INTERLEAVING FOR PACKET CHANNELS

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ABSTRACT

The growth in connectivity and types of data links of the global network has made transmission by packets the main communication framework, and it will hold on for coming years. Real packet communication channels are characterized by packet losses and delays (e.g. due to congestion, queuing, etc.) that both affect performance of real-time applications. Distributing information over different packets can increase transmission reliability at the price of further delays. Packet interleaving can improve communication performance when some delay is allowed and re-transmission is not. The trade-off between further delay and loss robustness must be accurately evaluated. Proper choice and evaluation of packet interleaving strategies must be based on knowledge of an appropriate channel model. Several studies showed how losses and delays in real packet channels present memory and correlation. A Hidden Markov Model (HMM), that can be trained on-line, is used to characterize loss-delay channel behavior. A number of preliminary results are presented. The optimum solution is related to the learned loss-delay statistics.

1. INTRODUCTION

Transmission by packets, as Internet is growing as the global communication structure, is becoming progressively the main communication framework. Packet losses, due to congestion, or other causes, characterize most communication links and can introduce significant limitations to performing reliable real-time communication. It is well known that distributing information [7][10] over different packets can increase transmission reliability. Also some coding strategies, such as Multiple Description Coding (MDC), have been devised to distribute source information among packets so that also partially received information can be utilized for partial source recovery. Interleaving and/or scrambling of source information can be used at the price of further delays. When some delay is allowed, as imposed by the specific application, and re-transmission is not possible, packet interleaving can significantly strengthen communication. However, proper choice and evaluation of packet interleaving strategies, must be based on knowledge of an accurate channel model.

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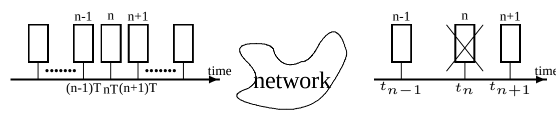


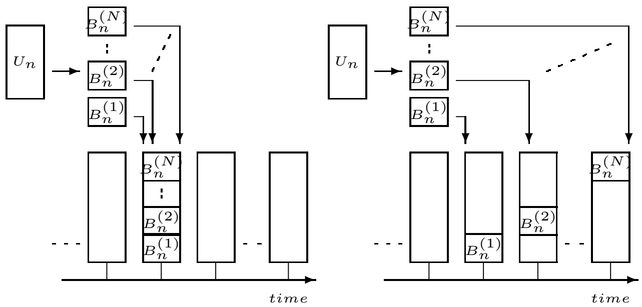
Fig. 1. End-to-end packet channel.

Several studies [1][3][4] showed how real packet channels are characterized by losses and delays that are strongly correlated. Some models have been proposed [5][6][11] in literature to account for loss and/or delay phenomena on real communication channels. If stationarity holds for sufficiently long time-frames, channel statistics can be estimated on-line by the receiver that, on a slow feedback link, can communicate them to the transmitter that adapts its coding strategy.

In this paper we assume that the channel is stationary and that channel statistics have been already estimated. For every specific application the maximum allowed delay (τ_{max}) has been chosen. Therefore, on the basis of a delay model, we have a total accounts of channel losses. By defining an appropriate distortion measure, which is a function of the total number of received bits, we study the results of optimizing packet interleaving proportions showing how channel loss statistics which are typically non uniform in time, affect total performance.

2. THE MODEL

Our reference model is shown in Fig. 1. Packets of size N_b bits are periodically transmitted over an Internet channel every T seconds. The network randomly cancels and delays packets according to current congestion.



(a) Minimum-delay loss-no-robust coding.

(b) No-minimum-delay loss-robust coding.

Fig. 2. Source coding.

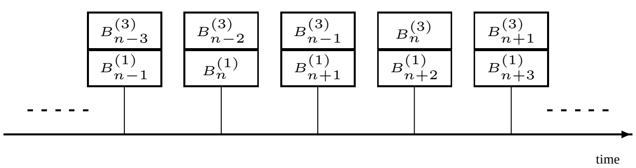


Fig. 3. Example of source flow where $N = 3$, and where $s_1 = N_b/2,\ s_2 = 0,\ s_3 = N_b/2$.

