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Coding for Multi-Hop Wireless Networks

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Abstract:
This deliverable is intended to survey the state of the art and to suggest open problems that give rise to joint research activities among the cooperating partners in work package 5. The area of activities in this work package is to identify and analyze instances where the application of coding theory in wireless multi-hop networks yields significant advantages. Furthermore, the development and implementation of novel coding theoretic algorithms and techniques is the goal in the medium and long term.

In particular, based on the areas of expertise, research intentions and visions of the contributing partners, we suggest grouping the activities in four different clusters.

1. Erasure coding on the network layer (mainly for video transmission)
2. Network coding as an enhancement of geographic routing
3. An information theoretic analysis of network coding
4. Coding for relay networks

We believe, that the suggested areas are both, theoretically challenging and at the same time highly relevant in practice and constitute a good basis for the joint research activities between the WPR.5 partners. A significant part of the WPR.5 effort is intended to go towards the better understanding of the relatively new field of network coding and its impact in a wireless multi-hop scenario. Indications are numerous that it can provide substantial gains and we want to capitalize on them.

Keyword list: Wireless Networks, Multi-Hop, Network Coding, Rateless Codes, Relay Networks, Multi-User Information Theory, Geographic Routing

TABLE OF CONTENTS

Table of Contents 2
Section 1 – Introduction 3
Section 2 – Erasure Coding on the Network Layer 5
2.1 State-of-the-art in erasure coding on the network layer 5
2.1.1 Erasure codes for packet protection 5
2.1.2 Binary block codes 7
2.1.3 Reed-Solomon codes 7
2.1.4 Digital fountain codes 7
2.2 Knowledge gaps in network-layer erasure coding 7
2.2.1 Reed-Solomon codes 7
2.2.2 Digital fountain codes 7
2.3 Suggestion for collaboration in network-layer erasure coding 7
Section 3 – Network Coding and Geographic Routing 7
3.1 Background on geographic routing 7
3.2 Background on network coding 7
3.2.1 Network coding for one source 7
3.2.2 Network coding for multiple sources 7
3.3 Proposal for a joint collaboration within the WPR.5 on the topic network coding enhanced geographic routing 7
3.3.1 Scenario 7
3.3.2 Open problems and proposals 7
3.3.3 Collaboration 7
Section 4 – Information Theoretic Aspects Behind Network Coding 7
4.1 A brief survey of cooperative communications 7
4.2 Fundamental limits of the relay channel 7
4.3 Fundamental limits of general multiterminal networks 7
4.4 Open problems 7
Section 5 – Practical Coding Schemes in Networks with Relays 7
5.1 Introduction 7
5.2 Advance in distributed turbo coding for the relay channel 7
5.3 First applications of network-channel coding in wireless environments 7
5.4 Open problems 7
6 Conclusions 7
References 7

SECTION 1 – INTRODUCTION

The objective of work package 5 is to investigate the application of coding to wireless multi-hop networks. The utilization of network resources to the fullest, practically feasible extent is a central research objective in today's information dependent society. It is becoming increasingly evident and has been demonstrated by many authors (we will defer the more technical results to the mains sections) that to achieve this goal coding is absolutely necessary.

In particular, recently, the emerging field of *network coding* has received much attention. In contrast to traditional modes of network operation that merely relay and replicate data at intermediate nodes, network coding calls for nodes to perform algebraic operations on the received data. It is similar in spirit to source coding in its use of encoded source data, but is augmented by the ability of network nodes to actively operate on the data, giving network coding additional and surprisingly powerful capabilities. Drawing on these capabilities, network coding has the potential to significantly reduce delay and power consumption while improving bandwidth utilization and overall network robustness for a diverse set of applications across a wide variety of network technologies, such as overlay, wireless, and sensor networks for concrete applications. These applications include unicast transfers of data along multiple paths, reliable multicast, and best-effort dissemination of control traffic.

Though certainly network coding will be a main topic to investigate in this work package, we plan a broader scope of our work and particularly plan to explore the application of erasure-correcting codes on the network layer and their interplay with network coding. Furthermore, an information theoretic investigation of the fundamental limits is important in order to be able to assess the performance of coding schemes and quantify how close they are to optimality.

In spite of the large body of research results on coding for wireless networks and network coding, there is still a gap to be bridged between theory and practice, before a broad deployment of these technologies is conceivable. We believe that to that end our work package can significantly contribute. We have already started the process of determining what impediments need to be overcome and what the true benefits and underlying tradeoffs are when using network coding. In fact, the challenges are manifold and generally require the combined expertise of people on the coding and information theoretic side as well as on the networking side. Work package 5 involves a balanced mix of people with both perspectives which in our opinion is important for successful joint research activities that can capitalize on theoretical results and yet yield practically implementable engineering solutions.

For example, many network coding strategies are not yet fully deployable due to a high computational complexity or a significant decoding delay. The quest for low complexity architectures with maybe slightly suboptimal and yet practical engineering advantages is one of the main challenges we are facing. Another question that impacts the utility of network coding is the amount of computation and buffering it requires from intermediate nodes. While theorems in the network coding literature focus on what is possible information theoretically, and while there are some initial results that bound the resources required, experimentation is necessary to determine actual performance in real systems.

Furthermore, though there are already a large number of scenarios where network coding has proven its merit, we believe that this list is far from complete. In fact, hardly a month passes without some new application of network coding being reported. Some of these applications are theoretically well understood. In some cases the performance gains are difficult to analyze mathematically and have to be determined by means of simulations and measurements. We intend to equally emphasize both directions. On one hand to strengthen the theoretical framework in order to quantify the gains that network coding can provide in practice and, on the other hand, to experimentally investigate the benefits in different scenarios. One example is the suggested joint research activity on network coding in conjunction with geographical routing, where theory can help in the choice of a coding strategy, but the overall system has to be tested by means of simulation in the end.

Since our emphasis is on wireless multi-hop networks, one has to be aware that the application of network coding in wireless and sensor networks presents implementation problems different from the wired network setting. The most salient aspect of wireless transmission is the common use of broadcast via omnidirectional antennae. Broadcasting may beneficially provide collateral transmission to neighboring nodes for free, but also causes interference. The use of network coding may alter the nature of the interplay between this advantage and drawback significantly. Another key example is the communication/computation tradeoff of using network coding within sensor networks. Because communication is relatively expensive compared to computation in sensor networks, network coding may offer substantial advantages in power consumption.

We believe that due to these factors wireless networks are the most promising instances to investigate the utility of network coding. The rich variety of phenomena in the wireless realm may cause problems but at the same time offers opportunities to be exploited. Our objective in this work package is to develop coding theory and network coding into a powerful and practical tool for the design and operation of wireless multi-hop networks.

In the remainder of this document we plan to summarize the state of the art in various important areas and identify joint research areas between the collaborators in the work package. The rest of this deliverable is organized as follows:

Chapter 2 deals with erasure-correcting codes on the network layer. The cooperating partners are CNIT Torino and UGent, primarily. In this chapter, firstly the basics of the most important classes of such codes, Reed-Solomon and Fountain codes, are reviewed and knowledge gaps are identified. At that stage a joint research project is outlined and possible avenues of further work are suggested.

Chapter 3 looks at network coding in conjunction with a low complexity routing mechanism such as geographic routing. It is primarily a cooperation between LNT-TUM and CNIT Catania. We review the state of the art in geographic routing as well as important network coding fundamentals. We then argue that it seems promising to use network coding as a tool to improve the performance of geographic routing schemes and suggest to investigate approaches in this direction.

Chapter 4 explores the information theoretic background of network coding and surveys the information theoretic tools that might be useful in our understanding the performance and limits of network coding.

Chapter 5 deals with cooperative communication in relay networks. Based on the results for a single relay channel it gives an outlook on the basic properties and communication strategies in networks with many relays.

Chapter 6 finally concludes the document, summarizes the basic insights and provides an outlook on the research agenda ahead. Furthermore, based in the preceding discussion it suggests intersections with other work packages in the NEWCOM++ program.

SECTION 2 – ERASURE CODING ON THE NETWORK LAYER

2.1 State-of-the-art in erasure coding on the network layer

2.1.1 Erasure codes for packet protection

When digital information is packetized and sent over a transmission medium, it is common practice to provide protection on the physical layer against transmission errors (e.g., by using a Forward Error Correcting (FEC) code [1]). Unfortunately, this protection cannot prevent the occasional *erasure* of packets caused by the presence of noise, fading and/or interference during transmission. Indeed, from time to time conditions on the transmission medium will be such that the code on the physical layer is no longer able to correct the bit errors within a packet, i.e., the packet will be corrupted. Whenever that happens, the packet is eliminated by the data link layer, which continuously monitors the integrity of all received packets by verification of the Cyclic Redundancy Check (CRC) sum of the packet (which serves as a check on the packet's integrity). Evidently, the occasional erasure of packets affects the quality of the information received at the destination.

