

# Reliable Communication and Latency Bound Generation in Wireless Cyber Physical System

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Low-power wireless communication has been widely used in cyber-physical systems which require time-critical data delivery. Achieving this goal is challenging because of link burstiness and interference. Based on significant empirical evidence of 21 days and over 3,600,000 packets transmission per link, we propose both routing and scheduling algorithms that produce latency bounds of the real-time periodic streams and accounts for both link bursts and interference. The solution is achieved through the definition of a new metric  $B_{\max}$  that characterizes links by their maximum burst length, and by choosing a novel least-burst-route that minimizes the sum of worst-case burst lengths over all links in the route. With extensive data-driven analysis, we show that our algorithms outperform existing solutions by achieving accurate latency bound with much less energy consumption. In addition, a testbed evaluation consisting of 48 nodes spread across a floor of a building shows that we obtain 100% reliable packet delivery within derived latency bounds. We also demonstrate how performance deteriorates and discuss its implications for wireless networks with insufficient high-quality links.

CCS Concepts: • **Networks** → **Network protocols**; **Network reliability**.

Additional Key Words and Phrases: Link Burstiness, Link Interference, Latency Bound, Reliable Transmission, Real-time Applications.

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## 1 INTRODUCTION

More and more cyber-physical systems (CPS) have been applied to for industrial process [19, 28], structural health monitoring [4, 12], and smart cities [10, 23, 26]. In those applications, low-power wireless communication technology gains rapid adoption due to its better scalability, lower maintenance costs, and flexibility in installation points for either sensing or actuation [21, 39, 42]. To support CPS for controlling or monitoring physical processes, wireless communication in CPS usually requires high reliability and hard end-to-end deadline. However, achieving these goals are difficult since wireless links are non-deterministic because of link burstiness and interference [11, 29, 37]. Link burstiness is a physical property which means that transmissions on a wireless link do not have an independent probability of failure; instead, they have periods of continuous message loss, i.e. they fail in a burst. Link interference is another physical property of the communication environment which causes packet transmission between different links to interfere with each other which results in packet loss. Due to these types of non-determinism, it is difficult to offer reliability and latency bounds for packet delivery over wireless networks.

In this paper, we propose a systematic approach for CPS communication to achieve reliable packet transmission with bounded latency. Our solution includes the following steps: first, we characterize physical properties like interferences and burstiness of a particular network. Then we compute the latency bound for reliable delivery of a certain number of packets on each link. Finally, we schedule packet transmissions of multiple streams in the network to achieve reliable end-to-end reliability within specified latency bound.

It is obvious that we cannot allocate only a single transmission time slot for a stream on each link, especially if we are dealing with bursty links. Because, if the transmission fails at that time slot due to a link burst, the node will need some additional time slots to transmit its packet. So, to provide the end-to-end latency bounds, we have to allocate more than one time slot per link for a stream. The number of time slots we need to allocate depends on the burstiness of the link. We don't want to allocate more time slots than we need, otherwise, we will increase the latency bound. In addition, because of interference, other streams cannot be scheduled in nearby links during that entire multiple slot allocation time, which may increase the overall latency bound of all streams. So, achieving reliable communication and minimizing latency bound by schedule is, therefore, a challenging goal.

The specific problem that we are addressing assumes that we are given a network topology and a set of periodic streams. The route of each stream is either given or assigned by our routing algorithm. After the routing phase, our algorithm outputs a packet transmission schedule and estimates a latency bound for each stream by taking into account both link burstiness and interference. For each stream, if the estimated latency bound is lower than or equal to the period, we offer reliable end-to-end delivery. We confirmed that traditional approaches based on packet reception rate can not bound latency because they do not account for link bursts. Our 21 day long empirical study shows the evidence that over 23% links having packet reception rate (PRR) as high as 0.99 lose more than 50 packets in a row, some lose even over 1000 packets in a row. Thus, we need to carefully design some other metrics to characterize the burstiness of the links that will allow us to allocate a sufficient number of time slots to produce a latency bound. The main contributions of this paper are listed as the following.

- Based on 21 days of empirical study over an 802.15.4 network we define a new metric *max burst length* ( $B_{\max}$ ) that allows us to classify links and allocate a sufficient number of time slots to produce a latency bound of the streams. Empirical analysis indicates that within a period of measuring  $B_{\max}$ , the measured  $B_{\max}$  values are consistent, robust, and can characterize link quality better than PRR.

- Our algorithm has two phases. In the routing phase, we use least-burst-routes to produce minimum latency bounds of the streams by taking into account link bursts and load-balancing. In the broadcast scenario, the algorithm generates a tree-structured route that takes the advantage of omnidirectional transmission. In the unicast scenario, the algorithm generates “least burst” route for each stream which considers load-balancing of each link. In the scheduling phase, the scheduling algorithm is a greedy-based one with the consideration of internal interference.
- The simulation results suggest that the routes generated from our algorithms have lower latency bounds compared with other baselines. By using testbed evaluation we also show that we can actually bound the latency by achieving 100% packet delivery ratio within the derived latency bounds. We also investigate how the performance degrades when we do not have sufficient high-quality links in the network.

An implication of these contributions is that if the transmission period of each stream is greater than or equal to its latency bound that we provide, then our scheduling algorithm allows reliable communication subject to our burstiness and interference assumptions. If the burstiness characterization used for creating the schedules is violated during the course of execution, then a packet might still be missed, but this is rare because the link characterization is performed under realistic operating conditions and an adaptive solution will reduce such scenarios subsequently. Note that our approach does not minimize latency to maximize throughput. Instead, our average delivery latency is higher than most other techniques. However, we do offer a reliable communication and latency bound, which makes it easier to engineer predictable systems. We verify this claim by evaluating our approach on a 48-node wireless testbed with 10 simultaneous and periodic packet streams. The result shows that our scheme has a 100% on-time delivery ratio when all links are available and 90% ones when top 25% PRR links are removed.

A preliminary version of the results in this paper [29] mainly focuses on the method of calculating end-to-end latency bounds with  $B_{\max}$ . In this paper, we provided additional analysis on the robustness of  $B_{\max}$  among links in different classes. Moreover, we proposed routing algorithms for unicast and broadcast scenarios which achieve minimum latency bounds. The evaluation of the routing algorithms and comparison with other baselines is provided in the simulation section.

The rest of this paper is structured as follows. Section 2 formally defines the problem. Section 3 describes related work on scheduling streams with real-time constraints in low-power wireless communication. Section 4 describes our model parameters, assumptions and link classification based on an empirical study. Section 5 describes the routing/scheduling algorithms to achieve minimum latency bounds for streams. Section 6 shows the simulation result of the algorithm performance and compares it with other baselines. Section 7 describes the experimental setup and the results of the experiments. We discuss the future work in Section 8 and conclude our work in Section 9.

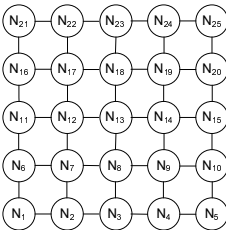


Fig. 1. An Example Topology

Stream	Period	ST	Source	Dest.	Route
$S_1$	20	1	$N_1$	$N_4$	$N_1, N_2, N_3, N_4$
$S_2$	20	1	$N_2$	$N_5$	$N_2, N_3, N_4, N_5$
$S_3$	20	1	$N_7$	$N_9$	$N_7, N_8, N_9$
$S_4$	10	1	$N_{17}$	$N_{19}$	$N_{17}, N_{18}, N_{19}$

Table 1. An Example Set of 4 Streams

## 2 PROBLEM DEFINITION

The problem that we are addressing is formally stated as follows: given a number of periodic streams and a fixed network topology, output a transmission schedule with an estimated end-to-end latency bound of each stream. Applications have a constraint on period for each stream. If the provided latency bound is less than or equal to its period, then our approach offers reliable communication.

We define a Stream Set  $SS$  that contains  $n$  periodic streams, where a periodic stream  $S_i$  has a source node  $SRC_i$ , a destination node  $DEST_i$ , a starting time  $ST_i$ , a period  $P_i$ , and a route  $RT_i$ (optional), where  $i = 1, 2, 3, \dots, n$  and a stream is represented as a 6-tuple  $(S_i, SRC_i, DEST_i, ST_i, P_i, RT_i)$ . After the starting time is elapsed, at every period, the source node has a packet that has to be transmitted to the destination node by following that route. The route  $RT_i$  can be either fixed or assigned by our algorithms.

Network topology is specified as a set of nodes  $N_1, N_2, N_3, \dots, N_k$  along with their connectivity matrix and interference matrix. For each pair of connected nodes  $N_i$  and  $N_j$ , we denote the intermediate link as  $L(i, j)$  that has burstiness parameters  $B_{\max}$  and  $B'_{\min}$  which are estimated based on empirical data. We will define these parameters formally in Section 4.

For example, consider the network topology shown in Figure 1. If two nodes are within their radio range, they are connected by an edge. Suppose that we are given 4 streams to calculate their latency bounds. The details of the streams are shown in Table 1. The route means stream  $S_1$  is going from node  $N_1$  to  $N_4$  using route  $N_1 \rightarrow N_2 \rightarrow N_3 \rightarrow N_4$ . So, we have

$$SS = \{(S_1, N_1, N_4, 1, 20, RT_1), (S_2, N_2, N_5, 1, 20, RT_2), \\ (S_3, N_7, N_9, 1, 20, RT_3), (S_4, N_{17}, N_{19}, 1, 10, RT_4)\}, \quad (1)$$

where  $RT_1 = \{N_1, N_2, N_3, N_4\}$ ,  $RT_2 = \{N_2, N_3, N_4, N_5\}$ ,  $RT_3 = \{N_7, N_8, N_9\}$ ,  $RT_4 = \{N_{17}, N_{18}, N_{19}\}$

Given this input set, our algorithm schedules all the streams and outputs latency bound  $LB_i$  for each stream  $S_i$ . We claim that for any integer  $k$ , the  $k$ -th packet of stream  $S_i$  will reach its destination no later than  $ST_i + k \cdot LB_i$  if packet transmission is performed according to our schedule.

