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Performance improvement of streaming media QoS based on behavior feature and content analysis

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Abstract. In order to improve the performance of streaming media QoS, an enhanced QoS-supported streaming media service system model based on behavioral characteristics and content analysis. By adding the media filter module and the feedback processing module, the improved system model can support different characteristics of the heterogeneous multimedia terminal, such as resolution, color display attributes and so on, and then realize the QoS support of the streaming media service. Because the traditional Internet network was originally designed for ordinary data traffic transmission, its best effort (best effort) features cannot provide real-time multimedia data transmission for the corresponding quality of service (QoS) guarantee. In order to improve QoS support for network streaming media applications, a streaming media service system model with enhanced QoS support is proposed based on the traditional streaming media application system model. By adding the media filter module and the feedback processing module, the improved system model can support different characteristics of heterogeneous multimedia terminals, such as resolution, color, display, attributes and so on. And then the support of streaming media service QoS is realized. The necessity and feasibility of the system model are demonstrated by setting up an experimental platform, designing test cases and analyzing experimental results. The performance of the core module algorithm in the system model is verified. The results show that the streaming media system model with enhanced QoS support can effectively improve the performance of streaming media QoS. For different broadcast terminals and their QoS requirements, QoS support for network streaming media applications has been effectively improved.

Key words. Streaming media, QoS, media filters.

1. Introduction

As streaming media transmission compared with traditional data transmission has its own characteristics, so the streaming media transmission has a special request to the network QoS (Quality of Service) compared to the traditional data transmission. Streaming media transmission on the network delay and delay jitter have more stringent requirements, end to end transmission delay and delay jitter

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in the transmission will affect the streaming media applications. Streaming media transmission on the packet loss rate has a higher requirement. Due to the inconsistency of the importance of streaming media data, the loss of a small amount of critical data can cause a sudden drop in service quality, and continuous packet loss will make the quality of service completely unacceptable. Streaming media transmission has a higher bandwidth requirement, requires adequate bandwidth and has a smooth throughput. Bandwidth, delay and packet loss rate have a great impact on the quality of streaming media service, but because of the traditional Internet transmission characteristics, it cannot provide the corresponding QOS guarantee [1]. Therefore, streaming media applications on the Internet often play pause, audio and video images are not synchronized or mosaic and other phenomena, seriously affect the streaming media application quality of service. To expand the application of streaming media in various fields, it is necessary to provide a corresponding guarantee mechanism for the QoS of the media, so as to improve the service quality of streaming media applications.

Compared with the real-time characteristics of streaming media, the traditional TCP protocol needs more overhead and is not suitable for streaming media transmission. Therefore, HTTP/TCP is used to transmit control information, and RTP/UDP is used to transmit real-time sound data [2]. Unfortunately, UDP does not have the TCP congestion control mechanism. Sally Floyd proposes an equal congestion control mechanism for unicast traffic. Internet The existing "best effort" traffic in Internet uses the traditional TCP protocol. In contrast to the streaming media, a TCP friendly (TCP-friendly) congestion control mechanism needs to be found to avoid the mechanism of using TCP to reduce the speed of packet loss when congestion is generated. For the same problem, some researchers have proposed a new congestion control algorithm and give a performance simulation. By analyzing the shortcomings of the Padhye'S TCP throughput model, a new congestion control mechanism for streaming media based on the dynamic TCP throughput model is proposed [3]. It can guarantee the TCP friendliness in the dynamic environment, and has good response to the change of the network status. In order to improve the QoS support for the application of network streaming media, the current research work is mainly focused on the following two aspects:

1. From the physical / network level to improve the service performance of network QoS, including the network equipment, network-protocol and network infrastructure transformation, the streaming media QoS needs mapping to the underlying network, thereby enhancing network support for streaming media applications QoS.

2. From the application layer to improve the service quality of streaming media application, the service quality optimization strategy is adopted in the server side and the client side, so as to improve the adaptability of the streaming media application to the network environment.

Based on the traditional streaming media application system model, this paper presents an enhanced QoS-supported streaming media service system model. By adding the media filter module and the feedback processing module, the improved system model can support different characteristics of the heterogeneous multimedia terminal, such as resolution, color display attributes and so on, and then realize the

QOS support of the streaming media service.

