TRANSPORT AND MAC CROSS-LAYER PROTOCOL FOR VIDEO SURVEILLANCE OVER WIMAX

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ABSTRACT

Video surveillance is an emerging application for activity and security monitoring. Outdoor surveillance applications can take advantage of a WiMAX network to provide installation flexibility and mobility. A WiMAX-based surveillance system can be implemented as a dedicated network which only serves surveillance nodes to ensure high reliability. However, wireless video transmission is prone to interferences which degrade video quality. This paper proposes a novel transport and MAC cross-layer (TMC) protocol which aims at reducing delay and increasing video quality by integrating a transport layer protocol and bandwidth allocation within WiMAX. The simulations show that the proposed protocol outperforms existing protocols.

INTRODUCTION

Most CCTV and IP surveillance systems use coaxial and Ethernet cable networks for indoor surveillance. Outdoor surveillance systems rely on wireless LAN and point to point radio technologies. Although research on the use of cellular networks for surveillance application exists, real implementations are hardly found since the channel bandwidth is limited. WiMAX (Worldwide interoperability for Microwave Access) is a wireless broadband technology that offers higher capacity than Wi-Fi networks and wider coverage than cellular networks. WiMAX has experienced intensive development from fixed wireless applications, mobile WiMAX, up to standard with 4G capabilities. This makes WiMAX a promising technology for video network infrastructures. Surveillance applications such as multi surveillance cameras placed on high rooftop buildings in urban areas and rural surveillance have the potential to be implemented in aWiMAX network.

Since bandwidth allocation in WiMAX is application dependent, various scheduling techniques have been proposed (Dhrona et al. 2008) to improve the application performance. Each application, including video surveillance, requires particular scheduling and bandwidth request methods. Suitable bandwidth allocation leads to high WiMAX link performance. On the other hand, the transport layer protocol determines the end to end performance as it provides packet transmission for host to host applications. A high performance link provided by WiMAX will not be optimal if the chosen transport layer protocol is poor. TCP (Transmission Control Protocol) provides high reliability data transfer which ensures that each packet is received successfully and sequentially. It guarantees the quality of delivered video. However, routine acknowledgements and retransmissions in TCP generate a significant delay which is not suitable for real time applications. Furthermore, interferences and signal disruption in the wireless channel may cause TCP to experience significant delay as it keeps trying to resend the lost packets. In contrast, UDP (User Datagram Protocol) is able to reduce delay in video delivery. The drawback is that UDP does not respond to network conditions as it keeps sending data regardless of network congestion. UDP potentially makes the congested network even worse. Various transport protocols have been proposed to enhance protocol performances. An overview of these protocols is given in the next section.

This paper combines a transport layer protocol and bandwidth allocation in WiMAX to achieve better video surveillance performance. The transport and MAC cross-layer term refers to a method that explores interactions between the MAC layer and the transport layer. Our proposed method is aimed at enabling the MAC layer to support retransmission in the transport layer. We avoid two-way interactions to prevent processing delays. Instead, the proposed method enables the MAC layer to read the transport layer header in order to provide service to the transport layer. Besides performance improvement, the proposed method requires only a minor change in the MAC layer and the WiMAX device is still compatible with other implementations.

The rest of the paper is organized as follows. After discussing related works, we present the proposed method. Then the proposed method and the existing solutions are compared using ns-2 simulations. The performance of the proposed protocol is initially compared to existing transport layer protocols using the same bandwidth allocation. The improvement is then examined for various bandwidth allocations. Finally, the conclusion section summarizes the contribution and suggests future work.

RELATED WORKS

The TCP/IP protocol stack defines four independent abstraction layers for IP based networks. . Data is passed from one layer to the other by using header encapsulation and de-capsulation. The idea of layer separation may work well for wired communication, but not in a wireless environment where device characteristics and channel quality often vary. Several cross-layer solutions have been proposed for wireless communication.

Raisinghani and Iyer (2004) outlined cross-layer possibility in different layers. In WiMAX implementation, cross-layer approaches mostly occur between PHY and MAC layers. For instance, Noordin and Markarian (2007) implemented a cross-layer optimizer between MAC and PHY layers to maximize WiMAX performance. The optimizer collects data from both layers and returns the optimized parameters for bandwidth allocation in MAC layer as well as coding selection in PHY layer. Cross-layer approach can also be performed between the MAC layer and the network layer. This was used in (Mohanty and Akyildiz, 2007) to provide seamless handover. Meddour et al. (2011) implemented a cross-layer approach between MAC and application layer to optimize unicast and multicast video streaming in WiMAX network. Our work completes the cross-layer schemes by proposing the cross-layer MAC and transport layer protocol. The proposed Transport MAC cross-layer protocol provides high performance end to end transport layer connection in WiMAX network that can replace the existing UDP protocol. Unlike the aforementioned cross-layer schemas, our proposed cross-layer schema does not require a new protocol data unit (PDU) or a separate layer entity. The proposed scheme uses existing PDU and entities. Therefore, the cross-layer design is much simpler and fast. However, the protocol does not aim to compete against the existing schemes, as each cross-layer design has a different emphasis. The MAC-PHY, MAC-Application, and the proposed cross-layer design could be combined to achieve the expected performances.

