Network-Embedded Channel Coding for Optimum Throughput of Multicast Packet Video

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Abstract— **Channel coding methods have played a key role in a wide range of video applications over unreliable networks. In particular, Forward Error Correction (FEC) schemes have been proposed and used successfully for packet loss recovery over the Internet, and especially for multicast applications. Traditional multicast video applications employ FEC on an end-to-end basis between the sender and the clients. However, the reliability and efficiency of end-to-end FEC-based packet video could suffer significantly over large video distribution networks. In this paper, we explore a new alternative for improving the reliability and efficiency (in terms of throughput) of packet video applications by optimum placement of few FEC codecs within large packet video distribution networks. We develop an optimization algorithm for the placement of FEC codecs within selected nodes of random packet-video networks. Based on extensive H.264 video simulations, we show that this approach provides significant improvements in video quality, both visually and in terms of PSNR values. These significant improvements are achieved while (a) maintaining a desired minimum source-video coding rate to all receiver nodes, and (b) avoiding any source-video rate shaping or complex transcoding within the network.**

Index terms- **FEC, Network Coding, Multicast, Peer-Peer, Video**

I. INTRODUCTION

Channel coding methods have played a key role in a wide range of video applications over unreliable networks. In particular, Forward Error Correction (FEC) schemes have been proposed and used successfully for packet loss recovery over the Internet, and especially for multicast applications. Traditional multicast video applications employ FEC on an end-to-end basis between the sender and the clients. However, the reliability and efficiency of end-to-end FEC-based packet video could suffer significantly over large video distribution networks. In this paper, we explore a new alternative for improving the reliability and efficiency (in terms of throughput) of packet video applications by optimum placement of few FEC codecs within large packet video distribution networks. We develop an optimization algorithm for the placement of FEC codecs within selected nodes of random packet-video networks. We show that this approach provides significant improvements in video quality, both visually and in terms of PSNR values. These significant improvements are achieved while (a) maintaining the desired source-video coding rate, and (b) avoiding any source-video rate shaping or complex transcoding within the network. Hence, our proposed approach is motivated, in part, by the following. First, for many practical realtime video applications, the sender needs to transmit and adhere to a minimum source-video rate. This, for example, could represent the bitrate of the base-layer of scalable video, or the rate of a minimum-acceptable quality non-scalable video stream. Second, for many applications, including ones with a large number of receivers, performing complex rate-shaping or transcoding operations may not be desirable or even feasible. Third, emerging and new network paradigms (e.g., [1]-[6]), such as overlay and peer-to-peer (p2p) systems, can facilitate the proposed framework for placing FEC codecs within realtime video distribution networks.

It is well known that the overall video quality is directly related to the effective video packet throughput that can be achieved with a given FEC channel-coding rate. However, for a given FEC coding rate (e.g., based on the popular Reed-Solomon FEC method), the packet loss ratio experienced by an end-to-end FEC-based video application could become very high when the number of nodes in the distribution tree increases. This naturally leads to a reduction in video packet throughput. One alternative for improving the reliability of end-to-end FEC solution is to lower the FEC coding rate (i.e., use more redundant packets and less video packets within an FEC block). However, this approach could lead to a significant reduction in the effective source-video rate[1]. Furthermore, and as highlighted above, the video application may need to adhere to a minimum source rate. This constraint could be expressed in terms of a rate-value k/n, i.e., the sender needs to maintain a transmission rate of k video packets over an n-packet transmission periods. Consequently, in the context of an FEC channel coding, a minimum of k message (video) packets must be included in an n-packet FEC block.

In this work, under a given FEC (n,k) block constraint (i.e., a k/n coding rate constraint), we seek to achieve optimum video-packet throughput by the placement of FEC codecs within selected (optimum) locations (nodes) of the video distribution network. In particular, we analyze and optimize the impact of *Network-Embedded FEC* (NEF) within real-time packet video networks. We develop a recursively optimum scheme for the placement of a small number of NEF codecs within any randomly-generated multicast video network of known (yet random) link loss rates. In essence, the proposed NEF codecs work as signal regenerators in a communication system, and hence, they can reconstruct the vast majority (and sometimes all) of the lost data packets without requiring retransmission and complex rate shaping and/or transcoding operations. Our theoretical analysis and simulation results show that a relatively small number of NEF codecs placed in (sub-)optimally selected intermediate nodes of a network can improve the throughput and overall reliability dramatically. This leads to the dramatic improvements in the overall video quality observed at the receiving nodes. Figure 1 shows an example of the proposed NEF framework.

As mentioned above, we envision the deployment of the proposed NEF framework in emerging networks such as overlay and p2p multimedia multicast systems (e.g., [1]-[6]). Overlay and peer-to-peer (p2p) networks are becoming increasingly popular for the distribution of shared content over the Internet. Most of the studies conducted for these networks have focused on multicast tree building. Further, these studies assume that reliable transport and congestion control are performed by the underlying end-to-end transport protocol such as TCP. However, this assumption is not appropriate for realtime multicast applications. More importantly, the deployment of FEC within these networks for realtime multimedia applications has received very little attention (if any).

Under the two types of networks considered here, "overlay" and "*p2p*" [1]-[6], multicast functions such as membership management and data replication are promoted to the application layer. Here, to distinguish it from a *p2p* network, an *overlay* network is equivalent to a *proxy-based* network[2] [4]. In a *p2p* multicast network, each node in the multicast tree can also be a multicast client (receiver). In an (proxy-based) overlay network, only the leaf nodes are clients. Within both networks, and at each intermediate node, data packets reach the application layer, and then get replicated and forwarded. Hence, in both cases (proxy-based or *p2p*), packet-loss recovery as an application level service can be placed in the intermediate nodes of the network.

