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REALTIME MUTIMEDIA VIDEO STREAMING FOR WIRELESS SENSORS NETWORK

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Abstract:

This article presents the design of a networked system for joint compression, rate control and error correction of video over resource-constrained embedded devices based on the theory of compressed sensing. The objective of this work is to design a cross-layer system that jointly controls the video encoding rate, the transmission rate, and the channel coding rate to maximize the received video quality. First, compressed sensing based video encoding for transmission over wireless multimedia sensor networks (WMSNs) is studied. It is shown that compressed sensing can overcome many of the current problems of video over WMSNs, primarily encoder complexity and low resiliency to channel errors. A rate controller is then developed with the objective of maintaining fairness among video streams while maximizing the received video quality. It is shown that the rate of compressed sensed video can be predictably controlled by varying only the compressed sensing sampling rate. It is then shown that the developed rate Controller can be interpreted as the iterative solution to a convex optimization problem representing the optimization of the rate allocation across the network. The error resiliency properties of compressed sensed images and videos are then studied, and an optimal error detection and correction scheme is presented for video transmission over loss channels.

Keywords: Wireless Multimedia Sensor Networks (WMSN), Network Optimizing.

Introduction:

Wireless Multimedia Sensor Networks (WMSN) [2] [3] are self-organizing systems of embedded devices deployed to retrieve, distributive process in real-time, store, correlate, and fuse multimedia streams originated from heterogeneous sources [4]. WMSNs are enablers for new applications including video surveillance, storage and subsequent retrieval of potentially relevant activities, and person locator services. In recent years, there has been intense research and considerable progress in solving numerous wireless sensor networking challenges. However, the key problem of enabling real-time quality-aware video streaming in large-scale multihop wireless networks of embedded devices is still open and largely unexplored. There are two key shortcomings in systems based on sending predictively encoded video (e.g., MPEG-4)

• **Encoder Complexity.** Predictive encoding requires complex processing algorithms, which lead to high energy consumption. New video encoding paradigms are therefore needed to reverse the traditional balance of complex encoder and simple decoder, which is unsuited for embedded video sensors. Recently developed *distributed video coding* [10] algorithms (aka Wyner- Ziv coding [11]) exploit the source statistics at the decoder, thus shifting the complexity to the decoder. While promising [2], most practical Wyner-Ziv codecs require end-to-end feedback from the decoder [12], which introduces additional overhead and delay. Furthermore, gains demonstrated by practical distributed video codecs are limited to 2- 5 dBs PSNR [13], [12]. Distributed video encoders that do not require end-to-end feedback have been recently proposed [14], but at the expense of a further reduction in performance. In addition, all of these techniques require that the encoder has access to the *entire video frame* (or even multiple frames) before encoding the video.

• **Limited Resiliency to Channel Errors**. In existing layered protocol stacks based on the IEEE 802.11 and 802.15.4 Standards, frames are split into multiple packets. If even a single bit is flipped due to channel errors, after a cyclic redundancy check, the entire packet is dropped at a final or intermediate receiver. This can cause the video decoder to be unable to decode an independently coded (I) frame, thus leading to loss of the entire sequence of video frames. Instead, ideally, when one bit is in error, the effect on the reconstructed video should be unperceivable, with minimal overhead. In addition, the perceived video quality should gracefully and proportionally degrade with decreasing channel quality.

EXISTING SYSTEM:

In existing layered protocol stacks based on the IEEE 802.11 and 802.15.4 standards, frames are split into multiple packets. If even a single bit is flipped due to channel errors, after a cyclic redundancy check, the entire packet is dropped at a final or intermediate receiver. This can cause the video decoder to be unable to decode an independently coded (I) frame, thus leading to loss of the entire sequence of video frames.

Disadvantages:

Instead, ideally, when one bit is in error, the effect on the reconstructed video should be unperceivable, with minimal overhead. In addition, the perceived video quality should gracefully and proportionally degrade with decreasing channel quality.

