國 立 交 通 大 學

資訊科學與工程研究所

碩 士 論 文

針對 H.264/SVC 使用 SCTP 之可靠影音傳輸演算法

A Reliable Video Transmission Algorithm for H.264/SVC Applications over SCTP

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摘要

近年來，在網路上收看各式各樣的電視節目及影片是一件相當普遍的行為。為了要配合異質終端的能力和網路壅塞程度提供不同種解析度的影片，可適性視訊編碼(scalable video coding)使視訊傳輸更能適應在異質的網路環境。為了使影片的播放能更為流暢，避免在撥放途中發生影片中斷，在此篇論文中，我們對可視訊編碼提出一個的資源排程演算法。此演算法基於串流控制傳輸協議(Stream Control Transport Protocol, SCTP)在網路上傳輸可適性視訊編碼串流，並且透過一個機率模型估算對於重要的資訊至少要預留多少重傳的次數才能確保整體影片有一定程度的可靠性，藉此犧牲些微影片解析度換取影片的流暢度。在實驗的部分，我們將提出的演算法透過 SCTP 傳輸並與被廣泛使用的 RTP/UDP 方法和基本的 SCTP 做比較。可由結果看出，與基本的 SCTP 作比較，我們點算法將能夠被撥放影片的比例提升 20%，畫質部分也提升 13.5%。

A Reliable Video Transmission Algorithm for H.264/SVC Applications over SCTP

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Abstract

In recent years, it is prevalent for people to watch television shows and videos on internet. Scalable Video Coding (SVC) comes as a solution that adapts to network congestion and terminals heterogeneity. In this paper, we proposed an adaptive resource scheduling algorithm to send SVC-based videos over Stream Control Transport Protocol (SCTP) in order to play the video smoothly without interruption. We developed a probabilistic model to estimate how much bandwidth should be preserved. The preserved bandwidth is used to retransmit the important information for a reliable video transmission. On the other hand, our approach can mitigate the reduction of video resolution when some bandwidth was reserved. Our transmission scheme is compared with the widely used RTP/UDP method and standard SCTP. Simulation results show that our algorithm has 20% improvement on decodable frame rate and 13.5% improvement on the video quality compared with the standard SCTP.

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Chapter 1
Introduction

Currently UDP employed communications systems for video streaming over IP networks. However the absence of congestion control from UDP can cause performance deterioration [1]. When the network is congested, the UDP sender continues to send packets at a constant rate resulting in a number of unavoidable losses. As the result, these packet losses between the video sender and the receiver significantly degrade the quality of the received signal [2]. Most Streaming solutions today utilize some type of retransmission mechanisms to allow recovery of lost packets. TCP is a transport layer protocol that is widely used. However, the retransmission scheme in TCP is not appropriate for multimedia streaming applications because it can increase the end-to-end delivery latency [3].

To overcome the above limitation, streaming control transport protocol (SCTP) was proposed by IETF SIGTRAN Working Group which was been standardized in RFC 4960 [4]. SCTP is a transport-layer protocol for multimedia streaming. SCTP holds some features of TCP/UDP, such as reliable transmissions and message-oriented boundary. SCTP can provide multi-streaming and multi-homing services for a single connection. SCTP supports non-simultaneous data transfer over multiple paths in multi-homed hosts. SCTP establishes connections between different interface pairs at two end points and routed over paths connecting them. During the normal operation, SCTP always uses at most one path at a time for communication, leaving the remaining paths for packet retransmission and for backup in case of link failure [5] – [7].

In this paper we aim at coupling the Scalable Video Coding (SVC) extension of H.264/MPEG-4 AVC video compression standard [8] which is the most well-known scalable standards over SCTP in order to enable the retransmission-based reliability while consuming the network resource wisely by investigating network congestion in real time. The SVC-based videos are encoded in multiple separate layers to supply different qualities. When network congestion occurs, the sender can adjust the qualities of the sending video to cope with network fluctuation dynamically without re-encoding the video [9]. SVC scheme divides a video stream into one base layer and several enhancement layers. Each enhancement layer is depended on all the lower quality levels. Transmitting more enhancement layers can produce a better output video, but requires more bandwidth. Base layer is the most important part of the video streaming. Without Base layers, the playback would be interrupt and the received enhancement layers are useless which wastes the bandwidth to transmit. So the major problem is how to guarantee the base layer can be received successfully before its playback deadline then the video can play smoothly.

We propose an adaptive source scheduling algorithm “Adaptive Transmission Control over SCTP” (ATC-SCTP) to send SVC-based videos over SCTP. Our system investigates the congestion of network by estimating the drop rate in real time. By calculating the CDF of successful transmission of base layers under the drop rate, we can find the retransmission number for base layers we should guarantee to support the best quality of the video. Then we adjust the number of the enhancement layers to ensure the base layers have enough resource to do retransmission. The remainder of this paper is organized as follows. In Section II, we give a state of the art about the existing related works and background. In section III, we propose our algorithm, including the methods of calculating the retransmission number to guarantee the reliability for base layers and adjusting the quality of sending video to deal with network congestion. Section IV shows the simulation results and finally, Section V concludes the paper.

Chapter 2
Related Work

2.1. The Stream Control Transmission Protocol

The Stream Control Transmission Protocol [10] was designed by the IETF SIGTRAN workgroup and is basically a reliable transport layer protocol that was initially designed for signaling transport. SCTP inherits the best features of TCP (reliability, congestion control) and UDP (message-oriented based). SCTP offers a connection-oriented and reliable communication channel on top of a connectionless-less packet-based network. It combines the benefits of TCP and UDP while cutting their drawbacks and further introduces a set of new features. In the following, a selection of features for our proposed approach is presented.

Message oriented delivery: SCTP is message oriented and preserves boundaries of application-layer messages, similar to UDP, which is substantially different from the byte-stream oriented TCP. Messages are encapsulated in chunks. This feature allows SCTP to decouple reliable delivery from message ordering by introducing the idea of streams.

