INTER-FRAME RETRANSMISSION FOR VIDEO SURVEILLANCE OVER WIMAX

Suherman
Faculty of Technology
De Montfort University
Leicester, UK
Email: suherman@usu.ac.id

Marwan Al-Akaidi
Arab Open University
P.O.Box: 3322 Safat
13033 Kuwait
Email:m.al-akaidi@arabou.edu.kw

Raouf Hamzaoui
Faculty of Technology
De Montfort University
Leicester, UK
Email: rhamzaoui@dmu.ac.uk

KEYWORDS

Transport layer protocol, inter-frame retransmission, video surveillance, WiMAX.

ABSTRACT

Video surveillance is an important application for activity and security monitoring. Surveillance application can take advantage of wireless infrastructure which provides installation flexibility and terminal mobility. However, wireless video transmission is prone to interferences which degrade video quality. This paper proposes an inter-frame retransmission protocol for video surveillance over WiMAX. The protocol reduces packet and frame delay compared to existing protocols.

INTRODUCTION

Worldwide Inter-operability for Microwave Access (WiMAX) technology that offers high bandwidth connectivity and user mobility is a potential network infrastructure for video surveillance application. The surveillance nodes may reach distance as WiMAX is able to cover up to 50 km (Scalabrino et al. 2007). Moreover, the mobility feature of WiMAX enables video surveillance to be attached on moving objects such as public transportation. This paper proposes a transport layer protocol to support the surveillance application over WiMAX.

Transmission Control Protocol (TCP) provides high reliability data transfer, ensuring that each frame is received successfully and sequentially. However, TCP is not suitable for real-time video transmission as wireless interferences and signal disruption may cause significant delay. User Datagram Protocol (UDP) is the most common transport protocol for real-time video transmission over wireless networks (Chughtai et al. 2009). However, UDP does not respond to network conditions which can cause network congestion (Wong et al. 2005).

In order to gain maximum performance for the intended application, the transport protocol should be able to deliver video with sufficient quality as well as maintain low delay connectivity. Many works have been done to improve transport layer protocol performance, whether employing retransmission or congestion control services. The details are reserved in related works section. Since delay is crucial parameter in real time video transmission, video frames should be received in order to avoid delay. The work focuses on how to reduce packet loss by retransmitting dropped packets within one frame before sending the next frame. The NS-2 simulations show that the proposed method is able to reduce packet loss without producing significant delay.

RELATED WORKS

Many works have proposed improvements on the transport layer protocol. RUDP (Reliable UDP), RBUDP (Reliable Blast UDP), UDT (UDP-based Data Transfer) and BVS (Broadband Video Streaming) improve the existing protocol performance by using retransmission. UDP-lite and DCCP (Datagram Congestion Control Protocol) do not retransmit lost packets.

RUDP uses acknowledgement (ACK) as in TCP and provides a congestion control mechanism (Bova and Krivoruchka 1999). However, since RUDP employs almost all features in TCP, RUDP may produce excessive delay as TCP (Tuong et al. 2009). RBUDP (He et al. 2002) and UDT (Gu and Grossman 2007) are datagram protocols that work for high speed bulk data transfer. Both protocols were aimed to solve TCP weakness which underutilize high speed network (Gu and Grossman 2007). RBUDP and UDT employ negative acknowledgement based retransmission. RBUDP waits an additional “DONE” packet before NACK is sent, while UDT uses periodical NACK packets to request retransmission. Such methods work well in high speed networks but not in competing networks. BVS is a semi reliable protocol which applies retransmission when the prioritized packet is lost (Ali et al. 2011). BVS uses a NACK packet to request the sender retransmits lost packets. Frequent prioritized packet loss in BVS results irregular retransmission.

