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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

چکیده : طراحی پروتکل مطمئن درلایه حمل، یکی از چالش های جدید درشبکه های حسگر بیسیم می باشد. پروتکل های سنتی لایه حمل (نظیر TCP,UDP) به طور مستقیم قابل اعمال به شبکه های حسگر بیسیم نمی باشند. عموما از پروتکل های لایه حمل برای کنترل ازدحام و تحویل مطمئن بسته به مقصد استفاده می شود. ازدحام یکی از مسائل کلیدی درشبکه های حسگر بیسیم است. این پدیده نه تنها به خاطر ارسال های مجدد زیاد و اتلاف بسته ها باعث اتلاف انرژی می شود بلکه باعث اختلال دراطمینان نیز می گردد. بنابراین برای تامین نیازمندی های کیفیت سرویس برنامه های کاربردی شبکه، پروتکل مطمئن و عادل درلایه حمل اجباری می باشد. دراین مقاله ، یک پروتکل مطمئن لایه حمل برای شبکه های حسگر بیسیم طراحی شده است که نه تنها ازدحام را کنترل می نماید بلکه تحویل مطمئن بسته ها به مقصد را تضمین می کند. مدل پیشنهادی از یک مکانیسم کنترل نرخ برای تعدیل نرخ ارسال هریک از نودهای حسگر براساس درجه ازدحام درشبکه می نماید. ازپارامتر زمان لازم برای بازیابی بسته های اتلاف شده به عنوان نشانه ازدحام استفاده می شود. برای استفاده بهینه از انرژی نود، از مدل قابلیت اطمینان مبتنی بر بسته های NACK به صورت پرش به پرش استفاده شده است. نتایج شبیه سازی کارایی و برتری روش پیشنهادی را تایید می نماید.

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 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.

Given 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 the overall energy consumption, most transport 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.

The end-to-end loss recovery approach is not very effective for WSNs. The main reason is that the control messages that are used for end-to-end loss detection would utilize a return path consisting of several hops, and this is not energy efficient. Furthermore, in the end-to-end loss recovery approach, control messages travel through multiple hops and could be lost with a high probability due to either link error or congestion. However, end-to-end retransmission consumes more energy than hop-by-hop retransmission. In hop-by-hop loss detection and notification, a pair of neighbouring nodes are responsible for loss detection, and can enable local retransmission that is more energy efficient, as compared to the end-to-end approach. The main drawback of the hop-by-hop approach is that, each node needs a high capacity memory to store all packets temporarily.

Rate-Controlled Reliable Transport (RCRT) [9] 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 Additive Increase Multiplicative Decrease (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 follows: Some related works in this area are described in section 2. In section 3, we explain the details of proposed model. Simulation results which confirm the superiority of the proposed model are given in section 4. Finally section 5 concludes the paper.

II. RELATED WORKS

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. QCCP_PS showed better performance than the PCCP and CCF protocols.

Different methods have been proposed to detect congestion in wireless sensor networks. Some protocols such as SCTP, Fusion and CODA use the queue length as a congestion indicator. CCF uses the packet service time to detect any possible congestion in the network. PCCP proposes the ratio of packet inter-arrival time and packet service time as a congestion indicator.

Different congestion control and reliability guaranty models have been proposed for WSNs. Popular examples include PSFQ [10], RMST [11], DTC [12], DTSN [13], ESRT [14], and STCP [3]. PSFQ consists of three different states includes: pump, fetch, and report. When the sink is in the pump state, it periodically broadcasts data packets to its neighbors. Upon detecting any gap in the received data packets, the sensor node goes to the fetch state. It also sends an NACK in reverse path to recover the missing fragment. In the report state, using a simple and scalable hop-by-hop reporting mechanism, the sink sends information on data delivery to the sensor nodes. ESRT protocol guarantees event reliability through end to-end source rate adjustment while RMST provides packet reliability through hop-by-hop loss recovery. In the ESRT protocol, when the achieved reliability exceeds a stated value, the source rates are decreased in a multiplicative manner. Otherwise, the source rates are additively increased.

Most existing transport protocols for wireless sensor networks address congestion control or reliable transport separately. When the congestion control and loss recovery algorithms are separated from each other, applications that need reliability can invoke only a loss recovery algorithm, or invoke a congestion control algorithm. The joint use of congestion control and loss recovery may provide the full functionality required by the transport protocols for wireless sensor networks. Thus, an important direction in WSNs is the design of such transport protocols, which not only provides congestion control but also supports mechanisms to provide reliability guaranty for different applications. To our knowledge, Rate-Controlled Reliable Transport (RCRT) [9] is the only published work that has attempted a joint consideration of congestion control and reliability guarantee in wireless sensor networks.

