ISSN 1000-9825, CODEN RUXUEW
Journal of Software, Vol.18, No.11, November 2007, pp.2882−2892
DOI: 10.1360/jos182882

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结合信源特性与网络拥塞控制的可靠性视频传输算法*

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Reliable Visual Transmission Method Combining Source Statistics and Network Congestion Control

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Xiao S, Wu CK, Zhou YX, Du JC. Reliable visual transmission method combining source statistics and network congestion control. *Journal of Software*, 2007,18(11):2882−2892. http://www.jos.org.cn/1000-9825/18/2882.htm

Abstract: An algorithm combining source statistics and network congestion control for robust video transmission over wireless network is proposed in this paper. Based on scene modeling and characteristic analyzing, all layers generated by scalable coding are classified into several types and two queuing are made respectively according to their contribution to network congestion and to the quality of reconstructed video. Then the source rate, unequal error protection level and congestion control strategy are dynamically adjusted according to different network status in which the packet loss is caused by network congestion or by unreliable transmission in wireless channel. The simulation results show that the proposed method can achieve better results than MPEG-4 video source coding with fixed rate Turbo coding and than the one which dynamically adjusting source coding and channel coding rates and using network congestion control method of selectively dropping *I*, *B* or *P* packets.
Key words: scalable video; joint source channel coding; source statistic; network congestion control; rate compatible punctured Turbo (RCPT) code

摘　要: 提出了一种用于在无线网络中传输视频的结合信源特性及网络拥塞控制的鲁棒性算法.通过场景建模以及特性分析,将分级编码产生的所有码流层划分成不同的类型,并根据它们对网络拥塞控制的贡献以及对重建图像质量的贡献不同,将其分成两个不同的队列.系统根据不同的网络丢包状态(即丢包是由网络拥塞引起还是由无线信道的不可靠传输引起)动态地调整信源速率、不等错误保护强度以及拥塞控制策略.仿真结果表明,该方法与 MPEG-4 信源编码加固定速率 Turbo 码方法以及动态调整信源、信道编码速率加选择性丢 *I*,*B*,*P* 包的网络拥塞控制方法相比,能够提供更好的性能.
关键词: 分级编码;联合信源信道编码;信源特性;网络拥塞控制;速率自适应截短 Turbo(rate compatible punctured Turbo, RCPT)码

* Supported by the National Natural Science Foundation of China under Grant Nos.60702058, 60532060 (国家自然科学基金); the Defense Key Laboratory Foundation of China under Grant No.51434050105DZ0101 (国防重点实验室基金)

Received 2006-03-31; Accepted 2006-08-17

中图法分类号: TP393 文献标识码: A

1 Introduction

Visual communication plays an increasingly important role in information infrastructures. Despite a large number of standards and commercial products for applications of visual communications, providing high quality visual information over resource-limited wireless network still remains a major challenge. This is because of the huge data size of visual information and the complicated characteristics of wireless channels. Due to the limited bandwidth of wireless channels, it is desired that visual information is compressed by highly efficient compression schemes while, due to the time-varying error prone environments of wireless channels, controlled redundancy is necessary to be added in the compressed bit streams in order to ensure reliable transmission. Therefore, there is a tradeoff between efficiency and reliability.

Furthermore, in order to achieve high compression efficiency, most of the standards including JPEG-2000, MPEG-4, H.264 etc. use variable length source coding schemes (VLC), such as Huffman, Lempel-Ziv, and arithmetic coding, which are very sensitive to channel errors. A single error can even blow up the whole scheme. Generally, the higher the compression factor the higher is the sensitivity to channel errors.