There are different ways to cope with packet erasures.

- In the case of delay-insensitive applications where the integrity of the received information is very important (e.g., file transfer), the destination sends to the source a request for retransmitting erased packets.
- In the case of delay-sensitive applications that can tolerate some amount of packet loss (e.g., audio streaming), no measures are taken to recover lost packets. It is assumed here that the error protection on the physical layer provides a packet erasure rate that is low enough to yield a sufficient Quality of Experience (QoE).
- In the case of delay-sensitive applications that can tolerate only a very low packet loss rate (e.g., video transmission, in particular HDTV [2]), the error protection on the physical layer is not sufficient to provide a packet erasure rate that yields a sufficient QoE. In principle this problem could be solved by improving the physical layer error protection. However, often the physical layer error protection cannot be altered because of standardization. Also, assuming the network provides a mix of applications, it is not efficient to dimension the physical layer error protection (to be used for all applications) according to the most demanding application. Further, requesting the source to retransmit erased packets is not convenient when the round-trip delays are too large to satisfy the delay constraints of the application. Hence, the only possibility left is the use of *erasure coding* [1, 3, 4], meaning that the source transmits redundant packets (in addition to the information packets) that allow the recovery of a limited number of erased packets at the destination.

In the following, we restrict our attention to packet protection by means of erasure coding, mainly in the context of video transmission.

Fig. 1 shows how the erasure coding for packet protection is organized. The packets are represented as rows, each containing L (binary or nonbinary) symbols. The source adds M = N-K parity packets to K information packets. The resulting N packets constitute a (N, K) *packet codeword* (PCW). Each of the L columns in Fig. 1 is a codeword from a (N, K) erasure code. In the j-th column, $\{B_{1,j}, ..., B_{K,j}\}$ are the K information symbols of the code, and $\{P_{1,j}, ..., P_{M,j}\}$ are the corresponding parity symbols. Hence, when ν packets from the PCW are erased, then in each of the L codewords from the erasure code the number of erased symbols equals ν as well.

The number of erasures that can actually be recovered depends on the particular code: a code with minimum Hamming distance d_H can recover any pattern of at most d_H-1 erasures (and might recover also some patterns consisting of more than d_H-1 erasures) [1]. Hence, when at most d_H-1 packets from the (N, K) PCW get erased (irrespective of their position within the PCW), the K information packets can still be reconstructed from the remaining correctly received packets.

From the above, it follows that codes with a larger minimum Hamming distance d_H yield a better error performance. The Singleton bound [1] states that for a (N,K) code, we have $d_H \leq N\text{-}K+1$; hence, the number of erasures that can be resolved (irrespective of their position) cannot exceed the number of parity symbols N-K. For binary codes, the Singleton bound holds with equality for the (N, N-1) single parity check code ($d_H = 2$) and the (N, 1) repetition code ($d_H = N\text{-}1$) only. For nonbinary codes, the Reed-Solomon (RS) codes satisfy the Singleton bound with equality.

Typically, the minimum Hamming distance d_H of practical codes increases with increasing codelength N and decreasing code rate K/N. However, there are some practical constraints on the values of N and K/N that can be achieved. First, the network resources used for the transmission of the N-K parity packets are no longer available to other applications. Therefore the overhead (N-K)/K should not exceed some maximum value; this implies a corresponding lower limit on the code rate K/N. Second, the erasure coding introduces an additional latency, which in a worst case situation equals the duration of the entire PCW (this happens when the first information packet of the PCW is erased, and we need the last parity packet in order to reconstruct the first information packet). In some applications the latency should be limited (in video applications, latency increases the zapping delay[1]), which imposes an upper limit on the length N of the codeword.

In the following we discuss several types of erasure codes for packet protection that have been considered in the literature or by standardization committees, namely standard *binary block codes*, *Reed-Solomon (RS) codes*, and *Fountain codes*.

[1] The zapping delay is the time that elapses between giving the command to change the TV channel and the appearance of the new TV channel on the screen [10].

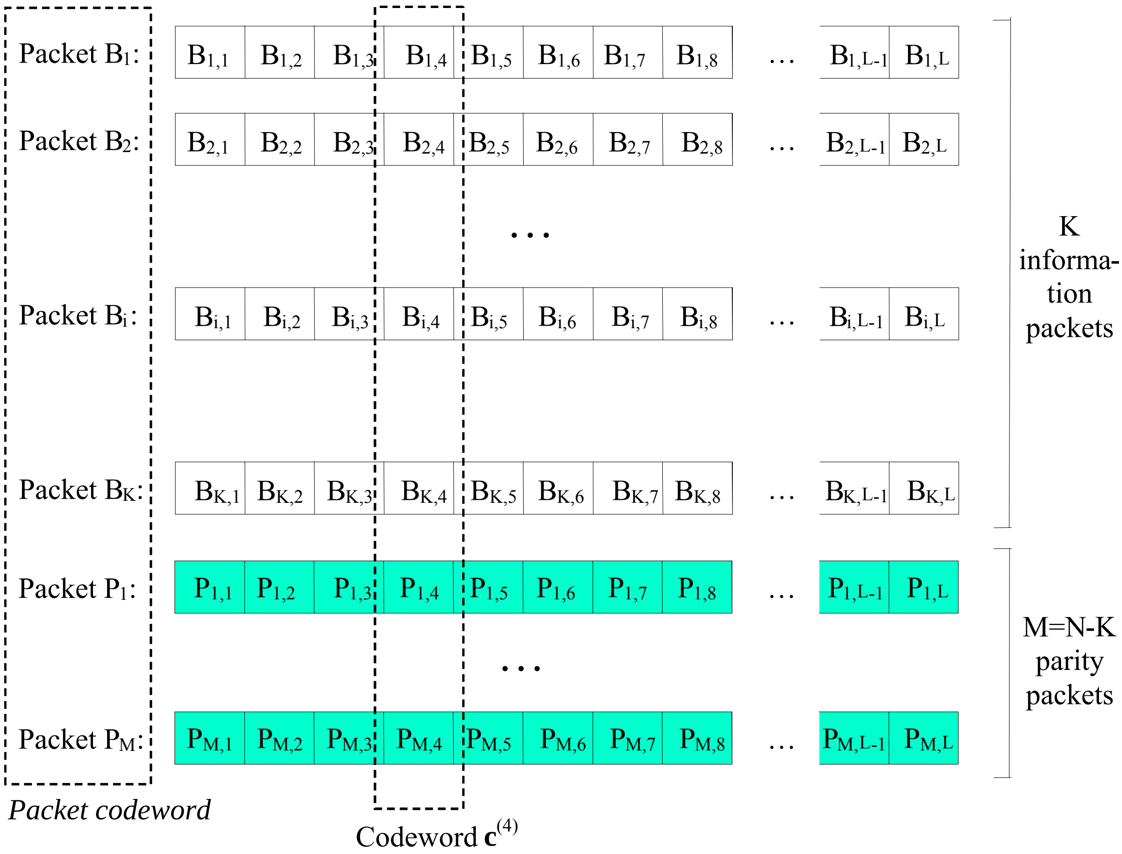

Figure 1: Construction of packet codewords.

2.1.2 Binary block codes

Binary codes [1] are very appealing candidates for the protection of information packets because of their simplicity, which allows the decoder at the receiver to remain of low hardware complexity. The drawback of binary codes is that they are not as powerful as the RS codes to be discussed in section 2.1.3. Below we briefly illustrate two types of binary erasures codes that have been proposed for the protection of video packets.

2.1.2.1 Pro-MPEG binary codes

In figure 2 we illustrate the construction of the binary Pro-MPEG code [5]. This code arranges a set of K information packets into a K_rxK_c matrix (green squares in figure 2, where $K_r = 7$ and $K_c = 5$). The information packets are protected by means of a row of K_c parity packets (red squares in figure 2). Each parity packet is the modulo-2 sum of the K_r information packets in the corresponding column. As such, all K_r information packets in a column are protected by a single parity packet. The packets are transmitted row by row (which corresponds to interleaving of the 'vertical' codewords).

The resulting $((Kr+1)K_c, K_c)$ PCW has a minimum Hamming distance $d_H = 2$. This implies that the Pro-MPEG code can correct one erasure, irrespective of its position. Moreover, the code can cope with any erasure pattern that has in each column at most one erasure; an example of such pattern is a burst of at most K_c consecutive erasures (figure 2 shows an erasure burst of length 3 in the first row). However, from the moment a column is affected by more than one erasure (as illustrated for two independent erasures in figure 2), erasure decoding fails and a permanent and unrecoverable packet loss results.

