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Improved FEC: Improving The Efficiency of Forward Error Correction Coding In Reducing The Network Packet Loss

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Abstract – In this paper we explores a method for measuring the performance of FEC coding combining with interleaving in reducing the packet loss in IP networks. In order to evaluate the performance of FEC data can be transfer from the source to destination and creates the packet loss voluntarily, at the destination the lost packets can recovered using FEC decoder. The performance of the FEC coding can be measured using an analytical method stated in this paper. Here we use the single multiplexer network model for transmission of the data from multiple sources to destinations. In this a unified approach provides an integrated framework for exploring the compromises between the various key parameters i.e. channel coding rates, interleaving depths, block lengths. It provides the selection of various optimal coding strategies with various QOS requirements and system constraints.

Index Terms- FEC Coding, interleaving, packet loss rates, multiplexer network model, multi session and single session.

1. INTRODUCTION

The packet transport service provided by representative packet-switched networks, including IP networks, is not reliable and the quality-of-service (QoS) cannot be guaranteed. The Packets may be lost on their route, Switching nodes requires more processing power as the packet switching protocols are more complex, Switching nodes for packet switching require large amount of RAM to handle large quantities of packets, A significant data transmission delay occurs - Use of store and forward method causes a significant data transmission.

Packets may be discarded due to excessive bit errors and failure to pass the cyclic redundancy check (CRC) at the link layer, or be discarded by network control mechanisms as a response to congestion somewhere in the network.

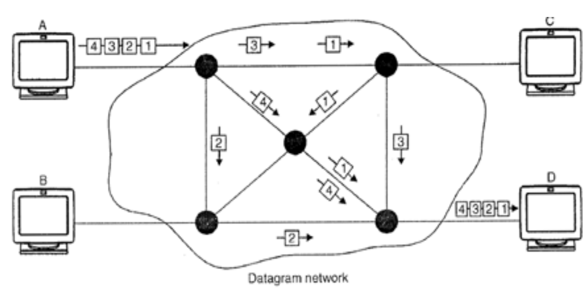

Fig1: Packet switched networks (datagram)

Forward error correction (FEC) coding has often been proposed for end-to-end recovery from such packet losses. FEC can be defined as
A class of methods for controlling errors in a one-way communication system. FEC sends extra-
information along with the data, which can be used by the re

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ceiver to check and correct the data. Codes that include the unmodified input in the output are **systematic**, while those that do not are **nonsystematic**.

A novel technique based on forward error correction (FEC) has been proposed that allows the destination to reconstruct missing data packets by using redundant parity packets that the source adds to each block of data packets.

Example: an analog to digital converter that samples three bits of signal strength data for every bit of transmitted data. If the three samples are mostly all zero, the transmitted bit was probably a zero, and if three samples are mostly all one, the transmitted bit was probably a one. The simplest example of error correction is for the receiver to assume the correct output is given by the most frequently occurring value in each group of three.

Triplet received	Interpreted as
000	0
_00	0
0_0	0
00_	0
0__	0
_0_	0
0__	0
111	1
_11	1
1_1	1
11_	1
1__	1
_1_	1
__1	1

This allows an error in any one of the three samples to be corrected by "democratic voting". This is a highly inefficient FEC. In practice FEC codes typically examine the last several dozen, or even the last several hundred, previously received bits to determine how to decode the current small handful of bits Such triple modular redundancy, the simplest form of forward error correction, is widely used. FEC works by adding check bits to the outgoing data stream. Adding more check bits reduces the amount of available bandwidth, but also enables the receiver to correct for more errors.The main advantage of the FEC is reduces the retransmission of the lost packet.

Forward Error Correction is particularly well suited for satellite transmissions, where bandwidth is reasonable but latency is significant.

The use of FEC in applications provides a double-edged sword. From an end user's perspective, FEC can help recover the lost packets in a timely fashion through the use of redundant packets, and generally adding more redundancy can be expected to improve performance provided this added redundancy does not adversely affect the network packet loss characteristics.

On the other hand, from the network's perspective, the widespread use of FEC schemes by end nodes will increase the raw packet-loss rate in a network because of the additional loads resulting from transmission of redundant packets. Therefore, in order to optimize the end-to-end performance, the appropriate trade-off, in terms of the amount of redundancy added, and its effect on network packet-loss processes, needs to be investigated under specific and realistic modelling assumptions.

