2015 IEEE International Conference on Computational Intelligence & Communication Technology

# A Study on Adaptive Rate Control Scheme

Shizhe Tan[1]
Department of Electronic Engineering
Ocean University of China
Qingdao, China, 266100
e-mail: tanshizhe@ouc.edu.cn

Fajiang Sun[2]
Department of Electronic Engineering
Ocean University of China
Qingdao, China, 266100
e-mail:fajiangsun@163.com

Jing Zhang[3]
Department of Electronic Engineering
Ocean University of China
Qingdao, China, 266100
e-mail:jingzhang_love@163.com

**Abstract—On the base of research on real time transport protocol and real time transport control protocol (RTP/RTCP), a flow adaptive control algorithm was proposed. The new algorithm chooses different growth factors according to the current state of the network. It aims at enhancing the stabilization of RTP flow, decreasing the jitter and utilizing network bandwidth efficiently.**

***Keywords: adaptive control algorithm, RTP/RTCP, network bandwidth***

## I. INTRODUCTION

With the improvement of digital home technologies, the digital home industry comes into real practice stage and begins to penetrate into people's lives. Through interconnection and resources sharing between devices, people can control all kinds of devices conveniently. Meanwhile, the new applications on smart terminal based on home network interconnection bring us a lot of fresh entertainment experience. Wireless transmission screen technology plays a pivotal role in the digital home. Screen transmission means that the screen display content of one terminal is instantly transmitted to another terminal to show via a wired or wireless network[1].

The complexity of network bandwidth, increases the difficulty of the multi screen interactive, the most current real-time internet services using the RTP/RTCP protocol. On the base of research on real time transport protocol and real time transport control protocol (RTP/RTCP)[2], a new flow control algorithm is proposed. This method sets four growth factors to adapt different network conditions.

## II. ADAPTIVE VIDEO STREAMING MECHANISM

Traditional transport layer protocols are not suitable for transmission of streaming media. TCP protocol is a connection-oriented transport protocol, provided reliable, error-checked and ordered transmission. Network based real-time multimedia transmission always shows characters of high bandwidth, real time, burst and so on. Using TCP to transmit real-time data and carry on congestion control, which can introduce high jitter and cause real-time issues, will severely affect service quality. Thus, TCP is not the appropriate transport protocol for real-time multimedia application. UDP protocol[3][4] provides connectionless, unreliable datagram service. UDP Protocols and TCP Protocols sharing the network bandwidth, because UDP Protocols have no congestion control mechanism and access to the most of the bandwidth. Therefore, UDP is not the appropriate transport protocol for real-time multimedia application.

The IETF(Internet Engineering Task Force) has devised several protocols support streaming media transmission. The Real-time Transport Protocol (RTP)[5][6] defines a standardized packet format for delivering audio and video over IP networks. RTP is used extensively in communication and entertainment systems that involve streaming media, such as telephony, video teleconference applications, television services and web-based push-to-talk features[7]. RTP is used in conjunction with the RTP Control Protocol (RTCP). While RTP carries the media streams (e.g., audio and video), RTCP[8] is used to monitor transmission statistics and quality of service (QoS) and aids synchronization of multiple streams.

In this paper, the transmission control scheme based on RTP / RTCP as shown in Fig1.

978-1-4799-6023-1/15 $31.00 
DOI 10.1109/CICT.2015.124

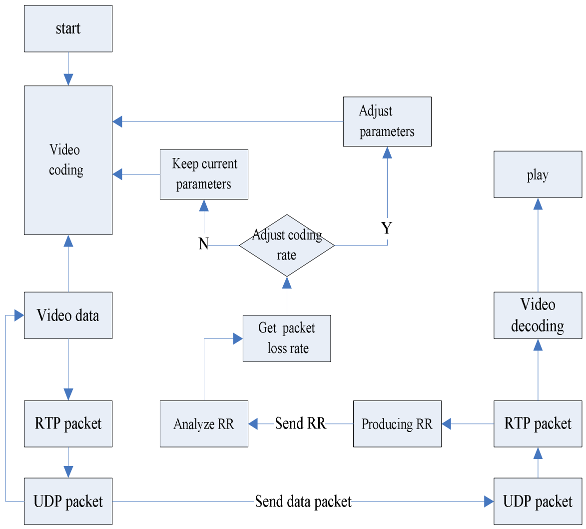


Figure1. Transmission model

The encoded data is packaged by RTP. The RTP data packet is transmitted to receiver by UDP. The receiver resolve video data from RTP packets and then send it to video decoder. There is another important task for the receiver that is computing the packet loss rate and producing receiver report periodically. The receiver send (Receiver Report)RR to the sender periodically. The sender can make the packet loss rate from the RR and adjust to the encoder's coding rate according it.