RCRT is a new transport protocol for wireless sensor networks. RCRT consists of four major components namely: congestion detection, rate adaptation, rate allocation and end-to-end retransmission. In the RCRT protocol a network is uncongested as long as end-to-end losses are repaired quickly which permits a few end-to-end losses caused by transient congestion, or by poor wireless links. RCRT uses the length of retransmission list as the congestion indicator. When there are too many packets in the 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. While RCRT uses AIMD, it adapts the total aggregate rate of all the flows as observed by the sink, rather than the rate of a single flow. RCRT places its congestion control functionality at the sink, whose perspective into the network enables better aggregate control of traffic, and affords flexibility in rate allocation. The rate allocation component of RCRT essentially assigns rates to each flow in keeping with a rate allocation policy. As RCRT decouples rate adaptation from rate allocation, it is possible to obtain this flexibility. RCRT implements an 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.

Although RCRT has taken the first step towards joint congestion control and reliability, the current RCRT protocol still has some major problems. First, RCRT uses an end-to-end retransmissions strategy.

All lost packets are detected in the sink node which implies a potentially long delay between packet loss occurrence and loss detection. As discussed earlier, hop-by-hop retransmission strategy provides a much better performance when compared with the end-to-end model. It can be shown that the RCRT protocol does not have good performance in high noisy channels. When the channel error rate is high, most packets are lost in the intermediate links. So the sink node will always detect out-of-order packets and can't repair losses quickly. Simulation results show that the performance of RCRT degrades quickly when the channel error rate is increased. Further, when there is a high channel error rate in the network, the number of lost packets increases, and the retransmission list keeps growing. When the sink node detects too many lost packets, it decreases the transmission rate of all traffic flows which causes a significant decrease in the network throughput. RCRT protocols always provides 100% packet delivery ratio for all transmitted packets. We believe that providing 100% reliability is not always necessary for some applications of sensor networks especially in multimedia applications. For example in the MPEG-4 standard a video stream is encoded to 3 different frame types I, P and B, where the P and B frames depend on the I frame. Loosing an I frame causes a noticeable worsening of the video quality of all the frames in the group. However, losing a B frame may cause significantly less degradation when compared with losing an I frame.

In this paper, we present a new protocol for congestion control and reliability guaranty for wireless multimedia sensor networks. The proposed protocol can be used for partial order services in WMSNs. For some applications such as multimedia services, there exists a genuine ability to tolerate packet losses. Losing one frame per second in a 30 frame per second video or losing a segment of its accompanying audio channel is usually not a problem. The proposed protocol provides partial reliability for this type of application.

III. PROPOSED PROTOCOL

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. Figure 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 $i-1$. The average delay can be estimated by measuring the time elapsed between sending a packet and receiving its ACK. Each node i computes its congestion degree CD_i as follows:

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

where α is a positive number less than 1 and CD_i^{old} denotes the previous value of CD_i. 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 children is set to 0. The sink node obtains the effective congestion degree CD_{eff} as follows:

$$CD_{eff} = \max\{CD_1, CD_2, \ldots, CD_N\} \tag{2}$$

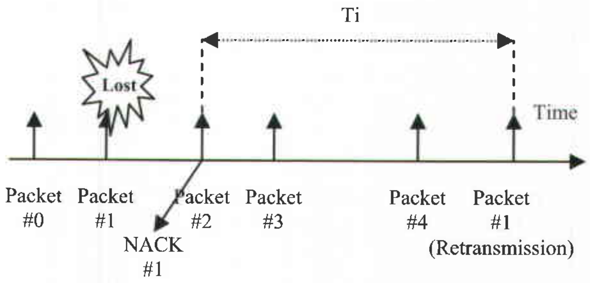


Figure 1: An example of T_i calculation

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:

$$r_{Total}(t) = r_{Total}(t-1) + \lambda_I \tag{3}$$

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

$$r_{Total}(t) = r_{Total}(t-1).\lambda_D(t) \tag{4}$$

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 [15]. 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:

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

The packet loss probability of traffic source i ($P^i{}_{loss}$) is obtained as follows [15]:

