UDDP: A User Datagram Dispatcher Protocol for Wireless Multimedia Sensor Networks

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***Abstract*—The quality of service in Wireless Multimedia Sensor Networks (WMSN) is related to packet loss rate. Recently different studies have been done on developing efficient protocols in the transport layer for controlling packet loss in WMSN. However, all of these protocols are independent of the characteristics of multimedia content. In this paper, a novel transport layer protocol, called User Datagram Dispatcher Protocol (UDDP), is proposed to minimize the packet loss ratio in WMSN by considering traffic characteristics, the inter-arrival pattern of packets and packet priority. UDDP is a new cross-layer transport layer that uses information of MAC and application layers to distribute packet arrivals in a GOP. The experimental results show that our protocol provides performance improvement in terms of high video quality, packet loss, energy conservation and delay compared to other protocols such as CCF and PCCP.**

***Index Terms*—WMSN, Energy aware, Transport layer, Inter-arrival pattern, Video transmission.**

I. Introduction

A Wireless Multimedia Sensor Network (WMSN) [1] is a network of wireless sensors that can collect multimedia content, such as audio and video. In WMSN, multimedia traffic is mainly transmitted from a number of sensor nodes to a base station (sink) node. According to the physically small sensor nodes with limited battery life and the requirements of multimedia applications, WMSN have various characteristics, such as bursty and unbalanced mixture traffic, resource constraints (for instance, bandwidth and memory), and power consumption. Therefore, developing new protocols to minimize energy consumption while satisfying the quality of service requirements is an important problem.

In WMSN sensor nodes produce snapshots or multimedia streaming. The video frames can be encoded to three different types: Intra frame (I), Predictive frame (P) and Bidirectionally predictive frame(B). Since the packet size of WMSN is too small (about 100 bytes [2]), the video frames are fragmented to different number of packets. Some of the packets belong to the I-frames, some to the P-frames, and others are related to the B-frames. These packets are called I-, P- and B-packets respectively.

The quality of multimedia applications in WMSN is related to packet loss rate. Packet loss in WMSN occurs due to interference in wireless environment and bursty traffic. Since transport layer is responsible for controlling packet loss, recently different studies have been done on developing efficient protocols in this layer for congestion control in WMSN. STCP [3], Fusion [4], CODA [5], CCF [6], PCCP [7] and DPCC [8] are the most well known protocols. All of these protocols are focused on avoiding the packet loss in WMSN independent of the characteristics of multimedia content.

In this paper, a new transport layer protocol, called User Datagram Dispatcher Protocol (UDDP), is proposed to minimize the packet loss ratio in WMSN by considering traffic characteristics, the inter-arrival pattern of packets and packet priority. There are two different types of traffic in WMSN, monitoring and event-driven traffic [9]. In UDDP, we estimate the probability of bursty traffic in the future based on the traffic type. In bursty traffic that leads to high packet loss ratio, the proposed protocol only sends packets with high priority. Therefore, not only a lot of energy in the source nodes is saved but also the video quality is improved. Moreover, since the packet size of WMSN is too small, the video frames are fragmented to different number of packets. This leads to a specific inter-arrival pattern of packets in WMSN. UDDP adjusts the inter-arrival pattern of packets by adding a queue in the transport layer to provide equal delay for packets in a Group of Pictures (GOP).

The rest of the paper is organized as follows; Related work are presented in section II. Section III introduces the proposed protocol. Section IV provides performance evaluation, and the concluding remarks are presented in section V.

II. Related Work

Recently several studies have been done on developing efficient protocols in transport layer for WMSN. STCP [3], CCF [6], PCCP [7] and DPCC [8] are the most well known protocols. In the following, we describe these protocols in detail.

Sensor Transmission Control Protocol (STCP) [3] is a generic transport protocol that only uses the information of this layer to control congestion in WMSN. Rate adjustment in STCP is done by Additive Increase Multiplicative Decrease (AIMD) scheme. STCP guarantees application requirements and improves energy consumption. However, this results in high packet loss during congestion in network. Fusion [4] uses prioritized Medium Access Control (MAC) to assign a higher priorities to packets of the sensor nodes with full buffer. This results preventing packet loss in these nodes. Therefore, buffer queue length is congestion index in Fusion.