UDP-Lite (Larzon et al. 1999) implements partial checksum for sensitive part of the packets and ignores error in non-sensitive part. However, passing error packets to application layer limits network observation capabilities (Welzl 2005). DCCP improves unreliable connection by providing congestion control mechanisms (Kohler et al. 2006). Two congestion control mechanisms were proposed: TCP-Like and TFRC-Like. DCCP employs acknowledgement service without retransmission which means it does not recover the lost packets.

Many protocols are designed for specific applications such BTP, Bidirectional Transport Protocol (Wirz et al. 2009) and ERT, Embedded Reliable Transport protocol (Wei and Chao 2010). But, only few that are designed for real time video transmission.

THE PROPOSED PROTOCOL

The proposed protocol is intended for video surveillance applications over WiMAX networks. The designed protocol is called as inter-frame retransmission protocol. It employs negative acknowledgements, inter-frame retransmission scheduling and congestion delay. The details are described in the remainder of this section.

Negative Acknowledgement

The proposed protocol uses negative acknowledgement (NACK) to inform the sender that packet loss has occurred and packet should be retransmitted. The NACK packet contains either a list of indices of lost packets or the start and end indices of the lost packets. As soon as the sender receives a NACK packet, it resends the requested packets (Figure 1).

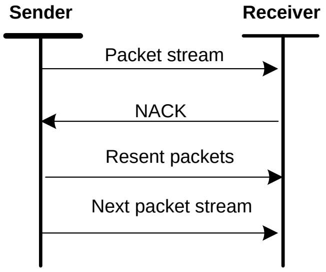


Figure 1: Negative acknowledgement

In RBUDP, NACK is used but delivered over a different connection (He et al. 2002). Moreover, RBUDP requires an additional "DONE" packet before sending a NACK. NACK is also implemented in BVS (Ali et al. 2011) which uses quick response scheduling as discussed in the next section.

Retransmission Scheduling

The NACK packet is sent by the receiver when packet loss is detected. The sending time is set according to the scheduling type. There are two NACK scheduling types, Quick Response (QR) and Inter-frame Retransmission (IR). QR requires the receiver to send a NACK packet as soon as packet loss is detected. The packet loss information is determined by two values, the current and previous successfully received packet indices. The sender will check these values to decide which packet to retransmit. For instance, if the current packet index is 7 and the previous one is 4, then packets with indices 5 and 6 should be retransmitted. The advantage of QR is small NACK overhead and responding loss quickly. However, the receiver may generate more than one NACK packet for a frame, which requires more bandwidth and interrupts the sender frequently.

The second scheduling strategy is called inter-frame retransmission (IR). Instead of sending a NACK packet for every detected lost packet, the receiver records indices of packet lost within one frame and sends a NACK packet after receiving the last packet in that frame. If no packet is lost in one video frame, then no NACK packet will be sent. The advantage of IR is that a NACK packet will be sent only once for all lost within one video frame. IR generates fewer NACK packets than QR.

Figure 2 shows the NACK scheduling. One video frame may be sent in several packets. The time distance between the last packet in one frame and the first packet of the next frame is called inter-frame gap (IFG). Figure 2a and 2b assume that the round trip time (RTT) is less than IFG. Packet A and C within frame 1 are lost. In QR, NACK packets will be sent as soon as the receiver receives packets B and D. NACK packets may interrupt the sender frequently and may cause additional delay or another packet loss. On the other hand, IR sends NACK and resends packets during inter-frame gap when the sender is idle.

