A Survey on Transport Protocols for Wireless Multimedia Sensor Networks

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Abstract

Wireless networks composed of multimedia-enabled resource-constrained sensor nodes have enriched a large set of monitoring sensing applications. In such communication scenario, however, new challenges in data transmission and energy-efficiency have arisen due to the stringent requirements of those sensor networks. Generally, congested nodes may deplete the energy of the active congested paths toward the sink and incur in undesired communication delay and packet dropping, while bit errors during transmission may negatively impact the end-to-end quality of the received data. Many approaches have been proposed to face congestion and provide reliable communications in wireless sensor networks, usually employing some transport protocol that address one or both of these issues. Nevertheless, due to the unique characteristics of multimedia-based wireless sensor networks, notably minimum bandwidth demand, bounded delay and reduced energy consumption requirement, communication protocols from traditional scalar wireless sensor networks are not suitable for multimedia sensor networks. In the last decade, such requirements have fostered research in adapting existing protocols or proposing new protocols from scratch. We survey the state of the art of transport protocols for wireless multimedia sensor networks, addressing the recent developments and proposed strategies for congestion control and loss recovery. Future research directions are also discussed, outlining the remaining challenges and promising investigation areas.

Keywords: Wireless multimedia sensor networks, transport protocols, congestion control, loss recovery, survey.

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1. Introduction

In recent years, wireless networks composed of resource-constrained sensor nodes have raised a lot of attention of the industry and the academic community. Such wireless sensor networks (WSNs) have addressed a series of monitoring and control applications by providing scalar information as humidity, pressure, temperature, luminosity, seismic variations, among others, all retrieved from deployed sensors in a monitored field. In order to allow monitoring in regions with absent infrastructure and to reduce costs for massive deployment, those sensors are expected to be battery-operated (finite energy supply) with limited processing and storage resources. As a result, issues as ad-hoc communication and energy-efficiency in wireless sensor networks have imposed many challenges, demanding specific research efforts in the last decade [1][2][3].

Sensor nodes in WSNs harvest information that has to be somehow accessible from outside the network. The limited communication ranges imposed by energy constraints of the nodes and the expected lack of infrastructure require an ad-hoc multi-hop communication model, where sensed data have to be transmitted to a central computer (sink) or a gateway using deployed sensors as intermediate transmission hops. Due to such inherent particularities of wireless sensor networks, traditional protocols from Internet can not be directed employed, requiring research in adapting TCP/IP protocols or even proposing new ones [4][5].

For an increasing group of applications, scalar data gathered from traditional wireless sensor networks are insufficient, even if a large number of sensors are deployed [6]. Applications as surveillance, environment monitoring, wildlife observation, localization and tracking, traffic control, automated assistance for elderly and disabled people, among others, can benefit from acquired information in the form of video, still images and/or audio (voice, animal sounds, noise, etc). Recent advances in CMOS technology have allowed the development of low-power cameras and microphones that can be embedded in wireless nodes for a new set of sensing functions, attending applications unassisted by Internet and even traditional wireless sensor networks. The resulting wireless multimedia sensors networks (WMSNs) define a particular coverage model (for camera-enabled source nodes) and have stringent communication and processing requirements [6][7][8]. Such challenging scenario has demanded new researches in communication protocols, architectures and paradigms for wireless networks comprised of multimedia sensors.

In typical wireless multimedia sensor networks, nodes are constrained in energy, processing and memory resources. Such constraints are the result of the effort to reduce the cost of the sensors, allowing massive deployment [9]: sensors are densely deployed in order to increase the sensing coverage, connectivity and network lifetime [10]. In fact, energy constraints limit the wireless communication range and inflict controlled use of transmission and processing functions. Furthermore, processing and memory constraints can impact the use of buffering techniques by communication protocols, as well as the execution of compression and entropy codes [11]. For WMSNs, additional challenges are imposed by real-time multimedia communication and minimum bandwidth demand, requiring specialized solutions to address such challenges. As a result, most transport protocols for wireless sensor networks have been shown to be inefficient for WMSNs [12].

WMSNs transport protocols are mainly designed to support congestion control, loss recovery or both services. In the first case, real-time multimedia streaming and event-triggered visual monitoring can potentially overload intermediate ad-hoc nodes and wireless links, incurring in energy waste, packet dropping and communication delay. Congestion control mechanisms could act on congested nodes or over an entire communication path to face the cause of congestion or even to relieve congested nodes applying some low-impact dropping police. Secondly, data loss on multimedia communications could affect the final data quality when high relevant data for the decoding process are lost. Typical approaches for loss recovery could retransmit lost information according to its relevance or send redundant packets through multiple paths. Moreover, correction codes could be used to reconstruct corrupted packets.

Several papers can be found on the literature surveying wireless sensor networks [1][2][3], wireless multimedia sensor networks [6][7][8] and multimedia communication in WMSNs [13][14][15]. In a different way, we surveyed the recent developments, challenging issues and open research areas of congestion control and loss recovery mechanisms in wireless multimedia sensor networks, where both services are usually addressed by protocols and algorithms conceptually located in transport logical layer. In fact, as cross-layer design is expected to enhance the overall performance of wireless multimedia sensor networks [16], congestion control and loss recovery services might be located in MAC, networking or even application layer. Furthermore, the notion of modularized protocol stacks can be disrupted, turning hard the classification of a service in a specific layer. Nevertheless, congestion control and loss recovery services are commonly assumed to be provided by transport protocols and they indeed are in many researches. In such way, our survey not only describe and compare approaches in the area of congestion control and loss recovery, but also present valuable transport protocols that can be employed in real-world wireless multimedia sensor networks applications. Moreover, the surveyed transport protocols are concerned with some specific issues of wireless multimedia sensor networks and thus bring significant contributions to the area, in a different way of recent works that address only transport protocols for scalar wireless sensor networks [17].

The rest of this survey is organized as follows. Section 2 discusses design issues of transport protocols for wireless multimedia sensor networks. In section 3, many transport protocols for WMSNs are surveyed. Future research directions are discussed in section 4, followed by conclusions and references.

2. Design Issues of WMSNs Transport Protocols

When designing transport protocols to support communications in wireless multimedia sensor networks, some design issues should be properly considered. Among them, the application requirements, the communication paradigms and the congestion control and loss recovery strategies are the most significant. Next subsections present these issues and discuss how they influence the design of transport protocols for WMSNs.

2.1. Application Requirements

In wireless multimedia sensor networks, it is expected energy-efficient transport protocols that are scalable and provide good performance in terms of transmission throughput, communication delay and fairness. However, the notion of "good performance" strongly depends on the application requirements and the nature of the deployed sensor network.

Many types of multimedia sensing applications have been allowed by WMSNs technologies. Regardless the purpose of the user application, each one will have communication requirements that will guide the choice of the appropriated protocols and communication technologies. For example, applications can be streaming for a prolonged period of time (requiring sustained information delivery) or may require event triggered observations (transmitting data bursts) [18]. They can also tolerate different levels of loss and delay in the communication. The logical architecture of the nodes, which can be typically flat or hierarchical [19][20], can create nodes with different roles for the sensor network, potentially impacting congestion and loss control. Moreover, event-driven, query-driven and data-centric issues can also influence the design of transport protocols [8]. Finally, while Internet defines one-to-one (unicast) and one-to-many (multicast) communication paradigms, transmissions in sensor networks flow in a many-to-one fashion, with majority of the communication flowing upstream [12].

Regarding the application requirements for multimedia communications, we can determine eight different classes of transmission requirements, as presented in **Table 1**. Although some of these traffic classes have limited practical application, they are a reasonable indication of the permissible level of congestion and packet loss in each type of communication.

Table 1. Transmission requirements for WMSN applications.

Real-time	Loss-intolerant	Stream/Data	Bandwidth	Example of application
yes	yes	stream	high	Monitoring, tracking and surveillance applications requiring high-quality multimedia streaming.
yes	yes	data	low	Critical control applications retrieving multimedia data with delay and loss constraints.
yes	no	stream	high	Multimedia data transmitted in real-time to a human or device, tolerating low or moderate loss.
yes	no	data	low	Snapshots, processed images and/or complementary scalar data that have to be timely received.
no	yes	stream	high	Multimedia streaming intended for storage and offline processing, with loss restrictions.
no	yes	data	low	Non-time-critical snapshot applications possibly associated with scalar data, with loss constraints.
no	no	stream	high	Multimedia streaming intended for storage and offline processing, tolerating moderate loss.
no	no	data	low	Non-time-critical snapshot applications possibly associated with scalar data, tolerating moderate loss.

