Delay Constrained Throughput-Reliability Tradeoff in Network-Coded Wireless Systems

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Abstract—We investigate the performance of delay constrained data transmission over wireless networks without end-to-end feedback. Forward error-correction coding (FEC) is performed at the bit level to combat channel distortions and random linear network coding (RLNC) is performed at the packet level to recover from packet erasures. We focus on the scenario where RLNC re-encoding is performed at intermediate nodes and we assume that any packet that contains bit errors after FEC decoding can be detected and erased. To facilitate explicit characterization of data transmission over network-coded wireless systems, we propose a generic two-layer abstraction of a network that models both bit/symbol-level operations at the lower layer (termed PHY-layer) over several heterogeneous links and packet-level operations at the upper layer (termed NET-layer). Based on this model, we propose a network reduction method to characterize the throughput-reliability function of the end-to-end transmission. Our approach not only reveals an explicit tradeoff between data delivery rate and reliability, but also provides an intuitive visualization of the bottlenecks within the underlying network. We illustrate our approach via a point-to-point link and a relay network and highlight the advantages of this method over capacity-based approaches.

Index Terms—Wireless networks, random linear network coding, delay, throughput, reliability, cross-layer optimization

I. INTRODUCTION

In wireless communication systems, much effort has been devoted to advanced forward error-correction coding (FEC) and signal processing techniques on the physical layer to pursue reliable transmission over each single hop. Even if data transmission over each individual link is reliable, the end-to-end transmission can still fail as a result of packet losses caused by higher-layer effects, such as congestion or buffer overflow. These missing packets are usually detected and recovered by some automatic repeat-request (ARQ) based mechanisms on data link layer and/or on transport layer. Many wireless networks are inherently heterogeneous in the sense that the channel quality, system implementation, and available resources can be vastly different from link to link. These parameters may interact with and be dependent upon one another, which can make the performance analysis a difficult task over networks of nontrivial size.

In this paper, we focus on delay constrained data transmission over wireless networks where end-to-end feedback is absent or performed at a higher layer, motivated by scenarios where the end-to-end feedback is unavailable, excessively delayed, or onerous. While, from solely a throughput perspective, coding at the physical layer should suffice if the only losses are due to deleterious effects at that layer, operational wireless communication systems differ from this paradigm in operation. Firstly, they are comprised of at least two layers, which we coarsely identify as PHY and NET; the latter operates on the packet level. Secondly, some losses are not due to physical layer effects (such as buffer overflow). Finally, throughput may not be the only parameter we seek to improve; in particular, we consider the tradeoff between throughput and reliability, under a delay constraint.

To model this behavior, we propose a generic two-layer abstraction of the wireless network incorporating the effects of bit/symbol-level operations at the PHY over multiple heterogeneous links and packet-level operations at the NET. Our model assumes that packets are either provided to the NET layer intact or erased, a simplification based on the operations practiced by state-of-the-art systems. PHY level techniques (such as adaptive coding and modulation), while imperative in achieving this effect, are not explicitly considered here. We focus on the scenario where intermediate nodes can perform random linear network coding (RLNC) [1] re-encoding based on the received packets. The interaction of parameters from the different layers is characterized by a clean and general interface; we assume that we always know the PHY-layer operations. Based on this model, we propose a network reduction method that works on the NET-layer of the wireless system to characterize the throughput-reliability function of the end-to-end transmission under the delay constraint. Our approach not only reveals an explicit tradeoff between data delivery rate and reliability, but also provides an intuitive visualization of the bottlenecks within the underlying network. We illustrate our approach via a point-to-point link and a relay network and highlight its advantages over capacity-based approaches.

The rest of this paper is organized as follows. We first give a brief overview of related work in Sec. II and then present the two-layer model in Sec. III. We analyze the probability of errors on both layers in Sec. IV and describe the network reduction method in Sec. V. In Sec. VI we establish the throughput-reliability function and illustrate via examples how it can be used for system design. Sec. VII is the conclusion.

