An error recovery transmission mechanism for mobile multimedia

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Abstract

This paper presents a channel coding transmission mechanism in wireless and mobile networks by using the cross-packet transmission method of the sub-block structure to solve the problem of the poor quality of the network mobile multimedia. The present channel coding technology has the protective effect aiming to the data transmission in the mobile network. However, the present channel coding technology faces many kinds of network transmission phenomenon under the limited bandwidth constraint, including those comprehensive problems caused by the network congestion and the signal error. The poor efficiency of the data recovery and the bandwidth utilization is still one of the channel coding's main defects. Compared with the present technologies, the experimental results demonstrate that the proposed method can achieve better bandwidth utilization by means of using sub-block coding structure to combat with small amount of transmission errors. Furthermore, the data recovery performance can be improved based on the two following facts about the cross -packet transmission: 1) the channel coding capacity is beyond a single packet and 2) both packet loss and signal error can be recovered simultaneously.

Keywords: cross-packet, channel coding, forward error correction, FEC, error recovery, mobile multimedia

1 Introduction

With the increasing of the Internet popularizing rate, the source of the digital contents presents a prosperous development tendency through the wireless access technology. According to the statistics from the ministry of economic affairs digital content industry office, the output in the mobile application service attained 2.816 billion dollars in 2012 and its annual growth rate is 13.45%. Adding to the emerging digital converge service, the future mobile users can receive several of data from the telecommunication, Internet and television. The investigating reports in the Google mobile network and the user's behavior also show that 90 percent of the smart phone users watch videos through the mobile phones, and 31 percent of users use the video function once everyday.

The traditional multimedia data transmission model easily causes the data loss under the influence of communication quality and the bandwidth. For example, the lack of the bandwidth under the network heterogeneity causes the packet loss and the poor efficiency of the communication environment causes the data error. The results make users can not obtain the complete data so that the correct information can not be showed. The present network does not possess the quality of service mechanism, the channel coding technology and the low latency method can provide the reliable data transmission. The typical channel coding applications include the wireless communication protocol such as 802.11 Wireless LAN and 802.16 WiMAX, and the video streaming service such as the video conference. The core technology of the channel coding is forward error correction. The original data is coded to the FEC data in the transmission terminal and the two data are transmitted together. The receiving terminal conducts the decoding action to recover the original data within the tolerant data error numbers in terms of the condition of the data reception. Although the FEC utilization can offer the error recovery function, the cost of adding the bandwidth should be paid off. When the FEC coding data cannot recover the network transmission error, the problem of wasting the bandwidth can be caused. According to the advantages and the disadvantages analysis of the network transmission in the FEC application, the high efficient FEC mechanism in the data transmission system should be designed to improve the service quality in the mobile network video in terms of the bandwidth resources whose wireless mobile network is relative precious.

The present known channel code has two types: the signal packet model and the cross packet model. The Figure 1 shows that the signal packet model merges the original data and the FEC data into a signal transmission packet, and its advantage is that the decoding procedure can be conducted after the receiving terminal gets a packet. Therefore, it time delaying is low. While its disadvantage is that it just can deal with the data loss caused by the signal error. What's worse, the whole packet loss has no protective ability and the efficiency of the channel coding can be limited by the length of the maximum transmit unit in the network, namely, the size of the data number in EFC is limited. When the condition of the signal error is severe, the data protective ability provided by the signal packet model will be lowed. The cross packet model regards the packet in the FEC coding & decoding as unit and the

