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Robust Live Unicast Video Streaming with Rateless Codes

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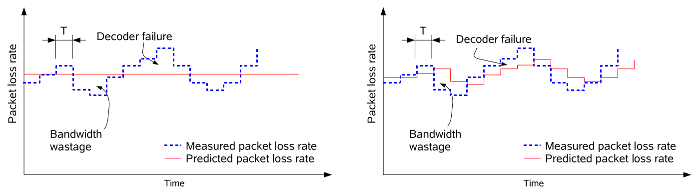
***Abstract*— We consider live unicast video streaming over a packet erasure channel. To protect the transmitted data, previous solutions use forward error correction (FEC), where the channel code rate is fixed in advance according to an estimation of the packet loss rate. However, these solutions are inefficient under dynamic and unpredictable channel conditions because of the mismatch between the estimated packet loss rate and the actual one. We introduce a new approach based on rateless codes and receiver feedback. For every source block, the sender keeps on transmitting the encoded symbols until it receives an acknowledgment from the receiver indicating that the block was decoded successfully. Within this framework, we provide an efficient algorithm to minimize bandwidth usage while ensuring successful decoding subject to an upper bound on the packet loss rate. Experimental results showed that compared to traditional fixed-rate FEC, our scheme provides significant bandwidth savings for the same playback quality.**

I. Introduction

Live video streaming over packet erasure channels requires error resilience mechanisms against packet loss and delay. One such mechanism is application-layer forward error correction (FEC) [1], [2], [3], [4]. However, previous work uses fixed-rate FEC, where the channel code rate is fixed a priori or updated adaptively (for example, by puncturing or shortening a Reed-Solomon code [3]) according to a prediction based on past observations of the packet loss rate. Unfortunately, the packet loss rate in many packet erasure channels, including the Internet and wireless networks, is hard to predict and can rapidly change over time. Thus, the performance of fixed-rate FEC schemes may be poor because of the unavoidable mismatch between the actual packet loss rate and the predicted one. Indeed, overestimating the packet loss rate would waste the bandwidth and underestimating it would result in decoding failure (Figure 1).



(a) (b)

Fig. 1. Mismatch between observed and predicted packet loss rate. The observed packet loss rate corresponds to transmission intervals of length T. (a) Static prediction. (b) Adaptive prediction.

To circumvent this problem, we propose to use rateless codes [5], [6] instead of fixed-rate codes. With rateless codes, also known as fountain codes, the code rate does not have to be fixed a priori as the encoder can generate on the fly a potentially infinite stream of encoded symbols. The most powerful rateless codes are the Raptor codes [6], which can recover k source symbols from any received $k(1+\epsilon)$ encoded symbols with high probability. Here ϵ is small compared to 1. For example, for $1000 \leq k \leq 8192$, ϵk is typically equal to two symbols [7]. Moreover both the encoding and decoding times of Raptor codes are much lower than those of standard fixed-rate erasure codes (e.g., Reed-Solomon codes).

Rateless codes have been previously used for video streaming in a broadcast/multicast scenario [8], [7], [9], [10]. In this paper, we propose to apply them for live unicast video streaming. The basic idea is that for every source block, the sender keeps on sending the encoded symbols until an acknowledgement is received from the receiver. However, as the acknowledgment needs time to reach the sender, the sender may transmit redundant encoded symbols. We show how to construct transmission strategies that minimize this overhead, while ensuring successful reconstruction of the video stream subject to an upper bound on the packet loss rate. We compared the performance of our strategies to that of traditional fixed-rate coding where the code rate is fixed a priori. Our results show that the benefits of our scheme become more important with increased available bandwidth or decreased round trip time.

The rest of the paper is organized as follows. In Section II, we describe our live video streaming system. In Section III, we present a class of appropriate transmission strategies for this system. Then we propose an efficient algorithm whose aim is to select among this class one that minimizes the expected overhead. Section IV contains experiments that compare our approach to fixed-rate FEC.

II. Live video streaming system

Figure 2 shows the proposed transmission system. The raw video stream produced by the camera from time $t = 0$ to $t = T$ is fed into the source encoder (e.g., H.264) to produce the first *source block*. For simplicity, we ignore the video encoding time, which is usually very small and depends on the particular implementation of the source encoder. At $t = T$, the sender applies FEC to

