# A Congestion Control Scheme for (m,k)-firm Real-Time Streams in Wireless Sensor Networks

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Abstract— In this paper, we design an (m,k)-firm[1] based realtime congestion control scheme to make the prior protocol be able to handle both link-level and node-level congestions during packet transmissions. Simulation results show that for real-time applications, the proposed congestion control scheme could effectively avoid dynamic failures and guarantee end-to-end QoS requirements.

Keywords—congestion control, (m,k)-firm streams.

#### I. Introduction

In our prior work, we have designed a novel real-time routing protocol for (m,k)-firm stream in [2]. This protocol is used to overcome the main drawback of existing works that it can provide firm real-time OoS instead of soft real-time OoS, with the help of G DBP (global-DBP) value and a newly employed local transmission status indicator called "L DBP (local-DBP)". L DBP can make the intermediate nodes be aware of local stream transmission status, and implement different fault recovery schemes according to the fault category it indicates such as congestion and link failure. In this paper, we present a real-time congestion control scheme for (m,k)-firm streams to improve the performance of our prior routing protocol. In accordance with the features of sensor networks, a new node model including two queuing buffers for both source traffic and transit traffic is used in this scheme. Each node can dynamically adjust its source/transit traffic rates according to the related L DBP and G DBP values.

## II. EASE OF USE

## A. Node Model

A new node provides two queuing buffers for 1) source traffic generated by node itself; 2) transit traffic that node receives from upstream nodes. By using this node model, one node i can adjust its source traffic sending rate  $r\_src_i$  and transit traffic forwarding rate  $r\_trs_i$  separately. The outgoing traffic rate of node i can be calculated by adding the two traffic rates  $(r\_out_i = r\_src_i + r\_trs_i)$ .

# B. Rate Adjustment

G\_DBP and L\_DBP values are used in two rate adjustment algorithms to limit the source traffic rate and source/transit traffic rates, for both sink-source node system and intermediate nodes system, respectively. Since end-to-end

QoS performance can be only evaluated at sink, the first metric for source rate adjustment should be G\_DBP value which is obtained by calculating the DBP value of each stream, using the following equation mentioned in [2].

$$G_{DBP_{s(x)}} = k_{s(x)} - l_{s(x)}(m_{s(x)}s) + 1$$
 (1)

where  $G\_DBP_{s(x)}$  is the measured DBP value of stream x at sink,  $k_{s(x)}$  comes from the required (m,k)-firm of stream x,  $l_{s(x)}(m_{s(x)},s)$  denotes the position (from the right) of the mth deadline meeting in the current state s of stream x.

Sink sends back the measured  $G_DBP_{s(x)}$  and an adjusted source traffic rate  $r_adj_src_i$  to the corresponding source node i in a small predefined time interval. When source node receives  $G_DBP_{s(x)}$ , it adds the value into the packets it generates. The adjusted source traffic rate  $r_adj_src_i$  is calculated according to  $G_DBP_{s(x)}$  and is supposed to adapt the traffic load to network capability and acceptable QoS.

ALGORITHM 1: SINK-SOURCE NODE SYSTEM RATE ADJUSTMENT

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G\_DBP_{s(x)}: evaluated G\_DBP value of stream x k_{s(x)}: k value from required (m,k)-firm of stream x r\_min\_src_i: minimum source traffic rate of node i r\_src_i: current source traffic rate of node i r\_adj\_src_i: adjusted source traffic rate of node i
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#### PSEUDO-CODE RUNS AT SINK IN EACH ROUND

1: if  $G\_DBP_{s(x)} \le 0$  //stream cannot meet requirement 2: if  $r\_src_i > r\_min\_src_i$  //current source traffic rate can be reduced 3:  $r\_adj\_src_i = \max\{r\_src_i \cdot (1 - \underbrace{k_{s(x)} - |G\_DBP_{s(x)} - 1|}_{k_{s(x)}}), r\_min\_src_i\}$ 

4: end if 5:end if

In order to reduce the network traffic load and satisfy required QoS performance at the same time, a minimum source traffic rate is defined to limit the multimedia distortion caused by low source traffic rate. Therefore, when sink monitors that  $G\_DBP_{s(x)}$  is no more than 0, which indicates stream x is in negative condition, it will adjust the corresponding source traffic rate to a particular level, but not less than the minimum rate. The calculation of adjustment is based on the ratio of deadline meetings to monitored consecutive packets. Then, the adjusted source traffic rate will be sent back to source node to implement traffic limitation.

