

The Comparison And Analysis Of The Streaming Media Transport Protocol In The Transmission System

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Abstract. In order to solve the the problem of poor video quality in the transmission process of the streaming media which is caused by transmission delay and packet loss, this paper analyzes and compares the streaming media transport protocols from the aspect of the streaming media transmission protocols. The comparative results show that: RTP protocol based on UDP can effectively improve the performance of streaming media's transmission.

Keywords: streaming media; protocol; transmission system;performance

1. Introduction

The rapid development and popularity of Internet provides the strong market momentum for the development of the streaming media business, the streaming media business is becoming increasingly popular. Streaming media technology is widely used in multimedia news publishing, live broadcast online, advertising online, e-commerce, video on demand, remote education, telemedicine, Internet radio, real-time video conference and other aspects of the Internet Information Services. The application of the streaming media technology will bring great changes on the network information exchange and have a profound impact on the people's work and life.

At present, protocols which support for streaming media transmission are TCP / UDP (Transmission Control Protocol / User Datagram Protocol); RTP (Real-Time Transport Protocol) / RTCP (Real-Time Transport Control Protocol). In practice, UDP / RTP is used for audio / video media, and TCP / RTCP is used for the transmission of data and control signaling. The graph of streaming media transport protocol is shown in Figure 1:

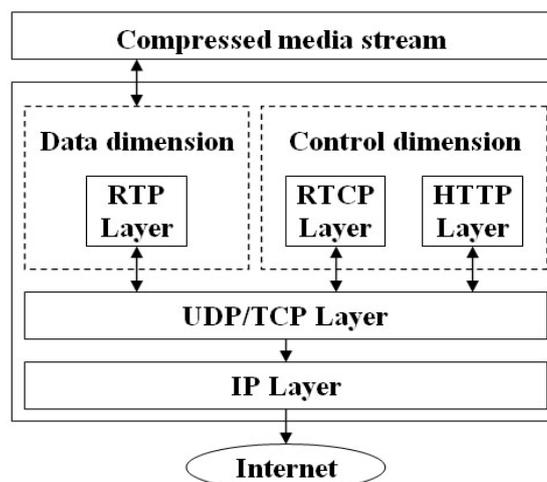


Figure 1. The graph of streaming media transport protocol

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2. Analysis Of TCP/UDP Protocol

There are TCP protocol and UDP protocol in the transport layer. TCP is a stream-based protocol, it provides the effective error control and congestion control algorithms, the users need not to consider the packet loss and the order of data arrival. UDP protocol is based on block transmission, it does not guarantee the loss of data, error, and the order of data arrival, at the same time, it has no congestion control algorithm.

UDP protocol is more suitable for using in the streaming media broadcast system. UDP protocol has the following advantages:

(1) UDP is oriented to connectionless, there is no TCP three-way handshake process, the speed is fast, and the cost of UDP session state maintenance is lower;

(2) Multimedia error only affects the quality of viewing, there is no need to retransmit after the error occurs, and TCP errors retransmission is not controlled;

(3) Under the condition of the instability in the network, TCP is easy to produce large fluctuations in the rate, which is determined by TCP's congestion control algorithm. When the packets have no response for a long time, TCP will reduce the rate, then transmission delay will increase;

(4) UDP can perform NAT traversal, while TCP is more difficult;

(5) UDP is more easier to control the size of the subcontract than TCP, because TCP is a stream-oriented communication protocol, there is no concept of packets at the application layer, and the size of the bottom's subcontract is controlled by the operating system.

3. Analysis Of RTP/RTCP Protocol

RTP is a transport protocol for the multimedia data stream on the Internet, it is released as RFC1889/3550 by the IETF (Internet Engineering Task Force). RTP a protocol which provides end-to-end transmission services for real-time data, and it has been widely used in Internet multimedia data communications. RTP typically uses UDP to transfer data, and can carry over on the other underlying protocol to provide one to one or one to many working mode. If the underlying network supports multicast technology, then can use the RTP protocol to provide multi-purpose address transmission. RTP protocol itself does not provide mechanism which assures real-time and service quality, and does not assure that the message transfers in order, the receiver can use the SN (Sequence Number) field in the message to restructure the message according to need.