COngestion Detection and Avoidance (CODA) [5], uses three mechanisms for congestion control: congestion detection (to detect any congestion in the network by employing the buffer length or channel occupancy), open-loop hop-by-hop backpressure (to broadcast backpressure signal to other nodes when a node detects congestion in the network) and closed-loop multi-source regulation (it starts when the source event rate is more than the maximum theoretical throughput of the channel). Congestion Control and Fairness (CCF) [6] is a hop-by-hop based congestion control that provides a scalable algorithm to ensure the fairness in packet delivery. However, it is limited to many-to-one topologies and the fairness mechanism leads to low throughput in WMSN.

Priority-based Congestion Control (PCCP) [7] introduces an upstream and priority-based congestion control protocol. PCCP uses three components for congestion control, Intelligent Congestion Control (ICD), Implicit Congestion Notification (ICN), and Priority-based Rate Adjustment (PRA). However, it doesn't have any mechanism to handle prioritized mixed traffic. Dynamic Priority Based Congestion Control (DPCC) [8] assigns a dynamic priority to each packet to localize traffic of closed nodes to the base station in highly congested WMSN.

Table I shows a brief comparison of the mentioned congestion control protocols. In summary, since multimedia applications in WMSN have specific characteristics, they should be considered in developing a transport layer protocol. However, all of the aforementioned congestion control protocols for avoiding packet loss in WMSN do not support multimedia content.

III. The Proposed Transport Layer Protocol

The Proposed protocol called User Datagram Dispatcher Protocol (UDDP) is a new cross-layer transport layer that uses information of MAC and application layers to distribute packet arrivals in a GOP (refer to Fig. 2). Application layer sends the number of packets in each GOP along with the frame type to transport layer. Moreover, MAC layer sends Maximum Transmission Unit (MTU) of packet to transport layer for fragmenting each video frame to suitable number of packets. In the following, we describes the proposed protocol in detail after discussion on fundamental concepts.

The video frames captured by a camera-enabled wireless sensor node can be encoded to three types of video frames (I, P and B) in application layer (refer to Fig. 1) [11]. A coded video streams includes a group of successive pictures (video frames) called GOP. A GOP contains one I-, multiple P- and B-frames. The number of video frames (e.g., 9, 12 and 24) and their orders (e.g., IBBPBBPBB, IBPBPBPB) in a GOP can vary. Moreover, the video frames have different importance for the overall quality. Since other P- and B- frames depend on I-frames, I-frames are the most important among all three frame types.No other frames depend on B-frames and these frames are at the least importance. Therefore I- and B- frame loss has the most and the least effect on video quality in multimedia applications respectively [11].

The encoded video frames are packetized in transport layer and transmitted over WMSN. Since packet size in WMSN is too small (about 100 bytes for IEEE 802.15.4 [2]), each video frame is packetized to different number of packets based on frame type. For example, I-frames that are the biggest frames in GOP are fragmented into the largest number of packets compared to other frame types (refer to Fig. 1).

We define *Packet Inter-Departure Time* ($PIDT$) for each packet type (I, P and B) as a measure that computes the inter-departure time of two consecutive packets in transport layer (refer to Eqs. (4), (5) and (6).

$$NP'_j = NP_{I,j} + NP'_{P,j} + NP'_{B,j} \tag{1}$$

$$NP'_{P,j} = b((1-a)\overline{NP}_P + aNP_{P,j-1}) + (1-b)(NP_{I,j} - (NP_{I,j-1} - NP_{P,j-1})) \tag{2}$$

TABLE I
Comparison of WMSN Congestion Control Protocols

Protocols	Congestion detection	Congestion notification	Congestion mitigation (rate adjustment)
STCP [3]	Queue length	Implicit	AIMD-like end-to-end
Fusion [4]	Queue length	Implicit	Stop-and-start hop-by-hop
CODA [5]	Queue length and channel status	Explicit	AIMD-like end-to-end
CCF [6]	Packet service time	Implicit	Exact hop-by-hop
PCCP [7]	Packet inter arrival time and packet service time	Implicit	Exact hop-by-hop
DPCC [8]	Packet scheduling rate and Packet service rate	Explicit	Exact rate control

