Quality Differentiation with Source Shaping and Forward Error Correction ⋆

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Abstract. The transmission of video traffic over the Internet is a fundamental issue of network design. Video applications require quality of service guarantees from the network in terms of limited packet loss, end-to-end delay, and delay variation. The question of today's research and development is how to provide these guarantees considering the architecture of the present Internet. In the last years a variety of admission control schemes based on per-hop or end-to-end measurements has been suggested to control delay and loss sensitive streams with very little or no support at the routers. Most of these solutions, however, have to apply the same acceptance threshold for all streams, a significant limitation considering the diverse quality requirements of the applications. In this work we investigate how source shaping and forward error correction (FEC) can be used together to achieve application specific quality differentiation in terms of end-to-end delay and packet loss probability. While source shaping and FEC have been proposed independently to decrease the probability of packet loss due to buffer overflow, their joint use has not been studied before.

As the two control functions use the same scarce resource, end-node delay, and their efficiency to decrease loss probability is proportional to the introduced delay but with a decreasing marginal gain, combining the two a better performance can be achieved than by using only one of them.

The performance evaluation focuses on the optimal delay allocation for shaping and FEC, such that the loss probability is minimized. We investigate how shaping can be used to substitute FEC redundancy and the sensitivity of the quality differentiation to the background traffic characteristics.

Keywords: Quality of Service, source shaping, FEC

1 Introduction

Internet is now considered as the universal network for future data, voice and video communications. It is recognized, however, that the best effort service implemented today is not satisfactory for delay and loss sensitive applications such as voice and video. It is widely accepted that quality provisioning for these applications requires (i) transmission and scheduling solutions that process best effort and QoS sensitive traffic in different ways, as reflected in both the IETF DiffServ and IntServ architectures; and (ii) call admission control for applications with strict QoS requirements, reflected in the controlled load and guaranteed service class proposed for the IntServ architecture. The question of today's research and development is how to implement these new functions through minimal changes in the architecture of the present Internet.

In the last years a variety of admission control schemes based on per-hop or end-to-end measurements has been published to provide admission control for delay and loss sensitive traffic with very little or no support at the routers. Measurement based admission control (MBAC) schemes base the acceptance decision on per-hop real-time measurements of the aggregate traffic intensities [1]. Endpoint admission control (EMBAC) schemes decrease the required router support even further, involving only the end systems in the admission control process [2–5]. The idea behind these schemes is to probe the transmission path from the sender to the receiver to experience the congestion level in the network and accept new streams only if the level of congestion is acceptable.

Most of these solutions, however, suffer from limited granularity, namely, the QoS guarantees (packet loss, delay and delay jitter) within a service class are the same for all streams [1, 3]. One way of quality differentiation is to define application specific service classes, but the management of a large number of classes would increase the complexity of the router operations. Instead, we believe that quality differentiation has to be achieved within one service class by using application specific traffic control functions at the end nodes.

In this paper we investigate how source shaping combined with forward error correction (FEC) can provide quality differentiation. Both of these functions exploit the streams' end-to-end delay limits looser than the one provided by the service class. Source shaping changes the traffic characteristics in a way that the expected packet loss probability of the stream decreases and the loss distribution becomes more even. FEC, in addition, recovers lost packets based on error coding, and consequently achieves lower perceived packet loss probability than the one ensured by the service class. The goal is then to share the delay available at the end-node – the difference between the acceptable end-to-end delay of the stream and the delay introduced by the network – between shaping and FEC such that the experienced information loss of the stream is minimized. The efficiency of source shaping and FEC for decreasing the probability of packet loss due to buffer overflow has been subject of extensive research, but the combined use of the two functions has not been investigated before.

In section 2 we discuss buffering strategies, explain the basic characteristics of source shaping and FEC and overview related work; in section 3 the combined use of shaping and FEC is described, in section 4 we evaluate the performance of combined source shaping and FEC and finally we conclude our work in section 5.

2 Network architecture and control functions

Buffering is the most straightforward solution to decrease packet loss probability in packet switched networks. Large buffers, however, introduce uncontrollable delay and delay variation and cause increasing burstiness on the transmission path. To utilize the advantages of large buffers when delay sensitive traffic is transmitted in the network scheduling solutions with per stream delay and jitter control have to be applied.

