# Adaptive FEC Technique for Multimedia Applications Over the Internet

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*Abstract*— **Forward Error Correction (FEC) is a common technique for transmitting multimedia streams over the Internet. In this paper we propose a new approach of adaptive FEC scheme for multimedia applications over the Internet. This adaptive FEC will optimize the redundancy of the generated codewords from a Reed-Solomon (RS) encoder, in–order to save the bandwidth of the channel. The adaptation of the FEC scheme is based on predefined probability equations, which are derived from the data loss rates related to the recovery rates at the clients. The server uses the RTCP reports from clients and the probability equations to approximate the final delivery ratio of the sent packets to the client after applying the adaptive FEC. The server uses the RTCP reports also to predict the next network loss rate using curve fitting technique to generate the optimized redundancy in-order to meet certain residual error rates at the clients.**

*Index Terms*— **Forward Error Correction (FEC), Reed-Solomon coder, network loss prediction, redundant bandwidth optimization.**

## I. Introduction

Streaming of multimedia over the Internet suffers many difficulties, because of the already installed equipments and protocols that support mainly data applications. Recently, many multimedia applications over the Internet infrastructure are taking place. Applications such as Voice over IP (VoIP), Video on Demand (VoD) and interactive gaming are utilizing the installed switches, routers and backbone of the Internet. Using the Internet infrastructure ensures multimedia applications with low cost to end users.

Although the TCP session guarantees the delivery of all the of the packets; it is not appropriate for on-line or interactive multimedia streams, because the out-of-order discard mechanism and NACK-retransmit technique generates unsuitable jitter of the play-back multimedia player at the client. The data of multimedia streams are attached to time, which means that the arrived packet is useful only if it arrived before the play-back time. So the Forward Error Correction (FEC) technique has been used for multimedia applications, which mainly depends on sending redundant packets that might be used to recover the lost packets [1-3].

### *A. Related Work*

Most of the FEC research concentrated toward bit error recovery, where the Reed-Solomon codec is extensively used for storage devices such as the Compact Disks (CD) [15]. Recent network installations suffer very low Bit Error Rates (BER), as low as $10^{-9}$ [15] in fiber networks. The main loss contributor in the Internet is due to buffering of packets and discard mechanisms in routers. Also multimedia applications suffer the out-of-time delivery of packets, due to the latency of arrived packets. The packet level recovery includes more complex computations due to the large arrays involved in the generation of large blocks of data. For a Reed-Solomon codec the computational complexity is still relatively low at the server, but it is much complex at the client side [7].

The FEC source techniques vary based on the application, Nonnenmacher *et al* [12] suggested a hybrid FEC and ARQ layer for time tolerance multicast applications, where the FEC is applied as the 1st layer, then the normal ARQ procedure will take place for lost packets after the FEC operation, Chan *et al* [4] also target the time tolerance video streams by introducing another selectively delayed replication stream rather than the FEC scheme, in order to achieve certain residual loss requirements. The work of Parthasarathy et. al. [13] presents another hybrid approach for high-quality video over ATM networks; by joining FEC technique at the sender side with simple passive error concealment at the client side, which in turn enhances the packet recovery even at high network loss rates. Yang et al [16] introduced a FEC scheme adapted for video over Internet, based on the importance of high spatial-temporal frame packets, and it its effect on further depending packets, so they send multiple redundancy levels based on the spatial content of the packets. Change, Lin and Wu [5] studied the FEC impact for CDMA 3-D systems over wireless ATM networks, so they presents two levels of FEC for header and payload packets, the header contains the Discrete Cosine Transform (DCT) information so it requires powerful FEC scheme to be reliably delivered, and the payloads will be transmitted with lower FEC protection. Song L., Yu M., Shaffer M. [15] present ideas for hardware designing blocks of Reed-Solomon coders.

### *B. Forward Error Correction*

In traditional FEC, the server adds n-k redundant (parity) packets to every k data packets, this yields n packets to be transmitted. At the client side, if any k packets from the n packets were received then the client


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can recover all the lost packets without retransmission. The amount of parities n-k is determined at the start of a session, where the redundancy is calculated based on a long-term average of network loss. The redundancy R is defined as the amount of parities n-k to the block of packets k as in equation 1 [12]

$$R = \frac{n-k}{k} \tag{1}$$

The generation of the extra parities requires mathematical codec. The codec must be reversible, so the client can reconstruct the lost data out of the received data. The Reed-Solomon codec is often used [6,7].