## 3 RELATED WORK

The challenge of providing real-time and reliable transmission in cyber-physical system has been addressed in [18] and researched in recent years [8, 25, 31, 45, 46]. The general framework includes link quality estimation [5, 22, 36] and routing/ scheduling algorithm design for real-time streams [35, 38, 44]. Particularly, the quality of wireless links fluctuates due to link burstiness which has been closely investigated by recent studies [1, 2, 32, 40]. The critical aspect in these works is to accurately capture the characteristic link burstiness and design algorithms to guarantee QoS.

### 3.1 Link burstiness

The effort of dealing or coping with link burstiness can be divided into offline approaches, which focus on the long-term characteristic of link quality, and online approaches, which aims to detect short-term link quality fluctuation. For offline approaches, K. Srinivasan et al. [36] present the  $\beta$  metric to measure link burstiness.  $\beta$  is calculated by using a conditional probability packet delivery function (CPDF). By giving a  $n$  previous packets delivered trace, the function determines the probability of successfully delivering the next packet. His empirical study shows that link burstiness is avoidable by having an inter-packet delay of at most 500 ms. Yet, it does not specify the inter-packet delays for links with different characteristics of link burstiness. Moreover, as M. Radi et al. [32] observe, measuring  $\beta$  requires a large amount of data to achieve a 95% confidence interval.

J. Wen et al. [40] propose an offline scheme that estimates the expected reliable transmission periods by using the conditional probability distribution function of the signal-to-noise ratio (SNR) of received acknowledgment packets. Their approach can detect link burstiness when the SNR of received packets is below a threshold. However, empirical studies show that the estimated burst periods might be over- or under-estimated since the SNR varies for most intermediate links.

In advance, Z. Ansar et al. [1] propose a transmission scheme which applies a two-stage Markov model to characterize link quality fluctuation. They cluster link quality into  $k$  different levels based on the values of Acknowledgement Reception Ratio (ARR) and SNR from a large amount of packet transmission traces. After clustering, the transition probabilities between the levels is computable by using a first-order Markov chain. Moreover, the expected number of successfully received packets at each level can be acquired by using another first-order Markov chain. Their experiments show that their approach improved the packet delivery ratio of the links by up to 40%.

For the online approaches, Brown introduces BurstProbe [5], a mechanism to measure link burstiness ( $B_{\max}$  and  $B'_{\min}$ ) online. The mechanism has an additional probing slot in the transmission schedule such that each node can probe and estimate link burstiness online and share the information among neighbor nodes. This approach is more reactive for capturing short-term burstiness but requires additional time slots to measure link quality. Ansar [2] proposes another online transmission scheme extended from his previous offline approach [1]. His scheme is two-fold. During the offline phase, the link quality has been categorized into 3 different stages (good, intermediate, bad) by applying a two-stage Markov model. In the online phase, the transmission scheme takes the short-term statistics of received acknowledgment packets to predict the most probable future state and the associated burst length. They implement the scheme on the TinyOS using the TelosB platform, and their experiments show that their scheme produces a higher throughput compared with the scheme  $\beta$ . However, it also shows that their scheme exhibits 4 to 10 percent higher packet losses than the  $\beta$ .

The methods of capturing the characteristics of link burstiness in these offline/ online approaches are all based on the probabilistic functions. Different from their works, the  $B_{\max}$  metric is focused on the characteristics of link burstiness in the worst case scenario. It aims to provide end-to-end latency bound on hard real-time applications.

### 3.2 Routing/ scheduling algorithm

On the other side, there are a number of scheduling algorithms of stream transmission in CPS. Here, we only provide some representative works and one can refer [24] for a comprehensive survey. In early stage, SPEED [14] maintains a desired delivery speed across the network by a combination of feedback control and non-deterministic geographic forwarding. It is designed for soft real time applications and it is not concerned with the reliability issues of individual streams.

For recent works, RDDS [15] provides a semantics-aware communication mechanism to guarantee QoS which mainly focus on the predictive sensor models. A multicast routing is provided in [20] which can control the data flow in real-time CPS but it has no reliability guarantee. C.T. Sony et al. [35] provide a method to cluster hops and give a TDMA schedule with a reliable transmission for each stream. However, there is no reliability guarantee on these researches and it is only suitable in LEACH (Lower Energy Adaptive Clustering Hierarchy) protocol [27, 43]. Without the requirement of specific protocols, [44] proposes a scheduling algorithm for reliable packet delivery with end-to-end delay constraint in the data-link layer. However, this work requires two steps: scheduling and extending, which makes it hardly scalable. In advance, [8] provides SchedEx, a low-complexity generic extension for existing slot-based scheduling algorithms which improves the calculation time of re-scheduling and make scheduling algorithms scalable.

H.T. Yang et al. [13] propose a cross-layer (MAC and network layer) scheduling algorithm to provide reliability guarantees for multi-stream. In their scenario, each stream has different reliability requirement and they aim to schedule streams such that each stream can meet its requirement while minimizing end-to-end latency. They propose a routing algorithm that does not take the internal interference into account. Compared with their work, we present our complete algorithm for scheduling multi-stream which provides end-to-end latency bounds. We also implement the system with real-world experiments.

## 4 EMPIRICAL STUDY

To see how burstiness affect packet transmission, we conducted a 21-day experiment on a testbed with 48 Tmote nodes running TinyOS. For each pair of two nodes, a total of 3,600,000 packets is transmitted to evaluate packet delivery traces. These nodes were deployed on the walls and ceilings of a building. The transmission rate was approximately 200 packets per second for all links. The transmissions of different links were scheduled at different time slots to avoid a collision. The receiving status of each packet at every node is recorded as a sequence of 0 and 1. These sequences are used for calculating  $B_{\max}$  and further analysis in the following subsections.

In Section 4.1, we show how to characterize link bursts with parameters  $B_{\max}$  and  $B'_{\min}$ . In Section 4.2, we classify links based on the value of  $B_{\max}$  and discuss the robustness of  $B_{\max}$ . We confirmed that the  $B_{\max}$  value reflects the maximum number of retransmission times in a link with more than 99% correctness while the link is affected by the burstiness. This suggests  $B_{\max}$  as a robust signature to characterize link bursts.

### 4.1 Model Parameters and Assumptions

To characterize link bursts we define five parameters:  $B$ ,  $B_{\max}$ ,  $B'$ ,  $B'_{\min}$ , and  $W$  for every link. We define  $W$  as a window of outcomes of packet transmission. As an example, when  $|W|$  is 3, we consider a sliding window of a size 3 over the outcomes of packet transmission. We define  $B$  as the number of time slots where packet transmission failed due to link bursts for a window  $W$ . We define  $B'$  as the number of time slots where packet transmission was successful for a window  $W$ . We define  $B_{\max}$  as the maximum value of  $B$  for all possible windows of size  $|W|$  and  $B'_{\min}$  as the minimum value of  $B'$  for all possible windows of size  $|W|$ .

The way these parameters are computed is as follows: after transmitting 3,600,000 packets over every link, we have a long sequences of data trace of 0s and 1s per link where 1 at  $i$ th index of the sequence means that packet with  $i$ th sequence number was successfully transmitted and 0 at that place means it failed. As an example, consider a sample data trace 0110010011 of length 10. We would like to estimate the  $B_{\max}$  of this link. To do that, we start with choosing  $B'_{\min}$ . Let's assume, we choose  $B'_{\min} = 1$ . We start with a window slightly bigger than  $B'_{\min}$ , which is  $|W| = 2$ . Now, for every window of size 2, we check if there is at least one good slot within the window since  $B'_{\min} = 1$ . Here, we have 9 different windows of size 2. The first window spans from index 1 to 2, the second one spans from 2 to 3, and so on. In the first window containing 01, where we have  $B = 1$ ,  $B' = 1$ . In the second window containing 11, we have  $B = 0$ ,  $B' = 2$ . In the third window containing 10, we have  $B = 1$ ,  $B' = 1$ . However, in the fourth window containing 00, we do not have a good slot as  $B = 2$ ,  $B' = 0$ . So, we increase the window to 3. Now we have 8 different windows to consider, each of length 3. We have  $B = 1$ ,  $B' = 2$  for the first window (011),  $B = 1$ ,  $B' = 2$  for the second window(110),  $B = 2$ ,  $B' = 1$  for the third window(100) and so on. At the end, we get  $B_{\max} = 2$  for this link for  $B'_{\min} = 1$ . Note that it means that there is at least one good slot for every window of size 3. In the rest of the paper, we use the terms  $W$ ,  $B_{\max}$ ,  $B'_{\min}$  when we discuss the general burstiness influence and link quality and  $W(i, j)$ ,  $B_{\max}(i, j)$ ,  $B'_{\min}(i, j)$  when we want to emphasize its calculated values on link  $L(i, j)$ .

Our proposed solution uses the same  $B'_{\min}$  for all the links. Choosing a higher  $B'_{\min}$  of a link can reduce latency bounds of the streams that use this link due to spatio-temporal overlapping as mentioned in Section 5.3. However, choosing a higher  $B'_{\min}$  of a link when such spatio-temporal overlapping does not exist can hurt latency bounds of the streams that use the link. We leave the selection of optimal  $B'_{\min}$  of each link for minimizing latency bound of each stream as future work. More details on the selection of  $B'_{\min}$  is described in Section 8.2.