2.2. Logical structure and communication paradigms

The characteristics of wireless multimedia sensor networks encourage the design of new protocols and communication technologies, which could be from scratch or adapting existing protocols [15]. This is due to the fact that traditional protocols and communication technologies from TCP/IP networks and scalar WSNs [21] are not suitable for WMSNs [15].

TCP was designed for wired links where bit errors in transmissions are uncommon (less than 1%). On the other hand, the major reason for packet dropping in wireless links is often bit errors in transmissions [22][23] (higher than 5% [24]), resulting in communication delay and undesired energy wasting when TCP is employed. Although some adaptations to TCP have been proposed [25][26], we have noticed that TCP-based protocols will not be considered by most wireless multimedia sensor networks. Finally, the lack of congestion control and error recovery mechanisms in UDP turns this protocol also unsuitable for most WMSNs.

Although important issues as energy preservation have been addressed by transport protocols of traditional wireless sensor networks [17][21], as PSQF (Pump Slowly, Fetch Quickly) [27] and CODA (COngestion Detection and Avoidance) [28], such protocols do not concern delay, jitter and minimum transmission rate requirements.

When designing transport protocols for WMSNs, well defined concepts as end-to-end transport-layer communication should be revised in order to achieve higher efficiency [16]. In some proposed transport protocols, intermediate nodes can process packets during transmission. For example, hop-by-hop processing can provide error recovery by 1-hop retransmission procedures [29] or even reconstruction of corrupted packets exploiting correction codes [30]. In-network congestion control can also outperform end-to-end mechanisms, achieving lower delay and higher energy preservation [31].

Other crucial design issue is cross-layer optimization, which may enhance the expected efficiency of WMSN applications disrupting the conventional information flow through the protocol layers [16]. The basic idea behind this approach is the reduction of the protocol overhead with the jointly design of network protocols, exploiting information in a way that would be prohibitive in strictly modularized protocol architectures. Some examples of cross-layer optimization in wireless multimedia sensor network can be found in [16][32][33][34].

2.3. Congestion Control and Loss Recovery Services

The high transmission rate of multimedia streaming applications may congest intermediate nodes and result in packet dropping. Additionally, idles nodes can suddenly wake up, producing a high data transmission rate [35]. In fact, retransmission of dropped packets due to congestion may rapidly deplete the energy of the overloaded nodes and incur in extra communication delay [36]. Congestion may also prejudice the event detection reliability of event-triggered applications, since real-time data will be delayed if intermediate nodes are congested. Finally, congestion degrades the link utilization, incurring in more delay and energy depletion.

When nodes receive more data than they can process, which means packet-arrival rate exceeding packet-service rate, theirs buffers overflow and packets are dropped. It is more likely to happen in nodes closer to the sink, since they receive more combined upstream traffic. We can also expect congestion when the shared wireless links are not able to bear the current transmission demands of the nodes, usually due to contention procedures, physical interference or bit-errors [37][38].

Congestion control usually comprises three distinct logical steps:

a) *Congestion detection*: the first step to treat congestion is to detect it, actively or proactively. Active methods can use timers or acknowledgement to detect congestion, demanding additional processing and energy resources of the sensor nodes. Proactive methods employ simpler congestion indicators observed from the network. Typical congestion indicators are queue length, packet-service time and ratio of packet-service time over packet inter-arrival time.

b) *Congestion notification*: after congestion detection, congestion notifications have to be propagated from the congested node to the upstream sensor nodes and/or to the source nodes. In fact, congestion notification can be performed in an implicit or explicit way. Implicit notification utilizes techniques as piggybacking, which may save energy avoiding an extra control message. On the other hand, the most suitable option for some transport protocols may be to use explicit notifications to inform congestion, defining a specific control message.

c) *Congestion mitigation*: when the sources and/or the intermediate nodes are aware about the congestion, some mechanism for congestion mitigation has to be applied as soon as possible. Typically, congestion mitigation is accomplished by transmission rate adjustment, which may follow some QoS requirements based on the relevance of parts of the encoded media or the significance of the source nodes for the application. Other approach maintains the source transmission rate employing load repartition over multiple paths, reducing the traffic through the congested path. In the case some action is taken before an effective congestion, we define the congestion control approach as congestion avoidance.

Wireless links have a high bit-error rate when compared with wired links, which can result in packet dropping along the time. Congestion can also result in packet dropping and data loss [36]. Transport protocols can ensure reliable packet delivery in WMSN applications by retransmitting lost packets or reconstructing corrupted packets. Loss recovery is necessary because some parts of the encoded media may have higher importance for the decoding process [39][40][41]. However, the adopted recovery procedures have to comply with delay and jitter requirements of the application.

Three distinct steps may compose the loss recovery mechanisms:

a) *Loss detection*: nodes in a wireless multimedia sensor network have to be able to detect lost packets. Loss detection is performed by the sender or the receiver nodes. The sender/relaying node can use timers or overhearing techniques to check packet dropping. On the other hand, receivers typically employ sequence numbers in received packets to discovery packet loss.

b) *Loss notification*: when some loss is detected, the source, the receiver and/or the proper intermediate nodes have to be notified. Explicit loss notification usually sends acknowledgement messages to confirm the correct reception of transmitted packets or to directly request retransmission. Loss notification can be also implicit when nodes overhear successful transmission from the next hop.

c) *Data recovery*: the most common approach to recover lost packets is retransmission. The retransmission procedures in WMSNs should follow a hop-by-hop approach, where intermediate nodes cache packets for faster retransmission. Doing so, the overall communication delay is reduced and undesired energy consumption is avoided. An alternative is to reconstruct corrupted packets using recovery codes as Forward Error Coding (FEC) or to transmit redundancy packets. According to the adopted approach, the reconstruction can be performed by intermediate nodes or at the sink.

Congestion control and loss recovery are indeed complementary subjects. Congestion mitigation approaches can cause packets dropping while loss recovery mechanisms as retransmission can congest intermediate nodes. Although frequently treated in a separate way, most proposed mechanisms surveyed herein will have some impact in both services. Nevertheless, we considered the services separately in order to outline the expected benefits and the specific imposed challenges, helping in the development of future transport protocols.

3. Transport Protocols for WMSNs

In wireless multimedia sensor networks, nodes and links can become overloaded when relaying/routing many packets, requiring congestion control mechanisms in order to avoid packets dropping, additional communication delay and energy wastes. Moreover, bit-errors are common in wireless links, also demanding support for loss recovery. Although we can find other topics concerning transport protocols, as reordering of received packets and QoS, congestion control and loss recovery in WMSNs are major issues that have addressed most of the research in transport protocols for sensor networks.

To the best of our knowledge, there is still no ideal transport protocol for WMSNs. However, many of them bring contributions that together can improve the communication performance of WMSNs, as multipath congestion control, packet prioritization, hop-by-hop retransmission, source synchronization, reconstruction of corrupted packets, among others.

3.1. Transport Protocols for Congestion Control

When intermediate nodes or wireless links get congested, multimedia communications may experience packet dropping, transmission delay and energy depletion. In fact, congestion may have different impact on wireless multimedia sensor networks according to the application requirements. For example, real-time monitoring applications may require smooth traffic variation, while loss-intolerant applications will demand reliable transmissions over the network. And such requirements will dictate the specific approaches that should be adopted for congestion mitigation.

We defined a taxonomy for transport protocols according to the different approaches usually employed for congestion control in wireless multimedia sensor networks. Such classification establishes three categories, where the first two of them refer to congestion mitigation and the last one stands for congestion avoidance:

a) *Rate adjustment*: when network is facing congestion, the basic mechanism for congestion mitigation is transmission rate adjustment. The idea is to reduce the current transmission rate in order to relieve congested nodes, the communication path(s) and/or wireless links, probably reestablishing the original rate after congestion.

b) *Multipath load distribution*: if one or more paths toward the sink are overloaded, the source node can increase the transmission rate in paths that are not congested, decreasing the load at the congested path(s). Alternative idle paths can also be employed to face congestion. Such approach maintains the quality of the transmitted data, since the overall source transmission rate is not adjusted.

c) *Network congestion avoidance*: some energy consumption and packet loss can be avoided if it is perceived that intermediate nodes may soon become congested if the current communication scenario is not changed. In such case, the transmission rate may be adjusted or alternative paths toward the sink may be employed. The queue occupation and the ratio of relayed packets over received packets in each node are good parameters for congestion prediction. Notice that congestion avoidance is indeed only a methodology, since it is likely that one of the two previously presented approaches for congestion mitigation will be considered to avoid (to face in advance) congestion.

