STUDY ON REAL-TIME VIDEO TRANSPORATION FOR NATIONAL GRAIN DEPOT

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Abstract: Because current remote monitor systems can’t deal with problems of real-time transmissions in the bad condition of network very well, this article presents a study which is combined with the evaluation of video streams, adjustment and sealing user-defined data pockets and discarding useless data pockets. The goal of this solution is to transmit video information of national grain depot. And its practical use shows that the system has good effect.

Keywords: code rate, adaptive transmission, RTP protocol.

1. INTRODUCTION

It’s known that, professional companies in China take responsibility of depositing the nation grain. But without strict supervision, someone will, aiming of private benefit, conduct corrupt behaviors, such as making the false bill of document and imitating good grain with bad grain and so on. On the view for this situation, government should construct a series of supervising system which can provide supervision at any time. The most important thing is how to copy with the quality and transmission of real-time videos. They are proof to accuse those criminal behaviors. This monitor system is operated depending on accuracy of real-time video data. However, the quality of video data is associated with conditions of network environment. If the bandwidth is stable and the speed of network transmission is ideal, the packet-losing rate will decrease to make sure video transmitted successfully, besides phenomena of frame-losing and time-delay will disappear. But the bandwidth is varying from time to time. If there are so many data pockets in the channel, the increasing rate of packet-losing will impact the transmission of video stream very seriously (B.J. Kim et al., 2000). The media streams compressed by MPEG4 technology become hard to transmit under such bad conditions. It's because that MPEG4 technology split video stream into several layers, losing of key frames will damage the quality of video and the network spending will become larger result from error inspection and retransmission protocol (Talley et al., 1996). In order to solving this kind of problem, this article presents a method. It is based on the speculate speed of network with AMID algorithm and adjusted the speed rate of video transmission automatically and sealed data packets in user-defined format and discarded some useless data packets.

2. GENERAL DESIGN

The system of real-time video transportation is composed by three components: sender, net medium and receiver. This paper concentrates on introducing the design of sender. The overall structure of transportation system is shown in Fig.1.

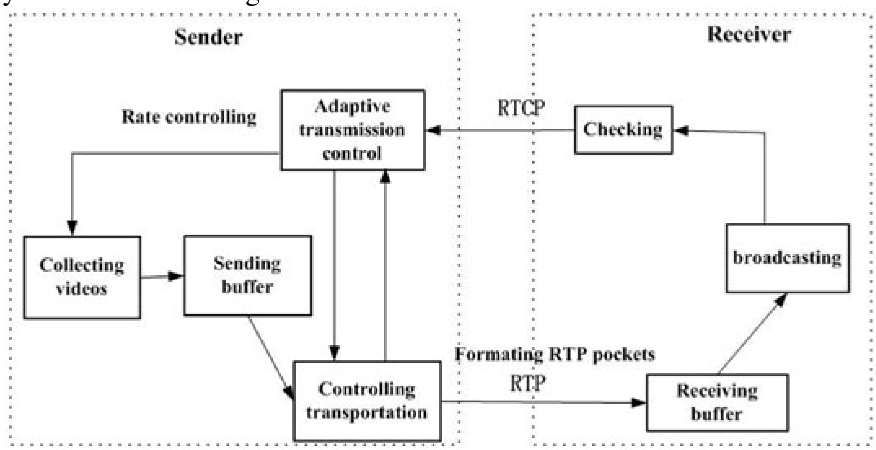

Fig. 1: Overall structure of transportation system

2.1 Obtaining and compressing real-time video streams

Video streams are received at sending-port through cameras devised in grain depots. These videos will be transported to the media server and compressed into MPEG4 video streams. The reason to choose MPEG4 for transporting format is that, it's apparently different with other video compressed technology. MPEG4 is based on objects and it spilt a video file into different objects that are formatted by a special object layer. (The structure of MPEG4 frame is shown in Fig.2.) Each layer contains much information about figure and texture and other aspects. Moreover, MPEG4 affords extension of time-field and space-field. It will operate some changes on basic layer according with current conditions of network. Furthermore, in order to perfect the quality of video, MPEG4 technology could insert some frame into basic layer and increase or decrease the resolution. In a word, it is most suitable to be used in the field of long distance net TV inspection.

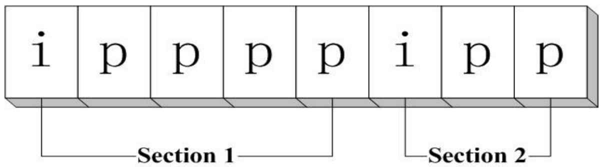

Fig.2: Map showing the location of the research area

2.2 Design of video transportation

Step1: Choosing RTP/RTCP protocol for transportation

RTP/RTCP protocol is a real time transmitting method. It is specially used for the transportation of media data. This protocol could achieve stream media data for singleness & group play in internet. It can rearrange the video stream frame by correct order, checks the integrity of frames, and supplies some services such as security guarantee and so on.