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

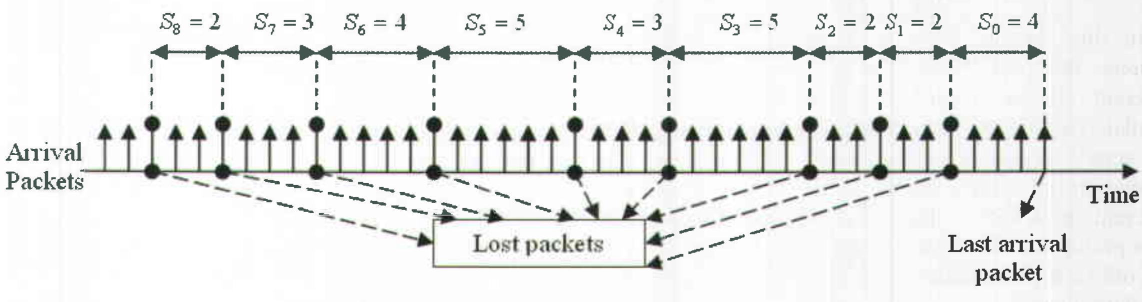


Figure 2: An example of average loss interval method used to compute packet loss probability

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 $\frac{1}{3.4} = 0.294$.

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:

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

Note that when packet loss is zero, there is not any lost packet in the network. Based on equation (7), in this case the value of $\lambda_D(t)$ is equal to 1 and there is not any change in the total rate. On the other hand, when the value of packet loss is more than 0, $\lambda_D(t)$ is less than 1 and so base on equation (4), the total rate is decreased. By decreasing the total rate, the probability of packet loss and the number of lost packet will also decreased. As we should consider the worst case, use the min operator to calculate the minimum value of $\lambda_D(t)$.

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 Figure 3. All sensor nodes have a random service time. The simulation parameters are given in Table 1. Note that different simulations with different value of parameters were performed. The value of parameters given in Table 1 is only a typical value which is used in the following simulation trials.

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 Table 2, for different value of channel Packet Error Rates (PERs) and for both RCRT and the proposed protocol the average total assigned source rate is given. Note than when PER=0, the RCRT protocol has a little bit better performance but the results confirm that 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 simulation results 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.

Table 2: The average total source rate of RCRT and the proposed algorithm at different value of PER

	RCRT (pkt/s)	**Proposed (pkt/s)**
PER=0	65.20841	65.14032
PER=1%	62.83134	66.04117
PER=3%	28.05573	65.24598
PER=5%	6.654369	64.58724

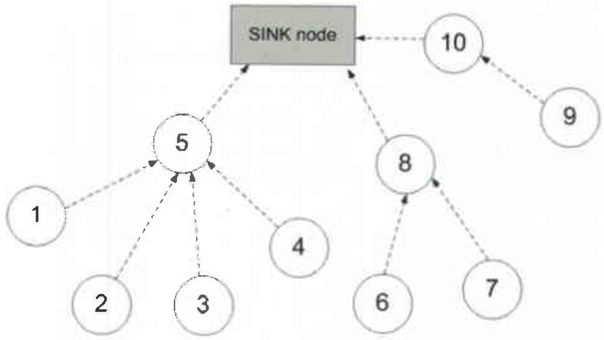


Figure 3: Network topology used in the simulation

Figure 4 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.

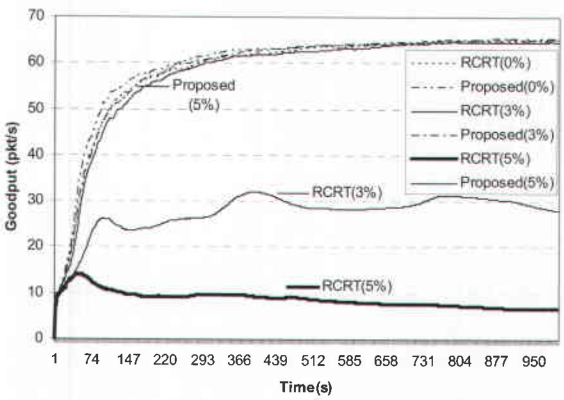
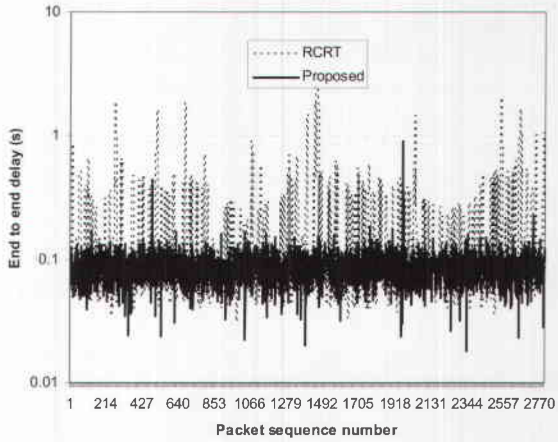