Multiple logical streams: A stream is an abstraction that allows applications to preserve in-order reliable delivery within a stream, but unordered delivery across streams. In this way, head-of-line (HOL) blocking is avoided at the receiver in case multiple independent data streams are flowing in the same SCTP session.

Congestion control: A major benefit of SCTP is its TCP-friendly congestion control. The SCTP protocol based the congestion control on TCP congestion control principles and uses SACK extensions for reception of acknowledgment at receiver side.

2.2 The Scalable Video Coding Extension of the H.264/AVC Video Coding Standard

Scalability has appeared as a relevant solution that addresses both video coding and network challenges. The term “scalability” refers to the removal of parts of the video bit stream in order to adapt it to the various needs or preferences of end users as well as to varying terminal capabilities or network conditions.

Efficient SVC provides a number of benefits in terms of applications [11] – [13]. For instance, the scenario of a video transmission service with heterogeneous clients, where multiple bit streams of the same source content differing in coded picture size, frame rate, and bit rate should be provided simultaneously. With the application of a properly configured SVC scheme, the source content has to be encoded only once for the highest required resolution and bit rate, resulting in a scalable bit stream from which representations with lower resolution and/or quality can be obtained by discarding selected data.

Another benefit of SVC is that a scalable bit stream usually contains parts with different importance in terms of decode video quality. This property in conjunction with unequal error protection is especially useful in any transmission scenario with unpredictable throughput variations and/or relatively high packet loss rates. Thus, the loss of important transmission units due to congestion can be avoided and the overall error robustness of the video transmission service can be substantially improved.

2.3 Related Works

SCTP is well suited to transmit different kinds of data including encoding video data. However there exists no standard which exploits the advanced features of SCTP for streaming H.264 or other encoded video data. (c.f. RFC 3984 [14] for streaming H.264 over RTP/UDP). In spite of this, there are several approaches for transporting encoded video data via SCTP have been evaluated.

Kim et al. [15] and Sanson et al. [16] use PR-SCTP to transmit H.264/AVC video. For each frame type (e.g. I, P or B), they assigned a different number of retransmissions. In [15], the delay constraint is more strictly for I-frames. I-frames have higher opportunities to be discarded. Although [15] can decrease the amount of transmissions, it causes the video fractionally. [16] develops a simple probabilistic model to assign a limited number of retransmission . [16] calculates the probability of having all n packets from the NAL unit successfully received after m retransmissions and find optimum values for the maximum number of retransmissions that offer the best trade-off between reliability and delay.

Cheng et al. [17] combines multi-streaming of SCTP and RED queue management mechanism which provides a differential retransmission priority for the encoding frame types of MPEG video stream and enable the modified RED to provide differential stream protection foe MPEG video stream.

Quaritsch et al. [18] uses SCTP as transport layer protocol and is designed for H.264 encoded video. They use multiple streams of SCTP features and multiplex the individual NAL units on the streams in a round-robin manner. The approach incorporates limited bandwidth and packet-loss regarding the number of received frames as well as quality of perceived video.

The approach presented in this paper uses the similar concept from the works described above as we try to guarantee the reliability of base layers (as I-frames). But the significantly difference is that we do not proceed from the number of retransmission to which needs to retransmit, we consider the real network environment and preserve the resource for possible retransmissions of base layers in order to archive reliability. And the results in the papers cited above are all found on simulations.

Chapter 3
An Adaptive Solution for SVC Streaming

The main objective of our SVC-streaming algorithm is to keep the perceived video by the end-user as smooth as possible during the entire streaming process. In the other words, the algorithm aims at saving up bandwidth for base layers to do retransmission when it is possible to drop during transmission. Consequently, SVC-streaming algorithm on the server decides to adapt its streaming strategy accordingly. Adaption is done through measuring the drop rate of the path and adjusting the number of the sent enhancement layers that economize on the available resource.

3.1. Drop Rate Estimation

Drop rate of a network path presents the congestion at the path's bottleneck. We use this metric to evaluate our path because it can determine whether a path has enough capability to transfer a video stream with high resolution. In our work, the server estimates the drop rate of the link according to the received SACK at server.

The drop rate sample is obtained on path during the k-th time SACK can therefore be written as

$$\hat{s}^{(k)} = 1 - \frac{the\ number\ of\ \ received\ packets\ of\ SACK^{(k)}}{the\ number\ of\ sent\ packets} \tag{1}$$

Fig. 1 shows the example for calculating $\hat{s}^{(k)}$. We can find that there are 15 packets have been sent and only 11 packets have been received. Then $\hat{s}^{(k)}$ is equal to 0.267 approximately.

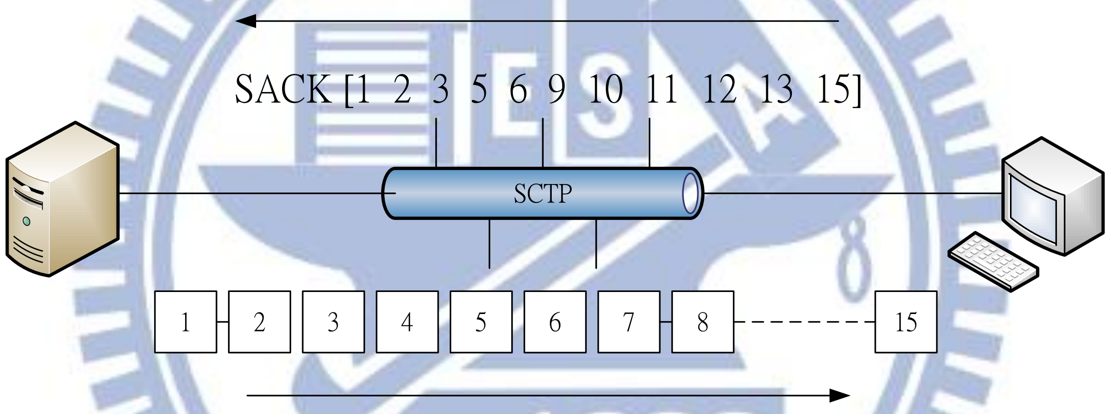

Fig 1. The drop rate calculated by SACK and the number of sent packets

The actual drop rate-estimation and sampling k, $\hat{p}^{(k)}$, is obtained as smoothed exponential average of drop rate estimation and sampling k – 1, $\hat{p}^{(k-1)}$, and the last drop rate sample k, $\hat{s}^{(k)}$, according to [22]

$$\hat{p}^{(k)} = \frac{7}{8}\hat{p}^{(k-1)} + \frac{1}{8}\hat{s}^{(k)} \tag{2}$$

3.2. Adaptive Transmission Control over SCTP (ATC-SCTP)

We suppose that the server is waiting for a client request of a specific video. Client sends the important parameter to the SVC server which is the bandwidth BW of the client for initial data-rate threshold. The parameter is sent only once before the streaming, so sever can use it during the streaming process.