On the other hand, in order to achieve high reliability when transmit real-time video over wireless networks, there will be many challenges. Firstly, bandwidth fluctuation is a serious problem. The throughput of a wireless channel may be reduced due to multi-path fading, co-channel interference and noise disturbances. The capacity of a wireless channel may fluctuate with the changing distance between the base station and the mobile host. When a mobile terminal moves between different networks, the available bandwidth may vary drastically. When a handoff takes place, a base station may not have enough unused radio resource to meet the demand of a new joint mobile host. All of the above cases result in serious packet loss at the receiver. Secondly, compared with the wired links, wireless channels are typically much more noisy and have both multi-path and shadow fades, making the bit error rate quite high. The resulting bit errors can have a drastic effect on reconstructing video quality. Therefore, there is a critical need to design a proper scheme for robust video transmission over wireless network.

Two main reasons result in packet loss at the receiver: one is network congestion, and the other is unreliable transmission. Then the methods used to combat the problems can also be divided into two categories: congestion control at network layer and source channel coding control at physical layer.

For congestion control, the methods make management on the data array in router buffer and discard some packets according to the network environment to relieve the network congestion. The approaches to deal with the problem include passive queue management methods such as the congestion control method based on rate[1], the delay-based congestion avoidance mechanism[2] etc. and active queue management methods such as random early detection (RED)[3] and explicit congestion notification (ECN)[4], etc.

For source channel coding control, several methods have been proposed[5−8]. Much of them dynamically allocate the source rate and channel rate according to the channel or network condition, which is called joint source and channel coding (JSCC). Optimal JSCC is usually realized in the form of unequal error protection (UEP). For the recent researches of JSCC with UEP, Source coding mainly focus on scalable video such as MPEG-4 FGS[5] and embedded image such as 3D-SPIHT[6] and JPEG2000[7], while channel coding includes Reed-Solomon, Turbo[7] and LDPC code[8], etc.

However, each category of methods has their own disadvantages. For congestion control methods, they often discard data packets randomly without taking into account of the source characteristics, which will have a drastic effect on the reconstruct video quality because different packets have different contributions to the video quality. For source channel coding control methods, they often adjust the source coding rate and channel coding rate according to the feedback parameters without taking into account that packet loss is resulted by not only unreliable transmission but also network congestion.

More recently, there have been some investigations[9−11] focusing on the joint design of different layers of network to achieve optimal performance of the system. Reference [9] combined congestion control with power control and proposed a jointly optimal algorithm to increase end-to-end throughput and energy efficiency of multi-hop transmissions in wireless multi-hop networks. Reference [10] proposed the methodologies to tackle coupling a non-convexity of the optimization problem and develop a joint congestion control and MAC protocol for ad-hoc wireless networks, which provides a globally optimal solution in a distributed way and accommodates a wide variety of utility functions. Reference [11] designed a joint routing, medium access control (MAC) and congestion control algorithm that stabilizes the buffers, and drives the mean flow rates to a system-wide fair allocation point. Although the above researches carried out joint design of network performance from different points of view, none of them take the source characteristics into considered, which also have a great effect on the network traffic and queuing behavior. As a result, in this paper, a new method combining source statistics and network congestion control for robust video transmission over wireless network is proposed. The method first makes statistic to the scene distribution according to the scene data modeling and find the segmentation point of the scene, assigning different transmission priority to different bit stream after performing scalable coding. During the encoding process, the method dynamically adjusts the source coding rate, channel coding rate and congestion control strategy according to the feedback parameter.

This paper is organized as follows. Section 2 presents the system architecture of the proposed method and describes the new method in detail. In Section 3, the performance of the proposed scheme is examined compared to other algorithms. The conclusion of the paper is presented in Section 4.

2 System Architecture

2.1 System overview

The system architecture is shown in Fig.1.

