Serially Concatenated Luby Transform Coding and Bit-Interleaved Coded Modulation Using Iterative Decoding for the Wireless Internet

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Abstract – **In Bit-Interleaved Coded Modulation (BICM) the coding and modulation schemes were jointly optimized for the sake of attaining the best possible performance when communicating over fading wireless communication channels. The iterative decoding scheme of BICM (BICM-ID) invoking an appropriate bit-to-symbol mapping strategy enhances its achievable performance in both AWGN and Rayleigh channels. BICM-ID may be conveniently combined with Luby Transform (LT) codes, which were designed for handling packetized wireless Internet data traffic in erasure channels without retransmitting the corrupted packets. By jointly designing a serially concatenated LT-BICM-ID code, an infinitesimally low Bit Error Rate (BER) is achieved for Signal to Noise Ratios (SNR) in excess of 7.5dB over wireless Internet type erasure channels contaminated by AWGN.**

1. INTRODUCTION

In both wireless and wired Internet based data communication, each data file is transmitted in terms of small packets. In the Medium Access Control (MAC) layer, an Acknowledgment (ACK) mechanism is used for requesting the source to retransmit lost or corrupted packets. Under hostile channel conditions, the retransmission overhead may become excessive, especially when using header-compression for reducing the extra overhead imposed by the Internet Protocol (IP). After the reception of an error-infested IP header, an uncompressed header has to be transmitted. Therefore in hostile wireless channel conditions the employment of the IP header compression and retransmission mechanism may substantially increase the transmission overhead and hence may in fact reduce the effective throughput.

For the sake of addressing these wireless Internet design problems, in this contribution we propose a rather different solution. The Internet packet may consist of a number of symbols, where each symbol contains a certain number of bits. These individual bits would be transmitted in terms of symbols over the wireless channel after the modulation. An error correcting code is employed for the sake of recovering the original symbols. However, to avoid the employment of a retransmission mechanism, in addition to using redundant data symbols for Forward Error Correction (FEC) coding, the source may transmit a limited number of redundant data packets, which helps to recover the original data file, even if the FEC code failed to remove all errors in some of the packets. Our design objective is then to minimize the number of redundant packets required for the complete recovery of the original data file, rather than aiming for the highest possible coding rate, which is the design objective of classic FEC codes. The associated design trade-off is that the number of corrupted packets can be reduced by reducing the code-rate of the FEC scheme, which clearly imposes a higher burden on the packet-recovery scheme. Our ultimate goal is hence to minimize the total redundancy imposed by both FEC and LT coding, while maximizing the overall integrity. Serially concatenated and iteratively detected coded modulation schemes are capable of attaining a near-capacity performance and hence in the proposed system Bit Interleaved Coded Modulation using Iterative Decoding (BICM-ID) is combined with a Luby Transform (LT) code [1] [2] [3].

To elaborate a little further, LT codes were proposed by Luby [3] based on the philosophy of the so-called Fountain codes [1] which were first proposed for the sake of reliably transmitting data files over the Ethernet-based Internet. The files are assumed to be constituted by K information packets, each containing n bits. For the sake of reliably recovering the original data file, K'=K+E encoded data packets have to be transmitted, where E is the number of redundant packets that has to be as low as possible.

BICM was proposed by Zehavi [4] for transmission over wireless fading channels. It invokes both a non-systematic convolutional code as well as independent interleavers for each of its bit. The bit interleavers assist in minimizing the effects of bursty errors as well as maximizing the attainable time-diversity order. Combining BICM with iterative decoding [5] leads to an improved performance for BICM-ID over both AWGN as well as Rayleigh channels. When using n original uncoded information bits per symbol, the number of coded BICM-ID output bits becomes $m = n + 1$. Assuming that the packets contain P number of symbols and that no tail bits are used by the BICM-ID scheme, we have a total of $P.n$ number of bits in the LT source packets and the total rate of the LT-BICM-ID code becomes R=$Kn/K'm$. Figure 1 illustrates the structure of the source packets as well as the encoded packets employed in the LT-BICM-ID scheme.

