Congestion Control Protocols in Wireless Sensor Networks: A Survey

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***Abstract*—The performance of wireless sensor networks (WSN) is affected by the lossy communication medium, application diversity, dense deployment, limited processing power and storage capacity, frequent topology change. All these limitations provide significant and unique design challenges to data transport control in wireless sensor networks. An effective transport protocol should consider reliable message delivery, energy-efficiency, quality of service and congestion control. The latter is vital for achieving a high throughput and a long network lifetime. Despite the huge number of protocols proposed in the literature, congestion control in WSN remains challenging. A review and taxonomy of the state-of-the-art protocols from the literature up to 2013 is provided in this paper. First, depending on the control policy, the protocols are divided into resource control vs. traffic control. Traffic control protocols are either reactive or preventive (avoiding). Reactive solutions are classified following the reaction scale, while preventive solutions are split up into buffer limitation vs. interference control. Resource control protocols are classified according to the type of resource to be tuned.**

***Index Terms*—Wireless sensor networks, transport protocols, congestion control, contention, resource control, traffic control.**

I. INTRODUCTION

A Wireless sensor network (WSN) is a set of tiny nodes that are equipped with embedded computing devices interfacing with sensors/actuators. They generally use short-range wireless transmitters and they act autonomously - but cooperatively - to route data, hop-by-hop towards a central node called sink, or base station.

A WSN comprises a large set of distributed nodes over a wide geographical (indoor or outdoor) area to monitor a physical or environmental event [1]. With the emergence of IoT (Internet of Things), WSN becomes more and more attractive by their integration in a real world of interconnected objects through internet [2]. As IoT consists of the perception and transmission of information for everything in many forms [3]–[5], sensing is the axis of concepts related to this paradigm like M2M (Machine To Machine) and CPS (Cyber Physical Systems) [6]–[11].

CPS tries to assist the interaction between the physical world and the virtual one through the integration of sensing, communication, computing and control, while the interfacing of M2M systems and WSN permit to take decisions with limited human intervention by emphasizing on the communications among machines and the practical applications to make appropriate actions [6]–[11].

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Some typical applications of WSN includes telemedicine monitoring, intelligent transportation, home automation, factory monitoring, energy conservation, target tracking and environmental monitoring, etc [1], [12]–[15].

Traffic patterns in sensor networks can be derived from the monitored physical processes. These applications might be interested in different sensory data and therefore create different requirements in terms of QoS (Quality of Service) and reliability. Further, depending on specific applications, the delivery of upstream traffic can be event-driven, continuous, query-driven, and hybrid. These types of applications are presented in the following.

Event-based applications: in this category, the network load is light but it unpredictably becomes active in response to a detected event. Depending on the application, the generated data may be large. For example, in the battlefield surveillance application, each node senses its surrounding in a continuous manner. When an event is detected (a tank entry), every node sends its samples to the base-station which can result in congestion [16]. Even the information generated at event happening causes congestion, its importance is vital for application fidelity. In practice, different combinations of traffic density derive from event-based applications. Some applications generate light occasional traffic from small regions, while others generate large frequent traffic across the covered sensing area [17].

Continuous sensing applications (Time-driven): some critical applications require continuous sending of sensing values to get real time values, e.g., nuclear stations monitoring. If the load of the network does not allow for continuous transmissions, periodic sensing can be used, but with an adequate periodicity that satisfies the application requirements.

Query-driven applications: contrary to event driven applications where the sensing nodes trigger the sending after the event detection, in query-driven applications it is the sink that invokes and queries sensing nodes to answer.

Hybrid applications: This kind of applications will be common in the future. In such applications, often bulk data is generated in addition to the constantly sensed data. For example, in structural health monitoring, each sensor measures structural vibration continuously at a certain rate. When the sensors detect a significant anomaly, they generate and send out data at a much higher rate [18], which will lead certainly to congestion happening.

Congestion occurs when the traffic load exceeds the available capacity on node level (buffer overflow) or link level (interference or contention) [19]. The delivery of traffic, even being well regulated, is hindered by the poor and time-varying channel quality, asymmetric communication channels, the need of multi-hop forwarding, and the hidden terminal problem [20], which make the congestion being severe. In the case of traffic load fluctuation, a high degree of unfairness is remarked at remote nodes, besides the previous problems. These circumstances lead that congestion causes the waste of the nodes' energy [21]–[23], but the most serious is that it degrades the event detection reliability [24].

Many papers dealing with transport layer issues have been published, e.g. transport protocols [16]–[18], [20], [24]–[77], comparative studies [3], [78], [79], cross-layer design that include the transport layer [80]–[82].

However, no standard transport layer protocol for WSN exists despite many efforts conducted by the IETF [5], [83]–[90] to adapt protocols of different layers in the context of 6LoWPAN to be suitable for WSN environment.

In [4], [19], [91]–[100], authors present surveys for transport layer protocols or congestion control based protocols. None of these surveys provide a deep and comprehensive taxonomy, neither cover a large category of protocols. This paper reviews various existing techniques for detecting and controlling congestion. The rest of the paper is organised as follows, in Section II, congestion control paradigms are presented with a discussion on their strategies. Section III presents some evaluation parameters to evaluate congestion control protocols. Some state-of-the-art protocols are presented in Section IV and V following our classification presented in section II. Section VI concludes the paper and presents some future works.

II. Congestion Control Paradigm

Beside the application type, the flow type is of high importance to guide a real congestion control. Flow types may include a single packet, few packets, a large number of packets, which require light control, medium level control, and tight control, respectively. When a large number of nodes transmit information, their flows will cross at intermediate nodes. This high number of sources increases the congestion but helps improving the reliability. For example, in tree architectures, every intermediate node can suffer from congestion causing packet loss, which in turn decreases network performance and throughput and cause energy waste. It is very difficult to predict the intersection points due to network dynamics (addition or removal of sensors or a change in the report rate), variability in radio channel quality over time. All these can transform uncongested parts of the network to under-provisioned and congested regions [20]. The area around the intersection will become a hotspot and there is a possibility of congestion (buffer overflow) and contention (links interference). For these reasons, a congestion control algorithm for data packet transmission is necessary.

Contention-based Congestion: when many nodes within range of one another attempt to transmit simultaneously, losses occur due to interference and packet loss is engendered. This reduces the throughput of all nodes in the area [101]. If the packet generation rate is sufficiently small, simultaneous transmission becomes independent of the rate. Rather, it depends on the exact time generation of the packet. Explicit local synchronization (or also named phase shifting) among neighbors can reduce this type of loss [35], but it cannot eliminate the problem as non-neighboring nodes can still interfere (hidden nodes). The contention may happen between different flows in the same area, and between different packets of the same flow, especially in the case of high density networks. Consequently, the nodes' channel capacity becomes time-variant.

Buffer-based Congestion: each node uses a buffer for the packets waiting to be sent. The overflow of this buffer causes congestion and packets loss. This is due to high reporting rate that varies in time due to dynamic channel conditions. The many-to-one nature (or converge cast) of WSNs causes congestion, in addition to the other causes shared with general wireless networks.

In [80], it is shown that when using large buffer sizes, the network load increase dramatically harms the event reliability, due to the limited capacity of the shared wireless medium. When buffer size is reduced, event reliability can be improved to some extent. For low buffer size values, buffer overflows lead to a larger number of packet losses but result in lower channel contention and lower end-to-end packet latency values compared to those values of higher buffer sizes. This result is opposite to the conventional thought that limited storage always leads to performance degradation. This property is advantageous for real time applications. The [80] study also shows the effect of maximum retransmission limit. Although moderate increase in this limit has a significant difference with low retransmission values, the excess in retransmission does not have positive impact on the overall network reliability.

The congestion control functionality follows, in general, three steps starting by its *detection*, which will be *notified* to the concerned node, so that an appropriate *control* will be taken. The following subsections treat in details these functionalities.

A. Congestion Detection Strategies

In literature [16]–[18], [20], [24]–[78] , many congestion detection mechanisms are used and tested. The most used are: packet loss, queue length, packet service time, the ratio between packet service time and packet inter-arrival time, delay. In many cases, a single parameter cannot indicate congestion accurately.

Packet loss: It can be measured at the sender if ACKs (Acknowledgements) are used; this suggests reliability to be ensured by the protocol [28]. It can also be measured at the receiver with sequence numbers use. Further, CTS (Clear To Send) packet loss can be used as congestion indication, as in [67].

Not overhearing the parent's forwarding on the upstream link, by a child node over the downstream link, can be used as an indication for packet loss [35], as well. The time to repair losses (if reliability ensured) can be used as a congestion indication [51]. Loss ratio is also used in some protocols [52], [60]. However, the losses can be caused by wireless errors rather than packets collision.

Queue length: as each node has a buffer; its length can serve a simple and good indication of congestion. In [20], [25], [29], [33], [36], [39], [42], [45], [46], [49], [50], [53], [56], [58], [59], [63], [69], a fixed threshold is used and the congestion is signalled as soon as the buffer length exceeds this threshold; while in [32], the remaining buffer length from the overall size is used. In [37], [43], the difference between the remaining buffer and the traffic rate is used as congestion indication. The traffic rate represents the excess rate, which is the difference between the output rate and the sum of sourced and forwarded rates. In [40], [62], the buffer length is used in addition to the difference of output and input time, which is quite similar to output and input rate. In [37], [61], buffer length and capacity of the node are used together.

The number of non-empty queues can indicate congestion level [65]. When there is a congestion, this number is larger than 0. This number increases with network load. If the link layer applies retransmissions, link contention will be reflected through buffer length [49].

Queue length and Channel load: In case of increase in packets collision, and after several unsuccessful MAC (Medium Access Control) retransmissions, packets are removed. Consequently, the decrease in buffer occupancy due to these drops may mean the absence of congestion when only buffer state is used for congestion detection.

Therefore, for accurate congestion detection, a hybrid approach is required using queue length and channel loading as a congestion indication [16], [17], [27], [34], [72]. Channel busyness ratio or channel load is the ratio of time intervals when the channel is busy (successful transmission or collision) to the total time.

In [16], the authors use the busyness channel ratio, similarly to channel load, but apply it to a subset of nodes, and queue length for another set of nodes. The node activates channel monitoring only when it receives a packet to forward. Therefore, there is no overhead to measure channel loading [17]. DST [54] uses node delay and buffer length as an indication of congestion. It depends on the used rate and channel load.

Channel busyness ratio and throughput measurement: In [66], the authors use throughput in addition to channel busyness to take into account the effects of hidden nodes problem in multi-hop environment. The throughput quantifies the number of successful transmissions.

Packet service time: as the inverse of packet service rate, it is the interval between packet arrival at the MAC layer and its successful transmission. It covers packet waiting, collision resolution, and packet transmission time at the MAC layer [18]. This value changes regarding to the queue length and channel load, so it is just another measure of them. It also represents the one hop node delay, as in [102]. In [47], the end-to-end delay is calculated in a similar way. But using only the service time may be wrong when the incoming traffic is equal or less than the outgoing one through the overloaded channel [70].

Packet service time and queue length: in [18], service time is used to continuously adjust the rate at which children send their packets. The diminution is performed based on the queue length.

Ratio of packet service time and packet inter-arrival time (scheduling time): A scheduler between network and MAC layer gives packets from network queues to the MAC layer. The scheduling time quantifies the number of packets scheduled per time unit. This ratio indicates both node level and link level congestion [24]. In [38], [44], [55], the ratio of rates instead of times is used and authors named it packet service ratio. In [73], the difference between service and scheduling rates is used instead of the ratio. In [31], in addition to the precedent ratio, buffer length is also used to detect congestion.