PROPOSED SYSTEM:

With the proposed controller, nodes adapt the rate of change of their transmitted video quality based on an estimate of the impact that a change in the transmission rate will have on the received video quality. While the proposed method is general, it works particularly well for security videos. In addition, all of these techniques require that the encoder has access to the entire video frame (or even multiple frames) before encoding the video.

MODULES:

1. CS Video Encoder (CSV)

The CSV video encoder uses compressed sensing to encode video by exploiting the spatial and temporal redundancy within the individual frames and between adjacent frames, respectively.

Sensing the channel: those that have the cost of sensing channel have higher energy consumption and so they are not suitable for WMSNs.

Using extra packets*:* Using retransmission time of dropped packets includes not only retransmission request but also transmission of dropped packet. These methods waste a great amount of energy for congestion
Detection in sensor nodes.

Low cost*:* Some methods do not necessitate extra cost for congestion detection. These methods are the most suitable for congestion detection in WMSNs.

2. Rate Change Aggressiveness Based on Video Quality:

With the proposed controller, nodes adapt the *rate of change* of their transmitted video quality based on an estimate of the impact that a change in the transmission rate will have on the received video quality. The rate controller uses the information about the estimated received video quality *directly* in the rate control decision. If the sending node estimates that the received video quality is high, and round trip time measurements indicate that current network congestion
Condition would allow a rate increase; the node will increase the rate less aggressively than a node estimating lower video quality and the same round trip time. Conversely, if a node is sending low quality video, it will gracefully

decrease its data rate, even if the RT T indicates a congested network. This is obtained by basing the rate control decision on the *marginal distortion factor*, i.e., a measure of the effect of a rate change on video distortion.

3. Video Transmission Using Compressed Sensing:

We develop a video encoder based on compressed sensing. We show that, by using the difference between the CS Samples of two frames, we can capture and compress the frames based on the temporal correlation at low complexity without using motion vectors.

4. **Adaptive Parity-Based Transmission:**

For a fixed number of bits per frame, the perceptual quality of video streams can be further improved by dropping error samples that would contribute to image reconstruction with incorrect information. Which shows the reconstructed image quality both with and without including samples containing errors? It assume that the receiver knows which samples have errors, to demonstrate that there is a very large possible gain in received image quality if those samples containing errors can be removed. We studied adaptive parity with compressed sensing for image transmission, where we showed that since the transmitted samples constitute an Unstructured, random, incoherent combination of the original image pixels, in CS, unlike traditional wireless imaging systems, no individual sample is more important for image reconstruction than any other sample. Instead, the number of correctly received samples is the only main factor in determining the quality of the received image.

Architecture:

