Protocol Coding for Two-Way Communications with Half-Duplex Constraints

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***Abstract*—The operation of communication protocols is conventionally independent of the information being transmitted. This paper puts forth the notion of *protocol coding* to refer to transmission strategies in which data can be encoded by modulating the protocol actions according to the information message. We focus on communication in the presence of half-duplex constraints, where the task of the protocol is to schedule the transmission/reception times of different nodes. Such a schedule is conventionally decided *a priori* in the form of time-sharing. While previous work has focused on protocol coding for standard relay channels, this paper tackles *two-way* communications aided by a relay. Since the techniques developed for standard relay channels cannot be applied to the scenario at hand, a novel simple strategy is proposed that is tailored to two-way communications. The proposed scheme is shown to significantly outperform conventional time-sharing for a deterministic two-way relay channel.**

I. Introduction

Design of communication systems has so far largely upheld a dichotomy between, on the one hand, *transmission techniques* and, on the other hand, *communication protocols*. For instance, consider a multi-hop model with a single source S communicating to a destination D via a string of half-duplex relays, as in Fig. 1. The *communication protocol* decides which subset of nodes (source or relays) has the right to transmit at a given time. The *transmission technique* refers to the selection of codebooks (i.e., modulation, codes) and decoders to be employed during the transmission times scheduled by the transmission protocol. The defining aspect of this dichotomy is that the decision made at the communication protocol level is independent of the current information sources (messages) being communicated: It is fixed a priori and made known to all nodes. A relevant question is: How much additional information can be sent if one does not enforce the separation between protocol and transmission technique? In other words, what is the information value of operating the communication protocol on the basis of the current messages?

A. Protocol Coding

Let us focus on the multi-hop network mentioned above (Fig. 1) and assume noiseless links between pairs of adjacent nodes along the path between source to destination. Each node can either send a "1" symbol or a "0" (silent) symbol to the following node. Each node is able to receive what is sent by the preceding node only if in receive mode (i.e., if sending a "0"), due to *half-duplex* constraints. The standard design approach would prescribe *time-sharing* on each link: The transmitter of each link is silent half of the time (i.e., it sends "0"), so that it can receive the signal sent by the preceding node, while transmitting the other half of the time (i.e., it sends either "0" or "1"). By appropriately scheduling the transmit and receive periods at successive nodes, in a fixed and pre-determined fashion, the end-to-end rate of this scheme is easily seen to be 0.5 bit/ channel use (bits/c.u.). Reference [3] demonstrates that, if one instead encodes information into transmission modes (e.g., transmit or receive mode), without fixing them a priori as in time sharing, the achievable rate can be largely improved. For instance, with two hops one can obtain 0.7729 bits/c.u., while for a large number of hops one achieves 0.6942 bits/c.u.. The key idea is that the receiving mode of a node can be interpreted as part of its transmission alphabet: "Transmitting" the silent symbol can thus convey information to the next node, while the node itself is still able to receive at the same time. This allows performance improvement over a pure time-sharing strategy.

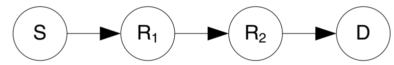

Fig. 1. Unidirectional transmission from the source S to the destination D through half–duplex relays R_i.

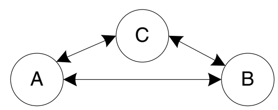

Fig. 2. Two–way communication between A and B assisted by the relay C. All nodes are half–duplex

This idea of encoding information via the *protocol-level* choice of whether to transmit or receive, can be seen as an instance of cross-layer design, and has been first put forth by [2] for a single-relay channel. We propose to generalize this approach via the notion of *protocol coding*: This refers to transmission strategies that convey additional information by utilizing the degrees of freedom available when deciding the set of actions to be followed by a communication protocol. Besides the work of [2] and [3], the literature features several instances of protocol coding as defined above. An early work that mentions the possibility to use data in order to impact the way random access is done is [4][1]. In [10], information is modulated in the arrival times of data packets. In a more recent work [11], protocol coding is performed by modulating additional data in the order in which the original data packets are sent or allocated to different users. The line of work of [5][6] is somewhat related, but conceptually different, in that it aims to quantify the protocol information necessary to ensure given delay requirements.