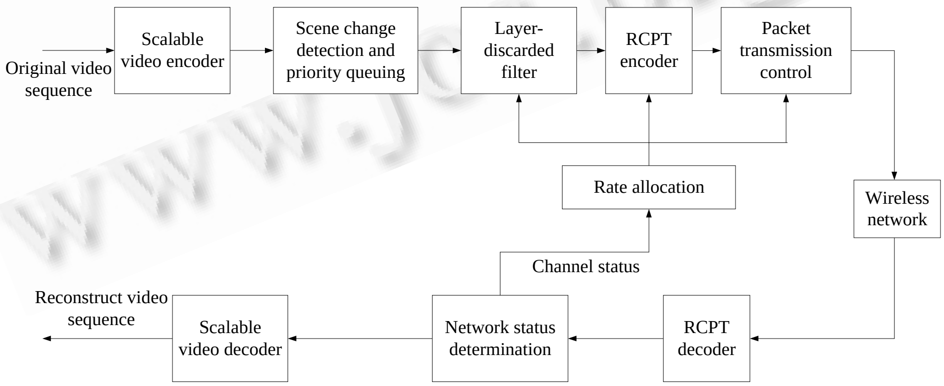

Fig.1 The system architecture of the proposed method

The original video sequences are firstly coded into base layer (BL) bit streams which contains basic information to reconstruct a coarse resolution of source and enhancement layer (EL) bit streams which contains more detailed information to improve the reconstructed quality, and then pass through scene modeling and priority queuing module. Based on scene modeling and characteristic analyzing, all layers are classified into several types and two queues are made respectively according to their contribution to network congestion and to the quality of reconstructed video. Secondly, all layers pass through the layer-discarded filter, Rate Compatible Punctured Turbo (RCPT) encoder and packet transmission controller module in turn under the control of rate allocation module, and sent through wireless network. In the whole procedure, the network status determination module at the receiver estimates the network status from packet loss rate and provides status information to the rate allocation module from feedback channel. Then rate allocation module will adjust source rate, RCPT rate and packet transmission strategy according to the feedback information and round-trip time.

2.2 Scene modeling and distribution statistic

The characteristic analyzing and modeling to the video streaming data is the foundation of studying their transmission performance in different networks. Previous researches[12] have found that the video streaming data has both the long-range dependence property in a wide range of successive scenes and the short-range autocorrelations within a scene state. These properties have a great effect on the video streaming transmission performance in the network, and also can help provide an accurate estimation to the queuing performance at a buffering node.

From the viewpoint of contents, a video sequence can be partitioned into many shots, where each video shot represents a meaningful event or a continuous sequence of actions. The boundaries between shots are generally known as scene changes or transitions, and the action of segmenting a video stream into different shots is called the scene-change detection. Scene transitions can be divided into two categories: abrupt transitions and gradual transitions. Gradual transitions include camera movements: panning, tilting, zooming and video editing special effects: fade-in, fade-out, dissolving, wiping. Both these transitions are used in narrative film and video to convey story structure. Abrupt transitions are very easy to detect, as the two successive frames are completely uncorrelated. Gradual transitions are more difficult to detect as the difference between frames corresponding to two successive shots is substantially reduced.

Different shots have different effects on the whole network traffic and queuing behavior. Once there emerge bit accumulations in the system buffer, the queuing length will increase, which will result in packet loss at the node. As a result, we partition the video frames into three categories: those belonging to scene transitions, those tightly related and those between the above two cases. Since the bit streams of most of the video compression standards at present are organized based on groups-of-pictures (GOP), such as MPEG-1, MPEG-2, MPEG-4, H.263, H.261 etc., all the analysis and conclusion in this paper are based on the GOP level. Assuming X_n is the “size” of the n-th GOP (i.e. the number of bits in the n-th GOP), then the partitioning strategy can be described as follows:

Firstly, we divide GOPs into three classes.

$$X_n \in \begin{cases} A,\ T_n > T_{rt} \\ B,\ T_{rb} \le T_n \le T_{rt} \\ C,\ T_n < T_{rb} \end{cases},$$

where $T_n = \dfrac{(X_{n+1} - X_n) - (X_n - X_{n-1})}{\frac{1}{2}(X_{n-1} + X_{n-2})}$. It is used to judge which scene a GOP belongs to. T_{rt} and T_{rb} are two specific thresholds which reflect the similarity degree of GOPs. When T_n is larger than T_{rt} the n-th GOP is the start point of a new scene or the end point of the current scene ($X_n \in A$). When T_n is smaller than T_{rb}, it exists large correlation between the past two GOPs and current GOP ($X_n \in C$).