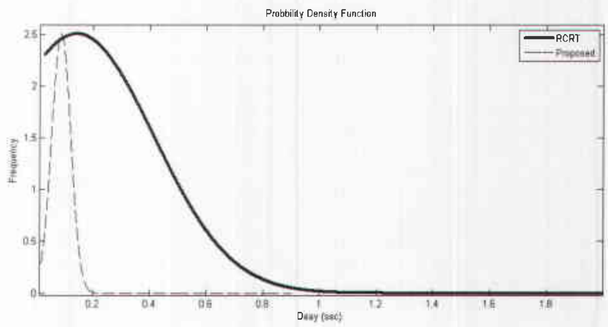
Figure 4: Total goodput versus simulation time at different values of PER

Figure 5 (a) 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 the proposed protocol are 0.142s and 0.078s,respectively. In Figure 5(b) for both protocols the Probability Density Function (PDF) is plotted. It can be observed that the proposed protocol has less mean delay and delay variation.

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



(a)



(b)

Figure 5: (a) End-to-end delay (b) Probability Density Function

The average packet loss probability of RCRT and the proposed protocol 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.

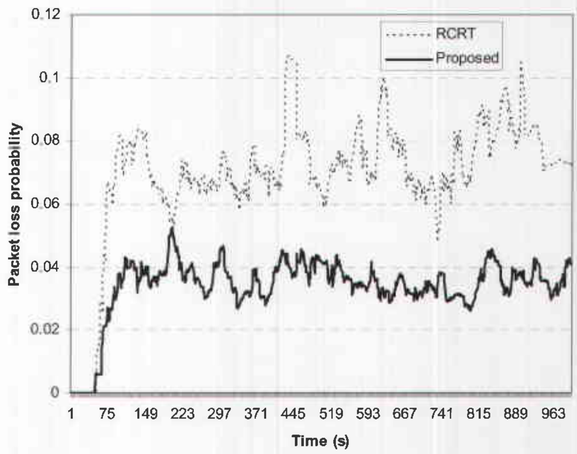


Figure 6: Packet loss probability versus simulation time

We also considered the variation of packet delivery ratio with the PER for both RCRT and the proposed protocol. The results is shown in Figure7.

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.

Figure 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 the 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.

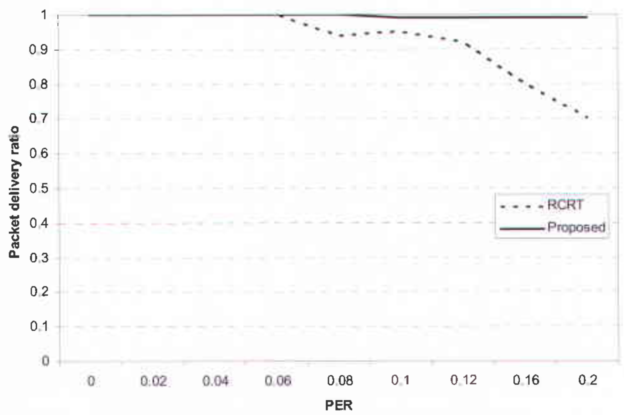


Figure 7: Packet delivery ratio versus PER

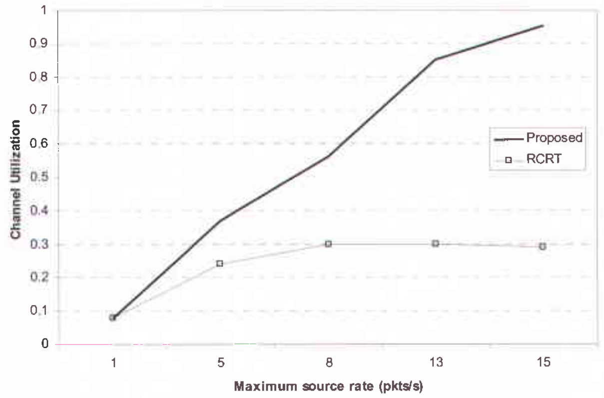


Figure8: Channel utilization versus maximum source rate (PER=3%)

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 occur. 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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