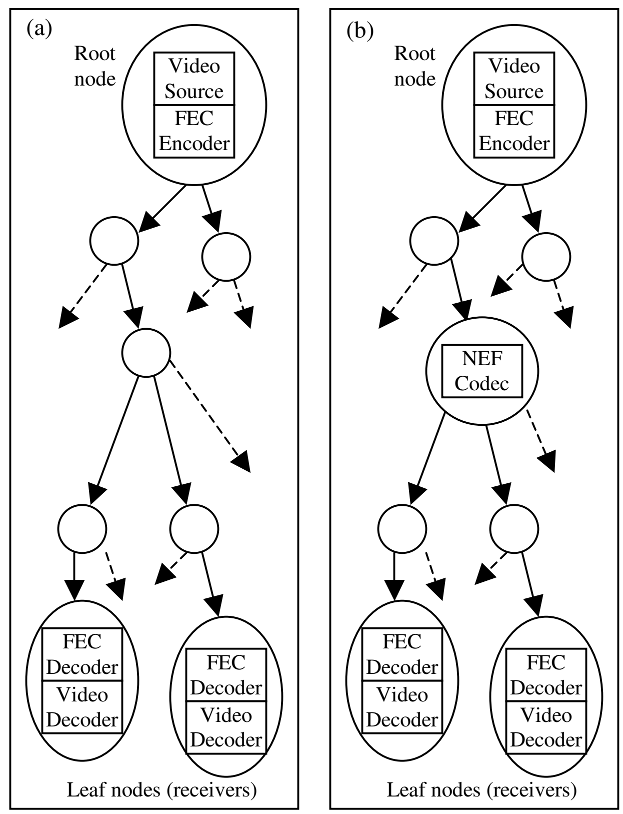


Figure 1 (a) Under traditional realtime video multicast, intermediate nodes and routers do not perform any FEC functions (b) A NEF codec in a video multicast tree can recover lost video and parity packets and transmit them downstream.

[1] Here, we are presenting arguments in the context of non-scalable packet video or the base-layer of a scalable video stream. However, the same arguments and motivations are applicable to each layer of a scalable video solution over packet networks.

[2] Please note that both *p2p* and *proxy-based* networks are forms of *overlay* networks [4]. In this paper, we use the term *overlay networks* to refer to proxy-based networks.

The remainder of the paper is organized as follows. Section II presents an analytical model for rate-constrained video throughput using Network-Embedded FEC within a multicast packet-video network. Section III describes and analyzes a recursive optimization NEF codec placement algorithm. Simulation results for reliability/throughput and for video quality measures are presented in Section IV and Section V, respectively.

II. Analysis Of Rate-Constrained Video Throughput Using Network-Embedded FEC

Analyzing the impact of FEC on packet losses has been an active research problem that was addressed by previous efforts. In particular, previous studies analyzed the packet-loss model for FEC-enhanced multicast trees (e.g., [7]-[8]). These studies are based on the IP multicast model, in which intermediate nodes do not participate in FEC. Here, we study the packet-loss model of a multicast tree when FEC codecs are placed in the intermediate nodes of a tree. In our analysis, we use the following notations:

T	A multicast tree with a root node r
$\lvert T \rvert$	The size (in terms of the total number of nodes) of a multicast tree T
T^c	A sub-tree rooted at some node $c \in T$ but does not include the node c.
T_l^c	The set of leaf nodes of T^c.
$\lvert T_l^c \rvert$	The total number of leaf nodes of a the sub-tree T^c
$P_v(i)$	Probability that node $v \in T$ receives exactly i packets.
$P_{v\vert v-1}(i,j)$	Probability that node v receives i packets given that its parent $v-1$ sends j packets.
p	the packet loss probability between the link from $v-1$ to v
(n,k)	The desired FEC block parameter pair (constraint) used by the system. n is the FEC block size, and k is the number of message (video) packets.
$RS(n,k)$	Reed Solomon code with k video packets and n-k parity packets.

Similar to previous studies, we assume a binomial distribution for the packet losses. For node v, if its parent $v-1$ sends j packets, the probability that it receives i packets is:

$$P_{v|v-1}(i,j) = \binom{j}{i}(1-p)^i p^{j-i} \tag{1}$$

When computing the probability $P_v(i)$ that a node v receives exactly i packets, we need to consider two cases; first, we consider the case when the parent node $v-1$ has no codec; second, we consider the case when the parent node $v-1$ has a NEF codec. If node v's parent does not have a codec, the probability that node v receives i packets is:

$$P_v(i) = \sum_{j=i}^{n} P_{v-1}(j) P_{v|v-1}(i,j) \tag{2}$$

Note that $P_{v\,|\,v-1}(i,j) = 0,\ \forall\, j < i$. In other words, node v can receive i packets only when its parent $v-1$ sends at least i packets. For the root node (r) of the tree, we define

$$P_r(i) = \begin{cases} 0 & 0 \le i \le n-1 \\ 1 & i = n \end{cases} \tag{3}$$

Equation (2) is a recursive function, and hence with the initial condition from (3), we can calculate the probability $P_v(i)$, for any node v in the multicast tree, that it receives exactly i packets. When a node has a codec for a $R(n,k)$ block, and if that node receives less than k packets and cannot decode the FEC block, it will just forward the received packets as usual; if it receives k or more packets, the node can decode the block and reconstruct the original data. It can also reproduce the lost parity packets. In fact, a codec can produce more or less than $n-k$ parity packets if desired; however, in this paper, we assume that the NEF codecs reconstruct the original data and reproduce the lost parity packets using the same $R(n,k)$ code. These packets are then multicasted downstream. (The design of NEF codecs with an adaptive FEC erasure codes is a problem that we are currently pursuing, and it is beyond the scope of this paper.)