Secondly, frames belongs to three cases above are partitioned into three categories respectively.

2.3 Estimation and determination of current network status

Since packet loss is resulted by not only network congestion but also unreliable transmission in wireless networks, two parameters can be evaluated at the receiver side to determine the network status, which are packet loss rate P_L and round-trip time (*RTT*). P_L reflects both the unreliable transmission and network congestion situations, while *RTT* mainly reflect the network congestion status[13].

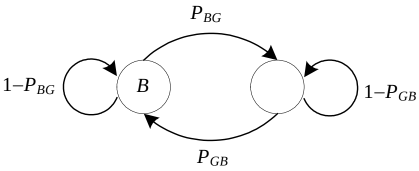

Fig.2 The Gilbert-Elliot model

We use the Gilbert-Elliot model to simulate the packet loss in wireless Internet. As shown in Fig.2, network status is either good (*G*) or bad (*B*). It is defined that P_{BG} and P_{GB} are the transition probabilities between two statuses, then they satisfy the equations $P_{BG}=\frac{n_{BG}}{n_B}$, $P_{GB}=\frac{n_{GB}}{n_G}$, where n_B and n_G denote respectively the number of *B* or *G* status in the observed time series, n_{BG} is the number of times in the observed time series when *G* status follows *B* status and n_{GB} is the number of times when *B* status follows *G* status. So P_L is formulated as follows:

$$P_L=\frac{p_{GB}}{P_{BG}+P_{GB}}=\frac{n_B n_{GB}}{n_B n_{GB}+n_G n_{BG}} \tag{1}$$

The round-trip time can be estimated as follows:

$$RTT=\alpha\times\overline{RTT}+(1-\alpha)\times(now-ST1-\Delta T) \tag{2}$$

where $\overline{RTT}$ denotes the current round trip time, *now* is the time stamp indicating the time at which the packet ACK is received at the sender, α is the weighting parameter that is usually set to 0.875, $ST1$ denotes the time stamp indicating the time at which the packet is sent, and ΔT is the time interval of a packet spent at the receiver side.

So current available network bandwidth W_{cur} can be estimated as follows:

$$W_{cur}=\frac{Const}{RTT+\sqrt{P_L}} \tag{3}$$

where *Const* is a constant that is usually set to either 1.22 or 1.31[14], depending on whether the receiver uses delay acknowledgement.

Assuming the variations of three variables in Eq.(3) are ΔW_{cur}, ΔRTT and ΔP_L respectively, and then there is an equation as shown in Eq.(4).

$$\begin{cases}\Delta W_{cur}^{RTT}=\dfrac{\partial W_{cur}}{\partial RTT}\Delta RTT=-\dfrac{1}{(RTT+\sqrt{P_L})^2}\Delta RTT\\ \Delta W_{cur}^{P_L}=\dfrac{\partial W_{cur}}{\partial P_L}\Delta P_L=-\dfrac{1}{2(RTT+\sqrt{P_L})^2\sqrt{P_L}}\Delta P_L\\ \Delta W_{cur}=\Delta W_{cur}^{RTT}+\Delta W_{cur}^{P_L}\end{cases} \tag{4}$$

where ΔW_{cur}^{RTT} denotes the bandwidth variation due to ΔRTT, and $\Delta W_{cur}^{P_L}$ denotes the bandwidth variation due to ΔP_L. So the rules to determine the current network status are presented as follows:

1) When W_{cur} drops, if $|\Delta W_{cur}^{RTT}|\ge|\Delta W_{cur}^{P_L}|$, then the current network is determined to be in congested situation, if $|\Delta W_{cur}^{RTT}|<|\Delta W_{cur}^{P_L}|$, then the current network is determined to be in unreliable transmission situation.

