Data Communication over Challenged Networks: Application of Error Control Schemes in the Delay Tolerant Network Architectures

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Abstract — **Nowadays, the realization and the deployment of network infrastructures able to guarantee connectivity to geographical areas characterized by hazardous access conditions, such as remote sensor networks, disaster areas and interplanetary networks, are assuming great importance. Under this view, the large access latencies along with the high percentage of lost packets makes the applicability of TCP/IP protocol suite almost unfeasible. A promising solution counteracting TCP/IP shortcomings is represented by the Delay Tolerant Network (DTN) architecture, which exploits the principle of the mail delivery service to guarantee the success of the information transfer. This paper applies the DTN concepts to data communications achieved in deep space environments and investigates the adoption of error control schemes implemented at the higher layers, in order to make the transmission more reliable without the need of a feedback link for the acknowledgment transportation.**

Index Terms — **DTN, Deep space, Satellite Communication, Low Density Parity Check codes.**

I. Introduction

THE exigence of converging different technologies and protocol architectures into an unique network infrastructure able to support different multimedia services independently of the medium characteristics, has become a hot research topic in the last years. In more detail, the scientific community has focused its attention on the design of a telecommunication infrastructure able to support multimedia services over challenged networks, such as interplanetary communication, military tactical scenarios, and sparse mobile ad hoc networks, where the intermittent end-to-end connectivity, the large propagation delays and the high link error rates make the use of TCP/IP stack unsuitable as it is [1]. In this perspective, the research activity carried on within the Delay Tolerant Network working group in IRTF (Internet Research Task Force) has given an effective solution to all these problems in terms of protocol architectures, through the definition of the Delay Tolerant Network architecture [2], which exploits the principle of the mail delivery service and thereby guarantees highly effective and reliable data communication. For this purpose, a novel convergence layer, named bundle layer [3], is placed at the higher layers and it is responsible of guaranteeing the custody of the data once the transmission medium is again reliable. In this paper, the attention is mainly paid to the advantages offered by this architecture when applied in deep space operations, such as interplanetary networks, where large delays and high bit error ratios (BER) make the employment of TCP-based approaches unfeasible [4]. Moreover, the introduction of the Licklider Transmission Protocol (LTP) [5] below the bundle layer and the application of highly effective error control schemes such as LDPC and its variants [6], are investigated. The remainder of the paper is structured as follows. Section II presents the scenario analyzed in this work, from the point of view of both the physical environment and the protocol architecture adopted. Section III addresses a preliminary performance analysis, while Section IV points out the advantages deriving from the application of error controls schemes in DTN scenarios.

II. The Investigated Scenario

A. Overview

The work addresses data communication achieved over deep space networks. In particular, the communication established between a satellite platform orbiting around Saturn and an earth station, responsible of collecting the transmitted data, is taken as reference scenario. The need for a protocol architecture alternative to TCP/IP suite is due to the large propagation delays and to the highly asymmetric nature of the interplanetary links, which experience a ratio of 1000:1 among the bandwidth availability of the down-link with respect to the up-link [5].

In order to counteract these physical impairments, the Delay Tolerant Architecture is assumed on all the nodes involved in the communication. In more detail, from the protocol stack point of view, the bundle layer is responsible of storing the data and transmitting them when the link is available [2]. On the other hand, recovery mechanisms required to assure the reliability of the communication are supposed to be implemented at the underlying layers. In particular, in this work we assume the presence of the Licklider Transmission Protocol (LTP) [5] working below the bundle layer, shortly discussed in the following. The overall protocol architecture is shown in Fig. 1, where the interconnection between two consecutive DTN nodes is sketched. More in detail, LTP protocol serves as convergence layer between the bundle layer and the datalink/ physical layers. As regards the lower layers, being our attention mainly focused on communication achieved over long-haul links, the CCSDS Telecommand (TC) [7] and Telemetry (TM) [8] protocols are considered.