A node that has a NEF codec and which receives $k \le j \le n$ packets will send n packets. If v is the immediate child of a codec, the probability that it receives i packets becomes

$$P_v^{'}(i)=\begin{cases}\sum_{j=k}^{n}P_{v-1}(j)P_{v|v-1}(i,n) & k\le i\le n\\ \sum_{j=k}^{n}P_{v-1}(j)P_{v|v-1}(i,n) & \\ \quad+\sum_{j=i}^{k-1}P_{v-1}(j)P_{v|v-1}(i,j) & 0\le i\le k\end{cases} \tag{4}$$

Once a node c is assigned a NEF codec, the probability $P_v(i)$ for all $v\in T^c$ will change and need to be recomputed. We use (4) to calculate $P_v(i)$ for the immediate children of the codec. For nodes that are not immediate children of a codec, the calculation of $P_v(i)$ is the same as equation (2).

Here we use P_v^{dec} to represent the probability that node v can decode a $RS(n,k)$ block:

$$P_v^{dec}=P_v(i\ge k)=\sum_{i=k}^{n}P_v(i) \tag{5}$$

We define the average decodable probability of a tree T for *p2p* and proxy-based overlay networks, respectively, as:

$$P_{avg}^{dec}=\frac{\sum_{v\in T-r}P_v^{dec}}{|T|-1} \tag{6}$$

and $$P_{avg-leaf}^{dec}=\frac{\sum_{v\in T_l^r}P_v^{dec}}{|T_l^r|} \tag{7}$$

If we use $r_d(v)$ to represent the number of received data packets (not including the parity packets received) of a FEC block at node v, then,

$$E[r_d(v)]=\sum_{i=k}^{n}kP_v(i)+\frac{k}{n}\sum_{i=0}^{k-1}iP_v(i) \tag{8}$$

Here we assume that for a $RS(n,k)$ block, if a node receives i packets, on average only $(k/n)i$ are data packets. For a *p2p* and overly networks, we define the data throughput as:

$$g=\frac{\sum_{v\in T-r}E[r_d(v)]}{(|T|-1)k} \tag{9}$$

and $$g_{leaf}=\frac{\sum_{v\in T_l^r}E[r_d(v)]}{|T_l^r|k} \tag{10}$$

III. Optimum Placement of Network Embedded FEC under a Rate-Constraint

In this section, we develop a mechanism for placing NEF codecs within a given network topology. In a large topology, identifying the optimum locations for the NEF codecs is not a trivial task. One objective is to place codecs in the intermediate nodes of a topology to maximize the average throughput. Assuming that the loss rate for each link in the topology and the number of codecs to be placed are known beforehand, the problem is similar to (but different from) the well-known *P*-median problem [9],[10]. A *P*-median problem is to find *P* locations in the network to place facilities in order to minimize the overall cost for servicing all of the nodes. Generally, in a *P*-median problem, the cost to serve a node is determined by the *weight* at the node and the distance between the node and its nearest available facility. The *P*-median cost has nothing to do with other facilities placed in the network. As we have seen in the previous section, in order to calculate the decodable probability and throughput, we need to know the locations of the codecs that have been placed on that path, not just the immediate codec that serves the node.

As the throughput at a node in a NEF network is impacted by all the codecs placed along the path from that node to the source (root), the dynamic programming approaches that have been used in previous network-placement problems (e.g., [10]) cannot be used to solve the NEF codec placement problem. Thus, we use a greedy algorithm to place *m* codecs in the multicast tree.

The greedy algorithm finds the best location for the first codec, then the next best location for the second one, and so on. Once a node is selected, an FEC codec is added to regenerate any lost data or parity packets. Let $T^c\subset T$ be the sub tree rooted at node $c\in T$ not including c. If c is set as a "codec node", only those nodes $v\in T^c$ will benefit from this selection; meanwhile, the "codec node" c itself will not be affected. For nodes $v'\in T-T^c$, everything remains unchanged. Let $E[r_d(v)]$ and $E'[r_d(v)]$ denote the average received packets for node $v \in T^c$ before and after node c is set as a codec node, respectively. We need to find $c \in T$ that maximizes the following:

$$\max_{c \in T}\left[\sum_{v \in T^c}(E'[r_d(v)] - E[r_d(v)])\right] \qquad (11)$$

A similar optimization objective function can be expressed for proxy-based overlay networks, except here the summation takes place over the leaf nodes only. Under the proposed greedy algorithm, we use an exhaustive search to find the best place for the first codec, after we find the optimum $c \in T$ node, we place the codec at that node. We use the same method to place the next codec; this process continues until all of the m codecs are placed.