Types of FEC: The two main categories of FEC are block coding and convolutional coding.1) Convolutional codes work on bit or symbol streams of arbitrary length. This code can be turned into a block code, if desired. Convolutional codes are most often decoded with the Viterbi algorithm, though other algorithms are sometimes used. 2) Block codes work on fixed-size blocks (packets) of bits or symbols of predetermined size. There are many types of block codes, but the most notable is Reed-Solomon coding because of its widespread use on the Compact disc, the DVD, and in computer hard drives. Block and convolutional codes are frequently combined in **concatenated** coding schemes in which the convolutional code does most of the work and the block code (usually Reed-Solomon) "mops up" any errors made by the convolutional decoder.

Advantages: EDAC has a number of advantages for the design of high reliability digital systems:

1) Forward Error Correction (FEC) enables a system to achieve a high degree of data reliability, even with the presence of noise in the communications channel. Data

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integrity is an important issue in most digital communications systems and in all mass storage systems.
2) In systems where improvement using any other means (such as increased transmit power or components that generate less noise) is very costly or impractical, FEC can offer significant error control and performance gains.
3) In systems with satisfactory data integrity, designers may be able to implement FEC to reduce the costs of the system without affecting the existing performance.
This is accomplished by degrading the performance of the most costly or sensitive element in the system, and then regaining the lost performance with the application of FEC.
In general, for digital communication and storage systems where data integrity is a design criterion, FEC needs to be an important element in the trade-off study for the system design. The introduction of the Per FEC line of FEC encoders and decoders makes powerful FEC implementation a realistic goal for most digital communication and storage systems. More than ever before, FEC is available for a wide range of applications.

Most telecommunication systems used a fixed channel code designed to tolerate the expected worst-case bit error rate, and then fail to work at all if the bit error rate is ever worse. However, some systems adapt to the given channel error conditions: hybrid automatic repeat-request uses a fixed FEQ method as long as the FEQ can handle the error rate, then switches to ARQ when the error rate gets too high; adaptive modulation and coding uses a variety of FEQ rates, adding more error-correction bits per packet when there are higher error rates in the channel, or taking them out when they are not needed.

Averaging noise: To reduce the errors FEC could be said to work by "averaging noise"; since each data bit affects many transmitted symbols, the corruption of some symbols by noise usually allows the original user data to be extracted from the other, uncorrupted received symbols that also depend on the same user data. Because of this "risk-pooling" effect, digital communication systems that use FEC tend to work perfectly above a certain minimum signal-to-noise ratio and not at all below it. This *all-or-nothing tendency* becomes more pronounced as stronger codes are used that more closely approach the theoretical limit imposed by the Shannon limit. Interleaving FEC coded data can reduce the all or nothing properties of transmitted FEC codes. However, this method has limits. It is best used on narrowband data.

This explores a framework for using Forward Error Correction (FEC) codes with applications in public and private IP networks to provide protection against packet loss. The framework supports applying FEC to arbitrary packet flows over unreliable transport and is primarily intended for real-time, or streaming, media. This framework can be used to define Content Delivery Protocols that provide FEC for streaming media delivery or other packet flows. Content Delivery Protocols defined using this framework can support any FEC scheme (and associated FEC codes) that is compliant with various requirements defined in this document. Thus, Content Delivery Protocols can be defined that are not specific to a particular FEC scheme, and FEC schemes can be defined those are not specific to a particular Content Delivery Protocol.

Interleaving:

Interleaving in computer science is a way to arrange data in a non-contiguous way in order to increase performance. It is used in: Time-division multiplexing (TDM) in telecommunications, Computer memory and Disk storage. Interleaving is mainly used in data communication, multimedia file formats, radio transmission (for example in satellites) or by ADSL. The term multiplexing is sometimes used to refer to the interleaving of digital signal data. Interleaving was used in ordering block storage on disk-based storage devices such as the floppy disk and the hard disk. The primary purpose of interleaving was to adjust the timing differences between when the computer was ready to transfer data, and when that data was actually arriving at the drive head to be read.

Interleaving was very common prior to the 1990s, but faded from use as processing speeds increased. Modern disk storage is not interleaved.

