Establishment and implementation of network congestion control algorithm based on real-time streaming transmission

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Abstract

In order to improve the network performance of computer and avoid the occurrence of network congestion better, this paper analyzed the problems faced by the integration of real-time streaming transport and network and their solutions in the perspective of the characteristics of real-time streaming transport. As to the congestion control of streaming media, TFRC algorithm was analyzed emphatically. Based on TFRC, network congestion was improved, monitored and predicted with parameter of real-time cache length; sending rate was corrected based on cache length when the network was saturate, in order to avoid congestion in time and improve fairness; at last, the test comparison proceeded by network simulation platform NS2. The result indicated that, the improved TFRC was fairer; meanwhile, it showed more friendliness to TCP.

Keywords: real-time streaming transport, congestion control, TFRC, cache length

1 Introduction

Network congestion is caused by the mismatching of network resources and data transmission, for example, the unreasonable network cache space, bandwidth capacity, processor performance, network structure; they detailed present as information packet reception delay, discard probability increasing, upper application system performance decreasing, etc. In the current stage, network protocol plan is widely applied in stream transmission [1,2]. Therefore, the research on network congestion control mainly focus on network TCP protocol, including improving TCP protocol, developing new congestion control protocol by simulating AIMD algorithm of TCP, or regulating transmission speed of media stream according to the TCP throughput capacity under stable state [3,4]. Wang Wenliang et al proposed a TCP congestion control method based on LwIP, detailed stated the implementation method of slow start algorithm, congestion avoidance algorithm, fast retransmission algorithm and restoring method, and optimized TCP congestion control method of LwIP according to correction of fast recovery algorithm of TCP. Lv Guanqiao [6] et al proved the effectiveness of TCP improvement as compared in the perspective of channel utilization, communication stability and fairness. Besides the above methods, TCP friendly congestion control mechanism is the most discussed and promising mechanism in recent years [7]. This paper first introduced TCP-friendly rate control protocol, then analyzed TFRC algorithm emphatically, adjust transmission speed through packet loss probability, round-trip time with the current formula. Meanwhile, smooth packet loss internal algorithm was introduced to precede weight average calculation on the newest eight packet loss rate to ensure smoothness of transmission speed. And then, based on TFRC, these papers introduced additional web cache parameters, proposed MulRFRC algorithm, estimate the behaviors of various TFRC data flow modify transmission speed and improved TCP friendliness and finally made experimental verifycation with network simulation platform NS2.

2 Real-time streaming transmission

2.1 DEFINITION OF STREAMING TRANSMISSION

Streaming transmission is process to transmit multimedia data from transmitting end to destination in the form of continuous data streaming and then play multimedia while receiving data in destination. It can be divided into real-time streaming transmission and progressive streaming transmission. Progressive streaming transmission refers to that, user only can visit the multimedia data that has been downloaded, moreover, the data is downloaded according to certain order. Real-time streaming transmission refers to transmit data into user side through certain network protocol, which is suitable for live event [8].

2.2 CHARACTERISTICS OF REAL-TIME STREAMING TRANSMISSION AND PROBLEM OF ITS INTEGRATING WITH INTERNET

The characteristics of real-time streaming transmission includes high bandwidth, burstiness, the same speed, voice frequency and video sampling rate while receiving end playing multimedia data in transmission.

In practical network, communication network is characterized by limited link bandwidth, unpredictable data packet delay, which is inconsistent with high bandwidth, real time and burstiness of real-time multimedia data strea-

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ming. Transmitting data by dedicated links is expensive and unpractical. As a resources-shared network system, some characteristics of internet is not suitable for multimedia data transmission. Normal transmission of multimedia data flow in internet should take some factors into account: internet has enough bandwidth, data packet can be transmitted to destination address of the same group at the same time, thus to reduce to burden of multicast; internet has relative mechanism and can meet the demand of enough bandwidth resources; internet transmission technology is based on data packet switching, and should ensure real-time data packet to reach receiving end in time according to certain order, thus play continuously and synchronously.

