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Cross-Layer Design for QoS Support in Wireless Multimedia Sensor Networks

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***Abstract*—The rapid growth in micro-electronics technology and research in wireless sensor networks (WSNs) has made it possible to realize multimedia delivery on wireless multimedia sensor networks (WMSNs), consisting of tiny sensing devices. The volume and characteristics of multimedia data produced by WMSNs is quite different from the scalar data generated by conventional WSNs. This has raised the need to explore efficient routing protocols for multimedia data delivery in WMSNs, particularly supporting stringent quality of service (QoS). In this paper, we propose a novel cross-layer framework for QoS support in WMSNs, which optimizes the functionalities of communication protocols to maximize the number of video stream requests to be delivered while the imposed distortion constraint on the streams are met. QoS requirements are mapped on joint operations of application, network, MAC and link layers of the underlying communication framework. The experimental results demonstrate that the proposed framework achieves its objective efficiently.**

I. Introduction

A Wireless multimedia sensor network (WMSN) is composed of tiny wireless sensing devices that allow fetching video, audio and still images and reporting to a sink node [1]. In traditional multimedia systems, the streaming server is capable of implementing complex encoding algorithms while the receiving clients just use a simple algorithm to decode the receiving multimedia data. In contrast, the multimedia sensors in WMSNs have severe resource constraints including bandwidth, energy, storage, computation, etc. Therefore, the traditionally complex encoding techniques are not applicable to WMSNs. Multimedia support in wireless networks has been widely explored. These include broadband wireless networks, satellite networks, mobile ad hoc networks [2], [11], [13], [14]. QoS support in WSNs has also been addressed in some studies [10], [8]. However, there are several main peculiarities that make multimedia content delivery in WMSNs challenging, which is largely unexplored. Most of these challenges are unique in WMSNs due to the limited resources imposed on sensor nodes, and transporting video with guaranteed QoS in WMSNs is of prime importance due to higher rate requirements on limited and variable capacity channels. By taking into account the resources in WMSNs, joint optimization of networking layers, i.e., cross-layer design, stands as the most promising alternative to inefficient traditional layered protocol architectures.

Recently, a cross-layer design for WMSNs is proposed [5] whose objective is to maximize the network lifetime and control the admission of low priority nodes. The framework neither incorporates the video source coding to dynamically adopt the source rate according to network conditions nor multirate link adaptation is provided to exploit multi-rate wireless interface for maximizing link capacity. This problem is partially addressed in [6] that jointly optimizes the source coding and routing functions to minimize the distortion with maximum network lifetime following multiple paths. However, this approach does not provide any source admission control and thus quality of service can not be guaranteed. In [4], the cross-layer QoS architecture for WMSNs uses a Time-Hopping Impulse radio MAC and dynamic channel coding algorithm [4]. This approach is based on end-to-end resource reservation and thus incurs higher overhead. Moreover, it does not consider source rate adaptation to dynamically adopt to the network conditions. In summary, the existing studies focused on guaranteed QoS through end-to-end resource negotiation and setting up routing paths. In contrast, the proposed framework adopts a stateless routing mechanism and performs localized resource negotiation to avoid from the high overhead.

In this paper we investigate the cross-layer optimization design for maximizing the number of video stream realizations while the quality of service imposed on each video stream is still preserved. More specifically, we deal with the bandwidth enhancement problem subject to the distortion perceived at the sink, minimum bandwidth reservation for each individual source, real-time packet delivery, and packet reliability constraints. Sensor nodes implement conventional encoding algorithms for key frame generation and Wyner-Ziv lossy distributed video coding for smaller size frames, where the ratio of key and distributed coded frames is maintained as a tradeoff between distortion and source data rate. These frames are routed using a novel source directed multipath routing (SDMR) algorithm. SDMR interacts with the underlying MAC protocol, which is based on QoS supporting IEEE 802.11e MAC standard with some enhancements in scheduling to address the fairness. At the link layer, we adopt multirate transmission modes of IEEE 802.11 and a particular transmission mode is chosen dynamically according to bandwidth requirement with constrained link error rate.

