Redundancy allocation in time-varying channels with long propagation delays

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Abstract

In this paper, we propose an optimal redundancy allocation over a binary symmetric channel by using recent results on a tight approximation of codeword error probability in the finite blocklength regime. We evaluate the performance of the proposed algorithm over experimental data where slowly time-varying channel conditions were measured. The rationale of this study is to enhance the performance of underwater acoustic communications, for which low bit rates are available. The performance of the proposed scheme is compared with those obtained for a constant rate allocation and a rateless scheme. Results confirm that in the presence of a time-varying channel with long propagation delays, the utilization of a robust codeword is inefficient in terms of bandwidth used and energy per bit. However, as it results in a small number of retransmissions, it also guarantees short delivery delays. Furthermore, in the case where a lower reliability can be tolerated, the proposed scheme is more efficient and also has shorter delivery delays.

Keywords: Underwater acoustic communications, redundancy allocation, reliability.

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1. Introduction

The recent development and employment of autonomous underwater vehicles, underwater sensors, and acoustic buoys motivate the interest in the design of reliable and energy-efficient underwater acoustic (UWA) communication strategies. Reliability translates into spectral efficiency. Spectral efficiency is important because the achievable data rate scales roughly with available bandwidth, which is a scarce resource in the UWA channel. At the same time, since transmission is the most energy consuming activity of an UWA modem [1, 2], spectral efficiency results in energy savings that are vital to keep the aforementioned devices operational for a longer time, given that both unsuccessful transmissions and unnecessary redundancy are reduced as much as possible.

Following this reasoning, in this paper we pose and solve a simple optimization problem, from which we then design a redundancy allocation scheme for packets transmitted over a time-varying channel with long propagation delays. Throughout this paper, we assume that the bit error process caused by this time-varying channel has a constant probability over the duration of a single packet, but it can change over different packets. Therefore, we indicate as non-stationary conditions those when the crossover probability changes over different packets. In particular, a Binary Symmetric Channel (BSC) model is assumed to be representative for the bit error process at the output of an adaptive equalizer that compensates for time-varying attenuation and intersymbol interference, typical of the underwater acoustic channel. Then, we compute the amount of redundancy that maximizes a metric that combines spectral efficiency and reliability. Consequently, we design a realtime algorithm that computes and allocates the amount of redundancy required to maintain the desired quality of service. Such redundancy is computed based on some channel state information that is estimated from a previously received control message, such as a probing sequence. The performance of the proposed scheme is compared with those obtained for a constant rate allocation and a rateless scheme. Results confirm that in the presence of a time-varying channel with long propagation delays, the utilization of a robust codeword is inefficient in terms of used bandwidth and energy per bit. However, as it guarantees a smaller number of

retransmissions, it also results in short delivery delays. When a lower reliability can be tolerated, the proposed scheme becomes more efficient and has shorter delivery delays.

As widely investigated in the past few years for terrestrial wireless communications, e.g., see [3, 4, 5, 6, 7], energy saving can be achieved by designing scheduling schemes characterized by low-power and long (low-rate) transmissions, as also proposed in [8]. In addition, these energy-efficient transmission schemes assume capacity achieving codes, which implies longer codewords, especially in poor channel conditions. The relatively low symbol rate supported by the underwater channel along with the decrease in the data rate (bits/symbol) to enable lower power operation and the attempt to get close to capacity at this reduced power level combine to give very long duration codewords. As a result, the channel is time-varying over the duration of a codeword while in the aforementioned literature it is assumed to be time-invariant. This is particularly likely in UWA scenarios, since the propagation of acoustic waves depends on stochastic processes that are in turn time-varying, such as surface waves, currents, and how the sound speed profile is statistically distributed in time and space. In this paper, we move towards a more realistic representation of the time-varying channel by assuming that the channel statistics can vary among different codewords, but it remains stationary during a packet transmission.

