Paper

Reliable and High QoS Wireless Communications over Harsh Environments

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Abstract—One of the most challenging research fields in which research community has taken a very active role is focused on trying to bring the features of wireless networks into line with the traditional wired solutions. Given the noisy and lossy nature of the wireless medium, it is more difficult to provide a comparable Quality of Service (QoS) and Reliability over wireless networks. This lack of reliability avoids the use of wireless solution in scenarios under harsh environment and mission-critical applications. In this paper we propose an inter-node collaborative schema with the aim of improving the achievable QoS level for multicast streaming, through the use of Network Coding and the algebra it is based on. We also present an implementation of the described algorithm on the OPNET discrete event simulation tool. Experimental results highlighting the performance achieved by the proposed algorithm and its improved efficiency as compared to other solutions are described.

Keywords—cooperative wireless networks, multicasting management techniques, next-generation networks, opportunistic wireless, quality and performance evaluation, QoS, reliability and performance modelling, reliability of networks, wireless and mobile networks.

1. Introduction

During the last decade, the research field related with improvements on the QoS (Quality of Service) of communications infrastructures for both in-home, in-building and WAN networks has been a very active topic ([1]–[3] and [4] describe good surveys on it). The research community has produced a large number of protocols and mechanisms enabling the use of networks with shared resources to provide time-sensitive services and distribute applications requiring a high QoS [3], [5]–[7].

Wireless communications have been introduced for applications such as web browsing or other services with low QoS requirements. In fact, the lower layers of the OSI model [8] for wireless environments do not offer a link quality comparable to that of wired networks, and they require the use of large buffers to compensate the effects of packet loss [9]. One of the most challenging research fields for the Future Internet therefore focuses on providing the same QoS level for a wireless link as for a wired link, facilitating the deployment of high-demand services in wireless networks, despite their noisy and lossy nature.

A new research branch emerges from new innovative Information Theory field based on network coding that has recently appeared, known as the *"Modern Theory of Communication"* [10], [11]. Our paper proposes a solution derived from the algebraic principles of network coding [12], and describes a basic implementation of the algorithm on a discrete event simulation platform. We propose a collaborative and opportunistic coding framework for this purpose.

This paper describes a collaborative algorithm between wireless nodes and an innovative queue management system that provides an increase in the quality of service and reliability of wireless networks as compared to already existing solutions. The broadcast nature of wireless networks makes this algorithm particularly effective, and considerably reduces the reliability difference with respect to wired solutions. This research paper describes an example of multicast streaming (e.g., for applications such as SVC, Scalable Video Coding) in a noisy and time-variable environment. The proposed algorithm increases both QoS and stability with regard to temporal variations in the environment, typical of the wireless nature of the links.

The rest of the paper is organized as follows. In the following paragraph (Section 2) the main fundamental algebraic principles are described, over which the algorithm proposed in this paper is based on. Afterwards, in Section 3 we describe the algorithm and in Section 4 we analyze the different configuration parameters that have been designed for it. In Section 5 the design and algorithms of the different nodes integrating the proposed protocol are described so that the complete communication network is defined. In the same section, the details of the used simulation environment and the generation of the harsh environment are analyzed.

Section 6 describes the experimentation process and the results of the performance measurement, highlighting the achieved milestones. Afterwards, Section 7 shows the comparison of the achieved performance among the proposed new algorithm versus the classical solutions for the multicast transmission over harsh environments. Finally, we describe the conclusions of the obtained results and future research lines are glimpsed.

2. Algebraic Fundamentals

2.1. Transmitted Information (vs) Sent Packets (Information Representation)

Lets consider an acyclic directed graph $G = (V,E)$ with unit capacity edges, a source node $s \in S$, and a multicast transmission to a set of receiver nodes $T \subseteq V$. R. L. Li *et al.* [13] and Koetter and Médard [12] proved that multicast capacity is achievable if linear network coding is used in a directed graph network, whereas using routing schemas it is generally impossible to guarantee that multicast capacity can be achieved and finding the best possible routing schema is computationally NP-hard.