Cross-layer design between MAC and transport layer protocol has been explicitly used in some existing reliable transport layer protocols which employ congestion control. Ye, Wang and Huang (2011) used the cross-layer method to provide fairness for some TCP flows. Work by Zhai et al.(2007) proposed WCCP (Wireless Congestion Control Protocol) which is effective only for static ad hoc network. WCCP adjusts sending rate based on channel utilization. However, reliable based protocols are not suitable for multi sources real-time video transmission over WiMAX as those protocols exert tremendous delay (Larzon et al., 1999). Our proposed cross-layer design does not explore channel quality to support congestion control as reliable protocols did. The protocol is intended to improve existing unreliable protocols; therefore the implemented methods should not change the nature of the unreliable protocol. The congestion avoidance is performed as simple as possible and the retransmission effort is performed only once.

Various works have proposed improvements on unreliable protocol performance. Reliable UDP (RUDP) adds congestion control mechanism, acknowledgement, and retransmission services to accommodate different transport protocol requirements (Bova and Krivoruchka, 1999). This protocol works between UDP and TCP. However, the excessive features make RUDP behave almost like TCP and remove the nature of unreliable protocol. UDP-Lite (Larzon et al., 1999) implements a partial checksum for the sensitive part and ignores errors in the non-sensitive part of the UDP packets. UDP-Lite performs better than UDP in terms of packet loss. However, it disables network supervision in upper layer as it masks error on transmission (Welzl, 2005). UDP-Lite requires additional processing time to determine whether data needs checksum, as well as to process it in the receiver. The ignored packet passed to application layer may not be acceptable. UDT (UDP-based Data Transfer) (Gu and Grossman, 2007) and RBUDP (Reliable Blast UDP) (He et al., 2002) are datagram protocols that work for high speed bulk data transfer link. Both protocols were aimed to solve TCP weakness which underutilize high speed network (Gu and Grossman, 2007). RBUDP employs negative acknowledgement which sends a TCP request-reply to acknowledge lost packets in a UDP based bulk transfer. UDT and RUBP are intended for single high speed link connection, which may perform worse in multiple traffics environment. BTP (Bidirectional Transport Protocol) modified UDP for tele-controlled robot application using inter-packet gap (IPG) congestion control (Wirz et al., 2009). The inter-packet gap determines the speed of data transfer. ERT (Embedded Reliable Transport protocol) added additional header on UDP to provide reliability for embedded application (Wei and Chao, 2010).

Kohler et al. (2006) proposed DCCP (Datagram Congestion Control Protocol) which employs two congestion controls; TCP-Like and TFRC-Like. DCCP is a potential transport protocol to replace UDP. However, DCCP does not retransmit lost packets and relies fully on client monitoring feedback. SCTP (Stream Control Transmission Protocol) (Stewart, 2007) was initially designed for reliable signalling and control transport protocol for telecommunications traffic running over IP networks. SCTP provides multi-homing features which enable alternative transmission path. However, SCTP performance is worse than that of DCCP for real-time video transmission (Chughtai et al., 2009). Ali etal(2011) proposed a semi-reliable transport protocol called Broadband Video Streaming (BVS). The protocol applies retransmission as soon as packet loss is detected. Our previous work (Suherman et al., 2011) has shown that inter-frame retransmission is able to improve the performance of video transmission in WiMAX. Inter-frame retransmission resends the lost prioritized packets at the end of each frame transmission. The transport layer part of our proposed protocol uses inter-frame retransmission.

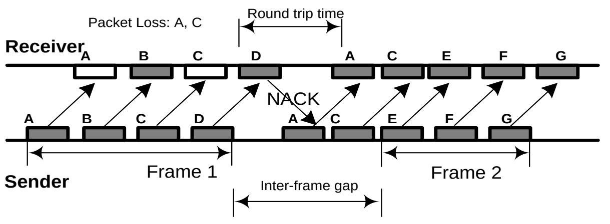
PROPOSED PROTOCOL

We assume a dedicated surveillance network which operates in non-saturated conditions and every node generates the same video bit rates. The proposed protocol aims at minimizing the delay and maximizing video quality. It consists of two parts, transport layer and MAC layer. The transport layer part uses inter-frame retransmission with congestion delay. The second part enables the MAC layer to assist the transport layer by providing sufficient bandwidth for the retransmitted packets.