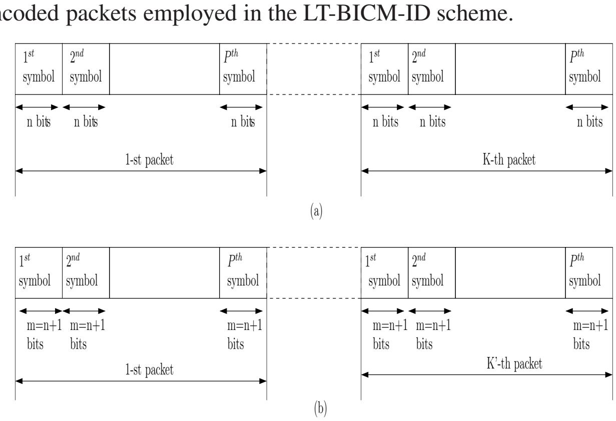

Figure 1: (a) Structure of the source packets before LT coding. (b) Structure of the BICM-ID encoded packets.

In this contribution, we propose a novel jointly designed serially concatenated LT-BICM-ID coding scheme, which is capable of improving the attainable Bit Error Rate (BER) performance in the wireless Internet. In Section 2 we describe the proposed LT code, while the entire system's architecture is outlined in Section 3. We discuss our simulation results in Section 4, while our conclusions are presented in Section 5.

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2. CODES FOR THE WIRELESS INTERNET

2.1. Fountain Codes

Let us consider the scenario that a Mobile Station (MS) would like to fetch a data file from a server using the wireless Internet constituted by an ad hoc network of MSs. The header-compression-based Internet Protocol using ACK-based Automatic Repeat Request (ARQ) may be expected to have an excessive transmission overhead in a hostile wireless ad hoc propagation environment due to the preponderance of error-infested and therefore retransmitted packets. Hence a more efficient design alternative is proposed here, which invokes the novel family of Fountain codes [1] [2] [3].

In 1998, Luby and Buyers proposed the employment of sparse-graph codes [6]. They packetized data files into small transmission packets and the number of bits per packet was defined based on the size of files [6] [7]. With the advent of these codes we are capable of transmitting as many encoded packets as necessary for recovering the source data, provided that the number of received packets is slightly higher than the number of original packets.

Again, the corresponding codes are referred to as digital Fountain codes based on the analogy of a fountain, which is capable of producing a sufficiently large supply of water drops. Each water drop may represent a data packet of $P.n$ original uncoded bits. Let us assume that the number of source packets in a file is K, where each source file has the size of $K.P.n$ bits. Metaphorically speaking, the MS which would like to receive the original data file will have to fetch it drop-by-drop i.e. packet-by-packet until the bucket recoding the received packets is filled. More explicitly, this bucket will collect a certain number of water drops, which is slightly higher than the number of packets K constituting the original file of $K.P.n$ bits. Again, when using the serially concatenated LT-BICM-ID coding scheme, a total of $K'.P.m$ bits are transmitted and we would like to simultaneously maximize both the concatenated scheme's coding-rate of $R = Kn/K'm$ and the achievable integrity expressed in terms of the Packet Loss Rate (PLR).

2.2. LT Codes

In 2002, a novel family of the Luby Transform (LT) codes [3] was proposed, which exhibited a higher source packet recovery probability and a lower complexity than the family of Fountain codes [1]. The encoding operation of LT codes is similar to that of the Fountain encoder [3], but for reasons of space economy it is not detailed here. Based on the source packets $s_1, s_2, ..., s_K$ seen in Figure 1, we are able to choose a degree d from the so-called *ideal soliton* distribution of $\rho(d)$ [1] which will be described in the next section. By definition, the degree d_c of a distribution within the interval c of generating a K-bit transmission packet represents the number of connections or *edges* between the received packets and the original source packets seen in Figure 2, which indicates with the aid of the LT-decoding rules of [3] how the source packets can be recovered from the received packets. More explicitly, the packet duration here refers to the time interval in which K random bits are generated by the encoder's generator matrix. As it will be detailed in the next section, the appropriate choice of $\rho(d)$ depends on the number of packets K in the source file to be transmitted. Figure 2 presents an example of the LT decoding process, where we observe three source packets S_i, each packet containing n=3 information bits. These packets are represented by hollow squares, which were previously encoded into four transmitted packets represented by the filled black squares. The value of each bit within the transmitted packet is the result of the modulo 2 operation of the corresponding source packets connected to it. The decoding process is implemented as follows.

