Reliable Bursty Convergecast in Multi-hop Wireless Sensor Networks

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***Abstract*— We address the challenges of bursty convergecast in multi-hop wireless sensor networks, where a large burst of packets from different locations needs to be transported reliably and in real-time to a base station. Via experiments on a 49 MICA2 mote sensor network using a realistic traffic trace, we determine the primary issues in bursty convergecast, and accordingly design a protocol, RBC (for Reliable Bursty Convergecast), to address these issues: To improve channel utilization and to reduce ack-loss, we design a window-less block acknowledgment scheme that guarantees continuous packet forwarding and replicates the acknowledgment for a packet; to alleviate retransmission-incurred channel contention, we introduce differentiated contention control. Moreover, we design mechanisms to handle varying ack-delay, to reduce delay in timer-based retransmissions, and to avoid queue overflow. We evaluate RBC, again via experiments, and show that compared to a commonly used implicit-ack scheme, RBC doubles packet delivery ratio and reduces end-to-end delay by an order of magnitude, as a result of which RBC achieves a close-to-optimal goodput.**

***Index Terms*— System design, experimentation with real networks and testbeds, sensor network, bursty convergecast, reliability, real-time, error control, contention control, flow control**

I. Introduction

A typical application of wireless sensor networks is to monitor an environment (be it an agricultural field or a classified area) for events that are of interest to the users. Usually, the events are rare. Yet when an event occurs, a large burst of packets are often generated and need to be transported reliably and in real-time to a base station, the interface between the sensor network and the external networks (such as the Internet). One exemplary event-driven application is demonstrated in the DARPA NEST field experiment "A Line in the Sand" (simply called *Lites* hereafter) [1]. In Lites, a typical event generates up to 100 packets within a few seconds and the packets need to be transported from different network locations to a base station, over multi-hop routes.

The high-volume bursty traffic in event-driven applications pose special challenges for reliable and real-time packet delivery. The large number of packets generated within a short period lead to high degree of channel contention and thus high probability of packet collision. The situation is further exacerbated by the fact that packets are transported over multi-hop routes: first, the total number of packets competing for channel access is increased by a factor of the average hop-count of network routes; second, the probability of packet collision increases in multi-hop networks due to problems such as hidden-terminals. Consequently, packets are lost with high probability in bursty convergecast. For example, with the default radio stack of TinyOS [2], around 50% of packets are lost for most events in Lites.

For reliability and real-time in packet delivery, hop-by-hop packet recovery is better than end-to-end recovery [3], [4]. Nevertheless, we find that existing hop-by-hop control mechanisms do not work well in bursty convergecast. By experimenting with a testbed of 49 MICA2 motes [2] and with traffic traces of Lites, we observe that the commonly used link-layer error control mechanisms do not significantly improve and can even degenerate packet delivery reliability. For example, when packets are retransmitted up to twice at each hop, the overall packet delivery ratio increases by only 6.15%; and when the number of retranmissions increases, the packet delivery ratio actually decreases, by 11.33%.

One problem with existing hop-by-hop control mechanisms is that they do not schedule packet retransmissions appropriately; as a result, retransmitted packets further increase the channel contention and cause more packet loss. Moreover, due to in-order packet delivery and conservative retransmission timers, packet delivery can be significantly delayed in existing hop-by-hop mechanisms, which leads to packet backlogging, queue overflow, and reduction in network throughput. (We examine the details in

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Section III-A.)

On the other hand, the new network and application models of bursty convergecast in sensor networks offer unique opportunities for reliable and real-time transport control:

- First, the broadcast nature of wireless channels enables a node to determine, by snooping the channel, whether its packets are received and forwarded by its neighbors.
- Second, high-precision time synchronization and the fact that data packets are timestamped relieve transport layer from the constraint of in-order packet delivery, since applications can determine the order of packets by their timestamps.
- Third, the fact that events are rare simplifies flow and congestion control, since packets are not generated continuously.

Therefore, techniques that take advantage of these opportunities and meet the challenges of reliable and real-time bursty convergecast are desired.

Related work. The performance of packet delivery in dense sensor networks is studied in [5]. The results show that, in the presence of heavy channel load, a commonly used loss recovery scheme at link layer (i.e., lost packets are retransmitted up to 3 times) does not mask packet loss, and more than 50% of the links observe 50% packet loss. The observation shows the challenge of reliable communication over multi-hop routes, since the reliability decreases exponentially as the number of hops increases.

The limitations of timers in TCP retransmission control are studied in [6]. The author analyzes the intrinsic difficulties in using timers to achieve optimal performance and argues that additional mechanisms should be used. Despite its focus on TCP, the study also applies to retransmission control in sensor networks, since timers directly affect the reliability as well as the delay in packet delivery (e.g., large timeout values of timers tend to increase packet delivery delay, whereas small timeout values tend to cause unnecessary retransmissions and thus decrease packet delivery reliability).

RMST [3] and PSFQ [4] show the importance of hop-by-hop packet recovery in sensor networks. Yet RMST and PSFQ are not designed for reliable and real-time bursty convergecast: first, they do not cope with retransmission-incurred channel contention; second, they both use timer-based NACK as the basis for retransmission control, but they do not design mechanisms to alleviate the delay incurred by the timers whose timeout values are conservatively chosen to reduce unnecessary retransmissions; third, they do not design mechanisms to reduce the probability of ack-loss. Moreover, the timers used in RMST are not adaptive and can introduce more retransmissions or delay than necessary. RMST and PSFQ also use in-network caching, which, if used in bursty convergecast, would consume significant amount of memory.

For packet-loss detection and retransmission control, MintRoute [7] uses synchronous explicit ack (SEA), and DFRF [8] uses stop-and-wait implicit ack (SWIA). Yet neither MintRoute nor DFRF addresses the issue of retransmission-incurred channel contention. Moreover, the retransmission timers in DFRF do not adapt to the varying ack-delay, which can introduce more retransmission or delay than necessary (as in RMST). (We study in detail the limitations of SEA and SWIA in Section III-A.)