Every T seconds the source generates an Information Unit (U_n) to be transmitted by the n-*th* packet. We suppose that U_n is composed by N blocks $\{B_n^{(1)}, B_n^{(2)}, \ldots, B_n^{(N)}\}$ respectively of size $\{s_1, s_2, \ldots, s_N\}$ such that $s_j \in \{0, 1, \ldots, N_b\}$ and $\sum_{j=1}^{N} s_j = N_b$ (note that a block can also be empty). Every block $B_n^{(j)}$ partially describes, with a certain level of accuracy, the corresponding U_n. This can be an MDC scheme or simply the partition of the original data block.

Transmission of the N blocks $\{B_n^{(1)}, B_n^{(2)}, \ldots, B_n^{(N)}\}$ by the n-*th* packet involves the risk of totally losing U_n if a loss occurs. The question is: what is the best distribution into a number of different packets to protect information from losses? The trade-off must be found between improved packet delivery reliability and increased delay in the transmission, see Fig. 2. In real-time communications high delays are equivalent to losses, so we address the problem of the trade-off between losses and delays and the determination of the optimal structure for transmission.

The n-*th* packet can contain information about $\{U_n, U_{n-1}, \ldots, U_{n-N+1}\}$. An example of distributed blocks is shown in Fig. 3. The structure of the n-*th* packet is shown in Fig. 4.

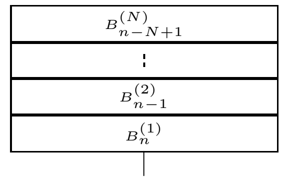


Fig. 4. Structure of the n-*th* packet.

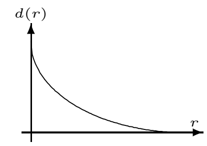


Fig. 5. Example of distortion function.

3. PROBLEM FORMULATION

For sake of simplicity we normalize bit-dimensions with respect to N_b, denoting $x_j = s_j/N_b$ the fraction of bits contained in the j-*th* block. We want to determine $\mathbf{x} = [x_1, x_2, \ldots, x_N]^T$, namely the *scrambling vector*, constrained to the following conditions,

$$x_j \in [0,1]_{\ j=1,\ldots,N} \quad , \quad \sum_{j=1}^{N} x_j = 1 \ , \tag{1}$$

such that the transmission is optimum, in the sense that it provides the best Quality of Service (QoS). The problem can be formulated as the constrained minimization

$$\mathbf{x}_{opt} = arg \max_{\{\mathbf{x}:Eq.(1)\}} \{D(\mathbf{x})\} \ , \tag{2}$$

where D is a measure of the QoS denoting the distortion affecting the information at the receiver. We suppose that the receiver can partially recover U_n if some of the N blocks $\{B_n^{(1)}, B_n^{(2)}, \ldots, B_n^{(N)}\}$ have been lost, and the distortion affecting decoded information depends only by the fraction of lost bits.

As previously noted, in real-time communications high delays are equivalent to losses, so in the following we say that the n-*th* packet has been erased if it has been lost during transmission or if its delay (τ_n) has grown higher than the maximum allowed delay (τ_{max}). Being $M = 2^N$, we will denote $\{\mathbf{e}(i) = [e_1(i), e_2(i), \ldots, e_N(i)]^T\}_{i=1}^{M}$ all the possible channel erasures configurations relative to transmission of N consecutive packets, where $e_j(i) \in \{0, 1\}$ indicates the presence (0) or the absence (1) of erasure for the j-*th* of the N packets. The fraction of received bits in case that the i-*th* erasures configuration occurred is

$$r(i) = \mathbf{x}^T \cdot \mathbf{e}(i) = \sum_{j=1}^{N} x_j e_j(i) \ . \tag{3}$$

The distortion function, characterizing the information coding, is a non-increasing function of the fraction of the

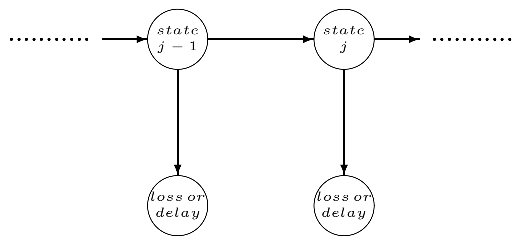


Fig. 6. An HMM for packet channels modelling.

received bits, $d = d(r)$. See Fig. 5 for an example. A reasonable choice for D is the average distortion at the receiver, $D = E\{d(r)\}$. Consequently Eq.(2) becomes

$$\mathbf{x}_{opt} = arg \max_{\{\mathbf{x}:Eq.(1)\}} \left\{ \sum_{i=1}^{M} \left[\pi_i \, d \left(\sum_{j=1}^{N} x_j e_j(i) \right) \right] \right\} , \tag{4}$$

where π_i is the probability that the i-*th* erasures configuration $\mathbf{e}(i)$ occurs.