Delay: In general, it quantifies the necessary time since the packet generation, at the sender, until its successful reception at the next hop receiver [54], [102], or end point receiver [47]. It can also be calculated as a part of the total delay, as in ATP [30] (queuing delay). However, the use of delay as a measure of congestion may be misleading. The largest amount of delay is caused by the sleep latency due to the use of duty-cycling at the MAC layer [103].

B. Congestion Notification

When congestion is detected, the information must be propagated to allow taking an appropriate decision. This information can be as small as a single bit (congestion notification bit) [20], [25], [27]–[29], [54], [72], as rich as a new data rate information [47], other values helping on the calculation of the new rate [18], [24], [30]–[33], [38], [41], or even the actual congestion level [31], [37], [39], [43].

Congestion information can be transmitted in data packets header (implicit notification) [18], [20], [24], [25], [27], [29], [30], [33], [37]–[39], [42], [44], [45], [49], or in separate control messages (explicit notification) [16], [17], [28], [31], [34], [36], [41], [47], [52], [62], [63].

C. Congestion Control

For some applications, applying the same type of congestion control at all nodes would not ameliorate the throughput. For example, in event-based applications with limited messages per event, congestion control by traffic regulation at the sources does not apply. Phase shifting may serve as the appropriate alternative in this case [35]. However, intermediate nodes can and have to regulate the rate at which they forward the event packets to the sink when a bottleneck happens, and the rate control will take place at intermediate nodes. When the event is reported in several messages (e.g. multimedia applications), congestion control extend to rate control at the sources. In this case, phase shifting is useful.

The congestion control cannot be decoupled from the MAC protocol, and adequate protocol should first be used to avoid congestion. In applications where the event cannot be known

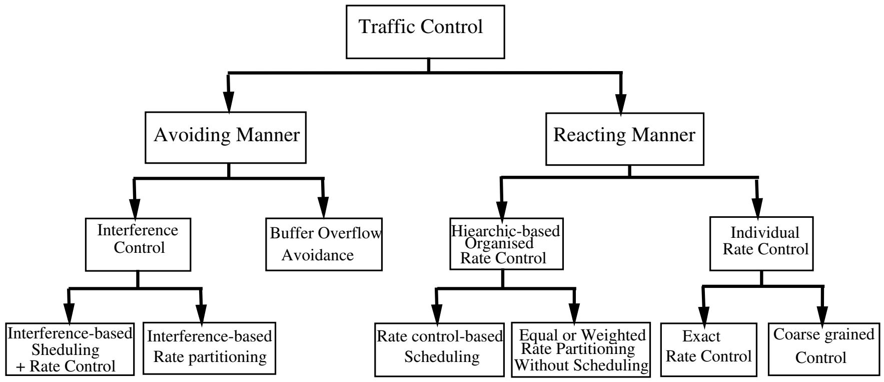

Fig. 1. Traffic control classes.

a priory, random access contention based MAC protocols are necessary (CSMA ”Carrier Sense Multiple Access”-based).

In continuous and periodic applications with high rate a TDMA ”Time Division Multiple Access”-like scheme is more appropriate.

Congestion control and fairness are two different but related aspects [69]. Fairness is the ability to ensure all data sources have equal access to the network bandwidth.

In a WSN monitoring and control applications, events may have different priorities and need to be reported at different rates. In this case, it is subject to weighted fairness instead of equality fairness. This paradigm is realized in different ways. In [20], [69], a token bucket scheme is used and each node can transmit only if it has a token. In [16], [18], [24], [29], [31], [35], [37]–[39], [42], [47], [51], [53]–[55], [58], [72], [73], the exact rate partitioning is used for both equal and weighted division, while in [40], [41], [49], [60]–[62], [68], [70], [76] scheduling is applied in addition to rate partitioning. Different metrics can be used for priority definition, depending upon application needs, e.g., event, node, region, or time [62]. In [24], [37], [47], [58], [69] the priority is defined at the node level according to the importance of its data. Further, routed packets (packets at intermediate nodes) are prioritized over sourced packets. In [38], [39], [42], [44], [53], [55], [62], the priority is defined at the data or event level. At the same node, different sensed events have different priorities. RAP [26] that has been proposed for query and event applications, gives more priority for packets originated from remote nodes from the sink over near sources, using packet Velocity Monotonic Scheduling (VMS). It chooses the forwarding order according to the distance and the end-to-end deadline. The priority is the ratio of the distance to the destination and the deadline value. RAP uses prioritized queues at each node. As packets from different prioritized senders may interfere on the same radio, RAP applies prioritized MAC to avoid collisions between different senders. Nonetheless, RAP does not present any rate control. It also requires localization, which comes at additional overhead. DST [54] uses the remaining deadline time as the packet priority. A packet gets higher scheduling priority with a decreasing value. In [74], a system rules is used to map data type to a transmission rate and a traffic class scheduling using phenomena’s priority and its location.

In end-to-end congestion control protocols, it is the end sink responsibility to detect the congestion [17], [25], [28], [30], [47], [66]. The sink may just receive the congestion indication and applies the control through an exact rate adjustment for each source [29], [54], [68], [76]. It can also be responsible for both the detection and the control [51], [52]. End-to-end control has a long latency, as at least one Round-Trip-Time (RTT) is needed to detect congestion. If congestion is transient and feedback latency is important, the notification may be much later than the congestion period. Thus, the solution may be inappropriate to WSN showing transient congestion [18]. The hop-by-hop back-pressure protocols [16]–[18], [20], [24], [27], [31]–[46], [49], [50], [53], [56]–[64], [67], [69]–[74], [102] react immediately to the congestion at the intermediate node, but they need more control at these nodes.

Many congestion control algorithms for WSNs are designed across the transport and MAC layers (and even the network layer) for efficient congestion detection and control. The cross layer design, by the interaction between different layers, that helps in enhancing sensor networks protocols has been investigated in [81]. [75] Shows through a case study how the cross layer helps in minimizing end-to-end delay, where [80], [82] investigate the usefulness of the cross layer design to congestion control.

Upon congestion detection, and depending on the application strategy, either traffic control is applied by throttling the node rates, or resource control is used by exploiting idle resources.

1) Traffic Control: The regulation or rate change of packets sending after the Congestion Notification (CN) can be assured in different ways. The AIMD (Additive Increase Multiplicative Decrease) scheme or its variants are usually applied [17], [35], [49], [66], [67], especially when using a single CN bit. In [20], [28], [32], [56], [69], [74], temporarily halt of packet sending is used to permit the congested nodes to empty their queues. The no embodying of event reporting nodes number, when calculating the increment/ decrement factor of rate change in AIMD schemes, leads to inappropriate values [62]. On the other hand, if detailed congestion information is available, exact and accurate rate adjustment can be implemented [16], [18], [24], [29]–[31], [37]–[39], [42], [43], [47], [50]–[55], [58], [60], [61], [64], [72], [73]. For adjusting the reporting rates, either a sink-based or in-network based solutions are used.

In [16], [46], packets are dropped not to propagate congestion. However, no congestion notification is used, which causes sources to waste their resources by continuing submission of traffic that will not achieve its final destination.

The traffic control can be performed in an avoiding or reacting manner, both of them can be based on interference or buffer overflow control. Avoiding interference is through scheduling the transmissions to avoid collisions [60], [68], [88], or partitioning the rate to prevent exceeding the interfering nodes capacity [49], [50], [61], [76]. The schemes that avoid the buffer overflowing are based upon limiting the sending [32], [56], [69].

With the reacting-based traffic control, both interference and buffer overflow are mitigated. The mitigation is based on either an organized hierarchic rate control, or individual control, so that only the rate of the concerned node is adjusted. The hierarchic-based organized rate control is applied through a rate-based scheduling [40], [62] or through an equal (or weighted) rate partitioning, without any schedule [16], [18], [20], [24], [29], [35], [37], [38], [42], [43], [47], [53]–[55], [58], [64], [70], [72], [73]. When applying individual traffic control, it is assured with either exact rate control [30], [39], [51], [52], [57], [66] or with a coarse grained control [17], [28], [31], [46], [67], [74]. Figure 1 illustrates explicitly our traffic control classification.

2) Resource Control: Resource provisioning techniques could be used when rate control methods cannot meet application's requirements, since reducing source traffic during a critical situation may violate application requirements. It is better to increase the capacity by turning-on more resources in order to face the high resulted traffic [34]. In presence of congestion, routing methods that employ alternative routes can be used to send data around the congested area [22], [33], [102], [104]. Load balancing the traffic between congested and uncongested routes upon congestion reaction has been used in [34], [44], [71], where preventive load balancing with an interference avoiding-based scheduling was used in [34], [63].

Resource control can be assured using clustering and multiple radios [27]. The cluster-heads are equipped with two radios; one is used to exchange packets with member nodes (short distance), where the other one is used to communicate with other cluster-heads and the sink (long distance). Some protocols adapt transmission power to ensure long distance sending [45]. Other protocols assure resource control by adapting duty cycling parameters, to balance between energy-efficiency (in low traffic scenarios) and traffic fidelity [41]. Figure 2 highlights resource control classes.

Some protocols neither apply traffic nor resource control upon congestion detection, but rather they apply aggregation strategies. Prioritized MAC schemes can also be used to give the congested nodes a prioritized channel access (making back-off length dependent on local congestion) permitting draining their buffer.

The choice of the control to be applied must answer applications requirements but at the same time has its consequences on the network lifetime. In [79], a comparative study between

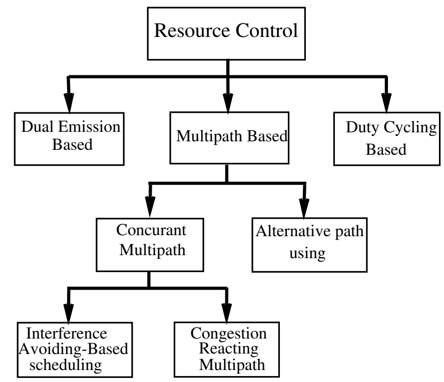

Fig. 2. Resource control classes.

traffic control (SenTCP [31]) and resource control (HTAP [33]) protocols in event-based networks has been presented. Parameters of comparison include: average node energy consumption, network lifetime, number of packets drops and source data rate. The results show that alternative path creation algorithms assure a high network lifetime while keeping the data rate stable, whereas data rate reduction algorithms present less power consumption per node and minimal packet drops. While spreading the traffic through different paths reduces the congestion, it increases the contention because of the crossing of these multiple routes toward the sinks.

In section IV, a resume of traffic control will be presented regarding the above classification, while in section V resource control protocols will be discussed. Through the literature, these protocols' performance is highlighted using simulations, experimentation or also by modelling their behaviour. This is done choosing specific metrics as evaluations parameters. The following section gives more details on theses parameters.

III. Evaluation Metrics

After the conception of a congestion control protocol, it must be evaluated in the purpose to show its efficiency in the presence of overload traffic. Through the literature, measures to evaluate the sensor networks performance under congestion are numerous. The measurement parameters allow comparing control strategies in specific cases. The commonest metrics used by the proposed protocols are: network efficiency, energy efficiency, sink received throughput, network fairness, and packet latency.

Network efficiency: quantifies the energy wasted on transmissions that do not deliver packets. The packets dropping cost varies depending on the distance from sink.

Energy efficiency: it is measured in joule (J). It includes energy spent in channel listening and packets transmissions and forwarding in the whole network. It is also measured per unit of successful communication or received packets [25], [34], [35], [45], [52], [55], [59], [67], [72]. In [44], [53], [62], residual energy is used as the ratio of final energy to initial energy. In [16], [55], [69], energy efficiency has been presented by the delivery ratio, which is defined in the following.

