On Determining Bandwidth Usage Threshold to Support Real-Time Multimedia Applications in Wireless Multimedia Sensor Networks

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Abstract—Real-time multimedia applications require Quality of Service (QoS) provisioning in terms of bounds on delay and packet loss along with soft bandwidth guarantees. The shared nature of the available bandwidth in Wireless Multimedia Sensor Networks (WMSNs) causes interference. Interference combined with the overheads associated with a Medium Access Control (MAC) protocol limit the available bandwidth in WMSNs. These overheads can result in congestion, even if transmission rates of nodes are well below the maximum bandwidth supported by an underlying communication technology. Congestion degrades the performance of admitted real-time multimedia flow(s). Therefore, in this paper, we experimentally derive the IEEE 802.15.4 channel capacity using an unslotted Carrier Sense Multiple Access Collision Avoidance (CSMA-CA) MAC protocol. Considering the experimental channel capacity estimation results, and the characteristics of real-time multimedia flows, we determine the threshold on bandwidth usage, so that the QoS requirements of real-time multimedia flows in terms of delay, bandwidth, and packet loss rate can be met. We performed several simulations, and results show that QoS requirements of applications are met operating within the bandwidth usage threshold determined in this paper. Simulation-based results further show that marginally exceeding the determined threshold value results in the performance degradation of at least one real-time multimedia flow.

Index Terms—Wireless Multimedia Sensor Networks (WMSNs); IEEE 802.15.4 ; Effective Bandwidth ; Quality of Service (QoS)

I. INTRODUCTION

A Wireless Multimedia Sensor Network (WMSN) is an extension of a Wireless Sensor Network (WSN). The availability of low power CMOS image sensors, microphones, advancements made in coding theory [1], and digital signal processing chipsets have resulted in the development of wireless multimedia sensor nodes [2]. Wireless multimedia sensor nodes are capable of retrieving, processing, and wirelessly transmitting/receiving multimedia content such as audio, video, and still images. Therefore, WMSNs assist in high-resolution and multidimensional sensing.

High-resolution and multidimensional sensing characteristics of WMSNs have lead to their application in many real-time scenarios: visual surveillance [3], assisted living [4], and intelligent transportation [5] to name but a few. The application domains of WMSNs suggest that such networks can generate real-time multimedia streams [6]. Real-time multimedia streams require Quality of Service (QoS) provisioning especially in terms of bounded delay and packet loss rate along with soft bandwidth guarantees. Bandwidth is a shared and scarce resource in WMSNs, and interference along with the overheads associated with a Medium Access Control (MAC) protocol further limit the available bandwidth. The stated phenomenon results in congestion, even if nodes’ transmission rates are well below the bandwidth supported by the underlying communication technology. Congestion increases delay and packet loss, hence results in degradation in the performance of admitted real-time multimedia flows. Therefore, it is crucial for a QoS provisioning framework to determine a threshold on bandwidth usage. Exceeding this threshold can result in the performance degradation of real-time multimedia flows. To achieve this task, a QoS provisioning framework must determine the wireless channel capacity based on an underlying communication technology. The channel capacity estimation result helps to determine the bandwidth usage threshold, keeping in-view the channel throughput, delay, and packet drop rate corresponding to a certain value of offered data load. It is therefore important to determine a threshold on bandwidth usage because it can help to avoid congestion, hence it increases the chances to meet the QoS requirements of real-time multimedia flows.

Part of this work focuses on estimating the wireless channel capacity for IEEE 802.15.4-based network using the unslotted CSMA-CA MAC layer protocol. In IEEE 802.15.4, two MAC layer protocols have been standardized: (i) Slotted CSMA-CA, in which there is small support for real-time applications in the form of Guaranteed Time Slots (GTSs). However, slotted CSMA-CA is designed for start topology, hence does not suit well in random network topologies. (ii) Unslotted CSMA-CA which can work without a central coordinator, hence is a better choice for random network topologies. A number of proposals exists on providing service differentiation support in an unslotted CSMA-CA protocol, and details can be found in [7], therefore focus of this work is on an unslotted CSMA-CA MAC layer protocol.
The remainder of this paper is organized as follows. Section II presents related work. An estimate of the IEEE 802.15.4-based wireless channel capacity is presented in Section III. Section IV presents performance evaluations to demonstrate that the threshold is tight, and finally we conclude this paper in Section V.