Before starting video streaming, we determine the initial number of the enhancement layers (*ELnum*) that client [19]. For the maximum data-rate that client can receive, *ELnum* must satisfy (3).

$$\mathrm{BW} \geq \delta \times \left(BL_{br} + \sum_{i=1}^{ELnum} EL_{bri} \right), \qquad \delta \geq 1 \tag{3}$$

BL_{br} and EL_{bri} are the bit-rate of base layer and of enhancement layer respectively. δ should be high enough to guarantee the streaming of the video without losses and low enough to allow other applications to share the bandwidth. This initial *ELnum* cannot be exceeded during the streaming process, because it has been chosen based on physical parameter that do not change during the streaming.

After choosing *ELnum*, the algorithm starts to stream the selected layers to the clients. As soon as the video start to send, the sever starts to evaluate the drop rate $\hat{P}$ of received SACK in real time. When the amount of change of drop rate multiplied by BW is reach to the half bit rate of base layer,

$$\mathrm{BW} \times \left|\hat{p}^{(k)} - \hat{p}^{(k-1)}\right| \geq 0.5 \times BL_{br} \tag{4}$$

the server re-calculates the CDF of successful transmission for base layers under drop rate $\hat{P}^{(k)}$ and adjust the number of the sent enhancement layers. Otherwise, the server continues the streaming keeping the same scheme of transmission. We use the concept that the server has at least ***Retx*** chances to retransmit base layer to calculate the successful probability of transmission. The formulate function is

$$\mathrm{CDF(Re)} = \sum_{0}^{Retx} \hat{P}^{Retx} \times \left(1 - \hat{P}\right) \tag{5}$$

Fig. 2 shows the CDF of successful transmission for base layer, when the drop rates are 0.17, 0.33 and 0.41 respectively as examples.

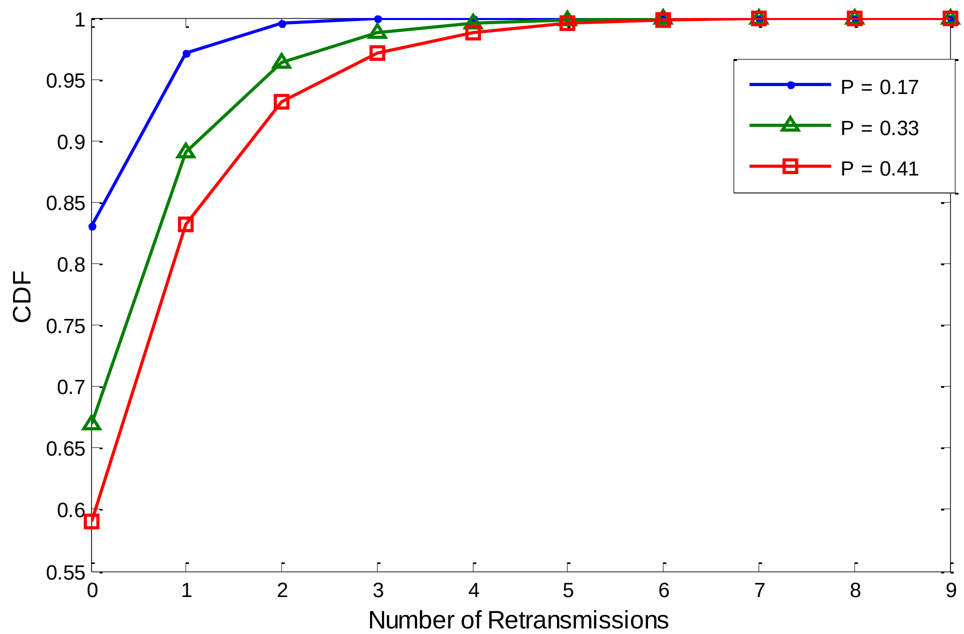

Fig. 2 The CDF of successful transmission probabilities

It is obvious that base layers have more opportunities to do retransmission, the higher probability to be received successfully. We can estimate the expected value which is the average number of transmission for base layer under the drop rate. If the number of transmission for base layers is lower than the expected value, the loss rate of base layers becomes higher. Therefore, we set the lower bound of the number of retransmission is the expected value of average number of transmission.

$$Retx_{lower} = \left\lceil \sum_{1}^{n} \hat{P}^{n-1} \left(1 - \hat{P}\right) \times n \right\rceil, \quad n \rightarrow \infty \qquad (6)$$

For example, we suppose the drop rate is 0.1, and there are 5 chunks with different number of retransmission as shown in TABLE I. Then we have the lower bound of the number of retransmission is 2.

