Optimization of Video Streaming Using SRT Protocol in Mobile Communication Network

Jeong-Hwan Lee and Jihun Cha
Autonomous UAV Research Section
Electronic and Telecommunications Research Institute
Daejeon, Republic of Korea
{jeonghwan.lee, jihun}@etri.re.kr

***Abstract*—**Interworking with 4G/5G mobile communication networks is essential for drones to perform missions in broadband environments, and high-quality video data from cameras such as EO/IR/Lidar, which are equipped on drones for mission status monitoring, must be transmitted to the control center using streaming protocols. In this paper, we aim to describe the optimization method for transmitting high-quality video data using the SRT transmission protocol in mobile communication network.

Keywords— SRT; Secure Reliable Transport; UAV; Unmanned Aerial Vehicle; Drone; Streaming Protocol

I. INTRODUCTION

Currently, UAV(Unmanned Aerial Vehicles)/drones are being utilized for various missions such as delivery, facility management, and reconnaissance by combining artificial intelligence, autonomous flight, and sensor technology. Operators monitor the drone's mission status using real-time video data from cameras such as EO and IR. Drones currently provide high-quality EO video above Full HD resolution and IR video above VGA resolution. Drones mostly use streaming protocols such as RTSP (Real-Time Streaming Protocol)/RTP (Real-time Transport Protocol), and RTMP (Real-Time Messaging Protocol) to transmit high-quality video streaming data. However, there have been no cases of using the SRT (Secure Reliable Transport) protocol in drones.

SRT is a royalty-free, open-source video streaming transport protocol that delivers secure low-latency streaming performance over noisy or unpredictable (lossy) networks such as the public internet. SRT uses an intelligent packet retransmit mechanism called ARQ (Automatic Repeat reQuest) on top of a UDP data flow to protect against packet loss and fluctuating bandwidth, as well as to ensure the quality of your live video.[1]

Therefore, this paper aims to describe the optimization method for transmitting high-quality video data using the SRT transmission protocol in 4G and 5G mobile communication network.

II. SRT STREAMING TEST APPLICATION

In this chapter, we will explain how to calculate SRT configuration options suitable for mobile communication network based on the SRT configuration steps presented by the SRT document. The SRT library version 1.5.1 was used in this study. I used the srt-stats-plotting utility as a tool to visualize SRT core statistics produced during experiments.[2]

A. RTT/PLR

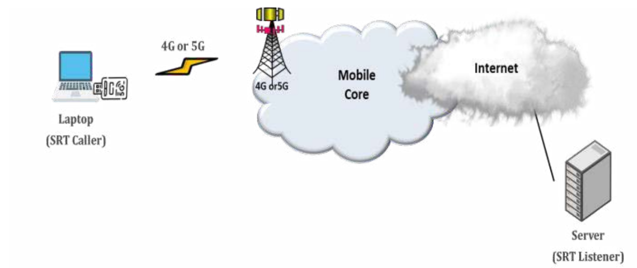

Fig. 1. SRT Test Environment

To configure SRT options, the first step is to calculate the Round-Trip Time (RTT) and Packet Loss Rate (PLR) between the SRT caller and SRT listener. While the ping command is recommended for calculating the RTT value, it sends small 64-byte data packets, which may not be suitable for the actual video data size and SRT transmission type. In this study, we measured the RTT and PLR by sending video data at a rate of bytes per second (Bps) of the actual video data to be transmitted, and distinguishing packet sizes of 108, 172, 556, 1068, and 1360 bytes, including SRT, IPv4, and UDP header sizes, respectively, for 3 minutes with 1 transmission per second. For example, assuming that video data of 5Mbps is to be transmitted, the number of bytes per second to be transmitted is 5,242,880/8 bits = 655,360. If the six packets above are transmitted every 5ms, approximately 200 transmissions can be made per second, making the test possible at approximately 5Mbps.

Every second, we measured the RTT using SRT statistics data and selected the largest value to use in option calculations, taking into account the mobile communication network environment. If the measured RTT was less than the minimum value of 20ms, we set the RTT to 20ms. For PLR, we calculated it every second using the formula (1) below, and also selected the largest value as the value to use in option calculations. Here, α represents the amount of bytes retransmitted and β represents the amount of bytes transmitted.

$$\text{PLR} = (\alpha/\beta) * 100.0 \qquad (1)$$

B. Latency and Bandwidth Overhead

The RTT Multiplier value is used in the calculation of SRT Latency, and it represents the relationship between network congestion and RTT. The range of RTT Multiplier values is set between 3 and 20, and values below 3 are inefficient while values above 20 indicate 100% packet loss on the network.