## III. ADAPTIVE FLOW CONTROL ALGORITHM FOR RTP/RTCP

RTP smooth adaptive algorithms[2] generally expressed as a (1)

$$x(t+1)=\begin{cases} x(\mathrm{t})+\mathrm{a} & r_{loss}\le TH_{loss} \\ b_D x(\mathrm{t}) & r_{loss}>TH_{loss} \end{cases} \quad (1)$$

Equation (1) is a general formula. However, in the RTP [9][10]stable adaptive algorithm, a is no longer constant, but variable, the more time near the point of congestion, the smaller the value of a. a is a growth factor, it has 4 kinds of value according to different circumstances. $\mathrm{X}_{cong}$ is the last velocity of the RTP stream when congestion occurs . The threshold of Packet loss rate is $TH_{loss}=0.04$ . $b_D=0.8$ is a coefficient. $TH_{cong}=0.1$ is a relative value, which reflects whether the current speed is close to the threshold( $\mathrm{X}_{cong}$ )of the last congestion. $\mathrm{X}_h$ is the maximum transmission speed under ideal bandwidth. $x(\mathrm{t})$ is the rate of RTP flow at the time t.

$$a_I=(1-\exp^{-0.5\times(1-\frac{x(\mathrm{t})}{R})})\times(\mathrm{X}_{cong}-\mathrm{x(t)}) \quad (2)$$

When the network is congested, RTP flow will decelerate and it is reflected in (3).

$$x(\mathrm{t}+1)=b_D x(\mathrm{t}) \quad (3)$$

At the same time, record the current congestion rate.

$$\mathrm{X}_{cong}=x(\mathrm{t})$$

If there is no congestion occurs, changes according to the following logic:

If the network congestion had not occurred, that RTP stream for the first time into the network the speed increases according to (4).

$$a=r_{add} \quad (4)$$

Then the RTP stream increases according to the $r_{add}$ , $r_{add}$ is a constant( $r_{add}=100k$ ).

If network congestion had occurred, then this is the following logic:

If $X_{cong}-x(\mathrm{t})<TH_{cong}\times X_{cong}$ (close enough to the last congestion points) and there is no congestion occurs.

$$a=a_I=(1-\exp^{-0.5\times(1-\frac{x(\mathrm{t})}{R})})\times(\mathrm{X}_{cong}-\mathrm{x(t)}) \quad (5)$$

else if $X_{cong}-x(\mathrm{t})\ge TH_{cong}\times X_{cong}$ and there is no congestion occurs, the increases of the speed according to (4).

There are two major drawbacks about this algorithm:

The first, through the analysis we can see that $(1-\exp^{-0.5\times(1-\frac{x(\mathrm{t})}{R})})\le 1$ , then $x(\mathrm{t}+1)=\mathrm{x(t)}+\mathrm{a}_I\le x(\mathrm{t})+(\mathrm{X}_{cong}-\mathrm{x(t)})\le X_{cong}$ . As it can be seen, even in the absence of congestion the current rate can not exceed the last point of congestion.

The second, in the case of no congestion, the increases of the speed according to (4) $x(\mathrm{t})>X_{cong}$ is possible, at this time $a_I=(1-\exp^{-0.5\times(1-\frac{x(\mathrm{t})}{R})})\times(\mathrm{X}_{cong}-\mathrm{x(t)})<0$ , the current rate is still reducing, although there is no congestion.

## IV. THE IMPROVED ALGORITHM

I. This paper focuses on these two points make improvements.

If network congestion had occurred, then this is the following logic:

If $X_{cong} - x(t) < TH_{cong} \times X_{cong}$ (Sufficiently close to or exceed the previous congestion points) and there is no congestion occurs, at this point there are two cases ($X_{cong} - x(t) > 0$ and $X_{cong} - x(t) < 0$);

The first is $X_{cong} - x(t) > 0$, in order to exceed the last point of congestion, we set the other threshold $b_D X_h$.

$$a_{II} = (1 - \exp^{-0.5\times(1-\frac{x(t)}{R})}) \times (X_h - x(t)) \qquad (6)$$

$$a = \begin{cases} a_I & X_{cong} - x(t) > 0, x(t) > b_D X_h \\ a_{II} & X_{cong} - x(t) > 0, x(t) \le b_D X_h \end{cases} \qquad (7)$$

The second, when $X_{cong} - x(t) < 0$, the current speed $x(t)$ exceeds the previous threshold $X_{cong}$, if the speed $x(t)$ does not exceed the desired bandwidth rate, the speed will slow down, at this time to ensure continue to increase, a is set to $a_{III}$,

$$a = a_{III} = \frac{1}{2}|a_I| \qquad (8)$$

else if $X_{cong} - x(t) \ge TH_{cong} \times X_{cong}$ and there is no congestion occurs, the increases of the speed according to (4).