A method for making data retrieval more efficient by rearranging or renumbering the sectors on a hard disk or by splitting a computer's main memory into sections so that the sectors or sections can be read in alternating cycles. i.e. Iinterleaving was used to arrange the sectors in the most efficient manner possible, so that after reading a sector, time would be permitted for processing, and then the next sector in sequence is ready to be read just as the computer is ready to do so. Matching the sector interleave to the processing

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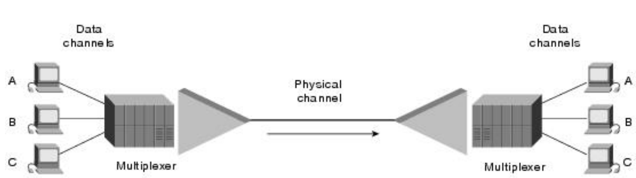
speed therefore accelerates the data transfer, but an incorrect interleave can make the system perform markedly slower.

Here we explores a brief study of the overall effectiveness of packet-level FEC coding, employing interlaced Reed-Solomon codes, in combating network packet losses and provide an information- theoretic methodology for determining the optimum compromise between end-to-end performance and the associated increase in raw packet-loss rates using a realistic model-based analytic approach. Intuitively, for a given choice of block length we expect that there is an optimum choice of redundancy, or channel coding rate, since a rate too high (low redundancy) is simply not powerful enough to effectively recover packet losses while a rate too low (high redundancy) results in excessive raw packet losses due to the increased overhead which overwhelms the packet recovery capabilities of the FEC code

1. *Single Source Model*

Single multiplexer model- The performance of the network is limited by a single bottleneck node, the network can often be modelled in terms of the single multiplexer. The single multiplexer is queuing system which consist of three parts one is the arrival process of the packets from N different sources S_i at arrival rate λ_i, second buffer that to hold up to the k packets and last one is an output link with averaging packet service rate μ, assume that packet service times are independent and identically Distributed (i.i.d) with an exponential distribution and average packet service time T=1/ μ, the normalized load to system is $\rho = \lambda / \mu$.

The packet arrives to the single multiplexer from single source or multiple sources. The single source corresponds to the per-flow control from the traffic (assigns fixed bandwidth). Whereas multiple source having no per-flow control is applied. Here the packet shares the bandwidth of the output and buffer.



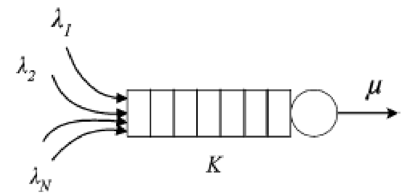


Fig. 2 single multiplexer network model

Source Model- Assume that the packet arrival process for each source S_i is renewal process. The packet interval times are i.i.d with probability density function a(t). erlang inter arrival time distribution is

$$a(t) = \frac{\beta(\beta t)^{h-1}}{(h-1)!} e^{-\beta t}, t \geq 0$$

For erlang distribution, the average arrival rate is $\lambda_i = \beta / h$ when h=1, a(t) becomes exponential and arrival process is Poisson when $h \to \infty$, the variance of a(t) converges to 0 and the interval time becomes deterministic with period $1/\lambda_i$

System model for FEC performance Evaluation–

Consider communication system illustrated in Fig.1 suppose there are N homogeneous and independent data sources sharing the single multiplexer and each source generates packets with average rate λ_i. The FEC coder for each source applies the Reed Solomon codes. *RS(n, k)* to the packets from the source, which means for every block of k source information packets it creates additional n-k parity packets to the network. The channel coding rates is given by R_C = *k/n*. because of this channel rate, the packet arrival rate into the network will increase $\lambda'_i = \lambda_i / R_c$.