2. Methods

2.1. Enhanced QoS supported streaming media system model

In order to increase the support of streaming media service QoS, this paper proposes an improved streaming media service system model, in which the media filter module is added in the existing streaming media system model to support the different characteristics of different multimedia terminals, such as resolution, color display attributes, etc., and increase the player feedback mechanism to enhance QoS support, as shown in Fig. 1.

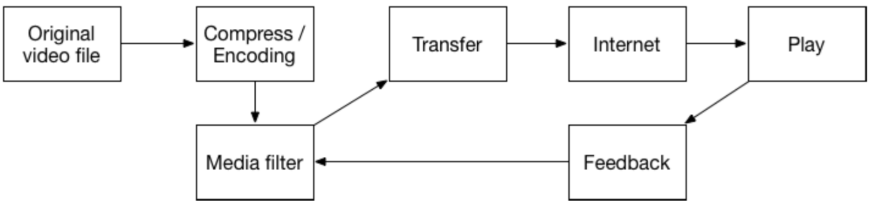

Fig. 1. Enhanced QoS supported streaming media model

2.2. Design of core algorithm for media filter

The main function of the media filter is to process the compressed data according to the feedback information (resolution, color display attribute, etc.) sent by the player, so as to adjust the transmission rate according to the different QoS requirements of the broadcasting end, in meeting the requirements of the playback side QoS is more conducive to network transmission [4]. As an example, for large screen resolution of the player to receive the data transmission rate faster than the screen resolution of the smaller player. In order to achieve this effect, the transmission rate can be increased by adjusting the priority of the transmitted data stream. To achieve the different QoS requirements for the player to adjust the sending rate of data timely, we can consider the congestion control mechanism applied to the streaming media application based on the implementation. Congestion control mechanism should be used to ensure smooth delivery rate (to meet the requirements of streaming media applications), but also for different data streams for different sending priority adjustment [5].

First, the feedback information processing module collects the feedback information from the broadcasting end, maps the feedback information to the sending priority of the transmission data according to the specific algorithm, and then the media filter module determines the transmission data priority according to the feedback information processing module, calculates the transmit data rate that is appropriate for the current playback QoS requirements [6]. Specifically, the media filter takes the following algorithm:

The media filter uses the same mechanism as the TFRC to detect the performance parameters of the current network, including the probability P_{e} of the lost event, the number of packets b received by an ACK, the average round-trip time RTT, the packet retransmission time T, and packet loss probability P_{r}.

According to the feedback information processing module, the data transmission priority n can be obtained, and the data transmission rate is calculated according to the following algorithm.

In the TFRC protocol, n TCP data stream average transmission rate is calculated as follows (unit: bit/s)

$$X_{\mathrm{Bps}} = \frac{S}{R\sqrt{2bp/3} + t_{\mathrm{RTO}} 3\sqrt{3bp/8}\, p(1+32p^2)}, \tag{1}$$

where S is the size (in bits) of each data fragment, R is the average round trip time in seconds, b is the maximum number of packets in a TCPACK advertisement, p is the loss event rate, whose value lies between 0–1, and t_{RTO} is the time of retransmission of TCP timeout.

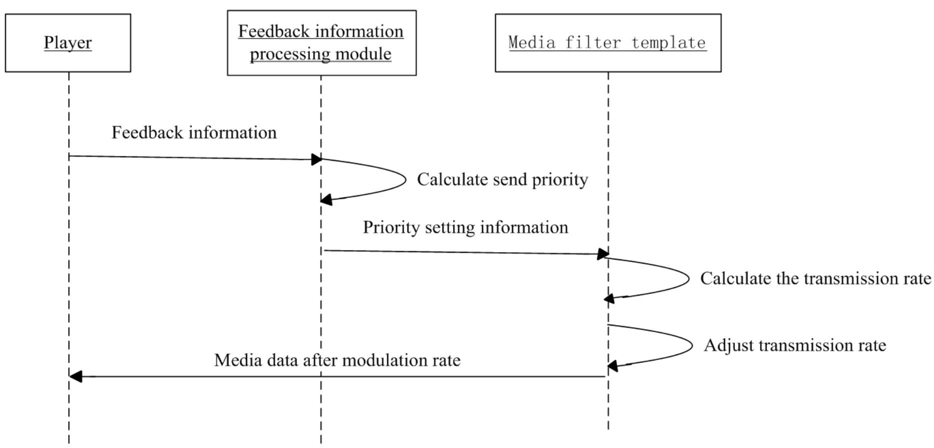

Fig. 2. Collaboration diagram of media filters with other modules

As shown in Fig. 2, the player sends the feedback information (video file resolution requested by the player) to the feedback information processing module, the feedback information processing module determines the priority of the transmission data according to the mapping algorithm and transmits the priority setting information to the media filter module. The media filter module runs the transmission rate algorithm to start the data transmission according to the adjusted transmission rate.

