무선 핸드오프 시 유디피 패킷의 전송

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UDP Packet Transfers on Mobile Handoffs

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요 약

In wireless UDP data transfers, the handoff is one of the most breaking down things. To polycast multimedia data to several clients a server may connect clients by UDP networks. Mostly, video data must be compressed based on differences between before and next packets. Thus, the packet losses on the video stream may cause the recomposing of video data to be corrupt. This proposal provides simply retransmitting of lost packets according to a sequence and buffering packets in a given order.

1. INTRODUCTION

Multimedia applications are getting popular in wire/wireless networks. By the developments of UWB, wireless networks may support wide bandwidths and better QoS. However, in mobile networks there are other problems to serve multimedia applications: narrow bandwidths; hand-offs; low quality of services; high costs on network resource usages, and so on. While these problems exist on mobile networks, there are requests on multimedia services more and more. Thus, now multimedia applications on mobile networks are deploying and offer some services. Mobile phone customers can watch TV programs through their mobile phones which serve these broadcasting of TV programs by Satellites or by terrestrial base stations of mobile networks [1].

For data transfers on wireless mobile networks, the handoff is one of the must-solved problems. During handoffs, data should be sent to both base stations, one already-connected base station and a new base station to connect. These duplicated data may request more network bandwidths instead of using for other data transfers. It will cause mobile network services to be kept high costs. Thus, handoffs must be processed as short as possible with correct data transferring.

Also, even though a handoff operation is finished in good procedures, most current video compression technologies do not always reproduce video images. If packets including a video image frame are lost on transfering, then next coming packets could be useless data even though they are arrived without any errors until the lost packet is re-transmitted and arrived or next I-frame data is received.

In this paper, we will introduce real-time video data transfers with end-to-end traffic control schemes. Thus, video displaying on clients will have less broken images and stuck image displaying in real-time.

2. REVIEW

A handoff in wireless mobile network systems is an automatic transfer of a wireless connection from one channel or a cell to another. Handoffs are initiated when the mobile station measures a neighbor cell's pilot signal power, and determines that this power level is above an acceptable threshold.

The application may delay the shutting down of the old connection and packets continue to arrive over the old connection and thereby, duplicated packets may cause the video player to be interrupted.

There also are two types of streaming for multimedia data transfers through networks: streaming by TCP or by UDP-based protocols.

For the available bandwidths of networks video streams, in general, consist of many PB-frames or B-frames, some P-frames and a few I-frames.

3. PROPOSED MODEL - TRAFFIC CONTROL FOR REAL-TIME VIDEO TRANSFER ON MOBILE HANDOFFS

When compressing video frames, for low bandwidth mobile network connections, the inter-frame compressed packets become more essential and popular and the intra-frame compressed ones used rarely. However, data losses during transmissions such as handoffs are a good reason for using the intra-frame compressed packet periodically. If there are errors in the compressed data, it prevents not only reconstruction of the current frame, but also of all of the next inter-compressed frames until an intra-frame compressed packet is received safely. The number of intra-frame compressed packets that must be inserted mostly depends on the quality of the mobile connections and the time that is considered as least painful for loosing the connection. Thus, the ratio of intra-frame compressed to inter-frame compressed packets should be dynamically changed on the mobile network conditions.

When data is packed as a packet, intra-frame compressed data packet is set to have sequence number zero. It can help on detecting packet loss and deciding whether or not request packet retransmitting to sender. Another consideration before describing the proposed algorithm is that applications do not necessarily take care of handoff control signaling.

Thus, we assume applications do not necessarily take care of handoff control signaling. We only consider video transports on wireless mobile network connections.

As wireless mobile network environments, a mobile station roams one access point (as a base station) to other APs, which are sharing the same frequency ranges between them and connected by a hub switch. The hub switch then connects these with another stationary station. Thus, the correct time on a handoff processing can be detected as accurately as possible since there is a wireless connection. As a mobile station, video images from a camera are compressed on the media server, then transmitted to an AP which is, so called, a base station. Video packets are based on the real-time transport protocols (RTP/UDP/IP). To decode video frames in simple, each video frame is ported in a packet. Thus, the length of each packet might be variable and it causes just a video frame loss when the packet is corrupted. In real mobile network system, there are usually shorter handoff time delays and better routing algorithms than these environments.

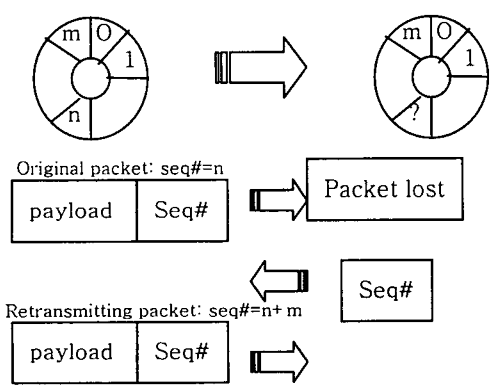

Figure 1. Retransmission and buffering of packets

The whole algorithms, Figure 1, which are proposed have two parts: Receiver and transmitter ones. Receiver and transmitter have each a buffer with cyclic I-frame period spaces, respectively. That is, the buffer has spaces for frames ranging from an I-frame to P-frames which are just before a next I-frame.

