A Reliable Transport Protocol for Wireless Sensor Networks

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***Abstract*—Designing a reliable transport protocol is a new challenging area in Wireless Sensor Networks (WSNs). Traditional transport layer protocols (such as TCP and UDP) can't directly be applied to the WSNs. Generally transport layer is responsible for congestion control and reliable packet delivery. Congestion is an essential problem in WSNs. It not only wastes the scarce energy due to a large number of retransmissions and packet drops, but also hampers the event detection reliability. Thus, to meet the Quality of Service (QoS) requirements of network applications, a reliable and fair transport protocol is mandatory. In this paper we present a reliable transport protocol for wireless sensor networks which not only controls the congestion in the network, but also provides reliability in packet delivery. The proposed model uses a rate control mechanism to adjust the transmission rate of each sensor node based on the congestion degree in the network. We use the time to recover packet loss as a congestion indicator. To use a node's energy efficiently, we use a hop-by-hop NACK based reliability guaranty model. Simulation results, confirm the superiority of the proposed model.**

Keywords: Wireless Sensor Networks; Congestion Control; Reliability; Transport Protocols; Rate based control

I.INTRODUCTION

Wireless Sensor Networks (WSNs) are a set of communication networks consisting of different independent sensors that cooperatively monitor some physical or environmental conditions, such as temperature, sound, vibration, pressure, motion or pollutants, at different locations. WSNs have many applications in different areas of technology. They are used in different applications including: commercial and industrial applications, environment applications, healthcare applications, home automation, traffic control and monitoring, object tracking and fire detection. Each node in a WSN is typically equipped with one or more sensors, a wireless communications device, a processor, and an energy source, usually a battery [1].

To achieve 100% packet delivery ratio in WSNs, having a reliable transport mechanism is important. In traditional communication networks, the transport layer is responsible for bridging the application and network layers using multiplexing and demultiplexing. It is also charged with providing end-to-end reliable data delivery and with performing congestion control by regulating the amount of traffic injected into the network. In addition to the challenges for reliable data transport in WSN, there exist additional challenges due to the unique requirements of the multimedia transport such as bounded delay and delay variation, minimum bandwidth demand, smooth traffic variation for multimedia streaming, and error control according to the specific requirements of the multimedia application. As argued in [2], the traditional TCP/UDP transport protocols cannot be directly implemented for WSN. Therefore, it is important to develop a reliable transport protocol for WSNs to ensure that the often differing QoS requirements of various applications can be met.

Congestion control is a critical issue in transport protocols. Congestion is an essential problem in wireless sensor networks. It not only wastes the scarce energy due to a large number of retransmissions and packet drops, but also hampers the event detection reliability. Congestion in WSNs has a direct impact on energy efficiency and application QoS. Not only can packet loss degrade reliability and application QoS, but it can also waste the limited node energy and degrade link utilization. In each sensor node, when the packet-arrival rate exceeds the packet-service rate, buffer overflow may occur. This is more likely to occur at sensor nodes close to the sink, as they usually carry more combined upstream traffic. Congestion control mechanisms typically consist of three important components: congestion detection, congestion notification, and rate adjustment. The past few years, different congestion control protocols have been proposed for WSNs. STCP[3], Fusion[4], CODA[5], PCCP[6],CCF[7] are the most well known congestion control protocols in WSNs. Recently we proposed QCCP-PS[8], a Queue based Congestion Control Protocol with Priority Support for wireless multimedia sensor networks which has better performance than the PCCP and CCF protocols.

In traditional TCP protocol, congestion is detected at the end nodes based on a timeout or redundant acknowledgments. In TCP protocol, both congestion and reliability are coupled with the receipt of an ACK from the receiver. TCP assumes the non-receipt of an ACK as a congestion problem and it slows down its transmission rate along with retransmitting the packet for reliability. TCP protocol has good performance in wired network where the channels are mostly reliable. However in wireless networks, this is a huge problem as error rates are usually high in wireless media. ACK/NACK based protocol can also be used in WSNs. This approach can easily

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detect errors, but a huge number of status report transmissions is required. On the other hand, though NACK based protocol spends less network resources, error detection is much harder than ACK/NACK based protocols and requires a high computational complexity.