CODA [9] and ESRT [10] study congestion control in sensor networks. But they do not consider bursty convergecast where the traffic bursts are well separated; instead, they consider a traffic model where multiple sources are continuously or periodically generating packets. Therefore, they do not focus on real-time packet delivery, and they do not consider retransmission-incurred delay as well as channel contention. Many transport protocols, such as ATP [11] and WTCP [12], are also proposed for wireless ad hoc networks. Again, they do not face the challenges of reliable bursty convergecast, and cannot be directly applied.

Block acknowledgment [13] is proposed for error as well as flow control in the Internet. It considers the problem of in-order packet delivery. Therefore, a lost packet blocks the delivery of all the packets that are behind the lost one but have reached the receiver, as a result of which packet delivery delay is increased. Moreover, a sender can send packets at most up to its window size once a packet is sent but unacknowledged, thus the channel resource may be under-utilized. Block acknowledgment also uses timers without addressing their limitations, which renders additional delay in packet delivery.

Contributions of the paper. We discover, to our surprise, that two commonly used hop-by-hop packet recovery schemes do not work well in bursty convergecast. The lack of retransmission scheduling in both schemes makes retransmission-based packet recovery ineffective in the case of bursty convergecast. Moreover, in-order packet delivery makes the communication channel under-utilized in the presence of packet- and ack-loss.

To address the challenges, we design an implicit-ack based protocol RBC (for Reliable Bursty Convergecast). RBC takes advantage of the unique sensor network models and overcomes the limitations of existing schemes via the following mechanisms:

- To improve channel utilization, RBC uses a windowless block acknowledgment scheme that enables con-

tinuous packet forwarding in the presence of packet- and ack-loss. The block acknowledgment also reduces the probability of ack-loss, by replicating the acknowledgment for a received packet.

- To ameliorate retransmission-incurred channel contention, RBC introduces differentiated contention control, which ranks nodes by their queuing conditions as well as the number of times that the enqueued packets have been transmitted. A node ranked the highest within its neighborhood accesses the channel first.

In addition, we design techniques that address the challenges of timer-based retransmission control in bursty convergecast:

- To deal with continuously changing ack-delay, RBC uses adaptive retransmission timer which adjusts itself as network state changes.
- To reduce delay in timer-based retransmission and to expedite retransmission of lost packets, RBC uses block-NACK, retransmission timer reset, and channel utilization protection.

We also design a lightweight flow control mechanism to avoid queue overflow.

We evaluate RBC by experimenting with an outdoor testbed of 49 MICA2 motes and with realistic traffic trace from the field sensor network of Lites. Our experimental results show that, compared with a commonly used implicit-ack scheme, RBC increases the packet delivery ratio by a factor of 2.05 and reduces the packet delivery delay by a factor of 10.91. Moreover, RBC achieves a goodput of 6.37 packets/second for the traffic trace of Lites, almost reaching the optimal goodput — 6.66 packets/second — for the trace.

RBC is independent of the underlying MAC protocol and works with any routing protocols that maintain routing structures (i.e., each node chooses a node as its next-hop to reach a destination).

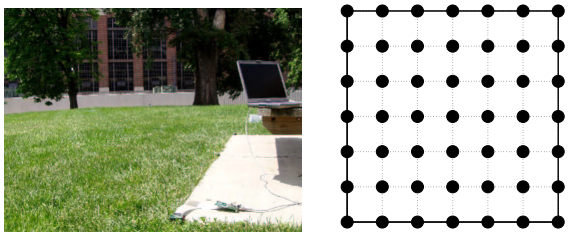
Organization of the paper. We describe our testbed and discuss the experiment design in Section II. In Section III, we study the limitations of existing hop-by-hop control mechanisms, and we evaluate the alternatives in reliable bursty convergecast. We present the detailed design of RBC in Section IV, then we present the analytical and experimental results in Section V. We make concluding remarks in Section VI.

II. Testbed and experiment design

Towards characterizing the issues in making bursty convergecast both reliable and timely, we conduct an experimental study. We choose experimentation as opposed to simulation in order to gain higher fidelity and confidence in the observations. Before presenting our study, we first describe our testbed and the experiment design.

Testbed. We setup our testbed to reflect the field sensor network of Lites, and we use the traffic trace for a typical event in Lites as the basis of our experiments.

The testbed consists of 49 MICA2 motes deployed in a grass field, as shown in Figure 1(a), where the grass is



(a) Environment (b) Grid topology

Fig. 1. The testbed

2-4 inches tall.The 49 motes form a 7×7 grid with a 5-feet separation between neighboring grid points, as shown in Figure 1(b) where each grid point represents a mote. The mote at the left-bottom corner of the grid is the base station to which all the other motes send packets.

The traffic trace (simply called *Lites trace* hereafter) corresponds to an event where each mote except for the base station generates two packets and, overall, 96 packets are generated. The cumulative distribution of the number of packets generated during the event is shown in Figure 2.

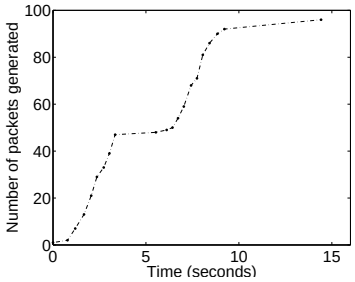


Fig. 2. The distribution of packets generated in Lites trace

If we define the burst rate up to a moment in the event as the number of packets generated so far divided by the time since the first packet is generated, the highest burst rate in Lites trace is 14.07 packets/second. Given that the highest one-hop throughput is about 42.93 packets/second for MICA2 motes with B-MAC (the latest MAC component of TinyOS) and that, in multi-hop networks, even an ideal MAC can only achieve $\frac{1}{4}$ of the throughput that a

single-hop transmission can achieve [14], the burst rate of Lites trace far exceeds the rate at which the motes can push packets to the base station, a serious challenge for reliable and real-time packet delivery.

Experiment design. To generate a multi-hop network, we let each mote transmit at the minimum power level by which two motes 10 feet apart are able to reliably communicate with each other, and the power level is 9 (out of a range between 1 and 255) in our case. We use the routing protocol LGR [15] in our testbed, since we can conveniently configure the routing structure in LGR.[1] In LGR, each mote forwards its traffic to a mote 2 grid-units closer to the base station, unless the mote is within 2 grid-units distance from the base station in which case the mote sends its traffic directly to the base station. Therefore, the number of routing hops a packet may go through is up to 6, and the average number of hops is 3.3.