The decoder performance can be improved by adding more parity packets to the code. One possibility shown in figure 2 is to add a column of K_r+1 parity check packets that protect the rows of the matrix (orange squares in figure 2). This results in a $((K_r+1)(K_c+1), K_rK_c)$ code with minimum Hamming distance $d_H = 4$, which allows restoring any pattern of up to 3 erasures. In addition, many other erasure patterns can be resolved as well (e.g., all patterns with at most one erasure in each row or column).

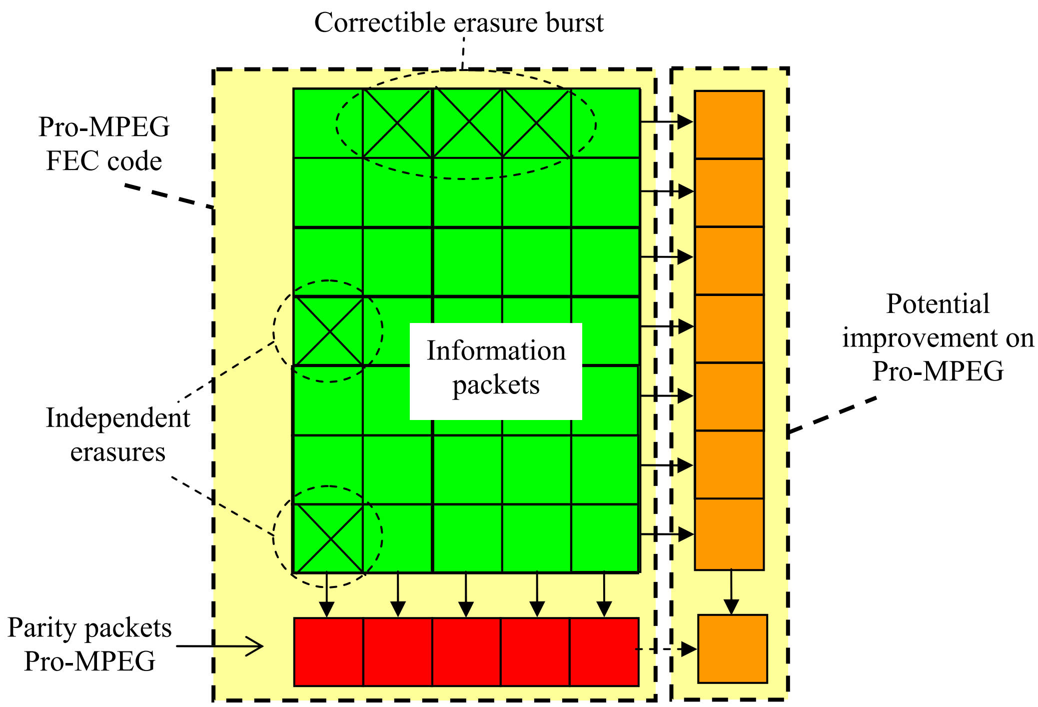

Figure 2: Pro-MPEG and proposed improvement.

The main advantage of the Pro-MPEG code (and its improved version as well) is the very simple decoding algorithm. However, there are several disadvantages as well.

- Because of the small minimum Hamming distance ($d_H = 2$ or $d_H = 4$) which is independent of the code parameters N and K, the resulting error performance might not be sufficient on adverse channels.
- In order to keep the overhead small, we need to make K_r large. Also K_c must be made large to provide protection against erasure bursts. Hence, the PCW is quite large, which increases the latency

These drawbacks are a motivation to investigate other erasure codes.

2.1.2.1 Binary block codes optimized for burst erasures

In [6], binary erasure block codes were introduced that are optimal (i.e., they are the best possible) for the protection against erasure bursts as well as independent erasures. These (N, K) codes are able to recover all erasure bursts of length up to N-K, and they maximize the number of other patterns of up to N-K erasures that can be recovered. Their (N-K)xN check matrix H can be written as H = (Z, Z, ..., Z), and Z is designed to have the maximum number of columns such that any N-K consecutive columns in H are linearly independent.

These binary codes were evaluated in [6] in a HDTV scenario with transmission over a DSL connection of 20Mb/s, where the latency and overhead were kept below 100ms and 5%, respectively. On the DSL line, bursty erasures occur when the physical layer codeword contains information from two video packets : in this case, a decoding error on the physical layer gives rise to two erased packets.

Assuming realistic traffic profiles and realistic error performance on the physical layer of the DSL line, it turned out that a PCW with only 3 parity packets achieved the target of less than 1 decoding failure in 12 hours, and performed only slightly worse than a RS code with the same values of N and K.

2.1.3 Reed-Solomon codes

RS codes are nonbinary block codes with symbols belonging to a Galois field GF(2^q) [7]. Typically, q = 8, in which case a symbol of the RS code can be represented by a byte. For given q, the maximum size of a codeword is 2^q-1. As (N, K) RS codes satisfy the Singleton bound with equality (i.e., d_H = N-K+1), they can restore up to (N-K) erasures in a codeword – which is the maximum number of erasures that can be recovered by any code with (N-K) parity symbols. Hence, RS codes are optimal erasure codes, and are by consequence preferable over any other code as far as error performance is concerned. However, as compared to binary codes, the RS codes have the drawback that their decoding is rather complex.

The performance of RS codes for the protection of video packets has been investigated in several papers.

- In the case of independent packet erasures, a (N, K) RS code performs considerably better than a binary (N, K) code, because the latter has the smaller minimum Euclidean distance. However, when the erasures tend to occur in bursts, a (N, K) RS code is only marginally better than a (N, K) binary code that has been designed to correct bursts of at most N-K erasures [6].
- In [8], the potential of RS codes in the presence of packet loss caused by buffer overflow in the aggregation network has been examined. When no erasure coding is used, buffer overflow yields irrecoverable loss of packets. When erasure coding is used, the transmission of parity packets on one hand increases the probability of buffer overflow, but on the other hand allows to recover some of the lost packets. The main result from [8] is that using a RS code with only 3 parity symbols yields better performance than no erasure coding, for channel loads up to about 95% and a buffer size of 40 packets.
- Another recent article [9] examined the efficiency of RS codes for packet protection in the context of variable rate video codecs. Two strategies were considered : in the first strategy the parameters (N, K) of the RS code remain constant, yielding a variable latency depending on the instantaneous bitrate; in the second strategy the latency was kept constant, yielding RS codewords with (N, K) depending on the instantaneous bitrate; for both strategies, the number N-K of parity packets is kept constant. It turns out that strategy 1 is better (worse) than strategy 2 when the erasures occur independently (in bursts).

2.1.4 Digital fountain codes

Digital fountain (DF) codes are a new class of low-complexity near-optimal erasure correction codes. As has been seen, a (N, K) RS code has the ideal property that if any K of the N transmitted symbols are received, then the original K source symbols can be recovered. However, RS codes are practical only for small values of K, N, and q; standard implementations of encoding and decoding have a cost of order K(N-K) log2 N packet operations. Moreover, the code rate must be chosen before transmission based on estimation of the packet loss probability. The code rate selection needs to be somewhat conservative, as the code cannot be extended if the loss rate is higher than expected.

DF codes have been designed to overcome these issues. The first DF codes, named LT codes, were invented by Luby in 1998 [11]. The idea of a DF code is as follows [12]. The encoder is a fountain that produces an endless supply of water drops (encoded packets); for example, the original source has a size of KL bits, and each drop contains L encoded bits (see Figure 3). A receiver holds a bucket under the fountain and collects drops until the number of drops in the bucket is a little larger than K, which allows to recover the original source. DF codes are said to be rateless, in that the number of encoded packets that can be generated from the source message is potentially limitless. Therefore, one can send as many encoded packets as are needed in order for the decoder to recover the source data, without the need to choose a code rate *a priori*. The source data can be decoded from any set of K' encoded packets, for K' slightly larger than K (e.g., about 5% larger). DF codes also have very small encoding and decoding complexities. With probability 1-d, K packets can be communicated with average encoding and decoding costs both of order K ln(K/d) packet operations. The overhead K'-K is of order $K^{1/2}(\ln(K/d))^2$.