B. Contributions

As discussed above, the scheme proposed in [3] is based on the idea that transmitters (source and relays) in the multihop network of Fig. 1 can encode the source message for forwarding to the following node via protocol coding, i.e., by deciding when to transmit or receive on the basis of the source message. The scheme works since any transmitter knows what will be sent by the downstream nodes, since it has already relayed the current source packet – This allows to construct coherent transmissions schedules, even with data-dependent channel access schedules. As detailed above, this protocol coding scheme outperforms simple time-sharing.

In this paper, we study the two-way relay scenario of Fig. 2, in which two sources A, B communicate to one another via a relay C. In this case, the concept of upstream and downstream nodes does not apply and, therefore, the argument above does not hold. As a result, one cannot use protocol coding as proposed in [3]. Our main contribution will be to show that, by modeling also the direct links between the two sources, a simple novel class of protocol coding strategies can be devised that outperforms time-sharing for two-way communications.

At this point it is important to make a clear distinction between protocol coding and physical-layer network coding in two-way relay channels (see, e.g., [7]). Assume that all three links on Fig. 2 are noiseless binary channels. The maximal two-way sum-rate at which A and B can communicate directly, without involvement of C is 1 bit/channel use (c.u.). This rate cannot be improved by exploiting the relay C via standard network coding[2]. We show that the strategy of protocol coding provide a higher two-way sum-rate.

II. THE CASE WITH BINARY ALPHABET

A. System Model and Reference Results

We consider the two-way relaying scenario in Fig. 2. Due to the half–duplex constraint, nodes cannot transmit and receive at the same time. As in [3], noiseless channels are assumed. We start by treating the binary-alphabet case, in which nodes can send either a "1" or a "0", where the latter is termed *silent symbol* and corresponds to the reception mode. Specifically, assuming that the transmissions of A and B are synchronized at the symbol level at the receiver of C, the received signal at C during a given time t is:

$$Y_C(t) = \begin{cases} X_A(t) + X_B(t) & \text{if } X_C(t) = 0 \\ X_C(t) & \text{otherwise} \end{cases}, \quad (1)$$

where $X_i(t)$ is the symbol sent by node i at time t. From the equation above, the relay is able to receive only when not transmitting (i.e., when $X_C(t) = 0$). This model may correspond, e. g., to power amplitude modulation where node C is able to distinguish collision ($Y_C(t) = 2$) from a single transmission $Y_C(t) = 1$. Notice that in the case of $Y_C(t) = 1$, C cannot determine whether A or B transmitted a 1. The signal received at A (and similar for $Y_B(t)$) is:

$$Y_A(t) = \begin{cases} X_B(t) + X_C(t) & \text{if } X_A(t) = 0 \\ X_A(t) & \text{otherwise} \end{cases} \quad (2)$$

When the traffic between A and B is symmetric, the standard approach of *time-sharing* would prescribe that nodes A and B agree on receiving half the time each, thus achieving a maximum sum-rate of 1 bit/ c.u., or 0.5 bits/ c.u. each. Notice that this rate can be achieved also in the absence of relay C. In fact, if one forces the reception times of each node to be fixed independently of the messages and known a priori, as in time-sharing, it can easily be seen by using cut-set arguments [8] that the relay cannot improve the achievable rates. We will show in the following that a better sum-rate is possible via protocol coding by exploiting the relay.

B. The Proposed Protocol Coding Scheme

With protocol coding, nodes A and B do not agree in advance on the transmission/ reception schedule. Therefore, since the time instants in which a node (A or B) is in receiving mode (i.e., sending 0) are not pre-determined, inevitably, not all transmitted symbols will be received by the recipient (B or A). Our proposed scheme exploits the relay to resolve this ambiguity. The communication between A and B is organized into *sessions*. In each sessions, node A (and B) conveys an independent message $W_A \in [1, 2, .., M_A]$ (and $W_B \in [1, 2, .., M_A]$) to node B (and A). Messages W_A and W_B are independent and uniformly distributed in their domain sets. A session consists of a multiple access (MA) phase followed by a broadcast (BC) phase. The MA phase consists of n symbol times (also referred to as slots or channel uses), during which C is in a receiving mode (i.e., "transmitting" 0), while the following broadcast (BC) consists of k slots, in which A and B are in receiving mode.

