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Improving transmission quality of MPEG video stream by SCTP multi-streaming and differential RED mechanisms

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Abstract SCTP is an emerging transmission protocol with the advanced advantage of supports multi-homing and multi-streaming, which can provide high availability and increase reliability. To improve the performance of MPEG video over Internet, the present study modifies the multi-streaming mechanisms of SCTP, which enable SCTP to use network resources efficiently and provide a differential transmission priority for the encoding frame types of MPEG video stream. Furthermore, to further enhance the decodable frame ratio, the present study also modifies the queue management mechanism of RED which enable the modified RED to provide differential stream protection for MPEG video stream. The simulation results presented in this study confirmed that the proposed scheme can improve the performance of MPEG video stream.

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1 Introduction

Stream Control Transmission Protocol (SCTP) [1] is a new transport layer protocol standardized by IETF which originally designed as a transport protocol for telephone signaling across IP networks. SCTP provides reliable and ordered transmission service similar to TCP, but it operates in a message-oriented fashion like UDP. SCTP inherits most of its functions from TCP (e.g. reliable and ordered transmission service). The main difference between SCTP and TCP is that SCTP allows each association to have multiple streams and can support multi-homing, congestion control and other functions that are similar to TCP, thus it could provide a wider range of applications, especially for Internet multimedia.

MPEG [2] is the most widely applied compression standard for video stream multimedia. Though the overhead of the multimedia data transmitted by UDP is smaller, however, every segment encapsulated from the video stream is seen as independent and unassociated in UDP. It is therefore easy to suffer from the error propagation problem by losing the important frame in a video stream. Though TCP provides reliable data transmission service, however, TCP supports single connection (or stream) and does not provide priority protection mechanism toward important data [3–5], thus transmission delay appears and leads to failure even the data is received.

For providing a better transmission service, SCTP has improved some disadvantages of TCP and adopted some advantages of UDP, and also supported functionalities of multi-streaming and partial reliability. But owing to SCTP continues using most transmission control mechanism adopted by TCP and does not provide any protection for a stream according its priority, when transmitting MPEG-based video stream, severe degradation of video quality due to error propagation is still inevitable.

Although dynamic queue management mechanism like RED [6] can drop packets randomly earlier before the network is full, RED mechanism drops packets without considering importance of data. For video transmission like MPEG, after coding of data, it generates frames assigned with different importance. If the network is full and has to drop packets, it has to consider how to drop packets. If importance of packets is considered, severe degradation of video quality due to error propagation can be avoided [7–9].

To prevent decoding failure of GOP frame caused by error propagation, this study modifies the multi-homing and multi-streaming mechanism of SCTP protocol and proposes a transmission control approach which can adjust transmission dynamically according to the quality of service requirement of an individual frame. When detecting bandwidth decrease or packet loss, SCTP adopts transmission control mechanism with importance grading and adjusts its data transmission rate according to the stream's service quality.

As described above, the media-aware procedure has a significant effect on the performance of MPEG-based applications. Accordingly, this study also proposes a differential RED scheme for MPEG-based video stream to grade the importance of MPEG video streams. With the proposed mechanism, when a router in network receives packets, it dynamically adjusts the drop threshold and packet drop rate of RED queue according to the coding and the priority of the data. This provides differential packet protection and improves degradation of video quality caused by packet loss when displaying video stream on the Internet.

The remainder of this paper is organized as follows. Section 2 provides a brief description of the SCTP protocol and discusses the problem on MPEG-based video stream over transport protocols. Section 3 introduces the modified multi-streaming mechanism and the differential RED mechanism for the SCTP protocol. Section 4 presents the numerical results. Finally, Sect. 5 provides some brief concluding remarks.