Energy Tax: the ratio between the total number of packets dropped in the sensor network and the total number of packets received at the sinks [17], [27], [33].

Packet Loss Ratio or Delivery Ratio: It is the ratio of the number of packets lost due to both buffer overflow and bit-error [18], [28], [31], [32], [37], [38], [40], [47], [55], [61], [62], [102] (respectively received [36]), to the number of packets generated. In [16], [33], [45], the number of packet drops is used, where in [73], the number of retransmissions per node is used.

Fairness: quantifies the variation degree in sending rates. A fair allocation of bandwidth delivered to the base-station from each node over multiple hops is desirable [16], [18], [20], [30], [35], [46], [49], [56], [60], [66], [69]. The weighted fairness regarding data priority is introduced in [37], [39], [70]. In [62], node throughput is used as fairness guaranty.

End-to-end delay (packet latency): It is measured as the time taken by a packet to reach the base-station from the time it was generated [20], [25], [28], [32], [33], [39], [46], [47], [57], [61], [64], [66], [67]. In [53], a per hop delay is used, where a weighted delay is used in [42], [70].

Control packet overhead: It quantifies the number of packets used by the protocol [102], or the ratio to total packets [43], [59].

The total throughput at the sink: It is the number of successfully received packets during time unit [16], [24], [30], [31], [36], [37], [39], [46], [49], [50], [55], [66], [67], [70], [72], [73]. In [51], [60], [64], network good-put is defined as the lowest observed packet reception rate at the base-station from any node in the network. In [38], [42], the throughput is weighted in respect to data priorities. In [53], [59], the total number of packets received by the sink during simulation time is used.

Instantaneous queue size: It shows the stability or fluctuation of queues [38], [44], [49], [50], [53], [61], [70]. In [55], the weighted queues notion is used, where the weight of a queue is determined by the importance of the events associated to it.

Memory requirements: It is generally based on buffer length, the code length, and the number of the considered sensing units [55].

Fidelity index: It is the fraction of the number of packets targeted to be received by the application, to that properly received [34].

Fidelity Penalty: It is the delivery of the required number of data event packets within a certain time limit [17].

Generated rate (or source rate): It is the total number of data packets generated by the sources per second [16], [43], [73].

In [78], a comparative study has been carried out using some metrics and different WSNs topologies. Different congestion control policies have been used (traffic control vs. resource control).

IV. Traffic Control Protocols

Existing transport protocols designed for Internet cannot be directly applied to WSNs as they either lack of reliability or flow control (UDP ”User Datagram Protocol”), or have high control overhead and inappropriate reactions to wireless losses (TCP ”Transmission Control Protocol”). These causes added to the specific WSNs characteristics are behind the motivation for new transport protocols. In TCP, ACK reception causes transmission window size increase. With a low-rate stream, this window inflation is artificial and does not mean that the indicated capacity is actually available. When an event occurs causing a sequence sending of packets, TCP supposes that the large window is usable, which is misleading and causes packet loss [74].

In this section many traffic control protocols are presented following the classifications presented in section II.

A. *Equal or Weighted Rate Partitioning*

In this category of protocols, congestion control is applied in a reactive manner, where nodes decrease their rates in response to congestion detection. This decrease is not applied independently to the detector node but in relation to its entire sub-tree, or even to the whole network. The decrease or the increase (when the congestion is eliminated) is performed by taking into account node priority, which results to an equal or weighted rate decreasing (respectively increasing). However, this control is completely decoupled with the scheduling. Table I, at the end of this section, summarizes some protocols of this class.

1) ARC [35]: Adaptive Rate Control treats contention in event and periodic applications by introducing a random delay (back off) at the application layer before transmitting packets (phase shifting). This way, it eliminates the hidden node problem without explicit control.
ARC uses packet loss as collision or congestion indication at each hop to adjust transmission rate of periodic applications. If the packet is successfully injected (overhearing parent forwarding), the node increases its transmission rate. Otherwise, it decreases its sourced rate and backs off for a phase change. An AIMD control is performed in a fair manner between sourced and routed traffic, using the sub-tree deep. Prioritized fairness has not been envisaged with ARC.

2) ESRT [29]: In Event to Sink Reliable Transport, sensors change their sending rate using the sinks feedback regarding the reliability level or congestion detection. Every node sets a CN congestion notification bit in the packets as soon as its buffer reaches a threshold. The sink periodically computes a new reporting rate for all sources based on reliability measurement, the received CN, and the old reporting rate. It broadcasts it with a high-powered amplification.
ESRT running includes in five states, No Congestion Low Reliability (NCLR), No Congestion, High Reliability (NCHR), Congestion High Reliability(CHR), Congestion Low Reliability (CLR), and Optimal Operating Region (OOR).
In NCLR, the reporting rate is increased to reach an acceptable reliability, while in both NCHR and CHR, the reporting rate is decreased. In CLR, the reporting rate decrease is sharper. In OOR the reporting rate is unchanged for the next decision interval.

Treating different characterized regions (event priority, node density) in the same way will decrease the system throughput. Moreover, ESRT does not give attention to interference.

3) EECC [72]: Energy Efficient Congestion Control protocol designs a source rate congestion control. Each node adds its current weight, which is defined as the product of channel busyness ratio and the buffer occupancy, to the packet received from its children, and passes the packet to its parent. The sum of such weights is then used. When buffer size and channel busyness ratio reach their higher threshold, the node sets the congestion notification bit in every data packet sent. By receiving this notification, the parent calculates the new rate and informs its children nodes.
After the sink collects enough data, it uses a clustering algorithm to partition nodes according to the sending rates and data similarity. Nodes within the same cluster work alternatively following an established schedule to save energy. However, cumulating weights does not directly reflect node channel busyness. The rate adjustment has not been clearly elaborated, as well as the rate sharing between children nodes.

4) FUSION [20]: uses three congestion control techniques: hop-by-by flow control, source limiting scheme, and prioritized MAC. It also uses buffer occupancy as a congestion indication level. When a node overhears a packet from its parent with the congestion bit set, it stops forwarding data to allow its parent emptying its queue. If the congestion persists, the hop-by-hop back-pressure reaches the source, which will consequently decrease its rate. By overhearing, each node determines the number of children nodes of its parent (N). The sending rate is regulated fairly between descendants using a token bucket scheme, where the node gains one token for N packet forwarded by its parent. In addition, FUSION uses a prioritized MAC scheme to give the congested nodes a prioritized channel access.

5) CCF [18]: Congestion Control and Fairness for Many to One Routing protocol modifies its rate using the packet service time which is the period between sending the packet at the transport layer to the network layer and the reception of successful transmission notification. Each node measures the average sending rate which is then divided by the number of children in its sub-tree. Each child compares this last with the rate sent from the parent to use and propagate the smaller one downstream by piggybacking in data packets. CCF also uses the queue length for requesting child nodes to reduce their rates if the defined threshold is reached. Maintaining a separate queue for every child is not memory efficient in dense networks. Further, in case of idle nodes or nodes with little traffic, the remaining bandwidth is not allocated to active nodes.

6) PCCP: Priority-based Congestion Control Protocol [24] considers that sensor nodes may have different importance and need different throughput. It uses then a weighted fairness defined with a node priority index. PCCP uses a scheduler between network and MAC layers, as well as two queues at each node; one for sourced traffic and another for transit traffic. It periodically detects congestion, using the ratio between packet service time and packet inter-arrival time at MAC layer. This ratio is used to achieve exact rate adjustment with priority-based fairness. PRA (Priority-based Rate Adjustment) algorithm is used to guarantee fairness between source and the sub-tree transit traffic. PRA adjusts the scheduling and the source rate using the priority weight of the sourced data and the global priority (sum of sub-tree weights) of the node to control link and node level congestion. In low congestion scenarios, PCCP increases scheduling and source rate of all traffic sources without priority index, and it implicitly uses the information about active nodes. While in case of high congestion, it decreases the sending rate of all traffic sources based on their priority index.

7) MCCH, APRC, PHTCCP: In these protocols, it is supposed that nodes sense different events and the sink assigns different priorities to data according to its importance. Multi-path routing is used in MCCH [38] (Multipath Congestion Control for Heterogeneous traffic) [44]. Both APRC [53] (Application Priority-based Rate Control) and PHTCCP [55] (Prioritized Heterogeneous Traffic-oriented Congestion Control Protocol) use single path routing.
Nodes dispose queues with different priorities for the different events. The queues are scheduled according to the inter queue priority and each node adapts its scheduling and output rates. In MCCH, where each node has multiple parents; the sum of parents scheduling rates gives the total scheduling rate.
MCCH and PHTCCP use the ratio of average packet service rate and packet scheduling rate as congestion indication (named as packet service ratio), similarly to PCCP, while APRC uses the average of queue lengths.

MCCH, PHTCCP and APRC apply a scheduler rate control when receiving a parent congestion indication or observing local measuring. They piggyback in the packet headers the scheduler rate and the number of child nodes. In addition to these parameters, MCCH piggybacks also the packet service rate, while APRC and PHTCCP adds the number of active child nodes, the average queue length of a node and its child nodes. PHTCCP uses traffic priority based MAC protocol (differentiating inter-frame-spacing and back-off mechanisms) by assigning short IFS and back-off to the higher priority traffic.
As MCCH and APRC do not apply any priority between children nodes for rate controlling adjustments, it is possible for a node with higher priority packets to have the same rate as other nodes having less priority packets. Further, queues scheduling at nodes are not elaborated.
In APRC, rate control defines priority depending on the actual queue length and not on the data. PHTCCP uses the same principle with application's weighting.
Even routed packets have higher priority than sourced packets; their location should also be considered. Further, this priority will penalize shallow nodes on the tree compared to deep nodes. Piggybacking control information in every packet causes a huge overhead.

8) QCCP-PS, NCC: Yaghmaee et al. propose QCCP-PS [58] (Queue based Congestion Control Protocol with Priority Support) for multimedia sensor networks and NCC [42] (Novel Congestion Control Protocol) for vital real-time signs monitoring in biomedical sensor networks. A separate queue for each child and another for local traffic are used, and the queue length is used as congestion indication. In NCC, packet class priority is applied within every queue.
When the queue length exceeds the defined max threshold, the rate is decreased, and it is increased once the length is bellowing another minimum threshold. Between the two thresholds, the congestion index is related to queue length linearly. Periodically, each node calculates the sending rate of its child sources and its local traffic source with considering the priority and the current congestion degree of the child nodes queues. This new rate is divided and sent to the child nodes, according to their total priority, i.e. the sum of the priorities of the sub tree rooted at the node, which is shared only between active nodes. The sending rate is the minimum of the parent assigned value and the local service rate. We think queue length alone is not sufficient for calculating congestion and service time may be useful for determining contention level.

9) CCF2 [73]: Congestion Control and Fairness protocol is a distributed congestion control algorithm proposed for tree based communications, using contention-based MAC that targets fair sending rate assignment. It is supposed that every node may have infinite data to send. It periodically monitors its output and input traffic rates. Based on the difference in these rates and the queued packets, it decides to increase or decrease the allocable bandwidth to sourced and forwarded traffic. It is similar to PCCP [24] with regards to congestion detection, where the service rate is substituted with the output rate, and the scheduling rate with the input rate, but instead of calculating the ratio, CCF2 calculates the difference between the rates. CCF2 uses an AIMD-like scheme and shares the increase/decrease between children nodes.