II. RELATED WORK

The focus of this paper is on estimating the IEEE 802.15.4-based wireless channel capacity and setting the threshold on bandwidth usage in order to meet the QoS requirements of real-time multimedia flows. The purpose of determining the threshold on bandwidth usage is to limit data traffic in the network so that the QoS requirements of already admitted flows can be met. Therefore, in this section we briefly discuss state-of-the-art work focusing on analysing the IEEE 802.15.4 throughput along with admission control algorithms.

Testbed-based throughput measurements of IEEE 802.15.4 are presented in [8] and [9]. Furthermore, [9] also presents simulation-based throughput measurements of IEEE 802.15.4. Both research papers focus on the overall channel capacity. The testbed results reported in [8] and [9] show that the upper limit on channel throughput is in the range of 35 to 40 kbps. The simulation-based results reported in [9] show that the maximum channel capacity is approximately 65 kbps. Both [8] and [9] focus on deriving the IEEE 802.15.4 throughput, none focuses on setting a bandwidth usage threshold exceeding which can degrade the performance of real-time multimedia flows.

In [10] an analytical throughput analysis of the slotted CSMA-CA MAC layer protocol of IEEE 802.15.4 is presented. An analytical model typically requires simplifying assumptions to produce results. In real WSNs, these assumptions may not be true [11], hence such analysis may not accurately predict achievable throughput. Furthermore, this work only focuses on maximum achievable throughput using the IEEE 802.15.4 standard. No consideration is given to determine the bandwidth usage threshold exceeding which can result in congestion inside a network.

Contention-Aware Admission Control Protocol (CACP) [12] provides admission control functionality in Mobile Ad-hoc Networks. As part of the protocol, CACP presents bandwidth estimation techniques for wireless ad-hoc networks. CACP suggests that bandwidth estimation can be done by monitoring the wireless channel i.e., bandwidth available to the node is the fraction of time when the wireless channel is idle. Moreover, CACP presents another technique for bandwidth estimation i.e., each node monitors its bandwidth usage and informs nodes within its two-hop distance about it. The drawback associated with this scheme is that the impact of a MAC layer protocol is not considered. E.g., if a CSMA-based MAC layer is used, an idle channel does not necessarily mean it is available for transmission as there is a possibility of all nodes backing off as part of the channel access attempt. Nodes share their bandwidth usage information with other nodes, but they neglect the time spent in the back-off mode, hence this can result in an overestimate of the available bandwidth. Neglecting MAC layer overhead can result in overestimating the available bandwidth, hence already admitted real-time flows experience a performance degradation when a new flow is admitted based on the bandwidth estimation technique used by CACP. Furthermore, it is possible that a bandwidth estimation technique informs that X bps are available, but the bandwidth estimation technique is not aware of an increased delay and packet drop rate associated with an additional offered load of X bps. This again can result in the performance degradation of admitted real-time multimedia flows, if a new flow is admitted.

Adaptive Admission Control (AAC) for ad-hoc and sensor networks providing Quality of Service is presented in [13]. A node running AAC keeps a record of the number of bits sent by the node and the number of bits sensed by the node by monitoring the wireless channel. The node adds the two listed quantities and subtracts the resultant number from the system’s available bandwidth. The downside of AAC is that it does not consider the impact of the MAC layer on the available bandwidth. Moreover, it does not consider the impact of admitting a new flow on delay and packet drop rate.

Stateless Wireless Ad-hoc Network (SWAN) [14] is a distributed network model designed to provide QoS in Mobile Ad-hoc Networks. SWAN uses a bandwidth-based admission control mechanism. SWAN’s admission control mechanism admits a new real-time flow if a node is operating below a bandwidth knee, i.e., a point beyond which congestion can occur in a network. Therefore, for proper functionality of SWAN, a node needs to estimate the bandwidth knee.

III. IEEE 802.15.4 CHANNEL CAPACITY ESTIMATION

To estimate the IEEE 802.15.4-based WSN channel capacity, and relationship of delay and packet loss rate with offered traffic load, we simulated a WSN with eleven nodes. All nodes are within the transmission range of each other. Ten nodes act as transmitters and one node acts as a receiver. We increase the total offered data load in the network from 20 to 200 kbps (offered data load is uniformly distributed among 10 transmitters). Each simulation scenario is repeated three times, and averaged results are reported to account for the random nature of the CSMA-CA protocol. Table I shows general simulation parameters. Simulations are performed using the Cooja WSN simulator.

From Table I it can be seen that we are using full size IEEE 802.15.4 packets to estimate the IEEE 802.15.4 channel capacity. If we use short IEEE 802.15.4 addressing mode, 102 bytes of application data is carried in MAC layer frame using the Rime protocol stack of Contiki operating system. Invariably, multimedia applications can generate bulk of data, therefore it is not rare that such applications utilize 102 bytes in each transmitted packet. Fig. 1 shows the average channel throughput w.r.t. the offered data load.