On the other hand, the choice of using small buffers for transmitting delay sensitive traffic allows simple (e.g., FIFO) scheduling at the network nodes, since delay and jitter are limited by the maximum buffer sizes, and makes the network tractable as stream characteristics do not change significantly at the network nodes [6, 7]. The size of the buffers has to be selected in a way that the contention of simultaneously arriving packets is resolved (i.e., packet scale buffering is provided instead of burst scale buffering [8]), that means a buffer size in the range of $\min\{C/p, n\}$ packets, where C is the transmission capacity of the link, p is the maximum bitrate of the streams and n is the number of input ports.

Traffic shaping at the source node is used to decrease the packet loss probability at the buffers inside the network by decreasing the burstiness of the traffic stream. As it is shown in e.g., [6, 9], (i) shaping even a part of the sources decreases average packet loss probability by orders of magnitude; (ii) shaped streams experience lower loss rates than unshaped ones, (iii) shaping, when applied to MPEG sources, decreases packet loss in loss sensitive I frames and (iv) makes the packet loss pattern more even as well, which in turn gives potential to FEC.

The performance of networks with source shaping and small buffers is analyzed in e.g., [6, 10, 11]. In [11] the performance of source shaping and buffering is compared for networks providing strict end-to-end delay bounds. The paper compares two solutions. In one of them, source shaping with the maximum acceptable delay is applied and nodes are equipped with small buffers, performing packet scale buffering. In the other solution the maximum acceptable delay is divided among the network nodes, thus nodes perform burst scale buffering with buffer size defined by the per node maximum delay. Nodes in this case apply jitter compensation. It is proved that source shaping outperforms buffering in the case of long transmission paths. In [10] the performance of these two solutions is compared considering video transmission, showing that shaping provides a visual quality similar to that of buffering even for short transmission paths. Source shaping in networks with small buffers is evaluated in [6] as well, proving that single buffer shapers are optimal in this case.

Proposals for source shaping algorithms address a variety of applications, like [12, 13] for streaming with known traffic pattern, [14] for lossy and [9, 15, 16] for lossless shaping for real-time traffic with unknown traffic pattern. The efficiency of shaping, in terms of decreasing the burstiness and consequently the packet loss probability depends significantly on the traffic stream itself. Considering MPEG coded video streams, shaping even with a very low, 20-40 ms delay is efficient, as it smoothes the data of large I and P frames. The efficiency increases with the introduced delay, but with decreasing marginal gain [9]. The above results motivate the use of source shaping combined with packet scale buffering for quality differentiation.

Forward error correction has been proposed to recover from information losses in real-time applications, where the latency introduced by retransmission schemes is not acceptable. FEC schemes increase the redundancy of the transmitted stream and recover losses based on the redundant information.

There are two main directions of FEC design to recover from packet losses due to buffer overflow. One solution, proposed by the IETF and implemented in Internet audio tools is to add a redundant copy of the original packet to one of the subsequent packets [17]. In the case of packet loss the information is regained from the redundant copy. This solution suits well interactive audio applications with low transmission rate and low delay limit. The efficiency of these schemes can be tuned by the number of redundant copies and the offset between the original packet and the redundant copy.

The other set of solutions uses block coding schemes based on, e.g., Reed-Solomon coding [18, 19]. In this case a block of packets is considered and error coding is applied for each bit position, generating a number of error correcting packets. The error correcting capability of Reed-Solomon codes with k data packets and c error coding packets is c if data is lost, which is the case if coding is used to regenerate lost packets. FEC based on block codes introduces an overhead of $(c+k)/k$ percent. Delay is introduced at the receiver only, where the error correcting packets have to be received for packet regeneration. The decoding delay is $(c+k)t_p$, where t_p is the packet interarrival time.

The error correcting capability of both classes of solutions increases with introduced decoding delay and overhead, with decreasing marginal gain.

The efficiency of FEC for correcting packet losses due to buffer overflow, however, is questionable due to the uneven distribution of packet losses and the additional load that FEC introduces in the network. Results considering different FEC schemes and based on analytical and simulation studies [17–19] show that the overall use of FEC does not always improve transmission quality, but FEC supports quality differentiation if only a part of the streams, requiring stringent QoS guarantees, applies error coding.

The above results indicate that both source shaping and FEC can be used for quality differentiation. The efficiency of the two functions is proportional to the introduced delay but with decreasing marginal gain. Consequently, combining shaping and FEC, by sharing the available end-node delay, a better performance may be achieved than by the use of only one of them.