3 Research on network congestion control

3.1 NETWORK CONGESTION ANALYSIS

Network congestion mainly refers to that, the demand on network resources of user exceed the current volume, thus cause overload condition, decrease the communication of network and network throughput capacity, increase packet loss rate, data packet delay, and even cause network collapse. The main reason is that, link bandwidth, data packet processing capacity of network node and node cache cannot meet the demand of user [9]. Limitation of link bandwidth is because the channel capacity of transmission media is limited and the exceeding data volume capacity can cause loss or error of data packet; the limitation of data at node refers to the congestion when processor cannot process data packet, update of routing table and retransmission of data packet; node cache is induced by overflow of data packet and change of performance in buffer memory space caused by limited link bandwidth and insufficient processor. Moreover, complex network structure and unreasonable routing algorithm principle are also possible to induce network congestion.

3.2 TFRC ALGORITHM STUDY AND PROTOCOL

TCP protocol based on AIMD algorithm is not suitable for all internet application. New transport layer protocol is needed to meet multimedia application in order to meet the requirements of stable bandwidth, low delay and small shake. TFRC ensures the service quality of network in the premise of ensuring friendliness of TCP. TFRC is a kind of congestion control algorithm suitable for traditional and TCP data flow-dominated network environment. It can ensure data flow to have smooth transmit speed, reduce transmit speed through theoretical formula and is suitable for real-time multimedia data flow. The relative function is:

$$T = \frac{s}{R\sqrt{\frac{2p}{3}} + t_{RTO}\left(3\sqrt{\frac{3p}{8}}\right)p\left(1+32p^2\right)}. \quad (1)$$

In the Equation (1), T is the maximum of transmit rate of data flow, s is the size of data packet, R is the round-trip time of data packet, p is the packet loss rate in stable state, t_{RTO} is the retransmission time of TCP packet loss. If compared the different data flow effect in the same network environment, it is better to choose same response function to control transmission speed and fairly use network resources. Therefore, good use of TCP response function can ensure fair competition of TFRC and TCP as well as smooth of transmit rate. The flow of TFRC congestion control mechanism is as shown in Figure 1:

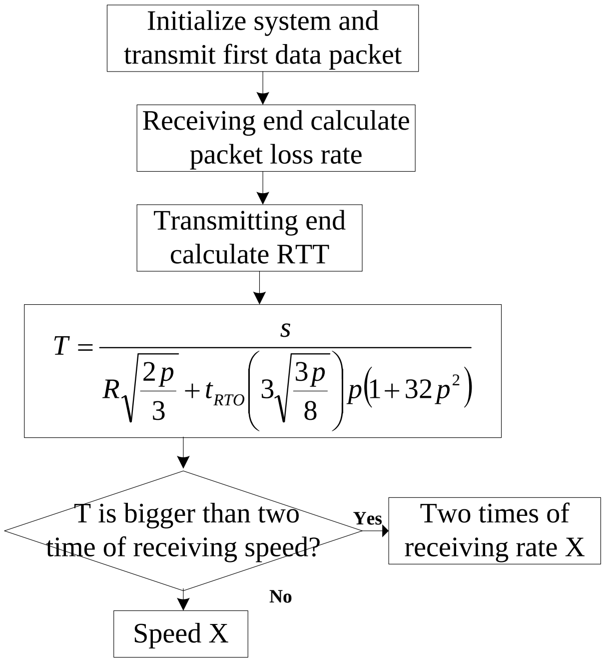

FIGURE 1 Flow chart of TFRC congestion control mechanism

Calculating packet loss rate is the most important part of TFRC algorithm. TFRC selects packet loss internal smooth algorithm to calculate the probability of packet loss event with weighted average, and can restrain some non-representative packet loss as well as reduce the shake of packet loss rate. Weighted average algorithm is as follows:

$$\hat{s}(1,n) = \frac{\sum_{i=1}^{n} w_i s_i}{\sum_{i=1}^{n} w_i}, \quad (2)$$

where is refers to the former number i packet loss event internal, which can be expressed as the number of data packets, $\hat{s}(1,n)$ refers to the smooth result of the former n packet loss event interval; the calculation of weight w_i is as follows:

$$\mathrm{w}_i = \begin{cases} 1, & 1 \le \mathrm{i} \le n/2 \\ 1 - \frac{i-n/2}{n/2+1} & n/2 < i \le n \end{cases}. \quad (3)$$