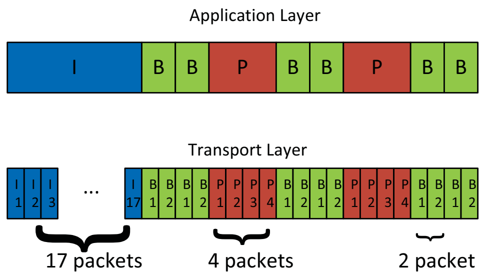


Fig. 1. Transmission of data in application and transport layer

$$NP'_{B,j} = b((1-a)\overline{NP}_B + aNP_{B,j-1}) + (1-b)(NP_{I,j} - (NP_{I,j-1} - NP_{B,j-1})) \quad (3)$$

$$PIDT_{I,j} = \frac{1/_{FR}}{NP_{I,j}} \quad (4)$$

$$PIDT_{P,j} = \frac{1/_{FR}}{NP'_{P,j}} \quad (5)$$

$$PIDT_{B,j} = \frac{1/_{FR}}{NP'_{B,j}} \quad (6)$$

Where $NP_{I,j}$ is the number of I-packets in GOP j that is known initially, $NP'_{P,j}$ is the predicted number of P-packets in GOP j, $NP'_{B,j}$ is the predicted number of B-packets in GOP j and FR is the frame rate of video (fps). Eq. (1) predicts the number of packets in GOP, based on I-frame size. In (2) and (3) we compute the number of P- and B-packets in current GOP respectively by Exponential Weighted Moving Averages (EWMA). It should be noted that a and b are constant numbers that adjust desired weights for previous GOP in EWMA. For example, assume that the application layer delivers video frames to transport layer with 30 fps and I-, P- and B- frames are packetized to 17, 4 and 3 packets respectively. Therefore, PIDT is equal to 0.0019, 0.0083 and 0.0111.

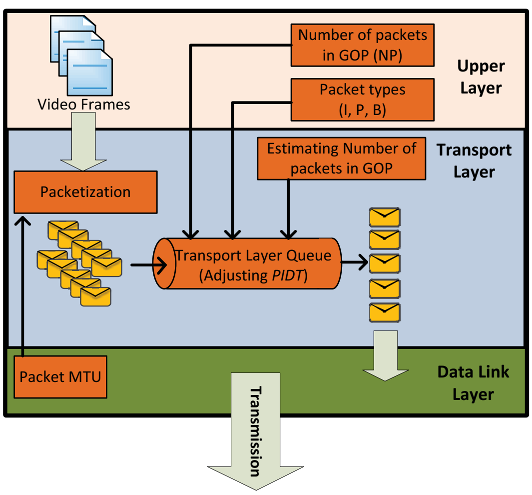


Fig. 2. Cross layer architecture of the proposed protocol

Algorithm 1 Pseudo code of UDDP protocol

for all GOPs of node i **do**
  Predicting the number of packets in GOP j by using Eqs. 1, 2 and 3.
  Finding the packet inter-departure time in GOP j ($PIDT_j$) by using Eq. 7.
  Ignoring IP_j packets with lower priority in GOP j by using Eq. 8.
  Sending packets in GOP j with $PIDT_j$ intervals.
end for

When PIDT of I-packets is lower than PIDT of P- or B-packets (the same as above example), this leads to bursty traffic during sending I-packets. To solve this problem, our idea in the proposed protocol is sending the video packets in distributed intervals in GOP. To this end, we send I-, P- and B- packets in equal packet inter-departure times that results equal PIDT for all these packets in GOP j (refer to Eq.(7)). As Fig. 2 shows, UDDP uses a queue in transport layer to adjust PIDT.

$$\begin{cases} PIDT_j = \frac{NF_j \times 1/_{FR}}{NP'_j} & \text{if } OR_s > \frac{PS \times NP'_j}{NF_j \times 1/_{FR}} \\ PIDT_j = \frac{PS}{OR_s} & \text{if } OR_s < \frac{PS \times NP'_j}{NF_j \times 1/_{FR}} \end{cases} \quad (7)$$

Where $PIDT_j$ is packet inter-departure time in GOP j, NP'_j is the number of predicted packets in GOP j, FR is the frame rate of video (fps), NF_j is the number of frames in GOP j, PS is the packet size and OR_s is the maximum output rate of the source s. $\frac{PS \times NP'_j}{NF_j \times 1/_{FR}}$ shows the predicted sending rate of the source s. In the second case of Eq. (7) where the predicted output rate is bigger than the maximum output rate, some of the frames in GOP with lower priority are ignored and not sent by source node to adjust sending rate. B-packets have the lowest priority. Moreover, the priority of P-packets that are close to the end of GOP is lower among all other P-packets. Eq. (8) computes the number of ignored packets in GOP j.

$$IP_j = NP'_j - \frac{OR_s \times NF_j \times 1/_{FR}}{PS} \quad (8)$$

By using this approach, PIDT of all three packet types are the same and is equal to 0.0069 in the above example. Therefore, our proposed scheme provides an increase in the PIDT of I-packets and lower bursty traffic during sending I-packets. The pseudo code of UDDP is shown in Algorithm 1.