In the literature, the problem of reliable UWA communications has been studied, e.g., in [9, 10, 11, 12, 13, 14]. In particular, the authors of [9, 10] investigate the performance of rateless coding schemes used for broadcast communications. In [11], the authors study the tradeoff between delay and reliability in a multi-hop sensor network. The authors of [12] perform a simulation study on the reliability achieved by the proposed Automatic Repeat reQuest (ARQ) technique, tailored to UWA communications. Even though these studies provide insight on the reliability associated to a multi-user channel in different networking scenarios, the presented results are limited to the case of time-invariant channel fading, since only the dependence on distance is considered.

In contrast to this previous work, we focus on the unreliability due to the time variability of the statistics for the underwater acoustic channel. In particular, we design and evaluate a scheme that allocates the redundancy required to protect UWA transmissions, based on some channel state information (CSI). This CSI is estimated at the

transmitter by decoding a probing sequence piggy-backed in the ACKs/NACKs sent by the receiver to the transmitter, upon the receipt of a packet.

In [14], the authors design a super-Nyquist modulation and rateless coding scheme
65 suitable for doubly-selective underwater acoustic channels. However, they do not explicitly study how to optimally allocate the redundancy to decrease the number of re-transmissions. The authors of [15] propose a rateless coding scheme, whose distribution adapts to the fading conditions and which runs based on limited CSI (ACK/NACK) available at the transmitter. They show that this adaptive rateless code outperforms the
70 standard rateless codes in terms of throughput. In this work a similar adaptive coding scheme is considered. However, the proposed scheme consists in adapting the coding rate depending on the time-varying communication performance at each single transmission.

The structure of the present paper is summarized as follows. In Sec. 2, we present
75 the considered scenario and the statement of the problem together with some related work. In Sec. 3, three allocation schemes are described: the proposed one, a constant allocation and a rateless scheme. In Sec. 4, we evaluate the redundancy allocations over experimental data, and therefore we first describe the data set, and then we present the results in terms of reliability and delay. Sec. 5 concludes the paper.

80 **2. Statement of the problem and related work**

In this section, we first describe the scenario by detailing the assumed properties of the wireless communication system under study. In particular, we discuss the channel and the interference models, and we comment on the traffic model, and on the feedback. We then present some relevant results found in the recent literature, which are leveraged
85 when defining the proposed allocation scheme. More specifically, the focus is on the tight upper bounds derived analytically for the average and maximal error probability for a codeword in the finite blocklength regime.

As a reminder, the main problem addressed in this work is the investigation of the most efficient redundancy allocation in the scenario considered. In this scenario, the
90 wireless communication channel is described by a constant crossover probability over

the duration of a codeword. However, as propagation delays are large relative to the correlation time of the channel fluctuations, the crossover probability can vary among different packets.

2.1. Description of the scenario

95 The considered scenario consists of a point-to-point wireless communication link between a transmitter and a receiver. The propagation delay, τ , between the two is assumed to be much larger than the duration of a packet, D_{pkt} , e.g., $\tau = 12.5D_{pkt}$. This is typical in an underwater acoustic communication system, where packets 1 kb long at 6.25 kbps lasts 160 ms and the propagation delay is 2 s, e.g., in a link that is 3
100 km long with a nominal sound velocity of around 1500 m/s.

Both transmitter and receiver are assumed static, thus making the assumption of a slowly time-varying channel acceptable, as the communication channel only varies with the same rate of the changes in the surrounding environment. A slow time-varying quality of the communication means that the random process for bit errors is constant
105 over the duration of a packet, but can change among different packets. In particular, the error process inside a packet is assumed to be binary, i.e., bits can be flipped, and symmetric, i.e., errors on 0 s and 1 s are equiprobable, and therefore it can be represented by one parameter, δ , which is the bit error probability. Furthermore, bit errors are assumed to be independent of each other, which is reasonable in the presence
110 of an interleaver that compensates for a possible memory in the error process due to inter-symbol interference.

Multiple-user interference is absent, as a medium access control has previously allocated the channel to the considered transmitter-receiver pair. This means that the communication performance depends solely on the channel quality. In fact, our main
115 focus is on making it efficient to transmit as many bits as possible. As typical applications of underwater networks, where a network is defined as two or more nodes communicating, require very few nodes, we decided to first address the problem of a reliable and efficient communication in a point-to-point link, and leave that in a multiple-users scenario as a next step in our future work.