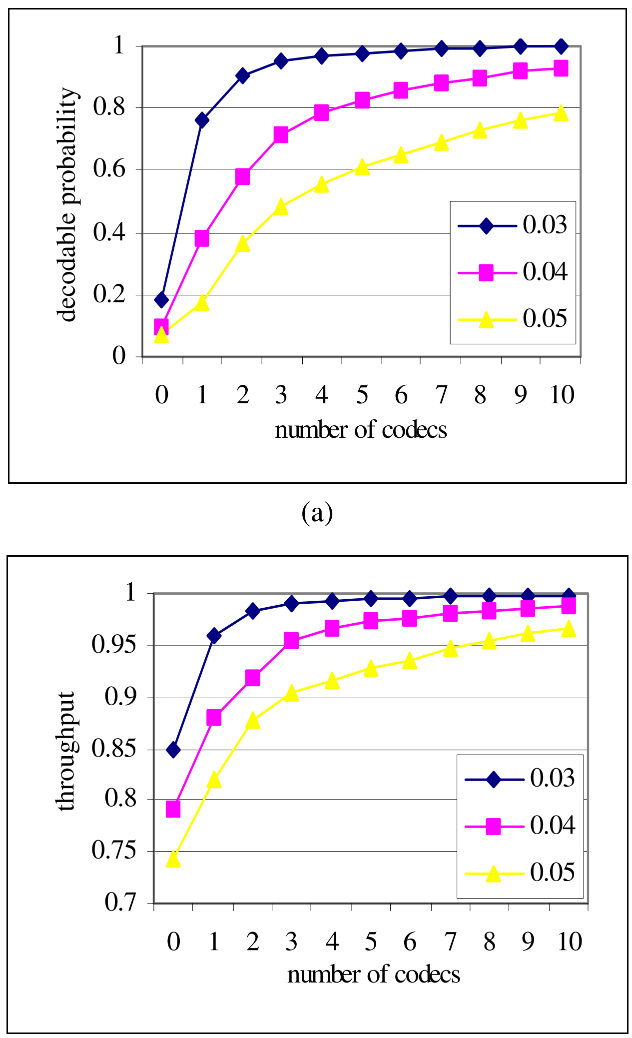
The proposed greedy algorithm does not guarantee a global optimum solution for the placement of the m FEC codecs. Nevertheless, its performance has been very close to the global optimum. Table 1 shows the performance (in terms of throughput) resulting from the placement of m=2 and 3 FEC codecs (within 100-node multicast trees) based on the greedy algorithm, and compares these numbers with the throughput of the actual optimum placement under three (average) packet-loss ratios (p) over the multicast trees' links. (More details on the simulations are presented in the next section.) It is clear from the table that the greedy algorithm provides an excellent set of (sub-)optimum solutions in all 6 cases covered in this example.

Table 1 Average Throughput: Comparison between Optimal and Greedy algorithm

Num of codecs	p=3%		p=4%		p=5%	
	opt %	greedy %	opt %	greedy %	opt %	greedy %
2	98.5	98.4	93.9	91.9	87.8	87.8
3	99.1	99.1	95.8	95.4	90.4	90.4

IV. THROUGHPUT ANALYSIS AND SIMULATION RESULTS

The throughput performance analysis presented above was applied to several random tree topologies. We use the popular Georgia Tech gt-itm [11] network topology generator to produce a set of ten 100-node transit-stub graphs. (Analysis and simulations with trees of larger sizes were also conducted. Here, we focus on the 100-node tree cases for brevity.) For each graph, we use Dijkstra's Shortest Path First (SPF) algorithm to produce a tree rooted at a randomly selected node. We used the greedy algorithm described in the previous subsection to place the NEF codecs in the multicast tree. The number of codecs was increased from 0 to 10. After each codec is placed, we calculate the improvement on average decodable probability and throughput. In addition to applying the above performance analysis on the ten 100-node trees, we used the network simulator2 (ns2) [12] with some modifications for the support of the proposed NEF codecs in intermediate nodes. We modified the simulator to allow packets to reach the UDP and application layers. We have implemented a FEC UDP agent and a FEC application in the simulator. The analysis and simulation results were virtually identical. Below, and due to space limitations, we only present the analysis results.



(a)

(b)

Figure 2 (a) Average decodable probability over all nodes. (b) Average data packets throughput over all nodes

As mentioned above, in a *p2p* overlay multicast network, nodes in the multicast tree are also end users, which often are placed at the edge of the Internet. Each hop in the overlay network often consists of several underlying physical hops. This implies that the loss rate of each hop could be higher than the loss rate of a backbone link in an IP multicast model. Here, we show results when the loss rate per-link is set to 3%, 4%, and 5%. These loss rates are in accordance with previous studies [13]. We studied the performance improvement under each of these loss rates for a variety of RS codes. Here, we present the results for $RS(255,223)$, which is a popular FEC code that has both hardware as well as software implementations. (The channel-coding rate[3] for $RS(255,223)$ is 87.5%.)

The average FEC block decodable probability and data throughput for each tree were evaluated. The results are the averages over all of the ten random trees that were analyzed. Figure 2(a) shows the average decodable probability (over all nodes in a *p2p* tree) when the loss rates are set to 3%, 4% and 5%. (Similar results where obtained for proxy-based overlay networks. These results are not shown in this section for brevity. Video simulations for both cases are shown in the next section.) When no codecs are added, the FEC block decodable probabilities are very low for all three loss rates. For example, if the link loss ratio is 3%, the average decodable probability is just 18.6%. As the codec number increases, we see a dramatic increase in the decodable probability. It can be observed that a relatively small number of codecs can increase the decodable probability significantly. For a 3% per-link loss rate, the first codec increase the decodable probability from 18.6% to 76%; the first 3 codecs increase the decodable probability to above 95%. When the number of codecs increases to 10, the decodable probability reaches 99.9%; this implies that we can use NEF to achieve a very high level of reliability while using a very high (i.e., efficient) channel-coding rate. The results for the throughput are shown in Figure 2(b). For an average *p2p* node, when no codecs are added, the throughput is about 85% with per-link loss rate of 3%. The first codec raises the throughput to over 95%. With only 3 NEF codecs, the throughput increases to 99%. For a typical video application, reducing the effective packet losses from 15% (85% throughput) to less than 1% (higher than 99% throughput) will naturally have dramatic improvements in the decoded video quality, both in terms of PSNR and visual perception.

Figure 3(a) shows the block decodable probability and packet throughput averaged only over the leaf nodes.