IV. Simulation Results

The performance of UDDP is analyzed in NS-2 simulator version 2.28 [10]. Evalvid version 2.7 [12] is used to enable video transmission simulation in NS-2. The NS-2 simulations are setup to use IEEE 802.15.4 for defining MAC and physical layer of each node that is the most common MAC and physical layer protocol for WMSN [13]. The sending rate of MAC layer is set to be 250 Kbps [13]. According to IEEE 802.15.4, the packet size is 100 Bytes that includes 72 bytes to carry

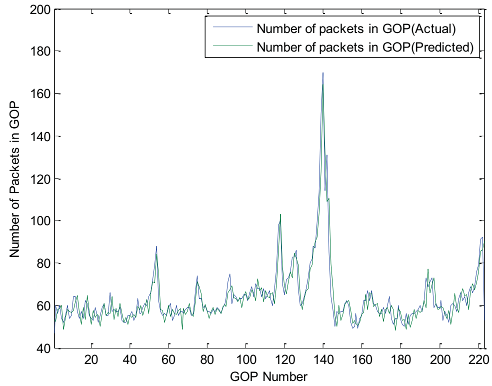


Fig. 3. Predicted number of packets in GOP

multimedia content and 28 bytes as header. The initial power of each sensor is set to be 100 joules and 0.3 watt is consumed for each sending and receiving operations.

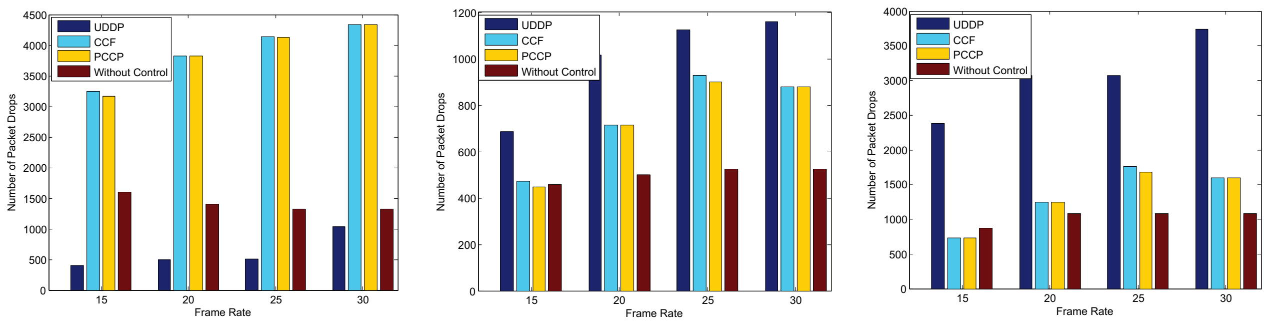
Simulations are done during 800 seconds over 30 WMSN sensors, and the maximum number of simultaneous video transmissions is set to be 4. All sensors are assumed to have 5 KB RAM memories and AODV [14] is employed as the routing protocol in our simulations.

We compare our proposed protocol with two other well known protocols, CCF [6] and PCCP [7], and a protocol that does not have any congestion control (called without control). The comparison is done in terms of packet loss, frame loss, video quality, energy consumption and throughput.

Two types of traffic are generated in WMSN: monitoring and event-driven traffic. Monitoring time and monitoring period are set to be 5 and 100 seconds respectively. Moreover, events are occurred in the network using NS-2 exponential random generator with μ=70 seconds. In each event, the sensors that are close to the event area, streams the video in 10 seconds. The video file used in simulation was Highway-QCIF with 2000 frames, and the simulation is done in four different frame rates, 15, 20, 25 and 30 fps.

