VARIABLE TIME-SCALE AUDIO STREAMING OVER 802.11 INTER-VEHICULAR AD-HOC NETWORKS

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ABSTRACT

This paper presents an analysis of audio streaming in an inter-vehicular network based on 802.11b wireless devices. In such a scenario characterized by strong link availability variations, we investigate the performance of an adaptive packet scheduling policy that adapts the inter-packet transmission interval to the channel conditions. Network simulations are used to evaluate the effects of varying the transmission time scale between zero, when the connection is not available, to as fast as possible when the channel is available and reliable. Results show that the proposed approach ensures high quality audio streaming among the nodes of the inver-vehicular network by heavily reducing the percentage of lost packets and with only a limited increase in the delay and jitter.

1. INTRODUCTION

IEEE 802.11 Wireless LAN (WLAN) products have become widely used because of their simple set-up and moderate cost. Potential uses of such equipment range from WLAN hot spots to direct connectivity of devices in ad-hoc mode.

Ad-hoc networks are a key factor in the evolution of wireless communications enabling data exchange between wireless hosts in absence of a centralized fixed infrastructure. In an inter-vehicular scenario, for instance, vehicles can operate as a pure ad-hoc network in which each individual vehicle broadcasts data to other vehicles.

Due to the relative novelty of the application, few effort have been devoted to study and simulate 802.11 inter-vehicular transmissions [1][2], and, to the best of our knowledge, exploitation of inter-vehicular 802.11 communications for real-time multimedia services received even less attention. In [3] performance of video communications has been evaluated while driving two cars equipped with 802.11b standard devices in urban and highway scenarios. The experiments show that each scenario presents peculiar characteristics in terms of average link availability and SNR which can be exploited to develop more efficient inter-vehicular applications. The strong link availability variations experienced in the highway scenario suggest, in fact, that a *variable time-scale transmission policy* may be investigated for multimedia streaming to mitigate the effect of frequent disconnection between the mobile stations.

Streaming implementations developed and tuned for wired connections or wireless environments with limited mobility are usually designed to cope only with limited variations in network latency and bandwidth [4]. Before starting the playout, a pre-roll delay is then used to fill the receiver buffer. Buffering, in fact, reduces system sensitivity to short-term fluctuations in the data arrival rate by absorbing variation in end-to-end delay. However, if the rate offered by the channel falls below that of the source, the buffer will soon underflow. In this case rate adaptive algorithms are used to adapt the source rate to the current state of the network so as to generate only the bandwidth that the network is capable of carrying. The assumption is, in fact, that the distortion introduced by lowering the source coding rate is smaller than the expectedly larger distortion due to packet losses.

To deal with the fast changing inter-vehicular wireless scenario, where the connection to other mobile nodes is frequently lost and the streaming flow is interrupted, in addition to the foregoing techniques appropriate streaming algorithms must be implemented so that the receiver has enough data to continue the playback until the connection is re-established. Then the buffer needs to be refilled to a level that provides sufficient protection for a subsequent disconnection [5].

In this paper we analyze the performance on audio streaming of an *adaptive packet scheduling (APS)* technique

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that enables transmission rate changes by varying the inter-packet transmission interval instead of the size of audio packets. Streaming systems usually transmit audio frames at fixed time intervals, that is with the same rate at which they will be decoded and presented to the user. In the proposed scheduling algorithm for audio streaming the packet scheduler is instead able to change its instantaneous transmission rate from zero (i.e., the transmission pauses) when the link is not available, to as faster than real-time as the channel bandwidth allows (to refill the receiver buffer to the right size). The original playback rate, however, is not changed, and remains constant.

Most of the research work performed on the idea of changing the packet transmission schedule has been previously focused on end-to-end congestion control [6], or bandwidth smoothing for VBR video [7]. The effects on the quality of inelastic multimedia traffic [8], especially with regards to its delay and jitter constraints for real-time playback, still remain to be investigated.