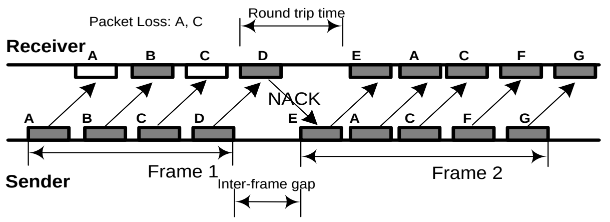
The transport layer part aims to improve the reliability of the protocol. Therefore, we employed a transport layer protocol with simple congestion control and retransmission scheme without repetition for the lost retransmitted packet. Additional bandwidth given by the cross-layer scheme in the MAC layer is the primary feature of the proposed protocol.

Transport layer part

Transport layer protocols that employ negative acknowledgement (NACK) use either quick or delayed response. In quick response, the receiver notifies the sender with a NACK packet as soon as packet loss is detected. The sender then retransmits the requested packets. For example, BVS (Ali et al., 2011) is a quick response retransmission protocol, while RBUDP (He et al., 2002) is a delayed response protocol. Inter-frame retransmission uses a delayed NACK response to acknowledge lost packets. The NACK packet is sent after receiving the last packet within one frame. The objective is to avoid multiple acknowledgements for multiple losses in one video frame. Inter-frame retransmission also aims at smoothing the network load by sending the NACK in idle time (inter-frame gap (IFG)).



(a) Inter-frame retransmission when RTT < IFG



(b) Inter-frame retransmission NACK when RTT > IFG

Figure 1: Inter-frame retransmission

Figure 1 shows how inter-frame retransmission works. We assume that packet A and packet C within frame 1 are lost. The receiver requests retransmission to the sender after receiving the last packet within frame 1 (packet D). Soon after receiving the NACK packet, the sender retransmits the requested packets. In case packet D is lost, the NACK packet is sent after receiving the next packet. If the round trip time (RTT) is smaller than IFG, the retransmission occurs in IFG. Otherwise, the retransmitted packets compete with the packets from the next frame. Therefore, inter-frame retransmission is suitable for real time video with a small number of intermediate nodes, as in video surveillance.

MAC layer part

The MAC layer part is responsible to ensure that the retransmitted packets have sufficient bandwidth. The additional MAC functionality detects NACK packet and reads its content. Based on the NACK information, MAC allocates additional bytes in bandwidth request to accommodate the retransmitted packets in the transport layer.

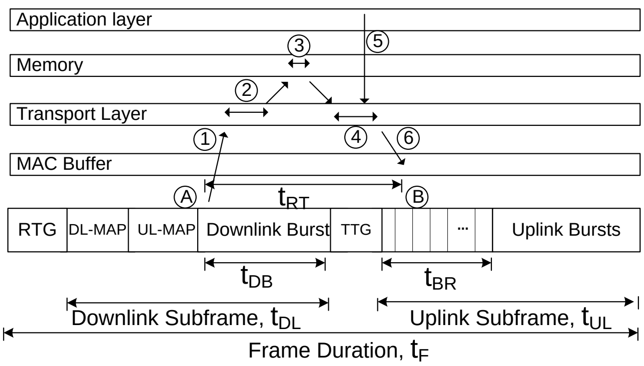


Figure 2: NACK and retransmitted packet flows

Figure 2 illustrates the need of the MAC layer part. We assume the NACK packet is received by subscriber station (SS) in the beginning of a downlink burst in a downlink sub-frame (A). MAC forwards the NACK packet to the transport layer (1). The transport layer protocol processes the packet (2), retrieves the requested packets from the memory (3) and encapsulates the retransmitted packets (4). Since the application layer periodically sends new packets (5), the transport layer may experience congestion that leads to retransmission failure. Packets which reach the MAC layer are queuing in the MAC buffer (6) before being transmitted. Bandwidth request is performed based on the number of bytes in the MAC buffer.
In order to minimize the delay for the retransmitted packets, the access time for the retransmitted packets, t_{RT}, should be as small as possible so that the packets has been in the MAC buffer by the time the nearest bandwidth request is made. Since SS bandwidth request opportunity is randomly chosen within the bandwidth request period t_{BR}, packets should be in the queue at the latest just after TTG (transmit/receive transition gap). In the best case when NACK is received in point A, t_{RT} should be less than $(t_{DB} + \mathrm{TTG})$, where t_{DB} is the downlink burst duration. In the worst case when NACK is received in point B, t_{RT} should be less than TTG. Otherwise, the retransmitted packet will miss the nearest bandwidth request opportunity and must wait for another bandwidth request opportunity which leads to an additional delay of at least one full frame duration, t_F.

By using MAC functionality, the nearest bandwidth request should not wait for the retransmitted packet arriving in the queue to add bandwidth allocation request. Instead, the MAC layer adds additional tasks. First, MAC reads the NACK packet content to determine the number of requested packets. Then, MAC informs the bandwidth request module to add additional bytes in incoming bandwidth request packet. As a result, the requested bandwidth includes the retransmitted packets although they do not appear in the MAC buffer yet.