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Hence, each application requirement is then mapped to the joint operations of application layer, network layer, MAC layer and link layer to achieve the desired QoS from the underlying communication framework.

The remainder of the paper is organized as follows. Section II introduces the system model and presents the formal problem definition. Section III proposes a source directed multipath routing protocol. Detailed performance evaluation and simulation results are discussed in Section IV, and the conclusions are given in Section V.

II. System Model

A WMSN consists of n stationary video sensor nodes, which are randomly and densely deployed in a region of interest, and a sink node that is used to gather the video data from the sensors. We assume that all the nodes as well as the sink are location-aware and the location of sink is known to all the sensor nodes. Moreover, the transmission range r of each sensor node is identical. Let A be the monitoring region area, then the node density $\rho(r)$ is $\frac{\pi r^2}{A}n$.

We employ distributed source coding [12] in which nodes encode videos with mixed encoding consisting of conventional video encoder to generate key frames, referred to as I-Frames, for side information computation and lossy Wyner-Ziv encoder exploiting side information to generate much smaller frames, referred to as W-Frames [18]. The routing algorithm is a stateless in which the selection of the forwarding node is purely based on the decision made by the relaying node. Thus, there are no fixed routing paths for unicast requests. To explore multiple disjoint paths, we partition the forwarding region into three disjoint zones with each accommodating one non-interfering (NI) path as shown in Fig. 2. Let N_1^i, N_2^i and N_3^i be the sets of neighboring nodes of node i in its three zones. Paths P_1, P_2 and P_3 are established corresponding to each zone, which are originated at nodes u, v, and w and destined to the sink, where $u \in N_1^i$, $v \in N_2^i$ and $w \in N_3^i$. Let $R_{in}^i(t)$ and $R_{out}^i(t)$ represent the total incoming data rate from the upstream nodes of node i and the total outgoing data rate from node i to its downstream nodes through paths P_1, P_2 and P_3, respectively at any time instant t. The values are measured in a discrete time fashion in a sequence $\{0, 1, ..., t, t+1, ...\}$ of sampling intervals of length Δ_t. In a stateless routing protocol that does not need to establish a routing path a priori, if per hop constraints can be meet under certain number of hops, then the end-to-end constraints can be guaranteed. The upper bound on the number of hops can be estimated in a densely deployed sensor network [9], [8]. Now, let τ be the end-to-end packet deadline from any source to the sink. We use the minimum advancement a meters per hop towards the destination and determine the expected hops count $E[h_{ix}]$ from source i to sink x. Having hops count information, we can compute the per hop delay budget $\tau^i = \tau/E[h_{ix}]$ for source node i. The multi-rate transmission modes of IEEE 802.11 are exploited in order to dynamically achieve the desired capacity and reliability. Assuming that the interface supports m transmission modes. Let $\chi_i(k)$ and $b_i(k)$ represent the bit error rate (BER) and the bandwidth at the k^{th} mode selected by node i, for $0 \leq k < m$. A node measures per hop mean delay $\bar{\tau}_j$, link reliability and available bandwidth during each sampling period Δ_t. We further assume that a source sends its stream *request* only once in a sampling interval and if the request is rejected in the current interval, then it attempts again in the next interval. The mean distortion introduced due to packet loss rate (P_L) is modeled by Stuhlmuller et. al. [17]. Likewise, higher motion-compensation disparity or inaccuracy of the side information function produces decoding errors. In practice, the larger the Group of Picture (GOP) size g, the higher the motion-compensation disparity between the side information and distributed coded W-Frame is predicted. Therefore, distortion introduced due to motion-compensated interpolation at the decoder using Wyner-Ziv distributed encoding is derived [18].