TABLE I. The retx of chunks

# of retransmission	5	4	1	2	3
Probability	$0.1^4 \times 0.9$	$0.1^3 \times 0.9$	0.9	0.1×0.9	$0.1^2 \times 0.9$
Expected value	0.00045	0.0036	0.9	0.18	0.027

And, the upper bound is the maximum number of retransmission for base layers than can be sent before its playback deadline. T_{DL} is the playback deadline. The figure 3 shows the example that the current time is T and the playback deadline for base layers is T_{DL}. We have four times of retransmission for base layers before their playback deadline.

$$Retx_{upper} = \left\lfloor \frac{T_{DL} \times BW}{BL_{br}} \right\rfloor \tag{7}$$

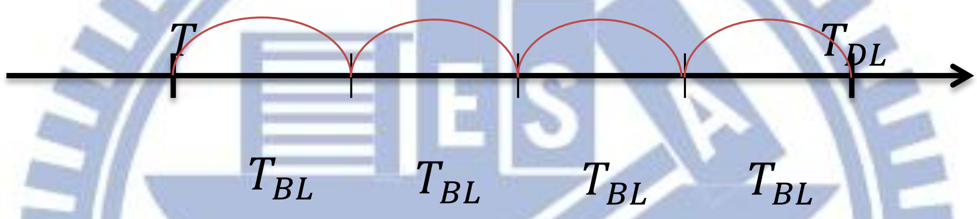

Fig. 3 The upper bound of Retx

Then we predict the quality of the received video with the number of retransmission of base layers between the lower bound and the upper bound of the number of retransmission. We choose the number of retransmission of base layers with the best quality and adjust the resolution of the streaming. First, we calculate how much bandwidth that needs to support to do $Retx$ times retransmission of base layers by equation (8). We define the preserved bandwidth for retransmissions as $\boldsymbol{PBW}$.

$$PBW = BL_{br} \times Retx \tag{8}$$

Second, we adjust the number the sent enhancement layers to cope with the network capability by equation (9).

$$\mathrm{BW} \geq \delta \times \left(PBW + BL_{br} + \sum_{i=1}^{ELnum} EL_{bri} \right) \tag{9}$$

By second step, we have the new resolution of the sent video. We can predict the video quality by calculating PSNR with the reliability of the number of retransmission by equation (5). We choose the number of retransmission with the highest PSNR by equation (10) and we can find the corresponded number of sent enhancement layer. Finally, we adjust the resolution of the sent video.

$$\underset{lower \leq Retx \leq upper}{\arg\ Max}\ PSNR(Retx) = \mathrm{Retx} \tag{10}$$

For example, we suppose T_{DL} is 1 sec, $BW = 1Mbps$, $BL_{br} = 229.7kbps$ and the drop rate is 0.17. We have $Retx_{upper}$ is 4 and $Retx_{lower}$ is 2. Then the result of the prediction of the video quality is shown in TABLE III. We can find that when the number of retransmission of base layers is 2, we have the highest quality of the received video. And, the corresponded number of the enhancement layers is 2.

Fig. 4 shows the flow chart of our SVC streaming algorithm which consists in several blocks:

TABLE II. The predict quality of video

# of retransmission of base layer	# of enhanced layer	Reliability	PSNR
4(Upper)	0	0.9992	27.5017
3	1	0.9951	31.3667
2(Lower)	2	0.9711	34.1923

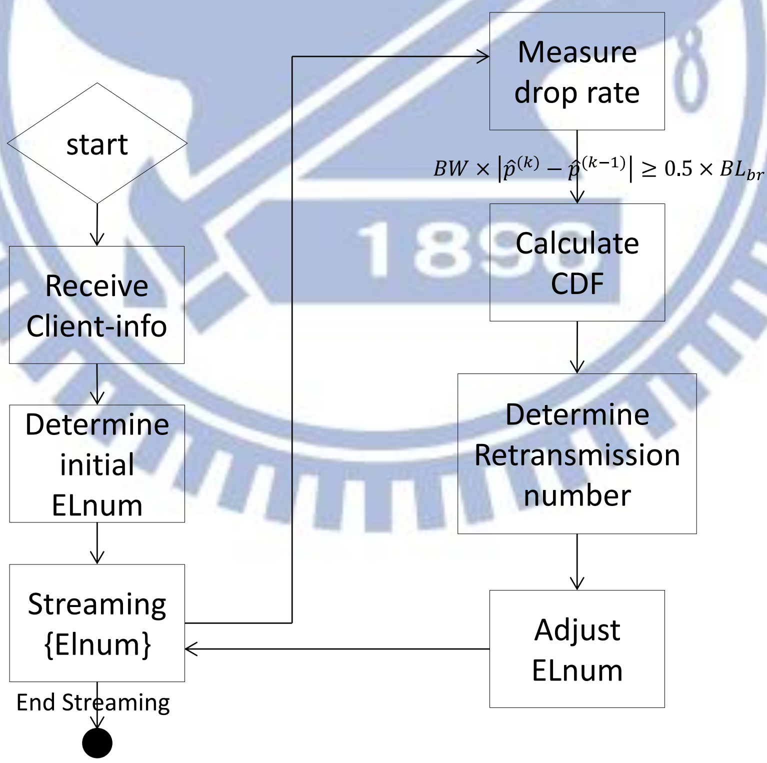

Fig. 4 Flow chart

Chapter 4
Simulation

This section shows the performance of our adaptable SVC-streaming algorithm. Our simulations are done by using NS2 [23] to simulate streaming application under the real network environment. The proposed transmission scheme is compared to the widely-used RTP/UDP method and standard SCTP for streaming video.

4.1. Simulation Environment

We use the well-known Elephants Dream video flow of 60 seconds duration [24]. This flow is encoded with SVC standard encoder [20] in one base layer and three SNR enhancement layers. We use a spatial resolution of NTSC (DVD) format (704 * 480 pixels/frames) and a temporal resolution of 24 (frames/sec). The parameters are summarized in TABLE III. The presented approach for transmitting H.264/SVC has been evaluated with a real-world application. The evaluation investigates the proposed transmission scheme in Ethernet networks. In simulation, a network topology is used as shown in Fig. 5. There are five streams of SCTP, four for SVC traffic, another one for retransmission and two intermediate nodes. In this network topology, SVC video traffic is transmitted from a SVC server to a client. At the same time, CBR background traffics are generated between two intermediate nodes. We simulated with the capacity of links set 3Mbps and 2Mbps. Link delay is set 10ms. CBR background traffic is set from 1Mbps to 1.9 Mbps. The simulation parameters is shown in TABLE IV. We choose a value of 1.5 for δ [19] was shown to be sufficient to stream the video without a significant packet loss and also to allow other applications to share the bandwidth with ours over the network in a preliminary experiment.