The proposed algorithm is based on the linear combination of information units (as an abstract representation of packets) [14], [15]. Each *"original"* packet (ingress to the combining node) is considered to be an independent variable in a system of equations [13]. This being the case, the packets can be transmitted *"systematically"*, so that they are not combined with other packets, and only the original variable is sent. However, an interesting alternative is to send equations with these variables – in other words, linear combinations of the original incoming packets – to the networked nodes [16], [17]. As it is a linear system, it can be represented as a matrix of equations, whose elements are the original packets and the coefficients used to generate the linear combinations among them. Each combined packet can be represented as [18], [19]:

$$P_{lc} = \alpha \cdot P_{in_1} + \beta \cdot P_{in_2}, \tag{1}$$

where $\alpha, \beta \in F_{q \equiv 2^8}$.

Being P_{in_1} and P_{in_2} the original incoming packets to be redistributed, α and β the coefficients from a Galois finite field F_q with a field size of ($q = 2^8$), so that the finite field is defined as F_{2^8}, and P_{lc} the outgoing packet as a result of linear combination of incoming packets. It must be taken into account that P_{lc} will be in general the concatenation of several symbols (*bytes*) obtained multiplying incoming symbols by random coefficients.

The complete representation of each system is therefore determined by the extended matrix, made up of the equation coefficients and the payload of the packets, whose bytes would be the free terms of each equation. By basic algebra, if a node obtains sufficient equations to complete the rank of the matrix, it has sufficient information to decode the whole system of equations, and consequently the original packets (information).

2.2. Galois Fields

Galois fields are the basis for coding and collaboration algorithms. Galois fields are sets of finite numbers with operations such as the sum and the product, and linearity properties. Galois fields are frequently used in areas such as cryptography and coding theory, and they have been implemented in the field of network coding since its creation [20]. The finite fields habitually used are of size 2^8 [21], which is very useful in computing as each element can be represented in exactly one byte.

2.3. Galois Matrices

Galois matrices are matrices whose elements belong to a Galois field. In this research work, we will use Galois matrices to perform linear operations over packets. The proposed network coding algorithm takes the original packets as variables in a system of equations, which can be represented by a matrix. This means that this system can be solved if a sufficient number of equations are available. As linear operations can be performed on the matrices, a Gauss-Jordan reduction can be applied to them and their rank can be calculated by receiver nodes. Moreover, the matrix can be even further reduced, and it can be determined which variables can be solved, even if the whole rank is not available. The number of these variables is referred to as *"solvable"*, and it determines the level of information decoded by a node.

2.4. Transmission Matrix

Generalizing the outgoing linearly combined packet generation statement, we have already mentioned that we can consider that the transmission is defined by a system of equations, also known as the *"Transmission Matrix"*. Let's consider P_{in_i} as the i-th original packet, we can consider a bucket of K original packets, being K the *"Generation Size"*, the following linear system equation defines the generation of the linearly combined outgoing packets P_{lc_i}:

$$\begin{bmatrix} P_{lc_1} \\ P_{lc_2} \\ P_{lc_3} \\ \dots \\ P_{lc_K} \end{bmatrix} = \begin{pmatrix} \xi_1(e_1) & \xi_2(e_1) & \dots & \xi_K(e_1) \\ \xi_1(e_2) & \xi_2(e_2) & \dots & \xi_K(e_2) \\ \xi_1(e_3) & \xi_2(e_3) & \dots & \xi_K(e_3) \\ \dots & \dots & \dots & \dots \\ \xi_1(e_K) & \xi_2(e_K) & \dots & \xi_K(e_K) \end{pmatrix} \cdot \begin{bmatrix} P_{in_1} \\ P_{in_2} \\ P_{in_3} \\ \dots \\ P_{in_K} \end{bmatrix} . \tag{2}$$

The K-by-K matrix of $\xi_j(e_i)$ elements corresponds to the coefficients randomly chosen in F_q, where $\xi_j(e_i)$ represents the random coefficient multiplying the symbol P_{in_j} incoming from the (e_i) artificial edge to s node [10], [18]. The generalized expression for Eq. (1) is therefore:

$$P_{lc_i} = \sum_{j=1}^{K} \xi_j(e_i) \cdot P_{in_j}. \tag{3}$$

The Transmission Matrix is therefore compounded of the linear combinations generated by multiplying the incoming packet information bytes and the random coefficients in the defined Galois finite field of F_q. It will be necessary to send K packets P_{lc_i}, being $\{i = 1, \dots, K\}$ and K the generation size, to send the whole original information.