In WSNs, the nodes use a radio channel to transmit their data toward a base station (sink node). Because of this, congestion is a very realistic concern in sensor networks. As the power consumption is an important issue in these networks, the cost of retransmission of a lost frame is very high. This makes the congestion control problem in WSN to be a more urgent concern.

Fiven that the links in WSNs are not reliable, the end-to-end reliability model is not suitable in this networks. To save energy at a node and to minimize overall the energy consumption, most transfer protocols in WSNs use a hop-by-hop reliability model. In the hop-by-hop reliability model, intermediate nodes are supposed to participate in data transport by caching and retransmitting data packets, generating or changing the contents of control packets. To minimize energy consumption, retransmission should be reduced. Retransmission can be reduced by using hop-by-hop error recovery schemes. Different congestion control and reliability guaranty models have been proposed for WSNs. Popular examples include PSFQ[9], RMST[10], DTC[11], DTSN[12], ESRT[13], and STCP[14].

Rate-Controlled Reliable Transport (RCRT) [15] is a new transport protocol for wireless sensor networks. RCRT consists of four major components including: congestion detection, rate adaptation, rate allocation and end-to-end retransmission. RCRT uses the length of retransmission list as the congestion indicator. When there are too many packets in retransmission list, it means that the congestion density is high. In this case the RCRT tries to adapt the transmission rate of each sensor node, using an AIMD rate control mechanism. RCRT implements a NACK-based end-to-end loss recovery scheme. The sink detects packet losses and repairs them by requesting end-to-end retransmissions from source nodes.

In this paper we propose a modified version of the RCRT protocol which is designed for reliable transport protocol in WSNs. Rather than the retransmission length, the proposed model uses the time to recover packet loss as a congestion indicator. Unlike the RCRT protocol, to use node's energy efficiently, the proposed model uses a hop-by-hop NACK based reliability guaranty model. Upon detecting a packet loss, the node sends a NACK message to its downstream node and asks it to retransmit the lost packet. Downstream node searches in its local cache memory for the lost packet and retransmits it to its upstream node.

The reminder of this paper is organized as follow. In Section 2, we explain the details of proposed model. Simulation results which confirm the superiority of the proposed model are given in Section 3. Finally Section 4 concludes the paper.

II. PROPOSED MODEL

As described in section 1, the hop-by-hop reliability guaranty model is more suitable than end-to-end model in wireless sensor networks. For lower energy consumption, the proposed model uses the hop-by-hop reliability guaranty model. Each intermediate node has two types of buffer, namely the receive buffer and retransmission buffer. The packets which are received in order are placed in the receive buffer. A copy of each received packet is also saved in a cache memory. When a node receives the ACK of its already sent packet, it removes the packet from its local cache. Packets which are received out of order are forwarded to the retransmission buffer. In the proposed model, each node on the forward path from source to sink caches the packets. When a node detects a lost packet, an NACK message is sent to the next hop on the reverse path toward the source. If the requested lost packet is found in the local cache, a copy of the lost packet is retransmitted. If not, the NACK message is forwarded to the next hop toward the source. Caching of packets along forward path is used to limit power waste due to end-to-end retransmission. To detect any gap in received packets, each packet must contain a sequence number. Each node uses a timer based loss detection mechanism. When the requested packet doesn't arrive in a predefined time interval, a NACK message is sent to the next hop in the reverse path. The value of timer could be dynamically tuned based on the degree of congestion in the network. Every out of order packet is located in the retransmission buffer. Each node maintains a list of missing packets per flow. When losses are detected, the sequence numbers of the lost packets are inserted into a list. Entries in this list of missing packets are sent as NACKs by the node to the downstream node. Upon receiving a NACK, the node retransmits the requested packets to repair the losses.

The proposed model measures the time to recover packet loss to calculate the congestion density. When the congestion density is low, in the case of packet loss, the lost packet would be recovered very soon, while when congestion is high, many packets (including retransmitted packets) would be lost which increases the packet lost recovery time. In the proposed model, each node measures the average time to recover lost packet. Suppose at each node i, T_i represents the time to recover lost packet which is calculated as the time between sending the NACK message and receiving the retransmitted packet. Fig. 1 shows an example for the calculation of T_i.