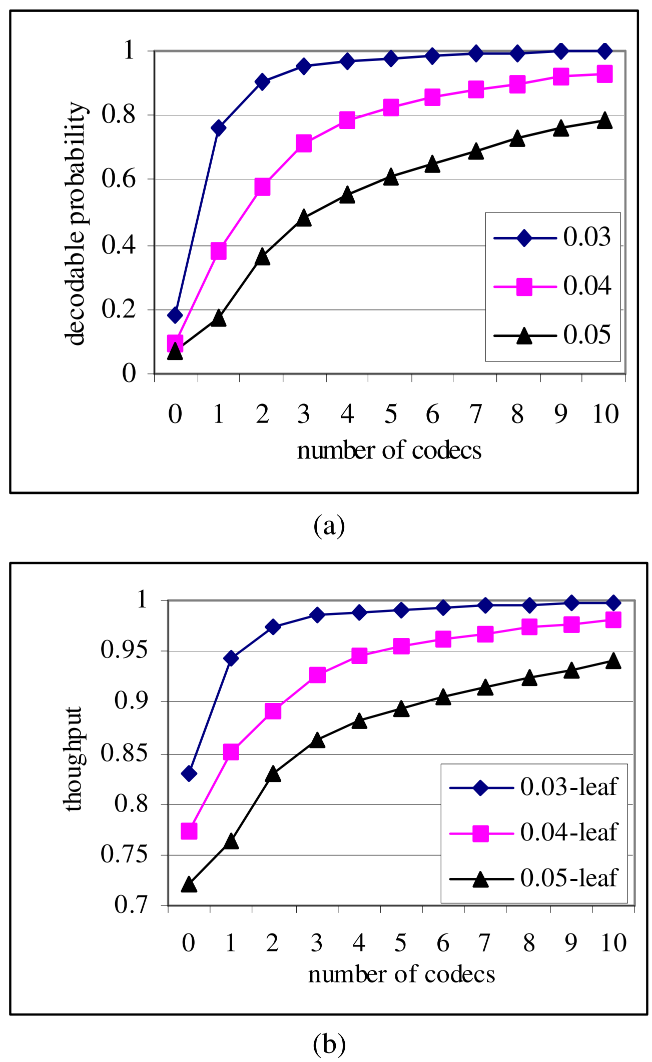


Figure 3 (a) Average decodable probability over leaf nodes. (b) Average data packets throughput over leaf nodes

[3] As we emphasized earlier, we target to achieve an optimum video-packet throughput solution under a given coding rate constraint. The above coding rate could represent one possible constraint value. Nevertheless, this channel coding rate may be high for some of the loss rates that are evaluated in this paper. However, it is important to note that the main conclusions of our study are valid regardless of the specific RS codes used. In particular, the proposed NEF framework can be used in one of two ways. Under one approach, a *given* RS code is already being used (on an end-to-end basis) prior to adding any NEF codecs. In this case, NEF can significantly improve the overall throughput as shown extensively by our analysis and simulations in this paper. Under another approach, a reliable communication infrastructure is already in place. This reliable infrastructure would be normally based on using very conservative (low) FEC rates (i.e., much lower than the effective end-to-end channel capacity). In this case, NEF can be used to significantly improve the efficiency of the RS codes by increasing its rate while maintaining the same level of reliability provided by the original infrastructure. In this paper, we focused on the first scenario to illustrate the benefits of the proposed NEF-based framework in the context of rate-constraint video applications.

The leaf nodes represent the set of nodes, which receive the source data after the maximum number of relays and thus receive the data with more losses than the overall average. It can observed that in the absence of a codec less than 20% packet blocks are received without any losses. However from Figure 3 (a) and (b) it can be observed that introduction of embedded FEC can provide remarkable improvement in performance even for the leaf nodes.

As we eluded above, under high losses, traditional end-to-end FEC could resort to a significantly lower FEC coding rate (to lower the packet losses and achieve high reliability). However, this reduces the effective source video rate significantly. In this case, NEF could be used to maintain the high reliability performance while increasing the FEC rate significantly (i.e., increasing the effective source video bitrate). Either way, NEF provides salient and dramatic improvements in the delivery of realtime video over *p2p* and overlay networks. Thus in the next section we provide H.264 based video simulations which will further substantiate benefits of NEF codecs.

V. VIDEO SIMULATIONS

Discussions in previous sections have concentrated on exhibiting the packet throughput improvements that can be achieved using NEF codecs. At this stage, it is necessary to clearly establish the advantage of using NEF in terms of the quality of video service available at the receivers. We use the emerging H.264/JVT [14] video standard for all the video simulations in this section. All the test sequences considered in this section have a "cif" frame size and are encoded at a frequency of 30 frames/sec. We use a constant quantization size of QP = 16 for all the video sequences. The results presented in this section are a subset of the examples we considered and thus it should be stated that the above choice of QP, frame frequency and frame size do not compromise on the generalness of the conclusions derived on the basis of the video analysis presented here. Unless specified no special error-resilience features are activated during the source encoding. Specifically, we do not turn on the "RD-optimization in presence of losses" feature in the JVT standard unless specified. Lastly, the encoded streams are made up of video packet of size 512 bytes each.