3. Principles of the Algorithm

3.1. Algorithm's Basic Operation

Given that the algorithm is located at OSI-Layer 2, also known as Link Layer, everything proceeding from the higher layer is considered to be encapsulated in a Layer-2 payload. As we have mentioned, each *"original"* packet is associated with a variable in a system of equations. This association is made by the transmitter node. As the total information to be transmitted will be potentially greater than the size of this system of equations, we propose that several systems of equations may co-exist throughout the transmission, not necessarily at the same time. Each of them must be identified with respect to the others. To designate each system of equations, we will call each one a different *"generation"*, being K the generation size or number of equations of the linear system. When performing operations such as adding or multiplying *"variables"* (packets), these operations are performed directly on the packet payload, taking each of the bytes as elements of the Galois field of size 2^8.

3.2. Packet Format

In order to provide the basic functionalities of the algorithm, a specific packet format has been defined (Fig. 1). The fields of the proposed packet format are:

Generation ID. A unique identifier of the system of equations (also known as generation). It is defined by the original transmitter of the packets.

Generation Size. The size of the system of equations (number of independent variables, or original packets involved in each generation). We have previously defined K as the number of equations in the linear system or the generation size.

Remaining. Number of remaining equations to be transmitted by the original transmitter. The receivers thus know the remaining amount of equations to be transmitted for the generation to be completed. Being P_{lc_i} the i-th current packet, remaining field will be equal to $K-i$, being K the generation size.

Type. The type of packet transmitted. In the present version of the algorithm there are two types of packet:

- *None*: packet transmitted by the original transmitter. These are combinations of original packets, not recombination of them.
- *Collaboration*: collaboration packets sent by intermediate network nodes. These are recombination of the packets received by the node.

3.3. Source Coding Mechanism

One of the most simple mechanisms for obtaining greater reliability in the wireless link is the use of coding in the source node, also known as source coding. The possibility of linearly combining the packets provides a very straightforward mechanism for this coding. The idea consists of the source node sending a system of equations containing the information corresponding to the whole original packet generation. The system of equations transmitted by the source may be composed of the minimum number of equations or may be over-determined. This over-determined linear system takes into account added packets with redundant information, so that it generates a system of equation

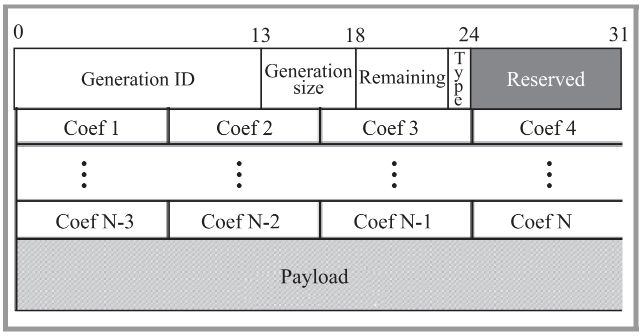

Fig. 1. Proposed packet format description.

of size ψ, being $\psi > K$. This system can be expressed by means of a matrix, and its structure affects the quality of reception by the nodes.

3.4. Inter-Node Network Coding Collaboration Schema

The *"Remaining"* field of the packet informs the nodes of how many more packets of the current generation will be sent by the transmitter. This packet field can be used as the trigger for the collaboration mechanism. To make use of the broadcast characteristics of the wireless medium, after a minimum threshold for the remaining field has been exceeded, the receiver nodes in turn transmit one or more collaboration packets, linearly combining those they have received from the generation in progress. As the information is multicast, one packet can provide information to several nodes at the same time. This collaborative concept is the basis of the network coding [17], [21], and although more sophisticated mechanisms may be designed, even this simple version provides added reliability to the network.