Considering the characteristic of multimedia application that big volumes of data are generated in a short period, it's possible that only sink-source node system rate adjustment is not sufficient to achieve congestion control. Thus, in the proposed congestion control scheme, local system such as intermediate nodes also participates in end-to-end QoS guarantee, by contributing a local congestion control mechanism.

Local congestion control mechanism is implemented at intermediate nodes, by reducing both source/transit traffic rates to adapt the local traffic load to node capability and eventually mitigate congestions. Due to the features of WSNs such as wireless natures and limited resources, there exist two types of congestions: link-level congestion and node-level congestion [3]. We use the novel transmission status indicator L\_DBP from our prior work [2] to detect the link-level congestion. In this letter, we directly use the measured L DBP value and assume that all degradations are caused by congestion. The transit traffic buffer status of node i, named as buff<sub>i</sub>, is used to monitor the node-level congestion. It could be sent as a beacon by node i to its neighbors. We argue that this beacon would not involve additional energy consumption since piggybacking scheme is used. The algorithm shown below is supposed to be able to detect both two types of congestions and then implement a 2-step mechanism to adjust source/transit traffic rates. If congestion is not mitigated after this 2-step mechanism, a congestion notification will be propagated to the one-hop further upstream node in a backpressure manner, to make it run the same algorithm to control the traffic.

**STEP 1**: similar to sink-source node system, in case of only link-level congestion happens, upstream node i will first decrease its own source traffic rate according to the local transmission status. Thus, the outgoing traffic rate of node i can be reduced to an acceptable level based on the value of L\_DBP and minimum source traffic rate.

**STEP 2:** if congestion is not mitigated after the source traffic rate is reduced to a minimum acceptable level, or nodelevel congestion happens at downstream node, the second step will be taken to limit the transit traffic from upstream nodes to the congested downstream node. The weight of each upstream node i is measured according to the total L\_DBP values of all streams passing by, and also the outgoing traffic rate of downstream node.

#### III. PERFORMANCE EVALUATION

Performance of the proposed scheme is evaluated by NS-2. During all simulations, we use a uniform topology which includes 100 nodes in an area of 200m x 200m. Propagation model is set to be Two-Ray Ground, protocols for physical and MAC layer are set to be wireless-phy and 802.11, respectively. Radio range is set to be 50 meters, for nodes to transmit 1000 bytes packets on the bandwidth of 2Mb/s. Fig. 3 shows the QoS performance of two schemes by representing the end-to-end dynamic failure ratios.

Since we have learned from the last simulation that serious congestions happen during the increasing of cross-traffic streams and lead to a large number of packets deadlines missing. At the meantime, decline of packet reception rate is caused by packet drops of the overloaded traffic, accordingly. Thus these two factors of real-time end-to-end dynamic failure largely degrade the QoS performance of the evaluated stream. In our prior protocol, this degradation is more critical and inevitable, since it doesn't have the functionality to adjust traffic rate and mitigate congestion by reducing traffic load. By contraries, the proposed real-time congestion control mechanism works well in the improved scheme, for the reason that it could adjust the traffic load according to the network capability and acceptable QoS performance. Simulation results reveal that for real-time applications, their challengeable issues such as bursty of traffic and timeliness could be well handled by the proposed real-time congestion control scheme.

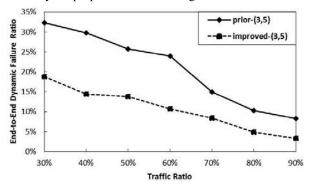


Fig. 1. Stream end-to-end dynamic ratio

#### IV. CONCLUSION

In this paper, we improved the prior work with a new realtime congestion control scheme. Based on (m,k)-firm, the proposed scheme can provide source/transit traffic rates adjustment for both sink-source node system and intermediate nodes system, to limit the traffic load of network. Simulation results demonstrate that the proposed congestion control scheme has greatly improved the performance of prior work.

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#### REFERENCES

- M. Hamdaoui and P. Ramanathan, "A dynamic priority assignment technique for streams with (m, k)-firm deadlines," IEEE Trans. Computers, vol. 44, pp. 1443-1451, Dec. 1995.
- [2] B. J. Li and K. Kim, "A novel routing protocol for (m,k)-firm-based real-time streams in wireless sensor networks," in Proc. of IEEE Wireless Communications and Networking Conference, Paris, Apr. 1-4, 2012
- [3] M. H. Yaghmaee and D. Adjeroh, "A new priority based congestion control protocol for wireless multimedia sensor networks," in Proc. of IEEE Int. Symp. World of Wireless, Mobile and Multimedia Networks, Newport Beach, 2008, pp. 1-8.