The random variable N_p denotes the number of lost packets within a block. If $N_p \leq n-k,$ then assume that all the lost packets are recovered by the channel decoder. Assume that $p(j,n)$ denotes the block error distribution, the expected number of lost packets within a block is

$$E[N_p] = \sum_{j=n-k+1}^{n} j * P(j,n)$$

And the expected number of the lost information packets within a block is

$$E[N_I] = E[N_p] * (k/n) = \left(\sum_{j=n-k+1}^{n} j * P(j,n) \right) * (k/n)$$

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Finally the effect information packet loss rate after channel decoding is

$$PLR_{eff} = E[N_I]/k = \left(\sum_{j=n-k+1}^{n} j * P(j,n) \right) / n$$

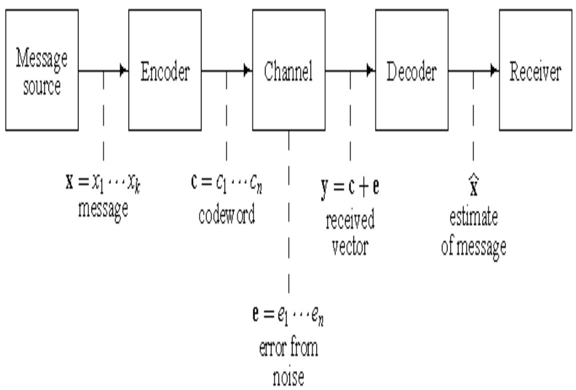

Fig3. Communication Channel

Evaluation Metric: Packet loss probability or Frame loss probability

The frame loss metric is more suitable than the packet loss probability for evaluation of FEC performance the frame lost probability FLR_{eff} is given by

$$FLR_{eff} = \left(\sum_{j=n-k+1}^{n} P(j,n) \right)$$

The differences between residual packet loss probabilities and frame-loss probabilities with decoding and without decoding denoted by FLR_{eff} – FLR_{wo} respectively.

A single-multiplexer model for this bottleneck node is widely used to analyze the associated queuing-related Packet losses, e.g., losses due to buffer overflows and excessive delays. Since the correlation level of the packet-loss process has great impact on the FEC efficacy, this dependence using the autocorrelation function of the packet-loss process

Autocorrelation function of packet loss processes- For this packet loss process we use the autocorrelation function to characterize the dependence between the packet loss events over the time, let $\{Y_i\}$ random sequence represent the packet loss process with 1 denoting the loss and 0 denoting the reception if $\{Y_i\}$ is stationary then autocorrelation function of $\{Y_i\}$ is given as

$$\rho y(l) = \frac{E[(Y_{i+1} - \mu_y)(Y_i - \mu_y)]}{E[(Y_i - \mu y)^2]}$$

Where l is the lag and μy is the expectation of the sequence $\{Y_i\}$. The performance of FEC in recovering network packet losses. Redundant parity packets was proposed to reconstruct lost data packets and the corresponding performance evaluation indicated that residual packet-loss rates can be reduced up to three orders of magnitude. ***FEC performance with single source FEC without Interleaving*** - Suppose there is only one user for multiplexer the key quantity in evaluating the residual packet loss rate after FEC decoding is $p(j,n)$, the block-error distribution for an arbitrary number n of consecutive packets. Queuing system there are two types of queuing system denoted as M/M/1/K queue: finite buffer queue with Poisson intervals and exponential service times and extension of G/M/1/K queue: the finite buffer with general independent and identically distributed interval times and exponential service times.

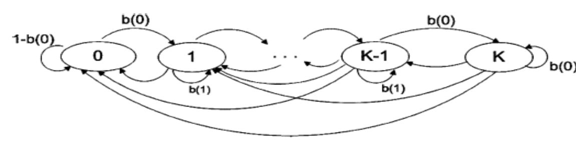

Fig. 4 state transition of arrival-epoch system size

Analysis of Block error Distribution: suppose there is only one source sharing multiplexer and the packet inter arrival times are i.i.d with arbitrary probability density function $a(t)$, single multiplexer can be modeled as a standard G/M/1/K queue. Discrete-time marcov chain: let Xn be the no.of packets in the buffer just before the nth packet arrives at the system because the memoryless property of the exponential service time { Xn} lets consider the above figure 4 with state space S={0,1,2,…K}.

Numerical example: The Fig. 4 shows the effective packet loss rates PLR_{eff} computed according to

$$PLR_{eff} = E[N_I]/k = \left(\sum_{j=n-k+1}^{n} j * P(j,n) \right) / n$$

With different coding block size n=63,127,255 and 511 as a function of coding rates $R_C = k/n$. having the Poisson arrivals (h=1), ρ =0.8, k=10.