Let D_i denotes the average delay (queuing and transmission delay) between each node i and its downstream node $i-1$. Each node i computes its congestion degree CD_i as follows:

$$CD_i = (1-\alpha)CD_i + \alpha \frac{T_i}{D_i} \tag{1}$$

where α is a positive number less than 1. At each node i, the value of congestion degree CD_i is forwarded to the sink node using a specific field in the packet's header. Suppose there are N different sensor nodes (except the sink node) in the network. Each sensor node i computes and forwards its congestion degree to the sink node. The value of CD_i for the end nodes with no childeren is set to 0. The sink node obtains the effective congestion degree CD_{eff}:

$$CD_{eff} = \max\{CD_1, CD_2, ..., CD_N\} \quad (2)$$

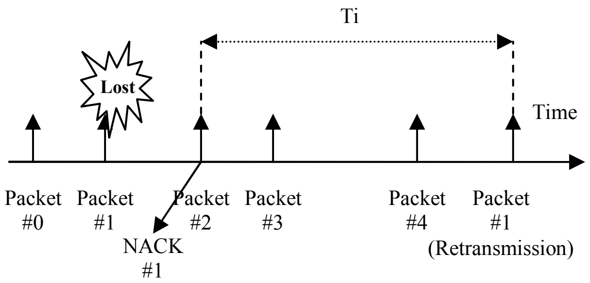

Fig1. An example of T_i calculation

Each sensor node i uses the exponential weighted moving average mechanism to calculate the value of D_i and T_i. Similar to the RCRT protocol, the proposed model uses a simple threshold mechanism. When the value of effective congestion degree, CD_{eff}, exceeds a maximum threshold, $TH_{\max}$, the network is congested. In this case, the source rates of all network nodes should be decreased. If CD_{eff} is less than a predefined minimum threshold, $TH_{\min}$, there is no congestion in the network. In this case, to use the network capacity efficiency the source rate of each node is increased. When CD_{eff} is between these two thresholds, the source rates aren't changed. Note that any change in the source rates is performed when the time elapsed from the previous change is more than 2 maximum Round Trip Time , $RTT_{\max}$. Let $r_{Total}(t)$ denotes the sum of the current assigned rates to all traffic sources. Similar to the other rate adaptation techniques, the proposed model uses an AIMD on $r_{Total}(t)$ as follows:

- When $CD_{eff} \le TH_{\min}$ and time elapsed from the previous adaptation is more than $2RTT_{\max}$ then:

$$r_{Total}(t) = r_{Total}(t) + \lambda_I \quad (3)$$

- When $CD_{eff} \ge TH_{\max}$ and time elapsed from the previous adaptation is more than $2RTT_{\max}$ then:

$$r_{Total}(t) = r_{Total}(t).\lambda_D(t) \quad (4)$$

where λ_I is a constant value and $\lambda_D(t)$ is a time-dependent multiplicative decrease factor. Note that the rate of each traffic source i could not be more than a predefined bound $r^i{}_{\max}$.

To estimate $\lambda_D(t)$, the packet loss of each traffic source i is measured using the Average Loss Interval (ALI) method given in [16]. For each traffic source i , suppose that $S_k (k = 1,..,8)$ be the number of packets in the k-th most recent loss interval. Let the most recent interval S_0 be defined as the interval containing the packets that have arrived since the last loss. Two variables $\hat{s}_0$ and $\hat{s}_1$ are used::

$$\hat{s}_0 = \frac{S_0 + S_1 + S_2 + S_3 + 0.8S_4 + 0.6S_5 + 0.4S_6 + 0.2S_7}{6}$$
$$\hat{s}_1 = \frac{S_1 + S_2 + S_3 + S_4 + 0.8S_5 + 0.6S_6 + 0.4S_7 + 0.2S_8}{6} \quad (5)$$

The packet loss probability of traffic source i ($P^i{}_{loss}$) is obtained following [16]:

$$P^i{}_{loss} = \frac{1}{\max(\hat{s}_0, \hat{s}_1)} \quad (6)$$

Figure 2 shows an example of ALI method. In this example the value of $\hat{s}_0$ and $\hat{s}_1$ are calculated as: $\hat{s}_0 = \frac{20.6}{6} \approx 3.4$ and $\hat{s}_1 = \frac{20}{6} \approx 3.3$, respectively. In this example, the packet loss probability is equal to $\frac{1}{3.4} = 0.294$.