RTCP is one kind of controlling telegram that is sent by sender. Its main function is to afford QOS quality feedback. As a part of RTP protocol, it is relevant with stream control supplied by other protocols. And feedbacks operate adaptive code-control directly. The function of feedback is executed by both sender and receiver.

At the conversation of RTP, all members send RTCP control packets periodically. Server can take advantage information to change transmitting speed optionally. The corporation between RTP and RTCP, can perfect the efficiency of transmission by necessary feedback and least cost. Therefore, this protocol is pretty suitable for real-time transmission. Considering information collected from feedbacks, it's easy to make a fitful strategy.

Step2: Evaluating the speed rate of transmission on the foundation of feedback data brought by RTCP pockets, the rate of video stream will adjust by itself.

The solution is based on adaptively adjusting the speed rate of video stream transmission; the sender analyzes the current conditions of network through feedback information brought by RTCP pockets (J.Y. Tham et al., 1998). Bandwidth can be calculated with the loss-pocket rate in the QOS report. After getting approximate value of rate, one can use programs at the sender to adjust transferring rate into a suitable value that is useful to real-time transmission.

There exits two algorithms to adjust the sending rate: the model algorithm and the detect algorithm. The foundation of model algorithm is loss-pocket rate and time for sending and receiving data pockets and maximum number of communicating data pockets, which should be used to calculate the sending rate (D. Mills, 1992). Detect algorithm is that sender evaluates the speed of network by frequent adjustment of sending rate. This solution chooses the detective and adaptive controlling algorithm AMID. Its description is as the follows:

if(P< Pth)

r=min((r+AIR),MaxR)

else

r=max(□r,MinR)

Where: P stands for the actual loss-pocket rate; Pth stands for a trigger value in a range of time; r means sending rate. MaxR and MinR separately stand for maximum value and minimum value of sending rate that has been set before. AIR means accelerating rate, □ is subtrahend factor and its value is between 0 and 1 (Li, 2006).

The meaning of this algorithm is that, in a defined range, one can increase the sending rate untill the loss-pocket rate is too great to assure affording an accurate play in the receiver. And the next step is to reduce the sending rate to make sure the loss-pocket rate in an acceptable range, following increasing sending rate gradually.

Step3: Analyzing MPEG4 pockets and making a RTP pocket in a particular format.

The chief difference between MPEG4 and other traditional video compressing standard is that MPEG4 is based on objects. This technology splits video data into many different objects and forms a layer for each object which contains figure and texture and other information (S. Palacharla et al., 1997).

Grammatical layers of MPEG4 stream have four aspects: video communication, video objects, video object layer, and video object plane (VOP).

VOP is a frame of video object. MPEG4 separately codes to every VOP and gender three different frame styles: I VOP , P VOP , B VOP. I VOP is very critical to the quality of video and it has no relations with other adjacent VOPs; P VOP needs I VOP in front of it as a consult to compensate movement (J. Shapiro, 1993); B VOP works with adjacent VOPs. Because

VOP is the basic unit for saving video information, so its construction will be introduced by blow picture.

Synchronous code is very important to video play, if once it misses, video will not play well. Report head of video pocket is also crucial, because the receiver need it to sort a correct order for video pockets. Besides as the quantity of information contained by different video pockets, some huge pockets will be lost during the transmission in the network especially under the terrible net conditions, this is one of main reasons that I use the particular defined data pockets to do transmissions.

The particular defined data pocket means that， after analyzing each VOP pocket and getting relative information, one can seal a new pocket by RTP pocket structure which will be introduced in the next section. The focus of this article is to design a new RTP pocket structure which is suitable for special requirement.

The structure of New RTP pocket is shown in Fig.3.

V=2	P	X	CC	M	PT	Sequence number
Timestamp						
SSRC						
CSRC						
Mpeg4 Video data						
RSN					CK	

Fig.3: A new RTP pocket structure is suitable of special requirement.

The meaning of each element:

V: version

Extension - X: defined by RTP structure

PT: a introduction of sort of load interpreting the style of code

Sequence Number: the number of each RTP data pockets. It's used to set up a correct order of data pockets and inspect whether there are errors and damage in pockets.

Marker - M: defined by operating structure

SSRC: help receiver identify the adscription of all streams with only one number that sender supplies. SSRC is a strict random number.

CSRC: identify the streams.

Besides (RSN) stands for synchronous code taken by this frame, and critical key (CK) stands that whether this frame is a key frame that takes great responsibility for video quality. This is because, under the conditions that bandwidth is terrible, and some huge pockets will be lost during transmission. And there will be many chances to form congestions. The receiver has to attempt many times to send repeat requests. In this situation, the spending of network will become so large, and the quality of real-time transmission will be impacted seriously (H. Schulzrinne, 1995).

So we can average huge video streams to many RTP pockets, this solution will alleviate the pressure of each pocket and fitful for transmission (B. Paul et al., 1999). Each pocket will take the same RSN to certificate that all pockets come from the same frame and the sequence number will increase by one to record their order.

Step4: Discarding frames which aren't key frames and adjusting the rate, if the condition of network is poor.