2 Background and related works

2.1 SCTP basis

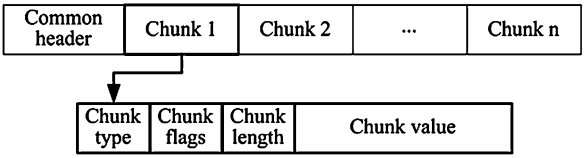
Stream Control Transmission Protocol (SCTP) is a new transport layer protocol with congestion control similar to that used by TCP. The most important enhancements in SCTP are the multi-homing and multi-streaming mechanisms. SCTP introduces the concept of an association that exists between two hosts, but can potentially collaborate with multiple interfaces at each host. Conceptually, an SCTP association is exactly the same as TCP connection, except that SCTP supports multiple streams within an association. All streams within an association are independent and unassociated, but are related to the association. Since a SCTP allows packets to be switched among multiple streams in an association. This powerful mechanism allows SCTP to provide increased availability and reliability.

Like TCP, a SCTP association must be established before delivering data segment. But SCTP provides not only reliable transmission service, but also partial reliable (configurable reliability from fully reliable to send once only), unreliable, ordered, partial ordered and unordered transmission service. In SCTP, if the partial reliability (PR) mode is not started, SCTP provides reliable transmission service equivalent to TCP, but under PR mode, SCTP provides partially reliable transmission service as in UDP. When transmitting the packet, PR-SCTP designates the lifetime of the packet; if the packet is not successfully transmitted within the scheduled time, SCTP would give up transmission of the packet and use FORWARD-TSN packets to notify the endpoint to omit this packet.

Data transmission in SCTP is message-oriented. As opposed to the byte-stream sequence number used in TCP, SCTP uses transmission sequence number (TSN) to govern the transmission of messages and the detection of message loss. The SCTP protocol bases the congestion control on TCP congestion control principles and uses SACK extensions for reception of acknowledgment at the receiver side. However, congestion controls of standard SCTP are applied to the entire association (RFC 2960) no matter that it uses multiple source/destination IP addresses or multiple paths. Hence, this kind of per-association congestion control limits the performance of SCTP as multiple paths are involved in an association.

As shown in Fig. 1, the SCTP packet is composed of a common header and a series of chunks. The chunks included in the SCTP packet may be control chunks

Fig. 1 SCTP packet format



or data chunks. Data chunks contain user messages, while control chunks contain control information. Each chunk begins with a type field, which is used to distinguish between data chunks and various types of control chunks followed by chunk specific flags and a chunk length field. In addition, for reliability and congestion control, each data chunk is assigned with a unique TSN, while stream identifier and stream sequence number are used to determine the delivery sequence of received data for multi-streaming support [10].

To reduce the overhead by packet header, SCTP binds several data chunks or control chunks in a packet. Basically, the amount of chunks is not limited, and the packet size only has to be limited under maximum transmission unit (MTU). Furthermore, the heartbeat chunks are sent periodically to all idle destinations (i.e. alternate addresses), and a counter is maintained on the number of heartbeats sent to an inactive destination without receipt of a corresponding heartbeat ACK. When this counter exceeds a configured maximum that destination address is also declared inactive.

2.2 Problem on MPEG video stream over transport protocols

MPEG (Motion Picture Experts Group) is the most widely applied video and audio compression standard of the Internet. Due to continuous video frames are very similar, pixel value of neighbor frames are usually similar. When transmitting video frames, MPEG estimates variation of neighbor frames and processes according to these variations, deleting familiar and repeating frame, compressing video frame, and finally, after coding via VLC (variable length coding) and combining with dynamic motion, it generates compressed video coding.

In MPEG coding, the coded video stream may be classified into three frames: I-frame, P-frame, and B-frame. I-Frame refers to the coded frame data, not referring to other frames. P-frame refers to previously coded I-frame or P-frame and its data to code. B-frame refers to previous and later I-frame or P-frame and its data being coded.

Generally, MPEG video can be divided into several GOP to code. As illustrated in Fig. 2, a GOP may be represented as $G(N, M)$, where N denotes the number of the frames from one I-frame to next I-frame, M denotes the number of the frames from one I-frame to next P-frame or one P-frame to next P-frame. For example, $G(12, 3)$ represents that a GOP contains 1 I-frame, 3 P-frames, and 8 B-frames.