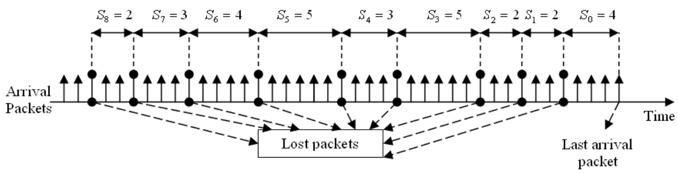

Fig. 3. An example of average loss interval method used to compute packet loss probability

Using the ALI method described above, the sink node calculates $P^i{}_{loss}$, the packet loss probability of each traffic source i (). The time-dependent multiplicative decrease factor $\lambda_D(t)$ is computed as follows:

$$\lambda_D(t) = \min_{i=1}^{N} \frac{1 - P^i{}_{loss}}{1 + P^i{}_{loss}} \quad (7)$$

When packet loss probability is zero, there is no congestion in the network. In this case $\lambda_D(t)$ is equal to 1 and the source rates aren't changed. When there is congestion in the network, the packet loss probability is increased. By increasing the packet loss probability, the value of $\lambda_D(t)$ is decreased toward zero which causes the decrease in the source rates. By decreasing the source rates, the congestion density is also decreased.

III. SIMULATION RESULTS

In this section, we use computer simulations to evaluate the performance of the proposed model at different channel error rates. For this purpose, we simulated a wireless sensor network shown in Fig. 3. All sensor nodes have a random service time. The simulation parameters are given in Table 1.

Table 1. Simulation parameters

Parameter	Value
Buffer size	100 Pkts
Mean service time	0.01 s
$TH_{\min}$	1
$TH_{\max}$	4
$r^i_{\max}$	9.5 pkts/s
λ_I	0.05 pkts/s
Simulation time	1000 s

In Fig. 4, the total assigned source rate is plotted versus simulation time, for different channel Packet Error Rates (PERs), for both RCRT and proposed protocol,. The figures shows tha when the channel error rate increases, the performance of RCRT degrades. Unlike the RCRT protocol, as the proposed protocol uses a hop-by-hop congestion control and reliability model, it can tune the source rate of each sensor node so that maximum channel utilization is achieved. The results shown in figure 4 confirm that the rate fluctuation in the proposed protocol is lower than RCRT, making it more appropriate for streaming applications which require constant video quality.

