Adaptive Erasure Coding for Interplanetary Networks with Incomplete Channel Side Information

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Abstract **— Providing effective telecommunication services over interplanetary networks poses a number of open challenges because of severe impairments exhibited in such environment.**

In this perspective, this work proposes a coding framework, integrated within the Delay Tolerant Network (DTN) architecture, implementing adaptive erasure coding schemes that rely upon partial observation of the transmission channel dynamics. Three main protocol solutions based upon Partial Observable Markov Decision Process (POMDP) and Incremental Redundancy Adaptation (IRA) have been devised and analysed under different scenario configurations. The performance analysis showed promising results and the superiority of the proposed solutions with respect to approaches based upon fixed-rate coding strategies.

Index Terms **– Interplanetary Networks, Adaptive Erasure Codes, Delay Tolerant Networks, LDPC codes**

I. INTRODUCTION

Since the end of nineties, the interest for space communications has increasingly grown especially in terms of analysis of communication technologies for transferring data over interplanetary networks [1]. In this perspective, particular emphasis has been put on the physical impairments introduced by deep space networks, which result in poor data communication performance

On the one hand, the long round trip times, typically ranging in space networks from few seconds up to hours severely affect data delivery time. Besides, also high error ratio plays a crucial role. More precisely, data communications are strongly affected by information loss that can be compensated for only through either specific retransmission or data protection techniques. The former approach can be hardly applied because of the long retransmissions rounds; concerning the latter, the reliability guarantees offered by adequate powerful schemes are traded off by the network bandwidth, power and processing consumption that they involve.

To tackle most part of these challenges, big effort has been made by standardisation bodies such as Consultative Committee for Space Data Systems (CCSDS) and Internet Research Task Force (IRTF). The former developed a complete protocol stack from the application down to the physical layer [2]. On the other hand, The Delay Tolerant Network (DTN) working group, as part of IRTF, has provided a novel overlay network architecture, suitable to enable data exchange in very challenged environments [3]. Furthermore, over the last years, the scientific community has focused its attention mainly on protocol solutions [4], able to mitigate the effect of noisy channels and to compensate for long latencies. In this light, Akyildiz *et al.* [5] proposed a transmission paradigm able to recognise link disruption events and to tune accordingly recovery procedures. As far as coding strategies are concerned, in [6] and [7] the advantages offered by fixed erasure codes are pointed out along with the power consumption issues that have to be accounted for. Finally, appropriate multimedia coding solutions have been proposed in [8], where the joint use of Tornado codes and probing packets is proposed and analysed.

This paper extends the concept of higher layer coding, by introducing adaptive erasure codes, relying upon partial observations (i.e., measures) of the channel state. This incomplete knowledge allows taking appropriate decisions about the coding strategy selection.

The remainder of this paper is organised as follows. Section II shows the general framework, by introducing the reference scenario and the protocol architecture. Section III presents the novel integrated coding framework, by proposing three different adaptive erasure coding strategies, whose effectiveness is assessed in Section IV. Finally, the discussion of the results and the conclusions are drawn in Section V.

II. GENERAL FRAMEWORK

A. *Delay Tolerant Network (DTN) architecture*

This work takes as reference the Delay Tolerant Network architecture [3]. It basically consists in the Bundle Protocol layer implemented just below the application layer (where present): it encapsulates the information messages into Bundle Protocol PDUs, hereafter referred to as *bundles*. The Bundle Protocol runs over the transport layer or directly over the network layer. In this work, we assume that the Bundle Protocol Layer lies directly over the network layer, implementing the CCSDS Space Packet Protocol, responsible for performing addressing and routing operations. Concerning the protocol specifications suited for the underlying layers two choices, depending on characteristics of the physical medium, have been considered. The CCSDS Telemetry/Telecommand Protocols (TC/TM) have been taken as reference for deep space links, whereas the CCSDS Proximity-1 Link Protocol

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has been considered for the case of proximity links (i.e., links connecting stations and orbiting aircrafts).