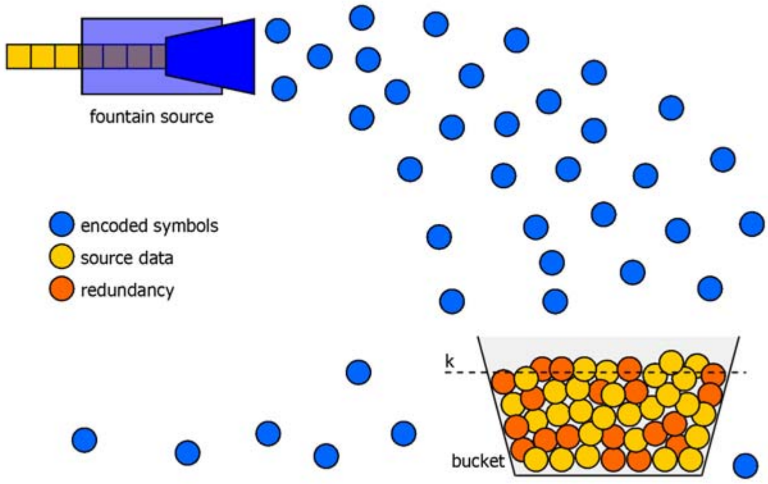

Figure 3 - Digital Fountain encoder and decoder

In practice, every encoded packet is obtained as the bit-wise XOR of a number r of source symbols taken uniformly at random from the source. An example of this is shown in Figure 4. Packet E1 is obtain as the XOR of symbols a, b and c, with r=3; packet E2 is obtained as the XOR of b with no other symbol, with r=1; packet E3 is obtained as the XOR of symbols b and d, with r=2. The degree r of each packet follows a specific statistical distribution (e.g., LT codes employ the robust Soliton distribution [11]) that is designed to minimize complexity for a given probability of successful decoding. Each encoded symbol represents a linear equation in Galois field GF(2), where the variables are the source symbols. Therefore, decoding amounts to solving a linear system, e.g., employing Gaussian elimination. In practice decoding is achieved through message passing as follows. First, all degree-1 equations (e.g., packet E2 in the example above) are identified, and the corresponding input symbol is solved. Then, the degree of all equations containing the known input symbol is reduced by one, generating new equations of degree one. The process is iterated until all input symbols have been resolved. The degree distribution must ensure that this process can be completed with very high probability, i.e., that it is very unlikely that, at any stage, there is no available equation of degree one and the decoding process has to be stopped.

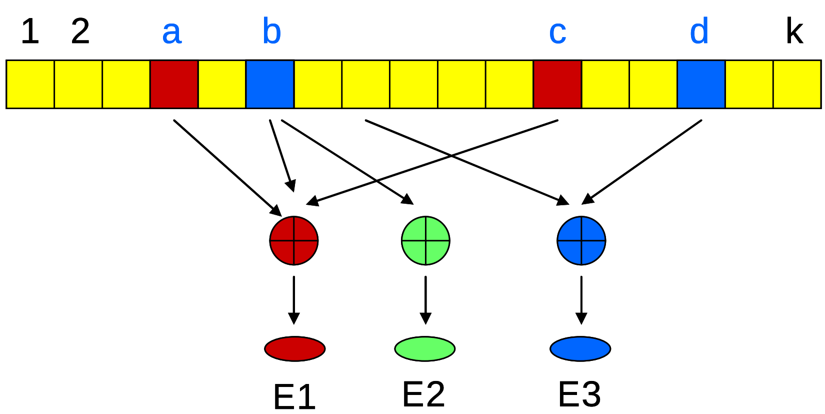

Figure 4 - Generation of encoded packets for a Digital Fountain code

Following the success of LT codes, Raptor codes were presented in [13]. Raptor codes build on LT codes by first applying a block pre-code, followed by a LT code. The use of the block precode allows to “weaken” the degree distribution of the LT code, as the undecoded symbols can be recovered using the outer block decoder. This allows to significantly decrease the code complexity, achieving with high probability a linear encoding and decoding cost. The success of Raptor codes is witnessed by their adoption in the most recent multimedia communication standards such as 3GPP MBMS (Multimedia Broadcast/Multimedia Service) [14] and DVB-H [15].

More recently, there has been an increasing interest in the application of DF codes to video communications. This has led to the proposal of a set of improved DF codes, as well as video protection systems based on DF codes.

As for the improved DF codes, in [16] distributed LT codes are proposed, in which the XOR operations can be done at different nodes of a network. In [17] a method is proposed to construct a DF code based on turbo codes. In [18] a sliding-window approach is proposed, which yields improved decoding probability, especially for short data blocks. In [19,20] it is proposed to modify LT and Raptor codes in order to achieve unequal error protection (UEP). UEP is very important in multimedia communications, as a layered image/video coder yields a compressed file where the first layers are more important than the last ones. In [19,20] UEP is achieved by not taking the symbols to be XORed uniformly at random; rather, the most important symbols are picked with a higher probability, making their correct decoding more likely. A similar approach is followed in [21], where windows of increasing size are considered; all windows start from the first symbol, and they encompass increasing numbers of symbols. In the DF code, the symbols are taken at random in a given window, but the shortest windows are used with higher probability than the longest ones; as a result, the most important symbols are picked more frequently, and can be decoded with higher probability than the least important ones.

Regarding video applications, in [22] a protocol for asynchronous video multicast is proposed, employing Tornado codes and LT codes. In [23] a protocol for live unicast video streaming is introduced, based on rateless codes, and its performance is evaluated. Similar ideas are described in [24], where the asymptotic behaviour of an asynchronous media streaming system based on DF codes is investigated. In [25] Raptor codes are used for streaming of stereoscopic video. An application of Raptor codes to H.264 scalable video is described in [26].

2.2 Knowledge gaps in network-layer erasure coding

2.2.1 Reed-Solomon codes

It has been pointed out above that RS codes satisfy the Singleton bound with equality. Hence, a (N, K) RS code can recover any pattern of up to N-K erasures, and no other (N, K) code can perform better. Therefore, as far as error performance is concerned, RS codes are the best codes for protection against erasures. However, the drawback of RS codes is their decoding complexity.

A first method to decode a RS codeword affected by erasures is based on the linearity of the RS code. Denoting the RS codeword by the row vector $\mathbf{c} = (c_1, ..., c_N)$, we have $\mathbf{cH}^T = \mathbf{0}$, where the (N-K)xN matrix $\mathbf{H}$ is the check matrix of the code, and $\mathbf{0}$ is an all-zero row vector. Note that the elements of $\mathbf{c}$ and $\mathbf{H}$ are in GF(2^q). Let us consider the case where ν coded symbols have been erased. The positions of the erased symbols in the codeword are denoted $n_1, n_2, ..., n_\nu$. We define by $\mathbf{x} = (x_1, ..., x_\nu)$ the ν erased symbols from $\mathbf{c}$, by $\mathbf{y} = (y_1, ..., y_{N-\nu})$ the remaining symbols from $\mathbf{c}$, by $\mathbf{A}$ the matrix containing the n_1-th, n_2-th, ..., n_ν-th row of $\mathbf{H}^T$, and by $\mathbf{B}$ the matrix containing the N-ν remaining rows of $\mathbf{H}^T$. Then we have

$$\mathbf{xA} = \mathbf{b}, \text{ with } \mathbf{b} = -\mathbf{yB} \quad (1)$$

As long as $\nu \leq$ N-K, the matrix **A** has full rank ν, and **x** can be obtained by solving (1), which boils down to solving a set of ν linear equations with ν unknowns in GF(2^q).

A second method to decode a RS codeword is the erasure decoding algorithm that is based on the specific structure of the RS codes. Because we know the location of every erasure that occurs in the PCW, the decoding algorithm is simplified as compared to the case where error locations are unknown. Consider the received word $r(z) = r_0+r_1z+...+r_{N-1}z^{N-1}$ that is affected by ν erasures at locations $n_1, n_2..., n_\nu$, with $0 < \nu \leq$ N-K. The erasure decoding algorithm is as follows :

1. Using the known erasure locations, compute the erasure locator polynomial defined by
$$\Gamma(z) = \prod_{m=1}^{\nu}(1 - z\alpha^{n_m})$$
where α is a primitive element of GF(2^q).

2. Compute the derivative $\Gamma'(z)$ of the erasure locator polynomial.

3. Replace the erased symbols in the received word r(z) by zeroes and compute the syndrome polynomial S(z) defined by
$$S(z) = \sum_{i=1}^{N-K} S_i z^{i-1}$$
with $S_i = r(\alpha^i), i = 1,..., N - K$.

4. Compute the modified syndrome polynomial defined by $T(z) = \Gamma(z)S(z) \bmod z^{N-K}$.

5. Then, the erasure values are equal to $e_{n_m} = \dfrac{T(\alpha^{-n_m})}{\Gamma'(\alpha^{-n_m})}, m = 1,..., \nu$ and the estimated erasure polynomial and corrected codeword are respectively equal to $\hat{e}(z) = \sum_{m=1}^{\nu} e_{n_m} z^{n_m}$ and $\hat{c}(z) = r(z) + \hat{e}(z)$.