Since the NACK packet flows through base station (BS) to SS, the MAC functionality for the proposed TMC protocol can be implemented in either BS or SS. The advantage of the SS implementation is that the additional bandwidth is allocated after NACK packet is safely received. On the other hand, BS can allocate additional bandwidth directly without waiting for bandwidth request from SS. However, BS implementation may decrease network performance as BS will have more tasks. Moreover, if NACK packet is lost, then bandwidth is wasted.

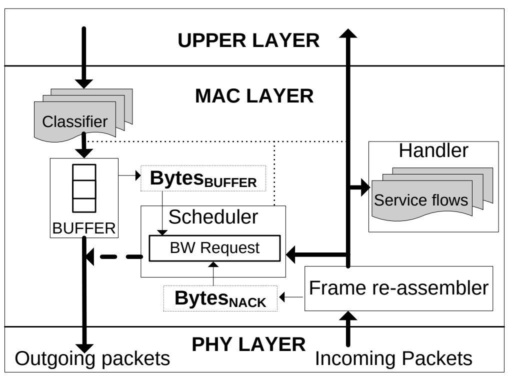
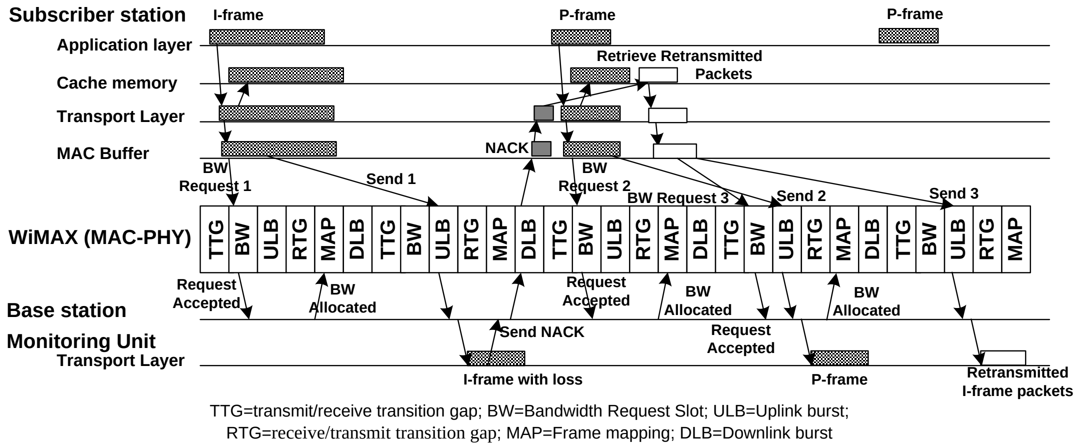


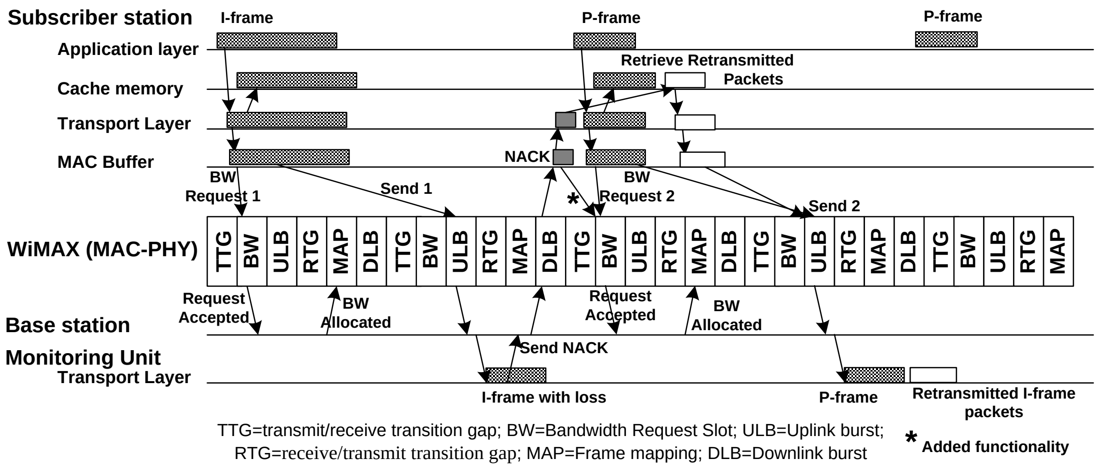
Figure 3: MAC layer implementation

Figure 3 shows the SS implementation of the proposed cross-layer design in the MAC layer part. The scheme is based on the NIST WiMAX module (NIST, 2007). The frame re-assembler in the MAC layer reads the NACK packet and notifies the scheduler to add the number of requested bytes (BytesNACK) in the bandwidth request. In turn, the scheduler sends a bandwidth request based on data size on MAC buffer (BytesBUFFER) and the retransmission bytes (BytesNACK).

