Improving Performance of MPEG-based Stream by SCTP Multi-Streaming Mechanism

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Abstract—**To prevent decoding failure of MPEG-based applications caused by error propagation, the present study modifies multi-streaming mechanism of SCTP protocol and proposes a transmission control approach which can adjust transmission policy dynamically according to the quality of service requirement of individual frame. The simulation results confirmed that the proposed scheme can improve the performance of MPEG-based video stream by modified SCTP multi-streaming mechanism.**

Keywords- SCTP; Multi-streaming; MPEG

I. 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 delivery service similar to TCP, but operates in a message-oriented fashion like UDP. SCTP inherits most of its functions from TCP (e.g. reliable, ordered delivery 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 is the most widely applied compression standard for video stream multimedia. Though overhead of 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 delivery service, however, TCP support single connection (or stream) and does not provide priority protection mechanism toward important data, 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.

To prevent decoding failure of GOP frame caused by error propagation, this study will modify multi-homing and multi-streaming mechanism of SCTP protocol and propose a transmission control approach which can adjust transmission dynamically according to the quality of service requirement of individual frame. While detecting bandwidth decrease or packet loss, SCTP adopts transmission control mechanism with importance grading and adjust 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 proposes a scheme for SCTP to grade the importance of MPEG video streams. With the proposed mechanism, the need for per-flow support in routers is avoided, and the available capability in the SCTP protocol can be fully exploited.

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

II. Background and Related Works

A. 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 end-host 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

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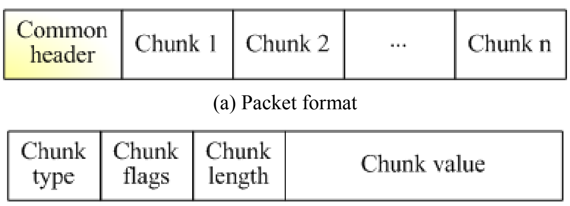
streams within an association are independent and unassociated, but are related to the association. Since a SCTP allows packets to be switched among multiple streams in an association. This powerful mechanism allows SCTP to provide increased availability and reliability.

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 that notify the endpoint to omit this packet.

Data transmission in SCTP is message-oriented. As opposed to 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), whether using 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 Figure 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 or data chunks, where 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 a unique TSN, while stream identifier and stream sequence number are used to determine the delivery sequence of received data for multi-streaming support [2].



(b) Chunk field

Figure 1. SCTP packet format

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.

B. 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, 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 its frame data to code without 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 to code.

Generally, MPEG video can be divided into several GOP to code. As illustrated in Figure 2, a GOP may be represented as $G(N, M)$, where N denotes number of frames from one I-frame to next I-frame, M denotes number of 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.

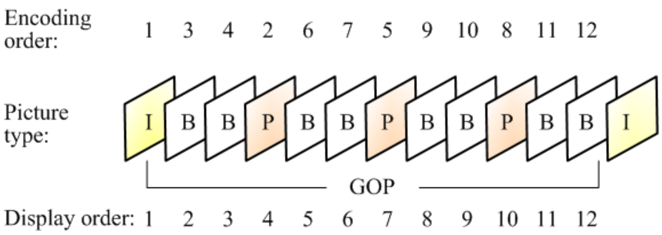


Figure 2. An illustration of MPEG GOP ($N = 12$, $M = 3$)

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

From Figure 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 loses during transmission and leads to decoding failure of receiver side, the later P-frame and B-frame are not able to be decoded and causes invalid frames of GOP, leading to significant degradation of video quality.

The present MPEG-based multimedia classifies display frame into three layers: I, P and B. When mistake happens in a transport layer, packet loss affects not only present frame, but also later referred frame because frame generated by codec 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 previous frame may lead to severe degradation of service quality of other frames. This phenomenon is called error propagation.

Though TCP has 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, present TCP/UDP protocol does not have concept of multiple stream and does not provide priority protection for important data. Though SCTP has improved partial disadvantage of TCP and adopted partial advantage of UDP, and also, has multiple stream, owing to it inherits most of its functions[3] [4] 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 and leads 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 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.

C. Relate Studies

MPEG is the most widely applied video stream standard to confirm which transmission protocol meets Qos of MPEG-4, [5] compares transmission effectiveness of IEEE 802.11 WLANs UDP, DCCP and SCTP. In [6], effect of 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 enhance transmission effectiveness of FTP application.