120 The traffic model consists of a backlogged queue at the transmitter side, i.e., a

continuous flow of data keeps arriving to a queue with infinite capacity. The receiver is allowed to feed back both an ACK upon the correct decoding of a packet and a probing sequence used by the transmitter to monitor the communication performance of the link. As the communication system is half-duplex, this results in a packet delivery
125 delay that lasts at least a round trip time, which is 2τ , plus the duration of both the data packet and the probing sequence.

2.2. Related work

Recently, the authors of [16] have derived new tight upper bounds for the average and maximal error probability achievable with a given rate and blocklength. In the
130 following, we limit the presentation to the results for the BSC, even though the scheme proposed in Sec. 3.1 is independent of the type of channel, as the only requirement is to have an analytical expression for the upper bound of the maximal error probability.

The codeword error probability is indicated with ϵ and depends on: *i*) the number of information bits, x , *ii*) the number of redundancy bits, y , *iii*) the channel condi-
135 tions, here for the BSC represented by $\delta \in]0, 1[$ and $\delta \neq \frac{1}{2}$. We indicate with n the blocklength of a codeword. The expression for the upper bound of the maximal error probability is derived from the computation of the achievable rate in the asymptotic regime once the blocklength, n , and the codeword error probability, ϵ , are given.

The asymptotic behavior, i.e., for $n \rightarrow \infty$ and $\epsilon \rightarrow 0$, is also compared with the
140 non-asymptotic upper and lower bounds derived in [17, Sec. III]. It turns out that the asymptotic expression is accurate even for short blocklengths and for rates much lower than the capacity, and for this reason we use the upper bound obtained in the asymptotic behavior. The achievable rate can be expressed as:

$$\log M(n, \epsilon) = nC(\delta) - \sqrt{n\delta(1-\delta)} \log\left(\frac{1-\delta}{\delta}\right) Q^{-1}(\epsilon) + \frac{1}{2} \log n + O(1). \quad (1)$$

$C(\delta)$ is the capacity of a BSC with crossover probability, δ , i.e., $C(\delta) = 1 - h(\delta)$,
145 where $h(\delta)$ is the entropy:

$$h(\delta) = -\delta \log(\delta) - (1-\delta) \log(1-\delta). \quad (2)$$

Figure 1: Representation of the communication system with the proposed redundancy allocation.

Q^{-1} is the inverse of the standard Gaussian complementary cumulative distribution function. Therefore, the codeword maximal error probability, ϵ , as a function of the number of information bits, x , the redundancy, y , and the crossover probability, δ , can then be written as:

$$\epsilon(x, y, \delta) = Q\left(\frac{(x+y)C(\delta) - x + \frac{1}{2}\log(x+y)}{\sqrt{(x+y)\delta(1-\delta)\log^2\left(\frac{1-\delta}{\delta}\right)}}\right). \quad (3)$$

3. Redundancy allocation techniques

In this section, we describe the three allocation techniques that are then evaluated using experimental data in Sec. 4. These are an adaptive allocation, a constant redundancy allocation, and a rateless scheme.

3.1. Proposed adaptive redundancy allocation

The idea underlying the proposed allocation consists in adjusting the blocklength, n , depending on the current crossover probability, so that the resulting codeword is long enough to reliably deliver x bits, but not too long so that bandwidth is not wasted. A representation is shown in Fig. 1. Packets consist of a constant blue portion, which contains the information bits, and of a green variable portion made of redundancy bits. The duration of a time slot is Round Trip Time (RTT) + D_{pkt} + D_{prob} , where D_{pkt} is

the duration of the codeword and D_{prob} is the duration of the probing sequence. Note that as D_{pkt} is allowed to vary, so is the time slot duration.