TABLE III. Video parameters

Fixed Parameter	**Value**			
Resolution	1920 * 1080 pixels/frames			
Frame Rate	24			
NAL number	1440			
Layer/Bit Rate	0/229.7	1/250.2	2/264.1	3/278.3
Layer/QP	0/34	1/33	2/31	3/28

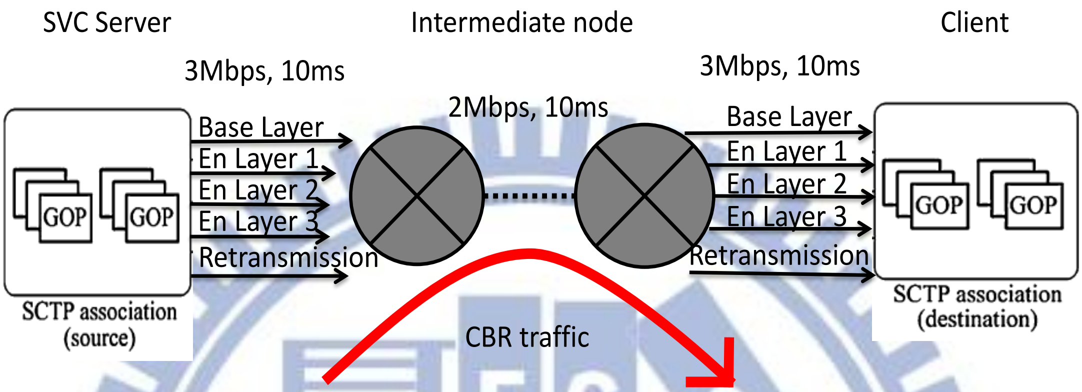

Fig. 5 Simulation topology

TABLE IV. Network parameters

Fixed parameter	**value**
Bottleneck Bandwidth	2Mbps
Link Bandwidth	3Mbps
Link Delay	10ms
Intermediate node	2
Queue Length	5

4.2. Simulation Results

Our simulation examines each scenario ten times as the samples. We calculate the average and the standard deviation of these samples and prune extreme samples under the following constraint,

$$samples \in \begin{cases} \bar{x} \pm \varepsilon, & included \\ otherwise, & excluded \end{cases} \tag{11}$$

where $\bar{x}$ is the average of the ten samples as shown in TABLE V and ε is the standard deviation of the ten samples. Here our standard deviation is 0.005. The first evaluation focuses on the number of lost frames under different network congestion conditions. Fig. 6 shows the obtained results with different background traffics. When using RTP/UDP, the number of lost frames increases significantly with background traffic up to 1.7Mbps and more. Considering the standard SCTP which has retransmission mechanism, it can retard the frame loss rate. However, with the increasing of network congestion, the standard SCTP is getting to lose the sent frames violently because it does not have any adaptive transmission mechanisms. In contrast to these, our algorithm is adaptive with network congestion. We calculate how much bandwidth should be preserved for base layers to do retransmission and reduce the number of the sent enhancement layers to cope with the network congestion. By the simulation results, the number of the lost frames can be controlled within 17% with ATC-SCTP even if the background traffic is up to 1.9 Mbps.

TABLE V. The $\bar{x}$ of different background traffic

$\bar{x}$ / Background traffic	**RTP/UDP**	**Standard SCTP**	**ATC-SCTP**
1Mbps	0.011737	0.03125	0.03125
1.1Mbps	0.029343	0.040307	0.040307
1.2Mbps	0.3169	0.070727	0.069583
1.3Mbps	0.093897	0.069583	0.062977
1.4Mbps	0.152582	0.07064	0.054632
1.5Mbps	0.172535	0.083333	0.07651
1.6Mbps	0.211268	0.117188	0.102426
1.7Mbps	0.306338	0.092652	0.083045
1.8Mbps	0.374413	0.131579	0.119792
1.9Mbps	0.442488	0.193277	0.172131

The decodable frame rate is also important in video transmission which is with respect to video quality. Fig 7 shows the decodable frame rate in RTP/UDP, standard SCTP and ATC-SCTP. When network congestion is increasing, the decodable frame rate of the flow is decreasing as expected. The loss of base layers results in significant quality degradation at the receiver side. The main objective of our algorithm is to keep the perceived video as smooth as possible at the client. We preserve the bandwidth for base layers to do retransmission in order to ensure that the base layer can be received successfully so the playback won’t be interrupted. Equ. 5 tell us that the CDF of successful transmission of base layers corresponds to the number of retransmission. ATC-SCTP can choose a specific number of retransmission for ensuring the successful transmission. As shown in Fig. 7, ATC-SCTP guarantees more than 90% reliability of the video even the background traffic (networks congestion) varies from 1 to 1.9 Mbps. It can be seen that ATC-SCTP indicates a much higher decodable frame rate compared with standard SCTP and RTP/UDP. On the other hand, RTP/UDP sends packets without any retransmission mechanism, so the base layers could be dropped when the background traffic increases and the reliability significantly degrades. Although standard SCTP has retransmission mechanism, standard SCTP sends data in constant bit rate without any adaptive mechanism for network congestion which would cause base layers lost when the congestion increased.