In Fig. 2, a simple example of the collaborative schema is depicted. On the upper side, it can be observed that the transmitter broadcasts 2 packets (P_1 and P_2). Both P_1 and P_2 are the result of linearly combining 2 "original" information packets (x and y), so that $P_1 = x+2y$ and $P_2 = 2x+y$. Therefore, the coefficients to generate P_1 have been $\xi_1(e_1) = 1$, $\xi_2(e_1) = 2$ and $\xi_1(e_2) = 2$, $\xi_2(e_2) = 1$ to generate P_2. However, due to the harsh environment (interference created by the jammer node) and the lossy wireless medium, R_2 and R_4 lose one packet each node. R_2 node lost packet P_2, whereas node R_4 lost packet P_1. Nodes R_1 and R_3 received the whole information transmitted by the transmitter (P_1 and P_2), therefore they can collaborate with the rest of the network to complete the information distribution to the remaining nodes. In the proposed algorithm, nodes that completed the whole information reception will be able to collaborate, in case they are configured to do so, in the redistribution of the information that some nodes have missed.

In the example shown in Fig. 2, as the packet that nodes R_2 and R_4 miss is a different one, transmitter would need to retransmit 2 packets in classical transmission schema. However, in the proposed algorithm, nodes that have received the whole information can collaborate with linear

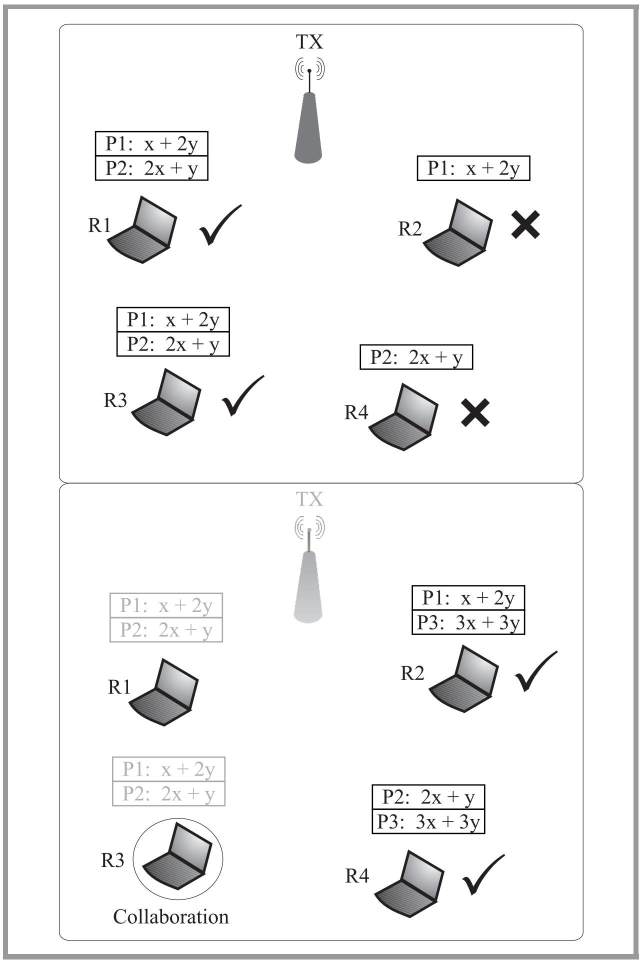

Fig. 2. Collaborative schema. Transmitter sends original packets (upper side) and some packets are lost due to harsh environment. Complete reception with collaborative schema (lower side)

combinations of received packets. Hence, as can be observed in the lower side of Fig. 2, node R_3 acts as a collaborative node and sends a packet (P_3) generated as a linear combination of the packets it has received (P_1 and P_2), for example: $P_3 = P_1 + P_2$, it therefore generates $P_3 = 3x + 3y$. This single packet helps nodes R_2 and R_4 to complete the whole original information reception after just 1 packet collaboration.

4. Algorithm Configuration Parameters

It can be observed that the proposed algorithm described up to now has a variety of configurable parameters, some of them vital for improving its performance, and also resulting in improved QoS and reliability.

4.1. Source Coding Structure

The source coding mechanism provided by the algorithm is defined as the linear combination of packets of the same generation, and can be fully represented by the matrix known in advance as the "transmission matrix".

Basic (*fundamental*) transmission matrix: by elementary algebra, even on a perfect scenario with no packet loss, the minimum matrix to be transmitted must have at least as many equations as variables, and it must also have the maximum matrix rank (equal to the generation size). These are the minimum conditions required for the information to be decoded by the receiver node(s).

We now introduce the concept of *"Coding Density"*, which is the number of different coefficients other than zero of each of the source-generated equations (Fig. 3). It can vary from density 1 (systematic coding) to a density equal to the size of the system of equations (dense coding). In other words, the coding density represents the number of original packets that have been combined to generate an outgoing packet. To ensure that the transmission matrix rank will always be the maximum, the main diagonal elements are always other than zero.