In an I-frame of GOP, if packets of this I-frame are received, this I-frame is considered decodable. When packets in a P-frame of GOP are correctly received and its previously referred I-frame or P-frame can be correctly decoded, this P-frame is considered decodable. Finally, when packets in a B-frame of GOP can be correctly received and its previously referred B-frame and previous and later referred I-frame or P-frame can be correctly decoded, this B-frame is considered decodable. Thus,

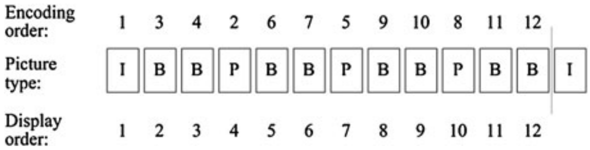


Fig. 2 An illustration of MPEG GOP ($N = 12$, $M = 3$)

when packets of a frame and packets of referred frame can be received correctly, this frame is considered decodable.

The “Decodable Frame Ratio” [11] is decodable frames ratio in all transmitted frames in a video. In other words, when the receiver side completely receives all packets, they belong to a certain frame. Not only does this frame need to be directly decodable, but also the frame referred by this frame has to be decodable. For P-frame and B-frame, owing to it needs to consider decoding of other frames, especially B-frame; it has to consider more frames than the P-frame. Hence, on the importance of coding/decoding, I-frame requires the highest, P-frame requires less, and B-frame requires the lowest.

From Fig. 2, it is known that in hierarchical coding, P-frame decodes according to I-frame content, while B-frame decodes according to I-frame or P-frame. For this reason, in a GOP, if an I-frame packet is lost during transmission and leads to decoding failure of the receiver side, the later P-frame and B-frame are not able to be decoded and causes invalid frames of GOP, leading to severe degradation of video quality.

The present MPEG-based multimedia classifies the coded video stream into three layers: I, P, and B. When a mistake happens in a transport layer, packet loss affects not only the present frame, but also the later referred frame because frame generated by coding may be referred by other later frames. If there is any mistake in an important frame (e.g. I-Frame), video quality after decoding of later frame referred to it might become worse. In other words, a mistake in a previous frame may lead to severe degradation of video quality of other frames. This phenomenon is called error propagation.

Though TCP has a concept of connection, it is not suitable to be used in video stream transmission (TCP only supports single connection, not multiple stream). Unfortunately, though multimedia data adopting UDP has smaller overhead, UDP does not have concept of stream or connection. To end-to-end host of UDP, every datagram is independent and unassociated. In other words, the present TCP/UDP protocol does not have a concept of multiple stream and does not provide priority protection for important data. Though SCTP has improved the partial disadvantage of TCP and adopted partial advantage of UDP, and also has multiple stream, owing to it inherits most of its functions [12, 13] from TCP and does not have importance grading towards data. Thus, when adopting SCTP protocol, error propagation is still inevitable.

To prevent loss of important frame leading to error propagation during displaying, the present study modifies partial transmission control mechanism of SCTP and proposes a transmission control approach that can dynamically adjust transmission policy according to the requirement of individual frame. When detecting available bandwidth decrease or packet loss, this modified SCTP applies retransmission mechanism with importance grading to retransmit the lost packet and adjust transmission rate.

2.3 Related studies

MPEG is the most widely applied video stream standard. To confirm whether transmission protocol meets QoS of MPEG-4, [14] compares transmission effectiveness of IEEE 802.11 WLANs UDP, DCCP, and SCTP. In [15], effect of the SCTP multi-streaming mechanism on transmission effectiveness of FTP application is explored. The results show that the multi-streaming mechanism is beneficial for eliminating transmission delay and enhancing transmission effectiveness of FTP application.