The receiving procedure in the receiver watches video frame packets by the sequence numbers and sender addresses which are included in the packet headers. When packets arrive in the proper sequence through mobile networks, packets are unpacked without any extra processing in the receiver and video frames can be displayed in a screen. However, when a packet lost happened our proposed algorithms have been monitoring sequence numbers and buffering next packets according to the sequence numbers. That is, we assume that the receiver has got packets up to the sequence number "n" and a lost packet is assumed to have the sequence number "n+1". Incoming packets with later sequence numbers such as "n+2","n+3", ... are checked out whether n+2 is 0, which is the sequence number of I-frame data packets, or a non-zero bigger number than n+1, which is the sequence number of a P-frame data packet. If the sequence number, n+2, is zero then image displaying may go up to this frame by accessing this video buffer. However, if the sequence number is not zero the retransmit-request procedure will send to senders with the lost packet sequence numbers. These retransmit-request packets have small packet size because there are only packets with packet headers. Thus each retransmit-request packet has one sequence number out of the lost packets. This may have more retransmit-request packets than one retransmit-request packet which includes all sequence numbers or the range of lost packets.

When sender has received packets from the receiver for retransmitting of video frame packets, the sender checks out whether the sequence number is proper or too old to send. The sender discards the request if the sequence number is older than the sequence number of the most recent I-frame packet. If not, the sender duplicates the sequence number and the host address from received packet into the retransmitting packet header and add some number to the sequence number to distinguish from the first sending packets. Then the image frame data from the buffer is attached to the packet payload. Since the sender has a buffer of an I-frame period capacity the corresponding image data to the sequence number is searched in ease and the buffer is flushed just after an I-frame is buffered. Also, the sender does not require much time on transmitting of packets because of UDP packet sending. It does not cause the sender not to send new image data packets.

4. PERFORMANCE EVALUATION

To implement this proposed algorithms there are some critical issues and these affect on the implementation and results: As the first, the I-frame periods are severely considered. The better image qualities on video compressions the bigger the size of an image frame data. That is, to keep good image qualities the sizes of compressed image data are much bigger than those of low quality compressed image data. The packet size of an I-frame compressed data with high quality may cause the network to be overloaded at a time. It may reduce the number of image frames arrived to the receiver. Thus, the rate of displaying images is decreased and the motions on the video seem not to be smooth. But for the recovery at P-frame losses the I-frame rate should not be low. Only for one-to-one communication automatically-variable rate control algorithm is the best fitted one. However for broadcasting or multicasting purposes, the rate is fixed to a proper value according to the recovery from packet losses and bandwidths of networks. As the second, circular queued buffer should be considered. Without buffering of incoming packets, received video frames may not be displayed properly on the monitor because there may be some lost packets when receiving packets from a remote host. Also UDP packets may be delivered out of orders. Thus there should be a buffering space to solve these problems. In this system a circular queue is proposed as the buffer of incoming video packets.

Again the capacity of the queue has memory spaces of one I-frame period and each of the spaces has the offset by the sequence numbers of receiving packets. It means that complicated data structures such as linked list or hashing tables are not needed and that the simple queue supports fast memory accesses. If there are missed packets the spaces of the queue buffers keep empty the corresponding spaces and displaying might be stopped on the missed space. After receiving the missed packets the empty spaces of the queue could be filled with the data and the whole processing will be going to go to the next sequence. As the third issue, broken time at a handoff affects severely on the smooth displaying of video frames and on deciding the size of the circular queue buffer. [3] compared a handoff to a server within the same LAN delay with 20 ms latency to a WAN-server with 2000 ms latency for their evaluations of delays on handoffs. If this time is short then the circular queue can be reused even in a I-frame period and thus the size can be much less than the size of a I-frame period. As the last issue, duplicated packets during handoffs should be processed clearly.

According to the above issues, we have the following results: Figure[2] shows the ratio of succes video displaying to the handoff time delays. Usually the disconnected time during handoff is much shorter time than this results. In this result we ignored packets after packet loss and before intra-fram compressed packet. Also Figure [3] shows the increases of bandwidth given on the network when the handoff time delay increased. Since the increases of bandwidths is caused mostly by the intra-fram compressed video data packets, retransmitting of inter-frame compressed video data packets do not affect much on the bandwidths of the networks.

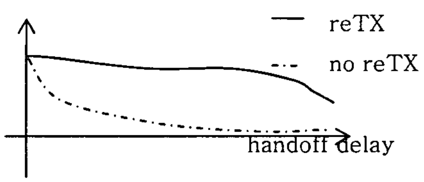

Figure 2. The number of correctly received packets after retransmission

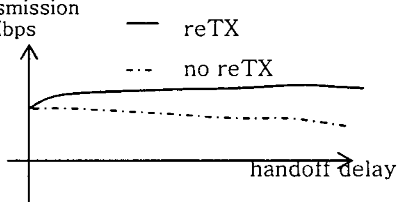

Figure 3. Bandwidth increases on handoff

5. CONCLUSION

We have proposed a recovery of compressed video data during handoff. Even if this is proposed simply with retransmitting and buffering packets during handoffs, it is assumed to have good recoveries for UDP based video packets on handoffs. Thus, we can conclude the recovery approach from the handoff on wireless networks must improve the quality of video streams on wireless UDP networks.

7. REFERENCES

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