Considering video packet transmission over the DSL line, the use of (N, K) RS codes with N-K = 3 gave rise to satisfactory performance. In this case, erasure decoding involves either solving a set of at most 3 linear equations in GF(2^q), or applying the erasure decoding algorithm where the erasure locator polynomial has a maximum degree of only 3. However, when considering video packet transmission over wireless channels, the channel is more hostile than the DSL channel. This implies that for wireless transmission larger values of N-K will be required, in order to resolve a higher number of erasures per codeword. When N-K increases, also the decoding complexity of the RS code increases. The high RS decoding complexity in the case of wireless transmission might be problematic, especially when considering small hand-held devices that are very limited in power consumption.

Therefore, the following should be further investigated :
- what erasure correction capability is needed in the case of wireless transmission ?
- what is the associated decoding complexity of the RS codes ?
- is a binary code with worse performance but smaller decoding complexity a better choice than an RS code ?

2.2.2 Digital fountain codes

Although DF codes have attracted a lot of interest, there are several aspects that require further investigation.

One aspect of the DF code design lies in the optimization of the degree distribution. For LT codes, the proposed robust soliton distribution [11] has been designed to achieve good erasure correction capabilities with limited computational complexity. Theoretically, however, there is no proof that this is the optimal distribution. For Raptor codes, the employed distribution is the results of a numerical optimization process. All modified LT and Raptor codes, e.g. [18,19,20,21] employ the original distributions, which are arguably suboptimal for the modified codes. Designing a good distribution for a DF code is not a simple process, and thus far this has limited the amount of work spent on this problem.

Similarly to other channel codes, DF codes have been developed for generic data. However, the most important practical applications are in the field of multimedia communications, especially video. In this case, it is known that channel coding has to be tailored to the structure of compressed video files, e.g., providing unequal error protection of different layers of scalable video. This can be done with an ad-hoc design of the code, as in [19,20], or at the system level, allocating more transmission rate or more retransmission opportunities for the most important layers. However, it is not clear which strategy would provide the best results.

Another issue lies in the protocols used for transmitting data using DF codes. The DF code is a flexible tool to generate encoded packets, but in a networked environment some communication between sender and receiver is needed to manage the DF code. Communication protocols are expected to have a significant impact on the performance of a DF code. Such protocols are not established, and are an important topic of current research in this field. Interestingly, such protocols may differ between RS and DF codes because of the different features of these codes. Unlike RS codes, DF codes require a non-zero overhead to guarantee correct decoding; however, their flexibility in generating the packets may eventually make up for this.

The remarks above highlight the need of a comparison between RS and DF codes in a relevant application scenario, e.g., wireless video. The main pros and cons of RS and DF codes are well known. RS codes for erasure channels are optimal, in that they require the minimum possible amount of encoded symbols to recover the whole data block. RS have been used for many years in several applications, are not proprietary, and co-decoding software is freely available. On the other hand, they require to set the coding rate in advance; therefore, RS are not adaptive with respect to varying network conditions. Moreover, they operate over Galois fields, and this limits the flexibility in defining symbol and block sizes. Consequently, their performance may be affected by the constraints on the symbol and block dimensions, as padding may be necessary to match the actual code constraints. Finally, RS codes have relatively high complexity. Although sub-quadratic algorithms have been proposed, for practical block lengths their complexity is not significantly reduced.

On the other hand, DF codes are rateless, so they can be adapted to varying network conditions. However, it should be noted that the delay in the decoding process is indeed dependent on the erasure probability, and should be kept under control if such codes are applied to delay sensitive data such as real time video. When using DF codes, a rate overhead must be accounted for, depending on the desired probability of decoding success. DF codes do not impose any limitation on the symbol and block sizes, and, for Raptor codes, the complexity is reportedly linear with the block size.

In the end, the advantages of either code depend on many application details such as the required degree of flexibility, the available computational resources, and the involved block and symbol sizes. A thorough comparison of RS and DF codes is not currently available for practical application case studies.

2.3 Suggestion for collaboration in network-layer erasure coding

Based on the remarks above, it is clear that an important knowledge gap lies in the comparison between RS and DF codes under realistic conditions. Therefore, a joint research activity will be set up between CNIT and UGent, aiming at performing such comparison. The activity will involve the following steps:

- Identification of one or more relevant wireless video case studies (e.g., 3GPP, WiFi, WiMax, ad-hoc networks, and so on).
- Identification of a software platform for simulation.
- For each case study, definition of a communication protocol using RS codes, and one using DF codes.
- Simulation of wireless video employing either protocol.
- Performance comparison in terms of overhead, latency, video quality, encoding and decoding complexity.

SECTION 3 – NETWORK CODING AND GEOGRAPHIC ROUTING

3.1 Background on geographic routing

On the topic of geographic forwarding, several techniques have been proposed. In these solutions the availability of location information is assumed by means of GPS or GPS-less techniques [27] and used for performing packet forwarding without requiring either the exchange of routing tables among network nodes or the explicit establishment of a route from a sender to a destination.

The simplest strategy for performing geographic data delivery is simply forwarding the packet to the neighbour node that is closest to the destination. However, if this greedy method is used, a solution to the problem of possible holes in the network must be found. Karp and Kung in GPSR [29] propose an efficient technique to recover from such a problem. GPSR is a non-energy-aware routing protocol that uses a planar subgraph of the wireless network to route around low-density network regions. This technique seems to be scalable and suitable for dynamic networks, although batteries of nodes in the planar subgraph tend to be quickly exhausted.

An alternative geographic routing protocol specifically designed for sensor networks is GEAR [37]. It saves energy by disseminating queries only to appropriate regions. Moreover, each node estimates the cost to reach the destination and then refines it when the transmission of the first packet has actually taken place, thus accounting for routing around holes.

In GeRaF [38, 39], the next relay node is not known a priori, since the current source node does not know which neighbors are active or sleeping. Consequently, upon need of forwarding a data packet, the source node sends a message which contains both its own coordinates and the coordinates of the destination. Then, a receiver contention scheme takes place and, ideally, the best available relay node is chosen based on geometrical considerations, being this relay the closest node towards the destination among the available slices considered in the forwarding area. This solution, while simple, does not guarantee data delivery since the sender node could also not find any awake relay node in its proximity, when trying to forward data. Moreover this solution does not include the possibility of transmitting with different power levels.

Concerning the power management, only in the recent past, few papers addressing the problem of using variable transmission power have appeared. PARO [28], for example, tries to minimize the amount of power needed to deliver data packets between nodes in the radio coverage of each others. This is achieved selecting some intermediate forwarding nodes, called *redirectors*, in the area between the source and the destination. In fact, since a large increase in the transmission power is needed every time the coverage area of a node must be enlarged, trying to exploit available neighbour nodes in the closest proximity allows to reduce the overall amount of transmission power required for data delivery. In such a way, PARO shows to be efficient and to drastically outperform traditional broadcast-based routing protocols which, instead, aim at reducing the number of hops between a sender and a receiver. Note, however, that PARO does not include routing facilities and therefore must be used along with appropriate routing protocols.

In [30] Akyildiz et al. introduce a variable range routing protocol for sensor networks which addresses the problem of determining the optimal transmission range of each node, say the *knowledge range* (*KR*), which allows to make geographic routing more energy-efficient. However, in case of large sensor networks, the algorithm may converge slowly given that the transmission ranges are tuned based on a backward learning process run end-to-end between the source and the destination.

Energy efficiency may be preserved at the link level too since techniques to put nodes periodically into sleep can help to reduce energy consumption. To this purpose a lot of work on energy efficient MAC protocols for sensor networks has been proposed [31,33,34,36]. Among them, one of the most interesting is STEM [32]. STEM aggressively puts nodes in sleep mode and ensures satisfactory latency for forwarding data, as it allows to trade latency for energy consumption. Each node is assumed to be equipped with two radios, the one for data transfer and the other for signaling. In the so-called *monitoring state* the data radio is in sleep, while the control radio wakes-up periodically. Once a node has a packet to forward to one of its neighbours, it starts polling this neighbour on the control channel. As soon as the neighbour wakes-up and receives one of these beacons, it acknowledges the receipt of this packet and turns ON its data radio, thus allowing the data transfer.

It is worth pointing out that the majority of the above schemes lack in integration between MAC and routing. This feature instead represents a key factor when designing approaches customized for sensor networks where the simplicity of the procedure, as well as the effectiveness in energy management, are required. In this perspective, a strategy for reducing energy consumption which combines routing with a technique to put nodes into sleep mode is GAF [35]. This protocol selects nodes equivalent under the routing capability point of view and turns them off; consequently, the whole area where nodes are distributed is divided into *virtual grids* where only one node at a time is elected to remain active. However, the major disadvantage of this protocol resides in the high degree of synchronization it requires so that simpler solutions would be much more desirable.