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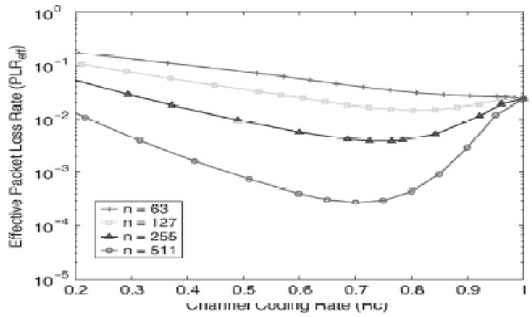


Fig5: effective packet loss rates With different coding block size n=63,127,255 and 511 as a function of coding rates R_C = k/n. having the Poisson arrivals (h=1), ρ =0.8, k=10

The above fig shows the FEC performance with different numbers of sources multiplexed, where the load from each source is fixed at λi = 0.02 with buffer size K = 10 and coding block size n=63 . It shows that, with an increase in the number of sources N, the effective packet-loss rates increase due to the increased system load. Suppose now the load from each source is again λi = 0.02 and the required effective packet-loss rate is 10^{-6}.

FEC with Block Interleaving- FEC performance is often limited by the bursty nature of typical packet-loss processes, and block interleaving techniques are frequently used to reduce the burstiness of the packet-loss processes in networks, thereby improving FEC performance. In this section, we analyse the efficacy of interleaving in reducing the burstiness of network packet-loss processes and in improving the FEC performance. *Interleaving Operation:* Before packets being transmitted into the network, packets are filled into an M1*M2 Row wise.

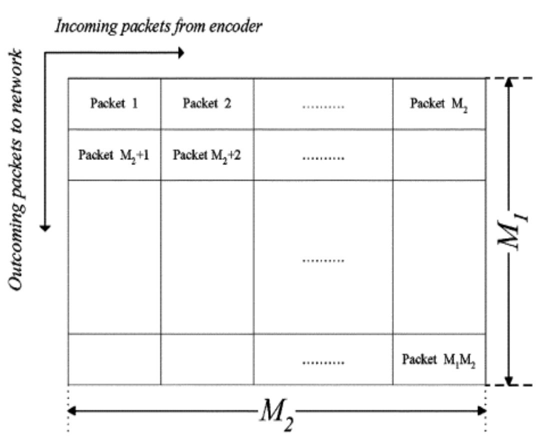
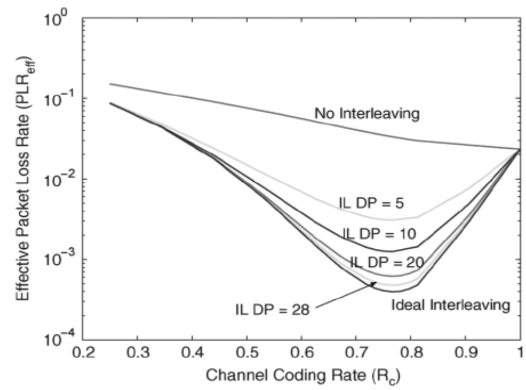
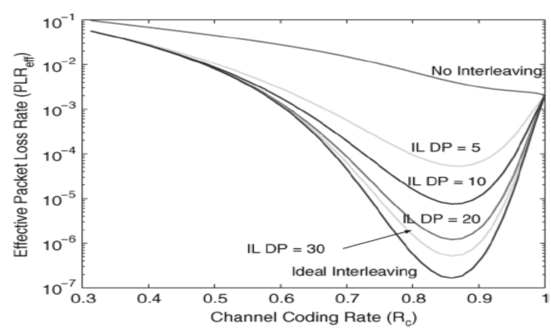


Fig 5: Illustration of block interleaving operation (interleaving depth=M1).

The above fig shows the case of deterministic arrivals(h=∞) with all other system parameters the same as in Fig. 6 Compared to Fig. 5 shows that for more deterministic source arrivals an increased coding rate Rc is required to achieve the optimum performance. Figure demonstrate that with interleaving the performance of FEC coding can be greatly improved, and interleaving with even larger depth can achieve increasingly improved performance.



(a)



(b)

Fig6: (a) Evaluate interleaving FEC performance Poisson arrival (b) Evaluate interleaving FEC performance deterministic arrival.

2. FEC Performance with Multiple Sources (N>1)

Here, we proceed to explore the FEC performance in case of multiple sources sharing the multiplexer. In order to facilitate the analysis, here assume in the packet arrival process seen by the multiplexer from each source is Poisson.