3 Combined Source Shaping and FEC

In this work we propose the combined use of FEC and source shaping to support delay and loss sensitive transmission. If both functions are used at the end-nodes, the available end-node delay has to be split such that both functions can work efficiently. In addition, the two functions are not independent, as source shaping, by smoothing the packet losses in the stream, improves the packet loss correcting capability of FEC. It has to be noted that shaping achieves performance improvements without increasing the resource requirements of the streams, while FEC may introduce significant overhead. Thus, for some networking scenarios, FEC can prove to be an expensive control solution.

To evaluate the performance of combined source shaping and FEC we consider the following networking scenario.

We assume that the service class for loss and delay sensitive transmission uses dedicated buffer and link transmission capacities, and the applied call admission control together with FIFO scheduling at the routers provides the same bound on the packet loss probability for all streams. We also assume that only small buffers are applied at the network nodes, providing buffering for simultaneously arriving packets only.

This system architecture thus provides the same, strict upper bound on the network delay and the same, stochastic upper bound on the average packet loss rate for all streams. Sources can then utilize the available end-node delay – the difference between their maximum acceptable end-to-end delay and the delay introduced by the network – to decrease the packet loss rate of the stream, using source shaping and FEC. Note, that packet loss happens due to buffer overflow only. All packets arrive within the defined delay limit to the destination due to the use of limited buffers at the network nodes and at the source shaper.

Given the stream specific end-node delay D, which is divided between shaping and FEC as $D = D_{sh} + D_{FEC}$, the parameters of the FEC and the shaper are calculated based on c/k, the required FEC redundancy and m, the mean transmission rate, including the redundant packets.

In the case of video transmission FEC blocks that are entirely within a video frame do not introduce any delay, since all packets of the frame have to be received to regenerate a picture, the ones that spread over more than one frame however do. The delay introduced by these blocks is the time between the arrival of the last packet from the frame where the FEC block started and the arrival of the last packet from the FEC block. Based on the delay assigned to FEC the maximum FEC block length $k+c$ is defined by $(k+c)/m < D_{FEC}$.

The shaper rate is adjusted as described in [9]. When frame i arrives to the shaper, its size is added to the amount of data in the buffer $b(t) = b'(t) + f_i(t)$, where $b'(t)$ denotes the buffer occupancy before the frame arrival and $f_i(t)$ the size of the arriving frame. The shaper rate is then set to ensure that all data leave the buffer within the specified delay D_{sh}, thus $r(t) = b(t)/D_{sh}$. To avoid delay bound violation for frames stored in the buffer, the shaper rate can be decreased only if the buffer was empty before the arrival of the new frame.

Based on the aboves, the combined shaping and FEC algorithm works as follows. Frames generated by the source coder are put into the shaper buffer, the redundant packets according to the FEC scheme used are added and the shaper rate is adjusted. If the shaper rate during an FEC block transmission is lower than the average rate m, the FEC block is shortened by inserting an error correcting packet before schedule, to avoid the violation of the maximum FEC decoding delay D_{FEC}.

4 Performance Evaluation

In this section we evaluate how FEC combined with source shaping supports the transmission of delay and loss sensitive video streams.

The presented results are based on simulation. The simulated network model is shown in figure 1. The system includes traffic sources, channel coders doing FEC, source shapers and a multiplexing node with a single output link, modeling the transmission capacity dedicated for the controlled traffic. The multiplexing node performs simple FIFO queuing. We argue that results obtained with this simple model can be extended to the multiple node case, based on the fact that the traffic characteristics of the streams do not change as they cross nodes with small buffers [20].

For the simulations we use an MPEG-4 coded talk show trace – since MPEG coding is often used to transmit video streams nowadays – with an average rate of 540 kbps and a peak rate of 2.5 Mbps. The trace is approximately 2700 seconds, thus 67000 frames long. The frames of the MPEG trace are packetized to 188 bytes, as given for the transport stream in the MPEG-2 standard [IEC61883]. The capacity of the output link is 22.5 Mbps. The buffer at the multiplexer can store up to 10 packets, which is the ratio of the output link capacity to the peak rate of the individual streams, thus the multiplexer provides packet scale buffering. At full utilization there are approximately 38 streams competing at the multiplexer, depending on the FEC schemes used. The considered available end-node delays run from 60 ms to 120 ms, where the lower delays correspond to conversational while the higher to on-line streaming applications. The confidence interval of the presented simulation results is 5% or less at 95% confidence level.