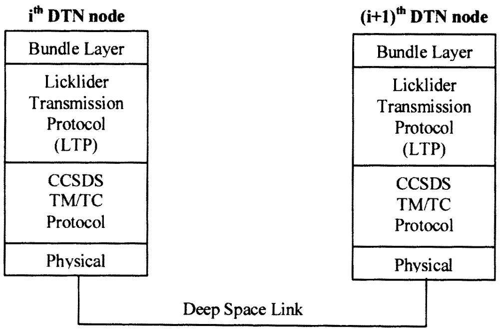

Fig. 1. The Delay Tolerant Network Architecture between two consecutive DTN nodes

B. Licklider Transmission Protocol (LTP)

The Licklider Transmission Protocol inherits the basic functionalities of CCSDS File Delivery Protocol in terms of recovery operations, but simplifying the protocol. In this work, given the high asymmetry exhibited by the deep space channel, the recovery schemes based on Automatic Repetition reQuest (ARQ) philosophy are not applicable. On the contrary, the focus of this paper is to propose the implementation of error controls schemes within the LTP protocol specification, in order to guarantee highly reliable transfer of data, even if at cost of network resources waste.

C. Error Control Schemes

Three different approaches are adopted: Repeated Transmission (RT), Reed Solomon Encoding (RSE) and Low Density Parity Check (LDPC) [6]. The former is based on the heuristics of repeated transmissions. When it is applied, every LTP block, scheduled for the transmission, is replicated N-1 times, giving rise to N transmissions. The second approach, is based on the adoption of Reed Solomon codes. In this case, each LTP block is firstly split into "k" data packets, whose dimension has to be properly tuned, and then encoded into "n" packets, n-k of which are redundancy packets. It is straightforward that an important role is played by the ratio between the "n" encoded and the "k" original packets. This parameter is referred as *Fec_ratio* in the remainder of the paper and it varies from 1.5 up to 5. The third solution adopted is based on a subclass of LDPC codes, defined in [9] and indicated in the following as LDGM. In this case the Fec_ratio value has been set to 1.5.

The first approach is referred in the following as LTP-RT and the other schemes as LTP-RSE and LTP-LDPC, respectively.

III. PERFORMANCE ANALYSIS

A. Testbed Configuration

In order to test the effectiveness of the envisaged solutions, an interplanetary satellite network composed of the Saturn orbiting platform, a earth satellite platform serving as relay and the earth station is considered. In this scenario, a maximum propagation delay of 5500 s is exhibited and available bandwidth values of 2.048 Mbit/s, 1.024 Mbit/s, 0.512 Mbit/s and 0.256 Kbit/s are considered. As regards the deep space links, in first approximation the AWGN (Additive White Gaussian Noise) channel model has been assumed and different operative conditions have been investigated. Provided that protocols working below LTP perform control error operations, the bit error ratio seen by the LTP entity may be assumed lower than 0.1, which typically is the rough BER value measured at the physical layer in deep space environments.

In the following, we will refer to BER values as the bit error ratio seen by the LTP protocol, once the underlying layer has performed the proper error control operations. Under this view, BER values ranging from 10^{-2} to 10^{-8} have been considered, where the values from 10^{-8} to 10^{-7} correspond to *almost clear sky condition*, from 10^{-6} to 10^{-4} correspond to *hard link intermittence*, and from 10^{-3} to 10^{-2} to *deep fade periods.*

The tests have been accomplished by considering a data transfer of 100Mbytes.