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**Algorithm 1** : ComputeBmax( $D, B'_{\min}$ )
 

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1: for  $i \leftarrow B'_{\min} + 1$  to  $|D|$  do
2:    $isSatisfied \leftarrow TRUE$ 
3:   for  $j \leftarrow 1$  to  $|D| - i$  do
4:      $B' \leftarrow \sum_{k=j}^{j+i-1} D[k]$ 
5:     if  $B' < B'_{\min}$  then
6:        $isSatisfied \leftarrow FALSE$ 
7:       break
8:     end if
9:   end for
10:  if  $isSatisfied = TRUE$  then
11:     $B_{\max} \leftarrow i - B'_{\min}$ 
12:    return  $B_{\max}$ 
13:  end if
14: end for
15: return  $-1$ 

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Algorithm 1 computes  $B_{\max}$  given a data trace  $D$  of 0s, 1s and  $B'_{\min}$ . It returns -1 if the data trace does not have a  $B_{\max}$  that satisfies the condition specified by the arbitrary value of  $B'_{\min}$ . The running time of the algorithm is  $O(|D|^2)$  which is large for long data trace, although  $B'$  at line 4 can be computed in  $O(1)$  time by using a dynamic programming based memorization approach. But if a link has a very high  $B_{\max}$ , it indicates that the link is prone to heavy bursts, and we avoid this link for real-time applications. So, for practical purposes, we limit the maximum  $B_{\max}$  to be  $C = 1200$  and constrain the loop at line 1 to run from 1 to  $C$ . Then, the running time of the algorithm becomes  $O(C|D|)$  which is linear with respect to the size of the data trace.

Our model assumes that after network characterization, i.e., after we compute  $B_{\max}(i, j)$  and  $B'_{\min}(i, j)$  of every link  $L(i, j)$ , if we consider  $B_{\max}(i, j) + B'_{\min}(i, j)$  time slots for packet transmission, we have at least  $B'_{\min}(i, j)$  time slots to transmit packet successfully. Although the assumption looks questionable, we have some strong arguments in favor of it. The assumption may not hold in a battlefield where link behavior may change drastically, but it seems to hold in a regular working environment like offices, universities, and industrial plants if we can characterize the links for a long period of time under all possible working environments. Obviously, this assumption will not hold for all the wireless links. We classify the links (in the next section) for which the assumption seems to be true. If the wireless links are not good enough to meet the assumption, we can move some nodes or add additional nodes in the network to create the “right” topology having good burst properties that satisfy the model assumption. Also, another work [33] has explored that bursts in the wireless link have scaling properties meaning that the bursts show self-similarity or other coherent structure over many time scales without having long-range dependence. The paper specifies an onset point of 640ms where random variations stop affecting the wireless link

and self-similarity starts to dominate. It clearly indicates the possibility of capturing the burstiness characteristics of the wireless links if we investigate the burstiness behavior of the wireless links over a long period of time.

## 4.2 Robustness of $B_{\max}$

Since the whole characteristics of link burstiness may not be observable in a short period of time, one realistic solution is to collect packet transmission traces in a period to calculate  $B_{\max}$  and only use links with robust  $B_{\max}$  value that can stand for a long time. In this subsection, we discuss the robustness of  $B_{\max}$  among links under two factors: the  $B_{\max}$  value and burstiness frequency. The first one would influence the robustness of  $B_{\max}$  since a link with low  $B_{\max}$  value may imply that the burstiness event with the longest length is not observed yet in the collected trace. Burstiness frequency would also impact the robustness of  $B_{\max}$  since a robust  $B_{\max}$  value might not be derivable with only an observation from few burstiness events. To explore the relationship of  $B_{\max}$  robustness with these two factors, we divided our collected traces into measuring set and testing set. In the measuring set, we calculated the  $B_{\max}$  value and burstiness frequency for each link and classified them into different categories based on the values of these two factors. Then, in the testing set, we compared the changes of  $B_{\max}$  values among different classes by observing the exceeded rate and exceeded  $B_{\max}$  lengths. The exceeded rate is defined as the ratio of the burstiness events in the testing set whose lengths are longer than the calculated  $B_{\max}$  value from the measuring set. The measuring and testing set is from the 21 days long traces which we mentioned earlier. We used our first 7 days of collected traces as measuring set and the rest of 14 days as the testing set.

For demonstration, we set  $B'_{\min} = 1$  for all links and we only used links whose PRRs are higher than 90%. It is because there are  $48 \times 47$  links in the traces and some of them have very low reception rates due to long transmission range or across multiple obstacles.

**4.2.1 Burstiness frequency v.s.  $B_{\max}$ .** The links are classified by the following methods. Firstly, we calculated  $B_{\max}$  and burstiness frequency for each link in the measuring set. We defined a burstiness event as a number of consecutive packets lost in the data trace. The burstiness frequency for a link is the average number of the burstiness events per hour in its data trace of the measuring set that is collected over the period of 7 days. Then, we divide links into 4 classes based on its  $B_{\max}$  values and burstiness frequency. We define links whose  $B_{\max}$  value is in the top 50% of all links are high  $B_{\max}$  (HB), low  $B_{\max}$  (LB) otherwise. Links whose burstiness frequency is in the top 50% are high burstiness frequency (HF), low burstiness frequency (LF) otherwise. Thus, there are total 4 different classes and the distribution is shown in Figure 2.

HFLB : High burstiness frequency and low  $B_{\max}$

HFHB : High burstiness frequency and high  $B_{\max}$

LFLB : Low burstiness frequency and low  $B_{\max}$

LFHB : Low burstiness frequency and high  $B_{\max}$

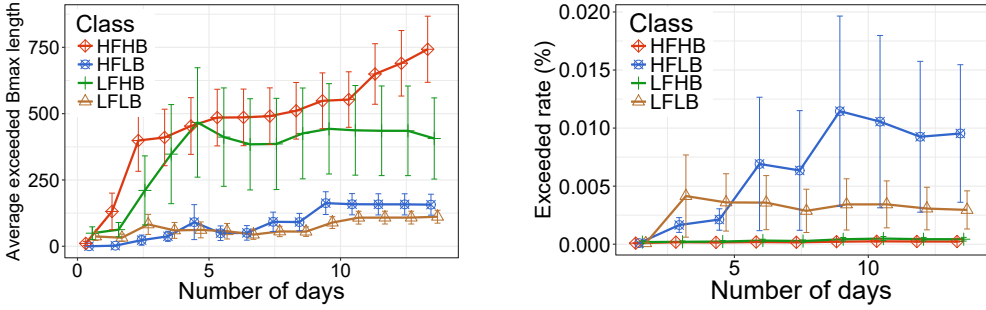
HFLB	HFHB	LFLB	LFHB
12.2%	37.8%	38%	12%

Fig. 2. Link classes distribution

This table suggests that most of the links are either in HFHB or LFLB class, which is reasonable from the testbed of our experiment. For example, most of HFHB links are near the stairs or hallways, where people's movements are frequent and varied. On the other hand, many LFLB links connect nodes in the same office room or storage, which is relatively less interfered.

**4.2.2 Exceeded values and times of  $B_{\max}$ .** To address  $B_{\max}$  robustness, we firstly focused on the exceeded  $B_{\max}$  values among links in the testing set. We aimed to find links with low exceeded value because it means the calculated  $B_{\max}$  of these links are still robust in the testing set. This





(a) Average exceeded  $B_{\max}$  vs. the number of days. (b) Exceeded rate v.s. the number of days. Exceeded rate is defined as the ratio of the burstiness lengths which are longer than the calculated  $B_{\max}$  value.

Fig. 3. The exceeded lengths and times of  $B_{\max}$  value among links from different classes in the testing set. The  $B_{\max}$  values are calculated from the measuring dataset.

result is shown in Figure 3a. The error bars are the standard error bounds for each day in the testing set. In the figure, HFHB and LFHB suffer high exceeded values rather than HFHB and HFLB. It suggests that the calculated  $B_{\max}$  values on links in HFHB and LFHB are unstable and cannot stand for long periods.

Our second interest is the exceeded rate of calculated  $B_{\max}$  value in the testing set, which is measured by the ratio of the burstiness events with the lengths more than the calculated  $B_{\max}$  value. This result is shown in Figure 3b. Different from the previous figure, links with low  $B_{\max}$  value have higher chances that the burstiness lengths would exceed the  $B_{\max}$  value. However, by referring the result of Figure 3a, since the exceeded  $B_{\max}$  values under these links are small, the exceeded rate of these links could possibly decrease in reality by setting the latency bound slightly higher than the calculated  $B_{\max}$  value. In addition, we would like to emphasize that the exceeded rate of  $B_{\max}$  value among all links is pretty low (at most 0.020%) according to Figure 3b, which suggests that the  $B_{\max}$  metric is a robust signature for real-time applications.

## 5 ALGORITHM

In this section, we show how to use  $B_{\max}$  information to design a schedule with minimum latency bound for each stream. We divide our algorithm into routing and scheduling parts. In Section 5.1 we explain how to choose routes with minimum latency bound for either unicast or broadcast transmission. In Section 5.2, we show how to deal with end-to-end latency in the presence of burstiness for a single stream. In Section 5.3, we then expand the schedule of a single stream to multiple streams where both burstiness and internal interference must be handled. In Section 5.5, we present our complete algorithm for scheduling multi-stream and provides end-to-end latency bounds.