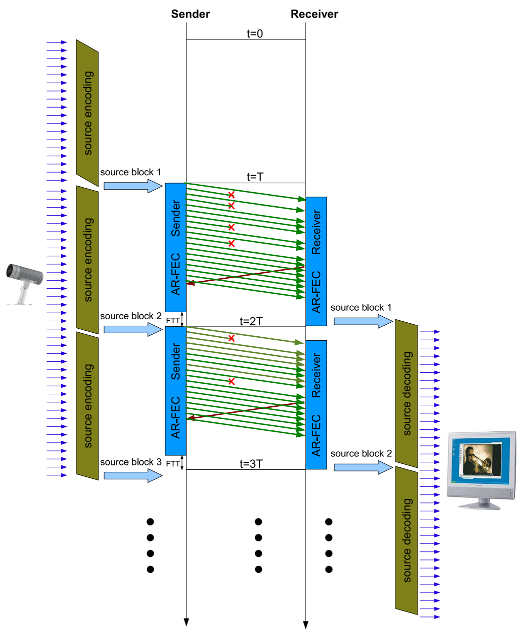


Fig. 2. Proposed live video streaming system.

the source block as will be explained in Section III. The encoded symbols are then transmitted according to the transmission strategy described in Section III-A. Some of the transmitted encoded symbols are lost or arrive at the receiver too late to be useful. The receiver tries to recover the source block. If it succeeds, then the source block is fed into the source decoder at $t = 2T$. Source decoding can be done with almost no delay providing the first byte of decoded video stream for playback at $t = 2T$, which ensures a maximum playback latency of $2T$. Increasing T will increase the size of the source block. This will lead to a more efficient rateless code, but also to a longer playback latency.

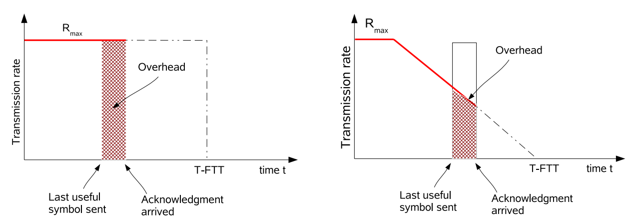
The same process is repeated. In this way, source block b corresponds to the video stream captured from $t = (b-1) \times T$ to $t = b \times T$, $b = 1, 2, \ldots$ The source blocks are encoded independently, which can be achieved, e.g., by starting each one with an I frame. We assume that all source blocks have the same number k of source symbols, which can be fulfilled by using a constant bit rate source encoder. Moreover, source block b has to be FEC encoded, transmitted, and FEC decoded from $t = b \times T$ to $t = (b+1) \times T$, so that it is available for playback at $t = (b+1) \times T$.

III. Proposed FEC scheme

Consider a source block b consisting of k symbols of size s each. The symbol size s may vary from one bit to several bytes, but is fixed for all source blocks. We encode the k source symbols by applying a rateless code to produce a potentially infinite stream of encoded symbols, each of size s. These encoded symbols are transmitted over the channel after encapsulating them in channel packets. A channel packet may contain one or more encoded symbols. For simplicity, we describe our system when a channel packet contains only one encoded symbol. We assume that a reliable feedback channel is available, and the channel bandwidth [11] (or capacity limit), which we denote by $R_{\max}$, is large enough to transmit $k(1+\epsilon)/(1-l)$ encoded symbols in a time period of length $T - FTT$. Here FTT is the forward trip time and l is the packet loss rate observed during a transmission interval. Some of the channel packets are lost or arrive at the receiver too late to be useful. We assume that the receiver can recover source block b correctly if and only if at least $k \times (1+\epsilon)$ encoded symbols for this block are received before time $(b+1) \times T$. Thus, a simple transmission strategy π to guarantee successful decoding of source block b is to send at least $C = k \times (1+\epsilon)/(1-l)$ encoded symbols from $t = b \times T$ to $t = (b+1) \times T - FTT$. Unfortunately, the transmitter does not know beforehand the value of C because l is unpredictable and varies from block to block. Overestimating l would result in bandwidth wastage and underestimating it would lead to decoding failure. However, when a feedback channel is available, this problem can be alleviated by making the receiver send an acknowledgment to the transmitter as soon as enough encoded symbols are received. Since the acknowledgement needs time to reach the transmitter, this approach introduces an *overhead* $H(\pi)$ equal to the number of unnecessary encoded symbols sent to the receiver.

The transmission strategy π is also characterized by an *outage rate* $\eta(\pi)$ equal to 0 if the source block is successfully decoded and 1, otherwise. Note that a source block can be decoded successfully if and only if $l \leq L(\pi)$, where $L(\pi) = 1 - k \times (1+\epsilon)/c_{\max}(\pi)$, and $c_{\max}(\pi)$ is the maximum number of encoded symbols that can be sent with the transmission strategy.