The probability of missing a LTP block, indicated as P_{loss} and defined as one minus the ratio among the transmitted and received blocks, neglecting the replicated ones as in the case of LTP-RT is the performance metrics together with the effective exploitation of the channel, indicated as *Effective Throughput*. The latter is measured as the product of $(1-P_{loss})$ and the ratio of the transfer amount with the time elapsed from the first byte until the last byte received, normalized to the reference bandwidth employed in the test. In facts:

$$P_{loss} = 1 - \frac{\text{Received Blocks}}{\text{Transmitted Blocks}}$$

$$\textit{Effective Througphut} = (1 - P_{loss}) \cdot \frac{\text{Transfer Amount}}{\text{Transfer Time}} \cdot \frac{1}{\text{Bandwidth}}$$

In order to characterize the different constraints of the traffic transported, namely data file, meteorological images and telemetry data, three classes of maximum P_{loss} have been defined. Class A includes the transfer of data file, which requires 100% of data delivery, and, hence, 0% of P_{loss}. Class B, may tolerate block loss up to 10^{-2} and Class C, may tolerate maximum P_{loss} of 10^{-1}.

B. Performance results

LTP-LDPC. The employment of LDGM codes, with Fec_ratio of 1.5, results to be powerful independently of the satellite channel state. For the sake of completeness, the tests have been performed by varying the size of the packets sent by the lower layers. The registered values of P_{loss} obtained by varying the BER configurations from 10^{-2} up to 10^{-8}, are equal to 0. In this case, the distinction among class A, B, and C, in terms of effective throughput is redundant, since no information loss is registered. As a consequence, all three kinds of data traffic are received by the destination users without information loss, confirming the power of such encoding scheme. As indicated in Fig. 2, as the dimension of the packet increases, higher values of effective throughput are registered.

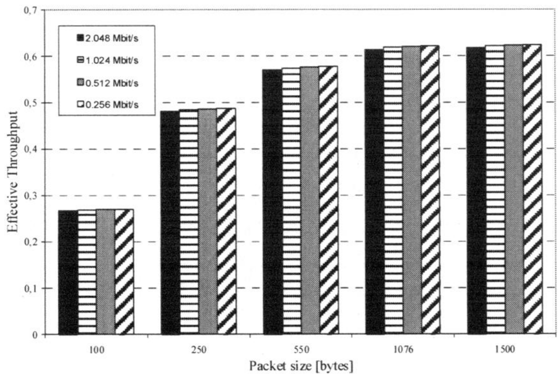

Fig. 2. Effective Throughput values for different available bandwidth values

This behavior is due to the fact that larger data units allow filling the channel pipe more effectively, since the information redundancy, caused by the LDPC encoding and by the headers imposed at the underlying layers, plays a minor role. A further consideration consists in the almost independence of the performance (as for LTP-RT and LTP-RSE) of the bandwidth availability. This aspect is determined by the following aspects:

- the only factors that may affect the performance are the extra-latencies introduced by the encoding/decoding operations;
- neither flow nor congestion control schemes are adopted.

Considering the values reported in [9], only a reduction of 0.1% of the performance is observed. Numerically, the maximum effective throughput registered (packet size of 1500 bytes) is about 0.61. Consequently, the channel bandwidth is not considered in the analysis of the other two approaches since its setting does not affect the performance.

LTP-RT. Performed tests show that, in general, the performance obtained is independent of the available channel bandwidth. So, only the registered values for the case of 2.048 Mbit/s are reported.

As far as the P_{loss} investigation is concerned, it is important to note that in presence of BER of 10^{-2} all the blocks are lost and P_{loss}=1 is measured. On the other hand, when BER values are lower than 10^{-6} (from 10^{-7} down to 10^{-8}), all the transmitted blocks are received correctly, giving rise to P_{loss}=0, independently of the number of performed transmissions. A particular attention has to be reserved to the intermediate cases (i.e. BER varying from 10^{-3} to 10^{-6}), in dependence of the number of repeated transmissions, indicated in Fig. 3, where P_{loss} versus the number of transmissions and size of the packets sent by the underlying layers is shown.

In correspondence of BER values ranging from 10^{-3} to 10^{-6}, the employment of 1-2 transmissions offers meaningful results. In general, with BER=10^{-3}, P_{loss} is higher than 0.1. Differently, when BER raises down, better performance values are registered in dependence of packet dimension. It is straightforward that increasing the number of transmissions, from 1 to 2, the probability of data blocks delivery increases too, at cost of the effective throughput, as pointed out in the following.