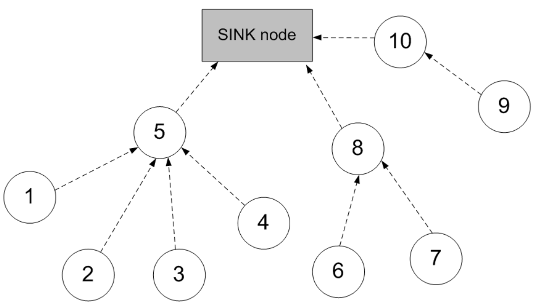

Fig3. Network topology used in the simulation

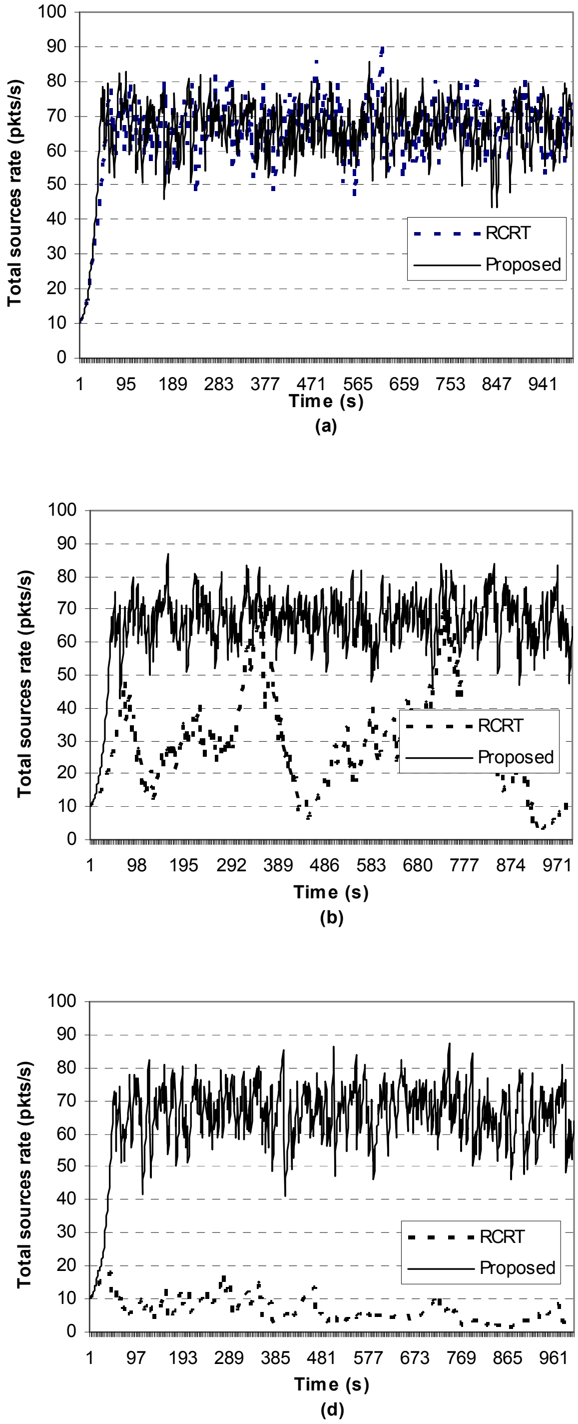

Fig 4. Total source rate versus simulation time at different value of PER (a) PER=0 (b) PER=3% (c) PER=5%

Fig. 5 shows the variation of the total goodput of traffic sources with simulation time, at different packet error rates, and for both RCRT and the proposed protocol. Total goodput is the application level throughput which is defined as the number of useful bits per unit of time forwarded by the network from all traffic sources to the sink node, excluding retransmitted packets. It can be seen that regardless of the value of PER, the proposed protocol can achieve more than 65% of total throughput of the network. Unlike the proposed protocol, the performance of RCRT is very dependent on the PER. As the figure shows, when PER=5%, the RCRT protocol can use less than 10% of the total network capacity.

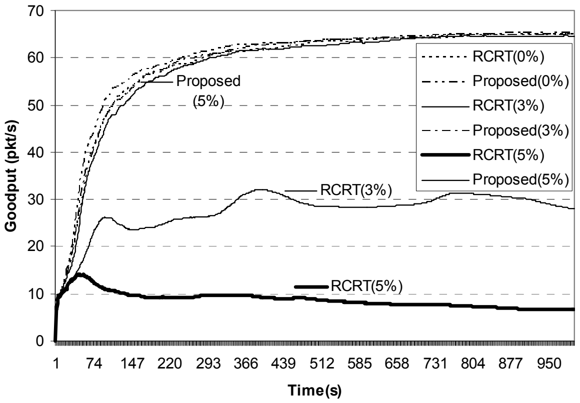

Fig 5. Total goodput versus simulation time at different values of PER

Fig. 6 shows the end-to-end delay plotted against packet sequence number. For the proposed protocol the end-to-end delay is always less than that of the RCRT protocol. In this case, the average end-to-end delay of RCRT and proposed protocol are 0.142s and 0.078s respectively.

The total packet loss probability of both RCRT and proposed protocol which has been calculated using ALI method is given in Fig. 6.

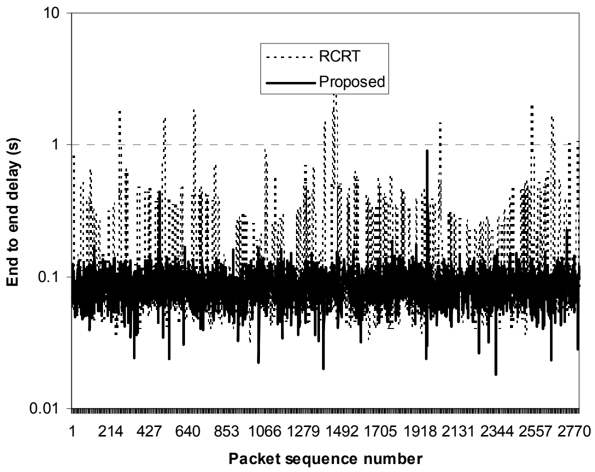

Fig 6. End-to-end delay

The average packet loss probability of RCRT and the proposed are 0.068 and 0.033 respectively. From the figure it is clear that the proposed protocol has better loss performance than the RCRT protocol.