Each RTP packet consists of a header and the variable length of media data, the first 12 bytes of its RTP packet's head are fixed, as is shown in Figure 2.

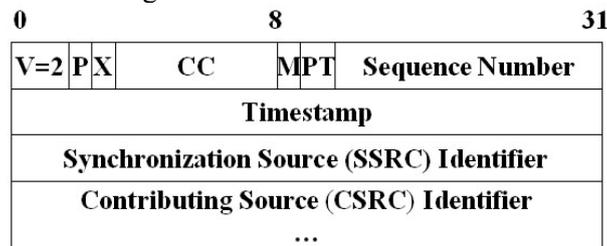


Figure 2. The structure of the RTP packet's header

RTP itself can not provide reliable transport mechanism for transmitting packets in order, and also does not provide flow control or congestion control, it relies on Combining with RTCP (RTP Control Protocol) to accomplish the control service. In the process of RTP session, each participant interacts with RTCP packets periodically, RTCP packets contain statistical data which include the quantity of packets which have been sent, and the quantity of loss packet, the session side can use this information to dynamically change the transmission rate, and even change effective load's type. RTP is combined with RTCP, can optimize the transmission efficiency with the effective feedback and minimal overhead, so it is particularly suitable for transmitting the real-time data on the Internet.

RTCP defines several kinds of packets' types which is used to carry a variety of control information. Each RTCP packet is similar with RTP packets, both all begin with a fixed header, and followed by a structural element which changes with the change of the packets' type. This element may have the variable length, but always ends in 32-bit boundary.

RTP/RTCP protocol has the following advantages:

(1) The Protocols are simple: RTP Protocol can be hosted above many kinds of transport layer protocols, the protocol itself does not support resource reservation and does not provide the mechanism of quality assurance;

- (2) Good extension: RTP can build on top of UDP, and make full use multiplexing service of UDP protocol, can extend the new type of load by defining the new Profile;
- (3) Data flow is separated from control flow: RTP / RTCP use a different port to control, which greatly improves the protocol's flexibility and independence.

The architecture of using RTP / RTCP to transmit video stream is shown in Figure 3.

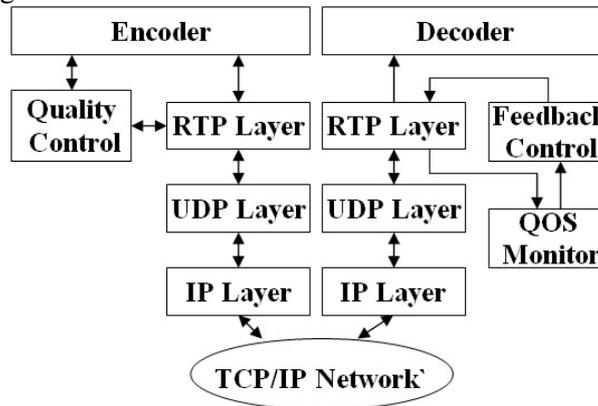


Figure 3. The architecture of using RTP / RTCP to transmit video stream

The previous MixCast System uses TCP protocol to transfer data. Due to its delay, it can not well support real-time playback, so use the RTP protocol based on UDP to transfer the streaming media data. RTP and UDP complete the functions of transport layer together. RTP packets are made up of RTP header and the load, one or more of the RTP can be transmitted in a UDP packet. RTCP packets are also made up of the header and a number of prescriptive data units, the content and format of data units varies according to need, usually a UDP packet can store multiple RTCP, thus can improve the bandwidth utilization. Two endpoints must negotiate the destination port number of receiving the RTP / RTCP packets well before using RTP to transmit media stream, which can make the RTP / RTCP data be properly sent. RTP data send to the even number of UDP ports, and the corresponding RTCP control signal's data is sent to the adjacent odd number of UDP ports (the number even UDP port +1), thus constitute a UDP-side pair. Both sending and receiving ends all use their own UDP port pair to receive RTP data and RTCP data.

4. The Comparison And Analysis Of The Streaming Media Transport Protocol In The Transmission System

The following are the comparison and analysis of RTP protocol and TCP protocol's specific performance in streaming media transmission system:

(1) Test Environment

AS is shown in Figure 4, in experiments, use 30 personal computers which are located in two local area network to test. The configuration of each PC is P43.0/1G/100M (CPU, memory, network card). Data source is deployed in one of the LANS, the structure of data transmission is network structure, bit rate is 400kbps, the output is 320 * 240, the system runs 10 hours.

