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一种基于流媒体的实时调度控制算法QFEC*

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QFEC: A Real-Time Scheduling Algorithm Based on Stream Media

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Abstract: With the development of network, the video applications based on Internet are growing rapidly. The real-time video applications become popular. Because of the complexity of the network and real-time stream video on demand, the scheduling algorithm has great influence on the QoS. In this paper, a scheduling algorithm based on real-time stream video, QFEC (QoS based on FEC) algorithm is proposed, associated the FEC and Kalman filter theories. According to the status of the receiver, the sending rate is adapted automatically by Kalman filter. The state of the scheduling algorithm is analyzed. This algorithm can maintain the continuity of the real-time video transmission. The simulation results are given, which indicate the scheduling algorithm can provide good video service.
Key words: stream media; QoS; forward error correction (FEC); Kalman filter

摘 要: Internet 视频业务的普及和用户越来越高的服务需求推动了实时流媒体业务的迅速发展,流媒体业务的服务质量(QoS)成为业界研究的热点.由于网络的复杂性,流媒体的实时调度控制算法是解决流媒体 QoS 的关键.结合 FEC 编码技术和 Kalman 数字滤波技术,提出一种基于 QoS 的改进 FEC 调度传输控制算法——QFEC.该算法根据接收方的状态合理调度流媒体业务,并结合 Kalman 滤波器原理完成传输速率控制.通过算法状态分析,以及实验数据和性能分析表明,该调度算法能够维持视频数据良好的连续传输,降低视频流的丢包率,显著改善流媒体业务的QoS.
关键词: 流媒体;QOS;前向纠错(FEC);Kalman 滤波器

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1 Introduction

Due to the development of broadband, to occupy a higher bandwidth, more Internet content will become more common, are found among the stream media content and technology for the development of the most rapidly. Stream media business is becoming telecom enterprise development in the field of new value-added services. Due to the special nature of network technology, users need to get data from the network to connect the server was in a small amount of data the user or small, will not be any problem, but when users doubled, the amount of data very expansion, the quality of service (QoS) into stream media business users and providers of the most concern.

Stream media transmission business is a weak real-time systems, time delay on the request is relatively high, that is, the transmission of video data packets arrive at the client must be allowed time frame; delay jitter at the same time, that is, video and audio streaming continuity with high demands. Stream media businesses usually require 25 to 30 frame/ s speed of processing and transmission. When the user, the system overload, the system can be appropriate to lower the quality of video streaming, such as video transmission only the basic level to ensure the continuity of the video player. However, lower-quality video streaming is limited, in order to effectively ensure the streaming media business, the quality of service, the system needs to provide effective real-time traffic scheduling control mechanisms.

In the streaming media server, each of the video or audio streams that can handle a task, and each media stream processing cycle as a mandate, as described in (C,P), C for the implementation of the mandate, P for the implementation of the mandate of the cycle. Such as video streaming to the processing speed of 30 frames / sec, to assume that every frame of the deal as an example of the task, each instance has the equivalent of the processing time for restraint, to be completed in 33ms, the video stream can be dealt with as the implementation of the cycle of 33ms to constitute a sequence of tasks. Each of the media stream of QoS control, is described by three parameters: S, Q, F, that is, for example to deal with S consecutive months, at least the successful completion of the Q, but also for the failure of a number of examples of not more than 000 F. Said Q / S task for the smallest success rate, F, said the task for the maximum allowable number of failures in a row. If the server scheduling lead to a particular stream S consecutive cases more than 000 examples of the failure of SQ, for example, or the failure of more than 000 F, that the task of scheduling server not meet the media flow QoS.

The classic real-time task scheduling algorithm EDF (earliest deadline first)[1], DM and LLF[12], and so can not overload the system when dealing with streaming media business, the quality of transmission. This article, based on the UDP protocol, combined with forward error correction coding technology (FEC), a QoS based on the FEC to improve the scheduling transmission control methods, combined with Kalman filter principle of rate control.