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original packet can codes FEC packet so that the problems of the packet loss and the data loss caused by the signal error can be solved simultaneously, its structure is shown as the Figure 2. The coding & decoding data numbers in the FEC are very large and the receiving terminal needs to wait for time to collect all transmitting packets, the time delaying is rather high. In addition, using the larger data number packet to recover the signal error whose data numbers are relative small in the bandwidth utilization is also not economic. Under the limited transmission bandwidth, the bandwidth cost in the cross packet model has low economic benefit so that its protective ability is limited. The two different kinds of the models have advantages and disadvantages. In general, the signal packet model is fit for the wireless communication protocol to maintain a certain communication quality and the cross packet model is widely applied to all kinds of the network application service. With the variety of the network. The channel coding must adapt to the changing of all network environments and the accumulation of the transmission conditions. For example, the numbers of the data loss may accumulate the final results formed by the different transmission factors. Therefore, no matter whether the effect of the signal packet model is not enough to handle the packet loss and the MTU limitations or low bandwidth utilization in the cross packet model, all of these problems are the needed improved objects.

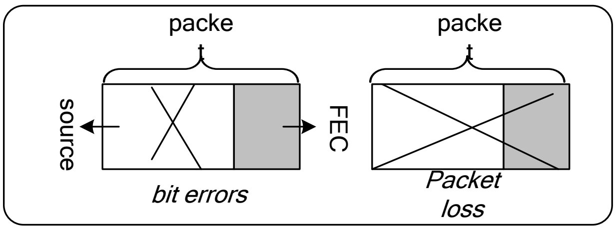

FIGURE 1 The packet structure in the signal packet model

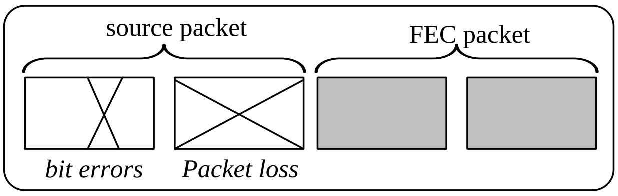

FIGURE 2 The packet structure in the cross packet model

The FEC coding mechanism with the sub block structure by connecting the wireless mobile network transmission system is connected is proposed to minimum the coding data numbers for protect the most common bit errors and the packet loss so that the efficiency of the bandwidth utilization and the efficiency of the data recovery can be considered simultaneously. The proposed mechanism in the operation can be divided into two parts: 1. the original data packet can be disassembled into a smaller source sub block and it regards FEC coding as the unit, and the needed coding sub blocks can be produced according to the present network transmission conditions. 2. The sub blocks can be assembled into the transmission packet, and the assembling method can merge the source sub blocks and the coding sub blocks to form the transmission packets, namely, two kinds of the sub blocks can be assembled into the transmission packet individually. The two different assembling methods allow the coding sub blocks to cross many transmission packets. When the mechanism connects to the wireless mobile network transmission system, the parameter and the formula can be set in terms of the target application's network environment. Moreover, the calculation of the sub block size, the numbers of the coding sub blocks and the assembling of the transmission packet can be done so that the proposed mechanism can have the optimal efficiency.

2 The error recovery mechanism with the sub block structure

Aiming to the signal packet model's and the cross packet model's efficiency limitation in the mobile data transmission, the error recovery mechanism with the sub block structure is proposed to adapt to the FEC coding unit in the wireless transmission environment, and the data's network transmission is conducted by the cross packet model. As shown in the Figure 3, the original data packet can be disassembled into a smaller source sub block and it regards FEC coding as the unit, and the needed coding sub blocks can be produced according to the present network transmission conditions. Then the sub blocks can be assembled into the transmission packet, and the coding sub blocks are allowed to cross many transmission packets during the assembling procedure. On the one hand, the mechanism can effectively improve the utilization of the bandwidth. On the other hand, the packet assembling method in the cross packet can avoid the MTU limitation and has a higher elasticity in the FEC utilization. In order to make the sub block structure apply into the transmission system, the mechanism further design the needed data packet format and formulate the data transmission procedure. The related details are shown in the following sub chapters.