In the first cycle of the decoding process seen in Figure 2a, the LT decoder determines, which of the received packets has a degree of one, indicating that this is a self-contained packet, which was not combined with any other source packet using the modulo 2 operation. Hence

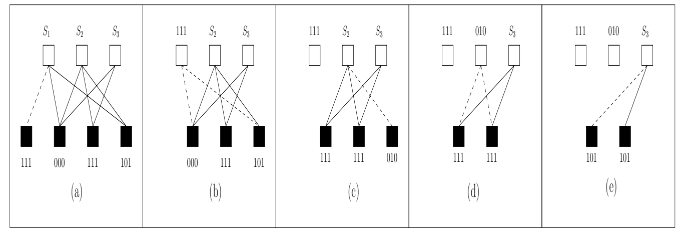

Figure 2: Decoding of a LT code having K=3 source packets and N=4 transmitted packets each containing 3 bits; adopted from [2].

the decoding operation is simply constituted by outputting the corresponding source packet as indicated by the dashed line in Figure 2a. At the same time, the decoder finds other transmitted packets, which are connected to this source packet Figure 2b. In order to further exploit the encoding rules of the specific LT-code used, as seen in Figure 2b, the decoder erases the already exploited connections. As the next decoding step, the received packets that have a link to the decoded degree-one packet are updated by modulo-2 adding their value to the value of each related,I.e. connected source packet. More explicitly, in this example the first transmitted degree-one packet found by the decoder is the packet having the value of "111" and the corresponding source packet is S_1. Hence, in the next decoding cycle seen in Figure 2b S_1 is decoded a value "111" and the connections drawn from this packet to the second and the fourth transmitted packets using dashed lines are erased. Correspondingly, the values of the second and fourth transmitted packets change from "000" in Figure 2b to "111" from "101" to "010", as observed in Figure 2c. As seen in Figure 2d and 2e, at the end of the LT decoding process all of the three source packets S_1, S_2 and S_3 are recovered.

2.3. Degree Of Distribution

An appropriate choice of the degree of distribution is crucial in the design of LT codes for the sake of maintaining a low PLR. Every source packet must have at least one $edge$ leading to the received packet, where an $edge$ is defined as the connection in Figure 2, indicating the reception of a source packet which was LT-encoded using the time-variant generator matrix defined for the corresponding packet interval. Ideally, upon the successful reception of a packet, a new self-contained degree-one received packet should appear for the sake of avoiding having redundant packets. This objective is achieved by the $ideal\ soliton\ distribution$ of [2]

$$\rho(d) = \begin{cases} 1/K & \text{for } d = 1, \\ \frac{1}{d(d-1)} & \text{for } d = 2, 3, ..., K, \end{cases} \tag{1}$$

which is plotted in Figure 3. However, the above-mentioned ideal behaviour is only achievable with a certain probability [2]. When no degree-one packet is recovered at any stage of the consecutive LT-decoding cycles exemplified in Figure 2, the decoding process will be suspended and therefore the remaining packets cannot be recovered, unless a sufficiently high number of redundant packets is received. Hence the so-called $robust\ soliton\ distribution$ was designed by Luby [3] for circumventing this problem by increasing the expected number of degree-one encoded packets to $S \equiv c \cdot log_e(K/\delta)\sqrt{K}$, where the parameter δ denotes the probability of decoding failure imposed by the lack of a degree-one packet, while c represents a constant. More explicitly, taking the above-mentioned parameters into consideration, Luby proposed the improved distribution of [3]:

$$\tau(d) = \begin{cases} \frac{S}{K}\frac{1}{d} & \text{for d=1, 2, } \ldots, \frac{K}{S} - 1, \\ \frac{S}{K}\log(\frac{S}{\delta}) & \text{for } d = \frac{K}{S}, \\ 0 & \text{for } d > \frac{K}{S}, \end{cases} \tag{2}$$

which is also plotted in Figure 3 as a function of the degree d. Finally, for the sake of improving both the distributions of Equations 1 and 2,