10) UHCC and HCCP: Both UHCC [37] (upstream hop-by-hop congestion control) and HCCP [43] (Hybrid Congestion Control Protocol), use packet delivery rate and buffer size to detect congestion. Each node uses the difference between the remaining buffer size and the net flow size to calculate and exchange its congestion degree. The net flow size is the difference between the sum of sourced and upstream neighbors flows rates, and the flows forwarded to downstream neighbors. Congestion happens when the sum of children and sourced rates is larger than the forwarding rate. It is controlled by reducing children and source rates. When the congestion degree is high, HCCP regulates rates by giving more incoming rate to upstream nodes with more data to send. If the rate is not sufficient, the congestion will be extended to other upstream nodes.
This rate sharing manner can be misleading as nodes with more data to send are not necessarily more important. In addition to this drawback, the protocol does not take the interference problem into account. In UHCC, the rate adjustment is assured using traffic priority ratio, which is the fraction between traffic priority at source or child node and the total traffic priority. It also uses the node congestion index to share the rate between the concerned nodes.

11) DST [54]: Delay Sensitive Transport protocol targets critical delay event applications where the late event notification at the sink leads to application failure. The event delay is the time between the event detection and the sink notification. DST uses a Time Critical Event First (TCEF) scheduling with prioritized MAC to ensure delay bounds. It measures the elapsed time to update the remaining time to the deadline at each node and piggybacks it in event packets. With decreasing values, the packets get higher priority. It also defines event reliability as the number of received packets in a decision interval. If the packets number is below a threshold, the reporting frequency is increased. DST detects congestion using average node packets delay and buffer level. Average node delay measures the contention around the node which varies depending on the used rate and channel load. Congested node having delay or buffer values higher than a threshold informs the sink using the notification (CN) bit in packet headers. Using reliability indicator and current network condition, the sink adjusts sensors reporting frequency, as in ESRT [29].
Neither DST nor ESRT try to avoid collision-based congestion, but they just decrease the source rates. No details for deadline attribution and TCEF scheduling have been given.

12) Prioritizing Information for QoS Control: This protocol [47] prioritizes the sensed information based on its nature. It uses end-to-end packet delay to evaluate congestion and update the rate at the sink. The new rate and the congestion level are forwarded to the sources including the congestion level. Intermediate nodes store data for a predefined time related to the estimated total packet drop probability. However, the congestion index is not well used, and no detail is given on how to use priority in rate calculating. Using end-to-end control may slow the appropriate control and west more time and energy, and it is prone to packet loss.

13) FACC [16]: Fairness-Aware Congestion Control protocol is a rate-based protocol dividing intermediate nodes into near-source and near-sink based on the application and QoS requirements. Near-source nodes record a per-flow state and allocate a weighted fair rate to passing flows based on available bandwidth. While near-sink nodes use probabilistic removing algorithm based on queue occupancy and hit frequency. Near-sink nodes send warning messages to the near-source nodes once a packet is dropped. Consequently, near-source nodes compare the incoming rate of each flow and the shared bandwidth to allocate a fair rate for passing flows, and notify the concerned sources to update their rate in an AIMD-like manner. FACC uses channel busyness ratio as congestion indication in near source nodes. The differentiation between near source and near sink nodes is not necessary and engender additional load. The use of busyness ratio and

TABLE I
EQUAL OR WEIGHTED RATE PARTITIONING PROTOCOLS.

Protocol	Congestion Detection	Congestion Notification	Congestion Control	H-by-H/ E-to-E	Application Type	Loss Recovery	Evaluation Type	Evaluation Parameters	Compared with
ARC [35]	Packet loss	-	Phase Shifting + AIMD Control	H-by-H	Event and Periodic	No	Implementation	Fairness,Energy Efficiency	Different CSMA Schemes
ESRT [29]	Queue Length	Bit in the header	Rate adjustment	E-to-E	Event	No	Simulation	Normalized Reliability, Average Power Consumption	Alone
EECC [72]	Cumulated Channel Busyness* Buffer length	Information in header	Rate control	H-by-H	Continuous, Event	no	Simulation NS2	Throughput, Energy Consumption, Packet Delivery Ratio	CODA, ESRT
FUSION [20]	Queue Length	Bit in the header	Stop Sending, prioritized MAC	H-by-H	Hybrid	No	Experimentation	Network Efficiency,Node Imbalance, Aggregate Sink Received Throughput, Fairness, Packet Latency	NCC, Rate Limiting
CCF [18]	Packet Service Time + Queue Length	Information in header	Rate adjustment	H-by-H	Event	Yes	Simulation, Implementation	Fairness, Number of Retransmissions Per Packet, Packet Generation Rate	Different components of the protocol
PCCP [24]	Packet Service Time/ Packet Inter-arrival Time	Information in header	Exact Rate Control	H-by-H	Event, Continuous	No	Simulation	Normalized System Throughput	CCF
MCCH [38]	Packet Service Ratio	Information in header	Traffic Control	H-by-H	Continuous	no	Simulation	Packet Drop, Queue Length, priority based Throughput	alone
APRC [53]	Queue Length	Information in header	Rate adjustment	H-by-H	Continuous	No	Simulation	Scheduling Rate, Queue Length, Node Delay, Packets Received, Residual Energy, Throughput	CCF, No Congestion Control
PHTCCP [55]	packet service ratio	Information in header	Rate adjustment	H-by-H	Periodic, event	no	Simulation NS2	Packet Drops, Weighted Queue Length, Memory Requirements, Throughput, Energy Efficiency	CCF, No Congestion Control
QCCP-PS, NCC [42], [58]	Queue Length	Information in header	Rate Adjustment	H-by-H	Multimedia	No	Simulation	Throughput, Achieved Priority, Packet Loss Probability	PCCP, CCF
CCF2 [73]	Service rate-scheduling rate-buffer length	Information in header	Periodic Rate Control	H-by-H	continuous	no	Simulation	Goodput, Fairness, Data Generation Rate, Link Layer Retransmissions	alone
HCCP [43]	Remaining Queue Length	Feedback msg, Information in header	Rate Control	H-by-H	continuous	No	Simulation, NS2	Source Rate, Control Overhead	AFA, BB
UHCC [37]	Remaining Queue Length- Excess Traffic Rate	Information in header	Rate Control	H-by-H	Periodic	No	Simulation	Throughput, Fairness, Loss Ratio	PCCP, CCF
DST [54]	Node Delay + Queue Length	Bit in the header	Rate Adjustment	E-to-E	Event	No	Simulation NS	Convergence Time, Energy Consumption	ESRT
Prioritizing for QoS [47]	E-to-E delay + queue length	Feedback msg	Rate Control	E-to-E	Event, Periodic, Continuous	Yes	Simulation NS2	E-2-E Latency, E-2-E Throughput, Data Loss, Priority Achieved	Different components of the protocol
FACC [16]	Channel busyness+ queue length	Feedback msg	Rate Control	H-by-H	continuous	No	Simulation NS2	Packet Loss, Source Rate, Fairness, Throughput	CODA, NCC
WRCP [64]		Information in header	Rate Control	H-by-H		No	Implementation	E-to-E Delay, Goodput	IFRC
DPCC [70]	Queue Utilization and Channel Quality	Information in ACK header	Rate Control, Adaptive back off	H-by-H			Simulation NS2	Queue Utilisation, Throughput Network Efficiency	CODA

buffer occupancy together may potentially achieve a better performance.

14) WRCP: [64] (Wireless Rate Control Protocol) tries improving the convergence time of a rate control using explicit capacity information in the purpose to overcome the long convergence time to the achievable rate, and frequent over capacity use that characterize AIMD schemes especially when the set of active flows in the system is continuously changing, which occur specially with short flows. WRCP applies a receiver capacity model in a tree like network by associating constant node capacities (one hop CSMA capacity), instead of links capacities, by assuming that the receiver capacity depends on the neighbors number rather than their transmission rates. WRCP models (by a linear equation) the relationship between the receiver capacity and passing flows' rates. Each node divides its allowable rate between traversing flows.

15) DPCC [70]: decentralized, predictive congestion control protocol controls congestion in a hop-by-hop manner through an adaptive flow and adaptive back-off interval selection schemes. It detects congestion using queue utilization and channel quality. The receiver regulates the rate in a weighted manner using the current and predicted congestion level, by estimating the outgoing traffic flow. The adaptive back-off interval is performed fairly regarding the number of neighbor nodes and fading channels in order to schedule adaptively the packet transmissions. DPCC uses MAC ACKs with piggybacking the envisaged rate.

B. Rate Control-based Scheduling

Like protocols of the previous category, those presented here after use a hierarchical reactive control, applied to the source and forwarder nodes. They differ, however, from the previous ones with respect to the applied method. The communications are re-scheduled by considering the rate change, without using any interference calculation. Table II summarizes the rate control-based scheduling protocols.

The operation of the proposed solution in [40] and MCCP [62] is based on successive data and schedule intervals. During data interval, nodes send events using the schedule they received from their next hop nodes. A single packet is sent per slot, whose length determines the reporting rate. Short slot length allows to forward more traffic per time unit. During the schedule intervals, nodes generate the schedule for the next data interval to obtain maximum throughput and avoid congestion. The nodes use an initial event request where they indicate to the next hop nodes the initial event reporting rate and the size of their sub-trees to permit the calculation of the schedule at the beginning of the schedule interval, which contains the slot length, the total number of slots, and the allocated number of slots.

During the schedule interval, nodes at one hop from the sink send the schedule packets to the nodes at the previous hop. Then every node compares the allocated slot length with the one it calculates and forwards the greater one. This schedule manner does not need a wide synchronization.

Time slots are dynamically assigned, depending on the per-hop average packet delivery time, and the buffer size. The average packet delivery time observed during the data interval is used as the slot length for the next data interval. Every node measures change in buffer occupancy between two consecutive data intervals and the predicted buffer occupancy (the actual interval buffer occupancy value + difference from the previous interval value). If the predicted value is not in the optimal range, then nodes adjust their slot length for the next interval by adding or subtracting a deviation factor. MCCP (Multi event Congestion Control Protocol) [62] accepts the following event reporting modes: general event reporting, per-node fair event reporting, and prioritized multiple event-reporting.

Slot attribution in this scheme does not show how to avoid contention, since no interference set establishment is used. On the other hand, if a purely sequential tree-based scheduling is used, performance will be very low.

C. Coarse Grained Rate Control Protocols

Unlike the previous two classes belonging to reactive control protocols, those presented in the following, limit the reaction to the concerned node, i.e. only those nodes causing the congestion (generally within one hop from the detector) to decrease their rates.

In this class, the AIMD paradigm is used to react to congestion. As the information contained in the congestion notification is limited, approximate rate adjustment is applied to the concerned nodes. Table III summarizes protocols of this class.

1) CODA [17]: Congestion Detection and Avoidance protocol uses a the present and past channel load conditions, and the current buffer state at each receiver, as congestion indication. CODA listens and measures channel load only at transmission moment, as carrier sensing is required before transmission. Once congestion is detected, the receiver broadcasts an explicit congestion notification back-pressure to its neighbors and adjusts locally the rate in order to avoid congestion spread. The neighbors consequently diminish their sending rates. The back-pressure upstream propagation is decided according to the local network conditions. If the congestion persists, the back-pressure is propagated up to the sources.

The source asks for constant feedback (ACK) from the sink, through setting ”regulate bit” in event packets, to preserve its rate. If the source does not receive ACKs, it reduces its rate. Also, low event packets rate reception at the sink is explained as a congestion indication that forces the sink to stop sending ACKs. CODA does not ensure fairness, and it does not detail how to change the rate after the congestion. The bandwidth may be badly used if the traffic control is not well designed.