In this paper we adaptively optimize the parity packets n-k, in order to save the redundant bandwidth without degrading the quality of the displayed media. In our approach, the source generates the maximum allowed redundant parities n-k using Reed-Solomon encoder, but it only sends r parity packets that are required to overcome the expected network loss.

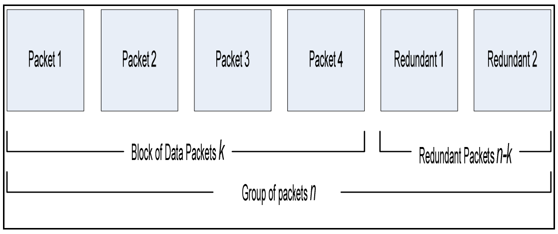


Figure 1. FEC group packets with $k$=4, $n$-$k$ = 2 and $n$ = 6

Our work of adaptive FEC shows a considered bandwidth saving over networks with low to medium loss rates without affecting the quality of the multimedia applications. That is, our scheme saves about 25% of the redundant bandwidth, which leads to more clients can subscribe to the same server, also the proposed scheme responds to networks with high loss rates by saturating to the maximum allowed redundancy, which corresponds to a best effort mode, where the adaptive FEC cannot save the bandwidth but can achieve the same quality of a Reed-Solomon FEC.

## II. BANDWIDTH OPTIMIZED ADAPTIVE FEC (BOAFEC)

The traditional FEC approach determines the redundant parities based on the long-term average of network loss, which is not suitable for multimedia applications where the loss is instantaneous. On the counterpart; generating on-the-spot parities is not possible since the source does not know the current loss rate at the client side, also the generation of adaptive parities involves more computational complexity. Also the long-term average of loss can miss lead the source to send more parities than required, which results in wasting the bandwidth.

To overcome the above weakness, we propose the BOAFEC approach where the source uses the long-term average only to determine the maximum allowed redundancy R. The BOAFEC predicts the current network loss using simple three-points curve fitting technique. The network loss prediction is used to determine the amount of parity packets r to be transmitted with the block of data packets k.

### *A. Probability Equations and Residual Loss Calculation*

The BOAFEC uses the Reed-Solomon encoder to generate the maximum allowed redundancy packets n-k, but only sends r parities. That is because the Reed-Solomon involves large arrays computations, and it is computationally efficient to design only one coder with fixed and maximum possible parities.

The BOAFEC predicts the network loss then uses the probability equations to calculate the expected loss at the client. And since packets suffer only two cases, whither delivered or lost, the Binomial Distribution is hold [7]. Assuming the loss of a packet is the event of success, and applying the Binomial distribution for a group of n packets. The probability of l packets to be lost from n packets, if the loss probability is πv will be:

$$P(Loss=l) = \binom{n}{l} * (1-\pi_v)^{n-1} * \pi_v^n \tag{2}$$

Since the group of n packets includes k data packets and r parity packets, the Reed-Solomon coder can recover up to r different packets from the lost packets. So the FEC coder has the property that if l packets were lost from the above group, then two cases apply:

l ≤ r: All packets will be recovered

l > r: None of the packets can be recovered

The FEC function of order r is defined as:

$$FEC_r(l) = \begin{cases} 0, l \le r \\ l, l > r \end{cases} \tag{3}$$

The Expectation function [8] of the number of lost packets is:

$$E(x) = \sum_{1}^{n} l \binom{n}{l} * (1-\pi_v)^{n-l} * \pi_v^{\,l} \tag{4}$$

And hence the Expectation function after applying the FEC is:

$$E(x) = \sum_{r+1}^{n} l \binom{n}{l} * (1-\pi_v)^{n-l} * \pi_v^{\,l} \tag{5}$$

The Expectation function presents the number of expected packets over a group of n packets, so new loss rate πv' which is the same as the residual loss rate ξr after applying the FEC operation is:

$$\pi_v^{'} = \xi_r = \frac{E}{n} \tag{7}$$

And so the delivery rate D is given as:

$$D = 1 - \pi'v \tag{8}$$

The above equations lead the source to calculate the residual loss rate ξr. The source will adaptively increase or decrease the redundant packets r, in order to meet the specified residual loss ξs at the client.

The number of packets k and number of maximum added codewords n-k will influent the Reed-Solomon encoder, the largest block size will result in more computational complexity. The L. Rizzo Reed-Solomon codec [7] is commonly used for software FEC coding.

### B. Network Loss Prediction

The network loss heavily fluctuates over time, so the average of long periods rarely presents the actual network status, Figure 2 shows the loss of a network running for 120 seconds, although Figure 2 presents the average of 500ms for each reading, it is still fluctuating over time. The long-term average loss of this run was 15.3%, while it is obviously deviate most of time on this average.