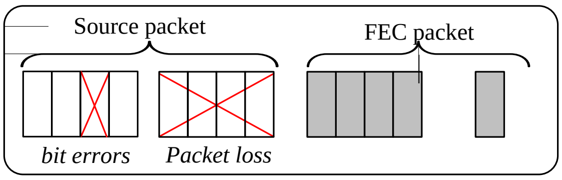

FIGURE 3 The cross packet model with the sub block structure

2.1 DATA PACKET FORMAT

The length of the sub block is allowed to be changed so that how to define the positions of some individual sub blocks for correctly conducting the E\FEC coding & decoding procedure in a packet is the designing key point. Take the applied layer as an example, the Figure 4 illustrates that the data format in the transmission packet is applied with the use of the sub block coding structure. In general, the FEC's error recovery technology is applied in the multimedia data network transmission. The kind of the transmission uses UDP (User Datagram Protocol) as the transmission layer protocol. And then connects with the application layer protocol in the RTP (Real-time Transmission Protocol) to control and supervise the data's transmission conditions. In order to carry all kinds of the data flow patterns, RTP has formulated the FEC's control header to conduct the FEC data's coding & decoding procedure. The packet can be divided into header fields and data fields with the present multimedia data transmission structure. The header fields include data header, sub block header and FEC header. The data header in the Figure 4 is compatible with the standard of the RTP/UDP protocol, and the FEC header is compatible with the related standards of the RFC5109 protocol.

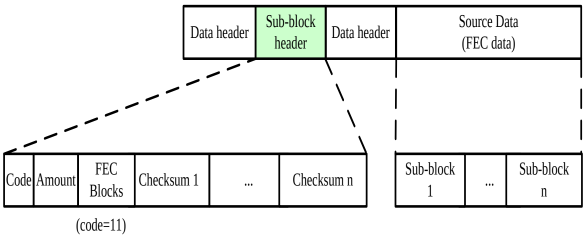

FIGURE 4 The packet format in the sub block

The sub block header can include the content source code segment which identify the composition source of the sub blocks in the packet, the amount segment which record and store the n numbers of the sub blocks in the packet, multi validation and checksum segments which individually record the validation and the value in the 1~n sub block. The validation and the value can be used to check whether the sub block can happen errors. The data fields put multi source sub block information or the FEC sub block information. There are three packet kinds through the sub block coding in terms of the difference of the packet contents: 1. source data, 2. FEC data and 3. Mixed source data with the FEC data. The former two packets can be regarded as the transmission packet assembled by the source sub block and the FEC sub block individually. Code segment can identify the packet types. For example, the Code segment is composed by two bits: the first bit is the source data bit. When the bit is remarked as I, the sub block in the data field includes the source sub block; the second bit is FEC data bit. When the bit is remarked as I, the sub block in the data field includes the FEC sub block. Therefore, when the Code segment is "10", the transmission packet is the source data packet; while the Code segment is "01", the transmission packet is the FEC data packet. When the Code segment is "11", the transmission packet is assembled by the source data packet and the FEC data packet. The Code segment "00" remains unused under the definition. When the Code segment is "11", the Amount segment needs a new extensional FEC blocks segment for remarking the numbers of the FEC data packet.

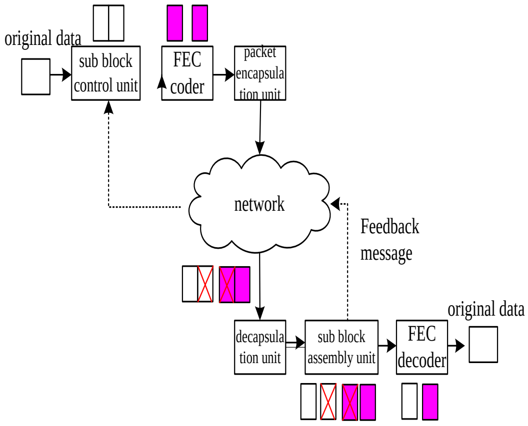

FIGURE 5 System structure and transmission procedure

The display order of the sub blocks in the data field is further defined under the model of the mixture source sub blocks and the FEC sub blocks, that is, the source sub block exists before the FEC sub block. Therefore, the former Amount FEC Blocks in the transmission packet are source sub block and the later FEC Blocks are the FEC sub block. When the Code segment is not "11", the FEC Blocks segment does not exist. The design considers that the mixture packet condition just appears in the convergence of the source sub blocks and the FEC sub blocks. Therefore, the FEC Blocks segment can be regarded as the need utilization for the effective use of the bandwidth.