In [40] the advantages of using different transmission power levels are investigated in the view of introducing an integrated MAC/Routing protocol. MACRO does not require location information to be exchanged in the network, thus reducing energy consumption. In fact the proposed protocol requires that each node only knows its own position and the position of the destination and forwarding is done according to geographic criteria. Consequently, in order to select the next relay node in the forwarding process, a competition is triggered at each hop in such a way that the most energy efficient choice is taken. To this aim, an analytical framework is derived to allow MACRO protocol to perform the optimal choice. Performance of MACRO are evaluated through ns-2 simulation and compared to the performance of other geographical forwarding protocols showing that MACRO both reduces energy consumption and is suitable for applications which perform data aggregation.

3.2 Background on network coding

In traditional networks, coding is used for two different purposes. At source nodes source coding is applied for data compression and therefore reduction of the required transmission bandwidth. On the other hand, channel coding is used at the link level to ensure reliable communication, thus enabling us to model links as essentially error-free channels subject to the channel capacity. Only recently, in the surprising and fundamental paper by Ahlswede et al. [41], it was shown that it is in general not optimal to restrict coding to these two applications. Even in a scenario where all redundancy at the sources has been removed (i.e., the sources emit independent, unit entropy bit streams) and channels are error-free, encoding data streams at intermediate nodes in general increases the capacity of the network. Traditionally, intermediate nodes in a network were restricted to merely routing or replicating information. We speak of *network coding*, when intermediate nodes perform mathematical operations on the incoming data streams to obtain the output.

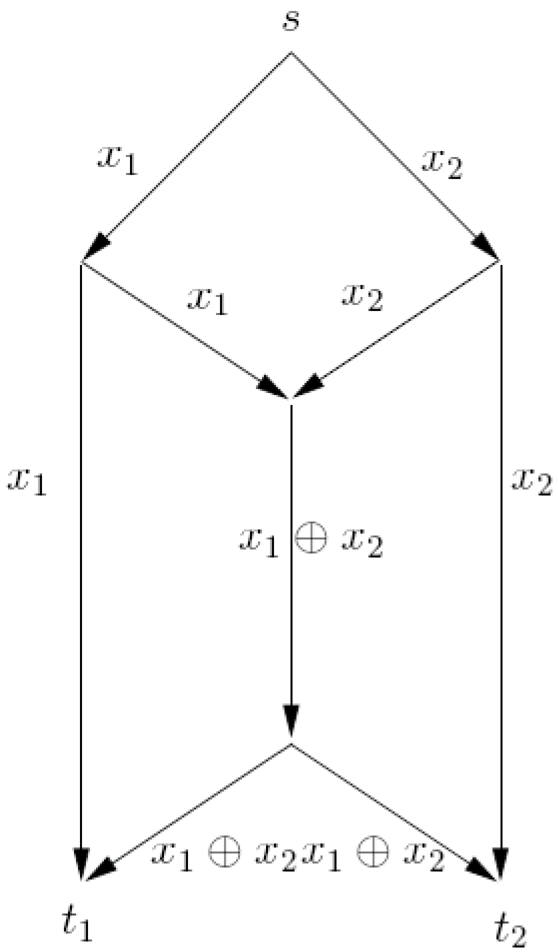

Figure 5 - The “butterfly” network for a multicast connection and the corresponding network coding solution.

To illustrate this abstract concept consider the (by now classical) example of a network in Figure 5. Every edge has unit capacity and the source at node s wishes to transmit at a rate of two bits per unit time (the bits are x_1 and x_2, respectively) to both receivers t_1 and t_2. This rate is not achievable using routing and replicating only, however the figure suggests a simple network code that indeed allows the desired rates of communication. The encoding takes place at the middle link, where the binary XOR of x_1 and x_2 is transmitted. Now, it is possible to reconstruct both bits at both destinations.

3.2.1 Network coding for one source

Ahlswede et al. in [41] not only introduced the idea of network coding, but also gave a complete characterization of the capacity of multicast networks. A multicast network consists of one sender transmitting the same information to a set of receivers. For the special case of one receiver, the maximum rate of information flow has long been known to be given by the capacity of the minimum cut separating source and destination [42]. Moreover, no encoding at intermediate nodes is necessary to achieve it. The new and surprising result was that, for multiple destinations, the achievable rate is the minimum over all minimal separating cuts between the source and one of the destinations.

However, in general this rate cannot be achieved with routing and replicating alone, in general network coding at intermediate nodes is necessary.

The multicast problem has received much attention recently and it is well understood. After the fundamental result, that a multicast network essentially has a max-flow min-cut interpretation, Li et al. in [43] showed that the maximal rate can be achieved using linear encoding functions at the nodes. Furthermore, Ho et al. in [44] proposed a scheme in which nodes select their linear encoding functions independently and randomly. They showed that if the coefficients are selected from a finite field of sufficient size, this random coding scheme is in fact capacity achieving. Choosing coefficients at random enables nodes to operate independently, without any central authority in the network, a highly desirable property for practical networks.

Subsequently, there has been considerable interest in network coding applied in a setting where the use of network resources is subject to a cost. Under this premise, we seek to find an optimal allocation (e.g., an allocation of minimal cost) of network resources to satisfy a specified set of connections. Lun et al. in [45] showed that finding a minimum-cost subgraph which can support a single multicast connection (i.e., it satisfies the max-flow min-cut conditions) is equivalent to solving a linear program. Thus, we see that the minimum-cost multicast problem essentially decomposes into two independent subproblems, namely the identification of the minimum-cost subgraph to code over and the construction of a network code, that achieves capacity.

In an extension of this work, the authors demonstrate that the resulting linear program can be solved in a decentralized way. Thus a combination of the decentralized optimization algorithm with a capacity achieving code construction technique, such as the inherently decentralized random coding, results in a completely decentralized minimum-cost multicast problem. This novel approach provides two advantages over the existing techniques. First, since network coding includes routing and replication of information as a special case, a multicast instance with network coding performs at least as well or better in terms of resource consumption. This is a gain that can be quantified using some appropriate cost function. Second, and more importantly the decentralized network coding algorithm is of greatly reduced complexity compared to the traditional approach. With routing only, the minimum-cost multicast corresponds to the Steiner tree problem, which is NP-complete [46] and thus has to be approximated in practice. This is not only computationally demanding, but also requires a central authority to carry out the computations. Thus, we see that network coding can be very useful in reducing the complexity of network operation, making it attractive from an engineering point of view.

3.2.2 Network coding for multiple sources

While for one source the multicast capacity of a network is known and practical, capacity-achieving codes are known, for multiple sources these questions are not answered yet. To indicate that network coding takes place only within the set of packets of a particular user we also speak of intra-session coding which is typically applied by forming linear combinations of packets with random coefficients at intermediate nodes of the subgraph.

As opposed to the multicast, which is well understood and for which practical capacity achieving schemes have been presented, we still know little about the case of multiple unicast connections sharing a network. This problem is of great practical significance, since most traffic in existing networks, such as the Internet, is point-to-point and not multicast. The multiple unicast problem significantly differs from the multicast and is considerably more complicated.

In the multicast setup, every receiver is interested in the entire information emitted by the source; therefore, all messages and linear combinations of messages are relevant to every receiver. Random coding works because, if the field size is sufficiently high, the random linear combination of messages arriving at every receiver will be linearly independent, making it possible for the receivers to deduce the original messages. In contrast, in the multiple unicast setup every receiver is only interested in the messages from its corresponding source. Thus, linearly combining messages of different unicast connections introduces interference, something undesired that has to be cancelled at a later point. This is the phenomenon that complicates the multiple unicast problem dramatically. There are a number of negative results on this problem - in fact most known results are negative - which we will outline next.

Dougherty et al. in [47] addressed the question whether linear network coding is sufficient not only for the multicast but also for the general network information flow problem, where different connections, possibly unicast or multicast, share a network. The authors disproved this conjecture by constructing a counterexample, which achieves capacity only when using nonlinear encoding functions. The network they consider is a large (46 nodes) and very sophisticatedly constructed graph and the network code is tailored specifically to it. This result, which is without doubt fundamental, unfortunately does not provide us insight into how to construct an optimal network code for an arbitrary network. A network coding solution like this apparently cannot be computed with an algorithm; instead careful examination of the underlying network is required. Thus the lesson learned from this particular example does not generalize to other networks. All we know is that examples exist where linear network coding is insufficient; however, the size and complexity of the proposed counterexample suggest that such instances are scarce. This intuitive argument is supported by the fact that, as of now, no ``simple'' structure that requires nonlinear network coding has been reported, though much research has been devoted to this problem.