Owing to SCTP is designed based on TCP, the problem appears during transmission. For example, loss recovery mechanism of packet loss of SCTP is exactly the same as TCP. A signal transmission has a strict requirement on transmission time, and the loss recovery mechanism without requiring transmission time like TCP may easily lose packets. To improve loss recovery mechanism of SCTP, a packet-based early retransmit algorithm (PBERA) is proposed in [16]. The empirical results show that under specific conditions, 62% of loss recovery time of SCTP can be reduced.

Due to partial reliability mechanism of SCTP allows packets to assign the priority and retransmission times, it is suitable for multimedia application. However, the additive-increase and multiplicative-decrease algorithm applied in SCTP is not suitable for being applied in multimedia transmission. Thus, in [17], a scheme called “Binomial congestion control” is proposed to solve this problem.

In [18, 19], redundant information of FEC is dynamically adjusted to adapt to Internet conditions and reduce degradation of transmission quality caused by packet loss, and source utilizes information provided by receiver to estimate redundant information, adding these redundant information to original data. Nevertheless, these response protocols change with Internet conditions, especially in the wireless network; the information transmitted by receiver side to server side may be non-real time. This causes accumulated redundant information and becomes a burden for Internet and softwares.

PTCP [20] establishes several TCP connections and congestion control governed by TCP-virtual for application. PTCP governs real data buffering and distribution of data transmission connection. When transmitting data, PTCP selects a connection without congestion to transmit and assign an available connection to retransmit when facing timeout. The advantage of this approach is that it prevents from repeating to implement buffering and acknowledgment mechanisms in application layer.

3 Description of the proposed method

3.1 Extension to SCTP multi-streaming mechanism

Owing to bandwidth of Internet is limited, it is necessary to make compressed encoding before transmission. Because I-frame, P-frame, and B-frame have different importance; this study modifies transmission mechanism of SCTP. Through the new module, SCTP protocol is equipped with the ability of distinguishing priority of packets. As illustrated in Fig. 3, when video stream enters transport layer, the extended SCTP module classifies the video stream into three patterns: Important I-frame is

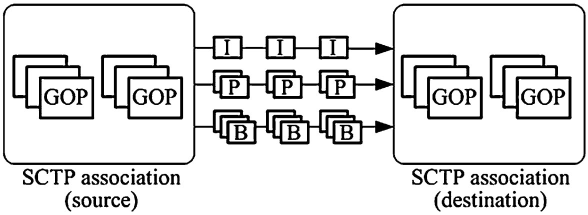


Fig. 3 Use of MPEG video streams within a SCTP association

classified to stream with the highest priority (called I stream); P-frame is classified to P stream with minor priority. Finally, if the received data is B-frame, then the generated data chunk is forward to B stream. The transmission order is the same as the display order.

The modified SCTP updates the size of the congestion window for each ACK received in accordance with the following function:

Receive ACK from SCTP destination:
Receipt of new ACK:
If $W < W_t$, set $W = W + 1$; Slow-Start phase.
Else set $W = 1 + 1/W$; Congestion Avoidance phase.
Adjust transmission rate by bandwidth estimation module
I-frame $\rightarrow$ I stream (high priority)
P-frame $\rightarrow$ P stream (medium priority)
B-frame $\rightarrow$ B stream (low priority)
Receipt of SACK:
Record the *TSN* number, and then retransmit the lost packet with differential priority.

When congestion window (W) is less than slow-start threshold (W_t), STCP is said to be in its slow-start phase and the size of the congestion window is increased exponentially. However, when the threshold limit is reached, STCP enters the congestion avoidance phase and increases the congestion window linearly. When detecting packet loss, the modified SCTP provides differential retransmission mechanism, I-frame is retransmitted firstly, while the P-frame is later. Priority of the B-frame is the same as that of the general data frame.

The bandwidth estimation mechanism has a significant effect on SCTP performance because SCTP sources receive no support from the network that would enable them to identify the link capacity of a bottleneck on the path to their destinations. Therefore, an additional mechanism is required with the capability of establishing an appropriate initial value of a slow-start threshold. Accordingly, this study develops a bandwidth estimation algorithm to estimate bandwidth currently available for SCTP sources. The details of the bandwidth estimation algorithm are described in the following.