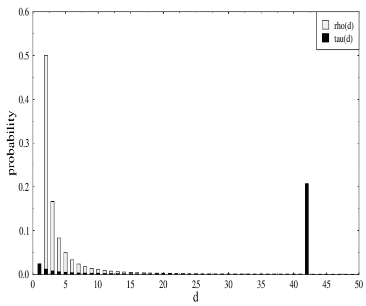

Figure 3: The distributions $\rho(d)$ of Equation 1, $\tau(d)$ of Equation 2 for the case of K=10000, c=0.2, δ=0.05 [3]

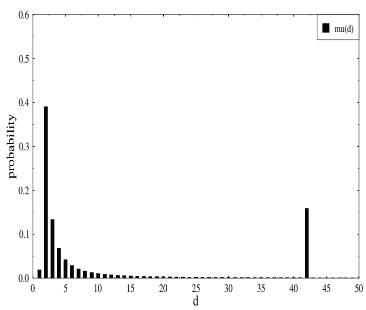

Figure 4: The robust distributions $\mu(d)$ of Equation 3 for the case of K=10000, c=0.2, δ=0.05

Luby introduced the so-called $robust\ soliton\ distribution$ μ, which is constituted by the superposition of the $ideal\ soliton\ distribution$ $\rho(d)$ and that of Equation 2 as [3]

$$\mu(d) = \frac{\rho(d) + \tau(d)}{Z}, \tag{3}$$

where the normalisation factor of the denominator is given by $Z = \sum_d \rho(d) + \tau(d)$, ensuring that all the probabilites sum to unity. The function $\mu(d)$ is portrayed in Figure 4. In practice, an LT code is capable of recovering the transmitted file having K packets, if it receives about 5% more packets than the number of original source packets K [2], provided that the channel conditions are adequate. The values of the parameters c and δ in $S \equiv c \cdot log_e(K/\delta)\sqrt{K}$ determine the number of packets required for recovering all the K source packets.

2.4. Improved Robust Distribution

In this section, we propose an Improved Robust Distribution (IRD). Referring to Luby's $robust\ soliton\ distribution$ characterized in Figure 4, we observe that there exist some degree distributions, which have such a low probability P_{di} that the number of packets given by the the product of P_{di} and K may be less than one, indicating the potential absence of packets having these degrees. It is undesirable to invoke this particular distribution in the computation of the LT encoder's generator matrix **G**. Therefore, in some cases Luby's $robust\ soliton\ distribution$ defined by Equation 3 may lead to premature decoding abortion and a concomitant loss of packets during the decoding process, unless the number of redundant packets is high. When the decoding process is suspended due to the absence of a degree one packet, we need more received packets for recovering all source packets.

In view of this, let us consider the case, when we have $P_{di} \cdot K < 1$ in Luby's $robust\ soliton\ distribution$. Our novel proposition is to improve the distribution's behaviour by introducing an extra factor

$$\nu = \sum \mu(d_i) \cdot K, \tag{4}$$

for the sake of creating a more beneficial degree distribution where d_i represents the degree-i term of the distribution, satisfying the following conditions

$$\begin{cases} \left(\frac{\frac{1}{d(d-1)} + \frac{s}{K} \cdot \frac{1}{d}}{Z}\right) \cdot K < 1 & \text{for } 2 \leq d \leq \frac{K}{S} - 1, \\ \frac{1}{d(d-1)} \cdot \frac{K}{Z} < 1 & \text{for } (\frac{K}{S}+1) \leq d \leq K. \end{cases} \tag{5}$$

