorithm's adjustment on queue length become limited.

Experiment 2: In this experiment, the throughputs of the two protocols under different wireless packet loss rates are simulated. In the simulation process the set of the wireless packet loss rate is between 0～10％, and the other set of the experiment is the same with the Experiment 1. In the simulation process, throughput the throughput for each receiver is recorded, and the total throughput rate is shown in Figure 7.

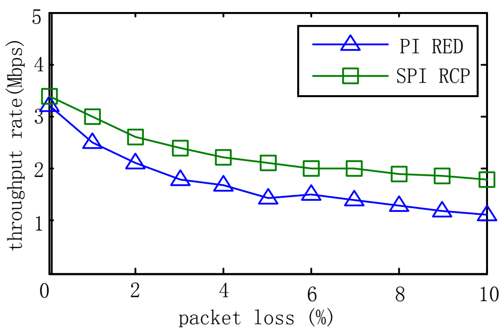

Figure 7. Total Throughput of the System

From Figure 7 it can be seen that the total throughput of the two protocols are decreasing with the increase of wireless packet loss rate, because with the increase of wireless packet loss, the system need to retransmit a large number of lost packets, which resulted in the bandwidth waste, so the throughput will be reduced. But on the whole, the throughput of the new protocol is better than the traditional protocol, and improvement degree will increase with the increase of packet loss rate. Because the higher the packet loss rate is, the greater the impact on transmitter's congestion window using the traditional protocol. However, the adjustment of the congestion window using the new protocol is not subject to wireless packet loss, so the new protocol can achieve higher throughput.

6. Conclusion

TCP plays a key role in the process of data transmission in computer network. As to the problem of low performance of traditional TCP in current network, the thesis establishes a end-to-end virtual router model, and presents a new end-to-end PI rate control algorithm.

The algorithm calculates and adjusts the data transmission rate with data transmitter using the classical control theory, and then makes control on the queue length of the router to stabilize it at the target location, thereby reduces network load fluctuations, increases the capacity of network anti-burst packet, reduces congestion packet loss, and reduces impact on network performance caused by wireless link packet loss. The simulation results show that there is a great improvement of new protocol on the stability control of the queue length, compare with the traditional routers' control algorithm of queue length PI RED, meanwhile, the properties is improved such as network throughput. The end-to-end design of the new protocol only need to increase a small amount of code on the transmitter side only, and then the satisfactory results can be achieved.

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