### 5.1 Unicast/ broadcast Routing with $B_{\max}$

From Section 2, we assume every stream is assigned to a single route before scheduling. However, this might not be true for different systems. Also, a stream might derive high latency if the selected route for this stream goes through some links with high  $B_{\max}$  values. To avoid these issues, in this subsection we provide routing algorithms for both unicast and broadcast scenarios. In the unicast scenario, there are multiple source-destination pairs in the network and the algorithm is aim to find a route for each pair such that the maximum latency bound is minimized. In the broadcast scenario, there is exactly one source that wants to broadcast the packets to all other nodes. The objective is

to minimize the maximum latency bound among all other nodes. The general idea of the routing algorithms is to refer  $B_{\max}$  information and select a route with links having the lowest  $B_{\max}$  values. In practice, these algorithms can be implemented in Network layer to provide routing decisions for multiple streams. Considering a weighted graph  $G = (V, E)$  which represents a sensor network, where sensors are represented by nodes and each edge represents a link between sensors. The weight of an edge  $(i, j) \in E$  is  $B_{\max}(i, j)$ , where  $i, j \in V$ . For the unicast scenario, we simply calculate the shortest path from source to destination in respect of the  $B_{\max}(i, j)$  value at each edge  $(i, j)$ . This path ensures the maximum retransmission times, i.e. latency bound, is minimal. In addition, if there are multiple unicast streams in a network, how to choose routes to achieve minimum latency bounds among all streams is considered as a NP-hard problem [45]. To provide an online routing algorithm for multi-stream, here we applied the load-balancing algorithm from J. ASPNES [3], which is to minimize the maximum traffic load while satisfying all the requests. The main idea is to introduce an exponential weight function on each edge and run the shortest path algorithm. In our case, for every assigned route, each edge  $e$  of the route would add an exponential cost. That is,  $w'(e) \leftarrow w(e) + a^{B_{\max}(e)}$  for some integer  $a$ , where  $B_{\max}(e)$  is the  $B_{\max}$  value of edge  $e$ ,  $w(e)$  is the current weight, and  $w'(e)$  is the updated weight of edge  $e$ . Then, the algorithm would schedule the route for next stream based on the updated weight in  $G$ . In this way, our unicast routing algorithm can provide  $O(\log n)$  approximation factor to achieve minimum latency bound for multi-stream.

For the routing in the broadcast scenario, there is only one source in the network. In addition, the end-to-end latency for a job is the time that all of the nodes in the graph receive the broadcast packet. Since each node only needs to receive the packet once, the broadcast route can be represented by a tree structure. The key idea to achieve minimum latency bound is to compute a minimum spanning tree in respect to the  $B_{\max}$  value. In this research we modify BIP [41], one of the minimum energy broadcast (MEB) routing algorithms, to achieve the minimum latency bound for a job. The main idea of BIP is to take the advantage of omnidirectional transmission to achieve broadcast mission with minimum energy. We notice that this advantage is also suitable to calculate the latency bound. We use Figure 4 as an example to explain this in detail. In this figure, there is a graph  $G$  with nodes  $N_1, N_2, N_3$ , links  $e_1, e_2, e_3$ , and the weight function  $w$ . Assume  $N_1$  is the source and it wants to broadcast a packet to  $N_2$  and  $N_3$ . It can either send a packet to  $N_2$  first and ask  $N_2$  to forward the packet to  $N_3$  or send the packet simultaneously to  $N_2$  and  $N_3$ . Now, suppose all links are suffering the burstiness at the same time, the first way would take at most  $w(e_1) + w(e_3)$  number of timeslots to broadcast the packet but the second way would only cost at most  $\max\{w(e_1), w(e_2)\}$  number of timeslots because of the omnidirectional transmission. Thus, the minimum end-to-end latency bound in this case is the minimum of  $\{w(e_1) + w(e_3), \max\{w(e_1), w(e_2)\}\}$ . Here, we provide a broadcast algorithm which takes advantage of omnidirectional transmission.

Our broadcast algorithm can be simplified into 2 steps: *select* and *add*. The basic idea is to iterate these 2 steps until the spanning tree is constructed. Starting from the source node, in *select* step, it selects an edge  $(i, j)$  with the minimum weight, where  $i$  is one of the nodes in the tree and  $j$  is one of the nodes not in the tree. In *add* step, our algorithm adds the selected link  $(i, j)$  into the tree and the weight of all links whose head is  $i$  would decrease by the weight of  $(i, j)$  due to the advantage of the omnidirectional transmission. Thus, by iterating these two steps, the algorithm would construct a spanning tree for broadcasting.

The time complexity of both algorithms is under  $O(|V|^2)$ . For the unicast algorithm, the running time of each stream is the time of finding the shortest path, which is  $O(|V|^2)$  by Dijkstra's algorithm, and the time of updating the weights, which is  $O(|V|)$ . For the broadcast algorithm, the running time is  $O(|V|d)$ , where  $d$  is the maximum degree in the graph.

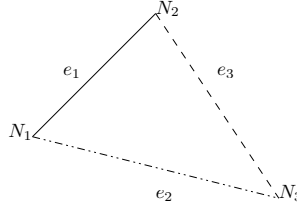


Fig. 4. The advantage of the omnidirectional transmission to provide lower latency bound.

## 5.2 Scheduling with a Single Stream

If there is only one stream in the network and there is no packet loss, then the end-to-end latency bound is the sum of per link latencies in all the intermediate links. This is the theoretical lower bound of end-to-end latency for a single stream. But in reality, links are not ideal and packet losses occur in a burst. If a stream has  $n$  intermediate links from source to destination and  $k$ -th intermediate link  $L(i_k, j_k)$  has burstiness parameters  $B_{\max} = b_k$  and  $B'_{\min} = b'_k$  ( $k = 1, 2, 3, \dots, n$ ), then on  $k$ -th link we have to allocate  $b_k + 1$  time slots for the stream. So the end-to-end latency

$$\text{bound } LB \text{ is } LB = \sum_{k=1}^n (b_k + 1).$$

Consider the topology in Figure 1. Assume that the links have  $B_{\max}$  and  $B'_{\min}$  as shown in Table 2. Now, assume that our SS consists of a single stream  $S_1$  from Table 1 having period 20, starting time slot 1 and it goes from  $N_1$  to  $N_4$  using route  $N_1 \rightarrow N_2 \rightarrow N_3 \rightarrow N_4$ . So,  $SS = \{(S_1, N_1, N_4, RT_1, 1, 20)\}$  where  $RT_1 = \{N_1, N_2, N_3, N_4\}$ .

Since,  $B_{\max}(1, 2)$  is 2 for  $L(1, 2)$ , if we allocate  $B_{\max}(1, 2) + 1$ , i.e., 3 time slots for  $S_1$  at node  $N_1$ , then the packet will reach to  $N_2$  even if there is a burst. Similarly, if we allocate 4 time slots at  $N_2$  to transmit over  $L(2, 3)$  and 4 time slots at  $N_3$  to transmit over  $L(3, 4)$  then it takes only  $3 + 4 + 4 = 11$  time slots to ensure that every packet of  $S_1$  will be delivered to its destination within that time even if there is a burst in a number of links. Hence,  $LB_1 = 11$ . The corresponding scheduling table is shown in Table 3 below where the column represents time slots and the row represents links. It shows which stream will be transmitted at which time slot using which link. For example,  $S_1$  will be transmitted at time slots 1, 2 and 3 using link  $L(1, 2)$ .

Link	$B_{\max}$	$B'_{\min}$
$L(1, 2)$	2	2
$L(2, 3)$	3	2
$L(3, 4)$	3	3
$L(4, 5)$	3	2
$L(7, 8)$	2	2
$L(17, 18)$	2	3
$L(18, 19)$	1	4

Table 2.  $B_{\max}$  and  $B'_{\min}$  of Links

Link/TimeSlot	1	2	3	4	5	6	7	8	9	10	11
$L(1, 2)$	$S_1$	$S_1$	$S_1$								
$L(2, 3)$				$S_1$	$S_1$	$S_1$	$S_1$				
$L(3, 4)$								$S_1$	$S_1$	$S_1$	$S_1$

Table 3. Scheduling Table with a Single Stream

Note that multiple time slots are reserved to ensure reliability. It does not mean that  $N_1$  is transmitting three packets of  $S_1$  in time slots 1, 2 and 3. Rather it means that  $N_1$  will try in these time slots to transmit a packet of  $S_1$  and it will stop as soon as the packet gets into the next hop node, i.e.,  $N_2$ . We assume that the radio transceiver supports hardware/software acknowledgment

so that the receiver can acknowledge its packet reception immediately to the sender. This is a reasonable assumption based on the radio transceivers available in the market.

Notice that to provide the latency bound, each transmission on each link is required to allocate  $B_{\max} + 1$  time slots, which has a potential drawback since the burstiness may not happen in every transmission and there would be a waste of additional allocated time slots. However, the routing algorithm prone to select links with low  $B_{\max}$  value such that the waste of time slots is minimal. In general, it is a dilemma to achieve high reliability and low latency for the scheduling algorithm.