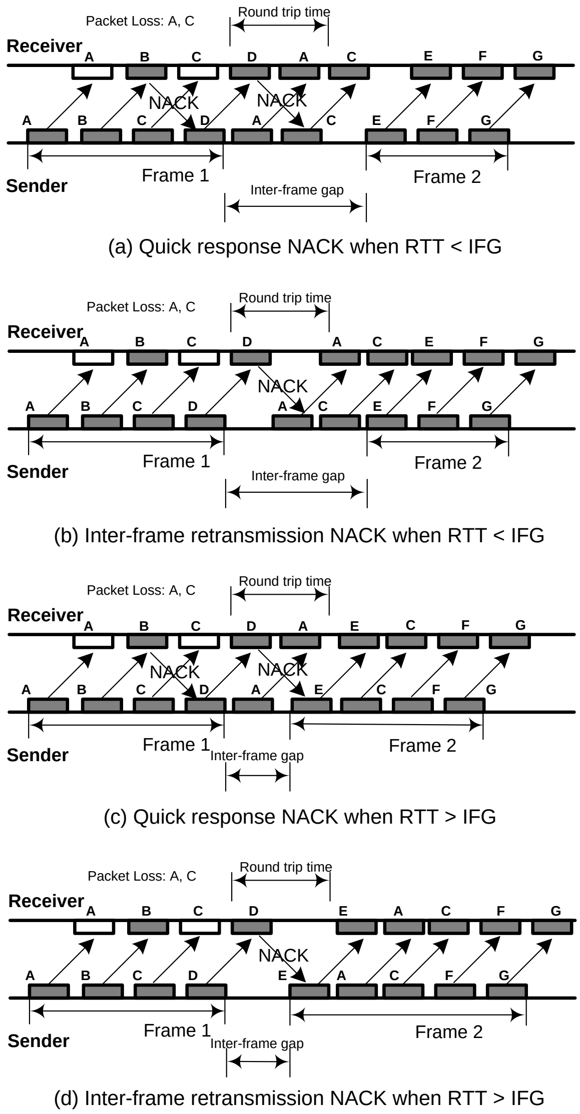


Figure 2: NACK scheduling

If RTT is greater than IFG as shown in Figure 2c and 2d, IR interrupts the sender only once. Although IR seems causing the next frame sending time longer, we found that the sender processing time is more sensitive to NACK reception than to packet retransmission. IR has additional requirement that the receiver should be able to detect the last packet in each frame. In case the last packet within a frame is lost, the lost packet will be retransmitted within the next frame.

Prioritized Packets

Unlike TCP which sends an acknowledgement for every received packet, the proposed protocol sends NACK packets only when packet loss occurs. However, if network congestion worsens, NACK packets may be sent more frequently as more loss appears. The frequent packet retransmissions may lead to high delay. To keep delay low, the NACK packet for a particular packet loss will be sent only once. The dropped retransmitted packet will be ignored.

Furthermore, the NACK packet reduction may be applied by sending NACK only for prioritized packets as video coding results non uniform frame significances. An additional packet header is required to flag whether a packet is prioritized or not. Simulation in this paper uses MPEG4 video coding with IPP frame sequence. The prioritized packets are set to be any packets corresponding to I-frames. The conducted simulations show that priority policy significantly reduces delay.

Congestion Delay

Congestion delay (CD) aims at reducing the effect of sender interruption and avoiding another packet loss by postponing the next packet transmission. Congestion delay also makes sure that the current frame arrives before the next frame.

Figure 3a shows retransmission without congestion delay. The sender sends packet E before retransmitting the lost packet C. Packet C which belongs to previous frame may arrive after packet E which belongs to the next frame. This situation results in higher frame delay. In the worst case, packet E can be lost during reception of a NACK packet. By using congestion delay, packet C will be retransmitted before packet E as shown in Figure 3b. This process results in lower delay on packet C and avoids loss of packet E. Although congestion delay introduces more delay for packet E, a small congestion delay value limits this additional delay.