The Figure 5 is the mechanism's structure and the operational procedure in the data transmission system, including six system units:

1. Sub block control unit: According to the recovery information in the sub block assembly unit, the size of the sub block can be decided and the numbers of the FEC sub block can be calculated.

2. EC coder: Conduct the coding procedure in the sub block.

3. Packet encapsulation unit: Encapsulate the sub block into the transmission packet.

4. De capsulation unit: De capsulate the transmission packet into sub block.

5. Sub block assembly unit: Filter the error sub block, collect the complete receiving sub blocks and recover the statistical information of the receiving data to the sub block control unit.

6. FEC decoder: Conduct the decoding procedure and recover the original data.

Compared with the traditional data transmission system, the mechanism adds the new specific units: Sub block control unit and Sub block assembly unit. The system can set the parameter and the formats in terms of the target application's network environment so that the method can obtain the optimal effect in all network transmission environments. The main calculation is to determine the length of the sub block, the number of the FEC sub block and the needed sub block numbers. The sub block length and the FEC sub block number are calculated by the sub block control unit in reference to the sub block assembly unit, while the packet encapsulation unit and the de capsulation unit are according to the former described packet format. In order to recover the packet loss caused by the network congestion, the FEC coder and decoder can adopt to the common Reed-Solomon (RS) code. As to the RS code, when the sub block happens error, it can be regarded as the sub block loss and combine with the cross packet's transmission model. Therefore, the data recovery in the transmission error and the packet loss can be handled simultaneously.

3 The optimal calculation of the sub block

The optimal calculation method is introduced and the performance verification can be conducted. If the current length of the sub block is L bit and the sub block error rate caused by the bit errors is P_b, the bit error rate is P_b. The Equation (1) is as follows:

$$P_B = 1-(1-P_b)^L \Rightarrow P_b = 1-\sqrt[L]{1-P_B} \tag{1}$$

In the Equation (1), the sub block error rate P_B can be obtained by the recovery information in the sub block assembly unit. According to the bit error rate is P_b, the optimal sub block length can be calculated (L_{opt}) and the present sub block length can be updated. If the needed data header length in the sub block is d bit, the transmission efficiency in the sub block can be defined as E:

$$E = (\frac{L}{L+d})\times(1-P_b)^{L+d}\ . \tag{2}$$

The L should be differential. If the differential result is set as 0, the L_{opt} can be obtained as shown in the Equation (3)

$$L_{opt} = \frac{-d\times In(1-P_b)-\sqrt{-4d\times In(1-P_b)+d^2\times In(1-P_b{}^2)}}{2\times In(1-P_b)}\ . \tag{3}$$

The new sub block error rate P_{B_new} can be obtained with the use of the sub block length L_{opt}. Later. The sub block error rate P_{B_new} caused by the newly obtained sub block length P_b can be predicted.

$$P_{B\_new} = 1-(1-P_b)^{L_{opt}}\ . \tag{4}$$

If the P_{B_new} adds the packet loss rate P_C caused by the packet loss in the sub blocks, the total loss rate P_{total} in all sub blocks can be obtained.

$$P_{total} = P_{B\_new} + P_c\ . \tag{5}$$

In the meantime, the packet loss rate P_C can be obtained from the recovery information in the sub block assembly unit. The recovery condition in the source data is that the numbers of the total sub blocks received from the FEC coder is larger than or equal to the k number in the source sub block. If the number of the present source sub block is k, the expected recovery rate in the source sub block is R. The following Equation can calculate the h number in the FEC sub block and its obtained total sub block number is $(k+h)$.

$$R = \sum_{i=k}^{k+h}\left[(k \underset{i}{+} h)(P_{total})^{k+h-i}(1-P_{total})^i\right]. \tag{6}$$

If the k number of the source sub block is 8, the packet total loss rate P_{total} is 0.2 and the expected data recovery rate R is 0.95, the minimun value of the h number sub block is 5 in terms of the Equation (6).