The performance analysis investigates how the packet loss probability depends on the applied control functions and on the network load. The network load is defined as the ratio of the sum of the mean rate of the streams including FEC redundancy, and the link transmission rate.

For the sake of simplicity we use the notation CF(k,c,d) for a control function with FEC of block length of k data packets and c redundant packets, and shaping with a delay of d ms. For example, 80 ms end-node delay and a FEC scheme with k=20 and c=2 leave 20 ms delay for shaping for the considered stream mean rate and packet size. This control function is thus denoted as CF(20,2,20).

Combined Source Shaping and FEC To analyze the efficiency of FEC combined with source shaping we consider a scenario where the multiplexer serves a combination of traffic streams using a variety of control functions. The FEC redundancy is 10% or 20% and the available end-node delay is 80 ms or 120 ms. 14% of the multiplexed streams do not apply any control function (CF(1,0,0)), 14%-14% of them use a FEC scheme with 10% and 20% redundancy without shaping (CF(20,2,0) and CF(20,4,0)), then the same FEC schemes are used with 80 ms (CF(20,2,20) and CF(20,4,14.5)) and 120 ms (CF(20,2,60) and CF(20,4,54.5)) end-node delays. Figure 2 shows how the average uncorrected packet loss probability depends on the network load. The results show that FEC achieves loss differentiation of 1 to 2 orders of magnitude at the considered redundancy levels. Adding shaping, the loss probabilities further decrease with 25-50%. The reason for this improvement is twofold. First, the loss probability decreases as streams get smoother, second, for shaped streams the loss distribution becomes more even, increasing the efficiency of FEC.

Optimal Delay Allocation As the efficiency of both shaping and FEC is proportional to the introduced delay, splitting the available end-node delay between the two functions is an optimization problem. In this part we show some simple examples how different delay allocations affect the probability of uncorrected packet loss. Figure 3 shows a scenario where the link is shared between streams having an end-node delay limit of

90 ms, split between FEC and shaping. The overhead of the streams using FEC is 10%, the block length varies from $k = 10$ to $k = 30$, introducing different coding delays. Note, that for block length $k = 30$ no delay remains left for shaping (CF(30,3,0)). The figure shows that CF(10,1,60) is outperformed by those using larger blocks, however it is hard to distinguish between the streams using CF(20,2,30) and CF(30,3,0).

Considering MPEG coded streams, the distribution of losses in an MPEG stream may have significant influence on the percepted visual quality. Losses in I frames propagate forward to the next group of pictures, up to the next I frame and backwards to the previous P frame, losses in P frames propagate forward to the next I frame. Consequently, losses in I and P frames have increased effect on the percepted visual quality. Figure 4 shows the weighted loss probabilities [21] for the same FEC schemes as figure 3. The graph shows similar characteristics as figure 3, but here the CF(20,2,30), which leaves some delay available for source shaping, achieves the lowest loss probability (by a factor of up to 2), due to the more even loss distribution. Figures 5 and 6 show a similar scenario for a delay of 120 ms. The optimal delay allocation in figures 4 and 6 is consistent in the sense that it allocates almost one frame interarrival time for shaping while the rest for FEC. It allows efficient error control, while shaping makes losses for large I frames and small B frames even. On the other hand simulations run with an end-node delay of 60 ms give CF(20,2,0) as the optimal solution, showing that the difference between the efficiency of FEC with a block length of 20 and 10 is higher than what shaping with a delay of 30 ms can compensate for.

In addition, comparing the simulation results we see that the width of the confidence intervals depends on the control scheme. The 95% confidence interval for CF(30,3,30) streams is approximately one third of that of the streams using CF(40,4,0), that is, the difference between the loss probability of the streams applying the same combination of control functions is lower. It shows that shaping makes the performance of FEC more predictable.

Simulation results with increased FEC redundancy show similar characteristics, though the difference between the packet loss probability of the streams with and without FEC is higher.

FEC Redundancy and Shaping Delay The use of FEC may decrease the effective load due to the introduced overhead. Figure 7 shows the number of accepted streams with an admission control limiting the loss probability without FEC at 0.1%, for an increasing ratio of streams using FEC. Two cases are compared. In one of them the end-node delay is 90 ms, a part of the streams use this delay for shaping (CF(1,0,90)), a part of them for FEC with 10% redundancy (CF(30,3,0)). In the other case the end-node delay is 120 ms, the used control functions are CF(1,0,120) and CF(40,4,0). As shown on the figure, the effective load decreases as higher ratio of streams uses FEC. Increasing the end-node delay the number of accepted streams, thus the effective load becomes higher.

