Impacts of Packet Scheduling and Packet Loss Distribution on FEC Performances: Observations and Recommendations*

Christoph NEUMANN
INRIA Rhône-Alpes, Planète Research Team
France
christoph.neumann@inrialpes.fr

Vincent ROCA
INRIA Rhône-Alpes, Planète Research Team
France
vincent.roca@inrialpes.fr

Aurélien FRANCILLON
INRIA Rhône-Alpes, Planète Research Team
France
aurelien.francillon@inrialpes.fr

David FURODET
STMicroelectronics, AST, Grenoble
France
david.furodet@st.com

ABSTRACT

Forward Error Correction (FEC) is commonly used for content broadcasting. The performance of the FEC codes largely vary, depending in particular on the code used and on the object size, and these parameters have already been studied in detail by the community. However the FEC performances are also largely dependent on the packet scheduling used during transmission and on the loss pattern introduced by the channel. Little attention has been devoted to these aspects so far. Therefore the present paper analyzes their impacts on the three FEC codes: LDGM Staircase, LDGM Triangle, two large block codes, and Reed-Solomon. Thanks to this analysis, we define several recommendations on how to best use these codes, depending on the test case and on the channel, which turns out to be of utmost importance.

Categories and Subject Descriptors

H.4 [**Information Systems Applications**]: Miscellaneous

General Terms

Performance, Reliability, Experimentation

Keywords

Multicast, Forward Error Correction (FEC), LDPC, Reed-Solomon, Loss Pattern, Packet Scheduling

*This work is supported by a research grant from STMicroelectronics.

1. INTRODUCTION

1.1 Context of the work

This work analyzes the impact of packet scheduling in the context of a content delivery systems like "IP Datacast" (IPDC) [13, 6] in DVB-H, the "Multimedia Broadcast/Multicast Service" (MBMS) [1] in 3GPP, or data broadcast to cars (e.g. [5]). These systems are characterized by the fact that there is no back channel, and therefore no repeat request mechanism can be used that would enable the source to adapt its transmission according to the feedback information sent by the receiver(s). The lack of feedback channel however enables an unlimited scalability in terms of number of receivers, who behave in a completely asynchronous way. Using a reliable multicast transmission protocol like ALC [9], along with the FLUTE [14] file delivery application, can turn out to be highly effective in this context [5].

Yet, in order to be efficient, these approaches largely rely on the use of a Forward Error Correction (FEC) scheme running at the application layer, within the ALC reliable transport protocol (we motivate the use of FEC in section 4.2). The channel is therefore a "packet erasure channel" and packets either arrive (with no error) or are lost (e.g. because of router congestion problems). After an FEC encoding of the content, redundant data is transmitted along with the original data. Thanks to this redundancy, up to a certain number of missing packets can be recovered at the receiver. The great advantage of using FEC with multicast or broadcast transmissions is that the same parity packet can recover different lost packets at different receivers.

Since we only consider file delivery applications in this work, the transmission latency has little importance, which would not be true with streaming applications. Therefore we will not consider the potential impacts of FEC codes and transmission scheme on the decoding latency at a receiver.

1.2 Goals of the work

The performance of the FEC code is largely impacted by the transmission scheduling. For instance, sending all source packets first and then parity packets does not necessarily yield the same efficiency as sending the packets in random order. The packet loss distribution observed by a receiver also largely impacts the decoding performances and a given transmission scheme may yield good results for a specific loss distribution and yield catastrophic results in other circumstances. This work analyzes the impact of packet scheduling and loss behaviors on the performances of three FEC codes: Reed-Solomon, LDGM Staircase and LDGM Triangle. Thanks to this analysis, we define several recommendations on how to best use these codes, which turns out to be of utmost practical importance. For instance it enables to optimize the FLUTE session to a specific download environment, or on the opposite, to find transmission schemes that will behave correctly (but may be not optimally) in a wide set of different environments.

In this work we do not consider FEC codes who are known to be covered by IPRs and for which no public domain, open-source, implementation exists. In particular we will not consider Tornado^{TM} and Raptor^{TM} codes [4].

The remainder of the paper is organized as follows: we first introduce the three FEC codes; section 3 explains and motivates the modeling method we used; section 4 presents and analyzes the performance of several transmission schemes while section 5 does the same with a reception model; finally section 6 explains how to use these results in practice, then we conclude.

2. INTRODUCTION TO RSE, LDGM STAIRCASE AND LDGM TRIANGLE CODES

2.1 Terminology

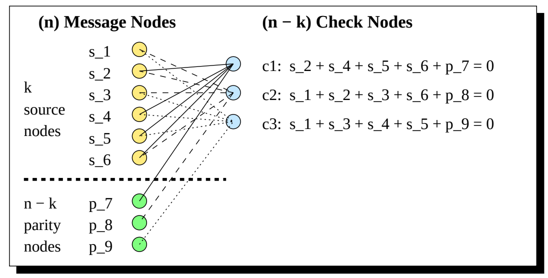
Forward Error Correction encoding of an object produces redundant data. Thanks to this redundancy, up to a certain number of missing packets can be recovered at the receiver. More precisely k *source packets* (A.K.A. data packets) are encoded into n packets (A.K.A. encoding packets). The additional $n-k$ packets are called *parity packets* (A.K.A. FEC or redundancy packets). A receiver can then recover the k source packets provided it receives any k packets (or a little bit more than k with LDGM/LDPC codes) out of the n possible. The *FEC expansion ratio* is the $\frac{n}{k}$ ratio and it defines the amount of parity packets produced. It is the inverse of the *code rate* (i.e. $\frac{k}{n}$). In the present paper we will only consider the FEC expansion ratio terminology.

2.2 RSE Code

The Reed-Solomon erasure code (RSE) is one of the most popular FEC codes. RSE is intrinsically limited by the Galois Field it uses [15]. A typical example is GF(2^8) where $n \leq 256$. With one kilobyte packets, a FEC codec producing as many parity packets as data packets (i.e. $n = 2k$) operates on blocks of size 128 kilobytes at most, and all files exceeding this threshold must be segmented into several blocks, which reduces the global packet erasure recovery efficiency (e.g. if B blocks are required, a given parity packet has a probability $1/B$ to recover a given erasure, and $B = 1$ is then the optimal solution). This phenomenon is known as the "Coupon Collector Problem" [3]. Another drawback is a huge encoding/decoding time with large (k, n) values, which is the reason why GF(2^8) is preferred to GF(2^{16}) in spite of its limitations on the block size. Yet RSE is optimal because a receiver can recover erasures as soon as it has received *exactly* k packets out of n for a given block. A code with this property is called a "Minimum Distance Separation" (MDS) code.

2.3 LDGM Codes

We now consider another class of FEC codes that completely departs from RSE: Low Density Generator Matrix (LDGM) codes, that are variants of the well known LDPC codes introduced by Gallager in the 1960s [7].