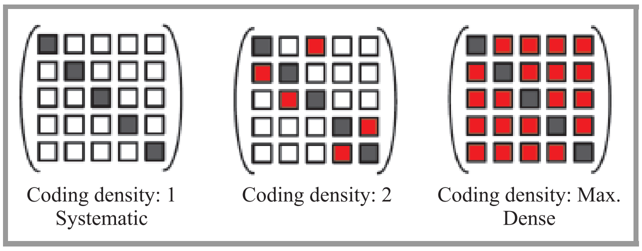

Fig. 3. Coding density options for the transmission matrix.

If the transmission matrix is created using systematic coding, the variables can be solved as soon as they are received, which can be an advantage on some time-sensitive scenarios. By the other side, increasing the coding density improves the probability of a single packet being able to help in the reconstruction of the original information when several nodes have lost different packets.

4.2. Redundancy Blocks

All the packets exceeding the minimum equation system size (rank of the system of equations or generation size) are considered as redundancy packets. A variable number of redundancy blocks can be configured. Each of these blocks can be configured in size and structure, in a similar way to the basic transmission matrix (Fig. 4).

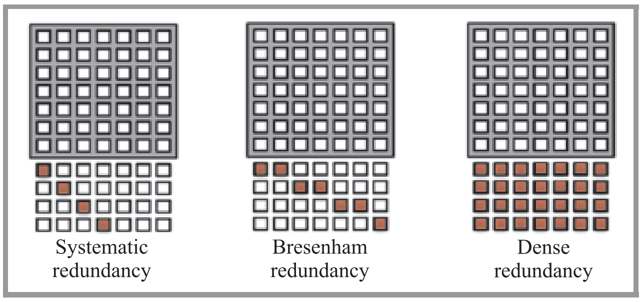

Fig. 4. Redundancy schema options.

Systematic redundancy: this is equivalent to repeating a limited number of packets, as only the main diagonal of a redundancy block is filled. This gives us a single non-zero coefficient in each redundant equation.

Random redundancy: the matrix is filled with variable density, in the same way as the basic matrix, starting by filling the main diagonal. In this case the density defines the number of non-zero coefficients of a given equation.

Bresenham redundancy: this is based on Bresenham's line-drawing algorithm. In this way we obtain the minimum number of non-zero coefficients in each equation, but all the variables are present in the redundancy block.

4.3. Collaboration

All the receiver nodes can be configured to send collaboration messages to help the rest of the receivers. The main parameters are:

- **Collaboration threshold**: this determines the value that needs to be reached by the *remaining* field for a node to decide to send its collaboration message (see packet format in Fig. 1).

- **The number of collaboration messages** sent by each node can be configured, from zero (only transmitter node is able to send packets) to an arbitrary number.

5. Simulation Environment

In the following paragraph we describe the simulation environment implemented to analyze the behaviour of the proposed algorithm, and we study in detail the differences in the QoS achieved by the different mechanisms in study.

5.1. Simulation Tool

A discrete event simulator (OPNET) has been chosen for implementing the proposed algorithm and for testing its performance. This is an event driven network simulator, widely used in the industry and academia to analyze the performance of a diverse range of communication networks.

5.2. Wireless Environment Simulation

OPNET has a specific high performance software package to simulate wireless networks and the effects of the wireless broadcast nature. It is based on the implementation of a series of stages (pipelines) which model each step of a wireless transmission/reception, from the antenna gains in transmission and reception, transmission and propagation delays, as well as wireless medium effects. For the purposes of this paper we have simplified the pipeline cascade, maintaining the pipelines that are the most critical (Fig. 5) for correctly representing a real environment.

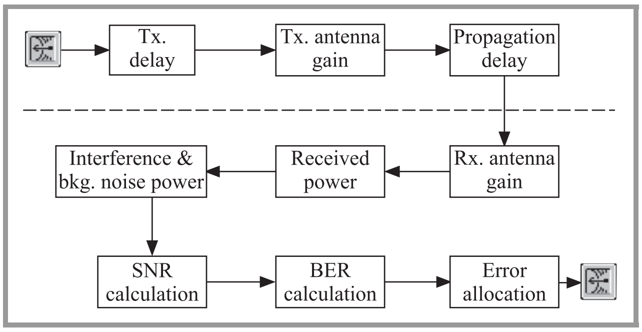

Fig. 5. Wireless pipeline stages.