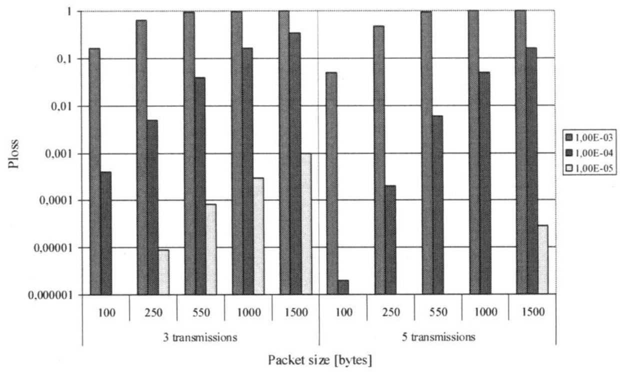

Fig. 3. P_{loss} obtained when 3 and 5 transmissions are performed

If the number of transmissions is increased, from 3 to 5 (in Fig. 3), when BER=10^{-6}, P_{loss}=0 is obtained (for the sake of the simplicity, this BER configuration has not been included in Fig. 3). For BER ranging from 10^{-3} to 10^{-5}, the performance is still strictly dependent on the packet size and on the number of transmissions performed. As highlighted in the previous case, better results are provided for the minimum packet size (i.e. 100 bytes) by performing 5 transmissions. In this case P_{loss}=0.05, $2\cdot10^{-6}$ and 0 is measured for BER=10^{-3}, 10^{-4} and 10^{-5}, respectively.

When there is a higher number of transmissions (from to 7 to 15), only BER=10^{-3} and 10^{-4} determine P_{loss} other than 0. In these cases, even the employment of 7 transmissions gives satisfying results with the minimum packet size (corresponding to 100 bytes). For 15 transmissions, the results are even better also for higher packet sizes, but at cost of wasted bandwidth.

The considerations reported above have a direct connection to the Effective Throughput results, depicted in Fig. 4. In particular, for data traffic belonging to Class A (P_{loss}=0), it is clear from Fig. 4 (reporting the best values) that, for BER=10^{-2} and 10^{-3}, registered P_{loss} values are 0, independently of the number of transmissions. With lower values of BER, the behavior is more satisfying even if a number of transmissions from 2 to 7 is required. Considering Class B ($P_{loss}\leq 0.01$), higher values of effective throughput are registered for each tested BER value, even if a low number of transmissions is performed. In particular, even in presence of BER = 10^{-3}, adopting a number of transmissions higher than 10, acceptable performance results are obtained (maximum Effective Throughput of 0.06 for 10 transmissions). Finally, analyzing Class C ($P_{loss}\leq 0.1$), the less stringent constraints on the P_{loss} allow registering higher results also in correspondence of low BER values.