List of Notation

- $\mathbb{F}_q^l$: set of all length-$l$ vectors over the finite field $\mathbb{F}_q$
characterized by the delay, rate, and the network capacity. The
with only point-to-point arcs (for wireline) and hyper-arcs
lossy unicast/multicast network into a lossless packet network
ensure non-zero throughput. Under such conditions, the mean
$k$
channels. They conclude that
$n$
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packet coding rather than decoupled codes and the regime
throughput-delay tradeoff, clarifies the embedding of error
concatenated code framework with a total delay constraint, where
error exponents for random coding are used to analyze both inner and outer codes. This framework investigates a
throughput-delay tradeoff, clarifies the embedding of error
functions in a multi-level approach, and models the fact that
data may be lost at either layer. Barg et al. in [3] also consider
error exponents, although the focus is on joint channel and
packet coding rather than decoupled codes and the regime
is asymptotic in block lengths at both layers. Berger et al.
in [4] focus on a flat block-fading channel, where perfect
channel decoding is assumed for all rates below capacity
and perfect packet-erasure coding is assumed when a fixed
overhead requirement is met. The tradeoff between spectral
efficiency and end-to-end probability of error is explored; one
particularly useful insight is the investigation of this trade-
off with few choices of physical layer transmission modes.
Courtade and Wesel in [5] also assume perfect channel and
erasure coding as in [4], but consider both fast and slow flat
block-fading with a delay constraint. They employ a Gaussian
approximation of end-to-end outage probability and optimize
for power consumption under end-to-end rate and reliability
constraints. In [6], Koller et al. consider RLNC operating on
top of a random code over a binary symmetric channel (BSC)
for a one-hop broadcast network. The expected number of
transmissions required to decode is minimized, and it is noted
that this optimization problem is not equivalent to optimizing
the expected number of accurate bits per transmission.

While [2]–[6] all elucidate different angles of the cross-
layer problem, they all consider only single-link networks
where generalization to larger topologies is nontrivial. We also
recognize some works that consider more complicated network
structures. Swappa et al. in [7] work on RLNC over $k$
packets broadcasting to $n$ users over independent time-correlated
erasure channels. They conclude that $k$ must scale at $O(\log(n))$ to
ensure non-zero throughput. Under such conditions, the mean
and variance of total transmission time depend on channel
correlation. Lun et al. in [8] propose a framework to translate a
lossy unicast/multicast network into a lossless packet network
with only point-to-point arcs (for wireline) and hyper-arcs
(for wireless). Assuming that the number of packets is large
and that the arrivals of packets at each node are independent
Poisson processes, the probability of RLNC decoding error is
characterized by the delay, rate, and the network capacity. The
average throughput of each individual link is the figure of merit
characterizing system performance [8]; additionally, delay is
based entirely on the depth of the network independently of
propagation time. Lower layers are not considered in [8]. Ming
et al. in [9] investigate the delay in packet erasure networks
where RLNC is used in a rateless fashion, and the delay is
optimized based on the tradeoff between codeword lengths on
physical layer and on network layer.

II. RELATED WORK

Numerous research efforts have been devoted to charac-
terizing the interaction between channel coding and packet-
erasure coding from different aspects; we only list a sampling
concatenated code framework with a total delay constraint,
where error exponents for random coding are used to analyze
both inner and outer codes. This framework investigates a
throughput-delay tradeoff, clarifies the embedding of error
functions in a multi-level approach, and models the fact that
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average throughput of each individual link is the figure of merit

III. SYSTEM MODEL: TWO-LAYER ABSTRACTION

A source node intends to transmit some messages to a
remote destination via a wireless network. The transmission
has to be completed within a certain delay constraint with
a rate of success no less than a predefined threshold. The
underlying wireless network connecting the source and the
destination suffers from both deleterious effects on the PHY-
layer and packet losses at the NET. The two layer abstraction
of the source node is shown in Fig. 1.