B. Higher Layer Coding approach

As pointed out in the introduction, this work develops around the concept and the application of higher layer erasure codes. As observed in [6] and [9] Reed Solomon codes, though being Maximum Distance Separable (MDS) codes, offer very poor performance because of processing times and high resource consumption. In alternative, a more promising candidate is represented by Low Density Parity Check (LDPC) codes that show great robustness against link errors with a lower processing overhead, provided that a large number of information symbols (> 1000) is available. By contrast, they show some inefficiency in terms of the minimum number of received symbols necessary to complete the decoding operations. In more detail, given k information symbols encoded into n symbols (i.e., code-rate equal to $R = k/n$), the decoding operations are likely to be successful unless $k(1+\varepsilon)$ symbols are received, where ε denotes the decoding inefficiency. However, an appropriate configuration of LDPC codewords may result in decoding inefficiency even lower than 0.05. Therefore, in this work, the use of LDPC-based code is assumed. In the following, the terms *symbols* and *bundles* will be used interchangeably as coding operations will be performed on Bundle Protocol PDUs.

C. Reference Scenario

The investigated scenario takes as reference and extends common environments in which ongoing space missions aimed at exploring the Mars surface, collecting images and measurements, and transmitting them to Earth's gathering centres are performed. From this viewpoint, it is possible to subdivide the whole scenario in four main areas, as sketched in Fig. 1: *Remote Planet*, *Earth's Gathering Centres*, *Relay Constellations*, and *Deep Space Links*.

All the network nodes implement full DTN protocol stack; the Bundle Protocol runs over the CCSDS Space Packet Protocol (network layer). Implementation of protocols at the underlying layers (i.e., datalink and physical) depends on the transmission link characteristics (i.e., proximity or deep space), as introduced in Section II-A.

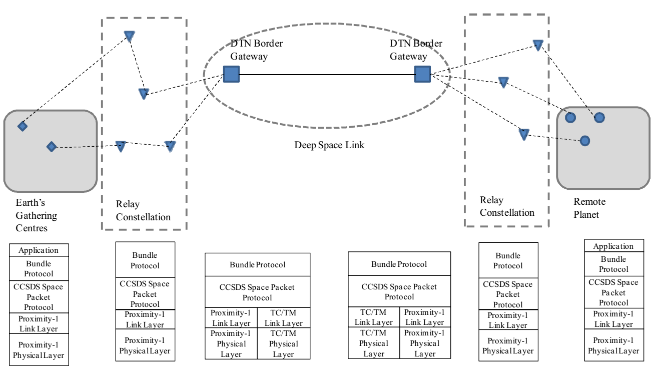

Fig. 1. Reference Scenario and DTN Protocol Stack.

As pointed out in the introduction, a crucial role in the system performance is played by physical medium impairments. Typical space missions register raw bit error rates as low as 10^{-1}, due to low signal to noise ratios observed in correspondence of shadowing and fading events. This work assumes a 3-states Discrete Time Markov Chain (DTMC) to model the deep space link. The Markov Chain is embedded at the beginning of each datalink layer transmission and the probability matrix is denoted by $\mathbf{P}$. The state i^{th} is characterised by the couple (FER_i, T_i), where FER_i is the datalink Frame Error Rate and T_i is the average permanence time, measured in state i^{th}, respectively. Let $\mathbf{FER}$ and $\mathbf{T}$ denote one-column vectors, representing the Frame Error Rate and the permanence time in each state. Thus, the channel model can be synthetically expressed by matrix C, where $C=[\mathbf{P}|\ \mathbf{FER}|\ \mathbf{T}]$. Finally, let τ_{ij} and B_{ij} denote the propagation delay and the bandwidth available on the transmission link established between nodes i and j. In this view, the whole deep space network can be represented by the 3-pla (C_{ij},τ_{ij}, B_{ij}), in the following referred to as A_{ij}.

III. The Integrated Coding Framework

Firstly, we assume that the encoding/decoding engine is directly implemented within the Bundle Protocol layer. Each processing operation is subdivided into encoding rounds, during which k bundles are buffered and then encoded into n bundles. The new encoded bundles implement an extended bundle header, in order to carry information about the code selection and the code-rate, necessary to the decoding operations. Secondly, we assume that the Bundle Protocol is configured in any node as working with the Bundle-Received-Report option: upon bundle reception, the Bundle Protocol entities issues an acknowledgment (ACK), notifying the DTN transmitting entities about the correct bundle reception. In more detail, the first acknowledgment is issued after the reception of k bundles, in order to reduce the load on the return link, and carries also information about the number of the received and already decoded bundles, and of the successful decoding rounds. These performance indicators are partial observations of the channel state that allows the DTN transmitting nodes, upon ACK reception, to properly tune the code-rate for the next encoding round. More precisely, the number of information symbols k is kept fixed, while the number of redundancy symbols n-k is varied.