2) SENTCP [31]: is a hop-by-hop congestion control protocol with three principles, computing congestion degree, sending feedback, and processing this feedback. It uses the ratio between the average packet service time and the average packet inter-arrival time, as well as the buffer occupancy ratio to estimate congestion degree at each node. If packet length is variable, it uses bit ratio rather than packet ratio. The feedbacks are sent to the neighboring nodes to adjust their sending rate. The later adjust their local rate, and they may relay the feedback to the next-hop. Every node substitutes the received congestion values by the ones it calculates. SenTCP may send feedback periodically, or when the buffer ratio exceeds a fixed threshold.

3) ART [28]: Asymmetric and Reliable Transport Mechanism provides event and query reliability, combined with congestion control. It classifies nodes to Essential nodes (E) and Non-essential ones (N). Higher energy level nodes are chosen as E nodes and form a topology toward the sink, ensuring end-to-end event and query reliability by recovering lost messages. E nodes send NACK (Negative ACK) to the sink when the query is lost, by using message’s sequence numbers as loss indication. For ensuring reliable event messages transfer, E nodes send an event alarm message to the sink and wait for the ACK. They retransmit this alarm if it is lost.

ART uses distributed congestion control handled by the E-nodes. It regulates the traffic by decreasing the active non-essential nodes. If an ACK is not received by E nodes during a time-out period, traffic of N nodes is reduced by sending them a Congestion Alarm message CA to stop their sending. If the congestion is not removed (ACK not received), the E node resends the CA by increasing the hop-count. When receiving the ACK, E nodes send Congestion Safe CS message to N nodes (with the hop-count value of the latest CA) to resume their normal sending.

Choosing E-nodes by only considering energy may be ineffective in some scenarios, where other parameters like the communication and event coverage assured by the nodes would be of high importance. Further, the fairness aspect

TABLE II
RATE CONTROL-BASED SCHEDULING PROTOCOLS.

Protocol	Congestion Detection	Congestion Notification	Congestion Control	H-by-H/ E-to-E	Application Type	Loss Recovery	Evaluation Type	Evaluation Parameters	Compared with
TDMA like, MCCP [40], [62]	Queue Length+ Packet Delivery Time	Scheduling msg	Slot Length Change	H-by-H	event	No	Simulation NS2	Packet Receive Ratio, Energy Consumption, Throughput	NCC, No Scheduled jittered Forwarding, Source based Congestion Control

between E-nodes has been completely ignored.

4) XLM [67]: is a cross-layer protocol fusing communication layers into a single protocol to minimize energy consumption, adapt communication decisions, and avoid congestion. XLM applies a receiver-based contention using routing level location, hop-by-hop congestion control, and distributed duty cycle. A node initiates a transmission by the broadcast of an RTS (Request To Send) with its location and that of the sink. By the reception of the RTS, each neighbor that is closer to the sink decides upon its participation according to the RTS Signal To Noise ratio (SNR), the remaining energy and available buffer space. If no CTS are received because of network congestion, the node multiplicatively decreases its generated rate. Otherwise, the generated rate is linearly increased for each received ACK. The overhead caused by this approach is heavy as each transmission at every hop must be preceded by a handshake message exchange. Also, the interpretation of CTS loss as a congestion is not accurate.

5) Bandwidth Management Architecture Protocol: This protocol [74] develops a rule system to specify how the generated traffic should be treated. It contains three components for bandwidth management: a rule system with priority queuing, a hop-by-hop flow control scheme, and a routing protocol. Each rule maps the data type and the generated value to a transmission rate, and a traffic class scheduling using phenomena's priority and its location. Nodes queue packets using the traffic class, and each node implements a rate-control mechanism. The packets are forwarded from the highest-priority not empty queue. When the queue size exceeds a threshold, the receiving node sends a synchronous NACK to slowing down the transmitter by momentary stopping sending. The transmitter waits by overhearing the congested nodes transmissions, and it resumes transmissions after hearing at least two packet transmissions from this node, which are as an indication of free queue space. However, sending from higher priority queues until empting them may penalize others.

6) PCC [46]: Priority-Based Coverage-Aware Congestion control protocol is a hop-by-hop mechanism at the network and MAC layers. Nodes generate periodic packets at a constant rate until event happening, where nodes generate event packets (indicated in the header) with higher rate and priority. Intermediate nodes forward packets with different priority using this indication. PCC uses queue scheduling with two queue thresholds to drop event and non-event packets at the network layer. When the queue length is less than the low threshold all packets are saved. When the queue length is between the two thresholds, non-event packets are stochastically dropped. If the queue value is greater than the high threshold, all non event packets are dropped, as well as some event packets. PCC defines packets cumulative survival probability and transmission failure probability in MAC/PHY(Physical) layer to quantify link quality and ensure fairness to remote nodes by cumulating their values.
PCC does not use any rate control and it performs static packets rate. It does not perform collision control, despite having information about channel state that is obtained through estimating link quality.

D. Exact Rate Control Protocols

This category uses tunable decrease, where the information contained in the congestion notification permit to the concerned node to decrease its rate in a precise manner depending on the degree of the congestion. Table IV summarizes protocols of this category.

1) ATP [30]: Ad-hoc Transport Protocol uses feedback for three purposes,i) initial rate feedback for start-up rate estimation, ii) progressive rate feedback for congestion detection, congestion avoidance, and congestion control, iii) and path failure notification feedback.
The intermediate nodes calculates available rate and piggyback it on the forwarded data packets. The receiver then collects and sends it periodically. Every node maintains two parameters. The average queuing delay of traversing packets, Qt, and the average packet transmission delay at that node, Tt. Tt depends on the contention between nodes in the same vicinity, and Qt depends on packets congestion of different flows at the same node. The node stamps the sum Qt+Tt if the previous value on the packet is smaller. Qt and Tt measures queue length and channel load, similarly to CODA [17], but in an accurate end-to-end manner.
The sender uses the feedback to increase, decrease, or maintain its rate. If the rate's feedback is lost (path failures), ATP performs a multiplicative decrease of the sending rate up to a maximum of two epochs. If no feedback is received for the third epoch, the sender moves to the connection initiation phase. ATP does not consider energy issue and it does not provide enough details with regards to fairness.

2) RCRT [51]: Rate-Controlled Reliable Transport is a centralized sink initiated transport protocol for loss-intolerant concurrent WSN applications. Each source initiates a flow by establishing an end-to-end connection with the sink, using

TABLE III
RATE CONTROL-BASED SCHEDULING PROTOCOLS.

Protocol	Congestion Detection	Congestion Notification	Congestion Control	H-by-H/ E-to-E	Application Type	Loss Recovery	Evaluation Type	Evaluation Parameters	Compared with
CODA [17]	Queue Length+ Channel Load	Back-pressure msg	AIMD	H-by-H, E-to-E	Event	No	Simulation, Experimentation	Energy Tax, Fidelity Penalty	No congestion control, open loop control
SENTCP [31]	Packet inter-arrival Time/ Service Time, Buffer Occupancy Ratio	Feedback msg	Rate Control	H-by-H	Event, Periodic	No	Simulation	Throughput, Packet Loss Ratio	TCP
ART [28]	Ack Loss	Feedback msg	Stop Sending	H-by-H	Event, Query	Yes	Simulation NS2	Residual Energy, Network Lifetime, E-to-E Delay, Loss Ratio	Alone
XLM [67]	CTS Packet Loss		AIMD	H-by-H	Event	Yes	Analytical, Simulation	Goodput, Consumed Energy, Latency	ESRT, CBR, RMST
PCC [46]	Queue Length	Locally	Packet Drop	H-by-H	Periodic, Event	No	Simulation	Throughput, E-to-E Delay, Fairness	FIFO

initial round-trip time (RTT) estimation and source desired rate. RCRT detects congestion and adapts the rate at the sink if the time to repair a loss is much more than RTT. It maintains a per-flow list of the out of order received packets. The list's length indicates the number of received packets after the first no recovered loss, which reflects the loss elapsed time. The number of RTTs elapse after the loss is a congestion indication. RCRT uses AIMD rate scheme with a time-dependent multiplicative decrease, based on loss rate. It uses a NACK scheme loss recovery, but it tolerate moderate end-to-end losses that may be caused by transient congestion or poor wireless links. So, sources transmit at a higher rate even with few losses. The new rate is piggybacked in NACK or sent in a separate packet. The use of an end-to-end scheme has slow reaction, and it causes high energy consumption.

3) PORT [52]: Price Oriented Reliable Transport protocol employs the termed node's price, which measures the communication cost from a node to the sink, in terms of path loss rate. This metric is increased when congestion happens. The sink continuously reports to the sources the desired reporting rate according to their price, the fidelity needed by the sink, and their contribution for enhancing this fidelity. PORT requests packets from less congested nodes to save energy, while maintaining the necessary level of reliability. Each node dynamically chooses its forwarder node, using the loss rates relating the node to its neighbors, and the prices of the latter. Sending end-to-end control information to every node may be difficult to achieve in multi-hop networks. Dynamic maintenance of a list of neighbors at every node with continuous update of loss rates has a significant memory footprint and communication overhead.

4) LATP [66]: Link Adaptive Transport Protocol is a transport layer end-to-end rate control scheme based on MAC layer feedback of the bottleneck node. It controls the offered load based on the contention degree. As the link capacity is time variant, the feedback provides the available path capacity for the sender to improve QoS. LATP uses the channel busyness ratio and throughput value (successful transmission) to predict the source's link contention. As link busyness ignores the hidden node problem, adding throughput estimation provides an accurate contention state. The throughput time includes RTS-CTS-DATA-ACK time. The sender controls the rate using periodical final-receiver feedback, as intermediate nodes piggyback contention information in outgoing packets. The receiver estimates contention degree on the path and informs the sender. The sender first uses a small rate until the reception of the first feedback, and then it controls the rate in an AIMD like manner.

5) ECODA: In ECODA [39] (Enhanced COngestion Detection and Avoidance protocol), packets are dynamically prioritized, using their initial static packet priority, delay and hop-count. The delay is the time from the packet generation to current time. ECODA defines the buffer weighted priority as the sum of terms, each of which is the number of packets in the appropriate priority class multiplied by the class priority. The weighted buffer difference of a node is the difference between its weighted buffer value and the maximum weighted buffer of its neighbors. ECODA uses buffer length and weighted buffer difference to deduce congestion status, which are piggybacked in packets. The packet delay value is also piggybacked on packets so that continuous neighbors delay estimation on a path gives the path delay to be used by the source as the rating send. When receiving a back-pressure message, the source node decreases its rate, or adjusts the rate for different paths if multiple paths exist. This is done by using the maximum delay. ECODA uses two queues, for sourced and forwarded traffic respectively. The scheduler selects the next packet in a round-robin way, and the forwarding queue contains separated priority ordered sources packets. It uses an AIMD scheme, but it differentiates forwarding and sourced rates. The weighted buffer concept can be ineffective when a node has few packets but with a higher priority than another one with more packets. Similar problem can be viewed in forwarding queue containing packets from the same source with higher priority than others, which must wait for the round-robin cycle.

6) CONSISE [57]: provides downstream Congestion control from the sink to the sensors. There are two categories of nodes:
-Receiver nodes that are concerned with the messages, i.e., buffering packets does not represent any overhead or additional cost, which simplifies the buffering of different

TABLE IV
EXACT RATE CONTROL PROTOCOLS.