Figure 4shows a comparison of simplified layer interactions of the inter-frame retransmission without and with MAC cross-layer functionality. In Figure 4 (a), the bandwidth for the retransmitted packets is separately requested as the packets are not available by the time the SS sends a bandwidth request to BS. Consequently, instead of sending the retransmitted data in the nearest uplink burst, SS will allocate it to the uplink burst after the next burst. This postponement increases the packet delay. On the other hand, the MAC cross-layer protocol accelerates packet retransmission as the earliest bandwidth request accommodates the retransmitted packets. The retransmitted and the current data are sent in the same burst.



(a) Inter-frame retransmission without MAC cross-layer functionality



(b) Inter-frame retransmission with MAC cross-layer functionality

Figure 4: Subscriber station based MAC cross-layer protocol

SIMULATION ENVIRONMENT

In order to evaluate the proposed methods for a dedicated video surveillance network, we conducted simulations using the ns-2 simulator with the WiMAX module taken from NIST (2007). The transmit power and receiver thresholds are set to provide 1000 m coverage radius. The modulation is 64 QAM, with a two-ray ground propagation model. The downlink/uplink ratio is 0.3.The simulated surveillance application has 4 mobile nodes. Node 0 is fixed (0 m/s). Node 1 is at walking speed of 1.39 m/s. Node 2 and Node 3 are assumed to be in a public transportation. Node 2 moves at 4.44 m/s and Node 3 at 6.67 m/s. The network configuration is shown in Figure 5.

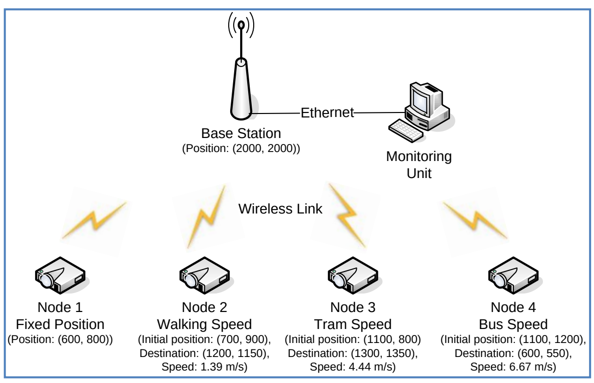
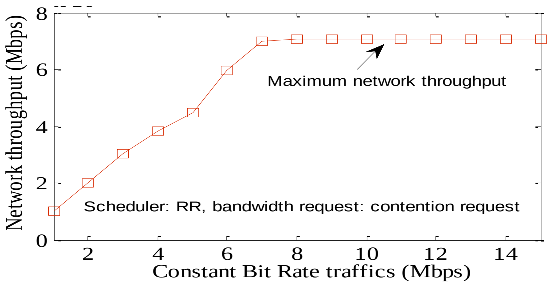
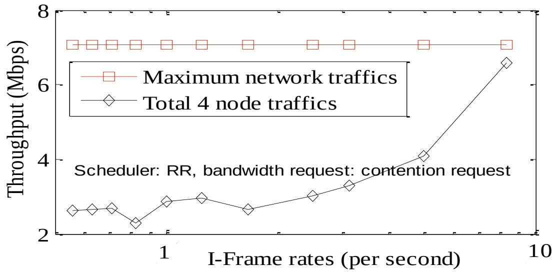


Figure 5: Network configuration

The number of mobile nodes was chosen to simulate a non-saturated network, which means that the traffic load is smaller than the network resources. This is important as the surveillance network should provide sufficient bandwidth in order to maintain video quality. By using constant bit rate (CBR) tests from 1 to 15 Mbps, we obtained a saturated uplink bandwidth of 7 Mbps (Figure 6a). Since the proposed methods deal with packet/frame types, we increase the traffic load based on I-frame rate (GOP) instead of number of SSs. The total video rate for each simulation is depicted in Figure 6b.



(a)



(b)

Figure 6: Traffic rate in simulation

The traffic sources were generated from the akiyo_cif.yuv video. Its video trace was used as simulated traffics in the ns-2 simulations, where the received patterns were reconstructed based on the original video. The traffic generation and reconstruction in the ns-2 simulator were based on the Evalvid video evaluation framework from (Klaue et al., 2003). The prioritized frames were set for I-frames and the transport layer protocol used was UDP. Table 1 shows the traffic parameters.