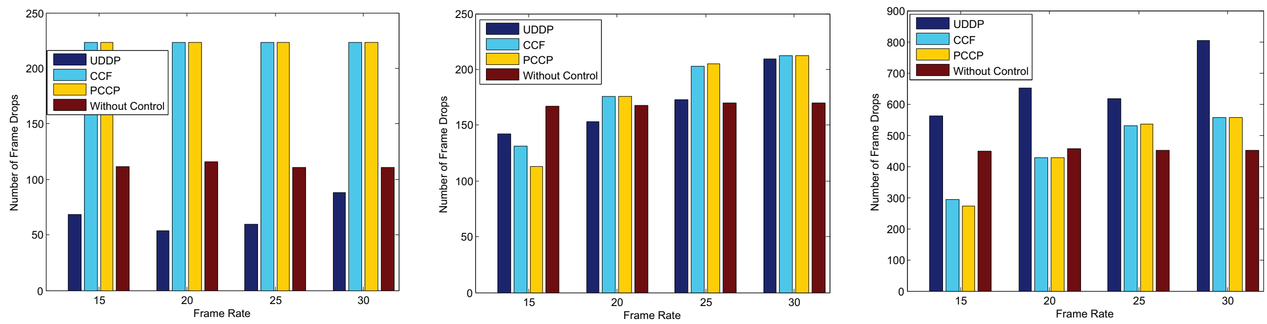
Fig. 3 shows the predicted number of packets in GOP compared to the actual number of packets in GOP when both a and b are 0.01. As we can see, our prediction is close to actual numbers. This demonstrates the validity of Eqs. (1), (2) and (3).

Fig. 4 shows the average number of packet loss in transmitted video vs. frame rates. As we can see in Fig. 4(a), the number of lost I-packets in UDDP is the lowest among all other protocols. However, Fig. 4(b) and (c) show that UDDP has the largest number of P- and B-packet loss. This is because of adjusting sending rate of source node in our proposed protocol by ignoring B-packets or sometimes P-packets in bursty traffic. Lower I-packet loss results in lower I-frame loss. Fig. 5 shows the results of simulations that support this fact. Without deploying a congestion control protocol, packet loss may occur at each video frame. Moreover, loss of each packet of a frame leads to drop that frame (frame loss). UDDP avoids



(a) (b) (c)

Fig. 4. Average number of packet loss for a) I-packets, b) P-packets, c) B-packets



(a) (b) (c)

Fig. 5. Average number of frame drops for a) I-frames, b) P-frames, c) B-frames

high frame loss rate by saving packets with higher priority. Fig. 4(b) shows that UDDP has higher number of P-packet loss. However, Fig. 5(b) shows that P-frame loss of our protocol is the same as other protocols. The reason for that is when UDDP adjusts the sending rate, it ignores the all packets in the less important frame. This leads to only one frame loss. It shows the benefit of using cross layer information to minimize the frame loss in transport layer.

It's clear that lower I-frame loss leads to better video quality in receiver node. Fig. 6 shows a comparison result between the UDDP protocol, CCF protocol, PCCP protocol, and without control in terms of video quality (the most important performance metric in WMSN). Peak Signal to Noise Ratio (PSNR) is used to measure video quality. It can be seen that UDDP has better PSNR and improves abut 7 db in received video quality compared to the without control method.

Moreover, both CCF and PCCP protocols provide too low video quality. Their video quality are even lower than result of without congestion control method. It is because of higher I-packets loss (refer to Fig. 4(a)) and consequently higher I-frames loss (Fig. 5(a)) in these protocols. Fig. 6 shows that it is not a good idea to use Wireless Sensor Network (WSN) congestion control protocols in WMSN. Therefore, congestion control protocol in WMSN should be content-aware.

Network throughput can be defined as the average successful message delivery rate over the network. Throughput basically shows bandwidth consumption through the network. Specifically more throughput in WMSN shows higher power consumption in sensor nodes. However, throughput is less important than the received video quality in WMSN. As shown in Fig. 7, UDDP protocol has the highest throughput compared to other protocols except for frame rates 25 and 30 fps. In these rates, UDDP has lower throughput than without control method.

To provide real-time video transmission in WMSN, controlling delay is crucial. Fig. 8 shows that UDDP, except when the frame rate is 15 fps, has at least same or better delay than CCF and PCCP protocols. In UDDP, the average delay of the network decreases as the frame rate of the video increases. This comes from the fact that UDDP protocol makes lower PIDTs when the frame rate of the video increases (refer to Eq. (7)).