The algorithm flow is shown in Fig. 2.

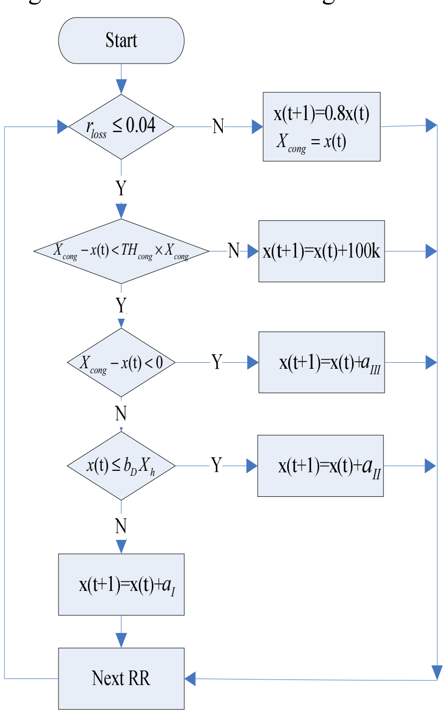


Figure2. The new algorithm flow

## V. Simulation Results

The network environment we established, there are 3 transmit terminal and 3 receive terminal. They were connected by two routers and the bandwidth is 2M. It was shown in Fig. 3

The algorithm flow is shown in Fig4. We can clearly see the current transmission rate and packet loss rate.

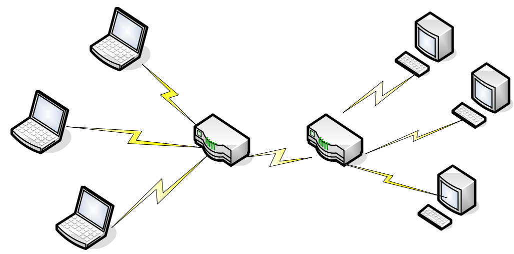

Figure3. The network environment

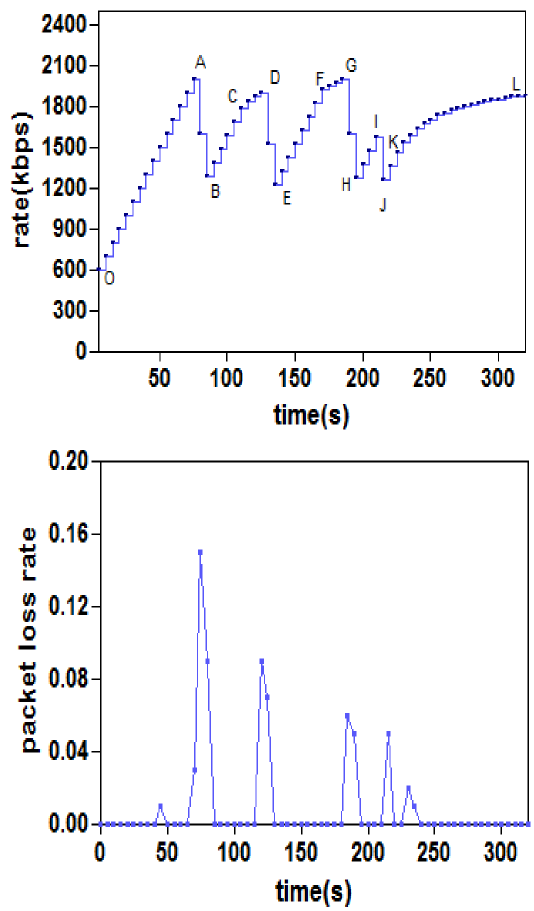


Figure4. Performance testing under the new algorithm

1)To test the improved algorithm

For [O-A], the speed increases according to the (4). At 45ms, packet loss rate is 0.01, less than the threshold（0.04）of packet loss rate，and therefore continue to increase. When reached A, the current speed exceeds the speed under ideal bandwidth, the speed reduces according to the (3) , the (3) is performed twice, after the first， the loss rate dropped to 0.09, greater than the threshold value（0.04），continues to decrease.