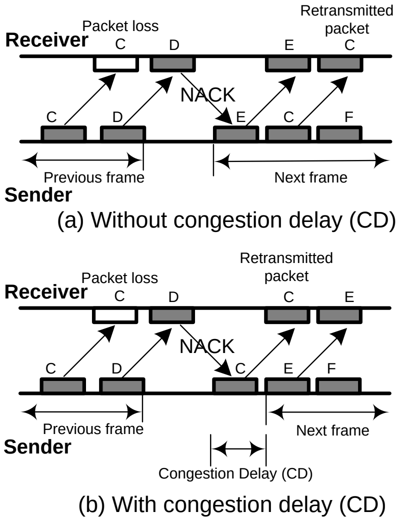


Figure 3: Congestion delay

Congestion delay also acts as instant congestion control by delaying next packet transmission in response to network congestion. Congestion delay produces temporary frame rate reduction as given by Equation 1.

$$FR_{CD} = \frac{1}{\frac{1}{FR_{init}} + CD} \qquad 1$$

For instance, if the initial frame rate (FR_{init}) is 25fps, and congestion delay (CD) is 0.01s, then the frame rate caused by congestion delay (FR_{CD}) is 20fps. This rate reduction gives the network time to reduce congestion, which potentially reduces packet loss. The congestion delay value should be smaller than inter-frame gap to avoid current frame competing with the next two frames.

SIMULATION ENVIRONMENT

WiMAX System

In order to evaluate the proposed protocol for video surveillance application over WiMAX, we conducted WiMAX simulation using the NS-2 simulator with the NIST WiMAX adds on module (National Institute of Standards and Technology 2007). The WiMAX transmit power and receiver threshold are set to provide 1000 m coverage radius. The modulation is 64 QAM, with two ray ground propagation model. By using this propagation model, measurement is in ideal line of sight path. The cyclic prefix is set to 0, which means no repeating frame preamble to avoid fading. The total cell bit rate is 10 Mbps, 7 Mbps are allocated for video traffics in uplink stream and 3 Mbps downlink are intended for negative acknowledgement services.

The simulated surveillance application has 4 mobile nodes (MN) within one base station. Each node has a different speed to represent some possible surveillance positions. Node 0 is fixed. Node 1 is set to a walking speed, 1.39 m/s. Node 2 and Node 3 are assumed to be attached in vehicles such as a bus or tram. Node 2 moves at 4.44 m/s and Node 3 speed is 6.67 m/s. The hand off process is not presented in this simulation. The single cell network configuration is shown in Figure 4.

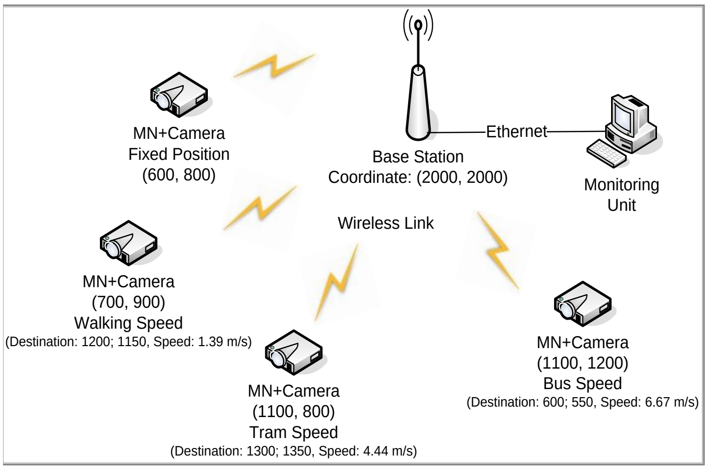


Figure 4: Network configuration

The Observed Protocols

We compared the proposed protocol to UDP, TCP, BVS, DCCP and RBUDP.

Traffic Source

The traffic source uses a video trace which contains a list of packet sequence number, byte length, frame types, and time stamp from real video source, akiyo.yuv with Common Intermediate Format (CIF) resolution 352 x 288. This video trace is used as simulated traffic in simulation, where the received pattern is reconstructed from the received packets based on the original video. The traffic generation and reconstruction in the NS-2 simulator use the Evalvid video evaluation framework (Klaue et al. 2003; Ke et al. 2008). Table 1 shows the simulation parameters.