We determine the parameter ν with the aid of the set of inequalities given by 5, so that we maximize the relative frequency of having packets of degree one, because this allows us to recover the source packets from a single received packet. In other words, the degree-one packets' reception probability was ignored by the original $robust\ soliton\ distribution$ [2] [3] [6], which was now taken into account by our improved robust distribution. The measure of increasing the relative frequency of degree-one packets has however not only benefits, but also disadvantages. The main benefit is that a degree-one packet is self-contained and hence may be decoded without reference to any other received packets. By contrast, the disadvantage is that it contains less information about other received packets in comparison with higher degree packets. Hence it may neither be used to recover other packets nor be recovered from other packets. Hence it may neither be used to recover other packets nor be recovered from other packets. Figure 5, 6 and 7 characterize the improved robust distribution. When using Luby's $robust\ soliton\ distribution$ of [2] and the LT coding scheme characterized in Figure 50.4 of [2], appromately 12000 packets were required for recovering 10000 packets with the probability at least $1 - \delta$ [2]. By contrast, the number of packets required was reduced to about 10800, when using the improved robust distribution. Having characterized the proposed degree distribution, let us now focus our attention on the proposed system architecture.

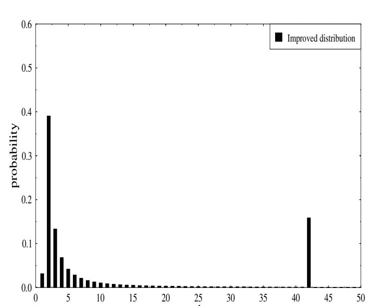

Figure 5: The improved robust distributions for the case of K=10000, c=0.2, δ=0.05

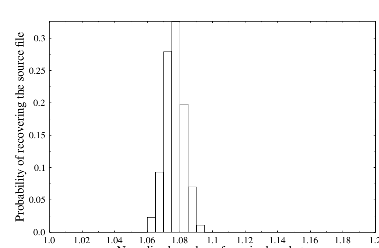

Figure 6: Improved Robust Distribution based LT code specified in Figure 50.4 of [2] for c=0.1 and δ=0.5. The actual number of packets N required to recover the original source file of size K=10000 packets is given by the product of K and the abscissa value

3. SYSTEM OVERVIEW

This section briefly outlines the design of our system seen in Figure 8, where the LT scheme is the outer code and the BICM-ID arrangement is the inner code. Our investigations are based on $P.n$=165

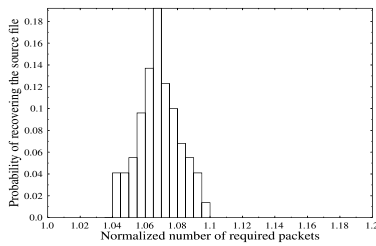

Figure 7: Improved Robust Distribution based LT code specified in Figure 50.4 of [2] for c=0.03 and δ=0.5. The actual number of packets N required to recover the original source file of size K=10000 packets is given by the product of K and the abscissa value

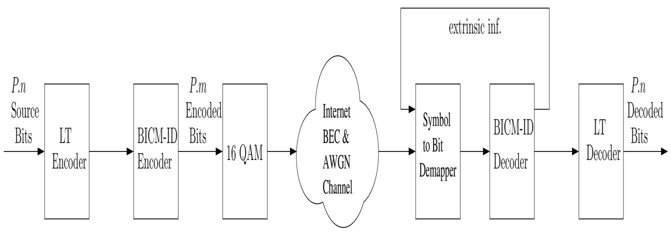

Figure 8: System structure of the proposed LT-BICM-ID scheme. The corresponding transmission frame structure is seen in Figure 1 and the associated system parameters are summarized in Table 1.

bits per original source packet, which is similar to the 164-bit packet size of a transport multimedia service packet in the Time Division Duplex (TDD) mode of the Third-Generation (3G) wireless system [8]. The success or failure of a packet is decided on the basis of Cyclic Redundancy Check (CRC) detection [8]. We transmitted a total of K=10^4 source packets corresponding to 1 650 000 bits. The number of 'parity' packets was 3000, rendering the total number of received packets N=13000. The BICM-ID encoder's output has $m = n +$ 1=3+1=4 encoded bits per symbol and therefore each packet consists of P=165/3=55 16QAM symbols. Hence the size of the BICM-ID encoded packet transmitted over the wireless channel is constituted by (165x4/3 = 220) bits/packet. Each symbol is modulated using 16QAM modulation and transmitted over an AWGN channel, where random data bits are assumed to be erased by the Internet modelled by the BEC. The received packet is demodulated and decoded at the receiver. The effective code rate of our system is R=$K.n/K'm$=(10000x3)/(13000x4)=30000/52000=15/26. The erasure channel imposes a packet erasure probability of P_e=0.1.