Protocol	Congestion Detection	Congestion Notification	Congestion Control	H-by-H/ E-to-E	Application Type	Loss Recovery	Evaluation Type	Evaluation Parameters	Compared with
ATP [30]	Queuing Delay + Transmission Delay	Information in header	Rate adjustment	E-to-E		Yes	Simulation	Throughput, Fairness	TCP default, TCP ELFN
PORT [52]	Packet loss rate	Feedback msg	H-by-H Resource control + E-to-E Traffic Control	E-to-E	continuous	No	Simulation NS2	Energy Consumption	Directed Diffusion + ESRT
RCRT [51]	High Time to Repair Losses	New Rate in NACK header, or Feedback Rate msg	AIMD Rate Control	E-to-E	All types	Yes	Implementation	Goodput, Rate, Packet Reception	IFRC
LATP [66]	Channel Busyness Ratio + Throughput Measurement	Information in header	Rate control	E-to-E	Multimedia streaming (Continuous)	No	Simulation NS2	Delay, Jitter, Throughput, Packet Loss Rate	TFRC, TCP NewReno
ECODA [39]	Weighted Queue length	Information in header	Rate adaptation	H-by-H	Periodic	No	Simulation NS2	Throughput, E-to-E Delay, Weighted Fairness	CODA
CONSISE [57]	Periodic Rate Control	Information in header	Receivers Rate	H-by-H	Sink to Sensors Information	Yes	Simulation NS2	Latency, Number of Retransmissions, Number of requests Sent	NCC, NACK based loss recovery

packets, and - No-receiver nodes that just act as forwarders, and for which buffer occupancy represents an overhead.
CONSISE periodically adjusts the downstream sending rate of sensors to minimize downstream congestion influenced by traffic, from the sensors to the sink. It also adjusts the contention caused by broadcast. CONSISE controls the receiving and sending rate for a receiver, the sending and receiving rate for a non-receiver. The non-receiver nodes that relay receiver nodes form a chain acting as a single virtual link, with the same sending and receiving rate. Every node maintains the maximum sending rate based on local channel conditions, and the current sending rate based on downstream channel conditions. A receiver determines the fastest route from the sink by choosing at each epoch the upstream node from which the maximum number of packets was received, and it notifies this node that it is selected as the preferred upstream receiver. The preferred node sets its sending rate using the receiving rate(s) of its downstream receiver(s). A node that does not get notification to be a preferred receiver gradually decreases its sending rate. Every node piggybacks in every packet it forwards the current and the maximum sending rates, as well as the Id of bottleneck downstream receiver. The sending rate is determined by the explicit feedbacks received from the downstream nodes. Every node maintains a separate list structure of the upstream receivers, including their sending rates, downstream dependency, and their required receiving rates.

This protocol does not give importance to collision mitigation, as it just tries controlling sending rate without any scheduling. It also causes control overhead through piggybacking control information.

E. Interference-based Rate Partitioning Protocols

Contrary to all previous solutions that are reactive, the once presented in the following are preventive and try to avoid the congestion caused by the interference. This is assured by exploiting the knowledge of the interfering nodes, which is used for a capacity sharing between the nodes. This category of protocols is summarized in table V.

1) IFRC [49]: Interference Aware Fair Rate Control protocol is a rate allocation scheme for tree-based wireless sensor networks. It uses a CSMA-like protocol with link layer retransmissions. Each node maintains a queue for both generated and routed packets and detects congestion using the queue length threshold. It shares this information and its source rate with potential interferers using overhearing. Every node adapts its rate in an AIMD manner not to exceed the channel capacity. A node n1 is a potential interferer of node n2 if a flow originating from n1 uses a link interfering with the link between n2 and its parent. The Potential interferers set of a node covers the node's sub-tree, its neighbors sub-trees, and also includes nodes in its parent's neighbors sub-trees. Each node allocates to its potential interferes a fair and efficient share of the nominal bandwidth. Finally, each node uses the minimum of all attributed values. When a node,i, is congested, all its descendants are notified of its congestion and reduce their rates. The node's neighbors, including its parent, set their originating rates to the originating rate of i; this provided that the latter is lower than their rates. The process is repeated at all the neighbors of is parent. Recursively, descendants of the parent's neighbors reduce their rate to i's rate.

2) FLUSH [50]: is conceived for applications handling large data. The sink schedules data transfers sequentially in a round-robin fashion, to avoid inter-flows interfering. After a sink request, Flush has four phases, i) topology query, ii) data transfer, iii) acknowledgement, and iv) integrity check. It uses end-to-end ACKs, implicit control, and hop-by-hop rate control. In the transfer phase, Flush dynamically chooses a sending rate for a path using a combination of local bandwidth measurements and interference estimation algorithm. It communicates this rate by piggybacking to every node between the bottleneck and the source. This is to avoid intra-path interference. The maximum sending rate without collisions and loss depends on the interference range at each node, as well as the path length (for short paths). Every node

TABLE V
INTERFERENCE-BASED RATE PARTITIONING PROTOCOLS.

Protocol	Congestion Detection	Congestion Notification	Congestion control	H-by-H/ E-to-E	Application Type	Loss Recovery	Evaluation Type	Evaluation parameters	Compared with
IFRC [49]	Queue length	Information in header	Rate adjustment	H-by-H	Continuous	No	implementation	Throughput, Packet Reception, Rate Adaptation, Instantaneous Queue size, Max/Min Goodput	alone
FLUSH [50]	Queue Length	Information in header	Rate control	H-by-H	Query	Yes	Implementation	Overall Throughput, Transfer Phase Throughput	Different fixed sending rate
CADT [61]	Queue length + Link Capacity	Information in header	Rate adjustment	H-by-H	Continuous	No	Simulation	Packet Delivery Ratio, Packet Delivery Latency, Queue Length	Alone

continually estimates and updates its interference range and its necessary sending time by using information it acquires by overhearing the channel. It then piggybacks them later on data packets. Flush rate control uses two rules:
-A node transmits when its successor is free from interference. i.e., each node waits the forwarding of its successor and all nodes that have interfering transmissions with the successor's reception.
-A node sending rate should not exceed its successor sending rate. As a result, the source does not send faster than the slowest node along the path. If a nodes queue reaches a threshold, it temporarily increases the advertised delay to avoid congestion.
The sink saves missing packets sequence numbers for recovery at the ACK phase. Flush uses end-to-end selective negative ACKs, but it relies on link layer retransmission. When the data is recovered, the sink verifies its integrity.
Flush divides time into slots and one packet can be sent per slot, and nodes cannot send and receive in the same slot. The maximum tolerated transmission rate of a node (that does not lead to collision) located N hops from the sink with i hops interference range is: r(N, I) = 1/ Min(N,2+I), supposing that the maximum rate is 1 Pkt /s. The best rate requires every node to determine the smallest safe inter-packet delay.
Flush is very restrictive as only one source at a time can transmit, which breaks intermediate nodes with source traffic from transmitting during the whole allocated transmission. Flush has been limited to capacity sharing, but with a high abstraction of slot scheduling. A duty cycle based mechanism cannot be used with Flush, as it bases on continuous measuring of interference.

3) CADT [61]: Capacity Aware Data Transport protocol is similar to FLUSH [50], but it permits to handling several flows at the same time. It tries using the maximum link capacity by performing rate control to the congested links, using link interference and buffer occupancy. Every node piggybacks control information (transmission rate, transmission interval, current buffer size) in its data packets. It overhears the transmission slot of its neighbors in the interference set. The capacity of bottleneck link reveals the sub-network capacity. The transmission interval for a link, l(i,j), is called transmission interval of node, i. It includes the total transmission time from node, i, to node, j, reception at node j, transmission/reception time of all the nodes in the interference set. This value varies over time and it is continually updated.
The minimum transmission interval of node, i, on the link,l(i,j), is the maximum transmission interval of all the links in the interference set, except l(i,j). The sum of interfering set rates must not exceed the link capacity to avoid interference. CADT estimates link capacity using aggregated transmission interval (the sum of all links transmission interval in the interference set).
However, the calculation of the capacity has not been justified. The rate control uses AIMD, by exploiting the buffer of immediate downstream node, and the link state of the concerning node as well.

4) Mesh Interference Protocol: This protocol [76] takes as input the topology of the network, the flows routing paths, and their desired sending rate. It captures the network's interference dependencies as an approximate conflict graph, and it uses an iterative process to estimate the (max-min fair) safe sending rate for each flow in the purpose to reflect the total network throughput.

F. Interference Aware Scheduling

Randomized access schemes are energy inefficient and witness reduced throughput due to the increased contention. This can be avoided by structured communication with bandwidth allocation and access scheduling. Protocols of this class share with the previous one the aspect of congestion avoidance. However, instead of merely calculating the rate sharing between interferers, a schedule taking into account interference dependencies is applied. Table VI summarizes protocols of this class.

1) Max-Min Fair Collision-Free Scheduling: This protocol [68] tries ensuring fairness for predictable, stable, large sized and high data rates flows with tree-based WSN scheme. It presents a linear programming formulation algorithm for establishing max-min fair bandwidth allocation, and a collision free distributed scheduling algorithm for time slot allocation using BFS (breadth first search).
No details on how to detect interfering nodes has been provided. The bandwidth capacity is changing during time, which requires continuous allocation update.

2) QCRA [60]: Quasi-static Centralized Rate Allocation protocol aims at determining (by the sink) optimal and fair sources transmission rates using information about topology, link loss rates, and communication pattern, to be piggybacked on data packets. Its algorithm is based on rate assignment heuristic, with nodes using CSMA.

TABLE VI
INTERFERENCE AWARE SCHEDULING

Protocol	Congestion Detection	Congestion Notification	Congestion Control	H-by-H/ E-to-E	Application Type	Loss Recovery	Evaluation Type	Evaluation Parameters	Compared with
QCRA [60]	Packet Loss Rate	Information in header	Periodic Rate Adjustment	H-by-H	Continuous	No	Implementation	Goodput, Rate	IFRC

It defines the network goodput as the minimum packet reception rate at the base station from any node in the network. Its heuristic computes a coarse-grained TDMA schedule between the node's neighbors to determine neighborhood traffic rates. A node and its neighbors are split into independent sets of nodes able to transmit simultaneously. From each set, the node that transmits more flows (sourced and forwarded ones) determines the number of flows related to the set. The sum of number of flows in all the sets defines the total bandwidth requirement at a node. The heuristic uses this sum to fairly divide the bandwidth among flows at a node, assuming an implicit coarse-grained time-division. The operation is repeated recursively. As it is based on a CSMA like protocol, losses will certainly happen and their rate is used to assign nodes sending rates.
QCRA allocation decisions are periodic and link loss rates are used to perform rate allocation decisions for the next epoch. Each epoch lasts for tens of minutes. QCRA measures the channel capacity by sending packets from one node to another, rather than using theoretical channel capacity. But the centralized approach at the sink does not bring an added gain. The bandwidth capacity is changing and needs to be recalculated.

3) TSCH [86]–[89]: TSCH (Time slotted Channel Hopping) mode of the IEEE802.15.4e is a medium access scheme used with LLNs (Low power and Lossy Networks). LLNs result in large mesh networks composed of resource constrained devices generally related to internet and serving in many industrial applications such as process control and home automation, characterized by its multipoint-to-point (MP2P) traffic [88]. In TSCH, schedule based approach with channel hopping is used, which requires time synchronization between nodes. The communication is resumed on the repetition of the frame time, which contains slots reserved to each node to send or shared by many nodes. 6TSCH is the entity responsible to run this mechanism, and may be seen as an adaptation layer. 6TSCH is responsible for controlling topology links resources through the schedule. It also controls the mechanisms that define how nodes join the network to ensure its good performance by avoiding interferences and ensuring synchronization between neighboring nodes. 6TSCH ensures also flow control by the administration of queues policies for arrived and sent packets, in order to inform TSCH to decline new ones. As TSCH guaranties frames authenticity, 6TSCH applies the necessary mechanism to ensure key monitoring at joining event and securing data transfer and control.