5.3. Simulated Losses

Packet losses have been simulated in the OPNET environment by means of *interfering nodes*, also known as *jammer nodes*. These jammer nodes have been designed to generate *noise packets* with configurable duration, power and temporal distribution for the noise packet inter-arrival. The implementation of these customized jammer nodes is essential to define a controlled harsh environment.

5.4. Algorithm Design for Simulated Nodes

Three types of nodes have been implemented in the aforementioned discrete event simulation tool:

- **Bridge/transmitter node**: this is the only node with two interfaces: a wired Ethernet interface and a radio interface for the wireless link. This bridge node takes the Ethernet frames that come from an standard Ethernet source, encapsulates them into packets with the previously described format and groups them into generations (systems of equations composed of set of packets whose number is defined by the generation size), running the described algorithm. This node is in charge of source coding mechanism, and therefore the transmission of the transmission matrix. Algorithm 1 describes the pseudo-code of the bridge/transmitter node's process.

Algorithm 1: Bridge/Transmitter node process pseudo-code

```
Insert packet into buffer.
if (# linearly indep. packets in buffer ≥ Gen. Size) then
    Create transmission matrix.
    for i = 0 to transmission matrix rows do
        Send packet or equation
    end for
end if
```

- **Receiver node**: all the receiver nodes are potential destinations for the multicast streaming. They attempt to solve the systems of equations, and, if they are thus configured, to send collaboration packets. These nodes are therefore able to combine packets. Algorithm 2 describes the pseudo-code of the receiver node's process.

Algorithm 2: Receiver node process pseudo-code

```
Receive packet from RF interface.
Read packet coefficients.
Insert coefficients into reception Matrix (Φ)
if (Rank of Matrix Φ has been increased) then
   insert packet into reception buffer
   if (Rank of received matrix == max. rank) then
      information received
      if (collaboration not sent) then
         for i = 1 to collaboration messages do
            send random collaboration packet
         end for
         set status as 'collaboration sent'
      end if
   end if
else
   delete packet
end if
```

- **Sniffer node**: to view the functioning of the algorithm, a simple sniffer node has been implemented, which prints the data of all the packets in the wireless medium into a file. This node provides very valuable information of the protocol performance for an off-line analysis.

6. Algorithm Performance Measurement

A test-bed scenario has been used for comparative measurements and performance analysis of different algorithms. Some of the previously described parameters have been varied, taking the degree of information received as the result. As the algorithm is geared to a multicast streaming, the average of the information received has been measured at all the receiver nodes, or, equivalently, the perceived loss rate has been calculated taking into account the average results over all the receiver nodes.

6.1. Generation Size Variation

The size of the transmitted system of equations has been varied. The transmission matrix is constructed using a basic dense coding matrix and a redundancy block of the same size, also dense. $2 \cdot N$ packets are thus transmitted, where $N = K$ is the generation size and where the coefficients of all the packets are a value other than zero.

It is observed (in Table 1) that the reception quality, and consequently the reliability of the network as a whole, is improved as the generation size is increased. This is due to the fact that there is a greater mathematical probability of receiving any 32 packets from 64 than any 2 from 4. However, the computational complexity (see the execution time required) of solving the coefficient matrices on decoding is of the type $\theta(n^3)$, being n the size of the transmission matrix (see the increasing required execution time as an indirect indicator of the computation complexity in Table 1).

Table 1
Traffic loss rate variation with the generation size

Generation size	2	4	8	16	32
Traffic loss rate [%]	1.79	1.08	0.47	0.15	0.05
Execution time [s]	16.3	17.9	22.4	38.7	119

This effect, together with a larger generation size, involves transmission of more coefficients, and a greater header overhead in each packet means a compromise value must be found. In the following simulations, the generation size of $K = 8$ is taken as the compromise value.

6.2. Coding Density Variation

In order to analyze the effect of the coding density variation, a total number of $2 \cdot N$ (where $N = K$) packets are sent again, but varying the density of their equations, i.e., the number of non-zero coefficients they have, in both the basic transmission matrix block and the redundancy block appended to it. It should be observed that the case of a redundancy with density 1 is identical to repeating each packet twice.