For each protocol we evaluate, we run the Lites trace 10 times and measure the average performance of the protocol by the following metrics:

- *Event reliability (ER)*: the number of unique packets received at the base station in an event divided by the number packets generated for the event.
  Event reliability reflects how well an event is reported to the base station.
- *Packet delivery delay (PD)*: the time taken for a packet to reach the base station from the node that generates it.
- *Event goodput (EG)*: the number of unique packets received at the base station divided by the interval between the start and the end of an event.
  Event goodput reflects how fast the traffic of an event is pushed from the network to the base station. By definition, the optimal event goodput for Lites trace is 6.66 packets/second, which corresponds to the case where the packet delivery delay is 0 and all the packets are received by the base station.
- *Node reliability (NR)*: the number of unique packets that are generated by a mote and received by the base station divided by the number of packets generated at the mote.

(Remark: The study in this paper applies to cases where network topologies other than grid and routing protocols other than LGR are used, since the protocols studied are independent of the network topology and the routing protocol.)

[1]To focus on transport issues,we disable the "base-snooping" in LGR so that the base station does not accept packets snooped over the channel.

III. Bursty convergecast: issues and alternatives

In this section, we first study the limitations of existing hop-by-hop packet recovery mechanisms. Then we evaluate the alternatives in reliable bursty convergecast.

A. Limitations of two commonly used mechanisms

Two widely used hop-by-hop packet recovery mechanisms in sensor networks are synchronous explicit ack and stop-and-wait implicit ack. We study their performance in bursty convergecast as follows.

1) Synchronous explicit ack (SEA): SEA is the default MAC layer acknowledgment mechanism in TinyOS [2] and many well-known MAC protocols such as S-MAC [16]. In SEA, a receiver switches to transmit-mode, immediately after receiving a packet, and sends back the acknowledgment without going through the procedure of channel access control. We evaluate SEA in our testbed, and the event reliability, the average packet delivery delay, as well as the event goodput is shown in Table I, where

Metrics	RT = 0	RT = 1	RT = 2
ER (%)	51.05	54.74	54.63
PD (seconds)	0.21	0.25	0.26
EG (packets/sec)	4.01	4.05	3.63

TABLE I
SEA in Lites trace

RT stands for the maximum number of retransmissions for each packet at each hop (e.g., RT = 0 means that packets are not retransmitted). The distribution of the number of unique packets received at the base station along time is shown in Figure 3.

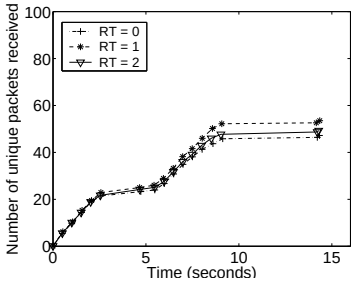


Fig. 3. The distribution of packet reception in SEA

Table I and Figure 3 show that when packets are retransmitted, the event reliability increases slightly (i.e., by up to 3.69%). Nevertheless, the maximum reliability is still only 54.74%, and, even worse, the event reliability as well as goodput decreases when the maximum number of retransmissions increases from 1 to 2.

The reason why retransmission does not significantly improve — and can even degenerate — communication reliability is that, in SEA, lost packets are retransmitted while new packets are generated and forwarded, thus retransmissions, when not scheduled appropriately, only increase channel contention and cause more packet collision. The situation is further exacerbated by ack-loss (with a probability as high as 10.29%), since ack-loss causes unnecessary retransmission of packets that have been received. To make retransmission effective in improving reliability, therefore, we need a retransmission scheduling mechanism that ameliorates retransmission-incurred channel contention.

2) Stop-and-wait implicit ack (SWIA): SWIA takes advantage of the fact that every node, except for the base station, forwards the packet it receives and the forwarded packet can act as the acknowledgment to the sender at the previous hop [8]. In SWIA, the sender of a packet snoops the channel to check whether the packet is forwarded within certain constant threshold time; the sender regards the packet as received if it is forwarded within the threshold time, otherwise the packet is regarded as lost. The advantage of SWIA is that acknowledgment comes for free except for the limited control information piggybacked in data packets.

We evaluate SWIA in our testbed, and the performance results are shown in Table II. The distribution of packet reception is shown in Figure 4.

Metrics	RT = 0	RT = 1	RT = 2
ER (%)	43.09	31.76	46.5
PD (seconds)	0.35	8.81	18.77
EG (packets/sec)	3.48	2.58	1.41

TABLE II
SWIA IN LITES TRACE

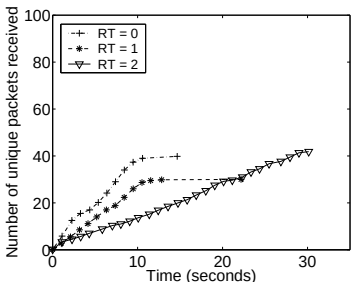


Fig. 4. The distribution of packet reception in SWIA

Table II and Figure 4 show that the maximum event reliability in SWIA is only 46.5%, and that the reliability decreases significantly when packets are retransmitted at most once at each hop. When packets are retransmitted up to twice at each hop, the packet delivery delay increases, and the event goodput decreases significantly despite the slightly increased reliability.

Unlike in SEA, we also observe that queues overflow at four nodes, and one of them drops 41% the packets it receives consequently.

The above phenomena are due to the following reasons. First, the length of data packets is increased by the piggybacked control information in SWIA, thus the ack-loss probability increases (as high as 18.39% in our experiments), which in turn increases unnecessary retransmissions. Second, most packets are queued upon reception and thus their forwarding is delayed. As a result, the piggybacked acknowledgments are delayed and the corresponding packets are retransmitted unnecessarily. Third, once a packet is waiting to be acknowledged, all the packets arriving later cannot be forwarded even if the communication channel is free. Therefore, channel utilization as well as system throughput decreases, and network queuing as well as packet delivery delay increases. Fourth, as in SEA, lack of retransmission scheduling allows retransmissions, be it necessary or unnecessary, to cause more channel contention and packet loss.