(a) Bipartite graph

$$
[\, H_1 \,|\, Id_3 \,] =
\begin{array}{c}
\begin{array}{cccccc|ccc} s_1 & & \dots & & & s_6 & p_7 & \dots & p_9 \end{array} \\
\left(\begin{array}{cccccc|ccc}
0 & 1 & 0 & 1 & 1 & 1 & 1 & 0 & 0 \\
1 & 1 & 1 & 0 & 0 & 1 & 0 & 1 & 0 \\
1 & 0 & 1 & 1 & 1 & 0 & 0 & 0 & 1
\end{array}\right)
\end{array}
\begin{array}{c} \\ c_1 \\ c_2 \\ c_3 \end{array}
$$

(b) H matrix

Figure 1: A regular bipartite graph and its associated parity check matrix for LDGM.

2.3.1 Principles

LDGM codes rely on a bipartite graph between left nodes, called *message nodes*, and right nodes, called *check nodes* (A.K.A. constraint nodes). The k source packets form the first k message nodes, while the parity packets form the remaining $n-k$ message nodes. The upper part of this graph is built following an appropriate left and right degree distribution (in our work the left degree is 3). The lower part of this graph follows other rules that depend on the variant of LDGM considered (e.g. with LDGM, figure 1 (a), there is a bijection between parity and check nodes). This graph creates a system of $n-k$ linear equations (one per check node) of n variables (source and parity packets).

A dual representation consists in building a parity check matrix, H. With LDGM, this matrix is the concatenation of matrix H_1 and an identity matrix I_{n-k}. There is a 1 in the $\{i; j\}$ entry of matrix H each time there is an edge between message node j and check node i in the associated bipartite graph.

Thanks to this structure, encoding is extremely fast: each parity packet is equal to the sum of all source packets in the associated equation. For instance, packet p_7 is equal to the sum: $s_2 \oplus s_4 \oplus s_5 \oplus s_6$. Besides LDPC/LDGM codes can operate on *very large blocks*: several hundreds of MBytes are common. However LDGM is not an MDS code and it introduces a decoding inefficiency: $inef\_ratio * k$ packets, with $inef\_ratio \geq 1$, must be received for decoding to be successful. The $inef\_ratio$, experimentally evaluated, is therefore a key performance metric.

2.3.2 Iterative Decoding Algorithm

With LDGM, there is no way to know in advance how many packets must be received before decoding is successful (LDGM is not an MDS code). Decoding is performed step by step, after each packet arrival, and may be stopped at any time.

The algorithm is simple: we have a set of $n-k$ linear equations of n variables (source and parity packets). As such this system cannot be solved and we need to receive packets from the network. Each non duplicated incoming packet contains the value of the associated variable, so we replace this variable in all linear equations in which it appears. If one of the equations has only one remaining unknown variable, then its value is that of the constant term. We then replace this variable by its value in all remaining equations and reiterate, recursively. As we approach the end of decoding, incoming packets tend to trigger the decoding of several packets, until all of the k source packets have been recovered.

2.3.3 LDGM Staircase Code

This trivial variant, suggested in [10], only differs from LDGM by the fact that the I_{n-k} matrix is replaced by a "staircase matrix" of the same size. This small variation affects neither encoding, which remains a simple and highly efficient process, nor decoding, which follows the same algorithm. But this simple variation *largely improves the FEC code efficiency* as we have shown in [16].

2.3.4 LDGM Triangle Code

In this variant of LDGM Staircase, the triangle beneath the staircase diagonal is now filled, following an appropriate rule [16]. This rule adds a "progressive" dependency between check nodes, as shown in figure 2. This variation further increases performance in some situations, while keeping encoding highly efficient (even if a bit slower since there are more "1"s per row). Here also decoding follows the same iterative algorithm.

Interested readers are invited to refer to [16]. An open source, GNU/LGPL implementation of these codes is also available at [12].

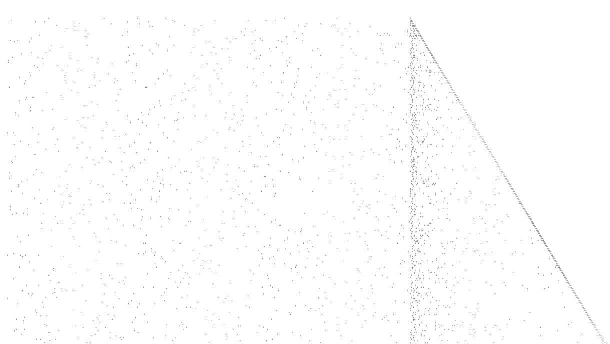


Figure 2: Parity check matrix (H) for LDGM Triangle (k=400, n=600).

3. MODELING OF THE WHOLE SYSTEM

3.1 The Three Models

Let's imagine that a server wants to broadcast a big file using a large scale content delivery system using FLUTE. The content is first FEC encoded which produces additional parity packets (we motivate the use of FEC in section 4.2). The sender must now decide in which order source and parity packets will be sent, and this choice will largely impact the whole system performances as we will see later. This packet scheduling constitutes the transmission model.

The channel is characterized by a packet loss distribution. It may be a lossy channel with long bursts of packet erasures, or it may be a channel where losses are completely independent from one another. This packet loss distribution constitutes the loss model.

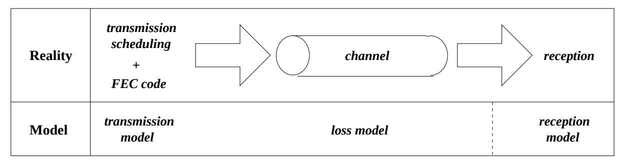


Figure 3: Modeling methods.

These two transmission and loss models together characterize a reception behavior at a receiver (Figure 3). In case of broadcast applications, there is no reason that different receivers experience the same loss model, so the reception behavior is receiver dependent. This is the approach that will be used for most experimental evaluations (section 4).

The reception behavior may also be modeled the other way round, by providing a reception model. This model defines which packets are received (and when) by a receiver. This approach can be complementary to the use of the transmission and loss models, for instance to study the FEC code performances in controlled situations. This is the approach that will be used in section 5.

3.2 The Channel Model

Finding an appropriate error model for a given channel is a complex task, especially with wireless networks where it is difficult to take into account all parameters (e.g. channel fading, reflections, refractions, diffractions, Doppler effects). Yet in this work we are only interested in a packet loss model (rather than a bit error model) that provides a high level abstraction of the channel parameters. The well known two state Markov model (A.K.A. Gilbert model) is such a simplified loss model, and it is widely used in the literature [2, 17].