End-to-end congestion control techniques are mainly based on two TCP-friendly rate control mechanisms: the rate adaptation protocol (RAP) [9] and the TCP friendly rate control (TFRC) [10][11]. Both these algorithms control the network status at the receiver and provide the sender with feedback information in order to adapt the output rate of the source to the channel available bandwidth.

The adaptive packet scheduling algorithm presented in this paper, although based on TFRC for the evaluation of the available bandwidth, relies instead on the variation of the packet sending rate and not of the source rate for matching the network status. The effectiveness of this choice has been previously demonstrated in [12], where the scenario was however limited to considering video streaming over wired Internet connections. Here we consider a wireless scenario where actual measurements from inter-vehicular transmissions are used to drive network simulations that analyze the performance of the adaptive packet scheduling approach in presence of high channel loss rates.

The paper is organized as follows. Section 2 presents the adaptive packet scheduling policy and its application to the vehicular scenario. Network simulation setup and results are described in Section 3. Finally conclusions are drawn in Section 4.

2. ADAPTIVE PACKET SCHEDULING

Streaming of real-time multimedia, as the name implies, heavily depends on timing constraints. Audio data, for example, must be played out continuously, so, if the data does not arrive in time to the client, the playout process must pause, with annoying effects for the human listener.

Before transmission over packet networks the compressed audio signal is divided into frames, with the property that all the data belonging to a single frame is played back at the same time during the decoding process. Each frame may be encapsulated in one or more packet that is then transmitted and stored at the receiver before decoding. All the frames must be received before their presentation time in order to decode the data without errors. Since the Internet introduces time-varying delays, a buffer is usually employed at the receiver to provide continuous playout. At the sender, on the other hand, frames are generally transmitted on the network with the same rate at which they will be decoded and presented to the user.

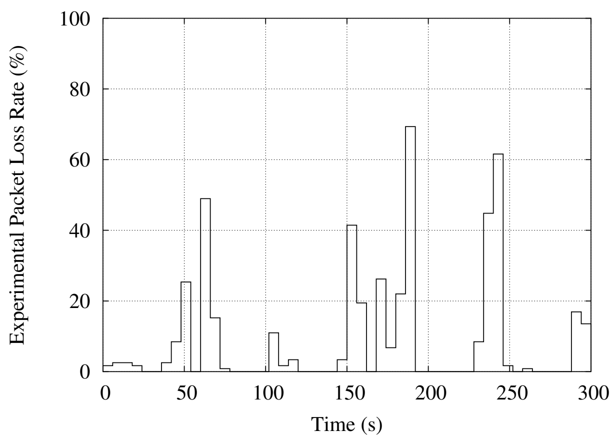

Fig. 1. Experimental packet loss rate measured transmitting UDP packets between two vehicles on an highway. Values are averaged on a six-second window.

In the inter-vehicular scenario considered in this paper a typical problem is the possible outage of the network connection between vehicles. Such channel outages may occur while the vehicles cross areas characterized by severe fading conditions or when vehicles are at a distance close to the limit of their wireless antenna coverage area. This is a very critical issue when dealing with streaming services, which experience possibly long (several seconds) interruption with highly negative impairments on the decoded multimedia quality. For instance, Fig. 1 illustrates the packet loss rate experienced in a highway scenario, as measured in transmission experiments performed during the activity reported in [3].

We argue that, rather than increasing the size of the playout buffer to allow additional caching and pre-fetching of audio data, an additional technique may be implemented to reduce the impact of channel degradation on the experienced audio quality. Instead of transmitting the audio frames always at the same rate, our adaptive packet scheduling technique is based on the concept of varying the transmission rate according to the instantaneous network conditions, while the original playback rate remains constant.

