Available online www.jocpr.com

Journal of Chemical and Pharmaceutical Research, 2014, 6(6):2286-2289

Research Article

ISSN : 0975-7384
CODEN(USA) : JCPRC5

A new feedback streaming media congestion control algorithm

Tao Kuang and Pei Tang

School of Computer and Information Engineering, Xinxiang University, Henan, China

---

ABSTRACT

Based on the feedback of streaming media congestion control algorithm (IFCA), from reduce system jitter, this paper proposes a new streaming media congestion control algorithm based on feedback. The simulation results show that the NFCA algorithm in delay jitter improved, more suitable for streaming media transmission.

Keywords: streaming media; Congestion control; New Feedback-Control Algorithm, NFCA; Delay jitter

---

INTRODUCTION

With the rapid development of Internet, streaming media applications are rapidly emerging. Network based on IP has become a streaming media distribution and transmission of the main platform. Streaming media transmission needs a certain quality of service QoS requirement. Nowadays, two main agreements on the transport layer are: TCP and UDP. TCP is the main transport protocol on the Internet, its congestion control mechanism is adopted by the Internet which plays an important role in promoting the stable. However, TCP protocol is not suitable for real-time data transmission. Due to that TCP provides reliable service, but its retransmission mechanism will attract large time delay and increase the packet delay jitter and out-of-order. Therefore, streaming media transmission are generally adopts the UDP protocol. They compete with TCP flow in unfair ways of bandwidth. The unfair situation may result in the TCP flow "starve", even lead to congestion collapse. Thus, streaming media transmission congestion control is very important, also is the key point to improve the quality of streaming media transmission service.

The real-time performance of streaming media transmission determines its real-time transport protocol based on UDP protocol RTP, and real-time transport control protocol RTCP, RTP and RTCP constituted a deal, RTCP And RTP provide flow control and congestion control services. During the period of RTP session, each participant periodically sends RTCP packets. RTCP packets contain the number of packets, the number of lost packets. Server uses this information to change the transmission rate dynamically, even change the payload type. RTCP's main function is to provide quality of the application session and broadcast performance quality information. Every RTCP packets do not encapsulate voice data or television, but encapsulate the sender and/or statistics. This information includes the sending number of packets, the number of lost packets and the packets of work etc. These feedbacks are very useful to the sender, receiver, or network administrators. RTCP specification does not specify what an application should use the feedback to do. For example, the sender can modify transmission rate according to the feedback information, the receiver can judge the problem whether it is local, regional or global according to the feedback information.

ACHIEVEMENT OF CONTROL MECHANISM

In order to adapt transmission changing in the environment, in the NFCA algorithm, system is divided into steady and unsteady. Steady state refers to that the network bandwidth is relatively stable, transmission rate has little change in the near time, image transmission rate and network bandwidth can be achieved better matching state. Rather than the steady state is in addition to the steady state of all, including the system image transmission rate is higher than the network band-width and system image transmission rate falls under the circumstance of network bandwidth. Under non steady state, the system transmission rate of the image is not good match the network band-width, we should take some measures to make the premise as soon as possible to ensure the quality of image transmission.

ROUND TRIP TIME

The interval of transmitting receiving terminal feedback is based on round trip time (RTT), which is measured by time stamp method. The first RTT calculation directly uses 2 times RTT sample, of which, RTTsample = Trcv-Tlsr-Tdlsr and is the actual RTT sample value, Trcv is the time of receiving RR package, Tlsr is the time of transmitting SR package last time and Tdlsr is the delay after transmitting SR package last time. The sender can carry out package's RTT estimation according to formula (9) based on feedback packet of receiving terminal:

$$RTT = \alpha \times RTT_{last} + (1-\alpha) \times RTT_{sample} \tag{1}$$

Here, RTTlast is the round-trip time of package last time, RTT is the round trip time of estimated package, αis weight value and plays smooth role to enhance the response of algorithm to delay, the paper selects 0.87.

TIMEOUT CLOCK VALUE

The calculation of acknowledgment timeout clock value is mainly responsible by clock control module. The timeout clock value Timeout will be calculated by formula (10) under the situation of no timeout.

$$Timeout = RTT + 4 * D \tag{2}$$

$$D = (1-\beta)D + \beta \cdot \left| RTT - M \right| \tag{3}$$

In the formula, α and β are constant, generally, α=1/8, β=1/4, M is the round trip time without timeout during recently measured transmission, D is offset, used to make RTT variation be smooth, at the same time, specify the minimum value of Timeout is 1 second to avoid unnecessary timeout.

TRANSMIT CONTROL MODULE

The sender adjusts transmission rate through window control mechanism to ensure the quality of streaming media business. The coder will transmit data to transmission control module when the sender needs to transmit data frame. The module decides whether to transmit data according to the relationship between pipe, the data frame number of sender having transmitted but not received acknowledgment at present, and current window size, namely, the maximum continuously transmitted frame number cwnd.

If pipe＜cwnd, a data frame can be transmitted and set pipe=pipe+1;

If pipe≥cwnd, this data frame can't be transmitted and will be abandoned or cached for some time according to certain rules and then transmit, depending on the requirements of specific business.

Every time when transmitting a data frame, will make timeout clock restart to timing if it fails to start, besides, if has started, the timeout clock will not need to restart timing.