Let μ denote the observed transmission rate of a stream within the association from source to destination. If a queue gradually accumulates at the bottleneck, the correlation between packets in-flight along the connection path and the window size during a measured round-trip period is given by:

$$\mu = (w - \rho)/t \tag{1}$$

where t denotes the measured minimal round-trip delay time. w and ρ refers to congestion window and the observed buffer size at the bottleneck, respectively.

Let μ' denote the measured bandwidth. Based on the Packet pair scheme [21], measured bandwidth μ' can be obtained by dividing the number of ACK packets by the sum of ACK inter-arrival times. In the proposed startup algorithm, the estimated bandwidth per round-trip time interval is given by:

$$\lambda = \nu \cdot \mu + (1 - \nu) \cdot \mu' \tag{2}$$

where ν is the ratio of the time required to transmit W packets to the measured *RTT* round-trip. In (2), estimated bandwidth λ varies with the bandwidth utilization of the SCTP stream in the measured round-trip. At the beginning of the startup procedure, the SCTP stream commences with small w and ν values, thus the estimated bandwidth is governed by the measured bandwidth μ'. However, when the sender bandwidth utilization achieves a higher value (indicated by ν), the expected bandwidth μ dominates.

3.2 Differential RED mechanism

RED is a queue management mechanism widely applied in Internet. The main difference between it and conventional Drop-tail is RED starts to drop the received packets randomly before the queue is full. This can control flow of source with congestion control like SCTP or TCP and prevent congestion. Average queue length of RED is calculated by the Exponentially Weighted Moving Average (EWMA) method. The equation is shown below:

$$q_{\text{avg}} = (1 - w) \times q_{\text{avg}} + w \times q \tag{3}$$

In this equation, q_{avg} is the average queue length, q is the present queue length, and w is the weighing factor of the present queue length with value ranged from 0 to 1.

According to (3), after calculating average queue length, q_{avg} is compared with minimum threshold (min_{th}) and maximum threshold (max_{th}). When average queue length is lower than minimum threshold, it means transmission condition of Internet is good, so the packet drop rate is 0 and all received packets can enter the queue. Along with increasing of the queue length, when value of q_{avg} is between min_{th} and max_{th}, packets are dropped randomly. And value of packet drop rate P_b increased with the queue length. P_b is calculated as below:

$$p_b = \max_p \times \frac{q_{\text{avg}} - \text{min}_{\text{th}}}{\text{max}_{\text{th}} - \text{min}_{\text{th}}} \tag{4}$$

In this equation, $\max_p$ denotes preset maximum packet drop rate. When q_{avg} is greater than max_{th}, value of the packet drop rate equals 1. At this time, all packets entering the queue will be dropped as shown in Fig. 4.

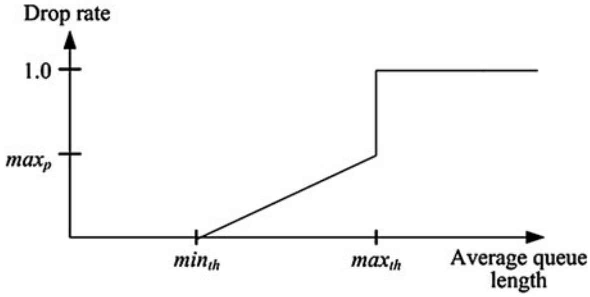


Fig. 4 Variation of packet drop rate of RED queue management mechanism

Though queue management mechanism like RED can randomly drop packets before the queue is full, this mechanism does not consider importance of the data. For transmission of video like MPEG, after coding, data will be graded as frames with different importance. If a network is full and has to drop packets, it has to drop packets according to the importance grading. If importance of packets is considered, and packets with lower importance are dropped first, error propagation due to failure of decoding frame caused by loss of important packet loss can be avoided.