Owing to SCTP is designed based on TCP, thus problem appears during transmission, for example, loss recovery mechanism of packet loss of SCTP is exactly the same as TCP. When a signal transmission has strict requirement on transmission time, 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 [7]. 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 applying in multimedia transmission. Thus, in [8] a scheme called "Binomial congestion control" is proposed to revise this problem.

In [9] [10], redundant information of FEC is dynamically adjusted to adapt to Internet conditions and reduce degradation of transmission quality caused by packet loss, source utilizes information provided by receiver to estimate redundant information and add these redundant information to original data. Nevertheless, these response protocols change with Internet conditions, especially in wireless network, the information transmitted by receiver side to server side maybe be non-real time. This causes accumulated redundant information and becomes a burden of Internet and softwares.

PTCP [11] 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 it prevents from repeating to implement buffering and acknowledgement mechanisms in application layer.

III. 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 has different importance, this study modify transmission mechanism of SCTP, through the new module, SCTP protocol are equipped with ability of distinguishing priority of packets. As illustrated in Figure 3, when video stream enters transport layer, the extended SCTP module classifies the video stream into three patterns: Important I-frame is 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 display order.

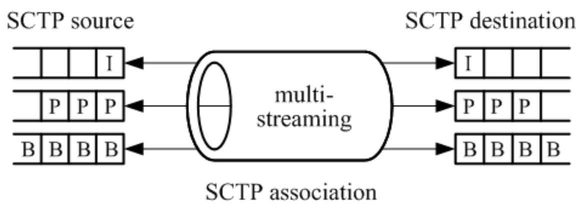


Figure 3. Use of MPEG video streams within a SCTP association

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

Receipt of SACK:

Record the TSN number, and then retransmit the lost with differential service 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 P-frame later. Priority of B-frame is same as that of general data frame.

The bandwidth estimation mechanism has 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 w and ρ refer to congestion window and the observed buffer size at the bottleneck, respectively.

Let μ' denote the measured bandwidth. Based on the Packet pair scheme [12], 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 Eq. (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 govern by measured bandwidth μ'. However, when the sender bandwidth utilization achieves a higher value (indicated by ν), the expected bandwidth μ dominates.

IV. NUMERICAL RESULTS

The performance of the modified multi-streaming scheme was evaluated using the *ns-2* network simulator [13]. The discussions commence with a simple network model shown in Figure 4. 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.

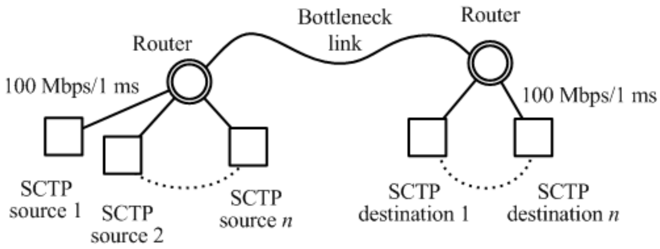


Figure 4. Simple simulation model

Figure 5 is partial results of simulation model, next field of TSN number represents type of encapsulated data, i denote I-frame, p denotes P-frame, and b denotes B-frame. Furthermore, SID (stream ID) denotes sequence number of stream. When processing 4-way handshake, source and destination of SCTP communicate to determine the type (in this study, it is MPEG) of application layer multimedia data. From the example, it is seen, when video stream arrives transport layer, modified SCTP may classify frames to various streams and transmit via multi-streaming mechanism.

```
        + 0.500000 init -home-media aware [mpeg]- addr=1
- 0.540006  2 initack
        + 0.540006 cookieecho addr=1
- 0.580010 11 cookieack
        + 0.580010 data tsn=1 i size=1024 sid=0 addr=1
        + 0.580010 data tsn=2 b size=1024 sid=2 addr=1
        + 0.580010 data tsn=3 b size=1024 sid=2 addr=1
- 0.620125  3 ack tsn=2 b size=16 -rtt- -slowstart-
        + 0.620125 data tsn=4 p size=1024 sid=1 addr=1
        + 0.620125 data tsn=5 b size=1024 sid=2 addr=1
        + 0.620125 data tsn=6 b size=1024 sid=2 addr=1
        + 0.620125 data tsn=7 p size=1024 sid=1 addr=1
- 0.660184  3 ack tsn=4 p size=16 -rtt- -slowstart-
        + 0.660184 data tsn=8 b size=1024 sid=2 addr=1
        + 0.660184 data tsn=9 b size=1024 sid=2 addr=1
        + 0.660184 data tsn=10 p size=1024 sid=1 addr=1
- 0.660297  3 ack tsn=6 b size=16 rate=36.36 -slowstart-
        + 0.660297 data tsn=11 b size=1024 sid=2 addr=1
        + 0.660297 data tsn=12 b size=1024 sid=2 addr=1
```

