2011 International Conference on Information Management and Engineering (ICIME 2011)
IPCSIT vol. 52 (2012) © (2012) IACSIT Press, Singapore
DOI: 10.7763/IPCSIT.2012.V52.91

The Implementation of an Adaptive Mechanism in the RTP Packet in Mobile Video Transmission

Xiaoyu Ma + and Ying Ma

[1] Department of Economic and Trade, Henan Institute Of Engineering, Zhengzhou, China
[2] School of Software, Henan Institute Of Engineering, Zhengzhou, China

Abstract. This paper firstly describes the advantages of packing mobile video stream before transmission, and then lists two systems of packet,and analyzes two factors of impacting the size of packet, on this basis, provides a design scheme of adaptive packet, and analyzes the transmission performance of the execute of the designing scheme in the actual monitoring system, The results show that the design of the new adaptive packet can improve the efficiency and quality of the video transmission.

Keywords: RTP, mobile video, packet, implementation.

1. Introduction

In any type of callback communication, the synchronization of sending and receiveing time are two key issues. There are two modes of synchronization, synchronous mode and asynchronous mode.Asynchronous mode sends the data stream by symbols of the predetermined number of bits, each symbol has the start bit and the parity bit followed , so each symbol has 2 bits overhead[1]. In synchronous mode, Transmiting characters haven't any start and end flags, in order to detect the start and end flags in the receiving end, each data segment begins from flag bit of the segment's header, and ends with the pattern of Postpositive bit, similar to the asynchronous system.This kind of data segment. The data segment is the data packet, which can be fixed length, such as ATM cell ,can also be variable length byte, such as the IP packet[2]. Fig. 1 is a typical packing flow chart of inputing video sources.

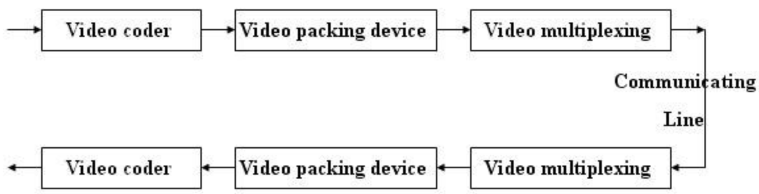

Fig. 1: Packing Schematic diagram of the video stream

2. The Advantages of Packing Mobile Video Stream before Transmission

Firstly, can efficiently use the bandwidth of network, the business type of net load is flaged by the type field's contents of each packet header, as unpacker can fast pack by the type field[3], Therefore, we can achieve multiplexing transmission of variety of data stream, and improve the utilization ratio. What's more, we can determine the appropriate packets to improve transmission efficiency according to network type and conditions.

+ Corresponding author. Tel.: + 0371-68723925.
E-mail address: hnzzmxy@yahoo.com.cn.

Secondly, can improve the robustness of video data about errors, using the packets,can make the impact of error and information's loss limited to a single packet, the video decoder will build up synchronously in the followed packet which has no error, and the MB(macroblocks) included in packet can be predicted independently by other MB[4]. Finally, can make the decoder interact dynamically,adjust the bit rate and error mechanisms to improve the quality of video.

3. Two Systems of Packet

The MPEG-4 data packages packed into the RTP packet, there are two mechanisms[5]:

(1) MPEG-4 for each is encapsulated into a RTP packet, As shown in Fig. 2.

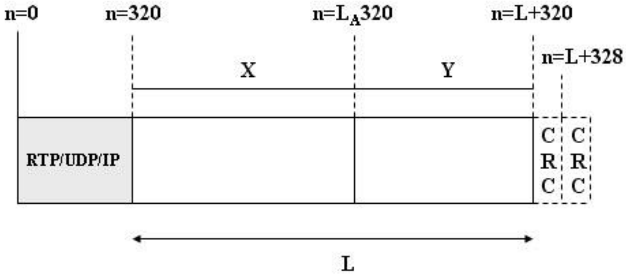

Fig. 2: A single MPEG-4 packet of each RTP packet

(2) Each RTP packet contains multiple MPEG-4 packet As shown in Fig. 3.