2) When W_{cur} increases, if $|\Delta W_{cur}^{RTT}|\geq|\Delta W_{cur}^{P_L}|$, then the network congestion is relieved compared with reliable transmission. If $|\Delta W_{cur}^{RTT}|<|\Delta W_{cur}^{P_L}|$, then the current packet loss rate drops down, and the channel transmission condition is better than the previous state.

2.4 Rate allocation combining network congestion control and FEC

As it has been described in Section 2.2, frames in the video sequence are partitioned into three classes *A*, *B* and *C* according to their correlations. According to different coding and prediction modes in the coding algorithm, frames can also fall into three types. i.e., intra frame (*I*), predictive frame (*P*), and bi-directionally predictive frame (*B*). These frames typically occur in deterministic periodic sequence, e.g. …*IBBPBBPBBPBBP*…, which form the GOPs. Since scalable video coding algorithm is concerned in this method, the bit stream of each frame can be further divided into base layer and enhancement layer. Then there are totally eighteen types of layers in our system, and the detail partition relation can be described in Fig.3.

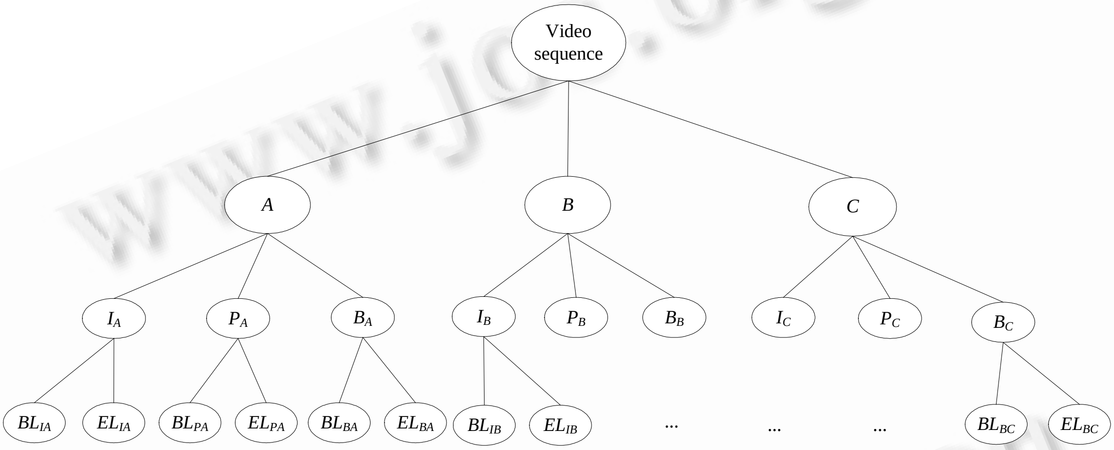

Fig.3 Scalable video bit stream partition

The contribution of each layer to network congestion is different with that to the reconstructed quality. So the method will assign different priority to each layer according to different cases. According to the feedback information that indicates the current network status, the method can adopt different strategies to relieve network congestion or to improve the transmission reliability.

1) According to the contribution to network congestion, six types of transmission priorities ($P_i,i\in[1,6]$) are assigned to all types of layers, which are queuing as follows:

$$\begin{cases} BL_{IC} & BL_{PC} & BL_{BC} & \in P_1 \\ BL_{IB} & BL_{PB} & BL_{BB} & \in P_2 \\ BL_{IA} & BL_{PA} & BL_{BA} & \in P_3 \\ EL_{IC} & EL_{PC} & EL_{BC} & \in P_4 \\ EL_{IB} & EL_{PB} & EL_{BB} & \in P_5 \\ EL_{IA} & EL_{PA} & EL_{BA} & \in P_6 \end{cases}.$$

The transmission priority is decreased from P_1 to P_6.

When the current network is determined to be in congested situation, packets will be discarded to relieve network congestion. The system will discard the packets of layers in the order from P_6 to P_1. Taking into account of the encoding importance of different layers at the same priority level, the system first discards the packets of *B* frame layers, then those of *P* frame layers and last those of I frame layers.