Table 1: Simulated traffic parameters

Parameter	**Value**
Video sequence	akiyo_cif.yuv
Frame rate/type	30fps/IPP
Video codec	MPEG4
Video bit rate	559.35 Kbps for GOP of 30 frames
Group of Pictures	3, 5, 8, 10, 15, 20, 25,30, 35, 40, 45

The performance evaluation was conducted by observing sending and receiving ports in each connection. The measurement in the ns-2 simulator refers to those in (Ke et al., 2008).The main performance metrics are the average of delay and PSNR (Peak Signal to Noise Ratio) of the four nodes. Measurement points are in SSs (sender) and in the monitoring unit (receiver). PSNR is obtained by reconstructing video from the received packets and comparing it to the original source.
First, we evaluated the performances of the TMC protocol using round robin scheduler with contention request. We did the same experiment for inter-frame retransmission (IR) without cross-layer (Suherman et al., 2011), BVS(Ali et al., 2011) and UDP. Afterwards, we applied the protocol for various scheduling algorithms to confirm the superiority of the proposed method.

RESULTS AND ANALYSIS

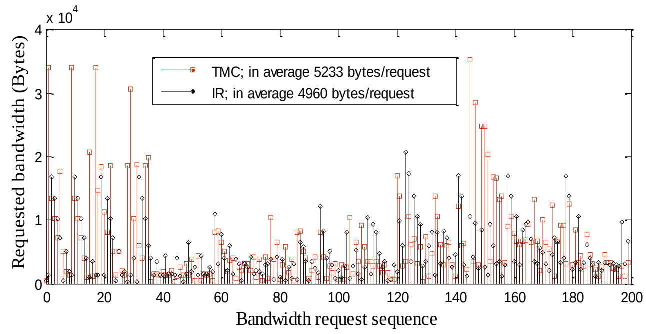
The impact of MAC cross-layer

Transport layer packets queue in the MAC buffer of the SS before being transported by the physical layer. MAC transfers the data to the uplink sub-frame based on the duration allocated by BS in UL-MAP. The duration itself is decided by BS based on SS bandwidth request and the available bandwidth. Since the main feature of the MAC cross-layer is additional bandwidth allocation for the retransmitted packets, the proposed protocol gains higher bandwidth than the basic IR protocol.

Table 2: Allocated bandwidth comparison (GOP 30)

Protocol	IR	TMC
Number of bandwidth requests	1270	1268
Average requested bandwidth	4960	5233
Number of uplink transmission	1530	1522
Average allocated bandwidth	2419	2430
Network utility	55.29%	55.54%

For the simulated traffics with GOP 30, TMC generates 1268 bandwidth requests, while IR produces 1270 requests (Table 2). In average, TMC requested bandwidth 273 bytes morethan IR. From those requests, BS allocates in average 2430 bytes/uplink transmission for TMC and 2419 bytes/uplink transmission for IR. TMC uses the network better than IR. Since the frame duration is 5ms and the maximum network throughput is 7Mbps, the network utility of the TMC protocol is equal to (2430x8/0.005)/7000000x100%=55.54%. IR utility is 55.29%. Figure 7 shows the requested and the allocated bandwidth for the first 200 bandwidth requests.



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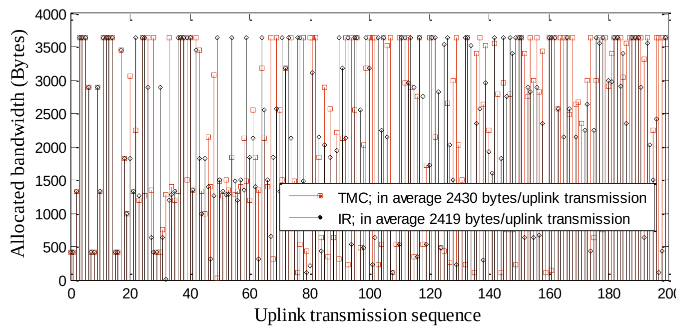
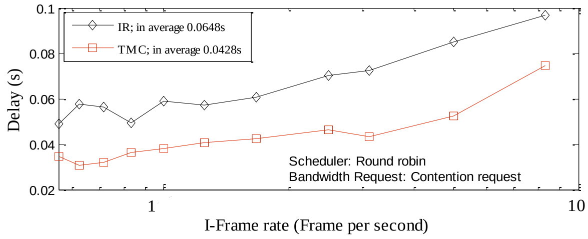


(b)

Figure 7: Bandwidth request comparison

Since the additional bandwidth is requested before the retransmitted packets available in MAC buffer, the allocated bandwidth can be used by regular data, even if the retransmitted packets failed to be retrieved. The higher bandwidth allocation and network utility in the proposed protocol produce lower delay and higher video quality. Figure 8 shows the performance comparisons between IR and TMC. TMC consistently reduces packet delay for all I-frame rates. Although the PSNR decreases when sending data with I-frame rate 1 fps, this is probably caused by the undecodable subsequent error frames.