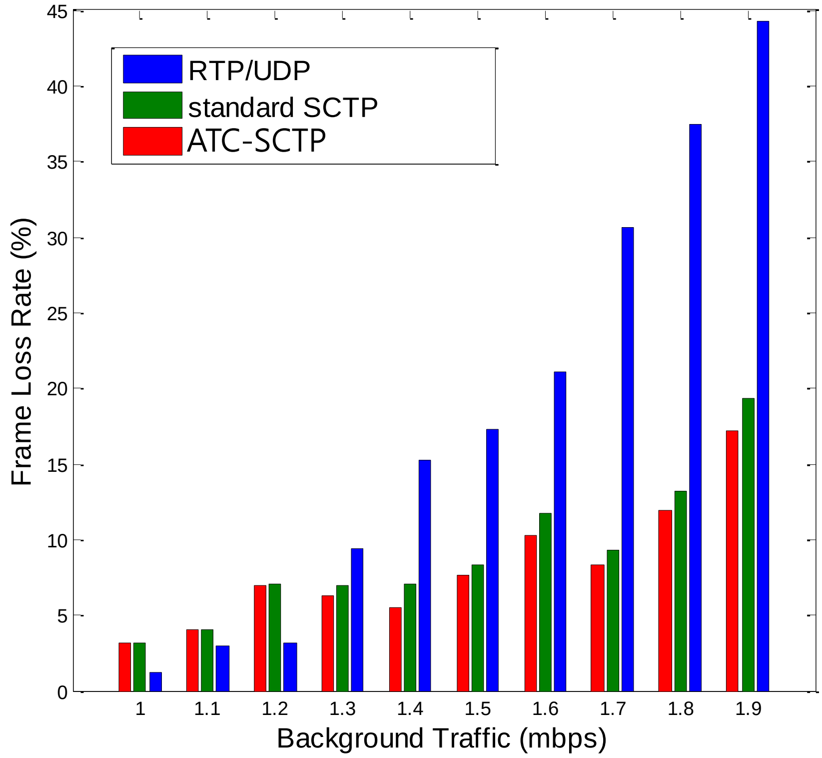

Fig. 6 Compared frame loss rate with background traffic increased

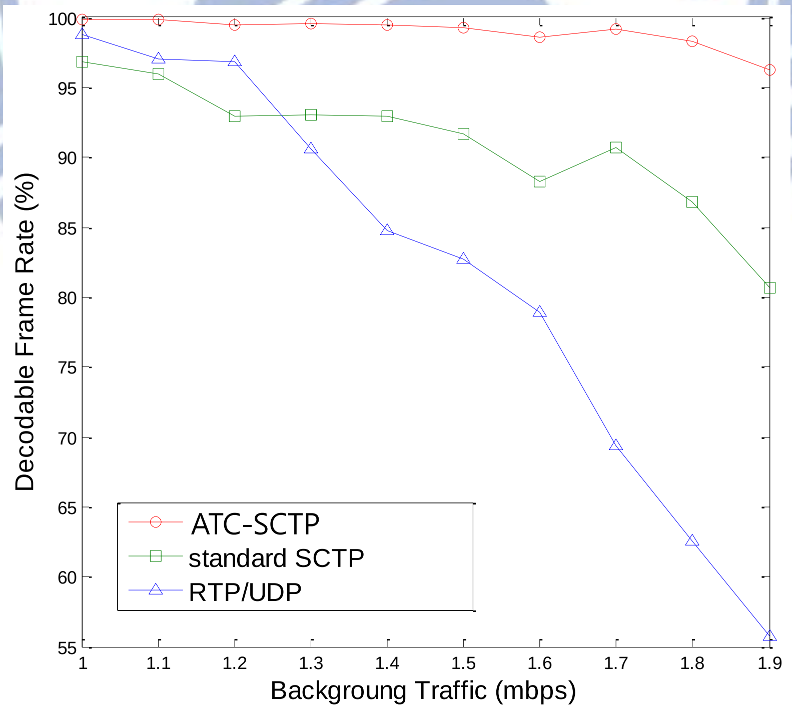

Fig. 7 Compared decodable frame rate with background traffic increased

Next, we use PSNR (Peak Signal to Noise Ratio) in Equ. 12 as performance metric to evaluate the quality of video streams. PSNR is one of most widely objective metrics for quality of video transmissions. It is utilized in literature measures quality of original image before and after compression.

$$\mathrm{PSNR} = 20 \times \log \frac{255}{\sqrt{MSE}} \tag{12}$$

where the mean square error (MSE) is

$$\mathrm{MSE} = \sum_{i=1}^{\text{NAL number}} (QP_{total} - \sum_{j=1}^{layer\ number} QP_j \times reliability_j)^2 \tag{13}$$

QP_{total} is the summation of the quantize parameters of each layer and QP_j is the quantize parameter of each layer. The reliability of base layer is the cumulative probability of successful transmission under specific number of retransmission shown in Equ. 5. And the reliability of enhancement layer is $1 - \bar{x}$ obtained by the simulation result. The more enhancement layers are received, the lower the mean square error is. The PSNR is a precise value of measuring video quality, however, human beings cannot tell the fine distinction from two different PSNR value which is close to each other. Therefore, in [21], PSNR can be translated into different levels of MOS (Mean-of-Score) as shown in Table VI. MOS is observed by human which is more close to the perception of human. We can know the video quality of PSNR simply by collating to MOS.

TABLE VI. PSNR v.s MOS

PSNR	MOS
>37	5 excellent
31-37	4 good
25-31	3 fair
20-25	2 poor
<20	1 bad

The video quality of ATC-SCTP is compared with RTP/UDP and stanrd SCTP in Fig. 8. It is known that the loss of the base layers affects quality. Fig. 8 depicts results for various background traffics ranging from 1Mbps to 1.9Mbps. The performance degradation of RTP/UDP-based protocol was something to be expected. Although standard SCTP has retransmission mechanism which can provide reliable transmission, it cannot adjust the resolution of the sent video dynamically to cope with the network congestion. As the background traffic is increased to 1.8Mbps, the quality of the sent video is decreased significantly. In contrast to these, our algorithm is adaptive with network congestion. We sent the video with the resolution under the ability of the network and ensure the successful transmission of the base layers. Even the network congestion is approaching to link capacity, our algorithm still can provide the video with the quality between “good” and “fair” because of the received base layers.