We present a simple network loss prediction based on the last three RTCP reports from the destination to the source. Every RTCP report contains the loss at the client at last transmission. The source constructs a matrix of the network loss, and using Gaussian Elimination method and Pivoting, the source can predict the next loss rate in a finite time. The source will then use the predicted loss to send the appropriate number of parity packets r, which can be used at the client to reconstruct the lost packets in order to meet the specified residual loss rate ξs.

### C. BOAFEC Procedure

The server and client negotiate at the start of the session to determine the network status, such as the round-trip-time and the long-term loss average. Also they negotiate to determine the parameters of the operation such as residual loss rate, number of packets k in each block and the maximum allowed redundancy. The procedure of operation for the BOAFEC is as follows:

***Server:***

- Allocate a certain bandwidth for each client at the start of a session based on it available resources and the multimedia application.
- Determine the k and n-k parameters, and hence the maximum allowable redundancy from equation (1) based on the long-term history of the network loss and the target residual loss ξs.

$$R_{\max} = \frac{n-k}{k}$$

- Start the transmission assuming the highest network loss, and so it uses the maximum redundancy.
- Wait for three RTCP reports in order to predict the next network loss rate.
- Calculate the optimal redundancy based on the probability equations then update the number of redundant packet for the next transmission.

***Client:***

- Ask for a reservation for a suitable bandwidth for the Multimedia application.
- Determine the acceptable residual loss rate ξs for the application.
- Specify a client window size based on the round-trip-time, the block of packets k and the Rmax
- Send RTCP reports

### D. The BOAFEC Parameter

The BOAFEC generates the redundant parities based on four parameters that are the Network Loss Rate (NLR), the maximum available redundant bandwidth, the maximum allowed jitter and the target residual loss rate ξs.

***Network Loss Rate (NLR):***

The NLR apparently is the main influence factor of the BOAFEC decision of the number of redundant parities. The BOAFEC cannot have control over the NLR, because the NLR corresponding to the network failures or congestions. Also the NLR is the only continuously changing variable of the four BOAFEC parameters, because the other three parameters are fixed at the start of a session.

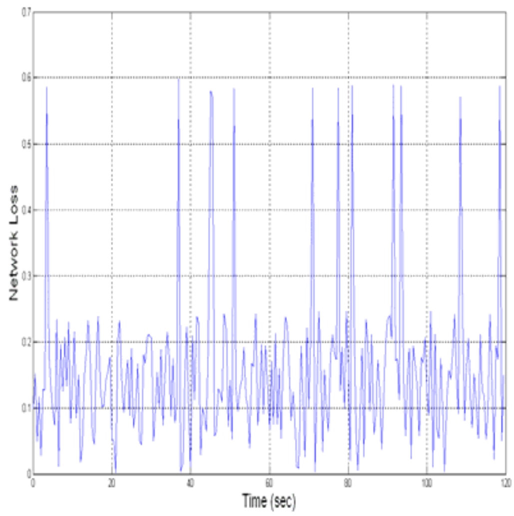


Figure 2: Raw Network Loss based on 120 seconds simulated network, every reading represents an average of 500ms.

***Maximum Redundant Bandwidth (MRB):***

The MRB determines how much redundancy can the server afford for the client as redundant codewords to be used later for recovering lost packets. The MRB inversely proportional to the residual loss rate. The MRB is defined as the number of redundant parity packets to the number of data packets, equation 1 represent the MRB.

The server determines the MRB based on the registered QoS for each client, the better QoS requirement needs more MRB to be associated to the data transmission, in –order for the receiver to reconstruct more lost packets.

***Maximum Allowed Jitter (MAJ):***

The MAJ represent the maximum jitter for a packet to be displayed in order for the media application to run smoothly. The MAJ indicates the scale of interactivity for the media application. Applications with higher inactivity requires less MAJ. The MAJ leads the media player to determine when the player should discard a delayed packet.

***Target Residual Loss Rate (ξs):***

The target residual loss rate ξs is the maximum tolerable loss rate for a multimedia application in order to run smoothly. Since the multimedia applications cannot tolerate the excess delay generated from the retransmission of the lost packets, there must be a residual loss even when using the FEC. The residual loss can further be reduced by using receiver based error correction techniques like the Interleaving or repetition of lost packets [14].