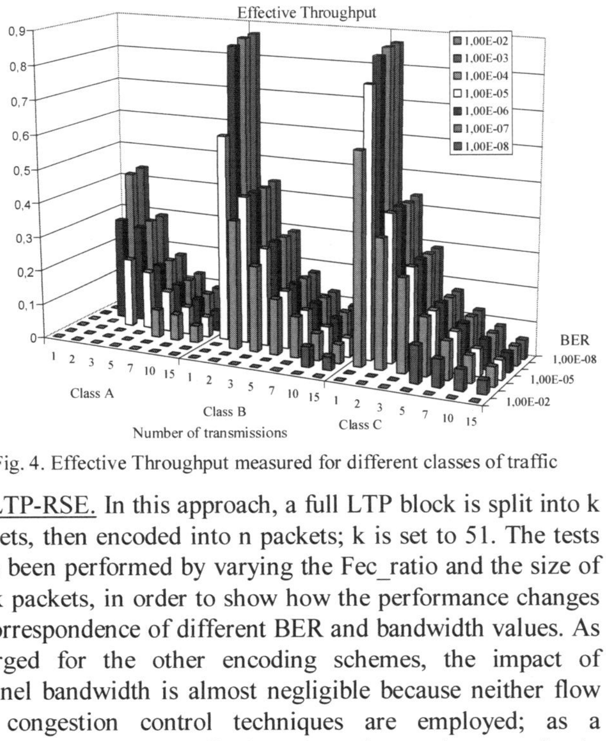

Fig. 4. Effective Throughput measured for different classes of traffic

LTP-RSE. In this approach, a full LTP block is split into k packets, then encoded into n packets; k is set to 51. The tests have been performed by varying the Fec_ratio and the size of the k packets, in order to show how the performance changes in correspondence of different BER and bandwidth values. As emerged for the other encoding schemes, the impact of channel bandwidth is almost negligible because neither flow nor congestion control techniques are employed; as a consequence P_{loss} and Effective Throughput values are simply ruled by Fec_ratio values along with the length of the encoded packets.

As far as P_{loss} analysis is concerned, independently of the Fec_ratio configurations, for BER of 10^{-2}, P_{loss}=1 is obtained, while for BER of $10^{-6}\div 10^{-8}$ P_{loss} falls down to 0.

For the other cases, two separated sets of Fec_ratio values can be individuated: low and strong *fec*, for values of 1.5-2 and 3-5, respectively. In the former case, for BER of 10^{-3}, the results are poor, since a limited number of redundancy packets is not able to recover a high number of errors, as exhibited for such BER configuration. For BER of 10^{-4} and 10^{-5}, the results are more encouraging: P_{loss} decreases down to 0 in the two cases, by employing a Fec_ratio of 2. For Fec_ratio of 1.5, acceptable results are provided too, by properly setting the packet size. In the latter case, P_{loss} observed for BER of 10^{-5} decreases to 0. For BER values of 10^{-3}, only a high number of redundancy packets (corresponding to Fec_ratio values of 4 and 5) allows obtaining P_{loss} results lower than 0.05. On the other hand, when a BER of 10^{-4} is investigated, very good performance results are experienced also with the application of Fec_ratio = 3.

As regards Effective Throughput, in Fig. 5, the results related to the different traffic classes are shown.

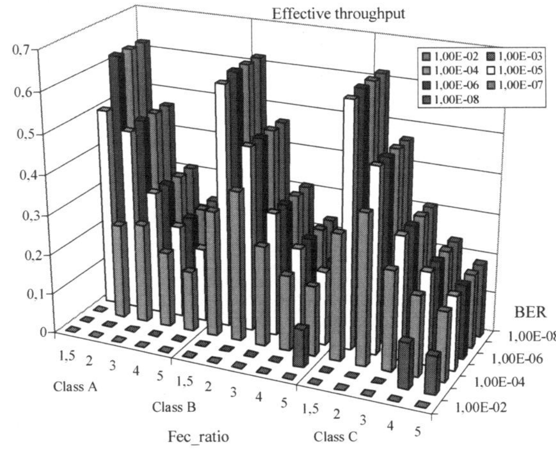

Fig. 5. Effective Throughput measured for different classes of traffic

In more detail, concerning Class A, which requires P_{loss} of 0, meaningful results can be observed only for BER lower than 10^{-3}. In this perspective, to obtain the maximum value a Fec_ratio=1.5 is enough for BER$\geq 10^{-5}$. Fec_ratio 3 for BER=10^{-4} is necessary. For class B, even if the loss constraints are less strong, , an effective throughput of 0 is given for BER=10^{-2}, while for BER=10^{-3} Fec_ratio=5 is required to get a performance result above 0. In the other investigated cases, the results are comparable with Class A ones. If Class C is considered, effective throughput values equal to 0 are again obtained for BER=10^{-2}, while for BER=10^{-3}, Fec_ratio=4 is sufficient for acceptable performance values. For the other analyzed BER configurations: higher values of Effective Throughput, if compared with Class A and Class B, are obtained. It is worth remembering that the extra-latency introduced for encoding/decoding operations has an important role in the protocol behavior: in facts, extra delays of tens of seconds are registered accordingly with [9]. However, this value impacts only for 1% of the protocol effectiveness.