3. Results

From the system model shown in Fig. 1, it can be seen that the media filter module is the key function module in the enhanced streaming media system model supported by QoS. Therefore, to verify the performance of the model, the key is to verify the media filter module for streaming media performance improvement contribution. In order to verify the value of the above system model, this paper builds an experimental test platform, and uses a self-developed tool with data transfer function, which implements MulTFRC protocol implementation code. The MulTFRC protocol was used to provide different transmission priorities for different transmit data streams compared to other congestion control protocols such as DCCP and TFRC [7]. Its priority is set by setting the aggression value of the data flow to achieve. In the transport layer is still using the UDP protocol implementation code, the relevant MulTFRC protocol control code is added to the tool's application layer code. After each data transmission, the receiver will calculate the transmission time; receive the packet size and other information. In addition, the sender can set the sending data of the aggression value, that is, set the parameter n value, it indicating that the transmitted data stream will simulate the performance of n TCP data streams, the default value of the parameter n is 1, representing a standard TCP data stream.

3.1. Test Scenario A - Verify the tunability of the media filter transmission rate

This experimental scenario mainly validates the tunability of the media filter module transmission rate in the enhanced streaming media system model supported by QoS. As the core module in the model, the function of the media filter is: when the media filter receives the feedback information from the player, it finds that the media quality of the player is high (such as the screen resolution is high, or high quality). Now it is possible to adjust the priority of the transmission data stream accordingly. By increasing the priority of the transmission, the high priority data stream obtains a higher throughput than the lower priority data stream, thereby satisfying the QoS requirement of the broadcasting end [8]. On the contrary, when the media filter module receives the player feedback information, if the media quality of the player is low, the media filter adjusts the priority of the transmission data to a lower value to make it more effective. Reasonable is here use of network bandwidth. In order to test the above performance, the experiment needs to test the performance of the data flow with different sending priority. The priority is set by the aggression value (n value in MulTFRC protocol). Table 1 and Fig. 3 show some important experimental data.

3.2. Test Scenario B - Verify the friendliness of the media filter

This experimental scenario mainly validates the TCP-friendliness of the media filter in the enhanced streaming media system model supported by QoS, i.e. whether TCP-friendly transmission data flow and TCP traffic adjusted by media filter are

TCP-friendly. In the enhanced QoS-supported streaming media system model, the media-adjusted transmitted data stream requires both a smooth delivery rate and a TCP-friendly nature. In order to study the TCP-friendly (competing with each other) between data streams with different transmission data priorities, the experiment uses a stream with an aggression value of 1 as the background traffic, other experimental scenario settings are the same as for test scenario A. Table 2 and Fig. 4 show some experimental data.

Table 1. Test Scenario A

File size (Mb)	Aggression value	Transmission time (s)	Average transmission rate (kb/s)
12.8	1	24.2	529
12.8	2	18.2	703
12.8	3	17.8	719
12.8	4	16.6	771
12.8	5	16.3	785
12.8	6	15.8	810
12.8	7	15.5	821
12.8	8	15.3	833
12.8	9	15.2	840
12.8	10	14.9	851