Table 1: Simulation parameters

Parameter	Value
Sequence	akiyo.yuv
Frame rate	30fps
Frame type	IPP
Video codec	MPEG4
Video bit rate	559.35 Kbps
Packet size	1024 bytes
Group of Pictures	30 frames

Performance Metrics

The performance evaluation was conducted by observing sending and receiving ports of each connection and noting the required values such as packet sequences, sending and receiving times, packet types: acknowledgement packets, and data size. The measurement in NS-2 follows those in (Ke et al. 2008).

The presented performance metrics were obtained as the averages of all nodes. The metrics are:

- Packet delay: one way delay, obtained by subtracting the sending time from the receiving time.
- Frame delay: the latest receiving time of the packets within one frame, subtracted by the frame time stamp.
- Jitter: the absolute value of subsequent delay differences.
- Fluidity: the frame distance, obtained from the difference of current and next frame' receive time.
- Packet loss: number of lost packets divided by the total transmitted packets (in percentage).
- Cumulative throughput: the total number of received bits.
- PSNR: peak signal to noise ratio, obtained by comparing the reconstructed video from the received packets and the original video source.

RESULT AND ANALYSIS

The Protocol Performance

We compared quick response and inter-frame retransmission scheduling. Figure 5 shows the results. Inter-frame scheduling generates lower packet delay and jitter, less packet loss, closer fluidity to the original video, higher cumulative throughput and PSNR than quick response scheduling. Although its frame delay is slightly higher than quick response scheduling, the overall performance of IR scheduling is better than QR scheduling.

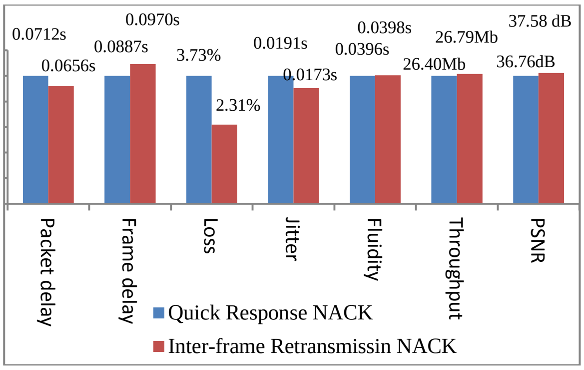


Figure 5: Comparison of QR and IR performance

By applying priority policy to IR scheduling (that is sending NACK packets only if lost packets are parts of the prioritized frames), the protocol is able to reduce packet and frame delays significantly (about 10ms and 32ms in average). Figure 6 shows the comparison of IR scheduling and prioritized IR scheduling. The prioritized one suffers higher packet loss which reduces throughput and video quality. However, in real time video transmission, low packet and frame delays are more important.

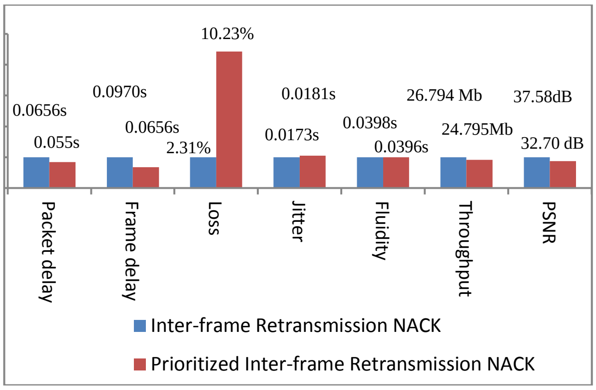


Figure 6: Comparison of IR and prioritized IR performance

The delay parameters gained by prioritized scheduling as shown in Figure 6 should be suppressed further to produce better characteristics for video transmission purpose. As described in Figure 3, congestion delay is expected to achieve the expected performance. Congestion delay should be less than the frame distance which means higher than 0 and lower than 0.004s. The smaller the value, the less the effects to next packet delay. We have tested various CD values as shown in Figure 7. The delay characteristics are relatively constant when CD values are less than 0.001s. However, they change alternately afterwards. In average, congestion delay successfully reduces delay of prioritized IR scheduling.