Fig. 1 depicts the surveyed works in each of the three defined categories.

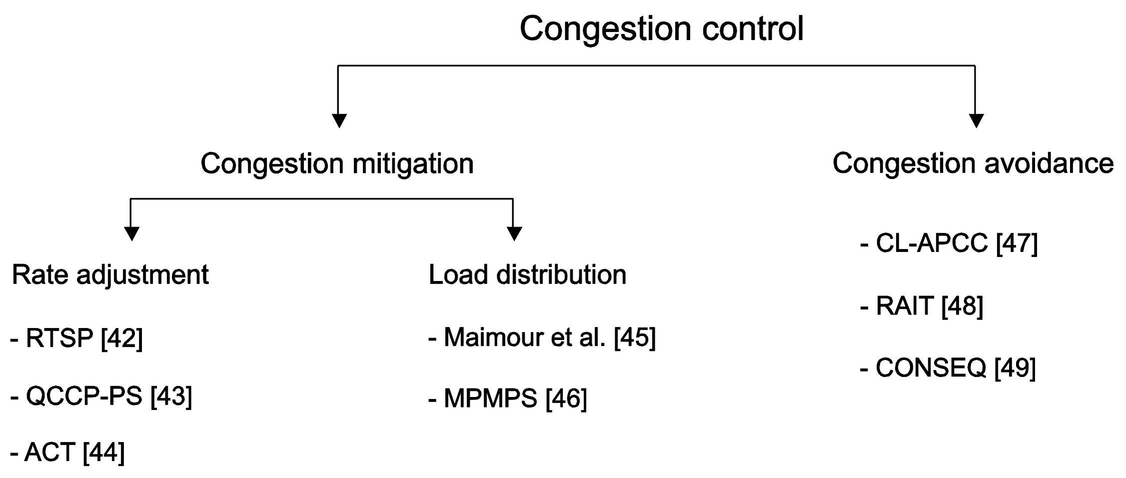

Fig. 1. The surveyed congestion control mechanisms.

3.1.1. Rate Adjustment

The Reliable Synchronous Transport Protocol (RSTP) [42] is a transport-layer protocol used for synchronization of image transmissions from multiple sources. The synchronization guarantees the same level of quality for the received images and a fairer utilization of the available bandwidth when images are coded using a progressive codec like JPEG. Only when all images of the same quality are received by the sink, source nodes are allowed to transmit higher quality images. This approach provides a soft load calibration which can be employed, with some adaptations, for congestion avoidance.

RSTP can directly manage congestion employing a mechanism based on TCP's congestion control facility. TCP congestion control is not feasible for wireless links, since it can not distinguish the nature of a packet loss (link-layer error or congestion), treating any packet dropping as congestion. An alternative is to use some extension as TCP-Explicit Loss Notification (TCP-ELN) [50] to explicitly inform the reason of the packet loss. In such case, TCP congestion control mechanism is used with no modification, but only when the source is notified about a congestion. RSTP incorporates TCP-ELN to its congestion control service.

The congestion control mechanism enabled in RSTP has some drawbacks. Even with the adaptation proposed in [50], TCP-based congestion control is not energy-efficient and result in higher communication delay than other available solutions. However, RSTP multisource transmission synchronization service may be beneficial for some types of WMSN applications, where a group of camera-equipped source nodes transmit still images to the sink.

Yaghmaee et al. propose the Queue based Congestion Control Protocol with Priority Support (QCCP-PS) [43]. That protocol extends the work in [51] defining a better treating of priority indexes, especially when random packet service time is considered. The idea of QCCP-PS transport protocol is to periodically assign a transmission rate to each transmitting node according to its priority (QoS) and the current congestion degree of the intermediate node. **Fig. 2** presents the conceptual operation of the proposed mechanism.

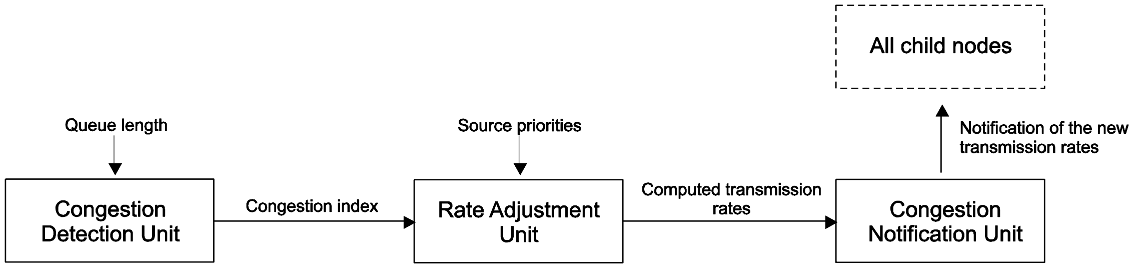

Fig. 2. Conceptual operation of QCCP-PS.

In [43] the congestion detection is performed by queue monitoring, where the level of perceived congestion is represented by a congestion index (a number between 0 and 1). Every node running QCCP-PS algorithm implements a separate FIFO queue for input packets from each direct child node. Additionally, one queue is created for local traffic of the sensor node. When detecting congestion, two thresholds are checked by the algorithm: max_{th} and min_{th}. If the current queue length is lower than min_{th}, the congestion index is very low and source node may increase its transmission rate. On the other hand, if the queue length is greater than max_{th}, source node should decrease its transmission rate to avoid any packet loss. When the current queue length is between min_{th} and $max_{th,}$ congestion index is a linear function of the current queue length.

Each node verifies its input queues in order to detect congestion. After that, the child nodes are notified of the computed desired transmission rate by an implicit mechanism, in order to reduce energy consumption. In that notification, it is indicated the new transmission rate that should be adopted by each child node.

The congestion mitigation mechanism in QCCP-PS is transmission rate adjustment based on the current congestion degree of the intermediate node and the priorities of the child nodes. The proposed algorithm maintains fairness among the child nodes, keeping the balance among the sending rates even after successive congestions.

The adoption of a priority for each node is very useful for wireless multimedia sensor networks. For example, sources streaming images (and nodes relaying them) may have a higher priority than audio streams, and audio nodes may have higher priority than sources sending scalar data. The outcome of the proposed congestion protocol in [43] is a fairer rate adjustment for heterogeneous sensor networks.

QCCP-PS protocol has some drawbacks for WMSNs. That protocol can bring very poor performance enhancement when most or all nodes have same priority for the application. An even more problematic characteristic of that protocol is its single-path nature. QCCP-PS considers only one path from the source to the sink. For some multimedia sensing applications with high bandwidth and low delay requirements, single-path routing may be inefficient.

Another protocol to mitigate congestion by reduction of the data transmission rate is presented in [44]. That work defined an Adaptive Compression-based congestion control Technique (ACT), a compression approach aimed at the congestion control with the reduction in the number of transmitted packets, but keeping quality of the received data. The proposed congestion mitigation approach discards packets containing less relevant data when the congested node may chose what packets must to be discarded. For that, the DWT (Discrete Wavelet Transform) technique indirectly defines priorities for the encoded data, while ADPCM (Adaptive Differential Pulse Code Modulation) reduces the amount of transmitted data from the source using the principle of quantization and RLC (Run Length Coding) generates a smaller number of packets for low-priority data. All these three techniques are combined to achieve congestion control with low impact to the application overall quality.

In [44] the source node assigns a priority value to each transmitted packet, according to the produced DWT subbands and theirs relevance to reconstruction of the original data. Such priorities are considered in the intermediate nodes in case of congestion, where packets with lower priorities are dropped instead of high-relevant packets. Moreover, ACT defines an adaptive operation of the queues, where control packets as congestion notifications have high priority and are relayed as soon as possible by the intermediate nodes.

Wavelet transforms provide data decomposition in multiple levels of resolution, where DWT is achieved discretely sampling the wavelets. DWT decomposes a signal (a series of digital samples) by passing it through two filters: a lowpass filter L and a highpass filter H. A 2D DWT processes an image considering the rows and columns, generating four subbands: LL, LH, HL and HH. The LL subband represents the lowest resolution and a half-sized version of the original image. In fact, it is the most significant information for the decoding process, while the remaining subbands contain vertical, horizontal and diagonal details of the image. Such processing produces two groups of relevance, but LL subband can be transformed again to generate more levels of resolution. In [44], the original image can be processed by a 2-level 2D DWT for the purpose of priority assignment. **Fig. 3** presents a visual representation of a 1-level and a 2-level 2D DWT.