When packets enter the routers, the modified differential RED mechanism will identify the packets according to algorithm procedure. First, the system will identify whether this packet is video packet according to the stream ID of the data chunk. If it is not a video packet, it will use the standard RED mechanism to process the packets; on the contrary, if it is a video packet, it will perform differential RED parameter settings ($\mathrm{min}_{\mathrm{th}}$, $\mathrm{max}_{\mathrm{th}}$, max_p) according to the original video pattern (I-frame, P-frame, or B-frame).

To provide differential transmission priority, threshold of the three data streaming varies with the coding importance. Because I-frame is the most referred, it has the frame with the highest importance and the highest threshold. It means it is not dropped easily by the modified differential RED queue during transmission; though P-frame is referred when decoding, its reference is lower than that of the I-frame, thus its threshold is lower than the I-frame; finally, because effect on B-frame dropping is low, its threshold is the lowest, thus it will be dropped by the queue first during transmission. When a differential parameter is set, the system will decide to drop packets or retransmit packets according to q_{avg} and P_b.

4 Numerical results

The performance of the modified multi-streaming scheme was evaluated using the *NS-2* network simulator [22]. The discussions commence with a simple network model shown in Fig. 5. In this model, each link is labeled with its corresponding bandwidth and propagation delay. Note that in the following simulations, the link capacity, propagation delay, and its packet error rate may be re-configured to investigate different evaluation scenarios.

Table 1 shows the variation of the SCTP goodput with the packet error rate (PER). The round-trip delay time is 20 ms and the bottleneck bandwidth is 155 Mbps. It is observed that the throughput achieved by the modified SCTP (mSCTP) is greater

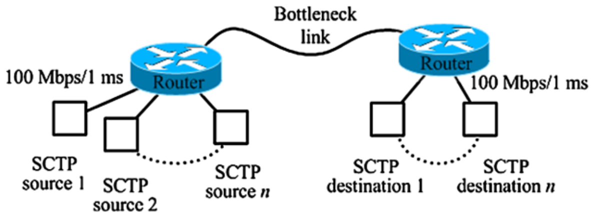


Fig. 5 Simulation model

Table 1 SCTP goodput with different packet error rate (RTT = 8 ms, bottleneck bandwidth 150 Mpbs)

Packet error rate	Goodput (Mbps)	
	SCTP	Modified SCTP
2.0×10^{-4}	76.72	81.52
2.5×10^{-4}	73.84	80.55
3.0×10^{-4}	70.29	79.28
3.5×10^{-4}	67.26	77.96
4.0×10^{-4}	66.15	76.68
4.5×10^{-4}	63.62	75.27
5.0×10^{-4}	61.49	75.02
5.5×10^{-4}	59.59	73.59
6.0×10^{-4}	57.25	72.85
6.5×10^{-4}	55.65	71.56

than that of the original SCTP. The reason for this is that the proposed per-stream differential mechanism and bandwidth estimation algorithm starts each round-trip by filling the bit pipe to its capacity, whereas conventional SCTP simply applies to the default threshold value and blindly halves the window size in the event of congestion.

From the discussions above, it is clear that modified SCTP provides an effective scheme for quickly retransmitting the important stream, thereby eliminating timeout and long idle time periods. Compared with conventional scheme, the modified SCTP significantly improves the end-to-end performance based on the proposed bandwidth estimation mechanism and allows SCTP to achieve an acceptable level of bandwidth utilization. Since the applied per-stream error recovery procedure and the bandwidth estimation mechanism operate at the transport layer level, it provides a viable incremental deployment for enhancing SCTP performance without MAC layer involvement.

Table 2 shows the variation of the SCTP goodput with various bottleneck link capacities. In this simulation configuration, the packet error rate is 1.5×10^{-4} and the minimal round-trip delay time is 8 ms. As shown in Table 2, when the bottleneck bandwidth increases, the modified SCTP increases its goodput to a level significantly higher than that of the conventional SCTP. It is because the sender can re-transmit the lost packets by per-stream retransmission scheme and scale to available bandwidth by the equipped bandwidth estimated mechanism.