Since shaping and FEC have similar effects, it is worth investigating if shaping can compensate for decreased FEC redundancy. Figure 8 shows the weighted loss probabilities in a scenario where the bandwidth is shared among sources using FEC only and sources using FEC with a lower level of redundancy combined with shaping. Comparing the streams using CF(20,2,60) and those using CF(20,3,0) indicates that by shaping the FEC redundancy can be decreased from 15% to 10% to achieve the same loss probabilities.

Sensitivity Analysis Finally, we evaluate the sensitivity of source shaping and FEC with respect to the background traffic characteristics at the multiplexer. The sensitivity analysis is important, since these functions themselves do not give loss guarantees. Guarantees are given only by the call admission process, while applications can have some expectations how the performance improves with additional control at the end-nodes. Figure 9 shows packet loss values for different background traffic characteristics. The background traffic characteristics are changed by shaping the background streams with delays up to 120 ms, resulting in smoother background traffic at high shaping delay values. FEC controlled streams have 120 ms end-node delay in all cases. Two scenarios are compared. In one of them 25% of the multiplexed streams use CF(30,3,30), in the other scenario 25% of the multiplexed streams use CF(40,4,0). For both cases 75% of the streams give the background traffic without FEC. The network load level is constant 0.82. The figure shows the uncorrected loss probabilities for the background traffic and the FEC controlled streams. The loss probability decreases for all traffic as the shaping delay of the background traffic increases, the gap between the background traffic and the FEC controlled traffic is 1.5 to 2 orders of magnitude, increasing slightly as the background traffic gets smoother. Figure 10 shows similar scenarios at a load level of 0.87. Comparing figures 9 and 10 we see that the gain achieved by FEC and shaping slightly decreases as the network load increases, but is still higher than one order of magnitude. These results indicate that sources can have some expectations on the minimum performance improvements without information on the network load and background traffic characteristics.

5 Conclusion

In this paper we examined how FEC combined with source shaping can decrease the uncorrected loss probability and thus add quality differentiation capability to admission control schemes that provide the same loss and delay thresholds for all the accepted streams. As both of these functions introduce end-node delay, the question is how to divide the delay between the two functions. The presented simulation based analysis, considering MPEG coded video streams provided the following results:

- Considering multiplexed streams applying different FEC schemes and allowing different end-node delays, FEC decreases the average loss probability with 1 to 2 orders of magnitude at reasonable network loads. Using shaping in addition to FEC, the loss probability is further decreased by 25-50%.
- By splitting the available delay between source shaping and FEC one can achieve better percepted quality than by applying FEC only. The optimal sharing of delay between the two functions depends on the available delay and the efficiency of shaping and FEC for the specific stream characteristics.

– Source shaping combined with FEC can reduce the level of FEC redundancy needed to achieve a given loss probability, thus contributing to higher effective network utilization.
– The gain achieved by using FEC and shaping does not considerably depend on the background traffic characteristics, rather on the average load.
– Source shaping makes the performance improvement due to FEC and thus the quality differentiation more predictable.

The above results indicate that admission control giving identical delay and loss guarantees for all streams combined with stream dependent source shaping and FEC provides a solution for transmitting audio-visual information with diverse quality requirements without introducing stream specific control functions inside the network. The results also assist to define an algorithmic solution for selecting the optimal FEC redundancy and assigning delay to shaping and FEC, which is subject of our further research.