Figure 3 shows two transmission strategies. In both cases, the transmitter keeps on sending the encoded symbols until an acknowledgement is received. In Figure 3(a), the transmission rate is fixed and equal to the available bandwidth $R_{\max}$. In Figure 3(b), it is variable and yields a smaller protocol overhead. Thus, the second transmission strategy (π_2) seems to be better than the first one (π_1). However, since $L(\pi_2) < L(\pi_1)$, the probability of successful decoding is greater with π_1 than with π_2.



(a) (b)

Fig. 3. (a) Transmission strategy with a fixed transmission rate. (b) Transmission strategy with a variable transmission rate. The shaded areas show the overhead.

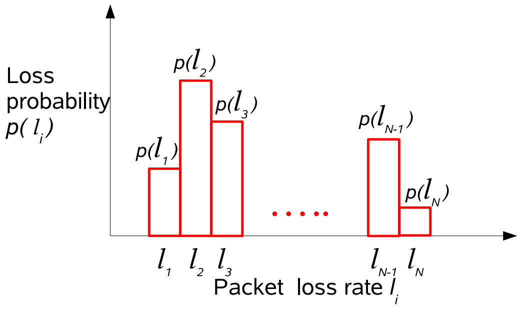


Fig. 4. Probability mass function of observed packet loss rate.

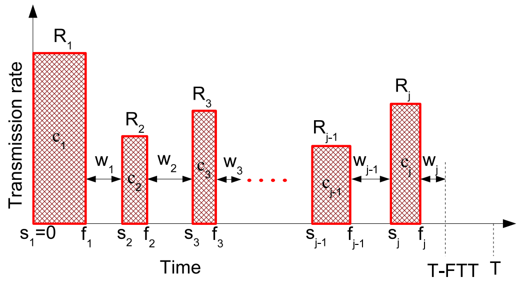


Fig. 5. Proposed transmission strategy. The encoded symbols are transmitted at rate R_i from s_i to f_i, followed by a waiting time of $w_i, \quad i = 1, \ldots, j$.

A. Proposed transmission strategy

Ideally, we would like to construct transmission strategies that minimize the expected overhead subject to a constraint on the expected outage rate. To simplify the problem, we assume that the channel is characterized by N packet loss rates $l_1 < \ldots < l_N$ with probabilities $p(l_1), \ldots, p(l_N)$ (Figure 4). Here a packet is considered to be lost if it is not available at the receiver within the transmission interval. Now we build for each $j \in \{1, \ldots, N\}$ a class of transmission strategies whose expected outage rate is equal to $1 - \sum_{i=1}^{j} p(l_i)$. In the next section, we provide an algorithm that selects among each class a transmission strategy that minimizes the expected overhead.

We make an optimistic guess by assuming that the packet loss rate l is minimum (equal to l_1) and start transmitting at rate R_1 from $t = s_1 = 0$ to $t = f_1$. Under this assumption, $c_1 = (k \times (1 + \epsilon))/(1 - l_1)$ is the number of encoded symbols that have to be transmitted to guarantee successful decoding. Thus we select R_1 to satisfy $R_1 \times (f_1 - s_1) = c_1$. If we denote by RTT the round trip time, an acknowledgement is expected to arrive at time $a_1 = f_1 + RTT$. Since any symbol transmitted from f_1 to a_1 may contribute to the overhead, we wait some time w_1 until $s_2 = f_1 + w_1$ before transmitting again at a rate R_2. An intuitive choice for w_1 would be $w_1 = RTT$. However, this choice may not be the best as it may not leave enough time to transmit the number of encoded symbols required to satisfy the target outage rate.

Similarly, we transmit at rate R_2 from s_2 to f_2 such that $R_2 \times (f_2 - s_2) = c_2 = (k \times (1 + \epsilon))/(1 - l_2) - c_1$. The same procedure is repeated giving transmission rates $R_1, \ldots, R_j$ $(0 < R_i \leq R_{\max},\ i = 1, \ldots, j)$ and waiting times $w_1, \ldots, w_j$ $(0 \leq w_i \leq RTT,\ i = 1, \ldots, j)$ where each transmission rate R_i, $1 \leq i \leq j$, starts at s_i and finishes at f_i (Figure 5) with

$$c_i = (k \times (1 + \epsilon))/(1 - l_i) - \sum_{m=0}^{i-1} c_m \tag{1}$$

$$R_i \times (f_i - s_i) = c_i \tag{2}$$

$$s_i = f_{i-1} + w_{i-1} \tag{3}$$

where $c_0 = f_0 = w_0 = 0$. It is easy to see that for each class j, a transmission strategy is completely defined by the transmission rates $R_1, \ldots, R_j$ and the waiting times $w_1, \ldots, w_j$.

Finally, we add the condition

$$f_j \leq T - FTT, \tag{4}$$

which states that all encoded symbols are sent within the available time budget.