It is evident from Fig. 1 that the rate at which the channel throughput increases till the offered data load reaches 100 kbps is almost linear with offered data load. When the offered data load ranges from 100 to 180 kbps, the slope of the line
TABLE I
GENERAL PARAMETERS FOR SIMULATION

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>MAC layer</td>
<td>CSMA-CA</td>
</tr>
<tr>
<td>MAC layer reliability</td>
<td>Disabled</td>
</tr>
<tr>
<td>Radio duty cycling algorithm</td>
<td>Null radio duty cycling</td>
</tr>
<tr>
<td>Radio Model</td>
<td>Undirected Graph Model</td>
</tr>
<tr>
<td>MAC layer queue size</td>
<td>30 packets</td>
</tr>
<tr>
<td>Bit rate</td>
<td>250 kbps</td>
</tr>
<tr>
<td>Node transmission range</td>
<td>50 meters</td>
</tr>
<tr>
<td>Node carrier sensing range</td>
<td>100 meters</td>
</tr>
<tr>
<td>Total packet size</td>
<td>127 bytes</td>
</tr>
<tr>
<td>Simulated node type</td>
<td>Tmote sky</td>
</tr>
</tbody>
</table>

Table 1 shows the general parameters for simulation, including MAC layer, MAC layer reliability, Radio duty cycling algorithm, Radio Model, MAC layer queue size, Bit rate, Node transmission range, Node carrier sensing range, Total packet size, and Simulated node type.

In Fig. 1, offered data load vs. throughput is shown, indicating that average channel throughput increases slowly. This is primarily due to the distributed nature of the CSMA-CA protocol. As data load in the WSN increases, each node has more data to send, hence nodes frequently contend for channel access, and in this process nodes go to back-off mode more frequently. It results in an increased delay in getting access to the channel, hence channel throughput increases slowly. In fact, in simulation results, we observed that an offered data load of 180 kbps acts as a threshold point in terms of offered data load, after which average throughput starts to decrease. This is primarily due to the CSMA-CA protocol and its back-off mechanism.

Fig. 2 shows the relationship of delay with the offered data load. It can be observed that average per-packet delay does not increase a lot as long as the offered data load ranges between 0 to 100 kbps. Beyond that point, average per-packet delay increases sharply. From Fig. 1 and Fig. 2, it can be concluded that channel capacity is 118 kbps, and it is achieved at an offered data load of 180 kbps. At an offered data load of 180 kbps, the per-packet delay is approximately 70 ms. This indicates the fact that operating below the bandwidth supported by the underlying communication technology results in congestion, hence increased delay and packet loss rate. Real-time multimedia flows require bounds on delay and packet loss rate, therefore an offered data load of 60 kbps can act as the bandwidth usage threshold as at this point per-packet delay is about 30 ms and per-hop packet drop rate is 8 percent.

IV. PERFORMANCE EVALUATION

In this section, we simulate a multi-hop WSN and we assume that nodes inside the network generate real-time multimedia flows. Real-time multimedia flows can tolerate some packet loss, but require bounded delay. Therefore, in our experiments, we assume that a real-time multimedia flow can tolerate approximately 8 percent per-hop packet loss and 30 ms of delay at each node. As per the experiments result shown in Fig. 1 and Fig. 2, offered data load inside the network should not exceed 60 kbps to provide an acceptable level of service to real-time multimedia flows. To determine the effectiveness of results presented in Fig. 1 and Fig. 2, we created three simulation scenarios. Each simulation scenario is repeated three times, and we report average results in this section. In each simulation scenario, average per-packet delay (total time spent by a packet in the MAC layer queue), and nodes average throughput between the simulation time interval of 20 to 100 seconds is measured at each node. Table I lists general simulation parameter. Simulations are performed on the Cooja WSN simulator. Moreover, Table III shows nodes within the interference range of each node present in the simulated network.