Even though the speed rate can be evaluated from Qos report, it's also difficult to operate a perfect transmission because of the abnormal variety of network. If the condition is much better than before, the transmission will be very successful. But if it's worse, the bandwidth will be narrower and loss-pocket rate will increase.

So it's hard to transmit video streams with the speed calculated before. In order to decrease the pressure of network, the reasonable approach is to adjust the rate again by discarding some useless RTP pockets that are not important for video play.

For MPEG4 video stream, I frame is compression of static imagine, P frame is formed depended on previous I frame. I frame is consulting frame whose losing will bring great damage to the quality of video as other frames can be sorted by a correct order. This solution is that discarding some frames that do little impact with video quality, by precondition that video could be play well at the receiver. (A. Kantarci et al., 2000).

Therefore, B frames which are relatively useless and P frames that are far away critical frames are should be discarded. This kind of operation should be done in the buffer. Buffers in this paper are sorted to two aspects, one part is buffer used to real-time transmission at the sender, and another part is used to save data pockets and wait for the command of retransmission.

3. IMPLEMENT

This experiment needs three personal computers to take responsibility of MPEG4 Collecting Server and Net Transportation Controlling Server and MPEG4 Receiver Client. Their functions are different: main functions of MPEG4 Collecting Server are collecting the spotted real-time video, compressing the video stream, making video files and saving some videos to the hard disk; Main functions of Net Transportation Controlling Server are providing sending buffer for stocking real-time video stream, analysis of current net situation, speculating the speed of transportation next time span, producing a RTP pocket in a particular format and checking information coming from receiver; Main functions of MPEG4 Receiver Client are receiving video streams, checking and rearranging the order of video stream pockets, sending requiring messages to sender and playing videos and so on. The structure of the circumstance of experiment is shown in Fig.4.

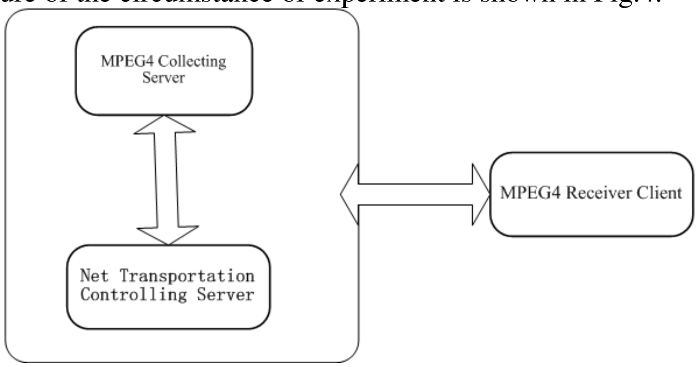

Fig. 4: The structure of the circumstance of experiment

The process of transportation is consisted by several steps as following:

(1)Collection of video streams:

Video streams are collected by cameras settled at grain depots, and sent to the MPEG4 Collecting Server in special line.

(2) Disposal of video streams:

Net Transportation Controlling Server speculate the speed of net transportation with accordance to RTCP data pockets, and send messages of adjusting rate to MPEG4 Collecting Server. According to evaluated result, system has different choices to do the transportation. MPEG4 streams can be transported steadily in the range of rate between 4.8kbit/s and 64 kbit/s (Zhang, 2006). But the situation of network is varying from time to time. So the condition may be very terrible for transportation in some time spans. It's hardly reach to the basic condition for video transportation. At this situation, system can call off these transportations, save video into the data base with unique mark. When the condition is suitable enough for video transportation, system will automatically search whether there are some videos without transportation, and send this kind of videos at prior. If current situation of net reaches the basic condition of transportation, system will adjust the compressing rate and send video streams to the buffer of Net Transportation Controlling Server.

(3)Transportation of video streams:

If there doesn't exist some barriers in the network, system will immediately execute transportation; If the condition is getting worse and the transporting speed is fewer that evaluated speed, the system will remove some useless data pockets in order to reduce the bulk of the whole RTP pockets, and identify each pockets with particular mark to avoid making mistakes in checking process in the receiver.

(4)Receiving video streams:

The receiver will check the order of data pockets. And if there are some phenomenons of disorder, the receiver will send out messages.

Comparing with other solutions dealing with the same problem, this solution has its advantages. This approach use a more positive way to copy with transportation of net. Because it constantly checks the condition of net and speculate the intending rate, so it's positive for its evaluation. It uses different methods to change the state of video, whether decreasing the bulk or changing the rate. Obviously, it's an efficient solution.

4. CONCLUSIONS AND FUTURE WORKS

It's critical for supervision of nation grain depot to obtain real-time video data that supply the valuable evidences to the administrative officer. Therefore, it's very important to assure the quality and uploading in time of video data which supervisors need to analyze and execute. In a view, this paper designs a series of solution which can deal with some difficult problems during the process of the real-time transmission of huge video information. But if the current condition is extreme terrible, it is hard to operate real-time transmission. In this situation, one had better cancel this transmission and save all data pockets into the hardware waiting for next transmission when the condition becomes better. Following with the further development of web technology and hard devices, you will have variety of solutions..

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