Two different approaches have been formalised: Incremental Redundancy Adaptation (IRA) and Partially Observable Markov Decision Process (POMDP). It is worth noting that the aim of these schemes is to optimise the communication reliability by applying the coding configurations suitable to use efficiently the available network resources (Goodput). It is also worth remarkable that link propagation delay τ (where indexes of link ends are omitted for simplicity) plays a topical role. Actually, the observations received at time instant t refer to the link state at time instant t-2τ and consequently the control decisions cannot result optimal. Nonetheless, results showed that availability of delayed observations helps improve significantly the overall performance.

A. IRA apprach

The Incremental Redundancy Adaption consists in properly tuning, at the beginning of each encoding round, the number of encoded symbols n. In practice, the indications about the

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number of received and decoded bundles are taken as reference to adapt the code-rate. It is worth noticing that a decoding round is successful if at least k(1+ε) bundles are received: in case a larger number of bundles is received, it is possible to increase the coding rate (i.e., decrease the number of redundancy symbols). On the contrary, in case a larger number of bundles is expected to complete the decoding process, a lower code-rate will be applied in the next encoding round. In math, it yields:

$$n_{i+1} = n_i + \alpha \cdot (d_i - m_i) \quad (1)$$

where k > 0, and n_j, m_j, d_j, represent the number of encoded, received and successful decoded symbols during the i^{th} encoding/decoding round, respectively. In particular, in case of $m_j > d_j$ (i.e., the number of received bundles is bigger than the minimum needed to the decoding operations), α has been set to less than 1, in order to have a conservative increase of the code-rate. On the other hand, when $m_j > d_j$, the decrease of the code-rate is quicker: α > 1.

B. POMDP apprach

The use of a Partially Observable Markov Decision Process is expected to bring benefits to the overall system performance, as already observed in [10]. The code-rate applied in the next encoding round is updated on the basis of *belief states*. In practice, the POMDP problem for the link connecting nodes i ,j has been formalised by considering the 4-pla (A_{ij}, $\wp$, $\Omega(\wp)$, $\Re$), where $\wp$ denotes the set of partial observations containing information about the channel reliability, $\Omega(\wp)$ the statistical distribution over the collected observations, and $\Re$ the set of possible code-rate updates, respectively. The set $\Omega(\wp)$ is progressively updated on the basis of set of the received partial observations $\wp$, leading to the identification of *belief states*.

In more detail, it is possible to distinguish three working phases (*states*):

State "Initialisation". At the beginning, no observations are available and, consequently, the code-rate is set to a default *Init Value*, denoted in the following as $R_0=k/n_0$ where k is the fixed number of information symbols, whereas n_0 indicates the initial number of encoded symbols.

State "Training". As first observations are available, statistics about bundle loss and unsuccessful decoding round rates are computed. Once these statistics become stable, it is possible to suspend the training phase and hence identify the i^{th} belief state in terms of the Bundle Loss Rate BLR_i and to evaluate the code-rate accordingly: $R_i=(1-BLR_i)/(1+\varepsilon)$. In this state, the code-rate is updated by applying Eq. (1), as defined in the IRA approach.

State "Normal Transmission". For each new received observation, the estimation of the instantaneous Bundle Loss Rate, denoted as EBLR, is performed. Thus, the system is considered as belonging to belief state *j* if the difference between BLR_j and EBLR, is the minimum among all the possible choices of belief states. In maths it means:

$$j = \operatorname{Arg} \min_{i \in \{1,2,3\}} \left| BLR_i - EBLR \right| \quad (2)$$

Afterwards, the code-rate for the next $(l+1)^{th}$ encoding round is computed as a function of the code-rate applied in the previous one. In particular, being fixed the number of information symbols *k*, it is sufficient to adapt the number of encoded symbols n_{l+1}, as follows:

$$n_{l+1} = n_l \cdot (EBLR - BLR_i) \quad (3)$$

It is immediate to note that in case the estimated bundle loss rate (EBLR) is larger than what assumed in the belief state, the number of redundancy symbols is increased in order to make the data communication more robust against link errors.