According to Fig. 9, B-packets are ignored more than P-packets and as the frame rate of the network increases, more packets are ignored. For frame rate 30 fps, 32.2% of all packets are ignored by source nodes. Ignoring packets leads to lower I-packet loss and lower energy consumption in the source nodes. Lower energy consumption is one of the main goals

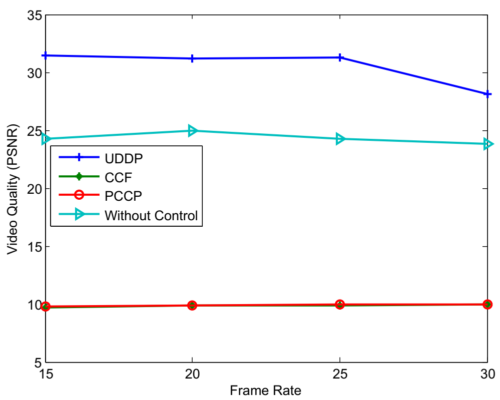


Fig. 6. Average received video quality

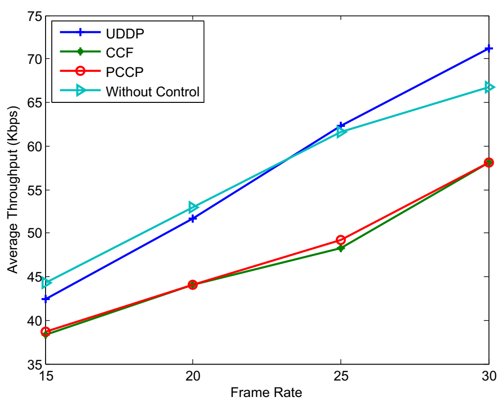


Fig. 7. Average Throughput (Kbps)

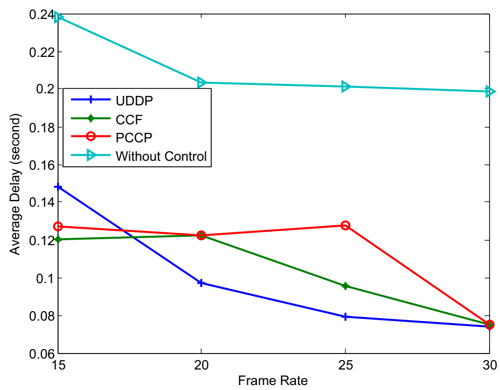


Fig. 8. Average Delay (sec)

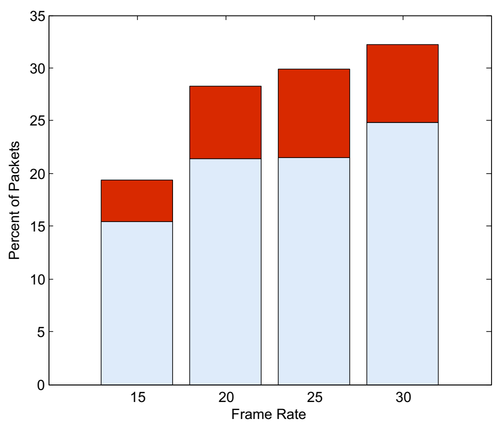


Fig. 9. Average ignored packets

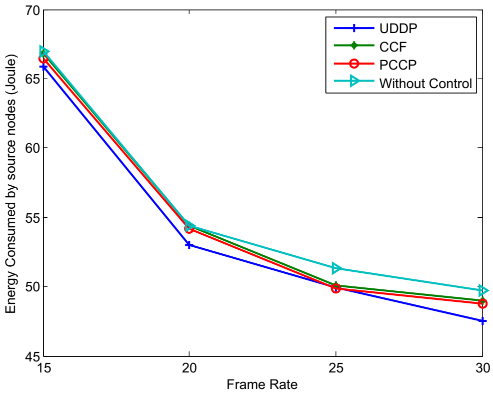


Fig. 10. Average energy consumption of source nodes

of developing a transport layer protocol for WMSN. Fig. 10 shows average energy consumption of source nodes at the end of simulation. In average, UDDP has about 5% improvement in terms of energy conservation in the source nodes compared to other protocols. Therefore, higher video quality in UDDP can be achieved, along with lower energy consumption.