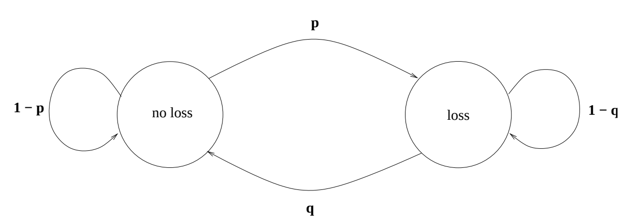


Figure 4: Two state Markov loss model.

The model is composed of two states: the *no-loss* state where no packet loss occurs, and the *loss* state where packets are lost (figure 4). p indicates the probability to go from no-loss state to loss state, and q from loss state to no-loss state. Having p and q we can calculate the global packet error probability [2]:

$$p_{global} = \frac{p}{p+q}$$

This probability is represented as a 3D-graph in figure 5. [1]

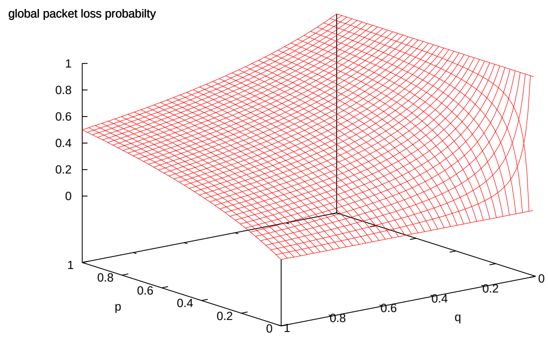


Figure 5: Global loss probability.

Depending on the channel modeled, the p and q values will differ. For a given channel it may be possible to determine p and q using packet loss traces. For instance, this has been done in [8] for traces coming from an GSM channel, and in [17] for traces coming from end-to-end connections in the Internet.

The Gilbert model also covers some specific loss behaviors that we want to emphasize:

- No loss: This perfect channel corresponds to $p = 0$.
- Independent and Identically Distributed (IID) losses (A.K.A. the Bernoulli model): This memoryless channel corresponds to $q = 1 - p$.

We are aware that the Gilbert model has some shortcomings in error modeling accuracy [8, 17]. We believe that it is however sufficient for our work since it already covers a very large set of loss behaviors. Moreover we took care to perform experiments with a very large set of p and q values (14×14 grid) in order to cover as many channel behaviors as possible. Other more complex models (e.g. the n-state Markov models), that may be required for specific channels, will be considered in future works.

When is Decoding Impossible?

In our work we want to know, given a certain FEC code, how many losses a receiver can support. With the Gilbert model we can calculate the maximum number of packet losses supported by any FEC code. Let's consider a FEC code producing $n - k$ parity packets from k source packets. We then transmit $n_{sent} \leq n$ packets over the network (sending all packets is not mandatory, see section 5). The number of packets actually received is equal to:

$$n_{received} = n_{sent} * (1 - p_{global}) \quad (1)$$

$n_{received}$ must be at least equal to $inef\_ratio * k$ (remember that LDGM codes are not MDS) for decoding to be successful. When $n_{received} = inef\_ratio * k$ we have:

$$q = \frac{-p * inef\_ratio}{inef\_ratio - \frac{n_{sent}}{k}}$$

These limits are shown in figure 6 for FEC expansion ratios 1.5 and 2.5, and assuming $inef\_ratio = 1$ (which is a lower bound for $inef\_ratio$). This figure shows that several areas of the (p, q) parameter space cannot enable a receiver to decode the object. This is not a flaw of the FEC code, it's merely a fundamental limitation.

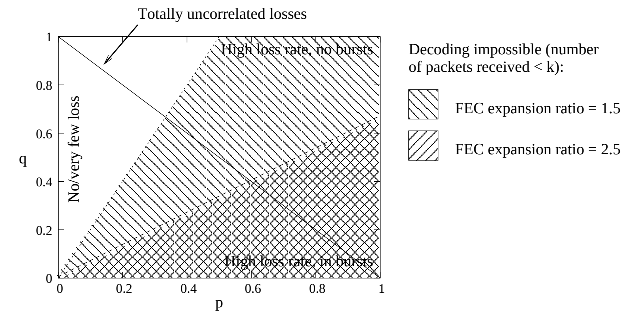


Figure 6: Loss limits.

4. SIMULATION RESULTS WITH SIX TRANSMISSION MODELS

4.1 Methodology

The methodology used to conduct performance tests is the following. We considered the RSE, LDGM Staircase and LDGM Triangle codes, and two typical FEC expansion ratios: 1.5 and 2.5 (they correspond to code rates 2/3 and 2/5 respectively). The object is composed of 20000 packets (i.e. $k = 20000$ with LDGM-* codes).

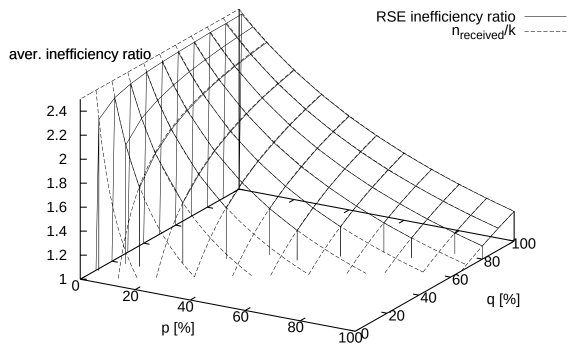
For RSE we use a Galois Field GF(2^8), i.e. $n \leq 256$. We use the maximum value for n (= 256) and derive the maximum block size supported. The object is then segmented into several blocks using the blocking algorithm described in [?].

The performance metric used is the average inefficiency ratio: $inef\_ratio = \frac{n_{necessary\_for\_decoding}}{k}$, which is the total number of packets received when decoding completes, divided by the number of source packets. Of course the optimal value is 1.0, and the higher this $inef\_ratio$, the more packets are needed in excess to k for decoding, which is not good[2].

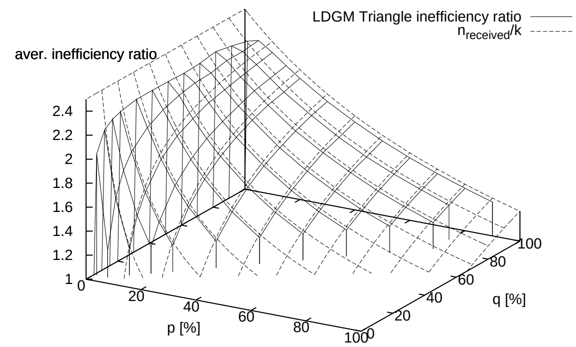
We considered six different *transmission models*. Although this is not an exhaustive study, we believe that these transmission schemes already cover a large set of possibilities and give good insights on the overall performances.

[1]This graph is helpful in order to have an idea on the p_{global} for the graphs in section 4. Note that for better visibility this figure is rotated compared to the figures in section 4.