C. The Protocol Solutions

Three different schemes are proposed. The use of IRA approach is identified as EDTN-IRA (standing for Extended DTN with IRA). The case of POMDP approach is split up into two protocol configurations, whose specifications basically differ in the coding scheme therein applied. First solution assumes that coding procedure performed in state "initialisation" implements an initial code-rate R_0. This solution will be referred to as EDTN-FPOMDP (standing for Extended DTN with Fixed POMDP). The second solution uses instead rateless code during the "initialisation" state, resulting in an indefinite amount of encoded symbols, which in turn can ensure, on the one hand, greater information protection, but, on the other hand, imply a significant waste of network resources. This solution will be referred to as EDTN-RPOMDP (standing for Extended DTN with Rateless POMDP). It is important to point out that rateless coding procedure takes place only in this state: afterwards the encoding procedure basically follows the same dynamics as EDTN-FPOMDP. In this configuration, the availability of an indefinite amount of encoded symbols (rateless code) is supposed to increase the probability of successful decoding and to improve the communication robustness unless first channel observations are available.

IV. Performance Analysis

A. Testbed Configuration

The performance analysis of the proposed solutions (termed EDTN-"scheme") was conducted by considering the case of two DTN nodes exchanging data through a deep space link. This configuration, though simplified with respect to the reference environment depicted in Fig. 1, does not limit the validity of the results achieved in the tests. In fact, in order to consider the different characteristics exhibited by both deep space and proximity links, various configurations have been considered. In more detail, forward channel bandwidth and propagation delays varying from 256 Kbit/s to 1 Mbit/s and from 2 s to 500 s, have been considered respectively. Bandwidth asymmetry has been accounted for as well, by assuming a ratio 1000:1 between down and uplink bandwidths. Moreover, as far the channel model is concerned (3-states DTMC, see Section II-C) the Markov Chain states are set with a Frame Error Rate equal to 10^{-3}, 10^{-1}, and $3 \cdot 10^{-1}$, corresponding to the cases of *almost line-of-sight*, *fading* and *shadowing*, respectively.

Regarding the protocol stack configuration, the application layer generates data messages carrying 1000 bytes. After bundle encapsulation and encoding operations, the overhead

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amounts to 66 bytes, resulting in a total length of 1066 bytes. The configurations of the encoding procedures have been varied in terms of the initial coding rate R_0. In particular coding rate values ranging from 2/3 to 10/13 have been considered. The number of encoded symbols *k* has been varied from 1000 to 1200.

Finally, in addition to the solutions explored in this work, for the sake of comparison, a fourth strategy relying upon fixed erasure coding schemes has been considered. The coding parameters are the same applied for the other solutions: number of encoding symbols *k* ranging from 1000 to 1200 and fixed coding rate from 2/3 to 10/13, respectively. This solution is referred to in the following as EDTN-Fixed.

The performance evaluation has been conducted by considering exchange of 100K messages, resulting in 100Mbytes; the Bundle Loss Rate (BLR) and the Normalised Goodput (NG) have been considered as reference metrics. The former is defined as ratio between the number of bundles missing at destination and the total number of transmitted bundles. On the other hand, the Normalised Goodput is defined as ratio between the amount of information bytes correctly received at destination and the equivalent number of bytes transmitted in the data delivery time. Firstly, the Bundle Loss Rate (BLR) has been compared to the performance offered by the static erasure coding rate algorithm, EDTN-Fixed. Secondly, EDTN-IRA, EDTN-FPOPMDP and EDTN-RPOMPD strategies have been analysed in terms of Normalised Goodput (NG).

All tests have been performed through ns-2, in a number of runs sufficient to guarantee the width of the confidence interval less than 1% for the 95% of the collected measures.

B. Results

The conducted tests have highlighted in particular that the key factors ruling the performance of the proposed solutions are the propagation delay, the network bandwidth and the code-rate. It is straightforward to expect that propagation delay undoubtedly affects the effectiveness of the implemented adaptive coding schemes because in case of large latencies, the received channel observations carry information about link reliability measured in past time instants. Also the code-rate plays a significant role: as it decreases, the probability of successful decoding operations increases in turn. As far as link bandwidth is concerned, it is immediate to point out that, being the propagation delay far larger than the bundle transmission time, the larger is the available bandwidth, the lesser is the number of received observations that can be used to select the most effective code-rate configuration. Finally, concerning the number of encoding symbols *k*, for the sake of simplicity, only the most effective configurations are reported here. Hereafter in this section, all the solutions will be denoted as "solution-initial code rate", expect for EDTN-RPMODP that does not implement an initial code-rate, but uses rateless codes.