B. Evaluating the alternatives

From the study above, we see that existing hop-by-hop packet recovery mechanisms do not meet the challenges of bursty convergecast. Therefore we need to re-think how to perform hop-by-hop control in bursty convergecast. Broadly speaking, the design space can be divided along the following two dimensions:

- **ACK vs. NACK.** In negative-ACK (NACK), the sender of a packet has to buffer the packet for certain threshold time whether or not the packet has been received, whereas in ACK the sender can safely release the packet whenever it is acknowledged. Therefore, NACK usually requires more buffering than ACK. This is especially true when more packets are received than lost, which is the normal case in sensor networks (e.g., the average link reliability is 77.3% in Lites trace). Given a buffer of constant size, higher degree of buffering also implies slower speed in delivering packets from one node to another.
  Moreover, loss of NACK leads to the loss of the corresponding packet, whereas loss of ACK only causes increased channel contention which does not imply 100% probability of packet loss.
  Considering the limited memory available in sensor nodes (e.g., 4KB for MICA2s) and the high reliability and real-time requirements of bursty converge-

cast, we use ACK as the basis of hop-by-hop control, and we only use NACK as an assisting mechanism.

- **Implicit vs. explicit ACK.** Due to physical constraints of the MICA2 radios, an acknowledgment packet in SEA takes 16 bytes, including the preamble, the synchronization-code, and the ack-code [2]. Yet only 3 bytes, used for the ack-code, in the 16-byte ack-packet carry information meaningful to reliability control. If we use implicit-ack where control information is piggybacked along data packets, all the 16 bytes can be used to convey information meaningful to transport control.

  Moreover, reliable and real-time data transport in bursty convergecast is essentially an issue above MAC layer, therefore it would be architecturally more appropriate to implement the related control mechanisms above MAC layer. In this case, implicit-ack is better than explicit-ack in reducing channel contention, since the latter increases (doubles at least) the number of packets competing for channel access.

  To make better use of every byte sent and to reduce channel contention, therefore, we use implicit-ack.

Challenges. As shown in Section III-A.2, protocols based on implicit-ack face the following special challenges:

- *Ack-loss*: longer packet length means higher probability of ack-loss.
- *Varying ack-delay*: queuing of data packets means varying delay in acknowledgment.

In addition, bursty convergecast also poses the following challenges to reliable and real-time transport control:

- *Inefficient channel utilization*: "in-order packet delivery"-oriented error control does not fully utilize network resources and introduces additional delay than necessary.
- *Retransmission-incurred-contention*: retransmission further increases the number of packets competing for channel access and thus increases the probability of packet collision and loss.
- *Timer-incurred-delay*: conservatively chosen retransmission timer increases delay in packet delivery.
- *Queue overflow*: high-volume bursty traffic and queue accumulation due to unacknowledged packets increase the probability of queue overflow.

IV. PROTOCOL RBC

To address the challenges of bursty convergecast, we design protocol RBC which consists of three components: error control, contention control, and flow control. Error control improves channel utilization and handles ack-loss, varying ack-delay, as well as timer-incurred-delay; contention control alleviates retransmission-incurred-contention as well as unbalanced queuing; and flow control prevents queue overflow. We elaborate on the three components as follows.

A. Error control

Error control handles reliable hop-by-hop packet delivery. We present the error control component by explaining how it addresses the challenges of inefficient channel utilization, ack-loss, varying ack-delay, and timer-incurred-delay. For clarity of presentation, we consider an arbitrary pair of nodes S and R where S is the sender and R is the receiver.

1) Improving channel utilization & reducing ack-loss: In traditional block acknowledgment [13], a sliding-window is used for both duplicate detection and in-order packet delivery.[2] The sliding-window reduces network throughput once a packet is sent but remains unacknowledged (since the sender can only send up to its window size once a packet is unacknowledged), and in-order delivery increases packet delivery delay once a packet is lost (since the lost packet delays the delivery of every packet behind it). Therefore, the sliding-window based block acknowledgment scheme does not apply to bursty convergecast, given the real-time requirement of the latter.

To address the constraints of traditional block acknowledgment in the presence of unreliable links, we take advantage of the fact that in-order delivery is not required in bursty convergecast. Without considering the order of packet delivery, by which we only need to detect whether a sequence of packets are received without loss in the middle and whether a received packet is a duplicate of a previously received one. To this end, we design, as follows, a *window-less block acknowledgment* scheme which guarantees continuous packet forwarding irrespective of the underlying link unreliability as well as the resulting packet- and ack-loss.

Window-less queue management. The sender S organizes its packet queue as $(M+2)$ linked lists, as shown in Figure 5, where M is the maximum number of retransmissions at each hop. For convenience, we call the linked lists *virtual queues*, denoted as $Q_0, \ldots, Q_{M+1}$. The virtual queues are ranked such that a virtual queue Q_k ranks higher than Q_j if $k < j$.

Virtual queues $Q_0, Q_1, \ldots,$ and Q_M buffer packets waiting to be sent or to be acknowledged, and Q_{M+1} collects the list of free queue buffers. The virtual queues are maintained as follows:

[2]Note that SWIA is a special type of block acknowledgment where the window size is 1.

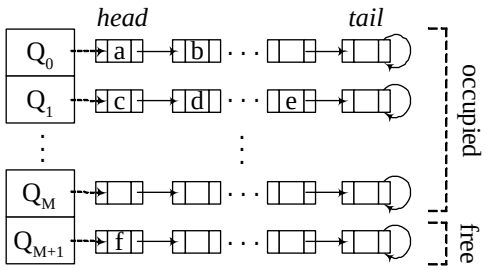


Fig. 5. Virtual queues at a node

- When a new packet arrives at S to be sent, S detaches the head buffer of Q_{M+1}, if any, stores the packet into the queue buffer, and attaches the queue buffer to the tail of Q_0.
- Packets stored in a virtual queue Q_k ($k > 0$) will not be sent unless Q_{k-1} is empty; packets in the same virtual queue are sent in FIFO order.
- After a packet in a virtual queue Q_k ($k \geq 0$) is sent, the corresponding queue buffer is moved to the tail of Q_{k+1}, unless the packet has been retransmitted M times[3] in which case the queue buffer is moved to the tail of Q_{M+1}.
- When a packet is acknowledged to have been received, the buffer holding the packet is released and moved to the tail of Q_{M+1}.