Figure 2 describes the block diagram of a video trans-

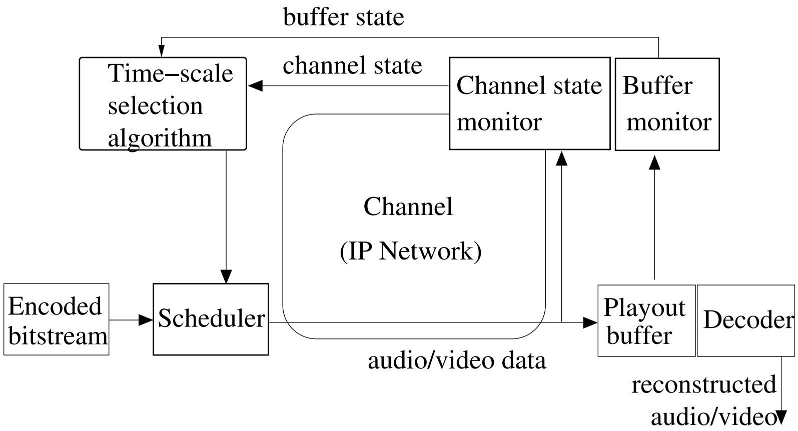

Fig. 2. Block diagram of the adaptive packet scheduling transmission system.

mission system using the adaptive scheduling approach. The pacing of the packets is determined by the scheduler in order to achieve the rate imposed by the time-scale selection algorithm. The channel state monitor determines the instantaneous channel state, then the status information is sent back to the time-scale selection algorithm which modifies the rate coefficient according to the estimate of the channel capacity. Additionally the buffer monitor may take into account the decoder buffer status and inform the scheduling algorithm with an estimate of the buffering delay experienced by the data before playout.

Because of the feedback loop between the source and the receiver the presented technique offers two advantages with respect to fixed scheduling transmission of audio frames. First, when a link outage occurs or when the channel is extremely bad, the number of dropped packets is considerably lower. In fact, the receiver feedback, that informs the source of the current channel loss rate, enables the sender to reduce the sending rate and so the number of packet transmitted during high error rate periods (if feedback reports are lost, the sender automatically cuts the rate in half after a given timeout expires). Second, the transmitting station may use the same feedback indication to increase the sending rate to match the available network bandwidth when the loss rate is low, thus enabling fast refill of the playout buffer. Both rate variations are obtained by varying the inter-packet gap, IPG_n, which is linked to the transmission rate, R_n, and to the packet size, s, as follows:

$$IPG_n = \frac{s}{R_n}. \qquad (1)$$

TFRC is used to control the maximum bitrate which can be used by the source to achieve TCP friendly behavior, this mode is based on the following equation

$$R = \frac{s}{RTT\sqrt{\frac{2p}{3}} + t_{RTO}(3\sqrt{\frac{3p}{8}})p(1+32p^2)}, \qquad (2)$$

which provides the output rate, as a function of the round-trip time, RTT, the loss rate, p, the packet size, s, and the timeout interval used to reveal losses, t_{RTO}.

In addition, monitoring the instantaneous conditions of the receiver-side playout buffer, the sender can realize when the bandwidth is not sufficient for the actual audio stream, i.e., a buffer underrun is likely to occur, so it can reduce the coding rate for the amount of time necessary to overcome the temporary channel degradation. This mechanism allow to combine the adaptive scheduling mechanism with the well known rate-adaptive approach.

3. NETWORK SIMULATIONS

The inter-vehicular scenario has been studied by means of simulations performed with the NS-2 network simulator (version 2.27) [13] for an 802.11b wireless LAN at 11 Mbps. Simulations aim at evaluating the adaptive packet scheduling algorithm in a context with mild concurrent traffic but with long periods of unstable connection as measured during actual experiments driving on a highway [3]. As regards the audio encoding technique, we used a stereo MPEG-1 Layer III CBR stream at 96 kb/s per channel that achieves CD quality playback in an error free scenario.

Results reported here summarize the observations made with various testing scenarios. All experiments have been carried out using a simple topology which consist of a single receiver node and a set of senders. Two nodes where instructed to send the same audio stream but with two different policies: the fixed packet scheduling (FPS) approach, where packets are regularly spaced in time, and the adaptive packet scheduling (APS) approach where packets are more apart from each other when the channel is bad and more closely set in time when the channel is good. The APS implementation is based on TFRC. TFRC, in fact, can vary its sending rate in response to network congestion and