### 5.3 Scheduling with Multiple Streams

When we have only one stream in the network, even in the presence of burstiness, the scheduling is not complex since the interference is not an issue. No two links can interfere with the transmission of each other in this scenario. But when multiple streams co-exist in the network, we have to take interference into account and ensure that no two interfering links transmit packets at the same time slot causing a packet loss. Note that finding the minimum average latency bound of all the streams by scheduling packet transmission considering all possible routes for every stream subject to link interference and link burstiness is an NP-Hard problem since it is analogous to the bin packing problem [6]. Hence we use a greedy solution based on the following principles:

- (1) Schedule packet transmission up to the least common multiple(LCM) of the periods of all the streams. If the estimated end-to-end latency bounds of all streams are lower or equal to their periods, we conclude that the stream set is schedulable.
- (2) Address packet loss due to link interference as follows: First, figure out the interference pattern of the network, represented by  $IM$  empirically. Then, schedule packet transmission in a way to make sure that no two interfering links ‘transmit’ packets at the same time slot.
- (3) Address packet loss due to link burst as follows:
  - (a) Allocate  $B_{\max} + 1$  contiguous time slots for packet transmission over a link. Note that different links have different  $B_{\max}$ . We are assuming that the route is *least-burst-route* as the route is the shortest path having the minimum sum of  $B_{\max}$  from the source node to the destination node.
  - (b) While allocating  $B_{\max} + 1$  time slots, overlap at most  $B'_{\min}$  streams’ time slot allocation. The reason is, within a window of  $B_{\max} + B'_{\min}$ , there are at least  $B'_{\min}$  good slots for packet transmission, and so, we should be able to transmit at least  $B'_{\min}$  number of streams. Hence, we allow slots of at most  $B'_{\min}$  streams to overlap. There are two conditions for it: (i). This overlapping is allowed only when packets are being transmitted over the same link. Note that we can not overlap time slots for neighboring nodes due to link interference. (ii). While overlapping time slots of multiple streams, no two streams are allowed to be allocated the same  $B_{\max} + 1$  time slots, i.e., complete overlapping is not allowed. The reason is, if two streams  $S_1, S_2$  are allocated the same  $B_{\max} + 1$  time slots, and if we lose  $B_{\max}$  time slots due to a burst, then we can not transmit packets of both  $S_1$  and  $S_2$  in the remaining time slot.

Link/TimeSlot	1	2	3	4	5	6	7	8
$L(1, 2)$	$S_1$	$S_1$	$S_1$	$S_1$				
$L(1, 2)$					$S_2$	$S_2$	$S_2$	$S_2$

Table 4. Schedule with non-overlapping

Link/TimeSlot	1	2	3	4	5
$L(1, 2)$	$S_1$	$S_1, S_2$	$S_1, S_2$	$S_1, S_2$	$S_2$

Table 5. Schedule with overlapping

To explain why overlapping is important and how it is done, consider the following example. Assume that Stream Set  $SS$  consists of two streams  $S_1$ , and  $S_2$ , both going from node  $N_1$  to  $N_2$  using route  $N_1 \rightarrow N_2$  through the link  $L(1,2)$ . Assume that both streams have period 20 and starting time slot 1. Also assume that  $B_{\max}(1, 2) = 3$  and  $B'_{\min}(1, 2) = 2$  for the link  $L(1,2)$ . A schedule without

overlap is shown at Table 4 where average latency bound per stream is  $(4 + 8)/2 = 6$  time slots. Compare it with a schedule with overlapping in Table 5 where average latency bound per stream is  $(4 + 5)/2 = 4.5$  time slots. Note that we could not do complete overlapping as in Table 6. Because, as it is mentioned earlier, if we lose  $B_{\max} = 3$  time slots due to a burst, then we can not transmit packets of both  $S_1$  and  $S_2$  in the remaining time slot.

Link/TimeSlot	1	2	3	4
$L(1, 2)$	$S_1, S_2$	$S_1, S_2$	$S_1, S_2$	$S_1, S_2$

Table 6. Schedule with Complete Overlapping

Link/TimeSlot	1	2	3	4	5	6
$L(1, 2)$	$S_1$	$S_1, S_2$	$S_1, S_2, S_3$	$S_2, S_3, S_4$	$S_3, S_4$	$S_4$

Table 7. Schedule with Maximum Overlapping

Overlapping time slots raises another issue. Since we can transmit only one packet at a time and the packet transmission process is completely deterministic, we have to prioritize the streams to be transmitted in the overlapped time slots. Our *prioritizing rule* works as follows: if multiple streams are scheduled to be transmitted at the same time slot, transmit the not-yet-transmitted stream that has the closest ending time slot. We will see its use in the next example.

To illustrate how many streams can be overlapped consider the following example. Assume that Stream Set SS consists of four streams  $S_1, S_2, S_3$ , and  $S_4$  all going from node  $N_1$  to  $N_2$  using route  $N_1 \rightarrow N_2$  through the link  $L(1,2)$ . Assume that all the streams have period 20 and starting time slot 1. Also assume that  $B_{\max}(1, 2) = 2$  and  $B'_{\min}(1, 2) = 4$  for the link  $L(1,2)$ . Table 7 demonstrates a schedule that shows within a window of size  $B_{\max}(1, 2) + B'_{\min}(1, 2) = 2 + 4 = 6$ , we can overlap at most  $B'_{\min}(1, 2) = 4$  streams. This schedule will always work if nodes transmit packets according to the prioritizing rule. For example, if packet transmission fails during time slots 1, and 2, packets of streams  $S_1, S_2, S_3$ , and  $S_4$  will be transmitted at time slots 3,4,5 and 6, respectively. If packet transmission fails at time slots 2, and 4 then packets of streams  $S_1, S_2, S_3$ , and  $S_4$  will be transmitted at time slots 1,3,5 and 6. So, even if packet transmission of any combination of size  $B_{\max}(1, 2)$  fails within a window of  $B_{\max}(1, 2) + B'_{\min}(1, 2)$ , all the packets of all the streams will get through to the next node if we allocate time slots according to the principles mentioned earlier and nodes transmit packets according to the prioritizing rule as it is proved in the next subsection.

#### 5.4 Correctness Proof

The correctness of our algorithm depends on the following theorem:

**Theorem:** *If we overlap packet transmissions of at most  $b'$  streams in  $(b + b')$  time slots, all going through the same link having  $B_{\max} = b$ ,  $B'_{\min} = b'$ , each stream having  $(b + 1)$  contiguous time slots allocated without complete overlapping with any other stream, then all of the streams will get through even if there is a burst of at most  $b$  time slots (not necessarily contiguous), if we transmit packets according to our prioritizing scheme.*

**Proof:** The proof is by contradiction. For a contradiction assume that stream  $S_i$  could not be transmitted due to a burst. Since we can lose at most  $b$  time slots due to a burst, there has to be a time slot, we call "good slot", when there was no burst, but still  $S_i$  was not transmitted. The reason for that can only be some other stream  $S_j$  was transmitted at that time slot. Note that the good slot can not be the last sending time slot of  $S_i$ . Because, at that time slot, stream  $S_i$  has the highest priority for transmitting a packet. So, no other stream  $S_j$  can be transmitted at that time slot. So,

now we have to consider one remaining possibility. There was a burst at the last sending time slot of  $S_i$  and among the previous  $b$  slots of  $S_i$ ,  $(b - 1)$  slots are gone due to a burst leaving one good slot and at that time slot some other stream  $S_j$  was transmitted for the priority scheme. Note that this can not happen either, because as  $S_j$  has higher priority than  $S_i$ ,  $S_j$  has started time slots before  $S_i$  did, and hence it should be transmitted even before the burst happens. Because we are overlapping at most  $b'$  streams in any time window of  $(b + b')$  slots and there has to be at most  $b'$  good slots between any consecutive bursts. So,  $S_j$  will be transmitted in the good time slots of the previous window and  $S_i$  will be transmitted within this time window. So, there is no way  $S_i$  will not be transmitted due to a burst.

### 5.5 Scheduling algorithm

Algorithm 2 schedules packet transmission up to the LCM of the periods of streams in a central sensor network based on the principles and conditions stated in Section 5.2. It takes the Stream Set  $SS$  of  $n$  streams (as defined in section 2), topology, Interference Matrix  $IM$ , burstiness parameters  $B_{\max}, B'_{\min}$  of every link as inputs and returns either *true* if all the streams are schedulable or *false* if they are not. If all the streams are schedulable, then  $LB_i$  holds the latency bound of stream  $S_i$ .

Let  $L_i$  be the last allocated time slot for  $S_i$  in the scheduling algorithm. So, initially we have  $L_i = (ST_i - 1)$  (at line 4) for every stream  $S_i$ . We need to adjust the starting time  $ST_i$  of stream  $S_i$  when its period  $P_i$  is over (at line 40) to correctly compute the latency bound  $LB_i$  (at line 29). Let,  $N_i$  be the node that has received the last transmitted packet of  $S_i$ . So, initially  $N_i = SRC_i$  (at line 5) which is the source node of stream  $S_i$ . Assume that  $N_{i+1}$  is the next node to which  $N_i$  has to transmit a packet of  $S_i$ . Note that  $N_{i+1}$  can be easily determined from the route of  $S_i$ .

We put *DeferThreshold* = 2 (used at line 20). If you use a higher value for it, the scheduler runs faster, but the generated latency bound may also rise. We are deferring time slot allocation to only those streams that can not take advantage of overlapping time slots with other streams. The reason for deferring time slot allocation is to increase parallelism over the link from  $N_i$  to  $N_{i+1}$ . This is a kind of lazy approach for allocating because we are hoping that it is possible that some other streams may show up and can be allocated in these time slots that can exploit parallelism.

Link/TimeSlot	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20
$L(1, 2)$	$S_1$	$S_1$	$S_1$																	
$L(2, 3)$				$S_1$	$S_1, S_2$	$S_1, S_2$	$S_1, S_2$	$S_2$												
$L(3, 4)$									$S_1$	$S_1, S_2$	$S_1, S_2$	$S_1, S_2$	$S_2$							
$L(4, 5)$														$S_2$	$S_2$	$S_2$	$S_2$			
$L(7, 8)$														$S_3$	$S_3$	$S_3$				
$L(8, 9)$																		$S_3$	$S_3$	$S_3$
$L(17, 18)$	$S_4$	$S_4$	$S_4$								$S_4$	$S_4$	$S_4$							
$L(18, 19)$				$S_4$	$S_4$									$S_4$	$S_4$					

Table 8. Scheduling with Multiple Streams

Table 8 shows the whole scheduling for the example problem we are working with from Section 2 and  $B, B'$  of Table 2. Here  $S_1$  and  $S_2$  share time slots at links  $L(2, 3)$  and  $L(3, 4)$ . The latency bound of the streams is shown in Table 9.