Therefore, the order in which unacknowledged packets have been sent is maintained in the virtual queues without any window-based control, providing the basis for window-less block acknowledgment. Moreover, newly arrived packets can be sent immediately without waiting for the previously sent packets to be acknowledged, which enables continuous packet forwarding in the presence of packet- and ack-loss.

Block acknowledgment & reduced ack-loss. Each queue buffer at S has an ID that is unique at S. When S sends a packet to the receiver R, S attaches the ID of the buffer holding the packet as well as the ID of the buffer holding the packet to be sent next. In Figure 5, for example, when S sends the packet in buffer a, S attaches the values a and b. Given the queue maintenance procedure, if the buffer holding the packet being sent is the tail of Q_0 or the head of a virtual queue other than Q_0, S also attaches the ID of the head buffer of Q_{M+1}, if any, since a new packet may arrive before the next enqueued packet is sent in which case the newly arrived packet will be sent first. For example, when the packet in buffer c of Figure 5 is sent, S attaches the values c, d, and f.

[3]Due to block-NACK, to be discussed in Section IV-A.3, a packet having been retransmitted M times may be in a virtual queue other than Q_M.

When the receiver R receives a packet p_0 from S, R learns the ID n' of the buffer holding the next packet to be sent by S. When R receives a packet p_n from S next time, R checks whether p_n is from buffer n' at S: if p_n is from buffer n', R knows that there is no packet loss between receiving p_0 and p_n from S; otherwise, R detects that some packets are lost between p_0 and p_n.

For each maximal sequence of packets $p_k, \ldots, p_{k'}$ from S that are received at R without any loss in the middle, R attaches to packet $p_{k'}$ the 2-tuple $\langle q_k, q_{k'} \rangle$, where q_k and $q_{k'}$ are the IDs of the buffers storing p_k and $p_{k'}$ at S. We call $\langle q_k, q_{k'} \rangle$ the *block acknowledgment* for packets $p_k, \ldots, p_{k'}$. When S snoops the forwarded packet $p_{k'}$ later, S learns that all the packets sent between p_k and $p_{k'}$ have been received by R. Then S releases the buffers holding these packets. For example, if S snoops a block acknowledgment $\langle c, e \rangle$ when its queue state is as shown in Figure 5, S knows that all the packets in buffers between c and e in Q_1 have been received, and S releases buffers between c and e, including c and e.

One delicate detail in processing the block acknowledgment $\langle q_k, q_{k'} \rangle$ is that after releasing buffer q_k, S will maintain a mapping $q_k \leftrightarrow q_{k''}$, where $q_{k''}$ is the buffer holding the packet sent (or to be sent next) after that in $q_{k'}$. When S snoops another block acknowledgment $\langle q_k, q_n \rangle$ later, S knows, by $q_k \leftrightarrow q_{k''}$, that packets sent between those in buffers $q_{k''}$ and q_n have been received by R; then S releases the buffers holding these packets, and S resets the mapping to $q_k \leftrightarrow q_{n''}$, where $q_{n''}$ is the buffer holding the packet sent (or to be sent next) after that in q_n. S maintains the mapping for q_k until S receives a block-NACK (to be discussed in Section IV-A.3) or a block acknowledgment $\langle q, q' \rangle$ where $q \neq q_k$.

In the above block acknowledgment scheme, the acknowledgment for a received packet is piggybacked onto the packet itself as well as the packets that are received consecutively after the packet without any loss in the middle. Therefore, the acknowledgment is replicated and the probability for it to be lost decreases significantly, by a factor of 2.07 in Lites trace (the detailed analysis is to be presented in Section V-A).

Duplicate detection & obsolete-ack filtering. Since it is impossible to completely prevent ack-loss in lossy communication channels, packets whose acknowledgments are lost will be retransmitted unnecessarily. Therefore, it is necessary that duplicate packets be detected and dropped.

To enable duplicate detection, the sender S maintains a counter for each queue buffer, whose value is incremented by one each time a new packet is stored in the buffer. When S sends a packet, it attaches the current value of the corresponding buffer counter. For each buffer q at S,

the receiver R maintains the counter value c_q piggybacked in the last packet from the buffer. When R receives another packet from the buffer q later, R checks whether the counter value piggybacked in the packet equals to c_q: if they are equal, R knows that the packet is a duplicate and drops it; otherwise R regards the packet as a new one and accepts it. The duplicate detection is local in the sense that it only requires information local to each queue buffer instead of imposing any rule involving different buffers (such as in sliding-window) that can degenerate system performance.

For the correctness of the above duplicate detection mechanism, we only need to choose the domain size C for the counter value such that the probability of losing C packets in succession is negligible. For example, for the high per-hop packet loss probability 22.7% in the case of Lites trace, C could still be as small as 7, since the probability of losing 7 packets in succession is only 0.003%.[4]

In addition to duplicate detection, we also use buffer counter to filter out obsolete acknowledgment. Despite the low probability, packet forwarding at R may be severely delayed, such that the queue buffers signified in a block acknowledgment have been reused by S to hold packets arriving later. To deal with this, R attaches to each forwarded packet the ID as well as the counter value of the buffer holding the packet at S originally; when S snoops a packet forwarded by R, S checks whether the piggybacked counter value equals to the current value of the corresponding buffer: if they are equal, S regards as valid the piggybacked block acknowledgment; otherwise, S regards the block acknowledgment as obsolete and ignores it.

2) Dealing with varying ack-delay: When the receiver R receives a packet m from the sender S, R first buffers m in Q_0. The delay in R forwarding m depends on the number of packets in front of m in Q_0. Since the number of packets enqueued in Q_0 keeps changing, the delay in forwarding a received packet by R keeps changing, which leads to varying delay in packet acknowledgment. Therefore, the retransmission timer at the sender S should adapt to the queuing condition at R; otherwise, either lost packets are unnecessarily delayed in retransmission (when the retransmission timer is too large) or packets are unnecessarily retransmitted even if they are received (when the retransmission timer is too small).