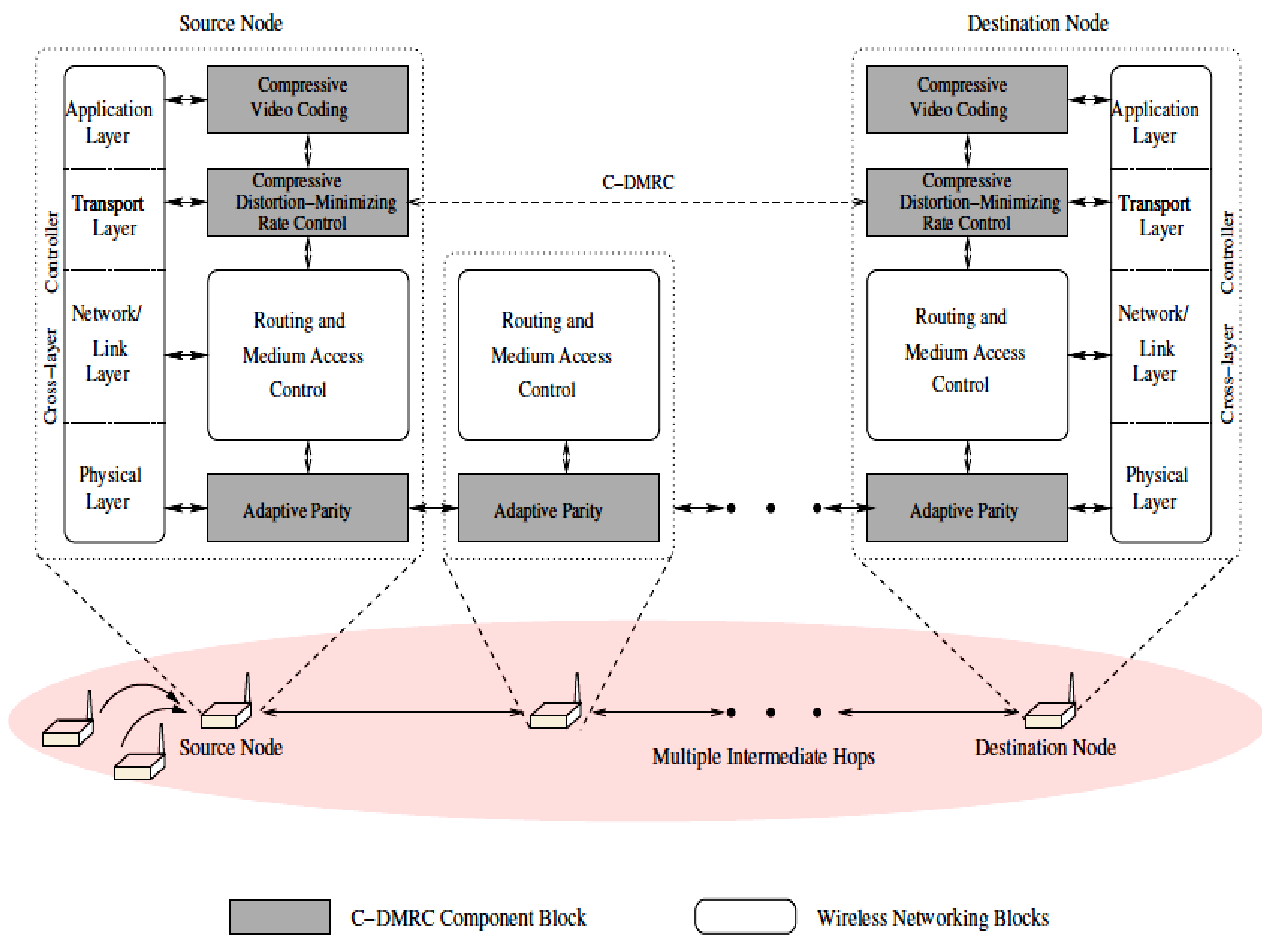

FIG 1.ARCHITECTURE OF C-DMRC SYSTEM

We describe the overall architecture of the compressive distortion-minimizing rate controller (C-DMRC). The system takes a sequence of images at a user-defined number of frames per second and wirelessly transmits video encoded using compressed sensing.

The end-to-end round trip time (RTT) is measured to perform congestion control for the video within the network, and the bit error rate (BER) is measured/estimated to provide protection against channel losses. The system combines functionalities of the application layer, the transport layer and the physical layer to deliver video through a multi-hop wireless network to maximize the received video quality while accounting for network congestion and lossy channels.