<u>Performance for diverse video sequences:</u>

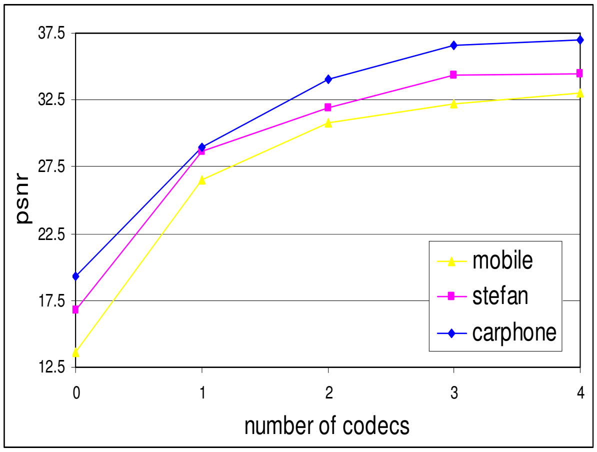
The test sequences that we consider are *mobile*, *stefan* and *carphone*. It should be mentioned that all the simulations are actually based on sequences that are multiple repetitions of the original test sequences. Our choice of test sequences represents a diverse set of source features, e.g. stefan is a sports sequence, carphone has comparatively high temporal correlation and mobile has multi-object motion. Thus based on the above described encoding parameters, a lossless stream of mobile, stefan, carphone exhibits a psnr value of 33.97, 35.47, and 37.75 dB, respectively.

Figure 4 shows the video quality of the above three sequences for a link loss probability of 0.03. It can be seen that significant improvement in video quality can be achieved by embedding codecs in the network. It can be observed that just adding 1-2 codecs can improve the performance by over 10db. The distortion levels continue to decrease as more codecs are introduced in the network. However, the distortion levels do not decrease by equal amounts on addition of a new codec. Thus in a practical scenario the number of codecs can be determined on the basis of the required quality of service guarantees. For example, in our simulations, although 4 codecs are unable to guarantee a 100% loss-less transmission, the distortion level is reduced to within 1dB of lossless when averaged over all nodes. As the leaf nodes represent peers that receive the data after maximum impairments, naturally the distortion observed by these receivers is more than the over all average. However compared with the quality of service received by leaf nodes in the absence of codecs, the improvement can still be termed as dramatic.

From Figure 4 it can be observed that the stefan sequence represents in some way the averaged behavior of mobile and carphone. On account of brevity of space we are unable to present the results for all the sequences and thus from this point onwards we concentrate all the analysis on the stefan sequence.

<u>Dependence on link loss probability:</u>

As done in previous sections, we evaluate the NEF performance under different link-loss probabilities, by evaluating the distortion in quality as seen by the receivers. Only results for the stefan sequence are presented. In order to describe the insight provided by Figure 5 we consider the following notation. Let m be the be number of codecs embedded in the network and let $Q(m,p)$ be the average video quality seen by the end user for a given link-loss probability p. Thus let $Q'(m,p) = Q(m,p) - Q(m-1,p)$ represents the utility of each incremental codec. It can be observed that $Q'(m,p)$ is a monotonically decreasing function with



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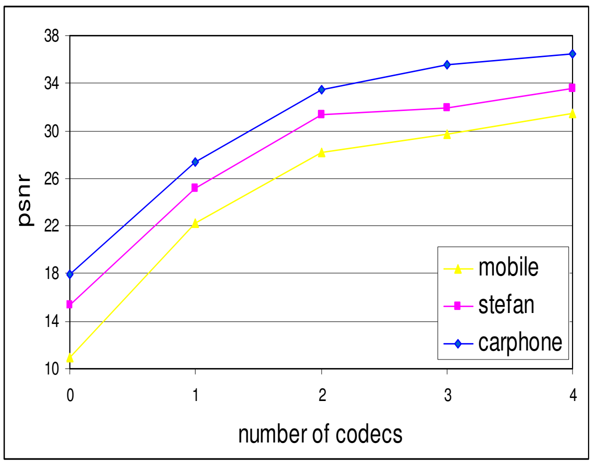

(b)

Figure 4 (a) Average psnr over all nodes. (b) Average psnr over leaf nodes. The simulations are based on a link loss probability is 0.03.

respect to *m*. That is the quality improvement on account of adding a new codec decreases as the number of already embedded codecs increase. Moreover, it should be noted from Figure 5 that the distortion decrement that can be achieved by a small number of codecs is more when the link-loss probability is low, e.g. $Q'(1,0.03) > Q'(1,0.04)$ for all nodes. However, as n increase, the distortion improvement for low link-loss probability saturates, but for high link-loss probabilities performance improvement on account of adding a new codec can still be substantial. e.g. $Q'(4,0.03) < Q'(4,0.04)$. Thus it can be concluded that utility of NEF even in very poor channel conditions can also be significant. However, as the channel conditions deteriorate the number of codecs have to be increased in order to maintain the quality of service.

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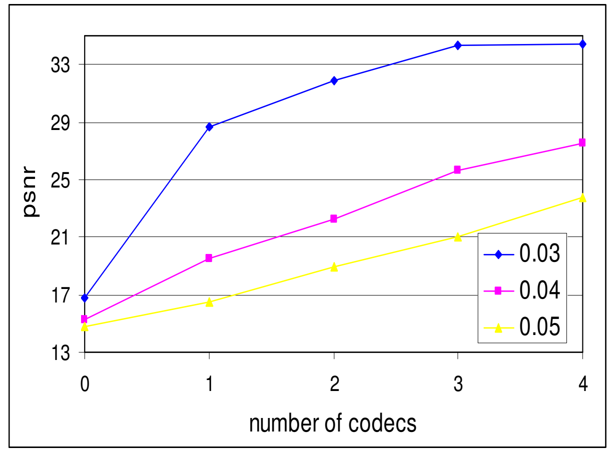
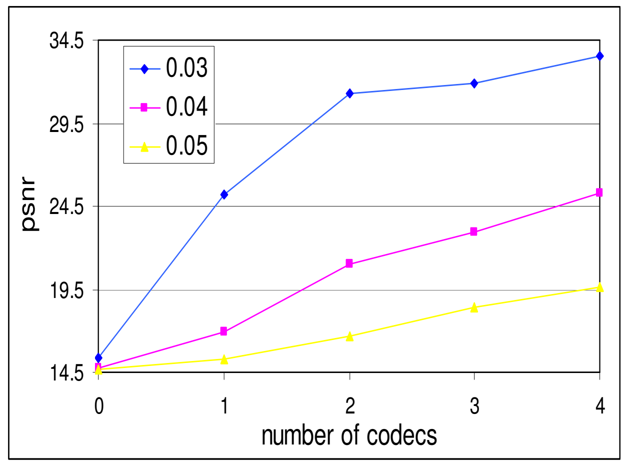