The MA phase duration is fixed to a value n for all sessions. Instead, the BC phase has a variable duration k, where k depends on the data sent by A and B, with $0 \leq k \leq n$. Protocol coding thus amounts here to selecting data-dependent transmission/ reception times at the nodes A and B and the data-dependent BC phase duration k. The critical issues in construct the transmission scheme are: (*i*) Nodes A and B should recover all the symbols that were not received during the MA phase (while transmitting 1) thanks to the BC phase; (*ii*) Nodes should operate according to coherent schedules,

[1]However, Massey used a model that *explicitly prohibits* this kind of protocols, stating that "The timing of a packet transmission by a station cannot depend on the values of the data packets to be communicated"

[2]Conventional network coding would be useful in this scenario in the absence of a direct link between the two sources.

A sends	B sends	C receives	C sends	k
00	00	00	-	0
00	01	01	0	1
00	10	10	0	1
00	11	11	00	2
01	00	01	0	1
01	01	02	1	1
01	10	11	00	2
01	11	12	01	2
10	00	10	0	1
10	01	11	00	2
10	10	20	1	1
10	11	21	10	2
11	00	11	00	2
11	01	12	01	2
11	10	21	10	2
11	11	22	11	2

TABLE I
THE PROPOSED PROTOCOL CODING SCHEME AND $n = 2$.

even without having agreed in advance to the value of k (this issue is similar to [3]).

We use the standard definition of achievable rates in terms of reliably transmitted bits per channel use. By considering the average over many sessions and using arguments from renewal theory [12], we obtain the achievable rates as

$$R_A = \frac{\log_2 M_A}{n + E[k]} \text{ and } R_B = \frac{\log_2 M_B}{n + E[k]}, \tag{3}$$

where $E[k]$ represents the average duration of the BC phase with respect to the message distribution.

Example ($n = 2$): To illustrate the main ideas, let us assume a MA phase of $n = 2$ symbols and each node A, B sends two bits. Each node sends an independent bit in each MA slot, such that $M_A = M_B = n$. Table I shows a possible design of a session according to our scheme. Consider the column which shows the two symbols that C receives during the MA phase. The key observation is that if C receives the silent symbol $Y_C(t) = 0$ in a given MA slot t, then both A, B are in the receiving state and hence A knows that B sent the silent symbol (and viceversa). In short, when $Y_C(t) = 0$ during the MA phase, all three nodes know that both A and B have sent the silent symbol. This implies that during the BC phase C only needs to send information about the "non-silent" symbols ($Y_C(t) > 0$) observed during the MA phase. The number s of such non-silent symbols can be inferred also by A and B based on their received signal, as discussed below. The column "C sends" in Table I refers to what is sent in the BC phase and k is its duration, which is equal to s. The encoding method applied by C and decoding applied at A, B is described in Proposition 1 for general n. Here, we evaluate the rates $R_A = R_B$ achieved by this scheme. To this end, from (3), we need to calculate the average BC duration $E[k]$ (over the message distribution), which from Table I is given by $E[k] = 6\frac{1}{16} + 9\frac{2}{16} = \frac{3}{2} = 1.5$, the achievable rates are $R_A = R_B = 2/(2 + 1.5) = 0.57$ bits/c.u. and the sum-rate is $R_A + R_B = 1.143$ bits/c.u., which is larger than the 1 bit/c.u. achievable by time-sharing.□

We will now generalize the principles used to construct the example above to a MA phase with any n symbols. Let us assume that C has observed s non–zero outputs in the MA phase. Then in the BC phase it needs to send at least s bits, since the observed output could imply that there was one node that sent s ones in the MA phase, and thus missed to receive the s transmissions from the other node. Another key observation is that s is perfectly known by the nodes A and B, since s is equal to the number of ones received by both A and B in the MA phase.

Let us number the slots in the MA phase by $t \in \mathcal{T} = \{1, 2, \dots n\}$. Let $\mathcal{T}_s = \{t_1, t_2, \dots t_s\} \subseteq \mathcal{T}$, where t_l is the $l-$th slot with non-zero $Y_C(t_i) > 0$ in the MA phase. We define $\mathcal{T}_0 = \emptyset$ and set the BC phase duration as $k = s$. If $k > 0$, then the slots of the BC phase are numbered $n + 1, n + 2, \dots, n + k$.