A. Problem Definition

Let $\Re$ be the set of video sensors that are sending their video data to the sink with meeting their individual constrained distortion, where $0 \leq |\Re| < n$. A new video stream request can be accommodated if each relaying node has adequate available bandwidth and also meets the desired quality of video. To make a new request admissible, the proposed strategy is to reduce the source coding rate to some extent, and increase the current link capacity to a certain amount by using some link adaptation function in multi-rate wireless interface, such as IEEE 802.11. Consider a video stream request sourced at i as a sequence of I-Frames and W-Frames for a GOP size g and encoded data rate r_i with mean packet length l. The source always keeps bandwidth reserved for its own video stream and its admission is controlled at the relaying nodes. If the request of source i was rejected by the relaying node j at time $t-1$ due to its bandwidth limitation, node i then increases the value of g to reduce its data rate and sends the request again at next interval t. Given a set $\Re_\delta$ of new requesting sources, the online routing problem is to maximize the admitted number of new requests at the relaying nodes. For a new request sourced at i, we formulate whether it is admissible at any relaying node j by the following linear program.

Maximize

$$\sum_{i \in \Re_\delta} (r_i(t-1) - r_i(t)) + b_j(k) - \left(R_{in}^j(t) + r_j(t)\right), \quad (1)$$

$$j \in \cup_{y=1}^3 N_y^i, 0 \leq k < m-1$$

subject to

$$D_i \leq \Theta, \quad \forall i \in \Re \quad (2)$$

$$\left(R_{in}^j(t) + r_j(t) + r_i(t)\right) \leq R_{out}^j(t), \quad i \in \Re_\delta, j \in \cup_{y=1}^3 N_y^i \quad (3)$$

$$\bar{\tau_j}(t) \leq \tau^i, \quad i \in \Re_\delta, j \in \cup_{y=1}^3 N_y^i \quad (4)$$

$$\left(1 - (1 - \chi_{ij}(k))^l\right) \leq P_L^{\frac{1}{E[h_{ix}]}}, \quad i \in \Re_\delta, 0 \leq k < m \quad (5)$$

In the objective function (1), we aim to accommodate source i by reducing its data rate and increasing the link capacity of

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relaying node j using higher transmission mode. The first term is the gain in bandwidth acquired by reduction in source data rate in which source i had higher data rate at time $t-1$ and it is reduced at time t. The request at time $t-1$ might have been rejected and attempts again at t with lower data rate. However, if it is the first attempt at time t, then r_i at $t-1$ is equal at t. The difference of the second and third terms gives the available bandwidth at k^{th} transmission mode.

III. Cross-layer QoS Framework

The QoS of multimedia data is generally characterized by four parameters: delay jitter, end-to-end deadline, bandwidth and reliability over unreliable medium. In multimedia applications, delay jitter is compensated by carefully choosing the playout time. However, packet deadline, bandwidth and reliability are dealt in the network at different communication layers jointly or separately, while distortion is the metric to measure the perceived QoS at sink node. In this framework, we consider per hop bandwidth (3), delay (4), and reliability (5) constraints such that the packets are delivered under probabilistic hops count limit as discussed in the following subsection. Fig. 1 illustrates the architecture of the proposed

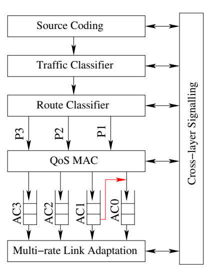

Fig. 1. Cross-layer QoS protocol architecture.

design. Traffic classifier module classifies the types of frames, i.e., I-Frames & W-Frames and stamps with their priorities to be dealt accordingly at the underlying layers and transfers it to the route classifier module. Route classifier implements the proposed source directed multiple path routing (SDMR) algorithm and chooses an appropriate path for frame delivery. Once, the packets reach at MAC interface, the MAC layer maps the packets to its ACs according to their priorities. The link adaptation algorithm dynamically chooses the appropriate transmission mode directed by the network layer to provide the required bandwidth subject to the link reliability.