We determined the RTT Multiplier and Bandwidth Overhead values corresponding to the measured PLR values based on the TABLE I, and if the measured PLR value was 20 or higher, we considered the mobile communication network to be unstable and stopped the test.

TABLE I. RTT MULTIPLIER AND BANDWIDTH OVERHEAD

PLR(%)	RTT Multiplier	Bandwidth Overhead (%)
<= 1	3	33
<= 3	4	25
<= 7	5	20
<= 10	6	17
< 20	7	14
The others	Ignore	

The bandwidth overhead, B, was calculated using the formula (2) below. Here, k represents the RTT multiplier value corresponding to the PLR in TABLE I.

$$B = 100.0 \,/\, \kappa \qquad (2)$$

According to [4], the recommended lowest SRT Latency is 3-4 times the average RTT. However, in this study, the SRT Latency, H was calculated using the following equation (3) by utilizing the maximum RTT value. γ represents the RTT value determined through measurement.

$$H = \kappa * \gamma \qquad (3)$$

C. Flow Control

Flow Control limits the maximum number of packets 'in flight'. This means the maximum number of payload packets that have been transmitted but have not been acknowledged by an ACK control packet. The Flow Control, N, is calculated using the formula (4) below. Here, the Input Bandwidth, I, represents the bitrate of the stream transmitted per second, and the Maximum Segment Size (MSS), M, is a constant set to 1,500 bytes. The number 44 represents the sum of the SRT header (16 bytes), IPv4 header (20 bytes), and UDP header (8 bytes) values.

$$N = (I/8) * ((\gamma * 1000.0) + (H/1000.0)) \,/\, (M-44) \qquad (4)$$

If the calculated Flow Control value is less than the minimum value of 32, we used the minimum value.

III. SRT STREAMING TEST & RESULTS

In this chapter, a test was conducted in a mobile communication network by transmitting data at a sending rate of 5Mbps for 3 minutes. The purpose of this test was to replicate the environment of transmitting video streaming data acquired by a drone to a ground control center. The drone played the role of SRT caller, and the ground control center acted as SRT listener. This test operates in the scenario of "Streaming using Caller & Listener modes"[3], and the SRT source(caller) behind the firewall is streaming to an SRT destination(listener) over the Internet, as illustrated in the following Fig.2.

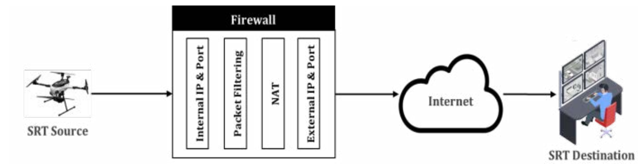

Fig. 2. SRT scenario

First, the default SRT settings as shown in TABLE II were used for the SRT Stream test.

TABLE II. SRT DEFAULT OPTION FOR STREAM TEST

SRTO_INPUTBW	652,800 bytes
SRTO_OHEADBW	25 %
SRTO_LATENCY	120 ms
SRTO_MSS	1,500 bytes
SRTO_PAYLOADSIZE	1,316 bytes
SRTO_FC	25,600 packets

As a result, as shown in Fig. 3, there were momentary maximum packet losses of about 500 packets and an average packet loss of 70 packets per second. Fig. 4 shows that packet loss reached a maximum of 28% at one point, with an average loss rate of 4.8% per second.

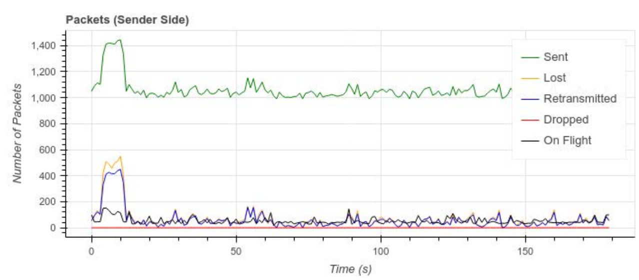

Fig. 3. Results of packet transmission with default option

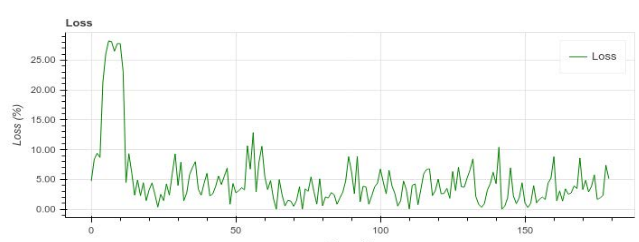

Fig. 4. Results of packet loss with default option

The results of the SRT Stream test measurements are shown in TABLE III, and the test results obtained by applying these values are as follows.