When sample s_i take i=0, s_0 refers to the number of data packet received during the last packet loss event to the current time. The mean value of s_0, s_1,…,s_{n-1} is:

$$\hat{\mathrm{s}}(0,n-1) = \frac{\sum_{i=0}^{n-1} w_{i+1} s_i}{\sum_{i=0}^{n-1} w_{i+1}}. \quad (4)$$

Finally, it is obtained that the average interval $\hat{s} = \max(\hat{s}(1,n),\ \hat{s}(0,n-1))$, and the probability of packet loss event is:

$$\mathrm{p} = \frac{1}{\hat{s}}. \tag{5}$$

The use of packet loss event probability can simulate "reduce transport protocol while coming across packet loss" [10]; meanwhile, use of TFRC algorithm can make transmit rate be smoother. However, TFRC responses slowly to packet loss, cannot be able to decrease transmit speed in time, thus occupy too much bandwidth. This paper aimed to improve based on this defect, thus enhance the fairness of TFRC.

3.3 TFRC IMPROVED ALGORITHM

Packet loss rate can be updates in a high speed through reduce sample number N, or increase weights of the newest samples, at the same time, shake of throughput rate increases and the redundant noise is introduced. This paper introduced extra network parameter - real-time cache length to correct packet loss interval, as shown in Equations (2)-(6):

$$s_m = \hat{s} \cdot f_{\mathrm{RealTime}}, \tag{6}$$

where S_m is the packet loss interval after correction, f_{RealTime} is the introduced network parameter, which is not related to network condition, but can reflect and predict the congestion and saturation of network. In network with coexisting of TFRC and TCP, one-way delay of TFRC data packet can monotonically increase or decrease within a large range, emerging lots of peak value. Transformation process of network state: free network, transmit speed increases, use efficiency improves – data packet increases, one-way delay grows – network congestion – congestion control – free network, in dynamic balance. One-way delay of TFRC is monotonous. When the delay value increases, data packet is considered to be cached, and the amount of data packet is the network cache length; when the delay value remains the same, then network is free and the cache is 0; when the delay value decreases, then cache length decreeses, as shown in Figure 2:

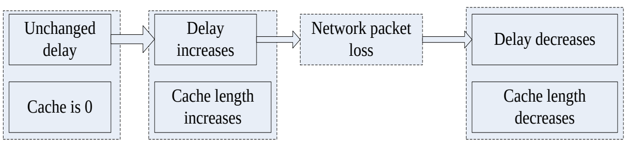

FIGURE 2 Change of cache length

When data packet reach receiving end, the one-way delay time of data packet is calculated. The equation is as follows:

$$\mathrm{stt\_sample}_i = t\_\mathrm{now}\text{-}t\_send_i. \tag{7}$$

In addition, instantaneous noise should be filtered, for example, the first sample takes $\mathrm{stt}_{i-1} = 0$. The calculation method is:

$$\mathrm{stt}_i = df * stt_{i-1} + (1-df) * stt\_sample_i \tag{8}$$

Smoothing factor can select the value same with RTT. When obtaining stt sequence, changes between samples are calculated, in order to deduce the current state of network. They are expressed by three parameters as follows:

Inc_count records the length of increased cache, indicating the network is in stage of gradual saturation. dec_count recorded the length of decreased cache, indicating the network is in stage of congestion control. equal_count records the amount of data packet when the cache is 0, indicating the network is free. Parameter zero_delay is used to filter the unnecessary delay shake, can be valued as 10-3ms. In addition, shake threshold value should be set to reduce shake and filter noise.

This paper mainly adopted RED [1] queue management mechanism, selected Minth as the shake threshold of parameter in gradual saturation state. For example, in free state, it is detected that inc_count is larger than Minth, then it is believed that network starts to enter gradual saturation state, that is, B_r =inc_count.