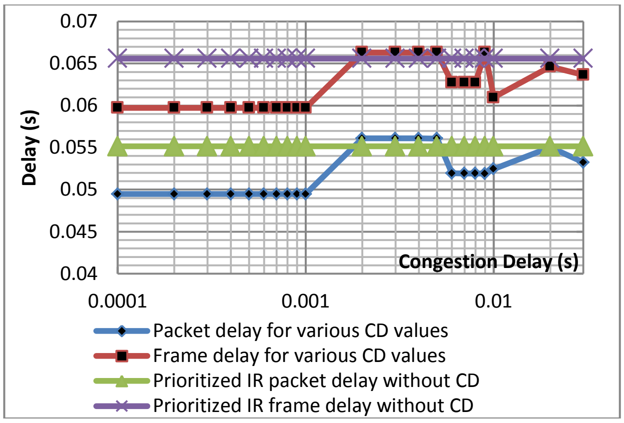


Figure 7: Congestion delay performances

Figure 8 depicts the performance enhancement of the proposed protocol by applying a 0.001s congestion delay. The average packet and frame delays plunge to 0.0495s and 0.0597s respectively. The average jitter is also reduced to 0.0169s. Even if packet loss increased causing a decrease in the cumulative throughput, the congestion delay preserved prioritized frames better. This is shown by the increase of the PSNR, which means that the protocol successfully avoids more loss on prioritized frames and produces better video quality.

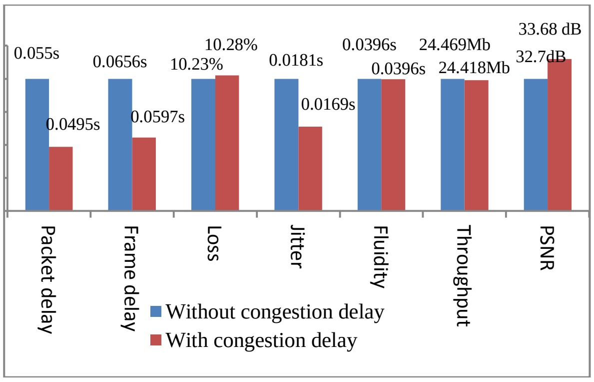
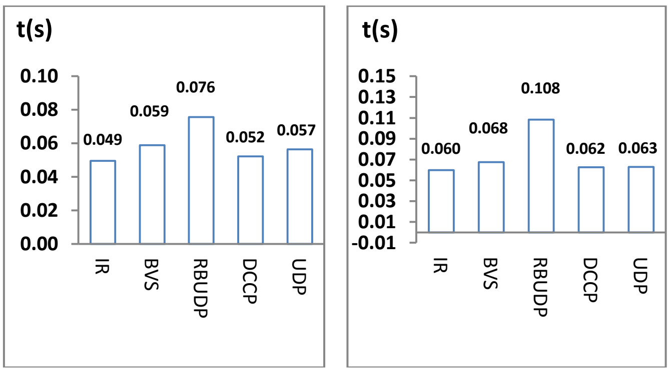


Figure 8: Performance comparison of prioritized IR without and with congestion delay