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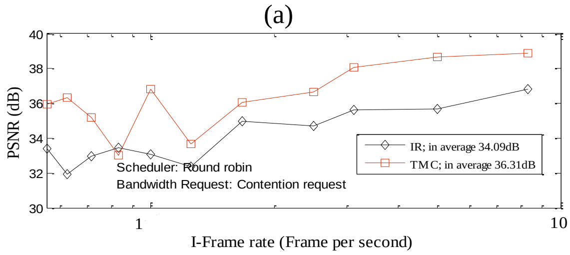
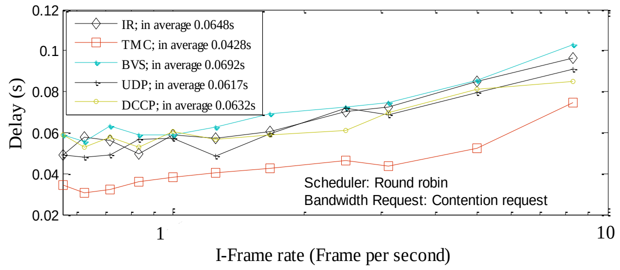


(b)

Figure 8: Performance comparison between IR and TMC

Transport layer protocol comparison

Figure 9 compares TMC to existing protocols. TMC was able to reduce UDP delay by 18 to 37%. The PSNR improvements were around 14.3 to 149.5%, 12.6 to 150.2%, 21.3 to 184.3% and 17.9 to 120.2.3% over IR, BVS, UDP and DCCP, respectively. Other existing protocols such as SCTP and RBUDP are not presented as they have been compared in (Chughtai et al., 2009). The result shows that TMC outperforms the existing protocols for surveillance application over WiMAX with uniform traffics.



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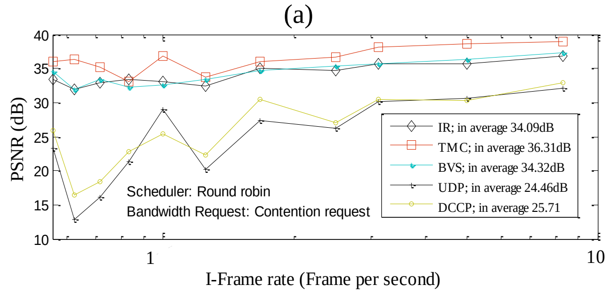
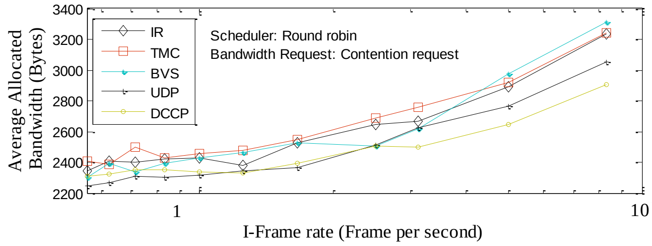


(b)

Figure 9: Performance comparison between TMC and other protocols

TMC has lower delay than UDP because it requested more bandwidth when loss occurred. As shown in Figure 10, TMC received more bandwidth than other protocols. TMC experienced lower allocation than BVS for high I-frame rates as the maximum network throughput (Figure 6b) limits the bandwidth for the retransmitted packets. However, the limited bandwidth does not reduce TMC performance as the cross-layer functionality still produces better allocation.



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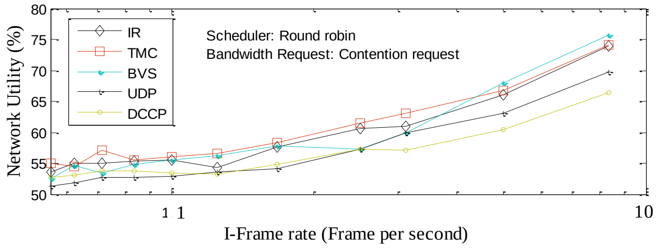


(b)

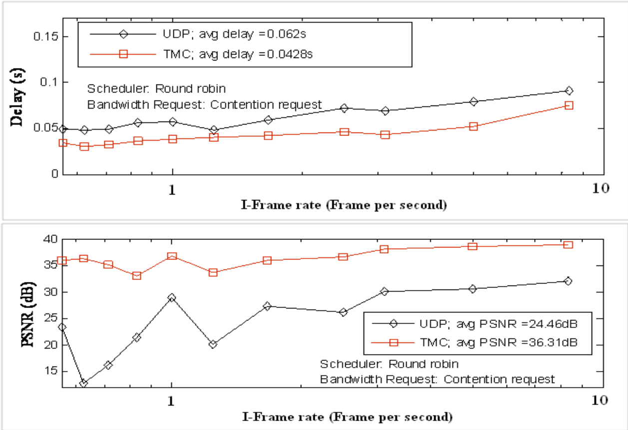
Figure 10: Comparison of the allocated bandwidth and network utility

On the other hand, although BVS received higher bandwidth for higher I-Frame rates, bandwidth may be wasted as multiple NACKs may disturb regular packet transmission. UDP and DCCP suffer low bandwidth allocation as both protocols do not retransmit lost packets. UDP does nothing to increase utility.