2 FEC Coding Technology

According to a domain FEC coding theory, in cell division or increase the level of information redundancy. It will a group of 000 m cell transformation for the m + k-cell transmission. At the receiving end, as long as any of them received the correct 000 m cell, anti-change operation can be the original 000 m cell[7]. FEC performance with the network load, data flow characteristics of the increase in the amount of information redundancy and other factors. FEC traditional methods may result in unnecessary waste of bandwidth, because regardless of whether the actual cell loss, it must be redundant transmission cell. To determine the increase in the number of redundant cell, the network must understand that the loss of properties and applications of the data flow characteristics, and they often change the dynamic. FEC simple method may also exacerbate the degree of congestion. FEC used in real-time semi-reliable transmission, and the FEC in the reliable transmission in particular, more reliable transmission is still relatively small.

FEC to address not only the traditional lost, and error-handling, and thus achieve greater cost. If he had known the location of the missing group, you can greatly reduce the encoding and decoding overhead. As the modern network of small bit error rate, error control of the main tasks is to restore the lost group. For the characteristics of the network protocol software for the FEC can remove the block encoder (block erasure codes) to achieve. The linear encoder block the use of linear transformation of data generated code can be expressed as a matrix in the form y = x G, which is the raw data x (m length of the vector), y is the vector encoded (for the length of the m + k),   ), G is m k coding matrix.

FEC coding can be broadly divided into two types of cell-level code, a message-level code. In this paper, the FEC coding is Based on message level, according to the different MPEG frame type (I, P, B frame) of the importance of data for FEC coding. I frame: independent coding frame, the frame encoding and decoding of frames do not need to rely on other data, can be independent of the encoding and decoding; I can frame as the encoding and decoding other frames of reference. P-frame: forward forecast frame, the frame encoding and decoding have to rely on its in front of the encoding and decoding the I-frame or P frame data can not be independent encoding and decoding; P frames can be used as other frame of reference to the encoding and decoding. Frame B: two-way forecast frame, the frame encoding and decoding not only in its reliance on in front of encoding and decoding the I-frame or P frame data to rely on in the back of its encoding and decoding the I-frame or P frame data can not be independent of the encoding and decoder, and can not be used as encoding and decoding other frames of reference. B relative to the frame of the more important I-frame and frame-P message we have adopted a code FEC transmission.

3 Scheduling Algorithm

The uses of traditional packet retransmission mechanisms consume too much bandwidth resources, at the same time a lot of feedback message will consume bandwidth resources. Improved scheduling QFEC control technology has effectively solved by the packet loss caused by retransmission caused by the consumption of bandwidth at the same time deal with the feedback control in order to campaign for the group to avoid too much feedback on the message bandwidth, To provide users with better video quality of service.

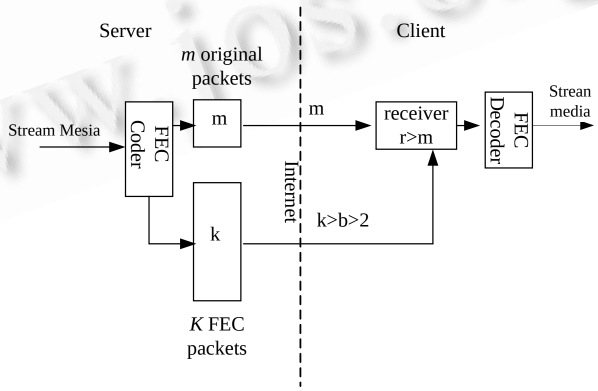

Fig.1 FEC coding diagram

Figure 1 will be shown in video encoding FEC data. 000 m by the raw data encoded to generate (m + k) data, in order to reduce redundancy send here do not send all the (m + k) data, but based on client needs a corresponding transfer of (m + b) a message (2 <Bm, will be able to successfully restore the video decoding.

We used Gilbert model[4] to carry out the packet loss rate is estimated. The packet loss rate, Gilbert model is a better description of the method.

The packets on the network transmission with three state Markov chain to describe, shown in Figure 2. 0 normal state of the system to send messages; state of a system that s in a row at the normal reception of message, this time to show a better environment for transmission, you can send the appropriate message to reduce the value of b; 2 state t times in a row that message is lost At this time to send the appropriate message b increase the value (k> b> 2). p, q transmission for the success of the state of transmission and loss of status between the probability of conversion. Gilbert-based model, the establishment of QFEC (QoS based FEC) of the state transition map shown in Figure 2.