On the other hand, it is well known, that network coding can also provide gains in the multiple unicast scenario. Techniques for inter-session network coding have been suggested in [48-51]. There, the code construction problem is formulated as a linear optimization problem, reminiscent of a flow formulation. Unfortunately, the resulting optimization problems are rather complex and therefore significantly further down the road of practical implementation. In contrast COPE [52] is a practical and successfully implemented network coding protocol. However, the opportunistic and heuristic nature of COPE is difficult to analyze rigorously. One such attempt is presented in [53] where COPE is modeled as a linear program.

3.3 Proposal for a joint collaboration within the WPR.5 on the topic network coding enhanced geographic routing

3.3.1 Scenario

Let us consider a scenario where static nodes exchange some data packets throughout the network. More specifically, we assume a certain source node S wants to send a packet to a multicast group. While in previous work existence of an underlying connected graph was assumed where no interference among simultaneously sending nodes was met, we propose to exploit a mechanism like MACRO to select time by time the best next relay which can allow improving a packet’s progress towards the destinations. This mechanism will take into account both a simple MAC mechanism to reduce collisions at the medium access level and a routing algorithm. For what concerns the kind of data processed by nodes, intra-session random network coding can be used according to the approach proposed in e.g. [44,45].

3.3.2 Open problems and proposals

Use of a more realistic and not collision-free MAC and routing protocols pose many problems in terms of:

- End-to-end delivery probability
- End-to-end delivery delay
- If nodes are mobile this can additionally decrease performance. Impact of mobility should be estimated

- What is the decrease in performance when comparing no MACRO combined and MACRO combined approaches?
- It would be desirable to design an algorithm specifically thought to allow proper multicast data forwarding through exploitation of both network coding and MACRO functioning. This algorithm should be optimized to give some guarantees in terms of delivery probability

3.3.3 Collaboration

The network coding background of LNT-TUM and expertise in terms of routing and MAC layer protocols of CNIT-Catania can be combined to investigate a more realistic scenario and estimate the impact of mobility on system behaviour.

SECTION 4 – INFORMATION THEORETIC ASPECTS BEHIND NETWORK CODING

4.1 A brief survey of cooperative communications

Network communications become more reliable and efficient when nodes support each other to transmit data. By supporting, we mean enabling neighbouring nodes to share their resource and their power with the hope that such a cooperative approach leads savings for the overall network resources and power consumption. Preliminary results over the past decade show that the benefit of user cooperation may be very real. Obviously, user cooperation in networks can potentially take place whenever the number of communicating nodes exceeds two. Therefore, the three-terminal network, first introduced by van der Meulen in 1968 [98], certainly constitutes a fundamental unit in user cooperation and deserves particular attention. A vast portion of the literature, especially in the realm of information theory, has already been devoted to the two-hop and relay channels, seen as special cases. Despite its seeming simplicity, the capacity of the general relay channel is still unknown. The first upper and lower bounds on the capacity discovered by van der Meulen in the early 70’s [99] were significantly improved by Cover and El-Gamal in 1979 [57]. Most of the results in their work have still not been superseded.

The difficulty to derive refined bounds on the relay channel capacity or to practically realize efficient signalling strategies most likely explains the moderate interest for relaying and user cooperation after the early 80’s. Recently, however, two seminal papers by Sendonaris et al. have greatly revived the enthusiasm of the community [95] [96]. In their work, the Authors propose user cooperation as a form of diversity in a cellular uplink scenario and show its benefit under various metrics. User cooperation in the sense of Sendonaris et al. can be seen as a special form of relaying where the source and the relay exchange their role. Also important are the contributions of Laneman and Wornell addressing the performance of important relaying protocols in wireless environments [54] [87] [88]. Another complementary line of thought comes in the form of novel information-theoretic results and new insights into information theoretic (random) coding for relays by Kramer et al. [85] and Chong et al. [74].

In parallel, the network coding paradigm has motivated intensive research after the landmark papers of Yeung et al. in 1999 [104] and Ahlswede et al. in 2000 [41]. Network coding must be seen as another form of user cooperation, where the nodes not only share their resource and their power but also their computation capabilities. It is also worth noting that the most general definition of network coding (i.e., whenever intermediate nodes perform mathematical operations on the incoming data streams to obtain the output) somehow comprises distributed coding for multihop or relay channels.

4.2 Fundamental limits of the relay channel

As pointed out in the previous section, the relay channel and its multiple variants (e.g., multiple relay channels or two-way relay channels) may be considered as the basis for network-coded or distributed coding schemes for error (or noisy) channels [54]–[56]. The study of relay channels dates back to Cover et al. [57]. For it, the authors presented two relaying protocols (namely, decode-and-forward (D&F) and compress-and-forward (C&F)) and compared their achievable rates with the max-flow-min-cut capacity upper bound [58]. Other relaying protocols such as partial decoding (PD), amplify-and-forward (A&F) and linear relaying (LR), helped to widen the analysis [59]–[61]. Among all, D&F and C&F turned out to be asymptotically optimal for the relay infinitely close to source and destination, respectively, while PD was shown to outperform all other techniques for the half-duplex relay [59]. A variety of contributions to relaying including new bounds, cut-set theorems, power control strategy and some the earlier results on half-duplex were also proposed in [KHO-04]. With more than one relay, the single-source single-destination channel is studied in [62]–[64] and references therein. In particular, the latter presents achievable rates with D&F, PD and C&F.

The extension of the previous results to multiple-access (MAC) and broadcast (BC) schemes, where multiple sources are present, has been recently considered in [65]–[70]. First, the MAC assisted by a relay is presented in [65]–[66] where, assuming time-invariant fading and channel state information (CSI) at the sources, the rate regions with D&F and C&F are derived. Additionally, LR is analyzed in [67] for the MAC without CSI at the sources, and shown to be optimal at the high multiplexing gain regime. On the other hand, the BC with relay is first studied in [68]. A network with a single source and two receivers, one acting as relay of the other, is analyzed and rate regions derived using D&F and PD. In parallel, LR for the two-hop BC (i.e., no connectivity between source and destinations) is studied by Jafar et al. in [70]. In that work, the optimal power allocation on the relays is presented, as well as the celebrated MAC-BC duality.

Finally, the bidirectional or two-way relay channel is studied in [71]. The two-way relay channel consists of two phases: a multiple access phase and a broadcast phase. It may be viewed as a generalization of the classical XOR network coding scheme to sources with different data rates. An achievable rate region for the two-way relay channel is derived in [71] for Gaussian channels with Gaussian signalling.

4.3 Fundamental limits of general multiterminal networks

Depending on the transmitter and receiver architectures and on the information flow, there are many different configurations of networks, e.g., the two-source/one-relay/one-destination configuration (four-terminal network) and its “butterfly network extension” with two destinations (five-terminal network). Those basic networks constitute two fundamental units in network coding. The information theoretic analysis of such scenarios is still in an early stage and would provide powerful communication strategies for user cooperation in networks. A general cut-set theorem for information flow in networks with multiple terminals is due to Cover and Thomas [58, chap. 14, section 10]. This theorem is analogous to the well-known max-flow min-cut theorem [42] and can be interpreted as follows: the rate of information flow across any cut dividing the set of communicating nodes in two parts cannot exceed the mutual information between the channel inputs at the transmitter side and the channel outputs at the receiver side conditioned on the knowledge of inputs at the receiver side. In theory, this theorem can be used to (upper) bound the capacity region of arbitrary multiterminal channels, including those mentioned above, e.g., the four-terminal multiple-access relay channel (MARC). Capacity results in network information theory are often difficult to prove, and in many important cases, such as MAC and degraded RC, the multiterminal channel capacity is known just because the upper bound provided by the max-flow min-cut theorem is achievable. Although there also exist many other examples where the capacity has been shown to be smaller than what is indicated by the cut-set bound (e.g. the Gaussian vector broadcast channel), the max-flow min-cut theorem (along with its recent extension to networks with multiple states [84]) remains a very powerful tool to derive information theoretic benchmarks for assessing the efficiency of our proposed network-channel codes.

On the other hand, [75] has proposed a novel approach towards finding the transport capacity of wireless networks with a large number of nodes, as opposed to the simple few-nodes channels studied by conventional network information theory. Scaling laws have been derived from this approach in various settings. Prominent among extensions of this work are [76], where the use of advanced multiuser schemes is shown to improve network transport capacity significantly and [101], where an achievable rate expression for a degraded Gaussian channel with relays is given, as well as bounds on its transport capacity (see also [102] and [69]).

4.4 Open problems

A joint research will be conducted by CNRS and CTTC, aiming at investigating the information theoretic aspects behind network coding. This activity will involve the following steps:

- Cast practical identified networking scenarios into an information-theoretic setting: rate region characterization for each case; derivation of bounds on outage capacities (proper metric for slow fading); extension of capacity theorems for correlated sources;
- Evaluate, in terms of capacity gain, the optimal trade-off between the benefits and limitations of cooperation in wireless communications, and including realistic constraints as delay and coordination requirements.