Note that the transmission is stopped as soon as an acknowledgment is received. Note also that equation (1) ensures successful decoding if the packet loss rate l is smaller than or equal to l_j. It therefore guarantees that the expected outage rate which we denote by η_j is equal to $1 - \sum_{i=1}^{j} p(l_i)$.

B. Expected overhead for proposed transmission strategy

We explain how to determine the expected overhead for a transmission strategy $\pi = (R_1, \ldots, R_j, w_1, \ldots, w_j)$ in the class of transmission strategies designed for $j \in \{1, \ldots, N\}$.

Let $H_i(\pi)$ be the overhead when $l = l_i$ $(1 \leq i \leq j)$. Then the expected overhead for π is

$$E_j(\pi) = \sum_{i=1}^{j} p(l_i) \times H_i(\pi) \tag{5}$$

When $l = l_i$, the smallest number of encoded symbols necessary for successful decoding of a block will be transmitted at time f_i, and an acknowledgement will be expected at $a_i = f_i + RTT$. For all $i < m \leq j$, R_m may contribute to $H_i(\pi)$ by transmitting extra encoded symbols if $s_m < a_i$. The overhead added by R_m to $H_i(\pi)$ is $R_m \times \Omega_{i,m}$, where

$$\Omega_{i,m} = \max((\min(f_m, a_i) - s_m), 0) \tag{6}$$

is the time for which we transmit at rate R_m before a_i.

Thus $H_i(\pi)$ is the sum of the overhead added by all R_m for $i < m \leq j$ under $l = l_i$ and can be given as

$$H_i(\pi) = \begin{cases} \sum_{m=i+1}^{j} R_m \times \Omega_{i,m} & \text{if } i = 1, \ldots, j-1; \\ 0 & \text{if } i = j \end{cases} \tag{7}$$

Combining equations (5) and (7), we can write $E_j(\pi)$ as

$$\begin{aligned} E_j(\pi) &= E_j(R_1, \cdots, R_j, w_1, \cdots, w_j) \\ &= \sum_{i=1}^{j-1} p(l_i) \sum_{m=i+1}^{j} R_m \times \Omega_{i,m} \end{aligned} \tag{8}$$

On the other hand, if we use fixed-rate coding with a code rate corresponding to the maximum loss rate l_j, then the expected overhead is

$$E_j = k(1+\epsilon) \sum_{i=1}^{j-1} p(l_i) \left\{1/(1-l_j) - 1/(1-l_i)\right\} \tag{9}$$

and the expected outage rate is $\eta_j = 1 - \sum_{i=1}^{j} p(l_i)$.

C. Proposed algorithm

Given a $j \in \{1, \ldots, N\}$ and the associated class of transmission strategies described in Section III-A, our goal is to find the transmission rates and the waiting times that minimize the expected overhead subject to the expected outage rate $1 - \sum_{i=1}^{j} p(l_i)$.

For $j \in \{1, \ldots, N\}$ and $t \in [0, T - FTT]$, let us denote by $E_j^*(t)$ the smallest expected overhead achievable within the time budget $[0, t]$ and providing an expected outage rate $1 - \sum_{i=1}^{j} p(l_i)$. Let $R_{1,j}^*(t), \ldots, R_{j,j}^*(t)$ be the transmission rates and $w_{1,j}^*(t), \ldots, w_{j,j}^*(t)$ be the waiting times corresponding to $E_j^*(t)$. Then the solution to the problem is given by the transmission rates $R_{1,j}^*(T - FTT), \ldots, R_{j,j}^*(T - FTT)$ and the waiting times $w_{1,j}^*(T - FTT), \ldots, w_{j,j}^*(T - FTT)$. We propose to compute these values in a greedy (nonoptimal) way, stage by stage from $i = 1$ to $i = j$. This is done as follows.

Let $\widehat{E}_i(t), \widehat{R}_{i,i}(t), \widehat{w}_{i-1,i}(t) (i = 1, \ldots, j)$ denote the approximations to $E_i^*(t), R_{i,i}^*(t), w_{i-1,i}^*(t)$ computed by our algorithm. Then for $1 \leq i \leq j$ and $t < \sum_{m=1}^{i} c_m / R_{\max}$, we set

$$\widehat{E}_i(t) = \infty \tag{10}$$

$$\widehat{R}_{i,i}(t) = \infty \tag{11}$$

$$\widehat{w}_{i-1,i}(t) = \infty \tag{12}$$

to avoid selecting $\widehat{R}_{i,i}(t)$ and $\widehat{w}_{i-1,i}(t)$ at any stage where t is not large enough to transmit $c_1 + \cdots + c_i$ even at the highest possible transmission rate.