RELATED WORK

The most common rate control scheme is the well-known transmission control protocol (TCP) [25][26]. Because of the additive increase/multiplicative-decrease algorithm used in TCP, the variation in the rate determined by TCP can be very distracting for an end user, resulting in poor end user perception of the video quality [27]. In addition, TCP assumes that the main cause of packet loss is congestion [28], and thus misinterprets losses caused by channel errors as signs of congestion. These considerations have led to a number of equation-based rate control schemes, which analytically regulate the transmission rate of a node based on measured parameters such as the number of lost packets and the round trip time (RT T) of the data packets. Two examples of this are the TCP-Friendly Rate Control [29] [28], which uses the throughput equation of TCP Reno [25], and the Analytical Rate Control (ARC) [30] [31]. Both of these schemes attempt to determine a source rate that is fair to TCP streams. However, in a WMSN, priority must be given to the delay-sensitive flows at the expense of other delay-tolerant data. Therefore, both TCP and ARC result in a transmission rate that is more conservative than the optimal rate. For this reason, in an effort to optimize resource utilization in resource-constrained WMSNs, our scheme does not take TCP fairness into account. Recent work has investigated the effects of packet loss and compression on video quality. In [32], the authors analyze the video distortion over lossy channels of MPEG-encoded video with both inter-frame coding and intra-frame coding. A factor β is defined as the percentage of frames that are an intraframe, or I frame, i.e., a frame that is independently coded. The 2SSIM considers three different aspects to determine the similarity between two images. If one image is considered the original, then the measure can be viewed as the relative quality of the second image. The SSIM index first calculates the luminance difference between the two images. Then it subtracts the luminance components out and measures the contrast difference between the two images. Finally, the contrast is divided out and the structural difference is measured as the correlation between the two remaining signals. These three measurements are then combined to result in the overall SSIM index, which is a normalized value between 0 and 1. SSIM is a more accurate measurement of error because the human visual system perceives structural errors in the image more than others. For example, changes in contrast or luminance, although mathematically significant, are very difficult to discern for the human eye. Structural differences such as blurring, however, are very noticeable. SSIM is able to weight these structural differences better to create a measurement closer to what is visually noticeable than traditional measures of image similarity such as mean squared error (MSE) or PSNR. These results have been shown for images [22] and for videos [23] [24] in the LIVE database authors then derive the value β that minimizes distortion at the receiver. The authors of [32] investigate optimal strategies to transmit video with minimal distortion. However, the authors assume that the I frames are received correctly, and that the

Only loss is caused by the inter-coded frames. In this paper, we assume that any packet can be lost, and rely on properties of CS video and on an adaptive parity mechanism to combat channel impairments and increase the received video quality. Quality of service (QoS) for video over the Internet has been

Studied in [33] and [34]. Both of these works deal with QoS of video over the Internet using TCP or a TCP-Friendly rate controller. In general, a WMSN will not be directly connected to the Internet, so requiring fairness to TCP may result in significant underestimation of the achievable video quality. Several recent papers take a preliminary look at video encoding using compressed sensing [35], [36], [37]. Our work is different in the following sense: (i) we only use information that can be obtained from a single-pixel camera [21] and do not use the original image in the encoding process at the transmitter. Hence, C-DMRC is compatible with direct detection of infrared or terahertz

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wavelength images, along with the ability to compress images during the detection process, avoiding the need to store the entire image before it is compressed; (ii) we look at the problem from a networking perspective, and consider the effect of joint rate control at the transport layer, video encoding, and channel coding to design an integrated system that maximizes the quality of wirelessly transmitted CS video. Finally, video encoding algorithms based on compressed sensing are also presented in [38] and further refined in [39] [40] [41]. In these works, the authors present a distributed compressive video sensing (DCVS) system which, like the encoder presented in this paper, does not require the source node to have access to the raw video data. The rate allocation is examined in [39]. However, unlike [39], the rate allocation scheme presented in this paper is intended to maximize the rate over multiple videos sharing a network, while [39] looks to determine the optimal rate of a single video session based on the sparsity of that raw video.