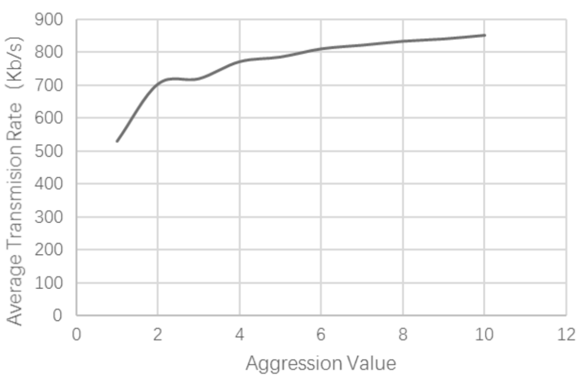

Fig. 3. Data of the test scenario A

4. Discussion

4.1. Verification of tunability of the media filter send rate

In the course of the experiment, each experiment is transmitted with a size of 12.8 M media files. When the priority of sending data is set to 1, the file transfer time is 24.25, the average transmission rate is 529 Kbs; when the priority of the transmission data is set to 2, the file transfer time is 18.25, the average transmission rate is increased To 703 Kbs. With the sending data priority increases, the file transfer time is getting shorter and shorter, the average sending rate is faster and faster. From this it can be concluded that the data flow with higher transmission priority in the same network environment can obtain higher throughput than the data flow with lower transmission priority. Corresponding to the enhanced QoS support streaming media system model, when the player needs higher QoS quality (such as the player screen is larger, higher resolution), the media filter by adjusting the priority of the data flow to a The higher the value to get a higher transmission throughput, and then meet the player QoS requirements; the contrary, when the player requires QoS quality is low, the media filter to reduce the priority of sending data streams, to meet the player QoS Requirements while more rational use of network bandwidth.

Table 2. Test Scenario B

File size (Mb)	Aggression value	Transmission time (s)	Average transmission rate (kb/s)
2.8	2	18.7	684
12.8	3	17.3	739
12.8	4	17.2	744
12.8	5	17.2	744
12.8	6	17.1	748
12.8	7	17.0	752
12.8	8	17.1	760
12.8	9	16.8	764
12.8	10	16.6	768

4.2. Verification of tunability of media filter's friendliness

In the course of the experiment, each experiment transmits the same media file with a size of 12.8 M, and sends the TCP data stream with a rate of 50 Kbs as the background traffic. The experimental results show that when the priority of the transmitted data is set to 1, the file transfer time is 29.95 and the average transmission rate is 427 Kb/s. When the priority of the transmitted data is set to 2,

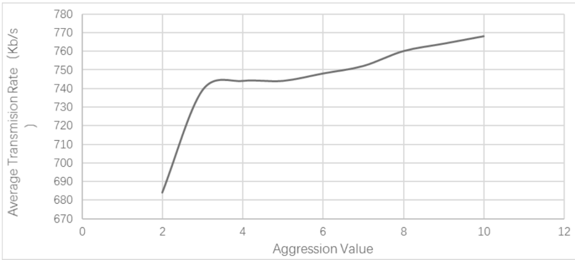

Fig. 4. Data of the test scenario B

the file transfer time is 18.75 and the average transmission rate is 624 Kb/s. As the priority of the transmitted data increases, the transmit data throughput also grows smoothly. It can be concluded that the data streams of different transmit priority and TCP data streams are friendly in the competitive bandwidth, and are still able to maintain the characteristics of the transmission rate as the transmission priority is improved. In the streaming media system model with enhanced QOS support, the data stream output through the media filter module can maintain good TCP friendliness while maintaining its inherent characteristics.

5. Conclusion

The results show that the QoS-supported streaming media system model can effectively improve the QoS performance of streaming media, and meet the QoS requirements of different playing terminals. Media filter module as the core module in the model, when the player needs higher QoS quality (such as the player screen is larger, higher resolution), the media filter by adjusting the priority of sending data to a more high value to get a higher transmission throughput, and then meet the player QoS requirements. On the contrary, when the quality of the QoS is required to be low, the media filter can reduce the bandwidth of the transmitted data stream and make more reasonable use of the network bandwidth while satisfying the QoS requirements. On the other hand, the transmission rate adjusted by the media filter can ensure smooth transmission rate and good TCP friendliness (data streams with different transmission priorities have good friendliness with each other, and TCP data flow is also friendly). Experimental results show that the improved QoS model built in this paper can effectively improve the performance of streaming media QoS, and meet the QoS requirements for different broadcast terminals. The main contribution of this paper is to study the traditional model of streaming media service system. On the basis of this, the concept of media filter is introduced, and a streaming media system model is proposed to enhance QoS support. By building an experimental platform, the performance of the core module in the system model is demonstrated. However, the system model is currently in the theoretical research stage. It only has a preliminary demonstration model that is not fully realized. In future research, this enhanced QoS support for streaming media system implementation and actual data validation also requires a lot of work.

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