A. Abstraction of the NET-Layer

On the NET layer, a new generation of source messages is
first evenly split into $\kappa$ segments $V_1, V_2, \ldots, V_\kappa$, each of $n\eta$
bits (appending zeros when necessary) where $0<\eta<1$. These
message segments are fed to the RLNC encoder to generate
$n$ RLNC-coded messages of the same size, which are then
passed through the packetization process where a header of
$(1-\eta)n$ bits is added to each of the RLNC-coded messages
to formulate packets of size $l$ bits. The above process can be
summarized as follows

$$\begin{align*}
(\mathbb{F}_2^n)^\kappa \xrightarrow{\text{RLNC}} (\mathbb{F}_2^l)^n \xrightarrow{\text{packetization}} (\mathbb{F}_2^l)^n.
\end{align*}$$

(1)

Here we assume that the packet length $l$ and format (overhead)
are long-term, system-wide parameters that cannot be adjusted
based on a single transmission task. These RLNC-coded
packets are then passed to the PHY layer for transmission.
After a period of time $\tau$, if the destination receives at least $\kappa$
packets and the decoding is successful, we say the transmission
task is accomplished. Otherwise the task fails. The benefits of
choosing RLNC as the erasure code are two fold: its decoding
error probability is easy to analyze, and its re-encoding process
at intermediate nodes is straightforward and facilitates the
network reduction approach we propose in this paper.

Our NET-layer model is generic in the sense that we do
not refer to any specific network protocols. Rather, we use the
abstraction of a “packet” as the atomic unit of information that
is exchanged between the two layers. Note that in canonical
Internet layering models, a packet has different meaning in dif-
f erent layers: a packet in an upper layer becomes the payload
of another packet in the adjacent lower layer. In our model, a
packet $[X_k, O_k]$ differs from the canonical concept, since its
payload $X_k$ refers to the RLNC-coded message (cf. service
data unit on transport layer), but its overhead $O_k$ refers to all
bits other than the payload: all the control/protocol information
(cf. overhead brought by data link layer, network layer, and
transport layer) plus the RLNC encoding coefficients. The
proportion of the payload ($\eta$) within each packet is a system
parameter that reflects the actual network protocol in use.
The effective data rates are and their channel coding rate are a channel via hierarchical modulation where schemes in the system. For example, when two users share facilitates flexible MAC, channel coding, and modulation for media access (abbr. MAC).

B. Abstraction of the PHY-layer

On the PHY-layer, packets from the NET-layer are grouped into frames of \( m r \) bits, and each frame is then mapped onto an \( m \)-symbol sequence (a.k.a., codeword) over \( \mathbb{F}_2^m \) by operations such as the channel coding and modulation for media access (abbr. MAC). The operation at the PHY can be described by the following mapping

\[
(F_2^l)^n \rightarrow (\mathbb{F}_2^m)^{\frac{m r}{l}} \rightarrow (\mathbb{F}_2^m)^{\frac{m}{l}}.
\]

Since each codeword carries a message of \( m r \) bits, or equivalently, \( \alpha=mr/l \) packets, we call \( \alpha \) the packet loading parameter as it decouples the PHY and the NET cleanly such that a packet is the basic unit at the NET and a message/codeword is the basic unit at the PHY.

The MAC block at the PHY does not specify or rely on any specific coding and modulation techniques. In effect, any physical layer operations that affect the transition from frames to codewords can be incorporated into the MAC block by adjusting the data rate \( r \) (bits per symbol-time, or equivalently, bits per channel-use). By focusing on \( r \) at the PHY, our model facilitates flexible MAC, channel coding, and modulation schemes in the system. For example, when two users share a channel via hierarchical modulation where \( z \)-QAM is used, and their channel coding rate are \( c_1 \) and \( c_2 \), respectively, the effective data rates are \( r_i = \frac{1}{2}c_i \log_2(z) \), \( i = 1, 2 \). The corresponding MAC operation at the PHY is performed by

\[
(F_2^m, F_2^{m_2}) \rightarrow (F_2^{m_1}, F_2^{m_2}) \rightarrow (F_2^{m_1'}, F_2^{m_2'}) \rightarrow (F_2^{m_2'}). \quad (3)
\]

C. Interface between Two Layers

Based on the two-layer abstraction, we can model and track the data flow over wireless networks as illustrated in Fig. 2 for transmission over a two-hop network. Note that at the relay a new RLNC-coded packet is generated in two steps: its payload is a random linear combination of the payloads of all the received RLNC packets stored in its buffer, and the coding vector in the header is likewise the corresponding linear combination of coding vectors in the buffer.