A. Scenario 1

Table II summarizes the flows in simulation scenario 1. It is evident from Fig. 5 that the average per-packet delay at each node is less than 30 ms corresponding to Scenario 1. Data load within the interference range of nodes 1, 2, 3, 4, 5, 6, 7, 8 and 9 is 40, 50, 50, 40, 30, 50, 30, 40, and 20 kbps respectively. Fig. 1 does not plot average throughput corresponding to the offered data load of 30 kbps. Considering Fig. 1 it can be
observed that the average throughput for an offered data load between 20 and 40 kbps increases almost linear. Therefore, we can use Equation 1 to estimate packet loss rate at an offered data load of 30 kbps via linear interpolation.

\[
\frac{x - x_1}{x_2 - x_1} = \frac{y - y_1}{y_2 - y_1}
\]

In this case, \(x_1 = 20, x_2 = 40, y_1 = 19.7,\) and \(y_2 = 38.\) Solving Equation 1 for the given values yields: \(y = 0.915x + 1.4.\) The average throughput for an offered data load of 30 kbps is 28.85 kbps, therefore per-hop packet drop rate is 3.3 percent. Similarly, Fig. 1 does not plot average throughput for an offered data load of 50 kbps, but we can solve Equation 1 using the closest two data points: \(x_1 = 40, x_2 = 60, y_1 = 38,\) and \(y_2 = 55.\) Solving Equation 1 for these values yields: \(y = 0.85x + 4.\) Therefore, the estimated per-hop packet drop rate is 7 percent for the offered data load of 50 kbps. The average number of bits that node 4 anticipates to receive as per the result present in Fig. 1 can be calculated as: \((10 \times 0.95 \times 0.93 \times 0.93) = 8.21\) kbps. Simulation results show that node 4 received 9.2 kbps. Hence, the per-hop packet loss rate for a flow originating from node 1 is certainly below 8 percent. Similarly node 9 must receive \((10 \times 0.95 \times 0.93) = 8.83\) kbps. Simulation results show that the average number of bits received by node 9 is 9.80 kbps. Therefore, the QoS requirements in-term of delay and per-hop packet loss rate are met for both flows.
Table IV summarizes the flows in simulation scenario 2. In this scenario, the maximum data load that can be ideally observed at nodes 1, 2, 3, 4, 5, 6, 7, 8, and 9 is 45, 60, 60, 40, 50, 40, 60, 40, and 20 kbps respectively. Data load within the interference range of nodes 2, 3, and 7 is maximum i.e., 60 kbps w.r.t. the chosen threshold level. In our experiments the average number of bits received at node 4 is 12.96 kbps. Node 4 receives 8.24 kbps for the flow originating from node 1, and 4.72 kbps for the flow originating from node 5. From Fig. 1 we can see that at the offered data load of 40 kbps, the packet drop rate is approximately 5 percent. Fig. 1 does not plot average throughput for the offered data load of 45 and 50 kbps, but if we consider the line joining points showing average throughput for the offered data load of 40 and 60 kbps, it is almost linear. Therefore, we can use Equation 1 to estimate average packet loss at an offered data load of 45 and 50 kbps, as done before.

The estimate of the average throughput for an offered data load of 45 kbps is 42.25 kbps, therefore in this case, we estimate that the per-hop packet drop rate is 6 percent. The per-hop packet drop rate for the offered data load of 50 kbps is 7 percent. If we use these derived values, the average number of bits that node 4 expects to receive for the flow originating from node 1 is: 
\( (10 \times 0.94 \times 0.92 \times 0.92) = 7.66 \) kbps, which is lower than 8.24 kbps, and the average per-packet delay at nodes 1, 2, and 3 is less than 30 ms as shown in Fig. 5, hence the QoS requirements of the flow originating from node 1 are met. Similarly, node 4 should at least receive 
\( (5 \times 0.93 \times 0.95) = 4.42 \) kbps. But our results show that node 4 has received 4.72 kbps, which is more than 4.42 kbps. Moreover, Fig. 5 shows that the average per-packet delay at nodes 5 and 6 is less than 30 ms. Hence, the QoS requirements of the flow originating from node 5 are also fulfilled. Node 9 is the destination of the flow originating from node 7, and as per our simulation results, it has on average received 9.80 kbps. Considering the results presented in Fig. 1 and the data load within the interference range of nodes 7 and 8, node 9 should have received 8.74 kbps on an average. In this case, node 9 has received 12 percent more data compared to the results presented in Fig. 1. The average per-packet delay as per Fig. 5 is less than 30 ms at nodes 7 and 8, hence the QoS requirements for the flow originating from node 7 are also met.

In this scenario, the average number of bits received at destination nodes is more than what is anticipated. In both scenarios we have observed that the packet drop rate is lower than what is shown in Fig. 1. One can consider this a good sign, because QoS requirements of real-time multimedia flows are fulfilled. But at the same time, one can argue that our threshold of 60 kbps is overly conservative, hence we are missing opportunity to admit more flows in a network. Therefore, to analyse the impact of operating marginally above the threshold, let us consider Scenario 3.