The family of LT codes has been originally designed for channels prone to packet loss, such as the Internet, where each packet can be readily identified by its packet ID. These channels are often referred to as 'erasure' channels. However, in order to correctly identify an erased packet, we have to make the assumption that the LT encoder and the LT decoder are perfectly synchronized and hence are capable of identifying the lost packet's ID.

RS codes	C(7,3,2)
LT codes	C(10000,13000)
LT-BICM-ID codes	C(30000,52000)
BEC erasure probability	0.1
Modulation	16QAM
Number of source packets K	10000
Number of transmitted packets K'	13000
Number of bits per source packets $P.n$	165

Table 1: System parameters.

At the demodulator, the depacketized bits are converted to soft bit values that are fed into the BICM-ID MAP decoder [9], which feeds the output $extrinsic$ information constituting the $a\ priori$ information to the input of the symbol-to-bit demapper seen at the input of the BICM-ID scheme in Figure 8. We used Set Partition (SP) based mapping [5] for the sake of achieving an iteration gain as the benefit accruing from having a maximized Euclidean distance amongst the BICM constellation points.

The BICM-ID-decoded hard bits are then used as the input bits of the LT decoder in Figure 8. Since a message-passing LT decoding algorithm is used [1], an excess number of errors found in the received packets may inflict error propagation and subsequently might degrade the reliability of the neighbouring check nodes, potentially imposing a low LT decoding performance. The output of the BICM-ID decoder ensures that the overwhelming majority of the erroneous bits has been corrected, hence reducing the probability of error propagation.

4. SIMULATION RESULTS

The BER performance of the system is characterized in Section 3 and summarized in Table 1. These results were extracted from Figure 9 and 10. Reed-Solomon (RS) codes have been widely used for transmission over erasure-type channels before the introduction of Fountain codes [1], hence similar-rate RS codes were used for benchmarking. Figure 9 shows the BER performance of the RS (7,3,2) code, the LT (13000,10000) code and the proposed LT-BICM-ID (30000,52000) code, when communicating over the BEC contaminated by AWGN, as shown in Figure 8. Again, the source file was constituted by K=10000 packets and a total of K'=13000 packets were transmitted. As it transpires from Figure 9, the stand-alone RS or LT code does not perform well, when communicating over the BEC contaminated by AWGN. The BEC employed has an erasure probability of 0.1. The LT-BICM-ID code also has a high BER for SNRs below 7.4dB, which however significantly improves above this point, as seen in Figure 9. This indicates that for SNR $>$ 7.4dB, the BICM-ID decoder becomes capable of correcting most of the bits in the P=55-symbol packets and the resultant number of corrupted bits becomes sufficiently low for the LT decoder to correct them.

Figure 9 also portrays the BER performance of the LT-BICM-ID scheme in conjunction with different number of inner BICM-ID iterations. The 2-iteration aided LT-BICM-ID code having a memory of eight states is only capable of achieving an error free BER performance for SNRs above 8.8dB, which is almost 1.5 dB higher than the corresponding threshold SNR of the 10-iteration based inner code. This illustrates that as the inner code's error correcting capability improves, a higher fraction of error-free bits is passed to the LT decoder, hence enhancing the overall BER performance of the system.

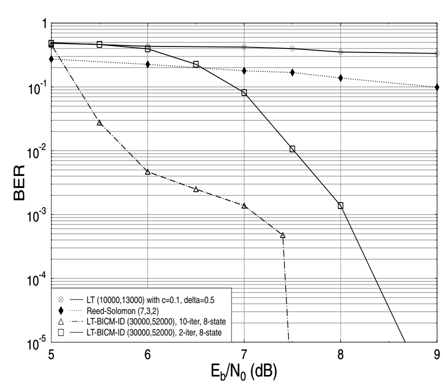

Figure 9: BER versus E_b/N_0 performance of the rate 15/26, 10/13 and 3/7 of LT-BICM-ID, LT and RS codes communicating over the BEC channel contaminated by AWGN, when using 16QAM.