References

1. L. Breslau, S. Jamin, and S. Shenker, "Comments on the performance of measurement-based admission control algorithms," in *Proc. of IEEE INFOCOM 2000*, pp. 1233–1242, March 2000.
2. G. Bianchi, A. Capone, and C. Pertioli, "Throughput analysis of end–to–end measurement-based admission control in ip," in *Proc. of IEEE INFOCOM 2000*, pp. 1461–1470, March 2000.
3. L. Breslau, E. W. Knightly, S. Shenker, I. Stoica, and H. Zhang, "Endpoint admission control: Architectural issues and performance," in *Proc. of ACM SIGCOMM 2000*, pp. 57–69, August 2000.
4. V. Elek, G. Karlsson, and R. Ronngren, "Admission control based on end-to-end measurements," in *Proc. of IEEE INFOCOM 2000*, pp. 623–630, March 2000.
5. R. B. Gibbens and F. P. Kelly, "Distributed connection acceptance control for a connectionless network," in *Proc. of the 16th International Teletraffic Congress*, pp. 941–952, June 1999.
6. M. Reisslein, K. Ross, and S. Rajagopal, "Guaranteeing statistical QoS to regulated traffic: The single node case," in *IEEE Infocom'99*, pp. 1061–1072, 1999.
7. J. Roberts, "Traffic theory and the internet," *IEEE Communications Magazine*, pp. 94–99, January 2001.
8. *Broadband Network Teletraffic, Final Report of Action COST 242*. Springer, 1996.
9. G. Dán and V. Fodor, "On the efficiency of shaping live video streams," in *SPECTS'02*, July 2002.
10. G. Dán and V. Fodor, "Comparison of shaping and buffering for video transmission," in *NTS 16*, August 2002.
11. T. Wu and E. Knightly, "Buffering vs. smoothing for end-to-end QoS: Fundamental issues and comparison," in *IEEE Performance'99*, August 1999.
12. E. Knightly and H. Zhang, "D-BIND: an accurate traffic model for providing QoS guarantees to VBR traffic," *IEEE/ACM Transactions on Networking*, vol. 5, pp. 219–231, April 1997.
13. J.-Y. Le Boudec and O. Verscheure, "Optimal smoothing for guaranteed service," *IEEE Transactions on Networking*, vol. 8, December 2000.

14. H. Zhang and E. Knightly, “RED-VBR: A renegotiation-based approach to support delay-sensitive VBR video,” *ACM Multimedia system Journal*, May 1997.
15. S. S. Lam, S. Chow, and D. K. Y. Yau, “An algorithm for lossless smoothing of MPEG video,” *ACM SIGCOMM Computer Communication Review*, vol. 24, pp. 281–293, October 1994.
16. J. Rexford, S. Sen, J. Dey, W. Feng, J. Kurose, J. Stankovic, and D. Towsley, “Online smoothing of live, variable-bit-rate video,” in *Proc. of International Workshop on Network and Operating Systems Support for Digital Audio and Video*, pp. 249–257, May 1997.
17. P. Dube and E. Altman, “Utility analysis of simple fec schemes for voip,” in *Proc. of Networking 2002*, May 2002.
18. I. Cidon, A. Khamisy, and M. Sidi, “Analysis of packet loss processes in high speed networks,” *IEEE Transactions on Information Theory*, vol. IT-39, pp. 98–108, January 1993.
19. K. Kawahara, K. Kumazoe, T. Takine, and Y. Oie, “Forward error correction in ATM networks: An analysis of cell loss distribution in a block,” in *Proc. of IEEE INFOCOM 1994*, pp. 1150–1159, June 1994.
20. M. Reisslein, K. Ross, and S. Rajagopal, “Guaranteeing statistical QoS to regulated traffic: The multiple node case,” in *IEEE Decision & Control’98*, pp. 531–538, 1998.
21. K. Mayer-Patel, L. Le, and G. Carle, “An MPEG performance model and its application to adaptive forward error correction,” *ACM Multimedia*, December 2002.

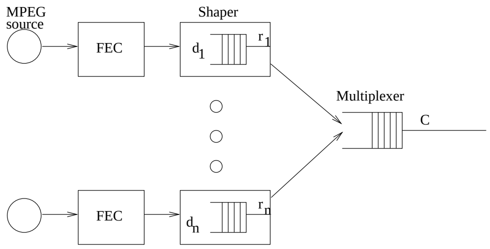

Fig. 1. The considered network model. MPEG source, FEC, delay limited shaping at the end nodes and packet scale buffering inside the network.

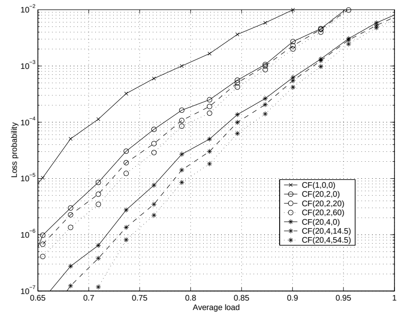

Fig. 2. Loss probability with FEC and source shaping, for FEC block size *k*=20, redundancies $c = 2,4$ and end-node delays 80-120 ms