For [B-C], the speed increases according to the (4).

For[C-D], $X_{cong} - x(t) < TH_{cong} \times X_{cong}$ and $X_{cong} - x(t) > 0$, the speed increases according to the (5). When reached D, the packet loss rate is 0.09, so to reduce the speed according to the (3).

For [E-F], the speed increases according to the (4).

For[F-G], $X_{avg} - x(t) < TH_{avg} \times X_{avg}$ and $X_{avg} - x(t) < 0$, the current speed $x(t) = 1916$ exceeds the last threshold D, and packet loss rate is 0, then the speed increased to G according to the (8) .

For [J-K], the speed up to the last congestion critical point I according to the (4), the packet loss rate is 0, $x(t) \le b_D X_h$, this time the (6) is executed to grow to K, breakthrough last congestion points.

For [K-L], the speed increases according to the (8).

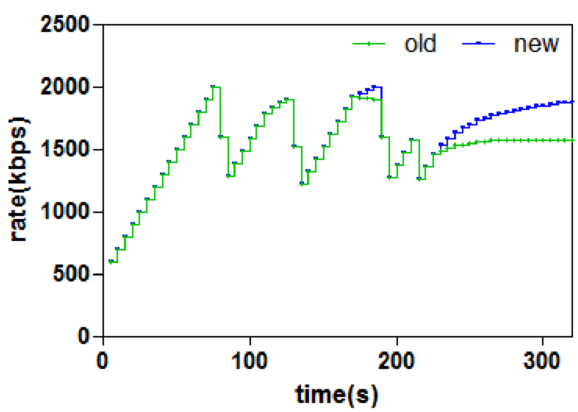


Figure5. Performance comparison.

As it can be seen from Fig5, the improved algorithm significantly improves the utilization of the channel, enhances the stabilization of RTP flow.

## VI. CONCLUSION

Wireless transmission technology plays an irreplaceable role in digital home. This paper proposed a flow control algorithm based on RTP/RTCP to control it. We set four growth factors to meet the different environment. Some experiments have been done to verify the new algorithm. In the future, it is essential to add more growth factors in order to meet the complex network environment.

## REFERENCES

[1] J. Yu, C. Liu, Y. He, S. Hu, "Adaptive video transmission using motion intensity," Journal of Computer-Aided Design and Computer Graphics, vol.21, no. 6, pp.847-852, June 2009 .

[2] Cheng Wanxiang, Zhang Yibo, Lei Zhenming," An Modified RTP Adaptive Algorithm," High Technology Letters, pp. 33 –38,2002

[3] Y. H. Gu and R. L. Grossman, "UDT: UDP-based data transfer for high speed wide area networks," Computer Networks, vol. 51, pp. 1777-1799, 2007.

[4] B. Eckart, X. B. He and Q. S. Wu, "Performance adaptive UDP for high speed bulk data transfer over dedicated links," in Proceedings of IEEE International Symposium on Praallel & Distributed Processing, pp.1511-1520, 2008.

[5] M. Wagner, I. Kopilovic, S. Dohla, N. Farber, and H. Fuchs, "Towards an rtp profile for iptv," Broadcasting, IEEE Transactions on, vol. 55, no. 2, pp. 343 –352, 2009.

[6] Uras Tos, and Tolga Ayav, "Adaptive RTP Rate Control Method," Computer Software and Applications Conf. Workshops (COMPSACW), 2011 IEEE 35th Annu. Munich, Jul. 2011, pp. 7-12.

[7] I. Kofler, R. Kuschnig, and H. Hellwagner, "In-network Real-time Adaptation of Scalable Video Content on a WiFi Router", ConsumerCommunication and Networking Conf., CCNC 2009, 6th IEEE, LasVegas, NV, Jan. 2009, pp. 1-2.

[8] R. Venkatesha Prasad, Richard Hurni, H S Jamadagni," A Scalable Distributed VoIP Conferencing Using RTCP," Proceedings of the Eighrh IEEE International Symposium on Computers and Communication, vol.1, pp.608-613, 2007.

[9] C. Perkins and T. Schierl: "Rapid Synchronisation of RTP Flows," Internet Draft, draft-ietf-avt-rapid-rtp-sync-09, Internet Engineering Task Force, January, 2010.

[10] B. VerSteeg, A. Begen, T. VanCaenegem and Z. Vax:"Unicast-Based Rapid Acquisition of Multicast RTP Sessions," Internet Draft, draft-ietf-avt-rapid-acquisition-for-rtp-08, Internet Engineering Task Force, March, 2010.