(a)



(b)

Figure 5 (a) Average psnr over all nodes. (b) Average psnr over all leaf nodes. All simulations are based on stefan test sequence.

However, as the channel conditions deteriorate the number of codecs have to be increased in order to maintain the quality of service.

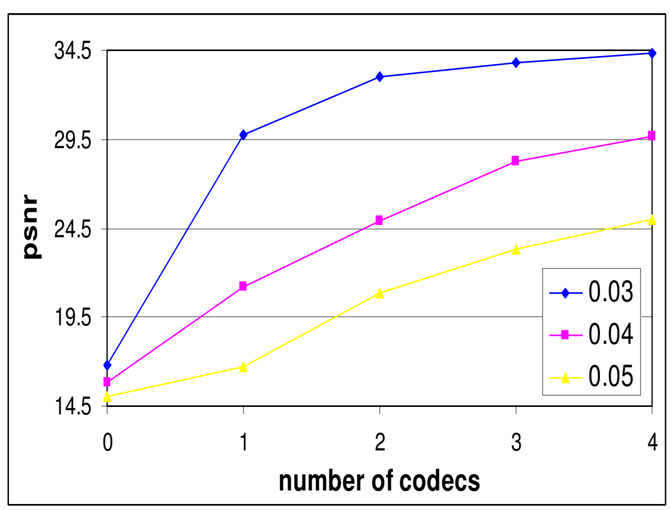
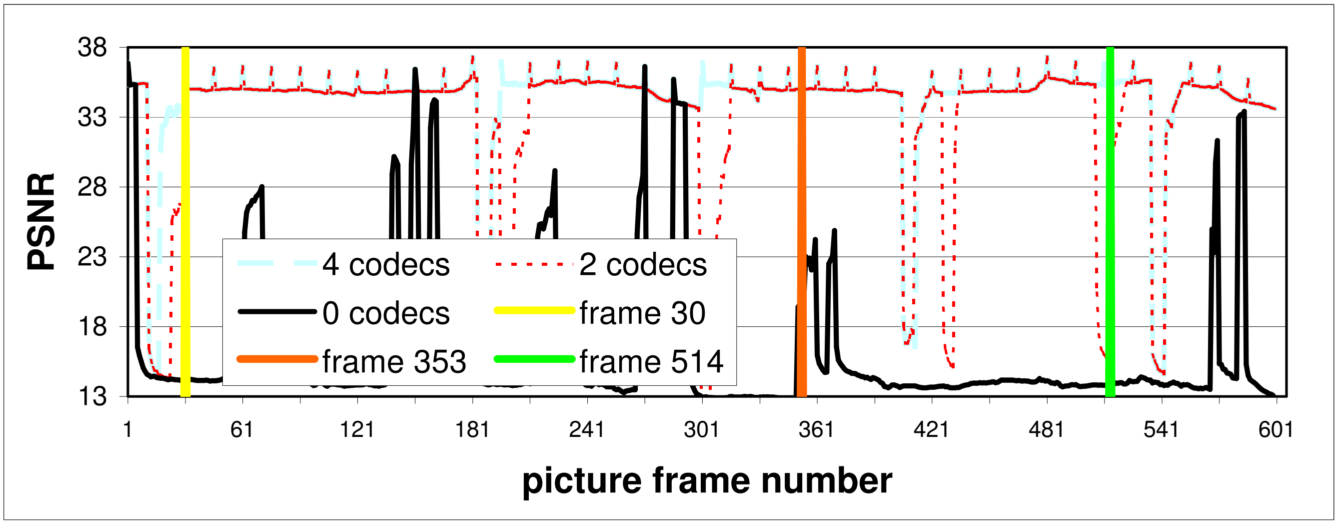
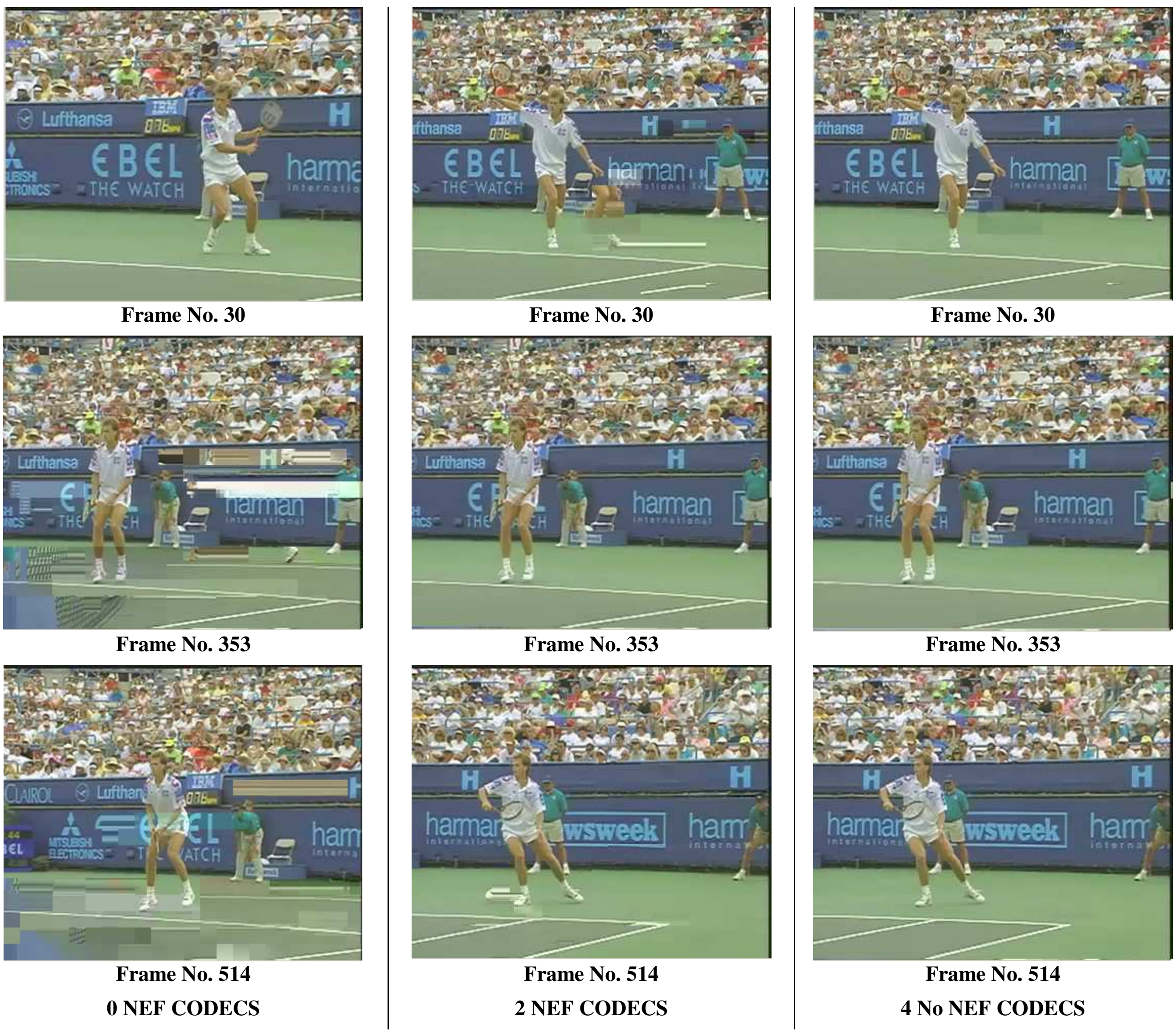