Figure 10 further characterizes a range of various LT-BICM-ID codes communicating over the BEC contaminated by AWGN and having different degree distributions [3]. For the general Poisson distribution, the degree of transmitted packets is defined by a function given in [3] as

$$\Omega_D(x) = \frac{1}{\mu+1}\left(\mu.x + \frac{x^2}{1.2} + \frac{x^3}{2.3} + ... + \frac{x^D}{(D-1).D} + \frac{x^{D+1}}{D}\right), \quad (6)$$

where D=$[\frac{4(1+\epsilon)}{\epsilon}]$ and μ=$(\epsilon/2)$ + $(\epsilon/2)^2$ and ϵ is defined as a small positive real-valued number.

We further investigated the truncated Poisson 1 (Ω_1) [10], the truncated Poisson 2 (Ω_2) [10], and the *robust* degree distributions, where the Poisson degree distributions [10] were used for the sake of reducing the number of corrupted bits inflicted by the AWGN channel at low SNRs. Here, we employed truncated Poisson distributions, which were defined as follows [11] :

Ω_1(d) = $0.007969d^1 + 0.5161d^2 + 0.166220d^3 + 0.072646d^4 + 0.082558d^5 + 0.056058d^8 + 0.037229d^9 + 0.0025023d^{65} + 0.003135d^{66}$;

Ω_2(d) = $0.007544d^1 + 0.5033d^2 + 0.166458d^3 + 0.071243d^4 + 0.084913d^5 + 0.049633d^8 + 0.043365d^9 + 0.0045231d^{19} + 0.010157d^{20} + 0.010479d^{66} + 0.017365d^{67}$; where d^n represents the degree order of each of the above items.

The *robust* degree distribution proposed in Section 2.4 of this paper was used. Figure 10 demonstrates that the Poisson degree distribution Ω_1 exhibited the worst BER performance, resulting in a constant BER around 6x10^{-2} for SNRs in excess of 7 dB. The BER performance of the Ω_2 degree distributions was slightly better due to its higher mean degree of five as opposed to the average degree of 4.63 for the Ω_1 distribution. Above the SNR of 7dB, all the Possion distributed LT-BICM-ID schemes maintain a constant BER with no further improvement in their error correction capability. This is due to the lack of connecting *edges* between the check nodes and the source nodes in the corresponding factor graph. In other words, the average degrees are insufficiently high for correcting the corrupted packets inflicted by the erasure channel. The LT-BICM-ID scheme associated with the *robust* distribution of Section 2.4, is however, capable of correcting all the errors, when the SNR reaches 7.5dB, since it contains a higher number of factor graph connections between the LT-coded packets and the source packets, which assists in correcting more erroneous packets. As the decoding process continues, we may encounter a lack of degree *one* connecting edges, thus forcing the decoding process to halt prematurely. This explains the phenomenon that the BER performance does not improve, upon increasing the SNR which is exhibited by all the Poisson distributed LT-BICM-ID scheme above the point of SNR=7dB.

We further investigate the performance of our proposed *robust* distribution compared to the original *robust* distribution of [1] in conjunction with the same value of c=0.1 and δ=0.5. We observe from Figure 10 that the LT-BICM-ID scheme using the original *robust* distribution of [1] performs better in the range of SNR $<$ 7.5dB. This is due to the lower number of *edges* or lower average number of degree distribution compared to our *robust* scheme of Section 2.4. When the mean degree of distribution is lower, resulting in a reduced number of connecting *edges* in the factor graph, there would be a reduced error propagation between the source packets to the LT-coded packets, especially at a lower SNRs when the inner BICM-ID code is less capable of correcting the bits corrupted by the AWGN. This means that, if a high number of received packets are corrupted, the corrupted packets are less likely to propagate its error bits due to the lower number of connecting *edges* between source packets and received packets. However, at higher SNRs, the proposed *robust* distribution of Section 2.4 outperforms the schemes using the remaining distributions due to their higher average degree distribution. When the inner BICM-ID code is capable of correcting those bit errors, the higher mean of the *robust* degree distribution scheme assists in recovering the corrupted packets, as a benefit of increased number of connecting *edges* in the factor graph. This shows that the proposed *robust* distributed LT outer codes of Section 2.4 exhibits a better BER performance at higher SNRs, as shown in Figure 10, while requiring a reduced number of extra packets for the sake of recovering the original source packets, as shown in Figures 6 and 7.