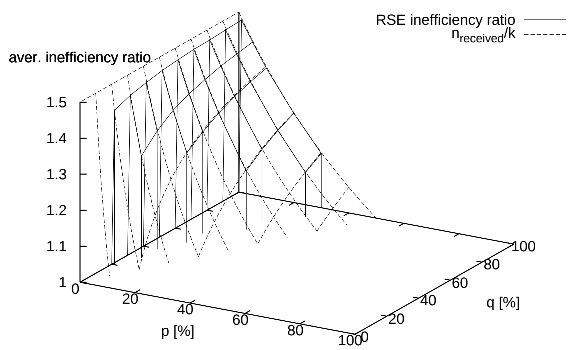
[2]With large block LDGM codes, an inefficiency ratio greater than 1.0 is caused by the non MDS nature of these codes, whereas with RSE, which is an MDS code, this is caused by the coupon collector problem (section 2.2). More information can be found in [16].



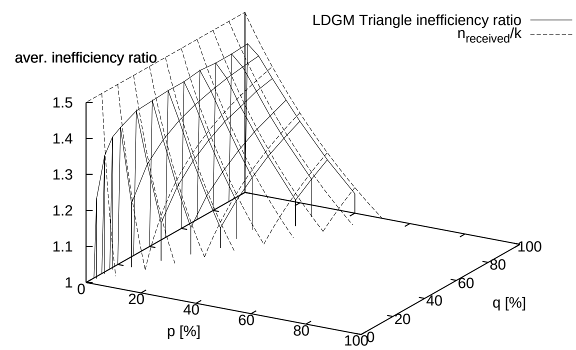
(a) RSE and a FEC expansion ratio of 2.5.



(b) LDGM Triangle and a FEC expansion ratio of 2.5.



(c) RSE and a FEC expansion ratio of 1.5.



(d) LDGM Triangle and a FEC expansion ratio of 1.5.

Figure 7: Tx_model_1

For each transmission model and FEC code, we study the impacts of the channel by varying the p and q probabilities in $[0; 1]$ (14×14 values are considered for $(p; q)$). Each point in the resulting 3-D graphs is the average $inef\_ratio$ value over 100 simulations. Yet if decoding fails for any of the 100 tests, we do not plot any point. This strict strategy enables us to better highlight the areas where the decoding probability is not acceptable. The numerical results of our simulation for the most interesting transmission schemes and codes are reported in the extended version of this paper [11].

For comparison purposes, we sometimes plot an additional curve, $\frac{n_{received}}{k}$, which corresponds to the total number of packets received (even after decoding stopped) divided by the number of source packets. This is the maximum value that the inefficiency ratio can achieve. It also gives an idea on the number of packets that a receiver may still receive after decoding ($n_{received} - n_{necessary\_for\_decoding}$). When $\frac{n_{received}}{k} = 1$ the curve corresponds to the limits described in section 3.2 and figure 6. Any additional packet loss will necessarily lead decoding to fail.

4.2 Why is FEC Needed?

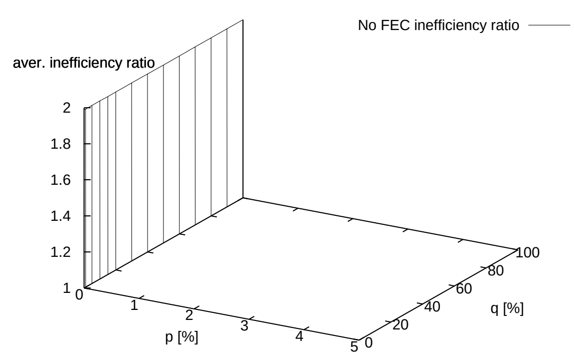
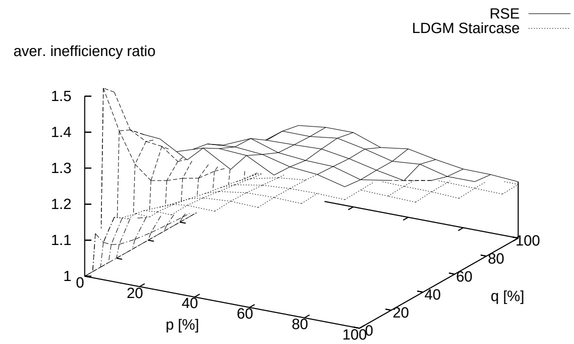
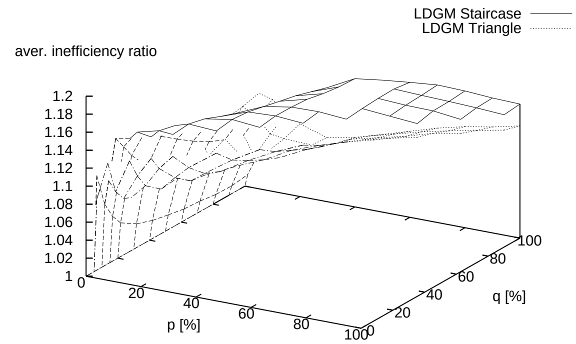


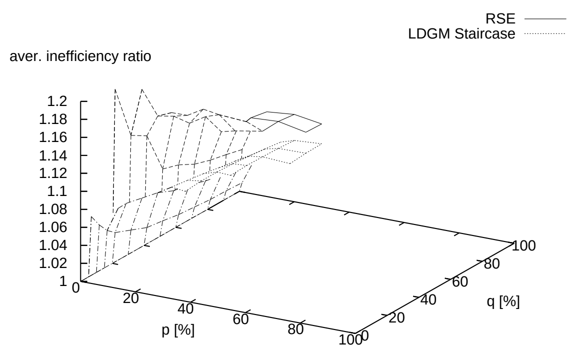
Figure 11: Performances without FEC but 2 repetitions.



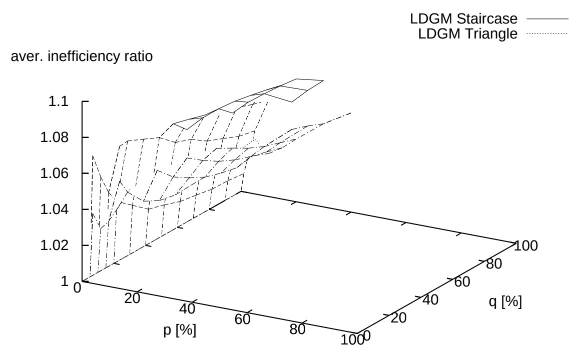
(a) RSE versus LDGM Staircase and a FEC expansion ratio of 2.5.



(b) LDGM Triangle vs. Staircase and a FEC expansion ratio of 2.5.



(c) RSE versus LDGM Staircase and a FEC expansion ratio of 1.5.



(d) LDGM Triangle vs. Staircase and a FEC expansion ratio of 1.5.

Figure 8: Tx_model_2.