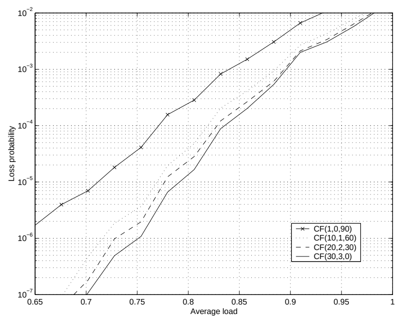

Fig. 3. Loss probability with FEC and source shaping for different FEC block sizes and 90 ms end-node delay

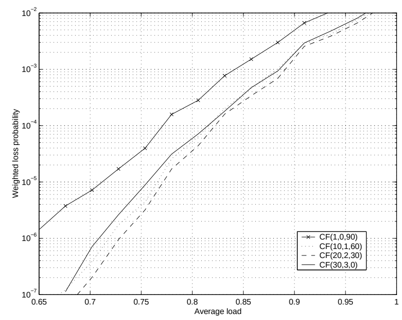

Fig. 4. Weighted loss probability with FEC and source shaping for different FEC block sizes and 90 ms end-node delay

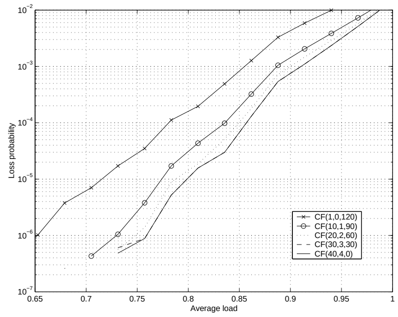

Fig. 5. Loss probability with FEC and source shaping for different FEC block sizes and 120 ms end-node delay

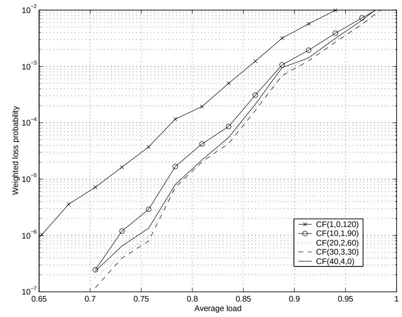

Fig. 6. Weighted loss probability with FEC and source shaping for different FEC block sizes and 120 ms end-node delay

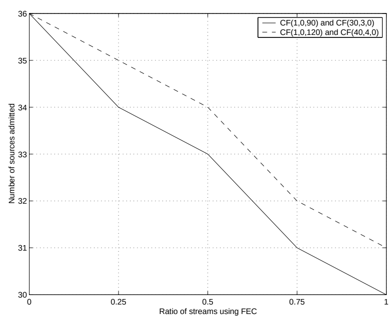

Fig. 7. Number of admitted sources vs the ratio of streams using FEC for different end-node delays

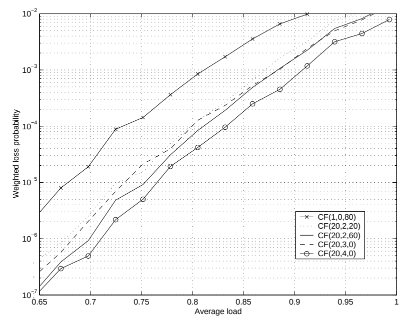

Fig. 8. Weighted loss probability for different combinations of redundancy and shaping delay

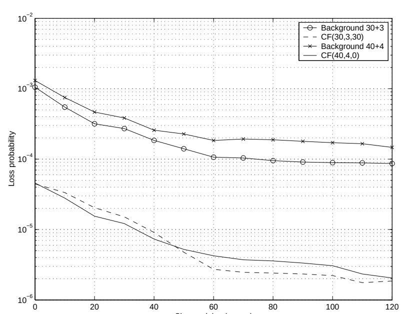

Fig. 9. Loss probability vs background traffic shaping delay for different FEC schemes, $\rho = 0.82$

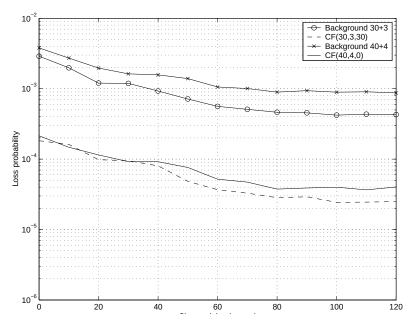

Fig. 10. Loss probability vs background traffic shaping delay for different FEC schemes, $\rho = 0.87$