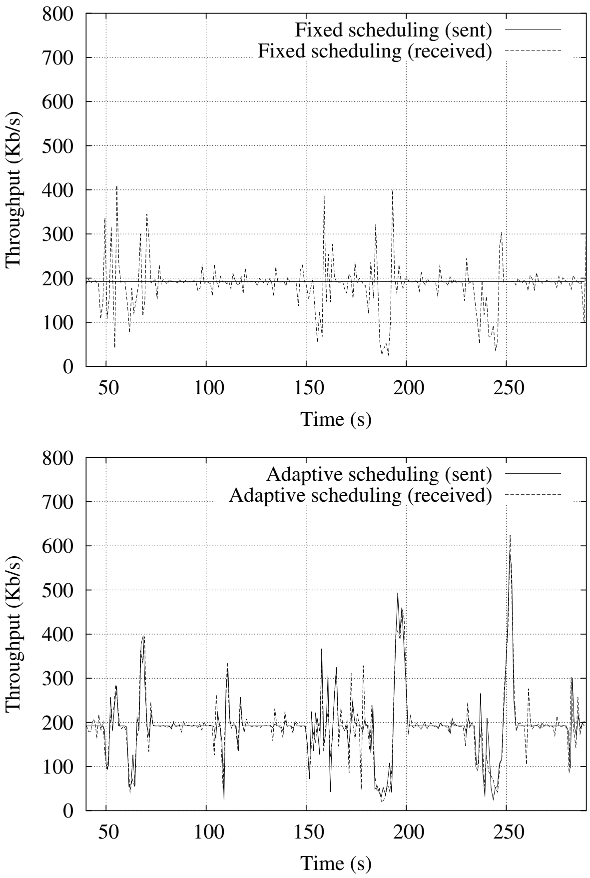

Fig. 3. Throughput measured at the sender and at the receiver for the fixed scheduling (top) and the adaptive scheduling (bottom) algorithms.

its sending data rate is calculated using the TCP throughput equation. Additional interfering traffic is modeled by four nodes sending constant bit rate UDP traffic at 2.3 Mb/s in packets of 1500 bytes.

The channel behavior experienced during the actual vehicular measurements has been reproduced by means of a uniform packet error model with time varying error rate corresponding to the values shown in Figure 1. A number of time periods with heavy loss rate lasting several seconds are present in the experimental trace. The loss trace frequently exceeds 50% of lost packets when the link availability between the transmitting and receiving vehicles is extremely bad. Important concerns in the evaluation of the adaptive policy are both its responsiveness to changes in network conditions and its ability to drop the packet sending rate in periods of deep errors to reduce the number of discarded packets. In particular, in the case of inelastic audio streaming, also the effects of delaying the transmission of time-sensitive data must be assessed because that practice can easily cause buffer underflows at the receiver. In the following we will present simulation results on the throughput, packet loss rate, and playout buffer fullness to validate the proposed scheduling policy with respect to the aforementioned constraints.

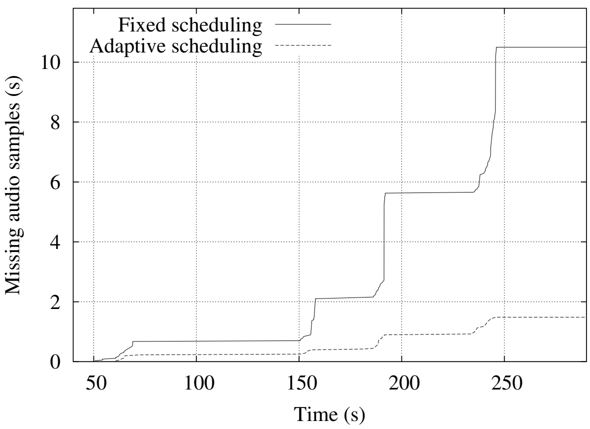

Fig. 4. Cumulative sum of missing audio seconds during the playout caused by packet losses.