2) According to the contribution to the reconstructed quality, the six types of FEC rates ($R_i, i \in [1,6]$) are assigned to all types of layers, which are queuing as follows:

$$\begin{cases} BL_{IA} & BL_{IB} & BL_{IC} & \in R_1 \\ BL_{PA} & BL_{PB} & BL_{PC} & \in R_2 \\ BL_{BA} & BL_{BB} & BL_{BC} & \in R_3 \\ EL_{IA} & EL_{IB} & EL_{IC} & \in R_4 \\ EL_{PA} & EL_{PB} & EL_{PC} & \in R_5 \\ EL_{BA} & EL_{BB} & EL_{BC} & \in R_6 \end{cases}.$$

The channel coding rate is increased from R_1 to R_6, which means the error protection level decreased from R_1 to R_6.

If the current network is determined to be in unreliable transmission situation, more error protection bits will be allocated to the important source. In order not to increase the burden of network traffic, some relative unimportant packets should be discarded. So the system will gradually increase the protection level of the first five types of layers and discard the packets of the last type of layers. The order of discard is: first, the packets of EL_{BC}, then, those of EL_{BB} and last, those of EL_{BA}.

3 Simulation Results

We use the network simulator (NS) version 2[15] to study the performance of our proposed protocol for MPEG-4 FGS video streaming over wireless Internet. A network topology shown in Fig.4 is used to simulate the Internet traffic. It is assumed that all links except for the bottleneck link are sufficiently provisioned to ensure that network congestion only happens at the bottleneck link. A simple selective ARQ protocol is applied to the wireless link to simulate the variable wireless environment.

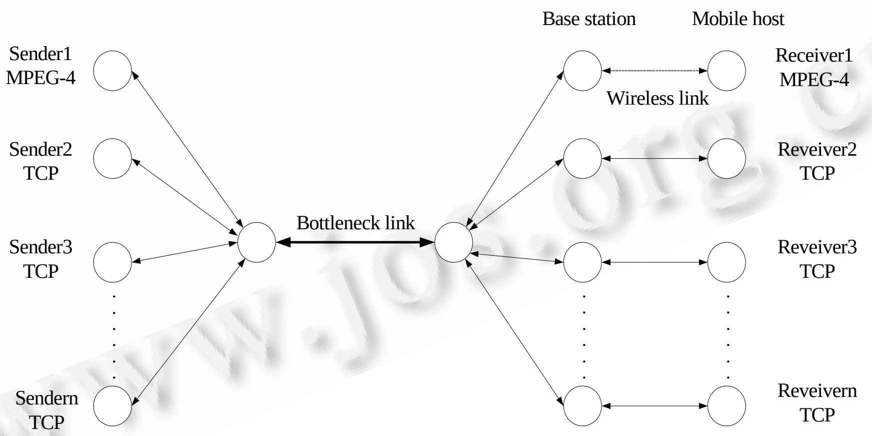

Fig.4 Simulation topology

The following parameters are used for the MPEG-4 encoder: The GOP size 12, the pattern *IBBPBBPBBPBB*, and the quantization values of frames are I=10, P=14 and B=18. In order to simulate the performance of the scheme sufficiently, three kinds of testing video sequences are used. They are *bridge*, *highway* and *Matrix* sequences respectively. *Bridge* is a sequence in which camera is fixed and there exists a large correlation between frames. *Highway* sequence is the scene of camera zooming. *Matrix* sequence is a clip of movie including abrupt transitions. All of the sequences are coded in CIF at a temporal resolution of 30 fps.

In this work, a set of RCPT codes are generated using the algorithm described in Ref.[16] employing two memory M=4 constituent encoders with generator $(1,23/35)_{octal}$, a R_c=1/3 mother code, a puncturing period P=8 and block length of 1024 information bits. The log-MAP algorithm is used for the SISO decoding element. It is assumed that the feed back channel is error free.