To motivate the use of FEC, we did a small test. Rather than using FEC to recover losses, a sender may decide to transmit each packet x times (remember that no back channel is available and therefore repeat request mechanisms are not an option in the context of this work). In figure 11 $x = 2$ and packets are sent in a random order. It shows that decoding is only possible with $p = 0$, and the average inefficiency ratio is then near 2.0 which means that the receiver waits almost systematically the end of the transmission to reconstruct the object. For all $p > 0$ at least one experiment failed, and therefore no inefficiency ratio is shown, as explained in section 4.1. This test highlights the *poor performances when using repetition only instead of FEC* for content broadcasting.

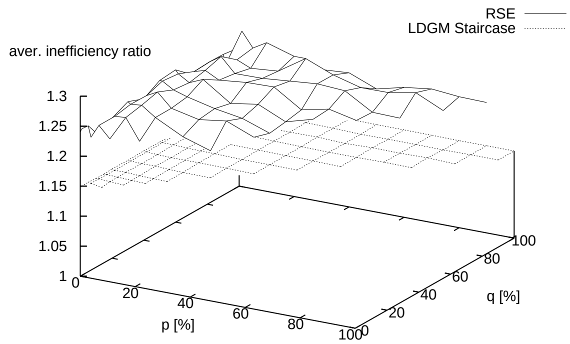
4.3 Tx_model_1: Send Source Packets Sequentially, Then Parity Packets

This scheme consists in sending all source packets sequentially, and then all parity packets, also sequentially. The results are shown in figure 7 (the LDGM Staircase curves are not shown since results are similar to LDGM Triangle).

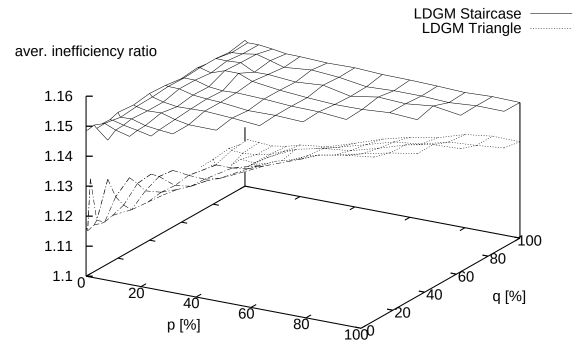
We notice a first obvious result: without loss ($p = 0$) the inefficiency ratio is 1.0 with all codes. Indeed, all source packets are received and the receiver does not need any of the following parity packets. We'll see that other transmission schemes do not necessarily have this good property.

With losses (in burst or not) all codes show a similar behavior. The inefficiency ratio curve is very close to the $\frac{n_{received}}{k}$ curve for nearly all values of p and q. It means that the receiver always needs to wait almost the end of transmission to reconstruct the object. With RSE this is not surprising since the object is segmented in blocks: if a source packet of the last block is lost, the receiver needs to wait the end of the transmission, when the associated parity packets will be sent.

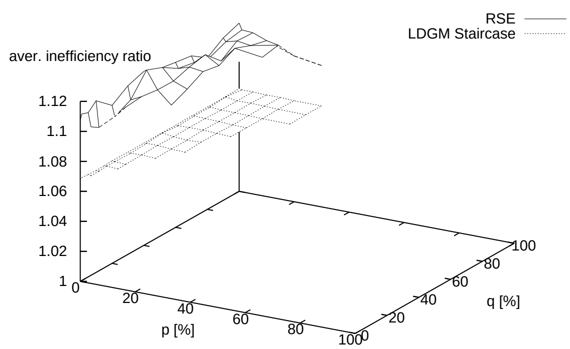
With LDGM codes this behavior is more surprising. These large block FEC codes encode the whole object directly, and parity packets are created using source packets coming from



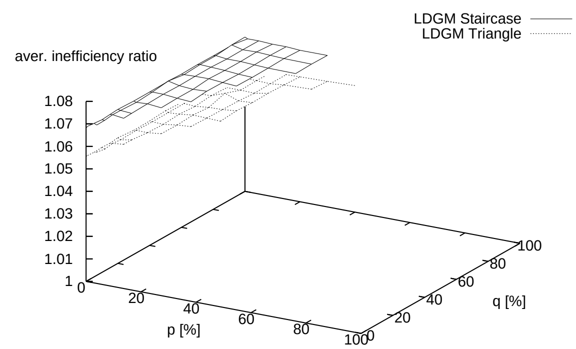
(a) RSE versus LDGM Staircase and a FEC expansion ratio of 2.5.



(b) LDGM Triangle vs. Staircase and a FEC expansion ratio of 2.5.



(c) RSE versus LDGM Staircase and a FEC expansion ratio of 1.5.



(d) LDGM Triangle vs. Staircase and a FEC expansion ratio of 1.5.

Figure 9: Tx_model_4

different parts of the content. However each parity packet depends on the previous one (because of the staircase in the parity check matrix, also present in LDGM Triangle codes). This dependency negatively impacts decoding performance when several sequential parity packets get lost.

Finally, RSE covers a smaller area (when $z - axis > 0$) than LDGM-* codes, which indicates that its erasure recovery capabilities are more limited. This is especially true for long packet loss bursts (small q). This is easy to understand: since packets are sent sequentially, and a single long burst results in the loss of a large number of packets of the same block. Decoding this block becomes difficult, especially with a small FEC expansion ratio (e.g. 1.5).

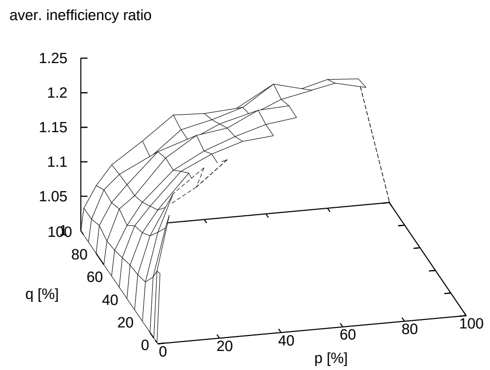
For all these reasons this transmission model is definitively bad, which was relatively foreseeable.

4.4 Tx_model_2: Send Source Packets Sequentially, Then Parity Packets Randomly

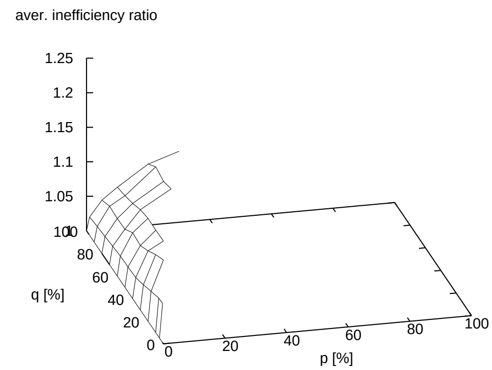
One idea to counteract the bad results of Tx_model_1 is to transmit the parity packets in a random order rather than sequentially. For RSE the advantage is clear: parity packets of the last few blocks can be transmitted earlier, so the receiver has on average less to wait before receiving parity packets for any block. This is confirmed in figures 8(a) and 8(c). The inefficiency ratio is much better than with Tx_model_1 and moreover this ratio is relatively constant.