Table 3 shows variations of the SCTP goodput act as a function of RTT for a conventional SCTP scheme and the modified SCTP scheme, respectively. The packet error rate is 1.5×10^{-4} and the round-trip delay time is 10 ms. The simulation results

Table 2 SCTP goodput with different bottleneck bandwidth (PER = 1.5×10^{-4}, RTT = 10 ms)

Bandwidth (Mbps)	Goodput (Mbps)	
	SCTP	Modified SCTP
80	55.04	61.03
85	66.14	70.52
90	69.42	73.52
95	73.93	75.37
100	73.44	81.85
105	77.34	81.28
110	77.35	82.52
115	76.67	82.46
120	79.01	82.19
125	79.93	83.60

Table 3 SCTP goodput with different round-trip delay (PER = 1.5×10^{-4}, Bottleneck = 15 Mbps)

RTT (ms)	Goodput (Mbps)	
	SCTP	Modified SCTP
5	86.30	87.28
10	73.04	80.79
15	58.49	77.51
20	44.65	75.70
25	37.17	72.60
30	29.75	70.87
35	24.35	69.60
40	20.73	64.51
45	18.82	56.41
50	16.43	53.41

show that, the goodput of both SCTP and mSCTP procedures decrease as the round-trip delay time increases because the SCTP sources reduce the transmission rates as ACK packets begin to return more slowly. It is also observed that the conventional SCTP scheme suffers from severe performance degradation as the round-trip delay time continues to increase. However, the performance of the m-SCTP scheme is far better due to efficient startup procedures.

Figure 6 shows emulation model after being equipped with differential RED. As the above-mentioned, after coding, video frame of MPEG-4 streaming generates frame with various importance (I-frame, P-frame, and B-frame). The modified SCTP will add different tags to label coding information of the packet before transmission and then, transmits them to the network. When the packets pass through the router, differential RED queue mechanism will grade the importance of the packets according to the information of the packet header and perform queue management according to the set value of the corresponding parameter. Finally, receiver side will receive video packets and decode video streaming data.

Fig. 6 Simulation/emulation model (with differential RED)

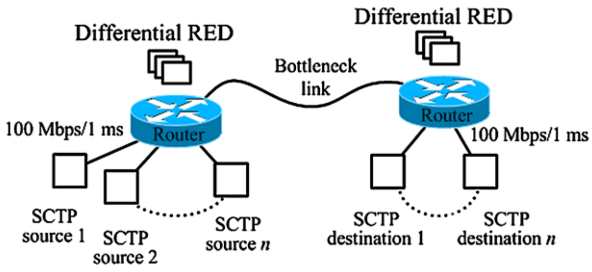


Table 4 Number of the test video frame

Frame number	I_{frame}	P_{frame}	B_{frame}
400	45	89	266

Table 5 Number of the test video packet

Packet number	I_{packet}	P_{packet}	B_{packet}
659	237	149	273

Table 6 Parameter configuration of differential RED

Data type	$\min_{\text{th}}$	$\max_{\text{th}}$	$\max_p$
I-frame	45	50	0.1
P-frame	15	30	0.1
B-frame	5	15	0.1

To verify whether the proposed differential RED mechanism improves video frame quality, the following experiment will utilize video trace file to compare performance of Drop-Tail, RED and differential RED. The test video sequence is Foreman [23]. The format of the test video frame is QCIF (176×144 pixel), and the maximal packet size is 1000 bytes. Tables 4 and 5 are the number of video frame and video packet, respectively.

Table 6 shows the threshold and drop rate configuration in differential RED. In performance evaluation, the decodable frame ratio and peak signal noise ratio (PSNR) are adopted as the metric of performance evaluation. Furthermore, the present study also utilizes YUV trace file to observe video frame quality of the real video streaming.

Table 7 shows the estimated packet drop rate when using Drop-Tail and RED queue management mechanism, where P_I, P_P, P_B are the packet drop rate of I-frame, P-frame, and B-frame, respectively. # SCTP indicates the number of SCTP association. The packet drop probability is increased with the number of SCTP associations during transmission. As shown in Table 7, the packet drop rate of different frame types in Drop-Tail and RED both approach the ratio of packets in the test video (I-frame: 36%, P-frame: 23%, B-frame: 41%) due to the reason of the less consideration of the priority in video frame.