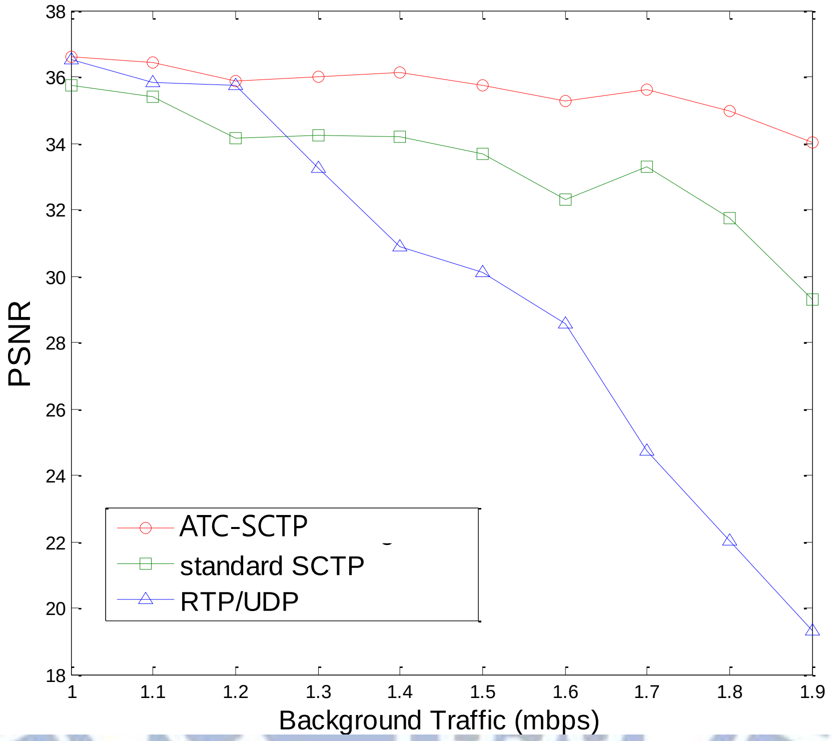

Fig. 8 Compared PSNR with background traffic increased

We evaluated the video quality during the streaming to observe the fluctuation of quality when the background traffics are 1.3Mbps, 1.6Mbps and 1.9 Mbps. We calculated the frame loss rate per sec by NS2 simulation and estimate the quality of the video per sec by PSNR above. Then we find the MOS corresponded to the estimated PSNR. When the background traffic is 1.3Mbps which causing the congestion is small, RTP/UDP, standard SCTP and SCTP with our algorithm, all of them can provide “good” quality of the video in figure 9. But it is obvious to see that ATC-SCTP can provide better quality of the video than the others.

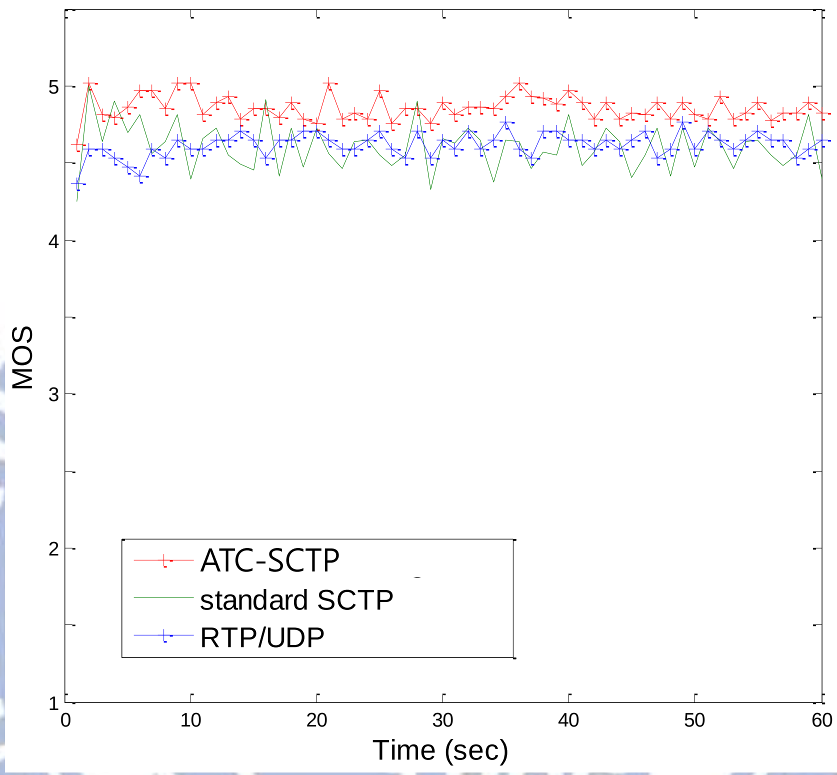

Fig. 9 Compared MOS with background traffic = 1.3Mbps during the streaming

In figure 10, we observe the fluctuation of quality during the streaming when the background traffic is 1.6Mbps. When the congestion is increased, the quality of the streaming of standard SCTP and RTP/UDP are getting unstable and decreased. And in figure 11, when the background traffic is increased to 1.9Mbps, the degradation of the quality of the video is more violent. But ATC-SCTP is designed to cope with the network congestion. We adjust the number of the sent enhancement layers under the network ability and ensure the successful transmission of the sent base layer. So we can offer stable quality of the sent video for users even the congestion is getting bad.