Table 2
Traffic loss rate variation with coding density

Coding density	1	2	3	4
Traffic loss rate [%]	2.30	0.76	0.44	0.44
Coding density	5	6	7	8
Traffic loss rate [%]	0.47	0.49	0.48	0.47

From a density greater than 2, the variations are mainly due to statistical variance. It can be seen that repeating packets is less efficient than sending dense coding, as with dense coding any 8 packets are valid, but when packets are repeated at least one of the two copies of each packet is required. The dense coding will be taken as standard, as apart from being one of the most efficient options it is the most simple schema to be implemented as no special precautions need to be taken to generate the maximum rank matrix.

6.3. Redundancy Type Variation

To compare the different methods of sending redundancy appended after the transmission matrix, different simulations have been made, sending from 2 to 6 redundancy packets using different methods to generate the redundancy block.

A larger redundancy size clearly implies a higher quality level at receiver nodes. It is interesting to observe its variation according to the type of redundancy in each case. In all cases, dense redundancy is more effective. This is an effect of the broadcast nature of the wireless medium. If

Table 3
Traffic loss rate variation with different redundancy type

Type	Systematic	Density 2	Bresenham	Dense
Redundancy size $\times 2$ [%]	7.86	7.04	6.20	6.26
Redundancy size $\times 4$ [%]	6.06	4.61	3.98	2.87
Redundancy size $\times 6$ [%]	4.17	2.77	3.13	1.00

the packets are combined, they can potentially help more receiver nodes and the QoS of the network considerably improves. Medium density and Bresenham solutions yield intermediate results.

7. Comparison with Existing Algorithms

After having tested the behaviour of the proposed algorithm in a variety of configurations, the configuration considered optimum has been selected for comparison with classical schemas in order to improve the QoS of a wireless network. To make a fair comparison, both the classical solution and the new algorithm should generate the same average use of the wireless medium. The comparative scenario contemplates the following alternatives:

- **Traditional/classical solution**: sending standard packets, without confirmation. The medium is used once for each packet transmitted.
- **Repetition of packets**: two copies of each packet are sent, without any coding or collaboration of any type. The medium is used twice for each transmitted packet.
- **Source coding**: dense source coding, with dense redundancy and generation size equal to 8, is used. The medium is therefore used 16 times for each 8 packets.
- **Collaborative algorithm**: each "relay" node in Fig. 6 sends two collaboration packets with a dense redundancy (coding density 8). In total, the medium is used 8 + 4 (*dense redund.*) + 2 (*collab.*) + 2 (*collab.*) = 16 times per each 8 packets.

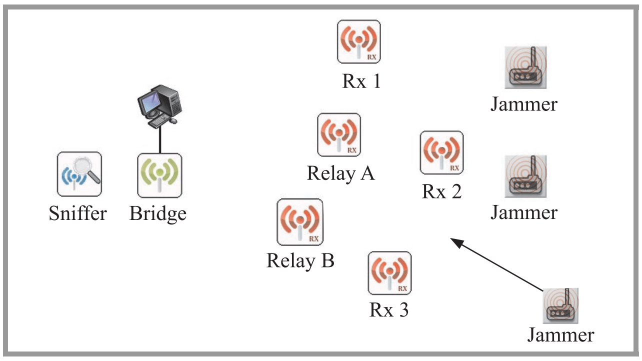

Fig. 6. Benchmark scenario with (moving) interfering nodes.

7.1. Temporal QoS Variation

An *interfering node* (see Fig. 6) is brought up to the network, which worsens the average quality of reception interfering packets sent by the bridge/transmitter or relay nodes. The interfering or jammer node travels a distance of 50 meters in the 60 seconds the simulation lasts. The reception quality of the R_{x3} node worsens considerably. This also entaile worsening the average network behaviour, and consequently its reliability as can be observed in Fig. 7.