To fix the ideas, in this paper, we use the distortion function $d(r) = 2^{-2r}$. Other similar function can be used as they depend on the application and on the specific coding strategy.

The optimum scrambling vector is found by the projected gradient algorithm, where the derivative of QoS measures are

$$\frac{\partial}{\partial x_k} D = -\ln(2) \sum_{i=1}^{M} \left[\pi_i e_k(i) 2^{-2\sum_{j=1}^{N} x_j e_j(i)} \right]_{k=1,\ldots,N} . \tag{5}$$

4. CHANNEL MODEL

In this paper we analyze the results of the optimization for a channel with correlated losses and delays with a Hidden Markov Model (HMM) structure [9][11]. Losses and delays are assumed to present memory and correlation, and they both are considered stochastically dependent on a hidden state, modelling the current congestion of the network, whose dynamic is ruled by a Markov chain. The channel is an HMM, see Fig. 6. The set of parameters characterizing the model is $\Lambda = \{\mathbf{A}, \mathbf{p}, \gamma, \vartheta\}$ where:

- $\mathbf{A} = [a_{i,j}]_{i,j=1}^{L}$ is the state transition probabilities matrix, i.e. $a_{h,k} = Pr\{current\,state\,is\,\text{“}k\text{”}|previous\,state\,was\,\text{“}h\text{”}\}$,
- $\mathbf{p} = [p_h]_{h=1}^{L}$ is the loss probabilities vector, i.e. $p_h = Pr\{loss\,occurs|current\,state\,is\,\text{“}h\text{”}\}$,
- $\gamma = [\gamma_h]_{h=1}^{L}$ and $\vartheta = [\vartheta_h]_{h=1}^{L}$ are the conditional delay vectors, i.e. in $state$ “h” delays are Gamma distributed with parameters γ_h and ϑ_h.

Due to the memory of the loss and delay phenomena we have

$$Pr\{\mathbf{e}(i)\} = Pr\{e_1(i)\} \prod_{j=2}^{N} Pr\{e_j(i)|e_{j-1}(i)\} , \tag{6}$$

where, denoting

$$\zeta_{h,j} = 1 - (1 - p_h) \int_0^{\tau_{max} - jT} f_h(t)\, dt , \tag{7}$$

and

$$\psi = [\psi_1, \psi_2, \ldots, \psi_L]^T , \tag{8}$$

the steady-state probability distribution, we have

$$Pr\{e_j = 0\} = \sum_{h=1}^{L} \psi_h \zeta_{h,j} , \tag{9}$$

$$Pr\{e_j = 0|e_{j-1} = 0\} = \frac{\sum_{h,k,=1}^{L} \psi_h \zeta_{h,j-1} a_{h,k} \zeta_{k,j}}{\sum_{h=1}^{L} \psi_h \zeta_{h,j-1}} , \tag{10}$$

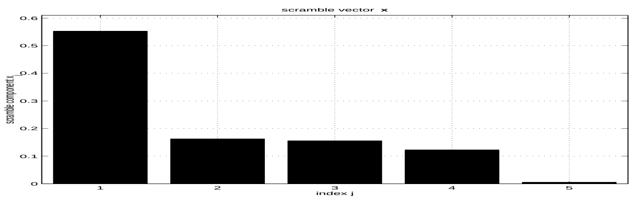
$$Pr\{e_j = 0|e_{j-1} = 1\} = \frac{\sum_{h,k,=1}^{L} \psi_h (1 - \zeta_{h,j-1}) a_{h,k} \zeta_{k,j}}{\sum_{h=1}^{L} \psi_h (1 - \zeta_{h,j-1})} . \tag{11}$$

Channel parameters can be estimated via the EM algorithm by use of measures of losses and delays performed on the link. More details can be found in [9][11].