V. Conclusion And Future Work

In this paper, a new transport layer protocol was proposed to minimize the packet loss in WMSN by considering traffic characteristics, the inter-arrival pattern of packets and packet priority. It is a cross-layer protocol that uses information of MAC (MTU of packets) and application layers (the number of packets in each GOP) to distribute packet arrivals in a GOP. The proposed protocol considers the priority of I-, P- and B- packets to send video frame in network. We have demonstrated the effectiveness of this protocol by performing simulations. The experimental results showed that the proposed protocol in despite of good improvement in terms of video quality (7 db in average), does not increase energy consumption. Moreover, our protocol has lower delay and approximately the same throughput compared to other protocols. WMSN sensors also have limited power supplies, therefore designing an efficient congestion control protocol seems to be necessary. In this work, we have introduced a new specific transport layer protocol for WMSN. The simulation results showed that WSN congestion control protocols are not suitable in WMSN and any congestion control protocol in WMSN should be content-aware to achieve a high video quality in the receiver side. As future work, we plan to work on an active queue management protocol to control node level congestions in WMSN. We also plan to adjust the packet size of WMSN to have an efficient video packet transmission in WMSN.

References

[1] S. Soro and W. Heinzelman. *"A Survey of Visual Sensor Networks"*. Advances in Multimedia, 2009, May 2009, pp. 1-21.

[2] Y. Sankarasubramaniam, I. F. Akyildiz, and S. W. McLaughlin, *Energy efficiency based packet size optimization in wireless sensor networks,* in 2003 IEEE International Workshop on Sensor Network Protocols and Applications, 2003. First IEEE, 2003, pp. 1-8.

[3] Y. G. Iyer, S. Gandham, and S. Venkatesan, *STCP: a generic transport layer protocol for wireless sensor networks,* in 14th International Conference on Computer Communications and Networks, 2005. ICACCN 2005. Proceedings, 2005, pp. 449- 454.

[4] B. Hull, K. Jamieson, and H. Balakrishnan, *Mitigating congestion in wireless sensor networks,* in 2nd international conference on Embedded networked sensor systems, Baltimore, MD, USA, 2004, pp. 134147.

[5] C. Y. Wan, S. B. Eisenman, and A. T. Campbell, *CODA: congestion detection and avoidance in sensor networks,* in 1st international conference on Embedded networked sensor systems, Los Angeles, California, USA, 2003, pp. 266279.

[6] C. T. Ee and R. Bajcsy, *Congestion control and fairness for many-to-one routing in sensor networks,* in 2nd international conference on Embedded networked sensor systems, Baltimore, MD, USA, 2004, pp. 148161.

[7] Chongang Wang, K. Sohraby, V. Lawrence, Bo Li, and Yueming Hu, *Priority-based congestion control in wireless sensor networks,* in IEEE International Conference on Sensor Networks, Ubiquitous, and Trustworthy Computing, 2006, vol. 1, 2006.

[8] Q.-min LIN, R.-chuan WANG, J. GUO, and L.-juan SUN, *Novel congestion control approach in wireless multimedia sensor networks,* The Journal of China Universities of Posts and Telecommunications, vol. 18, no. 2, Apr. 2011, pp. 1-8.

[9] I. F. Akyildiz, T. Melodia, and K. R. Chowdury, *Wireless multimedia sensor networks: A survey,* IEEE Wireless Communications, vol. 14, no. 6, Dec. 2007, pp. 32-39.

[10] *The Network Simulator - ns-2.* [Online]. Available: http://www.isi.edu/nsnam/ns/. [Accessed: 29-Dec-2010].

[11] H. Wang, L. Kondi, A. Luthra, and S. Ci, *"4G Wireless Video Communications"*. John Wiley and Sons, 2009.

[12] J. Klaue, B. Rathke, and A. Wolisz, *EvalVid A Framework for Video Transmission and Quality Evaluation,* in Computer Performance Evaluation. Modelling Techniques and Tools, vol. 2794, P. Kemper and W. H. Sanders, Eds. Berlin, Heidelberg: Springer Berlin Heidelberg, 2003, pp. 255-272.

[13] Almalkawi, I.T.; Guerrero Zapata, M.; Al-Karaki, J.N.; Morillo-Pozo, J. Wireless Multimedia Sensor Networks: Current Trends and Future Directions. Sensors 2010, pp. 6662-6717.

[14] C. E. Perkins and E. M. Royer, Ad-hoc on-demand distance vector routing, in Second IEEE Workshop on Mobile Computing Systems and Applications, 1999. Proceedings. WMCSA 99, 1999, pp. 90-100.