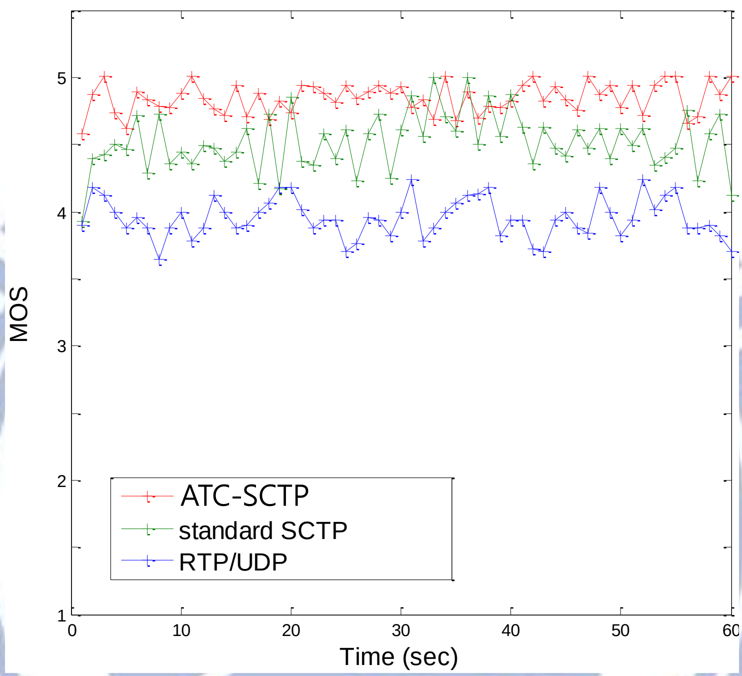

Fig. 10 Compared MOS with background traffic = 1.6Mbps during the streaming

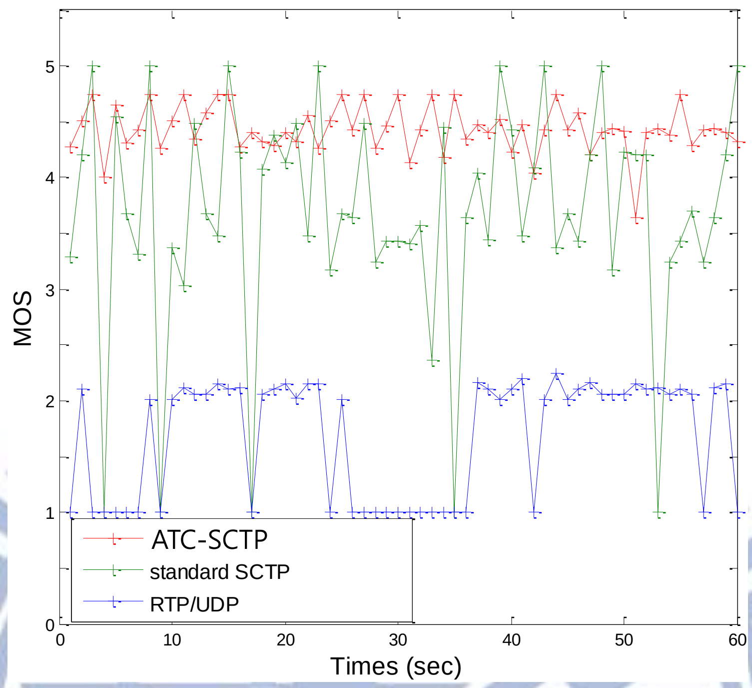

Fig. 11 Compared MOS with background traffic = 1.9Mbps during the streaming

Fig. 12 presents prove of the proposed algorithm adaptability to fluctuation of network congestion. In this experiment, we increase the network congestion from 1.3Mbps to 1.9Mbps at t = 20, in a way that the network congestion is getting worse. Then at t = 40, we decrease the network congestion from 1.9Mbps to 1.3Mbps gain to improve network condition. In Fig. 12, we can notice how the algorithm adapts to these changes. Consequently, server will re-measure the drop rate over the path to the client. As the drop rate has been increased and the amount of affect has reach to half bit rate of base layer, the algorithm decides to decrease the number of the sent enhancement layers in order to economize on network bandwidth and give base layers more chances to do retransmission. So at t = 21, it starts to stream one base layer and one enhancement layers. The algorithm will keep testing the drop rate over the path in order to improve the quality whenever it is possible. At t = 40, the algorithm observes an improvement in the estimated drop rate that enables the streaming of extra enhancement layers. Thus, after this second, it stream two enhancement layers again with the base layer, leading to a better video quality. Furthermore, without our algorithm, server continuous to stream the four layers even with the ascension in the drop rate that may affect base layer and prevent the decoder from decoding the remaining video streaming.

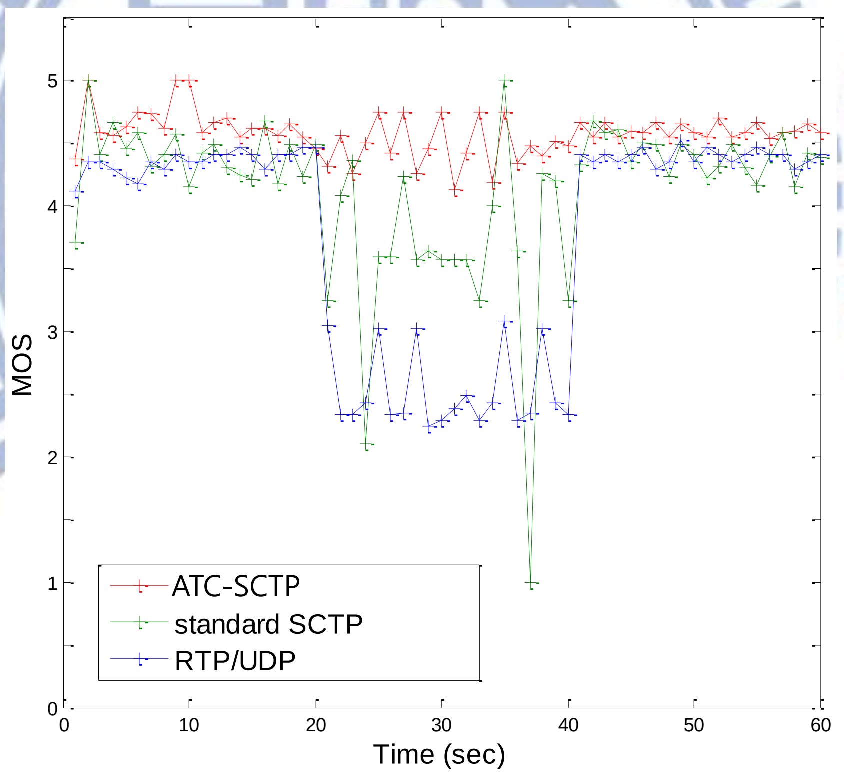

Fig. 12 Adaptation to network congestion between t=20 and t=40

Chapter 5
Conclusion

In this paper, we propose an adaptive algorithm for streaming SVC video flows over SCTP basing on the network congestion evaluation. We investigate the network congestion by measuring drop rate during the entire video streaming process. In order to play the video without interruption, we ensure the base layers can be received successfully by preserving the bandwidth for base layers to do more retransmissions and dynamically adjusting the number of the sent enhancement layers to cope with network condition. Finally, simulation results show that our algorithm keeps the perceived video with high reliability in order to playback smoothly and keeps the quality of video streams received by end-user as stable as possible.

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