This approach differs from using the achievable rate, as it indicates the maximum number of information bits that can be transmitted with a codeword n bits long when
 165 a maximal codeword error probability, ϵ , is tolerated. However, tracking the evolution of the codeword error probability, ϵ , for time-varying channels could be costly in terms of number of packets that need to be transmitted, and inaccurate in the case of long propagation delays as experienced in underwater acoustic communications. For this reason, we would rather adjust the blocklength based on the crossover probability, δ ,
 170 which requires the transmission of only a probing sequence.

Therefore, we propose the following framework. We first define an ‘efficiency’ metric:

$$\eta(x, y, \delta) = \frac{x(1 - \epsilon(x, y, \delta))}{x + y}, \quad (4)$$

where $\epsilon(x, y, \delta)$ is the codeword error probability obtained in Eq. (3). η represents the trade-off between reliability, indicated by the factor $1 - \epsilon$, and spectral efficiency,
 175 expressed by the ratio $x/(x + y)$. Therefore, η is a decreasing function of y , and an increasing function of $1 - \epsilon$. However, $1 - \epsilon$ itself is an increasing function of y , thus revealing a trade-off, which depends on how rapidly ϵ decreases as the redundancy increases. This trade-off gives rise to an optimum value y_{opt} that maximizes η . In other words, transmitting unnecessary bits for redundancy is a waste of power and
 180 bandwidth but providing insufficient redundancy results in a rise in packet failures and accompanying requests for retransmission which decreases efficiency.

The simple optimization problem is formalized as follows. Given the number of information bits x , and the bit error probability δ , find the amount of redundancy, y_{opt} maximizing η , i.e.,:

$$y_{opt} = \underset{y}{\operatorname{argmax}} \eta(x, \epsilon, y). \quad (5)$$

185 A maximizer is found by studying the sign of the first derivative of $\eta(x, y, \delta)$ with

Figure 2: η and its first derivative are represented by the green (dashed) and blue curves, respectively. In this case, $x = 200$ bits and $\delta = 0.046$.

respect to y , in symbols:

$$\frac{\partial \eta}{\partial y} = \partial_y \left(\frac{x(1 - \epsilon(x, y, \delta))}{x + y} \right) \quad (6)$$

$$= \frac{-(x + y)x\partial_y \epsilon(x, y, \delta) - x(1 - \epsilon(x, y, \delta))}{(x + y)^2}. \quad (7)$$

In the numerator, the first term has only positive factors, with the exception of $\partial_y \epsilon(x, y, \delta)$, which is negative, since ϵ is a strictly decreasing function of y . This makes the first term positive. Conversely, $x(1 - \epsilon(x, y, \delta))$ is non-negative, since $x > 0$ and $\epsilon(x, y, \delta) \in [0, 1]$, thus making the second term non-positive. These considerations highlight that there exists a trade-off between the two terms, which determine the sign of $\partial_y \eta(x, y, \delta)$ as a function of y .

The derived maximizers are here shown in the following for values of x lying in the interval $[200, 1000]$, with an increment of 50 bits, and for values of δ spanning the interval $[10^{-3}, 0.1]$, with increments of 0.005. In the basic case where no errors occurs, $\delta = 0$, then no redundancy is allocated. In particular, Fig. 2 represents η plotted with a dashed green line and its first derivative, plotted with a solid blue line, as a function of

Figure 3: The y -axis represents the optimal number of redundancy bits, obtained from the optimization framework, for varying x (represented by the different curves) and δ (in the x -axis).

y for $x = 200$ and $\delta = 0.046$.

Fig. 3 shows the maximizers of η as a function of δ in the aforementioned intervals. Each curve represents the optimal amount of redundancy for the considered values of x . In Sec. 4, we will use the values of y_{opt} obtained for $x = 1000$, i.e., the upper curve.

Algorithms 1 and 2 summarize the proposed scheme at the transmitter and at the receiver, respectively. In few words, we assume that a lookup table of δ and y_{opt} for the given x is stored at the transmitter. Then, at the beginning of each slot, the receiver sends a probing sequence, from which $\hat{\delta}$ is estimated by the transmitter. The smallest δ larger than $\hat{\delta}$, δ^* , is then found in the lookup table, and the codeword to be transmitted will have blocklength equal to $x + y_{opt}(\delta^*)$. If the bit error rate on the received codewords is not larger than δ^* , then the packet is successful.