FEC Performance without Interleaving: In order to evaluate the FEC performance for one of the N sources, the block-error distribution P(j,n). For a single isolated source is required. Here we discuss different method to compute

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P(j,n) for the N*M/M/1/K. queue, which can be extended easily to incorporate the analysis for interleaving.

Analysis of Block-Error Distribution for a Single Source: Assume the packets arriving at the single-multiplexer come from N independent sources: S1, S2……Sn. indicates that, for N homogeneous sources with a fixed overall load ρ, the loss process of a single source becomes less and less correlated with increasing N

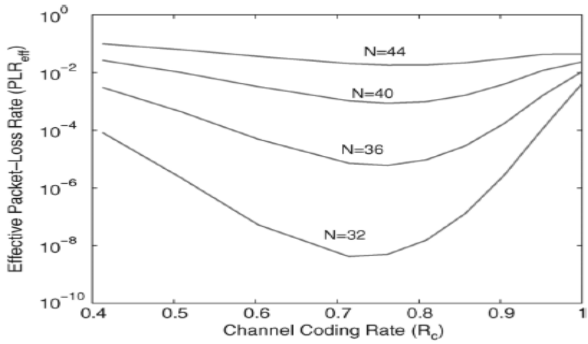

Fig7. FEC performance with N homogeneous sources; Poisson arrivals, load from each source fixed at ρi =0.02, K=10 block size n=63

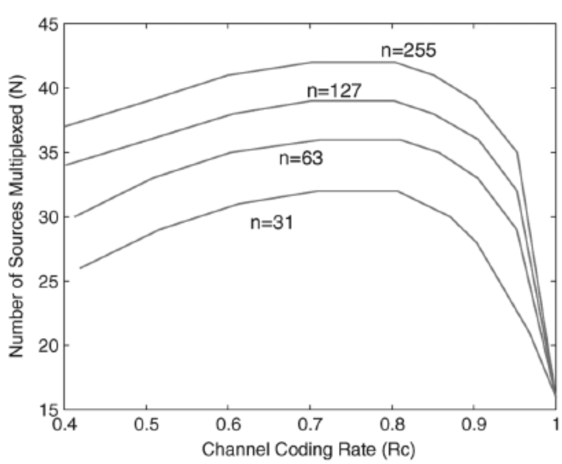

Fig8. Multiplexing gain achieved by FEC coding with different coding block sizes n; Poisson arrivals, the effective packet-loss rate fixed at PLReff= 10-6, load from each source fixed at ρi=0.02,K=10

Now look at the FEC performance in improving the Statistical multiplexing gain. Suppose the FEC coder for each homogeneous source applies an RS(n,k)code to the packets from the corresponding source coder. The channel coding rate remains Rc = k/n. As a result of the channel coding, the packet arrival rate into the single-multiplexer will increase to $\lambda^1 = \lambda i/Rc$. We assume that the average load from each source is fixed while the total load $\rho = \lambda/\mu$ changes with varying N.

FEC Performance with Interleaving: Now we suppose the packets from each homogeneous source are interleaved with the same interleaving depth before being transmitted into the network. The algorithm for computing the block-error distribution P(j,n) for a single source can be extended to include the interleaving procedure, as provided in previous Section. It can be expected that, compared to the case of a single source (N=1), the need for interleaving will be significantly reduced in a multiplexing environment (N ≥ 2), due to the already reduced packet-loss correlation as a result of the natural interleaving effect of multiplexing. Fig. 8 shows the to the already reduced packet-loss correlation as a result of the natural interleaving effect of multiplexing Depths, where the number of sources is N = 3 and the total system load is fixed at ρ =0.8 (Scenario 1) with buffer size K = 10. As expected, when, in order to optimize the FEC performance, an interleaving depth M ≥ 10 is required,

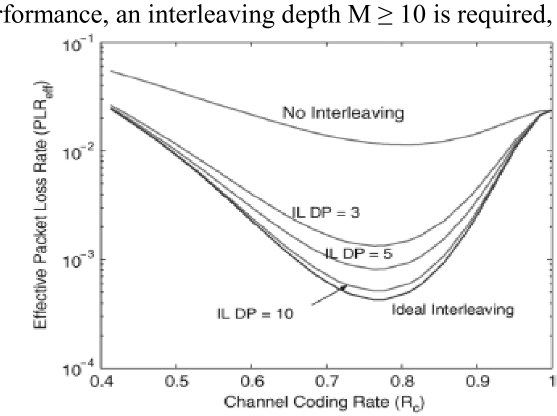

Fig9. Effect of interleaving on FEC performance with N=3 sources; Poisson arrivals, total load fixed at ρ = 0.8, K = 10, n=63.