4. Experimental result and efficient analysis

The validations of the experimental environment are as follows: The packet loss rate of the network is set as 1%, 2% and 3%, the loss number of the average packet is set as 3, and the bit error rate is set as 10^{-1}, 10^{-2}, 10^{-3}, 10^{-4}, 10^{-5} and 10^{-6}. According to the corresponding network from the 1%, 2% and 3% packet loss rate, the utilized bandwidth is 224 KB/s, 145 KB/s, 110 KB/s respectively. The fixed packet length is 1000 bytes, the original data source can be compressed into "Foreman" video whose format is MPEG4 QCIF and its frame per second is 30. The comparative object in the experiment is the traditional cross packet model. The size and the number of the sub block in the mechanism is determined by the above optimal method discussed in the former chapters. In the setting expected recovery rate aspect, the unequal error protection mechanism is adopted in terms of the method in the reference [11]. The important data can be distributed a large number of FEC packet and the unimportant data can be distributed a small number of FEC packet. The negative effects of the data recovery caused by the limited bandwidth can be reduced to the minimum. The efficient parameter adapts to the playable frame rate, and its definition is that if the present playable frame number per second id f, and the playable frame rate is (f/30). Therefore, the higher the playable frame rate is, the more the source data in the video has, namely, the recovery affect in the FEC better. The testing video in the experiment in compressed through the 30 frame numbers per second defined by the National Television Standards Committee, the conversion of the playable frame rate is 1.0.

The four kinds of the experiments are conducted successively. The former three mainly analyzes the difference of the playable frame rate by observing the 1%, 2% and 3% packet loss rate. The Figure 6 - 8 present the three experimental results. The forth experiment counts the total utilization of the bandwidth obtained from the former three experiments whose experimental result is as shown in the Figure 9. The result in the first experiment is as shown in the Figure 6 whose packet loss rate is 1%. The higher playable frame rate under the condition of the transmission bit error which is larger than 10^{-3} can be obtained in the paper. The 10^{-1} bit error usually represents the worst transmission environment so that nearly all transmission packet can happen errors. What's worse, the efficient difference can not be observed. The result in the second experiment is as shown in the Figure 7 whose packet loss rate is 2%. The higher playable frame rate under the condition of the transmission bit error which is larger than 10^{-3}can be obtained in the paper. Compared with the first experiment, the packet loss rate in the transmission environment can be increased and the utilized bandwidth can be reduced. The efficiency provided by the paper under the condition of the 10^{-2} bit error rate can be improved largely, in which the playable frame rate is 0.8 and the frame number per second is 25. In addition, the playable frame rate is 0.1 in the traditional cross packet model and its frame number per second is just 3.

The result in the third experiment is as shown in the Figure 8 whose packet loss rate is 3%. The higher playable frame rate under the condition of the transmission bit error which is larger than 10-4 can be obtained in the paper. The magnitude of the improvement can be increased with the increasing of the bit error rates. The efficiency parameter in the forth experiment is data overhead whose definition is as follows:

Overhead is equal to the packet header data size in all transmission packets plus to the data sizes of all FEC sub blocks.

The lower the Overhead s, the higher the efficiency of using the bandwidth by the FEC has. The Figure 9 shows the differences of the data usage between the two comparative methods. The data sources are from the first experiment to the third experiment, and the packet loss rate is respectively the data usage under the 1%, 2% and 3% packet loss rate.