In Sec. 4, we evaluate this scheme using experimental data and we also compare its performance with those obtained when a constant redundancy allocation and a rateless scheme are used. Therefore, for completeness we describe both these other schemes

Algorithm 1 Algorithm at the transmitter side for the proposed scheme.

Input: x information bits and look-up table (δ, y_{opt}) ;

Procedure:

- Decode the probing sequence;
- Measure $\hat{\delta}$ upon probing sequence receipt;

if $\hat{\delta} \leq 0.1$ **then**

- Find in the look-up table $\delta^* = \{\min \delta, \text{s.t. } \delta > \hat{\delta}\}$;
- Select the corresponding redundancy, $y_{opt}(\delta^*)$;

Output: send a packet of length $x + y_{opt}(\delta^*)$.

else

Output: repeat from the beginning of the procedure at the next slot.

end if

Algorithm 2 Algorithm at the receiver side for the proposed scheme.

Input: x number of information bits and look-up table (δ, y_{opt}) ;

Procedure:

- Send a probing sequence;
- Wait for a packet;

if A packet is received: **then**

- Select the used blocklength as indicated in the header of the packet;

Output: decode the packet.

else

Output: repeat from the beginning of the procedure at the next slot..

end if

Figure 4: Representation of the communication system with constant redundancy allocation.

in the following by detailing the assumptions that we made in the implementation of these schemes.

215 3.2. Constant redundancy allocation

A pictorial representation of the constant redundancy allocation is in Fig. 4. The transmitter sends a codeword with a constant blocklength. Such constant blocklength is determined by using the maximizer of η for a given bit error probability, δ_i and x , as described in Sec. 3.1. Therefore, whenever the bit error rate measured at the receiver, δ_{Rx} , is larger than the δ_i used to compute the constant blocklength, the packet will fail. Therefore, a probing sequence, piggybacked to an ACK if a transmission was made in the previous slot, is sent at the beginning of the slot by the receiver. If the bit error rate measured at the transmitter, δ_{Tx} , is not larger than δ_i , then a transmission is attempted, otherwise it is deferred until the next slot.

225 Also in Fig. 4 the blue portion of a packet contains information bits and the green part the constant number of redundancy bits. The red smaller packets sent by the receiver represent the probing sequence with a possible piggybacked ACK/NACK in case a transmission was attempted in the previous slot.

In Sec. 4, we consider two levels of robustness for the communication, i.e., when:
 230 a) redundancy is small, $y_{opt} = 507$, corresponding to $\delta_i = 10^{-3}$, and b) redundancy is large $y_{opt} = 3732$, corresponding to $\delta_i = 0.1$. Therefore the blocklength for case a) is

Figure 5: Representation of the communication system with a rateless coding scheme.

almost a third of that for case b), but it is also less robust. This comparison is meant to highlight the advantages and disadvantages of choosing low code rates in the context of underwater acoustic communications.

235 3.3. Rateless scheme

A rateless scheme consists in transmitting additional redundancy if the codeword made of the bits transmitted so far is not correctly received. As the amount of redundancy is adjusted on the fly and no rate is predefined, this scheme is called rateless. Its advantage over the constant allocation consists in transmitting as much redundancy as
240 required to counteract the time-varying levels of reliability seen at each transmission.

A representation can be found in Fig. 5. First, all the information bits are transmitted, and then if they have not been correctly decoded by the receiver additional redundancy is transmitted. We assume that the packet length is equal to the smallest amount of redundancy derived in Sec. 3.1, i.e., 507 bits. For this reason, a single infor-
245 mation frame, which is 1000 information bits long plus additional redundancy, requires the transmission of multiple packets¹. This choice of a short packet makes it possible to maintain a good resolution in adjusting the amount of redundancy, even though it taxes the delay with which a single information frame is delivered.

¹In the performed simulations, the information bits consist of 2 packets.