At the PHY, there are many dependent operating parameters even on a single link: channel \( h_i \), codeword length \( m_i \), data rate \( r_i \), channel coding rate \( c_i \), modulation order \( z_i \), packet loading factor \( \alpha_i \), and codeword error probability \( p_{e,i} \). Regardless of the PHY-layer operations, we only focus on two parameters that directly affect the NET: \( n_i \) (number of packets transmitted within the delay constraint \( \tau \)) and \( p_{e,i} \) (packet erasure probability, which may be different from the codeword error probability \( p_{e,i} \) as discussed in Sec. IV).

At the NET, the number of RLNC-coded packets that can be transmitted over a channel is limited by the PHY-layer settings. A packet can be either erased at the PHY with probability \( p_{e,i} \), or dropped by the NET-layer effects with probability \( p_{d,i} \). We denote the overall packet erasure probability as a function

\[
\xi_i = P_e(p_{e,i}, p_{d,i}, l).
\]

Now for each link, at the NET we only focus on \( n_i \) and \( \xi_i \) (the overall packet erasure probability).

IV. ERROR ANALYSIS

Given a generation of \( \kappa \) messages \( W = \{V_1, V_2, \ldots, V_\kappa\} \) each of \( n_l \) bits, by representing them as vectors over \( \mathbb{F}_q \), a RLNC-coded packet \( X_k \) is constructed by

\[
X_k = a_{k,1}V_1 + a_{k,2}V_2 + \cdots + a_{k,\kappa}V_\kappa,
\]

where the encoding coefficients \( \{a_{k,1}, a_{k,2}, \ldots, a_{k,\kappa}\} \) are drawn uniformly at random from \( \mathbb{F}_q \). The re-encoding process at intermediate nodes is the same as in (5) except that the linear combination is over the payloads of received packets. By the end of the delay constraint \( \tau \), some of the coded packets will have been received together with their encoding vectors.

1Note that our framing operation differs from canonical packet aggregation/segmentation in the sense that it incurs no change on the structure of a packet, which ensures independent packet erasures at a cost of overhead.

2As pointed out in [10], the buffer can be as small as one packet in size.
If these encoding vectors formulate a matrix of rank $\kappa$, we are able to recover $W$ through Gaussian elimination. If the rank is less than $\kappa$, the transmission fails. Through any link, a packet may be erased with probability $\xi$, which depends both on local parameters $p_e$ and $p_l$ as well as the packet length $l$ (a system parameter).

At the PHY, a codeword consisting of $m$ symbols is transmitted across the channel. The corresponding decoding error probability $\hat{p}_e$ depends on many parameters as explained in Sec. III-C. We may approximate the error probability by error exponent bounds as in [2], [3], by large deviation methods, Sec. III-C. We may approximate the error probability by error probability $\alpha < \hat{p}_e$ at the PHY equals the codeword error probability $p_e$. If one codeword contains several packets, each packet can still experience independent losses owing to their independent cyclic redundancy check (CRC) embedded in the packet headers. For the case $\alpha < 1$ or non-integer $\alpha$, we have $p_e > \hat{p}_e$ and their relationship depends on the actual framing process. It is therefore favorable to have integer-valued $\alpha$ in system design.