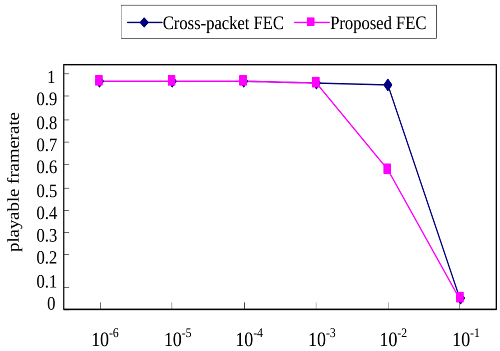

FIGURE 6 The first experimental result: the packet loss rate is 1%

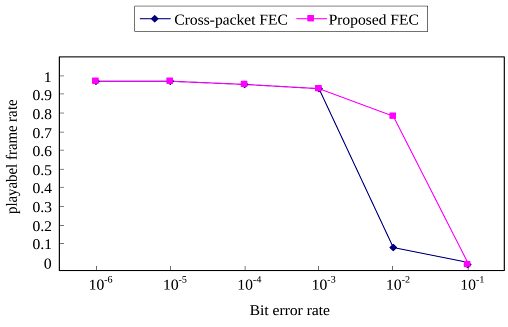

FIGURE 7 The second experimental result: the packet loss rate is 2%

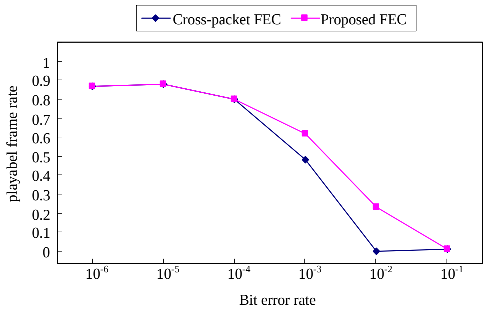

FIGURE 8 the third experimental result: the packet loss rate is 3%

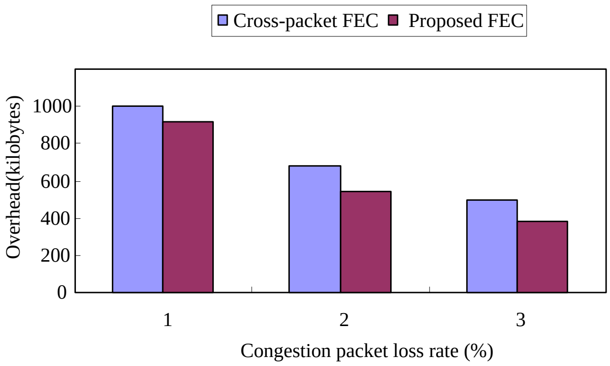

FIGURE 9 the overhead statistical result from the first experiment to the third experiment

Compared with the traditional packet model, the experiment results show that the data usage in the mechanism is lower. In fact, the mechanism can possessed better transmission efficiency in the former experimental results. In addition, the mechanism can adopt the time-saving FEC umber to finish the source data recovery so that the optimal efficiency under the limited bandwidth can be obtained.

5 Conclusion

Aiming to the limited bandwidth, the transmission error and the network congestion suffered from the multimedia data transmission in the wireless mobile network, the cross packet transmission mechanism with the sub block channel coding structure. Combining with the basic concepts between the sub block structure and the cross packet transmission, the data packet format in the application layer is further formatted and the transmission system structure and the optimal calculation in the sub block can be illustrated so that the mechanism in the implement interface is complete. After the comparison and analysis of the related experiments, the mainly efficiency in the mechanism can be applied in the time-saving FEC data sizes for recovering the error data so that the data recovery efficiency in the limited bandwidth can be largely improved. The advantage makes the mechanism be applied in the multimedia application whose bandwidth is limited in the mobile network.

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