We carry out simulation under wireless network by using four methods and comparing their performances: 1) MPEG-4 video source coding; 2) MPEG-4 video source coding with fixed rate Turbo coding which is denoted as MPEG-4+Turbo; 3) dynamically adjusting source coding and channel coding rates and using network congestion control method of the selectively dropping *I*, *B* or *P* packets[17,18], which we call AJSC_NCCIBP; 4) the proposed method combining scene modeling and network congestion controlling which is denoted as AJSC_SCNCCGOP. The difference between method three and our proposed method is that in method three, it makes packet marking in the field of TOS (type of services) at the network level in accordance with the pre-assigned importance of each video frame packet. In the event of network congestion, less important frame packets are dropped before important ones. For example, *B* frame packet is first dropped, then *P* frame packets, at last *I* frame packets. Our proposed method firstly makes priority queue and importance queue respectively according to different contributions of bit stream to network congestion and to reconstruct quality, and then dynamically adjusts the source coding rate, channel coding rate and congestion control strategy according to the feedback parameter of packet loss rate and round-trip time.

Figures 5 to 7 show respectively the performance of the above methods when the average packet loss of network equals to 20%, average bit error rate 0.001, available bandwidth 200Kbps and packet length 1000 Bytes. It can be seen that our proposed method outperforms the other schemes along the whole video sequences under different types of video sequences, and also we can observe that the higher the frequency of abrupt transition, the more efficient our system compared with MPEG-4+Turbo scheme. For *Matrix* sequence, our method can improve about 3.47dB and 7.23 dB in average PSNR respectively compared with AJSC_NCCIBP and MPEG-4+Turbo methods. For *Highway* sequence, it's about 1.8dB and 6.8 dB respectively. While for *Bridge* sequence, it's about 4.92dB and 6.04dB respectively. Since AJSC_NCCIBP method can change source coding rate and protection level according to the current network condition, it outperforms MPEG-4+Turbo method. However, our method outperforms AJSC_NCCIBP method because the latter can not differentiate congestive packet losses form erroneous packet losses so that it unnecessarily reduces its sending rate when packet loss is due to the errors in wireless channel.