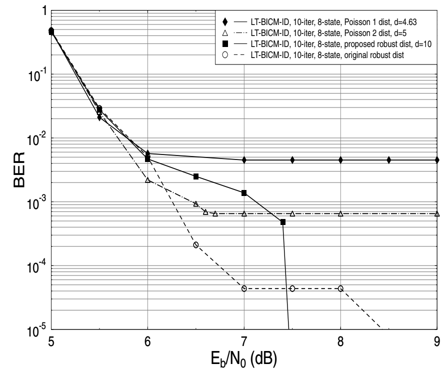

Figure 10: BER versus E_b/N_0 performance of various LT-BICM-ID codes communicating over the BEC contaminated by AWGN using various Poisson and *robust* distributions employing 16QAM.

5. CONCLUSIONS

A sophisticated serial concatenated LT-BICM-ID scheme was proposed, which is capable of improving the BER performance of packetized data transmission, when communicating over the AWGN-contaminated BEC. The concept of using BICM-ID to improve the attainable BER of the LT codes invoked for correcting the packet errors was the main benefit of the proposed scheme. With the advent of using our proposed improved *robust* distribution of Section 2.4 and a strong BICM-ID inner code, we demonstrated that our scheme is capable of achieving a better BER performance than that of the classic RS (7,3,2) and LT (10000, 13000) benchmarkers, when communicating over the BEC contaminated by AWGN. This scheme is particularly attractive in a wireless Internet scenario. We elaborated on the joint optimization of the inner and outer code-rates for the sake of maximizing both the effective throughput and the attainable integrity of the system. Our future research will consider the employment of both binary and non-binary LDPC codes as well as TTCM schemes in dispersive turbo-equalized channels [9].

6. REFERENCES

[1] D. J. C. MacKay, *Fountain Codes*. Cavendish Laboratory, University of Cambridge, 2004. http://www.inference.phy.cam.ac.uk/mackay/CodesTheory.html:

[2] D. J. C. MacKay, *Information Theory, Interference and Learning Algorithms*. http://www.inference.phy.cam.ac.uk/mackay/itila/: Cambridge University Press, 2003.

[3] M. Luby, "LT codes," in *Proceeding of the 43rd Annual IEEE Symposium on Foundations of Computer Science*, pp. 271–282, November 2002.

[4] E. Zehavi, "8-PSK trellis codes for a Rayleigh fading channel," *IEEE Transactions on Communications*, vol. 40, pp. 873–883, May 1992.

[5] X. Li and J. A. Ritcey, "Bit-interleaved coded modulation with iterative decoding," *IEEE Communications Letters*, vol. 1, pp. 169–171, November 1997.

[6] J. Buyers, M. Luby, M. Mitzenmacher and A. Rege, "A digital Fountain approach to reliable distribution of bulk data," *In Proceedings of ACM SIGCOMM'98*, September 1998.

[7] M. G. Luby, M. Shokrollahi, M.Watson and T.stockhammer, "Raptor Forward Error Correction Scheme," *Reliable Multicast Transport Internet-Draft*, June 2005.

[8] L. Hanzo, L-L. Yang, E-L. Kuan and K. Yen, *Single- and Multi-Carrier DS-CDMA*. New York, USA: John Wiley IEEE Press, 2003.

[9] L. Hanzo, T. H. Liew and B. L. Yeap, *Turbo Coding, Turbo Equalisation and Space Time Coding for Transmission over Wireless channels*. New York, USA: John Wiley IEEE Press, 2002.

[10] A. Shokrollahi, "Raptor Codes," p. 36, June 2004.

[11] A. Shokrollahi, *Raptor Codes*. http://algo.epfl.ch: Technical Report, June 2003.