G. Buffer Overflow Avoiding

This category of protocols has the preventive (avoiding) feature. But instead to avoid the interference, it is the buffer overflow that causes the congestion that is avoided. Table VII summarizes protocols of this class.

1) Congestion Avoidance Based on Lightweight Buffer Management, AFA: In [56] a tree based scheme is used, where AFA [69] (Aggregate Fairness Algorithm) generalizes the behavior to multipath networks. The authors behavior is avoiding congestion by permitting the parent to send at a rate matching the combined children rates. The sender transmits only when the receiver has enough buffer space. In [56], each nodes packet header piggybacks the buffer state. Before a child sends a packet, it checks parent's buffer. If full, it does not send until perceiving a non-full buffer state. In AFA, the weighted fairness notion is added. The node uses a token-bucket scheme similarly to Fusion [20] when the parent buffer is not full. As the child may still lose parent's buffer state due to the hidden terminal problem, the [56] protocol proposes a 1/6-buffer solution (advertising one sixth of reel buffer size). It also proposes an adaptive 1/k-buffer solution, where k is dynamically modified by the node advertising its buffer. Each node starts with k= 6 and reduces it in the absence of buffer overflow. After a buffer overflow, it increases k. In practice, it dynamically adjusts k without buffer overflow.
However, both AFA and [56] suppose that the collision problem is resolved by the MAC protocol using exponential back-off, which is not always effective in practice.

2) RBC [32]: Reliable bursty converge-cast protocol offers end-to-end packet based reliability in forwarding direction by scheduling retransmissions to reduce contention with newly generated packets. It implicitly detects losses, in a hop-by-hop way by hearing parent's packets header. The base-station uses the bloc ACK method, to enhance link utilization and adapts retransmission timer regarding network state. RBC ranks nodes by their queue size and the number of queued packets transmission attempts. Therefore, new packets are sent immediately to enable continuous packet forwarding. To reduce interference of the same rank packets and balance the network queuing and channel contention, inter-node packet scheduling uses packets number of a certain rank, permitting to nodes with more packets transmit earlier.
RBC implements a simple hop-by-hop flow control. Nodes piggyback their free queue size in packets and the sender detecting this number below a threshold stops sending for a certain period. Nevertheless, reducing interference by enabling the higher rank nodes is not efficient, as these nodes may interfere. Further, the fairness is not well ensured.

TABLE VII
BUFFER OVERFLOW AVOIDANCE PROTOCOLS.

Protocol	Congestion Detection	Congestion Notification	Congestion Control	H-by-H/ E-to-E	Application Type	Loss Recovery	Evaluation Type	Evaluation Parameters	Compared with
Lightweight Buffer Management, AFA [56], [69]	Queue Length	Information in header	Stop Sending	H-by-H	Event, Periodic	No	Analysis, Simulation	Packet Loss, Source Rate, Energy Expenditure	E-to-E Congestion Control, CODA, No Congestion Control (NCC)
RBC [32]	Remaining Queue Length	Information in header	Stop Sending	H-by-H	Event	Yes	Experimentation	Packet Delivery Delay, Loss Ratio	SEA, SWIA

V. RESOURCE CONTROL PROTOCOLS

A. Alternative Path Use

In these protocols, the application fidelity is considered, and the enhancement of resource use in the critical situation is adopted. Multi-path is used as a solution to augment resources. However, only a single path is used at the same time, and the other paths are used as alternatives in case of congestion.

1) SPEED [102]: tries maintaining a desired uniform delivery speed of real-time applications by diverting traffic through multiple routes and regulating sending rate, so, the end-to-end packet delay becomes proportional to the distance between the source and destination. It uses single hop sender delay estimation as a congestion indication. The sender timestamps queued packets and calculates the round trip single hop delay when receiving their ACK. SPEED searches for the next hop candidates that can support the desired delivery speed. If no candidate is found, the packet is buffered momentary or dropped not to propagate the congestion. It uses back-pressure re-routing to divert traffic. SPEED supposes the existence of location information, which is resource consuming. Further, no traffic control is used, which may result in excess capacity use. In [105], [106] enhancements of SPEED were proposed.

2) HTAP [33]: Hierarchical Tree Alternative Path algorithm is proposed for event-based sensor applications. It tries ensuring application reliability during overload periods without reducing the sources rate when sending critical events. HTAP combines two algorithms, Alternative Path Creation (APC) and Hierarchical Tree Creation (HTC), and it uses the network density to choose between them. When congestion takes place or a node's battery is about to draining, APC and HTC form alternative paths to the sink by unused nodes. APC uses these nodes by randomly exploiting neighboring table, while in HTC these nodes are placed in a hierarchical levelled tree starting from 0 for the leaves nodes. Every node piggybacks its buffer occupancy, reflecting its congestion state, when sending packets and the neighbors refresh their neighboring state tables when overhearing packets. A congested receiver sends a back-pressure packet to the sender in the purpose to remove congestion. The sender stops transmitting to this node and searches for a less congested receiver which leads to alternative paths creation.

B. Congestion Reacting Multipath

Unlike the alternative path protocols, the ones presented here use concurrent paths at the same time to enhance resource usage. Multiple-path rate balancing is used in response to congestion.

1) TADR [59]: Traffic-Aware Dynamic Routing protocol uses the idle or under-loaded nodes to remove congestion and enhance the throughput. It routes packets around the congestion areas and distributes them on multiple paths. It uses the depth to find the shortest paths and queue length to detect congestion. When there is no congestion, TADR chooses the shortest paths. In congestion case, TADR dynamically picks out multiple paths, so that the uncongested areas record or forward the excess of packets.

2) CAR [71]: Congestion-Aware Routing protocol discovers the congested zone between sources and the sink, that it preserves to forward high-priority traffic. CAR separates High Priority (HP) traffic from Low Priority (LP) traffic and uses multipath forwarding. HP traffic is the only routed through the shortest path nodes; while LP traffic is forwarded by uncongested nodes through longer paths. CAR follows three phases, starting by the HP network formation, congestion zone discovery, and differentiated routing. Combining these functions divides the network into congested zones and non-congested zones, where only the HP traffic is routed through the congested zone.

3) QOS-ACC: In QoS-ACC (QoS adaptive cross-layer protocol) [44], the authors suppose that nodes sense different events and the sink assigns different priorities to data according to its importance. They use multipath routing. QAC-ACC nodes send data to the appropriate next hop measuring QoS requirement of the packet (minimum delay, minimum service rate, reliability level), by using a distributed MAC manager. It considers a primary route and at least one alternative route. Nodes dispose priority queues for different events. A classifier at the network layer puts sourced and forwarded packets of "the same priority events" in the same queue with a higher priority to forwarded data as their loss results in more resource wastage. The scheduler plans the queues according to the inter queue priority. Therefore, adjusting the scheduling rate of a node adjusts its output rate. At the beginning, each node starts with a lower scheduling rate, and the originating rate depends on the scheduling rate and the data priority. QOS- ACC has two active queues, QRT and QNRT, for real time and non-real

time applications, respectively. A higher priority is assigned to real time applications and one back-up queue (QBACK-UP) is used for unacknowledged non-real time data.
QOS-ACC uses the ratio of average packet service rate and packet scheduling rate as congestion indication, (named as packet service ratio). It uses implicit congestion notification by overhearing. QAC-ACC applies a resource control method when receiving the parent congestion indication by splitting the real time-traffic to an alternate route. No detail concerning how to choose the scheduling rate is given, as despite of a resource control method use, a rate control has to be done in failure of resource control. Moreover, the use of the backup queue is not detailed.

C. Interference Avoiding Scheduling

In addition to the simultaneous use of multi-path, scheduling is also used in this category of protocols. Table VIII summarizes the protocols of this category.

1) TARA [34]: Topology-Aware Resource Adaptation strategy proposes using resource control for the aim of ensuring application fidelity. It uses a minimum number of nodes along the routing path during idle periods, to minimize energy consumption, and activates appropriate nodes forming new paths with sufficient capacity, to handle increasing traffic without congestion.
The authors study the influence of multiple paths on the end-to-end channel capacity and provide some guidelines for resource control schemes. In absence of interference, the capacity of a topology is the throughput of one-hop capacity. It becomes much smaller in the presence of link interference and a topology's throughput is limited by the bottleneck link(s) throughput. TARA calculates thus the capacity fraction for this portion. TARA defines the link congestion sum as the sum of link's traffic and interfering links' traffic. The chosen bottleneck has the largest congestion sum value. TARA uses graph-coloring approach for capacity estimation. It defines the topology interference degree and constructs the spatial interference graph, where each vertex is a wireless transmission, and the edges indicate that two transmissions are in the same interference range. Two links with an edge cannot transmit concurrently under the optimal schedule. Calculating the capacity fraction is to assign a coloring to the spatial interference graph, where each color is a time frame corresponding to transmission over a link. It represents the maximum throughput, or the delivery rate observed by the sink. When a hotspot node becomes congested (buffer occupancy and channel loading upper than the threshold), it chooses two nodes for constructing the detour path between the distributor and the merger nodes. If the creation of a detour path is not possible, traffic control mechanism is used by sending a back-pressure message to upstream neighbors.

2) I2MR [63]: Interference-Minimized Multipath Routing protocol is proposed for high rate streaming. It tries increasing the throughput by discovering disjoint paths for load balancing. It applies congestion control to load the paths at the highest possible rate. It uses conflict graphs to indicate interfering link groups that cannot be simultaneously active. Total Interference Correlation Factor (TICF) for a set of disjoint paths, derived from the conflict graph and defined as the number of links in the two paths that can interfere, describes the degree of interferences for all the paths in the set, and it is used to evaluate the quality of a path for multipath load balancing. I2MR records the interference zone of the first discovered path to avoid discovering another path within this interference zone. Each source node sends data concurrently using the primary and secondary path pair. It switches to the backup path only when either of the active paths fails.
Intermediate nodes detect long-term congestion using buffer length and notify the source to reduce its rate to the next predefined one. If the smallest predefined rate is reached and the congestion still happens, the source suspends packet loading (besides reducing the loading rate) for a predefined time, and sets a flag that is cleared later. If the congestion persists even with these two mechanisms, the source starts discovering new paths.

D. Dual Emission-based Protocols

In this class of protocols, the resource enhancement is envisaged either by using more than one radio, or different power emissions.

1) Siphon [27]: proposes distributing wireless dual radio virtual sinks to avoid congestion when it is persistently detected. Virtual Sinks (VS) form dynamically a secondary ad-hoc network, and route congested traffic to the physical sink to maintain the events rate, and avoid throttling or deleting packets. Siphon follows three phases starting by discovering the virtual sinks that will be selected. After the detection of congestion, the concerned node enables a redirection bit (in the network layer header) that permits to divert traffic out of the neighborhood, utilizing the VSs. Upon a VS reception of a redirected packet, it sends it to the near VS toward the sink. Every VS checks for its congestion level on the primary and secondary radios and does not advertise its existence if its radios is (are) overloaded. SIPHON adopts CODA congestion detection strategy in node-initiated detection.