FEC performance with different numbers of sources N for a given total load ρ=0.8. It shows that when the number of sources N increases, the need for interleaving depth decreases, which means reduced latency associated with the interleaving/de-interleaving operation. When N ≥ 14, interleaving makes an insignificant difference in FEC performance, because in this case the packet-loss process of each source is nearly independent.

3. Potential of FEC

Channel Model for Packet Transmission over Networks: Consider a channel model for packet transmission over a general packet-switched network. Assume a packet has M-bits. It is either transmitted and received by the receiver, or is lost due to network congestion or buffer overflow. For a

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received packet, bit errors may be introduced. Then packet transmission over networks can be modeled for coding purpose in terms of serial bit by bit transmission of M-bit symbols either over a binary symmetrical channel (BSC) with crossover probability ρ (state 0) or over a binary erasure channel (BEC) (state1),

Both of which are illustrated in Fig. 10 where ǿ is used to indicate the erasure symbol. A lost packet corresponds to the entire codeword symbol of m bits being erased, while a received packet means each of the m bits is sequentially transmitted over the BSC. This channel model belongs to the class of Block Interference Channels

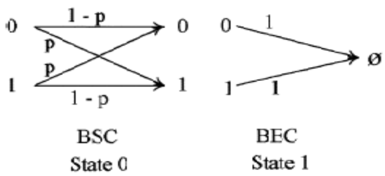

Fig10. Component channels of BIC corresponding to packet delivery and loss.

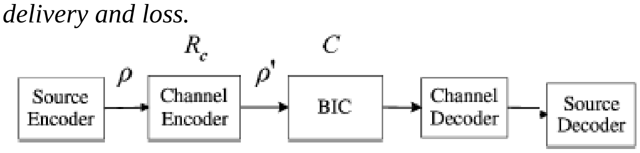

Fig11. Simplified communication system model.

Block Interference Channels (BIC), introduced by McEliece and Stark. Let S € {0,1} represent the state space of the BIC. If the state transitions are independent, then the Shannon capacity of the BIC is given as,

C = Es{Cs}; bits/transmission

Where Cs is the capacity of the component channel s € S, and the expectation is over the state space S. It Follows that

C = (1-ρ) * (1-H(p)); bits/transmission,

Where ρ is the probability of being in the loss state and H(p) is the binary entropy function,

H(p) = -p log p-(1-p) log (1-p); 0 ≤ p ≤ 1.

4. *Information Theoretic Bound On FEC*

Suppose the interleaving is ideal, and consequently the packet-loss process seen by the channel decoder is independent. If we consider the interleaver and the de-interleaver as components of the coding channel, then the channel, consisting of the interleaver, the single-multiplexer and the de-interleaver, can be modeled as a BIC with independent state transitions

Here we consider only the packet losses caused by the buffer overflows, and assume no bit errors, i.e., the BSC crossover probability p=0. Let PL be the packet loss rate of the single-multiplexer, so p= PL. Then

C=(1-p)*(1-H(p))=1- PL

Assume the source creates packets at rate λ and the packet service rate is μ. Then the normalized system load before coding is ρ = λ/μ. The channel encoder applies channel coding (not necessarily RS codes) with coding rate Rcto the source traffic.

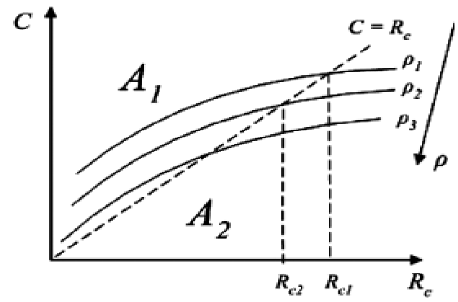

Fig12. Schematic illustration of the functional relationship C =1 – f (ρ / Rc), for different values of ρ with ρ1 ≤ ρ 2≤ ρ3.