SECTION 5 – PRACTICAL CODING SCHEMES IN NETWORKS WITH RELAYS

5.1 INTRODUCTION

With the significant advance in both coding theory (e.g., probabilistic coding and decoding, point-to-point capacity-achieving codes) and technologies over the last two decades, the promise of relaying and user cooperation (at large) seems very real. Research efforts are currently focused on developing practical user cooperation coding schemes to harvest the gain predicted by information theory. This, of course, includes the design of network-channel codes for noisy (wireless) links and distributed coding for relay or multihop channels, as special cases. Unfortunately, globally optimal design of channel codes for general network environments is hard. Therefore it is necessary to consider other avenues in the pursuit of good practical codes for simple network models, with the ultimate goal of understanding the fundamental principles of general networks.

5.2 Advance in distributed turbo coding for the relay channel

Before going deeper into details, it is worth stressing some basic notions on relaying which will extend to network coding in wireless environments. In the relay channel model, the three terminals, i.e., the source S, the relay R and the destination D, are conceptually divided into two subsets corresponding to the cuts of interest, namely the broadcast cut which separates S from {R,D} and the multiple-access cut, which separates {S,R} from D. This distinction is useful for analysis and code design but does not completely reflect the reality since both transmission phases (modes) could perfectly take place at the same time. Analysis and code design are also dependent on the nature of the relays (e.g., analog or digital, full-duplex or half-duplex), on the nature of the links (i.e., to the assumptions made on the fading statistics) and on the considered relaying protocol (in the sequel, we will mainly focus on the decode-and-forward protocol type of cooperation). Even though early information theoretic studies on relaying essentially considered full-duplex relaying, current research on practical protocols is primarily based on the premise of half-duplex relaying.

The idea of Distributing Turbo Coding (DTC) for the relay channel was first proposed by [107] [81]. In [107], the Authors consider digital relays, quasi-static Rayleigh fading and a simplified decode-and-forward protocol in which S does not transmit in the multiple-access cut (second mode) which, of course, facilitates the reception of the relays symbols at the destination (at the price of an inefficient use of spectral resource). In the first mode, the source S basically broadcasts the coded signals to R and D. The relay R then decodes the received source signals, interleaves them prior to the encoding (this is a mathematical operation) and transmits them during the second mode. From the destination viewpoint, the two kinds of received signals form a distributed turbo code which is iteratively decoded. This very simple concept was later refined, improved and even generalized in a number of ways. This very simple concept has been later refined, improved and generalized in a number of ways. Prominent contributions in this area include [105] for the most general form of relaying in full-duplex mode (i.e., the source is allowed to transmit in the second mode and the relay can simultaneously transmit and receive) and [72] [80] [73] [106] for the most general form of relaying in half-duplex mode (i.e., the source is allowed to transmit in the second mode and the relay can receive in the first mode and transmit in the second mode). Powerful distributed codes are built from either turbo codes or LDPC codes. In order to facilitate compound code design and optimization, most of works consider ergodic channels (i.e., fast Rayleigh fading) [94] [90].

DTC usually assumes that the relay can perform error-free decoding and thus error-free re-encoding (perfect DTC). Such an assumption is true if the S–R link is perfectly reliable which unfortunately rarely happens in realistic wireless communication scenarios, e.g., under slow Rayleigh fading. Of course, the S–R link can be made error-free using ARQ. However, an ARQ strategy will increase the delay (not tolerable for delay-sensitive applications) and reduce the system transmission throughput. To fight back this impediment, another very promising research avenue, directly inspired from [93] (Pearl's belief propagation algorithm), has emerged, e.g., in [97] [89], where soft information is used at the relay for both decoding and re-encoding. Unfortunately, [97] requires the relay to be equipped with digital and analog transmission chains, since the relaying signal (formally identified to A Posterior Probability (APP) pmfs on relay parity bits) is analog (continuous) in nature. This severe and probably unrealistic constraint is removed in [89] where an ad-hoc discrete equivalent model is proposed in order to convey the soft nature of the produced relay parity bits.
Clearly, further investigations are needed to tackle the issue of the S–R link imperfectness in a proper way, "oblivious" from the destination viewpoint. Interestingly, the paradigmatic notion of obliviousness can also apply to code design from the source perspective, as recently suggested by Katz and Shamai [82].

5.3 First applications of network-channel coding in wireless environments

The application of network coding to wireless packet networks has been already investigated from a network perspective in a variety of contributions. In particular, the asymptotic optimality of random distributed network coding for wireless networks with and without packet erasures has been demonstrated. More details about this subject can be found e.g., in [91] [92]. Here, we address the problem of wireless network coding but from the physical (PHY) layer perspective.

Joint channel/network coding was first proposed in [77] for the MARC. The authors considered a cellular based mobile communication system with two users sharing a common digital relay which performs network coding. Error-free transmission between the sources and the relay was assumed, while errors can occur at the links user-to-relay and user-to-destination. The authors considered the use of LDPC codes at the transmitter side. Each transmitter encodes its data and broadcasts the coded information to the destination and to the relay. At the receiver side, the destination can exploit the additional redundancy provided by the relay to improve performance. Since the relay provides additional information for both users at the same time, the decoder can use a joint decoding strategy exploiting the turbo principle. In particular, the authors showed that the channel and network codes can be seen as a single LDPC code spatially distributed. Therefore, at the destination, decoding of the channel and network codes can be done jointly on a single Tanner graph. Significant improvements in terms of diversity and coding gain with respect to non-cooperative systems were observed. A similar approach was considered in [79], where convolutional codes were assumed for channel coding at the transmitters, and in [78] for the two-way relay channel.
The error free assumption for the user-to-relay channel was removed in [103]. To fight error propagation in the relay the authors allow the relay to process soft information: instead of hard decoding the messages from the sources, the relay perform network coding at the "soft level", i.e., it transmits to the destination the likelihood ratio of the network coded messages. As in [79], the authors considered a network with two sources that share a single relay. Each source uses a convolutional code to encode the information bits. At the destination, decoding of the information of source 1 and source 2 is done jointly in an iterative manner, exploiting the extra information provided by the relay.
Network coding has also been considered for cooperative diversity in [100]. A system consisting of two source nodes A and B coupled as partners for the transmission of their data to a destination is considered. Each source transmits its own information and relays the information of the partner source as well. At time slot t node A proceeds as follows: if it has decoded correctly the information broadcasted by node B at time slot t–1, it transmits the codeword obtained from the XOR of the codeword containing its local information and the codeword containing the information of its partner. Otherwise, it simply transmits its own information. Clearly, a flag bit is needed to indicate which of the encoding methods is used. Decoding at the destination is then carried out by using an iterative algorithm that iterates between the decoders for nodes A and B, resulting in a significant gain compared with other approaches.

Although those previous works are all based on simplistic assumptions (e.g., orthogonal links), they have enlightened the potential benefits of joint network/channel coding for wireless networks and paved the way for further investigations.

5.4 Open problems

A joint research activity will be set up between CNRS and CTTC, aiming at investigating the potential of network coding in the presence of noisy links. In the very beginning, we will focus on very simple models (e.g., MARC). The activity will involve the following steps:

- Propose new powerful network-channel codes on graphs able to approach the theoretically promised rates and diversity gains;
- Formalize both the decoding/encoding problems at the relays and the joint decoding problem at the destinations in terms of factor graphs and message-passing algorithms (SPA) [86] [84];
- Propose a theoretical framework to solve the critical quantification issue of the intermediate (continuous) propagated messages at the relays;
- Propose new efficient joint source-network-channel coding and decoding strategies for correlated sources;
- Address the problem of channel uncertainty;
- Investigate of the relationship between network topology and coding gain. This includes a mathematical characterization of the network coding performance in network models with dynamic topologies (random graph topologies, mobility, and propagation models) and the identification of favorable topologies for network coding (i.e., fundamental characteristics and self-organized mechanisms able to lead the emergence of such topologies in an autonomous and distributed fashion).

6 CONCLUSIONS

In this first deliverable we have attempted to provide a meaningful survey of relevant results and challenges in our field. Four clusters have been identified, namely

1. Erasure coding on the network layer (mainly for video transmission)
2. Network coding as an enhancement of geographic routing
3. An information theoretic analysis of network coding
4. Coding for relay networks

After the kick-off meeting in Munich in February 2008, we plan to meet again in September 2008 for an update on the progress and coordination of the research activities.

Finally, we would like to point out interesting inter-package cooperations that arise naturally. In our opinion, we can particularly benefit from the exchange with packages 4 (Iterative receiver design), 6 (Relaying and cooperation in networks) and 11 (Opportunistic networks). A possible platform for this would be to hold co-located meetings.

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