Real-time caches B_r can be used to correct transmit speed of TFRC. When B_r fluctuates near 0, it means network is free and do not need to correct speed; when B_r increase, network begins to saturate; before packet loss event and after some data, the bigger B_r is, the higher network saturation is, the more packet loss interval need to reduce. At this moment, a correction threshold value B_{th} should be set. The correction threshold value selected in this paper is the cache length when the first data packet is abandoned. When B_r is larder than B_{th}, algorithm can adjust packet loss interval. When B_r is larger than or equal to B_{th}, then the correction is not needed. The value of f_{RealTime} proceeds according to the following equation:

$$f_{\mathrm{RealTime}} = \begin{cases} 1 & B_{\mathrm{r}} < B_{\mathrm{th}} \\ \sqrt{\dfrac{B_{\mathrm{th}}}{\mathrm{B_r}}} & B_{\mathrm{r}} \geq B_{\mathrm{th}} \end{cases} \tag{10}$$

Therefore, the Equation (10) for packet loss rate pm after improvement is as shown in Equations (2)-(11):

$$\mathrm{p}_m = \frac{1}{s_m} = \frac{1}{\hat{s} * f_{\mathrm{RealTime}}} \tag{11}$$

Detailed flow of improved TFRC is as shown in Figure 3:

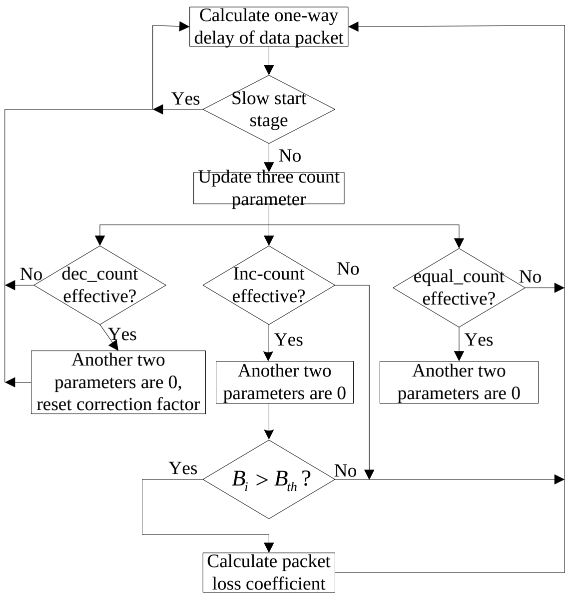

FIGURE 3 Algorithm flow figure

4 Performance analysis of algorithm

4.1 SET OF EXPERIMENTAL ENVIRONMENT

This paper selected NS-2 network simulation system as experimental environment. NS-2 is mainly used for solving problem of network research and simulates various network protocols such as TCP, routing and multicast in wireless or wire network. Network topological structure is this simulation is as shown in Figure 4:

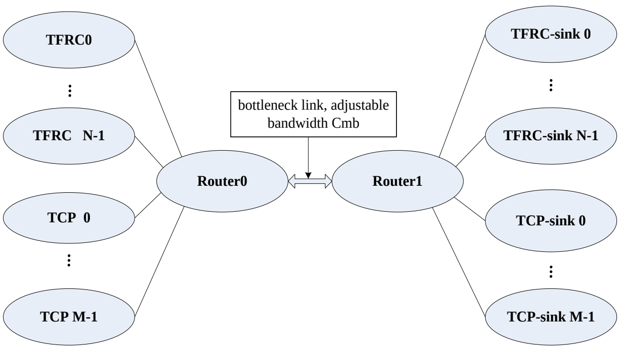

FIGURE 4 Simulation network protocol structures

R0 and R1 are two routers. Between them, it is the bottleneck of network. The delay is 15 ms. Queue scheduling adopts RED queue management. The minimum threshold of length is 5 packets and the maximum is 15 packets. Weight w is 0.002. The maximum temporary probability is 0.1. The link bandwidth between sending node and receiving node with router is 10Mb/s and the network delay is 0.

4.2 EXPERIMENTAL RESULT AND ANALYSIS

1) Analysis on fairness and throughput rate in network environment with different bottleneck bandwidth.

Set topology network according to the simulation scene of network environment with different bottleneck bandwidth. The fairness of simulation result is as shown in Figure 5 and the throughput rate of link is as shown in Figure 6. It can be found that, after the bottleneck bandwidth increases, packet loss event of TFRC data flow decreases, therefore, TFRC data flow tend to be more stable after bandwidth increasing. In addition, the improved TFRC can reduce resource occupation when congestion occurs, release more bandwidth for TCP data flow, improve the fairness; the average throughput rate of bottleneck is basically unchanged.