Every time a new packet is received, a decoding process starts anew with a code-
 250 word that consists of the bits sent up to the current slot juxtaposed with the additional
 redundancy received at the current slot. An information frame consists of at most 9
 packets, which correspond to the longest blocklength used in the allocations described
 in Secs. 3.1 and 3.2. A probing sequence is sent in order to estimate the bit error rate
 and in case this is larger than $\delta_{max} = 0.1$ (which can be corrected with an information
 255 frame of 9 packets), then a transmission is deferred to the next slot.

Algorithms 3 and 4 summarize the procedures performed at the transmitter and
 receiver, respectively.

Algorithm 3 Algorithm at the transmitter side for the rateless scheme.

Input: x information bits, packet size, δ_{max} , N number of transmissions for the
 current information frame);

Procedure:

- Decode the probing sequence;
- Measure $\hat{\delta}$ upon probing sequence receipt;

if $\hat{\delta} \leq 0.1$ **then**

if $N \leq 9$ **then**

- Transmit a packet for the same information frame;
- Increment N ;

else

- Transmit a packet for a new information frame;

end if

else

Output: repeat from the beginning of the procedure at the next slot.

end if

4. Evaluation over experimental data

In this section, we evaluate the described schemes using acoustic data collected at
 260 sea. In Sec. 4.1, we provide some details on how the data set was collected and the

Algorithm 4 Algorithm at the receiver side for the rateless scheme.

Input: N packets;**Procedure:**

- Send a probing sequence;
- Wait for a packet;

if A packet is received and $N \leq 9$: **then**

- Add in the current information frame the received packet;
- Increment N ;

Output: decode the packet and send ACK/NACK together with probing sequence in the next slot.**else****Output:** repeat from the beginning of the procedure at the next slot..**end if**

specifications of the communication system. Then in Secs. 4.2 and 4.3, we discuss the obtained performance in terms of reliability, efficiency, and delay.

4.1. Description of data set

The set of acoustic signals used in this paper were recorded during the KAM11
265 experimental trial [18]. Every two hours, bit streams made of nine one-minute long
sequences were transmitted with a BPSK modulation, at center frequency 13 kHz and
rate 6250 bps. We consider the signals recorded at 3 km from the transmitter. Both
the transmitter and the receiver were in the middle of the water column, i.e., at 45 m
below the surface. The receiver consisted of an array of 24 hydrophones, of which
270 we used 4 equally separated in space. The transmitter and the receiver were placed
along an isobaric line. The results shown are relative to two Julian dates (JD), 187 at
4 am GMT and 188 at 10 am GMT, where the channel impulse responses were quite
impulsive, as it can be observed from Figs. 6 and 7.

Figs. 6 and 7 contain examples of the time series of the squared magnitude of the
275 channel impulse response measured during the bit streams. The x-axis represents the
delay referenced to some arbitrary zero delay point, with which different paths arrive

Figure 6: Time series of the squared magnitude (in dB with respect to the maximum value indicated) of the channel impulse response during Julian date 187 at GMT 03:52 am.

at the receiver (and it is not related to the propagation delay, $\tau = RTT/2$). Since this stream of bits consists of trains of m-sequences, we estimated each channel impulse response by computing the auto-correlation of the 4095 bit long m-sequence. The
 280 colormap of these figures corresponds to the value in dB relative to the maximum.

These two epochs are examples of when the channel impulse response contains only a few significant arrival, whose amplitude however varies significantly over time. Such variability is more likely to be caused by the focusing effect of surface waves. Since the arrival structure is quite concentrated over a main arrival, the channel estima-
 285 tion requires a small number of symbols. In fact, the computed results are obtained by setting the length of the feedforward filter of the Decision Feedback Equalizer (DFE²) derived in [19], to 0.5 ms, and the length of the feedback filter to 1.5 ms.

²The equalizer is used in training mode. It combines the signals measured at four channels spanning half a meter of the water column.

Figure 7: Time series of the squared magnitude (in dB with respect to the maximum value indicated) of the channel impulse response during Julian date 188 at GMT 09:52 am.

The time series of bit error rates computed over the 4095 bit long sequences is shown in Fig. 8, where the black markers refer to JD 187 and the red ones to JD 188.