Our algorithm exploits parallelism in two ways. The first one is, multiple streams can be transmitted at the same time if there is no interference in their transmission as it is seen in  $S_1$  and  $S_4$  at time slots 1, 2 and 3 in Table 8. The second way is, when two streams are going to the same link,

**Algorithm 2** : Centralscheduler( $SS$ , topology,  $IM$ ,  $B_{\max}$ ,  $B'_{\min}$  of every link) - Part 1

---

```

1:  $n \leftarrow |SS|$ 
2:  $lcm \leftarrow LCM(P_1, P_2, P_3, \dots, P_n)$ 
3: for  $i \leftarrow 1$  to  $n$  do
4:    $L_i \leftarrow ST_i - 1$ ,
5:    $N_i \leftarrow SRC_i$ ,
6:    $LB_i \leftarrow 0$ 
7: end for
8: for  $t_1 \leftarrow 1$  to  $lcm$  do
9:   if All the streams are scheduled then
10:    break
11:   end if
12:   for  $i \leftarrow 1$  to  $n$  do
13:     if  $S_i$  is already scheduled then
14:       continue
15:     end if
16:     if  $t_1 = L_i$  then
17:       for  $t_2 \leftarrow t_1 + 1$  to  $ST_i + P_i$  do
18:          $(b, b') \leftarrow (B_{\max}, B'_{\min})$  of link  $L(N_i, N_{i+1})$ 
19:         if it is possible to allocate  $(b + 1)$  contiguous time slots starting from  $t_2$  by
           following the principles and conditions then
20:           if  $S_i$  is not making any overlap with any streams in these time slots AND
              $(t_2 - t_1) > DeferThreshold$  then
21:              $L_i \leftarrow t_2 - 1$ 
22:             break
23:           else
24:             Allocate these (b+1) time slots for  $S_i$  in  $N_i$ 
25:              $L_i \leftarrow t_2$ ,
26:              $N_i \leftarrow N_{i+1}$ 
27:             if  $N_{i+1}$  is the destination node for  $S_i$  then
28:               Consider that  $S_i$  is scheduled
29:                $LB_i \leftarrow \max(LB_i, t_2 + b + 1 - ST_i)$ 
30:             end if
31:             break
32:           end if
33:         end if
34:       end for
35:     end if

```

---

they can share at most  $B_{\max}$  time slots as it is observed between  $S_1$  and  $S_2$  in time slots 5, 6 and 7 in Table 8.

The running time of the algorithm is  $O(LP)$  where  $L$  is the LCM of the periods of the streams and  $P$  is the sum of the periods of the streams. The reason is, the *for* loop at line 8 takes  $O(L)$  time and the *for* loops at lines 12 and 17 together takes  $O(P)$  time. Note that  $O(LP)$  depends on mainly the periods of the streams, and it can be exponential if the streams have periods that are prime numbers.