A. Source Directed Multipath Routing (SDMR)

Higher channel coding rate and larger video data size need an excessive amount of bandwidth. It is, therefore, imperative to utilize the available bandwidth and other resources in such a way to efficiently support the QoS for video delivery. To meet these objectives, we propose a multipath routing algorithm to find geographically disjoint paths based on only local state information. Multipath routing is illustrated in Fig. 2. A source i determines three possible paths directed towards the destination sink x. P_2 is a primary path and obviously the shortest one among the three paths. The next hop is always chosen within the distance r (recall that r is the transmission range of a sensor) from the origin line, called path origin line (*PoL*) and is used as a primary path. The two secondary paths, P_1 & P_3, are also determined on the location basis and restricted above and below r unit of distance from *PoL*, respectively. Hence, the three paths, P_2, P_1 & P_3, are directed along *PoL*, $PoL^{'}$ (r units of distance above *PoL*) and $PoL^{''}$ (r unit of distance below *PoL*), respectively. However, the separation of r is loosely restricted that may be violated when the constraints have to be satisfied at the forwarding node. In the following we propose a heuristic for choosing the next hop forwarding node.

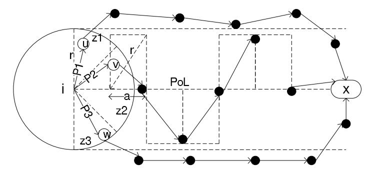

Fig. 2. Multipath routes selection from source i to the destination sink s.

1) Forwarding Node Selection: Due to random deployment, it is highly likely that the forwarding nodes do not lie exactly on *PoL* and there is always certain displacement from *PoL*. The relaying node i computes the displacement $\Delta d_{(j,PoL)}$ for the forwarding node j and uses it in the next hop selection. Node i computes the resistance (f_r) for each of the potential nodes that belong to the set of forwarding nodes compliance with Constraints (3), and (4). The resistance of a node with smaller displacement is lower and higher if it is far from *PoL*. On the other hand, a node will have lower resistance if its residual energy is higher than the other forwarding candidates and vice versa. Let E_j be the residual energy of node j, $f_r(j)$ for node j is computed as follows.

$$f_r(j) = \omega \frac{\Delta d_{(j,PoL)}}{r} - (1-\omega)\frac{E_j}{\max_{k \in \cup_{y=1}^{3} N_i^y}\{E_k\}}, \qquad (6)$$

where ω is a weighted factor that determines the contribution of each factor in computing the resistance force. The node $j \in \cup_{y=1}^{3} N_y^i$ is selected as a forwarding node if it has the lowest resistance, i.e., $f_r(j) = \min_{k \in \cup_{y=1}^{3} N_y^i}\{f_r(k)\}$. Thus, by combining the both factors allows to select nodes closer to PoL but with the balanced energy consumption. One can define a suitable value of ω according to the application requirements. If the end-to-end QoS support is far more important than the energy efficiency, then ω is kept lower to ensure more closer nodes along PoL will be selected for routing, and thus a shorter path follows, while a large value of ω would enhance the network lifetime. Note that this derivation is made for path P_2. Paths P_1 and P_3 differ by their origin lines and have longer distance. Moreover, in paths P_1 and P_3, a relay node ends up with a situation when it cannot find a forwarding node with minimum advancement (a) along its path origin line. The situation is assumed to be close to the sink at one or two hops away. Here, SDMR follows the directed routing from relaying nodes to the sink.

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2) Maximum Hops Limit: SDMR guarantees the constraints on a single hop and provides an estimation of the maximum number of hop on the basis of node density. Thus, such an assumption leads to the QoS support from source to destination subject to the certain node density. By defining minimum advancement of a units of distance in the forwarding node selection, a packet always progresses a unit of distance towards the sink. In the worst scenario, on path P_2 with minimum progress of a, the routing path will follow a see-saw pattern as shown in Fig. 2. The expected number of hops $E[h_{ix}]$ from source i to sink x can be simply determined by d_{ix}/a. However, it requires certain node density so that at least one forwarding node j is present at distance between a and r, i.e., $a \leq (d_{ix} - d_{jx}) \leq r$. For P_2, the forwarding region can be determined as $\frac{\pi(r^2-a^2)}{4}$. Therefore, in order to meet the constraint of minimum advancement a, it is required that the node density ensure the deployment of at least one node in the forwarding region. It implies that $\rho(z_2) \geq 1$, where $\rho(z_2)$ is the density in forwarding zone z_2.