In Fig. 3 we plot the throughput of the audio transmission as measured at both the source and the sink node. The key insight here is that in the APS case the number of packets sent through the network is clearly influenced by the various network conditions, i.e., the channel loss rate. The APS algorithm reduces the transmission rate by varying the inter-packet gap (IPG) around the 60th, 180th, and 230th second thus reducing the number of packet sent when they have a very high probability to be corrupted. On the other hand we can clearly note that the throughput increases well above the average bitrate of 192 kb/s just after those time periods when the sender takes advantage of the error-free connection to refill the receiver playout buffer. In the fixed scheduling case, instead, the sending rate is constant and always equal to 192 kb/s. We appreciate only sporadic spikes in the rate of the received data because of the varying percentage of dropped packets and of the flushing of networks buffers when the channel moves from a bad state to a good state.

More importantly, with the proposed algorithm the packet loss rate (PLR) drops significantly from 3.54% to 0.5%, a reduction by a factor of seven. This means that over the 300 second trace used for the simulations, the two techniques lost respectively 10.5 and 1.48 seconds of audio. Figure 4 shows the cumulative sum of the audio samples lost throughout the simulation. We observe that especially during very noisy channel conditions (i.e., when the wireless connection is not reliable), the fixed scheduling loses far more packets than the adaptive algorithm. This can be explained by the sensitivity of the proposed transmission protocol that reacts to changes in network conditions and

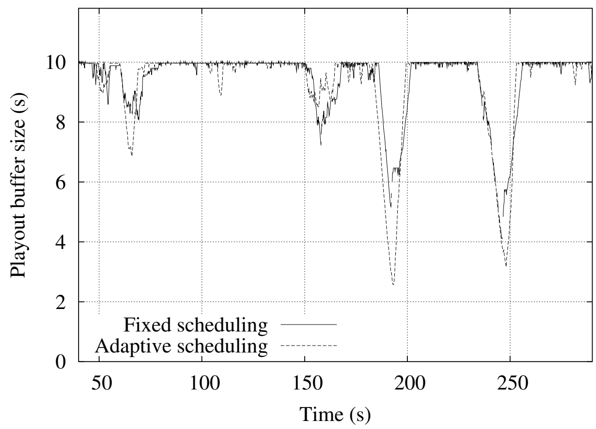

Fig. 5. Playout buffer size of the two transmission scheduling techniques as a function of time.

almost stops delivering packets when the feedback from the client warns of the bad channel behavior. In this cases the rate adaptation algorithm is not reacting to a congestion event but to a link failure, so it would be worthless to instruct the sender to change the source rate, instead the benefit comes from reducing the sending rate in terms of packets transmitted per second.

The playout buffer at the client allows the sender to vary the actual rate of the transmitted data. However, the buffer can accommodate only a certain amount of delay variation depending on its dimension and on the initial pre-buffering period. If this value is overrun, it cannot be guaranteed that the streaming can be continued without interruptions. So, it is important checking that the APS algorithm does not have a detrimental effect on the playout buffer. The buffer occupation in the client is shown in Fig. 5. In this case an initial pre-buffering period of ten seconds is chosen. The buffer fullness depends on the channel error rate and on the packet delay, so both the transmission scheduling algorithms are affected by the problem of buffer depletion. But, while the fixed scheduling algorithms mainly suffers from the number of lost packets (because its transmission rate is constant), the APS algorithm experiences also a reduction of the buffer size because it delays the transmission of audio packets when the channel is bad. The two algorithms, however, present almost the same behavior proving that, if the APS algorithm is able to appropriately adapt the IPG to the channel conditions, not only the PLR is reduced, but also the playout continuity is not damaged by the effect of postponing the transmission of the audio packets with respect to the fixed transmission policy.

4. CONCLUSIONS

We investigated the performance of an adaptive packet scheduling algorithm for audio streaming in an inter-vehicular network consisting of 802.11b wireless devices. The proposed technique can adapt the streaming rate by varying the inter-packet transmission interval as a function of the estimated link availability and bandwidth. Network simulation results, based on the network traces collected in our experiments, show that this mechanism is particularly good at reducing the packet loss rate and at providing continuous streaming to an inter-vehicular user with only a limited increase in the packet delay and jitter compared to widely deployed fixed transmission policies.

5. ACKNOWLEDGMENTS

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