For $i = 1$ and $t \geq c_1 / R_{\max}$, we set

$$\left\{\begin{array}{l} \widehat{E}_1(t) = 0 \\ \widehat{R}_{1,1}(t) = c_1/t \\ \widehat{w}_{0,1}(t) = 0 \end{array}\right\} \tag{13}$$

For $1 < i \leq j$ and $t \geq \sum_{m=1}^{i} c_m / R_{\max}$, we set

$$(\widehat{R}_{i,i}(t), \widehat{w}_{i-1,i}(t)) = \arg \min_{\substack{0 < R_i \leq R_{\max} \\ 0 \leq w_{i-1} \leq RTT}} R_i \sum_{m=1}^{i-1} p(l_m) \Omega_{m,i} + \widehat{E}_{i-1}(t - \frac{c_i}{R_i} - w_{i-1}) \tag{14}$$

$$\widehat{E}_i(t) = \min_{\substack{0 < R_i \leq R_{\max} \\ 0 \leq w_{i-1} \leq RTT}} R_i \sum_{m=1}^{i-1} p(l_m) \Omega_{m,i} + \widehat{E}_{i-1}(t - \frac{c_i}{R_i} - w_{i-1}) \tag{15}$$

We see (14) and (15) as discrete optimization problems. This is done by constraining R_i to take only integer values as noninteger transmission rates are not admissible. More generally, when a packet contains more than one symbol, R_i should be constrained to take values in $M, 2M, \ldots, R_{\max}/M$, where M is the number of symbols per packet. Also, we consider time as a discrete variable with a small step size, so that w_{i-1} can take only a finite number of values.

Given j, $R_{\max}$, $l_i, p(l_i)$, $i = 1, \ldots, j$, k, ϵ, T, FTT, and RTT, the algorithm first determines $c_1, \ldots, c_j$ using (1). Then $\widehat{E}_i(t)$, $\widehat{R}_{i,i}(t)$, and $\widehat{w}_{i-1,i}(t)$ are computed using the above equations for $i = 1, \ldots, j$ and $t = 0, \ldots, T - FTT$. In the next step, the algorithm tries to find another transmission strategy that gives the same expected overhead and expected outage rate, but which uses less time to complete the transmission (i.e., it has a smaller f_j or, equivalently, a greater w_j). If one such solution can be found, it is selected because it gives the receiver more time for decoding.

A pseudo-code of the algorithm is given in Algorithm 1. The algorithm has $j \times RTT \times Q \times (T - FTT) \times Q \times R_{\max}$ time complexity and $3 \times j \times (T - FTT) \times Q$ space complexity, where Q is the number of steps per second used to quantize time. In contrast, exhaustive search has $(RTT \times R_{\max} \times Q \times Q)^j$ time complexity.

IV. Experimental results

In the first experiment, we compared our approach to fixed-rate coding for the packet loss rate probability mass function of Figure 6. For all source blocks, the number of symbols was $k = 10000$, the transmission interval was $T = 1$ s, and the maximum transmission rate was $R_{\max} = 20000$ symbols/s. A hypothetical rateless code with $\epsilon = 0.05$ was assumed. Figure 7 shows the results. Each curve corresponds to the $N = 11$ expected outage rates $\eta_j = 1 - \sum_{i=1}^{j} p(l_i)$ $(j = 1, \ldots, 11)$. The expected overhead was computed according to (8) for the optimized transmission strategy and (9) for fixed-rate coding. We used $Q = 1000$ time steps per second to quantize time. Our approach achieved lower expected overhead for the same expected outage rate. The gain increased with decreasing round trip time because short

Algorithm 1

Input: j, $R_{\max}$, $l_i, p(l_i)$, $i = 1, \dots, j$, T, FTT, RTT, k, ϵ
Output: Optimized transmission strategy $(R_1, \dots, R_j, w_1, \dots, w_j)$, expected overhead E_j, f_i, $i = 1, \dots, j$
for $i = 1$ to j **do**
  for $t = 0$ to $T - FTT$ **do**
    Compute $\widehat{E}_i(t)$, $\widehat{R}_{i,i}(t)$ and $\widehat{w}_{i-1,i}(t)$
  end for
end for
Set $f_j = T - FTT$ and $E_j = \infty$
for $t = 0$ to $T - FTT$ **do**
  if $\widehat{E}_j(T - FTT - t) \leq E_j$ **then**
    $E_j = \widehat{E}_j(T - FTT - t)$ and $f_j = T - FTT - t$
  else
    break
  end if
end for
$w_j = T - FTT - f_j$
for $i = j$ to 1 **do**
  $R_i = \widehat{R}_{i,i}(f_i)$ and $w_{i-1} = \widehat{w}_{i-1,i}(f_i)$
  $f_{i-1} = f_i - c_i/R_i - w_{i-1}$
end for