290 The y-axis represents the time in seconds whereas the x-axis represents the bit error rate (or crossover probability δ). It can be noticed how the bit error rate drops from second 250 on in the black time series, i.e., as soon as the intensity of the main arrival increases as shown in Fig. 6. Similarly, during JD 188, the bit error rate increases when the main arrival fades around second 250.

295 In light of these observations, we now evaluate the three systems described earlier using these time series. These bit streams are therefore downsampled in order to reproduce the same evolution in time, i.e., accounting for the proper RTT that would be experienced at sea. This also means that as each scheme has different packet lengths, it can happen that transmissions occur at different times, and therefore at different channel conditions. Even though comparing the performance obtained by this study

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Figure 8: Time series of the bit error rate (or crossover probability) per packet estimated over packets 4095 bits long. Black and red markers correspond to JD 187 and 188, respectively.

might not be completely fair, as the three systems can downsample the time-varying error process at different times, it however helps in understanding the evolution and the behavior of the performance that would occur in a real scenario.

To better understand this, we show in Fig. 9 the time series of the SNR at the
 305 output of the equalizer, estimated over the packets that were successfully transmitted
 for each scheme. Blue and red markers refer to the constant redundancy allocation
 with a small and large robustness, respectively. As the blue markers correspond to less
 robust transmissions, they basically occur only at high SNR (in fact no blue markers can
 be found between 60 and 250 s), whereas more robust codewords can better cope with
 310 deep fades, as those around 60 and 250 s for both epochs. The black markers represent
 the SNR estimated over the packets with variable robustness, as shown by whether
 these markers are higher or lower than the red ones for some periods of time. This
 effect is caused by the slightly different sampling time of the time series, and by the

Figure 9: Time series of the SNR measured at successful transmissions made with the discussed allocation schemes. Circles and stars correspond to JD 187 and 188, respectively.

fact that the estimate of the SNR at the output of the equalizer decreases as the channel
 315 impulse response varies, which is more likely during larger packets. This effect is even
 stronger for the SNR estimated over the information frames of the rateless scheme.
 In fact, when the SNR is low, more redundant transmissions are requested. However,
 these packets are separated in time by at least 4 s (which is the RTT). This means that
 the assembled information frame contains more diversity in the channel conditions, and
 320 the DFE performs worse, thus giving rise to lower SNR.

Figure 10: Time series of the reliability per delivered packet for the allocation schemes.

4.2. Efficiency

In this section, we analyze the performance in terms of reliability, effective transmission rate, and efficiency, per successful packet. In particular, the reliability is computed as the inverse of the number of transmission attempts made before the successful delivery of x bits of information. In other words, retransmissions are performed until the packet is delivered successfully. Fig. 10 shows the time series of the reliability for

325

the different schemes. When a transmission is attempted³ and successful at the first shot then the reliability is 1. This happens for the majority of the transmissions made by the scheme with constant reliability and large redundancy. Since the CSI estimated
330 at the transmitter can become outdated, some retransmissions are also requested in the proposed scheme, thus reducing the reliability. For the rateless scheme, the reliability is computed as 2 over the number of transmissions attempts. In fact, in order to transmit 1000 bits, the transmitter needs to send at least 2 packets. For this scheme many more attempts per successful packet are performed due to the lower SNR experienced
335 by the information frames, as previously explained with Fig. 9.

The efficiency per delivered packet is then computed as the product of the reliability and the ratio between x and the overall number of bits transmitted until the delivery of the packet. Its time series is shown in Fig. 11 for the different allocation schemes. The constant rate allocations have mostly a constant efficiency equal to the rate, as the first
340 transmission attempt is also successful in the most cases. Instead, the proposed scheme presents a higher efficiency on average, except in those cases where outdated estimates of the bit error rate mislead the rate selection process and more transmissions have to be performed. The rateless scheme is on average much less efficient than the others as it requires many transmissions. Therefore, if reliability is the most important metric,
345 then a robust fixed rate allocation would be the right choice for underwater acoustic communications. However, if also efficiency is important, i.e., if the amount of energy put in the water should also be minimized, then the proposed scheme outperforms the others at the cost of a slightly lower reliability, caused by outdated estimate of the crossover probability.