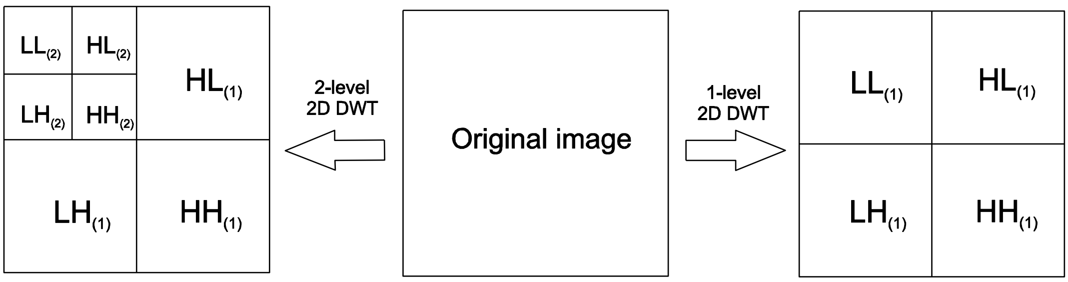

Fig. 3. DWT decomposition of an original image.

The compression approach proposed in [44] requires additional processing resources and energy of the nodes. However, authors argue that the energy spent for transmission is higher than the energy required for data compressing and processing. Thus, the reduced amount of data after compression can potentially decreases the energy consumption over the path, putting the proposed adaptive compression approach as a conceivable solution for congestion control in WMSNs.

3.1.2. Load Distribution

A cross-layer multipath congestion control algorithm is presented in [45]. In order to avoid direct reduction of the transmission rate as in [43] and packet dropping in congested nodes as in [44], which could potentially deplete the application quality, congestion is mitigated by load repartition over multiple paths. For that, the proposed congestion control algorithm expects some support of multipath routing from the network layer. For example, the network protocol SLiM [52] can be used to provide information on each node about the available paths.

The congestion detection in [45] is queue-based and the congestion notification is explicit. For congestion mitigation, that work defines three load balance strategies (modes 1, 2 and 3). In mode 1, source uses all available paths to the sink and the traffic is uniformly load balanced, even when there is no congestion. For modes 2 and 3, congestion is detected when the occupancy level in any intermediate node queue is higher than 80% or the collision rate in the link is higher than a given threshold. A node only reacts if it receives a notification and it is in the congested path (each node creates a table indicating the available paths and what are active). The difference between modes 2 and 3 is the way load balance is performed: mode 2 uniformly distributes the traffic among the available paths, while mode 3 considers the repartition of the current rate in the congested path over the available paths. For all modes, the original flow is only split in the source.

The experiments conducted in [45] showed that mode 1 (repartition without congestion) had a better performance (load fairness and packet dropping) than the other two modes, since it reduced the probability of congestion. However, it is showed that mode 1 is not energy-efficient, what can be prohibitive for WMSNs. Experiments also have demonstrated that mode 2 performs better than mode 3 in terms of packet dropping and mean consumed energy, but mode 3 presents a fairer load distribution. For all experiments, video-based source nodes were considered to be strategically positioned and intermediate nodes are randomly deployed.

The objective of the algorithm proposed in [45] is not to find the best paths, but achieve fast and fair load balancing after congestion. The load repartition strategies in that work reduced the packet dropping (the experiments were also compared with a single-path congestion control approach) without decreasing the source transmission rate.

Following the multipath structure adopted in that work, one node may belong to more than one path, depending on the nodes position after deployment and the available paths discovered by the routing protocol. In such case, nodes process traffic from more than one source, which could result in congestion, defining braided paths [35]. A different approach is to model node-disjoint paths, where intermediate nodes will be dedicated to a specific traffic. In other words, two (or more) paths are node-disjoint if they have no common intermediate nodes, strongly reducing the probability of node congestion (but we can still have link congestion). Fig. 4 presents an example of node-disjoint and braided paths composed using the same configuration of nodes.

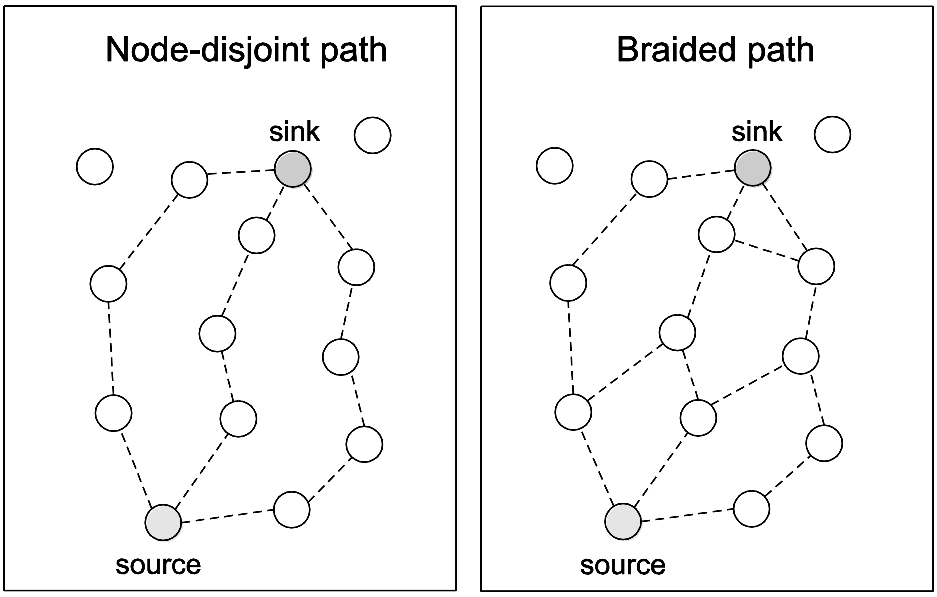

Fig. 4. Node-disjoint and braided paths.

The MPMPS (Multi-Priority Multi-Path Selection) algorithm finds the best paths for multimedia streaming in WMSNs, considering a set of available node-disjoint paths [46]. The best paths are those with lower end-to-end transmission delay, empirically measured by the number of hops. Additionally, that work proposes the splitting of video streaming in image and audio, giving to each resulting substream a particular priority. The best paths (lower delay) are used by the higher priority substream, letting the remaining paths to the lower priority substream.

Packets for video streaming generally are large in size and the transmission requirements can be several times higher than the maximum transmission capacity of the deployed sensor nodes. To address the application requirements, the source stream is load balanced, using the available paths discovered by routing protocols as the Two Phase geographical Greedy Forwarding protocol (TPGF) [53]. Since the end-to-end delay of the available node-disjoint paths can be significantly different, the algorithm presented in [46] reserves the paths with lower delays to the media stream (image or audio) with higher importance for the application. For example, in a WMSN deployed for fire monitoring, visual information is more relevant for the application and should be delivered with minimum transmission delay. The audio stream could be transmitted over the remaining paths.

Experimental results of MPMPS show a better performance (throughput) when compared with an algorithm that does not split the original stream in image and audio substreams. Since some available paths can have a transmission delay that does not fit the time constraints of particular streams, they are not used by traditional applications when they are sending only a combined video stream (audio and image together). On the other hand, for the multi-priority algorithm proposed in [46], even available paths with high delay could be used by the application for transmission of the lower priority substream, maximizing the attainable communication throughput.

The work in [46] does not directly propose a congestion control protocol, but just an algorithm for load balancing that can potentially reduce congestion and achieve a higher communication throughput. It resembles the mode 1 defined in [45], which uses all the available paths from the beginning and reduce the packet dropping on detriment of a greater energy consumption. As MPMPS regards node-disjoint paths, the congestion probability is potentially very low when the application knows the transmission capacity of each node of the paths.

Some drawbacks can be found when discussing MPMPS benefits. Initially, real-world WMSNs will not be so homogenous and the assumption that the transmission capacity of the nodes will be the same is unlike. Furthermore, congestion is not only a function of overload in intermediate nodes. In fact, congestion in only one link may prejudice the overall communication through the path, requiring proper congestion control mechanisms. Other unfeasible consideration for WMSNs is the use of GPS (Global Positioning System) to provide network topology discovery. GPS should not be employed for network topology and coverage discovery in WMSNs due to the lack of information about the orientation of the cameras, besides the cost and energy waste [54][55].