SYSTEM DESIGN

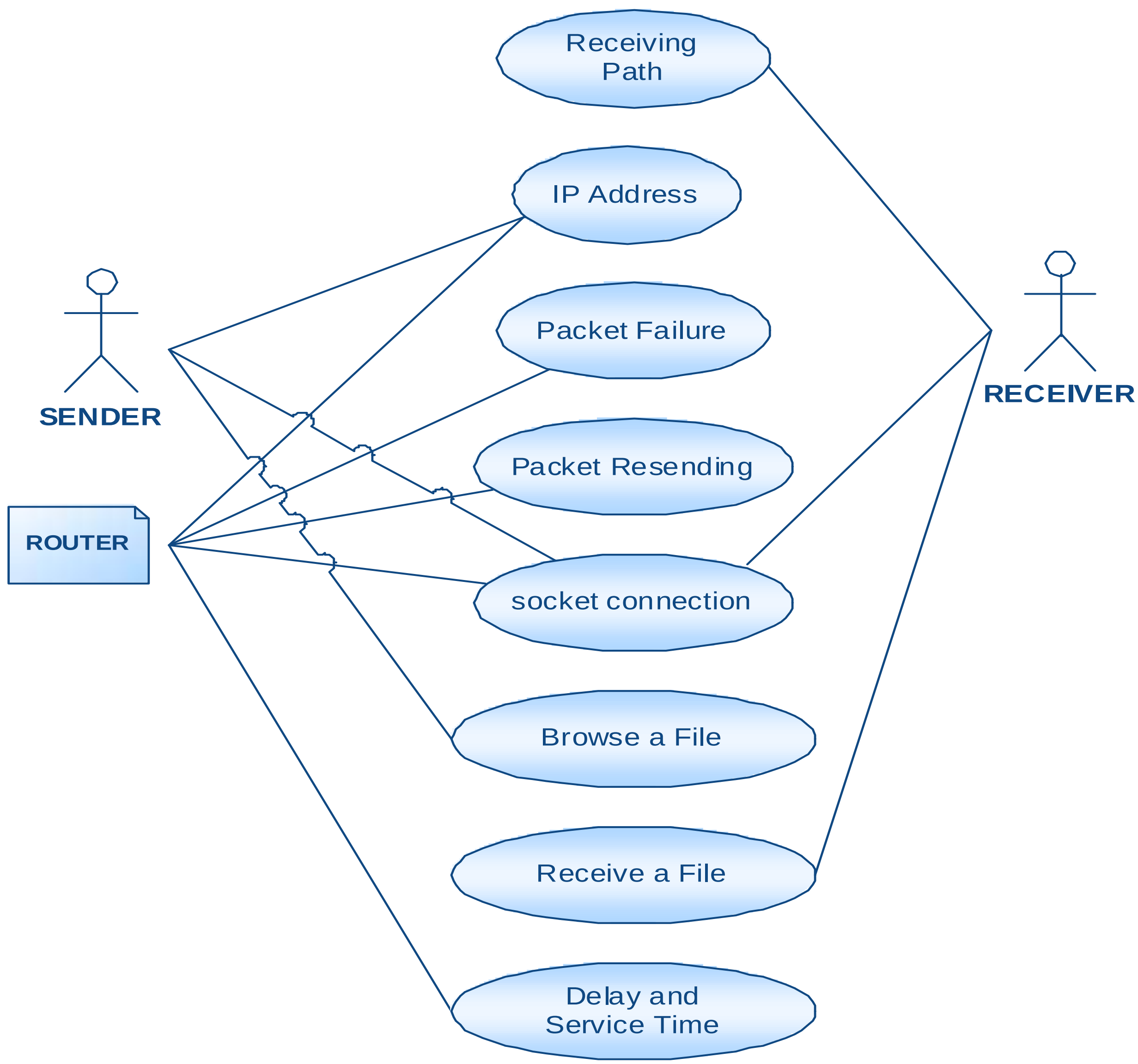

FIG 2.USE CASE DIAGRAM

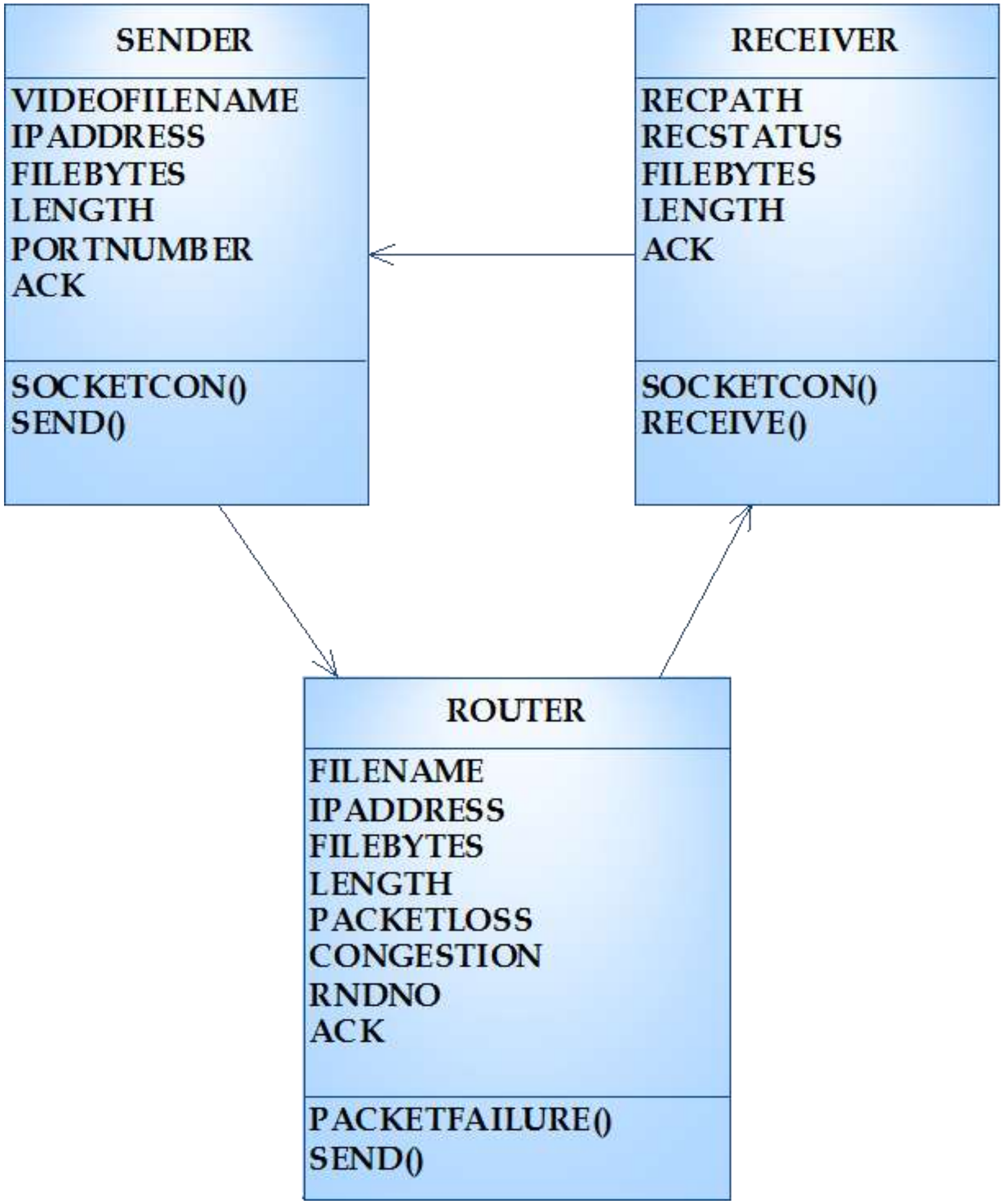

FIG 3 CLASS DIAGRAM

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CONCLUSION AND FUTURE WORK

We have introduced a new wireless video transmission system based on compressed sensing. The system consists of a video encoder, distributed rate controller, and an adaptive parity channel encoding scheme that take advantage of the properties of compressed sensed video to provide high-quality video to the receiver using a low-complexity video sensor node. The rate controller was then shown to be an implementation of an iterative gradient descent solution to the optimal rate allocation optimization problem. Simulation results show that the C-DMRC system results in a 5%-10% higher received video quality in both a network with a higher load and a small load. Simulation results also show that fairness is not sacrificed, and is in fact increased, with the proposed system. Finally, the video encoder, adaptive parity and rate controller were implemented on a USRP2 software defined ratio. It was shown that the rate controller correctly reacts to congestion in the network based on measured round trip times, and that the system works over real channels. We intend to implement the remaining portions of the C-DMRC system on the USRP2 radios, including image capture and video decoding.

Future Scope

- We believe that this research area will attract the attention of many researchers and that it will push one step further our ability to observe the physical environment and interact with it.
- We pointed out how recent work undertaken at the application layer, specialized spatio-temporal transport layer solutions, delay bounded routing, multi-channel MAC protocols, and UWB technology, amongst others, seem most promising research directions in developing practical WMSNs
- The Application can be extended on web.

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