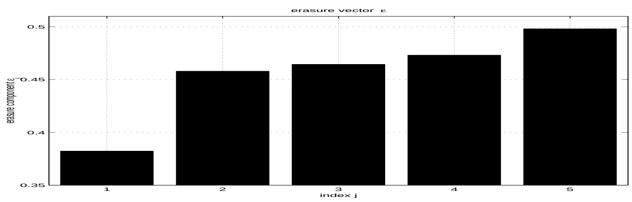
5. SIMULATION RESULTS

Optimum scrambling vectors were found by use of a projected gradient algorithm based on Eqs.(5). Simulations showed how a sort of *water filling* emerges as the result of the optimization. The scrambling vector $\mathbf{x}$ plays the role of the power resource while the role of the noise power is played by *the erasure probability vector* $\epsilon = [\epsilon_1, \epsilon_2, \ldots, \epsilon_N]^T$, where $\epsilon_j = Pr\{packet\,\text{“}j\text{”}\,is\,erased\,in\,\,a\,group\,of\,N\,packets\}$. The erasure probability vector can be easily computed by use of Eq. (6). Fig. 7 shows an example where the solution can be viewed as an almost-water-filling.

The dependency of the scrambling vector components from maximum allowed delay is seen as an increasing of allowed delay introduces more scrambling in the solution (there are less null components in the scrambling vector).



(a) Optimum scrambling vector.



(b) Erasure probabilities.

Fig. 7. Almost-water-filling solution for $N = 5$, $T = 50ms$, $\tau_{max} = 500ms$, and a Channel 1 with $\Lambda = \{0.1, 5, 30ms\}$.

This clearly shows the advantage of in distributing information in different packets. Fig. 8 shows an example.

Furthermore when maximum allowed delay is high enough, simulations showed how the scrambling vector assumes a configuration that could be associated to the formula $\mathbf{x} = \left[\nu, \frac{1-\nu}{N-1}, \ldots, \frac{1-\nu}{N-1}\right]^T$, where ν is decreasing with respect to loss rate p.

Loss rate of the channel also influences the optimum solution. It can be noted how increasing loss rate of the channel introduces more scrambling in the optimum solution, in the sense that information scrambling appears more sensible with respect to allowed delay. Fig. 9 shows an example.

The average distortion (D) versus loss rate was evaluated also for the simple scrambling configurations

- *no scrambling*:
  $\mathbf{x} = [1, 0, \ldots, 0]^T$,
- *split 1-2*:
  $\mathbf{x} = [1/2, 1/2, 0, \ldots, 0]^T$,
- *split 1-N*:
  $\mathbf{x} = [1/2, 0, \ldots, 0, 1/2]^T$,
- *flat scrambling*:
  $\mathbf{x} = [1/N, 1/N, \ldots, 1/N]^T$,
- *antilog scrambling*:
  $\mathbf{x} = [2^{-1}, 2^{-2}, \ldots, 2^{-N+1}, 2^{-N+1}]^T$,
- *log scrambling*:
  $\mathbf{x} = \frac{[log_2(N+2), log_2(N+1), \ldots, log_2(2)]^T}{log_2((N+2)!)}$.

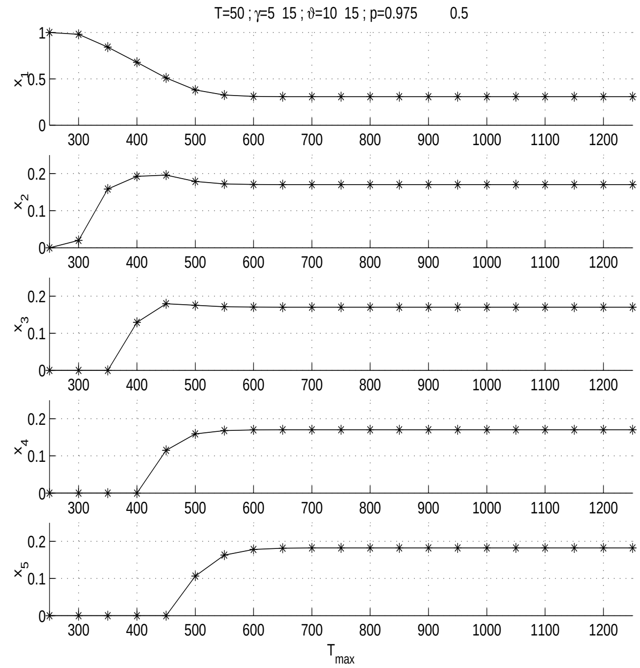


Fig. 8. Analysis of optimum scramble vector $\mathbf{x}$ depending on τ_{max} for $N = 5$, $T = 50ms$, and a Channel with $a_{1,2} = 0.1$,$a_{2,1} = 0.01$, $\gamma = [5, 15]^T$, $\vartheta = [10, 15]^T ms$, and $p = [0.025, 0.5]^T$.