TRANSMISSION CONTROL MECHANISM

The window module controls window according to feedback situations to adjust transmission rate, which is the core of achieving congestion control. The window adjustment algorithm is as follows:
It will be thanked that frame loss occurs if exceeding timeout clock value but not receiving acknowledgment information, what shows that, network congestion caused package loss. At this time, set pipe=0, the window adjustment threshold ssthreshold=3/4 * cwnd and cwnd=cwnd/2 to steadily reduce transmission rate or quickly into congestion avoidance state.

If the sender receives acknowledgment frame before the acknowledgment timeout clock, the total number r of actually received and lost frame will be calculate according to the last data frame number feedback by receiving terminal.The algorithm code will be shown as follows :

```
if (r != b)                // Package loss occurs, recover fastly
{pipe -= r;
ssthreshold=cwnd/2;
cwnd=cwnd/2;
```

```
}
else                              // r==b
{if (cwnd<= ssthreshold)      // Slow start
cwnd += 1;
else                          // Congestion avoidance
cwnd += 1/cwnd;
pipe -= b;
}
```

CONGESTION CONTROL ABILITY

In order to test bandwidth utilization and congestion control ability of TCP-like algorithm, we set two software terminals to mutually transmit video streaming with format of H261CIF as well as the highest rate of 400Kbps, and carry out comparison test with RTP/RTCP algorithm. During the experiment, we use interference program transmit package simultaneously on the test link by a certain average bit rate. The variation of transmission rate of two algorithms under interference bit stream with different bit rate is shown in Figure 6-7. Of which, the horizontal axis is the time axis (s) and vertical axis is transmission rate (Kbps). During the video transmission, the two algorithms' terminals will adjust average rate of interference program for one time every 15 minutes. The adjustment process of interference program bit rate is: 64Kbps-128Kbps-256Kbps- 128Kbps-64Kbps.

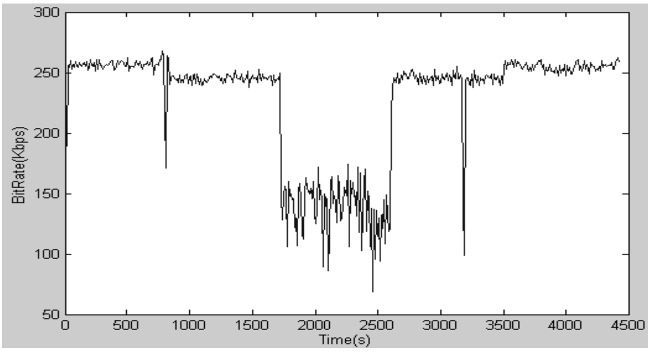

Fig. 1 Variation Process of TCP-like Algorithm Transmission Rate

In order to compare the FCA and NFCA performance of the algorithm, using the NS2 network simulation software to construct the network simulation environment. The system configuration is as follows:

bottleneck link bandwidth is 3 Mb, delay for 10 ms; beyond the bottleneck link, bandwidths on the other links is 5 Mb, delay for 5 ms.

a total time of experiment is 1200 ms, which results in the congestion of temporary flow is the flow of the FTP, opening and closing time is 180 ms and 750 ms respectively; Transmission of the RTP packet size is 1064 Byte, FTP data packets is 1040 Byte.

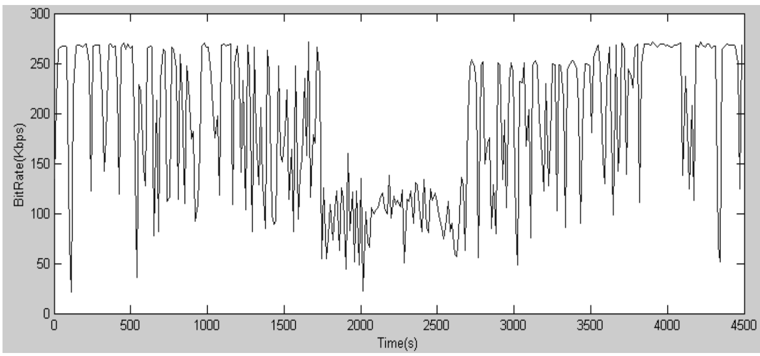

Fig. 2: Variation Process of RTP/RTCP Algorithm Transmission Rate

NFCA algorithm improved compare with the FCA algorithm in terms of average delay jitter. Average delay jitter is defined as follows: (1) the transmission time-the last time)/(the transport package serial number-the last transfer package serial number).Due to constantly adjusting the sending rate, and simulation is the drop tail queue management algorithm, When congestion occurs can produce phenomenon of packet loss, so the jitter still exist, but a slight improvement on dither amplitude

CONCLUSION

This paper puts forward an improved algorithm based on the feedback of FCA NFCA.The proposed algorithm can share with TCP flow fair link bandwidth, meet the friendly. At the same time, in view of the streaming media service quality improvement, it can make the terminal streaming playing more smoothly. In addition, the algorithm is simple, the optimization effect is obvious. Simulation experiment shows that the improved algorithm is more suitable for streaming media stream.

REFERENCES

[1]C. V. Hollot, V. Misra, D. Towsley, W. Gong. On Designing Improved Controllers for AQM Routers Supporting TCP Flows. *Proceedings of IEEE INFOCOM,* 1726-1734(**2001**).
[2]S. Athuraliya, V. H. Li, S. H. Low, Q. Yin. *IEEE Network Magazine*, 15(3): 48-53(**2001)**.
[3] S. Kunniyur, R. Srikant. *ACM Computer Communication Review*, 31(4): 123-134(**2001)**.
[4] S. H. Low, D. E. Lapsley. *IEEE/ACM Transactions on Networking*, 7(6): 861-875 (**1999).**