The first case ($\tau = 2$ s), depicted in Fig. 2, clearly shows the superiority of all the proposed solutions with respect to EDTN-Fixed, owing to the powerful adaptive coding schemes implemented. In particular, independently of the link bandwidth, EDTN-fixed exhibit a Bundle Loss Rate (BLR) of 0.042, while the other strategies stay below 0.015. In detail, the more effective configurations are given by EDTN-IRA working with code-rate of 2/3, resulting in BLR values of about 0.002.

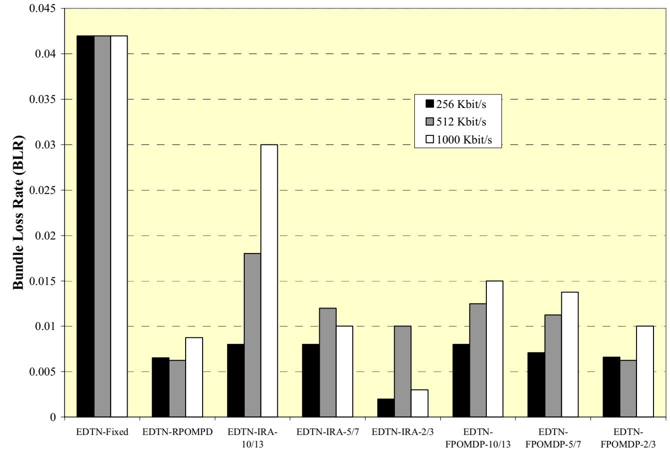

Fig. 2. Bundle Loss Rate performance for $\tau = 2$ s.

An intermediate case is represented by $\tau = 25$ s (shown in Fig. 3). Also in this case the performance offered by EDTN-Fixed is below the other solutions. However, the performance gap is reduced in this case: EDTN-Fixed attains 0.042 whereas, for instance, EDTN-IRA-5/7 gives 0.036. Finally, it is possible to remark that the most effective solution is EDTN-IRA-2/3 with a BLR of 0.006. It is worth observing that unlike previous case where application of rateless codes (EDTN-RPOMDP) was fruitful (BLR = 0.006), in this case the performance drops to 0.044.

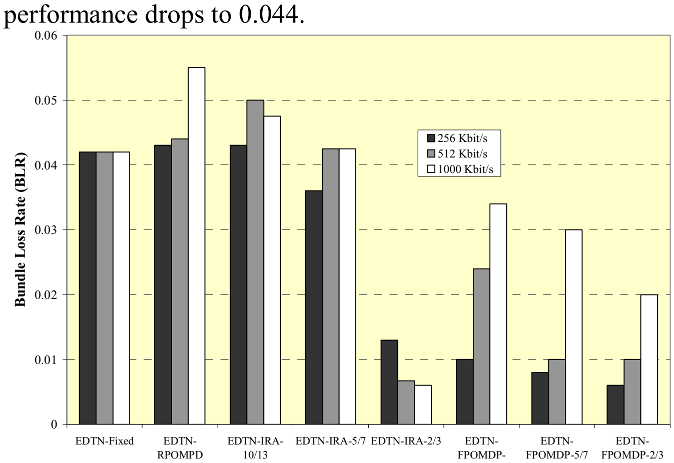

Fig. 3. Bundle Loss Rate performance for $\tau = 25$ s.

Finally, the case of $\tau = 500$ s, depicted in Fig. 4, highlights even lower performance difference between EDTN-Fixed and the other solutions. In fact, only EDTN-IRA-2/3 is able to provide more effective BLR values (0.016), while the other solutions give very poor performance (BLR ranging from 0.046 up to 0.078).

C. Comparison

In order to further characterise the effectiveness of each devised protocol solution, the attention has been finally paid to the Normalised Goodput (NG) offered by EDTN-RPOMDP, EDTN-FPODMP and EDTN-IRA. To this end, the protocol configurations in terms of code-rate and number of encoding symbols (*k*), guaranteeing the most satisfactory Block Loss Rate performance (discussed in the previous section) have

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been taken as reference. In particular, the analysis has been carried out by comparing the different solutions in dependence of propagation delay and link bandwidth (Fig. 5). It is possible to observe from Fig. 5 the strict relationship between Normalised Goodput (NG) and latency: not surprisingly, as the delay increases, from 2 s up to 500 s, NG values reduce accordingly, because high propagation delay affects both data transfer time and efficacy of code-rate selected upon receipt of channel observations, thus leading to reduced performance.