3.1.3. Network Congestion Avoidance

The Cross-Layer Active Predictive Congestion Control (CL-APCC) scheme has been proposed to avoid congestion by analyzing the memory status of single nodes, the current transmission trends and the average occupied memory of the local network [47]. Such analyzes are useful for the forecast of network congestion and for the dynamic adjustment of the transmission rate: the conditions in period t are used to predict the inputting and outputting rates of each node within the next period $t + 1$.

The overall network is divided in multiple grids, and CL-APCC is employed to avoid congestion in each grid. Every grid has an A-type node, a randomly selected sensor node that promiscuously monitors all nodes of the grid. The A-type is concerned with node-level congestion (packet dropping due to overflow in congested nodes) and system-level congestion (when packets from different sources collide). The monitoring by the A-type node will then be used to avoid congestion in the next transmission period, controlling the transmission rate of the nodes in the same grid.

The work in [47] defines an active and dynamic congestion control mechanism performed by intermediate nodes. However, it makes no mention to multimedia constraints, as high bandwidth and low delay requirements. Moreover, IEEE 802.11 is considered as the MAC-layer technology in that work, which is not an energy-efficient protocol. In fact, congestion avoidance mechanisms as proposed in [53] could benefit congestion control in WMSNs, but further analyses are still required, since we can not assure that such proposed method is suitable for real-world multimedia sensing applications.

In [48] authors define the Reliable Asynchronous Image Transfer Protocol (RAIT), a double sliding window method to avoid packet discarding due to congestion. In that work, each intermediate node implements a receiving queue and a sending queue. If the sending queue of an intermediate node is full, new packets should not be received by it for relaying, since such new packets are likely to be dropped (even if its receiving queue is not full). The packet flow is controlled by a token-bucket mechanism.

In order to avoid packet dropping due to congestion, the transmitting node has to know the current status of the next hop. Thus, the transmitting node must send a request to the next hop if it has packets to transmit but it has no token. Based on the available free space in the sending queue, the next hop sends tokens to the requesting node, enabling the transmission. If the replied token is equal to zero, the requesting node waits for a random time period before a new attempt. As different concepts are related in the proposed solution, the double sliding mechanism is based on cross-layer design, coupling MAC, network, transport and application layers.

Fig. 5 visually describes an example of how the proposed mechanism can avoid congestion. Node *i* have 4 packets to transmit and the next hop (node $i + 1$) has 4 free spaces in its receiving queue. However, as the sending queue of the next hop has 3 free spaces, there are only 3 tokens available. Node *i* must request tokens and at most 3 tokens will be assigned to it. Such operation avoids packet transmission bursts that may incur in congestion and packet dropping.

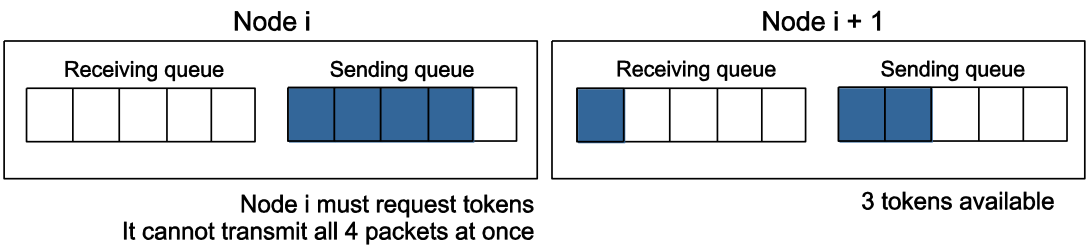

Fig. 5. RAIT token-based congestion avoidance mechanism.

The congestion avoidance mechanism in [48] is very promising but it may be unfeasible in some communication scenarios. Authors indeed argue that the proposed mechanism is intended for image transmission, where the transmission rate and bandwidth demands are less stringent than the video streaming requirements. Moreover, the token-based control can inflict in additional end-to-end delay, harming real-time sensing applications.

In a similar way, the CONtrol of SEnsor Queues (CONSEQ) [49] estimates the degree of congestion in order to avoid unnecessary energy depletion and delay. Authors advocate that 1-hop congestion estimation is more appropriated for wireless sensor networks, which is accomplished computing the virtual queue length of the next 1-hop nodes. The information about the queue current capacity is piggybacked to the previous hop in an ACK message, as depicted in **Fig. 6**. The virtual queue length is then computed using the piggybacked information, the node own queue length and the number of observed packets drops since the last successful transmission. With such information, the node can send packets to the next hop(s) with lower congestion probability, considering that there is more than one relaying node in its neighborhood in the path toward the sink.

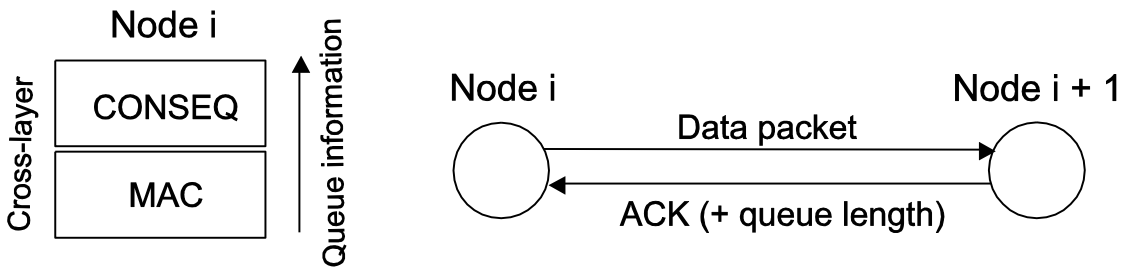

Fig. 6. Notification of the queue length in CONSEQ.

The probabilistic load balancing in [49] is very promising when wireless sensor networks are composed of braided-paths. When camera-equipped source nodes are deployed, the proposed congestion avoidance mechanism is even more beneficial, although more complexity is required for load balancing. Authors address such complexity applying fuzzy control theory and cross-layer design to efficiently schedule the packet transmission.

Information about the queue length of next hops is valuable for congestion avoidance, potentially benefiting wireless sensor network [48][49]. For WMSNs, congestion avoidance is even more beneficial, reducing the probability of packet losses and saving energy avoiding retransmissions. However, for applications that demand minimum communication throughput, the additional complexity and control imposed by the need for queue information may impact some multimedia sensing applications. As mentioned before, the application requirements will guide the choice of the more appropriate congestion control approaches.

3.1.4. Summarizing the Congestion Control Approaches

All proposed mechanisms for congestion control bring valuable contributions to congestion control in wireless multimedia sensor networks. Techniques that exploit the coding algorithms for higher efficiency are specially promising, as DWT for still image and predictive coding for video streaming, where energy saving or lower delay is achieved with a controlled loss of data quality.

Table 2. summarizes the surveyed transport protocols for congestion control. Note that queue-based congestion detection and explicit congestion notification are the most commonly adopted approaches for congestion control.

Table 2. Congestion control in WMSNs.

Protocol/Method	Congestion Detection	Congestion Notification	Proposed approach
RSTP [42]	Receiver-based	Explicit	TCP-ELN based congestion control. Image transmission synchronization based on progressive coding.
QCCP-PS [43]	Queue-based	Implicit	Rate adjustment considering source node priority and queue occupancy.
ACT [44]	Queue-based	Explicit	Congested nodes are relieved discarding low-relevant packets. Priorities based on DWT assure a controlled loss of quality in case of congestion.
Maimour et al [45]	Queue-based	Explicit	Multipath load distribution over braided paths regarding three different modes.
MPMPS [46]	-	-	Multipath load distribution over node-disjoint paths, with different priorities for image and audio streams.
CL-APCC [47]	Queue-based	Explicit	Rate adjustment for network congestion avoidance.
RAIT [48]	Queue-based	Explicit	Token-based flow control for congestion avoidance.
CONSEQ [49]	Queue-based	Explicit	The current queue occupancy is piggybacked allowing nodes to properly adjust the transmission rate.

3.2. Transport Protocols for Loss Recovery

Typically, packets are dropped when the network faces congestion, when bit-errors occur during packet transmission over wireless links (usually due to interference, signal attenuation or frame collision) or when intermediate nodes fail or run out of energy. Real-time multimedia communications on Internet backbones can generally tolerate some packet loss, but it is not commonly true for WMSNs.