Figure 6 Average psnr over all nodes for a robust source encoding of the stefan sequence.



(a)



(b)

Figure 7 (a) Picture frame wise PSNR for a stefan sequence with robust source encoding. (b) A subjective comparison of distortion level with 4 codecs (right column), 2 codecs (center column) and with out any codecs (left column).

Dependence on source encoding robustness:

It could be argued that if a robust enough source code is used, then the advantage of using a NEF codecs might not be significant. Our results show that this is not at all the case. We use a robust encoding of the stefan sequence to present our results. We use the H.264 RD optimization feature to optimize the source encoding for a loss rate of 30%. The rest of the simulation setup is maintained as before. Observing Figure 6 it can be seen that NEF codecs continue to provide improvement over 10-15 dbs. However by comparing Figure 6 with Figure 5 it can be observed that the improvements are not as dramatic as in the case of a non-robust source encoding.

Subjective Video Evaluation:

As the relative improvement of the NEF scheme for the robust source encoding is lesser than that for non-robust encoding, we, on purpose, chose the "robust" stefan sequence to present our subjective results. This represents a minimal distortion reduction (i.e., minimum advantage) that can be achieved by using NEF. Figure 7 (a) is temporal video quality plot. When losses are incurred, the distortion in a video sequence suddenly increases, this is represented by the downward spikes in Figure 7 (a). It can be seen that as the number of codecs are increased the frequency of these downwards spikes are decreased. In fact, the overall video quality in absence of video codecs is so low, that we have upward spikes of quality improvements on account of intra-refresh or some other error resilience feature, instead of downward spikes.

Figure 7 (b) compares actual video frames to facilitate a clear subjective comparison. It is important to note that the choice of frames is not at all biased in favor of NEF (in fact it slightly biased against). Again it can be clearly seen that the block distortion level or loss of motion prediction is very high in the absence of codecs. As error concealment features use data from previous frames to reduce block distortion the primary artifact causing loss in video quality is not block-distortion but in fact jerkiness. The video was observed to be very discontinuous in absence of codecs and this discontinuity decreases as codecs are increased. As psnr is primarily determined by block distortions, it should be appreciated that improvement in video quality in terms of the viewing experience is even greater than as predicted by the psnr plots.

VI. CONCLUSION

In this paper, we explored a new approach for improving the reliability and throughput of packet video by optimum placement of FEC codecs within large packet video distribution networks. We developed an optimization algorithm for the placement of FEC codecs within selected nodes of random packet-video networks. We also demonstrated that this approach provides significant improvements in video quality, both visually and in terms of PSNR values. Our proposed approach has been motivated by (a) the need to adhere to a minimum source-video rate constraint that many practical video application require, (b) the need for avoiding complex transcoding operations that may not be feasible over large video networks, and (c) exploiting new and merging peer-to-peer and overlay networks that facilitate the proposed framework for placing FEC codecs within realtime video distribution networks.

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