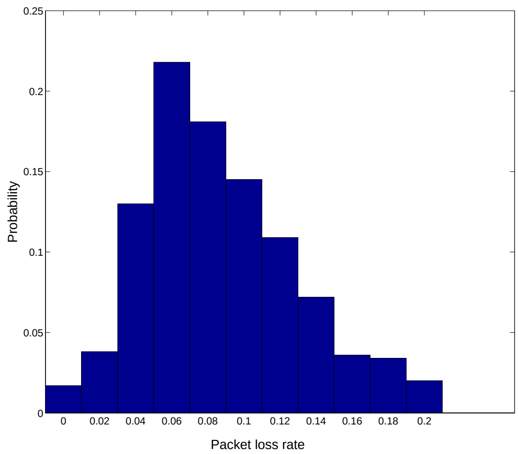


Fig. 6. Probability mass function of packet loss rate.

trip times allow the sender to quickly know the status of the receiver and stop transmitting redundant encoded symbols.

Figure 8 shows the transmission strategy that our algorithm selected to ensure zero expected outage rate when $RTT = 0.2$ s. Note how the first transmission rate was the highest one, which is a reasonable choice since the first c_1 symbols have to be sent in the shortest time to minimize overhead.

To study the efficiency of our algorithm, we compared it with exhaustive search. The comparison was done for small instances of the problem since, otherwise, exhaustive search would not be feasible. In most cases, our

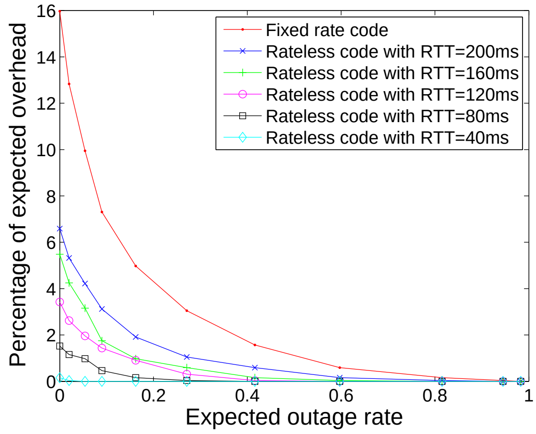


Fig. 7. Fixed-rate coding vs. proposed approach. For each curve $FTT = RTT/2$.

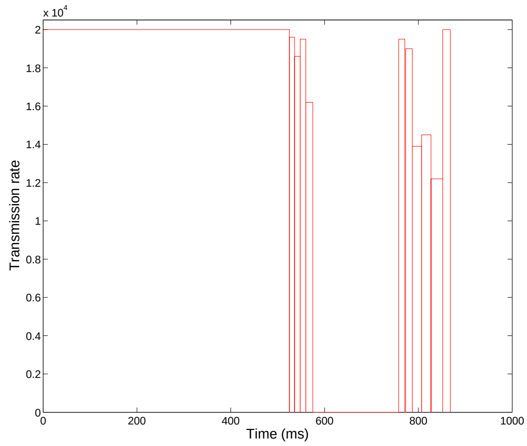


Fig. 8. Optimized transmission strategy showing transmission rates $R_1, \dots, R_{11}$ and waiting times $w_1, \dots, w_{11}$ as computed by Algorithm 1. Apart from w_5, all other waiting times are negligible.

algorithm produced an almost optimal solution. The negligible quality loss was compensated for by a significant speed up. Figure 9 shows results for $k = 130$, $T = 1$ s, $R_{\max} = 200$ symbols/s, $\epsilon = 0.05$, $RTT = 0.12$ s, and $FTT = 0.06$ s. On a PC running an Intel P4, 3 GHz processor and 1 GB RAM, exhaustive search needed 1.37 hours to compute a solution, while our algorithm took only 637 ms.

In a second experiment, we considered a real Internet connection. The connection consisted of a path Konstanz-Lahore-Konstanz. The Lahore site was used as a packet reflector server, which echoes all received UDP packets back to Konstanz. This allowed us to do all the channel measurements in Konstanz. The average forward trip time and the backward trip time were about 400 ms and 300 ms, respectively. Packet loss rates corresponding to intervals of length $T - FTT$ ($T = 2$ s) were measured during day time over periods of length 30 mn. The resulting histogram is shown in Figure 10. All packet loss

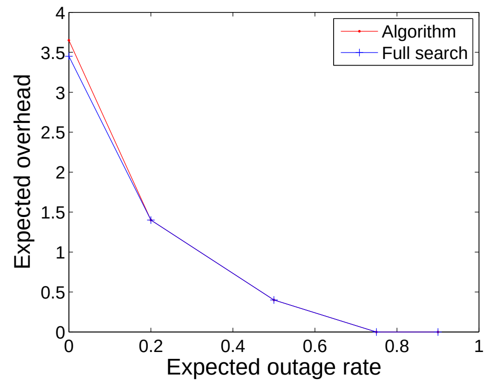


Fig. 9. Comparison of the proposed algorithm with exhaustive search.