In order to increase the bandwidth, SDMR also supports two more nodes disjoint paths P_1 and P_3. Unlike P_2 that is directed along the *PoL*, the lengths of paths P_1 and P_3 are a bit longer, which is obtained $(d_{ix} + 2r(\sqrt{2} - 1))$. Moreover, the forwarding zones for P_1 and P_3 are the half of z_2 that requires twice the density in z_2. Thus, per hop delay budget τ_i for source i can be computed by $\frac{\tau}{E[h_{ix}]}$. The value of τ_i in first case is smaller due to larger route length, which is based on the higher node density. However, if the density is lower as conditioned in the second case then the value of τ_i is higher that relaxes the per hop delay requirement due to lower node density. Hence, the routing algorithm adopts to the given node density.

B. Source Admission Control

In order to meet the bandwidth constraint, the framework incorporates source admission control at each node. Each source node i stamps the packet with its encoding bit rate r_i and delay budget τ_i, and transmits it over to the first hop forwarding node. A source node is generally able to find the first hop forwarding node because it keeps the bandwidth reserved at outgoing links by ensuring that $(R^i_{in} + r_i) \leq R^i_{out}$, where the source traffic r_i may not necessarily be in progress and may start at any moment. A source only transmits its first packet, representing the new source *request*, and waits until network feedback is received. If the *request* packet is successfully received at the sink, no matter which path is followed, then the sink generates the *Ack* packet and sends it to the source node. If a source receives an *Ack*, then it resumes the transmission of subsequent packets. However, if the relaying node j is unable to meet the bandwidth requirements of its upstream source i, then it runs the link adaptation algorithm to accommodate the *request* by switching to a higher transmission mode. If there is not any higher rate transmission mode at j to provide extra bandwidth under the given link reliability constraint, it then sends back the *reject* message to i. Node i forwards the request to another node j' in its forwarding region. In this situation, there are three possibilities: first, i will find a relaying node successfully; second, there is no such relaying node; and third, the attempt time exceeds the hop delay τ_i. The first case lets the request to proceed further at downstream nodes. The second and third cases trigger a *reject* message. When the *reject* message is received by the source, it reduces its encoding data rate by selecting higher value of g and sends the request again until either the *request* is successful or the tolerable distortion threshold has reached in reducing the encoding rate. In the former the source is admitted and has acquired sufficient resources to deliver its video, while in the latter the source can not be accommodated at the moment and will resume at the next bandwidth sampling interval.

C. IEEE 802.11e based MAC

In this framework, the MAC layer is based on the enhanced distributed coordination function (EDCF) provided by IEEE 802.11e wireless LAN standard. We exploit this prioritized scheduling to support QoS in an efficient way. The standard also defines the size of burst called $TXOP_{limit}$ that allows to transmit $TXOP_{limit}$ number of frames simultaneously through a single RTS/CTS frame exchange. In addition to the prioritized scheduling, we also define a maximum burst size γ to set different $TXOP_{limit}$ for each AC in which AC_VO is the allowed largest burst size and AC_BK has the smallest value according to their priorities. Hence, the value of $TXOP_{limit}[AC]$ for access category AC is set to $\frac{\gamma}{\alpha}$, where α takes the values 1, 2, 4, and 8 for AC_VO, AC_VI, AC_BE and AC_BK respectively. The value of γ should be greater or equal to the highest value of α so that at least one frame can be transmitted from the lowest priority AC in each round. The values of α are not fixed and can be set according to application requirement. However, such a scheme would prevent starvation of the lower priority traffic and intuitively provides the fairness corresponding to the values of α. To minimize the number of delayed packets, frames are sorted according to their deadlines and frames with shorter deadlines are selected in burst for transmission. This ensures that nodes at higher hops from the sink, thus shorter deadlines, get higher priorities within the same AC and are transmitted earlier. Such a scheme would balance the packet delivery of farther and closer nodes and implicitly provides bandwidth fairness. The difference between the average delay of ACs is due to the permitted burst size. The larger the burst size, the lower the average delay will be. However, a new frame that can not be transmitted in the current burst of any AC experiences the delay as the sum of the delays in all ACs or time taken to complete a round. Thus, to increase the size of bursts by choosing a larger value of γ, the more time will be required to complete the round.