Figure 5. Snapshot of partly simulation process

Table 1 shows the variation of the SCTP goodput with the packet error rate (PER). The round-trip delay time is 20ms and the bottleneck bandwidth is 155 Mbps. It is observed that the throughput achieved by modified SCTP (mSCTP) is greater 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 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, 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 I. SCTP GOODPUT WITH DIFFERENT PACKET ERROR RATE (RTT = 8ms, bottleneck bandwidth 150 Mpbs)

Packet error rate	Goodput (Mbps) SCTP	Goodput (Mbps) mSCTP
1.5×10^{-4}	78.98	82.90
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
7.0×10^{-4}	53.48	70.81
7.5×10^{-4}	52.16	69.80
8.0×10^{-4}	51.31	69.06

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 increased, the modified SCTP increases its goodput to a level significantly higher than that of the conventional SCTP. It is because the sender can retransmit the lost packets by per-stream retransmission scheme and scale to available bandwidth by the equipped bandwidth estimated mechanism.

TABLE 2. SCTP GOODPUT WITH DIFFERENT BOTTLENECK BANDWIDTH (PER = 1.5×10^{-4}, RTT = 10 ms)

Bandwidth (Mbps)	Goodput (Mbps) SCTP	Goodput (Mbps) mSCTP
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 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 is 10 ms. The simulation results 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.

TABLE 3. SCTP GOODPUT WITH DIFFERENT ROUND-TRIP DELAY (PER = 1.5×10^{-4}, Bottleneck = 150 Mbps).

RTT (ms)	Goodput (Mbps) SCTP	Goodput (Mbps) mSCTP
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

V. CONCLUSION

This paper proposes an extended multi-streaming scheme for MPEG-based video stream over SCTP transport protocol. The objective of the proposed scheme is to provide the differential stream service of SCTP protocol. In the scenarios considered in this study, it has been shown that the extended modification allows SCTP to achieve an effective goodput than regular SCTP. The simulation results also exhibits that the bandwidth estimation mechanism also makes it feasible to extend SCTP for competing available bandwidth.

REFERENCES

[1] R. Stewart et. al., "Stream Control Transmission Protocol," RFC 2960, 2000.

[2] L. Ong, J. Yoakum, "An Introduction to the Stream Control Transmission Protocol (SCTP)," RFC 3286, May 2002.

[3] W. Stevens, TCP Slow-Start Congestion Avoidance, Fast Retransmit, and Fast Recovery Algorithm, RFC 2001, 1997.

[4] V. Paxson, M. Allman, and W. Stevens, TCP Congestion Control, RFC 2581, 1999.

[5] S. Nosheen, S.A. Malik, Y. B. Zikria, M.K Afzal, Performance Evaluation of DCCP and SCTP for MPEG4 Video over Wireless Networks, IEEE INMIC'07, pp. 1-6, Dec. 2007.

[6] S. Ladha, P. D. Amer, Improving Multiple File Transfers Using SCTP Multistreaming, IPCCC'04, 2004.

[7] P. Hurtig and A. Brunstrom. Enhancing SCTP Loss Recovery: An Experimental Evaluation of Early Retransmit. Elsevier Computer Communications, Vol. 31, No. 16, pp. 3778–3788, Oct. 2008.

[8] D. Bansal, H. Balakrishnan. Binomial Congestion Control Algorithms. IEEE INFOCOM'01, Apr. 2001.

[9] K. Park, W. Wang, AFEC: An Adaptive Forward Error Correction Protocol for End-to-End Transport of Real-Time Traffic, IC3N'98, pp. 196 – 205, Oct. 1998.

[10] H. Ma and M. EI Zarki, A New Transport Protocol for Broadcasting/Multicasting MPEG-2 Video over Wireless ATM Access Networks, Wireless Networks Vol. 8: 317 – 380, Jul. 2002.

[11] H.Y. Hsieh and R. Sivakumar, A Transport Layer Approach for Achieving Aggregate Bandwidths on Multi-Homed Mobile Hosts, ACM/Springer Wireless Networks, Vol. 11 (1-2): 99-114, Jan. 2005.

[12] S. Keshav, A Control-Theoretic Approach to Flow Control, ACM SIGCOMM (1991) 3–15.

[13] Network Simulator, NS-2, http://www.isi.edu/nsnam/ns.