LDGM Triangle codes exhibit relatively good performances and largely outperform RSE codes (figures 8(b) and 8(d)). Yet the LDGM Staircase 3D-curve has a hole around $p = 50$ and $q = 70$ (Figure 8(b)), where exactly one test failed. This is not acceptable in practice, and therefore Tx_model_2 is not recommended with LDGM Staircase and higher loss ratios. Conversely LDGM Staircase may be used with small loss ratios where the code yields very good results.

This transmission model confirms that with LDGM-* codes, parity packets should not be sent sequentially but randomly, to prevent the loss of sequences of parity packets. All the following tests will confirm this observation.



(a) RSE and a FEC expansion ratio of 2.5 (NB: the figure is rotated for better visibility).



(b) RSE and a FEC expansion ratio of 1.5 (NB: the figure is rotated for better visibility).

Figure 10: Tx_model_5

4.5 Tx_model_3: Send Parity Packets Sequentially, Then Source Packets Randomly

We now consider the case dual to Tx_model_2. We first transmit all parity packets sequentially. We can then transmit the source packets either sequentially or randomly. We only consider the latter case in this paper since tests (not included in this paper) have shown that sending the source packets sequentially yields uninteresting results (it makes no difference with LDGM-* codes, and we experience a worse behavior with RSE).

So called *non-systematic* codes, are codes that can decode using only parity packets (no source packet). RSE can be used as a non-systematic code if the number parity packets is high enough (i.e. $n - k \geq k$). This is not the case with LDGM Triangle and Staircase who need at least a small number of data packets to start decoding. Our tests have shown that with $p = 0$ the LDGM-* codes need exactly one source packet to decode the content, and therefore the inefficiency ratio is ≈ 1.5 for a FEC expansion ratio of 2.5.

With RSE, the receiver needs 29903 packets, that is to say decoding is possible when k packets of the last block have been received. Therefore the inefficiency ratio is ≈ 1.5 for FEC expansion ratios of 2.5.

Otherwise performances are not that interesting, and that transmission scheme may only be interesting for some specific loss patterns. Because of that and for paper length considerations, the figures are not included in this paper but in the extended version of this paper [11].

4.6 Tx_model_4: Send Everything Randomly

In this transmission model, source and parity packets are sent in a fully random order.

Results are shown in figure 9. RSE offers the worst performances with an inefficiency ratio around 1.25. LDGM Staircase performs better and offers a ratio of 1.15. Finally LDGM Triangle yields the best results with ratios going from 1.12 to 1.14.

For the RSE and LDGM Staircase codes, the performances are relatively stable, independently of the packet loss behavior. This is not true with LDGM Triangle codes, that show better results with smaller p_{global}. This observation can generally be made when LDGM Triangle sends it parity packets randomly: it achieves better inefficiency ratios with smaller p_{global}. On the opposite, LDGM Staircase codes are not sensitive to p_{global} in this case.

4.7 Tx_model_5: Interleaving (RSE only)

Packet interleaving is a commonly used solution with small block FEC codes like RSE to increase their robustness against packet erasure bursts. The idea is to spread the transmission of one block over an interval that is longer that the loss burst duration. The maximum distance between two packet transmissions of the same block is achieved by sending successively one packet of each block, until reaching the last block, and then continuing with the following packet of each block, and so on.

With LDGM codes this interleaving scheme is not feasible since there is only one block. What we call interleaving for LDGM consists in sending successively one source packet and $\frac{n}{k}$ parity packet (because the FEC expansion ratio is not necessarily equal to 2), and so one.

Results are shown in figure 10. As suspected, results are excellent with RSE, actually the best ones compared to all other transmission schemes. The packets are optimally aligned using interleaving, and for all loss patterns this solution give optimal results. This is not a surprise since interleaving has been intensively used along with RSE codes.

4.8 Tx_model_6: Send Randomly a Few Source Packets Plus Parity Packets

In this transmission model we transmit only a few source packets in addition to all parity packets: we first pick randomly 20% source packets and schedule them randomly with all parity packets. This transmission model requires that the FEC expansion ratio be high enough (otherwise less than k packets will be received), and we chose 2.5.

The results are shown in figure 12. If all codes have constant performances, LDGM Staircase largely outperform other codes. Note that the fact that LDGM Staircase performs better than Triangle is rather unusual.

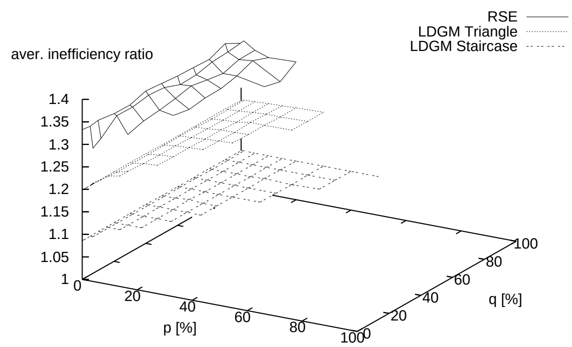


Figure 12: Tx_model_6 with LDGM Triangle, LDGM Staircase and RSE and a FEC expansion ratio of 2.5.

5. SIMULATION RESULTS WITH A RECEPTION MODEL

In this section we directly specify a reception model, without any consideration for the transmission and loss models that may generate it, i.e. there is no explicit transmission and loss model (no p and q) in that case. The goal is to better analyze an FEC code performances in a completely controlled environment.

5.1 Rx_model_1: Receive a Few Source Packets, Then Parity Packets Randomly

Section 4.8 has shown that sending only a few source packets along with the parity packets may be interesting. In this reception model, we further study this phenomenon. The difference with Tx_model_6 is that we now guarantee that these source packets arrive and are used for decoding. To do so the receiver first gets the source packets, and then, randomly, all parity packets.

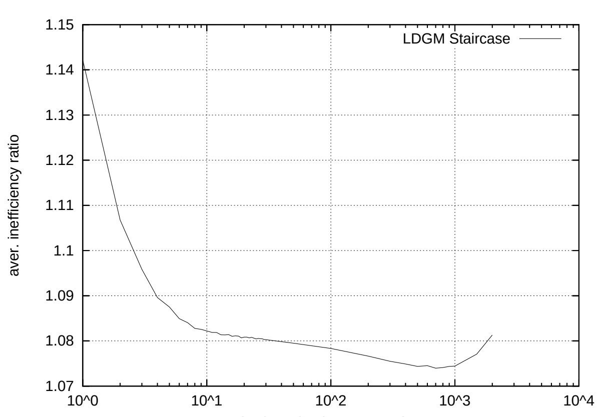


Figure 13: Rx_model_1 with LDGM Staircase.