Performance Comparisons

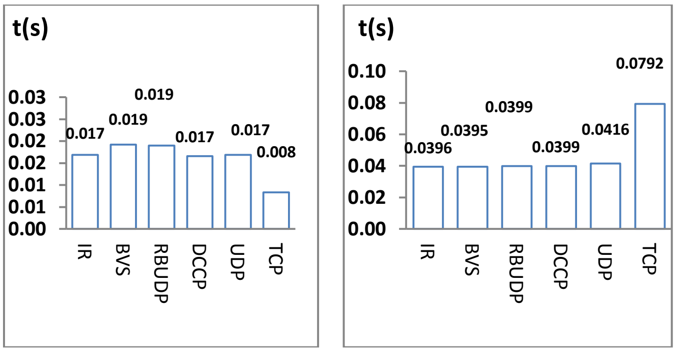
Figure 9 shows the delay characteristics of the examined protocols. Unlike UDP, BVS and RBUDP, the proposed protocol reacts to network congestion by postponing the next packet transmission. This response helps the proposed protocol to reduce network queue and suppress end to end delay. On the other hand, although DCCP and TCP implement congestion control to deal with congestion, these protocols require certain observation periods before reducing or increasing transmission rate. DCCP requires feedback packet containing receiver observation, while TCP implements time out before detecting network congestion. By arranging retransmission time and quickly responding to packet loss, the proposed protocol successfully reduces packet and frame delay. TCP experiences significant packet and frame delays, 0.468 s and 6.4 s respectively (which are not shown in Figure 6).



(a) Avergae packet delay (b) Average Frame delay

Figure 9

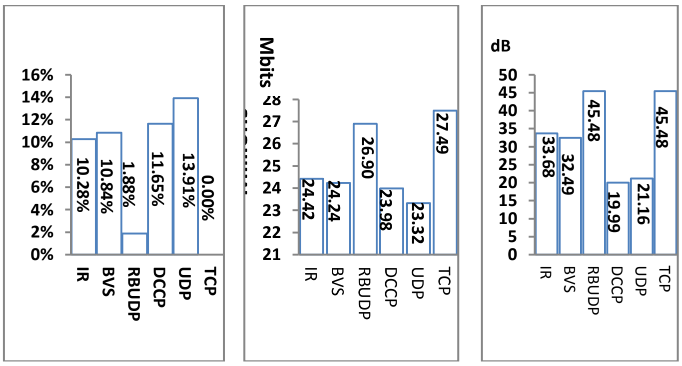
Congestion control is proven effective to avoid rough delay caused by congestion. As shown in Figure 10, TCP and DCCP produced low jitter. On the other hand, BVS and RBUDP failed to gain minimum jitter as these protocols inject retransmission traffics without dealing with congestion problem. Although the proposed protocol also streams additional retransmitted packets, congestion delay which deal with congestion is able to hold jitter as low as UDP. TCP yields the worse fluidity as it experiences high packet delay. Other protocols produce almost similar fluidity which shows that the received frames flow is similar.



(a) Average jitter (b) Average fluidity

Figure 10

In comparison to UDP, DCCP and BVS, IR reduces packet loss significantly. The loss is 3.5% lower than UDP, 1.37% lower than DCCP and 0.56% lower than BVS. Therefore the proposed protocol has higher throughput than those protocols. Furthermore, IR is able to preserve priority packets better than BVS which also retransmits priority packets. Consequently, IR produces better video quality as shown in Figure 11c. Although IR has higher packet loss and lower PSNR value than TCP and RBUDP, its low delay characteristics are more desirable for real-time video transmission.



(a) Packet loss (b) Cumulative throughput (c) PSNR

Figure 11

CONCLUSION AND FUTURE WORK

We have proposed an inter-frame retransmission (IR) protocol to reduce packet loss in video surveillance over WiMAX. The prioritized inter-frame scheduling with congestion delay method is able to make the proposed protocol perform better than existing protocols such as BVS, DCCP and UDP. Packet and frame delays as well as packet loss are reduced significantly. The protocol is also able to preserve prioritized frames so that video quality can be maintained.

Since mobile nodes in video surveillance move dynamically, congestion delay should also be dynamically analyzed and updated to enhance performance. Future work may optimize congestion delay values in response to various network conditions. Protocol deployment in other network may lead to different results as the implemented methods within the proposed protocol are optimized only for surveillance scenario over WiMAX.

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