C. *Performance Comparison*

As sketched in Fig. 6, LTP-LDPC offers a constant performance result, equal to 0.61, that gets over the other configurations for BER higher than 10^{-4}. As BER goes down, LTP-RT offers the best results, with a maximum of 0.868 for BER equal to 10^{-8}. Considering Class per Class:

- Class A: LTP-RT gives the lowest results, while LTP-RSE offers results progressively more satisfying as BER decreases and for BER $\geq 10^{-6}$, proves to be better than LTP- LDPC too.
- Class B: the advantage offered by LTP-RSE with respect to LTP-RT is clear only for BER $\geq 10^{-5}$. For lower BER values, LTP-RT overcomes the other solutions because of the relaxed constraint on P_{loss} (Class B: $P_{loss} \leq 0.01$).
- Class C: similar results are offered by all the solutions, but LTP-RT gives more satisfying results for BER of 10^{-2}. This more performing behavior, if compared to LTP-RSE, is due to the fact that $P_{loss} \leq 0.1$ is achieved by means of repeated transmission schemes, which allow obtaining higher probability of almost complete delivery of data.

IV. CONCLUSIONS

The problem of assuring reliability to data communications achieved over challenged environments has been addressed in this work, taking into account different channel conditions, namely "almost clear sky" (tolerable BER values ranging from 10^{-8} to 10^{-7}), "hard link intermittence" (experiencing BER values ranging from 10^{-6} to 10^{-4}) and "deep fade periods" (characterized by BER values of 10^{-2} and 10^{-3}).

The paper has been focused on the adoption of the Delay Tolerant Architecture and the application of error control schemes (Repeated Transmission RT, Reed Solomon Encoding RSE and Low Density Parity Check LDPC), implemented within the LTP protocol core. Three classes of data traffic have been assumed, namely Class A for transfer of data file, Class B for meteorological image transmission and Class C for telemetry data. They have been characterized by different constraints on the maximum probability of data block loss (i.e. 0 for Class A, 10^{-2} for Class B, 10^{-1} for Class C).

In the case of "deep fade periods" LTP-LDPC offers the best results, thanks to the very robust coding technique adopted. In the case of "hard link intermittence", also LTP-RSE offers encouraging results, while LTP-RT gives less satisfying performance values. On the other hand, LTP-RT employment is really promising when applied to "almost clear sky" conditions.

As next steps of this research: considering a wider scenario composed of several DTN nodes, in order to show how the combined employment of error control schemes, implemented within the LTP, and the exploitation of store and forward functions working in the bundle layer may help improve the overall performance, when even long outage periods are exhibited by the environment.

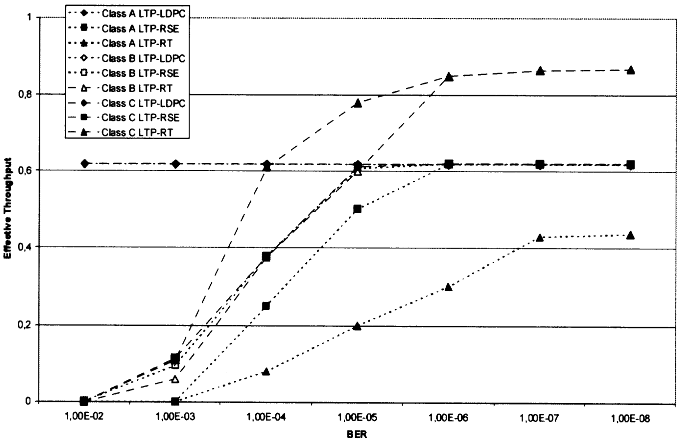

Fig. 6. Performance Comparison

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