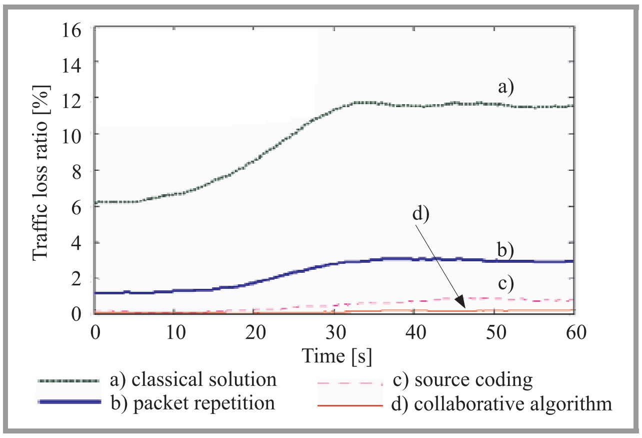

Fig. 7. Traffic Loss Ratio measurement (average among all nodes).

The traditional solution represents the result of a *gross* use of the wireless link, and thus has the worst quality of all. Repeating the packets notably increases this quality, but even higher quality (lower traffic loss rate ratio) is yielded by Galois finite field coefficient-based source coding. The mathematical bases for this improvement have already been presented, and they are clearly shown in this example. Among all the solutions, inter-node collaboration shows the best performance, as it is based on the network coding principle: one single help packet can serve to increase the information received at all nodes at the same time. The network QoS achieved with the proposed schema remains more stable than in the other cases, on the same scenario.

7.2. Interference Power Variation

The second comparison scenario proposed consists of maintaining the protocol configuration stable throughout each simulation run, varying the power of the interference nodes between each run. However, the PER (Packet Error Rate) value is represented on the horizontal axis, given that it is a more objective measurement than the *gross* quality of the wireless link. The environment thus becomes increasingly harsher. For fairer comparison, single repetition of packets (solution b in Fig. 8) is used, together with the protocol configuration that appears to achieve the best results (solution d), as both generate the same use of the wireless medium.

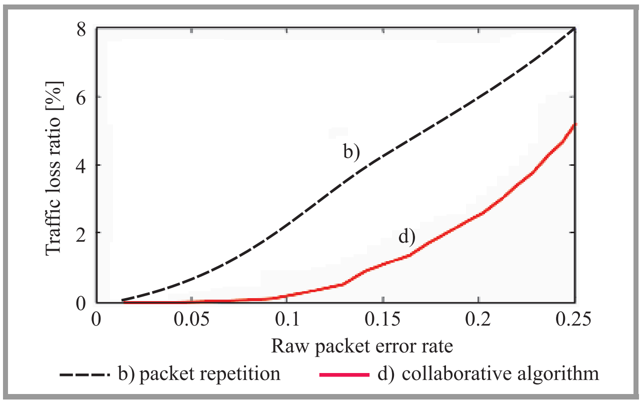

Fig. 8. Traffic loss ratio with raw packet error rate (average among all nodes).

Again, the result of combining the coding and collaboration among nodes using network coding mechanisms achieves clearly better performance than automatic packet repetition. This proofs that the proposed algorithm provides higher network reliability and reaches higher QoS and stable link behavior compared to classical solutions.

8. Conclusions and Future Lines of Work

We have analysed the theoretical bases of network coding and Galois arithmetic-based source coding. We proposed a simulation environment with a simple coding and inter-node collaboration algorithm to improve the network performance under harsh environments. We have tested the influence of several algorithm's main parameters, selecting the solution that optimizes the performance and network overall behavior.

Using the OPNET network simulator and its wireless simulation package, we have designed and implemented the proposed algorithm with a series of solutions and configurable parameters in order to increase the network reliability, demonstrating that the combination of the coding and collaboration mechanisms is the algorithm providing the most stable reliability, regardless of how harsh the medium is.

The presented algorithm increases the QoS and improves its immunity to changes or variations in the network. However, the collaboration mechanism is somewhat basic. We are currently working on more sophisticated mechanisms, based on adaptivity of the nodes collaborating in the network. The aim of all this work is to fully optimise the wireless link's efficiency, seeking the best possible QoS with the same bandwidth use.

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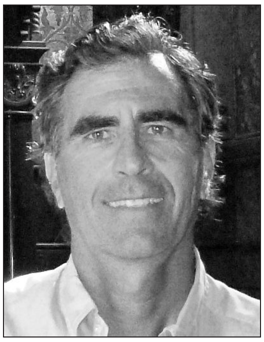

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