Then the normalized system load after coding will increase to ρ1 = ρ/Rc. Given the buffer size K, the average raw packet loss rate PL depends only on the load ρ1, as expressed by PL = f(ρ1) = f(ρ / Rc)

Where the function f can be determined by queuing analysis of the single-multiplexer model C = 1 - PL = 1 – f(ρ / Rc)

For a given load, we can plot the functional relationship of C with Rc. The channel coding theorem establishes that any rate less than the channel capacity can be supported with arbitrary low error probability. In other words, with regards to our model discussed here, as long as the channel coding rate Rc is smaller than the BIC capacity C, the source rate can be supported with arbitrarily high reliability.

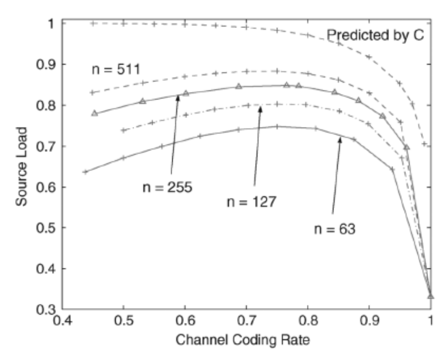

Fig13. The upper bound on the source loads ρ_{max} predicted by the channel capacity considerations, compared to the

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maximum source loads ρ_{max} that can be supported at a fixed effective packet-loss rate $PLR_{eff} = 10^{-5}$ using FEC coding; Poisson arrivals, ideal interleaving,

For the M/M/1/K model shows the upper bound on the source loads that can be supported as predicted by the preceding channel capacity considerations. It shows that with increasing coding block size the end-to-end performance achieved by FEC coding approaches that predicted by channel capacity.

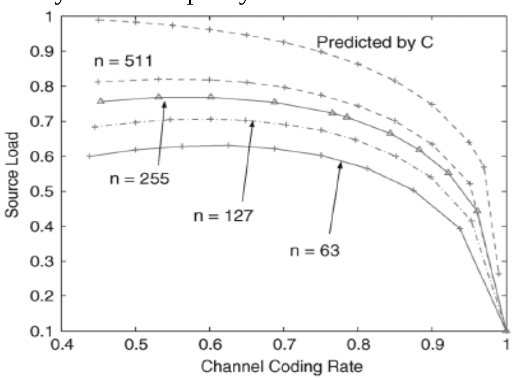

Fig14. The upper bound on the source loads ρ_{max} predicted by the channel capacity considerations, compared to the maximum source loads ρ_{max} that can be supported at a fixed effective packet-loss rate $PLR_{eff} = 10^{-5}$ using FEC coding; Poisson arrivals, ideal interleaving, using FEC coding; Poisson arrivals, ideal interleaving, K = 5.

The case of a smaller buffer (K = 5). The two figures indicate that, generally, the system with a larger buffer has a larger capacity. Note that in these two figures the capacity C is the capacity of the single-multiplexer combined with an ideal interleaver/de-interleaver, and not the capacity of the single-multiplexer itself. Actually, the capacity of the single-multiplexer channel can be greater than C described here since the capacity of the memory less interleaved channel is generally lower than the capacity of the original channel.

5. Conclusion

As the above analysis on the efficiency of FEC in packet losses in networks based on a single-multiplexer network model and explored potentiality of the FEC in recovering the packet losses occurred due to congestion at a bottleneck node of a packet-switched network, provided that the coding rate and other coding parameters are appropriately chosen. A discrete-time Markov chain model is used to analyse the efficacy of interleaving in improving the FEC performance and determined how much interleaving depth is required for FEC to approach the optimum performance. The implementation complexity of FEC coding and the corresponding coding/decoding delay also need to be considered, which an issue particularly important for real-time applications. One objective for future work is the analysis of the additional delay caused by the FEC coding, perhaps combined with interleaving/de-interleaving. Likewise, the application of FEC for network transport is limited by the time-varying and often uncertain error characteristics of the channel, which makes the appropriate choice of FEC coding rate difficult to determine. In real-world applications, FEC coders are required which can adapt the channel code rate to the time-varying channel conditions.

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