---

**Algorithm 3** Centralscheduler( $SS$ , topology,  $IM$ ,  $B_{\max}$ ,  $B'_{\min}$  of every link) - Part 2
 

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```

36:   if  $\text{mod}(t_1, P_i) = 0$  then
37:     if  $S_i$  is not scheduled yet then
38:       return false
39:     else
40:        $ST_i \leftarrow ST_i + P_i$ ,
41:        $N_i \leftarrow SRC_i$ 
42:     end if
43:   end if
44: end for
45: end for
46: if all the streams are not scheduled within the  $lcm$  then
47:   return false
48: end if
49: return true

```

---

Stream	Latency Bound
$S_1$	12 i.e. $\max(12 - 1 + 1, 0)$
$S_2$	17 i.e. $\max(17 - 1 + 1, 0)$
$S_3$	20 i.e. $\max(20 - 1 + 1, 0)$
$S_4$	5 i.e. $\max((5 - 1 + 1), (15 - 11 + 1), 0)$

Table 9. Latency Bound for Each Stream

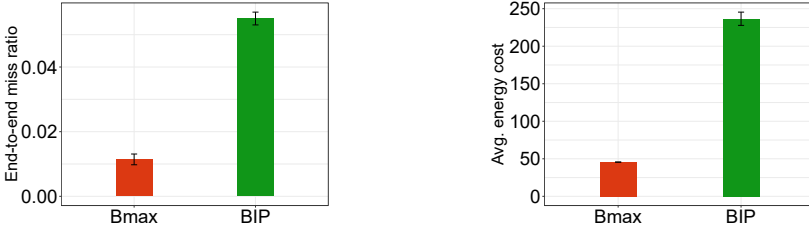
## 6 SIMULATION

In this section, we simulated the routing algorithms for broadcast and unicast scenarios, which we proposed in Section 5.1, and compared the performance with other baseline algorithms. This comparison includes end-to-end deadline miss ratio, energy cost, and the estimated latency bound. For each transmitted packet, there is a deadline, which is the value of the estimated end-to-end latency bound. During the transmission, if a packet is not delivered to its destination node within its latency bound, it is considered to miss its deadline. We use BIP [41] as the baseline for the broadcast scenario and ETX [7] for the unicast scenario. BIP is the algorithm we introduced in Section 5.1 and ETX is basically finding the shortest path in respect to the expected number of transmission counts. The internal interference between nodes is measured by interference matrix and we give a detail explanation in Section 7.2. In addition, all the nodes are using a TDMA-based schedule that has both sleep and awake modes.

The simulation dataset we used is the 21 day long transmission traces we collected in Section 4. We set the first 7 days of the data as the measuring set which is used for measuring  $B_{\max}$  values (with  $B'_{\min}$  selected to 1) and ETX values. The rest of data is used for the testing set which cannot be observed by the algorithms.

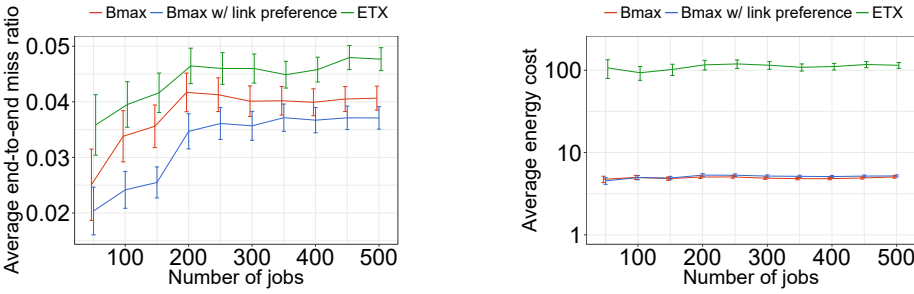
For the broadcast scenario, we set the source as one of the nodes in the testbed and used the routes which are generated from either our algorithm or BIP. Then, we simulated the broadcast transmission 1000 times and used the testing set to determine whether the delivered packet is received at each node. During each transmission, every link is capable of retransmitting the packet multiple times until it meets the  $B_{\max}$  value of the link. If a node cannot receive the packet before





(a) Average ratio of non-received nodes for broad- (b) Average energy cost (retransmission times) for casting a packet in all 47 different sources. broadcasting a packet in all 47 different sources.

Fig. 5. Compare the deadline miss ratio and energy cost under  $B_{max}$  and BIP in broadcast scenario.



(a) Average end-to-end deadline miss ratio per job (b) Average energy cost (retransmission times) per from 50 to 500 jobs. job from 50 to 500 jobs.

Fig. 6. Compare the deadline miss ratio and energy cost under  $B_{max}$ ,  $B_{max}$  with link preference on LFLB and HFLB links, and ETX metric.

the deadline we count this packet as a lost and this node cannot forward this packet to other nodes as well.

Figure 5 shows the performance comparison between our algorithm (which we labeled it as  $B_{max}$  in the figure) and BIP. In Figure 5a, it shows the average end-to-end miss ratio per broadcast transmission in all 47 sources. Overall, our algorithm has 5.02 times lower miss ratio than BIP. In addition, the routes chosen by our algorithm are more energy efficient than BIP as well. This result is shown in Figure 5b, where the average energy cost is defined as the number of (re)transmission times per broadcast transmission in all 47 sources. The average energy cost from the routes chosen by our algorithm is 48 transmission times per broadcast but 232 transmission times from the routes chosen by BIP.

On the other hand, for the unicast setting, we simulated a number of jobs from 50 to 500 with random sources and sinks. Similar to the broadcast simulation, for each job we simulated the stream transmissions 1000 times from the testing set to determine whether a packet transmission is successful. The setting of link retransmission ability is the same as the broadcast setting. However, unlike broadcast scenario that broadcasts one source at a time, all the jobs would transmit at the same time in unicast scenario. Therefore, the algorithms need to consider the load-balance of the network to avoid congestion. Lastly, besides comparing  $B_{max}$  and ETX, we also add one more algorithm “ $B_{max}$  with link preference”, which tries routing in graph  $G' = (V, E')$  first, where edge set  $E'$  includes LFLB links only. If it cannot find a route for a job in  $G'$ , it then try routing in the original graph  $G$ . The reason to do so is because links in this class have both low and robust  $B_{max}$  value according to our analysis in Section 4.2.

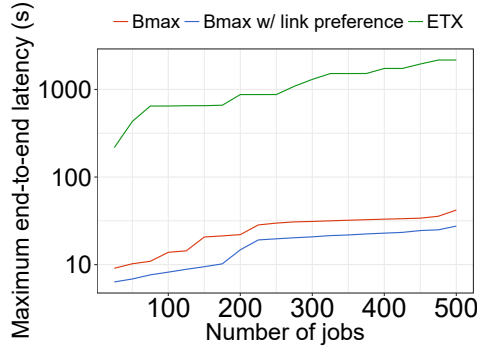


Fig. 7. The maximum latency (second) on streams after scheduling jobs from 50 to 500.

Figure 6 and 7 show the performances with different routing algorithms. Figure 6a reports the end-to-end miss ratio.  $B_{\max}$  with link preference shows most reliable transmission but it is not significant since the miss-ratio difference is only 1.5% compared with ETX. However, the routes ETX uses are not all energy efficient, which is reported in Figure 6b. For each job, routes from ETX takes 110 transmission times on average, where the routes from our algorithms take only 5 transmission times.

The scalability of our algorithms is also evaluated in this simulation. Figure 7 reports the maximum latency with 1000 samples after scheduling hundreds of streams. We have two observations in this figure. Firstly, the routes of our algorithms have low maximum end-to-end latency (from 6 to 40 seconds) compared with the routes of ETX (from 200 to 2000 seconds). Secondly, our algorithms are scalable since the maximum latency increases little when the number of scheduled jobs increase to 500.

To give another aspect of the difference in routing decisions between these algorithms, we provide the composition of links for routes in Table 10. We find out that there are 42.4% of HFHB links being chosen in the routes from ETX, whereas there are only 14% of HFHB links chosen from  $B_{\max}$  and 6.7% from  $B_{\max}$  with link preference. According to our previous analysis, the latency bound of HFHB links is high and causes high retransmission rate when these links are in the burst. Since our algorithms take account  $B_{\max}$  values when finding the routes, we avoid using these high latency bound links such that each scheduled job can have a lower energy cost and latency bound.

Routing Algo.	HFLB	HFHB	LFLB	LFHB
$B_{\max}$	24.2%	14%	54.3%	7.5%
$B_{\max}$ w/ preference	18%	6.7%	70.1%	5.2%
ETX	14.1%	42.4%	31.2%	12.3%

Table 10. Link classes distribution of the routes from different algorithms.

## 7 EXPERIMENT

In this section we evaluate our algorithm in terms of end-to-end deadline miss ratio (which is defined in Section 6) and latency bound. At first we describe the experimental setup. Then we describe how we measure burstiness parameters  $B_{\max}$  and  $B'_{\min}$  of every link and the interference matrix,  $IM$ . Then the effect of  $B_{\max}$  and  $B'_{\min}$  to estimated latency bound and end-to-end deadline miss ratio are evaluated.

## 7.1 Measuring Burstiness

To understand the effect of link burstiness on packet transmission and to calculate the model parameters  $B_{\max}$  and  $B'_{\min}$  of every link, we transmit 3,600,000 packets per link. The packet transmission rate is around 200 packets per second where every node tries to transmit the next packet as soon as possible with a zero inter packet interval. To avoid possible collisions and interference, we allow only one node to transmit packet at a time. There are 48 nodes, each node takes turn for packet transmission and at each turn a node transmits around 1200 packets continuously. We are supposed to transmit  $21 \times 24 \times 60 \times 60 \times 200/48 = 7560000$  packets per link in 21 days with the specified packet transmission rate. But we could only transmit 3,600,000 packet per link, because it takes time to fetch the sequence number of the received packets from all other nodes when a node finishes its turn of packet transmission. After collecting sequence numbers of received packets on every link we have a long data trace that is used to calculate model parameters  $B_{\max}$  and  $B'_{\min}$  of every link using algorithm 1. The spatial distribution of  $B_{\max}$  of the links at the testbed is shown in Figure 9, where green zones represent links having low  $B_{\max}$ , and red zones represent links having high  $B_{\max}$ . We find that most of the red zones fall within places where peoples' movement varies a lot, e.g., stairs, restrooms, copy-room, conference rooms, and kitchen.

## 7.2 Measuring Interference

$$IM(i, j) = \begin{cases} 1 & \text{if link } L_i \text{ and link } L_j \text{ are in interference range,} \\ 0 & \text{otherwise.} \end{cases} \quad (2)$$

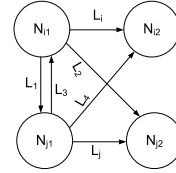


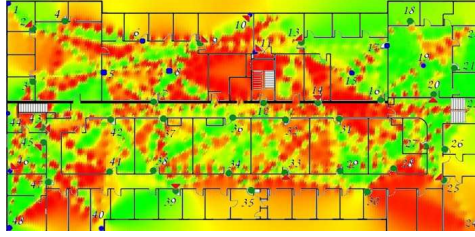
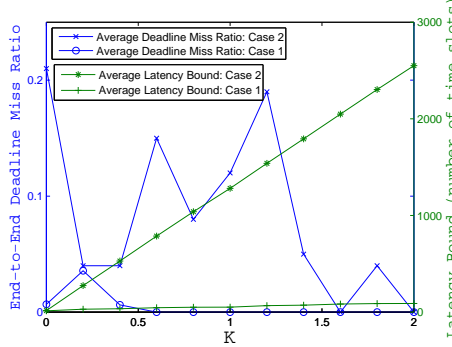
Fig. 8. Measuring  $IM$

The external interference, e.g. interference caused by WiFi networks, is captured through the  $B_{\max}$  and  $B'_{\min}$  parameters. To address internal interference, i.e. interference caused by packet transmission through neighboring links at the same time, we define a  $k \times k$  interference matrix,  $IM$  for a network of  $k$  links that specifies which links potentially interfere with others:

If  $IM(i, j) = 1$ , then scheduler will make sure that two packets are not scheduled for transmission over links  $L_i$  and  $L_j$  at the same time. The computation of  $IM$  is based on Packet Reception Ratio ( $PRR$ ), and  $PRR$  is computed from the same data trace that has been used to measure the burstiness parameters. We explain how  $IM$  is computed by using Figure 8. Assume that link  $L_i$  spans from node  $N_{i1}$  to node  $N_{i2}$  and link  $L_j$  spans from node  $N_{j1}$  to node  $N_{j2}$ . We compute  $PRR$  of every link and then set  $IM(i, j)$  to 1 if any of the links  $L_1, L_2, L_3$  or  $L_4$  have a  $PRR$  greater than  $PRR_t$ , a threshold. We set  $PRR_t$  to 0.3 in our experiment. This is a conservative model that hurts the latency bound, but improves the miss ratio.

## 7.3 Effect of $B_{\max}$

In this section we describe the effect of  $B_{\max}$  on end-to-end deadline miss ratio and latency bound. In our problem definition described in Section 2 the streams did not have any deadline associated with them and we define the deadline of the streams to be their generated latency bound. We evaluate it on the testbed. We compute the burstiness parameters  $B_{\max}, B'_{\min}$  and the interference matrix  $IM$  of the actual testbed network exactly the way specified in Sections 7.1 and 7.2. To ensure packet transmission experiences the same burst-behavior, at this experiment we maintain the same

Fig. 9. Spatial Distribution of  $B_{\max}$ Fig. 10. Effect of  $B_{\max}$  on End-to-end Deadline Miss Ratio and Latency Bound

packet transmission rate of around 200 packets per second with a zero inter packet interval that we use to measure link burstiness as described in Section 7.1.

To disable the effect of  $B'_{\min}$  we select  $B'_{\min}$  to 1 for all links. We define a multiplying factor  $K$  to demonstrate a trade off between latency bound and end-to-end deadline miss ratio and instead of allocating  $B_{\max} + 1$  time slots per link, we are allocating  $B_{\max} \times K + 1$  time slots for different values of  $K$  ranging from 0 to 2.

We run the experiment by considering two cases. In case 1, we use the whole testbed for evaluation. This case represents an actual deployed system. But in case 2, we remove the links which have the top 25% of lowest  $B_{\max}$ . This case represents another system where links are not as good and we are forced to use some non-stationary links. Since we are not using the top 25% of the links in case 2, the workloads for these two cases are different. But for each case, for each value of  $K$ , we generate 10 different workloads and the average values are plotted in Figure 10.