<table>
<thead>
<tr>
<th>Source Node</th>
<th>Destination Node</th>
<th>Start Time (Sec)</th>
<th>Pkts/Sec</th>
<th>Total Packets to Transmit</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>4</td>
<td>4</td>
<td>10</td>
<td>1000</td>
</tr>
<tr>
<td>7</td>
<td>9</td>
<td>10</td>
<td>10</td>
<td>1000</td>
</tr>
<tr>
<td>5</td>
<td>4</td>
<td>15</td>
<td>5</td>
<td>500</td>
</tr>
</tbody>
</table>

Table V summarizes the flows in simulation scenario 3. In this scenario, the maximum data load that can be ideally observed at nodes 1, 2, 3, 4, 5, 6, 7, 8, and 9 is 47, 64, 64, 44, 54, 44, 64, 40, and 20 kbps respectively. Data load within the interference range of nodes 2, 3, and 7 is maximum i.e., 66 kbps w.r.t. the chosen threshold level. In our experiments the average number of bits received at node 4 is 12 percent more data compared to the results presented in Fig. 1. The average per-packet delay as per Fig. 5 is less than 30 ms at nodes 7 and 8, hence the QoS requirements for the flow originating from node 7 are also met.

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<tr>
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<td>10</td>
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<td>1000</td>
</tr>
<tr>
<td>5</td>
<td>4</td>
<td>15</td>
<td>7</td>
<td>700</td>
</tr>
</tbody>
</table>

B. Scenario 2

C. Scenario 3

Table V summarizes the flows in simulation scenario 3. In this scenario, the maximum data load that can be ideally observed at nodes 1, 2, 3, 4, 5, 6, 7, 8, and 9 is 47, 64, 64, 44, 54, 44, 64, 40, and 20 kbps respectively. For simplicity we assume that the packet loss rate corresponding to the offered data load of 44, 47, 54, and 64 kbps is the same as for the offered data load of 45, 45, 55, and 65 kbps respectively. Per-hop packet drop rate corresponding to the offered data load of 45 kbps is 6 percent, as derived in Scenario 2. We need to solve \( y = 0.85x + 4 \) to estimate the per-hop packet loss rate corresponding to the offered data load of 55 kbps, which is 7.72 percent. Similarly, for the offered data load of 65 kbps we need to solve Equation 1 with values: \( x_1 = 60, x_2 = 80, y_1 = 55, \) and \( y_2 = 65 \) resulting in \( y = 0.5x + 25 \). Hence, per-hop packet loss rate when the offered data load is 65 kbps is 11.5 percent. Node 4 has received 13.41 kbps, 6.96 kbps for the flow originating from node 1 and 6.46 kbps for the flow originating from node 5.

The average number of bits that node 4 should expect to receive for the flow originating from node 1 is 
\( (10 \times 0.94 \times 0.885 \times 0.885) = 7.36 \) kbps. Similarly, as per Fig. 1, the average number of bits that node 4
The expected data rate is: 

\[ \text{Expected rate} = (7 \times 0.923 \times 0.94) = 6.10 \text{ kbps} \].

Our results show that node 1’s flow has suffered more packet loss, moreover Fig. 5 shows that the average per-packet delay at nodes 2 and 3 has significantly increased. Therefore, in this case the flow originating from node 1 experiences degradation in its performance. Node 9 can expect to receive: 

\[ (10 \times 0.885 \times 0.95) = 8.41 \text{ kbps} \]

but in simulation node 9 receives 9.65 kbps, which is 15 percent more than what is expected. Nevertheless, we have shown that exceeding 60 kbps deteriorates the performance of at least one real-time multimedia flow. Furthermore, Fig. 5 shows increased delay especially at nodes 2 and 3 which further supports the tightness of the chosen threshold presented in this paper.

V. Conclusions

It has been shown in this paper that determining the threshold on bandwidth usage keeping in view the wireless channel capacity is essential for QoS provisioning in WMSNs. We experimentally determine the IEEE 802.15.4-based wireless channel capacity using the unslotted CSMA-CA MAC protocol. The threshold on bandwidth usage is determined based on the plots showing the relationship of offered data load with delay and throughput, and considering the characteristics of real-time multimedia flows. It has been shown that the bandwidth usage threshold of 60 kbps meets the QoS criteria that we set for real-time multimedia flows. Furthermore, it has been shown that marginally exceeding 60 kbps results in the performance degradation of an already admitted flow. Moreover, using our plotted relationship of offered data load with throughput and delay, the bandwidth estimation threshold for other types of data traffic can also be determined. The results presented in this paper can serve as an input to an effective and efficient admission control algorithm for WMSNs.

References