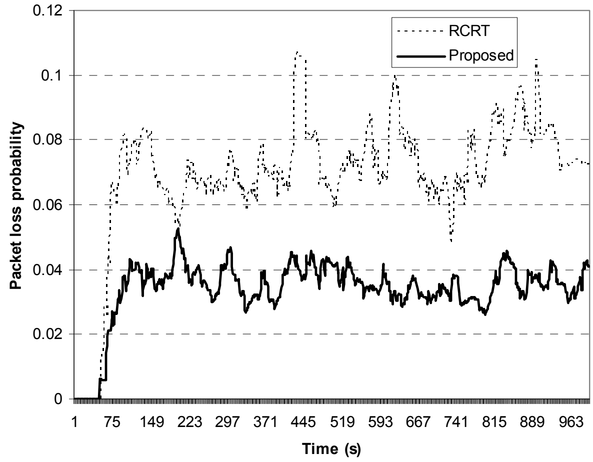

Fig 7. Packet loss probability versus simulation time

We also considered the variation of packet delivery ratio with the PER for both RCRT and the proposed protocol.

RCRT uses the end-to-end NACK mechanism which means that packet loss is detected only in the sink node. When the PER increases, the probability of packet loss is also increased. Since the proposed protocol uses a-hop-by-hop reliability model, any packet loss is detected and retransmitted in the intermediate node. As the figure shows, when the PER increases, the RCRT fails to provide 100% packet delivery ratio. Fig. 8 shows the channel utilization at different values of $r^{i}_{\max}$, the maximum source rate at PER=3%, the channel utilization is plotted. The results confirm that the utilization performance of the proposed protocol is better than that of RCRT protocol.

Overall, the results show that when the packet error rate in the network is low, both RCRT and proposed protocol have a high performance. However, with increasing channel error rates, the performance of RCRT degrades considerably while the proposed protocol still has acceptable performance.

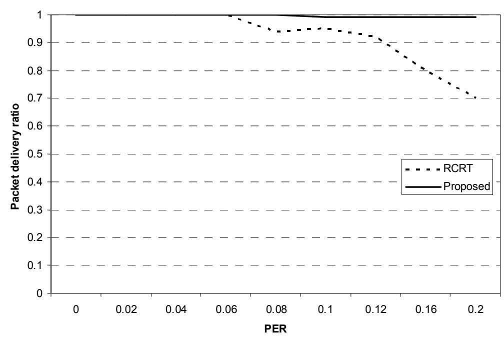

Fig 7. Packet delivery ratio versus PER

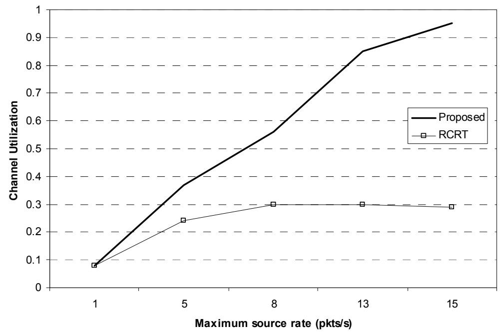

Fig 8. Channel utilization versus maximum source rate (PER=3%)

IV. CONCLUSION

The transport protocol enables end-to-end reliable message transmission. Its main functions are: orderly transmission, flow and congestion control, loss recovery, and possibly QoS guarantees such as timing and fairness. Due to limited wireless bandwidth in WSNs, congestion may occurr. Wireless channel introduces packet loss due to bit error rate, which not only affects reliability, but also wastes energy. In this paper we proposed a reliable transport protocol for WSNs. The proposed model uses the time to recover packet loss as the congestion indicator. We use hop-by-hop reliability model to decrease the number of NACK message. Simulation results validate the performance of the proposed scheme.

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