To adaptively setting the retransmission timer for a packet, the sender S keeps track of, by snooping packets forwarded by R, the length s_r of Q_0 at R, the average delay d_r in R forwarding a packet after the packet becomes the head of Q_0, and the deviation d'_r of d_r. When S sends a packet to R, S sets the retransmission timer of the packet as

$$(s_r + C_0)(d_r + 4d'_r)$$

where C_0 is a constant denoting the number of new packets that R may have received since S learned s_r the last time (C_0 depends on the application as well as the link reliability, and C_0 is 3 in our experiments). The reason why we use the deviation d'_r in the above formula is that d_r varies a lot in wireless networks in the presence of bursty traffic, in which case the deviation improves estimation quality [17].

At a node, each local parameter α (such as d_r for node R) and its deviation α' are estimated by the method of *exponentially weighted moving average (EWMA)* as follows:

$$\alpha \leftarrow (1-\gamma)\alpha + \gamma\alpha''$$
$$\alpha' \leftarrow (1-\gamma')\alpha' + \gamma'|\alpha'' - \alpha|$$

where γ and γ' are weight-factors, and α'' is the latest observation of α. As suggested in [17], we set $\gamma = \frac{1}{8}$ and $\gamma' = \frac{1}{4}$ in RBC.

3) Alleviating timer-incurred delay: The packet retransmission timer calculated as above is conservative in the sense that it is usually greater than the actual ack-delay [17]. This is important for reducing the probability of unnecessary retransmissions, but it introduces extra delay and makes network resources under-utilized [6].

To alleviate timer-incurred delay, we design the following mechanisms to expedite necessary packet retransmissions and to improve channel utilization:

- Whenever the receiver R receives a packet m from buffer n of the sender S while R is expecting (in the absence of packet loss) to receive a packet from buffer n' of S, R learns that packets sent between those in buffers n' and n at S, including the one in n', are lost. In this case, R piggybacks a block-NACK $[n', n)$ onto the next packet it forwards, by which the block-NACK can be snooped by S immediately.
  When S learns the block-NACK $[n', n)$ from R, S resets the retransmission timers to 0 for the packets sent between those in n' and n (including the one in n'), and for each of these packets, S moves the corresponding buffer to the tail of Q_{k-1} if the buffer is currently at Q_k. Therefore, packets that need to be retransmitted are put into higher-ranked virtual queues and are retransmitted quickly.[5]

[4]Given the small domain size for the counter value as well as the usually small queue size at each node, the duplicate detection mechanism does not consume much memory. For example, it only takes 36 bytes in the case of Lites.

[5]Note that the movement of NACKed packets do not disrupt the buffering order required by block-acknowledgment.

- Whenever S learns that the virtual queue Q_0 of R becomes empty, S knows that R has forwarded all the packets it has received. In this case, S resets the retransmission timers to 0 for those packets still waiting to be acknowledged, since they will not be (due to either packet-loss or ack-loss).
  Similarly, when S snoops the acknowledgment for a packet m, S resets the retransmission timer to 0 for those packets that are sent before m but are still waiting to be acknowledged.
- When a network channel is fully utilized, it should be busy all the time. Therefore, if the sender S has packets to send, and if S notices that no packet is sent by any neighboring node in a period of $C_1 \times T_{pkt}$ time, S sends out the packet at the head of its highest-ranked non-empty virtual queue, without considering the retransmission timer even if the packet is to be acknowledged.[6] C_1 is a constant reflecting the degree of channel utilization we want and T_{pkt} is the time taken to transmit a packet at the MAC layer. T_{pkt} is estimated by the method of EWMA.

4) Aggregated-ack at the base station: In sensor networks, the base station usually forwards all the packets it receives to an external network. As a result, the children of the base station (i.e., the nodes that forward packets directly to the base station) are unable to snoop the packets the base station forwards, and the base station has to explicitly acknowledge the packets it receives. To reduce channel contention, the base station aggregates several acknowledgments, for packets received consecutively in a short period of time, into a single packet and broadcasts the packet to its children. Accordingly, the children of the base station adapt their error control parameters to the way the base station handles acknowledgments.

B. Differentiated contention control

In multi-hop wireless sensor networks with reliable per-hop connectivity, most packet losses are due to collision, which in turn is caused by heavy channel contention. To enable reliable packet delivery, lost packets need to be retransmitted. Nevertheless, packet retransmission may cause more channel contention and packet loss, thus degenerating communication reliability. Moreover, there also exist unnecessary retransmissions due to ack-loss, which only increase channel contention and reduce communication reliability. Therefore, it is desirable to schedule packet retransmissions such that they do not interfere with transmissions of other packets.

[6]This mechanism improves channel utilization without introducing unnecessary retransmissions because of the "differentiated contention control" in RBC (to be discussed in the next subsection).

The way the virtual queues are maintained in our window-less block acknowledgment scheme facilitates the retransmission scheduling, since packets are automatically grouped together by different virtual queues. Packets in higher-ranked virtual queues have been transmitted less number of times, and the probability that the receiver has already received the packets in higher-ranked virtual queues is lower (e.g., 0 for packets in Q_0). Therefore, we rank packets by the rank of the virtual queues holding the packets, and higher-ranked packets have higher-priority in accessing the communication channel. By this rule, packets that have been transmitted less number of times will be (re)transmitted earlier than those that have been transmitted more, and interference between packets of different ranks is reduced.

Window-less block acknowledgment already handles packet differentiation and scheduling within a node, thus we only need a mechanism that schedules packet transmission across different nodes. To reduce interference between packets of the same rank and to balance network queuing as well as channel contention across nodes, inter-node packet scheduling also takes into account the number of packets of a certain rank so that nodes having more such packets transmit earlier.