We only consider LDGM Staircase and FEC expansion ratio of 2.5, since section 4.8 has shown that it yields the best results. We analyze the performance of the FEC codes as a function of the number of source packets received. Figure 13 shows that excellent performances are achieved when around 400 to 1000 source packets received. This is a very small number of packets compared to the object size ($k = 20000$ packets). Receiving more (or fewer) packets will degrade performances! To the best of our knowledge, we never saw such results mentioned in the literature, and we think this promising (and surprising) result deserves some complementary studies.

6. DISCUSSIONS AND RECOMMENDATIONS

6.1 Summary of the Results

We now discuss the previous results and draw some recommendations. Regarding FEC codes performances:

- RSE: RSE should always be used along with Interleaving (which is not a new result). Yet, even in that case, RSE shows lower performances than the best LDGM codes, for instance LDGM Triangle with Tx_model_2 or Tx_model_4.

- LDGM Triangle and LDGM Staircase: In most cases (with some exceptions though, like with Tx_model_6), *LDGM Triangle yields better results.* For both codes it makes no difference whether source packets are sent randomly or sequentially. However sending parity packets sequentially *must to be avoided*, since loss bursts will severely degrade performances. It can also be observed that LDGM Triangle is often more sensitive to packet losses, probably because of higher dependencies between parity packets.

Regarding the transmission models:

- Tx_model_1 and Tx_model_3 are of little or no interest in all cases.

- *Packet interleaving (Tx_model_5) is unavoidable with RSE, no matter the loss model.*

- The following models using totally or partially random transmissions have good results: sending sequentially source packets then randomly parity packets (Tx_model_2), sending everything randomly (Tx_model_4), and sending randomly a few source packets plus parity packets (Tx_model_6). More precisely:

  - Tx_model_2 is the preferred scheme for LDGM Triangle and LDGM Staircase if we want good results when *there are few packet losses.* With LDGM Staircase great care has to be taken of possible decoding failures with higher loss ratios.
  - Tx_model_4 (along with LDGM Triangle) and Tx_model_6 (along with LDGM Staircase) are the schemes that are the less dependent on the loss distribution. Therefore they are the *preferred solutions when the loss model is unknown, but Tx_model_4 will perform better with very high loss rates.*

6.2 In Practice

The previous results allow us to select an appropriate transmission scheme for a given use case. If the channel characteristics are unknown, LDGM Triangle and Tx_model_4 are excellent choices. If on the opposite the channel features (and its loss model) are known, we can compare all the FEC performances for all transmission schemes around the $(p; q)$ point, and make a choice.

The transmission can further be optimized by adapting n_{sent}, the number of packets actually sent. Remember that $n_{received} - n_{necessary\_for\_decoding}$ is the number of packets the receiver receives after decoding has finished. Our goal is now to have:

$$n_{received} = n_{necessary\_for\_decoding} + \epsilon \quad (2)$$

($n_{received}$ should be a little bit greater since some tolerance is required [3]). By doing so, the number of packets received will be very close to the number of packets that are actually needed by a given receiver to successfully decode the object. This is achieved by reducing n_{sent} (Equation 1), that is to say by stopping transmissions after n_{sent} packets, without changing the scheduling. The last $n - n_{sent}$ packets will never been sent.

Note that selecting a smaller FEC expansion ratio to reduce the number of packets sent in front of a certain loss model, is not always feasible, especially when there are only a small number of predefined ratios possible (e.g. because the FEC codec has been optimized for some FEC expansion ratios). For instance, if only 1.5 and 2.5 ratios are possible, with RSE and Tx_model_5, a FEC expansion ratio of 2.5 is needed when $p = 40\%$. With 1.5, decoding would fail.

Besides we omit an essential performance metric: *encoding and decoding speed*. From this point of view, *LDGM codes are an order of magnitude faster than RSE codes*, as shown in our previous work [16]. This can be an essential criteria of choice when broadcasting big objects, for which the encoding and decoding times may be non negligible.

We now detail how to proceed in two specific use cases.

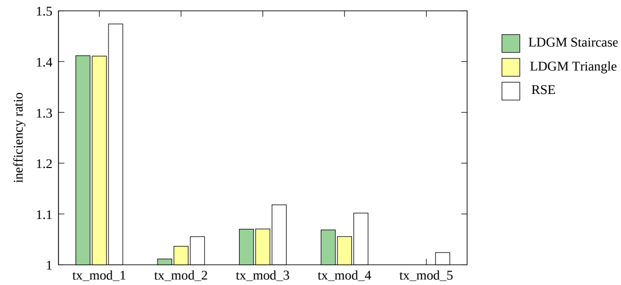
6.2.1 Homogeneous Receiver(s) and Known Channel

If there is only one receiver, or a set of receivers behind the same channel, the loss model may be identified and the p and q values determined (e.g. this can be the case when a content is broadcast in a cooperate network, or in some specific environments). Thanks to the p and q values, we can identify the best FEC code and transmission model for *this* environment.

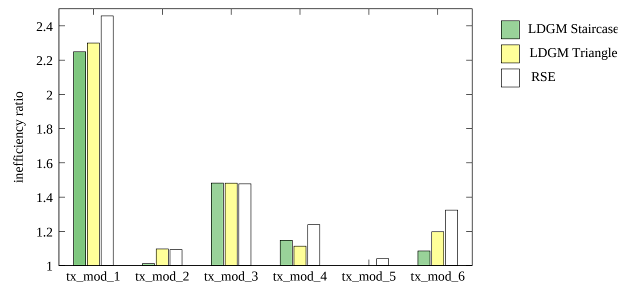
Once they have been chosen, we can adapt n_{sent}. The inefficiency ratio is given by the associated curve. Then, according to formulas 1 and 2, the optimal n_{sent} is achieved when:

$$n_{sent} = \frac{n_{necessary\_for\_decoding}}{1 - p_{global}} \quad (3)$$

Let's consider the following example: a 50 MByte object is sent to a single receiver from Amhers Massachusetts to Los Angeles. [17] traced the loss behavior on this link and obtained $p = 0.0109$ and $q = 0.7915$ ($p_{global} = 0.0135$). Figure 14 shows the performances of the various FEC Code and transmission models, for the FEC expansion ratios 1.5 and 2.5. We clearly see that *Tx_model_2 with LDGM Staircase and a FEC expansion ratio of* 1.5 give the best results ($inef\_ratio \approx 1.011$).



(a) FEC expansion ratio = 1.5



(b) FEC expansion ratio = 2.5

Figure 14: Example: Inefficiency ratios on the link Amher Massachusetts to Los Angeles.