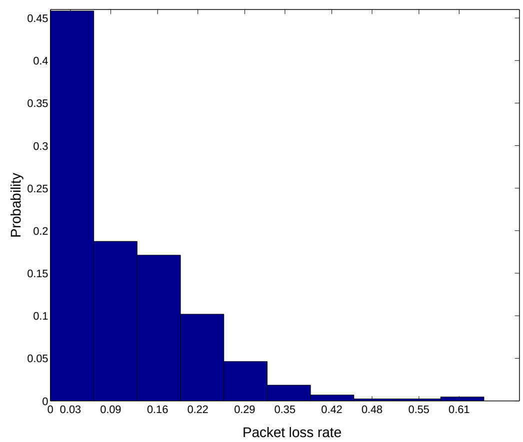


Fig. 10. Histogram of packet loss rate for a link Konstanz-Lahore-Konstanz.

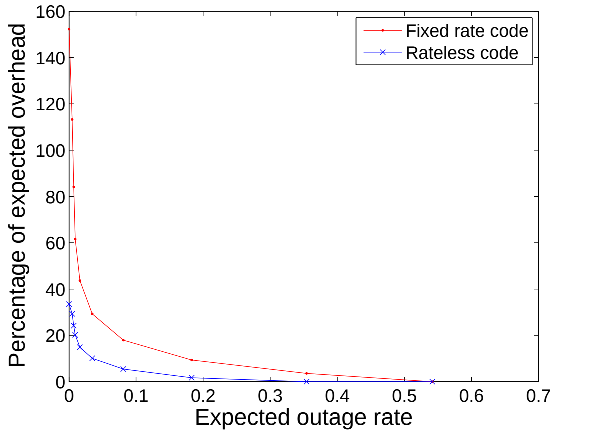


Fig. 11. Fixed-rate coding vs. proposed approach for the link Konstanz-Lahore-Konstanz.

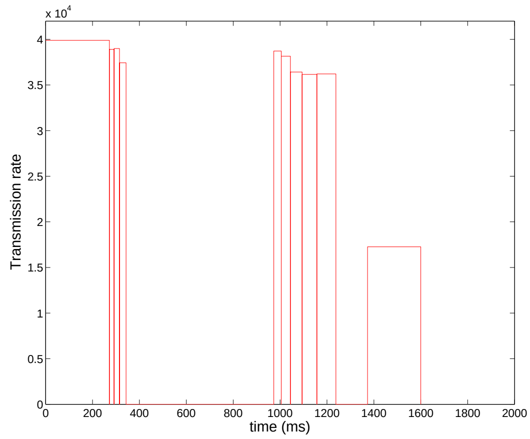


Fig. 12. Optimized transmission strategy computed by Algorithm 1 for the link Konstanz-Lahore-Konstanz. The strategy consists of 10 transmission rates and 10 waiting times. Apart from w_4 and w_9, all other waiting times are either zero or negligible.

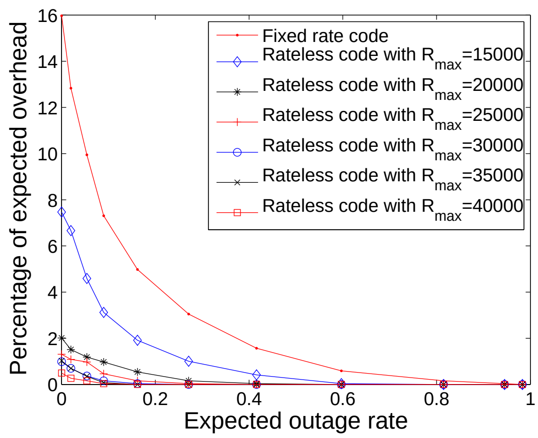


Fig. 13. Performance for various maximum transmission rates.

rates in a histogram bin were quantized to the bin center. The available bandwidth was estimated to be about 40 kilobytes/s. Figure 11 compares the performance of fixed-rate coding and our approach when $k = 10000$, $T = 2$ s, $R_{\max} = 40000$ symbols/s, and $\epsilon = 0.05$. Figure 12 shows the transmission policy selected by our algorithm to ensure a zero expected outage rate.