We classify the loss recovery approaches in three categories:

a) *Retransmission of lost packets*: when packets are dropped, the sink or intermediate nodes can request the retransmission of the packets carrying the lost data. Retransmission can be performed in an end-to-end or hop-by-hop fashion.

b) *Correction codes*: although retransmission of lost packets is widely exploited by transport protocols, WMSNs can also benefit from loss recovery based on correction codes. FEC codes can be employed to protect the entire encoded data or more relevant bits can receive higher protection exploiting techniques as Unequal Error Protection (UEP) [56].

c) *Partial reliability*: a reliable communication service can be partially provided according to the application requirements. In such case, depending on the network condition, some part of the transmitted packets may be lost with low impact to the application quality. Reliable transmission will only be required for specific packets.

Our taxonomy for transport protocols that provide loss recovery is depicted in **Fig. 7**, where the surveyed works are outlined.

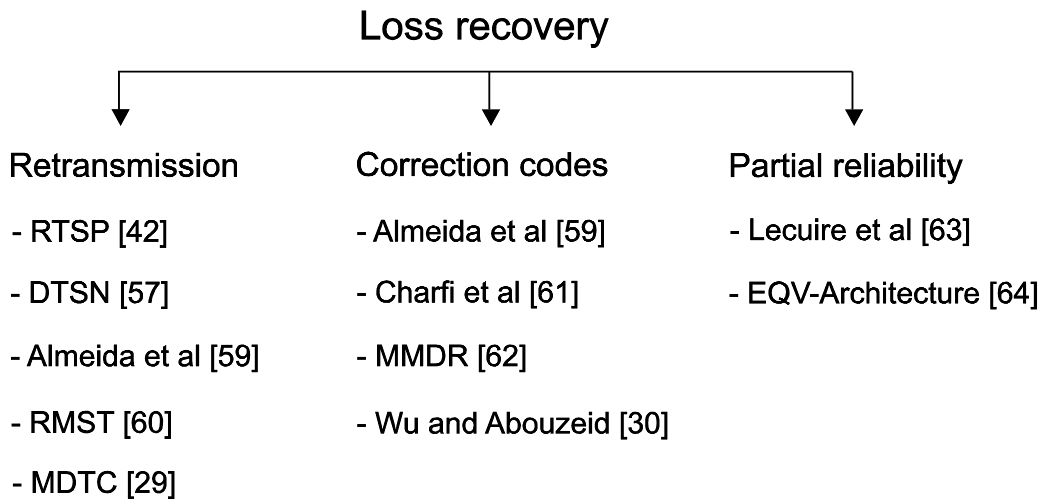

Fig. 7. The surveyed loss recovery mechanisms.

3.2.1. Retransmission of Lost Packets

As presented before, RSTP [42] is a transport protocol used for synchronization of image transmission from multiple sources, also providing a TCP-based congestion control. Additionally, RSTP provides a basic loss recovery service.

The loss recovery mechanism defined in RSTP is controlled by the receiver (sink). The receiver verifies the sequence number of received packets searching for any packet loss. If a gap in the sequence number is found, an explicit retransmission request is sent to the source node in the form of a negative acknowledgment (NACK). To properly initialize the sequence numbers, RSTP employs a three-way handshake to establish a communication, which is indeed initiated by the receiver.

A sequence number in the header of received packets is a conceivable inexpensive way to detect packet loss, but it is ineffective when the last packet of the stream is lost. In such cases, the sink sends a retransmission request when a specific counter timeouts. Nevertheless, a handshake is always necessary to properly initiate the sequence number, which could result in an undesired initial delay for real-time multimedia sensing applications.

Loss recovery in RSTP follows the end-to-end design concept, which can unnecessarily deplete energy of the intermediate nodes and incur in extra delay. End-to-end retransmission is only suitable for communications over wired links, where bit-errors rate is kept on a very low level. In fact, if RSTP is used as the transport protocol for a particular wireless multimedia sensor network application, it would probably be because of its source synchronization facility (its reliable communication service should be disabled for most WMSN applications).

In WMSNs context, retransmission of lost packets can also be accomplished by hop-by-hop packet caching. The work in [57] proposes the Distributed Transport for Sensor Networks (DTSN). That transport protocol provides reliable communications through hop-by-hop retransmissions, in a different way of [42]. The idea in [57] is that intermediate nodes can cache relayed packets for the purpose of data recovery in case of packet loss. When the sink identifies that there are missing packets, an NACK control message is transmitted to the source node. Intermediate nodes can intercept that message and transmit to sink the cached packet that match to the indication in the retransmission request. Doing so, the retransmission distance is reduced and energy is saved over the used path. Some theoretic and experimental verifications of DTSN are performed in [64].

An extension to DTSN is proposed in [58], where a hybrid solution (retransmission and correction code) for loss recovery is defined. In that work, the original data stream is divided on several logical blocks. For each block, the user application establishes its size (number of packets) and the percentage of received packets that represents the minimum required reliability. When sink receives more packets than the minimum reliability level, lost packets are not requested for retransmission. In such case, application can handle the received data even with missing information. Otherwise, sink notifies the sender through the reverse path and a hop-by-hop retransmission is performed.

The Reliable Multi-Segment Transport (RMST) is a transport protocol that also defines in-network caching of transmitted packets for the purpose of loss recovery [59]. However, intermediate nodes can detect packet loss and request retransmissions in a different way of RSTP and DTSN, where only sink requests retransmissions of lost packets. Packet losses in [59] are detected by proper timers enabled in intermediate nodes and notifications are performed by explicit NACK messages sent to the next hop on the reverse reinforced path toward the source. Additionally, RMST can also reassembly fragmented packets.

Fig. 8 presents typical uses of ACK and NACK messages for hop-by-hop retransmission. Packet transmission flows from node 1 to node *n*, and a packet corruption happens during transmission from node 3 to node 4. In the first example, gaps in the sequence of received packets are identified and an explicit NACK message is transmitted to request the missing data (packets). This is the case of RMST. In the second example, every transmitted packet is acknowledged. When no ACK message is received before a timeout, the packet is assumed to be corrupted/discarded and a retransmission takes place. In general words, the first option is more energy-efficient, while the second option is more robust and easier to implement.

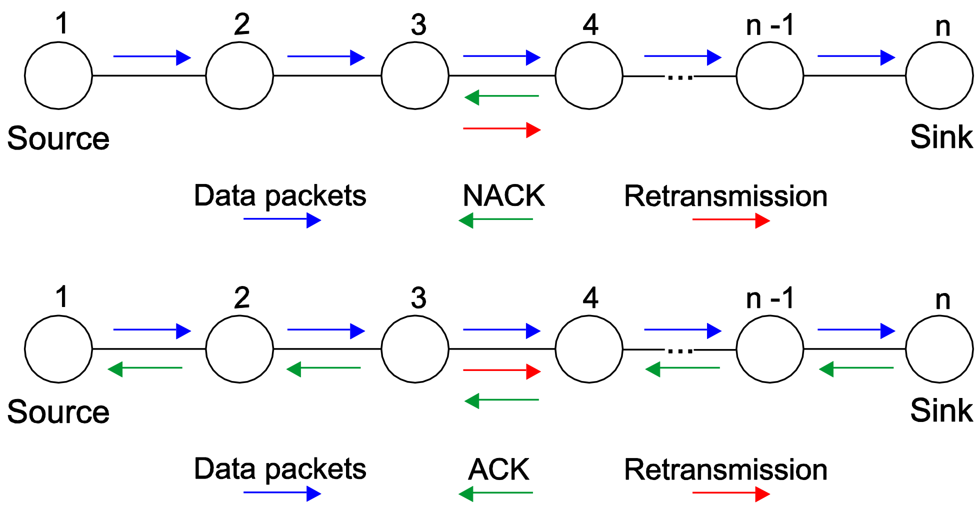

Fig. 8. NACK and ACK messages in hop-by-hop packet retransmission.

Hop-by-hop retransmission (also referred as local retransmission) can significantly reduce energy consumption and retransmission delay, but requires additional processing and energy consumption in intermediate nodes. Nevertheless, the cost of end-to-end retransmission and the impact of high delay are more harmful than the implementation of in-network caching by intermediate nodes [13][14][16].

The work presented in [59] also defines reliability guarantees expected from MAC layer and compares the performance of both approaches, showing that transport layer reliability usually performs better than loss recovery in MAC layer. However, authors argue that combining loss recovery in both MAC and transport layers can potentially result in higher performance for the application.