To generate a workload, we randomly select 10 pairs of nodes as sources and destinations to generate 10 random streams, and choose the route of those streams as the shortest path from the source node to the destination node having minimum sum of  $B_{\max} \times K + 1$  time slots. The starting time for every stream is set to its 1st time slot and period is randomly selected from a pool of even numbers up to 800 time slots for case 1 and 6000 time slots for case 2. If we consider a packet transmission rate of 200 packets per second, and allocate 5 msec per time slot, then the generated latency bound at  $K=1$  is  $51 \times 5 = 255$  msec for case 1, which is practical for the implementation of control loops in factory automation.

To time synchronize the nodes, we use a RBS style synchronization technique [9]. Since nodes may experience clock drift, to keep the nodes synchronized over time we need to allocate time slots for periodic broadcasting of the time-synchronization message.

Figure 10 shows the effect of  $B_{\max}$  on latency bound. From the figure we observe that as  $K$  increases, the average latency bound increases linearly for both cases. The reason is, as  $K$  increases,  $B_{\max}$  of all the links increases linearly and since  $B'_{\min}$  is one for all the links, there is no overlapping of time slots to reduce latency bound.

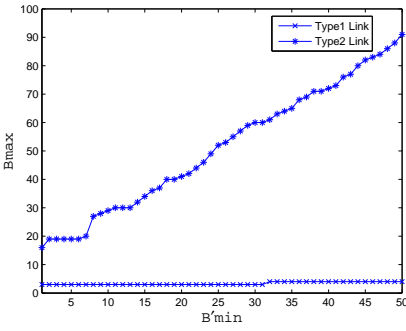
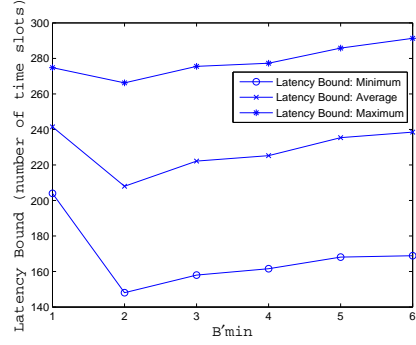

 (a) Variation of  $B_{\max}$  for Different  $B'_{\min}$ 

 (b) Effect of  $B'_{\min}$  on Latency Bound

 Fig. 11. The effect of  $B'_{\min}$  on the latency bound

Figure 10 also shows the effect of  $B_{\max}$  on end-to-end deadline miss ratio in the testbed. For case 1, with the full testbed, we observe that although there are some fluctuations in the end-to-end deadline miss ratio for different values of  $K$  up to 0.6, the end-to-end deadline miss ratio becomes 0 after  $K = 0.6$ . This implies that when there are many good links in a network, our solution obtains 100% packet delivery within the latency bounds. Surprisingly, we observe that even before  $K = 1.0$ . Note that our generated latency bound (at  $K = 1$ ) is within 14.2% of the minimum latency (at  $K = 0.6$ ) for which we observe zero end-to-end deadline miss ratio. So, the generated latency bound is relatively tight. Also, if we allocate  $0.6 * B_{\max} + 1$  time slots instead of  $B_{\max} + 1$  time slots, we can save 12.4 % of average latency and can still make all the deadlines. We may not want to set  $K < 1$  for hard real time applications, but it can be very useful to control the trade off between end-to-end deadline miss ratio and latency bound for soft real time applications.

For case 2, with the top 25% links removed, we observe a different result. Here, missing of deadlines continues beyond  $K \geq 1$  even though we are allocating  $B_{\max} \times K + 1$  time slots. The reason behind this is, links having high  $B_{\max}$  are typically susceptible to the changes in the physical environment as shown in Figure 9 and to accurately characterize these links, empirical data should be collected over more than 21 days. Since we are choosing least-burst-route for packet transmission, in case 1, we have sufficient number of links having low  $B_{\max}$  for packet transmission while in case 2, we are forced to choose some links having high  $B_{\max}$ . The burst size may become bigger than the measured one no matter how long the empirical characterization is. In this case, we can address the problem with two different approaches: packet recovery and link adaptation as discussed in Section 8.

#### 7.4 Effect of $B'_{\min}$

In this section we discuss the effect of  $B'_{\min}$  on the latency bound. For a particular stream, its latency bound will either increase or remain same if  $B'_{\min}$  is increased for every link that the stream goes through. The effect depends on the quality of the link. For high quality links, either stationary or asymptote stationary, the response to the increase of  $B'_{\min}$  is of two types. So, we call these links as Type1 and Type2 links. The response of the two types of links to different values of  $B'_{\min}$  is shown in Figure 11a.

In a Type1 link, as we see from Figure 11a,  $B_{\max}$  increases very slowly with the increase of  $B'_{\min}$ . As a result, the latency bound remains the same for some time and increases very slowly for the increase of  $B'_{\min}$ . In the case of a Type2 link,  $B_{\max}$  increases rapidly with the increase of  $B'_{\min}$  and that's why the latency bound for a particular stream also increases rapidly. The reason behind this behavior depends on the link quality. After transmitting 3,600,000 packets over every link, we

compute  $B_{\max}$  for different values of  $B'_{\min}$  by using algorithm 1. We observe that Type1 links are so good that even if we increase  $B'_{\min}$ ,  $B_{\max}$  remains almost the same and increases very slowly with keeping at least  $B'_{\min}$  number of good slots for packet transmission in every possible window of size  $B_{\max} + B'_{\min}$ . In the case of Type2 links, as  $B'_{\min}$  increases, to keep at least  $B'_{\min}$  number of good slots in every possible window of size  $B_{\max} + B'_{\min}$ , we need to increase the window size by increasing  $B_{\max}$  rapidly.

However, for a set of streams, the average latency bound may increase, decrease or remain same as a result of the increase of  $B'_{\min}$  of every link depending on quality of the links, either Type1 or Type2, and spatio-temporal overlapping of the streams. If a large number of streams pass through a dense network with some spatial and temporal overlapping in transmission, we observe a decrease in average latency as  $B'_{\min}$  increases, but after all the overlapping advantage is exploited, the average latency bound starts to rise again. We observe a similar result with 10 randomly generated streams for the testbed for which average latency bound decreases up to  $B_{\min} = 2$  and then it starts to rise again as is shown in Figure 11b. We run the experiment 5 times for each value of  $B'_{\min}$  and the minimum and maximum values of the latency bound of 5 runs are also plotted in the figure. It clearly indicates that we can minimize the average latency bound of a set of streams by an intelligent selection of  $B'_{\min}$  of the links.

## 8 FUTURE WORK

In this section, we discuss the possible improvements of our mechanism and routing/ scheduling algorithm. We consider the evaluation of these following approaches as future work.

### 8.1 Update $B_{\max}$ at runtime

Although  $B_{\max}$  and  $B'_{\min}$  are adaptive to capture worst-case link reliability for networks that have to support time-critical data delivery, it cannot be guaranteed that link characteristics are invariant. For example, links may have interference due to an interferer which appears temporarily in some areas of the network. The burst-behavior of the wireless links may change due to a change in the physical environment, e.g. node failure, node replacement, or unexpected obstacles. Therefore, it is necessary to measure and update link quality after a period of time. There are some works such as Burstprobe [5] that measure link burstiness online. However, their work needs additional time slots to remeasure max burst in the schedule, which increase redundancy of the network. Here we propose a method to update  $B_{\max}$  according to the transmission results of each link. We argue that  $B_{\max}$  values can keep being updated and it is not necessary to allocate large and long data traces every time to calculate new  $B_{\max}$  values.

Specifically, assume all streams are scheduled according to our proposed algorithm, each sensor would monitor the transmission results at its adjacent links. If a transmission failure happened at a specific link, we update its  $B_{\max}$  to  $B_{\max} + t$ , where  $t$  is an arbitrary integer, and reschedule the stream for its updated  $B_{\max}$ . We can also apply the doubling technique [17] to find the updated  $B_{\max}$  sooner. Although it is costly to reschedule streams, we argue there are only a few links that need to update their  $B_{\max}$  in one time if we only select links with robustness  $B_{\max}$ , as shown in Section 4.2.

### 8.2 Selection of $B'_{\min}$

In this paper, we set  $B'_{\min}$  to 1 for all the links during our evaluation of Section 7.3. In Section 7.4, we set  $B'_{\min}$  of all the links to 1, 2, 3, 4, 5, 6 to show how  $B'_{\min}$  affects latency bound. However, we did not set different  $B'_{\min}$  to different links. The minimum value of  $B'_{\min}$  is 1 and the maximum value of  $B'_{\min}$  is  $n$ , which is total number of streams in the system. One could compute  $B_{\max}$  values of all possible  $B'_{\min}$  values (1, 2, 3, ...  $n$ ) of all the links and call Algorithm 2 with different input

configurations repeatedly to find a better latency bound. However, this approach is computationally very expensive and we leave the selection of optimal  $B'_{\min}$  of each link to provide a better latency bound as future work.

### 8.3 Energy efficiency

Although the scheduler allocates redundant time slots for packet transmission, since we use least-burst-routing, most of the selected links are very good and hence single packet transmission suffices in most of the cases. Consider a case that the sensor network using a TDMA-based schedule with sleep and active modes, the radio needs to be only used for single packet transmission time in most of the cases and then goes to sleep mode to save the energy. In other cases of packet loss, the transmitter and receiver stay on until the packet is received.

On the other side, there are also other approaches which use statistical models such as Markov chain to predict burstiness. The transmitter would turn off the transmission temporary when they foresee the incoming burst [36] [16]. Their opportunistic approaches can save energy but do not provide any reliability guarantee and energy bound. Our solution can also potentially explore duty cycle based transmission schedules for more energy savings.

### 8.4 Specific reliability with latency bound

$B_{\max}$  metric considers stream latency when links are all in the worst-case scenario, which provides a latency bound for reliability guarantee. In addition, if we combine with the probability theory, it is possible to provide the latency bound with probability reliability guarantee. For example, we can consider a stream which only requires 90% success rate and derive a new minimum window  $B'_{\max}$  of each link on its route which has an equal or lower value compared with the original  $B_{\max}$  of each link. To calculate the new  $B'_{\max}$  we can use the algorithm which is provided in [13]. For the scheduling, we can keep using our algorithm since the input is still a route with a number of allocated time slots on each link. In this way, we can also provide routing and scheduling algorithms among streams with specific reliability guarantee with lower latency bounds.

### 8.5 System Integration

$B_{\max}$  metric and our proposed algorithms can be applied to any systems based on TDMA transmission. For example, WirelessHART [30, 34] is one of the systems using time slotted channel hopping (TSCH). It is adaptive to integrate with our scheduling algorithm under data-link layer. In addition, the routing schema in WirelessHART supports both broadcast and unicast routings which can be applied to our routing algorithm with a little modification. In graph routing of WirelessHART, the edges are created by the network manager and downloaded to each device. Thus, each device has full knowledge of the connectivity of the graph. To integrate with our routing algorithm, each device needs to provide  $B_{\max}$  information of its links to the network manager such that all devices can have the  $B_{\max}$  information of links in the graph. Therefore, each device can further generate the routes based on our broadcast/ unicast algorithm.

## 9 CONCLUSIONS

We have presented a new analysis technique that provides exact characterization and classification of the network links subject to link burstiness and interference. By considering link burstiness before scheduling, our routing algorithm provide reliable routes with minimum latency bounds for streams which are suitable for CPS. The scheduling algorithm is also novel since it accounts for both link burstiness and interference and offers a schedule for packet transmission that produces an upper bound on the latencies of the streams. We expect that this approach will improve the use of wireless networks in CPS.

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