To implement the above concepts, we define the rank $rank(j)$ of a node j as $\langle M - k, |Q_k|, ID(j)\rangle$, where Q_k is the highest-ranked non-empty virtual queue at j, $|Q_k|$ is the number of packets in Q_k, and $ID(j)$ is the ID of j. A node with a higher rank value ranks higher. Then, the distributed transmission scheduling works as follows:

- Each node piggybacks its rank to the data packets it sends out.
- Upon snooping or receiving a packet, a node j compares its rank with that of the packet sender k. If k ranks higher than j, j will not send any packet in the following $w(j,k) \times T_{pkt}$ time. T_{pkt} is the time taken to transmit a packet at the MAC layer, and $w(j,k) = 4 - i$, when $rank(j)$ and $rank(k)$ differ at the i-th element of the 3-tuple ranks. $w(j,k)$ is defined such that the probability of all waiting nodes starting their transmissions simultaneously is reduced, and that higher-ranked nodes tend to wait for shorter time.
- If a sending node j detects that it will not send its next packet within T_{pkt} time, j signifies this by marking the packet being sent, so that the nodes overhearing the packet will skip j in the contention control. (This mechanism reduces the probability of idle waiting, where the channel is free but no packet is sent.)

C. Flow control

In implicit-ack based bursty convergecast, two factors are most likely to cause queue overflow: high-volume bursty traffic and queue accumulation due to unacknowledged packets. The window-less block acknowledgment scheme alleviates the latter in the following ways: first, it enables quick acknowledgment by forwarding newly received packets earlier than those to be retransmitted; second, it reduces ack-loss by replicating acknowledgments for received packets. Therefore, we only need to design a mechanism for coping with high-volume bursty traffic.

In the presence of bursty traffic, the queue at a node may overflow if the corresponding senders send too many packets in a short time. To avoid this, we design as follows a flow control mechanism which tolerates packet losses:

- When forwarding packets, a node piggybacks the number of free queue buffers at its place.
- Whenever a sender S detects that the number L_r of free queue buffers at the receiver R is below a threshold L, S will stop sending any packet in the following $(L - L_r) \times d_{e,R}$ time. L is a constant chosen such that the probability of losing L packets in succession is negligible (by which the sender will not fail to detect the congestion state at the receiver), and $d_{e,R}$ is the average interval between R releasing one buffer and the next one while there are packets enqueued at R. (R estimates $d_{e,R}$ by the method of EWMA.)
- After learning the number L_r of free buffers at the receiver R each time, the sender S will send at most L_r packets to R in the following $L_r \times d_{e,R}$ time unless S snoops another packet forwarded by R.
- To help relieve queue congestion, the nodes having less than L queue buffers are not subject to the differentiated contention control.

V. Analytical and experimental results

In this section, we first analyze the ack-loss probability in RBC, then we present the experimental results.

A. Ack-loss probability

For convenience, we define the following notations:

p	:	the probability of losing a single (data) packet;
N	:	the number of packets received in succession without any loss in the middle;
N'	:	the number of packets lost in succession;
B	:	the number of packets received in succession without any loss in the middle, after a packet is already received;
A	:	the number of times that the acknowledgment for a packet is received at the sender.

Assuming that packet losses are independent of one another, we have the probability mass functions for random variables N and N' as follows.

$$\begin{aligned} P[N = k] &= p(1-p)^k \\ P[N' = k] &= (1-p)p^k \end{aligned}$$

In RBC, when a packet m is received at a receiver R, the acknowledgment for m can reach back to the sender S in two ways: S snoops m when it is forwarded by R later, with probability P_{self}; or S does not snoop m but snoops a packet whose block acknowledgment acknowledges the reception of m, with probability P_{ba}. Therefore, the probability P_{rbc} of S receiving the acknowledgment for m can be derived as follows:

$$\begin{aligned} P_{self} &= 1-p \\ P_{ba} &= p\sum_{k=0}^{\infty} P[B=k]P[A \geq 1|B=k] \\ &= p\sum_{k=0}^{\infty} P[B=k](1-P[A=0|B=k]) \\ &= p\sum_{k=0}^{\infty} P[N=k+1](1-P[N'=k]) \\ &= \frac{p(1-3p+4p^2-2p^3)}{1-p+p^2} \\ P_{rbc} &= P_{self} + P_{ba} \\ &= 1-p+\frac{p(1-3p+4p^2-2p^3)}{1-p+p^2} \end{aligned}$$

Then, the probability P'_{rbc} of losing the acknowledgment for a packet in RBC is $1 - P_{rbc}$.

In the case of Lites trace and implicit-ack, $p = 22.7\%$. Therefore $P'_{rbc} = 8.89\%$, reducing the ack-loss probability of SWIA by a factor of 2.07.

B. Experimental results

We have implemented protocol RBC in TinyOS, where the control logic takes 185 bytes of RAM when each node maintains a buffer capable of holding 16 packets, and the control information piggybacked in data packets takes 14 bytes.[7] We have successfully applied RBC in the field sensor network of Lites to support reliable and real-time convergecast [1]. We have also evaluated RBC in our testbed, and the performance results are shown in Table III. Figure 6 shows the distribution of packet reception in RBC. From Table III and Figure 6, we observe the following properties of RBC:

Metrics	RT = 0	RT = 1	RT = 2
ER (%)	56.21	83.16	95.26
PD (seconds)	0.21	1.18	1.72
EG (packets/sec)	4.28	5.72	6.37

TABLE III
RBC IN LITES TRACE

[7] We are currently optimizing our implementation to further reduce the memory consumption and the length of the piggybacked control information.

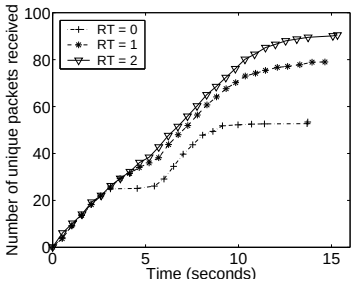


Fig. 6. The distribution of packet reception in RBC

- The event reliability keeps increasing, in a significant manner, as the number of retransmissions increases. The increased reliability mainly attributes to reduced unnecessary retransmissions (by reduced ack loss and adaptive retransmission timer) and retransmission scheduling.
- Compared with SWIA which is also based on implicit-ack, RBC reduces packet delivery delay significantly. This mainly attributes to the ability of continuous packet forwarding in the presence of packet- and ack-loss and the reduction in timer-incurred delay.
- The rate of packet reception at the base station and the event goodput keep increasing as the number of retransmissions increases. When packets are retransmitted up to twice at each hop, the event goodput reaches 6.37 packets/second, quite close to the optimal goodput — 6.66 packets/second — for Lites trace.