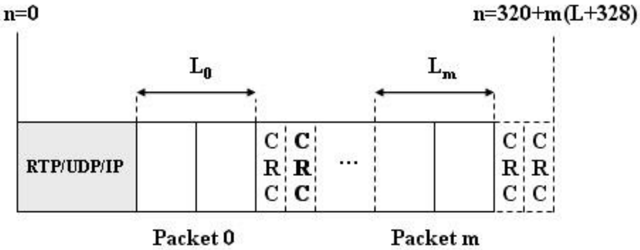

Fig. 3: Multiple MPEG-4 packets of each RTP packet

In comparison, choosing the long packet means when each RTP packet is damaged, there will be more loss of data, but, at the same time the packet header uses less packet ratio, can also reduce the possibility of damage to the header; choosing the short packet is opposite.

4. Two Factors of Impacting the Packet of Size Transmission of Parameter Set

4.1. Network status

Using RTP / RTCP transmission of video, RTCP can periodically feedback the information of current network status, and send it to the server.the ratio of packet loss and the time delay of transmission are two indexes which can react the status of network[6].

In the case of Good status of network , the ratio of packet loss will be reduced,at this time, using a long packet can improve the efficiency of transmission, and in the case of bad status of network, using a short packet, can reduce the loss of massive data which is caused by the loss of the long packets, and also reduces the error probability of packets.

4.2. Contents of video data

when the image sequence of video runs intensely, we can select the short packet to improve the quality of video, because the error probability of short packets is reduced, and packets can transfer data much faster, furthermore， can faster get the feedback information of channel, and adjust the bit rate[7].

when the image sequence of video runs calmly, we can select the long packet to improve the efficiency of transmission, because at this time the size of each MPEG-4 packet is relatively small, packing many short packets can reduce the channel ratio of occupied by the protocol header,and increase the efficiency of transmission.

5. Design of the Adaptive Packet

The determination of the traditional RTP packet's is implemented by the packet loss rate, the advantage is simple to implement, the shortcoming is not with the new video compression algorithm and the combination of fault-tolerant technology. Packing mechanism based on video contents, is proposed on the basis of MPEG-4, it is organically integrated with the MPEG-4 object-oriented thinking, and this is also the research direction of video transmission and compression in future[8]. The advantage of this packet is that it can improve the error robustness.

Combination of the two applications,can make up for the shortcomings of each other, thereby enhancing the robustness of video errors and the utilization ratio of channel. So，this article proposes an adaptive system of RTP packet, according to current network conditions and the contents of the current video data to dynamically adjust the size of packet. The following is the algorithm process:

5.1. Calculate the Packet loss rate,and determine the network status

The calculating process of the Packet loss rate is divided into three steps:

Step1: Calculate the number of lost packets of sending time slot L(n): According to the difference of the cumulative number of lost packets included into two receivers'report (RR) packets before or after a time slot of the send end is the number of the packet loss of sending time slot, as in (1)

$$L(n)=C(n)-C(n-1) \tag{1}$$

Thereinto,C(n) is the number of lost packets after the nth sending time slot;

Step2: Calculate the number of packets received in this time slot R(n): The difference between expansion of the highest packet sequence number in two packets received is the number of packets to be received in this time slot, as in (2)

$$R(n)=H(n)-H(n-1) \tag{2}$$

Thereinto, H(n) is the highest packet sequence number of lost packets after the nth sending time slot;

Step3: Calculating packet loss rate in this time slot F(n):the ratio of L(n)and R(n) is packet loss rate in this time slot, as in (3)

$$F(n)=L(n)/R(n) \tag{3}$$

Packet loss rate is an important index of ing the conditions of network channels,but wo can not directly use packet loss rate to judge the conditions of twork channels,and adjust the sending of video according to it,because this will make sending data change too frequently,and also make the image's quality of receiveing end unstable[9].so,before using packet loss rate to estimate the conditions of network channels,we shoud firstly make it smoothing.we can suppose that t(n) is the packet loss rate smoothed,its smoothing relationship is as in (4):

$$t(n)=(1-a)\times t(n-1)+a\times F(n),(0\leqslant a\leqslant 1) \tag{4}$$

When the value of a increases, the current packet loss rate F(n) increases the influence to the smoothing results t (n) ; When the value of a decreases, previous smoothing results t (n-1) increases the influence to the current smoothing results t (n). According to the experimental results to determine the value of a, the smoothing result will weaken the QoS oscillation of network channels, using the packet loss rate smoothed to judge the conditions of network channels is good to reasonably adjust the transmission of video.