2) TALONet [45]: uses traffic and resource control to avoid congestion. It uses two different transmission levels to alleviate link-level congestion, and buffer management to alleviate node-level congestion. It uses a multi-path detouring to increase the channel capacity (so can be classified in the previous class, too). TALONet works on three phases, network formation phase where it creates a virtual grid framework where each node is normal or talon. A talon node is the nearest to a certain grid point, which collects and relays the sensing data during the data transmission phase. The second phase is data dissemination, where the normal node transmits its sensing data to a neighboring talon approaching to the sink. This process is repeated until the data reaches the sink. The normal node uses the low power level, while the talon uses the higher one. TALONET applies the congestion avoiding

TABLE VIII
RESOURCE CONTROL PROTOCOLS.

Protocol	Congestion Detection	Congestion Notification	Congestion Control	H-by-H/ E-to-E	Application Type	Loss Recovery	Evaluation Type	Evaluation Parameters	Compared with
HTAP [33]	Queue length	Information in header	Resource Control (Alternative Path)	H-by-H	Event	No	Simulation-MATLAB	Network power, Packet drops, E-to-E Delay	alone
SPEED [102]	Single hop Packet Delay	MAC layer feedback	Resource+ Traffic control	H-by-H	Real time	no	Simulation (GloMossim), Experimentation	E-to-E Miss Ratio, Control Packet Overhead, Energy Consumption	AODV, DSR
QOS-ACC [44]	Packet Service Ratio	Information in header	Resource control (Alternative Routes)	H-by-H	Continuous	Yes	Simulation	Queue Length, Residual Energy	CCF, No Congestion Control
TADR [59]	Queue length		Resource Control	H-by-H	Event		Simulation TOSSIM	Receiving Packets Rate, Throughput Ratio, Energy Consumption per Received packet	MintRoute
TARA [34]	Queue length + Channel load	Feedback msg	Resource Control (Detouring Path)	H-by-H	Continuous	No	Simulation	Fidelity Index, Total Energy Consumption, Bit Energy Consumption	NCC, Traffic Control ideal resource control, topology-unaware resource control
I2MR [63]	Queue length	Feedback msg	Rate Control	H-by-H	Continuous	No	Simulation Glomosim	Throughput, Energy Consumption	AODV, NDMR
SIPHON [27]	Queue length+ Channel load	Bit in the header	Traffic Redirection	H-by-H	Event	No	Simulation, Experimentation	Energy tax, Energy tax Savings, Fidelity ratio, Residual energy	CODA
Talonet [45]	Queue length	Information in header	Rate control+ Detouring paths	H-by-H	Continuous	No	Simulation NS2	Dropped packets, Power Consumption	TARA, Back-pressure, NCC
ADCC [41]	Active period - Required Service Time	Feedback msg	Resource+ Rate Control	H-by-H	Periodic	No	Simulation NS2, Implementation	Packet Reception Rate, Loss Rates	NCC, Traffic Control
STCP [25]	Queue Length	Bit in the header	Rate control or Traffic Redirection	E-to-E	Event, Continuous	Yes	Simulation	Packet Latency, Energy Spent	alone
PLR [36]	Queue Length	Feedback msg	Traffic control + Resource control (Alternative Path)	H-by-H	Continuous	No	Simulation NS2	E-to-E Throughput, Packet delivery ratio	PCCP, LWBM

1/k node buffer management approach as proposed by [56]. Talon nodes piggyback congestion information in data packets. It uses detouring paths to distribute the traffic in case of congestion and decreases the source rate, in addition to its two level transmitting. If all receiving candidate nodes closer to the sink have no buffer spaces, the source considers the nodes located one-hop from the sink as receiving candidates.

Since talon nodes use higher power level to relay data, they exhaust their energy faster than normal nodes. To prevent this TALONet uses conditional or periodic topology update. However, using two levels for transmitting does not automatically avoid link level contention, as two normal neighboring nodes may interfere. The 1/k buffer method is restrictive.

E. Duty Cycling-based Protocols

Resource enhancement for critical event handling is envisaged in these protocols by adapting the duty cycle. This is to balance energy consumption and application fidelity. Table VIII summarizes the protocols of using Ressource control strategy.

ADCC [41] (Adaptive Duty-cycle based Congestion Control) protocol controls congestion using MAC adjustment. It uses resource control by increasing reception rate and traffic control by decreasing transmission rate. ADCC periodically calculates the required service time using children's packets inter arrival time to detect congestion. This is using the difference between the required service time and the duration of the active state in the duty-cycle. When incoming traffic is low, the active time of the receiving node is reduced to save energy. If congestion degree is below the higher threshold, it adjusts its duty cycle to reduce congestion by only applying resource control. If the degree is above the threshold and the active time reached the high limit, it notifies the children to reduce their rate by a calculated ratio. This duty cycle scheme is not appropriate for event-based applications when the events are unpredictable. The inter arrival time does not give the required service time but the scheduling time, this affects the correctness of the service time equation.

VI. CONCLUSION AND FUTURE WORKS

The wireless nature of WSNs expose their utilisation to harsh environment conditions like contention (links interference) and congestion (buffer overflows) which impact the overall system performance. Transport protocols play a pivotal role in improving the network reliability and throughput.

In this survey, these protocols are analysed in terms of their suitability to detect congestion and notify the concerned nodes so that an appropriate control will be taken. Also, many evaluation parameters to show protocols efficiency in overload traffic circumstances are presented.

Our study on congestion control protocols has shown that the application and flow types –characterized by the many-to-one nature communications– influence and guide the control applied to the traffic.

Depending on the application types, different mechanisms are used to handle the congestion. Either traffic control by throttling the node rates or resource control by exploiting idle resources are used to meet the application requirements.

From the flow type point of view, applying the same type of congestion control at all nodes may be a wrong decision to ameliorate the throughput. For example, phase shifting the source nodes is of best control solution for small packet event-based applications, while rate regulation is an adequate solution at intermediate nodes when a bottleneck happens. With voluminous packet event-based applications, the rate control extends to the sources. These applications rely on an contention based MAC protocols.

In periodic and continuous applications, reacting to the congestion can lead to performance degradation due to the elevated frequency of packets sending. So, an avoiding congestion control strategy through scheduling the transmissions to eliminate collisions and partitioning the rate to prevent exceeding the interfering nodes capacity seems to be a promising solution.

As sensor networks are energy constrained, upper bounding the sending rates of sources by only congestion limits may result in reducing the network lifetime. Applying an upper bound related to the application fidelity is thus of high importance. ESRT [29] and PORT [52] are examples of solutions using this principle.

A scheme similar to CTP [48] (A Configurable and Extensible Transport Protocol) can be proposed to handle congestion and other transport layer properties in WSN. In CTP, each transport protocol property is implemented as a separate micro-protocol and can be chosen based on the requirements of the application and executed when a specific event occurs.

The use of the standard IEEE 802.15.4e [107] and different IETF proposals [5] in the context of LLNs based IPv6 protocol stack, starting by TSCH at the MAC layer, 6TSCH [86], [87], [89] and 6LoWPAN [108] as adaptation layer, RPL [90], [109], [110] at the network layer, and COAP [83] at the application layer, may enhance the performance of the whole application and can be viewed as the IoT communication based solution meeting industrial requirements and needs.

In the LLN [88] paradigm, subnets of mesh networks are composed with constrained resource devices (such as WSN used in industrial and home automation) and attached to specialized routers. This brings the isolated WSN based applications and protocols towards the IoT based systems [2], which became more attractive notably by the use of IETF proposals cited previously.

A deep comprehension of the mechanisms above and their efficiency permit the design of a comprehensive transport protocol that also deals with reliability.

Reliability is habitually dealt with at the transport layer and it is essential to assure effective and dependable applications. Reviewing transport protocols that control the reliability aspect represents a perspective to this work.

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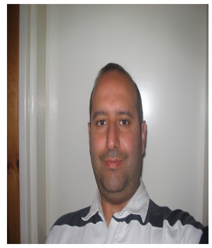

Dr. Ben-Othman received his B.Sc. and M.Sc. degrees both in Computer Science from the University of Pierre et Marie Curie, (Paris 6) France in 1992, and 1994 respectively. He received his PhD degree from the University of Versailles, France, in 1998. He was an Assistant Professor at the University of Orsay (Paris 11) and University of Pierre et Marie Curie (Paris 6), in 1998 and 1999 respectively. He was an Associate Professor at the University of Versailles from 2000 to 2011. He is currently full professor at the University of Paris 13 since 2011. Dr. Ben-Othmans research interests are in the area of wireless ad hoc and sensor networks, Broadband Wireless Networks, multi-services bandwidth management in WLAN (IEEE 802.11), WMAN (IEEE 802.16), WWAN (LTE), security in wireless networks in general and wireless sensor and ad hoc networks in particular. His work appears in highly respected international journals and conferences, including, IEEE ICC, Globecom, LCN, VTC, PIMRC etc. He has supervised and co-supervised several graduate students in these areas. He is widely known for his work on wireless ad hoc and sensor Networks, in particular, security. He is an editorial board member of Wiley Wireless Communications and Mobile Computing, Inderscience Int. J. of Satellite Communications Policy and Management, IEEE comsoc Journal of Communications and Networks (JCN) and International Journal On Advances in Networks and Services IJANS. He is also an Associate Editor of Wiley International Journal of Communication Systems. He has served as a member of Technical Committees of more than 40 international IEEE/ACM conferences and workshops including ICC, Globecom, MSWIM, LCN. He is a member of IEEE and ACM. He served as Local Arrangement Chair for the 13th IEEE International Symposium on Computer Communication (ISCC 09). He served as a TPC Co-Chair of IEEE Globecom Wireless Communications Symposium (Globecom 2010) and 9th international Workshop on Wireless local Networks (WLN09) and 10th international Workshop on Wireless local Networks (WLN10). He served as a publicity chair of several conferences such as the 12th ACM International Conference on Modeling, Analysis and Simulation of Wireless and Mobile Systems (MSWIM 09), IEEE International Symposium on a World of Wireless Mobile and Multimedia Networks (WOWMOM 2010), 25th Biennial Symposium on Communications. He has served as TPC Co-Chair for IEEE Globecom Ad hoc and Sensor and and Mesh Networking (Globecom 2011, 2014), 6th ACM International Symposium on QoS and Security for Wireless and Mobile Networks (Q2SWinet 2010, 2011, 2012), Wireless Networking Symposium of The 7th International Wireless Communications and Mobile Computing Conference (IWCMC 2011, 2012, 2013, 2014), IEEE International Conference on Communications Ad hoc and Sensor and and Mesh Networking (ICC 2012, ICC 2014). He has served for other conferences in ICNC, WSCP, CNIT. He is the secretary and will become Vice chair in january of the IEEE Ad Hoc and sensor networks technical committee since january 2012. He is an active member of IEEE CIS-TC, and WTC.

Nadjib Badache was with CISTTT (Centre d'Information Scientifique et Technique et de Transfert Technologique) now CERIST (Centre de Recherche sur l'Information Scientifique et Technique) at Algiers, from 1978 to 1983, working on applied research projects in information retrieval, operating systems and library management. In 1983, he joined USTHB University of Algiers, as Assistant Professor and then Professor, where he teaches operating systems design, distributed systems and networking with research mainly in distributed algorithms and mobile systems. From 2000 to 2008, he was head of LSI (Laboratoire des Systmes Informatiques of USTHB University) where he conducted many projects on routing protocols, energy efficiency and security in Mobile Ad-hoc Networks and Wireless Sensor Networks. Since March 2008, he is CEO of CERIST and Professor at USTHB University.