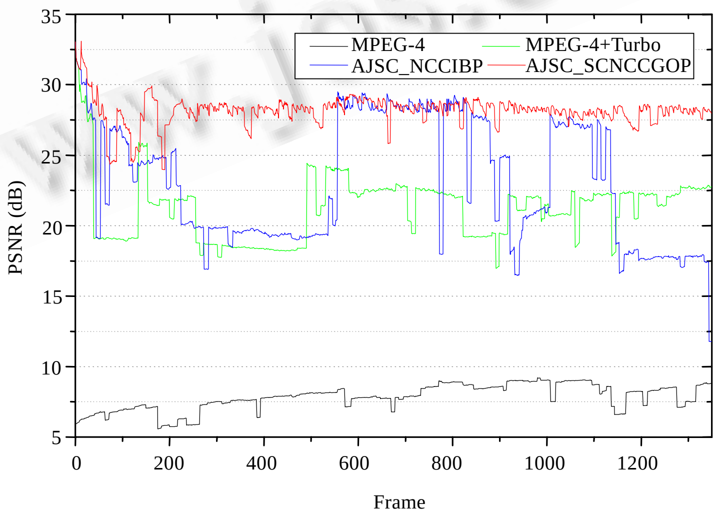

Fig.5 *Bridge* sequence performance

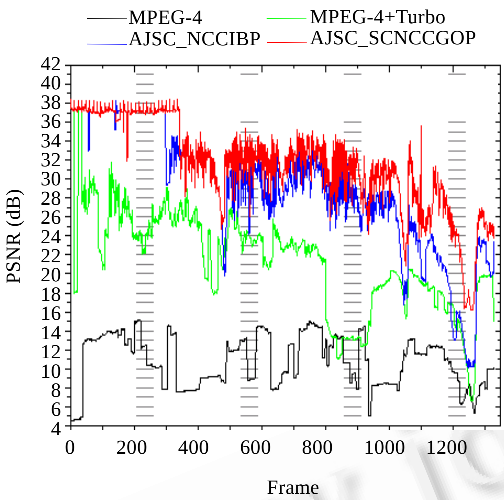

Fig.6 *Highway* sequence performance

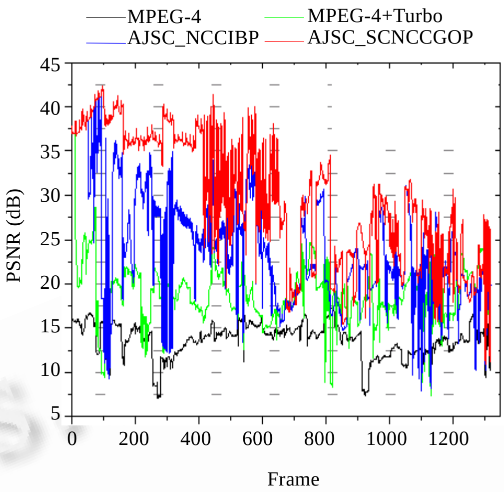

Fig.7 *Matrix* sequence performance

Table 1 shows the comparison results of PSNR for the test sequence *Matrix* under different packet loss rates (PLR) using different schemes. It can be seen that our AJSC_SCNCCGOP method achieves the best performance. Since our method adopts different strategies according to the packet loss patterns and source characteristics, the base layer are less likely to be corrupted during transmission. While for other schemes, base layer may be lost when the loss burst length is longer under the same packet loss rate. It may result in poor quality and sometimes leads to decoder crash.

Table 1 Average PSNR with different PLR (*MATRIX*)

	10%	20%	30%	40%	50%
MPEG-4	14.191	13.718	12.352	11.165	10.951
MPEG-4+Turbo	25.325	23.951	22.156	21.968	21.902
AJSC_NCCIBP	27.879	27.695	26.926	26.642	26.635
AJSC_SCNCCGOP	33.956	31.186	29.327	28.465	27.034

Figures 8 to 19 show respectively the subject decoded quality of MPEG-4, MPEG-4+Turbo, AJSC_NCCIBP and AJSC_SCNCCGOP methods for *Bridge*, *Highway* and *Matrix* sequences under the condition of 20% network packet loss rate and 0.001 channel error rate. Because of the randomicity of error, the decoded quality may be different every time. We just demonstrate here the typical frames which appear with larger probability in 50 times of experiments. They can also reflect the whole situation in the decoding procedure. Obviously we can see that our method can offer the best decoded quality among four schemes.

Fig.8 MPEG-4+noise bridge

Fig.9 MPEG-4+turbo bridge

Fig.10 AJSC_NCCIBP bridge

Fig.11 AJSC_SCNCCGOP bridge

Fig.12 MPEG-4+noise highway

Fig.13 MPEG-4+turbo highway

Fig.14 AJSC_NCCIBP highway

Fig.15 AJSC_SCNCCGOP highway

Fig.16 MPEG-4+noise matrix

Fig.17 MPEG-4+turbo matrix

Fig.18 AJSC_NCCIBP matrix

Fig.19 AJSC_SCNCCGOP matrix

4 Conclusions

This paper addresses the issue of how to effectively and robustly transport scalable video over wireless network. A new method combining source channel coding with scene modeling and network congestion control is proposed in this paper.

The main novelties of this paper can be summarized as follows.

(1) Source channel coding and network congestion controlling are often considered separately in the previous systems, however, in this paper, the system takes both of them into account, which makes rate controlling more reasonable and accurate.

(2) By analyzing the statistic characteristic of coded bit stream, the proposed system makes priority queue and importance queue respectively according to different contributions of bit stream to network congestion and to reconstruct quality, and then applies different control strategy.

The simulation results demonstrate that our proposed method can achieve better results than MPEG-4 video source coding with fixed rate Turbo coding and than the one which dynamically adjusts source coding and channel coding rates and uses network congestion control method of the selectively dropping *I*, *B* or *P* packets.

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