Proposition 1: Let C observe s non-zero symbols during the MA phase and apply the following encoding:

- If $s = 0$, then there is no BC phase. i.e., $k = 0$;
- If $s > 0$, then set $k = s$ and for each observed $Y_C(t_l) > 0$, relay C sends:

$$X_C(n + l) = Y_C(t_l) - 1, \text{ for } l = 1, ..., k. \tag{4}$$

At the end of the session A (and B) can uniquely decode the information sent by B (and A), achieving rates:

$$R_A = R_B = \frac{H_2(p)}{2 - p^2} \text{ [bits/c.u.]} \tag{5}$$

for any $0 \le p \le 1$ and $n \to \infty$. Rates (5) are maximized for $p = 0.63$ and we have $R_A = R_B = 0.593$ bits/c.u..

Remark 1: Parameter p in (5) represents the percentage of 0's transmitted by a node in the MA phase. It is noted that the optimal value of $p = 0.63$ favors transmission of 0. This is because this choice decreases the duration of the BC phase and thus increases the rates. Moreover, the maximal sum-rate equals $R_A + R_B = 1.1861$ bits/c.u., improving by over 18% the maximum sum-rate achievable by standard time-sharing thanks to protocol coding.

Proof: We first show that, given the encoding scheme of Proposition 1, A can reconstruct $X_B(t)$ for all $t \in \mathcal{T} = \{1, 2, \dots n\}$. Similar considerations hold for B. Let $\hat{X}_B(t)$ denote the decoded signal at A for the symbol $X_B(t)$ sent by B. We note that A knows the set $\mathcal{T}_s$ (and thus $s = k$) defined above. In fact, for all $t \in \mathcal{T}_s$ we have $Y_A(t) = 1$. Now, if $k = 0$, then A knows that there is no BC phase and can uniquely decode that $X_B(t) = 0$ for all $t \in \mathcal{T}$. Instead, if $k > 0$ then A sets the following:

$$\hat{X}_B(t) = \begin{cases} Y_A(t) & \text{if } X_A(t) = 0 \\ Y_A(n + l) & \text{if } X_A(t) = 1, t = t_l \in \mathcal{T}_s \end{cases} \tag{6}$$

It can be readily seen that $\hat{X}_B(t) = X_B(t)$. In fact, when $X_A(t) = 0$ in the MA phase it follows from (2) that $Y_A(t) = X_B(t)$ since $X_C(t) = 0$; Furthermore, if $t \in \mathcal{T}_s$ and $X_A(t) = 1$, we have $Y_C(t_l) = 1 + X_B(t_l)$, so that:

$$\hat{X}_B(t_l) = Y_A(n + l) = X_C(n + l) = Y_C(t_l) - 1 = X_B(t_l)$$

Two remarks are in order. First, with the decoding rule (6) the node implicitly removes the self–interference, i. e. the

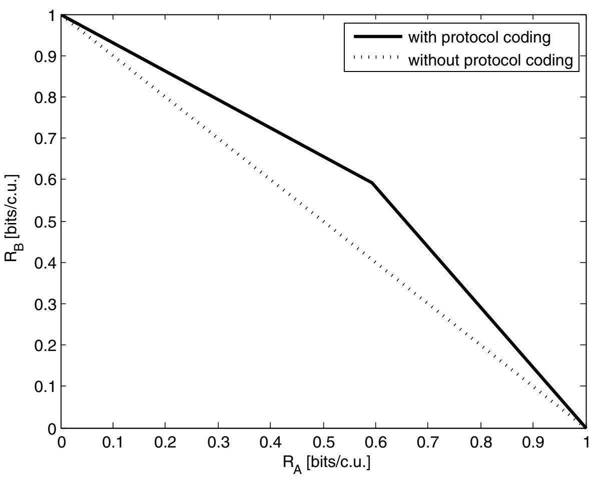

Fig. 3. Rate region achievable by using the described scheme for protocol coding. The reference line without protocol coding is obtained by time-sharing over the direct link $A - B$.

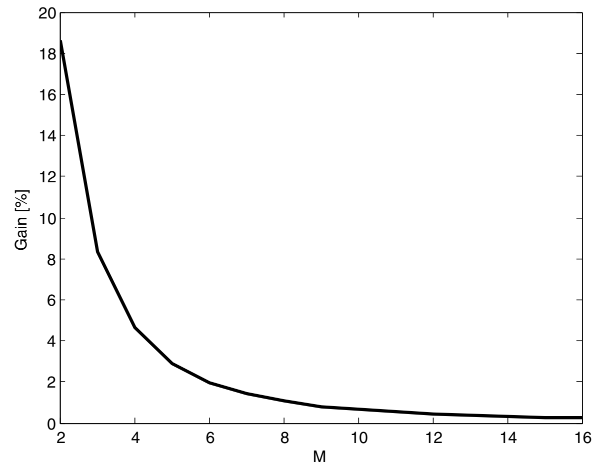

Fig. 4. Sum-rate gain of the proposed protocol coding scheme with respect to standard time-sharing versus the alphabet size M.

interference from its own signal sent during the MA phase. Second, if $t_k \in \mathcal{T}_s$ and $X_A(t_k) = 0$, then A can decode the symbol both directly as $Y_A(t_k)$, but also using $Y_A(n+k)$.