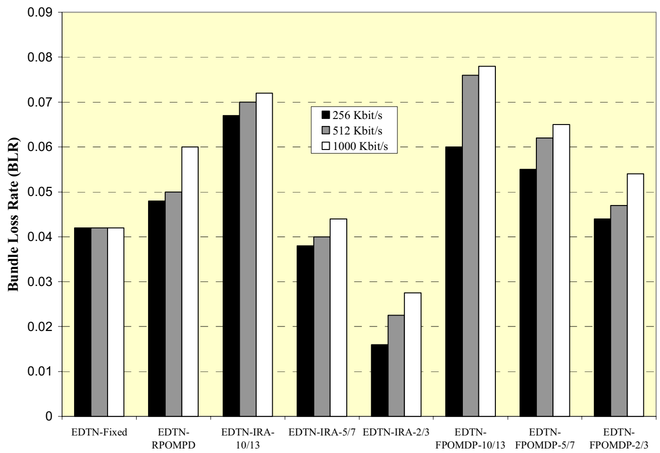

Fig. 4. Bundle Loss Rate performance for $\tau = 500$ s.

From the quantitative viewpoint, it is possible to recognise from Fig. 5 that EDTN-IRA outperforms the other solutions when the propagation delay is below 25 s, independently of link bandwidth: Normalised Goodput values of approximately 0.79 and 0.74 are registered for delays of 2 s and 25 s, respectively. On the other hand, when delay further increases to 50 s, EDTN-IRA performs poorly. EDTN-FPOMDP offers the most satisfactory results, attaining 0.69, while EDTN-IRA keeps below 0.64. Also EDTN-RPOMDP (NG is 0.66) is quite promising, though less effective than EDTN-FPOMDP. Finally, when propagation delay is set to 500 s, the performance difference among the three protocol solutions is less evident: NG varies from a minimum of 0.58 (at 1 Mbit/s) up to 0.66 (at 256 Kbit/s). In particular, also in this case, EDTN-FPOMDP is the most effective, showing NG values ranging from 0.601 (at 1 Mbit/s) up to 0.662 (at 256 Kbit/s). EDTN-RPOMDP offers good results as well: NG values range from 0.601 (at 1 Mbit/s) up to 0.661 (at 256 Kbit/s).

V. Conclusions

This paper addressed the study of adaptive erasure coding techniques for interplanetary networks. In this perspective, two approaches relying upon imperfect knowledge of the transmission channel have been devised. The former applies the concept of Incremental Redundancy adaptation, whereas the latter applies the concepts of Partially Observable Markov Decision Process. The results show that the proposed approach outperforms erasure coding solution based on fixed coding rate (EDTN-Fixed), independently of the network characteristics. Furthermore, comparative analysis of the proposed solutions in terms of Normalised Goodput shows that EDTN-FPOMDP and EDTN-RPOMDP are more effective in terms of Normalised Goodput in case of very large delay networks (> 50s), whereas EDTN-IRA attains the highest performance degree in correspondence of delays lower than 50 s.

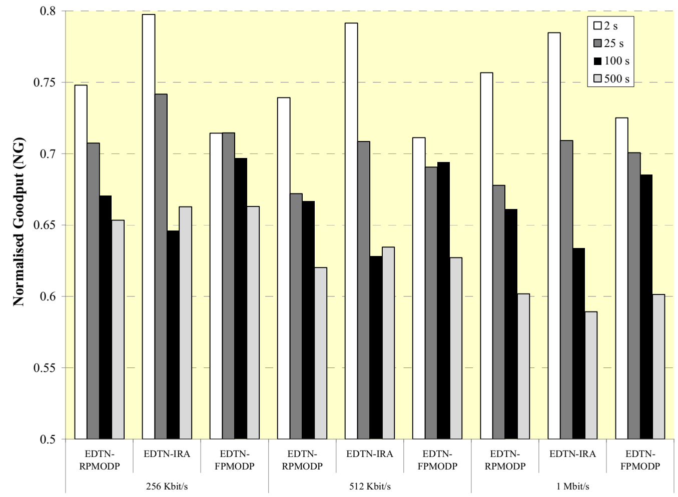

Fig. 5. Normalised Goodput performance

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