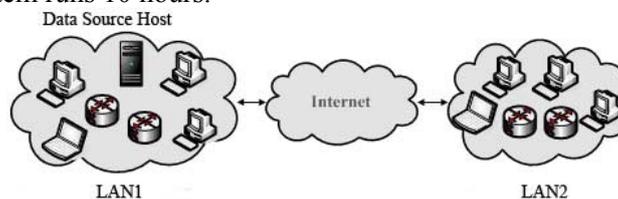


Figure 4. The testing topological graph

(2) The comparison of performance

Data integrity: TCP uses three-way handshake, and there is no packet loss problem. RTP underlying layer uses UDP protocol, UDP provides the service of unreliable transmission, and is easy to occur packet loss, which leads to poor video quality. In the test system, use PUSH / PULL mode, which reduces the packet loss rate. The experimental results (Figure 5, Table 1) prove that RTP data integrity maintains at 97% -99%, and the proper amount of packet loss has little effect on the playback quality.

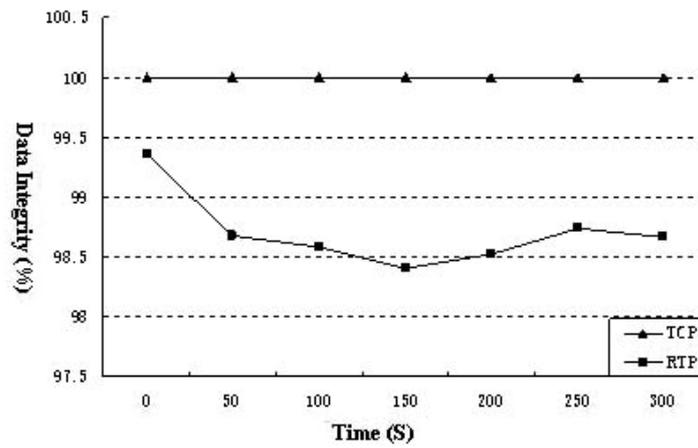


Figure 5. The comparison of RTP and TCP's data integrity

Table 1 The comparison of data integrity of RTP and TCP

Time(S)	325	350	375	400	425	450	475	500	525	550	575	600
TCP data integrity (%)	100	100	100	100	100	100	100	100	100	100	100	100
RTP data integrity (%)	98.7	98.5	98.2	98	97.8	97.72	97.91	97.8	97.66	97.48	97.5	97.8

The average transmission delay: Transmission delay is the data flow's transmission time from data source to the receiver. Compared with the tree network structure, the network structure greatly reduces the transmission delay. The experimental results show (Figure 6, Table 2): The TCP transmission delay is between 500ms and 620ms, The RTP transmission delay is between 20ms and 70ms. Therefore, using RTP protocol to transmit can effectively reduce the transmission delay, and improve the broadcast efficiency.

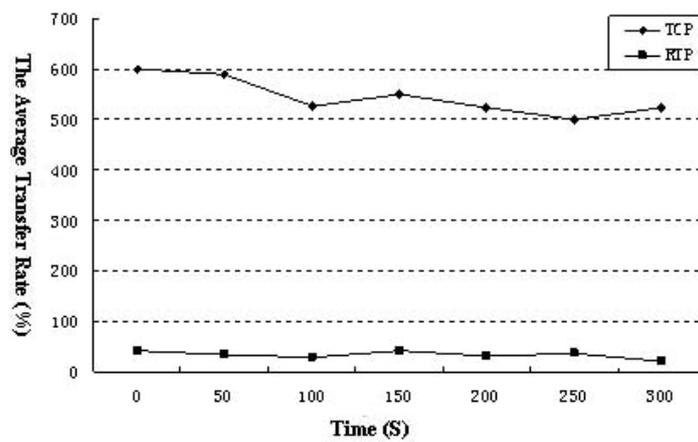


Figure 6. The comparison of RTP and TCP's transmission delay

Table 2 The comparison of RTP and TCP's transmission delay

Time(S)	325	350	375	400	425	450	475	500	525	550	575	600
TCP transmission delay (ms)	502	505	510	506	501	520	521	534	550	541	551	530
RTP transmission delay (ms)	23	28	29	23	25	28	27	25	32	40	41	38