TABLE III. MEASURED SRT OPTION

Measured Results	
RTT	39.02 ms
PLR	28.21 %
RTT Multiplier	6
SRT Option	
SRTO_LATENCY	234 ms
SRTO_OHEADBW	16 %
SRTO_FC	122 packets

The results of the SRT measured options through the SRT Stream test are shown in TABLE III, and the results of the test applied with these values are shown below in Fig. 5. As shown in the figure, there were momentary maximum packet losses of about 300 and an average of 55 packet losses per second. Fig. 6 shows that the packet loss reached a maximum of 21% at one point, with an average loss rate of 4% per second.

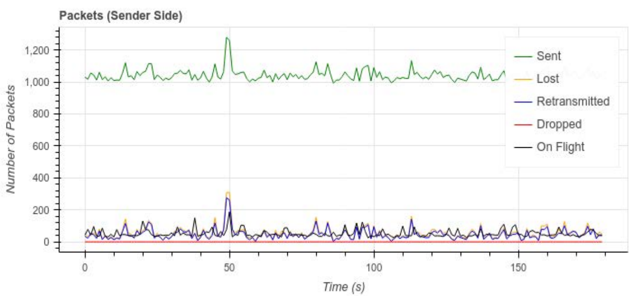

Fig. 5. Results of packet transmission with measured option

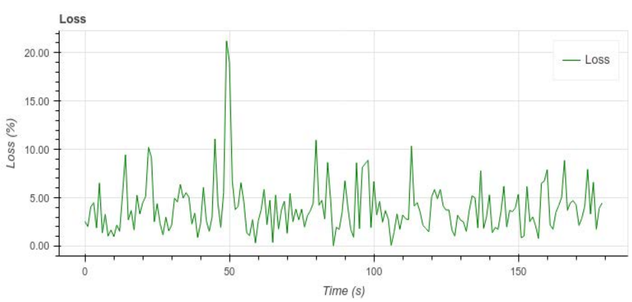

Fig. 6. Results of packet loss with measured option

The following is the result of a 20-minute test conducted at 5Mbps, assuming that EO/IR streaming data is transmitted during drone flight.

As a result, momentary maximum packet losses of about 200 packets and an average of 13 packet losses per second occurred, as shown in Fig. 7. In Fig. 8, packet loss reached a maximum of 13% at one point, showing an average loss rate of 1% per second. However, when looking at the entire transmitted data from the perspective of the receiving end, all of the data was received without any drop.

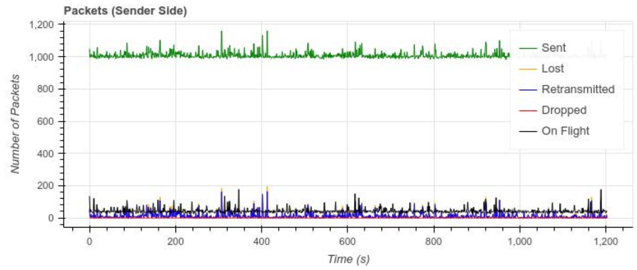

Fig. 7. Results of packet transmission with measured option for 20 minutes

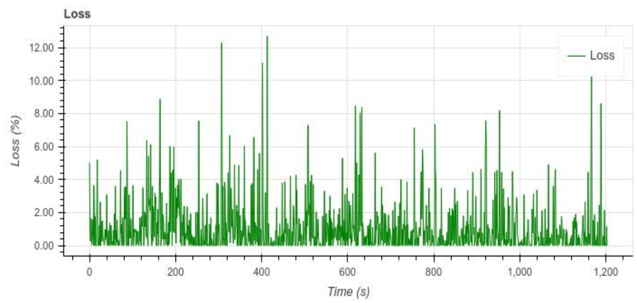

Fig. 8. Results of packet loss with measured option for 20 minutes

IV. CONCLUSION

A software for testing SRT streams was developed to determine the options between SRT Caller and Listener, and SRT options were measured on a mobile communication network. It was confirmed that the SRT communication with the measured options was more stable than the SRT communication before the measurement. After this study, we plan to conduct repetitive tests in mobile communication network and apply the software to the target board to be used in actual drones for testing. We also plan to conduct SRT communication research that can be applied to changes in mobile communication network while the drone is in flight.

ACKNOWLEDGMENT

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