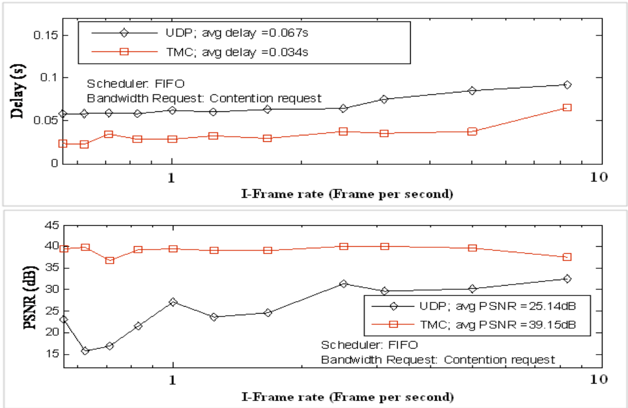
Protocol performance over various schedulers

In order to ensure that the proposed protocol is suitable for various WiMAX schedulers, we evaluated it with Round Robin (RR), First In First Out (FIFO) (Dhrona et al., 2008), Frame based (Kang and Zakhor, 2002), and the Earliest Deadline First (EDF) (Ferrari and Verma, 1990) schedulers for dedicated video surveillance over WiMAX (Figure 11).

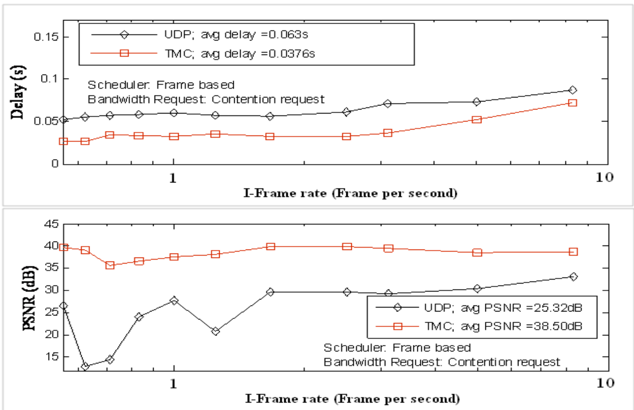
The proposed protocol applied with RR, FIFO and frame based schedulers significantly reduced the delay and increased the PSNR. On the other hand, the implementation of the protocol with the EDF scheduler experienced irregular delays. The reason is that the EDF scheduler is not suitable for applications with uniform traffics as the traffics have similar behaviour and deadlines, while the EDF scheduler classifies the allocated data based on traffic deadlines. As a result, BS performs unnecessary sorting which introduces delay. Although TMC failed to reduce the delay for several I-frame rates, it consistently increased the PSNR values.



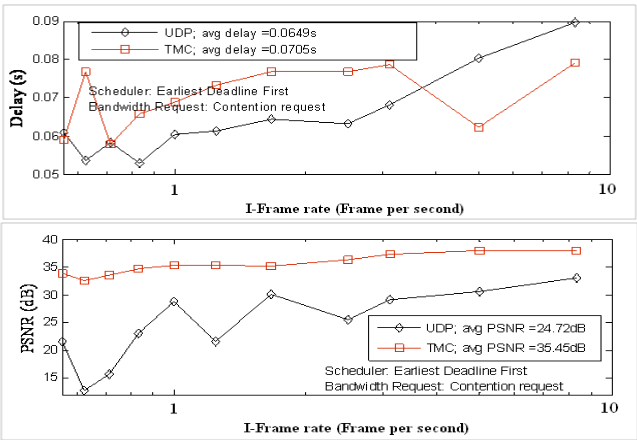
a. Round robin scheduler



b. FIFO scheduler



c. Frame based scheduler



d. EDF scheduler

Figure 11: TMC performance over various schedulers

CONCLUSION AND FUTURE WORK

This paper has proposed a transport and MAC cross-layer (TMC) protocol for a dedicated video surveillance network using WiMAX. The proposed protocol has two components that work separately in two layers. The inter-frame retransmission is used in the transport layer, while the MAC layer adds the capability to read the NACK packet content and uses the information to increase the number of bytes in bandwidth request. The simulations show that the proposed protocol outperforms existing transport layer protocols for WiMAX based video surveillance. It is able to achieve lower delay than UDP and better video quality than other protocols.

The proposed protocol is proven to work well with various scheduling algorithms. The use of the proposed protocol combined with a suitable scheduler improves a WiMAX application for a dedicated surveillance network with real-time video traffic. Further work will be carried out to assess the protocol performance for more general network settings as the simulations in this paper have limited bandwidth.

ACKNOWLEDGEMENTS

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