The available bandwidth $R_{\max}$ should be enough to transmit the number of encoded symbols corresponding to the worst-case packet loss rate. Otherwise, one has to decrease the source rate accordingly. Figure 13 shows the expected overhead when $R_{\max}$ is decreased. The results are given for the histogram of Figure 6, $RTT = 0.1$ s, $FTT = 0.05$ s, $k = 10000$, $\epsilon = 0.05$, and $T = 1$ s. The performance of our system improves with increasing $R_{\max}$ because an increase in $R_{\max}$ permits lower duration of R_i $(i = 1, \ldots, j)$, leaving more margin for w_i.

We considered another real Internet connection. This time, the remote reflector machine was situated in Minsk, Belarus. The average forward trip time and backward trip time were about 70 ms and 65 ms, respectively. Packet loss rates corresponding to intervals of length $T - FTT$ $(T = 1s)$ were measured during day time over periods of length 30 mn. The results are shown in Figure 14 and Figure 15. The available bandwidth was estimated to be

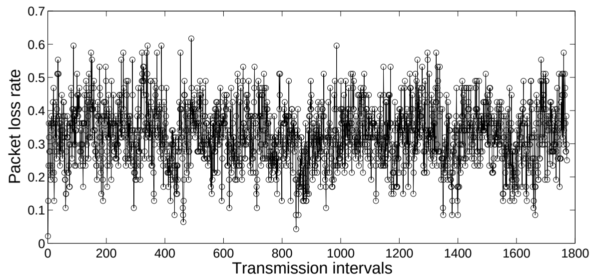


Fig. 14. Packet loss rate for the link Konstanz-Minsk-Konstanz.

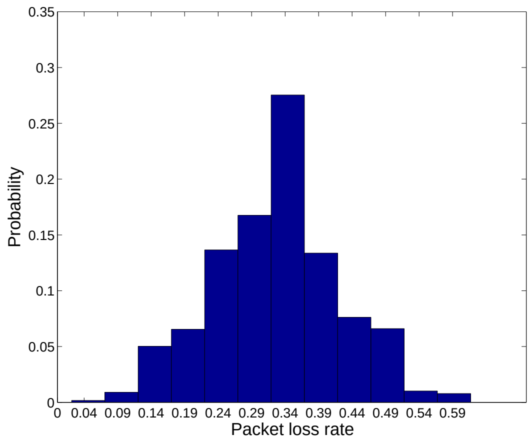


Fig. 15. Histogram of packet loss rate for the link Konstanz-Minsk-Konstanz.

about 30 kilobytes/s. Figure 16 compares the performance of fixed-rate coding and our approach when $k = 10000$, $R_{\max} = 30000$ symbols/s, $T = 1$ s, and $\epsilon = 0.05$.

V. Conclusion

We proposed a live unicast video streaming system where error control is based on forward error correction. To cope with the problem of fluctuating and unpredictable packet loss rates, we used rateless codes in conjunction with acknowledgement through a feedback channel. We developed transmission strategies that guarantee decoding success subject to a bound on the packet loss rate and gave an efficient algorithm to optimize the transmission strategies. Experiments indicate that our algorithm finds almost optimal solutions and that for the same expected outage rate these solutions use significantly less bandwidth than a scheme based on fixed-rate coding. The bandwidth saving increased with decreasing round trip times. This makes our scheme particularly appropriate for wireless channels where the round trip times are typically short.

We have assumed that a reliable feedback channel is available. In practice, this may not always be the case, and a late or lost acknowledgement would increase the overhead. In the worst case, when the packet loss rate of the feedback channel gets close to 1, the performance of the system would approach that of fixed-rate coding.

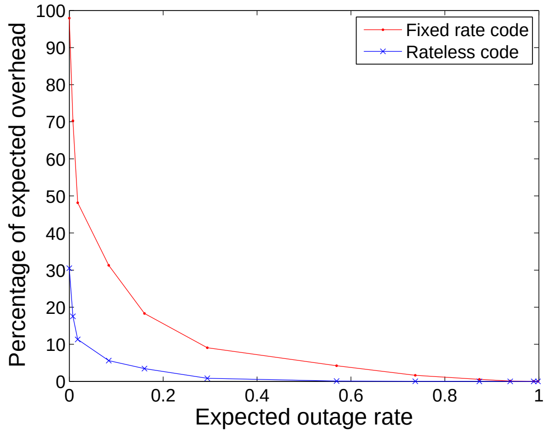


Fig. 16. Fixed-rate coding vs. proposed approach for the link Konstanz-Minsk-Konstanz.

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