Table 7 Packet drop rate (Drop-Tail and RED)

# SCTP	Drop-tail			RED		
	P_I	P_P	P_B	P_I	P_P	P_B
5	32%	12%	56%	37%	21%	42%
10	27%	16%	56%	28%	27%	45%
15	29%	17%	54%	34%	24%	42%
20	32%	14%	54%	34%	24%	42%
25	30%	17%	54%	33%	21%	45%
30	35%	14%	52%	33%	24%	43%

Table 8 Packet drop rate (with differential RED)

# SCTP	**Differential RED**		
	P_I	P_P	P_B
5	28%	15%	57%
10	21%	24%	55%
15	23%	23%	54%
20	18%	17%	65%
25	18%	15%	67%
30	18%	13%	69%

Table 9 Comparison of decodable frame ratio

# SCTP	Drop-tail	RED	Differential RED
5	0.90	0.78	0.96
10	0.85	0.69	0.95
15	0.79	0.62	0.93
20	0.76	0.61	0.92
25	0.74	0.55	0.91
30	0.71	0.49	0.92

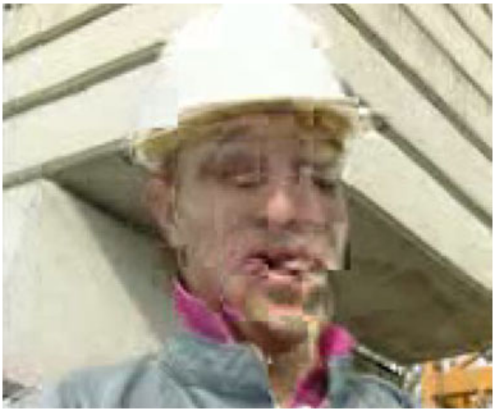
From the results shown in Table 8, the differential RED sets transmission priority according to coding importance of the frames. Thus, when congestion occurs and needs to drop packets selectively, packet such as B-frame will be dropped first and P-frame next. I-frame packet has higher protection priority, and thus its packet drop rate is reduced.

Table 9 shows decodable frame ratio. It is significant that decodable frame ratio obtained in differential RED is better than that in Drop-Tail and RED because differential RED mechanism provides different priority protection according to coding importance setting of video data, especially when amount of data flow increases, decodable frame ratio obtained by using drop-tail or RED drops significantly, however, with the proposed differential RED scheme, it still maintains above a 90% decodable frame ratio. Similar results can be found in comparison of PSNR (see Table 10).

Finally, Fig. 7 shows video frame quality when displaying videos by using different queue management mechanisms. Comparing drop-tail, RED and the proposed

Fig. 7 Comparison of the video frame quality (video frame)

(a) Drop-Tail



(b) RED

(c) Differential RED

differential RED mechanism, we found that video frame quality obtained through differential RED mechanism is obviously better than another two queue management mechanisms.

Table 10 Comparison of PSNR

# SCTP	Drop-tail	RED	Differential RED
5	33.50	32.02	34.19
10	32.91	30.88	33.97
15	32.13	29.84	33.73
20	31.76	29.47	33.73
25	31.43	28.81	33.69
30	31.19	27.65	33.69

5 Conclusion

This study combines multi-streaming of SCTP and RED queue management mechanism, proposing a scheme improving MPEG video stream transmission. Through classification of stream, the modified SCTP can provide differential packet retransmission and bandwidth detection service to reduce occurrence of error propagation. Meanwhile, when packets in networks are dropped, the proposed differential RED mechanism will drop video packet with lower importance, enhance decodable frame ratio and improve video transmission quality. In the scenarios considered in this study, it has been shown that the proposed schemes allows SCTP to achieve an effective goodput than regular SCTP and provides better video frame quality.

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