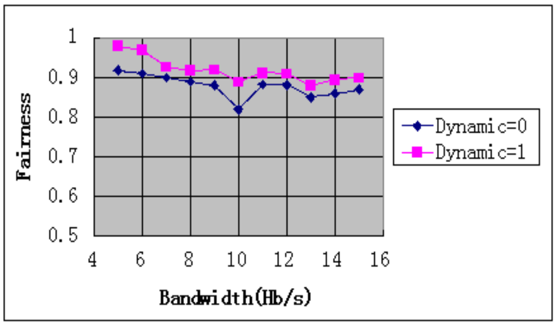

FIGURE 5 Fairness of TFRC in condition of different bandwidth

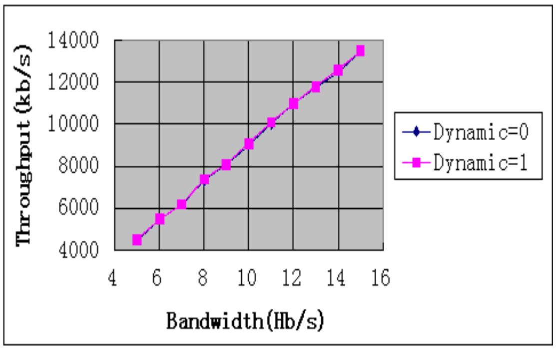

FIGURE 6 Throughput rate of TFRC in condition of different bandwidth

2) The analysis of fairness and throughput in network environment with different amount of TFRC data flow.

Set topology network according to the simulation scene of network environment with different amount of TFRC data flow. The fairness parameter of the simulation result is as shown in Figure 7 and the link throughput rate is as shown in Figure 8. It can be seen that the improved TFRC data flow cannot improve all sampling points because of the randomness of network; however, it can decrease sending rate in a high speed during network congestion, thus network resources can be released in time and its fairness also improves. With the increase of TFRC data flow, the improvement of TFRC algorithm improves fairness by sacrificing the average throughput rate of link.

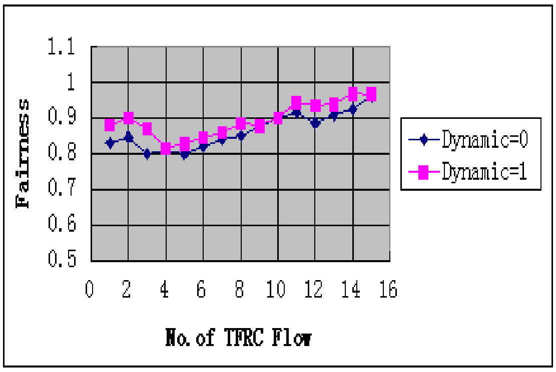

FIGURE 7 Fairness in the situation of different TFRC data flow

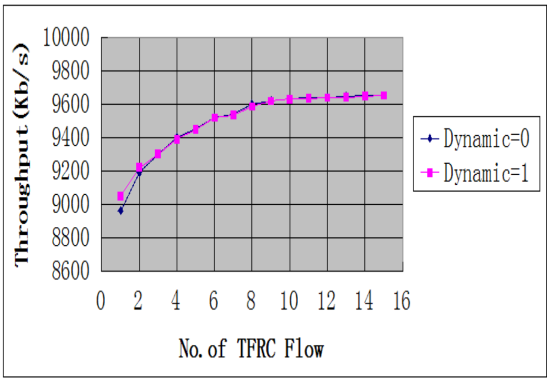

FIGURE 8 Throughput rates in the situation of different TFRC data flow

5 Conclusions

The key point of TFRC achieving good effect is to respond timely and accurately when the network occur congestion. TFRC algorithm can improve the quality of stream transmission service based on IP to some extent as well as the friendliness of TCP. This paper made improvement aiming at the insufficient fairness, introduced network parameter - real-time cache length to estimate network condition and improve the fairness of TFRC. In addition, it preceded test through network simulation platform, veryfying that the improved TFRC had higher fairness and showed good friendliness. Although real-time streaming transmission does not require the complete correction of data, proper packet loss retransmission can improve the transmission quality of multimedia. Therefore, the following research is to speculate network congestion degree according to real-time cache degree and precede retransmission of packet loss, in order to provide better transmission service for network.

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