We now analyze achievable rates using the random coding argument. Assume that nodes A and B select $M_A = M_B = 2^{nH(p)}$ MA-phase codewords $X_A(t)$ and $X_B(t)$, $t = 1, ..., n$, by generating each variable randomly and independently with binary distribution $P(X_A = 0) = P(X_B = 0) = p$, where $H(p) = -p\log_2(p) - (1-p)\log_2(1-p)$ is the binary entropy function [13]. Notice that this approach allows transmissions with different fractions of 0 and 1's by the two nodes. We now need to calculate the average duration $E[k] = E[s]$ of the BC phase. Averaging over the randomly generated codebooks and messages, using standard argument, we get for n large enough and arbitrary $\epsilon > 0$:

$$E[k] \leq \epsilon n + n(1-p^2)(1+\epsilon), \tag{7}$$

where the first term account for non-typical sequences $Y_C(t)$, $t = 1, 2, ..., n$, and it is assumed that the length of the BC phase is maximal $k = n$. The second term follows from the definition of (strongly) typical sequences [13]. The proof is concluded by taking $\epsilon \to 0$. ■

Remark 2 (Achievable Rate Region): Proposition 1 provides symmetric rates $R_A = R_B$ achievable via protocol coding. Using standard time-sharing strategies, one can instead achieve rate points $(R_A, R_B) = \{(0,1), (1,0)\}$ and all rate points connecting these two. Combining time-sharing with protocol coding, one can then achieve the rate region shown in Fig. 3. Notice that this combination is obtained by using a predefined schedule for the sessions, where a only fraction of sessions uses protocol coding as described above, while the other sessions operate at the rate points $(R_A, R_B) = \{(0,1), (1,0)\}$.

III. The M-ary Alphabet Case

We have so far considered the binary-alphabet case. We now generalize to the case in which a transmitting node can send M different symbols, including the silent symbol 0. By setting the input/output alphabets to $X_j \in \{0, 1, \ldots M-1\}$ and $Y_j \in \{0, 1, \ldots 2M-2\}$, we use the same channel models as in (1) and (2). The following Proposition generalizes Proposition 1 from $M = 2$ to any M.

Proposition 2: Let C observe s non-zero symbols during the MA phase and apply the following encoding:

- If $s = 0$, then there is no BC phase, i.e., $k = 0$;
- If $s > 0$, then set $k = s$ and set for each observed $Y_C(t_l) > 0$:

$$X_C(n+l) = Y_C(t_l) \mod M. \tag{8}$$

Then, at the end of the session A (and B) can uniquely decode the information sent by B (and A). Moreover, this scheme achieves the following rates, for any $0 \leq p \leq 1$.:

$$R_A = R_B = \frac{H(p) + (1-p)\log_2(M-1)}{2-p^2} \text{ [bits/c.u.]} \tag{9}$$

Remark 3: Parameter p, as in Proposition 1, accounts for the fraction of silent symbols 0 transmitted in the MA phase by two nodes. It can be seen that the optimal p is always larger than $1/M$: The silent symbol is favored. However, the gain in sum-rate rate compared to the ordinary time-sharing on the direct link $A - B$, for which the sum-rate is $\log_2(M)$ bits/c.u., decreases as M increases, as shown on Fig. 4. This is to be expected, since the "saving" that the silent slot produces during the BC phase becomes less significant as M increases. Similar conclusions are also reached in [2][3].