The concept of hop-by-hop retransmission can be further improved. The Multi-path-based Distributed TCP Caching (MDTC) [29] is a reliable and energy-efficient protocol that benefits from redundant paths and hop-by-hop retransmission for recovery of lost packets. The local retransmission algorithm is the same of [24], where TCP is improved to support hop-by-hop retransmission. For that algorithm, TCP segments should be cached at nodes as close to the receiver as possible, with extra care to cache segments that are likely to be dropped further along the path towards the receiver. Acknowledgement is performed by Selective ACK (SACK) messages, indicating the packets that are acknowledged even if the packet reception is scrambled.

A current path may be down if a single intermediate node gets congested or run out of energy. The multipath algorithm defined in [29] utilizes unused available paths from the source to the sink as redundant paths to mitigate congestion, avoiding additional packet loss. It is an interesting approach, but it is not indeed a load balance solution. For wireless multimedia sensor networks, redundant paths may be helpful for some applications, but their using have to be still carefully investigated.

Hop-by-hop retransmission proposed in [29][59] are very similar. Those works mainly differ in the using of multiple paths [29] and in the cross-layer interaction with MAC layer [59]. However, neither of them addresses directly high bandwidth and low delay requirements of typical wireless multimedia sensor networks, demanding additional investigation to adapt them to real-world WMSNs. However, their contributions influence the design of transport protocol for multimedia sensing applications.

3.2.2. Correction Codes

A correction mechanism for corrupted packets is described in [58]. The last packet of each logical block defined by the application provides a correction code to be processed at the sink. If only one packet of the block is lost, the missing information is reconstructed using such especial packet, even if the minimum reliability of the block has been assured. This is a soft way to increase the quality of the received media avoiding packet retransmission.

The use of blocks' minimum reliability is only feasible for real-world WMSN applications if proper multimedia codecs are employed, since only appropriate encoding can handle missing information in the received stream. Further, the correction mechanism may become useless if the packet containing the correction code is dropped during the transmission or when more than one packet is lost in the same logical block (due to error bursts or congestion).

Other strategies for data reconstruction in case of packet dropping can also be found on the literature. Charfi et al. [60] provide an end-to-end reliable communication service that does not require any retransmission from the source or intermediate nodes. In order to achieve loss recovery, N packets are transmitted to the sink, with L packets originated from the source stream and $N - L$ computed redundancy packets. The redundancy packets are obtained using a Reed-Solomon coder, based on the original multimedia data. To assure a reliable communication, the sum of received redundancy packets and ordinary data packets at the sink have to be at least equal to L. In such case, reconstruction of the original stream can be properly performed.

The authors in [60] propose the combination of multipath transmission with Forward Error Correction (FEC) code in redundancy packets to provide a reliable communication service with low delay in expense of additional energy consumption. More energy consumption is expected from the processing (coding and decoding) of the correction codes and the transmission of each redundancy packet over the selected paths, but retransmission is not required (saving energy). The tradeoff between the achievable reliability and energy consumption is widely investigated in that work through many experiments, which regard different packet error rates, algorithms for route selection and configurations for N and L variables.

The loss recovery mechanism specified in [60] is an end-to-end reliable service, since it is not expected any correction procedure from intermediate nodes or even from MAC error control facilities. In fact, in order to achieve the expected benefits by the adoption of that end-to-end loss recovery mechanism, retransmission should not be performed by MAC algorithms.

The proposed reliable communication in [60] is very promising, but it still needs further investigation to be widely used by real-world WMSN applications. One of the most critical issue is congestion, since redundancy packets increase the throughput over the sensor network. If not properly treated, congestion can deplete energy resources of intermediate nodes and disable routing paths, besides additional packet dropping. Particularly, excessive packet drops may turn useless the proposed reliable communication service.

A different correction-code approach is proposed in [61], where original video stream is split in multiple substreams according to their relevance for the decoding process. The authors define the Multipath Multi-stream Distributed Reliability framework (MMDR) to exploit multipath routing along with multi-stream packets to provide reliable video delivery in WMSNs, where the substreams are balanced over the available paths. Doing so, MMDR increases the probability that important video frames reach the sink.

One of the most relevant contributions of [61] is the use of distribute progressive error recovery. The idea is to use partial decoding at intermediate nodes to recover from channel induced errors, employing progressive error recovery algorithms. Since intermediate nodes can self-recover from link-layer errors, retransmission is not required. The channel coding of video data is performed using Low Density Parity Check (LDPC).

The recovery strategies adopted by [58][60][61] are based on reconstruction of original data after packet/frame drop/corruption, but they work in different ways. While the works presented in [58][60] follow an end-to-end reconstruction paradigm, Qaisar and Radha [61] define a hop-by-hop recovery mechanism. All those approaches may have different levels of impact on WMSNs, according to the application requirements and the nature of the deployed sensor network.

Wu and Abouzeid [30] consider local source encoding and in-network error recovery based on the transmission of redundancy packets. As retransmissions may incur in additional undesired end-to-end delay, authors propose correction mechanisms based on the transmission of redundancy packets through multiple paths and the use of FEC codes. Such codes are computed by each intermediate node instead of only in the source node and in the sink. Figure 9 presents an example of video streaming where intermediate nodes create and transmit redundant packets for reliability. The red arrow represents a packet corruption during transmission, but information is not lost since a redundant packet is received from the alternative path. The dashed circles define logical grids for the communication.

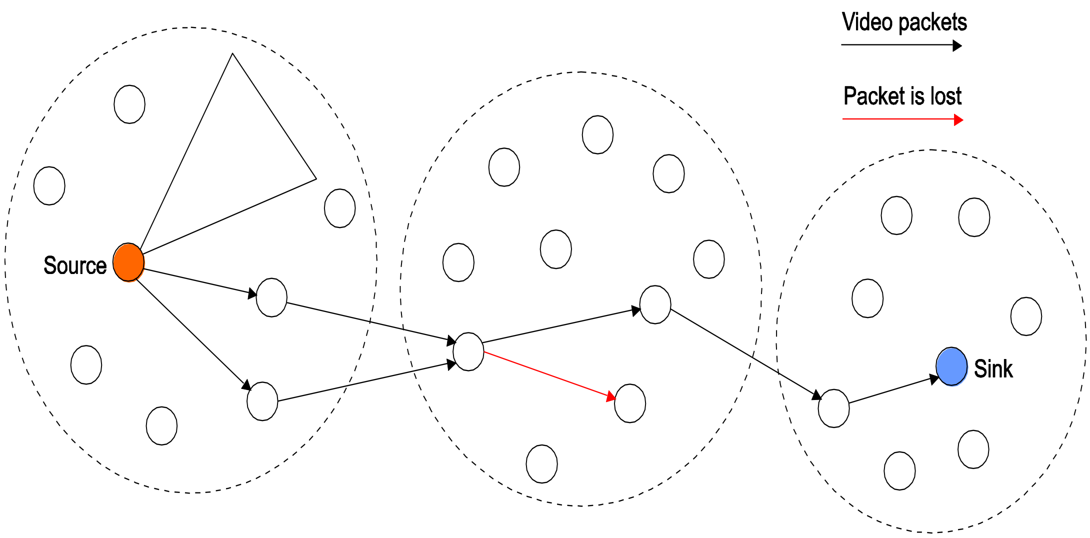

Fig. 9. Video streaming with in-network transmission and processing of redundant packets.

Source nodes encode the gathered multimedia data following quality requirements specified by the sink. After transmission, multiple copies of the same packet may be generated in the network. Such redundant packets may also be combined and processed, resulting in new multiple copies of the same packet.

Employing the proposed solution, errors are corrected as soon as possible or the packet is discarded if the correction can not be performed. In other words, it is performed hop-by-hop decoding and encoding: entire packets that are lost during transmission are recovered exploiting redundancy packets while bit errors are corrected exploiting the FEC code.

If only end-to-end correction would be employed, early corrupted uncorrectable packets could still be transmitted throughout the path, resulting in undesired energy wasting.

3.2.3. Partial Reliability

DWT technique can decompose an image into separated subbands for multi-resolution representation. Lecuire at al. [62] employ DWT to create packets with different levels of priority, allowing differentiated reliable transmission services over the sensor network. Intermediate nodes can decide to drop packets according to the packet priority and the remaining energy of the node, providing a reasonable tradeoff between the image quality and the network lifetime. In that proposed solution, dropped packets are not retransmitted.