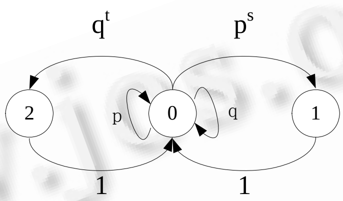

Fig.2 QFEC state model

If i-xi the first time that the packet transmission of the state, p, q, ps-qt can be expressed as (1), (2), (3)-indicated. One, xi = 1, said the success of transmission, xi = 0 said that the failure of the transmission.

$$p=P(x_i=1,q=P(x_i=0)) \quad (1)$$

$$p^s=P(x_i=1|x_{i-1}=1,x_{i-2}=1,\ldots,x_{i-s}=1) \quad (2)$$

$$q^t=P(x_i=0|x_{i-1}=0,x_{i-2}=0,\ldots,x_{i-t}=0) \quad (3)$$

Sender using the literature[6] based on the formula in the rate calculation method of calculating the initial rate, the rate of transmission KALMAN filtering[9,10], according to the receiving feedback on the rate adjustments shown in Figure 3. Each side does not receive feedback message, but a message for a feedback. So as to reduce the feedback network to reduce waste of resources. If the packet loss feedback from Kalman's forecast set to send rate.

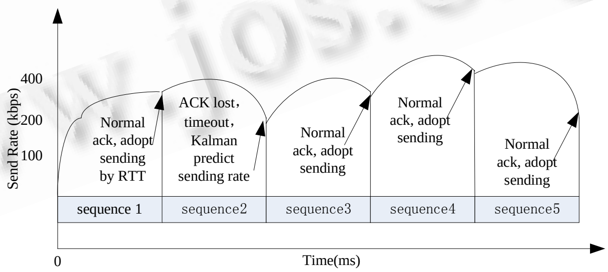

Fig.3 Using principles of Kalman filter to set sending rate

Here to MPEG[5] The FEC encoding video streaming transmission, for example, MPEG for the GOP structure of the relative importance of the I-frame data, P frame data transmission FEC coding. When a user to receive the packet loss rate, the sender will send more than the corresponding number of Message FEC coding; On the other hand if the packet loss rate, the FEC to send coded messages to a corresponding reduction in the number. By the recipient to receive the message according to the current calculation of the RTT (Round Trip Time), feedback through the feedback message to the sender. According to the sender of the feedback information to adjust the rate to send to address the problem of congestion. The continued low rate of loss of the line, scheduling algorithm stable and sustained advantage.

Sender based on feedback and forecast information to the appropriate speed to send data. To receive feedback on the analysis, the adjusted rate adjustment. As shown in Figure 4.

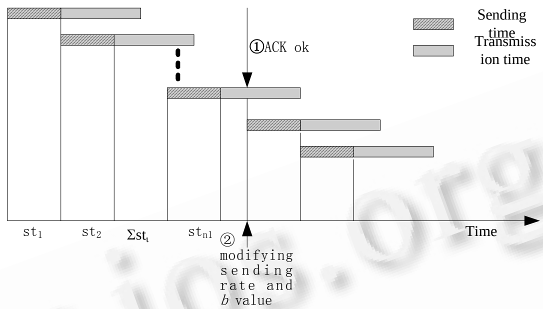

Fig.4 QFEC the normal process of sending

Reported to packet_size said that if the size of the text, and use that to send send_speed rate, it sent a message of the time (the task execution time): .

Based on the state to change the timing of feedback and information processing, information can be sent to a breakdown waiting to send and continued to send two. Send to wait, as shown in Figure 5, may provide a reliable service and to ensure that the received data completely reliable and accurate; continue to send, as shown in Figure 6, the user is able to tolerate the QoS, at the expense of a certain quality to Guarantee the uninterrupted transmission service.