They give some indications on how various distributions may work without solving the problem of Eq.4.

The distortion/loss plots are shown in Fig.10. In all cases scrambling the information among the various packets gives an improvement in performance. Split 1-2 and split 1-N seem to be reasonable but their results are not so impressive. Flat and log scrambling seem to achieve better QoS, while antilog scrambling is a little worst.

6. CONCLUSION AND FUTURE WORK

The presence of losses in real-time communication channels suggests the use of appropriate "packetizations" to increase transmission reliability. In this paper we showed some partial and preliminary results on how distributing information in time can improve the transmission QoS. Time, however, is a precious resource, especially in real-time communications, so that the trade-off between further delay introduced and loss robustness must be carefully evaluated within the maxim allowed delay imposed by a specific application. On the basis of a HMM-based channel model, that explicitly takes into account losses and delays and their strong correlation, we have have shown how interleaving may be a good strategy to fight against losses.

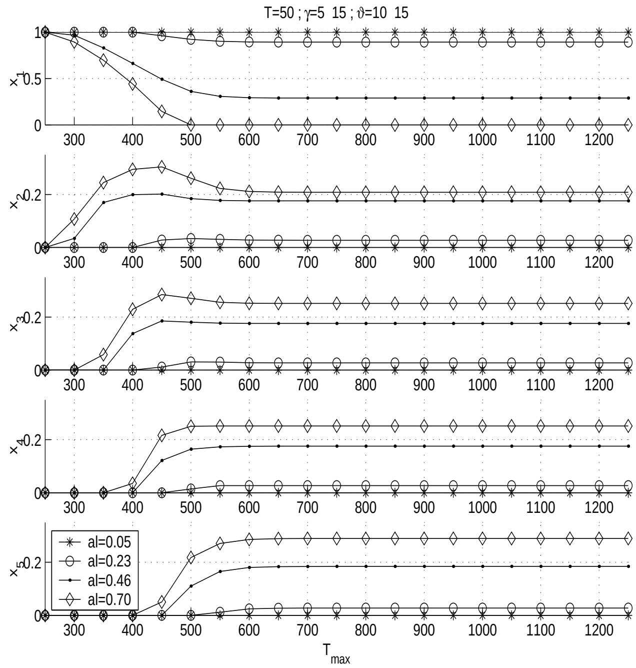


Fig. 9. Analysis of optimum scrambling vector $\mathbf{x}$ depending on p and τ_{max} for $N = 5$, $T = 50ms$, and Channel with $a_{1,2} = 0.1$,$a_{2,1} = 0.01$, $\gamma = [5, 15]^T$, $\vartheta = [10, 15]^T ms$.

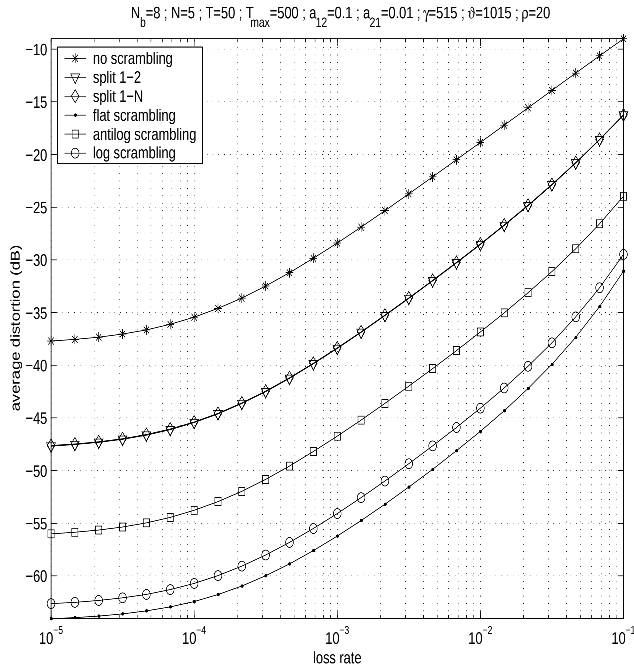


Fig. 10. Average distortion vs loss rate for $N = 5$, $T = 50ms$, $\tau_{max} = 500ms$, and a Channel 3 with $a_{1,2} = 0.1$,$a_{2,1} = 0.01$, $\gamma = [5, 15]^T$, $\vartheta = [10, 15]^T ms$, and $\rho = p_2/p_1 = 20$.

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