We now calculate the optimal n_{sent} (formula 3):

$$n_{sent} = \frac{1.011 * 50MBytes}{1 - 0.0135} \approx 51.24MBytes$$

With 1024 byte packet payloads, it means that the sender may stop transmitting after $\approx$ 50041 packets. In order to add some tolerance to this result we can decide to set n_{sent} to 55000 packets. This is significantly less than the $n = 73243$ packets that would have been sent otherwise, while preserving transmission reliability.

6.2.2 Heterogeneous Receivers and/or Unknown Channel

Imagine now that an object is broadcast over a wireless channel. All receivers observe different packet loss distributions, depending on many external parameters (e.g. movement, obstacles, distance to the source).

If the loss pattern of each receiver is known we may proceed as in the previous section, by choosing the (FEC code; transmission scheme; FEC expansion ratio) tuple that yields the best results for all receivers.

If the loss pattern of one or several receivers is unknown, which will probably be the general case, we need a universal scheme that shows the best possible performances in all situations.

Sending everything randomly (Tx_model_4) and sending only a few parity packets (Tx_model_6) are the schemes that

[3]To have an idea on the confidence interval around the average inefficiency ratios for the FEC codes used in this paper please read [16]

are the less dependent of the loss pattern, and are therefore the preferred solutions in this context for LDGM codes. With RSE interleaving should be used, but performances will be lower.

As before n_{sent} can be adapted. We proceed as before with the main difference that we now have to look at the borders of the inefficiency ratio curve. The aim is to get $\frac{n_{received}}{k}$ as near as possible to the inefficiency ratio curve. This can be done by choosing some values of p and q that are located in the border region of the inefficiency ratio curve, evaluate the inefficiency ratio, and then proceed like in section 6.2.1.

7. CONCLUSIONS

This work investigated the impacts of packet scheduling and packet loss distributions on FEC performances, for content broadcasting applications. This work should be of great help for FLUTE-based file broadcasting systems, when there is no backward channel and where transmission reliability is achieved through the massive use of FEC and complementary techniques (e.g. cyclic transmissions within a carousel). Since we only consider file delivery systems, the transmission latency has little importance and large block FEC codes, like LDPC/LDGM codes, are good candidates.

The experiments carried out provide good insights on which (FEC code; FEC expansion ratio; packet transmission scheduling) tuples yield the best results for a given channel. Non surprisingly, experiments have shown that interleaving should always be used with RSE, no matter the loss pattern (which is not a new result). *Yet LDGM Staircase and Triangle codes usually perform significantly better than RSE*. For environments where the channel (and its packet loss distribution) is *unknown*, which is probably the general case, LDGM codes require a random transmission scheme: usually either (LDGM Triangle; Tx_model_4) or (LDGM Staircase; Tx_model_6). For environments where the loss distribution is *known*, our work helps to identify the best (FEC code; transmission scheme; FEC expansion ratio) tuple. Our results are therefore of utmost practical importance for optimizing broadcasting systems.

Of course these results omit an essential performance metric: encoding and decoding speed. From this point of view, *LDGM codes are an order of magnitude faster* than RSE codes, as shown in our previous work [16]. This can be an essential criteria of choice when broadcasting very large objects, for which the encoding and decoding times is non negligible. Said differently, LDGM codes are more suited to devices with small processing power than RSE which relies on complex mathematical operations.

Future works will consist in studying new transmission schemes and FEC codes. More elaborated channel models may also prove to be useful for specific target environments. Other performance metrics will also be added, like the maximum memory requirements needed in each case.

8. REFERENCES

[1] 3GPP. *3rd Generation Partnership Project; Technical Specification Group Services and System Aspects; Multimedia Broadcast/Multicast Service; Protocols and Codecs (Release 6)*, Nov. 2004. 3GPP TS 26.346 v1.5.0.

[2] H. Bai and M. Atiquzzaman. Error modeling schemes for fading channels in wireless communications: A survey. *IEEE Communications Surveys and Tutorials*, 5(2), 2003.

[3] J. Byers, M. Luby, M. Mitzenmacher, and A. Rege. A digital fountain approach to reliable distribution of bulk data. In *ACM SIGCOMM'98*, Aug. 1998.

[4] Digital Fountain Inc. http://www.digitalfountain.com/.

[5] H. Ernst, L. Sartorello, and S. Scalise. Transport layer coding for the land mobile satellite channel. In *59th IEEE Vehicular Technology Conference (VTC'04), Milan, Italy*, May 2004.

[6] ETSI. *IP Datacast Baseline Specification: Specification of Interface I MT (a080)*, Apr. 2004. DVB Interim Specification.

[7] R. G. Gallager. Low density parity check codes. *IEEE Transactions on Information Theory*, 8(1), Jan. 1962.

[8] A. Konrad, B. Zhao, A. Joseph, and R. Ludwig. A markov-based channel model algorithm for wireless networks. In *Proceedings of Fourth ACM International Workshop on Modeling, Analysis and Simulation of Wireless and Mobile Systems*, 2001.

[9] M. Luby, J. Gemmell, L. Vicisano, L. Rizzo, and J. Crowcroft. *Asynchronous Layered Coding (ALC) protocol instantiation*, Dec. 2002. Request for Comment 3450.

[10] D. MacKay. *Information Theory, Inference and Learning Algorithms*. Cambridge University Press, ISBN: 0521642981, 2003.

[11] C. Neumann, V. Roca, A. Francillon and D. Furodet. Impacts of packet scheduling and packet loss distribution on fec performances: Observations and recommendations. Research Report 5578, INRIA, May 2005.

[12] C. Neumann, V. Roca, J. Labouré, and Z. Khallouf. *An Open-Source Implementation of a Low Density Parity Check (LDPC) Large Block FEC Code*. URL: http://www.inrialpes.fr/planete/people/roca/mcl/.

[13] T. Paila. Mobile internet over ip data broadcast. In *10th IEEE Int. Conference on Telecommunications (ICT'03)*, Jan. 2003.

[14] T. Paila, M. Luby, R. Lehtonen, V. Roca, and R. Walsh. *FLUTE - File Delivery over Unidirectional Transport*, Oct. 2004. Request for Comments 3926.

[15] L. Rizzo. Effective erasure codes for reliable computer communication protocols. *ACM Computer Communication Review*, 27(2), Apr. 1997.

[16] V. Roca and C. Neumann. Design, evaluation and comparision of four large block fec codes, ldpc, ldgm, ldgm staircase and ldgm triangle, plus a reed-solomon small block fec codec. Research Report 5225, INRIA, June 2004.

[17] M. Yajnik, S. B. Moon, J. F. Kurose, and D. F. Towsley. Measurement and modeling of the temporal dependence in packet loss. In *IEEE INFOCOM'99*, 1999.