## III. NETWORK LOSS BEHAVIOR

The network loss behavior over the Internet is very complex to be defined, because of the many variables that cannot be predicted nor defined. Although the loss over a channel is simply figured by the number of lost packets related to the total number of sent packets, but the sequence or probability of a packet to be lost is not simply defined.

### *A. Gilbert Model*

A well known approximation of network loss is the Gilbert-Model, which uses Two-State Markov chain to represent the end-to-end loss. The Gilbert model is widely used to simulate the Internet loss, due to its simplicity and mathematical traceability [9][12][16]. The Two-State Markov is shown in Figure 3. The 0 state represents a packet was lost, where the 1 state represents a packet reached the destination.

Let p denote the transition from the 0 state to state 1, and q denote the transition from state 1 to state 0, so the probability of losing a packet is (1-q), and the probability to lose n consecutive packets equals (1-q)qn-1. From the Markov Chain transition matrix [16], the long run loss probability π can be defined as:

$$\pi = \frac{p}{p + q} \tag{9}$$

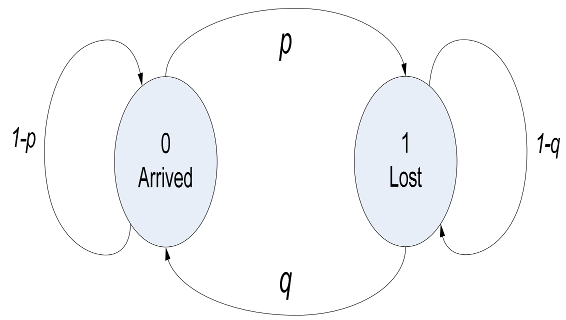


Fig ure 3: Two-State Markov Model

## IV. SIMULATION RESULTS

In this section we present the simulation results for the BOAFEC scheme, we show the response of the BOAFEC with different MRB and how it effects the overall residual loss ξs and the used redundancy. Also we show the BOAFEC results and compare it with the pure Reed Solomon FEC (RS FEC) and a replicated stream approach.

The MRB is very important factor for improving the recovery rates at the clients, whereas the server must bound the MRB in order to determine the number of clients that can be attached to it at once. The MRB relates directly with recovery rates at the clients, and hence it relates inversely with the residual loss. Using the BOAFEC can let the server to assign more MRB to clients since the BOAFEC scheme optimizes that redundancy.

Figures 4 and 5 show the response of the BAOFEC to different NLR when the MRB are varying. Figure 4 shows that increasing the MRB results in better residual loss rates at the clients, but it also shows that increasing the MRB up to a certain level for networks with low loss rates, such as when the NLR = 0.05; where increasing the MRB over the 0.2 results in slight reduction of the residual loss at clients, this response was due to the BOAFEC response when the residual loss matches the target loss ξs. Figure 5 presents the relation of the MRB to the real used redundancy, note that the BAOFEC optimizes the redundant bandwidth so the parity packets are used as much as possible.

We finally compare the BOAFEC with the traditional Reed-Solomon FEC and with the replicated stream FEC scheme. The replicated stream approach simply sends every packet twice, this increases the probability that at least one of the two packets could arrive. Apparently the replication stream requires 100% of redundancy, with the lowest processing overhead over the known FEC. The replicated stream is very useful for devices with low processing resources, also it is suitable for networks with high loss rates.

The Reed-Solomon codec was simulated using the parameters k = 30, n-k = 15 and hence n = 48. The BOAFEC parameters was residual loss rate ξs = 1%, k = 30 and maximum redundancy = 0.6.