Moreover, our experiments show that there is no queue overflow at any node due to the flow control in RBC.

Compared with SWIA, RBC improves reliability by a factor of 2.05 and reduces average packet delivery delay by a factor of 10.91. Compared with SEA, RBC improves reliability by a factor of 1.74, but the average packet delivery delay increases by a factor of 6.61 in RBC. Interestingly, however, RBC still improves the event goodput by a factor of 1.75 when compared with SEA. The reason is that, in RBC, lost packets are retransmitted and delivered after those packets that are generated later but transmitted less number of times. Therefore, the delivery delay for lost packets increases, which increases the average packet delivery delay, without degenerating the system goodput. The observation shows that, due to the unique application models in sensor networks, metrics evaluating aggregate system behaviors (such as the event goodput) tend to be of more relevance than metrics evaluating unit behaviors (such as the delay in delivering each individual packet).

To further understand protocol behaviors in the presence of packet retransmissions, we conduct, as follows, a comparative study of RBC, SWIA, and SEA for the case where packets are retransmitted up to twice at each hop.

Figure 7 compares the distribution of packet generation

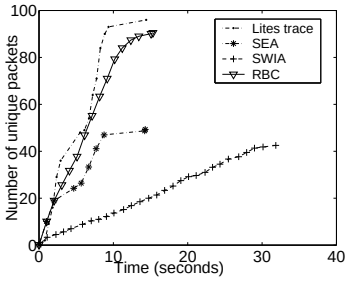
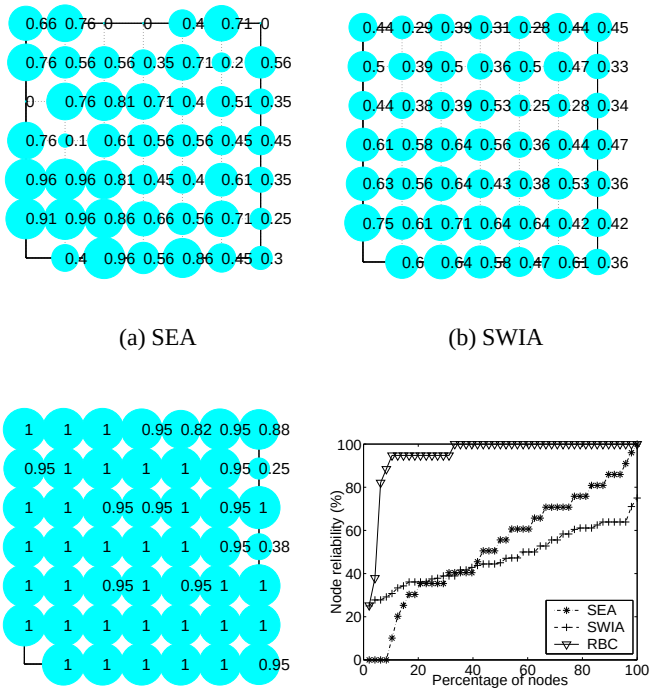


Fig. 7. The distributions of packet generation and reception

in Lites trace with the distributions of packet reception in SEA, SWIA, and RBC. We see that the curve for packet reception in RBC smooths out and almost matches that of packet generation. In contrast, many packets are lost in SEA despite the fact that the rate of packet reception in SEA is close to that in RBC; packet delivery is significantly delayed in SWIA, in addition to the high degree of packet loss.

Figures 8(a)-(c) show the node reliability in SEA,



(a) SEA (b) SWIA

(c) RBC (d) Reliability distribution

Fig. 8. Node reliability

SWIA, and RBC, and Figure 8(d) shows the cumulative distribution of node reliability in SEA, SWIA, and

RBC. We see that node reliability improves significantly in RBC: only 4.17% of nodes have a node reliability less than 80% in RBC; yet in SEA and SWIA, above 80% of nodes have a node reliability less than 80%.

Figure 9 shows the average node reliability in SEA, SWIA, and RBC as the number of routing hops (to the base station) increases. We see that the node reliability in RBC is much higher than that in SEA and SWIA at every routing hop, and that the reliability at the farthest hop in RBC is even greater than that at the closest hop in SEA and SWIA.

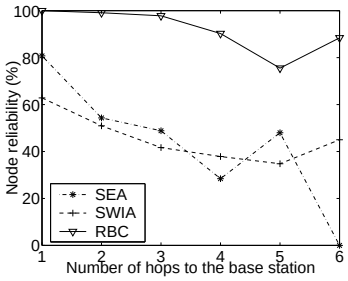


Fig. 9. Node reliability as a function of routing hops

VI. Concluding remarks

Unlike the existing literature on reliable transport in sensor networks that focuses on periodic traffic, we have focused on the problem of bursty convergecast where the key challenges are reliable and real-time error control and the resulting contention control. To address the unique challenges, we have proposed the window-less block acknowledgment scheme which improves channel utilization and reduces ack-loss as well as packet delivery delay; we have also designed mechanisms to schedule packet retransmissions and to reduce timer-incurred delay, which are critical for reliable and real-time transport of bursty traffic. With its well-tested support for reliable and real-time transport of bursty traffic, RBC has been used in the sensor network of Lites and is being used in another field sensor network experiment *ExScal*, where 10,000 XSM motes and 300 StarGates are being deployed [18].

From protocol RBC, we see that bursty convergecast not only poses challenges for reliable and real-time transport control, it also provides unique opportunities for protocol design. Tolerance of out-of-order packet delivery enables the window-less block acknowledgment, which not only guarantees continuous packet delivery in the presence of packet- and ack-loss but also facilitates retransmission scheduling. Well-separated traffic bursts allow us to design light-weight congestion control mechanisms. Overall, the unique network as well as application models in sensor networks offer opportunities for new methodologies in protocol engineering and are interesting areas for further exploration.

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