Proof: The proof of the first part is similar to Proposition 1, the only difference is that when node A receives $X_C(n+l)$ that corresponds to $Y_C(t_l) > 0$, it finds $X_B(t_l)$ uniquely as:

$$\begin{aligned} \hat{X}_B(t_l) &= [M + Y_C(t_l) - X_A(t_l)] \mod M = \\ &= [M + (X_A(t_l) + X_B(t_l))\text{mod}M - X_A(t_l)] \mod M \\ &\overset{(a)}{=} [M + X_A(t_l) + X_B(t_l) - X_A(t_l)] \mod M \\ &= [M + X_B(t_l)] \mod M = X_B(t_l) \end{aligned} \tag{10}$$

where (a) is due to $(a \mod M + b \mod M) \mod M = (a + b) \mod M$. As for achievable rates, here nodes select $M_A = M_B = 2^{nH(\mathbf{p})}$ codewords $X_A(t)$ and $X_B(t)$, $t = 1, ..., n$, with independent entries with distribution $p_i = P(X_A = i)$, where $H(\mathbf{p}) = -\sum_{i=0}^{M-1} p_i \log_2(p_i)$ [13]. It can be easily seen that, for given $\mathbf{p} = [p_0, .., p_{M-1}]$, the amount of data is maximized for $p_i = \frac{1-p}{M-1}$, $i = 1, ..., M-1$. The average duration $E[k]$ of the BC phase is calculated as for Proposition 1. ■

IV. Discussion

The protocols introduced so far are designed under assumption of errorless links and they are not only "gracefully degraded" when the probability of erroneous reception increases from zero. In fact, a single error may potentially affect the following transmissions. To see this, let us assume again binary alphabet $M = 2$, $n = 2$ and $p = 0.5$ for both nodes and the encoding as described in Table I. The proposed scheme strongly relies on the assumption that all three nodes A, B, C perceive the same number $k = s$ of "non-silent" inputs during the MA phase, which determines the duration of a session to be $n + k$. Now, assume that there is a single error in the reception of C where 0 is interpreted as 1. Then, according to C, the session has a duration of $n + k + 1$ slots, while it has duration $n + k$ for A and B. This means that the start of the next session does not occur in the same slot for all three nodes A, B, C, which destroys any subsequent communication.

A possible solution to this problem is to make the duration of the BC phase independent of the data, thus using protocol coding only in the selection of transmit/receive times at A and B in the MA phase. Using the proof of Proposition 1 and the random coding argument, it is possible to find MA-phase codes at A and B such that the number of non-silent symbols s seen at C is almost certainly $s \leq n(1-p^2)(1+\epsilon)$, for arbitrary small $\epsilon > 0$ and with probability approaching one as $n \to \infty$. For finite n, if one fixes a BC duration $k_\epsilon = n(1-p^2)(1+\epsilon) < n$ for some $\epsilon > 0$, then, depending on the data sent in the MA phase, the BC phase duration may not be enough to carry all the non-zero symbols observed by C during the BC phase, giving rise to an outage. Such outage events cannot simply be resolved by resending the packets in the next session, as outage will deterministically occur again.

A way to address this problem is to define a periodic session structure, such that after K_ϵ "short" sessions of duration $n+k_\epsilon$, there is one "long" session with duration $2n$. Then each node operates as follows: If it detects packet error, buffers the packet and retransmits it in the long session. Parameter K_ϵ depends on ϵ and is chosen such that the system should be stable (finite queueing time for the buffered packets).

A final remark is in order. There is an easy strategy to further optimize the described strategies for protocol coding. In the binary alphabet case, note that during the MA phase, C observes a sequence of symbols $0, 1, 2$. Now C may try to compress maximally this sequence, reckoning that both A and B have side information about that sequence. Due to the lack of space, details will be presented in a future work.

V. Conclusion

We have proposed a new approach in the class of *protocol coding* strategies that is tailored to two-way communications with half-duplex nodes. The main idea is that, rather than fixing a priori the duration of transmit/receive phases, these can be made dependent on the data. Although related, the proposed approach is significantly different from the "one-way" strategies discussed in [2][3] and exploits the signals overheard by each node when not transmitting. We have shown that such strategies outperform standard time sharing.

A final remark regarding the *optimal* transmission strategies for the scenario at hand. Unlike the model of [3], here capacity results are not available, and upper bounds, based on cut-set arguments, are not immediately achievable. Moreover, one could consider alternative achievable schemes based on specializing general strategies such at the noisy network coding scheme of [9]. We do not claim that the proposed schemes outperform such strategies. The goal of this paper is to show that time-sharing can be improved by simple protocol coding schemes. The issues mentioned above will be part of future work, along with the analysis of noisy communication scenarios.

Acknowledgement

P. Popovski was partially supported by the EliteForsk program of the Danish Research Council. O. Simeone was partially supported by the U.S. National Science Foundation under Grant CCF-0914899.

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