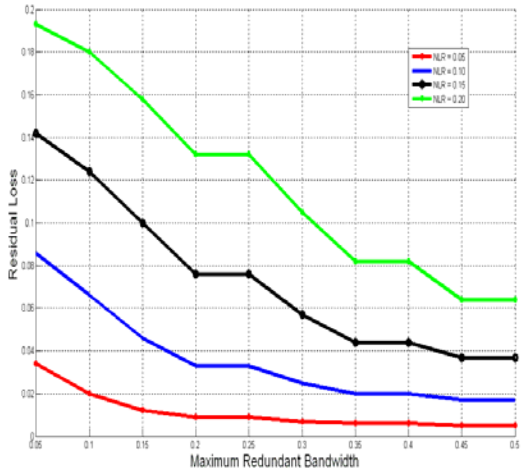


Figure 4: Residual loss versus the MRB for different NLR, k = 20 and ξs = 1%

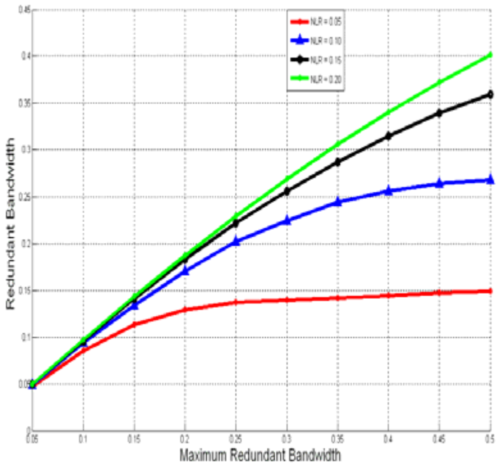


Figure 5: The used redundant bandwidth versus the MRB for different NLR, k = 20 and ξs = 1%

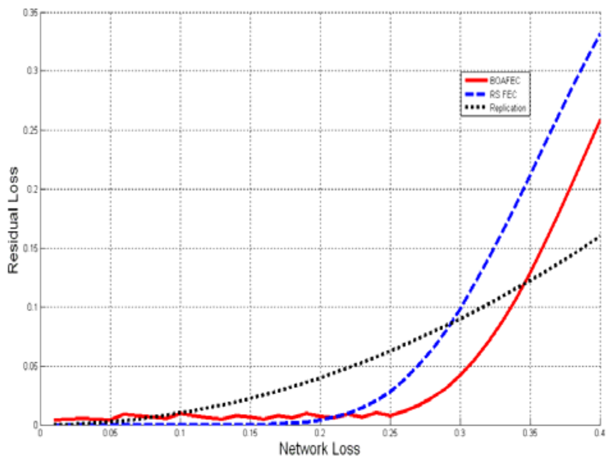


Figure 4: Residual loss versus the network loss. k = 30, n-k = 18, ξs = 1% and maximum redundancy = 60%.

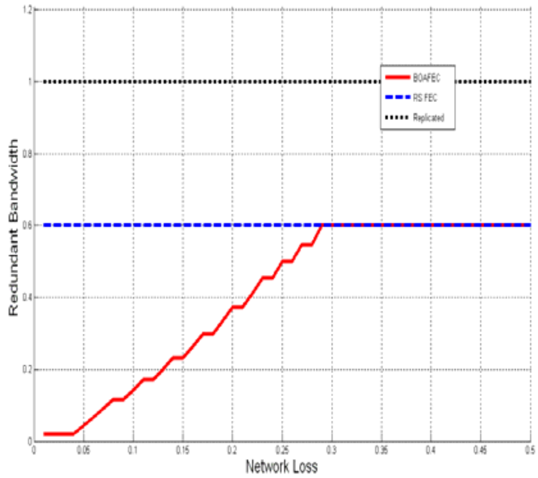


Figure 5: Redundant Bandwidth versus Network Loss. k = 30, n-k = 18, ξs = 1% and maximum redundancy = 60%.

From Figure 4, the residual loss rate for the BOAFEC scheme is very close to the RS FEC in the region of loss of 0% to 20%, where both the BOAFEC and RS FEC performs much better than the replicated stream in the above region, but it is still known that the replication stream performs better than both in networks with high loss rates (greater than 40%). Figure 4 shows also a swing results for the BOAFEC under the specified residual loss ξs, which is 1% in our example. The residual loss ξs is considered to be compensated, whither it is objectively tolerable or the destination uses other client based repair techniques, such as Interpolation or regeneration of lost packets [14].

Figure 5 shows the redundant bandwidth required for each scheme. Obviously the BOAFEC uses the lowest bandwidth while maintain close residual loss rates. The BOAFEC achieves its best results for networks with low to medium loss rates. Note when the loss rate exceeds the 25%; the BOAFEC saturates to the limit of its maximum allowed redundant bandwidth in order to reduce the residual loss rate.

## V. CONCLUSION

The packets loss is inevitable in networks, data networks can tolerate the latency but not the loss, where multimedia networks can tolerate the loss but cannot tolerate the latency, due to the interactive nature of multimedia applications. The FEC presents the least latency recovery technique. The FEC is a very promising technique for developing Multimedia applications over the Internet without scarifying the QoS of the media applications.

In this paper, we proposed and study a bandwidth optimized FEC approach, by predicting the loss at the client, while optimizing the amount of redundancy in order to achieve a certain residual loss rate.

The BOAFEC achieves very close recovery rates to the pure FEC, while saving 25% (on average) of the bandwidth, when the network loss rates are in the range of 0% to 20%. In networks with high loss rates, the BOAFEC saturates on the maximum allowed redundancy in order to achieve the best possible quality.

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