350 Fig. 12 represents the actual transmission rate for each value of the reliability per successful packet. The actual transmission rate is the ratio between x and the overall number of bits transmitted until the packet is correctly decoded. This figure is useful to understand whether the decrease of the actual rate is due to the loss of packets or to the allocation of more redundancy. The reliability of a constant redundancy alloca-

³We recall that if the $\hat{\delta}$ estimated over the probing sequence is too large then a transmission is deferred, as mostly happens for short packets (blue markers) between 50 s and 250 s.

Figure 11: Time series of the efficiency per successful packet for the analyzed schemes.

tion is mostly 1, when the transmission is attempted as we saw in Fig. 10. Therefore most of the realizations fall in the pair $(1, x/n)$ for both blue and red markers. The two other blue markers and the red marker corresponding to a smaller reliability are seldom obtained when the transmission attempt fails. For the proposed scheme, the multiple values of rate for reliability 1 indicate that the system is actually adapting to the channel conditions in an efficient way. Most of the points are in this area of the plot. Similarly, the several values of the rate for a given level of reliability indicates that even though the system adapts to the channel conditions, it overestimates the quality of the communication performance by relying on an outdated information. The smaller the

Figure 12: Actual transmission rate vs reliability per successful packet for the considered schemes.

reliability (which implies a moment of low SNR), the fewer the options in selecting
 365 the rate, which implies that the vertical lines collapse to one point. The rate obtained
 with the rateless scheme is linearly dependent on the reliability, since an information
 frame fails due to a low SNR at the output of the equalizer. Such decrease of SNR is
 due to the fact that packets with different channel conditions (at least 4 s apart) are jux-
 taposed, thus taxing the performance of the equalizer. This effect remains until enough
 370 redundancy is transmitted to balance this loss of communication performance due to
 the channel variability.

Figure 13: Time series of the packet delivery delay per successful packet for the analyzed allocation schemes.

4.3. Packet delivery delay

Another important metric is the time measured between two successful consecutive packets, i.e., the delivery delay. Fig. 13 shows the time series of this delay per successful packet for the different allocation schemes. The fixed allocation with large redundancy, indicated with red markers, has small delay as the first attempt is also successful. The proposed scheme has slightly lower delays when the first attempt is successful, but since the RTT (4 s) is much larger than the packet length, having adaptive blocklengths is not much advantageous in terms of delay. The rateless scheme has the largest delay as 2 packets need to be transmitted just for the information bits.

Figure 14: Efficiency vs delay for the analyzed allocation schemes

Fig. 14 represents the results for the measured efficiency given the delivery delay for all schemes. Note that each point represents different occurrences of the specific (η, delay) pair, and for this reason, this plot should be interpreted by also referring to the previously presented time series in Figs. 13 and 11. In particular, the red and blue markers corresponding to small values of delays are more frequent as shown in the time series, and therefore more representative. From this figure, we can conclude that the proposed scheme has higher efficiency than the other schemes without affecting the delivery delay. The considered rateless scheme indicated with green markers has instead the worst performance in terms of delay, but when the channel conditions are

390 favorable, it can be more efficient than a constant rate allocation scheme.

5. Conclusions

In this paper, we have proposed a redundancy allocation that leverages recent results on the maximal error probability for fixed rate and finite blocklength. We have then evaluated the performance of the proposed scheme in terms of reliability, efficiency, and delay. Such evaluation was performed over experimental data collected
395 over a link 3 km long.

The performance of the proposed scheme has been compared with those obtained for a constant rate allocation and a rateless scheme. Results confirm that in the presence of a time-varying channel with long propagation delays, the utilization of a robust
400 codeword is inefficient in terms of used bandwidth and energy per bit. However, as it results in fewer retransmissions, it also guarantees short delivery delays. Therefore, in the case where a lower reliability can be tolerated, the proposed scheme provides higher efficiency and shorter delivery delays.

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Photo Daniele Munaretto