Control load: Control load refers to the rate of the number of control information:the total number of information in the data transmission process. When using TCP to transfer, because the TCP uses three-way handshake, then the ACK, PSH-ACK packets, etc, and the control message of the interaction between nodes are much more. Nevertheless, when using RTP protocol, regularly interact a number of control packets between the nodes. When the initial connection begins, need TCP to interact more information, after the transmission is stable, will send 3-4 control messages while transmitting 60-70 RTP data. As is shown in Figure 7, Table 3, the control load of TCP is between 34% -50%. When using RTP to transmit, there is more interactive control information in the former 2-3 seconds, stable control load is between 10.2% -16%. Therefore, when use RTP to transmit, the control load is much less, and effectively reduces the control load of the nodes.

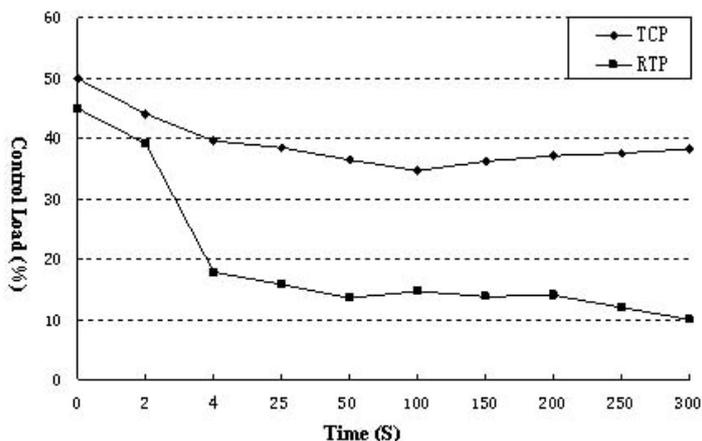


Figure 7. The comparison of RTP and TCP’s control load

Table 3 The comparison of RTP and TCP’s control load

Time(S)	325	350	375	400	425	450	475	500	525	550	575	600
TCP control load (%)	37.4	36.9	36.4	35.6	35.1	36.4	36.1	37.5	37.9	36.1	34.4	34
RTP control load (%)	11.8	13.4	15.3	14	15.6	13.6	13.9	13.8	12.4	14.7	13.6	13.1

5. Concluding Remarks

This paper first analyzes TCP / UDP protocols and RTP / RTCP protocols, then compares and analyzes the performance of TCP protocol and RTP protocol in the actual system, and the experimental results show that UDP-based RTP protocol can effectively reduce the delay and the control load, to achieve better streaming media services.

6. References

- [1] Xinyan Zhang, JC Liu, Bo Li, and Tak-Shing Peter Ynm. CoolStreaming/ DOnet: A data-driven overlay network for efficient live media streaming [C]. In Proceedings of IEEE INFOCOM, pp.121-134, Feb, 2010.
- [2] Liao X, Jin H, Liu Y, Ni L, Deng D. Anysee: Peer-to-Peer Live Streaming[C]. IEEE INFOCOM, pp.253-264, Jun, 2010.
- [3] R. Frederick, Blue Coat Systems Inc., V. Jacobson. RTP: A Transport Protocol for Real-Time Applications[S].RFC3550, IETF, pp.141-121, Aug, 2010.
- [4] Li Fan, Pei Cao, Jussara Almerida. Summary cache: A scalable wide-area web cache sharing protocol [J].IEEE/ACM Trans Networkin, pp.281-293, Apr, 2010.
- [5] Charu Aggarwal, Joel L.Wolf, and Philip S.Yu. Caching on the World Wide Web [J].IEEE Transactions on Knowledge and Data Engineering, pp. 94-107, Aug, 2010.
- [6] R.Wooster and M.Abrams. Proxy Caching the Estimates Page Load Delays [C].In the 6th International World Wide Web Conference, April 7-11, 1997, Santa Clara, CA, pp.977-986, May, 2010.