IV. Performance Evaluation

Performance of the proposed cross-layer design is evaluated using network simulator *ns-2* and traffic scenario files are generated by implementing the source coding algorithm in MATLAB. The performance metric is the quality of video measured as the signal to noise ratio (SNR) and packet delay.

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The simulation scenario consists of 100 sensors and a sink uniformly deployed in $200m \times 200m$ square field, where the sink is placed at the center of the field. The quality of video is measured for 20 sources sending their video data to the sink.

Fig. 3 shows the SNR value at different encoding rates and Group of Picture (GOP) size g. The value of g is optimal for the distortion threshold and encoding bit rate as given in [18]. However the video quality drops if packet losses are incorporated in measuring the distortion. For higher packet losses, we reduce the value of g to achieve the desired quality.

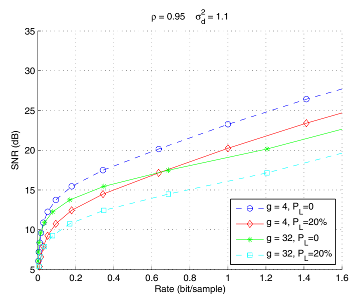

Fig. 3. Video quality measured at different GoP size g and packet loss rate P_L for 20 sources.

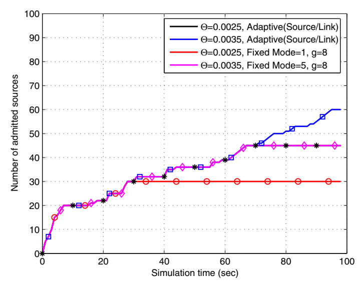

Fig. 4. Number of sources admitted at different time instant.

The adaptivity of the cross-layer design is depicted in Fig. 4, which shows that the number of admitted sources are increased upto 40% with adaptive source and link as compared to the fixed source encoding and single mode transmission. The value can be increased further if the tolerable distortion threshold Θ is set to some higher value, which is increased from 0.0025 to 0.0035. Thus by compromising the video quality to some extent, the number of admitted sources can be maximized. Fig. 5 shows the average packet delay. It can be seen that increasing the number of sources does not significantly increase the average end-to-end packet delay. This is due to the fact that the proposed routing algorithm SDMR follows multiple paths. If the higher delay is observed in one path, then the subsequent packets will be transmitted through alternative paths to distribute the load among three disjoint paths. Hence, the admission of a new source does not necessarily increase the delay of existing admitted sources.

V. Conclusion

This paper proposed a cross-layer design architecture to maximize the number of video sources admitted without

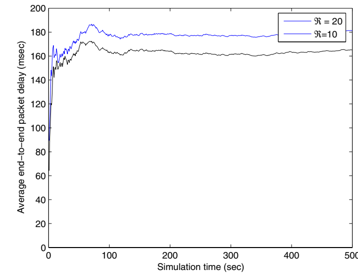

Fig. 5. Average packet delay for different number of source nodes.

affecting the quality of existing sources in WMSNs. In this architecture, application layer, routing layer and data link layer interact adaptively with each other to achieve the objective. Unlike previous studies, the proposed cross-layer design is capable of interacting with the application layer to choose an appropriate GoP size according to network conditions and the feedbacks received from the sink.

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