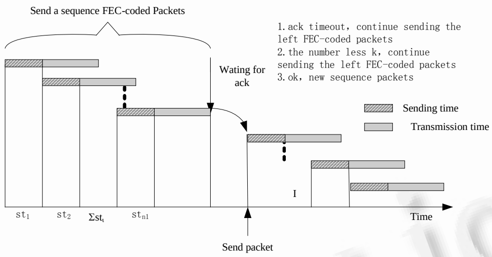

Fig.5 Reliable Wait-Transmission

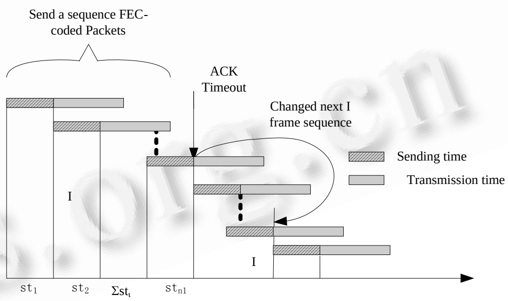

Fig.6 Adopted continue transmission

As shown in Figure 6, after sending wait to receive the feedback confirmed that the continuous water transmission. When the feedback received, in accordance with changes in the state change the FEC to send messages to send and b value of the rate. If the feedback overtime, the use of Kalman prediction process[10], and the rate of sending messages to send to set up.

4 Performance Analysis

Network topology, in Figure 7, the use of NS2 network simulator [2,3]. s0, s1 and s2 is the server, d0, d1 and d2 is the client, R1 and R2 is a router in the middle. S0 use QFEC scheduling algorithm, s1 for the ordinary time-sharing system. D0 corresponding QFEC used to configure the client, D1, D2 for the normal client.

Fig.7 Network topology

1) the successful transmission rate

Figure 8 is the average system load increases, were used in this article based on the FEC's QoS scheduling algorithm (hereinafter referred to as QFEC scheduling algorithm) and the literature of the time-scheduling algorithm, the video data transfer rate of failure of the comparison. Streaming media for the task of the total number of 15, the average server load and the flow of each media QoS parameters were set to S = 30, Q = 24, F = 4, Ci = 8ms, Pi = 30ms, time-scheduling algorithm Time for the film 1ms.[11]

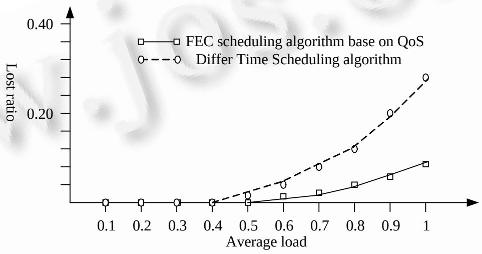

Fig.8 Comparison to QFEC and DTSA

As can be seen from Figure 8, when the average load were increased to 0.4 and 0.5, the two are scheduling algorithm so that the video data transmission failures occur, but this article QFEC scheduling algorithm so that the load-bearing capacity of the system significantly increased, which In this paper, the algorithms that improve the utilization of the system.. In addition, when the system load increases to a certain extent, the ordinary time-scheduling algorithms, QFEC scheduling algorithm allows video data transmission success rate of increase (as shown in Figure 8, the failure rate), QFEC This shows that the scheduling algorithm more Suited to deal with continuous media stream task, especially when the server load more, the task scheduling algorithm for the failure of the low rate, better able to meet the task of streaming media QoS requirements.

2) to send image quality comparison

s0 and s1 configured to the same bandwidth and resources to send video streams a, in the D0 and D1 above to receive video surveillance data. S0 one on D0 and QFEC corresponding to the configuration of the server and client nodes off procedures. I frame of the data FEC (12,6) code transmission, m = 6, t = 3, s = 6.

Figure 9 is the server using layered video transmission, the sending frames in a row sequence (IBBBPBBB), the system used in this article QFEC the scheduling algorithm and do not use scheduling algorithm, the receiving end of the video quality of the results of the comparison. Figure in the assumption that the size of the buffer zone to receive fixed. Figure 9 in a server for the data source; b system is not scheduling algorithm used, the receiving end of the decoding of data; c is the scheduling algorithm used in this article, received the receiving end of the image. Obviously, there is no scheduling algorithm using the receiver can decode the right, and the scheduling algorithm used in this article, you can get a better video image.

(a) (b) (c)

Fig.9 Comparison of the received video

5 Conclusion

According to the characteristics of video data, to distinguish between different data, such as I-frame, P frame and B frame. On the server-side streams of different media, using different mechanisms to ensure that a reasonable schedule to achieve video data transmission. Kalman filter theory and the FEC coding technology integration to address the high rate of packet loss in the line of video services. Comparison of experimental data through the use of this article dispatching algorithm for improved video performance.

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