Let $S$ be the number of packets successfully received by the end of transmission, and let $D$ be the number of degrees of freedom (DoF) available at the decoder (i.e., the rank of the matrix composed by encoding vectors). Given i.i.d. packet erasure with probability $\xi \in [0, 1]$, the probability of success of the RLNC coded transmission can be written as (see [8], [11])

$$Pr\{D = \kappa\} = \sum_{s=\kappa}^{n} \binom{n}{s} (1-\xi)^s \xi^{n-s} \prod_{u=0}^{\kappa-1} (1-q^{u-s}).$$  \hspace{1cm} (6)

This formulation is exact, but it is also difficult to precisely compute when $\hat{p}_e$ is well below the channel capacity $(1-\xi)$. With the help of an inequality given by Liva et al. in [11],

$$1 - \frac{1}{q-1} q^{s-s} < \prod_{u=0}^{\kappa-1} (1-q^{u-s}) \leq 1 - q^{s-s-1},$$  \hspace{1cm} (7)

can get a lower bound for probability of error as follows

$$P_e(\kappa) = 1 - Pr\{D = \kappa\} \geq 1 - \sum_{s=\kappa}^{n} \binom{n}{s} (1-\xi)^s \xi^{n-s} (1-q^{s-s-1}).$$  \hspace{1cm} (8)

This lower bound is very tight when $q$ is large.

V. NETWORK REDUCTION

In the following discussion we assume that our RLNC re-encoding operations are non-degenerate (i.e., $q$ is sufficiently large); if the coding vectors of the packets received at any node span a subspace $\mathcal{V}$ of $\mathbb{F}_q^n$, where $\kappa$ is number of information messages within a generation over which RLNC is performed, then any $\kappa$ of the new vectors of the re-encoded packets also span $\mathcal{V}$. In such cases, the probability of success defined in (6) is essentially the complementary cumulative distribution function (CCDF) evaluated at $S = \kappa$.

$A$. Tandem Link Equivalence

Let our network consist of two directed links connecting three nodes, as in the right part of Fig. 3. We will call these links 1 and 2. Link $i \in \{1, 2\}$ permits us to send $n_i$ packets per delay period $\tau$. Now define a random variable $S_i$ describing the number of successfully delivered packets over the link $i$ within the generation. Assuming that we have an explicitly defined channel model, we can get a probability mass function (PMF) for $S_i$, which will be described by the vector $\lambda_i \in [0, 1]^{n_i+1}$, where

$$\bar{\lambda}_i[s] = Pr\{S_i = s\}, \quad s \in \{0, 1, \ldots, n_i\},$$  \hspace{1cm} (9)

or equivalently by the CCDF of $S_i$, $\bar{\Lambda}_i \in [0, 1]^{n_i+1}$;

$$\bar{\Lambda}_i[s] = Pr\{S_i \geq s\} = \sum_{j=s}^{n_i} \bar{\lambda}_i[j], \quad s \in \{0, 1, \ldots, n_i\}.$$  \hspace{1cm} (10)

Because our assumption permits us to ignore the possibility of a degenerate decoding matrix, the number of DoF $D$ at the tail of the second link is equal to the minimum number of packets received across either link. For the distribution of $D$, we note that the following property holds for $\kappa \leq \min\{n_1, n_2\}$,

$$Pr\{D \geq \kappa\} = Pr\{\{S_1 \geq \kappa\} \cap \{S_2 \geq \kappa\}\} = Pr\{S_1 \geq \kappa\}Pr\{S_2 \geq \kappa\},$$  \hspace{1cm} (11)

where the second equality follows from the independence of the two links. We have now found the CCDF of a single link equivalent to that of the two-link tandem network. Then, if we zero-pad the shorter of the two single-link CCDFs and denote this operation with a $\prime$, we have

$$\bar{\Lambda}_{EQ}^c[d] = \bar{\Lambda}_1^c[d] \bar{\Lambda}_2^c[d],$$  \hspace{1cm} (12)

or equivalently,

$$\bar{\Lambda}_{EQ}^c = \bar{\Lambda}_1^c \circ \bar{\Lambda}_2^c,$$  \hspace{1cm} (13)

where $\circ$ denotes element-wise multiplication of two vectors.