The classification algorithm of network status: Set two thresholds k1, k2 (the value of test experience), the smoothed RTP packet loss rate is t (n):

① when t(n)< k1,As a light load;
② when t(n)> k1,As a full load;
③ when t(n)> k2,As a jam;

In order to test,we can take k1 and k2,and set the values 2% and 4%.

5.2. Calculating the exercise amount of current video sequence

Each MPEG-4's video packet data has two parts, one is made up of head and motion data, and the other is mainly made up of the texture data. The algorithm of current simple and better exercise mount of video sequence is calculating the the first part's proportion for each MPEG-4 packet. Calculated as in (5):

$$A = \frac{\overline{Y}_{MB}}{\overline{X}_{MB} + \overline{Y}_{MB}} \tag{5}$$

Here, the variable A is the proportion of the first part in group, characterize the exercise amount of video image sequence, $\overline{Y}_{MB}$ is the average number of bits per MB in the first part, $\overline{X}_{MB}$ is the average number of bits per MB in the second part. Similarly, we also set some appropriate threshold, which can generally be divided into three intervals in the application, and be adjusted concretely according to the different application. We can take 0.1 and 0.4 as the two thresholds, the value of A is divided into three intervals.

5.3. Determine the appropriate packet size

According to the Packet loss rate based on the calculated value of t (n) and A's value of characterizing exercise amount of video image sequence,we can determine the appropriate packet size, select a monitoring system, values of specific application are as the table I:

Tab. 1: values of determining the appropriate packet size of the selected monitoring system

value range of t(n)	**value range of A**	**value of L**
0<t(n)<=0.02	0.1<A<0.4	1200-1000*A
	A>0.4	800
	A<0.1	1100
0.02<t(n)<=0.04	0.1<A<0.4	1200-2000*A
	A>0.4	400
	A<0.1	1000
t(n)>0.04	0.1<A<0.4	1000-2000*A
	A>0.4	200
	A<0.1	800

Thereinto, L is the length of RTP packet, t(n) is the packet loss rate in the time of n,A is the proportion of the first part in MPEG-4's packet.

6. Analysis of Transmission Performance

The proposed packet loss rate and video content based on the adaptive grouping approach, were used in monitoring system tested, According to the experimental results, we can see, whether the transmission efficiency or the quality of the video is obviously improved.

In the aspect of transmission efficiency, the use of adaptive packing transmission frame rate is bigger than the other two packing systems, and the video images are more fluent. In the monitoring system tested , the table II is the results of statistics about frame rate running in the ten minutes:

Tab. 2: frame rate's comparison of three Systems

frame rate	**maximum**	**average**
Packet loss rate	9 frames/s	6.8frames/s
Video content	9frames/s	6.5frames/s
adaptive	10frames/s	7.2frames/s

7. Concluding Remarks

This paper analyzed factors of impacting the packet of size,and proposed a design scheme of adaptive packet, and analyzed the transmission performance of the execute of the design scheme in the actual monitoring system, According to the experimental results, we can see that the design of the new adaptive packet can improve the efficiency and quality of the video transmission.

8. References

[1] DAHLMAN E WCDMA - The radio Interface for future mobile multimedia communication [J], IEEE Trans. On. Veh. Tech.

[2] Third-generation Project, Technical Specification Group, Services and System Aspects; Service Aspects; Mobile multimedia service including mobile intranet and internet service, 3G TR 22.960, Version 3. 0. 1

[3] The 3rd Generation Partnership Project(3GPP) Available from. http://www.3gpp.org.

[4] MPEG-4 Video Verification Model version 15.0, ISO/IEC JTC1/SC29/WG11 N3093, Maui, Dec 1999.

[5] Wiegand T, Sullivan G J, Bjntegaard G, Luthra A. Overview of the H.264/AVC Video Coding Standard. IEEE Trans, Circuits Syst, Video Technol., 2010. 7

[6] Xiubo Feng ,Yingjian Xie. The new progress of the research in fault-tolerant algorithms for wireless video transmission, Communications journal,2010.11

[7] FFC3016. RTP Payload Format for MPEG-4 Audio/Visual Streams[S]. 2010.1

[8] Gary J.Sullivan.Thomas Wiegand,Thomas Stockhammer. using the draft H.26L video coding standard for mobile applications.[J]. System[C].ICIP,2010,10:573-576.

[9] Vetro and Huifang Sun. An overview of MPEG-4 object-based encoding algorithms.Proceedings of Information Technology: Coding and Computing, p.,2010.6