A maximum priority value is defined in [62] to assure that every node forward very important packets for the decoding process, guarantying a minimum level of quality for the application. The packets with the remaining priorities are forwarded according to the available energy in the nodes and thresholds defined by the application.

Two general schemes are defined in [62]. In the open-loop scheme, it is only considered the energy of the node. On the other hand, close-loop also regards the available energy in the next intermediate nodes to the sink, which could help in prediction of the dropping probability of the transmitted packets.

The experimental results in [62] showed a good performance of the proposed solution (reasonable image quality reducing the energy consumption). In fact, energy saving is about 70% (open-loop) to 90% (closed-loop) with a soft reduction in the quality of the received image.

Following this same investigation line, the work in [63] proposes a video compression logical sub-layer which defines a new compression model able to prioritize frames. The proposed solution is defined as the Energy-efficient and high-Quality Video transmission Architecture (EQV-Architecture). The main idea of that work is to produce more relevant encoded video frames that should be transmitted through highly reliable schemes, while less relevant frames flow using a semi-reliable transmission service.

The type of the employed transmission service (reliable or semi-reliable) is structured over two packet dropping strategies: energy aware dropping and random early dropping. In the first one, the priority levels for packet dropping are computed regarding normalized energy level of the intermediate nodes, and each node has a particular priority level. In such approach, all received packets with priority level equal or lower than the current node priority are discarded. For this approach, the energy consumption over the network is considered, avoiding dropping packets that are closer to the sink. In the second dropping scheme, less relevant packets have a probability to be early dropped (before transmission from the source node), avoiding undesired energy consumption with low prejudice to the quality of the application.

Considering the proposed architecture in [63], it is expected that the energy consumption thresholds, the desired video quality and the bandwidth usage can be directly adjusted by the user application.

Partial reliability can considerably enhance the performance of wireless multimedia sensor networks. However, proper coding techniques as wavelet transform for image and predictive coding for video have to be employed at source nodes. Furthermore, the user application must tolerate the expected quality loss.

3.2.4. Summarizing the Loss Recovery Approaches

Scalar wireless sensor networks monitor information employing a large set of redundant source nodes. As almost the same information may be transmitted by many source nodes, loss recovery is not mandatory for most WSN applications. On the other hand, wireless multimedia sensor networks will typically have few camera-equipped source nodes that follow a directional sensing model [10]. In practical means, source nodes will typically have a unique view of the monitored field, putting loss recovery as a key service. Moreover, some parts of the encoded data (e.g. lower frequency DWT subbands and MPEG I-frames) have higher relevance for the reconstruction of the original data, demanding some type of reliable transmission service.

Table 3 summarizes the surveyed protocols for loss recovery.

Table 3. Loss recovery in WMSNs.

Protocol/Method	**Loss Detection**	**Loss Notification**	**Proposed Approach**
RSTP [42]	Sink	Explicit (NACK)	End-to-end retransmission.
DTSN [57]	Sink	Explicit (NACK)	Hop-by-hop retransmission.
Almeida et al. [58]	Sink	-	Hop-by-hop retransmission. Differentiated reliability service with a correction packet for each logical block.
RMST [59]	Intermediate nodes	Explicit (NACK)	Hop-by-hop retransmission.
MDTC [29]	Intermediate nodes	Explicit (SACK)	Hop-by-hop retransmission regarding redundant paths.
Charfi et al. [60]	Sink	-	Reconstruction of original data using redundancy packets.
MMDR [61]	Intermediate nodes	-	In-network error recovery at intermediate nodes.
Wu and Abouzeid [30]	Intermediate nodes	-	In-network error recovery at intermediate nodes.
Lecuire at al. [62]	-	-	Dropping of lower priority packets based on the available energy. Packets carry DWT encoded images.
EQV-Architecture [63]	-	-	Dropping of lower priority packets based on the available energy. Packets carry video frames.

4. Research Directions

In this paper we surveyed the state of the art of transport protocols for wireless multimedia sensor networks, covering congestion control and loss recovery issues. They are indeed very active research areas, but with many unsolved challenges. We outlined some promising investigations that should be addressed by the academic community in the upcoming years.

Congestion can rapidly degrade the energy resources of intermediate nodes and incur in packet dropping and communication delay. Among the proposed solutions, we can remark some promising approaches as multipath load balancing and packet prioritization according to the multimedia coding algorithm. But some research challenges still remain. For example, most of the investigations on the literature regard homogenous nodes with identical communication range and same initial energy supply. Deterministically deployed sensors are also considered by some works, what is unlikely for real-world WMSNs deployed in wide areas [10].

The current topology and logical structure of the deployed sensor may impact the adopted congestion control mechanism, also demanding further investigation. The work in [65] discusses how congestion control can be affected by the sensor deployment and network topology. Strategies for coverage preservation and energy saving as presented in [66] may also impact the way congestion is mitigated, since the current available paths may change when nodes become idle. Wireless multimedia sensor networks monitoring wide areas for a prolonged period of time will result in changeable network topologies, due to node failure, energy depletion or algorithms for energy preservation and balanced energy consumption. In fact, congestion control mechanisms have been treated as an isolated problem, leading to unreal assumptions that can turn unfeasible the proposed solutions for practical wireless sensor networks. Future works should address such issues.

Packets loss in wireless multimedia sensor networks can also strongly impact the end-to-end perceived quality of the received media. It is very clear for us that hop-by-hop retransmission is a reasonable basis for loss recovery in WMSNs, but inflicted additional delay has to be properly considered. Correction codes and transmission of redundancy packets can also play an important role in reliable communications.

In general, packet dropping in wireless multimedia sensor networks is resulted from network congestion or bit-errors during transmission. When packets are transmitted through error-prone wireless links, there is a probability for bit-errors. Many works expect a linear error probability over a single bit, but such consideration is unreal. In fact, it is more appropriate to assume that bit-errors appear in bursts and not in isolated bits. In such context, large packets have higher probability to get corrupted, when compared with small packets [23][67]. As smaller packets lead to additional protocol header overhead, future research should be worried about the ideal size of transport data units.

Future works should be also concerned in how to put together congestion control and loss recovery mechanisms, since they can benefit from each other. Transport protocols that offer both services and additionally exploit cross-layer design may bring high performance for wireless multimedia sensor networks. For example, a transport protocol may verify the current usage of receiving queues in link-layer and go beyond considering information from multi-channel coding and link-layer contention mechanisms, benefiting both congestion control and loss recovery procedures. Furthermore, congestion control and loss recovery approaches designed for scalar wireless sensor networks can be adapted to the transmission requirements of multimedia sensing applications, opening new investigation possibilities [68].

To the best of our knowledge, there is no performance comparison among the available transport protocols for WMSNs. Such analysis is strongly required to show the best methodologies for each communication scenario and to point out rewarding approaches. However, eventual performance comparisons should also regard MAC and routing protocols more adapted to WMSNs requirements. For example, the analysis of transport protocols conducted in [12] considered Wi-Fi as the link-layer protocol and Direct Diffusion in the routing layer [69]. Both IEEE 802.11 technologies and routing protocols as Direct Diffusion are not assumed to be employed in near future WMSNs [13][14][15].

We noticed that ideal multi-purpose protocols are missing and they should not indeed be proposed by the academic community. There are a lot of different applications concerning distinct monitoring tasks and employing a variable number of sensors of different types. We believe that high efficiency is only achieved when very specialized protocols are employed [16]. The transmitted requirements presented in Table 1 are a good indication of that.

Strategies for delay tolerant sensing applications may be adapted for WMSNs and new protocols from scratch may also be proposed. Moreover, other promising investigations will still arise in the upcoming years, for example changing the way multimedia data is sensed from the monitored field by the deployed sensor network [70][71]. Nevertheless, open research areas will often have to deal with energy and processing constraints of the nodes and the minimum required bandwidth of the applications.

5. Conclusions

Transport protocols for wireless multimedia sensor networks are expected to support congestion control and/or loss recovery services. In short, congestion can deplete energy of relaying nodes and result in packet dropping and undesired transmission delay. On the other hand, loss recovery is designed to guarantee a reliable communication service, once packets can be lost due to congestion or bit-errors during transmission over wireless links. In fact, the stringent requirements of WMSNs impose many challenges to the design of transport protocols.

In this paper, we surveyed many works comprising congestion control and loss recovery in transport layer or following a cross-layer design. The state of the art was presented and future research directions were discussed.

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