$B$. Parallel Link Equivalence

Now we consider the scenario of two links in parallel, as shown in the left part of Fig. 3. This problem can be regarded as a kind of a dual to the tandem link problem. Let $n_i$ and $S_i$ retain their definitions from earlier. If we expand our assumption so that packets received from either link are linearly independent, the DoF $D$ at node B becomes

$$D = S_1 + S_2.$$  \hspace{1cm} (14)

Because $S_1$ and $S_2$ are independent, we can get a distribution for $D$ by convolving the distributions of $S_1$ and $S_2$:

$$\bar{\lambda}_{EQ} = \bar{\lambda}_1 \ast \bar{\lambda}_2, \quad \bar{\lambda}_{EQ} \in [0, 1]^{n_1+n_2+1},$$  \hspace{1cm} (15)

Notice that we can quite easily switch between $\bar{\lambda}$ and $\bar{\lambda}_{EQ}$ using a 1-to-1 linear operator.

![Fig. 3. Basic two-link parallel network and two-link tandem network.](image-url)
the bottleneck in a network; by condensing large pieces of data into smaller blocks, we can make the system more robust to errors and failures. Also, the end-to-end CCDF makes it easy to identify the maximum generation size for a minimum reliability constraint. The throughput-reliability curve also facilitates the identification of the maximum generation size for a minimum reliability constraint. With the throughput-reliability curve, system designers have the option of selecting objective functions more informative than throughput alone. The throughput-reliability curve can be used to select the best parameters for system design. Since the effective rate is limited by the delay constraint and the RS coding rate, we consider the tradeoff between these parameters.

Fig. 5. Results of optimization for $\kappa(1 - P_c(\kappa))$ on a single link with one modulation scheme and choice of 7 RS codes producing a range of $n$ from 20 to 140. The underlying channel BSC($\epsilon$) varies on the abscissa, while $p_2$ is fixed at 0.1. Optimal RS coding rate is shown in (a), NET-layer coding rate ($\kappa/n$) in (b), and the goodput $\kappa(1 - P_c(\kappa))$ in (c).

A. Illustrations

For the purpose of illustration, we assume all PHY-layer symbols consume the same transmission time $\tau_N$. Given the delay constraint $\tau$, the maximum number of symbols that can be transmitted on the NET layer is bounded by

$$n_s \leq \left(\frac{\kappa}{\tau}\right) / \tau_N,$$

where $\tau_N$ is the round trip time.

In Fig. 5, we show the theoretical results of code optimization over a practically-motivated scenario for one link where the channel at the PHY is a BSC($\epsilon$) where $\epsilon$ is the bit-crossover probability. We envision a Reed-Solomon (RS) code with fixed block length $m=255$ coding symbols (each of one-byte) and only 7 choices for coding rate. Over this, we run RLNC over $\mathbb{F}_{2^8}$ (i.e., one-byte encoding coefficients) where the number of RLNC-coded packets $n$ is limited by the delay constraint and the RS coding rate. While varying bit-crossover probability on the underlying channel and fixing $p_2$, we evaluate $\kappa(1 - P_c(\kappa))$, the “goodput”, for every configuration of $(r, \kappa)$, recording the highest value. The top line shows the optimal RS code rate, the second line shows the optimal RLNC rate, and the bottom line shows the goodput. Interestingly, we see that the network code rate drops down when the RS code begins to fail and erasures due to decoding on the lower level become more prevalent than dropped packets. This creates a smoothing effect on throughput.

A similar setup is used on the relay network in Fig. 6 to generate results for Fig. 7; each link has the same selection of RS codes with slightly different, but fixed, channel parameter and NET packet erasure rate. To simplify simulation we assume that $\xi_i = p_{e, i} + p_{l, i}$. With the chosen BSC channel parameters ($\epsilon_1, \epsilon_2, \epsilon_3$) as shown in Fig. 6, the corresponding RS codeword error probability $p_{e, 1} = 0.1$ at RS rate 7/8, $p_{e, 2} = 0.25$ at rate
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